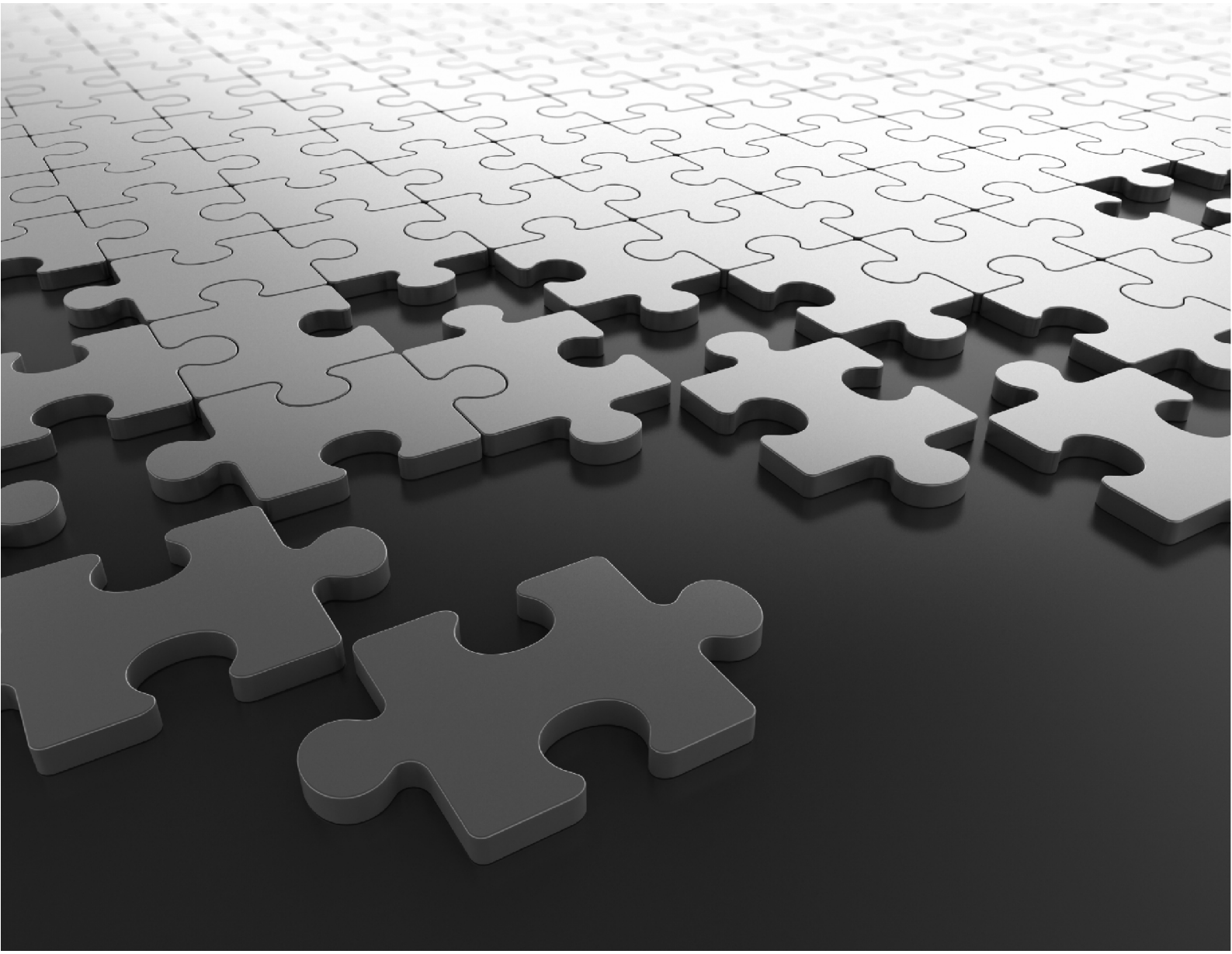
SARVAM UCS
System Manual



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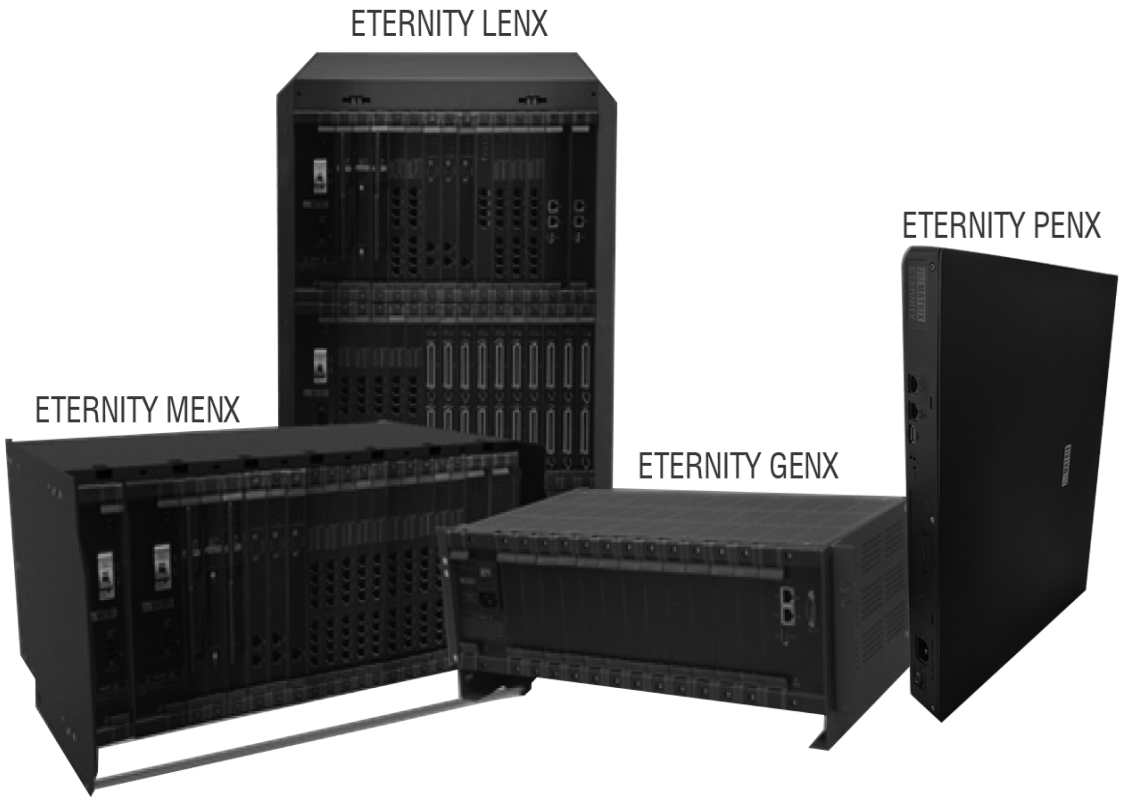
SARVAM UCS

The Unified Communication Server

System Manual



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CHAPTER 1 *Overview*

About the Product

Matrix provides different platforms to run the SARVAM Application.

- The ETERNITY GENX platform - This is the common hardware platform for SARVAM UCS SME and SARVAM UMG Application.
- The ETERNITY LENX/MENX platform - This is the hardware platform for SARVAM UCS ENT Application.
- The ETERNITY PENX platform - This is the hardware platform for SARVAM UCS SMB Application.

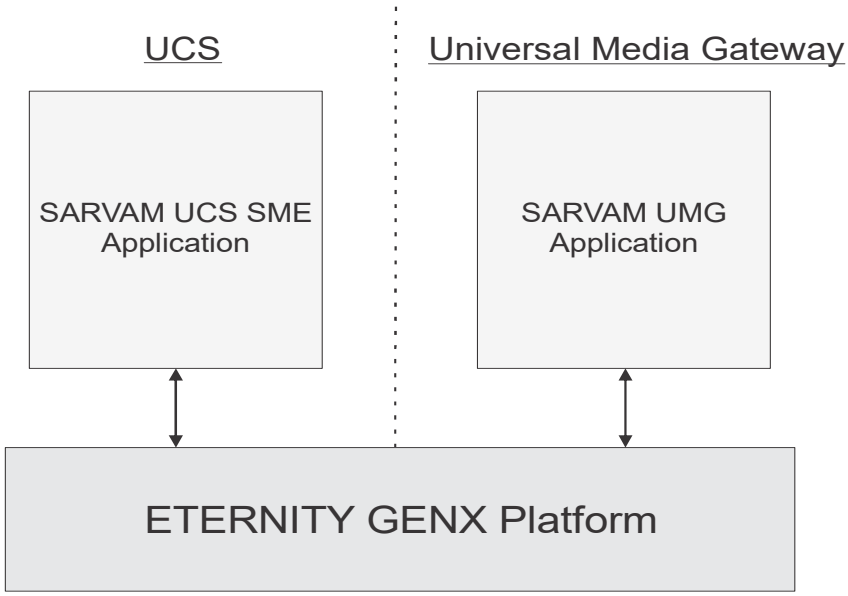
ETERNITY GENX Platform

The ETERNITY GENX is the common hardware platform for SARVAM UCS and SARVAM UMG Application.

You can use the ETERNITY GENX as the Unified Communication Server or the Universal Media Gateway depending upon the Application License you purchase.

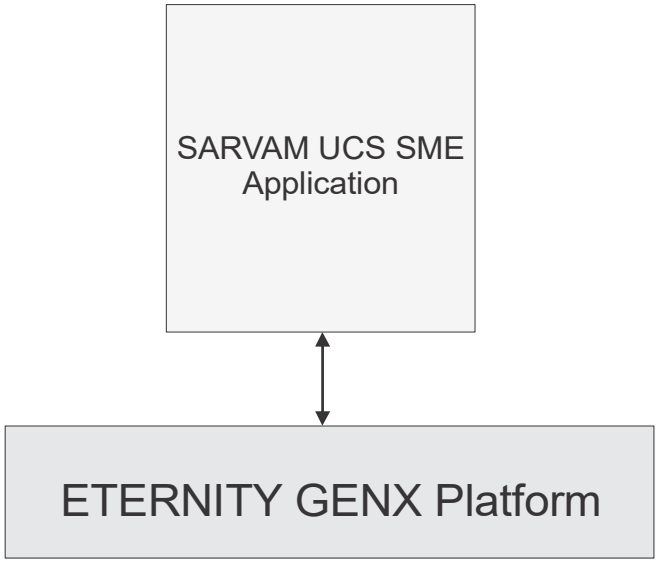
To run the ETERNITY GENX as the Unified Communication Server, you need to purchase the **SARVAM UCS SME** Application[1] license and to run it as the Universal Media Gateway, you need to purchase the **SARVAM UMG SME** Application license.

1. Refer *"Pre-activated Licenses"*.



ETERNITY GENX as the Universal Media Gateway acts as an IP-based system providing value added voice services. The system enables you to route calls from the Source port to the Destination port using Destination Number determination and Destination Port determination methods. It also offers a robust SIP Stack which gives a thorough interoperability and usability with various SIP trunk providers and IP-PBX. The system provides high performance and high reliability in a compact, modular design.

ETERNITY GENX as the Unified Communication Server acts as a fully hosted and managed Unified Communication Server. It delivers the convergence of voice, data, wired communications for small and medium sized businesses. It also offers UC features, Voice over IP Integration, Voice Mail, Computer Telephony Integration and Switching functions. The system provides reliable, efficient and unrestricted simultaneous communication (incoming and outgoing) by all users.

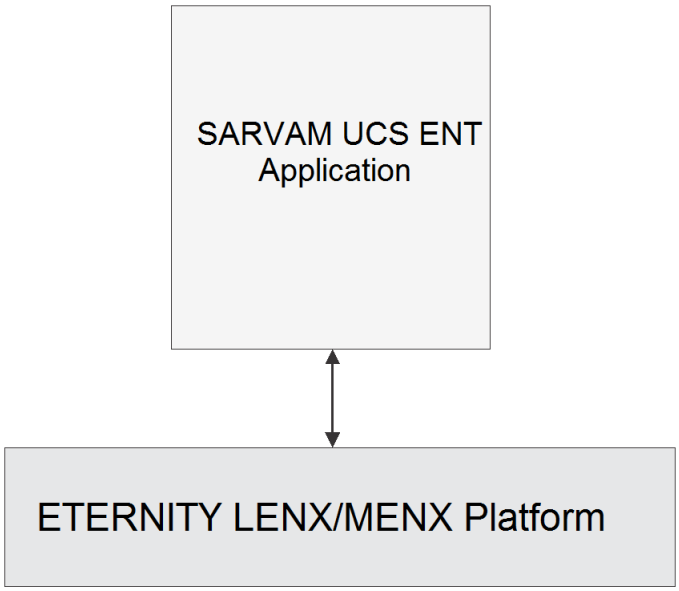


ETERNITY LENX/MENX Platform

The ETERNITY LENX/MENX is the hardware platform for SARVAM UCS ENT Application.

To run the ETERNITY LENX/MENX as the Unified Communication Server, you need to purchase the **SARVAM UCS ENT** Application[2] license.

ETERNITY LENX/MENX as the Unified Communication Server acts as a fully hosted and managed Unified Communication Server. It delivers the convergence of voice, data, wired communications for small and medium sized businesses. It also offers UC features, Voice over IP Integration, Voice Mail, Computer Telephony Integration and Switching functions. The system provides reliable, efficient and unrestricted simultaneous communication (incoming and outgoing) by all users.



ETERNITY PENX Platform

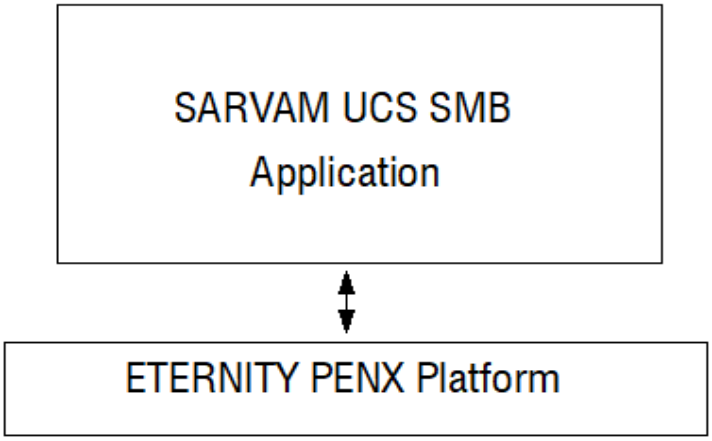
The ETERNITY PENX is the hardware platform for SARVAM UCS SMB Application.

To run the ETERNITY PENX as the Unified Communication Server, you need to purchase the **SARVAM UCS SMB** Application[3] license.

ETERNITY PENX as the Unified Communication Server acts as a fully hosted and managed Unified Communication Server. It delivers the convergence of voice, data, wired communications for small and medium sized businesses. It also offers UC features, Voice over IP Integration, Voice Mail, Computer Telephony Integration and Switching functions.

The system provides reliable, efficient and unrestricted simultaneous communication (incoming and outgoing) by all users.

2. *Refer "Pre-activated Licenses".*
3. *Refer "Pre-activated Licenses".*



The manual henceforth will describe in detail — the application, installation, configuration and features of SARVAM UCS.

CHAPTER 2

Introduction

Welcome

Thank you for choosing our product SARVAM UCS! We hope you will make optimum use of this intelligent, fully hosted and managed Unified Communication Server. Please read this document carefully to get acquainted with the product before installing and operating it.

About this System Manual

This System Manual provides information and instructions for installing, configuring and using the SARVAM UCS.

You may also refer to the SARVAM UCS *Quick Start* for quick installation. To view or download the Quick Start, scan the QR Code printed on the Product Label/Packaging Label. Similarly, the instructions and guidelines for setting up and operating SARVAM UCS in hotels and health care establishments is documented in *SARVAM UCS Hospitality System Manual.*

You may download the documents from https://www.matrixcomsec.com/support/telecom-product-manuals/

For instructions on using the features of the SARVAM UCS by the clients, refer to the respective User Guides The documentation can be found at the link shared above.

For product registration and warranty related details, please visit https://www.matrixcomsec.com/warranty/#telecom

This is a common documentation for all the platforms and the SARVAM UCS Application. This document is written with reference to the ETERNITY GENX platform and SARVAM UCS SME Application.

The configuration commands are written with respect to ETERNITY MENX.

Intended Audience

This System Manual is aimed at:

- **System Engineers**, who will install, maintain and support the system. System Engineers are persons who customize the system configuration to meet the requirements of the organization/users. It is assumed that they are experienced in installing the Unified Communication Server, are familiar with telecom wiring technology, how it works, and the various technical terms and functions associated with it. The SE must have undergone training in configuring the SARVAM UCS.

No one, other than the System Engineer is permitted to make any alterations to the configuration of the SARVAM UCS.

- **System Administrators**, who are the people administering the system. Generally an operator/receptionist in an organization, or the staff manning the reception or front desk area of the establishment are selected as System Administrators.

  It is assumed that the System Administrators have some previous experience in administering the system and its Terminals and Consoles. The System Administrators are not expected to install and program the Unified Communication Server, but only the routine jobs and features that are specific to them like generating SMDR reports, Setting report filters, configuring the features, Setting Alarms, reminders, etc.

- **Users**, people/organizations who will use the resources of the SARVAM UCS. They may be executives, include personnel of small and medium businesses, large enterprises, front desk and service staff of Hotels/Motels, hospitals, and other commercial and public organizations/institutions.

Organization of this Document

This system manual is broadly divided into the following:

- **Know Your SARVAM UCS** - introduces the system features and benefits, describes the application scenarios, hardware overview and the interfaces supported.

- **Installing the ETERNITY LENX** - gives step-by-step instructions for preparing for and installing the ETERNITY LENX Platform in general, like setting up the main distribution frames for the wiring, the safety measures for protecting the system and persons handling the installation and maintenance. It also provides step-by-step instructions for installation, inserting the cards, connecting the cables and powering the system.

- **Installing the ETERNITY MENX** - gives step-by-step instructions for preparing for and installing the ETERNITY MENX Platform in general, like setting up the main distribution frames for the wiring, the safety measures for protecting the system and persons handling the installation and maintenance. It also provides step-by-step instructions for installation, inserting the cards, connecting the cables and powering the system.

- **Installing the ETERNITY GENX** - gives step-by-step instructions for preparing for and installing the ETERNITY GENX Platform in general, like setting up the main distribution frames for the wiring, the safety measures for protecting the system and persons handling the installation and maintenance. It also provides step-by-step instructions for installation, inserting the cards, connecting the cables and powering the system.

- **Installing the ETERNITY PENX** - gives step-by-step instructions for preparing for and installing the ETERNITY PENX Platform in general, like setting up the main distribution frames for the wiring, the safety measures for protecting the system and persons handling the installation and maintenance. It also provides step-by-step instructions for installation, inserting the cards, connecting the cables and powering the system.

- **Configuring ETERNITY GENX** - describes the basic configuration to select the application you wish to run on the ETERNITY GENX platform.

- **Configuring SARVAM UCS** - contains description of the different tools and options available to configure SARVAM UCS. It provides detailed description of how to configure the various extension and trunk port

types - SIP Extensions, VoIP-SIP, T1E1PRI, E1FO, CO, Radio, Data, Mobile, BRI, E&M, SLT, DKP, ISDN Terminal and Magneto, Virtual Extensions - supported by SARVAM UCS.

- **Features and Facilities** - describes in detail, each feature and facility offered by the SARVAM UCS. This includes a description of the feature/facility, how it works, and how to program the feature/facility.

  The feature description is arranged alphabetically by Feature Name to make it easy for you to locate the description you want to look up.

- **System Maintenance** - provides instructions for back-up, generating reports and debugging.

How to Read this System Manual

This document is organized in a manner to help you get familiar with the ETERNITY GENX Platform and SARVAM UCS SME Application, learn how to install the platform, load application on platform, connect it to the various networks and interfaces, start the system and use the features.

This System Manual is presented in a manner that will help you find the information you need easily and quickly.

You may use the table of contents and the Index to navigate through this document to the relevant topic or information you want to look up.

All the commands given in the System Manual can be performed from DKP. However, these commands may or may not work from the IP Phones.

Cross-references are provided in blue font with hyperlinks. You can look up the source by clicking the links.

Instructions

The instructions in this document are written in a numbered, step-by-step format, as follows. Each step, its outcome and indication/notification, wherever they occur, have been described.

Access Codes

Access codes are strings of digits dialed by an extension to

- Call another extension, Department Group,
- Grab a trunk line
- Use a Feature, like Call Block, Call Forward.

The Access Codes provided in the instructions throughout this document, are default access codes. It is possible to change the Access Codes according to user requirement and preferences. Verify with the Installer/System Engineer, if the default Access Codes have been changed, and use the codes programmed by the System Engineer. For more information, read the topic “Access Codes” in this document.

Notices

The following symbols have been used for notices to draw your attention to important items.

Important: *to indicate something that requires your special attention or to remind you of something you might need to do when you are using the system.*

Caution: *to indicate an action or condition that is likely to result in malfunction or damage to the system or your property.*

Warning: *to indicate a hazard or an action that will cause damage to the system and or cause bodily harm to the user.*

Tip: *to indicate a helpful hint giving you an alternative way to operate the system or carry out a procedure, or use a feature more efficiently.*

Terminology used in this System Manual

The technical terms and Acronyms used in this Manual are standard terms, commonly used in the telecommunications and data communications industry. Considering the broad group of intended users of this manual, wherever possible, use of jargon has been avoided.

Acronyms have been defined in the text and a list of the same is appended.

Some of the terms specific to this Manual that you will encounter are defined below:

The words **'UCS'**, **'SARVAM UCS'** and **'System'** are used interchangeably and synonymously to mean Unified Communication Server.

- **IP Phone:** a phone that uses Voice over IP (VoIP) Technologies for placing and transmitting telephone calls over an IP network, such as the Internet.
- **SIP Extensions:** any SIP-enabled device — a Matrix UC Client (Matrix VARTA WIN200, Matrix VARTA ADR100, Matrix VARTA AMP100), an IP-phone, an Extended IP-phone, a Soft phone or an Analog Phone Adapter — registered with the system, from which you can make/receive calls to any extension or external number.
- **SIP Trunking:** VoIP or streaming media service based on Session Initiation Protocol (SIP) by which Internet Telephony Initiation Protocols (ITSPs) deliver telephonic services and Unified communications to the customers equipped with SIP-based System.
- **Port:** the physical interfaces on the cards for trunk lines and extension lines.
- **Extension:** it is the port of the system to which a telephone instrument (DKP/SLT/ISDN/SIP) is connected.
- **Mobile Extension:** a mobile/landline phone used as a remote extension of SARVAM UCS. You can access all the features of an extension of SARVAM UCS from the mobile/landline phone.
- **RUIM Card**: Removable User Identity Module is a card that supports the CDMA services just like the SIM card that supports GSM services. In this manual, consider SIM as RUIM, if CDMA Mobile Card is installed in your system.
- **Station:** same as extension.
- **Service Provider:** the providers of telecom network lines/Internet - POTS, PSTN, CDMA, GSM, ISDN PRI, ISDN BRI, and Internet Telephony Service Providers (ITSP).
- **CO trunks:** two-wire trunks, that is, analog trunk lines from the POTS network.

- **System Administrator Commands/SA Commands:** number strings dialed from the System Administrator access/mode to operate features or set/cancel features for other extensions.
- **System Commands/SE Commands:** number strings dialed from the System Engineer access/mode to program the system features/functions.
- **CO Network:** the public telephone exchange.
- **CO Lines:** the lines subscribed from the CO Network. These may be Two-wire Trunk Lines, ISDN BRI, ISDN PRI, etc.
- **Digital Key Phone (DKP):** refers to EON, the proprietary digital key phone of Matrix that can be connected with the system. The term 'Digital Key Phone' refers to all models of EON.
- **Single Line Telephone (SLT):** any standard two-wire telephone attached as extensions of the system.
- **Called party/Callee:** the person to whom the call is made.
- **Calling party/Caller:** the person who makes a call.
- **Enterprise Application/Features:** pertaining to the general and special telephone and call management features required by business establishments, public and private organizations.
- **Internal numbers:** same as extension numbers.
- **External Numbers:** numbers of parties/individuals outside the UCS network. The unique number string given to subscribers of PSTN, PLMN, ITSP, etc.
- **Internal Calls:** calls made from and received by one extension to another extension of the SARVAM UCS.
- **External Calls:** calls made by users of SARVAM UCS to subscribers of PSTN, PLMN, ITSPs, etc.
- **Hospitality Application/ Features:** pertaining to the special telephone and guest/patient management features required by accommodation establishments like hotels and hospitals.

Using this Manual, we hope, you will be able to set up, operate and make optimum use of this feature packed Unified Communication Server. If you encounter any technical problems, please contact your Dealer/reseller or the Matrix Customer Care.

CHAPTER 3

Know Your SARVAM UCS

Introduction

When ETERNITY GENX platform is used with SARVAM UCS application, it acts as a fully hosted and managed Unified Communication Server. It delivers the convergence of voice, data, wired and wireless communications for small and medium sized businesses.

SARVAM UCS offers UC features, Voice over IP Integration, wireless communications, voice mail, computer telephony integration and switching functions.

This Unified Communication Server creates a collaborative environment by:

- Integrating UC Features like unified messaging, video conferencing etc.
- Addition of more features and functions
- Simplifying management
- Providing services and support
- Centralizing troubleshooting to help solve problems quickly
- Securing IP communications for the mobile workforce

Features and Benefits

Features - UC Features like video calling, instant messaging, IM to SMS, Email to SMS etc, Caller ID, Call Forwarding, Pushkit support, Conference calling and voice messaging.

SIP Trunking - SIP Trunking to Internet Telephony Service Providers. SARVAM UCS allows users to make or receive SIP Trunk calls.

Trunk Interfaces - A variety of network trunk interfaces including CO/Two-wired trunk, ISDN BRI, ISDN T1E1PRI, E&M, Mobile and SIP Trunks.

Extensions - Provides support for a range of extensions — Digital Extensions, Analog Extensions, Radio Extensions, Virtual Extensions and SIP Extensions using Voice over IP technology to provide sophisticated voice performance for new and growing businesses.

UC Clients - Provides support for Matrix UC Clients — VARTA WIN200 Desktop UC Client, VARTA ADR100 and VARTA AMP100 Mobile UC Clients.

Telephones - A variety of telephones including analog, digital, IP hard and soft phones (wired and wireless) to provide an appropriate desktop or device telephone for every need.

Advanced Call Routing - Various options for routing incoming and outgoing calls on trunks are provided.

Q-Sig Networking - With Q-Sig, you can network SARVAM UCS with another UCS or any other ISDN QSIG supported system to expand the system resources. This provides the feature transparency between the systems to function as a single unit.

Universal Connectivity

SARVAM UCS supports UC communication which includes Instant Messaging, Call Management, Smart Directory and Video Calling using a variety of *IP phones such as the Standard SIP Phones and Matrix UC Clients — VARTA WIN200 Desktop UC Client, VARTA ADR100 and VARTA AMP100 Mobile UC Clients.*

SARVAM UCS offers Universal Connectivity, working with all major telecom interfaces: VoIP, POTS, ISDN BRI and PRI (T1/E1), GSM/3G/CDMA, E&M, Radio and Magneto. So, you have access to multiple telecom networks on a single platform. The system's intelligent Least Cost Routing logic diverts your calls through the appropriate network, ensuring least possible call cost.

SARVAM UCS also supports Video Conferencing over IP using any standard SIP Video Conferencing device. Video Conferencing and data connectivity is also supported over ISDN interface.

Flexibility and Scalability

SARVAM UCS is designed to support both pure IP and Hybrid IP solution. Non-IP interfaces can be added by inserting non-IP homogeneous and Mix Port Slave Cards in Universal Slots. The Hybrid Cards offer all possible communication interface options to match your communication requirement.

The system is also designed to provide very high level of flexibility and scalability to meet your future communication needs. The flexible licensing allows you to start with desired number of SIP Users, VoIP Channels and VARTA UC Users. If required, you may increase the number of users or channels supported by purchasing the respective licenses.

The Universal Slot platform and the modular design of the Cards allow you to start with the minimum required configuration and expand the system capacity later, by adding more Cards to the Universal Slots. So, you can invest progressively in scaling up the system as the communication needs of your organization grow.

Cost Saving and Productivity Enhancing Features

Key features like Multi-party Conference, Auto Attendant, Remote programming, SMDR Buffer with a large capacity, that normally warrant additional investment in most other brands, are built into the system.

Intelligent features like Auto Attendant, CLI/DDI based Routing and Dial by Name ensure efficient call management and prompt response to callers.

Least Cost Routing and Call Budgeting help reduce communication cost and enhance productivity.

The system can route a VoIP call to GSM, CO or T1E1PRI port. In the same way, a call on a T1E1PRI port can be routed to VoIP, GSM or CO ports. Further, you can select Fixed or Least Cost Routing to route outgoing calls. SARVAM UCS can handle calls on all ports simultaneously, allowing full traffic on all ports.

SARVAM UCS can work as an adjunct to your existing telephony infrastructure, as a Gateway, saving you the cost of equipment replacement, wiring and installation, while giving you Universal Connectivity and a host of intelligent features.

Redundancy[4]

To reduce down time and provide uninterrupted communication, the ETERNITY LENX/MENX supports redundancy option for the two cards that are critical to its functioning: The Power Supply Card and the CPU Card. There are two cards of each on the platform. When the active card fails, the standby card takes over.

Hot Swap[5]

With the Hot Swap feature you can remove a card and insert it back without switching off the system. So, you can replace a faulty card with a functional one without affecting the functioning of the system. The ETERNITY LENX/MENX supports Hot Swap.

UC Features

- Set/View Presence
- Presence based call controlling (make call/reject call)
- Video Calling
- IM using VARTA WIN200, VARTA ADR100 and VARTA AMP100
- IM using third party SIP Phones/soft clients
- IM to SMS and vice versa using SMS Server
- SMS campaigning using SMS Server and SMS Gateway(in collaboration with third party SMPP application)
- Smart Handover from extension to mobile
- Smart directory access using VARTA WIN200, VARTA ADR100 and VARTA AMP100 for easy and quick access to the extensions and other contacts
- Mobile and Remote workers support
- Outlook integration using CTI Interface
- Auto-attendant with configurable call-flow

Other Key Features

- Account Codes
- Auto Black List IP Address for Web access, VoIP access, etc.
- Call Accounting Software interfaces supported
- Call Billing
- CAS Interface
- Class of Service
- CLI Based Routing
- Closed User Group
- Conference upto 21 party
- Emergency Call Detection and Reporting
- Firmware and Configuration zip file upload using browser
- Hospitality features with Front Desk Web Interface (for Hotels and Hospitals)
- Hot Desking
- Internal SMDR Report buffer
- Least Cost Routing
- Logical Partitioning for restricting toll bypass
- PMS Interface
- Priority
- Q-Sig
- Remote(ISDN) Programming

4. ETERNITY GENX and PENX do not support Redundancy.
5. ETERNITY GENX and PENX do not support Hot Swap.

- SNMP interface for error/event notification and status check
- Syslog (for easy maintenance)
- System Activity Log
- System Fault Log
- Toll Control
- Trusted IP Address to avoid unauthorized access
- Universal Slots
- Voice Applications
- Voice Mail
- Web-based Programming (HTTP/HTTPS)
- White List IP Address

Applications of SARVAM UCS

The Matrix SARVAM UCS can be deployed in small to large enterprises and institutions: manufacturing units, corporate offices, banking and financial institutions, software firms, shopping malls, hospitals, hotels-motels, in power line carrier communication of electric utilities, call centers, in institutions and, power line carrier communication (PLCC) networks.

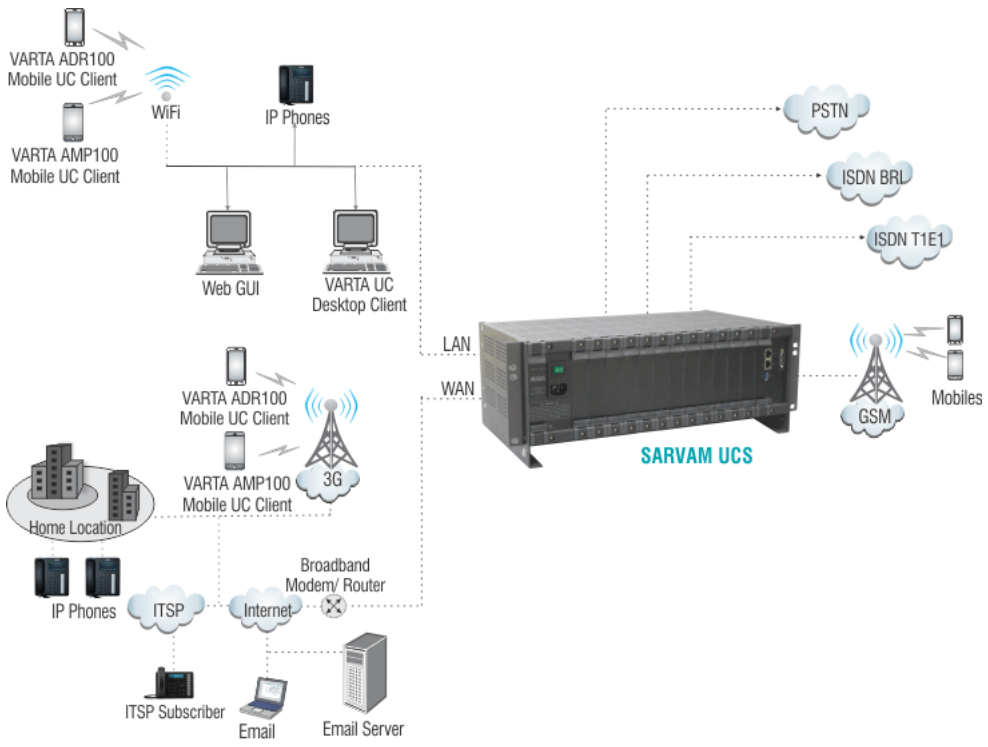
Illustrated in the following are various scenarios where the SARVAM UCS finds application.

Enterprise Application

UC Applications

If connecting to the Public IP Network,

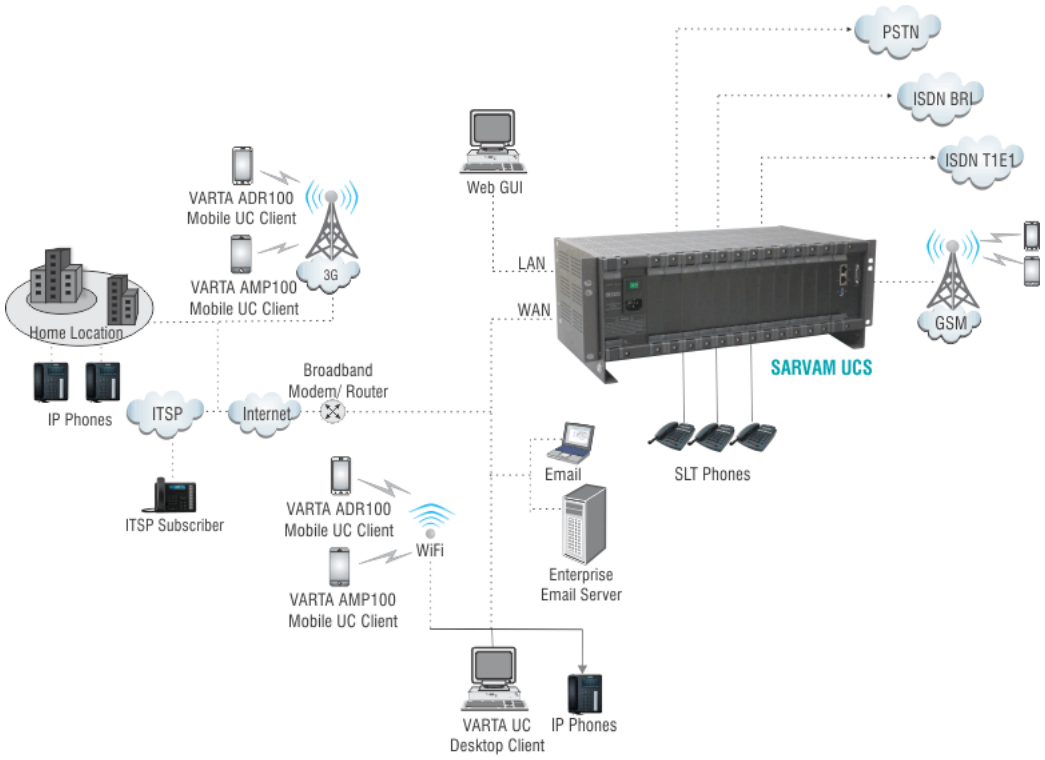
Plug one end of the RJ45 Ethernet cable into the WAN Port of the system and the other end into the Broadband Router/Modem.



Connecting SARVAM UCS to the Public IP Network

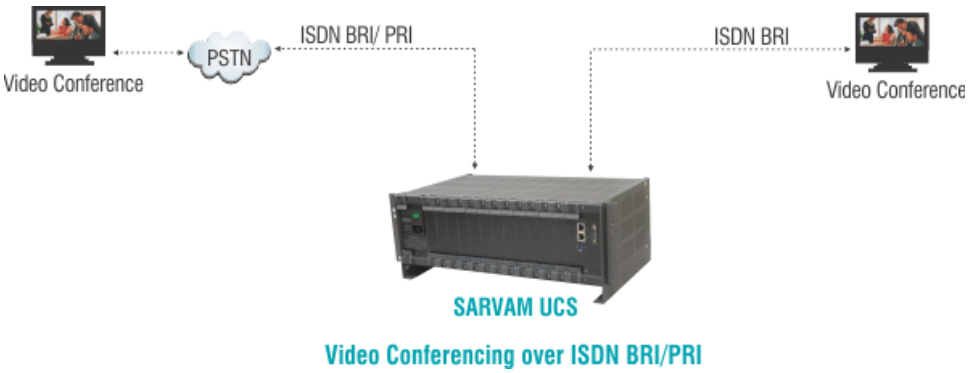
If connecting to a Private Network (Behind a NAT Router),

Plug one end of the RJ45 Ethernet cable into the WAN Port of the system and the other end into the LAN Switch/ Hub.

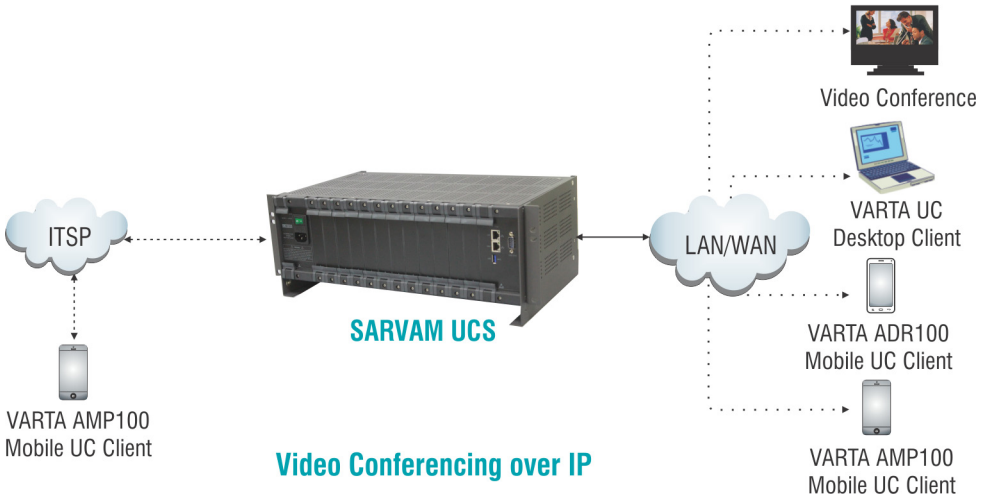


Connecting SARVAM UCS to the Private IP Network

Video Conferencing

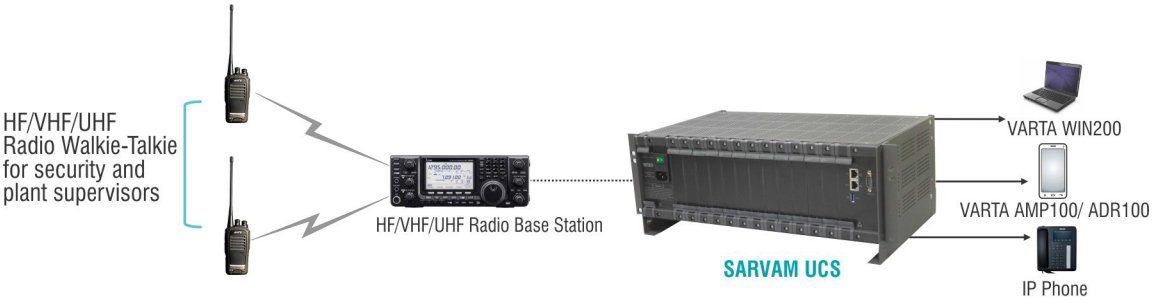


Video Conferencing over ISDN BRI/PRI



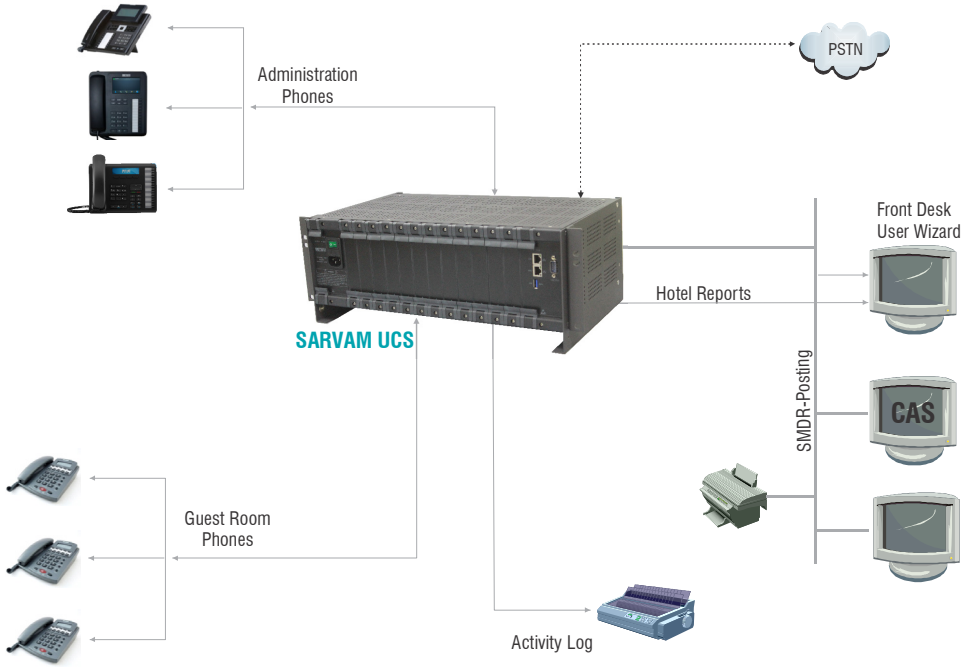
Video Conferencing over IP

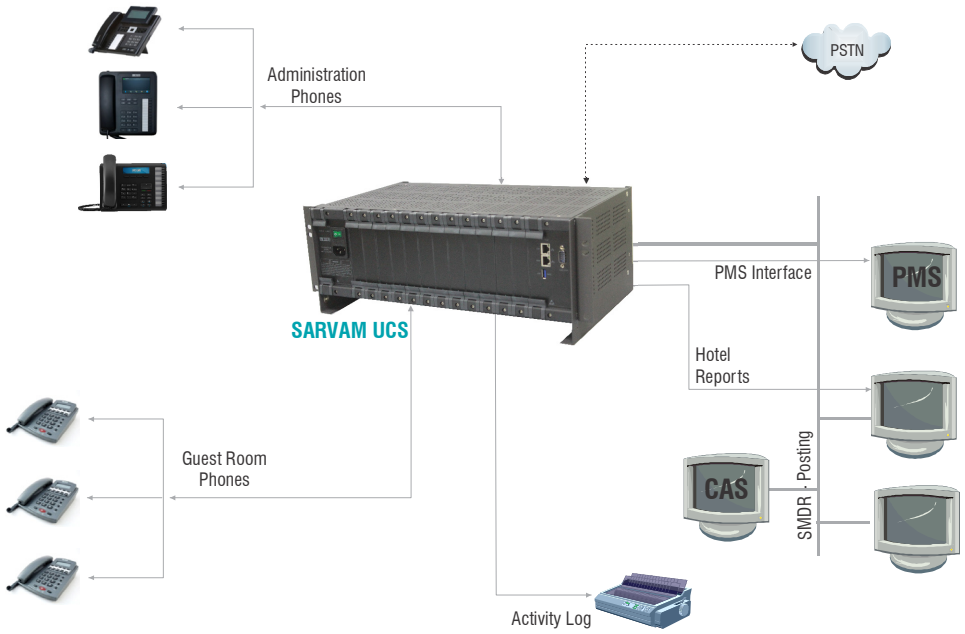
Mobility



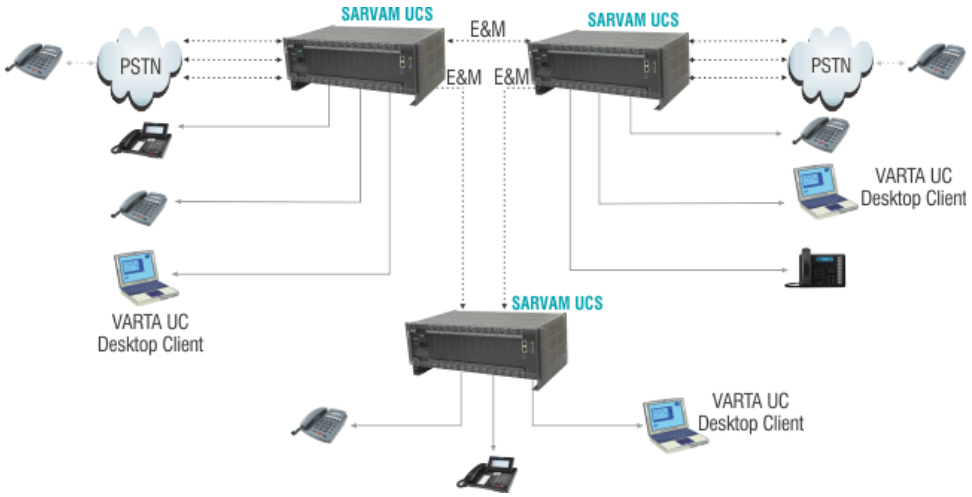
SARVAM UCS - MOBILITY

Hotel Application

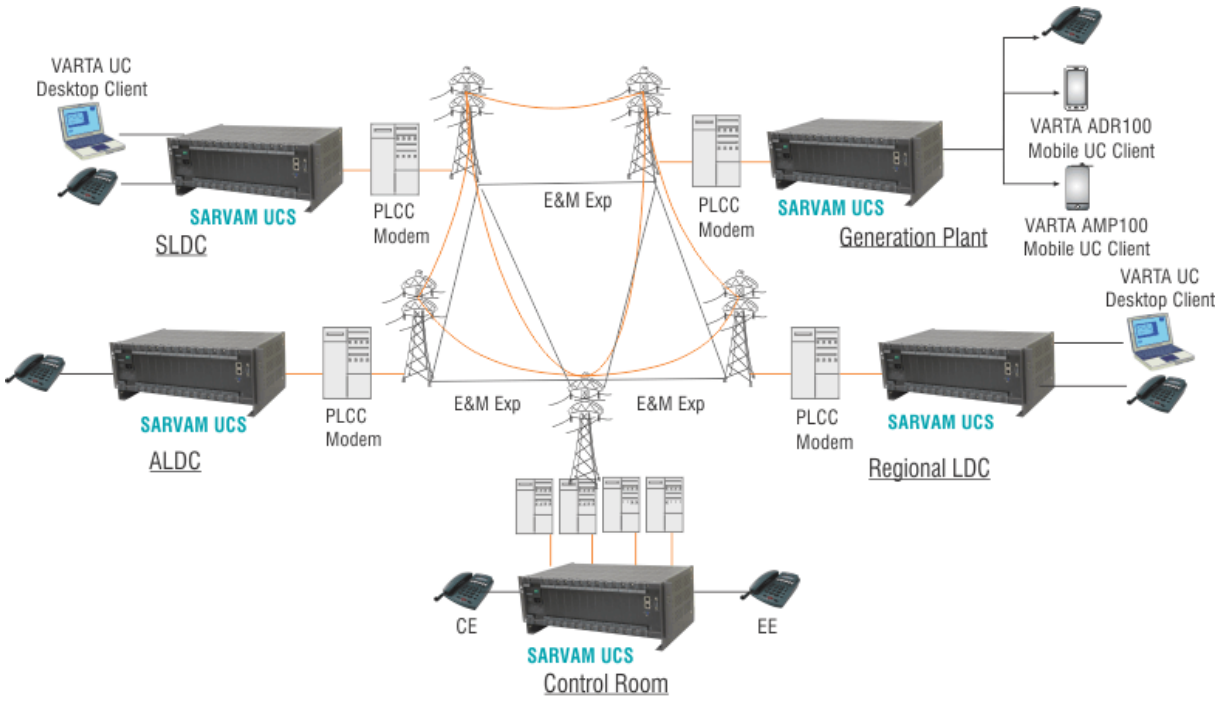




Closed User Group Application



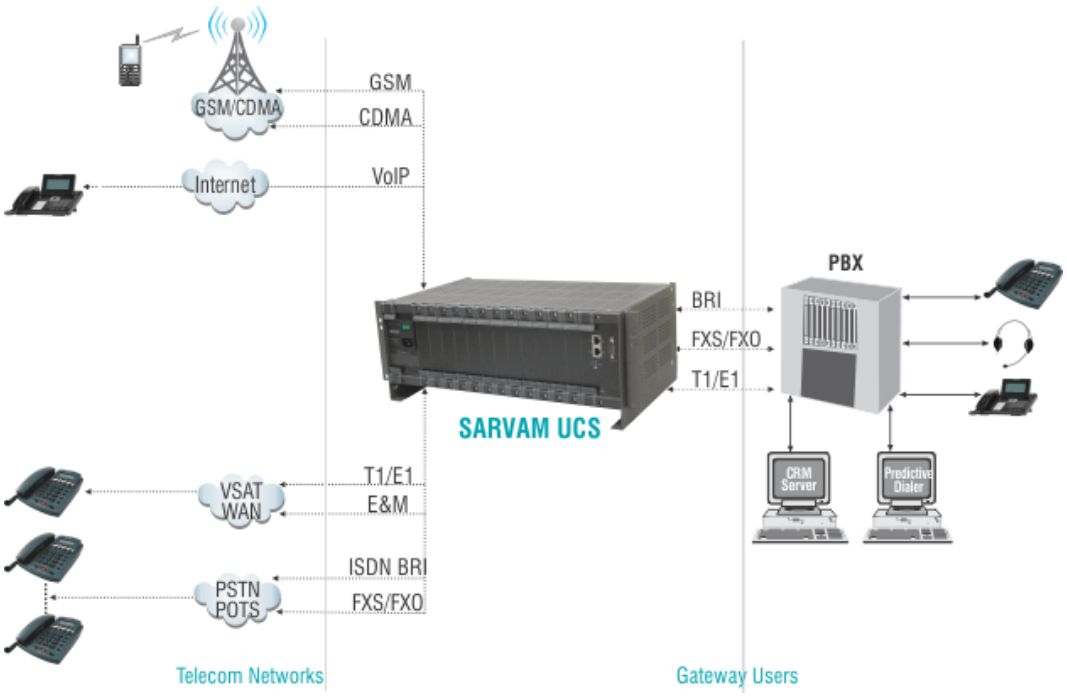
Power Line Carrier Communication Application



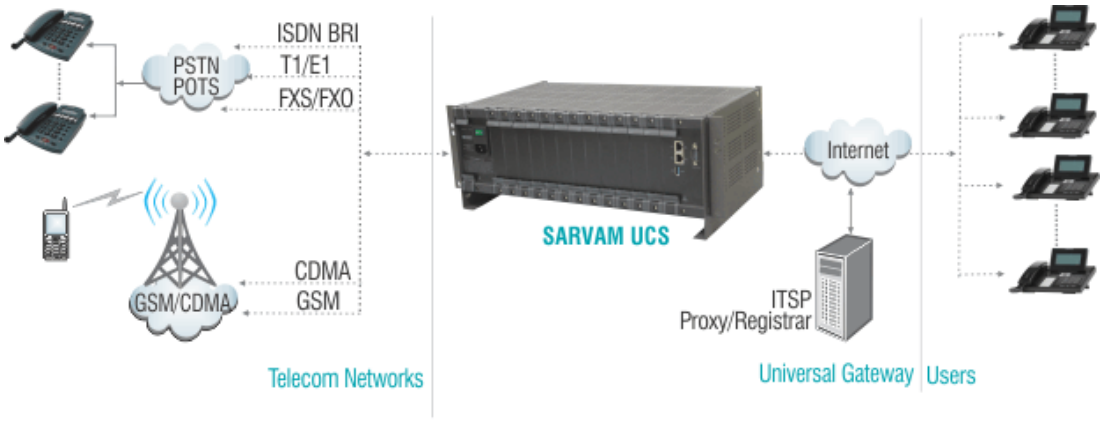
CE: Chief Engineer
EE: Executive Engineer
LDC: Load Despatch Centre

E&M - Express Lines

Universal Gateway Application



SARVAM UCS - Universal Gateway for Call Centres



SARVAM UCS- Universal Gateway for ITSP

Hardware Overview

ETERNITY LENX

The Enclosure

The enclosure of ETERNITY LENX consists of fixed and universal slots. The fixed slots are occupied by specific cards - Power Supply Cards and CPU Cards - and cannot be changed, whereas in the universal slots, you can install any of the various card.

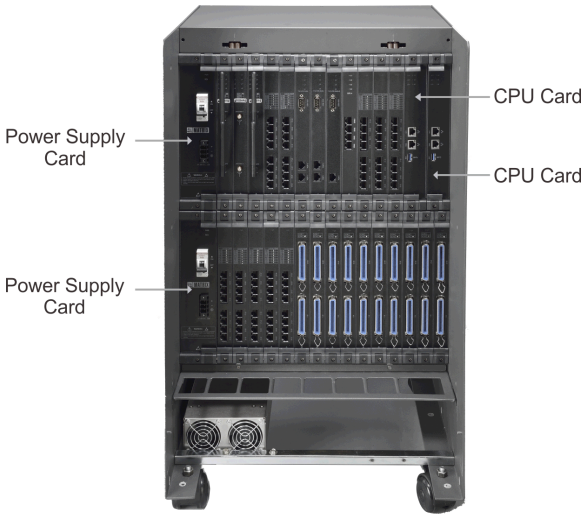
The slot connectors are located on the motherboard on the backplane of the enclosure. Each slot has guide rails for inserting the cards.

ETERNITY LENX has two racks and a total of 27 Universal slots. The first rack has a total of 15 slots. The first slot from the left is a fixed slot and the last two slots to the right are fixed slots and the remaining 12 are universal slots.

The second rack has a total of 16 slots. The first slot from the left is a fixed slot and the remaining 15[6] are universal slots.

ETERNITY LENX can be wall mounted, rack mounted or placed on a table. You can also affix wheels to the system; to move it like a trolley. After you have inserted the required cards, you must affix the front top cover.

Illustrated below are design of the enclosure and the position of the slots in ETERNITY LENX.



The Cards

ETERNITY LENX houses the following Cards:

1. CPU Card
2. Power Supply Card - PS48V
3. SLT Card
4. CO Card
5. CO+SLT Card

6. If you have systems with Hardware Version V1R4, the CPU Cards have 1K IDT Switch (PCB Version V1R3), hence out of the 15 universal slots, in four slots from the right the following cards cannot be affixed — ETERNITY LE CARD SLT48, ETERNITY ME CARD E1FO Dual and ETERNITY ME CARD T1E1PRI Dual. In the remaining slots any card can be affixed.The maximum number of SLT Ports supported will be 1232.

6. DKP Card
7. E&M Card
8. BRI Card
9. T1E1PRI Card
10. GSM/3G Card
11. Magneto Card
12. Radio Card
13. Data Card

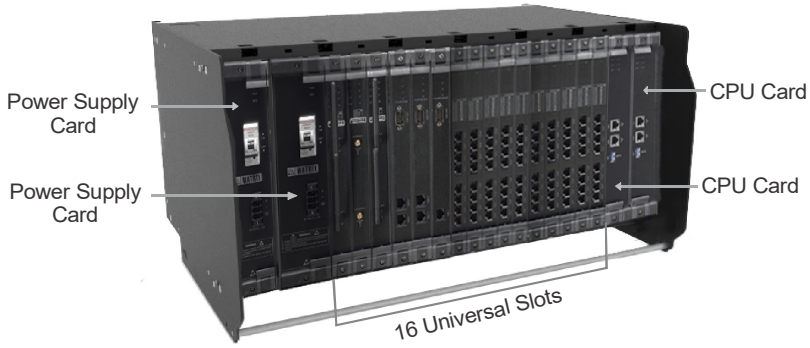
ETERNITY MENX

The Enclosure

The enclosure of ETERNITY MENX has fixed and universal slots. The fixed slots are occupied by specific Cards - Power Card and the CPU Card - and cannot be changed, whereas in the universal slots you can install various Slave Cards. ETERNITY MENX has 16 universal slots.

Inside the enclosure of ETERNITY MENX are slot connectors located on the motherboard on the backplane of the enclosure. Each slot has guide rails for inserting the Cards.

Illustrated below are the design of the enclosure and the slots positioning in ETERNITY MENX.



The Cards

ETERNITY MENX houses the following Cards:

1. CPU Card
2. Power Supply Card - PS48V/ PS UNI
3. SLT Card

4. CO Card
5. CO+SLT Card
6. DKP Card
7. E&M Card
8. BRI Card
9. T1E1PRI Card
10. GSM/3G/CDMA Card
11. Magneto Card
12. Radio Card
13. Data Card

ETERNITY GENX

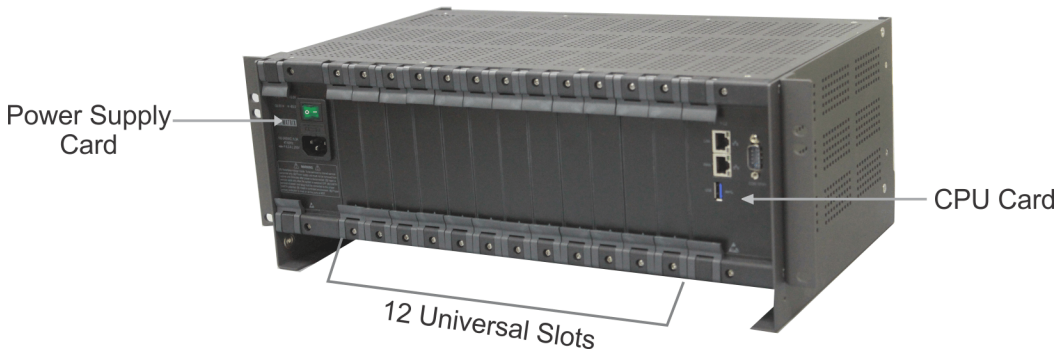
The Enclosure

The enclosure of ETERNITY GENX has fixed and universal slots. The fixed slots are occupied by specific Cards - Power Card and the CPU Card - and cannot be changed, whereas in the universal slots you can install various Slave Cards. ETERNITY GENX has 12 universal slots.

Inside the enclosure of ETERNITY GENX are slot connectors located on the motherboard on the backplane of the enclosure. Each slot has guide rails for inserting the Cards.

Illustrated below are the design of the enclosure and the slots positioning in ETERNITY GENX.

ETERNITY GENX



The first slot from the left is reserved for the Power Supply Card and the first slot from the right is reserved for the CPU Card.

The Cards

The ETERNITY GENX houses the following types of cards:

1. Power Supply Card - AC Input Card or DC Input Card
2. CPU Card
3. SLT Card
4. CO Card
5. DKP Card
6. CO+SLT(Combo) Card
7. DKP+SLT(Combo) Card
8. CO+DKP+SLT(Combo) Card

9. Intercom Line Card
10. E&M Card
11. E1FOPRI Single Card
12. BRI Card
13. T1E1PRI Single Card
14. Magneto Card
15. Radio Interface Card
16. GSM/3G/CDMA Card

For an at-a-glance view of the Cards available with ETERNITY GENX Platform when used with SARVAM UCS Application, refer “Technical Specifications - SARVAM UCS”.

ETERNITY PENX

The Enclosure

The ETERNITY PENX has a different design. The enclosure of ETERNITY PENX consists of a top plate, which functions as the cover and can be removed.

The Power Supply unit and the CPU are in-built, and fixed on the bottom plane of the ETERNITY PENX.

The CPU card is fixed on the bottom plate of the enclosure, with the 2-row connectors for the card slots facing up. Illustrated in the following are the design of the enclosures and the position of the slots in each model of ETERNITY PENX.

ETERNITY PENX

The Power Supply unit and the CPU are in-built, and fixed on the bottom plane of the ETERNITY PENX.

Universal slots are located on the CPU. The connectors of the slots are located on the CPU.

Cards are mounted on the CPU, and secured on the three studs labeled as H1, H2, H3 on the CPU, with the screws are provided for this purpose.

The Cards

The ETERNITY PENX houses the following Cards:

1. CPU Card
2. SLT Card
3. DKP Card
4. CO Card
5. DKP+ SLT Card

6. CO + SLT Card
7. DKP + CO Card
8. DKP + CO + SLT Card
9. T1E1PRI Card
10. GSM4 Card

The Interfaces

The SARVAM UCS supports the following interfaces for connecting with different telecom networks, digital key phones, standard telephones and other external devices.

The VoIP Interface

The Voice-over-IP (VoIP) Interface connects SARVAM UCS with the internet or the private IP network. It enables the SARVAM UCS to handle incoming and outgoing calls over the IP network. It offers great flexibility and mobility to users for making or receiving calls across the world.

The VoIP Interface supports UC Clients, SIP Extensions and SIP Trunks.

With SIP Trunks users can make IP calls using the SIP Server of the Internet Telephony Service Providers (ITSPs).

Make sure the NX DBM VOCODER64 module is installed and the channel licenses — SARVAM VOCODER CHNL4 or SARVAM VOCODER CHNL16 — purchased as per your requirement.

The in-skin Registrar Server allows any SIP enabled device like a UC/SIP Soft Client or an IP-Phone to be registered with it and function as the 'SIP Extension' of SARVAM UCS. The SIP Extension users can make and receive calls to any extension user of the SARVAM UCS as well as any external numbers over VoIP, PSTN, GSM and other trunks. With SIP Extensions, organizations can communicate and stay connected at the lowest cost without any geographical restrictions.

The VoIP Interface supports adaptive jitter buffer for reducing delay and improving speech quality.

The key features of the VoIP Interface are:

- Up to 99[7] SIP Trunks - for Proxy as well as Peer-to-Peer (non-Proxy) calls.
- Up to 999[8] SIP Extensions.
- Upto 550[9] simultaneous IP-to-IP calls (with Direct RTP/ RTP Relay).
- 64 simultaneous IP-to-IP call legs / IP-to-TDM calls (with Transcoding) for one NX DBM VOCODER64 module installed.
- 128 simultaneous IP-to-IP call legs/ IP-to-TDM calls (with Transcoding) for two NX DBM VOCODER64 modules installed.
- Selectable Network Assignment (Connection Type) - Static IP, DHCP, PPPoE.
- Selectable DNS - Automatic and Static.
- Dynamic DNS for SIP devices in public network.
- STUN.

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7. ETERNITY LENX/MENX also supports 99 SIP Trunks and ETERNITY PENX supports 50 SIP Trunks.
8. ETERNITY LENX/MENX supports 2000 SIP Extensions and ETERNITY PENX supports 100 SIP Extensions.
9. This is the total capacity supported by SARVAM UCS system. ETRENITY PENX supports 64 simultaneous IP-to-IP calls.

- TCP and UDP NAT Keep Alive.
- VLAN
- Symmetric RTP Selection.
- MAC Address Cloning option.
- Fax over IP - T.38 (UDPTL), T.38 (RTP) and Pass Through.
- Send CLI Option for outgoing calls
- Selectable DTMF - RTP (RFC 2833), SIP Info, InBand
- Flash Detection using SIP INFO and RFC2833.
- Broad Voice Codec Selection: G.723, G.729 AB, GSM FR, iLBC - 30 ms, iLBC - 20 ms, G. 711 μ-Law, and G. 711 A-Law.
- Quality of Service - SIP DiffServe/ToS, RTP DiffServe/ToS
- VoIP Silence Detection and Disconnection.
- Voice Mail Subscription on SIP Extensions.
- Busy Lamp Field Subscription on SIP Extensions and SIP Trunks.
- Upto 10 Call Appearances on SIP Extensions.
- Registration of SIP Extensions from 3 different locations and Shared Call Appearance.
- Sharing the same SIP Extension number among a maximum of 3 different SIP devices using Share Call Appearance functionality.
- Programmable Loop Current

SIP Trunks

The ETERNITY GENX[10] platform when used with SARVAM UCS application supports a maximum of 99 SIP Trunks, allowing you to subscribe to as many as 99 different Internet Telephony Service Providers (ITSP) or peer-to-peer connectivity with multiple end SIP devices.

Make sure the NX DBM VOCODER64 module is installed.

SIP Extensions

ETERNITY GENX platform when used with SARVAM UCS application supports 999[11] SIP Extensions which can be registered with the in-skin SIP B2BUA of CPU Card.

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10. ETERNITY LENX/MENX also supports 99 SIP Trunks and ETERNITY PENX supports 50 SIP Trunks.
11. ETERNITY LENX/MENX supports 2000 SIP Extensions and ETERNITY PENX supports 100 SIP Extensions.

Any SIP-enabled device like an IP-phone, a Softphone, analog phone adapter, can be registered with the CPU Card and function as the 'SIP Extension' of the ETERNITY GENX platform when used with SARVAM UCS application.

The SIP Extensions function in the same way as other extensions of the SARVAM UCS. SIP Extension users can make and receive calls from and to other extensions of System and external numbers over PSTN, GSM, VoIP and E&M lines[12]. You can also connect the Standard and Extended IP Phones offered by Matrix as SIP Extensions.

Make sure the NX DBM VOCODER64 module is installed and the channel licenses — SARVAM VOCODER CHNL4 or SARVAM VOCODER CHNL16 — purchased as per your requirement.

A SIP Extension can be registered with SARVAM UCS from three different locations. This helps organizations overcome geographical distances and reduce call costs.

Five SIP Extensions are free, for additional extensions you require an IP Subscriber license. To know more about Licensing requirements and how to acquire and activate a license key, see the topic "License Management".

You can connect ETERNITY GENX platform when used with SARVAM UCS application to the IP network, which may be Public Internet or a LAN.

The ISDN PRI Interface

The ISDN PRI Interface enables SARVAM UCS to be connected to digital networks over T1[13] and E1[14] carrier lines.

On T1 carrier lines, the Interface supports the following signaling types:

- PRI
- Robbed Bit Signaling (RBS)
- Q-Signaling (QSIG)
- E&M

On E1 carrier lines, SARVAM UCS supports the following signaling types:

- PRI
- Channel Associated Signaling (CAS)
- Q-Signaling (QSIG)
- E&M

The ISDN T1E1PRI Interface supports the following features:

- Terminal (TE) mode and Network (NT) Mode
- Voice Calling
- Video conferencing and data connectivity

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12. *Only if there are no restrictions on calls from VoIP to other Public Networks in your country. If the telecom regulations of your country prohibit call traffic between the public telephony networks and IP networks, you must configure Logical Partition in your system. To know more, see "Logical Partition".*
13. *T1 PRI (T-Carrier) offers 23 Bearer Channels and one Signaling Channel (23B+D). It is used in North America and Korea.*
14. *E1 PRI (E-Carrier) offers 30 Bearer Channels and two Signaling Channels (30B+D). It is used in all countries, except North America, Japan and Korea.*

ISDN BRI[15]

The ISDN BRI Interface enables SARVAM UCS to be connected to ISDN BRI Lines and connect ISDN BRI compatible devices with the SARVAM UCS.

The ISDN BRI Interface has the following features:

- Signaling types - 2B+D Signaling
- Terminal (TE) mode
- Video conferencing and data connectivity

Depending on the requirement, each BRI Port can be configured in the TE/NT mode.

It is possible to feed power from the SARVAM UCS to the terminal equipment connected to the ETERNITY (on its BRI port configured as NT).

The Mobile Interface

The Mobile Interface enables the SARVAM UCS to be connected to 2G/3G/4G/CDMA[16] network operators worldwide. For example, the Frequency Band supported by the GSM networks varies across countries. In some countries, the network operators may use 850MHz while network operators in some countries may use 1900MHz, in some countries network operators may use both frequency bands.

SARVAM USC's Mobile Interface supports full Quad-Band Operation (GSM850, 900, 1800, 1900MHz) for world-wide use, for Global, Inter and Intra country roaming.

The Mobile Interface supports the following features:

- CDMA, GSM 2G, GSM 3G, GSM 4G network support.
- Selectable GSM Frequency Bands - 900, 1800, 1900, 850 + 1900, 900+1800 MHz.
- Selectable CDMA Frequency Bands - 800MHz CDMA cellular; 1900MHz PCS.
- Programmable Network Selection for GSM - Manual and Automatic. For CDMA, only Automatic is applicable.
- Programmable Network Operator Codes in order of priority (from 1 to 9) in case of Manual Network Selection.
- Programmable Speech Tx Gain
- Programmable Speech Rx Gain
- Selectable Incoming Call Modes - Allow, Ignore, Reject.
- SIM Card protection with a Personal Identification Number (PIN)[17].
- Single Rooftop (RT) antenna for 'High Gain'

If CDMA module is installed in your system, following features and facilities will not be applicable.

- Band Selection
- CLIR
- BCCH Selection
- Emergency Number
- Network Selection

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15. ETERNITY PENX does not support ISDN BRI Interface.
16. ETERNITY PENX does not support CDMA.
17. Not applicable for CDMA Mobile Card.

- Preferred Network Mode
- RCOC (Route Call to Original Caller)
- SIM PIN
- SIM Card Balance and Recharging
- SIM Recharge
- SMSC
- SMS Parameters
- SMS on No Reply
- SMS Server
- SMS Gateway

If CDMA module is installed in your system, DTMF Detection might not work properly.

The SARVAM UCS Mobile Interface does not offer GPRS features, Fax and Data services, and network supported services, except CLIR and USSD.

The CO Interface

The CO Interface enables the SARVAM UCS to be connected to the POTS Network. The POTS Networks across the world support various standards and differ in features. For example, some networks support Caller ID Presentation using DTMF signaling, while some support Caller ID Presentation using FSK signaling; some networks offer 600 Ohms Impedance, while others offer complex impedance.

SARVAM UCS's versatile architecture allows it to be connected to such networks differing in their characteristics. The CO Interface supports following features:

- Programmable AC Impedance - 600Ω, 900Ω and various complex impedances
- Answer Supervision/Polarity Reversal
- Selectable Disconnect Supervision - Polarity Reversal, Open Loop Disconnect
- Selectable Caller ID Presentation - DTMF, FSK
- Programmable Dialing method - Pulse/Tone (with programmable Pulse Ratio/DTMF On-Off period)
- Programmable Speech Tx Gain
- Programmable Speech Rx Gain
- Programmable Disconnect Tone Sensing
- Programmable Flash Timer

The Radio Port Interface[18]

The Radio Interface Card offers connectivity to the Radio transceiver devices/Combat-Net Radio devices. The Radio Interface Card (RIC) adds the two-way Radio functionality in SARVAM UCS, wherein the speech can be transmitted as well as received by the radio device.

Radio Transceiver or CNR radio devices interfaced with the Radio port can be operated on High Frequency (HF), Very High Frequency (VHF) or Ultra High Frequency (UHF).

Using the Radio Interface, the user can:

- Set the frequencies for receiving and sending audio messages.
- Playback the audio received over the air through the Speaker.
- Send the audio message over the air through the Microphone.
- Activate radio transmission over air using Push-to-Talk (PTT) button.

18. ETERNITY PENX does not support Radio Interface.

The Radio Transceivers remains in receiving (Rx) mode, so that the audio messages broadcasted over the air can be heard.

The E&M Interface[19]

The E&M Interface of the SARVAM UCS supports analog trunking to connect various communication equipment telephone switches, Routers, Leased Lines, etc.

Often, E&M connectivity is used to expand the system capacity (by connecting a second system with the main system) or to connect two or more remotely located Systems, forming a network of Systems.

The E&M Interface can be used for the following applications:

- Power Line Carrier Communication (PLCC) Networks, where several EPAXs are connected with each other through E&M tie lines. Refer *"PLCC-An Introduction"* to know more.
- Closed User Groups, where several System are connected with each other through E&M tie lines[20].
- System expansion, where two System are connected with each other with E&M tie lines.
- Connecting remote Systems/Satellite equipment over E&M tie lines

Also, refer the topics *"E&M Connectivity"* and *"E&M Feature Template"* to know more.

The E&M Interface can be programmed to provide Trunk Interface, a Subscriber (Station) Interface or both, as a Tie Line with the dual personality of a Trunk and a Subscriber.

The E&M Interface of the SARVAM UCS has the following features:

- Programmable Orientation Type - trunk, station, tie-line.
- Selectable E&M Interface Types - IV and V.
- Selectable Speech Interface (Audio Interface)[21] - Two-wire and four-wire.
- Selectable E&M Trunk Seizure Type[22] - Immediate, Immediate + Wink, Seizure Pulse, Seizure Pulse + Wink, Express, and Compander Control Signal.
- Selectable Address Signaling - Pulse dial (Pulse 10PPS, Pulse 20PPS) and Tone Dial (DTMF).

The Magneto Interface[23]

Magneto Telephones[24] can be connected to the Magneto Interface of SARVAM UCS. Magneto telephones are widely used by the defense establishments as field phones in front lines, and by other establishments such as railroad companies (signaling emergencies, crossings, etc.), electric utilities, pipeline companies, who need to have their networks at places that are too remote to be serviced by public telephone networks.

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19. *ETERNITY PENX does not support E&M Interface.*
20. *The Systems in a Closed User Group can be connected over ISDN T1/E1 Lines as well. Refer the topic "Closed User Group (CUG)" to know more.*
21. *The number of wires used to transmit audio signals.*
22. *This is the line protocol that defines how the equipment seizes the E&M trunk. Also, referred to as Start Dial Supervision Signaling Protocol.*
23. *ETERNITY PENX does not support Magneto Interface.*
24. *A magneto telephone is a local battery telephone set, in which signaling current is provided by a magneto hand generator. The hand generator, commonly referred to as 'crank', is located on the right hand side of the telephone set and is turned to produce energy to ring other phones or to signal the CO. The magneto, also called the generator, is used to convert the mechanical motion via the crank to produce sufficient energy to ring other phones or to signal the CO.*

SARVAM UCS can land calls from magneto field telephones on the extensions (SLT, DKP, ISDN Terminal) of the SARVAM UCS and place calls from the extensions of the SARVAM UCS on magneto telephones.

To know more about how the Magneto Card works, refer the topic *"Configuring Magneto Interface"*.

The Digital Key Phone Interface

The Digital Key Phone Interface of SARVAM UCS is for the purpose of connecting the proprietary Digital Key Phones of Matrix Comsec of the *EON* series, the proprietary Direct Station Selection (DSS) Consoles and the PC-based phone, *EONSOFT*, with SARVAM UCS.

The Digital Key Phone Interface has the following features:

- Programmable Call Capacity - up to 10 Call Appearances/Call Loops on a single DKP.
- Selectable Headset and Speaker (Hands-free) modes.
- Selectable Auto Answer mode.
- Programmable Headset Speech Tx and Rx Gains.
- Programmable Hands-free (Speaker) Speech Tx and Rx Gains.
- Programmable Key Click Volume levels.
- Programmable Key Maps for Operator, Executive, Hotel Attendant and Guests.
- 4-Level Adjustable Back light and Contrast Controls
- Selectable Ring Types.
- Ringer Volume Control.
- Programmable Ringer Modes - Ring Immediately, Ring if Idle, Ring after Delay, Ring Off.
- Ring On Speakerphone and Ring on Headset options.
- Language Support - English, French, German, Portuguese, Spanish.

The Single Line Telephone Interface

The Single Line Telephone (SLT) Interface allows any standard, two-wire, analog single line telephone instrument - rotary, pulse-tone, cordless, feature phones with or without Calling Line Identification - to be connected to the SARVAM UCS as extension phone.

The SLT Interface has the following features:

- Selectable Caller ID Presentation - DTMF, FSK
- Programmable CLIP Digit Pad Count (0-9).
- Programmable Ring Type - Trapezoidal, Sinusoidal, Low Trapezoidal, Low Sinusoidal.
- Programmable AC Impedance - 600Ω, 900Ω, 350Ω + (100Ω || 0.21uF), 220Ω + (820Ω || 120nF), 270Ω + (750Ω || 150 nF)
- Selectable Answer Signaling.
- Selectable Disconnect Signaling - Polarity Reversal, Open Loop Disconnect.
- Programmable Speech Rx Gain.
- Programmable Speech Tx Gain.
- Programmable Flash Timer.
- Programmable Loop Current.
- Fax machine connectivity.

Voice Mail System (VMS)

The SARVAM UCS application supports a full-fledged, in-skin Voice Mail System module to provide mailbox facility to all its extensions users. The Voice Mail System also forms the basis of other features like Conversation Recording and Call Taping.

Each Mailbox has the capacity of storing 15,000 voice messages. The maximum size of each Mailbox is 60,000 minutes. By default, the size of each Mailbox is set to 5 minutes. The maximum Message Length for each Mailbox is 9,999 seconds. By default, the Maximum Message Length for each Mailbox is set to 15 seconds.

The VMS Module in the SARVAM UCS is an optional module. It must be purchased separately. The VMS in ETERNITY GENX/MENX/LENX/PENX is delivered as a module. The default factory fitted 8GB Pen Drive on the CPU Card contains the VMS configuration files, voice messages for prompts and greetings. The Pen Drive is also the storage device for mailbox messages.

If required, you may use a Pen Drive of upto 64GB by replacing the factory fitted pendrive with a new one.
The key Voice Mail and Auto Attendant features supported by SARVAM UCS's Voice Mail System are:

- Programmable Mailbox Size.
- Programmable Message Length.
- Welcome Greetings according to the time of the day.
- Different voice greetings for different time zones.
- Special greetings for holidays.
- Five call transfer types: none, blind, wait for ring, wait for answer, and screened.
- Dial by extension.
- Dial by name.
- Personalized greetings for each mailbox.
- Individual mailbox size.
- Call forward to voice mail.
- Message forwarding.
- Distribution lists.
- Broadcast message.
- Message Wait Notification.
- Redirecting messages.
- Message Wait Notification via Email and Call.

SARVAM UCS's Voice Mail System also forms the basis of other features like:

- Conversation Recording
- Call Taping
- Voice-guided Wake-up Calls and Reminders
- Message Wait Notification
- Call Transfer to Mailbox
- Call Forward to Voice Mail
- Department Calls - Mailbox for Department Groups

The Data Port Interface[25]

The Data Card supports four Ethernet 10/100mbps interfaces. Ethernet data coming to ports can be mapped to 2Mb streams. It can map data to 60 channels of E1 interface. The remaining channels of E1 can be used for voice

25. ETERNITY PENX does not support Data Port Interface.

applications. The Data Card has a 4-port Ethernet switch on board, which can aggregate multiple data streams to PCM streams of the system.

Computer

You can connect a standalone computer with the SARVAM UCS over the RS232 serial Communication Port/s[26] of the system.

You can also connect SARVAM UCS to a standalone computer or to a LAN Switch over the LAN/WAN Port of the SARVAM UCS.

PC connectivity is required to:

- access the web-based configuration interface JEEVES.
- set up and run software applications such as Property Management Software (PMS), Call Accounting Software (CAS) in the Hotel Application from Web JEEVES interface.
- to run the Front Desk User Wizard of the Hotel Application.
- capture and download Station Message Detail Records (SMDR): SMDR reports, SMDR Online and SMDR Posting.
- capture and download System Activity Log and System Fault Log, and Hotel Motel Activity Log in the Hotel Application.

26. *ETERNITY GENX supports two Communication Ports — COM and USB to COM. ETERNITY MENX/LENX/PENX supports only one communication port —USB to COM. For details, see "Communication Ports".*

CHAPTER 4

Before You Start

Before you begin the installation of the system, make sure that the required telecom wiring has been done.

The number of extensions you require and their location will determine the type of cabling you require on your premises.

We recommend that you plan the wiring and the installation of the system according to your current and expected future requirements.

Before you begin to install and set up the hardware of the system, make sure you have the following items:

- A Main Distribution Frame (MDF)
- A suitable location to install the Main Distribution Frame and the system.
- Cables for trunk lines and extensions.
- The Cards supported.
- One or more Single Line Telephone or Digital Key Phones for testing.
- Power supply.
- One or more active Two-wire trunk lines for test calls.
- A modem for the ISDN T1E1PRI line.
- An NT1 termination device for the ISDN BRI line.
- Appropriate cables and connectors to set up and test the Ethernet interface of the system and the LAN connection.
- A standalone PC or a PC connected in a LAN that can PING the system's IP Address.
- A SIM card to test mobile network connectivity.
- A SIP Account to test VoIP connectivity.

Well begun is half done; plan your hardware installation well.

The Main Distribution Frame (MDF)

The MDF connects outside telephone lines coming from the local exchange, on one side and the internal (System) lines on the other.

In simple form, the MDF is a special metallic frame designed and constructed with columns of receptacles to firmly hold the termination modules for the trunk and extension cables.

The cables or trunk lines to/from the Public Telephone Exchange terminate on the line side and cross connections (jumpers) run to the opposite (System) side of the MDF. From those terminals, a multi-core cable runs from a second set of terminals into the System.

A multi-core cable runs from the System into the MDF. From the distribution frame, the smaller cables run into each individual extension telephone outlet or socket (RJ11 or RJ45).

In a multi-storied building or on a widely spread out premises, it is common to have more than one distribution frame, called the Intermediate Distribution Frame (IDF) on each floor, to provide the connection between the MDF and the individual telephone wiring. IDFs function as wiring points to gather and distribute wiring. IDFs are used when a large number of extensions are to be connected and the wire runs extend over hundreds of feet; hence the distance is too great to economically terminate every extension individually to the MDF.

- Select a suitable MDF (and IDF, if required) with the standard lead-in cable termination KRONE modules.
- Ensure that the MDF complies with the local building telecom wiring Guidelines, Rules and Regulations.
- Select an appropriate site to install the MDF.

Location of the MDF

- The MDF is normally installed inside the building in a location and position which is free from the ingress of dust and moisture, and which is not subject to damp or humid conditions.
- This also applies to MDF installed outside the building. It must be protected from exposure to weather conditions, dust, dampness and humidity
- Do not install the MDF in any of the following locations:
    - In washing or toilet facilities, boiler/plant/machine rooms or any area subject to corrosive fumes and fluids;
    - In fire escape stairways;
    - Within a cupboard containing a fire hose reel;
    - Within any refrigeration room or sauna heater room;
    - Near any water feature or water body like fountains, sprinklers, a bath, shower or other fixed water container, a swimming pool, paddling pool, spa pool or tub; or any area where hosing down operations are carried out.
    - In a high voltage electrical switch room or near a heavy voltage transformer.
- The MDF should be robust and securely attached to a permanent building element such as a wall, floor or column. Do not mount the MDF on movable elements such as hinged panels or wheeled trolleys.
- Provide adequate lighting in the place where the MDF is located.
- Provide adequate space around the MDF where any person is required to pass to enable safe and convenient access to the MDF and ready escape from the vicinity under emergency conditions.
- Any room containing the MDF must not require the use of a tool, key, card, number pad or the like to exit the room. Ensure a quick hurdle-free exit from such a room.
- The MDF or the enclosure in which it is located should have the provision for securing with a key, lock or tool. External MDF should be adequately secured against vandalism and access by children or unauthorized persons.
- The MDF enclosure should be designed so as to prevent access to live parts by unqualified persons and should be free of exposed sharp edges.
- The MDF enclosure should be protected from insects and rodents.

Selecting the Installation Site for the system

The system maybe be mounted on a table or wall. Refer to the mechanical dimensions of the model you have, when selecting the site for installation, and deciding whether to mount the system on a table top, or on the wall, or on a rack.

Select an appropriate site to install the system taking into consideration the following recommendations and precautions:

- The site of installation should be well-ventilated, moisture and dust free, and not exposed to direct sunlight, heat, excessive cold or humidity.
- The site should be equidistant from all the extensions to simplify cabling network and reduce cabling costs.
- The system should be installed at a height of at least 3.5 feet from the ground. Installation at this height makes preventive or corrective maintenance tasks easy.
- The system should be installed away from any source of electromagnetic noise such as any radio equipment, heavy transformers, faulty electric chokes of tube-lights, any device having faulty coil, etc.

Read “Protecting the System and Yourself”.

Selecting Cables

- Select standard good quality telephone cables with 0.5 mm conductor diameter for the internal as well as over-head cabling.
- Use twisted pair wires to reduce interference.
- Use separate cable conduits for electrical and telephone cables.
- The length of the cables must not be too long. They must have minimum number of joints. This will help you detect cable faults easily.
- Maintain cable records so that cables and cross-connections on the MDF can be correctly identified and connected. The records should be in a clear, legible and updatable format.

Selecting Extension Telephones

Select appropriate telephone instruments to be connected as extension phones. You may connect:

- any of the models of the proprietary Digital Key Phone (DKP) of the EON series.
- Any standard telephone instrument like rotary phone, Pulse/tone switchable push-button phone, Feature phone or Cordless phone. So, you can also use your existing telephone instruments.

You can connect DKPs with DSS of the EON series for Operator/Receptionist/ Front Desk/Senior management extensions.

Providing Power Supply Source

- ETERNITY LENX operates on 48VDC (+20% to -15%).
- The ETERNITY GENX/ MENX/PENX work with input voltages ranging between 100-240VAC or 48VDC (250W for GENX and 500W for MENX).
- Arrange for a separate power point and switch, close to the system.
- Power supply for the system must be separate from other heavy electrical loads like Air-conditioners, heaters, welding machines, electrical motors, etc.

Terminating Trunk and Extension Cables on the MDF

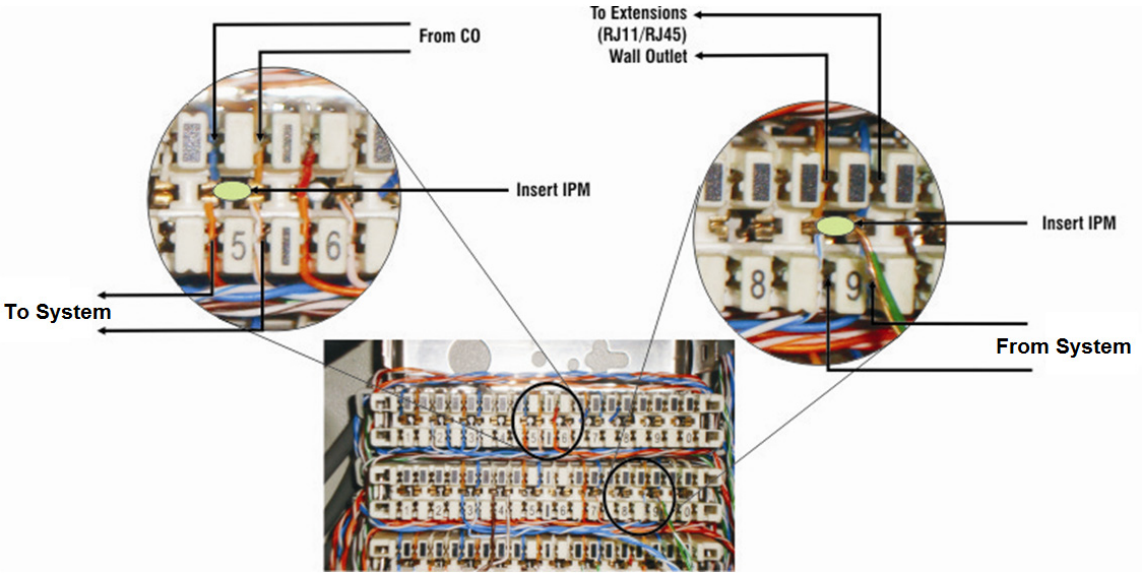
- Terminate the CO Trunk Line cables from the CO (public telephone exchange) and E&M cables into the 'Trunk Lines' side of the MDF using the punch tool for Krone modules.
- Terminate all the extension cables (connected to the wall sockets/outlets) into the 'Station Lines' side of the MDF using the punch tool for Krone modules.
- Label the trunk and extension line cables for easy identification and keep a record of the trunk and extension lines in an updatable format.

  Where multiple wiring cabinets/distribution frames are used, label each frame and reference its number on the corresponding outlet.
- Install Primary Protection modules with Gas Discharge Tubes (GDT) and fuses on entry points for all trunk lines. This is to protect the system from heavy voltages from trunk lines and overhead stations.

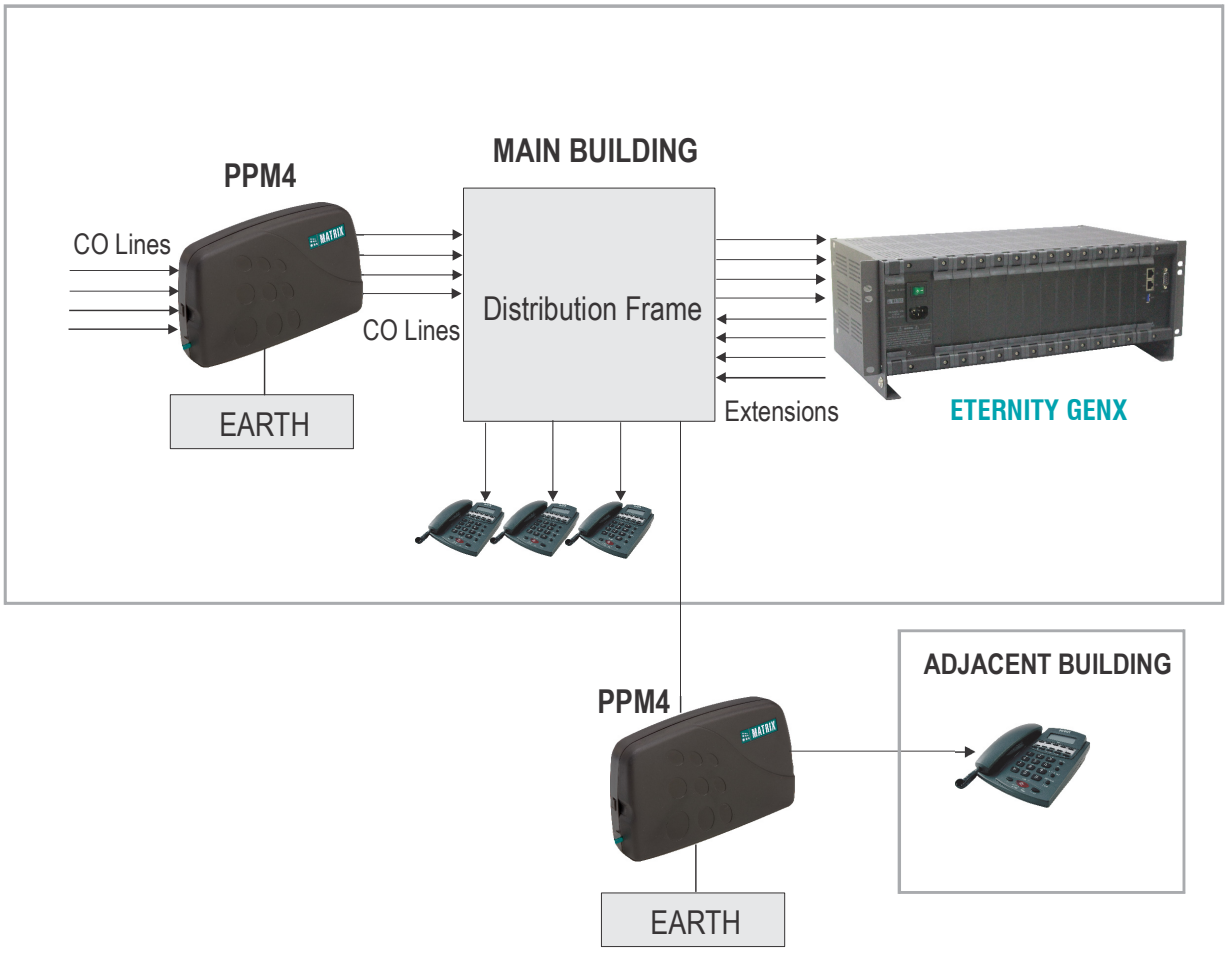
  The product warranty does not cover damages resulting from lack of primary protection on trunk lines.
- It is recommended that you also install Primary Protection modules with GDT and fuses on all Extension lines, particularly off-premise extensions, and E&M ports.

  For this, you are recommended to use the Primary Protection Module (PPM4) supplied by Matrix.

- A typical connection between a System and the MDF is illustrated in the figure below.



You are recommended to use the "Primary Protection Module - PPM4" supplied by Matrix.



Connecting Primary Protection Modules to Trunk and Overhead/Off-Premises Lines

It is necessary to protect the system from heavy voltages entering the system from the Trunk Lines and overhead stations.

The protection can be in the form of surge suppressor devices like Gas Discharge Tubes (GDT), MOVs, Fuses, etc.

Input Protection Modules (IPM)

Install IPM on the Krone Modules of the MDF. Input protection modules are for analog input channels to protect against over-voltages that may be applied between any two input connectors or between an input connector and the ground.

Primary Protection Module - PPM4

Matrix provides Primary Protection Modules (PPM) consisting of four PPM circuits. The PPM4 contains Gas Discharge Tubes and Fuses.

The Gas Discharge Tube is an over voltage protection device. It has three terminals. It is connected parallel to the CO Line or the overhead station cable. The third terminal is connected to a telecom earth. When the voltage between any of the two terminals exceeds the permissible limit (general 150V), the gas in the device begins to conduct and the terminals with the earth terminal. Heavy voltage passes to the earth instead of entering the system, thereby protecting the system.

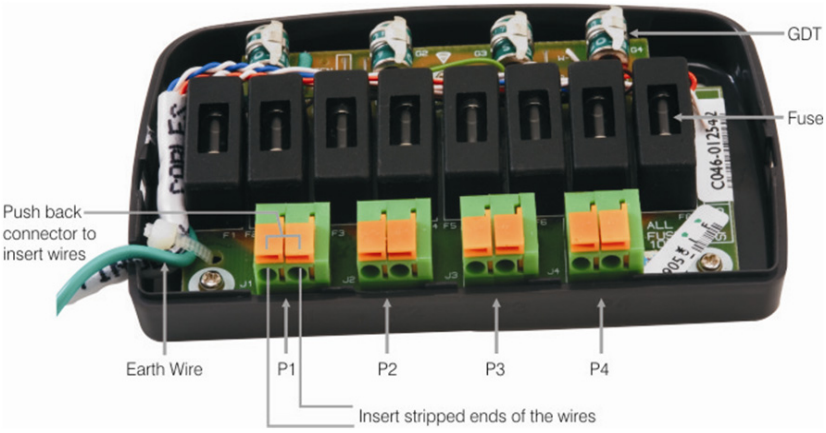
The Fuses in the PPM4 are an over current protection device. Whenever the current builds up beyond the permissible limit, (generally 100mA), the fuse opens to protect the circuit ahead.

PPM4 must be properly earthed to work well. It is recommended that PPM4 be connected to a separate telecom earth (ground).

Telecom earth is a dedicated earth (ground) only for the System. A dedicated earth greatly reduces the risk of back voltage.

Installing PPM4

Refer the block diagram above for the location of the PPM4.



1. Unpack the PPM and check the package contents.

2. Select an appropriate location for the PPM4. Refer the block diagram above when deciding where to place the PPM4. Also, take into consideration the length of the cables of the PPM4

.

3. Use the Mounting Template supplied with the PPM4 to drill holes on the wall to fix the PPM in the selected location. Fix the screws supplied with the PPM4 into the drilled holes, with their heads protruding from the wall.

4. You may mount the PPM4 first and connect the cables OR you may connect the cables first and then mount the PPM4.

5. To connect cables, press the snap fits on both sides of the PPM4 to release the cover. Remove the cover.

6. Connect the Earth wire (green wire) to the Telecom Earth.

7. Now connect the CO Trunk wires from the CO side into the PPM4 port connectors marked as P1, P2, P3 and P4.

8. To do so, strip off about half a centimeter of the insulation of the wire ends of the first pair of CO Trunk you want to connect to the PPM4.

9. Push back the (orange-color) levers of the connector of port P1, using a blunt pin or a small flat screw driver or your thumbnail.

10. Insert the stripped ends of the two wires into the two (green-color) openings of the connector, with one wire in each opening.

11. Release pressure on the levers. Both wires will be held in place by spring clamp action.

    To remove the wires,
    - push back the levers.
    - pull out the wires gently.
    - release pressure on the levers.

12. Now, repeat the above steps to connect the other CO Trunk wires from the CO side into the connectors of the ports P2, P3, and P4.

13. Now, terminate the wire pairs emerging from the PPM4 multi-pair cable into the 'Trunk Lines' side of the MDF using the punch tool for Krone Modules.

    Refer the following table for connection details of the wires:

PPM4 Port	Color
P1	Blue and White
P2	Orange and White
P3	Green and White
P4	Brown and White

14. Replace the cover of the PPM4 by pressing back the snap fits on both sides.

15. Mount the PPM, if not done already.

Protecting the System and Yourself

The System is an electronic device. When you handle any electrical or electronic equipment, you are in a situation that could cause you bodily harm, besides damage to the product. When handling any electronic equipment, you must be aware of the safety hazards involved in electrical circuitry and the standard practices for accident prevention.

When using any telephone equipment, take every safety precaution to reduce the risk of fire, electric shock and injury to persons. Read and understand the precautions, dos and don'ts of handling this product listed below.

These instructions are by no means exhaustive. So, take all the necessary precautions for handling electronic and electrical appliances. Your safety and that of the others lies in your hands.

Location

- Do not place this product in any of the following locations:

    - Near a water source like a wash bowl, kitchen sink, laundry tub, near a swimming pool, or in a wet basement.
    - In places where dust, oil, corrosive fumes may come in contact with the system.
    - Any area where it is exposed to direct sunlight, heat, excessive cold or humidity.
    - On movable or unstable surfaces, which may cause the product to fall and get damaged.
    - Any area where shocks or vibration are frequent or strong.
    - Near High-Frequency generating devices such as Electric Welder, Sewing Machine or and Microwave Oven.

- Do not leave cables exposed on the ground where they may be trampled upon, or get damaged by entangling with feet or pressure from other heavy objects.

Power Supply

- This product should be operated with proper supply voltage. If you are not sure about supply voltage, contact authorized dealer.

- The System does not work in isolation from the environment. Power is fed to the system for functioning of the system. It has several interfaces like trunk lines and extensions, PC interface, etc. So there are chances of heavy voltages entering the system through these interfaces. Also, static charges could find their way through the system components.

Protect the system from heavy voltages from the mains

- The System is designed to work with input voltages ranging between 100-240VAC. The Power Card of System have a 'switch mode' design to support such a wide range of operating voltage.

Protect the system from heavy voltages on the trunk lines and the overhead stations

- The System may be damaged by heavy voltages entering the system from trunk lines or from overhead stations. These heavy voltages may enter the trunk lines and from overhead stations due to:
    - Heavy voltage line falling on the CO line or on the overhead stations cable. A dangerous surge can occur if a telephone line comes in contact with a power line.
    - Lightning/Thunderbolts.
- To protect System from these voltages, use Primary Protection/Surge Protectors on the trunk and long distance extension lines to protect the system from lightning and electrical surges.
- Install any standard Input Protection (punch down protection) on the Krone Modules of the MDF or the "Primary Protection Module - PPM4" supplied by Matrix at entry points for all CO trunks lines and all overhead stations. The product Warranty does not cover damages resulting from heavy voltage on CO lines and overhead stations!
- It is recommended that you install the PPM on the MDF, as MDF cables from the CO are terminated on the System MDF.

Protect the system from Lightning

- To protect System from extremely high voltage currents associated with lightning strikes, install a lightning protector on an outside (CO) line.

Protect the system from static charges

- Every person carries some static charge in his/her body depending upon body composition and the environment around them. Most of the times, this charge finds its way to the earth when the person touches any object which is grounded, or when the person is barefoot.
- Generally, persons installing or handling electronic and electrical equipment take precaution to wear appropriate footwear to get protection from electric shocks. Doing so, the static charge accumulates in his/her body and does not find its way to the ground. But when such a person touches any of the electronic cards, the static charge finds its way through the electronic components thereby causing damage to the cards.
- So, the person installing or servicing the system must provide a path to the static charges, by wearing an antistatic belt, which is properly earthed.

Protect the system from heavy voltage on the communication cable

- The Communication Port (COM Port) is provided on the System for connecting a PC.
- If an electrical wire carrying heavy voltage accidentally shorts with this cable, heavy voltages can damage the communication port.
- It is recommended that the communication cable (connecting System and the PC) be run through the conduit carrying telephone cables or through a separate conduit.

Telecom Earth (Ground)

- The Earth (Ground) is the most important safety procedure to prevent electrical shocks and fires. It protects from lightning strikes, electrical transients, static discharges, electromagnetic interference and electrical hazards.
- Ensure that a proper electrical earth and a telecom earth are in place for the safety of people and the system. Telecom earth is a dedicated earth for the System/any other telecom equipment.
- Provide a separate Telecom Earth (Ground) to the system installation. Providing a separate earth to the telecom equipment eliminates the possibility of any back-voltage on the earth.
- Refer “How to Make the Telecom Earth” for instructions on making the perfect earth (ground).

Shock and Fire Hazard

- Always wear a properly earthed, electrostatic discharge preventive wrist strap/belt while handling the system and its cards to prevent damage to the system and harm to yourself.
- Do not open the system in power On condition.
- Slots and openings in the cabinet and the back or bottom are provided for ventilation, to protect the system from overheating. These openings must not be blocked or covered.
- Never insert or push objects of any kind into this product through the cabinet slots as they may touch dangerous voltage points or short out parts which may result in fire or electric shock.
- Do not allow anything to rest on the power cord. Do not locate this product where the cord will be trampled upon or get entangled.
- This product is equipped with a plug having a third (ground) pin, which fits only into a grounding-type outlet. This is a safety feature. So, if the existing outlet is not a three-pin and or if you are unable to insert the plug into the outlet, have the outlet replaced by the electrician.
- Do not overload wall outlets and extension cords as this can result in the risk of fire or electric shock.
- Do not disassemble this product. Opening or removing covers may expose you to dangerous voltages or other risks. Incorrect reassembly may cause electric shock when the appliance is used. Take the product to a qualified technician when service or repair work is required.
- Avoid using a telephone (other than a cordless type) during a storm, to prevent electric shock from lightning.
- Do not use the telephone to report a gas leak in the vicinity of the leak so as to prevent the risk of fire.

External Devices

- When you connect external devices like headset, sensors, paging devices, telephone instruments, cables, connectors, etc., ensure that they are of standard make and good quality, so that the functioning of the system is not affected.
- Matrix does not guarantee the performance of external devices that are not supplied by it.

Cleaning and maintenance

- Unplug this product from the wall outlet before cleaning.
- Do not use liquid cleaners or aerosol cleaners. Never spill liquid of any kind on the product.
- Use a dry cloth for cleaning.

Service and Repair

- Unplug this product from the wall outlet and refer servicing to a qualified service person under the following conditions:
    - When the power supply cord or plug is damaged or frayed.
    - If liquid has been spilled into the product.
    - If the product has been exposed to rain or water.
    - If the product has been dropped or the cabinet has been damaged.
    - If the product exhibits a distinct change in performance.

Battery

System contains a 3VDC/15mAh Manganese Lithium Coin Battery (ML 1220 - Rechargeable) of diameter 12.5mm and height 2.0mm. The Battery is located on the CPU Card. The Battery should be replaced only by authorized dealers of Matrix. End Users must not attempt to replace it.

Caution: *There is risk of explosion if the Battery is replaced in an incorrect manner. Please dispose-off used Batteries.*

Disposal

- This product must be disposed according to the national laws and regulations prevailing in the country where it is installed.

Warning for RF Safety

The product complies with RF exposure guidelines as per standard FCC 47 CFR part 2. However, please observe the following precautions.

- Make sure that the RF Antenna is installed at least 20 cm away from other electronic and radio transmission devices.
- Make sure that the RF Antenna is installed at a place at 20 cm away from people's vicinity.
- Do not place magnetic storage media near the product.
- People carrying medical implants like cardiac pacemakers are advised to maintain appropriate distance from the system. They are also advised to avoid being in the vicinity of the product for a long time.

CHAPTER 5

Installing ETERNITY LENX

- *The Matrix ETERNITY LENX is to be installed by persons trained and experienced in telecom wiring.*
- *The person installing the ETERNITY LENX must be familiar with trunks, physical wiring of the MDF on both the exchange (System) side and the line side (CO).*
- *When installing any equipment, make sure that you take all the necessary precautions for handling electronic and electrical appliances. Follow proper procedures for static electricity, while handling the system and its cards to prevent damage to the system and harm to yourself.*
- *Use a grounding mat and wear an anti-static strap/belt. Read the do's and don'ts listed in "Protecting the System and Yourself".*
- *If you have complied with the requirements and instructions described in "Before You Start", you may now begin the installation of your ETERNITY LENX.*

Hardware Version V1R2 and earlier

ETERNITY LENX can be wall mounted, rack mounted or placed on a table. You can also affix wheels to the system; to move it like a trolley.

ETERNITY LENX has two racks and a total of 27 Universal slots. The first rack has a total of 15 slots. The first slot from the left is a fixed slot and the last two slots to the right are fixed slots and the remaining 12 are universal slots.

The second rack has a total of 16 slots. The first slot from the left is a fixed slot and the remaining 15 are universal slots.

The Matrix ETERNITY LENX is shipped factory fitted with the Power Supply Card, the CPU Card in their respective fixed slots (refer the section "Know Your SARVAM UCS"). VoIP and VMS are in-skin to the CPU Card. Hence, separate VMS and VoIP Cards are not required.

The Slave Cards - BRI, T1E1PRI, GSM, DKP, CO, SLT, E&M, Radio, Data, Magneto, E1FO - are shipped separately as per the order placed by individual customers. These Cards are installed in any of the Universal slots.

ETERNITY LENX supports ETERNITY LE PS48VDC card, ETERNITY MENX-LENX CPU card, ETERNITY LE SLT48 card, ETERNITY LE ILC48 card and all other cards of ETERNITY ME (except ETERNITY ME Power Supply card.)

If you upgrade the system firmware to V1R3, the Expansion Slots license will be applicable for the universal slots. No universal slots will be functional by default. You must purchase the SARVAM EXP4 ENT license to activate the universal slots as required.

For details, see “Expansion Slots” under “License Management”.

Hardware Version V1R3

ETERNITY LENX can be wall mounted, rack mounted or placed on a table. You can also affix wheels to the system; to move it like a trolley.

ETERNITY LENX has two racks and a total of 27 Universal slots. The first rack has a total of 15 slots. The first slot from the left is a fixed slot and the last two slots to the right are fixed slots and the remaining 12 are universal slots.

The second rack has a total of 16 slots. The first slot from the left is a fixed slot and the remaining 15 are universal slots.

If you have upgraded the old ETERNITY LENX system to V1R3, the Expansion Slots license will be applicable for the universal slots. No universal slots will be functional by default. You must purchase the SARVAM EXP4 ENT license to activate the universal slots as required.

If you have purchased the new ETERNITY LENX system with the firmware V1R3, the Expansion Slots license will be applicable for the universal slots. The first eight universal slots after the power supply card in the first rack will be functional by default. If you require more functional universal slots, you must purchase the SARVAM EXP4 ENT license.

Each SARVAM EXP4 ENT license will provide the activation for next four universal slots in sequence.

For details, see “Expansion Slots” under “License Management”.

Hardware Version V1R4 and later

ETERNITY LENX can be wall mounted, rack mounted or placed on a table. You can also affix wheels to the system; to move it like a trolley.

ETERNITY LENX has two racks and a total of 27 Universal slots. The first rack has a total of 15 slots. The first slot from the left is a fixed slot and the last two slots to the right are fixed slots and the remaining 12 are universal slots.

The second rack has a total of 16 slots. The first slot from the left is a fixed slot and the remaining 15 are universal slots. However, since the CPU Card/s have 1K IDT Switch, out of the 15 universal slots in four slots from the right the following cards cannot be affixed — ETERNITY LE CARD SLT48, ETERNITY ME CARD E1FO Dual and ETERNITY ME CARD T1E1PRI Dual. In the remaining slots any card can be affixed.

The first eight universal slots after the power supply card in the first rack will be functional by default. If you require more functional universal slots, you must purchase the SARVAM EXP4 ENT license.

Each SARVAM EXP4 ENT license will provide the activation for next four universal slots in sequence. For details, see “Expansion Slots” under “License Management”.

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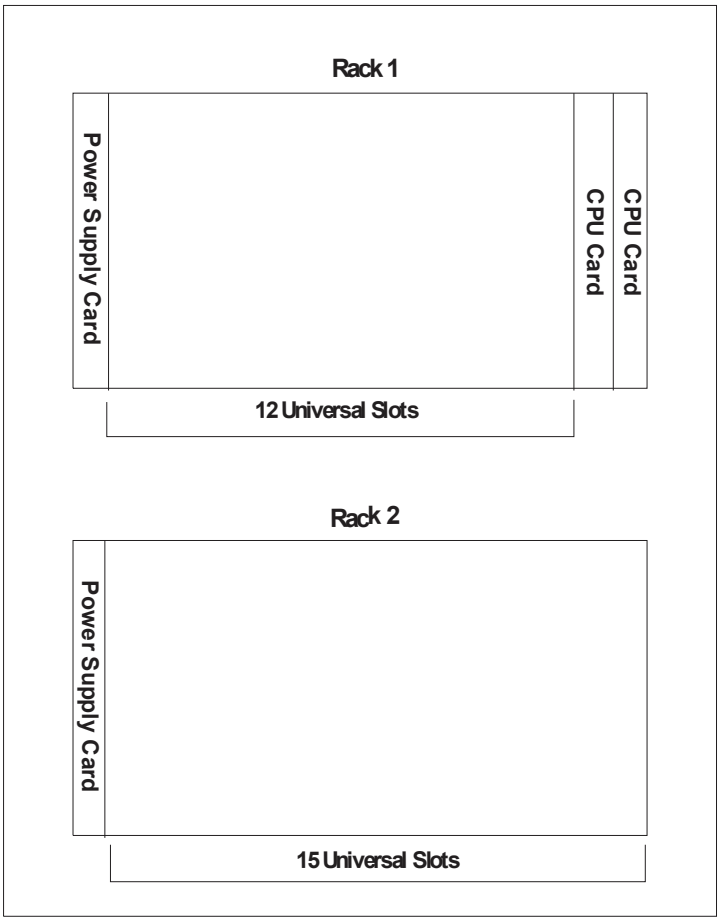
The Matrix ETERNITY LENX is shipped factory fitted with the Power Supply Card, the CPU Card in their respective fixed slots (refer the section "Know Your SARVAM UCS"). VoIP and VMS are in-skin to the CPU Card. Hence, separate VMS and VoIP Cards are not required.

The Slave Cards - BRI, T1E1PRI, GSM, DKP, CO, SLT, E&M, Radio, Data, Magneto, E1FO - are shipped separately as per the order placed by individual customers. These Cards are installed in any of the Universal slots.

ETERNITY LENX supports ETERNITY LE PS48VDC card, ETERNITY MENX-LENX CPU card, ETERNITY LE SLT48 card, ETERNITY LE ILC48 card and all other cards of ETERNITY ME (except ETERNITY ME Power Supply card.)

Illustrated below is the position of the fixed and universal slots in ETERNITY LENX.

ETERNITY LENX



The extreme left slot of each rack is reserved for the Power Supply Cards and the two extreme right slots of the first rack are reserved for the CPU Cards.

Follow the installation instructions for Cards described here also when you expand the system (add more Cards) or remove or swap Cards for maintenance and repair.

1. Unpack the box. Check the package contents (see "Packing List"). Contact your Dealer/Distributor if any of the items is missing, faulty or damaged. Do not discard the packaging material.

Mounting the System

2. Decide where to mount the ETERNITY LENX - on a table or wall - taking into consideration the mechanical dimensions and the weight. If mounting the system on a wall, you may refer the mechanical dimensions and the Mounting Template for drilling holes at appropriate places on the wall. Make sure the system orientation is horizontal.

3. When installing the system in a rack, allow adequate space between the system and other units for air circulation. Make sure the system orientation is horizontal.

4. Mount the system at the selected site. Make sure that the system is placed such that you have full access to the front and back panels. The holes in the panels are provided for ventilation; Make sure that these are not blocked, to prevent overheating.

Connecting Input Power Supply

5. Ensure that a proper electrical earth and telecom earth are in place.

6. Check the voltage at the power point from where the supply is to be given to the system. It should be as per the specifications. Earth the system properly. (Refer “How to Make the Telecom Earth”).

Inserting Cards

7. Make sure that the ETERNITY LENX power is off and the power cord is unplugged.

8. Select a free slot from the universal slots.

9. Unscrew and remove the filler bracket that covers the card-slot opening of the slot you intend to use.

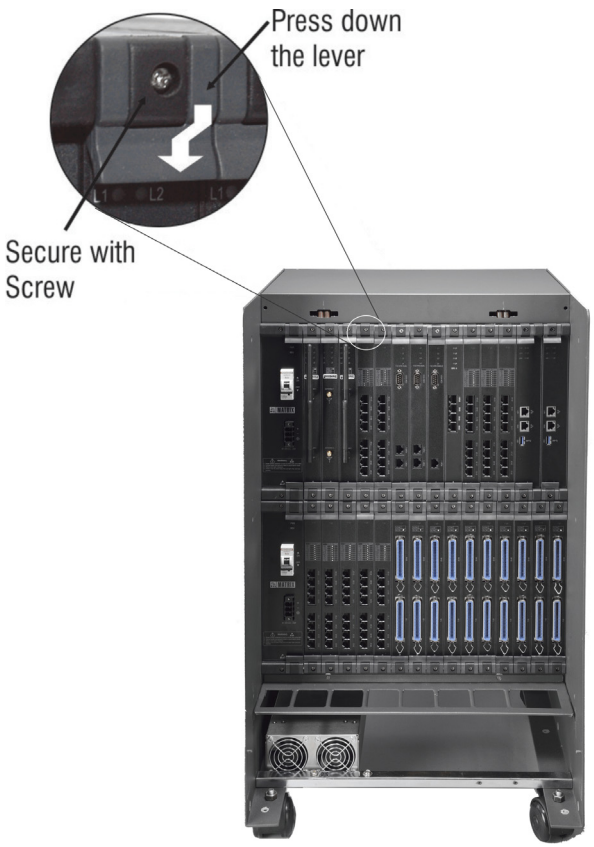
10. Hold the card with the connectors facing you. Do not grab the card from both ends.

11. Slide the card into the slot, along the guide rails provided for each slot at the top and bottom planes.

12. Ensure that the cards are inserted deep enough for all the connector pins on the cards make complete contact with those of the motherboard on the backplane.

Do not force the card into the slot. Doing so can damage the card or the slot connector.

13. When the card is firmly seated in the connector, push down the levers on the card mounting bracket and secure the card with the screw provided.



14. Tighten the screws on either side of the bracket.
15. Following the above steps, install each card into the universal slots.

    Detailed installing instructions are provided for each card - DKP, SLT, CO, ISDN BRI, ISDN T1E1PRI, GSM, E&M, E1FO, Magneto, Radio - later in this section. Refer to them when installing each card type.
16. To remove a card:

    - Switch off power supply, unplug the power cord.
    - Disconnect any cables connected to the card.
    - Remove the screws from the card-mounting bracket.
    - Lift the levers on the mounting bracket to release the card.
    - The card will emerge out of the slot.
    - Grasp the card by its mounting bracket, and ease it out of its slot.

    - *If you are removing the card permanently or for a certain period of time, install a filler bracket over the empty card opening in the chassis.*

    - *Installing filler brackets over empty card-slot openings is necessary to protect the system from dust, dirt, insects and damage.*

17. Using the cables supplied with each card, and terminate the cables in the Main Distribution Frame (SLT, DKP, CO, and E&M lines), the NT1 device (ISDN BRI lines), ISDN Modem (ISDN PRI Lines), as applicable.

    Lead the cables neatly and tangle-free into the MDF.

18. After you have completed inserting and connecting the cards, power ON the system and observe the Reset cycle and the LED pattern of each card, where applicable.

The Power Supply Card

Matrix ETERNITY LENX supports PS48V Card with 48VDC (+20% to -15%) as Input DC Power Supply Voltage. A Float cum Boost Charger (FCBC) or 48VDC power supply is required to feed 48VDC power (Emerson Power Supply supplied by Matrix or any other) to the card. The FCBC works on input AC mains. This card is available in DC to DC 1000W power.

The card has two LEDs, a power ON/OFF Switch, protection MCB and a 3-way termination block for connecting the power cord. The PS48V Card provide DC output voltages as: +3.5V, +5.0V, -27V and -85V. These are indicated by LEDs.

The ETERNITY provides Redundancy and Hot Swap option for the Power Supply card.

The ETERNITY LENX supports two PS48V power supply cards. Whenever there is a fault in one, the other takes over the control, providing uninterrupted communication.

Installing the Power Supply Card

The Power Supply Card is located in a fixed slot. No other card can be inserted in this slot.

The Power Supply Card is delivered factory fitted, when you buy the system. However, if you want to remove the card for the purpose of maintenance or replace it with a new one, please follow the instructions below:

1. Unpack the Power Supply Card and verify the package contents.

   If already installed, switch OFF power supply, unplug the power cord. Remove the screws securing the card. Lift the levers on the mounting bracket to release the card. As the card emerges from the slot, ease it out of the slot.

2. Insert the Power Supply card into the guide rails into the slot designated for the Power Supply Card. Make sure that the card is inserted deep enough to make perfect contact with the connectors on the motherboard at the backplane.

3. Now, press down the levers on the card mounting bracket to secure the card in its slot.

4. Secure the card in the slot by screwing the bracket on both ends.

5. If installing the **PS48V card**, connect the Float cum Boost Charger (FCBC) or Emerson PS or any 48VDC power source. Terminate the power cord from the FCBC output or the Emerson PS into the 3-way termination block (female connector) on the PS48V card.

   Polarity is critical. Ensure that the wires are connected with the correct polarity. Follow the standard color codes used by FCBC manufacturers:

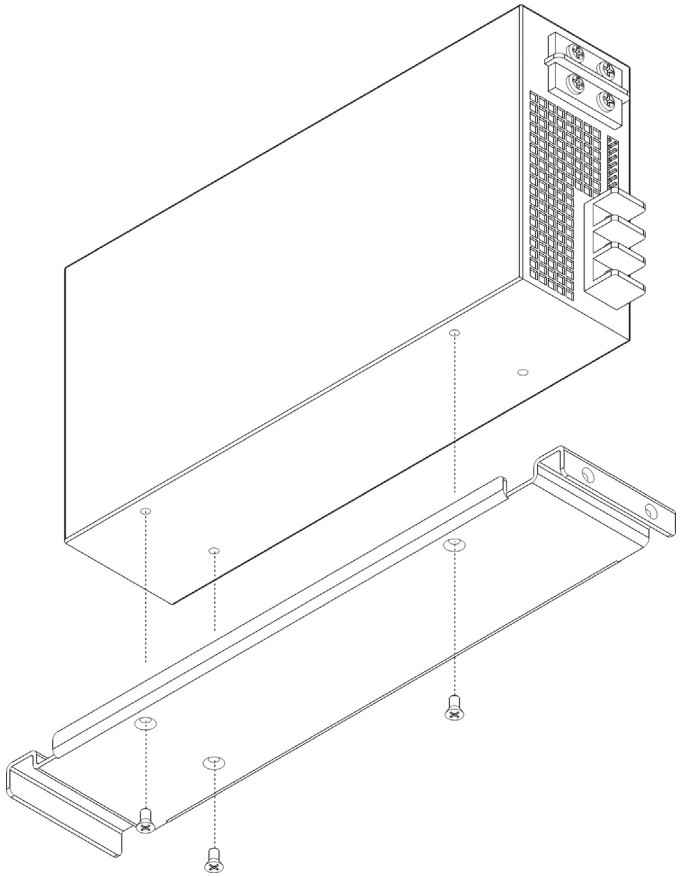
Color	Signal
Red	+48VDC
Black	GND
Yellow-Green	Earth

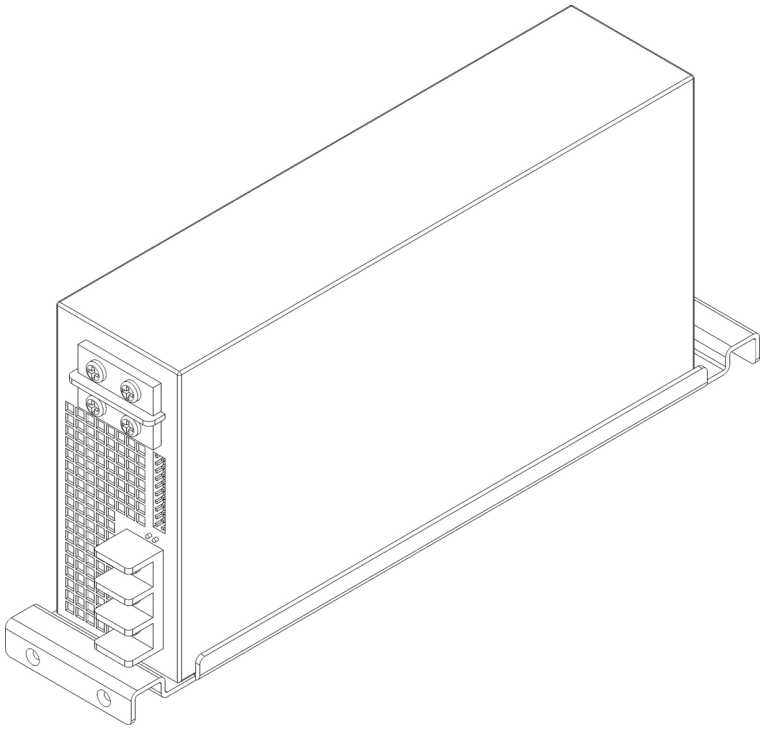
If connecting the Emerson PS, refer “Connecting an Emerson Power Supply to the ETERNITY LENX”.

It is recommended that you measure the voltage before connecting the power cable to the power supply card. Ensure that the earth is connected.

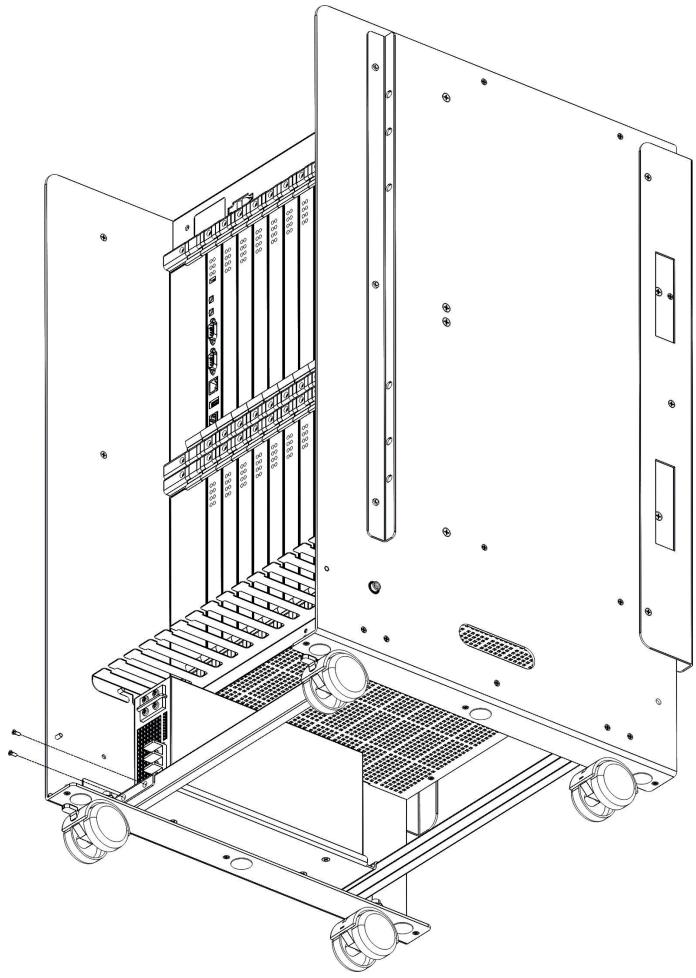
Connecting an Emerson Power Supply to the ETERNITY LENX

- Unpack the Emerson PS and the Emerson PS Tray. Check the package contents. Refer “ETERNITY LENX”.
- Affix the Emerson PS unit onto the Emerson PS tray supplied by Matrix with the help of three screws.





- Now insert the tray into the last slot of the system rack and affix the tray onto the rack with the help of screws.



- The Emerson PS has an Input Terminal and an Output Terminal. The Input Terminal is used to connect the Emerson PC to the Mains Power Supply. The Output Terminal is used to connect the Emerson PS to the 48VDC Power Supply Card.

- The Input Terminal has three screws. Unfasten and remove the screws. Use the Power cable (not supplied by Matrix) to connect the Emerson PS to the Mains. The cable has three wires — Red, Black and Yellow-Green. At one end of the cable a plug is connected and the wires are free on the other end. A lug is affixed to each wire.

- Insert the lugs onto the screws as per the color codes mentioned below and fix the screws back.

Color	Label on the Emerson PS
Red	L (Line)
Black	N (Neutral)
Yellow-Green	⏚ (Earth)

- Insert the Plug into the Main power source.

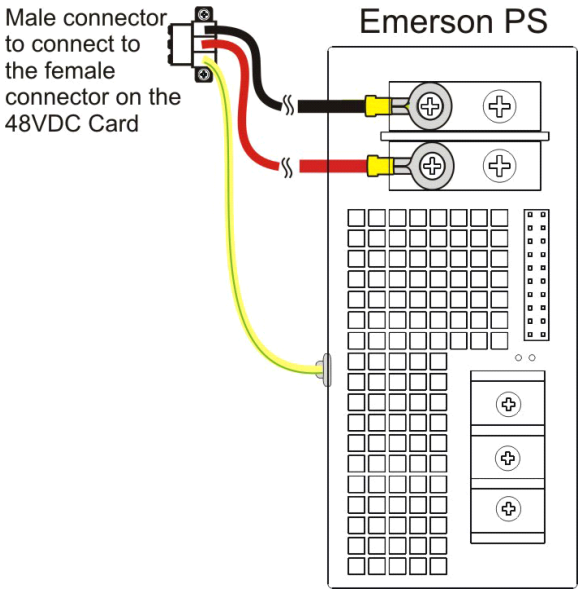
- The Output Terminal has four screws — two to carry V+ signaling and two to carry V- signaling as the system supports two 48VDC PS cards.

- A cable is supplied by Matrix with each 48VDC Power Supply card to connect the card to the Emerson PS. The cable has three wires — Red, Black and Yellow-Green. At one end of the cable a male connector is connected and the wires are free on the other end. A lug is affixed to each wire. A separate screw is placed on the top cover of the Emerson PS for earthing.

- To connect one card, unfasten and remove two screws, one that carries V+ signal and the other one that carries V- signal. Similarly, also remove the screw provided for earthing. Insert the lugs onto the screw as per the color code given below:

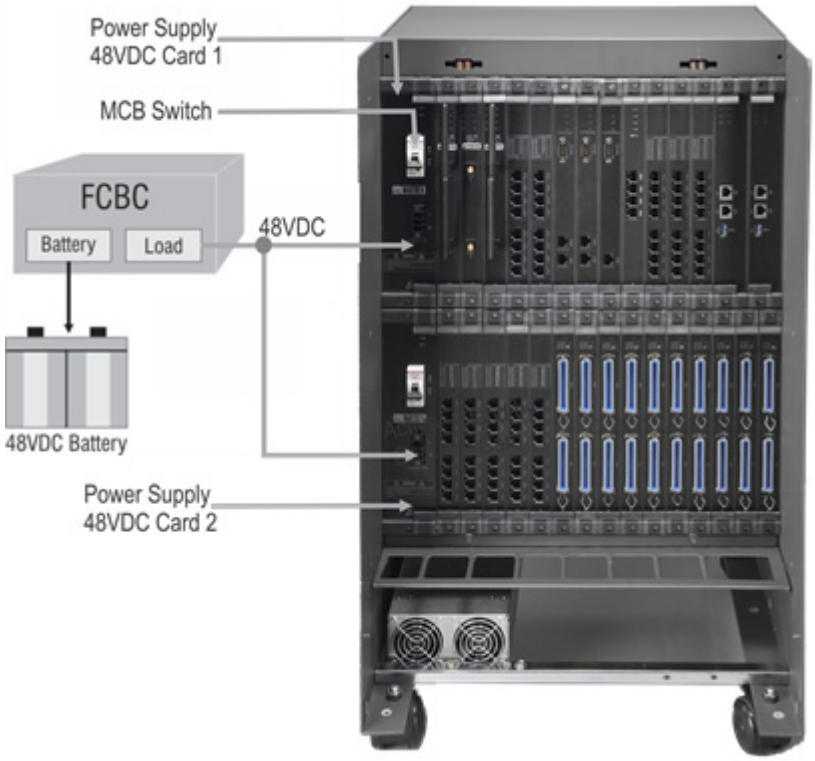
Color	Label on the Emerson PS
Red	V+
Black	V-
Yellow-Green	Earth

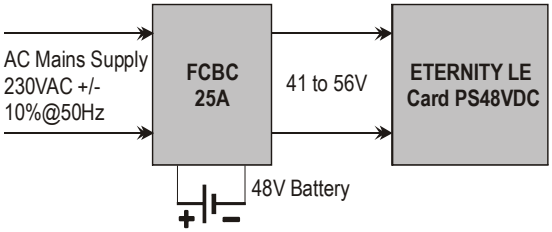
- Fix the screws back.



- Follow the same steps to connect another 48VDC Power Supply card with the Emerson PS.
- Now insert the male connector into the female connector (3-way termination block) on the card.

Connecting FCBC to the ETERNITY LENX





6. Connect Battery back up to the FCBC[27].

   Battery backup time depends upon the total load. The total load is the sum of system's load and load of active extensions.

   The Battery back up time depends on the 'Ah' rating of the battery connected to the FCBC. If 48V/26Ah batteries are connected to the FCBC for the system then backup time of 2.5 to 3[28] hrs can be ensured. The FCBC uses the constant voltage charging method. So, the batteries get charged faster if less power is consumed by the system when in mains mode.

7. Switch on power supply, after completing all other installation.

LED Pattern

LED L1 is a dual color LED for Power.

LED 2 is a single color LED that indicates Reverse Input Voltage.

Condition	LED	Color	Cadence
All modules operating inside the card are OK	L1 (PWR)	GREEN	Continuously ON
Any one or more modules operating inside the card are faulty	L1 (PWR)	RED	Continuously ON
Reverse Input	L2 (REV)	RED	Continuously ON

27. When the batteries are drained, the FCBC goes into the boost mode and begins to charge the batteries at higher current. When the batteries reach a preset voltage level (typically set to 56.0 volts), the FCBC goes to float mode. In the float mode the FCBC keeps charging the battery but at lower current. The FCBC monitors the voltage level of the batteries. As soon as the battery voltage goes below preset voltage (typically set to 50.4 volts), FCBC goes from float mode to boost mode. The change over from mains to battery and vice-versa is automatic. The advantage of using an FCBC is that batteries get charged faster, since the batteries are charged with higher current initially.

28. This is calculated taking into consideration 25% SLT load of the System.

The CPU Card

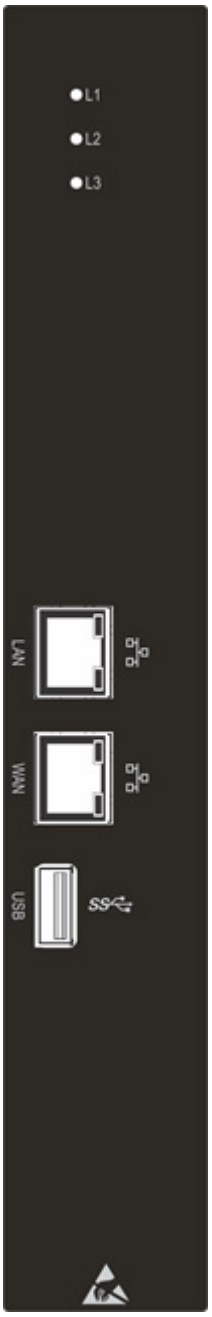
The ETERNITY MENX-LENX CPU Card hosts the SARVAM UCS Application. It supports four VOCODER modules and one VMS module. Both the modules — NX DBM VOCODER64 and NX DBM VMS64 are optional and can be purchased separately.

This card hosts Communication Manager, Feature Server, VoIP Server, VMS Server and other important servers and modules which controls all other slave cards (SLT, DKP, CO+SLT, DKP+SLT, E&M, BRI, T1E1, E1FO, GSM etc.). All the configuration and programming information is stored on this card.

ETERNITY LENX supports redundancy along with Hot Swap for the CPU Card. An additional CPU Card can be installed for redundancy. If the main CPU Card fails, the other card takes over, providing uninterrupted communication. To know more, refer to “Redundancy”.

The CPU Card occupies fixed slots, the first two slots from the extreme right of the first Rack, with a unique arrangement of connectors. So no other card can be inserted in the slots of the CPU Card.

The CPU Card has a WAN Port, LAN Port and USB Port on the front panel. It also has an Internal USB Port with a factory fitted pendrive.



Ports and Connectors:

Port	Connector	Description
LAN	RJ45	Used for connecting the Ethernet cable into LAN Port to connect to a PC or a LAN Switch.
WAN	RJ45	Used for connecting the Ethernet cable into WAN Port to connect to a Broadband Router/Modem.

Port	Connector	Description
USB	USB to COM Converter (Optional)	The External USB can be used as COM Port by connecting the USB to COM Converter. The USB to COM Port can be used to: • set up and run software applications — PMS and CAS. • capture System Activity Log, System Fault log and Hotel Motel Activity logs. • generate SMDR reports.

If you buy a spare CPU Card separately, the default pendrive will not be provided along with it.

LAN Interface

The LAN Port is provided to connect:

- the system to a PC or a LAN. This port is used for operating the web-based programming software Jeeves.
- the CPU Card to the Local Area Network to register SIP extensions through the LAN Port.
- set up and run software applications such as PMS and CAS on any PC on the LAN.
- generate Station Message Detail Record (SMDR) Reports on any PC on the LAN.
- capture "System Activity Log", "System Fault Log" and Hotel Motel Activity Log.

WAN Interface

The WAN Port is provided to connect:

- a LAN Switch/Hub/Router/Modem.
- the CPU Card to the public network over a Router/Modem. Any user on the public network can be registered as SIP Extension through the WAN Port.
- set up and run software applications such as PMS and CAS on any PC on the LAN.
- generate Station Message Detail Record (SMDR) Reports on any PC on the LAN.
- capture "System Activity Log", "System Fault Log" and Hotel Motel Activity Log.

VoIP Interface

The CPU Card supports four NX DBM VOCODER64 modules. You must purchase the module separately for VoIP functionality.

VOCODER Channels

The system supports four NX DBM VOCODER64 Modules. Each module supports 64 VOCODER Channels[29]. You must purchase the modules separately. The system provides 4 pre-activated VOCODER channels by default which can be used after installing NX DBM VOCODER64 module. If you require more channels, you can purchase the licenses accordingly. Matrix provides two licenses — SARVAM VOCODER CHNL4 and SARVAM VOCODER CHNL16.

If you require more than 64 VOCODER channels, you can install another NX DBM VOCODER64 Module.

29. The number of VOCODER channels that will be supported would be as per the license you purchase.

A call made from a SIP Extension or SIP Trunk to another SIP Extension or SIP Trunk will consume two VOCODER channels, whereas a call made from a SLT or DKP extension to a SIP Extension or SIP Trunk will consume one VOCODER channel. Thus, the number of speech paths available to make simultaneous calls will depend not only on the number of VOCODER channels, but also on the number of channels consumed by such SIP-to-SIP and Analog/Digital extension to SIP Trunk/SIP Extension calls.

VMS Interface

The system supports a full-fledged, 'in-skin' Voice Mail System module to provide mailbox facility to all its extensions users. The Voice Mail System also forms the basis of other features like Conversation Recording and Call Taping.

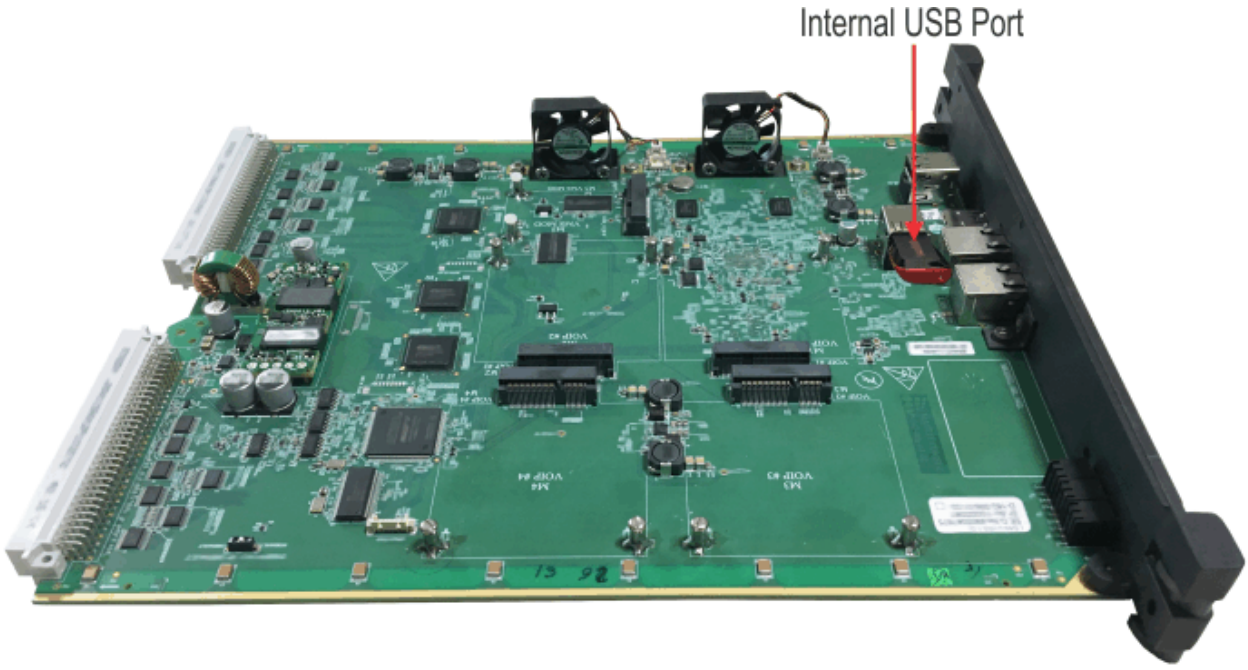
Each Mailbox has the capacity of storing 15,000 voice messages. The maximum size of each Mailbox is 60,000 minutes. By default, the size of each Mailbox is set to 5 minutes. The maximum Message Length for each Mailbox is 9,999 seconds. By default, the Maximum Message Length for each Mailbox is set to 15 seconds.
The NX DBM VMS64 Module is an optional module. It must be purchased separately. The factory fitted Pen Drive provided also contains the VMS configuration files, and voice messages for prompts and greetings along with the SARVAM UCS ENT Application. The Pen Drive is also the storage device for mailbox messages.

If required, you may use a Pen Drive of upto 64GB by replacing the factory fitted pendrive with a new one.

The system supports a maximum of 64 channels out of which 4 channels are provided by default. If you require more channels, you can purchase the licenses accordingly. Matrix provides two licenses — SARVAM VMS CHNL4 and SARVAM VMS CHNL16.

Internal USB Port

The CPU Card has an Internal USB Port with a pendrive inserted into it.



The pendrive supports FAT32 file format. It contains the SARVAM UCS Application, VMS greetings, messages, Matrix Extended IP phone firmwares and SMS Server firmware.

Do not remove the pendrive.

When you select the SARVAM UCS ENT Application, the system fetches the application from the pendrive.

External USB Port (Device Port) 3.0

The CPU Card has an External USB Port on the fascia. This can be used as a COM Port by connecting the USB to COM Converter.

The USB to COM Converter will not be provided by Matrix.

The following USB to COM Converters are supported:

- Prolific PL2303 by BAFO
- CH341 by Winchiphead

If you use any other USB to COM Converter, it may not function properly.

The USB to COM Port has a DB-9 connector.

The port allows you to connect a PC to the system, so that you can install and operate the following features:

- set up and run software applications such as PMS and CAS on any PC on the LAN.
- generate Station Message Detail Record (SMDR) Reports on any PC on the LAN.
- capture "System Activity Log" and "System Fault Log", Hotel Motel Activity Log.

LED

The CPU Card has three dual color LEDs:

- Heart beat (L1)
- Layer Communication indications (L2)
- Slave Switch indications (L3)

LED for both Active and Standby Card:

State	Color	Cadence
At Power ON		
	L1 and L3 – RED	Continuous ON for 15 sec (approximately)
	L1 and L3	OFF for 30 sec (approximately)
When L1 and L3 has been OFF for 10 sec	L2 – ORANGE	Continuously ON for 3 sec
	L2 – OFF	OFF

LED for Active Card:

State	Color	Cadence
L1 Pattern		
During initialization process	ORANGE	Continuous ON
After initialization process	-	OFF
In normal condition	GREEN	Toggle
L2 Pattern		
During initialization process	GREEN, RED, ORANGE	500 ms GREEN – 500 ms RED – 500 ms ORANGE
After initialization process	-	OFF
In normal condition	-	OFF
L3 Pattern		
When sync is received by Active card from Standby Card	RED	Continuous ON
When Sync is completed	-	OFF
When Data transfer done by Active card	RED	Continuous ON

LED for Standby Card:

State	Color	Cadence
L1 Pattern		
During initialization process	ORANGE	Continuous ON
After initialization process	-	OFF
In normal condition	RED	1 sec ON – 1 sec OFF
L2 Pattern		
During initialization process	GREEN, RED, ORANGE	500 ms GREEN – 500 ms RED – 500 ms ORANGE
After initialization process	-	OFF
PLL Lock (standby card clock is in sync with active card)	GREEN	Continuous ON
PLL is not lock (standby card clock is not in sync with active card)	-	OFF
L3 Pattern		
When sync is sent by Standby card to Active card	RED	Continuous ON
When Sync is completed	-	OFF

The CPU Card has a Password IP Default Switch (SW1) on the board:

Switch	Position	Function
SW1	ON	Password IP Default
	OFF (default)	Normal

Installing the VOCODER Module

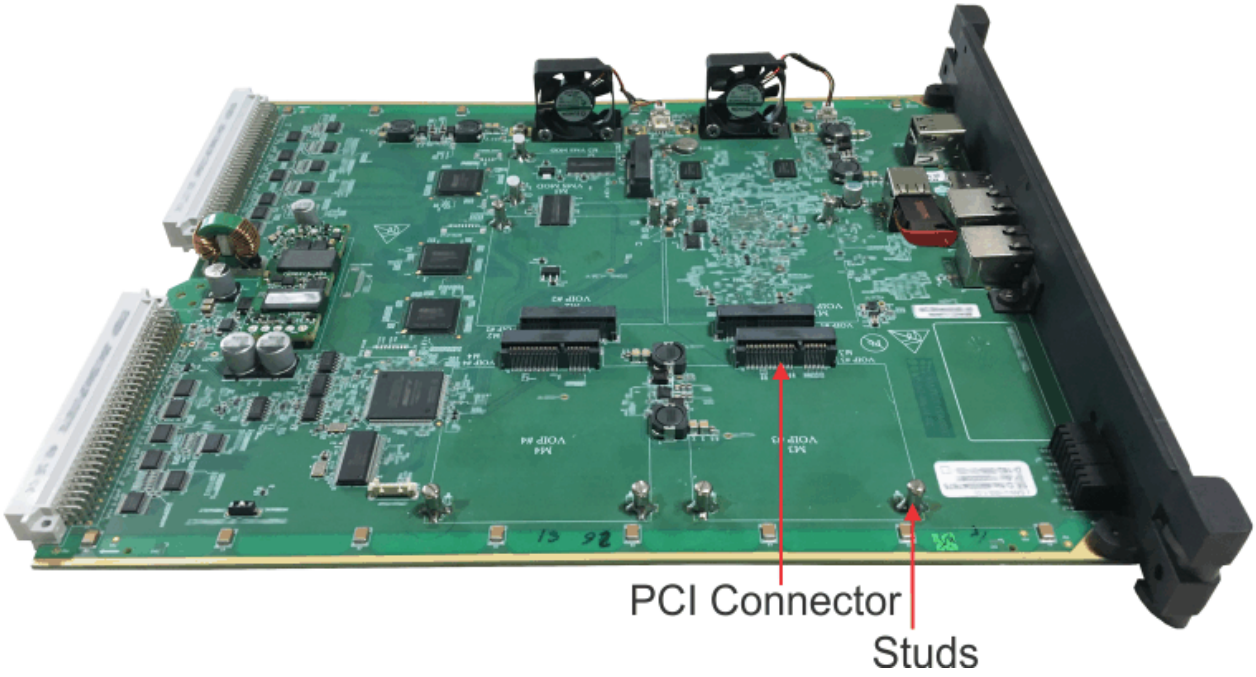
To install,

- Unpack the NX DBM VOCODER64 Module.

- If the CPU Card is already installed, switch off power supply, unplug the power cord. Remove the screws securing the card. Lift the levers on the mounting bracket to release the card. As the card emerges from the slot, ease it out of the slot.

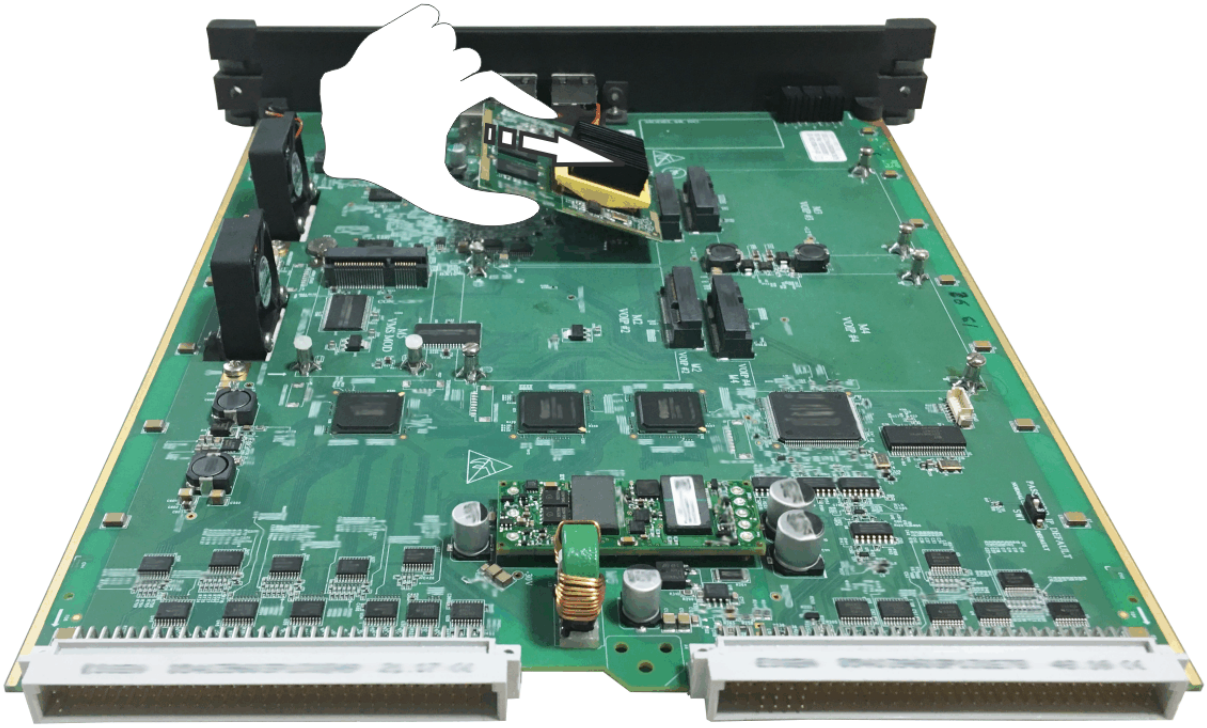
- Place the card carefully on a table with some packing underneath it. Avoid any physical contact with the PCB part of the card as this could cause Electrostatic discharge (ESD) and may damage the hardware.

- There are 5 PCI Connectors (4 for VoIP and 1 for VMS) and five pairs of studs on the CPU board.



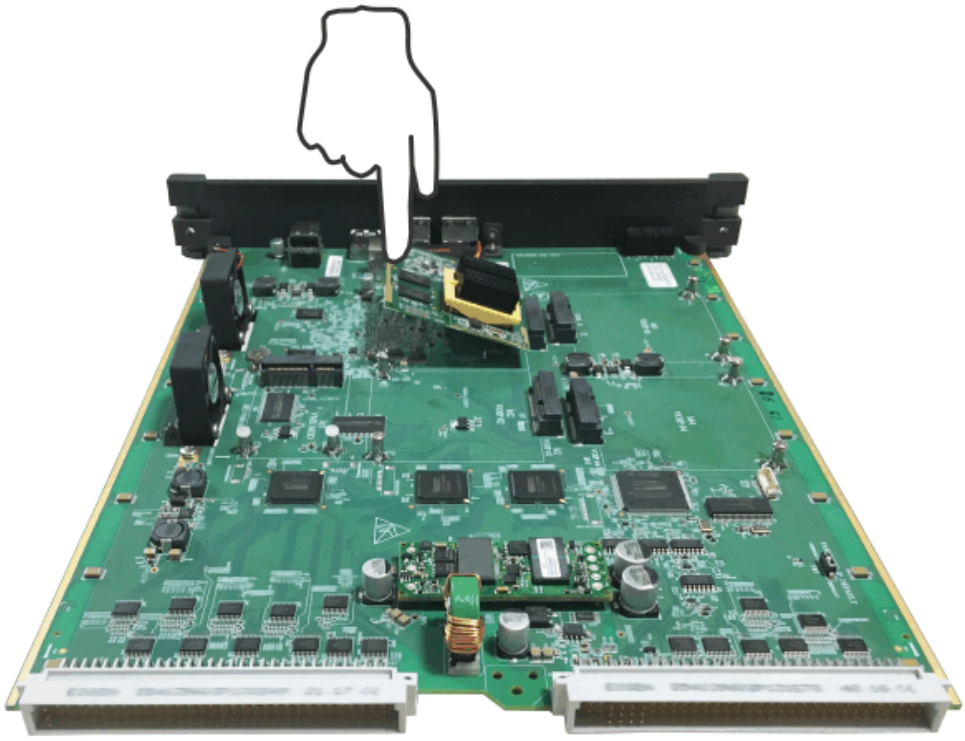
- Locate the **VOIP #1** label on the CPU board. The first NX DBM VOCODER64 Module should be mounted here.
- Remove the screws on the studs for the module and keep them aside.
- Carefully hold the NX DBM VOCODER64 Module from the edges. Make sure you do not touch the PCB area.

- Insert the NX DBM VOCODER64 Module into the PCI Connector socket.



- Press the Module with a finger and match the mounting holes perfectly with the stud holes. Make sure you do not touch the PCB area of the module except the yellow line provided for grounding at the front end of the module.

  Do not apply excessive pressure.

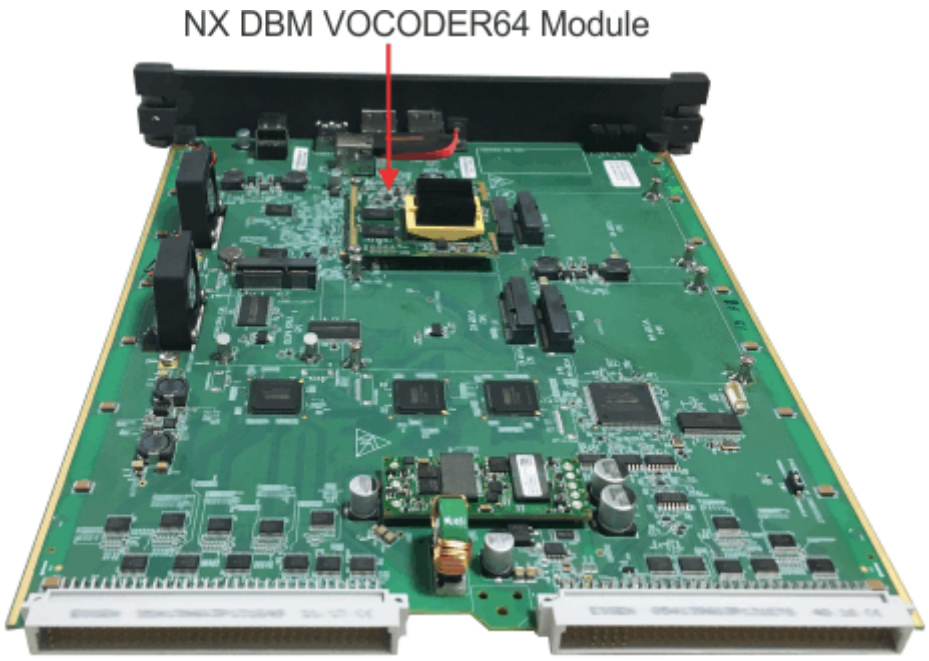


- Secure the module with the screws on the studs.

You may install the second NX DBM VOCODER64 Module by locating the **VOIP #2** label on the CPU board and follow the same steps as above. Similarly, install the other modules if required.

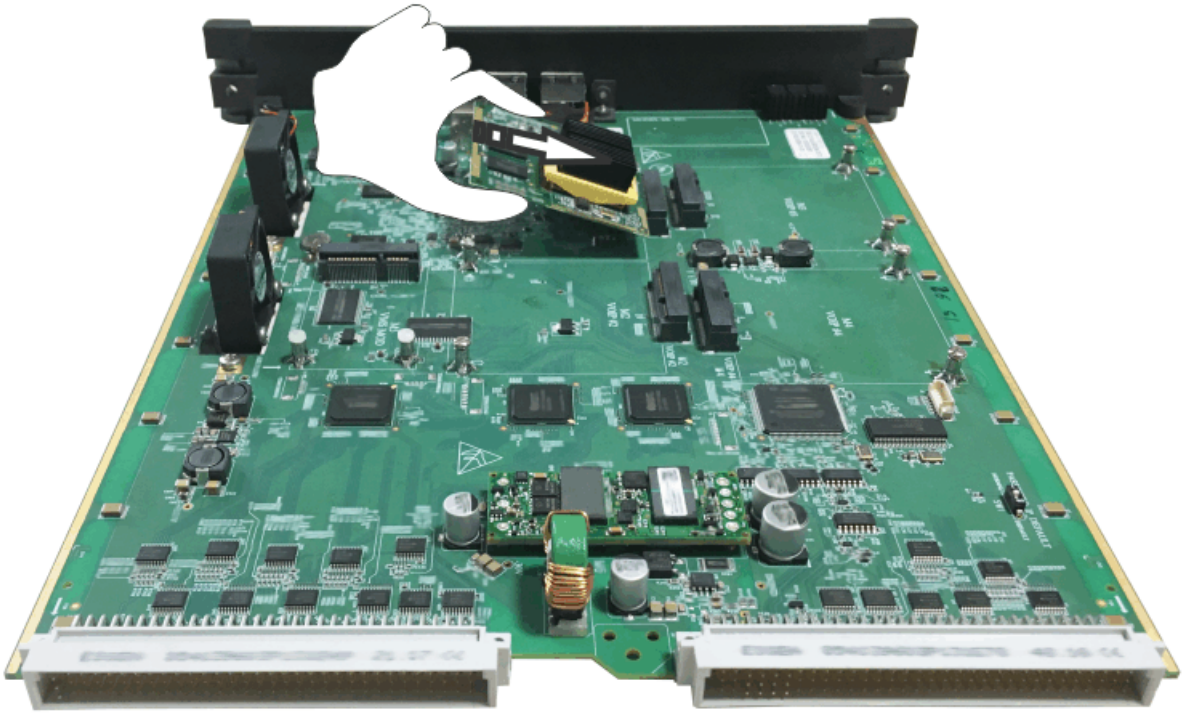
Removing the VOCODER Module

- Locate the VOCODER Module you want to remove from the CPU Card.



- Remove the screws from the module and keep them aside.

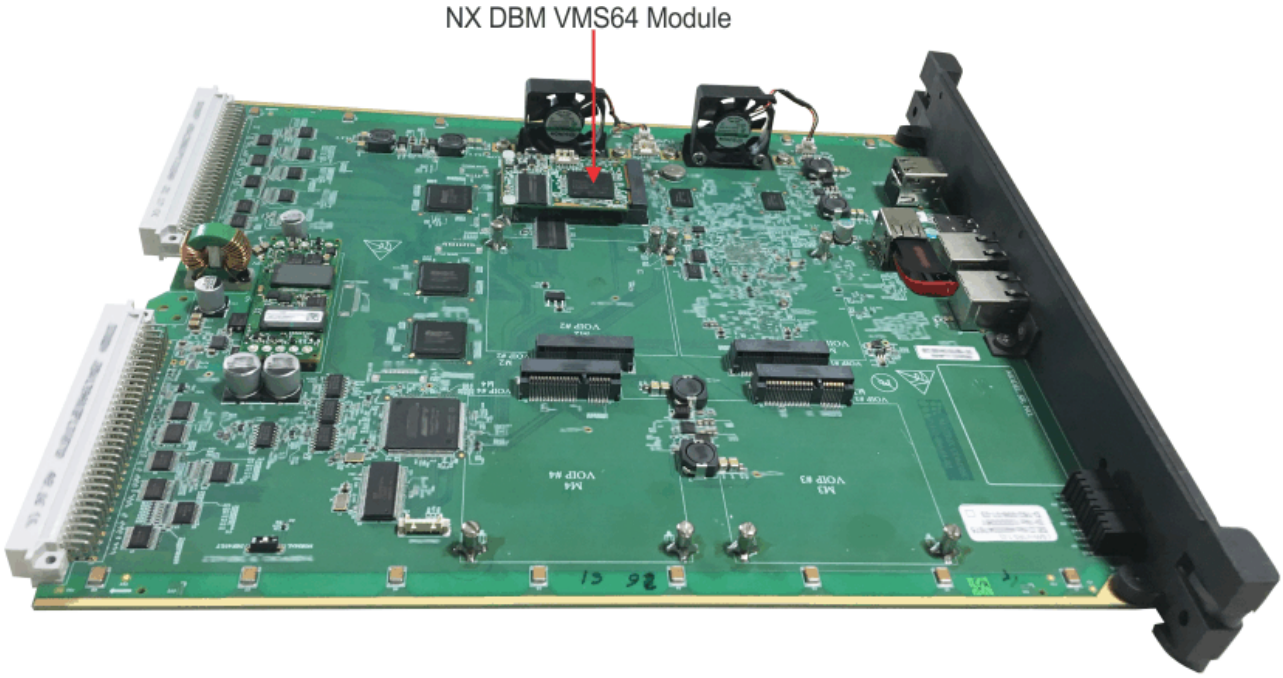
- Firmly hold the module and ease it out of the PCI connector carefully.



- Replace the screws on the studs.

Installing the VMS Module

To install the NX DBM VMS64 Module, locate the **VMS MOD** label on the CPU board and follow the same steps as described for “Installing the VOCODER Module”.



Make sure you use the plastic screws provided to affix the VMS Module over the studs.

- The pendrive which is provided to you by default contains VMS data and VMS firmware. You will be able to use the VMS features once you activate the VMS License.

  If you want to store more voice mail messages or greetings then you will need more space to store the same. You can replace this default pendrive with a new one having more space.

  To do so, you need to format your new pendrive with FAT32 file format and then copy all the contents of the factory fitted pendrive into the new pendrive.

  Switch-off the system and then replace the pendrive. The system will not detect the new pendrive if you do not restart the system after replacement.

Make sure while replacing the pendrive, you insert it in the same USB Port where the factory fitted pendrive was inserted.

- If you have no other modules to install, insert the card back into the ETERNITY LENX.
- Connect a computer to the LAN/WAN Port of the system with the ethernet cable supplied for the port.
- Open a Web browser on the computer to access the embedded Web server, Jeeves.
- Activate the VMS License Voucher. See “License Management” for instructions.
- Configure VMS. For detailed instructions, see “Configuring Voice Mail System”.

For removing the VMS Module, follow the same steps as described for removing the VOCODER Module. See “Removing the VOCODER Module”.

The Single Line Telephone Card

The Single Line Telephone (SLT) Card provides the interface to connect as extension phones, any standard, two-wire, analog single line telephone instrument - rotary, pulse-tone, cordless, feature phones with or without Calling Line Identification.

The SLT Card is available in the following configurations. SLT interface also is available in combination with Two-wire trunk interfaces on a single card.

SLT Cards for ETERNITY LENX

Card Name	Configuration and Application
ETERNITY LE Card SLT48[a]	48-port card to connect 48 Single Line Telephones
ETERNITY ME Card SLT32	32-port card to connect 32 Single Line Telephones
ETERNITY ME Card SLT16	16-port card to connect 16 Single Line Telephones
ETERNITY ME Card SLT8	8-port card to connect 8 Single Line Telephones
ETERNITY ME Card CO8+SLT24	Combination card, with 8-ports to connect to 8 Two-wire Analog trunk lines and 24 Single Line Telephones This Card supports Power Fail Transfer. To know more, see “Power Fail Transfer”.

a. *If you have CPU Cards with 1K IDT Switch (PCB Version V1R3) out of the 15 universal slots in the second rack, in four slots from the right the following cards cannot be affixed — ETERNITY LE CARD SLT48, ETERNITY ME CARD E1FO Dual and ETERNITY ME CARD T1E1PRI Dual. In the remaining slots any card can be affixed.*

The maximum number of SLT ports supported are 1296[30].

Connectors

The SLT Cards have RJ45 connectors, with each connector having 4 SLT ports. A multi-pair, MDF cable is supplied for each connector.

Only the SLT48 card has a 50-pin Centronics connector for the ports.

LEDs

The SLT cards have a single, tri-color LED to indicate:

- the health of the card during the Reset Cycle.
- the status of any one extension during normal functioning of the system.

  You may monitor any of the SLT Extension ports by assigning the LED to that port[31].

30. *If you have CPU Cards with 1K IDT Switch (PCB Version V1R3), the maximum number of SLT ports supported will be 1232. Using commands you can configure upto 999 ports and the remaining ports can be configured from Jeeves only.*
31. *To do this, enter SE mode, and dial the SE Command 7902-Slot-LED Number-Port, where Slot is the number of the universal slot in which the card is installed and Port is the port on the card to which the LED is to be assigned to monitor its functioning. LED Number is the number of the LED on the card, which will monitor the port.*

Installing Single Line Telephones

To be able to connect Single Line Telephones as Extensions to your system, you must install at least one of the aforementioned SLT Cards in the System.

1. Decide the number of SLT extensions required and arrange for as many telephone instruments.

   You may use any standard telephone instrument like a rotary phone, a pulse-tone switchable push-button phone, a feature phone or a cordless phone.

Use SLTs equipped with a 'Flash' key, as several of the features and facilities of the system require you to press Flash. If any of the SLTs you have selected does not have a Flash key, tap the Hook switch of the phone to dial Flash.

2. Unpack the SLT Card and check the package contents. Ensure that the power supply is switched off, before you begin the installation of the card. Always wear an electrostatic discharge prevention wrist strap/ belt and use a grounding mat.

3. Unscrew and remove the filler card mount bracket of any of the free (empty) Universal Slots. Do not discard the filler bracket! You may require it at a later stage.

4. Insert the SLT Card into the guide rails of the free slot you selected for the card.

   Make sure that the connectors on the card make perfect contact with those on the motherboard on the backplane.

5. Press down the levers on the mounting bracket to secure the card in its slot. Now, secure the mounting bracket with the two screws provided.

If you are installing more than one SLT Card, you can install the second card in any other free slot. It is not necessary to install the second/third card in the subsequent slots.

6. Use the cables supplied with the SLT Card to connect the SLT wires with the Main Distribution Frame.

   For each connector on the SLT Card, there is a separate 4-pair cable with an RJ45 jack on one end and free at the other end.

   Refer the illustrations below for pin out details of each connector.

ETERNITY LE Card SLT48

Connector 1

Port Type	Port Number	Pin Number	Signalling	Wire Colour
SLT	Port 1	1	Tip	Blue
		26	Ring	White
	Port 2	2	Tip	Orange
		27	Ring	White
	Port 3	3	Tip	Green
		28	Ring	White
	Port 4	4	Tip	Brown
		29	Ring	White
	Port 5	5	Tip	Gray
		30	Ring	White
	Port 6	6	Tip	Blue
		31	Ring	Red
	Port 7	7	Tip	Orange
		32	Ring	Red
	Port 8	8	Tip	Green
		33	Ring	Red
	Port 9	9	Tip	Brown
		34	Ring	Red
	Port 10	10	Tip	Gray
		35	Ring	Red
	Port 11	11	Tip	Blue
		36	Ring	Black
	Port 12	12	Tip	Orange
		37	Ring	Black
	Port 13	13	Tip	Green
		38	Ring	Black
	Port 14	14	Tip	Brown
		39	Ring	Black
	Port 15	15	Tip	Gray
		40	Ring	Black
	Port 16	16	Tip	Blue
		41	Ring	Yellow

Port Type	Port Number	Pin Number	Signalling	Wire Colour
SLT	Port 17	17	Tip	Orange
		42	Ring	Yellow
	Port 18	18	Tip	Green
		43	Ring	Yellow
	Port 19	19	Tip	Brown
		44	Ring	Yellow
	Port 20	20	Tip	Gray
		45	Ring	Yellow
	Port 21	21	Tip	Blue
		46	Ring	Violet
	Port 22	22	Tip	Orange
		47	Ring	Violet
	Port 23	23	Tip	Green
		48	Ring	Violet
	Port 24	24	Tip	Brown
		49	Ring	Violet
	Port 25	25	NC	Gray
		50	NC	Violet

Connector 2

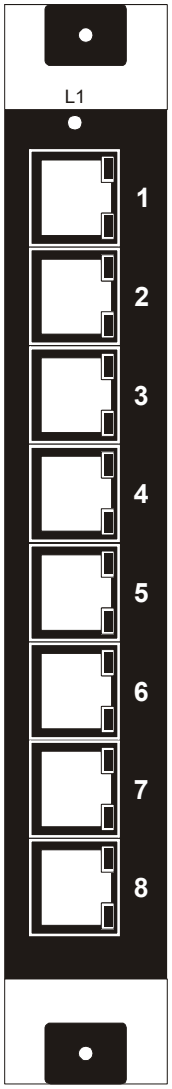
SLT	Port 25	1	Tip	Blue
		26	Ring	White
	Port 26	2	Tip	Orange
		27	Ring	White
	Port 27	3	Tip	Green
		28	Ring	White
	Port 28	4	Tip	Brown
		29	Ring	White
	Port 29	5	Tip	Gray
		30	Ring	White
	Port 30	6	Tip	Blue
		31	Ring	Red
	Port 31	7	Tip	Orange
		32	Ring	Red
	Port 32	8	Tip	Green
		33	Ring	Red
	Port 33	9	Tip	Brown
		34	Ring	Red
	Port 34	10	Tip	Gray
		35	Ring	Red
	Port 35	11	Tip	Blue
		36	Ring	Black
	Port 36	12	Tip	Orange
		37	Ring	Black

SLT	Port 37	13	Tip	Green
		38	Ring	Black
	Port 38	14	Tip	Brown
		39	Ring	Black
	Port 39	15	Tip	Gray
		40	Ring	Black
	Port 40	16	Tip	Blue
		41	Ring	Yellow
	Port 41	17	Tip	Orange
		42	Ring	Yellow
	Port 42	18	Tip	Green
		43	Ring	Yellow
	Port 43	19	Tip	Brown
		44	Ring	Yellow
	Port 44	20	Tip	Gray
		45	Ring	Yellow
	Port 45	21	Tip	Blue
		46	Ring	Violet
	Port 46	22	Tip	Orange
		47	Ring	Violet
	Port 47	23	Tip	Green
		48	Ring	Violet
	Port 48	24	Tip	Brown
		49	Ring	Violet
		25	NC	Gray
		50	NC	Violet

Jumpers on the Main Board (SLT48)

Jumper Number	Position	Function
J1	AB (default)	Normal Operation
	BC	For uploading software using COM Port
J2 & J3	AB (default)	Normal Operation
	BC	For uploading software using COM Port

ETERNITY ME Card SLT32



Connector	Color	Connection	H/w Port Offset
RJ45-1	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	01 02 03 04
RJ45-2	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	05 06 07 08
RJ45-3	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	09 10 11 12
RJ45-4	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	13 14 15 16
RJ45-5	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	17 18 19 20
RJ45-6	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	21 22 23 24
RJ45-7	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	25 26 27 28
RJ45-8	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	29 30 31 32

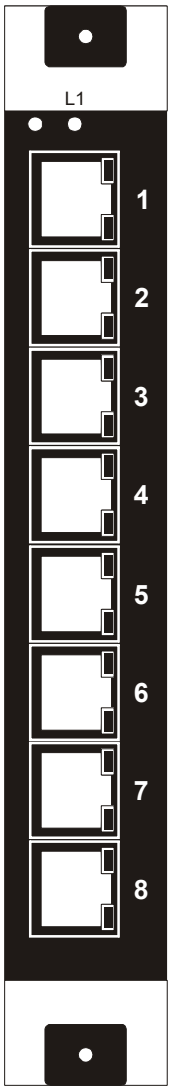
ETERNITY ME Card SLT16

Connector	Color	Connection	H/w Port Offset
RJ45-1	Blue - (Blue & White)	SLT	01
	Orange - (Orange & White)	SLT	02
	Green - (Green & White)	SLT	03
	Brown - (Brown & White)	SLT	04
RJ45-2	Blue - (Blue & White)	SLT	05
	Orange - (Orange & White)	SLT	06
	Green - (Green & White)	SLT	07
	Brown - (Brown & White)	SLT	08
RJ45-3	Blue - (Blue & White)	SLT	09
	Orange - (Orange & White)	SLT	10
	Green - (Green & White)	SLT	11
	Brown - (Brown & White)	SLT	12
RJ45-4	Blue - (Blue & White)	SLT	13
	Orange - (Orange & White)	SLT	14
	Green - (Green & White)	SLT	15
	Brown - (Brown & White)	SLT	16

ETERNITY ME Card SLT8

Connector	Color	Connection	H/w Port Offset
RJ45-1	Blue - (Blue & White)	SLT	01
	Orange - (Orange & White)	SLT	02
	Green - (Green & White)	SLT	03
	Brown - (Brown & White)	SLT	04
RJ45-2	Blue - (Blue & White)	SLT	05
	Orange - (Orange & White)	SLT	06
	Green - (Green & White)	SLT	07
	Brown - (Brown & White)	SLT	08

ETERNITY ME Card CO8+SLT24



Connector	Color	Connection	H/w Port Offset
RJ45-1	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	01 02 03 04
RJ45-2	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	05 06 07 08
RJ45-3	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	09 10 11 12
RJ45-4	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	13 14 15 16
RJ45-5	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	17 18 19 20
RJ45-6	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	21 22 23 24
RJ45-7	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	TWT TWT TWT TWT	01 02 03 04
RJ45-8	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	TWT TWT TWT TWT	05 06 07 08

7. Plug in the RJ45 end of the MDF cables supplied with the card into the respective connectors. Refer to the pinout details of the connectors of each SLT Card type illustrated above.

8. Terminate the open end of the cables into the punch down blocks of the Krone modules designated for 'Station Lines' in the “The Main Distribution Frame (MDF)”.

   Each wire-pair from the SLT Port must be terminated to the bottom of the Krone Connector, while the wire-pair of the extension line to be connected to this port must be terminated on the top of the Krone connector. Refer the topic “The Main Distribution Frame (MDF)” for illustration.

9. Repeat the same steps to install another SLT Card.

10. If you have completed all installation tasks, power ON the system, observe the Reset Cycle and the LED Pattern of the SLT Card.

LED Pattern of the SLT Card

Stage	LED Color	Cadence
Auto Upgradation[a]		
Card waiting for application	RED	ON-200ms-OFF 200ms
Card is up, loaded with new application	GREEN	ON-200ms-OFF 200ms
Initialization		
	RED	ON 500ms-OFF 500ms
	GREEN	ON 500ms - OFF 500ms
	ORANGE	ON 500ms - OFF 500ms
Stand-by task	ORANGE	1 sec Orange -1 sec Green
Errors		
Flash Failure	None	None
RAM Failure	None	None

a. *Done by the Boot Loader Application.*

Status of Selected Port

PORT Status	LED Color	LED Cadence
Selected port data transmitted to Master Card	RED	Toggle[a]
Selected port data received from Master Card	RED	Toggle

a. *The current LED state will remain the same until the next command is received from the application on the SLT Port. For example, if the current LED state is Green/Red ON, on the next command received, the LED will be turned OFF. It will remain OFF until the next command is received. When the next command is received it will be turned Green/Red ON again. This process continues.*

Jumpers on the Main Board

Jumper Number	Position	Function
J1	AB (default)	Normal Operation
	BC	For uploading software using COM Port
J2 & J4	AB (default)	Normal Operation
	BC	For uploading software using COM Port

Connecting SLT instruments

11. Connect the SLT instruments you have arranged for. Plug in the SLTs into the wall socket/outlets.

- *For the purpose of testing, you may connect one or two Single Line Telephone instruments by plugging in the phone cables into the RJ45 connectors on the card.*

- *When you plug the RJ11 connector of SLT into an RJ45 connector on the SLT Card, the SLT will be connected on the first port on the connector.*

The Intercom Line Card[32]

For the Building Intercom application, the system supports the Intercom Line Card (ILC).

You can connect any standard, two-wire, analog single line telephone instrument - rotary, pulse-tone, cordless, feature phones with or without Calling Line Identification to the Intercom Line Card.

ILC Cards for ETERNITY LENX

Card Name	Configuration and Application
ETERNITY LE Card ILC48	48-port card to connect 48 Single Line Telephones
ETERNITY ME Card ILC32	32-port card to connect 32 Single Line Telephones

Choose an ILC Card with the configuration that meets your requirement for intercom ports. Also, consider the maximum Port capacity of the system you are installing. The maximum number of intercom ports supported are 1296.

Connectors

The ILC32 Cards have RJ45 connectors with four ports on each connector. A multi-pair, MDF cable is supplied for each connector.

Only the ILC48 card has a 50-pin Centronics connector for the ports.

LEDs

The ILC cards have a single, tri-color LED to indicate the health of the card during the Reset Cycle.

Installing Intercom Telephones

To be able to connect intercom telephones to the system, you must install at least one of the aforementioned intercom line cards in the system.

1. Decide the number of intercom extensions required and arrange for as many telephone instruments.
2. Ensure that the extension wiring is completed according to your requirements. The extension cables from the wall jack are terminated in the Main Distribution Frame and the telephones are connected to the wall jacks.
3. Always wear an electrostatic discharge prevention wrist strap/belt and use a grounding mat to prevent damage to the components of the card.
4. Unpack the ILC card and check the package contents. You may switch off power supply before you install the card. Since, ETERNITY LENX supports Hot Swap, you can install the card in power on condition.
5. Unscrew and remove the filler card mount bracket of any of the free (empty) Universal Slots. Keep the filler bracket for future use.

32. This card has been phased-out. if required, only support will be provided for the same.

6. Insert the ILC Card into the guide rails of the free slot you selected for the card.

   Make sure that the connectors on the card make perfect contact with those on the motherboard on the backplane.

7. Press down the levers on the mounting bracket to secure the card in its slot. Now, secure the mounting bracket with the two screws provided.

8. Repeat these steps to install another card.

9. Now, use the cables supplied with the ILC Card to connect the card to the Main Distribution Frame to which the intercom phones are connected.

   For each connector on the card, there is a separate 4-pair cable with an RJ45 jack on one end and free at the other end. Refer the illustrations of the pinout of the intercom cards to connect the wires.

ETERNITY LE Card ILC48

Connector 1

Port Type	Port Number	Pin Number	Signaling	Wire Colour
SLT	Port 1	1	Tip	White
		26	Ring	Blue
	Port 2	2	Tip	White
		27	Ring	Orange
	Port 3	3	Tip	White
		28	Ring	Green
	Port 4	4	Tip	White
		29	Ring	Brown
	Port 5	5	Tip	White
		30	Ring	Slate
	Port 6	6	Tip	Red
		31	Ring	Blue
	Port 7	7	Tip	Red
		32	Ring	Orange
	Port 8	8	Tip	Red
		33	Ring	Green
	Port 9	9	Tip	Red
		34	Ring	Brown
	Port 10	10	Tip	Red
		35	Ring	Slate

Port Type	Port Number	Pin Number	Signaling	Wire Colour
SLT	Port 11	11	Tip	Black
		36	Ring	Blue
	Port 12	12	Tip	Black
		37	Ring	Orange
	Port 13	13	Tip	Black
		38	Ring	Green
	Port 14	14	Tip	Black
		39	Ring	Brown
	Port 15	15	Tip	Black
		40	Ring	Slate
	Port 16	16	Tip	Green
		41	Ring	Orange
	Port 17	17	Tip	Green
		42	Ring	Brown
	Port 18	18	Tip	Green
		43	Ring	Slate
	Port 19	19	Tip	Blue
		44	Ring	Orange
	Port 20	20	Tip	Blue
		45	Ring	Green
	Port 21	21	Tip	Blue
		46	Ring	Brown
	Port 22	22	Tip	Blue
		47	Ring	Slate
	Port 23	23	Tip	Orange
		48	Ring	Brown
	Port 24	24	Tip	Orange
		49	Ring	Slate
	Port 25	25	NC	Slate
		50	NC	Brown

Connector 2

SLT	Port 25	1	Tip	White
		26	Ring	Blue
	Port 26	2	Tip	White
		27	Ring	Orange
	Port 27	3	Tip	White
		28	Ring	Green
	Port 28	4	Tip	White
		29	Ring	Brown
	Port 29	5	Tip	White
		30	Ring	Slate
	Port 30	6	Tip	Red
		31	Ring	Blue
	Port 31	7	Tip	Red
		32	Ring	Orange
	Port 32	8	Tip	Red
		33	Ring	Green
	Port 33	9	Tip	Red
		34	Ring	Brown
	Port 34	10	Tip	Red
		35	Ring	Slate
	Port 35	11	Tip	Black
		36	Ring	Blue
	Port 36	12	Tip	Black
		37	Ring	Orange
	Port 37	13	Tip	Black
		38	Ring	Green
	Port 38	14	Tip	Black
		39	Ring	Brown
	Port 39	15	Tip	Black
		40	Ring	Slate
	Port 40	16	Tip	Green
		41	Ring	Orange
	Port 41	17	Tip	Green
		42	Ring	Brown

SLT	Port 42	18	Tip	Green
		43	Ring	Slate
	Port 43	19	Tip	Blue
		44	Ring	Orange
	Port 44	20	Tip	Blue
		45	Ring	Green
	Port 45	21	Tip	Blue
		46	Ring	Brown
	Port 46	22	Tip	Blue
		47	Ring	Slate
	Port 47	23	Tip	Orange
		48	Ring	Brown
	Port 48	24	Tip	Orange
		49	Ring	Slate
		25	NC	Slate
		50	NC	Brown

Connector 1

Port Type	Port Number	Pin Number	Signalling	Wire Colour
SLT	Port 1	1	Tip	Blue
		26	Ring	White
	Port 2	2	Tip	Orange
		27	Ring	White
	Port 3	3	Tip	Green
		28	Ring	White
	Port 4	4	Tip	Brown
		29	Ring	White
	Port 5	5	Tip	Gray
		30	Ring	White
	Port 6	6	Tip	Blue
		31	Ring	Red
	Port 7	7	Tip	Orange
		32	Ring	Red
	Port 8	8	Tip	Green
		33	Ring	Red

Port Type	Port Number	Pin Number	Signalling	Wire Colour
SLT	Port 9	9	Tip	Brown
		34	Ring	Red
	Port 10	10	Tip	Gray
		35	Ring	Red
	Port 11	11	Tip	Blue
		36	Ring	Black
	Port 12	12	Tip	Orange
		37	Ring	Black
	Port 13	13	Tip	Green
		38	Ring	Black
	Port 14	14	Tip	Brown
		39	Ring	Black
	Port 15	15	Tip	Gray
		40	Ring	Black
	Port 16	16	Tip	Blue
		41	Ring	Yellow
	Port 17	17	Tip	Orange
		42	Ring	Yellow
	Port 18	18	Tip	Green
		43	Ring	Yellow
	Port 19	19	Tip	Brown
		44	Ring	Yellow
	Port 20	20	Tip	Gray
		45	Ring	Yellow
	Port 21	21	Tip	Blue
		46	Ring	Violet
	Port 22	22	Tip	Orange
		47	Ring	Violet
	Port 23	23	Tip	Green
		48	Ring	Violet
	Port 24	24	Tip	Brown
		49	Ring	Violet
	Port 25	25	NC	Gray
		50	NC	Violet

Connector 2

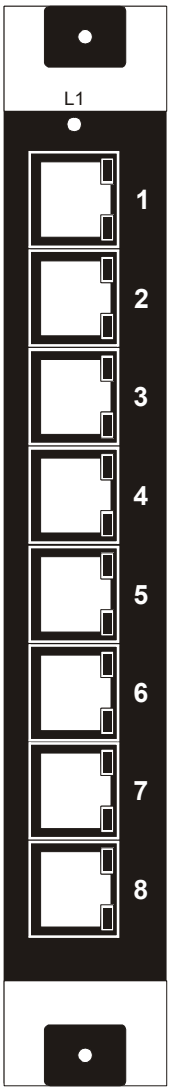
SLT	Port 25	1	Tip	Blue
		26	Ring	White
	Port 26	2	Tip	Orange
		27	Ring	White
	Port 27	3	Tip	Green
		28	Ring	White
	Port 28	4	Tip	Brown
		29	Ring	White
	Port 29	5	Tip	Gray
		30	Ring	White
	Port 30	6	Tip	Blue
		31	Ring	Red
	Port 31	7	Tip	Orange
		32	Ring	Red
	Port 32	8	Tip	Green
		33	Ring	Red
	Port 33	9	Tip	Brown
		34	Ring	Red
	Port 34	10	Tip	Gray
		35	Ring	Red
	Port 35	11	Tip	Blue
		36	Ring	Black
	Port 36	12	Tip	Orange
		37	Ring	Black
	Port 37	13	Tip	Green
		38	Ring	Black
	Port 38	14	Tip	Brown
		39	Ring	Black
	Port 39	15	Tip	Gray
		40	Ring	Black
	Port 40	16	Tip	Blue
		41	Ring	Yellow
	Port 41	17	Tip	Orange
		42	Ring	Yellow

SLT	Port 42	18	Tip	Green
		43	Ring	Yellow
	Port 43	19	Tip	Brown
		44	Ring	Yellow
	Port 44	20	Tip	Gray
		45	Ring	Yellow
	Port 45	21	Tip	Blue
		46	Ring	Violet
	Port 46	22	Tip	Orange
		47	Ring	Violet
	Port 47	23	Tip	Green
		48	Ring	Violet
	Port 48	24	Tip	Brown
		49	Ring	Violet
		25	NC	Gray
		50	NC	Violet

Jumpers on the Main Board (ILC48)

Jumper Number	Position	Function
J1	AB (default)	Normal Operation
	BC	For uploading software using COM Port
J2 & J3	AB (default)	Normal Operation
	BC	For uploading software using COM Port

ETERNITY ME Card ILC32



Connector	Color	Connection	H/w Port Offset
RJ45-1	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	01 02 03 04
RJ45-2	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	05 06 07 08
RJ45-3	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	09 10 11 12
RJ45-4	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	13 14 15 16
RJ45-5	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	17 18 19 20
RJ45-6	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	21 22 23 24
RJ45-7	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	25 26 27 28
RJ45-8	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	29 30 31 32

10. If you have completed all other installation tasks, power ON the system, observe the Reset Cycle and the LED indication of the card.

LED Indication of the ILC Card

Stage	LED Color	Cadence
Auto Upgradation[a]		
Card waiting for application	RED	ON-200ms-OFF 200ms
Card is up, loaded with new application	GREEN	ON-200ms-OFF 200ms
Initialization		
	RED	ON 500ms-OFF 500ms
	GREEN	ON 500ms - OFF 500ms
	ORANGE	ON 500ms - OFF 500ms
Stand-by task	ORANGE, GREEN	1 sec Orange -1 sec Green

Stage	LED Color	Cadence
Errors		
Flash Failure	None	None
RAM Failure	None	None

a. *Done by the Boot Loader Application.*

Status of Selected Port

PORT Status	LED Color	LED Cadence
Selected port data transmitted to Master Card	RED	Toggle[a]
Selected port data received from Master Card	RED	Toggle

a. The current LED state will remain the same until the next command is received from the application on the SLT Port. For example, if the current LED state is Green/Red ON, on the next command received, the LED will be turned OFF. It will remain OFF until the next command is received. When the next command is received it will be turned Green/Red ON again. This process continues.

Jumpers on the Main Board

Jumper Number	Position	Function
J1	AB (default)	Normal Operation
	BC	For uploading software using COM Port
J2 & J4	AB (default)	Normal Operation
	BC	For uploading software using COM Port

The Digital Key Phone Card

The Digital Key Phone (DKP) Card provides the interface to connect the proprietary digital key phones of the EON series, the proprietary PC-based phone EONSOFT, the Direct Station Selection (DSS) Consoles, with the system.

DKP Cards for ETERNITY LENX

Card Name	Configuration and Application
ETERNITY ME DKP32	32-port card to connect 32 DKP/DSS Consoles
ETERNITY ME DKP16	16-port card to connect 16 DKP/DSS Consoles.
ETERNITY ME DKP8	8-port card to connect 8 DKP/DSS Consoles

Select a DKP Card with the configuration that meets your requirement for DKP Ports.To connect the proprietary digital key phones with the system, you must have at least one of the above mentioned DKP Cards installed in the system.

The maximum number of DKP Ports supported by the system are 128.

Connectors

The DKP Cards have RJ45 connectors, with each connector having 4 DKP ports. A multi-pair MDF cable is supplied for each connector on the card.

LEDs

The DKP cards have a single, tri-color LED to indicate:

- the health of the card during the Reset Cycle.
- the status of any one of the ports during normal functioning of the system. By default, it is assigned to DKP Port 1.

  You may monitor any of the DKP ports by assigning the LED to that port[33].

Installing the Digital Key Phone Card

Decide the number of DKP extensions and DSS Consoles required and arrange for as many EON, EONSOFT and DSS Consoles.

Decide the locations of the DKP extensions and make sure that the necessary wiring for the DKP extensions, from the wall jack to the MDF, is done.

1. Unpack the DKP Card and check the package contents[34]. Before handling the card, make sure that power supply is switched off and you are wearing an antistatic-wrist strap/belt and have a grounding mat.
2. Unscrew and remove the filler card mount bracket of any of the free (empty) Universal Slots. Do not discard the filler bracket, keep for future use to cover empty slots.

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33. You can do this from the SE mode, by dialing the SE Command 7902-Slot-LED Number-Port, where Slot is the number of the universal slot in which the card is installed and Port is the port on the card to which the LED is to be assigned to monitor its functioning. LED Number is the number of the LED on the card, which will monitor the port.

34. See "ETERNITY GENX Cards" under 'Packing List' of Appendix topic.

3. Insert the DKP Card into the guide rails of the free slot you have selected for the card. All the pins on the connector of the card should make perfect contact with those on the connector of the slot on the backplane motherboard.

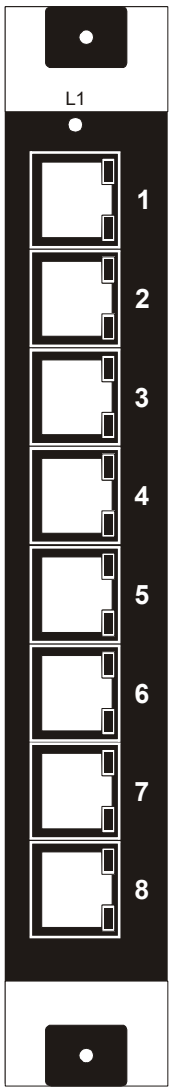
4. Press down the levers on the mounting bracket to secure the card in its slot. Now, fix the card in its slot with the two screws provided.

If you are installing more than one DKP Card, it is not necessary to install the next card in the subsequent slot.

5. Using the MDF Cables supplied with the DKP Card connect the DKP ports to the Main Distribution Frame.

   Refer the connector pin details for each DKP Card type given in the following.

ETERNITY ME Card DKP32



Connector	Color	Connection	H/w Port Offset
RJ45-1	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	DKP DKP DKP DKP	01 02 03 04
RJ45-2	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	DKP DKP DKP DKP	05 06 07 08
RJ45-3	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	DKP DKP DKP DKP	09 10 11 12
RJ45-4	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	DKP DKP DKP DKP	13 14 15 16
RJ45-5	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	DKP DKP DKP DKP	17 18 19 20
RJ45-6	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	DKP DKP DKP DKP	21 22 23 24
RJ45-7	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	DKP DKP DKP DKP	25 26 27 28
RJ45-8	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	DKP DKP DKP DKP	29 30 31 32

ETERNITY ME Card DKP16

Connector	Color	Connection	H/w Port Offset
RJ45-1	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	DKP DKP DKP DKP	01 02 03 04
RJ45-2	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	DKP DKP DKP DKP	05 06 07 08
RJ45-3	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	DKP DKP DKP DKP	09 10 11 12
RJ45-4	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	DKP DKP DKP DKP	13 14 15 16

ETERNITY ME Card DKP8

Connector	Color	Connection	H/w Port Offset
RJ45-1	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	DKP DKP DKP DKP	01 02 03 04
RJ45-2	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	DKP DKP DKP DKP	05 06 07 08

6. Plug in the RJ45 end of the MDF cables provided with the DKP card into the respective connectors.

7. Terminate the free end of the cables into the punch down blocks of the Krone modules designated for 'Station Lines' in the Main Distribution Frame (MDF).

   Each wire-pair from the DKP Port must be terminated to the bottom of the Krone Connector, while the wire-pair of the extension line to be connected to this port must be terminated on the top of the Krone connector. Refer the topic “The Main Distribution Frame (MDF)” for illustration.

8. Connect the Digital Key Phones to the wall jacks at their respective locations. Detailed installations instructions for EON, EONSOFT are provided separately.

If you have completed all installation tasks, power on the system and observe the Reset Cycle and the LED Pattern of the DKP Card.

LED Pattern DKP Card

Stage	LED Color	Cadence
Auto Upgradation		
Card waiting for application	RED	ON-200ms-OFF 200ms
Card is up, loaded with new application	GREEN	ON-200ms-OFF 200ms
Initialization		
	RED	ON 500ms-OFF 500ms
	GREEN	ON 500ms-OFF 500ms
	ORANGE	ON 500ms-OFF 500ms
Stand-by task	ORANGE, GREEN	1 sec Orange-1 sec Green
Errors		
Flash Failure	None	None
RAM Failure	None	None

Status of Selected DKP Port

PORT Status	LED Color	LED Cadence
Selected DKP's data are transmitted to CPU Card	RED	Toggle[a] on each event
Selected DKP's data are received from CPU Card	RED	Toggle[b] on each request from Master

a. The current LED state will remain the same until the next event is received from the application on the DKP Port. For example, if the current LED state is Green/Red ON, on the next event, the LED will be turned OFF. It will remain OFF until the next event occurs. When the next event is received it will be turned Green/Red ON again. This process continues.

b. Same as the above note.

Jumpers on the Main Board

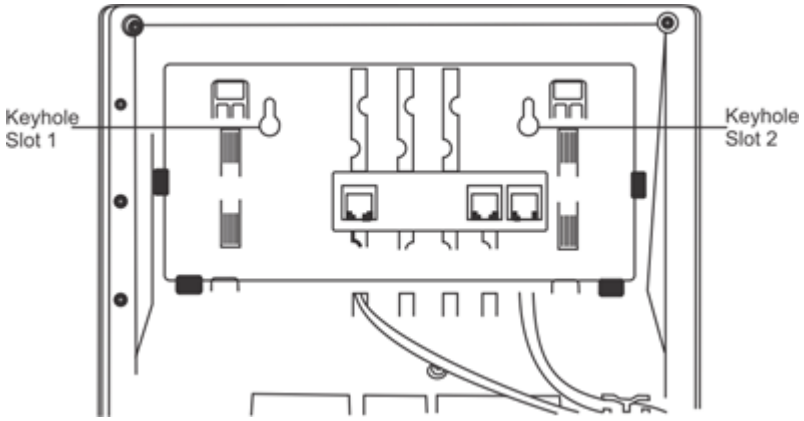
Jumper Number	Position	Function
J1	AB (default)	Normal Operation
	BC	For uploading software using COM Port
J2 & J4	AB	NA
	BC (default)	Normal Operation/ Debug

Installing EON48

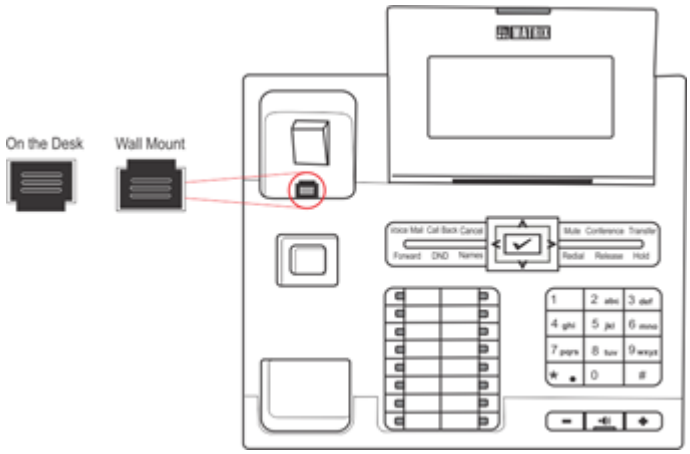
- Unpack the box and verify the package contents.
- You can mount the phone on a wall or on desk.

Mount the phone on a Wall

- Use the mounting template to drill holes of appropriate size and distance.
- Fix the screw grips in the holes you drilled.
- Fix two screws in the holes on the wall, ensuring that they are aligned with the Keyhole Slots 1 and 2. The screws should protrude from the wall to fit into the keyhole slots.

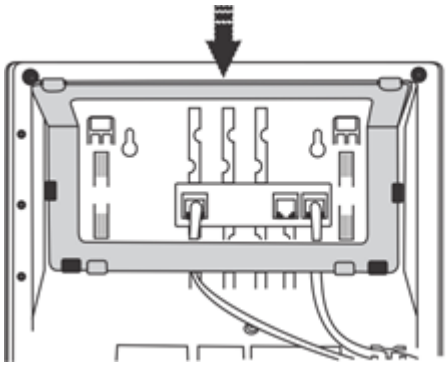


- Now, mount the phone with the screws fitting into the keyhole slots.
- Reverse the handset wall mount tab to make sure that the handset remains intact when you mount the phone. Push the handset wall mount tab upwards to remove it from the slot. Rotate it 180 degrees clockwise and push it downwards into the slot.

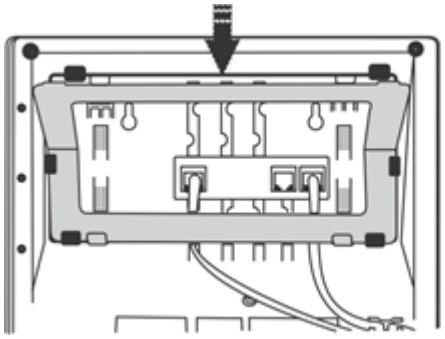


Mount the phone on the Desk

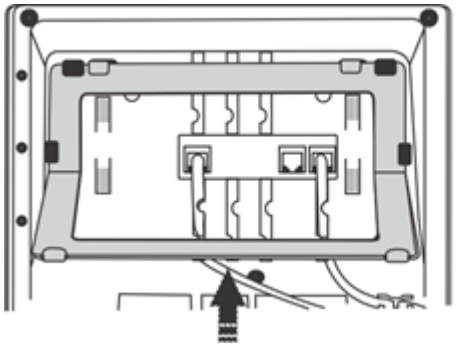
- You can attach the Foot Stand in the following ways—at an angle of **20° Angle** or at **30° Angle** or at **50° Angle**.



Foot Stand at
20 degree angle



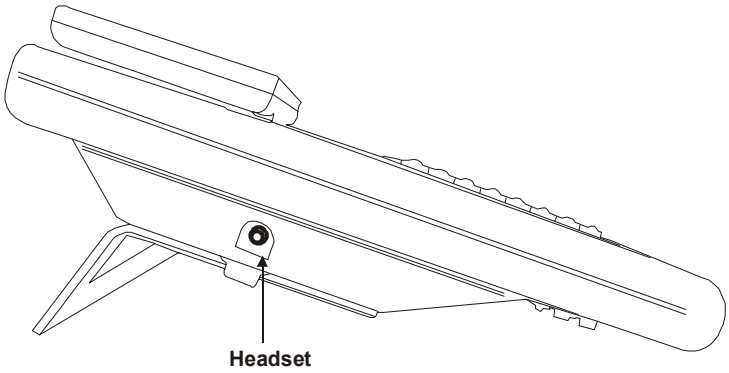
Foot Stand at
30 degree angle



Foot Stand at
50 degree angle

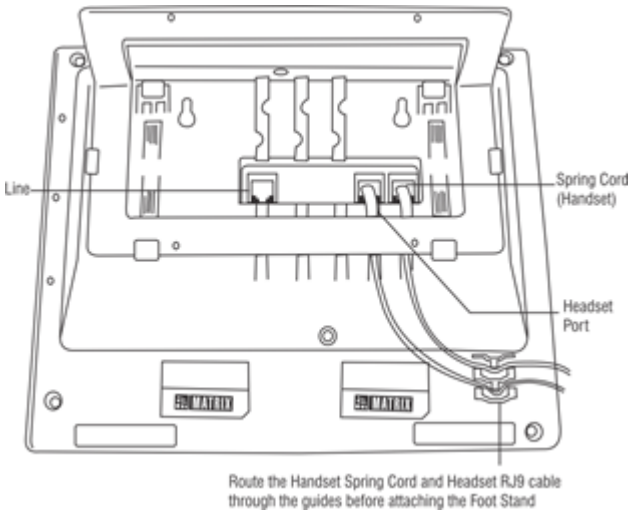
- If you attach the Foot Stand at 50°, the phone will be placed in an almost upright position on your desk.
- Decide which of these positions would work for you best and accordingly attach the Foot Stand.
- Connect the handset of the EON48 to the phone body using the spring cord.

- To use a Headset (not supplied with the phone), plug any standard stereo headset with 2.5mm single connector into the headset jack with the symbol on the left side panel of the phone.

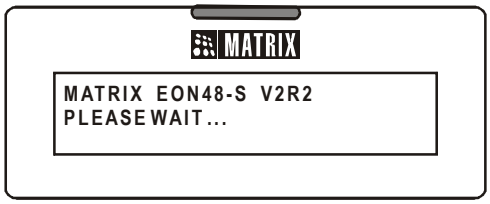


You may also plug in a stereo headset with an RJ9 connector into the headset port at the bottom of the phone, marked with the symbol .

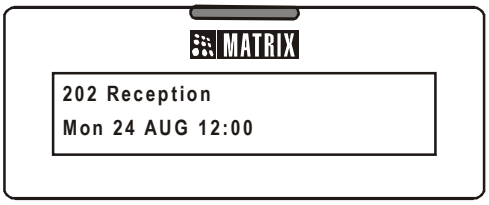
- Plug one end of the RJ11 cable supplied with the phone into the RJ11 connector of the phone labeled as '**LINE**' and the other end into the wall jack/DKP Port.



- When the system is powered ON, the EON will reset. The EON communicates with the system. The handshaking lasts for 5-6 seconds. The EON model, version and revision number, along with the message *'Please wait'…* appear on the LCD display.



- After successful handshaking and reset cycle, if the DKP Parameters have been configured, the LCD display of the EON will show the extension number and the extension name in one line and the day, date and time and the time zone in the other line.



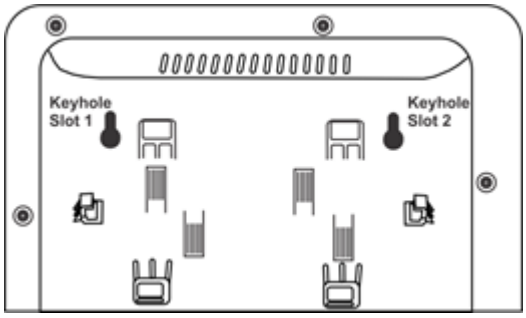
- You may adjust the LCD for brightness, contrast and backlight. Refer the topic, “Digital Key Phone-Operation” for instructions.

Installing EON310

- Unpack the box and verify the package contents.
- You can mount the phone on a wall or on desk.

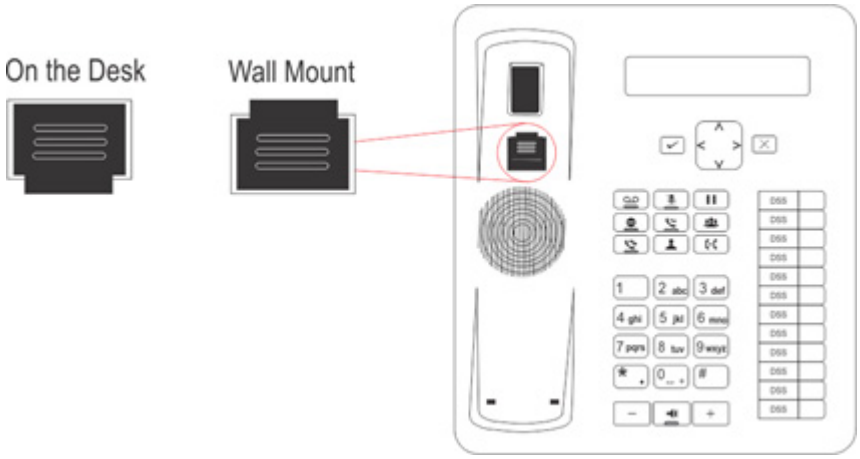
Mount the phone on a Wall

- Use the mounting template to drill holes of appropriate size and distance.
- Fix the screw grips in the holes you drilled.
- Fix two screws in the holes on the wall, ensuring that they are aligned with the Keyhole Slots 1 and 2. The screws should protrude from the wall to fit into the keyhole slots.



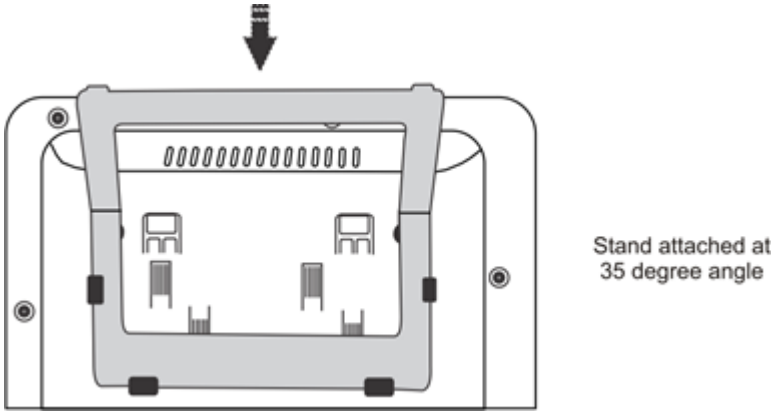
- Now, mount the phone with the screws fitting into the keyhole slots.

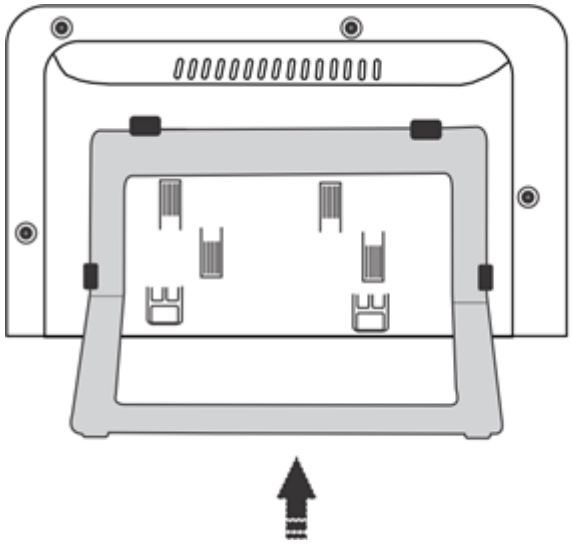
- Reverse the handset wall mount tab to make sure that the handset remains intact when you mount the phone. Push the handset wall mount tab upwards to remove it from the slot. Rotate it 180 degrees clockwise and push it downwards into the slot.



Mount the phone on the Desk

- You can attach the Foot Stand in the following ways—at an angle of **35° Angle** or at **50° Angle**.

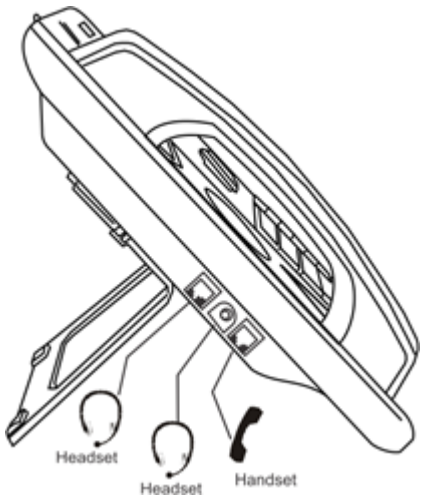




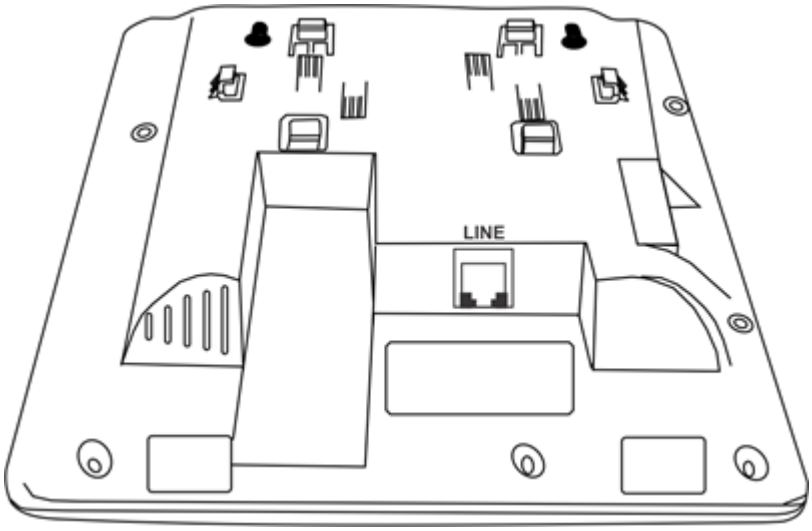
Stand attached at
50 degree angle

- Decide which of these positions would work for you best and accordingly attach the Foot Stand.
- Connect the handset of the EON310 to the phone body using the spring cord.
- To use a Headset (not supplied with the phone), plug any standard stereo headset with 3.5mm single connector into the headset jack with the symbol on the left side panel of the phone.

  You may also plug in a stereo headset with an RJ9 connector into the headset port marked with the symbol , on the left side panel of the phone.



- Plug one end of the RJ11 cable supplied with the phone into the RJ11 connector of the phone labeled as '**LINE**' and the other end into the wall jack/DKP Port.



- When the system is powered ON, the EON will get reset and the message "Welcome to Matrix. Booting"…appears on the LCD display.

- The EON communicates with the system. The handshaking lasts for 5-6 seconds. The EON model, version and revision number, along with the message "Please Wait"…appears on the LCD display.

- After successful handshaking and reset cycle, the extension number, day, date and time will appear on the LCD of the phone. If you have already assigned extension number and name, in the DKP Parameters, these will appear on the LCD.

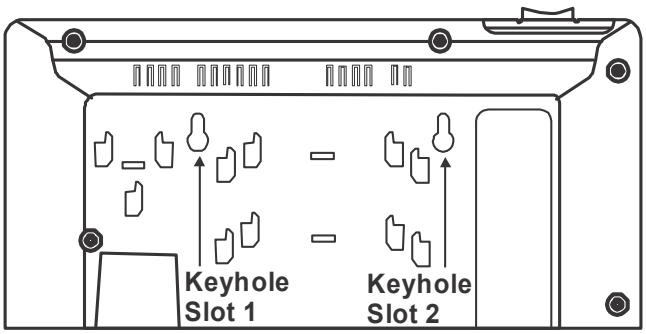
- You may adjust the LCD for brightness, contrast and backlight. Refer the topic, “Digital Key Phone-Operation”.

Installing EON510

- Unpack the box and verify the package contents.
- You can mount the phone on a wall or on desk.

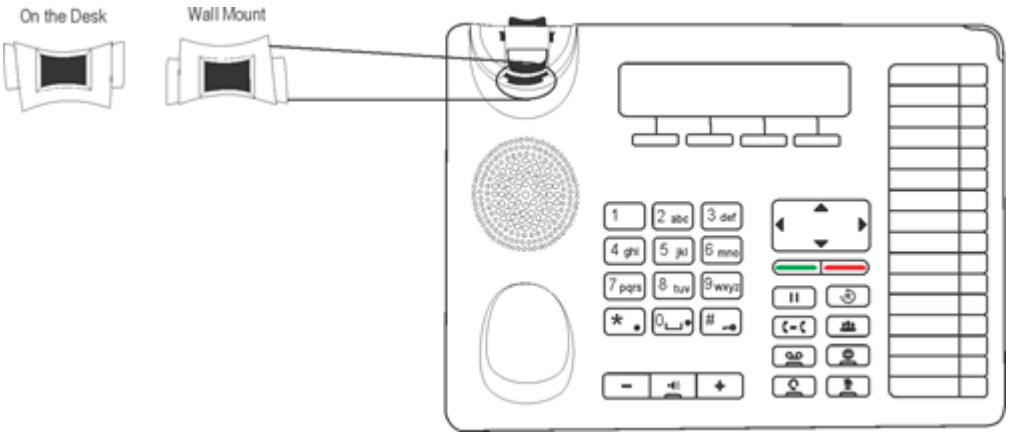
Mount the phone on a Wall

- Use the mounting template to drill holes of appropriate size and distance.
- Fix the screw grips in the holes you drilled.
- Fix two screws in the holes on the wall, ensuring that they are aligned with the Keyhole Slots 1 and 2. The screws should protrude from the wall to fit into the keyhole slots.



- Now, mount the phone with the screws fitting into the keyhole slots.

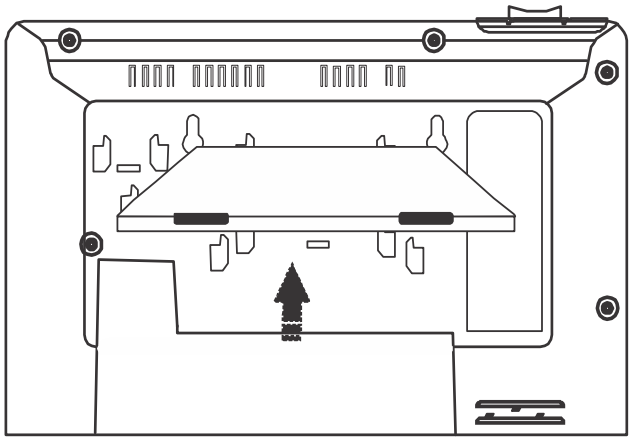
- Reverse the handset wall mount tab to make sure that the handset remains intact when you mount the phone. Push the handset wall mount tab upwards to remove it from the slot. Rotate it 180 degrees clockwise and push it downwards into the slot.



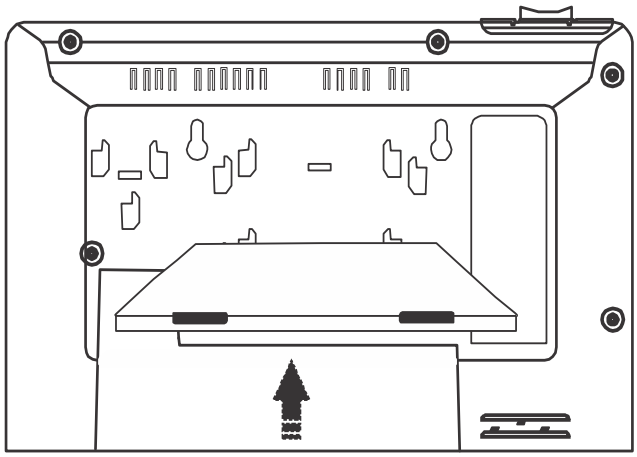
If you are unable to remove the wall mount tab, you may use a tool like a minus screw driver to remove it.

Mount the phone on the Desk

- You can attach the Foot Stand in the following ways—at an angle of **45° Angle** or at **55° Angle**.

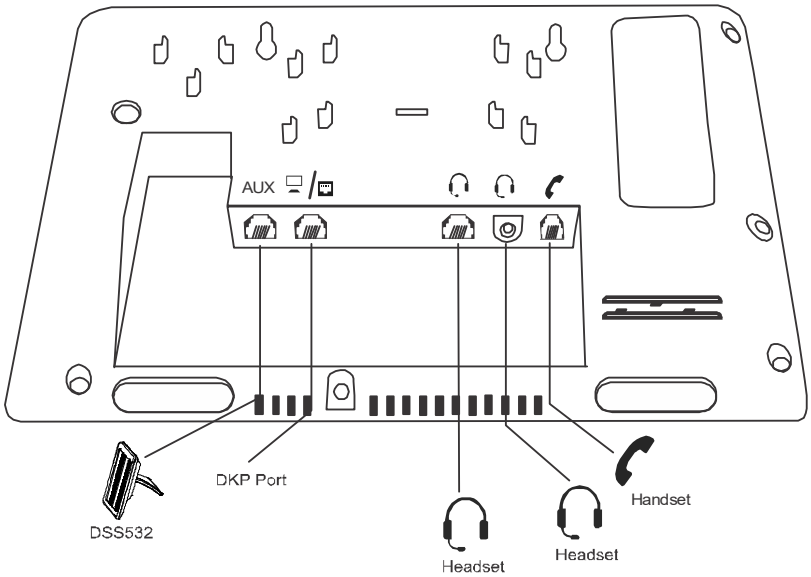


Stand attached at
45 degree angle



Stand attached at
55 degree angle

- Decide which of these positions would work for you best and accordingly attach the Foot Stand.



- Connect the handset of the EON510 to the phone body using the spring cord.
- To use a Headset (not supplied with the phone), plug any standard stereo headset with 3.5mm single connector into the headset jack with the symbol on the left side panel of the phone.

  You may also plug in a stereo headset with an RJ9 connector into the headset port marked with the symbol , on the left side panel of the phone.
- Plug one end of the RJ11 cable supplied with the phone into the RJ11 connector of the phone marked with the symbol and the other end into the wall jack/DKP Port.
- To connect DSS532 with the phone, plug one end of the RJ11 cable into the AUX Port of the phone and the other end into the IN Port of the DSS532. For installation, see “Installing DSS532 with EON510”.

- When the system is powered ON, the EON will get reset.and the message 'Phone is Booting; Please wait…' appear on the LCD display.

- The EON communicates with the system. The handshaking lasts for 5-6 seconds. The message 'Loading Firmware Version-Revision; Please wait…' appear on the LCD display.

- After successful handshaking and reset cycle, the extension number, day, date and time will appear on the LCD of the phone. If you have already assigned extension number and name, in the DKP Parameters, these will appear, as illustrated below.

You may adjust the LCD for brightness, contrast and backlight. Refer the topic, “Digital Key Phone-Operation”.

Installing DSS Consoles

Installing DSS64

Once you have installed EON48/310 with SARVAM UCS, installing the DSS Consoles can be done in a few simple steps, very much similar to those involved in the installation of EON.

1. Unpack the box and verify the package contents[35].
2. Place the DSS Console next to the DKP to which it is to be attached.
3. Decide which DKP Ports on the DKP Card are to be assigned to the DSS Consoles. You may select any free (unused) port on the card for DSS Consoles. It is not necessary for the DSS Console ports to be in a sequence with the DKP ports to which they are attached.

35. See “Packing List” of Appendix topic.

For example: you have connected DKP1 to Port 1 on the first RJ45 connector of the DKP8 card. You want to attach two DSS Consoles to DKP1. The two DSS Consoles may be connected to any port on the second connector of the card, not necessarily to Port 2 and Port 3 on the first connector.

4. The wire-pairs from the DKP Ports designated for DSS Consoles should be terminated on the bottom of the Krone Connector (of 'Station Lines' on the MDF).

5. The wire-pairs of the DSS Consoles should be terminated into the top of the Krone Connector (of 'Station Lines' on the MDF). Refer the topic "The Main Distribution Frame (MDF)" for illustration.

   You can connect maximum two DSS64 with a single EON48/310.

6. The system automatically detects the DSS Console you connect and it will be will appear under **Unassigned DSS64** in "DSS Status". You must first assign these DSS Consoles to the respective DKP Ports and thereafter you will be able to configure the DSS Keys.

7. To assign the DSS Consoles, see "DSS Status" and to configure the DSS Keys, see "Programming DSS Console Keys".

Installing DSS532 with EON510

The instructions for installing DSS532 with EON510 or SPARSH VP510 are same. For detailed instructions, refer to "Installing DSS532 with SPARSH VP510".

Installing EONSOFT

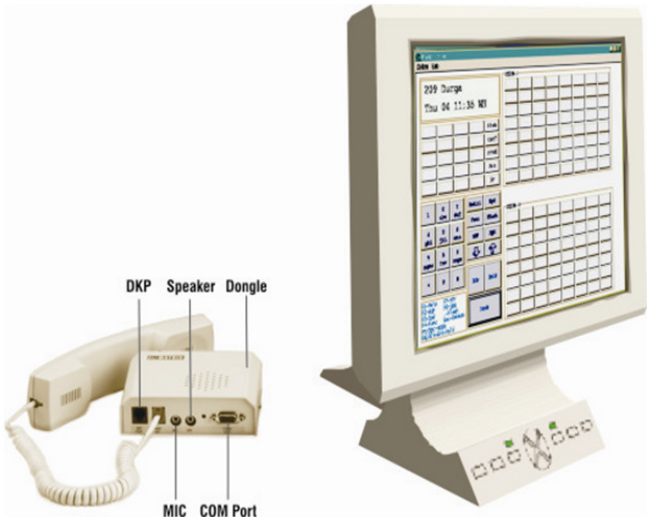
To install EONSOFT, you must have a computer with Windows as the operating system. The EONSOFT is compatible with the following Operating Systems of Windows:

- Windows 98
- Windows XP
- Windows NT
- Windows 2003
- Windows Vista
- Windows 2007

1. Unpack the box and verify the package contents[36].

36. See "EONSOFT" under 'Packing List' of Appendix topic.

2. Connect the Handset to the dongle in the handset jack. If using a headset, connect the microphone and the speaker connectors into the dongle.



3. Connect one end of the Communication cable to the COM port of the dongle. Connect the other end of the communication cable into the COM port of the computer.

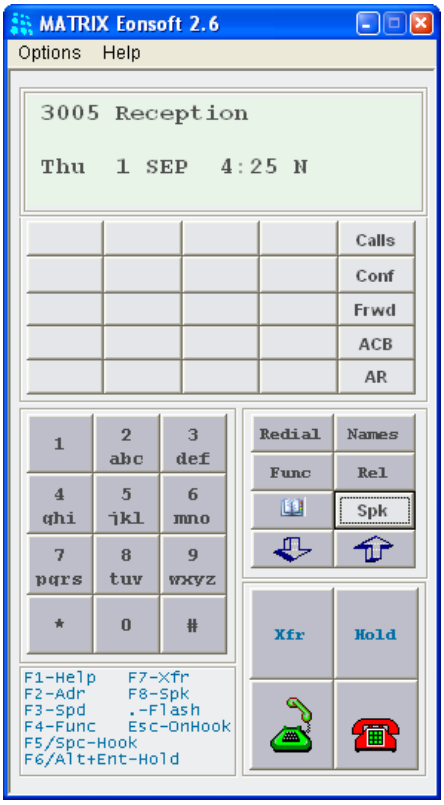
4. Connect a wire-pair of a DKP port to the RJ11 port marked 'DKP' on the dongle.

5. Switch ON the computer. The computer must have Windows Operating System installed on it.

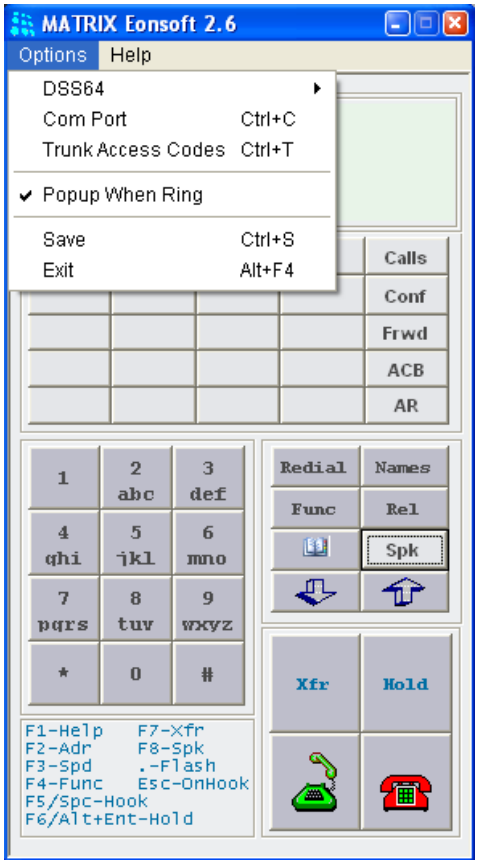
6. Copy the EONSOFT Application Software provided by the Support Team onto your PC and install the application.

7. After the program has been installed and run, a shortcut will be automatically created and appear on your desktop.

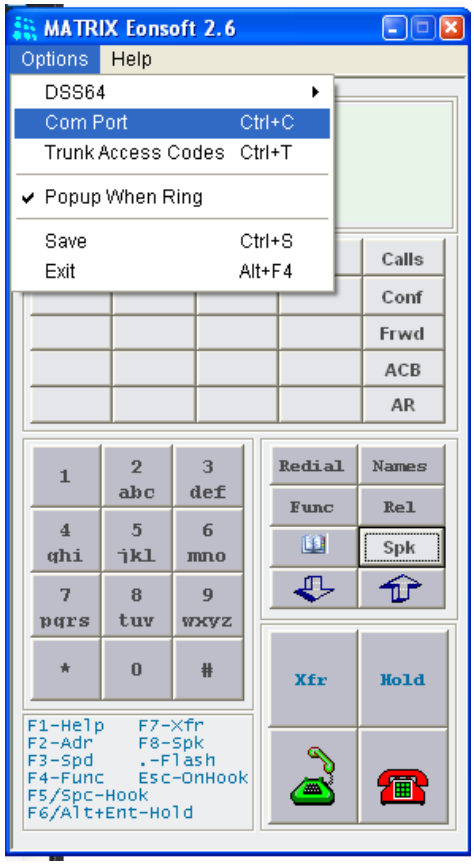
8. Click the shortcut to open the program. The EONSOFT window will open:



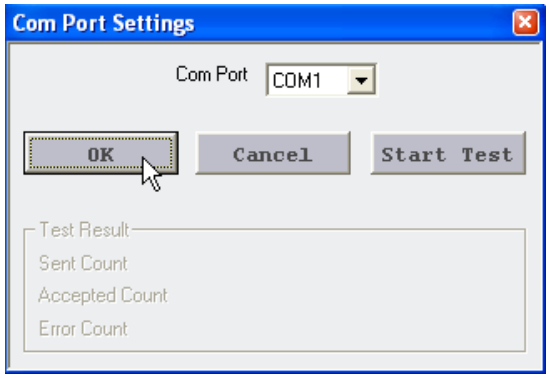
9. Click **Options** at the top left of the window. A drop down menu will appear.



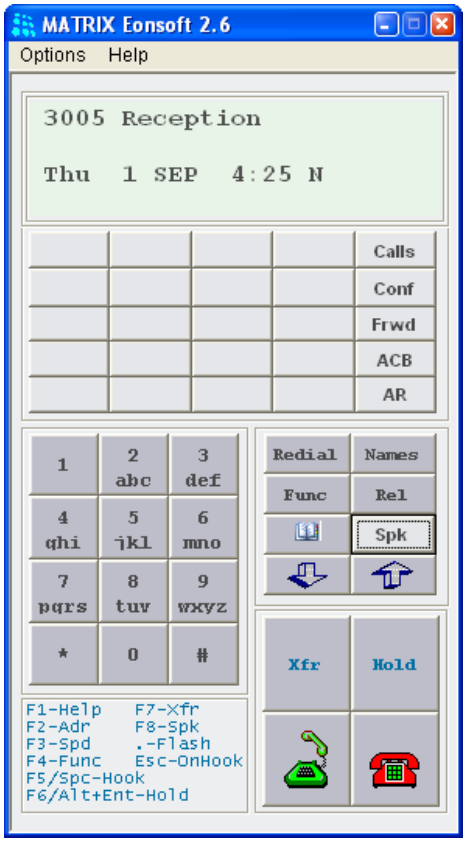
10. Click the option **COM Port**.



11. Select the COM Port to which the communication cable is connected.



12. EONSOFT is now connected. If you have already configured the DKP parameters like Access Code and Name for the port to which EONSOFT is connected, these will appear.



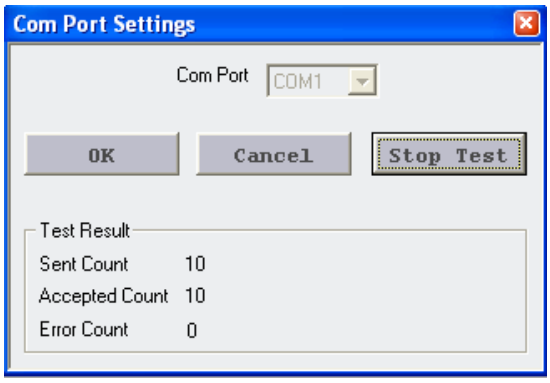
- *If this window does not appear after you have selected the COM Port Option, test the COM Port for data transfer.*
- *If the wrong COM port has been selected, a dialog box will pop up on your screen with the message: "COMx is invalid or busy, please select another COM Port". Select the correct COM Port.*

Test the functioning of the COM Port of the PC and the communication cable, before you install the EONSOFT.

Testing the COM Port

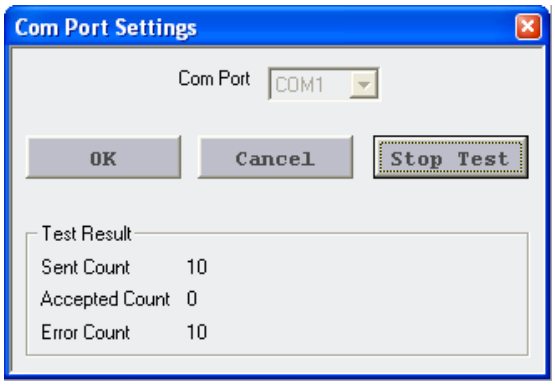
- From the drop down menu of **Options**, select the **COM Port** to which you have connected the communication cable.
- The COM Port Settings dialog box will open.
- Connect the communication cable to the COM Port of the PC.
- Short pin2 and pin3 of the DB-9 connector at the free end of the cable.
- Click the button labeled **Start Test** in the COM Port Settings dialog box.
- After clicking this button, observe the Test Result section on the dialog box.

- The **Error Count** value shows zero, if both the communication cable and the COM port are working.



The above screen shows that the COM Port/communication cable is working.

- If the **Error Count** shows a value other than zero, it means that either the communication cable or the COM port of the PC is faulty.



Above screen shows the faulty COM Port/Communication Cable.

- Remove the communication cable from the COM Port of the PC.
- Short pin2 and pin3 of the communication port of the computer and click 'Start Test' in the COM Port Settings dialog box.
- Now, if the error count is zero, please check the Communication Cable.
- If the error count is not a zero, the COM Port of the PC is faulty. Try another communication port.

The CO Card

The CO Card provides the interface to connect the system with the Two-Wire Analog Trunk lines from the CO Network. The CO Card supports the different standards and features of CO Networks across the world.

The CO Card is available in the following configurations. CO interface is also available in combination with SLT ports on a single card.

CO Cards for ETERNITY LENX

Card Name	Configuration and Application
ETERNITY ME Card CO16	16-port card to connect 16 Two-wire Trunk lines from the CO network
ETERNITY ME Card CO8	8-port card to connect 8 Two-wire Trunk lines from the CO network
ETERNITY ME Card CO8+SLT24	Combination card, with 8 CO ports to connect 8 CO analog trunk lines and 24 SLT ports to connect 24 Single Line Telephones This Card supports Power Fail Transfer. To know more, see “Power Fail Transfer”.

Choose a CO Card with the configuration that meets your requirement for CO trunk ports, keeping in mind the maximum CO Trunk Port capacity of the system you are installing.

The maximum CO Trunk Ports supported are 128.

Connectors

The CO Card has RJ45 connectors, with 4 CO ports on each connector. A multi-pair, MDF cable is supplied for each connector on the card.

LED

The CO Cards have a single tri-color LED to indicate:

- the health of the card during the Reset Cycle.
- the status of a selected Trunk port during normal functioning of the system.

You can assign the LED to any CO port on the card which you want to monitor[37].

Installing the CO Card

For CO connectivity, you must install at least one of the above mentioned CO Cards in the system.

1. Take all the necessary precautions prescribed for handling the cards and electronic equipment. Make sure that power supply is turned off before you begin the installation of the card. Put on an electrostatic-discharge preventive wrist strap/belt and use a grounding mat.

37. *To assign the LED to a selected port for monitoring its functioning, you must enter SE mode and dial the SE Command 7902-Slot-LED Number-Port, where Slot is the number of the universal slot in which the card is installed and Port is the port on the card to which the LED is to be assigned to monitor its functioning. LED Number is the number of the LED on the card, which will monitor the port.*

2. Unpack the CO card and check the package contents.

3. Select any free (empty) slot from the Universal Slots. Unscrew and remove the filler bracket of the empty slot. Preserve the filler bracket for future use!

4. Insert the CO Card into the guide rails of the free slot you selected for the card. The connectors on the card should make perfect contact with those of the slot on the backplane motherboard.

5. Press down the lever on the card mounting brackets to secure the card in its slot. Fix the mounting bracket in place with the two screws provided.

If installing more than one CO Card, it is not necessary to insert the other cards in subsequent slots. Any card can be inserted in any of the Universal Slots.

6. Use the cables supplied for each connector on the CO Card to connect the Trunk Lines with the Main Distribution Frame.

   Refer the illustrations below for the pinout details of the connectors on each card.

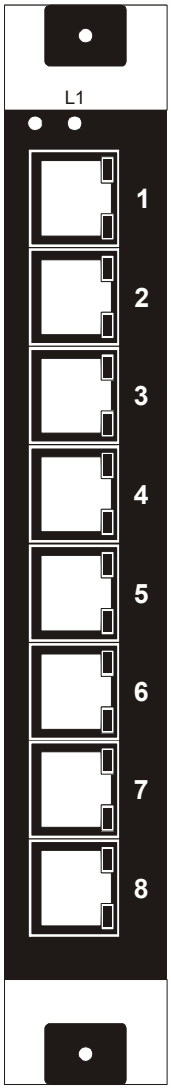
ETERNITY ME Card CO16

Connector	Color	Connection	H/w Port Offset
RJ45-1	Blue - (Blue & White)	CO	01
	Orange - (Orange & White)	CO	02
	Green - (Green & White)	CO	03
	Brown - (Brown & White)	CO	04
RJ45-2	Blue - (Blue & White)	CO	05
	Orange - (Orange & White)	CO	06
	Green - (Green & White)	CO	07
	Brown - (Brown & White)	CO	08
RJ45-3	Blue - (Blue & White)	CO	09
	Orange - (Orange & White)	CO	10
	Green - (Green & White)	CO	11
	Brown - (Brown & White)	CO	12
RJ45-4	Blue - (Blue & White)	CO	13
	Orange - (Orange & White)	CO	14
	Green - (Green & White)	CO	15
	Brown - (Brown & White)	CO	16

ETERNITY ME Card CO8

Connector	Color	Connection	H/w Port Offset
RJ45-1	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	CO CO CO CO	01 02 03 04
RJ45-2	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	CO CO CO CO	05 06 07 08
RJ45-3	Unused		
RJ45-4	Unused		

ETERNITY ME Card CO8+SLT24



Connector	Color	Connection	H/w Port Offset
RJ45-1	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	01 02 03 04
RJ45-2	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	05 06 07 08
RJ45-3	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	09 10 11 12
RJ45-4	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	13 14 15 16
RJ45-5	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	17 18 19 20
RJ45-6	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	21 22 23 24
RJ45-7	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	CO CO CO CO	01 02 03 04
RJ45-8	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	CO CO CO CO	05 06 07 08

7. Plug in the RJ45 end of the Trunk Card cables into the respective connectors.

8. Terminate the free end of the CO Card cable into the punch down blocks of the Krone modules designated for 'Trunk Lines' on “The Main Distribution Frame (MDF)”.

   Trunk cables from the system are to be connected with the Trunk Lines from the PSTN/CO terminated on the MDF. Each wire-pair form the CO Port must be terminated on the bottom of the Krone Connector, while the wire-pair of the trunk line from the CO Network to be connected to this port must be terminated on the top of the Krone Connector.

   Refer the topics “The Main Distribution Frame (MDF)” and “Terminating Trunk and Extension Cables on the MDF”.

9. Repeat these steps to install other CO Cards, if applicable.

10. If you have completed all other installation tasks, power ON the system.

LED Pattern of the CO Card

Stage	LED Color	Cadence
Auto Upgradation[a]		
Card waiting for application	RED	ON-200ms-OFF 200ms
Card is up, loaded with new application	GREEN	ON-200ms-OFF 200ms
Initialization		
	RED	ON 500ms-OFF 500ms
	GREEN	ON 500ms - OFF 500ms
	ORANGE	ON 500ms - OFF 500ms
Stand-by task	ORANGE	1 sec GREEN - 1Sec Orange
Errors		
Flash Failure	None	None
RAM Failure	None	None

a. Done by the boot loader application.

Status of Selected CO Port

PORT Status	LED Color	LED Cadence
Selected CO's data are transmitted to CPU Card	RED	Toggle[a] on each event
Selected CO's data are received from CPU Card	RED	Toggle[b] on each request from Master

a. The current LED state will remain the same until the next event is received from the application on the CO Port. For example, if the current LED state is Green/Red ON, on the next event, the LED will be turned OFF. It will remain OFF until the next event occurs. When the next event is received it will be turned Green/Red ON again. This process continues.

b. Same as above note.

Jumpers on the Main Board

Jumper Number	Position	Function
J1	AB (default)	Normal Operation
	BC	For uploading software using COM Port
J2 & J4	AB	NA
	BC (default)	Normal Operation/ Debug

The BRI Card

The BRI Card provides the interface to connect system with ISDN BRI Lines. The BRI lines may be from a public ISDN exchange, a private ISDN exchange.

BRI Cards of ETERNITY LENX

Card Name	Configuration and Application
ETERNITY ME Card BRI8	8-Port card to connect 8 ISDN BRI Lines or ISDN Compatible Devices
ETERNITY ME Card BRI4	4-Port card to connect 4 ISDN BRI Lines or ISDN Compatible Devices

The maximum number of BRI lines supported are 32.

Connectors

The BRI cards have RJ45 connectors. The ETERNITY ME BRI8 card has 8 RJ45 connectors for 8 BRI ports.

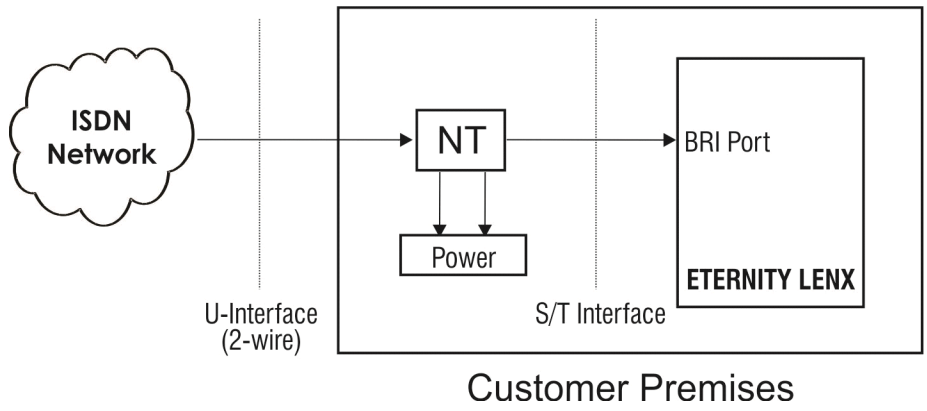
The ETERNITY ME BRI4 card has 4 RJ45 connectors for 4 BRI ports. A separate cable is supplied for each connector.

LEDs

The ETERNITY ME BRI8 and ETERNITY ME BRI4 has 4 LEDs.

ISDN BRI - Installation Scenarios

Most ISDN Service Providers also provide the NT1 device along with the BRI line. The BRI Line from the ISDN central office is terminated on the NT1 on the Customer's Premises, as illustrated below.



Where,

- U Interface = between the NT1 equipment and the ISDN central office.
- S/T Interface = between the ISDN user equipment, in this case, ETERNITY LENX and the Network Interface Equipment (NT1).

The BRI line is terminated on the NT1. The S/T interface of the NT1 is connected to BRI port of the ETERNITY LENX.

TE and NT Modes

In this illustration, the BRI line from ISDN Service Provider is directly connected to BRI port of the ETERNITY LENX via the NT1 device. Here, the ETERNITY LENX is the Terminal Equipment, so the BRI Port must be programmed to work in the TE mode.

When an ISDN Phone is to be connected to the BRI port of ETERNITY LENX, the BRI port must be programmed to work in NT mode.

When a BRI port of another ISDN System is to be connected to the BRI port of the ETERNITY LENX, in such a configuration, you may configure

- the BRI port of the other ISDN System in the TE mode and the BRI Port of the ETERNITY LENX in the NT mode.

  OR

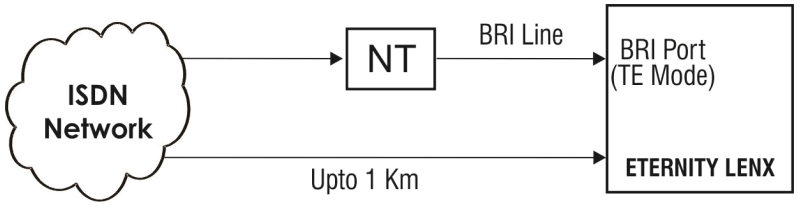
- the BRI port of the other ISDN System in the NT mode and the BRI Port of the ETERNITY LENX in the TE mode

Also refer the topic “Configuring BRI Trunks” to know more.

Types of BRI Configuration

There are two types of configurations in BRI: Point-to-Point Configuration and Point-to-Multipoint Configuration. Each of these is discussed below.

Point-to-Point Configuration



The maximum distance between the NT (Network Termination, NT1 or NT2) and a single Terminal Equipment, in this case ETERNITY LENX, can be up to 1 kilometer.

Point-to-Multipoint Configuration

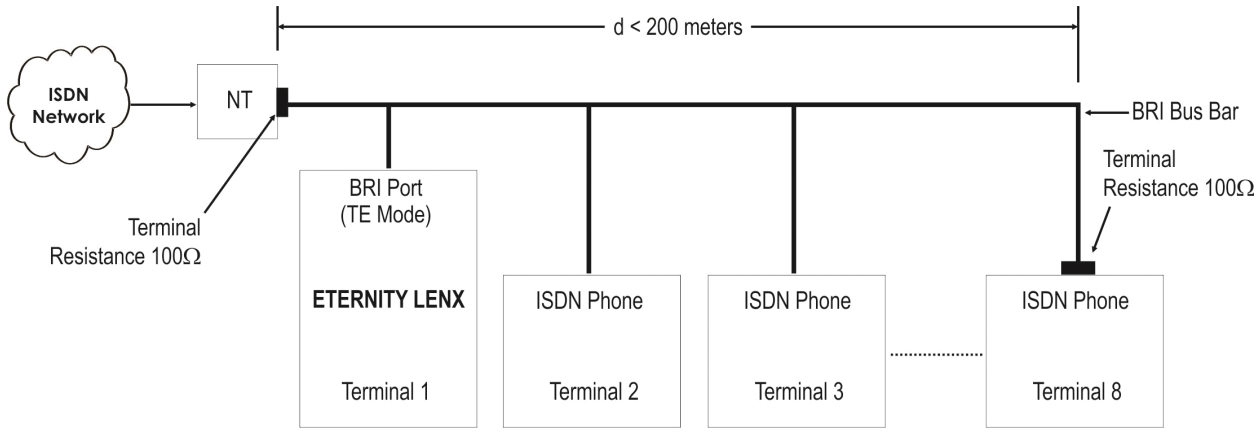
A maximum of 8 ISDN equipment can be connected on a single BRI Bus line in a Point-to-Multipoint configuration.

Further, two configurations are possible in a Point-to-Multipoint configuration:

a. Short Passive Bus Configuration

b. Extended Passive Bus Configuration

Short Passive Bus Configuration

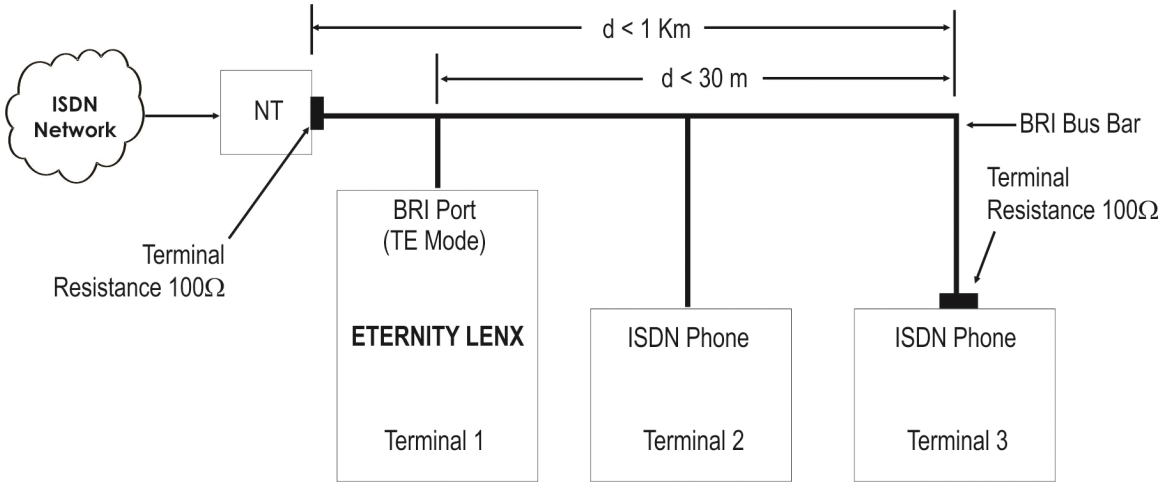


Where,
TE = Terminal Equipment or ISDN device (End user device)
NT = Network Termination provided by the ISDN Service Provider
d = distance from NT to the last TE equipment.

In a Short Passive Bus Configuration,

- A maximum of 8 TEs or ISDN devices can be connected to a single NT on a bus up to 200 meters from the NT.
- 100Ω Terminal Resistance is required to be inserted at the NT side as well as the last TE Equipment as shown in the figure.
- Using this configuration, any subscriber from ETERNITY LENX can access a BRI line and can make outgoing calls. At the same time, another subscriber from ETERNITY LENX or any ISDN phone shown in the figure can make outgoing call from the same BRI. In the same way, incoming calls are possible on the same BRI.
- Only two simultaneous speech paths can be established, as BRI supports 2 voice channels only.
- This configuration is useful on the smaller premises, where a single BRI line and multiple ISDN devices are used.

Extended Passive Bus Configuration



Where,
TE = Terminal equipment of any ISDN Equipment
NT = Network Termination provided by Service Provider
TR Terminal Resistance 100Ω
d = distance from NT to the last TE Equipment
d1 = the total distance from first TE equipment and the last TE equipment.

In an Extended Passive Bus Configuration,

- You can connect only 3 Terminal Equipment or ISDN devices. These devices are grouped together at one end of the bus, with may extend to a distance of up to 1 kilometer from the NT.
- However, all the 3 Terminal Equipment/ISDN devices must be located within a range of 30 meters, as shown in the figure.
- Using this configuration, any subscriber from ETERNITY LENX can access the BRI line and make outgoing calls. At the same time, another subscriber from the ETERNITY LENX or any ISDN phone shown in the figure can make outgoing calls from the same BRI. In the same way, incoming calls are possible on the same BRI.
- Only two simultaneous speech paths can be established, as BRI supports 2 voice channels only.
- This configuration is useful on large premises where a limited number of ISDN devices (maximum 3) are to be used within a range of 30 meters.

Installing the BRI Card

1. Take all the necessary precautions prescribed for handling the cards and electronic equipment: turn off power supply, always wear an electrostatic-discharge preventive wrist strap/belt and use a grounding mat.
2. Unpack the BRI Card and check the package contents.
3. Select any free (empty) slot from the Universal Slots. Unscrew and remove the filler bracket of the empty slot. Do not discard the filler bracket! Preserve it for future use!

Setting Orientation Type of BRI Port

4. The BRI Ports can be configured for different applications and can be interfaced directly with the BRI Network with Terminal Equipment like an ISDN Phone, with an ISDN-System.

   To connect the BRI Port to the public network, BRI Port must configured in the TE mode.

   To connect ISDN phones, an ISDN System or any ISDN equipment, the BRI Port must be configured in the NT mode.

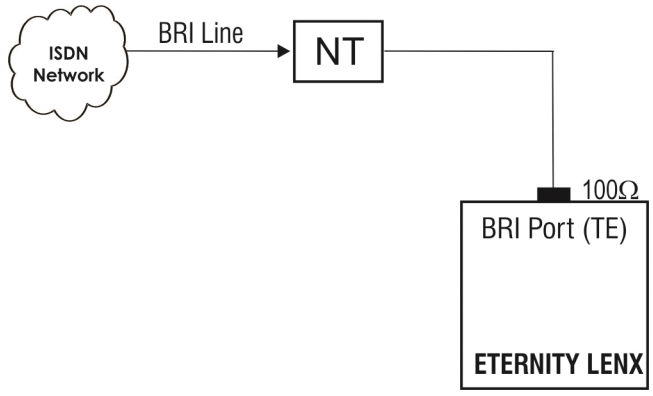
   By default, BRI Ports are configured in the TE mode.

   To set Orientation Type of the BRI Port, under **Configuration**, open the **BRI Configuration** link, and under **BRI Parameters**, set the **Orientation Type**.

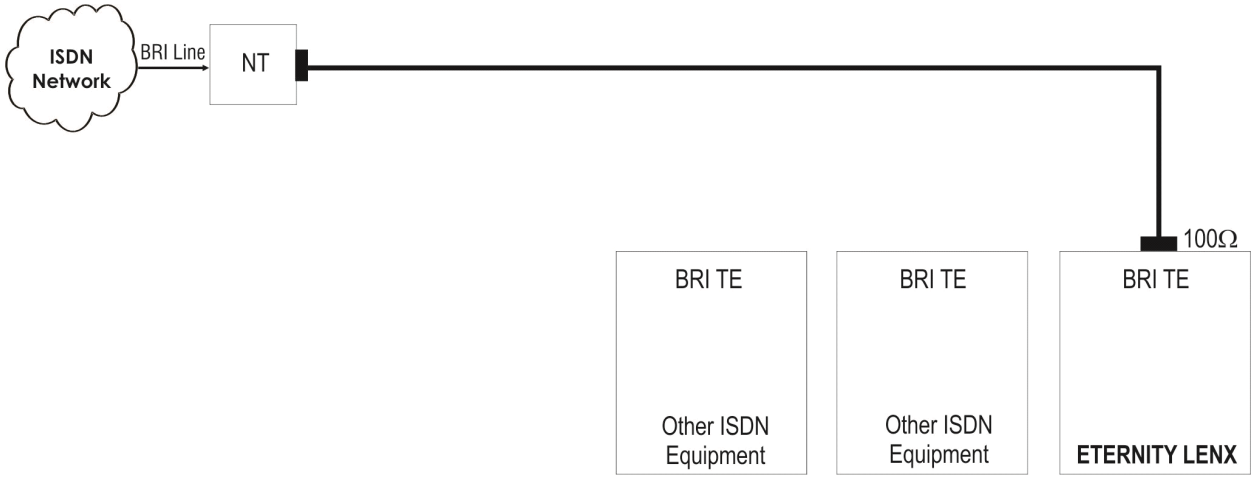
Inserting Termination Resistance

5. Termination of 100Ω should be inserted in the following cases:

   - When the BRI port is configured in the TE mode and connected in a Point-to-Point configuration as shown below.



   - When the BRI port is configured in the TE mode in a Point-to-Multipoint configuration as shown below. 100Ω Termination is required on the last Terminal connected on the S0 bus to terminate calls properly.



In a Point-to-Multipoint configuration, 100Ω termination can be provided on either of the following:
- Last TE equipment
- Last point of the bus bar where the last TE equipment is connected.

- When BRI port is configured in the NT mode.

- *If the S0 bus itself supports Terminating resistors, Termination Resistance need not be inserted when*
  - *BRI Port is configured as TE and connected in a Point-to-Point Configuration as illustrated above.*
  - *BRI Port is configured as NT.*

- *Termination need not be inserted if the BRI port of ETERNITY LENX (configured in TE mode) is connected as any terminal other than the last terminal on the S0 bus (in a Multi-point configuration).*

Termination in TE Equipment (BRI Port)

6. To set the 100Ω termination on the BRI port set the Jumpers on the BRI Module (daughter-board) to the position described below:

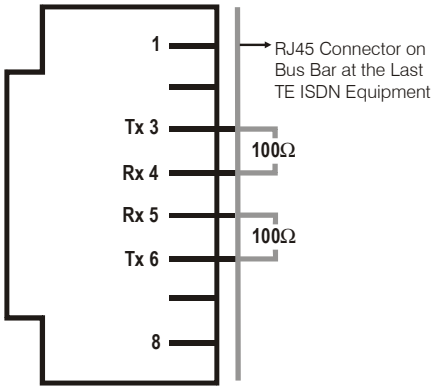
Function	**Module 2 (M2)**				**Module 3 (M3)**			
	BRI Port 1		BRI Port 2		BRI Port 3		BRI Port 4	
	Jumper Position		Jumper Position		Jumper Position		Jumper Position	
	J6	J8	J7	J9	J6	J8	J7	J9
To insert 100Ω termination	AB	AB	AB	AB	AB	AB	AB	AB
To remove 100Ω termination	BC	BC	BC	BC	BC	BC	BC	BC

Function	**Module 4 (M4)**				**Module 5 (M5)**			
	BRI Port 5		BRI Port 6		BRI Port 7		BRI Port 8	
	Jumper Position		Jumper Position		Jumper Position		Jumper Position	
	J6	J8	J7	J9	J6	J8	J7	J9
To insert 100Ω termination	AB	AB	AB	AB	AB	AB	AB	AB
To remove 100Ω termination	BC	BC	BC	BC	BC	BC	BC	BC

By default, Termination Resistance of 100Ω is set on the BRI port (the Jumpers are in AB position).

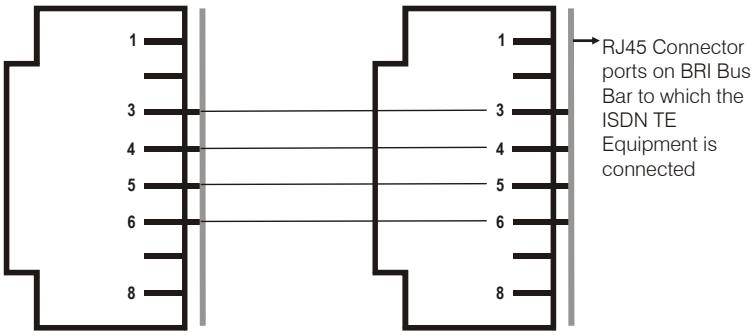
Termination in the Bus Bar

7. 100Ω termination resister can be connected between TX and RX, between pin number 3-4 and 5-6 in the RJ45 connector as illustrated below.



As shown in the application diagrams for Point-to-Multipoint connectivity, each ISDN TE device is connected in a Bus Bar, which may be Short Passive Bus Bar configuration or an Extended Passive Bus Bar configuration.

Illustrated below is the connection diagram of two ports connected with each other on the same BRI bus bar.



- The above figure shows the connection details of two ports on the BRI Bus Bar. Similarly, you can connect 8 ports on the Bus Bar, keeping in mind the Termination Resister for the NT and the Last TE on the Bus bar.
- Pin number 3, 4, 5 and 6 of the RJ45 connector are used for connectivity.
- Pin number 3 and 6 are used for Transmit (Tx) and pin number 4 and 5 are used for Receive (Rx) from the ISDN TE side.
- Pin number 3 and 6 are used for Receive (Rx) and pin number 4 and 5 are used for Transmit (Tx) from the NT side.

Feeding Power to the Terminal

8. When the BRI Port of the ETERNITY LENX is used as BRI-NT, you can feed power to the terminal equipment connected to the BRI-NT Port from the ETERNITY LENX.

   To do this,

   - Enable Feed Power on the BRI Port. For instructions see Power Feed under “Configuring BRI Trunks”.

- By default, the Jumpers are set in AB position to feed power through Tx and Rx wires (Phantom Power). If you want to feed power through a separate pair of wires, you may change the position of the Jumpers on the BRI module as mentioned in the table below.

Function	**Module 2 (M2)**				**Module 3 (M3)**			
	BRI Port 1		BRI Port 2		BRI Port 3		BRI Port 4	
	Jumper Position		Jumper Position		Jumper Position		Jumper Position	
	J4	J5	J2	J3	J4	J5	J2	J3
To feed power on Tx and Rx wires (Phantom Power)	AB	AB	AB	AB	AB	AB	AB	AB
To feed power on separate pair of wires	BC	BC	BC	BC	BC	BC	BC	BC

Function	**Module 4 (M4)**				**Module 5 (M5)**			
	BRI Port 5		BRI Port 6		BRI Port 7		BRI Port 8	
	Jumper Position		Jumper Position		Jumper Position		Jumper Position	
	J4	J5	J2	J3	J4	J5	J2	J3
To feed power on Tx and Rx wires (Phantom Power)	AB	AB	AB	AB	AB	AB	AB	AB
To feed power on separate pair of wires	BC	BC	BC	BC	BC	BC	BC	BC

- *The maximum power that can be fed to a single BRI port is 50mA.*
- *From signaling point of view, a maximum of 8 terminal equipment can be connected on the BRI port configured in the NT mode.*
- *The number of ISDN Terminals that can be connected on the BRI port configured in the NT mode depends on the power consumed by the ISDN terminals.*

9. Insert the BRI Card into the guide rails of the free slot you selected for the card. The connectors on the card should make perfect contact with those of the slot on the backplane motherboard.

Press down the lever on the card mounting brackets to secure the card in its slot. Fix the mounting bracket in place with the two screws provided.

If installing more than one BRI Card, it is not necessary to insert the other cards in subsequent slots. Any card can be inserted in any of the Universal Slots. Remember to set the Orientation Type, Termination Resistance and Power Feed, as required.

10. Use the straight cables supplied for each connector on the BRI card to connect the BRI Ports to the NT1 device supplied by your ISDN service provider. Refer the configuration and pinout details given below for guidance.

Configuration details of the U interface (RJ-45) at NT1

Pin Number	Pin Details
4	Tx
5	Rx

Configuration details of the S/T interface (RJ-45) on NT1

Pin Number	Pin Details
3	Rx1
4	Tx1
5	Tx2
6	Rx2

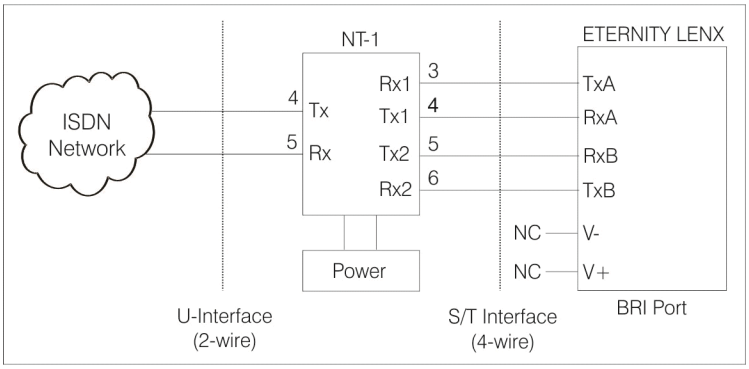
Pinout and Cable Details of BRI4 Port in TE Mode

Pin	Color	Connection
1	Orange-White	Not connected
2	Orange	Not connected
3	Green-White	TxA
4	Blue	RxA
5	Blue-White	RxB
6	Green	TxB
7	Brown-White	V-
8	Brown	V+

Pinout and Cable Details of BRI4 Port in NT Mode

Pin	Color	Connection
1	Orange-White	Not connected
2	Orange	Not connected
3	Green-White	RxA
4	Blue	TxA
5	Blue-White	TxB
6	Green	RxB
7	Brown-White	V-
8	Brown	V+

The following diagram shows how to connect a BRI Line to the BRI port in the TE mode.



11. If you have completed all other installation tasks, you may turn ON the system and observe the Reset Cycle and the LED pattern of the BRI Card.

LED Pattern of the BRI Card

- The BRI8 Card has 8 LEDs: L1[38], L2, L3, L4, L5[39], L6, L7 and L8. These display the status of each port.
- The BRI4 Card has 4 LEDs: L1[40], L2, L3 and L4. These display the status of each port.

  The LEDs show the Status of the Ports as summarized in the table below:

Port Status	LED Color	LED Cadence
Port is not active	RED	Continuously ON
Port is active	GREEN	Continuously ON

38. This LED keeps blinking. It displays the system heart bits, at the rate of 100ms. It will remain OFF for 100ms and will show the status of port 1 for the next 100ms.

39. This LED keeps blinking. It displays the system heart bits, at the rate of 100ms. It will remain OFF for 100ms and will show the status of port 1 for the next 100ms.

40. This LED keeps blinking. It displays the system heart bits, at the rate of 100ms. It will remain OFF for 100ms and will show the status of port 1 for the next 100ms.

The T1E1PRI Card

The ETERNITY ME T1E1PRI Card provides the interface to connect the system to ISDN Network.

When connected to T1 carrier lines, the Card supports the following signaling types:

- PRI
- Robbed Bit Signaling
- Q-Signaling (QSIG)
- E&M

When connected to E1 carrier lines, the Card supports the following signaling types:

- PRI
- Channel Associated Signaling (CAS)
- Q-Signaling (QSIG)
- E&M

The T1E1PRI Card is available in the following configurations:

T1E1PRI Card for ETERNITY LENX

Card Name	Configuration and Application
ETERNITY ME Card T1E1PRI Dual[a]	2-Port card with QSIG support to connect 2 ISDN T1/E1 PRI Lines or ISDN Compatible Devices
ETERNITY ME Card T1E1PRI Single	1-Port card with QSIG support to connect 1 ISDN T1/E1 PRI Line or ISDN Compatible Device

a. *If you have CPU Cards with 1K IDT Switch (PCB Version V1R3) out of the 15 universal slots in the second rack, in four slots from the right the following cards cannot be affixed — ETERNITY LE CARD SLT48, ETERNITY ME CARD E1FO Dual and ETERNITY ME CARD T1E1PRI Dual. In the remaining slots any card can be affixed.*

The maximum number of PRI Lines supported are 24.

Connectors

The T1E1PRI card has an RJ45 Connector for each port. The ETERNITY ME T1E1PRI Dual card has 2 RJ45 Connectors for the two ports, while the ETERNITY ME T1E1PRI Single card has a single RJ45 Connector.

A cable with RJ45 plugs on both ends is supplied for each connector.

LEDs

The ETERNITY ME T1E1PRI Dual Card has four LEDs: L1, L2, L3 and L4.
The ETERNITY ME T1E1PRI Single Card has two LEDs L1 and L2.

Installing the T1E1PRI Card

1. Before installing the card, take the necessary precautions prescribed for handling the cards. Always wear an electrostatic-discharge preventive wrist strap and use a grounding mat. Make sure the power supply is turned off.

2. Unpack the T1E1PRI Card and check the package contents.

3. Select any free (empty) slot from the Universal Slots. Unscrew and remove the filler bracket of the empty slot. Do not discard the filler bracket.

Setting Line Termination Resistor

4. The default positions of SW3, SW4, SW6, SW7 switches should be as follows:

Pin1	Pin2	Pin3	Pin4	Termination Resistance (Ω)
OFF	OFF	OFF	ON	0 (Default)

Do Not Change the positions of any of these switches!

5. By default, termination resistance of PRI port is set as 120Ω, which is for E1 connectivity.

   - To use the PRI Port for T1 connectivity, termination resistance must be changed to 100Ω.

   - Use DIP Switch SW5 to change the Termination Resistance of PRI Port 1. Set the Pins of SW5 as shown below:

Pin-1	Pin-2	Pin-3	Pin-4	Resistance
OFF	OFF	ON	OFF	120Ω (for E1)
OFF	ON	OFF	OFF	100Ω (for T1)

   - If using the ETERNITY ME T1E1PRI Dual Card, use DIP Switch SW2 to change the Termination Resistance of PRI Port 2. Set the Pins of SW2 as shown below:

Pin-1	Pin-2	Pin-3	Pin-4	Resistance
OFF	OFF	ON	OFF	120Ω (for E1)
OFF	ON	OFF	OFF	100Ω (for T1)

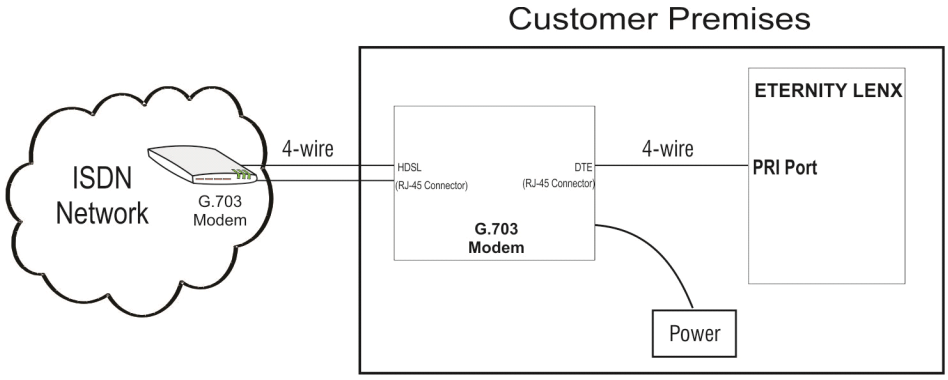
6. Insert the T1E1PRI Card into the guide rails of the free slot you selected for the card. Make sure that the connectors on the card make perfect contact with those of the slot on the backplane motherboard.

7. Now, press down the levers on the card mounting brackets to secure the card in its slot. Fix the card in place with the two screws provided.

Connecting ISDN T1/E1 PRI Lines

8. Use the cable supplied with the T1E1PRI Card to connect the system to the T1/E1 PRI network interface equipment (modem), which is usually supplied by your ISDN Service Provider along with the PRI line.

The diagram below illustrates this.



- Most Service Providers insist on connecting an ISDN modem at both the ends of the PRI line—one at the Local Exchange and other at the Customer's Premises.
- At the Customer's Premises, the PRI line is terminated on the HDSL interface of the modem.
- The DTE interface of the modem is to be connected to the PRI port (RJ-45 connector on the ETERNITY ME T1E1PRI Card).

9. Plug in one end of the RJ45 cable supplied with the card into the card's connector. Plug the other end of the RJ45 cable into the Network Termination Unit.

10. Refer the following pin details for connecting the Network Termination Unit with the system.

Pin details of HDSL Interface of the G.703 Modem. (HDSL Network Termination Unit)

Pin Number	Pin Details
1	Line A
2	Line A
3	Not used
4	Line B
5	Line B
6	Not used
7	Not used
8	Not used

Pin details of DTE Interface of G.703 Modem. (HDSL Network Interface Unit)

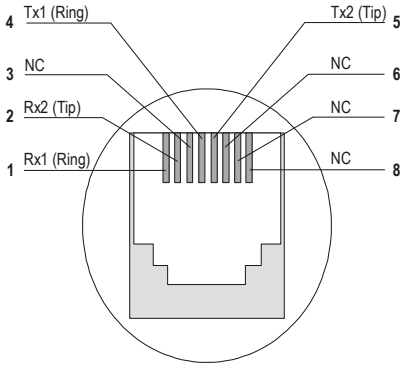
Pin Number	Pin Details
1	TX1 (Tip)
2	TX2 (Ring)
3	Not used
4	RX1 (Ring)

Pin Number	Pin Details
5	RX2 (Tip)
6	Not used
7	Not used
8	Not used

Most of the HDSL Network Termination Unit manufacturers use these connectors. But you are advised to read the installation guide of the HDSL Network Termination Unit being used by you.

Pin details of the System's T1E1PR1 Port

The T1E1PRI Port of the system terminates in an 8-pin RJ45, female connector and is wired according to the table below.



The cable wires may have to be crossed depending on the pinout of the DTE Interface of the modem.

11. Repeat the same steps to install another card. It is not necessary to install the other T1E1PRI Cards in a sequence. Any card can be installed in any of the slots.

12. If you have completed all other installation tasks. Power the system. After the Reset Cycle, observe the LED patterns of the T1E1PRI Card.

LED Patterns

The ETERNITY ME T1E1PRI Dual Card has four LEDs: L1, L2, L3 and L4.

- L1 and L2 are assigned to PRI Port 1 (PRI#1)
- L3 and L4 are assigned to PRI Port 2 (PR1#2)
- L1[41] shows Card Heart Bit as well as status of Port 1.

The ETERNITY ME T1E1PRI Single Card has two LEDs: L1 and L2.

Given below are the LED Patterns defined for indicating port states in the signaling types supported by the ETERNITY LENX.

41. This LED keeps blinking. It displays the system heart bits, at the rate of one second. It will remain OFF for one second and will show the status of port 1 for the next one second.

1. Port Active Mode

Signaling Type: E1-PRI

LED1/LED3 Pattern:

Port Status	Color	Cadence
Layer 1 established successfully	GREEN	Continuous ON
CRC4 Alarm	GREEN	100ms ON-100 ms OFF
BFA Alarm	RED	500ms ON-500 ms OFF
LOS Alarm	RED	Continuous ON

LED2/LED4 Pattern:

Port Status	Color	Cadence
Layer 1 established successfully	GREEN	Continuous ON
RAI Alarm	RED	500ms ON-500 ms OFF
AIS or LOS Alarm	RED	Continuous ON

Signaling Type: E1-CAS

LED1/LED3 Pattern:

Port Status	Color	Cadence
Layer 1 established successfully	GREEN	Continuous ON
CRC4 Alarm	GREEN	100ms ON-100 ms OFF
MFA Alarm	RED	100ms ON-100 ms OFF
BFA Alarm	RED	500ms ON-500ms OFF
LOS Alarm	RED	Continuous ON

LED2/LED4 Pattern:

Port Status	Color	Cadence
Layer 1 established successfully	GREEN	Continuous ON
Y-Bit Alarm	GREEN	100ms ON-100 ms OFF
AIS16 Alarm	RED	100ms ON-100 ms OFF
RAI Alarm	RED	500ms ON-500 ms OFF
AIS or LOS Alarm	RED	Continuous ON

Signaling Type: T1-RBS or T1-PRI

LED1/LED3 Pattern:

Port Status	Color	Cadence
No Alarm	GREEN	Continuous ON
BFA Alarm or MFA Alarm	RED	500ms ON-500 ms OFF
AIS Alarm	RED	100ms ON-100 ms OFF
LOS Alarm	RED	Continuous ON

LED2/LED4 Pattern:

Port Status	Color	Cadence
Layer 1 established successfully	GREEN	Continuous ON
RAI or LOS Alarm	RED	Continuous ON

2. Port Maintenance Mode

LED1/LED3 Pattern:

Port Status	Color	Cadence
Maintenance Mode	RED -GREEN	500 ms RED-500 ms GREEN

LED2/LED4 Pattern:

Port Status	Color	Cadence
Near end loop back wait before activate	RED	100ms ON-100 ms OFF
Near end loop active	RED	Continuous ON
Near end loop back wait before deactivate	RED	500ms ON-500 ms OFF
Far end loop back wait after activate	GREEN	100ms ON-100 ms OFF
Far end loop active	GREEN	Continuous ON
Far end loop back wait after deactivate	GREEN	500ms ON-500 ms OFF

3. Port Disable Mode

LED1/LED3 Pattern:

Port Status	Color	Cadence
Port Disable	RED	Continuous ON

LED2/LED4 Pattern:

Port Status	Color	Cadence
Port Disabled	OFF	OFF

Jumpers on the Main Board

Jumper Number	Position	Function
J5[a]	AB	Dual T1E1
	BC	Single T1E1
J6	AB	JTAG Mode
	BC (default)	Embedded ICE
J7	AB	Internal Boot
	BC (default)	External Boot
J8 & J12	AB (default)	Debug
	BC	UART
J9, J10 & J11	BC (default)	Normal
	AB	Boot

a. In T1E1 PRI Dual Card, default jumper position will be AB. In T1E1 PRI Single Card, default jumper position will be BC.

The E1FO Card

The ETERNITY LE E1FO Card provides the interface to connect the ETERNITY to the ISDN PRI Network. For E1 carrier lines the card supports the signaling types PRI, Q-Sig and CAS. For T1 carrier lines the card supports the signaling types PRI, Q-Sig and Robbed Bit Signaling. The T1/E1 ports can be set in Terminal or Network mode.

The E1FO Card supports Copper and Fiber Optic (FO) interfaces. At a time, either the Copper interface or the FO interface can be used. The FO interface supports only Single-mode (Mono mode) fiber connectivity and will work within a range of 30km.

E1 connectivity is supported over, the Copper interface as well as the Fiber Optic interface. The T1 connectivity is supported over the Copper interface only.

The E1FO Card is available in the following configurations:

E1FO Card for ETERNITY LENX

Card Name	Configuration and Application
ETERNITY ME Card E1FO Dual[a]	2-Port card to connect 2 ISDN T1/E1 Lines.
ETERNITY ME Card E1FO Single	1-Port card to connect 1 ISDN T1/E1 Line.

a. *If you have CPU Cards with 1K IDT Switch (PCB Version V1R3) out of the 15 universal slots in the second rack, in four slots from the right the following cards cannot be affixed — ETERNITY LE CARD SLT48, ETERNITY ME CARD E1FO Dual and ETERNITY ME CARD T1E1PRI Dual. In the remaining slots any card can be affixed.*

Connectors

The E1FO Dual card has two RJ45 Connectors and two Fiber Optic connectors. The E1FO Single card has one RJ45 Connector and one Fiber Optic connector.

LEDs

The ETERNITY ME E1FO Dual Card has four LEDs: L1, L2, L3 and L4.
The ETERNITY ME E1FO Single Card has two LEDs L1 and L2.

Installing the E1FO Card

1. Before installing the card, take the necessary precautions prescribed for handling the cards. Always wear an electrostatic-discharge preventive wrist strap and use a grounding mat. Make sure the power supply is turned off.

2. Unpack the E1FO Card and check the package contents.

3. Select any free (empty) slot from the Universal Slots. Unscrew and remove the filler bracket of the empty slot. Do not discard the filler bracket.

Setting Line Termination Resistor

4. The default positions of SW3, SW4, SW6, SW7 switches should be as follows:

Pin1	Pin2	Pin3	Pin4	Termination Resistance (Ω)
OFF	OFF	OFF	ON	0 (Default)

Do Not Change the positions of any of these switches!

5. By default, termination resistance of PRI port is set as 120Ω, which is for E1 connectivity.

   - To use the PRI Port for T1 connectivity, termination resistance must be changed to 100Ω.

   - Use DIP Switch SW5 to change the Termination Resistance of PRI Port 1. Set the Pins of SW5 as shown below:

Pin-1	Pin-2	Pin-3	Pin-4	Resistance
OFF	OFF	ON	OFF	120Ω (for E1)
OFF	ON	OFF	OFF	100Ω (for T1)

   - If using the ETERNITY ME E1FO Dual Card, use DIP Switch SW2 to change the Termination Resistance of PRI Port 2. Set the Pins of SW2 as shown below:

Pin-1	Pin-2	Pin-3	Pin-4	Resistance
OFF	OFF	ON	OFF	120Ω (for E1)
OFF	ON	OFF	OFF	100Ω (for T1)

6. Insert the E1FO Card into the guide rails of the free slot you selected for the card. Make sure that the connectors on the card make perfect contact with those of the slot on the backplane motherboard.

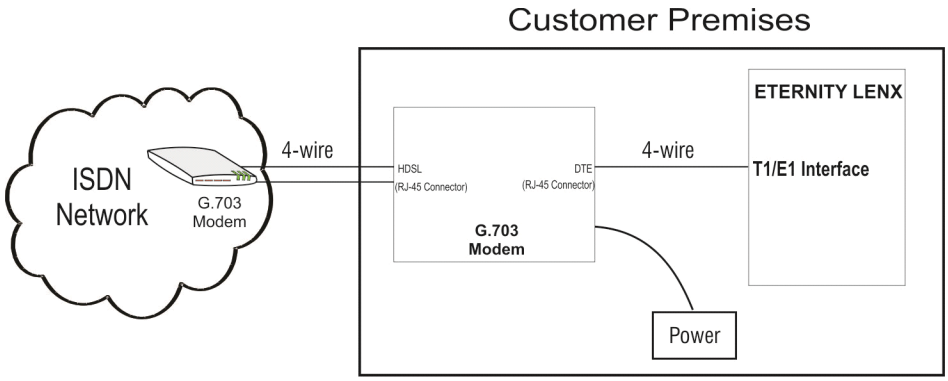
7. Now, press down the levers on the card mounting brackets to secure the card in its slot. Fix the card in place with the two screws provided.

Connecting the T1/E1 Lines

Copper Interface Connectivity

8. Use the cable to connect the T1/E1 Port to the T1/E1 network interface equipment (modem), which is usually supplied by your ISDN Service Provider along with the PRI line.

The diagram below illustrates this.



- Most Service Providers insist on connecting an ISDN modem at both the ends of the PRI line—one at the Local Exchange and other at the Customer's Premises.
- At the Customer's Premises, the PRI line is terminated on the HDSL interface of the modem.
- The DTE interface of the modem is to be connected to the PRI port (RJ45 connector on the Matrix ETERNITY ME CARD E1FOPRI Single).

9. Plug in one end of the RJ45 cable into the card's connector. Plug the other end of the RJ45 cable into the Network Termination Unit.

10. Refer the following pin details for connecting the Network Termination Unit with the system.

Pin details of HDSL Interface of the G.703 Modem. (HDSL Network Termination Unit)

Pin Number	Pin Details
1	Line A
2	Line A
3	Not used
4	Line B
5	Line B
6	Not used
7	Not used
8	Not used

Pin details of DTE Interface of G.703 Modem. (HDSL Network Interface Unit)

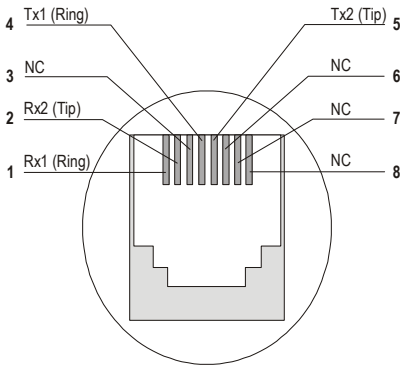
Pin Number	Pin Details
1	TX1 (Tip)
2	TX2 (Ring)
3	Not used
4	RX1 (Ring)

Pin Number	Pin Details
5	RX2 (Tip)
6	Not used
7	Not used
8	Not used

Most of the HDSL Network Termination Unit manufacturers use these connectors. But you are advised to read the installation guide of the HDSL Network Termination Unit being used by you.

Pin details of the T1/E1 PRI Interface

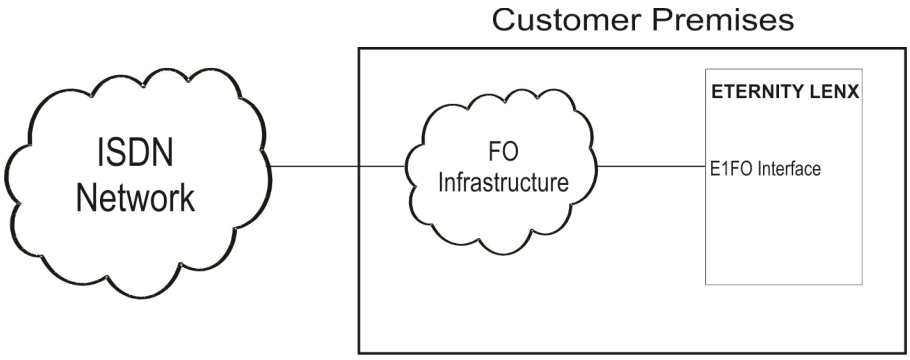
The T1/E1 Interface of the ETERNITY ME CARD E1FOPRI Single terminates in an 8-pin RJ45, female connector and is wired according to the table below.



The cable wires may have to be crossed depending on the pinout of the DTE Interface of the modem.

Fiber Optic Interface Connectivity

- *Fiber Optic interface supports E1 connectivity only.*
- *The Fiber Option (FO) interface supports only Single-mode (Mono mode) fiber connectivity and will work within a range of 30km.*



11. If your service provider provides Fiber Optic connectivity or you have an existing Fiber Optic infrastructure, you can connect the FO line to the E1FO interface of the E1FO Card.

    To use Fiber Optic interface, make sure you select **Line Coding Mechanism** as **NRZ (Fiber Optic)**. For details, see “Configuring E1 Trunks”.

12. Repeat the same steps to install another card. It is not necessary to install the other E1FO Cards in a sequence. Any card can be installed in any of the slots.

13. If you have completed all other installation tasks. Power the system. After the Reset Cycle, observe the LED patterns of the E1FO Card.

LED Patterns

The ETERNITY ME E1FO Dual Card has four LEDs: L1, L2, L3 and L4.

- L1 and L2 are assigned to PRI Port 1 (PRI#1)
- L3 and L4 are assigned to PRI Port 2 (PR1#2)
- L1[42] shows Card Heart Bit as well as status of Port 1.

The ETERNITY ME E1FO Single Card has two LEDs: L1 and L2.

Given below are the LED Patterns defined for indicating port states in the signaling types supported by the ETERNITY LENX.

1. Port Active Mode

Signaling Type: E1-PRI

LED1/LED3 Pattern:

Port Status	Color	Cadence
Layer 1 established successfully	GREEN	Continuous ON
CRC4 Alarm	GREEN	100ms ON-100 ms OFF
BFA Alarm	RED	500ms ON-500 ms OFF
LOS Alarm	RED	Continuous ON

LED2/LED4 Pattern:

Port Status	Color	Cadence
Layer 1 established successfully	GREEN	Continuous ON
RAI Alarm	RED	500ms ON-500 ms OFF
AIS or LOS Alarm	RED	Continuous ON

42. *This LED keeps blinking. It displays the system heart bits, at the rate of one second. It will remain OFF for one second and will show the status of port 1 for the next one second.*

Signaling Type: E1-CAS

LED1/LED3 Pattern:

Port Status	Color	Cadence
Layer 1 established successfully	GREEN	Continuous ON
CRC4 Alarm	GREEN	100ms ON-100 ms OFF
MFA Alarm	RED	100ms ON-100 ms OFF
BFA Alarm	RED	500ms ON-500ms OFF
LOS Alarm	RED	Continuous ON

LED2/LED4 Pattern:

Port Status	Color	Cadence
Layer 1 established successfully	GREEN	Continuous ON
Y-Bit Alarm	GREEN	100ms ON-100 ms OFF
AIS16 Alarm	RED	100ms ON-100 ms OFF
RAI Alarm	RED	500ms ON-500 ms OFF
AIS or LOS Alarm	RED	Continuous ON

Signaling Type: T1-RBS or T1-PRI

LED1/LED3 Pattern:

Port Status	Color	Cadence
No Alarm	GREEN	Continuous ON
BFA Alarm or MFA Alarm	RED	500ms ON-500 ms OFF
AIS Alarm	RED	100ms ON-100 ms OFF
LOS Alarm	RED	Continuous ON

LED2/LED4 Pattern:

Port Status	Color	Cadence
Layer 1 established successfully	GREEN	Continuous ON
RAI or LOS Alarm	RED	Continuous ON

2. Port Maintenance Mode

LED1/LED3 Pattern:

Port Status	Color	Cadence
Maintenance Mode	RED -GREEN	500 ms RED-500 ms GREEN

LED2/LED4 Pattern:

Port Status	Color	Cadence
Near end loop back wait before activate	RED	100ms ON-100 ms OFF
Near end loop active	RED	Continuous ON
Near end loop back wait before deactivate	RED	500ms ON-500 ms OFF
Far end loop back wait after activate	GREEN	100ms ON-100 ms OFF
Far end loop active	GREEN	Continuous ON
Far end loop back wait after deactivate	GREEN	500ms ON-500 ms OFF

3. Port Disable Mode

LED1/LED3 Pattern:

Port Status	Color	Cadence
Port Disable	RED	Continuous ON

LED2/LED4 Pattern:

Port Status	Color	Cadence
Port Disabled	OFF	OFF

The E&M Card

The E&M Card of the system provides the interface for analog trunking to connect various communication equipment telephone switches, Routers, Leased Lines, etc. using Tie-Lines.

The E&M Card is required for the following applications:

- Power Line Carrier Communication (PLCC) Networks, where several systems are connected with each other through E&M tie lines. Refer "PLCC-An Introduction" to know more.
- "Closed User Group (CUG)", where several systems are connected with each other through E&M tie lines[43].
- System expansion, where two systems are connected with each other with E&M tie lines.
- Connecting remote systems over E&M tie lines.

Refer the topic "E&M Connectivity" to know more.

An E&M Port can be programmed to behave as a Trunk Interface, a Subscriber (Station) Interface or both, as a Tie Line with the dual personality of a Trunk and a Subscriber.

The E&M Card supports

- E&M Interface - Types IV and V
- Speech Interface - Two-wire and four-wire.
- E&M Trunk Seizure Type[44]: Immediate, Immediate + Wink, Immediate with Ack, Immediate with Ack+Wink, Seizure Pulse, Seizure Pulse + Wink, Express, and Compander Control Signal.
- Address Signaling: Pulse dial (Pulse 10PPS, Pulse 20PPS) and Tone Dial (DTMF).

The E&M Card is available in the following configurations:

E&M Card for ETERNITY LENX

Card Name	Configuration and Application
ETERNITY ME Card E&M8	8-port card to connect 8 E&M Tie Lines
ETERNITY ME Card E&M4	4-port card to connect 4 E&M Tie Lines

The maximum number of E&M ports supported are 64.

Connectors

The E&M Card has RJ45 Connectors. A separate MDF cable is supplied for each connector.

43. *The Systems in a "Closed User Group (CUG)" can be connected over ISDN T1E1PRI Lines as well. Refer the topic Closed User Groups to know more.*
44. *This is the line protocol that defines how the equipment seizes the E&M trunk. Also referred to as Start Dial Supervision Signaling Protocol.*

LEDs

The ETERNITY ME Card E&M8 has eight tri-color LEDs. The ETERNITY ME Card E&M4 has 4 LEDs, to indicate the functioning of the ports.

Installing the E&M Card

An E&M port can be programmed to take on the function of:

- **a Station** - works like an extension interface, receiving incoming calls.

  OR

- **a Trunk** - works like a trunk interface when any of the extensions of the system makes an outgoing call through it.

  OR

- **a Tie Line** - takes on a dual personality: functioning as both as an extension and a trunk. The E&M port works like an extension interface for incoming calls. It works like a trunk interface when any extension makes an outgoing call through it.

  This dual function is used in systems that are used as Transit Exchanges as in a PLCC Network. Read "PLCC-An Introduction" to know more.

You cannot connect a trunk line or an SLT / DKP to an E&M port.

1. Have the necessary wiring for the E&M Analog trunk in place. Take the necessary safety precautions before you begin handling the card; switch off power supply and always wear an antistatic wrist strap and use a grounding mat.

2. Unpack the E&M Card and check the package contents.

3. The E&M Card supports E&M Interface Type IV and Type V connection. To select the appropriate Interface Type out of the two, you need to change the Jumper Settings.

   Refer the table below to select the desired Interface Type and Speech Interface.

Jumper Number	Position	Function
J1 and J2	AB	Type IV E&M Interface
	BC	Type V E&M Interface

   - By default all the E&M Ports are set to support Type-IV.

   - To select the Type-V connection for the E&M Port, set Jumpers J1 and J2 (located on the E&M module) in BC Position.

4. Select the speech interface - 2-wire speech or 4-wire speech - as required, by changing the jumper settings. Refer the table below.

Jumper Number	Position	Function
J3 and J4	AB	4-wire speech interface
	BC	2-wire speech interface

- By default all the E&M Ports are set to support 2-wire Speech Interface.
- To select 2-wire speech interface for the E&M Port, set Jumpers J3 and J4 (given on E&M module) to BC Position.
- To select 4-wire speech interface for the E&M Port, set Jumpers J3 and J4 on E&M module to AB Position.

Keep Jumper number J5 in BC position and Jumper number J6 in BC/Open position.

5. Now, select a free slot for the E&M Card. Unscrew and remove the filler bracket by pushing up the levers on the bracket. Preserve the filler bracket for future use.

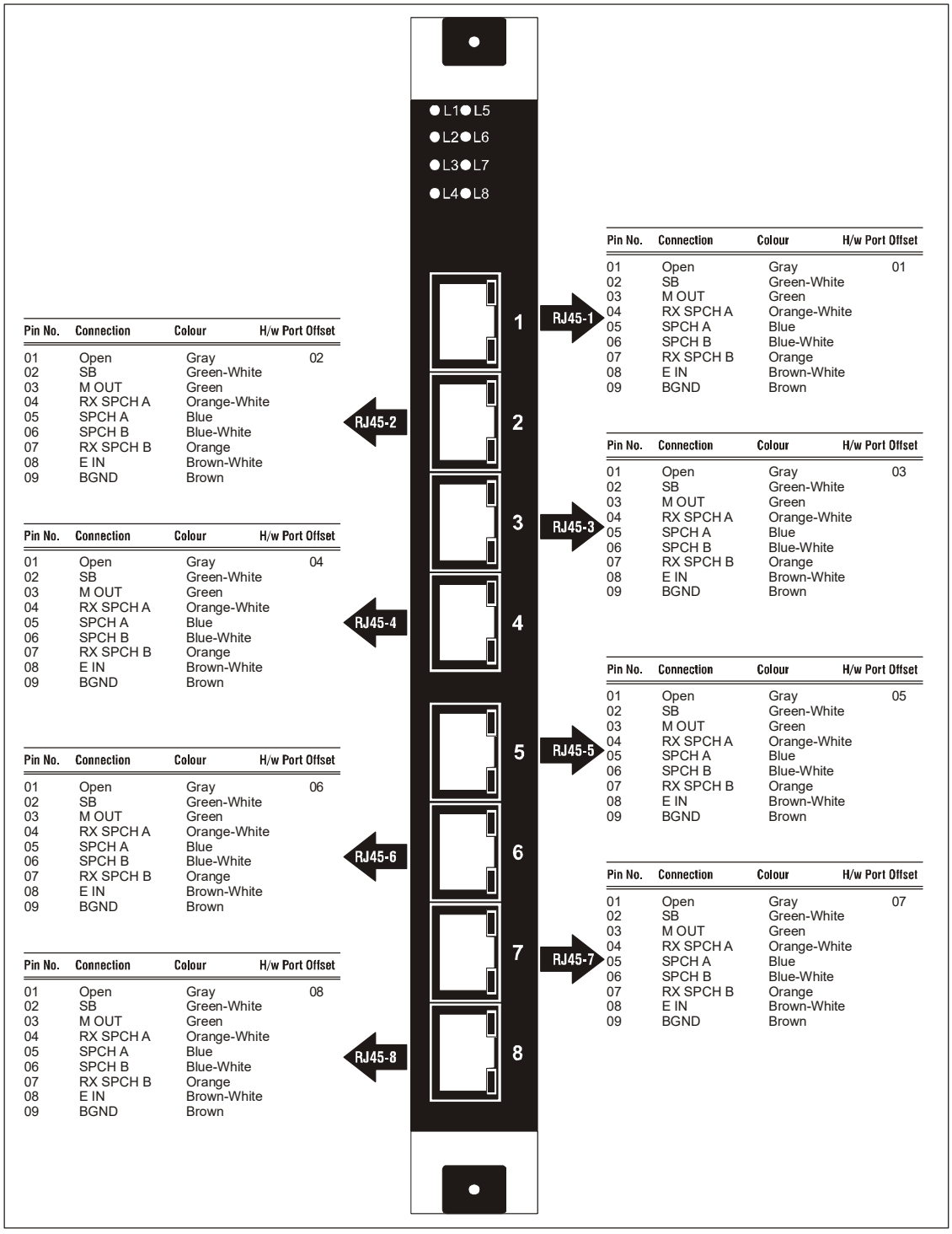
6. Insert the E&M Card into guide rails of the empty slot. Make sure the connectors on the card make perfect contact with those on the backplane motherboard. Secure the card by pressing down the levers and fix the bracket with the screws provided with the card.

7. Connect the cables supplied with the E&M Card into the RJ45 connectors on the E&M Card.

8. Connect the free ends of the cables into the E&M Ports of the other System/PBX/Router/Tie Line equipment by appropriate crossing of the wires.

   Refer the following pin-out details for the E&M Card and for each Interface and Speech Interface Type.

Pinout details of ETERNITY ME Card E&M8 - RJ45 Connectors



Pin No.	Connection	Colour	H/w Port Offset
01	Open	Gray	01
02	SB	Green-White	
03	M OUT	Green	
04	RX SPCH A	Orange-White	
05	SPCH A	Blue	
06	SPCH B	Blue-White	
07	RX SPCH B	Orange	
08	E IN	Brown-White	
09	BGND	Brown	

Pin No.	Connection	Colour	H/w Port Offset
01	Open	Gray	02
02	SB	Green-White	
03	M OUT	Green	
04	RX SPCH A	Orange-White	
05	SPCH A	Blue	
06	SPCH B	Blue-White	
07	RX SPCH B	Orange	
08	E IN	Brown-White	
09	BGND	Brown	

Pin No.	Connection	Colour	H/w Port Offset
01	Open	Gray	03
02	SB	Green-White	
03	M OUT	Green	
04	RX SPCH A	Orange-White	
05	SPCH A	Blue	
06	SPCH B	Blue-White	
07	RX SPCH B	Orange	
08	E IN	Brown-White	
09	BGND	Brown	

Pin No.	Connection	Colour	H/w Port Offset
01	Open	Gray	04
02	SB	Green-White	
03	M OUT	Green	
04	RX SPCH A	Orange-White	
05	SPCH A	Blue	
06	SPCH B	Blue-White	
07	RX SPCH B	Orange	
08	E IN	Brown-White	
09	BGND	Brown	

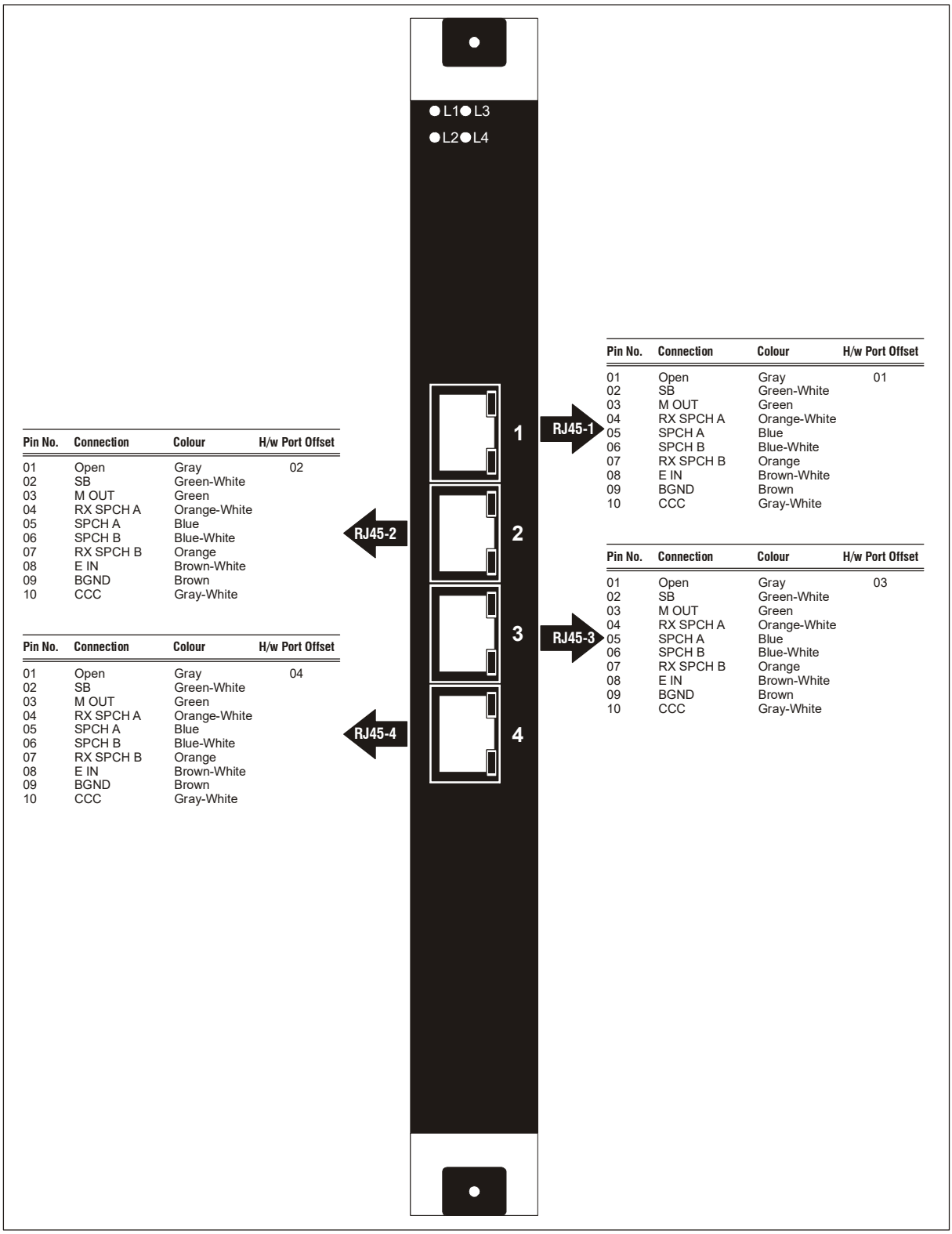
Pin No.	Connection	Colour	H/w Port Offset
01	Open	Gray	05
02	SB	Green-White	
03	M OUT	Green	
04	RX SPCH A	Orange-White	
05	SPCH A	Blue	
06	SPCH B	Blue-White	
07	RX SPCH B	Orange	
08	E IN	Brown-White	
09	BGND	Brown	

Pin No.	Connection	Colour	H/w Port Offset
01	Open	Gray	06
02	SB	Green-White	
03	M OUT	Green	
04	RX SPCH A	Orange-White	
05	SPCH A	Blue	
06	SPCH B	Blue-White	
07	RX SPCH B	Orange	
08	E IN	Brown-White	
09	BGND	Brown	

Pin No.	Connection	Colour	H/w Port Offset
01	Open	Gray	07
02	SB	Green-White	
03	M OUT	Green	
04	RX SPCH A	Orange-White	
05	SPCH A	Blue	
06	SPCH B	Blue-White	
07	RX SPCH B	Orange	
08	E IN	Brown-White	
09	BGND	Brown	

Pin No.	Connection	Colour	H/w Port Offset
01	Open	Gray	08
02	SB	Green-White	
03	M OUT	Green	
04	RX SPCH A	Orange-White	
05	SPCH A	Blue	
06	SPCH B	Blue-White	
07	RX SPCH B	Orange	
08	E IN	Brown-White	
09	BGND	Brown	

Pinout details of ETERNITY ME Card E&M4 - RJ45 Connectors



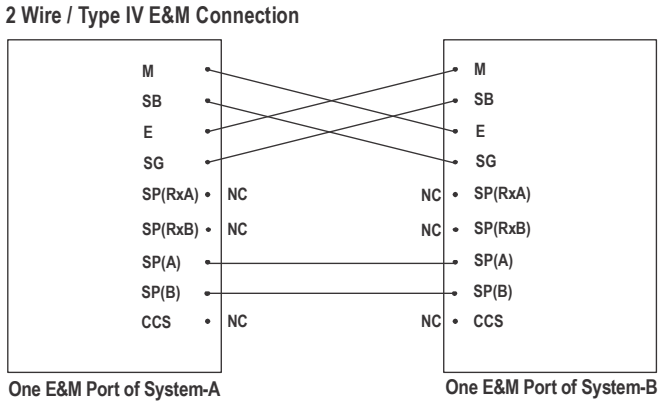
Pin No.	Connection	Colour	H/w Port Offset
01	Open	Gray	01
02	SB	Green-White	
03	M OUT	Green	
04	RX SPCH A	Orange-White	
05	SPCH A	Blue	
06	SPCH B	Blue-White	
07	RX SPCH B	Orange	
08	E IN	Brown-White	
09	BGND	Brown	
10	CCC	Gray-White	

Pin No.	Connection	Colour	H/w Port Offset
01	Open	Gray	02
02	SB	Green-White	
03	M OUT	Green	
04	RX SPCH A	Orange-White	
05	SPCH A	Blue	
06	SPCH B	Blue-White	
07	RX SPCH B	Orange	
08	E IN	Brown-White	
09	BGND	Brown	
10	CCC	Gray-White	

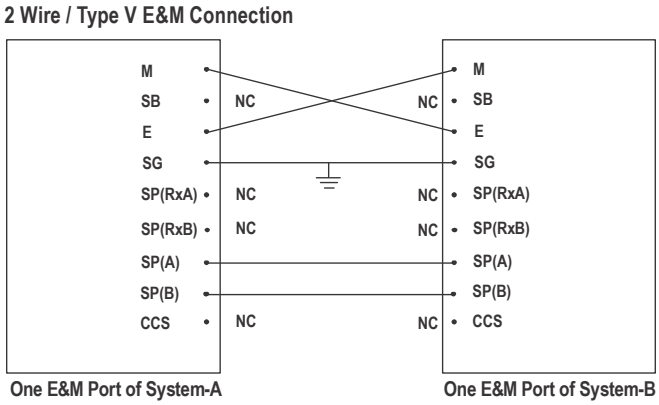
Pin No.	Connection	Colour	H/w Port Offset
01	Open	Gray	03
02	SB	Green-White	
03	M OUT	Green	
04	RX SPCH A	Orange-White	
05	SPCH A	Blue	
06	SPCH B	Blue-White	
07	RX SPCH B	Orange	
08	E IN	Brown-White	
09	BGND	Brown	
10	CCC	Gray-White	

Pin No.	Connection	Colour	H/w Port Offset
01	Open	Gray	04
02	SB	Green-White	
03	M OUT	Green	
04	RX SPCH A	Orange-White	
05	SPCH A	Blue	
06	SPCH B	Blue-White	
07	RX SPCH B	Orange	
08	E IN	Brown-White	
09	BGND	Brown	
10	CCC	Gray-White	

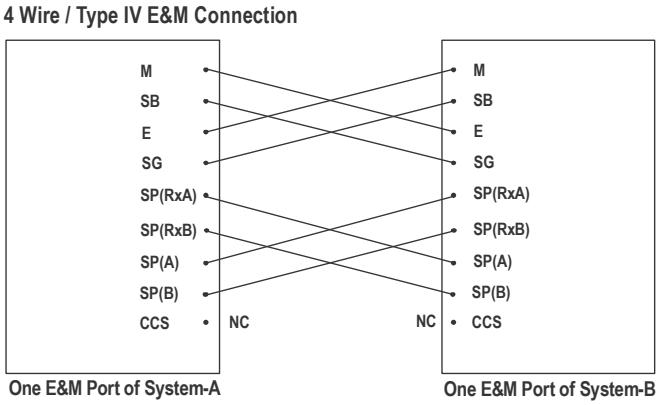
2 Wire, Type IV E&M Connection



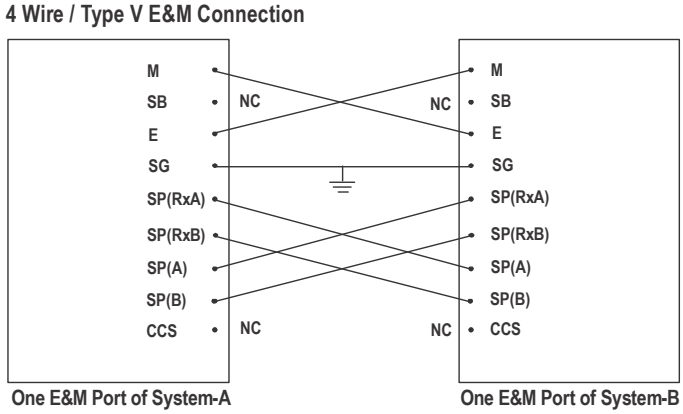
2 Wire, Type V E&M Connection



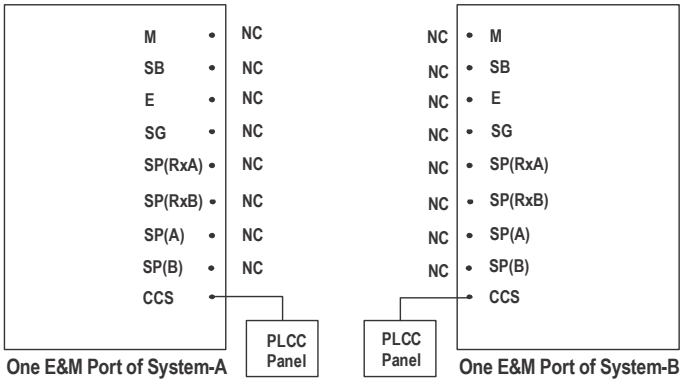
4 Wire, Type IV E&M Connection



4 Wire, Type V E&M Connection



9. If you are connecting two PLCC EPAX in a Power Line Carrier Communication Network Compander Control Signal (CCS) Connection should be made as illustrated in the block diagram below for any of the four combinations of E&M and Speech Interfaces illustrated in the previous step.



Compander Control Signal (CCS) is a special type of signal used by Power Line Carrier Communication Networks to improve quality of speech transmission. The PLCC network expects this signal from the system when speech is established and the E&M Card supports this facility. The system sends CCS signal to the PLCC panel.

- *When the E&M port is used as an Endpoint; the system sends a CCS to the PLCC panel while making an outgoing call through the E&M port or when a call is received at the E&M port.*
- *When the E&M port is used for Transit Exchange; the system sends a CCS to the PLCC panel while there is a Transit call through the E&M port.*

10. If you have completed all installation tasks, power ON the system, observe the Reset Cycle and the LED pattern of the E&M Card.

LED Pattern of the E&M Card

Stage	LED Color	LED Cadence
Initialization		
At Power ON	RED	L1 to L8 glow one after the other in a sequence, ON 100ms -OFF. L1 ON 100ms-OFF, L2 100ms ON-OFF, L3 100ms ON-OFF, L4 100ms ON-OFF……L8 100ms ON-OFF
	GREEN	L1 to L8 glow one after the other in a sequence, ON 100ms -OFF. L1 ON 100ms-OFF, L2 100ms ON-OFF, L3 100ms ON-OFF, L4 100ms ON-OFF……L8 100ms ON-OFF
Stand-By[a]	GREEN, ORANGE	L1 toggles GREEN ON 1 sec, ORANGE ON 1 sec
Normal (Port Event)		
M-Wire High	GREEN	LED of the Port continuously ON
M-Wire Low		LED of the Port continuously OFF
E-Wire High	RED	LED of the Port continuously ON
M-Wire Low		LED of the Port continuously OFF
E-Wire and M-Wire High	ORANGE	LED of the Port continuously ON
Errors		
Controller RAM failure	ORANGE	All LEDs Toggle at 1 sec
External RAM failure	ORANGE	All LEDs Toggle at 2 sec
Eprom failure	ORANGE	All LEDs Toggle at 3 sec
Invalid Slot detected	ORANGE	All LEDs Toggle at 6 sec

a. Waiting to be detected by Master Card.

Jumpers on the Main Board

Jumper Number	Position	Function
J1	AB (default)	Normal Operation
	BC	For uploading software using COM Port
J4 & J5	AB (default)	Normal Operation
	BC	For uploading software using COM Port

The Mobile Card

The Mobile Card interfaces the system with 2G/3G/4G networks. It routes calls made and received over mobile networks, like a mobile handset.

The Mobile Cards are available in 2G, 3G and 4G variants.

The Mobile Card does not support GPRS features, Fax and Data services, network supported services, except CLIR and USSD.

For compatibility and use of Matrix GSM products (2G/3G/4G) in Russia and Iran Province connect with Matrix Sales or Technical Support Team.

The Mobile Card for ETERNITY LENX

Card Name	Configuration and Application
ETERNITY ME Card GSM8	8-port card to connect to 8 GSM networks (8 SIM Cards can be installed). To know more, refer to "ETERNITY ME Card GSM8/GSM8 3G without SIM Hot-swap". For Hardware Design V3R2, CPLD V3R2 and PCB Version Revision V3R1 This version onwards SIM Hot Swap is supported, that is the SIM card can be removed and inserted in the SIM Slots without turning off the system. To know more, refer to "ETERNITY ME Card GSM8/GSM8 3G/GSM8 4G with SIM Hot-swap".
ETERNITY ME Card GSM8 3G	8-port card to connect to 8 GSM networks with 3G support (8 SIM Cards can be installed), refer to "ETERNITY ME Card GSM8/GSM8 3G without SIM Hot-swap". For Hardware Design V3R2, CPLD V3R2 and PCB Version Revision V3R1 This version onwards SIM Hot Swap is supported, that is the SIM card can be removed and inserted in the SIM Slots without turning off the system. To know more, refer to "ETERNITY ME Card GSM8/GSM8 3G/GSM8 4G with SIM Hot-swap".
ETERNITY ME CARD GSM8 4G	8-port card to connect to 8 GSM networks with 4G support (8 SIM Cards can be installed) with SIM Hot Swap. To know more, refer to "ETERNITY ME Card GSM8/GSM8 3G/GSM8 4G with SIM Hot-swap".

Just like mobile handsets, each Mobile Port has a unique IMEI (International Mobile Equipment Identity) number, pasted on the mobile engine.

The maximum Mobile ports supported are 64.

SIM cards from different service providers can be used.

Antenna

There is a single rooftop (RT) antenna for four GSM ports. A splitter connects all the four ports on the card into a single antenna. An antenna cable is also provided, giving you the flexibility to move the antenna to another position (in case of weak signal).

Personal Identification Number (PIN)

The SIM cards can be protected from unauthorized use by programming a Personal Identification Number (PIN) on the SIM. If the wrong SIM PIN is entered thrice in a row, by a user, the SIM card suspects the user and asks for the Personal Unlock Keyword (PUK).

LEDs

There is a tri-color LED for each mobile port on the card to indicate the functioning of the card and the status of the ports.

Installing the Mobile Card

To be able to connect the system to 2G/3G/4G networks, you must have one of the above mobile cards installed in the system.

1. To install the Mobile Card,

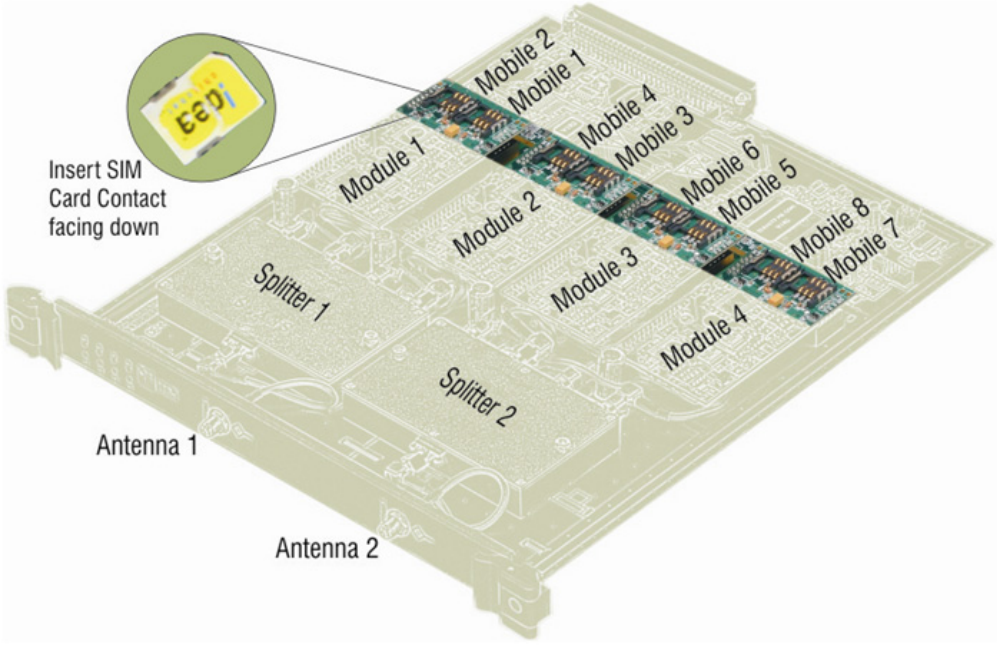
    - If using a 2G/3G/4G card, get the SIM Card from the 2G/3G/4G service provider of your choice ready. Use SIM PIN protection, if required.

Disable Call Waiting in the SIM, else it may result in call disconnection.

2. Make sure that the system is installed at a location where sufficient network coverage is available. The power supply should be turned off, and you must be wearing an electrostatic discharge preventive wrist strap and must have a grounding mat, before you begin handling the card.

3. Unpack the Mobile Card and verify the package contents.

ETERNITY ME Card GSM8/GSM8 3G without SIM Hot-swap



Enabling PIN Protection on SIM

4. For the 2G/3G Card, enable SIM PIN before installing the SIM card in the system.

   - insert the SIM into a mobile handset first.
   - enable PIN Protection from the mobile handset.
   - change the SIM PIN to 1234 (this is the default PIN for all SIM cards used in the system). Changing the SIM PIN to '1234' enables you to change the SIM PIN from the Jeeves later (Refer SIM PIN under “Configuring Mobile Trunks” for instructions).
   - remove the SIM from the mobile handset.

If you do not want to use PIN protection, insert the SIM in the mobile handset and disable PIN protection. Remove the SIM Card from the mobile handset.

5. Insert the SIM card (PIN changed to 1234), with its connector side down into the SIM holder on the Mobile card. You can insert multiple SIM cards of the same GSM service provider or of different service providers.

6. Insert the Mobile Card into the guide rails of the Universal Slot you have selected for this card. Make sure that the card is inserted deep enough to make perfect contact with the connectors in the backplane. Now, press down the levers on the card mount bracket to secure the card in its slot.

7. Connect the antenna provided with the card on the splitter connector on the front panel of the card. You may also use the antenna cable to place the antenna at another position.

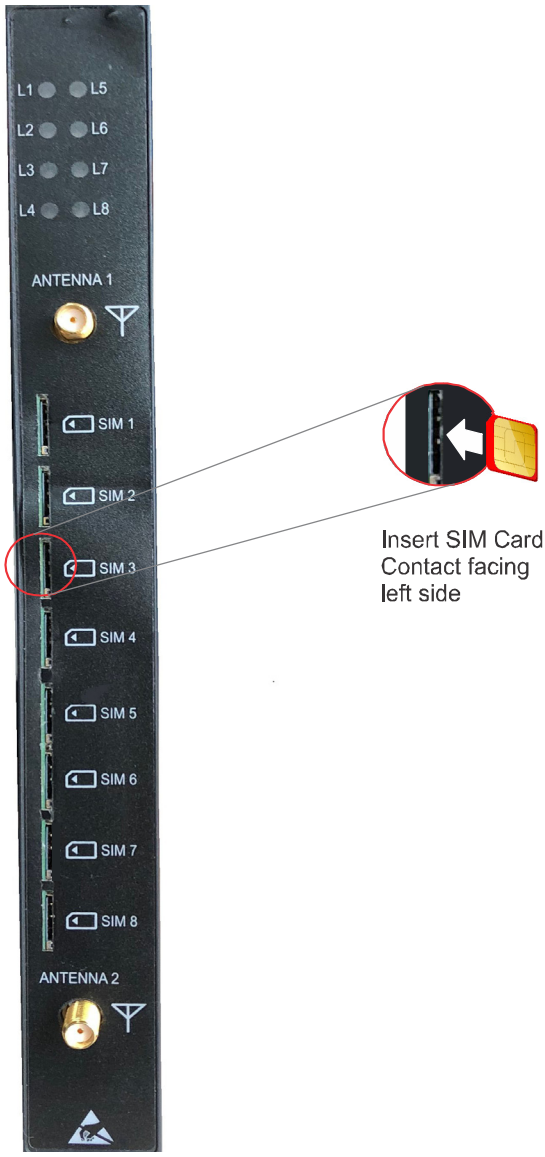
8. Repeat Steps 1-7 to insert another Mobile Card.

9. If you have completed all installations tasks, power the system.

10. Wait for the system to register with the Mobile network. By default, the Mobile ports are set to select and register with the Mobile networks automatically. Now, observe the LED Patterns of the Mobile Ports.

- *At every power up of the system, it takes about 3 minutes for the Mobile ports to get registered with the network. Once registration with the GSM network is completed, the mobile port can be used.*
- *Each time the Mobile Port sends a request, such as Registration Request, the system waits for the duration of the Network Response Timer. This Timer signifies the time for which the Mobile Port waits for a response from the Mobile network. It is fixed for 150 seconds for all Mobile ports.*

ETERNITY ME Card GSM8/GSM8 3G/GSM8 4G with SIM Hot-swap



Enabling PIN Protection on SIM

1. For the 2G/3G/4G Card, enable SIM PIN before installing the SIM card in the system.
   - insert the SIM into a mobile handset first.
   - enable PIN Protection from the mobile handset.

- change the SIM PIN to 1234 (this is the default PIN for all SIM cards used in the system). Changing the SIM PIN to '1234' enables you to change the SIM PIN from the Jeeves later (Refer SIM PIN under "Configuring Mobile Trunks" for instructions).
- remove the SIM from the mobile handset.

If you do not want to use PIN protection, insert the SIM in the mobile handset and disable PIN protection. Remove the SIM Card from the mobile handset.

2. Insert the SIM with its contact side facing left into the SIM slot located on the fascia of ETERNITY ME Card.

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3. Push the SIM backwards into the slot until you hear a click and the SIM is locked in place.
4. To unlock the SIM, push the protruded portion of the SIM backwards again and release it.

The Mobile cards with SIM Hot - swap are designed keeping in mind the Standard Nano SIM size. In case, you face any issues due to the SIM size, contact your respective Service Provider for assistance.

5. Repeat the same steps to insert another SIM Card. You can insert multiple SIM cards of the same GSM service provider or of different service providers.

6. Insert the Mobile Card into the guide rails of the Universal Slot you have selected for this card. Make sure that the card is inserted deep enough to make perfect contact with the connectors in the backplane. Now, press down the levers on the card mount bracket to secure the card in its slot.

7. Connect the antenna provided with the card on the splitter connector on the front panel of the card. You may also use the antenna cable to place the antenna at another position.

8. Repeat Steps 1-7 to insert another Mobile Card.

9. If you have completed all installations tasks, power the system.

10. Wait for the system to register with the Mobile network. By default, the Mobile ports are set to select and register with the Mobile networks automatically. Now, observe the LED Patterns of the Mobile Ports.

- *At every power up of the system, it takes about 3 minutes for the Mobile ports to get registered with the network. Once registration with the GSM network is completed, the mobile port can be used.*

- *Each time the Mobile Port sends a request, such as Registration Request, the system waits for the duration of the Network Response Timer. This Timer signifies the time for which the Mobile Port waits for a response from the Mobile network. It is fixed for 150 seconds for all Mobile ports.*

LED Pattern of Mobile Ports

The number of tri-color LEDs on the GSM card corresponds with the number of mobile ports on the card.

At Power On: All LEDs will blink 1 second ON and 1 second OFF in the color sequence: Red-Green-Orange until the Reset cycle is complete.

In the Stand-by state: All LEDs will glow Orange for a second and turn Green for a second, repeatedly.

During normal functioning: The LEDs will various events on the Mobile port in the color and cadence described in the table below:

Event	Color	Cadence in msec (1 cadence is of 3000 msec)
Port disabled	-	LED OFF
Port idle	-	LED OFF
Port Active (All States other than Ring and Speech)	Red	Continuous ON
Ring Event	Green	400ms ON-200ms OFF400ms ON-200ms OFF
Speech	Green	Continuous ON
GSM initialization	Orange	200ms ON-200ms OFF-200ms ON-200ms OFF-200ms ON-200ms OFF-200ms ON-200ms OFF-200ms ON-1200ms OFF (5 blinks)
PUK required	Orange	200ms ON-200ms OFF-200ms ON-200ms OFF-200ms ON-200ms OFF-200ms ON-1600ms OFF-
SIM PIN faulty	Orange	200ms ON-200ms OFF-200ms ON-200ms OFF-200ms ON-2000ms OFF (3 blinks)
SIM Absent	Orange	200ms ON-200ms OFF-200ms ON-2400ms OFF (2 blinks)
Network Link Down (Absence of GSM Network)	Orange	200ms ON-2800ms OFF

The Magneto Card

The Magneto Card is used for connecting the system to Magneto Telephones[45], which are widely used by the defense establishments as field phones in front lines, and by other establishments such as railroad companies (signaling emergencies, crossings, etc.), electric utilities, pipeline companies, who need to have their networks at places that are too remote to be serviced by public telephone networks.

The Magneto Card lands calls from magneto field telephones on the extensions of the system and places calls from the extensions of the system on magneto telephones.

Magneto Card for ETERNITY LENX

Card Name	Configuration and Application
ETERNITY ME Card Magneto8	8-port card to connect 8 Magneto Phones

The maximum number of magneto ports supported are 16.

Connectors

The Magneto Card has RJ45 connectors. A multi-pair cable is provided for each connector.

LED

The ETERNITY Magneto8 has 8 LEDs for each magneto port supported by the card.

The LEDs indicate the health of the cards during the Reset Cycle and the status of the ports during the normal functioning of the system.

Installing the Magneto Card

1. Have the necessary wiring for the Magneto Ports in place.

   You may install an MDF to connect the Magneto Ports with the Field Telephone wires.

   OR

   You may connect the wires from the Magneto Field Telephones directly to the Magneto Port.

   You are advised to use a separate set of Krone Modules for connecting the Magneto phones to the Magneto ports of the system

2. Prepare for the card installation by switching off power supply and wearing an electrostatic discharge preventive wrist strap and use a grounding mat.

3. Unpack the Magneto Card and check the package contents.

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45. A magneto telephone is a local battery telephone set, in which signaling current is provided by a magneto hand generator, usually a magneto. The hand generator, commonly referred to as 'crank', is located on the right hand side of the telephone set and is turned to produce energy to ring other phones or to signal the CO. The magneto, also called the generator, is used to convert the mechanical motion via the crank to produce sufficient energy to ring other phones or to signal the CO.

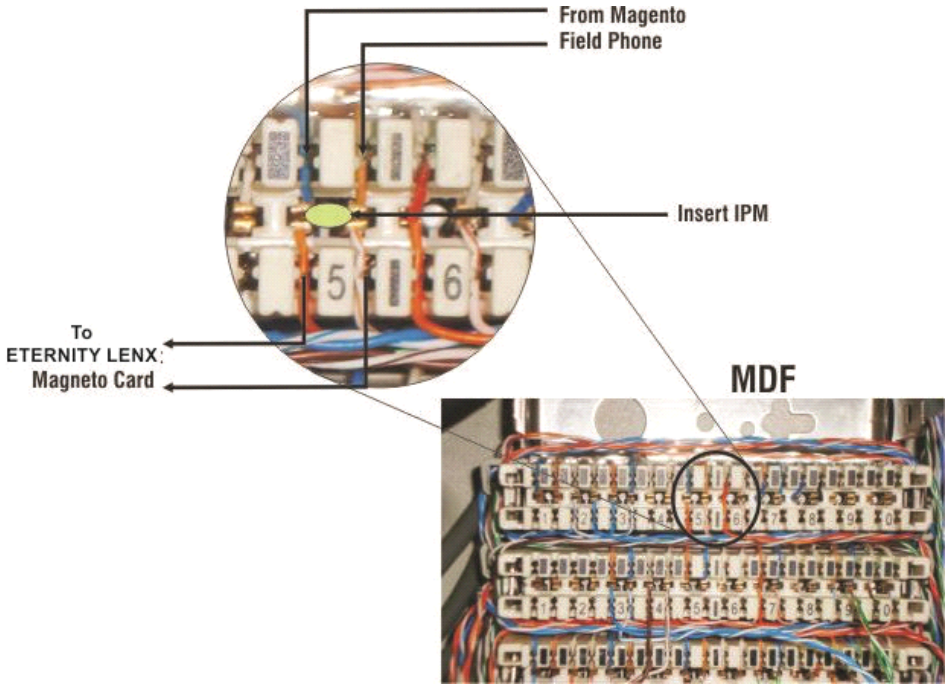
4. Select any universal slot to insert the card. Unscrew the filler bracket and remove it by pushing up the levers on the bracket.

5. Insert the Magneto card into the guide rails of the free slot. The card's connectors must make perfect contact with the connectors on the backplane motherboard. Press down the levers of the mounting bracket to secure the card in its slot and fix the two screws provided with the card on the mounting bracket.

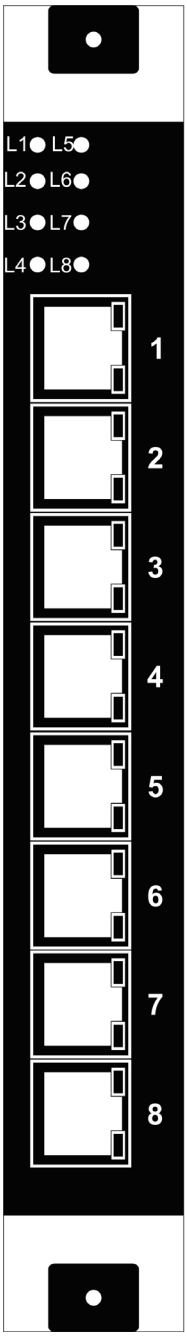
6. Now, plug in the cables supplied with the Magneto Card into the connectors on the card. Terminate the free ends of the cables into the MDF, if applicable.

   Refer to the following block diagram for terminating the cables from the Magneto Card and the wires from the Magneto Field Telephones.

Connecting Magneto Telephones to the Magneto Card



7. Connect the pairs of wires from the Magneto Field Phones to the appropriate pairs emerging from the Magneto Card on the MDF. Refer the cable diagram below.



Connector	Color	Connection	H/w Port Offset
RJ45-1 (Blue)	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	Magneto — — —	01 — — —
RJ45-2 (Orange)	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	Magneto — — —	01 — — —
RJ45-3 (Green)	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	Magneto — — —	01 — — —
RJ45-4 (Brown)	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	Magneto — — —	01 — — —
RJ45-5 (Blue)	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	Magneto — — —	01 — — —
RJ45-6 (Orange)	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	Magneto — — —	01 — — —
RJ45-7 (Green)	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	Magneto — — —	01 — — —
RJ45-8 (Brown)	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	Magneto — — —	01 — — —

8. Repeat the same steps to install the other Magneto Cards.

9. If you do not have any other Card to insert and have completed the installation procedures, power on the system and observe the Reset Cycle and the LED Pattern of the Magneto Card.

LED Pattern of the Magneto Card

Stage	LED Color	LED Cadence
Auto Upgradation / Auto Downgradation		
Card waiting for application	RED	L1 turned ON for 200ms, turned OFF simultaneously.
Card is up, loaded with new application	GREEN	L1 turned ON for 200ms, turned OFF simultaneously.
Initialization		
	RED	L1 to L8 turned ON 500ms-OFF 500ms
	GREEN	L1 to L8 turned ON 500ms-OFF 500ms
	ORANGE	L1 to L8 turned ON 500ms-OFF 500ms
Stand-By[a]	ORANGE, GREEN,	L1 toggles 1 sec Orange and RED
Normal (Port Event)		
Ring (incoming/outgoing call)	RED	LED of the Port continuously ON
Port Disabled		LED of the Port continuously OFF
OFF-Hook (in Speech) For Outgoing call from Magneto: When called extension answers the call For Incoming call to Magneto: When stopping Ring on Magneto using MRE key or #	GREEN	LED of the Port continuously ON
Port Idle		LED of the Port continuously OFF
Errors		
Invalid Card Configuration Jumper	ORANGE	All LEDs Flash (250ms ON-250ms OFF) twice, OFF 3 sec.
Invalid Slot detection[b]	ORANGE	LEDs Flash (250ms ON-250ms OFF) 6 times, OFF 3 sec.

a. Waiting to be detected by CPU Card.
b. Waiting to be detected by CPU Card.

Jumpers on the Main Board

Jumper Number	Position	Function
J1 & J2	AB	NA
	BC (default)	Normal Operation
J3, J5 & J6	AB (default)	Normal Operation
	BC	For uploading software using COM Port

The Radio Card

The Radio Interface Card (RIC) adds the Two-way Radio functionality in the system. In Two-way radio, the speech can be transmitted as well as received by the radio devices such as Radio Phone, Radio Repeater. Such devices are called Radio Transceivers. The Two-way radio works on High Frequency (HF), Very High Frequency (VHF) or Ultra High Frequency (UHF).

Radio Cards for ETERNITY LENX

Card Name	Configuration and Application
ETERNITY ME Card Radio8	8-port card to connect 8 Radio devices.
ETERNITY ME Card Radio4	4-port card to connect 8 Radio devices.

The maximum number of radio ported supported are 16.

Connectors

The Radio Card has RJ45 connectors. A multi-pair cable is provided for each connector.

LED

The Radio Card has one tri color LED.

Installing the Radio Card

1. Have the necessary wiring for the Radio Ports in place.

   You may install an MDF to connect the Radio Ports with the Radio device wires.

   OR

   You may connect the wires from the Radio device directly to the Radio Port.

   You are advised to use a separate set of Krone Modules for connecting the Radio devices to the Radio ports of the system.

2. Prepare for the card installation by switching off power supply and wearing an electrostatic discharge preventive wrist strap and use a grounding mat.

3. Unpack the Radio Card and check the package contents.

4. Select any universal slot to insert the card. Unscrew the filler bracket and remove it by pushing up the levers on the bracket.

5. Insert the Radio Card into the guide rails of the free slot. The card's connectors must make perfect contact with the connectors on the backplane motherboard. Press down the levers of the mounting bracket to secure the card in its slot and fix the two screws provided with the card on the mounting bracket.

6. Now, plug in the cables supplied with the Radio Card into the connectors on the card. Terminate the free ends of the cables into the MDF, if applicable.

7. Connect the pairs of wires from the Radio devices to the appropriate pairs emerging from the Radio Card of the system on the MDF. For more details, see “The Main Distribution Frame (MDF)”.

Refer to the Pin-out details given below.

ETERNITY ME Card Radio8

Connector	Color	Pin Number	Signaling	H/w Port Offset
RJ45-1 to RJ45-8	Orange & White	1	PTT	01 to 08
	Orange	2	PTT_RTN	
	Green & White	3	Rx-	
	Blue	4	Tx+	
	Blue & White	5	Tx-	
	Green	6	Rx+	
	Brown & White	7	Unused	
	Brown	8	Unused	

ETERNITY ME Card Radio4

Connector	Color	Pin Number	Signaling	H/w Port Offset
RJ45-1 to RJ45-4	Orange & White	1	PTT	01 to 04
	Orange	2	PTT_RTN	
	Green & White	3	Rx-	
	Blue	4	Tx+	
	Blue & White	5	Tx-	
	Green	6	Rx+	
	Brown & White	7	Unused	
	Brown	8	Unused	

8. Repeat the same steps to install the other Radio Cards.

9. If you do not have any other Card to insert and have completed the installation procedures, power on the system. Observe the Reset Cycle and the LED Pattern of the Radio Card.

LED Pattern

Stage	**LED Color**	**LED Cadence**
Auto Upgradation / Auto Downgradation		
Card waiting for application	RED	200ms ON - 200ms OFF

Stage	LED Color	LED Cadence
Card is up, loaded with new application	GREEN	200ms ON - 200ms OFF
Initialization		
	RED	500ms ON - 500ms OFF
	GREEN	500ms ON - 500ms OFF
	ORANGE	500ms ON - 500ms OFF
Stand-By	ORANGE, GREEN	Toggles 1 sec Orange 1 sec Green
Port Status		
Selected Port's data are transmitted to master card	RED	Toggle on each event
Selected Port's data are received from master card	GREEN	Toggle on each event from master

Jumpers on the Main Board

Jumper Number	Position	Function
J3	AB (default)	Normal Operation
	BC	For uploading software using COM Port
J4 and J5	AB (default)	Normal Operation
	BC	For uploading software using COM Port

In PCB-P-200-80-01-01, for 4-wire speech the Jumpers J1 and J2 on the modules must always be set in AB position.

In PCB-P-200-80-01-02 onwards only 4-wire speech is possible therefore there are no Jumpers on the modules.

The Data Card

The Data Card supports four Ethernet 10/100mbps interfaces. Ethernet data coming to ports can be mapped to 2Mb streams. Each data port can be mapped to one 2Mb - E1 stream, that is 30 channels. The remaining channels of E1 can be used for voice applications. The Data Card has a 4-port Ethernet switch on board, which can aggregate multiple data streams to PCM streams of the system.

The Data Card can be installed in any of the Universal Slots of the system.

Ports and Connectors

The Data Card has an RJ45 Connector for each port. Use the cables supplied with the card for connectivity.

You can connect the cables from the LAN switch to these connectors.

LEDs

The Data Card has one dual color LED.

Installing the Data Card

1. Unpack the Data Card and check the package contents. It is recommended that you switch off the power supply, before you begin the installation of the card. Always wear an electrostatic discharge prevention wrist strap/belt and use a grounding mat.

2. Select any free (empty) slot from the Universal Slots. Unscrew and remove the filler bracket of the empty slot. Do not discard the filler bracket! Preserve it for future use!

3. Insert the Data Card into the guide rails of the free slot you selected for the card. The connectors on the card should make perfect contact with those of the slot on the backplane motherboard.

   Press down the lever on the card mounting brackets to secure the card in its slot. Fix the mounting bracket in place with the two screws provided.

4. Use the cable supplied with the card to connect the Data Ports to the Ethernet Network (Switch/PC).

5. If you have completed all other installation tasks, you may turn ON the system and observe the Reset Cycle and the LED pattern of the Data Card.

   LED Pattern of the Data Card

Status	Color	Cadence
Waiting for Master to start Up-gradation procedure (waits for 10 seconds)	Red	Blinking 200ms ON - 200ms OFF
Up-gradation in progress	Green	Blinking 200ms ON - 200ms OFF
Application run error after up-gradation procedure	Red	Continuous ON
Auto upgradation completed successfully and in process to start new application	Green	Continuous ON

Status	Color	Cadence
Main application is being executed	Red	Blinking 1 sec ON - 1 sec OFF

Jumpers

Jumper Number	Position	Function
J9	AB (default)	External Boot - Normal
	BC	Internal Boot

SIP Extensions

SARVAM UCS supports up to 2000 SIP/UC Users. The SIP/UC Users function in the same way as DKP/SLT extensions of the system. SIP/UC Users can make and receive calls to any extension user of the system and to external numbers over various telecom networks like CO, Mobile, ISDN PRI, BRI, and VoIP[46].

You may register any SIP-enabled device — a Matrix UC Client, an IP-phone, a Soft phone, Analog Phone Adapter — as the SIP User of the system.

The Matrix UC Clients also offer UC functionalities in addition to the SIP functionalities.

The SIP Users register with the CPU Card of the system. Five free SIP Users are provided by default. You may register any of the SIP-enabled devices except the Matrix UC Clients with these free SIP Users. For registering the Matrix UC Clients, you must purchase the Matrix VARTA User License. If you require additional SIP Extensions you must purchase the IP Subscribers License.

The system supports two NX DBM VOCODER64 Modules. You must purchase the module separately. Each NX DBM VOCODER64 module supports a maximum of 64 VOCODER channels. The Vocoder channels are required for — VoIP to Non-VoIP calls, VoIP to VMS calls and VoIP to VoIP calls — where transcoding is required.

The system provides 4 pre-activated VOCODER channels by default. To use these channels make sure you have installed atleast one NX DBM VOCODER64 module. If you require more channels, you can purchase the channel licenses according to your requirement.

For more information on Licenses — Matrix VARTA User License, IP Subscribers License and VOCODER Channel License, see “License Management”.

You may connect any Standard Phone or Extended IP Phones of Matrix as SIP Users.

Matrix VARTA WIN200, VARTA ADR100 and VARTA AMP100 can be registered as SIP Users, also offering the support for UC functionalities.

You may also connect/register[47] the following as SIP Extensions of the system:

- Connect SPARSH VP248, the Extended IP Phone. For instructions, see “Connecting SPARSH VP248 as Extended SIP Extension”.
- Connect SPARSH VP310, the Extended IP Phone. For instructions, see “Connecting SPARSH VP310 as Extended SIP Extension”.
- Connect SPARSH VP330, the Touch Screen Extended IP Phone. For instruction,“Connecting SPARSH VP330 as Extended SIP Extension”.
- Connect SPARSH VP510, the Premium IP Phone. For instruction, “Connecting SPARSH VP510 as Extended SIP Extension”.
- Connect Extended SPARSH VP210, the Entry Level IP Phone. For instruction “Connecting SPARSH VP210 as Extended SIP Extension”.
- Connect Extended SPARSH VP710, the Smart Video IP Phone. For instruction,“Connecting Extended SPARSH VP710 as Extended SIP Extension”.

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46. Calls between VoIP, Public and Private Networks may be subject to Regulation in your country. You may have to configure your system to allow or restrict call traffic between networks to comply with the telecom regulations of your country. To know more, read “Logical Partition”.

47. SARVAM UCS supports only IPv4 Addresses for registering SPARSH VP248, SPARSH VP310, SPARASH VP330, Extended SPARSH VP710, Mobile UC Clients as well as VARTA WIN200.

You can register following UC Clients as SIP Users of the system:

- Matrix VARTA WIN200, Unified Communication Client for Windows. For instruction, refer to the *MATRIX VARTA WIN200* User Guide.
- Matrix Mobile UC Clients, as given below:
  - Matrix VARTA AMP100, the Mobile UC Client for iPhones. For instruction, refer to the *Matrix VARTA AMP100* User Guide.
  - Matrix VARTA ADR100, the Mobile UC Client for Android Smartphones/Tablets. For instruction, refer to the *Matrix VARTA ADR100* User Guide.

Refer to “SARVAM UCS Features Supported in Terminals” to know the features supported in each client.

The SIP Users may be registered over ***WAN*** or over ***LAN*** according to your preference and your IP network installation scenario. Extended SIP Phones and UC Clients can be registered with SARVAM UCS, for details refer to the specific topic.

You can register the same SIP User from three different locations.

If you register the Extended IP Phone outside the Region/Country selected for SARVAM UCS, the time and Time Zone dependant features, such as Alarms, Reminders, Time Zone Display, of the phone at each location will operate according to the Real Time Clock of SARVAM UCS. Also, Access Codes and Emergency Numbers will work according to the Region/Country selected for SARVAM UCS.

- Connect the Extended IP Phone, or any Standard IP Phone to the LAN Switch.

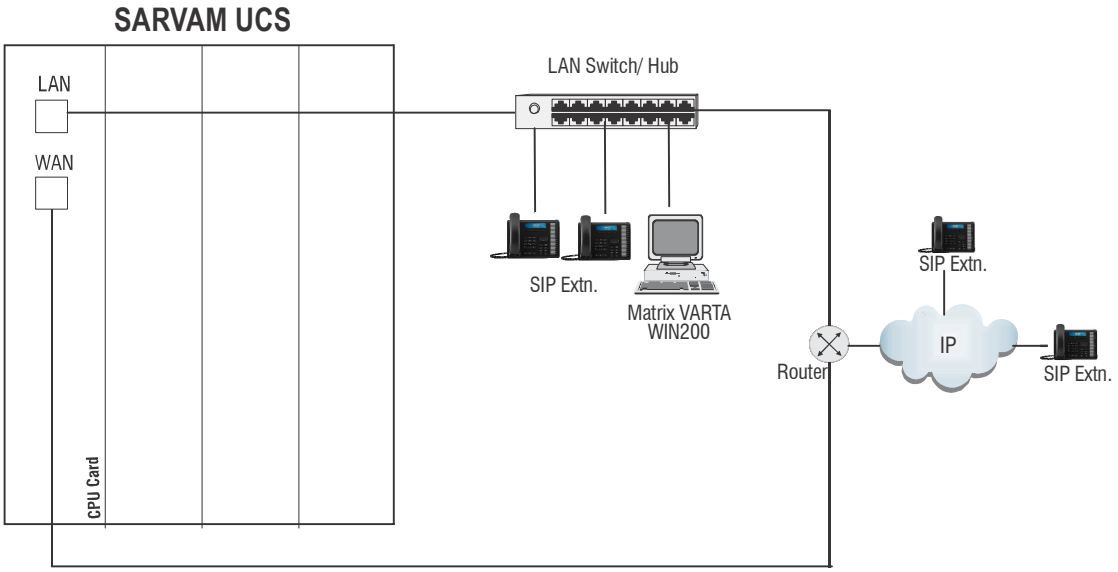
- Register any SIP device (Extended IP phone/ Soft clients or Standard IP phone) on the public network as SIP Extension.

- When you register the Matrix Extended IP Phone with the system, the WAN/LAN port is used for Auto Configuration as well for Registration of the Extended IP Phones.

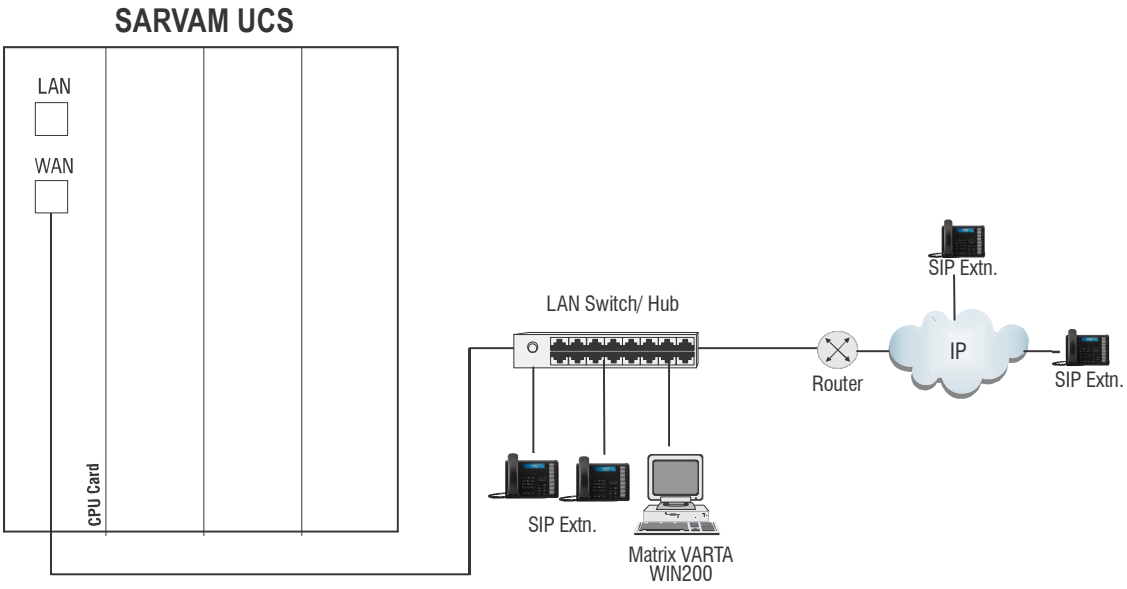
- When you register a SIP device other than the Matrix Extended IP Phone on the public network as SIP Extension, do the following:
  - In this SIP device configure the following:
    - the Registrar Server Address of SARVAM UCS
    - the Registrar Server Port
    - the SIP ID
    - Authentication ID and Password.
  - Configure Port Forwarding for the WAN Port of SARVAM UCS on the Router.

If the SARVAM UCS is connected to a ***Public Network***,

- Connect the Matrix VARTA WIN200, Extended IP Phone, or any Standard SIP device to the LAN Switch.
- Register any SIP device (Matrix VARTA UC Clients, Extended IP phone or Standard SIP phone) on the public network as SIP extension.



If the SARVAM UCS is connected to a ***Private Network (Behind the NAT)***,



- Connect Matrix VARTA WIN200, Extended IP Phones or Standard SIP Phones to the LAN Switch
- You may also register any SIP device (Matrix VARTA UC Clients, Extended IP Phone or Standard SIP phone) on the public network as SIP Extension.

  When you register the Matrix Extended IP Phone with SARVAM UCS, configure **Port Forwarding** for the **WAN port of the CPU Card** on the Router. The WAN Port is used for Auto Configuration of the Extended IP Phones.

Connecting SPARSH VP248 as Extended SIP Extension

You are recommended to complete the following steps before connecting the Matrix Extended IP Phone to SARVAM UCS:

- Decide the location of the Extended IP Phone, whether within the same network or outside, according to your installation scenario.

*If you want to use the **DHCP Server** for assigning IP Address to the Extended IP Phone, select **DHCP option 224** and **Data Type** as **'String'** and program the LAN or WAN IP Address /Domain Name of SARVAM UCS and SPARSH Port in the format **"IP_Address:Port"** in your DHCP Server as per your installation scenario.*

- Log in to *Jeeves*. For instructions, read the topic "Configuring SARVAM UCS".

- Assign SIP User ID (will work as an extension number) to the Extended IP Phone. For instructions on assigning SIP ID, see "Configuring SIP Extensions".

  For the SIP User ID you assigned to the Extended IP Phone, you must configure the necessary parameters in SARVAM UCS so that Extended IP Phone can register as a SIP Extension. For instructions, see the topic "Configuring SIP Extension Settings as per the Extended Phone Type" under *Configuring SIP Extensions*.

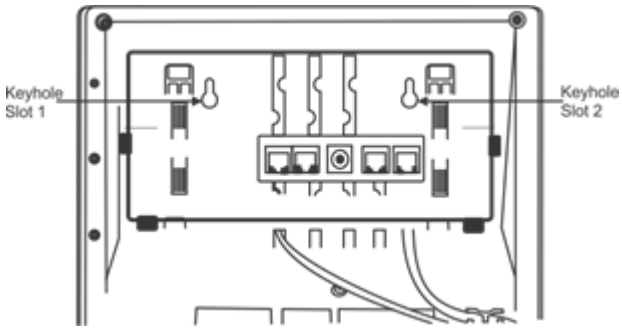
Now, follow the steps described below to install the Extended IP Phone. The instructions are common for all models of the SPARSH VP248. For the purpose of illustration, the premium model, SPARSH VP248P, has been used.

1. Unpack the SPARSH VP248 box and verify package contents.

2. Mount the phone on a desk or wall at a location convenient to you.

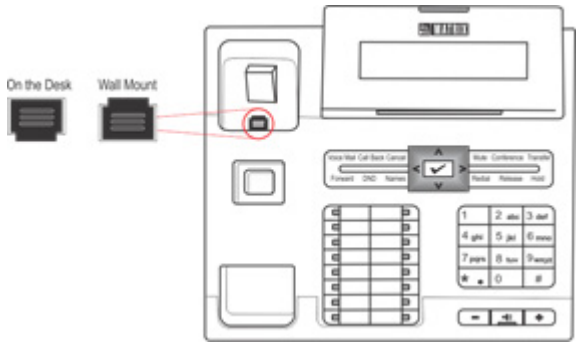
   - When mounting the phone on the wall,

     - Use the mounting template to drill holes of appropriate size and distance. Fix the screw grips in the holes you drilled.
     - Fix two screws in the holes on the wall, ensuring that they are aligned with the Keyhole Slots 1 and 2.
     - Use wall plugs, if required, to fix the screws. Leave the screw heads protruding from the wall to fit into the Keyholes.



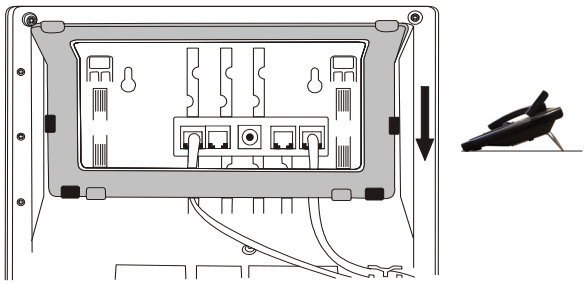
   - Now, mount the phone on the wall, with the screws fitting into the Keyhole slots.

- Reverse the handset wall mount tab to make sure the handset remains intact when you mount the phone. Push the handset wall mount tab upwards to remove it from the slot. Rotate it 180 degrees clockwise and push it downwards into the slot.

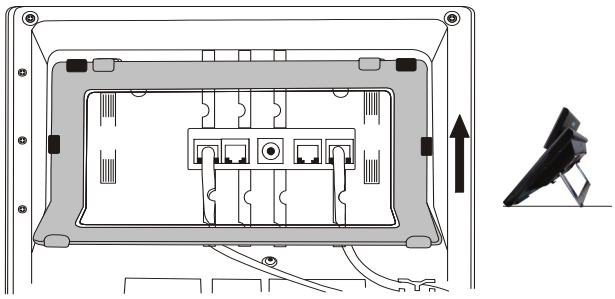


- When you mount the phone on a desk,
  - You can attach the Foot Stand in two ways as illustrated in the following.

Foot Stand attached at 30° Angle



Foot Stand attached at 50° Angle

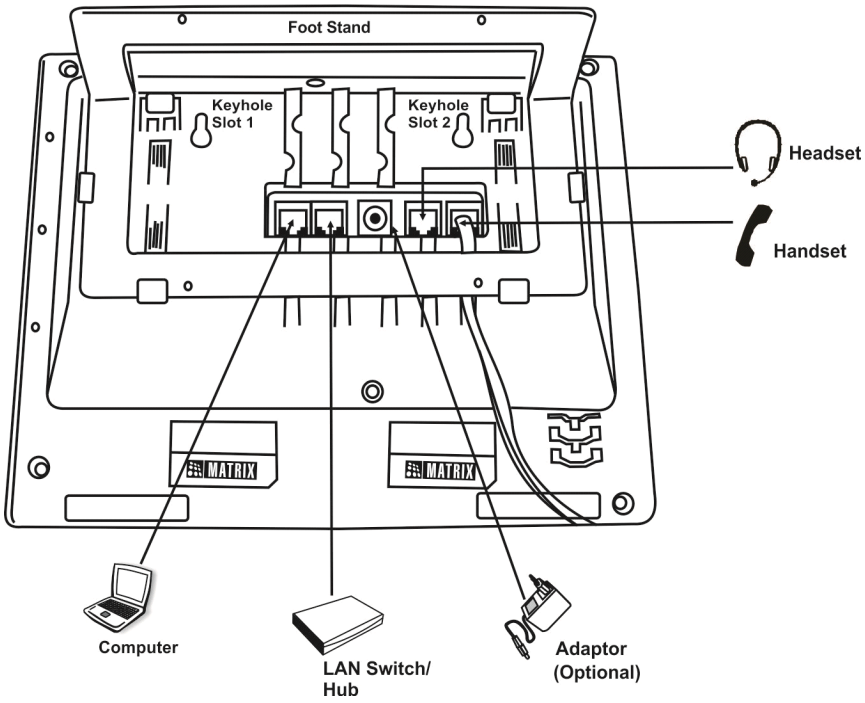


If you attach the Foot Stand at 50°, the phone will be placed in an almost upright position on your desk.

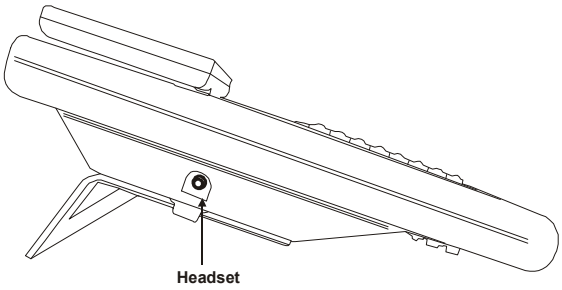
  - Decide which of these positions would work for you best and accordingly attach the Foot Stand.

3. Connect the Handset to the Phone body.
   - Plug the long straightened end of the phone cord into the handset jack at the bottom of the phone marked with the handset symbol.

- Plug the other (short straight) end of the phone cord into the jack at the bottom of the handset.

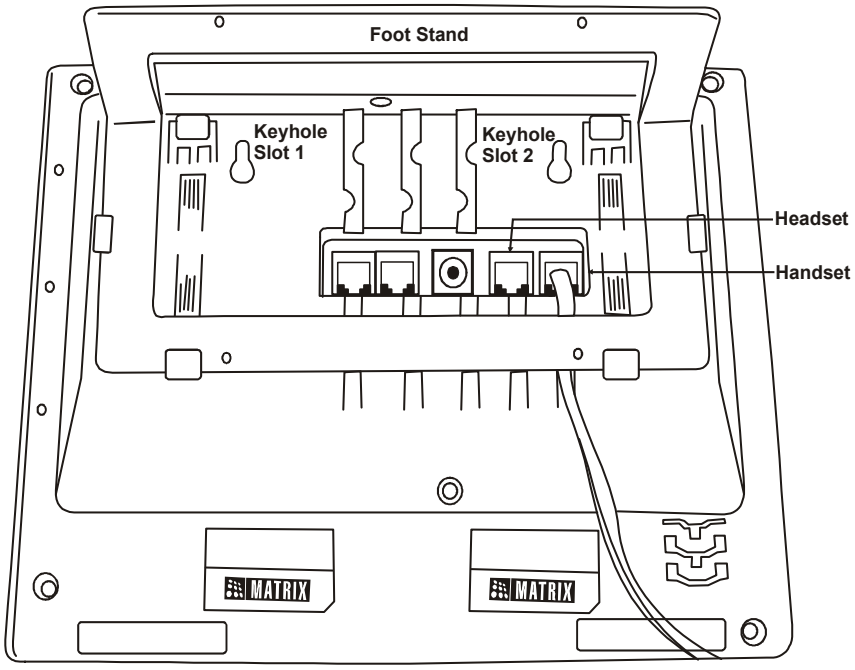


4. If you want to use a Headset (not supplied) with your phone, you may plug a headset with a 2.5 mm single connector into the headset jack headset jack with the symbol on the left side panel of the phone, as illustrated in the figure below.



OR

You may plug a headset with an RJ9 connector into the headset port at the bottom of the phone, marked with the symbol , as illustrated in the figure below.



5. Connect the LAN Port of SPARSH VP248 to the LAN Switch/Hub or a Router, according to your installation scenario.

6. To connect your phone to a computer on your desk, use an Ethernet cable (not supplied with this phone) to connect the PC Port of the phone to the LAN Port of the computer.

7. It is a PoE enabled phone and can be powered over Ethernet by connecting it to a PoE enabled LAN Switch (IEEE 802.3af Compliant). In this case you need not connect the Power Adapter.

   If you do not want to use PoE, plug in the connector of the Adapter into the power jack (DC Jack) at the bottom of the phone. Plug in the Power Adapter into a power outlet.

If both the power options, that is, PoE as well as Power Adapter are available to the phone, then the phone will derive power from the PoE enabled LAN Switch.

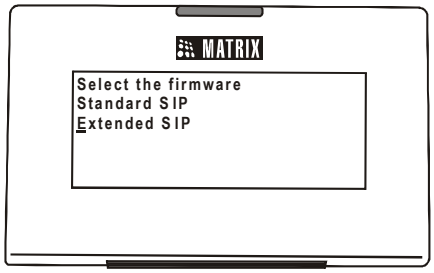
The IP phone should be used with Matrix original power adapter (5V/2A) only. The use of any third-party power adapter may cause damage to the phone.

8. Switch ON power supply.

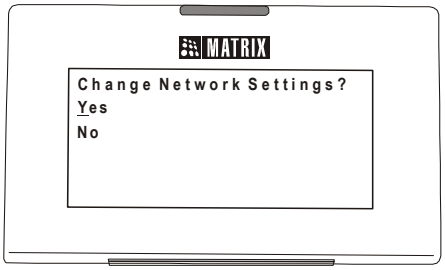
   When you power the phone, the boot process will be initiated in the following sequence.

   - All keys with LED, including the Speaker key, and the Ringer LED, will glow.
   - The LCD display will light up and booting message appears.
   - As soon as the ‘Loading...’ message appears on the phone display, press **#** key.

- Select the firmware **Extended - IP Phone**. Move the cursor by pressing the DOWN navigation key **V**.
- When the cursor is placed under the Extended IP Phone, press Enter key.



- The phone will start loading the Extended IP Phone Firmware. It will display current firmware being loaded.
- After loading the firmware, the phone will prompt you to change Network settings.



- Wait for a few seconds.

If you want to change the Network Settings or Server Settings, press the Enter key. Detailed instructions for changing the Network Settings of the phone are provided at the end of this topic. See "Network Settings"at the end of this topic.

- The phone makes DHCP Discovery and fetches its IP Address and Server Address from the DHCP Server.

  On getting the IP Address and Server Address, the phone initiates Auto Configuration to download the configuration files from SARVAM UCS.
- As the phone downloads the configuration files, the file names will appear one by one.
- On successful download of all configuration files, the phone attempts to register with SARVAM UCS.
- On successful registration, the phone will display the current day, date and time, the extension number and name assigned to the Extended IP Phone.

Network Settings

You can change the network settings of the Extended IP Phone by accessing the Local Menu of the phone. To move the cursor and scroll through the menu and submenu options, use the following touch sense navigation keys on your phone.

- The Enter key ✓ to make a selection or to complete an action.
- The Up key ∧ to move up the Menu.

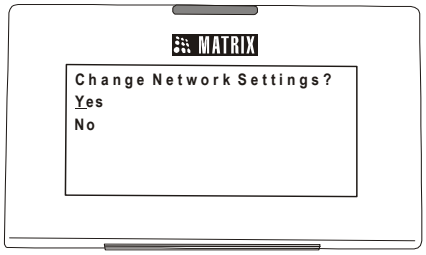
- The Down key ⋁ move down the Menu.
- The Forward key > move the cursor one character.
- The Back key < to move the cursor one character and to return from the submenu to the main menu.

The cursor is a non-blinking underscore that appears under the first letter of the first option in the menu. To make a selection in the menu, you must move the cursor in the desired direction using the Up, Down, Forward and Back key. When the cursor is at the desired position, press Enter key to make a selection.

Accessing Network Settings

You can access the Network Settings of the Extended IP Phone in any of the following stages:

1. During start-up, when the phone prompts you to change the network settings after loading the firmware.

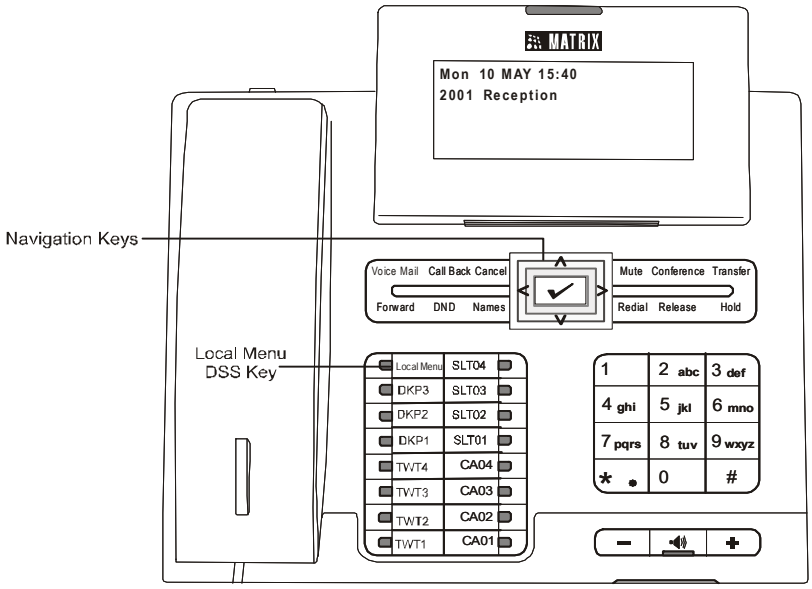


   You must press the Enter Key to select Yes and access network settings.

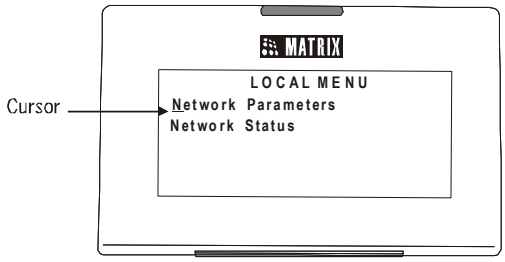
2. When the phone is making Network discovery, downloading configuration files, attempting registration.

   You must press the Enter Key ✓ to access network settings,

3. When the phone is in idle state. You must press the DSS key assigned to ‘Local Menu’.



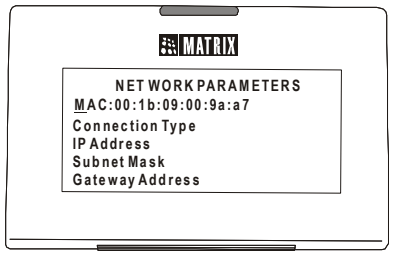
4. When you press the Local Menu DSS Key (in idle state) or when you press the Enter key during any process, the Local Menu appears on your phone display.



You can configure Network Parameters and view Network status from the Local Menu.

Configuring Network Parameters

- In the Local Menu of the phone, select Network Parameters by pressing the Enter Key.
- The Network Parameters submenu appears.



- Use the Down/Up key to reach the desired network parameter and press Enter key to select and change the settings.
- You can configure all network parameters described below, except the MAC Address.

- *To enter a dot in the editable fields — IP Address, Subnet Mask, Gateway Address, DNS Address, Server Address — press * (Star) key.*
- *If you want to clear a single digit/character, move the cursor to the right of the digit/character you wish to clear. Now press the Cancel key. The digit to the left of the cursor will be deleted. If the cursor is to the extreme left and you press the Cancel key, you will go to the previous menu.*

Connection Type

- Select the Connection Type as DHCP, PPPoE or Static according to the IP Addressing scheme of your network.

  If you select DHCP or PPPoE, the phone will be assigned IP Address, Subnet Mask and Gateway Address, DNS Address Server Address, automatically by the DHCP/PPPoE server.

  For PPPoE Connection Type, you must configure the PPPoE User ID and Password provided by the Internet Service Provider.

  If you select Static, you must assign the IP Address, Subnet Mask and Gateway Address to the phone.

IP Address

- If you select Static as Connection Type, enter the static IP Address to be assigned to the phone.

  Enter the desired Static IP Address by pressing the digit keys.

  To enter the dot/period in the IP Address, press the Star '*' key.

Subnet Mask

- If you select Static as Connection Type, enter the Subnet Mask to be applied on the phone by pressing the digit keys.

  To enter the dot/period in the IP Address, press the Star '*' key.

Gateway Address

- If you select Static as Connection Type, enter the Gateway Address here. This is the IP Address of the LAN Port of the Router.

DNS Server

- If you select Static as Connection Type, select the DNS Server option **Static** and configure the DNS Address.

- If you select DHCP or PPPoE as Connection Type and your Internet Service Provider provides DNS Address, select the DNS Server option **Automatic**. However, if your Internet Service Provider does not provide DNS Address, select **Static** and configure the DNS Address.

DNS Address

- If you select DNS Server as Static, enter the DNS Address here.

  To enter dot/period in the IP Address, press the Star '*' key.

DNS Domain Name

- If you select DNS Server as Static, enter the DNS Domain Name here. DNS Domain Name is optional.

PPPoE User ID

- If you have selected PPPoE as Connection Type, you must enter the User ID provided to you by your Internet Service Provider.

PPPoE Password

- This is the password provided by your Internet Service Provider for the PPPoE User ID. If you have selected PPPoE as Connection Type, you must enter the password provided by your Internet Service provider here.

PPPoE Service Name

- If your Internet Service Provider has provided a Service Name, enter the Service Name here. If your Internet Service Provider has not provided a Service Name, do not configure this parameter.

Server Address

- The system works as the Auto Configuration Server for the phone. Enter the LAN or WAN IP Address/ Domain Name of SARVAM UCS here. Default: blank. The phone sends the request for configuration files to this Server Address.

  If you have selected DHCP as Connection Type, the phone will get the Server Address and Port automatically from the DHCP Server. For this, use ***DHCP option 224*** and ***Data Type*** as '***String***' to provide Server Address and Port from the DHCP Server.

  For PPPoE and Static Connection Types, you need to enter the Server Address.

Server Port

- Enter the SPARSH Port of SARVAM UCS here. The phone sends the request for configuration files to this port.

  Valid range of the port is: 80 or 1025–65535. Default: 80.

VLAN Setting

If your phone is connected to a virtual LAN, you need to configure VLAN Settings.

To enable the VLAN switch to correctly route packets generated by the phone and the computers (on the LAN) to each other, the packets must be tagged with a VLAN header.

The VLAN header consists of the VLAN ID (12-bit) and Class of Service (CoS, 3-bit) for prioritization of traffic[48].

The meaning of CoS bits with respect to traffic type is as follows:

CoS	Traffic Type
0	Best Effort
1	Background
2	Spare
3	Excellent Effort
4	Controlled Load
5	Video
6	Voice
7	Network Control

- Select **Phone VLAN/COS** to add VLAN header to the packets generated by the phone, and add VLAN header to the packets relayed from the PC to its LAN port (packets generated by the PC connected to its PC port).

  - To configure Phone VLAN/COS, select **Enable?**. The VLAN ID will be tagged on all packets generated by the phone (SIP, RTP, DNS, ARP, etc.). Default: Disabled.

48. The IEEE 802.1P standard allows Layer2 switches to prioritize the traffic, thus providing Quality of Service (QoS), that is, better handling of data that pass over a network, thereby resulting in greater reliability and quality. Quality of Service (QoS) on Layer2 is referred to as Class of Service (CoS) which is defined by IEEE 802.1P.

    - Select **VLAN ID** and enter the VLAN ID that you have assigned to the VLAN in which the IP Phones are connected. Valid range: 0-4094. Default: 1.
    - Select **SIP CoS** and define the CoS (priority) bits in all SIP packets. Valid range: 0-7. Default: 3
    - Select **RTP CoS** and define the CoS (priority) bits in all RTP packets. Valid range: 0-7. Default: 6.
- Select **PC/VLAN CoS** to add VLAN header to all packets entering the PC Port and leaving the LAN port of the phone. Default: Disabled.
    - To configure PC VLAN/COS, select **Enable?**.
    - Select **VLAN ID** and enter the same ID as you have assigned to the VLAN in which the computers are connected. Valid range: 0-4094. Default: 1.
    - Select **CoS** and define the Layer 2 CoS (priority) bits. Valid range: 0-7. Default: 0.

PCAP

To capture packets sent and received from and by the phone for monitoring and troubleshooting, you can enable PCAP on the phone. The phone captures up to 1 MB of packets. For more information and for instructions on how to use PCAP Trace on the phone, see “Using PCAP Trace for Matrix SPARSH VP248 Extended IP Phone”, under *PCAP Trace*.

When you change the Network Settings, the phone will restart.

MAC Cloning

If you require cloning of the MAC Address, configure the following:

- Select **Enable?**.
- In **Enter Clone MAC Address**, enter the address you wish to clone.

802.1x Authentication

If you want to restrict unauthorized clients from connecting to your LAN, you need to enable 802.1x Authentication. Using EAP MD5 protocol the PC connected to the LAN port of the IP Phone is first authenticated and then it gets connected to LAN.

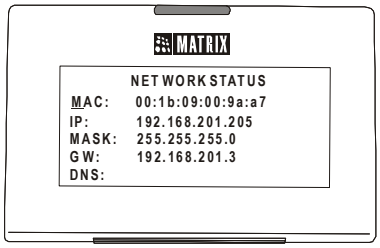
You need to configure the following 802.1x Authentication parameters:

- Select **Enable?**.
- Enter the 802.1x Authentication **Identity** provided by you network administrator.
- Enter the 802.1x Authentication **MD5 Password** associated with identity provided by your network administrator.

Viewing Network Status

- In the Local Menu of the phone, place the cursor on Network Status and press the Enter key.

- The Network Status submenu appears.



Use the Down/Up key to view the status of the various network parameters. The status of the following parameters appear on your display as you scroll.

- **MAC:** This is the MAC Address of the phone.
- **IP:** The current IP Address assigned to the phone.
- **MASK:** The current Subnet mask assigned to the phone.
- **GW:** The current Gateway IP Address assigned to the phone.
- **DNS:** The Domain Name Server address assigned to the phone.
- **S. ADD:** The LAN or WAN IP Address / Domain Name of the SARVAM UCS.
- **S. PORT:** The SPARSH Port SARVAM UCS.
- **DOMAIN:** The Domain Name assigned to the phone.
- **802.1x Authentication:** The 802.1x authentication status is displayed—Success, Failure, Authenticating or Disabled.
- **FIRM:** The version of the current Firmware of the phone.
- **UBOOT**: The UBOOT release date.
- **KERNEL**: The KERNEL release date.

Connecting SPARSH VP310 as Extended SIP Extension

You are recommended to complete the following steps before connecting the Matrix Extended IP Phone to SARVAM UCS:

- Decide the location of the Extended IP Phone, whether within the same network or outside, according to your installation scenario.

*If you want to use the **DHCP Server** for assigning IP Address to the Extended IP Phone, select **DHCP option 224** and **Data Type** as '**String**' and program the LAN or WAN IP Address /Domain Name of SARVAM UCS and SPARSH Port in the format "**IP_Address:Port**" in your DHCP Server as per your installation scenario.*

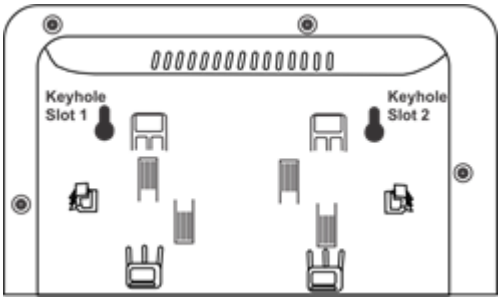
- Log in to *Jeeves*. For instructions, read the topic "Configuring SARVAM UCS".

- Assign an extension number (**SIP ID**) to the Extended IP Phone. For instructions on assigning SIP ID, see "Configuring SIP Extensions".

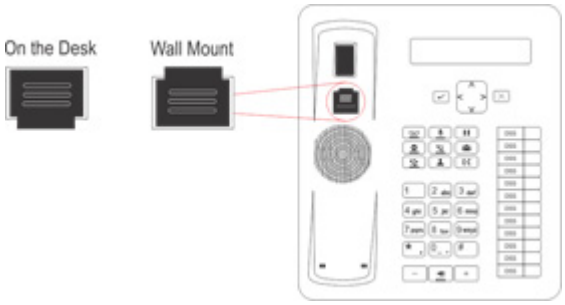
  For the SIP extension number you assigned to the Extended IP Phone, you must configure the necessary parameters in SARVAM UCS so that Extended IP Phone can register as a SIP Extension. For instructions, see the topic "Configuring SIP Extension Settings as per the Extended Phone Type" under *Configuring SIP Extensions*.

Now, follow the steps described below to install the Extended IP Phone.

1. Unpack the SPARSH VP310 box and verify package contents.
2. You can mount the phone on a wall or on the desk.
3. When you mount SPARSH VP310 on a wall,
   - Use the mounting template to drill holes of appropriate size and distance.
   - Fix the screw grips in the holes you drilled.
   - Fix two screws in the holes on the wall, ensuring that they are aligned with the Keyhole Slots 1 and 2 of SPARSH VP310. The screws should protrude from the wall to fit into the Keyhole Slots.

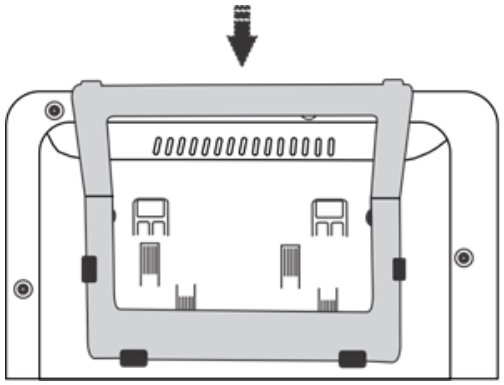


   - Now, mount the phone with the screws fitting into the Keyhole Slot.
   - Reverse the handset wall mount tab to make sure the handset remains intact when you mount the phone. Push the handset wall mount tab upwards to remove it from the slot. Rotate it 180 degrees clockwise and push it downwards into the slot.

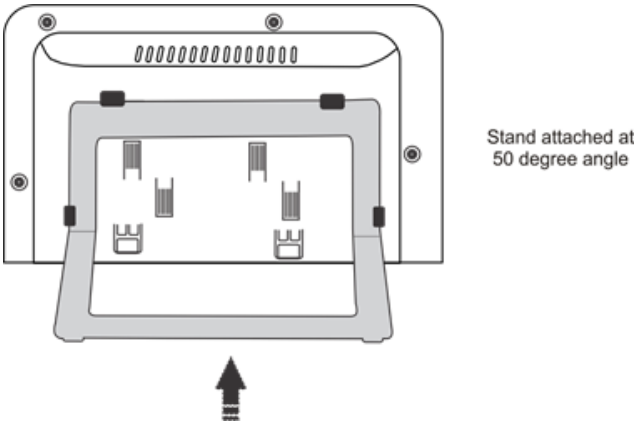


- When you mount the phone on a desk,
  - You can attach the Foot Stand in two ways as illustrated in the following.

Foot Stand attached at 35° Angle

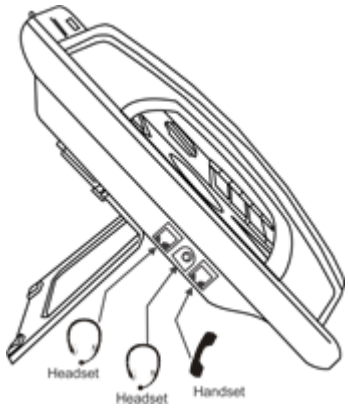


Foot Stand attached at 50° Angle



If you attach the Foot Stand at 50°, the phone will be placed in an almost upright position on your desk.

- Decide which of these positions would work for you best and accordingly attach the Foot Stand.



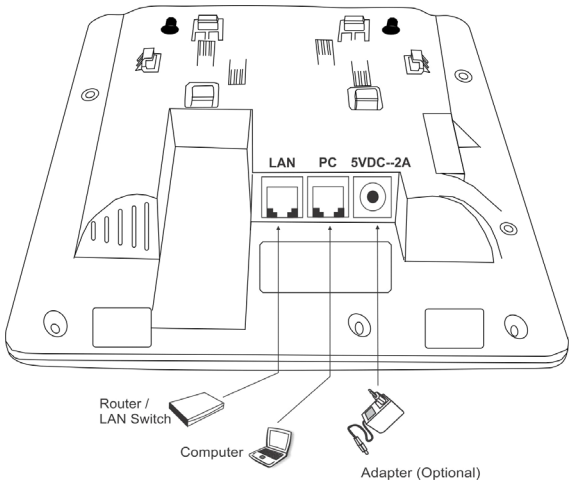
4. Connect the Handset to the Phone body.

- Plug the long straightened end of the phone cord into the handset jack on the left side panel of the phone marked with the handset symbol .
- Plug the other (short straight) end of the phone cord into the jack at the bottom of the handset.

5. If you want to use a Headset (not supplied) with your phone, you may plug a headset with a 3.5 mm single connector into the headset jack headset jack with the symbol on the left side panel of the phone, as illustrated in the figure above.

   OR

   You may also plug in a headset with RJ9 connector into the headset port on the left side panel of the phone, marked with the symbol .



6. Connect the LAN Port of SPARSH VP310 to the LAN Switch/Hub or a Router, according to your installation scenario.

7. To connect your phone to a computer on your desk, use an Ethernet cable (not supplied with this phone) to connect the PC Port of the phone to the LAN Port of the computer.
8. It is a PoE enabled phone and can be powered over Ethernet by connecting it to a PoE enabled LAN Switch (IEEE 802.3af Compliant).In this case you need not connect the Power Adapter.

   If you do not want to use PoE, plug in the connector of the Adapter into the power jack (DC Jack) with the label 5VDC-2A at the bottom of the phone. Plug in the Power Adapter into a power outlet.

If both the power options, that is, PoE as well as Power Adapter are available to the phone, then the phone will derive power from the PoE enabled LAN Switch.

The IP phone should be used with Matrix original power adapter (5V/2A) only. The use of any third-party power adapter may cause damage to the phone.

9. Switch ON power supply.

   When you power the phone, the boot process will be initiated in the following sequence.

   - All keys with LED, including the Speaker key, and the Ringer LED, will glow.
   - The LCD display will light up and the booting message appears.
   - Then the ‘Loading...’ message appears on the phone display.

- The phone will start loading the Extended IP Phone Firmware. It will display current firmware being loaded.

- After loading the firmware, the phone will prompt you to change Network settings.

- Wait for a few seconds.

If you want to change the Network Settings or Server Settings, press the Enter key. Detailed instructions for changing the Network Settings of the phone are provided at the end of this topic. See "Network Settings" at the end of this topic.

- The phone makes DHCP Discovery and fetches its IP Address and Server Address from the DHCP Server.

  On getting the IP Address and Server Address, the phone initiates Auto Configuration to download the configuration files from SARVAM UCS.

- As the phone downloads the configuration files, the file names will appear one by one.
- On successful download of all configuration files, the phone attempts to register with SARVAM UCS.
- On successful registration, the phone will display the current day, date and time, the extension number and name assigned to the Extended IP Phone.

Network Settings

You can change the network settings of the Extended IP Phone. Press the Down key ∨ when the phone is in idle state. To move the cursor and scroll through the menu and submenu options, use the following navigation keys on your phone.

- The Enter key ✓ to make a selection or to complete an action.
- The Up key ∧ to move up the Menu.
- The Down key ∨ move down the Menu.
- The Forward key > move the cursor one character.
- The Back key < to move the cursor one character and to return from the submenu to the main menu.
- The Cancel key ✕ to exit a menu.

The cursor is a non-blinking underscore that appears under the first letter of the first option in the menu. To make a selection in the menu, you must move the cursor in the desired direction using the Up, Down, Forward and Back key. When the cursor is at the desired position, press Enter key to make a selection.

Accessing Network Settings

You can access the Network Settings of the Extended IP Phone in any of the following stages:

1. During start-up, when the phone prompts you to change the network settings after loading the firmware.

   You must press the Enter key ✓ to select Yes and access network settings.

2. When the phone is making Network discovery, downloading configuration files, attempting registration.

   You must press the Down key ⋁ to access network settings.

3. When the phone is in idle state. You must press the Down key ⋁ to access the Network Settings.

Configuring Network Parameters

- When the phone is in idle state. You must press the Down key ⋁ to access the Network Settings.
- Press Enter key to select Network Parameters.
- The Network Parameters submenu appears.
- Use the Down/Up key to reach the desired network parameter and press Enter key to select and change the settings.
- You can configure all network parameters described below, except the MAC Address.

- *To enter a dot in the editable fields — IP Address, Subnet Mask, Gateway Address, DNS Address, Server Address — press * (Star) key.*
- *If you want to clear a single digit/character, move the cursor to the right of the digit/character you wish to clear. Now press the Cancel key. The digit to the left of the cursor will be deleted. If the cursor is to the extreme left and you press the Cancel key, you will go to the previous menu.*

Connection Type

- Select the Connection Type as DHCP, PPPoE or Static according to the IP Addressing scheme of your network.

  If you select DHCP or PPPoE, the phone will be assigned IP Address, Subnet Mask and Gateway Address, DNS Address, Server Address, automatically by the DHCP/PPPoE server.

  For PPPoE Connection Type, you must configure the PPPoE User ID and Password provided by the Internet Service Provider.

  If you select Static, you must assign the IP Address, Subnet Mask and Gateway Address to the phone.

IP Address

- If you select Static as Connection Type, enter the static IP Address to be assigned to the phone. Enter the desired Static IP Address by pressing the digit keys.

To enter the dot/period in the IP Address, press the Star '*' key.

Subnet Mask

- If you select Static as Connection Type, enter the Subnet Mask to be applied on the phone by pressing the digit keys.

  To enter the dot/period in the IP Address, press the Star '*' key.

Gateway Address

- If you select Static as Connection Type, enter the Gateway Address here. This is the IP Address of the LAN Port of the Router.

DNS Server

- If you select Static as Connection Type, select the DNS Server option **Static** and configure the DNS Address.

- If you select DHCP or PPPoE as Connection Type and your Internet Service Provider provides DNS Address, select the DNS Server option **Automatic**. However, if your Internet Service Provider does not provide DNS Address, select **Static** and configure the DNS Address.

DNS Address

- If you select DNS Server as Static, enter the DNS Address here.
  To enter dot/period in the IP Address, press the Star '*' key.

DNS Domain Name

- If you select DNS Server as Static, enter the DNS Domain Name here. DNS Domain Name is optional.

PPPoE User ID

- If you have selected PPPoE as Connection Type, you must enter the User ID provided to you by your Internet Service Provider.

PPPoE Password

- This is the password provided by your Internet Service Provider for the PPPoE User ID. If you have selected PPPoE as Connection Type, you must enter the password provided by your Internet Service provider here.

PPPoE Service Name

- If your Internet Service Provider has provided a Service Name, enter the Service Name here. If your Internet Service Provider has not provided a Service Name, do not configure this parameter.

Primary Server Address

- The system works as the Auto Configuration Server for the phone. Enter the LAN or WAN IP Address/ Domain Name of SARVAM UCS here. Default: blank. The phone sends the request for configuration files to this Server Address.

  If you have selected DHCP as Connection Type, the phone will get the Primary Server Address and Port automatically from the DHCP Server. For this, use ***DHCP option 224*** and ***Data Type*** as '***String***' to provide Server Address and Port from the DHCP Server.

For PPPoE and Static Connection Types, you need to enter the Server Address.

Primary Server Port

- Enter the SPARSH Port of SARVAM UCS here. The phone sends the request for configuration files to this port.

  Valid range of the port is: 80 or 1025–65535. Default: 80.

Secondary Server Address

- If required, you can also configure the Secondary Sever Address as a fallback option. If the registration with the Primary Server fails the phone will send the registration and configuration requests to the Secondary Server Address. Speech-cut or unclear speech may be observed during on-going mature calls.

Secondary Server Port

- Enter the Secondary Server Port. The phone sends the request for configuration files to this port if the Primary Server fails.

  Valid range of the port is: 80 or 1025–65535. Default: 80.

VLAN Setting

If your phone is connected to a virtual LAN, you need to configure VLAN Settings.

To enable the VLAN switch to correctly route packets generated by the phone and the computers (on the LAN) to each other, the packets must be tagged with a VLAN header.

The VLAN header consists of the VLAN ID (12-bit) and Class of Service (CoS, 3-bit) for prioritization of traffic[49].

The meaning of CoS bits with respect to traffic type is as follows:

CoS	Traffic Type
0	Best Effort
1	Background
2	Spare
3	Excellent Effort
4	Controlled Load
5	Video
6	Voice
7	Network Control

- Select **Phone VLAN/COS** to add VLAN header to the packets generated by the phone, and add VLAN header to the packets relayed from the PC to its LAN port (packets generated by the PC connected to its PC port).

49. The IEEE 802.1P standard allows Layer2 switches to prioritize the traffic, thus providing Quality of Service (QoS), that is, better handling of data that pass over a network, thereby resulting in greater reliability and quality. Quality of Service (QoS) on Layer2 is referred to as Class of Service (CoS) which is defined by IEEE 802.1P.

- To configure Phone VLAN/COS, select **Enable?**. The VLAN ID will be tagged on all packets generated by the phone (SIP, RTP, DNS, ARP, etc.). Default: Disabled.
- Select **VLAN ID** and enter the VLAN ID that you have assigned to the VLAN in which the IP Phones are connected. Valid range: 0-4094. Default: 1.
- Select **SIP CoS** and define the CoS (priority) bits in all SIP packets. Valid range: 0-7. Default: 3
- Select **RTP CoS** and define the CoS (priority) bits in all RTP packets. Valid range: 0-7. Default: 6.

- Select **PC/VLAN CoS** to add VLAN header to all packets entering the PC Port and leaving the LAN port of the phone. Default: Disabled.
  - To configure PC VLAN/COS, select **Enable?**.
  - Select **VLAN ID** and enter the same ID as you have assigned to the VLAN in which the computers are connected. Valid range: 0-4094. Default: 1.
  - Select **CoS** and define the Layer 2 CoS (priority) bits. Valid range: 0-7. Default: 0.

PCAP

To capture packets sent and received from and by the phone for monitoring and troubleshooting, you can enable PCAP on the phone. The phone captures up to 1 MB of packets. For more information and for instructions on how to use PCAP Trace on the phone, see “Using PCAP Trace for Matrix SPARSH VP310 Matrix Extended IP Phone”, under *PCAP Trace*.

MAC Cloning

If you require cloning of the MAC Address, configure the following:

- Select **Enable?** and press the Enter key. Select Yes to enable.
- In **Enter Clone MAC Address**, enter the address you wish to clone.

802.1x Authentication

If you want to restrict unauthorized clients from connecting to your LAN, you need to enable 802.1x Authentication. Using EAP MD5 protocol the PC connected to the LAN port of the IP Phone is first authenticated and then it gets connected to LAN.

You need to configure the following 802.1x Authentication parameters:

- Select **Enable?** and press the Enter key. Select Yes to enable.
- Enter the 802.1x Authentication **Identity** provided by you network administrator.
- Enter the 802.1x Authentication **MD5 Password** associated with identity provided by your network administrator.

When you change the Network Settings, the phone will restart.

Viewing Network Status

- When the phone is in idle state. You must press the Down key ⋁ to access the Network Settings.

- Again press Down key ⋁ to select Network Status and press the Enter key ✓.

  Use the Down/Up key to view the status of the various network parameters. The status of the following parameters appear on your display as you scroll.

  - **MAC:** This is the MAC Address of the phone.
  - **IP:** The current IP Address assigned to the phone.
  - **MASK:** The current Subnet mask assigned to the phone.
  - **GW:** The current Gateway IP Address assigned to the phone.
  - **DNS:** The Domain Name Server address assigned to the phone.
  - **Active Server:** This displays the Server that is active — Primary, Secondary — with which the phone is currently registered.
  - **S. ADD:** This displays the IP address of the Active Server. It may be the LAN or WAN IP Address / Domain Name of the SARVAM UCS or the Secondary Server IP Address (if configured) or any Fallback Server.
  - **S. PORT:** This displays the port of the Active Server. It may be the SPARSH Port of SARVAM UCS or the Secondary Server Port (if configured) or the Fallback Server Port.
  - **DOMAIN:** The Domain Name assigned to the phone.
  - **802.1x Authentication:** The 802.1x authentication status is displayed—Success, Failure, Authenticating or Disabled.
  - **FIRM:** The version of the current Firmware of the phone.
  - **UBOOT**: The UBOOT release date.
  - **KERNEL**: The KERNEL release date.

Connecting SPARSH VP330 as Extended SIP Extension

You are recommended to complete the following steps before connecting the Matrix SPARSH VP330 to SARVAM UCS:

- Decide the location where you want to place SPARSH VP330 within your LAN.
- By Default, in SPARSH VP330, the Connection Type selected is DHCP.
- If you want to use the **DHCP Server** for assigning IP Address to the Extended IP Phone, select **DHCP option 224** and **Data Type** as '**String**' and program the LAN or WAN IP Address /Domain Name of SARVAM UCS and SPARSH Port in the format "**IP_Address:Port"** in your LAN DHCP Server as per your installation scenario.
- Log in to *Jeeves*. For instructions, read the topic "Configuring SARVAM UCS".

- You must configure the necessary parameters in SARVAM UCS so that SPARSH VP330 can register as a SIP Extension. For instructions, see “Configuring Matrix SPARSH VP330”.

Now, follow the steps described below to install SPARSH VP330.

1. Unpack the SPARSH VP330 box and verify package contents.

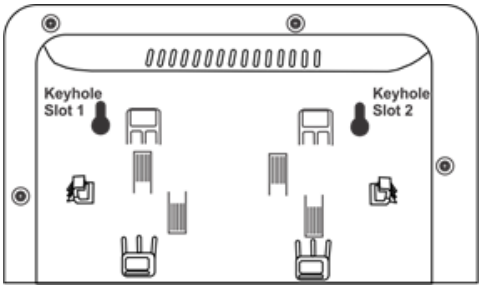
2. Mount the phone on a desk or wall at a location convenient to you.

    - When mounting the phone on the wall,

        - Use the mounting template to drill holes of appropriate size and distance. Fix the screw grips in the holes you drilled.

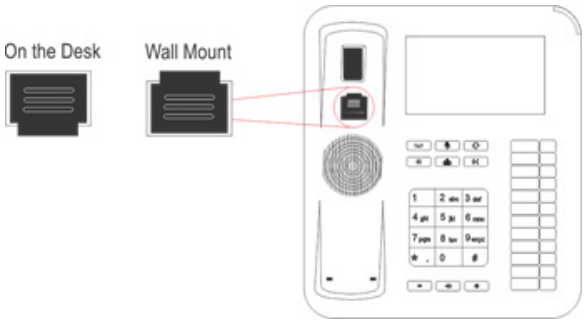
        - Fix two screws in the holes on the wall, ensuring that they are aligned with the Keyhole Slots 1 and 2.

        - Use wall plugs, if required, to fix the screws. Leave the screw heads protruding from the wall to fit into the Keyholes.

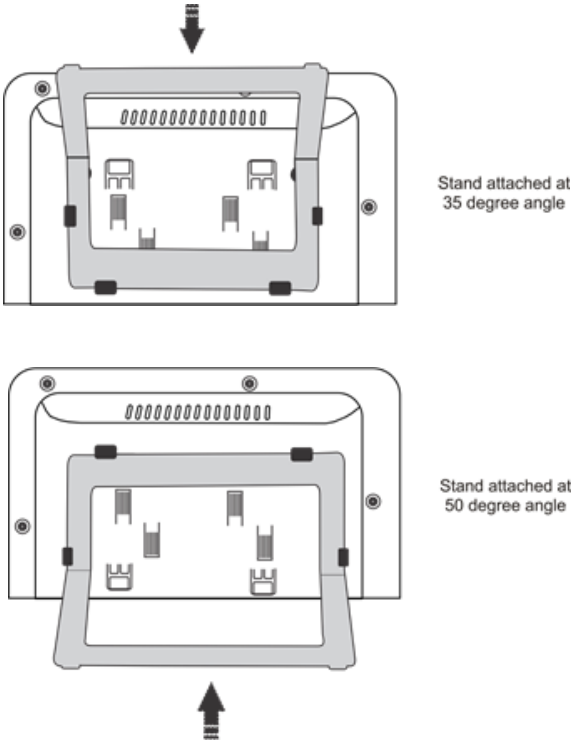


        - Now, mount the phone on the wall, with the screws fitting into the Keyhole slots.

        - Reverse the handset wall mount tab to make sure the handset remains intact when you mount the phone. Push the handset wall mount tab upwards to remove it from the slot. Rotate it 180 degrees clockwise and push it downwards into the slot.

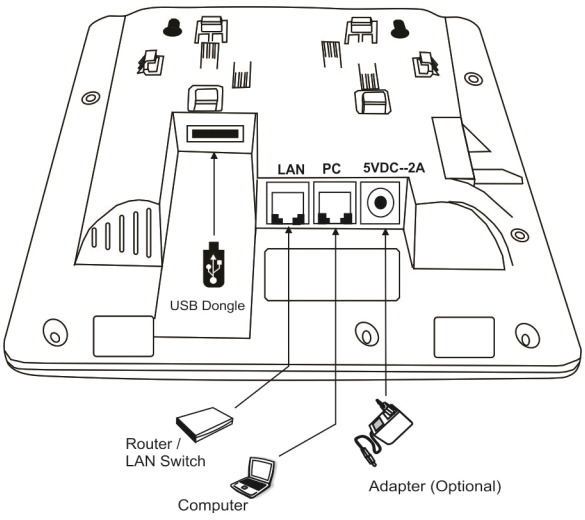


3. When you mount the phone on a desk, you can attach the Foot Stand in two ways at **35° Angle** or at **50° Angle**.



- If you attach the Foot Stand at 50°, the phone will be placed in an almost upright position on your desk.
- Decide which of these positions would work for you best and accordingly attach the Foot Stand.

Refer to the diagram below for connectivity.



4. Connect the Handset to the Phone body.

- Plug the long straightened end of the phone cord into the handset jack on the left side panel of the phone marked with the handset symbol.

- Plug the other (short straight) end of the phone cord into the jack at the bottom of the handset.

5. If you want to use a Headset (not supplied) with your phone, you may plug a headset with a 3.5 mm single connector into the headset jack headset jack with the symbol on the left side panel of the phone.

   OR

   You may plug a headset with an RJ9 connector into the headset port on the side panel of the phone, marked with the symbol .

6. Connect the LAN Port of SPARSH VP330 to the IP Network — A Router or LAN Switch — using the Ethernet Cable.

   OR

   Connect the Wi-Fi USB Adapter into the USB port of the phone.

You can purchase the Wi-Fi USB Adapter from Matrix as an optional peripheral device to support wireless network.

7. To connect your phone to a computer on your desk, use an Ethernet cable (not supplied with this phone) to connect the PC Port at the bottom of the phone to the LAN Port of the computer.

8. It is a PoE enabled phone and can be powered over Ethernet by connecting it to a PoE enabled LAN Switch (IEEE 802.3af Compliant). In this case you need not connect the Power Adapter.

   If you do not want to use PoE, plug in the connector of the Adapter into the power jack (DC Jack) with the label 5VDC-2A at the bottom of the phone. Plug in the Power Adapter into a power outlet.

If both the power options, that is, PoE as well as Power Adapter are available to the phone, then the phone will derive power from the PoE enabled LAN Switch.

The IP phone should be used with Matrix original power adapter (5V/2A) only. The use of any third-party power adapter may cause damage to the phone.

9. Switch ON power supply.

   When you power the phone, the boot process will be initiated in the following sequence.

   - All keys with LED, including the Speaker key, and the Ringer LED, will glow.
   - The LCD display will light up and booting message appears.
   - While loading the application then the loading message appears on the phone display,
   - The phone makes DHCP Discovery and fetches its IP Address and Server Address from the DHCP Server.

If you want to change the Network Settings/Server Settings or want to use Wi-Fi for connectivity, press Settings .

Refer to the SPARSH VP330 User Guide, for detailed instructions:

- *To change the Network Settings of the phone and configure the network parameters.*
- *To use Wi-Fi for connectivity and configure its parameters.*

- On getting the IP Address and Server Address, the phone initiates Auto Configuration to download the configuration files from SARVAM UCS.
- On successful download of all configuration files, the phone attempts to register with SARVAM UCS.
- On successful registration, the Home screen appears.

The phone will register successfully, only if the SIP Extension parameters in SARVAM UCS have been correctly configured as per your installation scenario.

Refer to the ***SPARSH VP330 User Guide*** to know more.

Connecting SPARSH VP510 as Extended SIP Extension

You are recommended to complete the following steps before connecting the Matrix Extended IP Phone to the system when used with SARVAM UCS application:

- Decide the location of the Extended IP Phone, whether within the same network or outside, according to your installation scenario.

*If you want to use the **DHCP Server** for assigning IP Address to the Extended IP Phone, select **DHCP option 224** and **Data Type** as '**String**' and program the LAN or WAN Port IP Address /Domain Name and SPARSH Port in the format "**IP_Address:Port**" in your DHCP Server as per your installation scenario.*

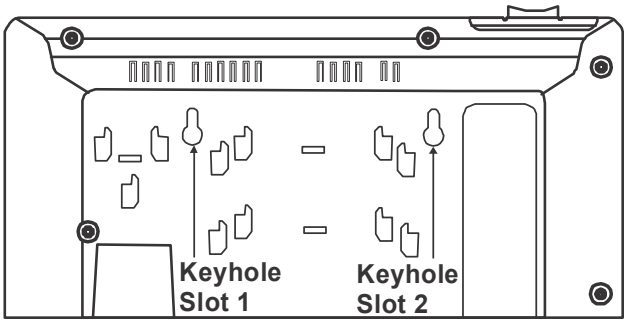
- Log in to *Jeeves*. For instructions, read the topic "Configuring SARVAM UCS".
- Assign an extension number (**SIP ID**) to the Extended IP Phone. For instructions on assigning SIP ID, see "Configuring SIP Extensions".

  For the SIP extension number you assigned to the Extended IP Phone, you must configure the necessary parameters in SARVAM UCS so that Extended IP Phone can register as a SIP Extension. For instructions, see the topic "Configuring SIP Extension Settings as per the Extended Phone Type" under *Configuring SIP Extensions*.

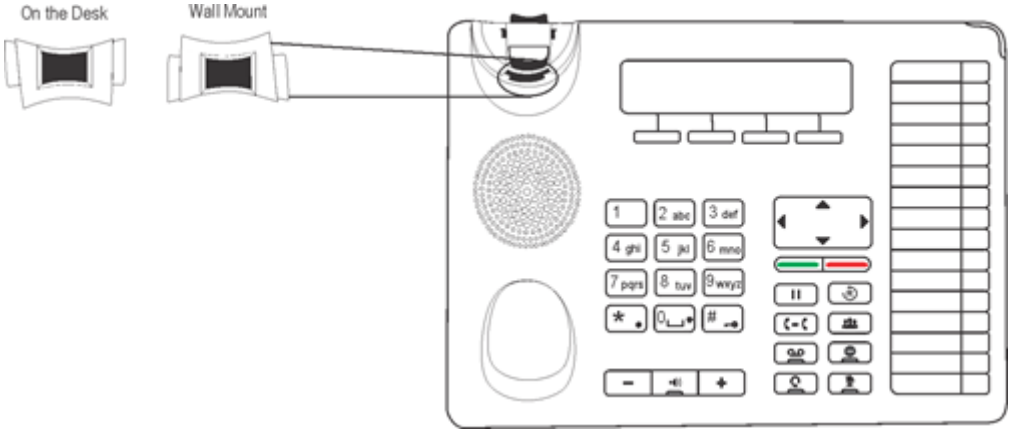
Now, follow the steps described below to install the Extended IP Phone:

1. Unpack the SPARSH VP510 box and verify package contents.
2. You can mount the phone on a wall or on the desk.
3. When you mount SPARSH VP510 on a wall,

- Use the mounting template to drill holed of appropriate size and distance.
- Fix the screw grips in the holes you drilled.
- Fix two screws in the holes on the wall, ensuring that they are aligned with the Keyhole Slots 1 and 2 of SPARSH VP510. The screws should protrude from the wall to fit into the Keyhole Slots.



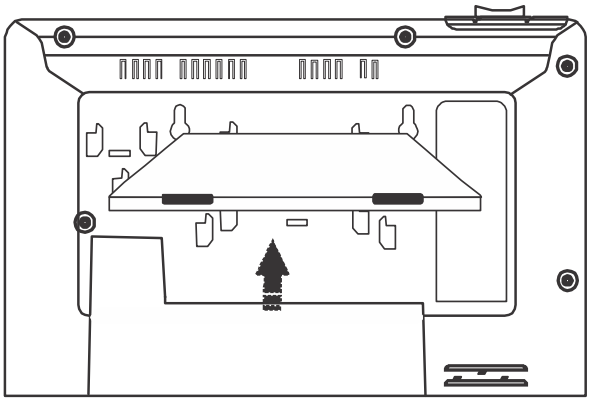
- Now, mount the phone with the screws into the Keyhole Slots.
- Reverse the handset wall mount tab to make sure that the handset remains intact when you mount the phone. Push the handset wall mount tab upwards to remove it from the slot. Rotate it 180 degrees clockwise and push it downwards into the slot.



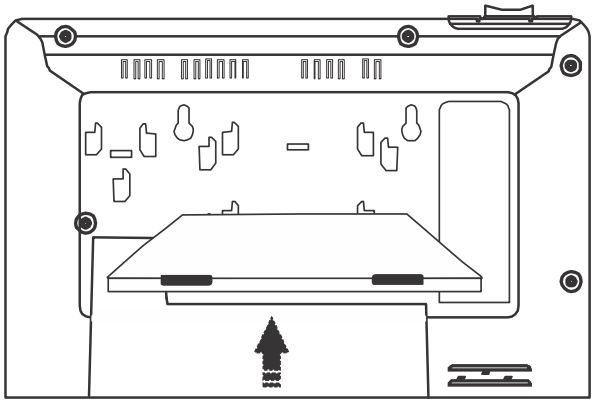
If you are unable to remove the wall mount tab, you may use a tool like a minus screw-driver to remove it.

- When you mount the phone on a desk,

- You can attach the Foot Stand in the following ways — at an angle of 45 degrees or 55 degrees



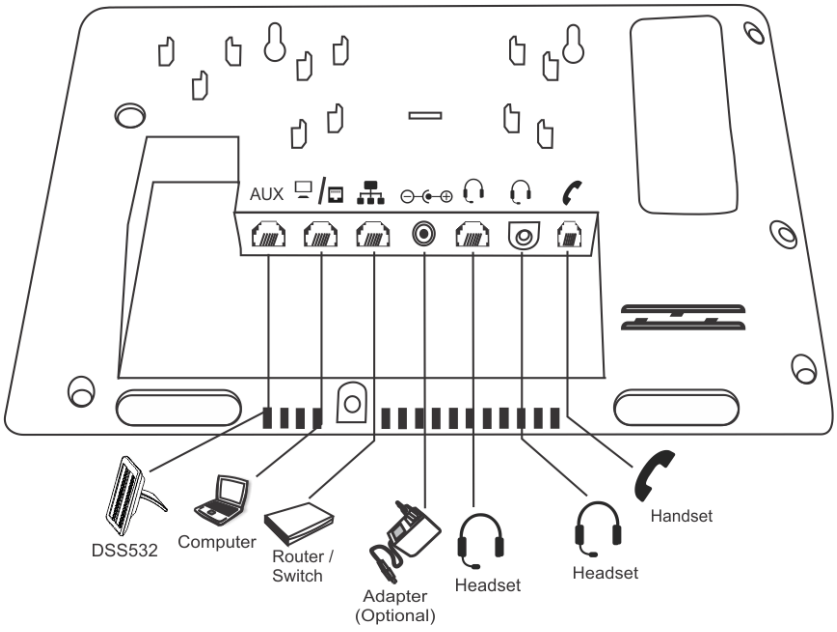
Stand attached at
45 degree angle



Stand attached at
55 degree angle

- Decide which of these positions would work for you best and accordingly attach the Foot Stand.

Refer to the diagram below for connectivity.



4. Connect the Handset.

   - Plug the long straightened end of the Spring Cord into the handset jack at the bottom of the phone, marked with the handset symbol .
   - Plug the other (short straight) end of the Spring Cord into the jack at the bottom of the handset.

5. Connect the Headset (not supplied by Matrix).

   - To use a Headset (not supplied with the phone), plug any standard stereo headset with 3.5mm single connector into the headset audio jack at the bottom of the phone, marked with the symbol .

     OR

     You may also plug in a headset with an RJ9 connector into the headset port at the bottom of the phone, marked with the symbol .

6. Connect to the IP Network.

   - Plug one end of the Ethernet Cable into the LAN Port at the bottom of the phone, marked with the symbol and the other end to the IP Network — A Router or LAN Switch.

7. Connect a PC to the Phone.

   - Plug one end of the Ethernet Cable into the PC Port at the bottom of the phone, marked with the symbol and the other end into the LAN Port of your PC/LAN Switch.

8. Connect DSS532 with the Phone.
   - To connect DSS532 with the phone, plug one end of the RJ11 cable into the AUX Port of the phone and the other end into the IN Port of the DSS532. For installation, see “Installing DSS532 with SPARSH VP510”.

9. Connect the Power Supply.

   - It is a PoE enabled phone and can be powered over Ethernet by connecting it to a PoE enabled LAN Switch (IEEE 802.3af Compliant).

     If you do not want to use PoE, plug in the connector of the Adapter into the power jack (DC Jack) at the bottom of the phone, marked with the symbol . Plug in the Power Adapter into a power outlet.

If both the power options, that is, PoE as well as Power Adapter are available to the phone, then the phone will derive power from the PoE enabled LAN Switch.

The IP phone should be used with Matrix original power adapter (5V/2A) only. The use of any third-party power adapter may cause damage to the phone.

10. Switch ON power supply.

When you power the phone, the boot process will be initiated in the following sequence.

- All keys with LED, including the Speaker key, and the Ringer LED, will glow.
- The LCD display will light up and the booting message appears.
- Then the 'Loading...' message appears on the phone display.
- The phone will start loading the Extended IP Phone Firmware. It will display current firmware being loaded.
- After loading the firmware, the phone will prompt you to change Network settings.
- Wait for a few seconds.

If you want to change the Network Settings or Server Settings, press Yes key. Detailed instructions for changing the Network Settings of the phone are provided at the end of this topic. See "Network Settings" at the end of this topic.

- The phone makes DHCP Discovery and fetches its IP Address and Server Address from the DHCP Server.

  On getting the IP Address and Server Address, the phone initiates Auto Configuration to download the configuration files from SARVAM UCS.

- As the phone downloads the configuration files, the file names will appear one by one.
- On successful download of all configuration files, the phone attempts to register with SARVAM UCS.
- On successful registration, the phone will display the current day, date and time, the extension number and name assigned to the Extended IP Phone.

Network Settings

You can change the network settings of the Extended IP Phone. Press the Down key ⋁ when the phone is in idle state.

To navigate the menu,

- Press the Menu Key when the phone is idle.
- Scroll by pressing the Up/Down Navigation Key to reach the desired Menu option.
- Press the Select / OK Key to select the desired Menu option.
- Scroll by pressing the Up/Down Navigation Key to reach the desired sub-menu option.
- Press the Select / OK Key to select the desired sub-menu option.

To exit menu,

- Press Cancel Key.
  or
  Go ON-Hook.

The cursor is a non-blinking underscore that appears under the first letter of the first option in the menu. To make a selection in the menu, you must move the cursor in the desired direction using the Up, Down, Forward and Back key. When the cursor is at the desired position, press Enter key to make a selection.

Accessing Network Settings

You can access the Network Settings of the Extended IP Phone in any of the following stages:

1. During start-up, when the phone prompts you to change the network settings after loading the firmware.

   You must press **Yes** key and access network settings.

2. When the phone is making Network discovery, downloading configuration files, attempting registration.

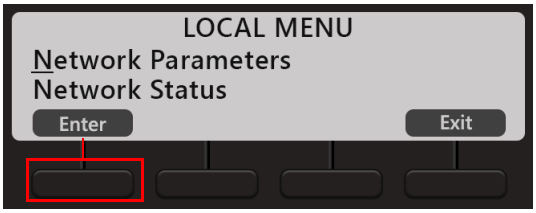
   You must press the **Menu** key to access network settings.

3. When the phone is in idle state, press the Down key ⋁ .

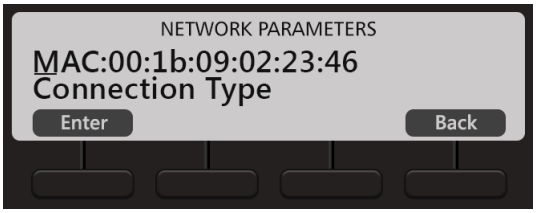
You can configure Network Parameters and view Network status from the Local Menu.

Configuring Network Parameters

- In the Local Menu of the phone, select Network Parameters by pressing the Enter Key.



- The Network Parameters submenu appears.



- Use the Down/Up key to reach the desired network parameter and press Enter key to select. Change the settings as per your requirements.

- Press **Save** key, to save the changes you make.
- You can configure all network parameters described below, except the MAC Address.

- *To enter a dot in the editable fields — IP Address, Subnet Mask, Gateway Address, DNS Address, Server Address — press * (Star) key.*
- *If you want to clear a single digit/character, move the cursor to the right of the digit/character you wish to clear. Now press the Delete key. The digit to the left of the cursor will be deleted.*

Before you start configuring the Network Parameters, get acquainted with following context keys:

Context Keys	Description
Enter/OK	To select a particular parameter
Save	To save the changes
Back	To move a step backwards without saving the changes
Delete	To delete previous characters from the cursor position
2Ab/123	2Ab - Alphanumeric Mode 123 - Numeric Mode

Connection Type

- Select the Connection Type as DHCP, PPPoE or Static according to the IP Addressing scheme of your network.

  If you select DHCP or PPPoE, the phone will be assigned IP Address, Subnet Mask and Gateway Address, DNS Address, Server Address, automatically by the DHCP/PPPoE server.
  For PPPoE Connection Type, you must configure the PPPoE User ID and Password provided by the Internet Service Provider.

  If you select Static, you must assign the IP Address, Subnet Mask and Gateway Address to the phone.

IP Address

- If you select Static as Connection Type, enter the static IP Address to be assigned to the phone.

  Enter the desired Static IP Address by pressing the digit keys.

  To enter the dot/period in the IP Address, press the Star '*' key.

Subnet Mask

- If you select Static as Connection Type, enter the Subnet Mask to be applied on the phone by pressing the digit keys.

  To enter the dot/period in the IP Address, press the Star '*' key.

Gateway Address

- If you select Static as Connection Type, enter the Gateway Address here. This is the IP Address of the LAN Port of the Router.

DNS Server

- If you select Static as Connection Type, select the DNS Server option **Static** and configure the DNS Address.

- If you select DHCP or PPPoE as Connection Type and your Internet Service Provider provides DNS Address, select the DNS Server option **Automatic**. However, if your Internet Service Provider does not provide DNS Address, select **Static** and configure the DNS Address.

DNS Address

- If you select DNS Server as Static, enter the DNS Address here.

  To enter dot/period in the IP Address, press the Star '*' key.

DNS Domain Name

- If you select DNS Server as Static, enter the DNS Domain Name here. DNS Domain Name is optional.

PPPoE User ID

- If you have selected PPPoE as Connection Type, you must enter the User ID provided to you by your Internet Service Provider.

PPPoE Password

- This is the password provided by your Internet Service Provider for the PPPoE User ID. If you have selected PPPoE as Connection Type, you must enter the password provided by your Internet Service provider here.

PPPoE Service Name

- If your Internet Service Provider has provided a Service Name, enter the Service Name here. If your Internet Service Provider has not provided a Service Name, do not configure this parameter.

Primary Server Address

- The system works as the Auto Configuration Server for the phone. Enter the LAN or WAN IP Address/ Domain Name of SARVAM UCS here. Default: blank. The phone sends the request for configuration files to this Server Address.

  If you have selected DHCP as Connection Type, the phone will get the Primary Server Address and Port automatically from the DHCP Server. For this, use ***DHCP option 224*** and ***Data Type*** as '***String***' to provide Server Address and Port from the DHCP Server.
  For PPPoE and Static Connection Types, you need to enter the Server Address.

Primary Server Port

- Enter the SPARSH Port of SARVAM UCS here. The phone sends the request for configuration files to this port.

  Valid range of the port is: 80 or 1025–65535. Default: 80.

Secondary Server Address

- If required, you can also configure the Secondary Sever Address as a fallback option. If the registration with the Primary Server fails the phone will send the registration and configuration requests to the Secondary Server Address. Speech-cut or unclear speech may be observed during on-going mature calls.

Secondary Server Port

- Enter the Secondary Server Port. The phone sends the request for configuration files to this port if the Primary Server fails.

  Valid range of the port is: 80 or 1025–65535. Default: 80.

VLAN Setting

If your phone is connected to a virtual LAN, you need to configure VLAN Settings.

To enable the VLAN switch to correctly route packets generated by the phone and the computers (on the LAN) to each other, the packets must be tagged with a VLAN header.

The VLAN header consists of the VLAN ID (12-bit) and Class of Service (CoS, 3-bit) for prioritization of traffic[50]. The meaning of CoS bits with respect to traffic type is as follows:

CoS	Traffic Type
0	Best Effort
1	Background
2	Spare
3	Excellent Effort
4	Controlled Load
5	Video
6	Voice
7	Network Control

- Select **Phone VLAN/COS** to add VLAN header to the packets generated by the phone, and add VLAN header to the packets relayed from the PC to its LAN port (packets generated by the PC connected to its PC port).

  - To configure Phone VLAN/COS, select **Enable?**. The VLAN ID will be tagged on all packets generated by the phone (SIP, RTP, DNS, ARP, etc.). Default: Disabled.
  - Select **VLAN ID** and enter the VLAN ID that you have assigned to the VLAN in which the IP Phones are connected. Valid range: 0-4094. Default: 1.

  - Select **SIP CoS** and define the CoS (priority) bits in all SIP packets. Valid range: 0-7. Default: 3

  - Select **RTP CoS** and define the CoS (priority) bits in all RTP packets. Valid range: 0-7. Default: 6.

- Select **PC/VLAN CoS** to add VLAN header to all packets entering the PC Port and leaving the LAN port of the phone. Default: Disabled.

  - To configure PC VLAN/COS, select **Enable?**.

50. The IEEE 802.1P standard allows Layer2 switches to prioritize the traffic, thus providing Quality of Service (QoS), that is, better handling of data that pass over a network, thereby resulting in greater reliability and quality. Quality of Service (QoS) on Layer2 is referred to as Class of Service (CoS) which is defined by IEEE 802.1P.

- Select **VLAN ID** and enter the same ID as you have assigned to the VLAN in which the computers are connected. Valid range: 0-4094. Default: 1.

- Select **CoS** and define the Layer 2 CoS (priority) bits. Valid range: 0-7. Default: 0.

PCAP

To capture packets sent and received from and by the phone for monitoring and troubleshooting, you can enable PCAP on the phone. The phone captures up to 1 MB of packets. For more information and instructions on how to use PCAP Trace on the phone, refer to the *EON510_SPARSH VP510 User Guide*.

MAC Cloning

If you require cloning of the MAC Address, configure the following:

- Select **Enable?** and press the Enter key. Select Yes to enable.

- In **Enter Clone MAC Address**, enter the address you wish to clone.

802.1x Authentication

If you want to restrict unauthorized clients from connecting to your LAN, you need to enable 802.1x Authentication. Using EAP MD5 protocol the PC connected to the LAN port of the IP Phone is first authenticated and then it gets connected to LAN.

You need to configure the following 802.1x Authentication parameters:

- Select **Enable?** and press the Enter key. Select Yes to enable.

- Enter the 802.1x Authentication **Identity** provided by you network administrator.

- Enter the 802.1x Authentication **MD5 Password** associated with identity provided by your network administrator.

When you change the Network Settings, the phone will restart.

Viewing Network Status

- When the phone is in idle state. You must press the Down key ⋁ to access the Network Settings.

- Again press Down key ⋁ to select Network Status and press the Enter key.

  Use the Down/Up key to view the status of the various network parameters. The status of the following parameters appear on your display as you scroll.

  - **MAC:** This is the MAC Address of the phone.

  - **IP:** The current IP Address assigned to the phone.

  - **MASK:** The current Subnet mask assigned to the phone.

  - **GW:** The current Gateway IP Address assigned to the phone.

  - **DNS:** The Domain Name Server address assigned to the phone.

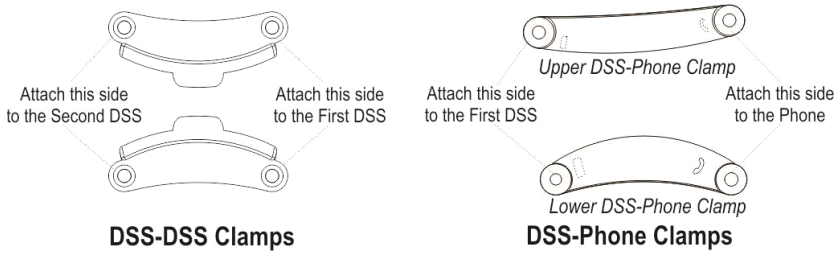
- **Active Server:** This displays the Server that is active — Primary, Secondary — with which the phone is currently registered.
- **S. ADD:** This displays the IP address of the Active Server. It may be the LAN or WAN IP Address / Domain Name of the SARVAM UCS or the Secondary Server IP Address (if configured) or any Fallback Server.
- **S. PORT:** This displays the port of the Active Server. It may be the SPARSH Port of SARVAM UCS or the Secondary Server Port (if configured) or the Fallback Server Port.
- **DOMAIN:** The Domain Name assigned to the phone.
- **802.1x Authentication:** The 802.1x authentication status is displayed—Success, Failure, Authenticating or Disabled.
- **FIRM:** The version of the current Firmware of the phone.
- **UBOOT**: The UBOOT release date.
- **KERNEL**: The KERNEL release date.

Refer to the ***EON510_SPARSH VP510 User Guide*** to know more.

Installing DSS532 with SPARSH VP510

Once you have installed SPARSH VP510 with SARVAM UCS, you can install the DSS532 by following the steps given below:

1. Unpack the box and verify the package contents[51].
2. Four clamps are provided with the phone — 2 DSS-Phone Clamps and 2 DSS-DSS Clamp.

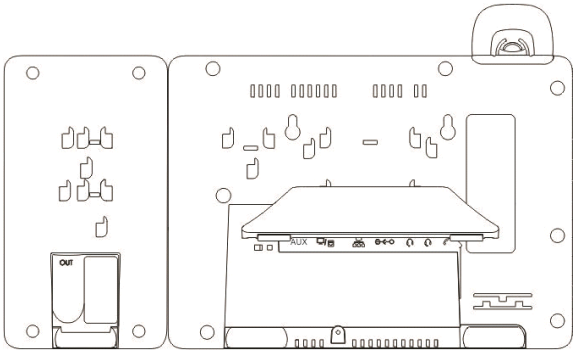


51. *See "Packing List" of Appendix topic.*

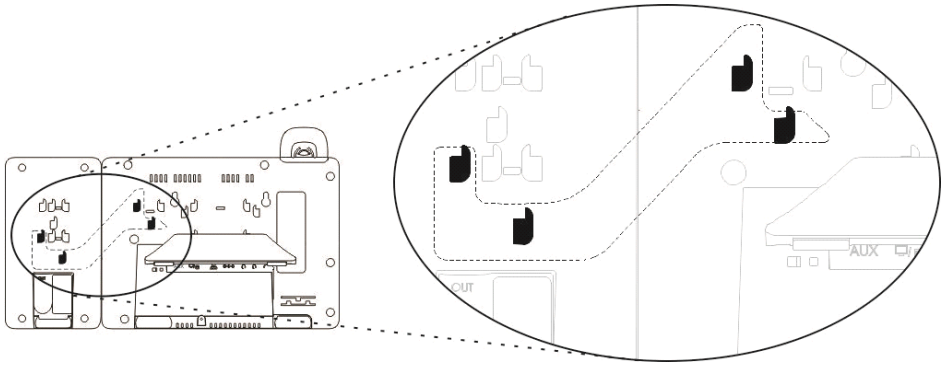
Connecting the First DSS532

Connecting the Extender

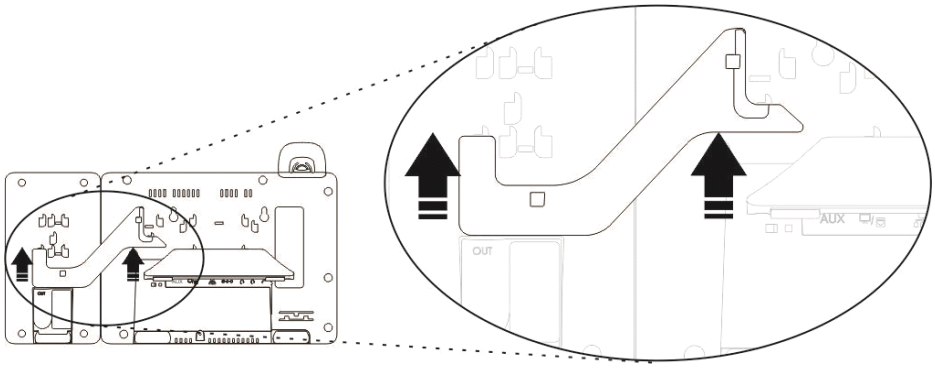
3. Turn the phone upside down on the table and place the inverted DSS532 adjacent to it.



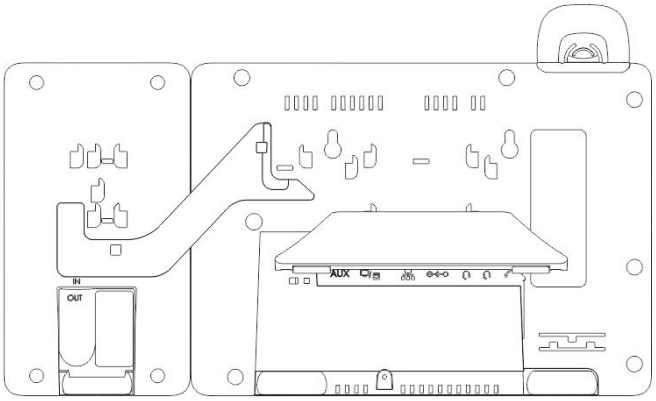
4. To attach the DSS532 with the phone, place the DSS Extender as illustrated below.



5. Insert the hooks on the Extender into the slots provided on the phone and the DSS532.

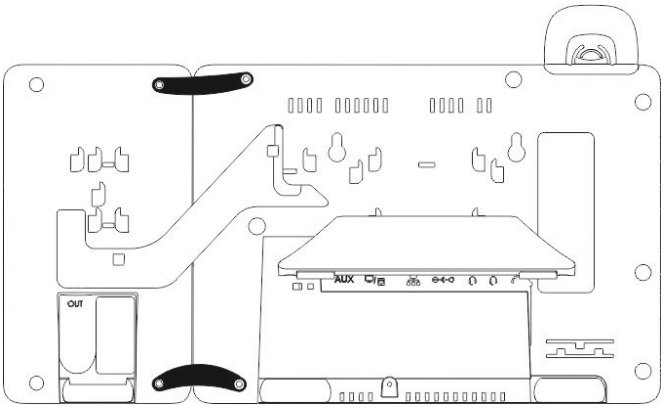


6. Firmly slide the DSS Extender upwards to lock them in place.



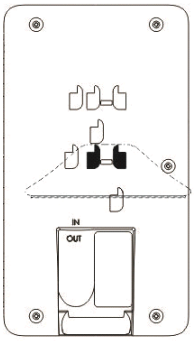
Attaching the Clamps

7. Now attach the clamps. To do so,
   - Remove the screws to attach the clamps.
   - Place the DSS-Phone Clamps between the DSS532 and the phone.
   - Insert the screws back to fix the clamps.

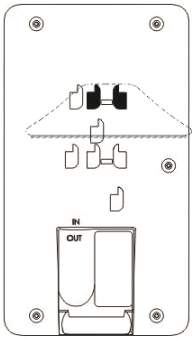


Attaching the Footstand

8. You can mount the DSS532 with the phone on the desk at two angles — **45 degrees** or **55 degrees** by attaching the Foot Stand.



Stand attached at 45 degree angle

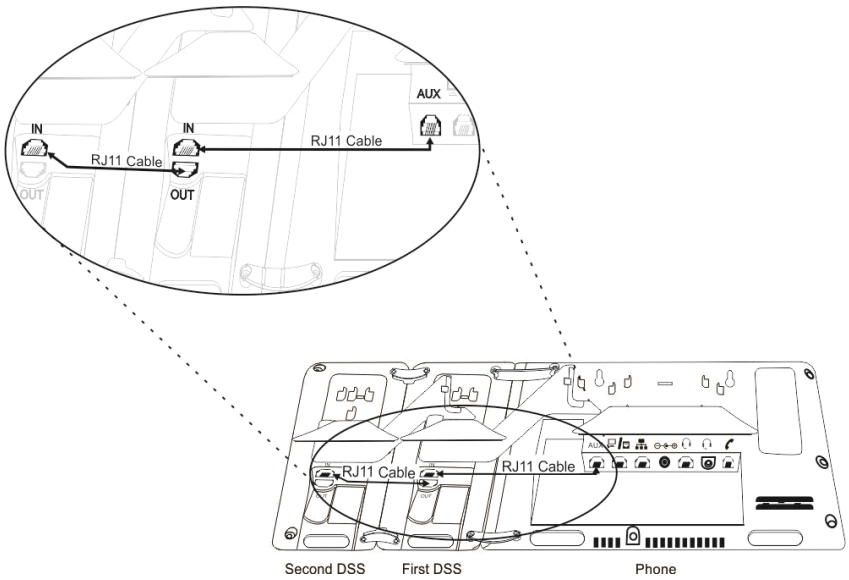


Stand attached at 55 degree angle

Make sure both, the DSS532 and phone are mounted at the same angle.

Connecting the Cables

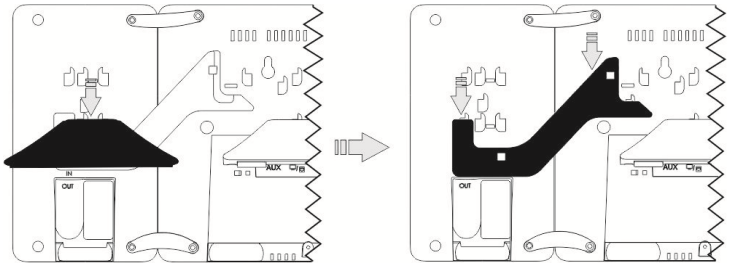
9. To connect the DSS532 with phone, plug one end of RJ11 Cable into **Auxiliary(AUX) Port** of the phone and the other end into the **IN Port** of the DSS532.



Connecting Multiple DSS532

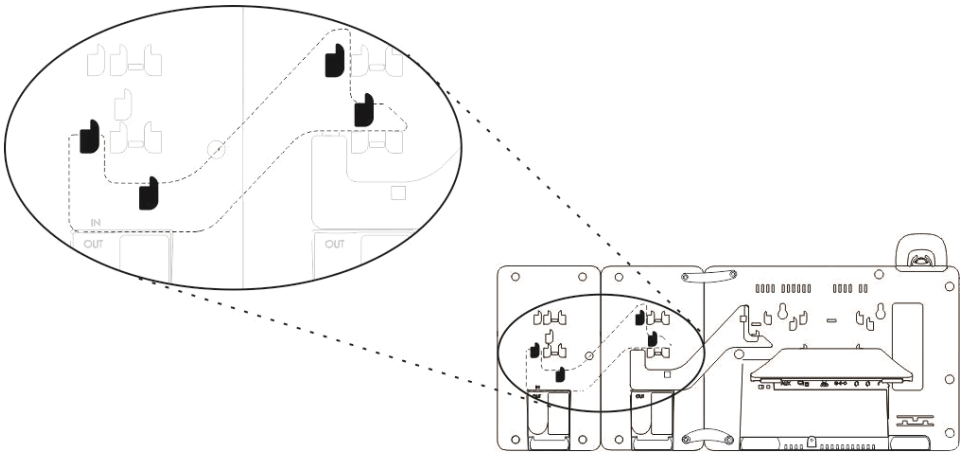
Remove the Foot Stand

10. Remove the Foot Stand of attached DSS532. To do so,
    - Firmly slide the Foot Stand of the attached DSS532 downward to unlock.
    - Now, slide down the attached DSS Extender in downward direction.

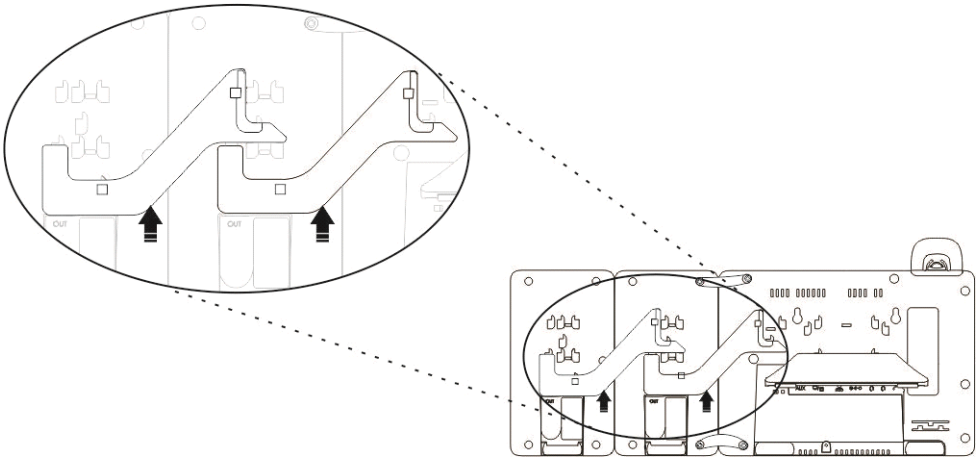


Attach the second DSS Extender

11. To attach the second DSS Extender,
    - Place another inverted DSS532 adjacent to the existing assembly.
    - Place the DSS Extender as illustrated in the diagram below.
    - Insert the hooks on the Extender into the slots provided on both the DSS532.

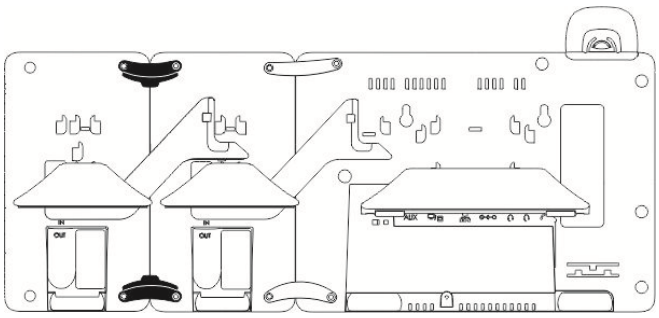


12. Firmly slide both the DSS Extenders upward consecutively (attach the second extender first followed by the existing one attached to the phone) and lock them in place.



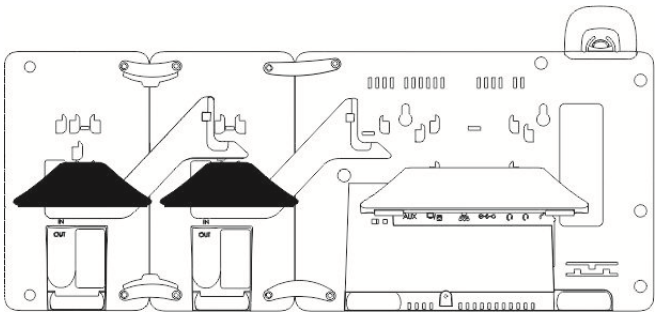
Attach the Clamps

13. Attach the DSS-DSS Clamps between both the DSS532.



Attach the Foot Stand

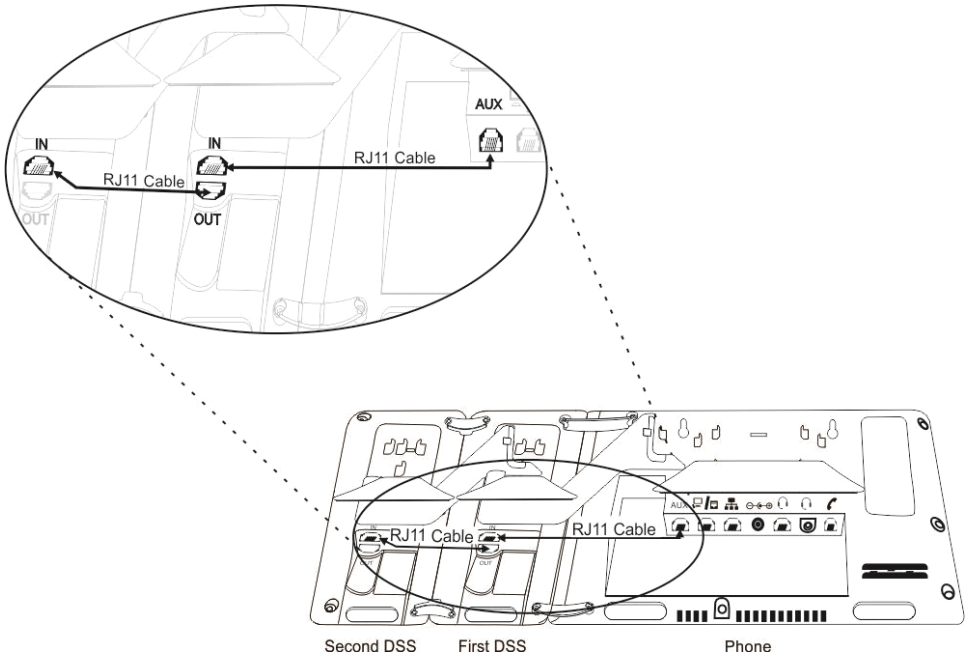
14. Attach the Foot Stand of both the DSS532.



Make sure both, the DSS532 and the phone are mounted at the same angle.

Connect the second DSS532 to the existing assembly

15. Plug one end of the RJ11 Cable into the OUT Port of the existing DSS532 (already connected with the phone) and the other end into the IN Port of the second DSS532.



You can install a maximum of four DSS532 with a phone.

16. After you have connected the DSS532 with the phone, you can configure the DSS Keys. For instructions, see “Programming DSS Console Keys”.

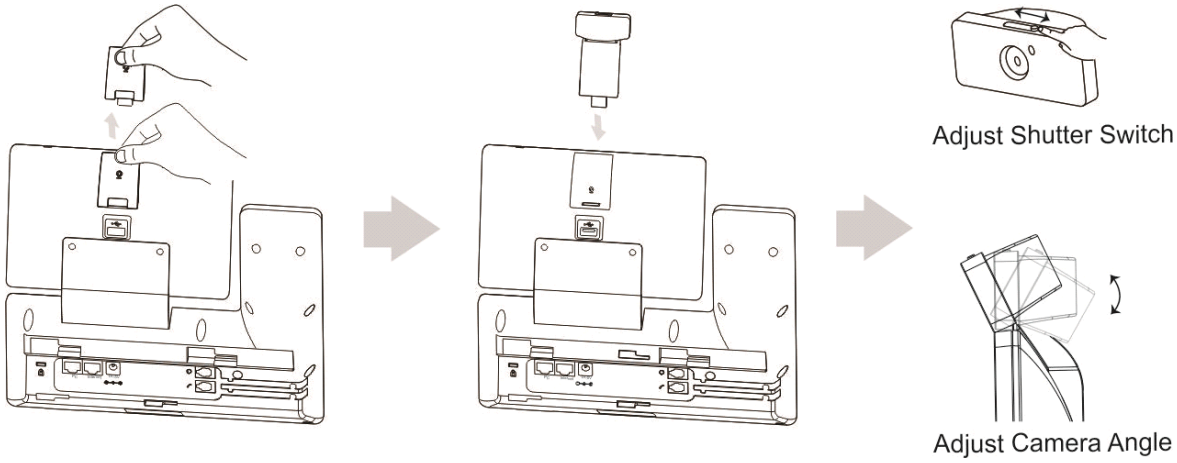
Connecting Extended SPARSH VP710 as Extended SIP Extension

You are recommended to complete the following steps before connecting the Matrix Extended SPARSH VP710 to SARVAM UCS:

- Decide the location where you want to place Matrix Extended SPARSH VP710 within your LAN.
- Log in to *Jeeves*. For instructions, read the topic “Configuring SARVAM UCS”.
- You must configure the necessary parameters in SARVAM UCS so that Extended SPARSH VP710 can register as a SIP Extension. For instructions, see “Configuring Matrix Extended SPARSH VP710”.

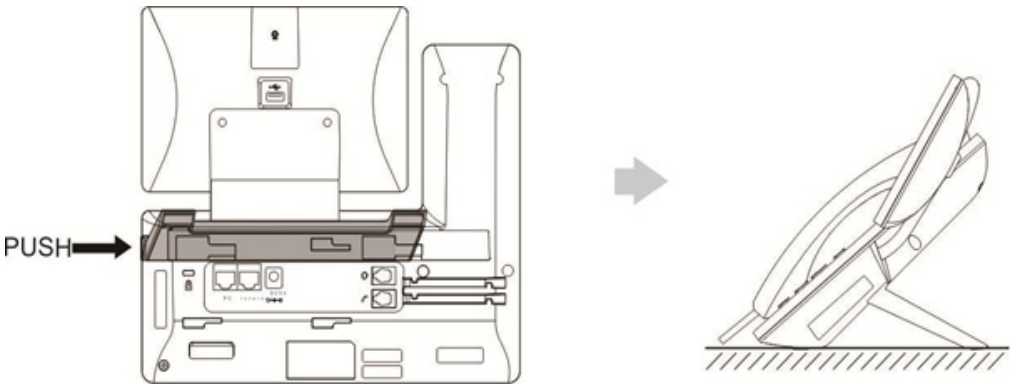
Now, follow the steps described below to install Extended SPARSH VP710.

1. **Inserting the camera**

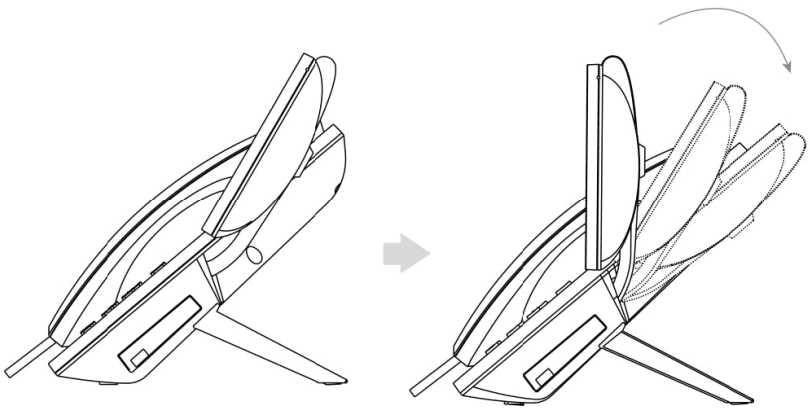


It is recommended to use only the Matrix original Camera, supplied with the IP Phone for video calling. The use of any third-party camera may cause damage to the phone. Damages to the phone caused by using third-party camera is not covered by Matrix warranty.

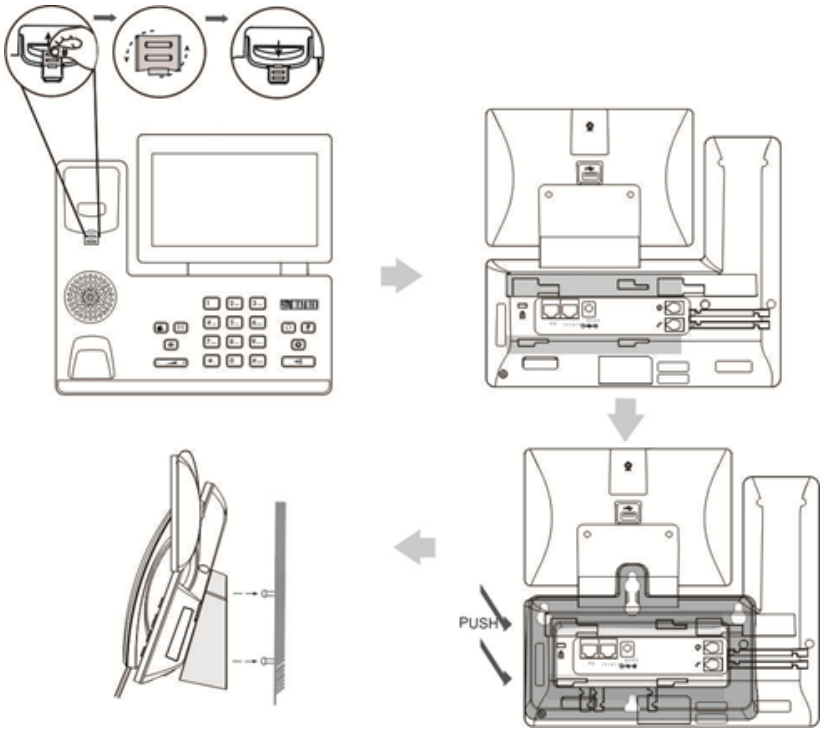
2. **Attaching the stand**



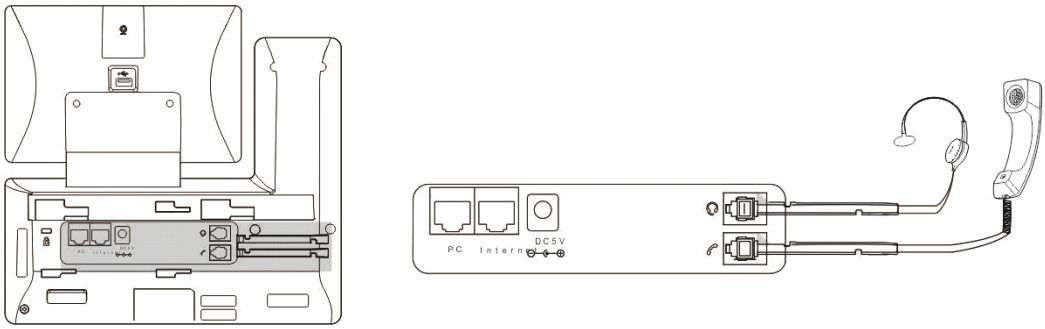
3. **Adjusting the angle of the touch screen**.



4. **Attaching the optional wall mounting bracket**

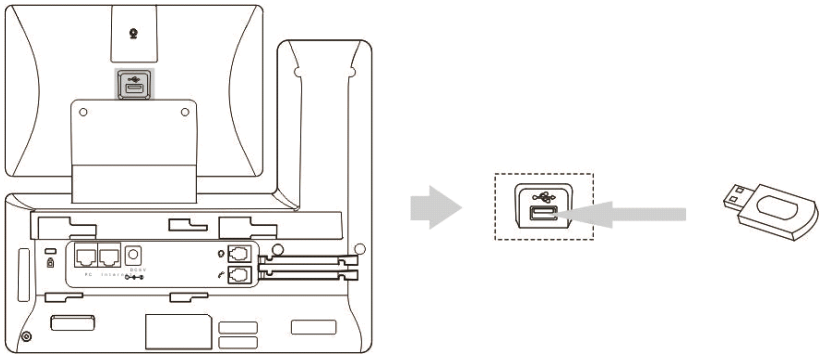


5. **Connect the handset and optional headset.**



A headset is not included in the packaging contents. Contact your dealer/reseller for more information.

6. **Connect the optional USB Flash drive.**



7. **Connect the network and power.**

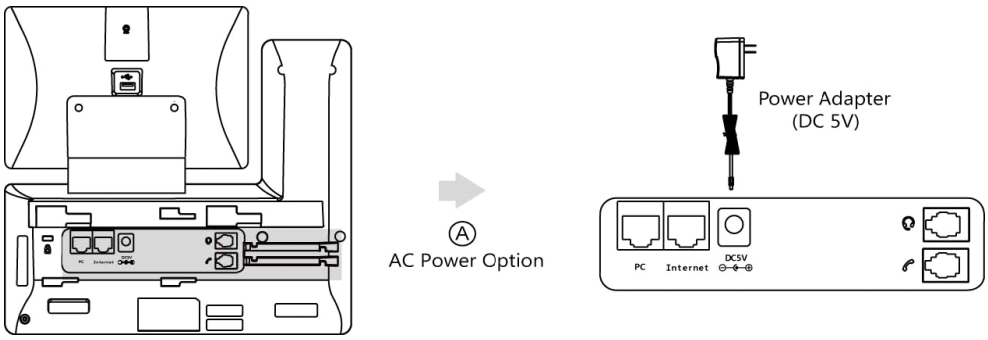
There are two options to connect the power and the network.

- AC power
- Power over Ethernet (PoE)

AC Power

To connect the AC power:

- Connect the DC plug on the power adapter to the DC5V port on the phone and connect the other end of the power adapter into an electrical power outlet.

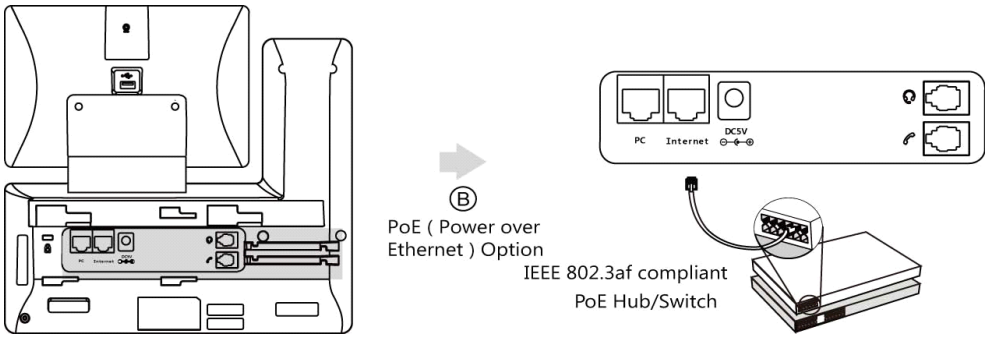


Power over Ethernet (PoE)

With the included or a regular Ethernet cable, the IP Phone can be powered from a PoE-compliant switch or hub.

To connect the PoE:

- Connect the Ethernet cable between the Internet port on the phone and an available port on the in-line power switch/hub.



If in-line power switch/hub is provided, you don't need to connect the phone to the power adapter. Make sure the switch/hub is PoE-compliant.

Do not unplug or remove power while the phone is updating firmware.

After the IP Phone is assembled and connected to the power supply, it automatically begins the initialization process.

During this process, the IP Phone displays the start up screen "Welcome Initializing…please wait".

Once the IP Phone is initialized, it displays two different phone modes:

- Standard SIP
- Extended SIP

- Select Extended SIP, to operate the IP Phone in the extended mode. As soon as you select this mode, the booting process initiates again and the start up screen displays "Welcome Initializing…please wait". After the IP Phone is initialized, it attempts to contact a DHCP Server in your network to obtain valid IPv4 network settings (example: IP address, Subnet Mask, Gateway address, DNS address).You need to configure the basic network parameters of the IP Phone manually, if these are not provided by the DHCP Server or if your network does not support DHCP.

  Refer to the *EXTENDED SPARSH VP710 User Guide*, for detailed instructions:

  - To change the Network Settings of the phone and configure the network parameters.
  - To use Wi-Fi for connectivity and configure its parameters.

- On getting the IP Address and Server Address, the phone initiates Auto Configuration (when DHCP is selected) to download the configuration files from SARVAM UCS.
- On successful download of all configuration files, the phone attempts to register with SARVAM UCS.
- On successful registration, the Home screen appears.

The phone will register successfully, only if the SIP Extension parameters in SARVAM UCS have been correctly configured as per your installation scenario.

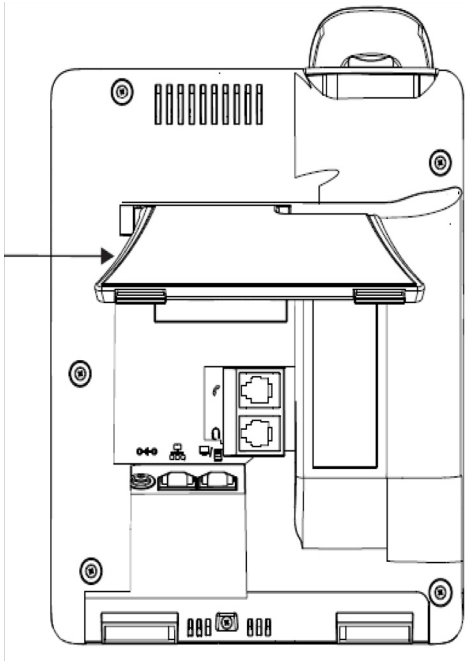
Connecting SPARSH VP210 as Extended SIP Extension

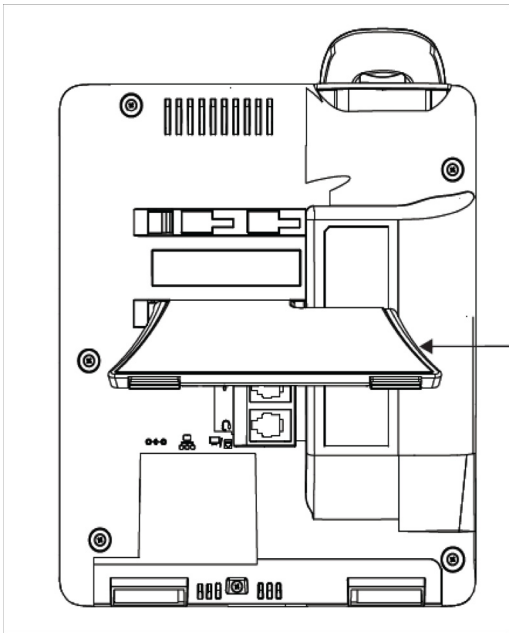
You are recommended to complete the following steps before connecting the Matrix SPARSH VP210 to SARVAM UCS:

- Decide the location where you want to place SPARSH VP210 within your LAN.
- By Default, in SPARSH VP210, the Connection Type selected is DHCP.
- If you want to use the **DHCP Server** for assigning IP Address to the Extended IP Phone, select **DHCP option 224** and **Data Type** as '**String**' and program the LAN or WAN IP Address /Domain Name of SARVAM UCS and SPARSH Port in the format **"IP_Address:Port"** in your LAN DHCP Server as per your installation scenario.
- Log in to *Jeeves*. For instructions, read the topic "Configuring SARVAM UCS".
- You must configure the necessary parameters in SARVAM UCS so that SPARSH VP210 can register as a SIP Extension. For instructions, see "Configuring Matrix SPARSH VP210".

Now, follow the steps described below to install SPARSH VP210.

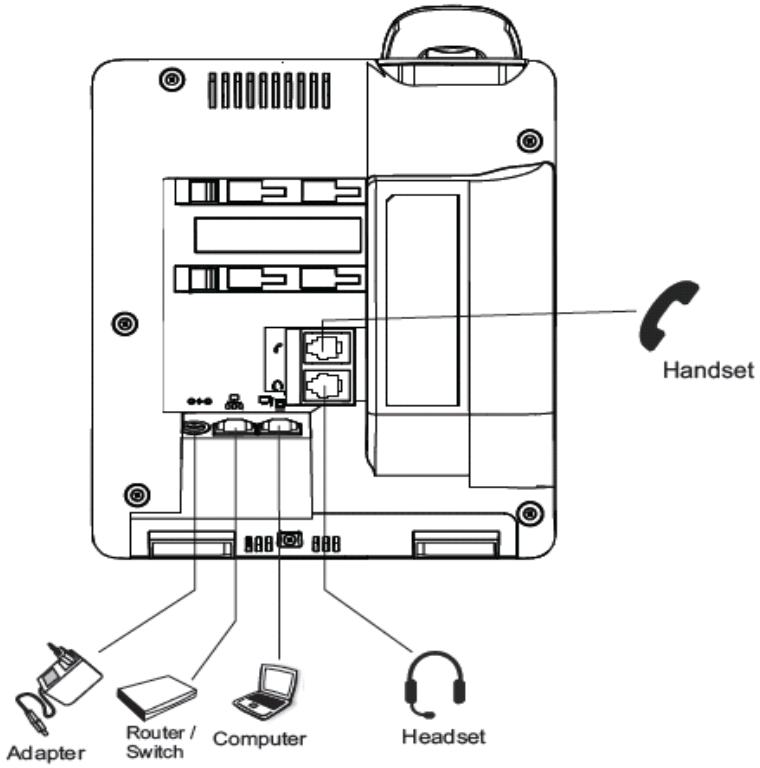
1. Unpack the SPARSH VP210 box and verify package contents.
2. When you mount the phone on a desk, you can attach the Foot Stand in two ways at **45° Angle** or at **55° Angle**.





- Decide which of these positions would work for you best and accordingly attach the Foot Stand.

  Refer to the diagram below for connectivity.



3. Connect the Handset to the Phone body.

- Plug the long straightened end of the Spring Cord into the handset jack at the bottom of the phone, marked with the handset symbol .
- Plug the other (short straight) end of the Spring Cord into the jack at the bottom of the handset.

4. If you want to use a Headset (not supplied) with your phone, You may plug in a headset with an RJ9 connector into the headset port at the bottom of the phone, marked with the symbol .
5. To connect the LAN, Port , plug one end of the Ethernet Cable into the LAN Port at the bottom of the phone marked with the symbol and the other end to the IP Network — A Router or LAN Switch.
6. To connect your phone to a computer on your desk, plug one end of the Ethernet Cable (not supplied with this phone) into the PC Port at the bottom of the phone, marked with the symbol and the other end into the LAN Port of your PC/LAN Switch.
7. It is a PoE enabled phone and can be powered over Ethernet by connecting it to a PoE enabled LAN Switch (IEEE 802.3af Compliant). In this case you need not connect the Power Adapter.

   If you do not want to use PoE, plug in the connector of the Adapter into the power jack (DC Jack) at the bottom of the phone, marked with the symbol . Plug in the Power Adapter into a power outlet.

If both the power options, that is, PoE as well as Power Adapter are available to the phone, then the phone will derive power from the PoE enabled LAN Switch.

The IP phone should be used with Matrix original power adapter (5V/0.6A) only. The use of any third-party power adapter may cause damage to the phone.

8. Switch ON power supply.

   When you power the phone, the boot process will be initiated in the following sequence.

   - The LCD display will light up and booting message appears.
   - While loading the application then the loading message appears on the phone display.
   - The phone makes DHCP Discovery and fetches its IP Address and Server Address from the DHCP Server.

If you want to change the Server Settings, press Settings.

Refer to the SPARSH VP210 (Extended) User Guide, for detailed instructions, to change the Network Settings of the phone and configure the network parameters.

- On getting the IP Address and Server Address, the phone initiates Auto Configuration to download the configuration files from SARVAM UCS.
- On successful download of all configuration files, the phone attempts to register with SARVAM UCS.
- On successful registration, the Home screen appears.

The phone will register successfully, only if the SIP Extension parameters in SARVAM UCS have been correctly configured as per your installation scenario.

Refer to the ***SPARSH VP210 (Extended) User Guide*** to know more.

Starting Up ETERNITY LENX

Power ON

- If you have completed all the installation tasks, switch on power supply. Keep the MCB Switch ON of your PS48V card installed in the system and power the FCBC.
- Observe the Reset Cycle.

Reset Cycle

- Reset Cycle (Power-ON Self Test) takes about 2 minutes to finish.
- All the LEDs of the system, the cards and the keys of the DKP/SIP devices attached to the System are turned on.

Interpreting LEDs

The functioning of the LEDs of the system and the various cards and their meaning are summarized at the end of the installation instructions for each Card Type.

Refer to the LED Patterns described for each Card Type to verify if the system is operating properly and locate faults, where they occur.

When the reset cycle is successful, the default Extension "Access Codes" loaded by the system and the date and time of the "Real Time Clock (RTC)" of the system will appear on the LCD display of the DKPs/ IP Phones you have connected with the system.

CHAPTER 6

Installing ETERNITY MENX

- *The Matrix ETERNITY MENX is to be installed by persons trained and experienced in telecom wiring.*
- *The person installing the ETERNITY MENX must be familiar with trunks, physical wiring of the MDF on both the exchange (System) side and the line side (CO).*
- *When installing any equipment, make sure that you take all the necessary precautions for handling electronic and electrical appliances. Follow proper procedures for static electricity, while handling the system and its cards to prevent damage to the system and harm to yourself.*
- *Use a grounding mat and wear an anti-static strap/belt. Read the do's and don'ts listed in "Protecting the System and Yourself".*
- *If you have complied with the requirements and instructions described in "Before You Start", you may now begin the installation of your ETERNITY MENX.*

Hardware Version V1R2 and earlier

ETERNITY MENX can be wall mounted, rack mounted or placed on a table.

ETERNITY MENX has a total of 16 Universal slots.

The Matrix ETERNITY MENX is shipped factory fitted with the Power Supply Card and the CPU Card in their respective fixed slots (refer the section "Know Your SARVAM UCS"). VoIP and VMS are in-skin to the CPU Card. Hence, separate VMS and VoIP Cards are not required.

ETERNITY LENX has a total of 16 Universal slots. The cards - BRI, T1E1PRI, GSM/CDMA, DKP, CO, SLT, E&M, Magneto - are shipped separately as per the order placed by individual customers. These cards can be installed in any of the Universal slots.

If you upgrade the system firmware to V1R3 and later, the Expansion Slots license will be applicable for the universal slots. No universal slots will be functional by default. You must purchase the SARVAM EXP4 ENT license to activate the universal slots as required.

For details, see "Expansion Slots" under "License Management".

Hardware Version V1R3 and later

ETERNITY MENX can be wall mounted, rack mounted or placed on a table.

The Matrix ETERNITY MENX is shipped factory fitted with the Power Supply Card and the CPU Card in their respective fixed slots (refer the section “Know Your SARVAM UCS”). VoIP and VMS are in-skin to the CPU Card. Hence, separate VMS and VoIP Cards are not required.

ETERNITY MENX has a total of 16 Universal slots. The cards - BRI, T1E1PRI, GSM/CDMA, DKP, CO, SLT, E&M, Magneto - are shipped separately as per the order placed by individual customers. These cards can be installed in any of the Universal slots.

If you have upgraded the system firmware to V1R3 in the old ETERNITY MENX system, the Expansion Slots license will be applicable for the universal slots. No universal slots will be functional by default. You must purchase the SARVAM EXP4 ENT license to activate the universal slots as required.

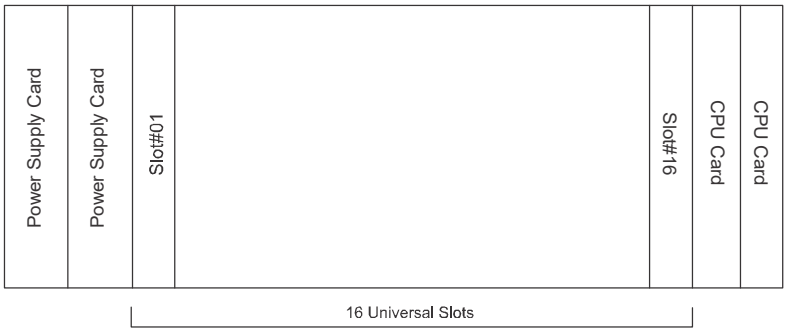
If you have purchased the new ETERNITY MENX system with the firmware V1R3 and later, the Expansion Slots license will be applicable for the universal slots. The first eight universal slots after the power supply card will be functional by default. If you require more functional universal slots, you must purchase the SARVAM EXP4 ENT license.

Each SARVAM EXP4 ENT license will provide the activation for next four universal slots in sequence.

For details, see “Expansion Slots” under “License Management”.

Illustrated below is the position of the fixed and universal slots in ETERNITY MENX.

ETERNITY MENX



The two extreme left slots are reserved for the Power Supply Cards and the two extreme right slots are reserved for the CPU Cards.

Follow the installation instructions for Cards described here also when you expand the system (add more Cards) or remove or swap Cards for maintenance and repair.

1. Unpack the box. Check the package contents (see “Packing List”). Contact your Dealer/Distributor if any of the items is missing, faulty or damaged. Do not discard the packaging material.

Mounting the System

2. Decide where to mount the ETERNITY MENX - on a table or wall - taking into consideration the mechanical dimensions and the weight. If mounting the system on a wall, you may refer the mechanical dimensions and the Mounting Template for drilling holes at appropriate places on the wall. Make sure the system orientation is horizontal.

3. When installing the system in a rack, allow adequate space between the system and other units for air circulation. Make sure the system orientation is horizontal.

4. Mount the system at the selected site. Make sure that the system is placed such that you have full access to the front and back panels. The holes in the panels are provided for ventilation; Make sure that these are not blocked, to prevent overheating.

Connecting Input Power Supply

5. Ensure that a proper electrical earth and telecom earth are in place.

6. Check the voltage at the power point from where the supply is to be given to the system. It should be as per the specifications. Earth the system properly. (Refer “How to Make the Telecom Earth”).

Inserting Cards

7. Make sure that the ETERNITY MENX power is off and the power cord is unplugged.

8. Select a free slot from the universal slots.

9. Unscrew and remove the filler bracket that covers the card-slot opening of the slot you intend to use.

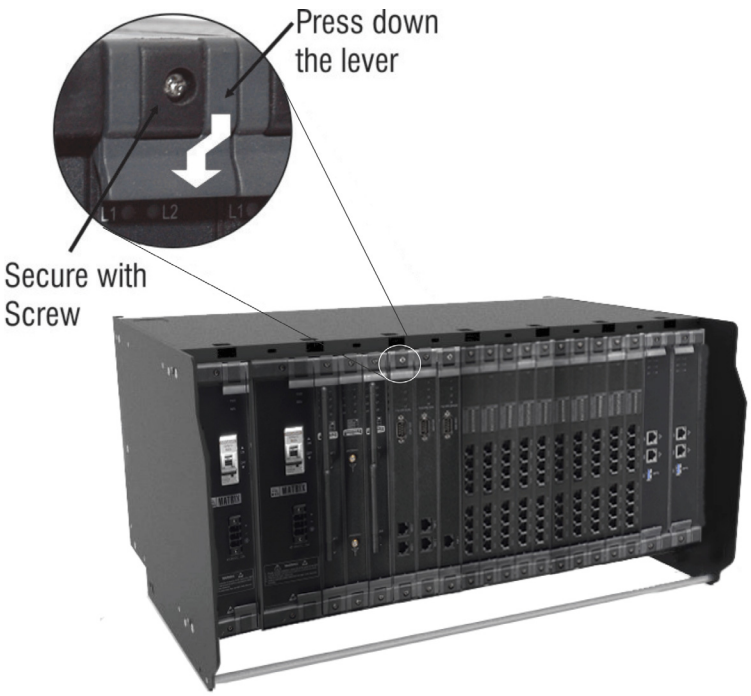
10. Hold the card with the connectors facing you. Do not grab the card from both ends.

11. Slide the card into the slot, along the guide rails provided for each slot at the top and bottom planes.

12. Ensure that the cards are inserted deep enough for all the connector pins on the cards make complete contact with those of the motherboard on the backplane.

Do not force the card into the slot. Doing so can damage the card or the slot connector.

13. When the card is firmly seated in the connector, push down the levers on the card mounting bracket and secure the card with the screw provided.



14. Tighten the screws on either side of the bracket.

15. Following the above steps, install each card into the universal slots.

    Detailed installing instructions are provided for each card - DKP, SLT, CO, ISDN BRI, ISDN T1E1PRI, GSM/CDMA, E&M, E1FO, Magneto, Radio - later in this section. Refer to them when installing each card type.

16. To remove a card:

    - Switch off power supply, unplug the power cord.
    - Disconnect any cables connected to the card.
    - Remove the screws from the card-mounting bracket.
    - Lift the levers on the mounting bracket to release the card.
    - The card will emerge out of the slot.
    - Grasp the card by its mounting bracket, and ease it out of its slot.

    - *If you are removing the card permanently or for a certain period of time, install a filler bracket over the empty card opening in the chassis.*

    - *Installing filler brackets over empty card-slot openings is necessary to protect the system from dust, dirt, insects and damage.*

17. Using the cables supplied with each card, and terminate the cables in the Main Distribution Frame (SLT, DKP, CO, and E&M lines), the NT1 device (ISDN BRI lines), ISDN Modem (ISDN PRI Lines), as applicable.

    Lead the cables neatly and tangle-free into the MDF.

18. After you have completed inserting and connecting the cards, power ON the system and observe the Reset cycle and the LED pattern of each card, where applicable.

The Power Supply Card

Two types of Power Supply Cards are supported by the Matrix ETERNITY MENX: PS UNI Card and PS48V Card.

- **PS UNI Card** with 100-240VAC, 47-63Hz Mains as Input AC Voltage Power Supply.

  This card is designed on the SMPS scheme. As this card does not have any provision for battery backup, it is recommended that a UPS be connected to keep the system powered during outages.

  This card has four LEDs, a Mains Switch, and a Socket assembly for connecting the mains cord.

- **PS48V Card** with 48VDC as Input DC Power Supply Voltage. A Float cum Boost Charger (FCBC) is required to feed 48VDC power to the card. The FCBC works on input AC mains.

  This card is available in one variant - DC to DC 500 W.

  The card has four LEDs, an MCB Switch, a power ON/OFF Switch, and a 3-way termination block for connecting the power cord.

  Both, the PS UNI card and the PS48V Card provide DC output voltages as: +3.5V, +5.0V, -27V and -85V. These are indicated by LEDs.

- *Redundancy option is provided for the Power Supply Card and the CPU Card.*
- The ETERNITY MENX supports two power supply cards. Whenever there is a fault in one, the other takes over the control, providing uninterrupted communication.
- It also support 'Hot Swap' for the Power Supply Card.
- The maximum number of ports supported by the GSM, SLT and DKP Cards may vary according to the type of Power Supply used. Refer the following table for maximum ports supported with Universal Power (PSUNI) and DC Power Supply.

ETERNITY MENX - Universal Power Supply (PSUNI)

Type of Port	Maximum Number of Ports Supported
GSM	32 ports in Talk mode
SLT	128 ports in Talk mode for short loop 10 extensions in Talk mode for long loop 472 extensions permanently connected
DKP	96 ports maximum

Analog SLT ports supported for Short Loop with Loop Current programmed

Loop Current Programmed	25mA	30mA	35mA	40mA
Number of ports supported in talk mode (OFF-Hook short loop) according to loop current programmed	200	172	150	128

ETERNITY MENX DC to DC 500W Power Supply

Type of Port	Maximum Number of Ports Supported
GSM	112 ports in Talk mode
SLT	256 ports in Talk mode for short loop 20 extensions Talk mode for long loop 512 extensions permanently connected
DKP	128 ports maximum

Installing the Power Supply Card

The Power Supply Card is located in a fixed slot. No other card can be inserted in this slot.

The Power Supply Card is delivered factory fitted, when you buy the system. However, if you want to remove the card for the purpose of maintenance or replace it with a new one, please follow the instructions below:

1. Unpack the Power Supply Card and verify the package contents.

   If already installed, switch OFF power supply, unplug the power cord. Remove the screws securing the card. Lift the levers on the mounting bracket to release the card. As the card emerges from the slot, ease it out of the slot.

2. Insert the Power Supply card into the guide rails of the first slot on the extreme left, designated for the Power Supply Card. Make sure that the card is inserted deep enough to make perfect contact with the connectors on the motherboard at the backplane.

3. Now, press down the levers on the card mounting bracket to secure the card in its slot.

4. Secure the card in the slot by screwing the bracket on both ends.

To install a second card for redundancy insert the second card on the next slot.

5. If installing the **PSUNI card**, connect the three-pin power cord into the socket of the PS UNI card and plug in the cord into the mains supply.

   You may connect the PSUNI Card to a UPS to keep the system live during power outages.

   Select a UPS considering the typical power consumption of ETERNITY MENX presented in the table below:

Model	Power Consumption (Typical)
ETERNITY MENX16S	100 watts

6. If installing the **PS48V card**, connect the Float cum Boost Charger (FCBC). Terminate the power cord from the FCBC output into the 3-way termination block on the PS48V card.

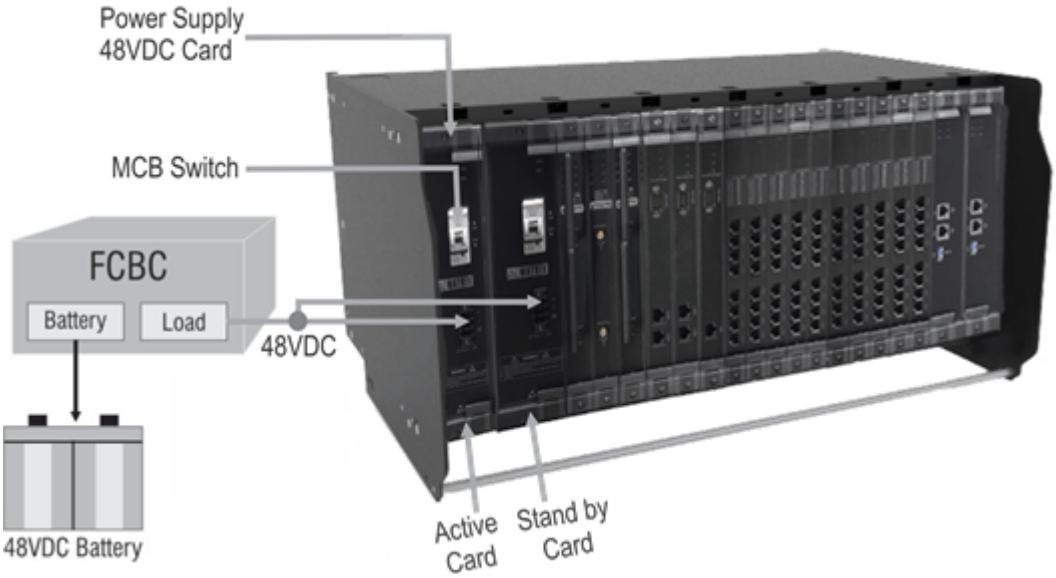
   Polarity is critical. Ensure that the wires are connected with the correct polarity. Follow the standard color codes used by FCBC manufacturers:

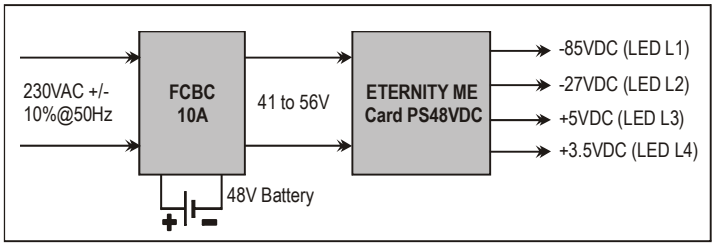
Color	Signal
Red	+48VDC

Color	Signal
Black	GND
Green	Earth

It is recommended that you measure the voltage before connecting the power cable to the power supply card. Ensure that the earth is connected.

Connecting FCBC to the ETERNITY MENX





If two PS48V cards are installed for redundancy, each must be connected to a separate FCBC and each FCBC must be connected to a separate source of power supply.

7. Connect Battery back up to the FCBC[52].

52. *When the batteries are drained, the FCBC goes into the boost mode and begins to charge the batteries at higher current. When the batteries reach a preset voltage level (typically set to 56.0 volts), the FCBC goes to float mode. In the float mode the FCBC keeps charging the battery but at lower current. The FCBC monitors the voltage level of the batteries. As soon as the battery voltage goes below preset voltage (typically set to 50.4 volts), FCBC goes from float mode to boost mode. The change over from mains to battery and vice-versa is automatic. The advantage of using an FCBC is that batteries get charged faster, since the batteries are charged with higher current initially.*

Battery backup time depends upon the total load. The total load is the sum of system's load and load of active extensions. The power consumed by the variants of ETERNITY MENX is given in the table below:

Model	Power Consumption (Typical)
ETERNITY MENX16S	100 watts

The Battery back up time depends on the 'Ah' rating of the battery connected to the FCBC. The FCBC uses the constant voltage charging method. So, the batteries get charged faster if less power is consumed by the system when in mains mode.

8. Switch on power supply, after completing all other installation.

The CPU Card

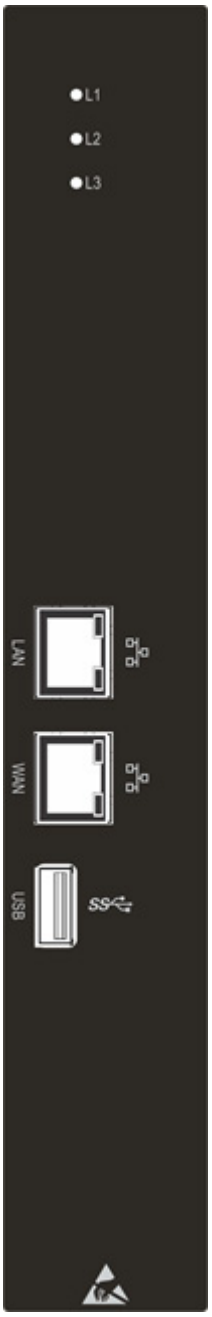
The ETERNITY MENX-LENX CPU Card hosts the SARVAM UCS Application. It supports four VOCODER modules and one VMS module. Both the modules — NX DBM VOCODER64 and NX DBM VMS64 are optional and can be purchased separately.

This card hosts Communication Manager, Feature Server, VoIP Server, VMS Server and other important servers and modules which controls all other slave cards (SLT, DKP, CO+SLT, DKP+SLT, E&M, BRI, T1E1, E1FO, GSM etc.). All the configuration and programming information is stored on this card.

ETERNITY MENX supports redundancy along with Hot Swap for the CPU Card. An additional CPU Card can be installed for redundancy. If the main CPU Card fails, the other card takes over, providing uninterrupted communication. To know more, refer to “Redundancy”.

The CPU Card occupies fixed slots, the first two slots from the extreme right of the first Rack, with a unique arrangement of connectors. So no other card can be inserted in the slots of the CPU Card.

The CPU Card has a WAN Port, LAN Port and USB Port on the front panel. It also has an Internal USB Port with a factory fitted pendrive.



Ports and Connectors:

Port	Connector	Description
LAN	RJ45	Used for connecting the Ethernet cable into LAN Port to connect to a PC or a LAN Switch.
WAN	RJ45	Used for connecting the Ethernet cable into WAN Port to connect to a Broadband Router/Modem.

Port	Connector	Description
USB	USB to COM Converter (Optional)	The External USB can be used as COM Port by connecting the USB to COM Converter. The USB to COM Port can be used to: • set up and run software applications — PMS and CAS. • capture System Activity Log, System Fault log and Hotel Motel Activity logs. • generate SMDR reports.

If you buy a spare CPU Card separately, the default pendrive will not be provided along with it.

LAN Interface

The LAN Port is provided to connect:

- the system to a PC or a LAN. This port is used for operating the web-based programming software Jeeves.
- the CPU Card to the Local Area Network to register SIP extensions through the LAN Port.
- set up and run software applications such as PMS and CAS on any PC on the LAN.
- generate Station Message Detail Record (SMDR) Reports on any PC on the LAN.
- capture "System Activity Log", "System Fault Log" and Hotel Motel Activity Log.

WAN Interface

The WAN Port is provided to connect:

- a LAN Switch/Hub/Router/Modem.
- the CPU Card to the public network over a Router/Modem. Any user on the public network can be registered as SIP Extension through the WAN Port.
- set up and run software applications such as PMS and CAS on any PC on the LAN.
- generate Station Message Detail Record (SMDR) Reports on any PC on the LAN.
- capture "System Activity Log", "System Fault Log" and Hotel Motel Activity Log.

VoIP Interface

The CPU Card supports four NX DBM VOCODER64 modules. You must purchase the module separately for VoIP functionality.

VOCODER Channels

The system supports four NX DBM VOCODER64 Modules. Each module supports 64 VOCODER Channels[53]. You must purchase the modules separately. The system provides 4 pre-activated VOCODER channels by default which can be used after installing NX DBM VOCODER64 module. If you require more channels, you can purchase the licenses accordingly. Matrix provides two licenses — SARVAM VOCODER CHNL4 and SARVAM VOCODER CHNL16.

If you require more than 64 VOCODER channels, you can install another NX DBM VOCODER64 Module.

53. The number of VOCODER channels that will be supported would be as per the license you purchase.

A call made from a SIP Extension or SIP Trunk to another SIP Extension or SIP Trunk will consume two VOCODER channels, whereas a call made from a SLT or DKP extension to a SIP Extension or SIP Trunk will consume one VOCODER channel. Thus, the number of speech paths available to make simultaneous calls will depend not only on the number of VOCODER channels, but also on the number of channels consumed by such SIP-to-SIP and Analog/Digital extension to SIP Trunk/SIP Extension calls.

VMS Interface

The system supports a full-fledged, 'in-skin' Voice Mail System module to provide mailbox facility to all its extensions users. The Voice Mail System also forms the basis of other features like Conversation Recording and Call Taping.

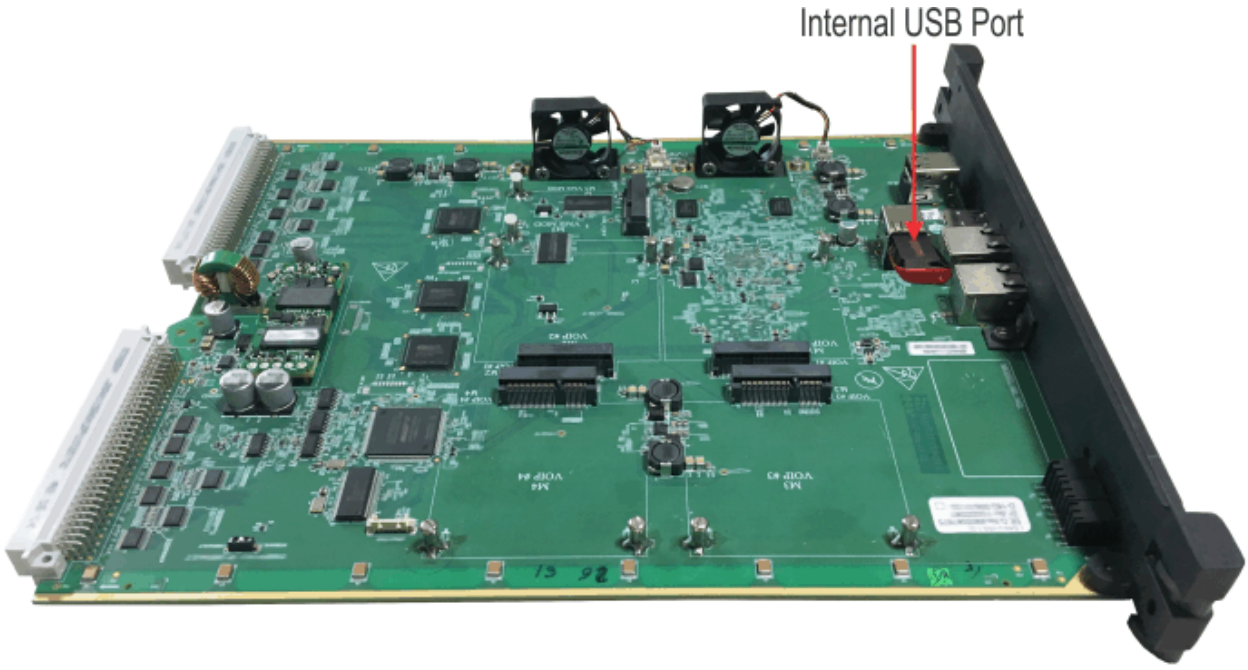
Each Mailbox has the capacity of storing 15,000 voice messages. The maximum size of each Mailbox is 60,000 minutes. By default, the size of each Mailbox is set to 5 minutes. The maximum Message Length for each Mailbox is 9,999 seconds. By default, the Maximum Message Length for each Mailbox is set to 15 seconds.
The NX DBM VMS64 Module is an optional module. It must be purchased separately. The factory fitted Pen Drive provided also contains the VMS configuration files, and voice messages for prompts and greetings along with the SARVAM UCS ENT Application. The Pen Drive is also the storage device for mailbox messages.

If required, you may use a Pen Drive of upto 64GB by replacing the factory fitted pendrive with a new one.

The system supports a maximum of 64 channels out of which 4 channels are provided by default. If you require more channels, you can purchase the licenses accordingly. Matrix provides two licenses — SARVAM VMS CHNL4 and SARVAM VMS CHNL16.

Internal USB Port

The CPU Card has an Internal USB Port with a pendrive inserted into it.



The pendrive supports FAT32 file format. It contains the SARVAM UCS Application, VMS greetings, messages, Matrix Extended IP phone firmwares and SMS Server firmware.

Do not remove the pendrive.

When you select the SARVAM UCS ENT Application, the system fetches the application from the pendrive.

External USB Port (Device Port) 3.0

The CPU Card has an External USB Port on the fascia. This can be used as a COM Port by connecting the USB to COM Converter.

The USB to COM Converter will not be provided by Matrix.

The following USB to COM Converters are supported:

- Prolific PL2303 by BAFO
- CH341 by Winchiphead

If you use any other USB to COM Converter, it may not function properly.

The USB to COM Port has a DB-9 connector.

The port allows you to connect a PC to the system, so that you can install and operate the following features:

- set up and run software applications such as PMS and CAS on any PC on the LAN.
- generate Station Message Detail Record (SMDR) Reports on any PC on the LAN.
- capture "System Activity Log" and "System Fault Log", Hotel Motel Activity Log.

LED

The CPU Card has three dual color LEDs:

- Heart beat (L1)
- Layer Communication indications (L2)
- Slave Switch indications (L3)

LED for both Active and Standby Card:

State	Color	Cadence
At Power ON		
	L1 and L3 – RED	Continuous ON for 15 sec (approximately)
	L1 and L3	OFF for 30 sec (approximately)
When L1 and L3 has been OFF for 10 sec	L2 – ORANGE	Continuously ON for 3 sec
	L2 – OFF	OFF

LED for Active Card:

State	Color	Cadence
L1 Pattern		
During initialization process	ORANGE	Continuous ON
After initialization process	-	OFF
In normal condition	GREEN	Toggle
L2 Pattern		
During initialization process	GREEN, RED, ORANGE	500 ms GREEN – 500 ms RED – 500 ms ORANGE
After initialization process	-	OFF
In normal condition	-	OFF
L3 Pattern		
When sync is received by Active card from Standby Card	RED	Continuous ON
When Sync is completed	-	OFF
When Data transfer done by Active card	RED	Continuous ON

LED for Standby Card:

State	Color	Cadence
L1 Pattern		
During initialization process	ORANGE	Continuous ON
After initialization process	-	OFF
In normal condition	RED	1 sec ON – 1 sec OFF
L2 Pattern		
During initialization process	GREEN, RED, ORANGE	500 ms GREEN – 500 ms RED – 500 ms ORANGE
After initialization process	-	OFF
PLL Lock (standby card clock is in sync with active card)	GREEN	Continuous ON
PLL is not lock (standby card clock is not in sync with active card)	-	OFF
L3 Pattern		
When sync is sent by Standby card to Active card	RED	Continuous ON
When Sync is completed	-	OFF

The CPU Card has a Password IP Default Switch (SW1) on the board:

Switch	Position	Function
SW1	ON	Password IP Default
	OFF (default)	Normal

Installing the VOCODER Module

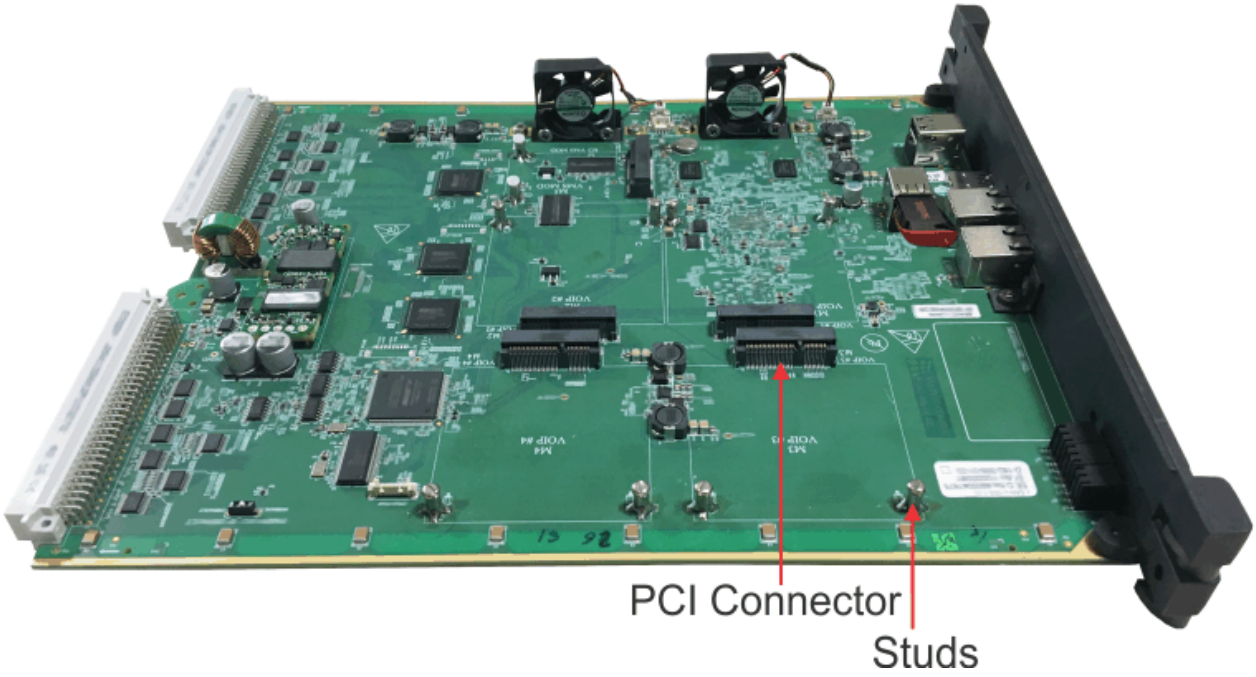
To install,

- Unpack the NX DBM VOCODER64 Module.

- If the CPU Card is already installed, switch off power supply, unplug the power cord. Remove the screws securing the card. Lift the levers on the mounting bracket to release the card. As the card emerges from the slot, ease it out of the slot.

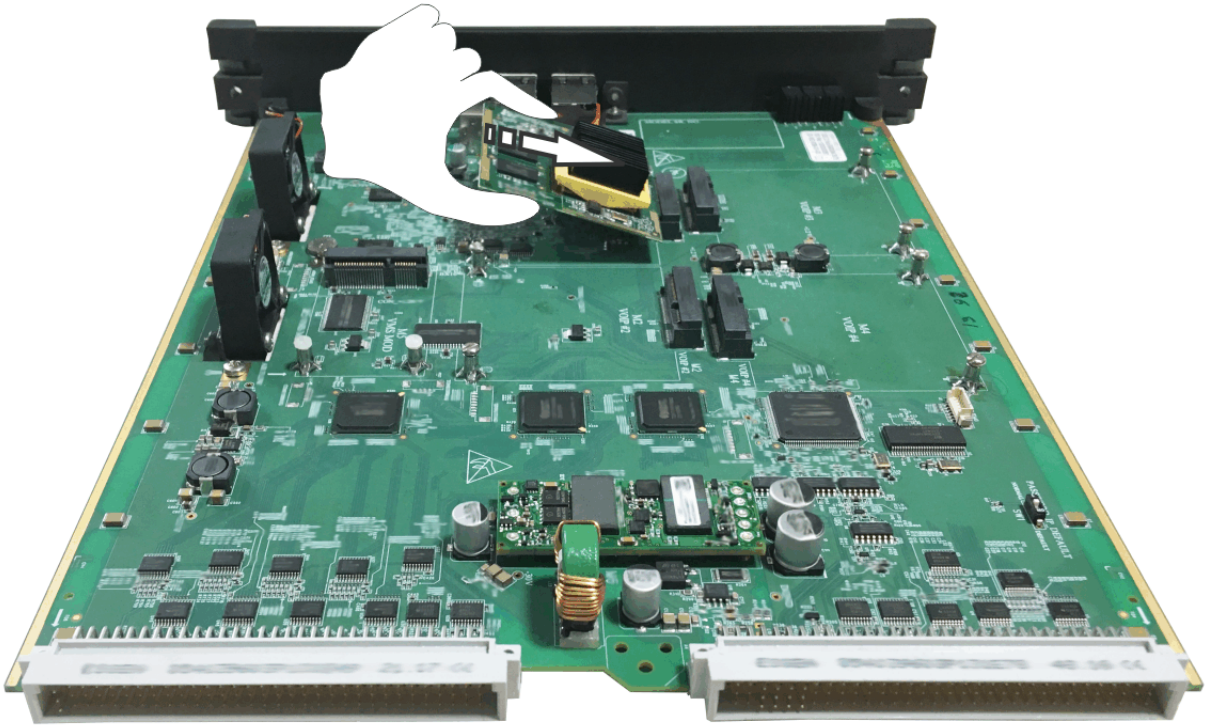
- Place the card carefully on a table with some packing underneath it. Avoid any physical contact with the PCB part of the card as this could cause Electrostatic discharge (ESD) and may damage the hardware.

- There are 5 PCI Connectors (4 for VoIP and 1 for VMS) and five pairs of studs on the CPU board.



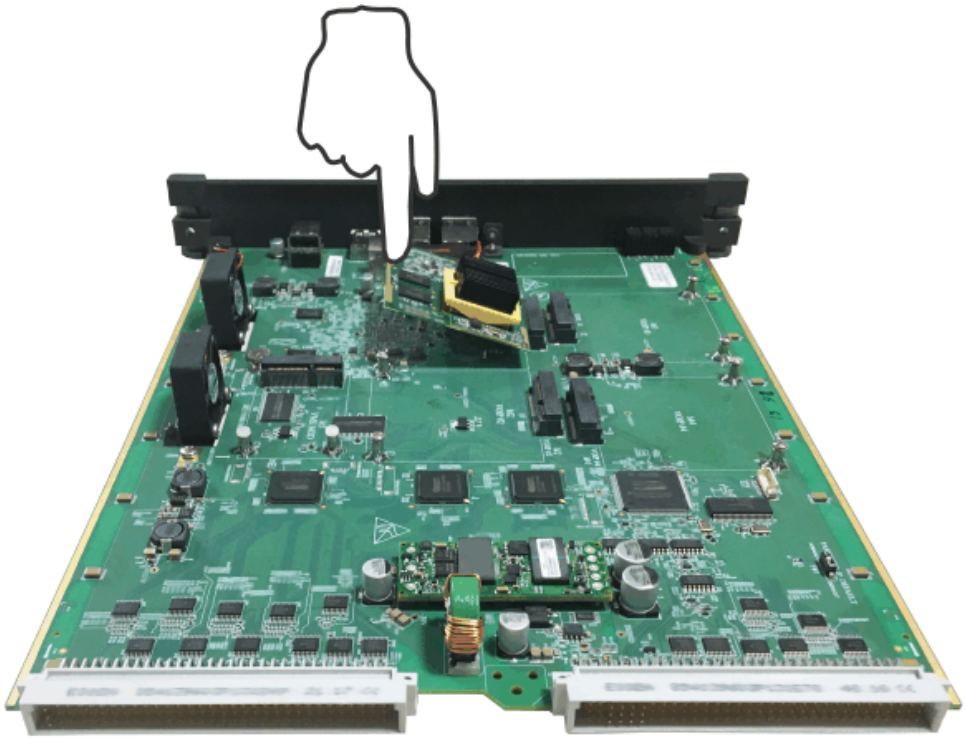
- Locate the **VOIP #1** label on the CPU board. The first NX DBM VOCODER64 Module should be mounted here.
- Remove the screws on the studs for the module and keep them aside.
- Carefully hold the NX DBM VOCODER64 Module from the edges. Make sure you do not touch the PCB area.

- Insert the NX DBM VOCODER64 Module into the PCI Connector socket.



- Press the Module with a finger and match the mounting holes perfectly with the stud holes. Make sure you do not touch the PCB area of the module except the yellow line provided for grounding at the front end of the module.

  Do not apply excessive pressure.

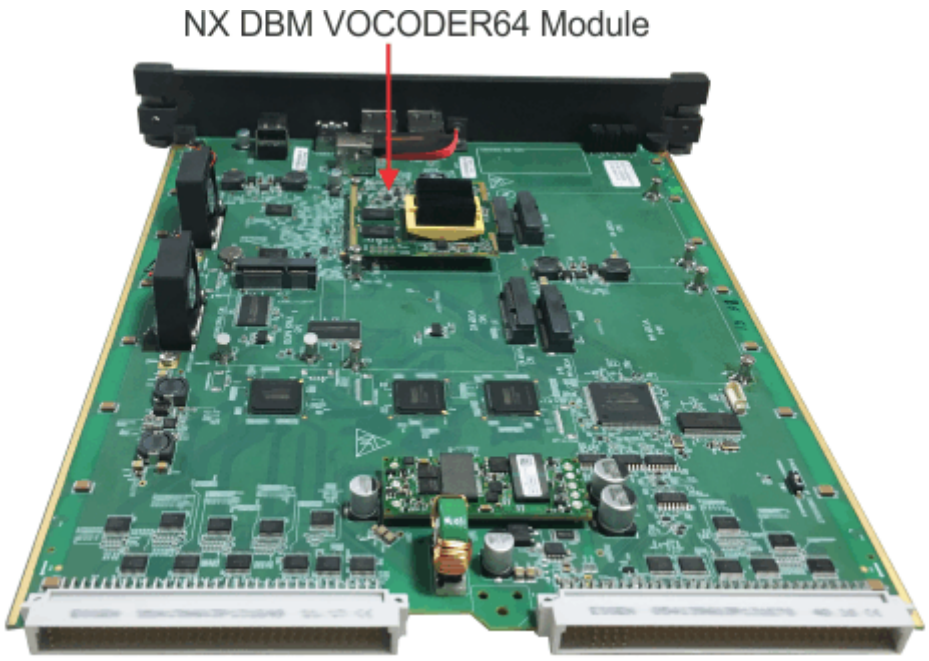


- Secure the module with the screws on the studs.

You may install the second NX DBM VOCODER64 Module by locating the **VOIP #2** label on the CPU board and follow the same steps as above. Similarly, install the other modules if required.

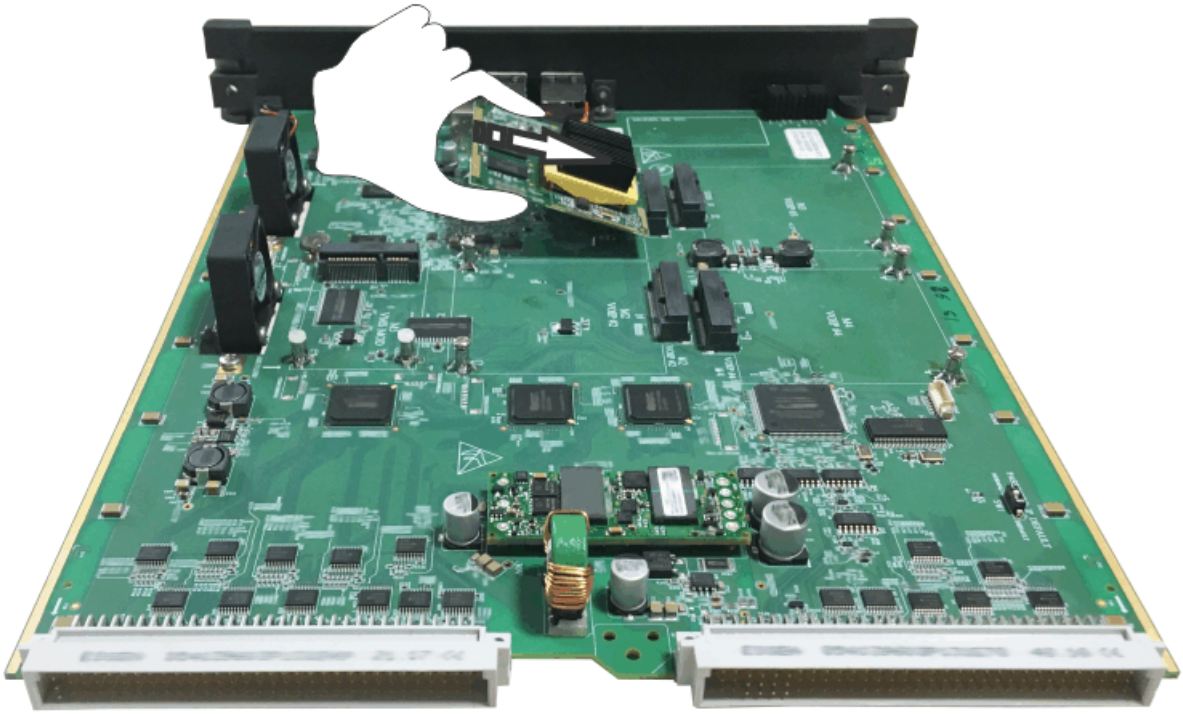
Removing the VOCODER Module

- Locate the VOCODER Module you want to remove from the CPU Card.



- Remove the screws from the module and keep them aside.

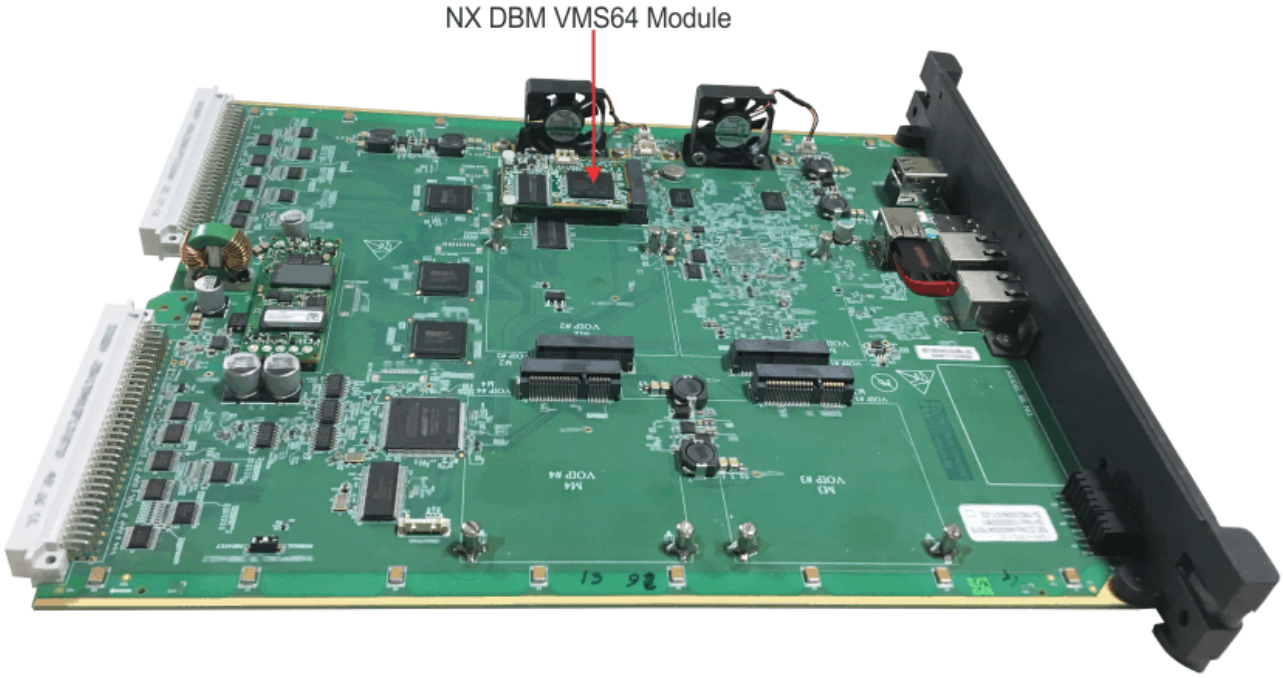
- Firmly hold the module and ease it out of the PCI connector carefully.



- Replace the screws on the studs.

Installing the VMS Module

To install the NX DBM VMS64 Module, locate the **VMS MOD** label on the CPU board and follow the same steps as described for “Installing the VOCODER Module”.



Make sure you use the plastic screws provided to affix the VMS Module over the studs.

- The pendrive which is provided to you by default contains VMS data and VMS firmware. You will be able to use the VMS features once you activate the VMS License.

  If you want to store more voice mail messages or greetings then you will need more space to store the same. You can replace this default pendrive with a new one having more space.

  To do so, you need to format your new pendrive with FAT32 file format and then copy all the contents of the factory fitted pendrive into the new pendrive.

  Switch-off the system and then replace the pendrive. The system will not detect the new pendrive if you do not restart the system after replacement.

Make sure while replacing the pendrive, you insert it in the same USB Port where the factory fitted pendrive was inserted.

- If you have no other modules to install, insert the card back into the ETERNITY MENX.
- Connect a computer to the LAN/WAN Port of the system with the ethernet cable supplied for the port.
- Open a Web browser on the computer to access the embedded Web server, Jeeves.
- Activate the VMS License Voucher. See “License Management” for instructions.
- Configure VMS. For detailed instructions, see “Configuring Voice Mail System”.

For removing the VMS Module, follow the same steps as described for removing the VOCODER Module. See “Removing the VOCODER Module”.

The Single Line Telephone Card

The Single Line Telephone (SLT) Card provides the interface to connect as extension phones, any standard, two-wire, analog single line telephone instrument - rotary, pulse-tone, cordless, feature phones with or without Calling Line Identification.

The SLT Card is available in the following configurations. SLT interface also is available in combination with Two-wire trunk interfaces on a single card.

SLT Cards for ETERNITY MENX

Card Name	Configuration and Application
ETERNITY ME Card SLT32	32-port card to connect 32 Single Line Telephones
ETERNITY ME Card SLT16	16-port card to connect 16 Single Line Telephones
ETERNITY ME Card SLT8	8-port card to connect 8 Single Line Telephones
ETERNITY ME Card CO8+SLT24	Combination card, with 8-ports to connect to 8 Two-wire Analog trunk lines and 24 Single Line Telephones This Card supports Power Fail Transfer. To know more, see “Power Fail Transfer”.

The maximum number of SLT ports supported are 288.

Connectors

The SLT Cards have RJ45 connectors, with each connector having 4 SLT ports. A multi-pair, MDF cable is supplied for each connector.

Only the SLT48 card has a 50-pin Centronics connector for the ports.

LEDs

The SLT cards have a single, tri-color LED to indicate:

- the health of the card during the Reset Cycle.
- the status of any one extension during normal functioning of the system.

  You may monitor any of the SLT Extension ports by assigning the LED to that port[54].

Installing Single Line Telephones

To be able to connect Single Line Telephones as Extensions to your system, you must install at least one of the aforementioned SLT Cards in the System.

1. Decide the number of SLT extensions required and arrange for as many telephone instruments.

54. *To do this, enter SE mode, and dial the SE Command 7902-Slot-LED Number-Port, where Slot is the number of the universal slot in which the card is installed and Port is the port on the card to which the LED is to be assigned to monitor its functioning. LED Number is the number of the LED on the card, which will monitor the port.*

You may use any standard telephone instrument like a rotary phone, a pulse-tone switchable push-button phone, a feature phone or a cordless phone.

Use SLTs equipped with a 'Flash' key, as several of the features and facilities of the system require you to press Flash. If any of the SLTs you have selected does not have a Flash key, tap the Hook switch of the phone to dial Flash.

2. Unpack the SLT Card and check the package contents. Ensure that the power supply is switched off, before you begin the installation of the card. Always wear an electrostatic discharge prevention wrist strap/ belt and use a grounding mat.

3. Unscrew and remove the filler card mount bracket of any of the free (empty) Universal Slots. Do not discard the filler bracket! You may require it at a later stage.

4. Insert the SLT Card into the guide rails of the free slot you selected for the card.

   Make sure that the connectors on the card make perfect contact with those on the motherboard on the backplane.

5. Press down the levers on the mounting bracket to secure the card in its slot. Now, secure the mounting bracket with the two screws provided.

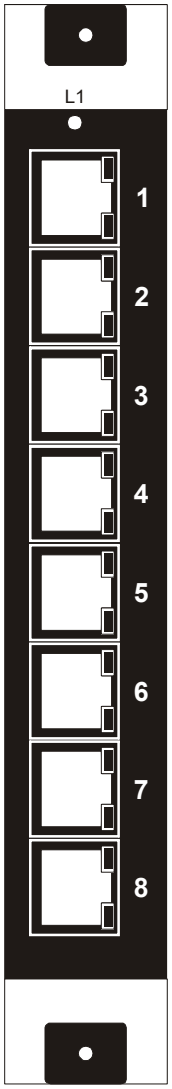
If you are installing more than one SLT Card, you can install the second card in any other free slot. It is not necessary to install the second/third card in the subsequent slots.

6. Use the cables supplied with the SLT Card to connect the SLT wires with the Main Distribution Frame.

   For each connector on the SLT Card, there is a separate 4-pair cable with an RJ45 jack on one end and free at the other end.

   Refer the illustrations below for pin out details of each connector.

ETERNITY ME Card SLT32



Connector	Color	Connection	H/w Port Offset
RJ45-1	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	01 02 03 04
RJ45-2	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	05 06 07 08
RJ45-3	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	09 10 11 12
RJ45-4	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	13 14 15 16
RJ45-5	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	17 18 19 20
RJ45-6	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	21 22 23 24
RJ45-7	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	25 26 27 28
RJ45-8	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	29 30 31 32

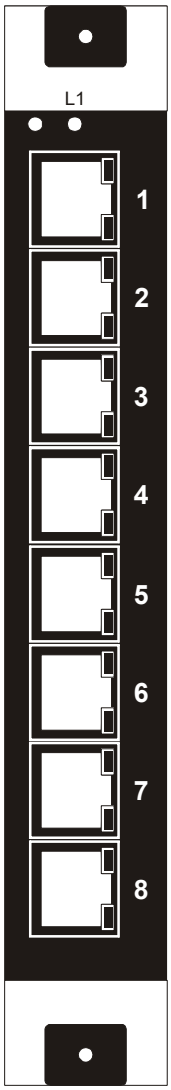
ETERNITY ME Card SLT16

Connector	Color	Connection	H/w Port Offset
RJ45-1	Blue - (Blue & White)	SLT	01
	Orange - (Orange & White)	SLT	02
	Green - (Green & White)	SLT	03
	Brown - (Brown & White)	SLT	04
RJ45-2	Blue - (Blue & White)	SLT	05
	Orange - (Orange & White)	SLT	06
	Green - (Green & White)	SLT	07
	Brown - (Brown & White)	SLT	08
RJ45-3	Blue - (Blue & White)	SLT	09
	Orange - (Orange & White)	SLT	10
	Green - (Green & White)	SLT	11
	Brown - (Brown & White)	SLT	12
RJ45-4	Blue - (Blue & White)	SLT	13
	Orange - (Orange & White)	SLT	14
	Green - (Green & White)	SLT	15
	Brown - (Brown & White)	SLT	16

ETERNITY ME Card SLT8

Connector	Color	Connection	H/w Port Offset
RJ45-1	Blue - (Blue & White)	SLT	01
	Orange - (Orange & White)	SLT	02
	Green - (Green & White)	SLT	03
	Brown - (Brown & White)	SLT	04
RJ45-2	Blue - (Blue & White)	SLT	05
	Orange - (Orange & White)	SLT	06
	Green - (Green & White)	SLT	07
	Brown - (Brown & White)	SLT	08

ETERNITY ME Card CO8+SLT24



Connector	Color	Connection	H/w Port Offset
RJ45-1	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	01 02 03 04
RJ45-2	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	05 06 07 08
RJ45-3	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	09 10 11 12
RJ45-4	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	13 14 15 16
RJ45-5	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	17 18 19 20
RJ45-6	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	21 22 23 24
RJ45-7	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	TWT TWT TWT TWT	01 02 03 04
RJ45-8	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	TWT TWT TWT TWT	05 06 07 08

7. Plug in the RJ45 end of the MDF cables supplied with the card into the respective connectors. Refer to the pinout details of the connectors of each SLT Card type illustrated above.

8. Terminate the open end of the cables into the punch down blocks of the Krone modules designated for 'Station Lines' in the “The Main Distribution Frame (MDF)”.

   Each wire-pair from the SLT Port must be terminated to the bottom of the Krone Connector, while the wire-pair of the extension line to be connected to this port must be terminated on the top of the Krone connector. Refer the topic “The Main Distribution Frame (MDF)” for illustration.

9. Repeat the same steps to install another SLT Card.

10. If you have completed all installation tasks, power ON the system, observe the Reset Cycle and the LED Pattern of the SLT Card.

LED Pattern of the SLT Card

Stage	LED Color	Cadence
Auto Upgradation[a]		
Card waiting for application	RED	ON-200ms-OFF 200ms
Card is up, loaded with new application	GREEN	ON-200ms-OFF 200ms
Initialization		
	RED	ON 500ms-OFF 500ms
	GREEN	ON 500ms - OFF 500ms
	ORANGE	ON 500ms - OFF 500ms
Stand-by task	ORANGE	1 sec Orange -1 sec Green
Errors		
Flash Failure	None	None
RAM Failure	None	None

a. *Done by the Boot Loader Application.*

Status of Selected Port

PORT Status	LED Color	LED Cadence
Selected port data transmitted to Master Card	RED	Toggle[a]
Selected port data received from Master Card	RED	Toggle

a. The current LED state will remain the same until the next command is received from the application on the SLT Port. For example, if the current LED state is Green/Red ON, on the next command received, the LED will be turned OFF. It will remain OFF until the next command is received. When the next command is received it will be turned Green/Red ON again. This process continues.

Jumpers on the Main Board

Jumper Number	Position	Function
J1	AB (default)	Normal Operation
	BC	For uploading software using COM Port
J2 & J4	AB (default)	Normal Operation
	BC	For uploading software using COM Port

Connecting SLT instruments

11. Connect the SLT instruments you have arranged for. Plug in the SLTs into the wall socket/outlets.

- *For the purpose of testing, you may connect one or two Single Line Telephone instruments by plugging in the phone cables into the RJ45 connectors on the card.*

- *When you plug the RJ11 connector of SLT into an RJ45 connector on the SLT Card, the SLT will be connected on the first port on the connector.*

The Intercom Line Card[55]

For the Building Intercom application, the system supports the Intercom Line Card (ILC).

You can connect any standard, two-wire, analog single line telephone instrument - rotary, pulse-tone, cordless, feature phones with or without Calling Line Identification to the Intercom Line Card.

ILC Cards for ETERNITY MENX

Card Name	Configuration and Application
ETERNITY ME Card ILC32	32-port card to connect 32 Single Line Telephones

Choose an ILC Card with the configuration that meets your requirement for intercom ports. Also, consider the maximum Port capacity of the system you are installing. The maximum number of intercom ports supported are 288.

Connectors

The ILC32 Cards have RJ45 connectors with four ports on each connector. A multi-pair, MDF cable is supplied for each connector.

LEDs

The ILC cards have a single, tri-color LED to indicate the health of the card during the Reset Cycle.

Installing Intercom Telephones

To be able to connect intercom telephones to the system, you must install at least one of the aforementioned intercom line cards in the system.

1. Decide the number of intercom extensions required and arrange for as many telephone instruments.
2. Ensure that the extension wiring is completed according to your requirements. The extension cables from the wall jack are terminated in the Main Distribution Frame and the telephones are connected to the wall jacks.
3. Always wear an electrostatic discharge prevention wrist strap/belt and use a grounding mat to prevent damage to the components of the card.
4. Unpack the ILC card and check the package contents. You may switch off power supply before you install the card. Since, ETERNITY MENX supports Hot Swap, you can install the card in power on condition.
5. Unscrew and remove the filler card mount bracket of any of the free (empty) Universal Slots. Keep the filler bracket for future use.
6. Insert the ILC Card into the guide rails of the free slot you selected for the card.

55. This card has been phased-out. if required, only support will be provided for the same.

Make sure that the connectors on the card make perfect contact with those on the motherboard on the backplane.

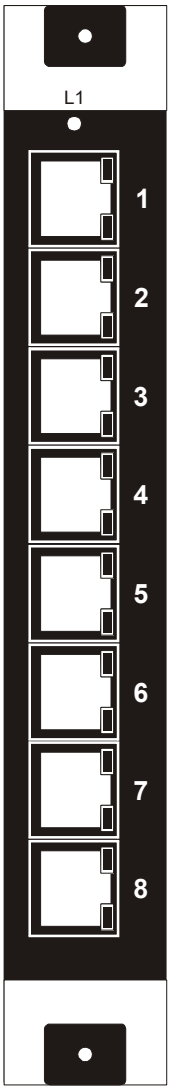
7. Press down the levers on the mounting bracket to secure the card in its slot. Now, secure the mounting bracket with the two screws provided.

8. Repeat these steps to install another card.

9. Now, use the cables supplied with the ILC Card to connect the card to the Main Distribution Frame to which the intercom phones are connected.

   For each connector on the card, there is a separate 4-pair cable with an RJ45 jack on one end and free at the other end. Refer the illustrations of the pinout of the intercom cards to connect the wires.

ETERNITY ME Card ILC32



Connector	Color	Connection	H/w Port Offset
RJ45-1	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	01 02 03 04
RJ45-2	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	05 06 07 08
RJ45-3	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	09 10 11 12
RJ45-4	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	13 14 15 16
RJ45-5	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	17 18 19 20
RJ45-6	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	21 22 23 24
RJ45-7	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	25 26 27 28
RJ45-8	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	29 30 31 32

10. If you have completed all other installation tasks, power ON the system, observe the Reset Cycle and the LED indication of the card.

LED Indication of the ILC Card

Stage	LED Color	Cadence
Auto Upgradation[a]		
Card waiting for application	RED	ON-200ms-OFF 200ms
Card is up, loaded with new application	GREEN	ON-200ms-OFF 200ms
Initialization		
	RED	ON 500ms-OFF 500ms
	GREEN	ON 500ms - OFF 500ms
	ORANGE	ON 500ms - OFF 500ms
Stand-by task	ORANGE, GREEN	1 sec Orange -1 sec Green
Errors		
Flash Failure	None	None
RAM Failure	None	None

a. *Done by the Boot Loader Application.*

Status of Selected Port

PORT Status	LED Color	LED Cadence
Selected port data transmitted to Master Card	RED	Toggle[a]
Selected port data received from Master Card	RED	Toggle

a. The current LED state will remain the same until the next command is received from the application on the SLT Port. For example, if the current LED state is Green/Red ON, on the next command received, the LED will be turned OFF. It will remain OFF until the next command is received. When the next command is received it will be turned Green/Red ON again. This process continues.

Jumpers on the Main Board

Jumper Number	Position	Function
J1	AB (default)	Normal Operation
	BC	For uploading software using COM Port
J2 & J4	AB (default)	Normal Operation
	BC	For uploading software using COM Port

The Digital Key Phone Card

The Digital Key Phone (DKP) Card provides the interface to connect the proprietary digital key phones of the EON series, the proprietary PC-based phone EONSOFT and the Direct Station Selection (DSS) Consoles with the system.

DKP Cards for ETERNITY MENX

Card Name	Configuration and Application
ETERNITY ME DKP32	32-port card to connect 32 DKP/DSS Consoles
ETERNITY ME DKP16	16-port card to connect 16 DKP/DSS Consoles
ETERNITY ME DKP8	8-port card to connect 8 DKP/DSS Consoles

Select a DKP Card with the configuration that meets your requirement for DKP Ports.To connect the proprietary digital key phones with the system, you must have at least one of the above mentioned DKP Cards installed in the system.

The maximum number of DKP Ports supported by the system are 128.

Connectors

The DKP Cards have RJ45 connectors, with each connector having 4 DKP ports. A multi-pair MDF cable is supplied for each connector on the card.

LEDs

The DKP cards for ETERNITY ME have a single, tri-color LED to indicate:

- the health of the card during the Reset Cycle.
- the status of any one of the ports during normal functioning of the system. By default, it is assigned to DKP Port 1.

  You may monitor any of the DKP ports by assigning the LED to that port[56].

Installing the Digital Key Phone Card

Decide the number of DKP extensions and DSS Consoles required and arrange for as many EON, EONSOFT and DSS Consoles.

Decide the locations of the DKP extensions and make sure that the necessary wiring for the DKP extensions, from the wall jack to the MDF, is done.

1. Unpack the DKP Card and check the package contents[57]. Before handling the card, make sure that power supply is switched off and you are wearing an antistatic-wrist strap/belt and have a grounding mat.

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56. You can do this from the SE mode, by dialing the SE Command 7902-Slot-LED Number-Port, where Slot is the number of the universal slot in which the card is installed and Port is the port on the card to which the LED is to be assigned to monitor its functioning. LED Number is the number of the LED on the card, which will monitor the port.

57. See "ETERNITY GENX Cards" under 'Packing List' of Appendix topic.

2. Unscrew and remove the filler card mount bracket of any of the free (empty) Universal Slots. Do not discard the filler bracket, keep for future use to cover empty slots.

3. Insert the DKP Card into the guide rails of the free slot you have selected for the card. All the pins on the connector of the card should make perfect contact with those on the connector of the slot on the backplane motherboard.

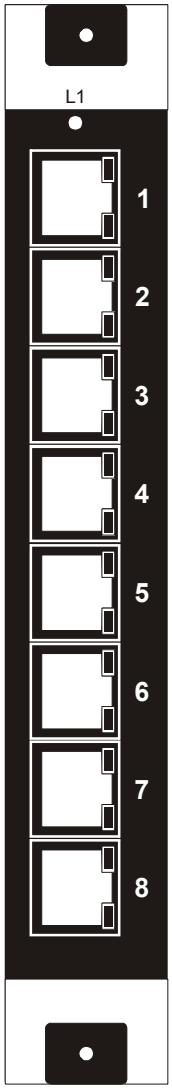
4. Press down the levers on the mounting bracket to secure the card in its slot. Now, fix the card in its slot with the two screws provided.

If you are installing more than one DKP Card, it is not necessary to install the next card in the subsequent slot.

5. Using the MDF Cables supplied with the DKP Card connect the DKP ports to the Main Distribution Frame.

   Refer the connector pin details for each DKP Card type given in the following.

ETERNITY ME Card DKP32



Connector	Color	Connection	H/w Port Offset
RJ45-1	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	DKP DKP DKP DKP	01 02 03 04
RJ45-2	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	DKP DKP DKP DKP	05 06 07 08
RJ45-3	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	DKP DKP DKP DKP	09 10 11 12
RJ45-4	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	DKP DKP DKP DKP	13 14 15 16
RJ45-5	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	DKP DKP DKP DKP	17 18 19 20
RJ45-6	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	DKP DKP DKP DKP	21 22 23 24
RJ45-7	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	DKP DKP DKP DKP	25 26 27 28
RJ45-8	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	DKP DKP DKP DKP	29 30 31 32

ETERNITY ME Card DKP16

Connector	Color	Connection	H/w Port Offset
RJ45-1	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	DKP DKP DKP DKP	01 02 03 04
RJ45-2	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	DKP DKP DKP DKP	05 06 07 08
RJ45-3	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	DKP DKP DKP DKP	09 10 11 12
RJ45-4	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	DKP DKP DKP DKP	13 14 15 16

ETERNITY ME Card DKP8

Connector	Color	Connection	H/w Port Offset
RJ45-1	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	DKP DKP DKP DKP	01 02 03 04
RJ45-2	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	DKP DKP DKP DKP	05 06 07 08

6. Plug in the RJ45 end of the MDF cables provided with the DKP card into the respective connectors.

7. Terminate the free end of the cables into the punch down blocks of the Krone modules designated for 'Station Lines' in the Main Distribution Frame (MDF).

   Each wire-pair from the DKP Port must be terminated to the bottom of the Krone Connector, while the wire-pair of the extension line to be connected to this port must be terminated on the top of the Krone connector. Refer the topic “The Main Distribution Frame (MDF)” for illustration.

8. Connect the Digital Key Phones to the wall jacks at their respective locations. Detailed installations instructions for EON, EONSOFT are provided separately.

If you have completed all installation tasks, power on the system and observe the Reset Cycle and the LED Pattern of the DKP Card.

LED Pattern DKP Card

Stage	LED Color	Cadence
Auto Upgradation		
Card waiting for application	RED	ON-200ms-OFF 200ms
Card is up, loaded with new application	GREEN	ON-200ms-OFF 200ms
Initialization		
	RED	ON 500ms-OFF 500ms
	GREEN	ON 500ms-OFF 500ms
	ORANGE	ON 500ms-OFF 500ms
Stand-by task	ORANGE, GREEN	1 sec Orange-1 sec Green
Errors		
Flash Failure	None	None
RAM Failure	None	None

Status of Selected DKP Port

PORT Status	LED Color	LED Cadence
Selected DKP's data are transmitted to CPU Card	RED	Toggle[a] on each event
Selected DKP's data are received from CPU Card	RED	Toggle[b] on each request from Master

a. The current LED state will remain the same until the next event is received from the application on the DKP Port. For example, if the current LED state is Green/Red ON, on the next event, the LED will be turned OFF. It will remain OFF until the next event occurs. When the next event is received it will be turned Green/Red ON again. This process continues.

b. Same as the above note.

Jumpers on the Main Board

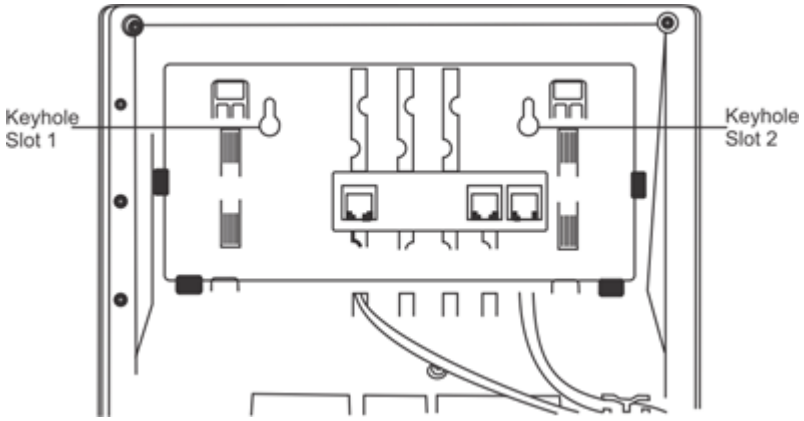
Jumper Number	Position	Function
J1	AB (default)	Normal Operation
	BC	For uploading software using COM Port
J2 & J4	AB	NA
	BC (default)	Normal Operation/ Debug

Installing EON48

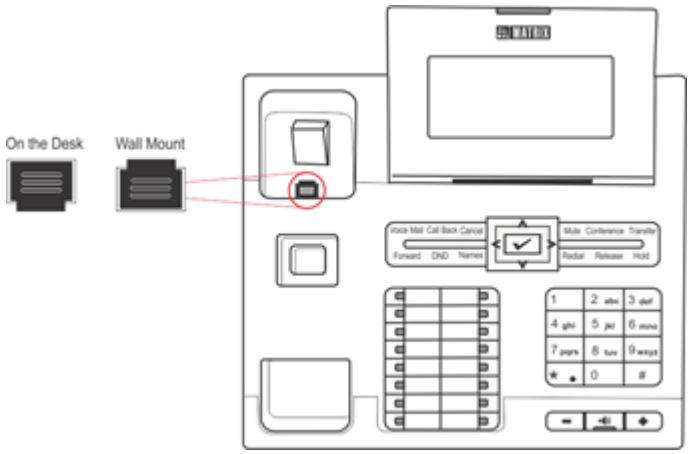
- Unpack the box and verify the package contents.
- You can mount the phone on a wall or on desk.

Mount the phone on a Wall

- Use the mounting template to drill holes of appropriate size and distance.
- Fix the screw grips in the holes you drilled.
- Fix two screws in the holes on the wall, ensuring that they are aligned with the Keyhole Slots 1 and 2. The screws should protrude from the wall to fit into the keyhole slots.

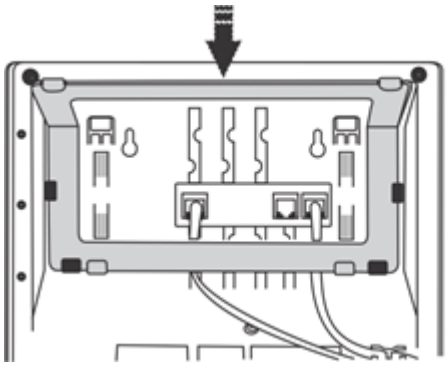


- Now, mount the phone with the screws fitting into the keyhole slots.
- Reverse the handset wall mount tab to make sure that the handset remains intact when you mount the phone. Push the handset wall mount tab upwards to remove it from the slot. Rotate it 180 degrees clockwise and push it downwards into the slot.

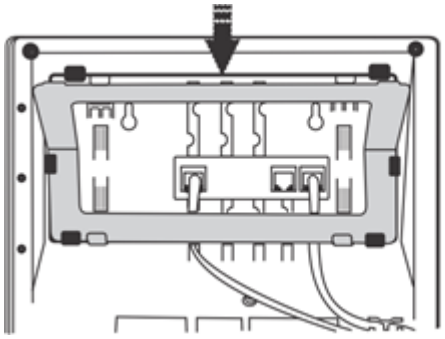


Mount the phone on the Desk

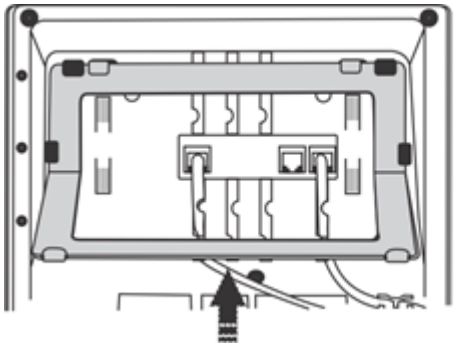
- You can attach the Foot Stand in the following ways—at an angle of **20° Angle** or at **30° Angle** or at **50° Angle**.



Foot Stand at
20 degree angle



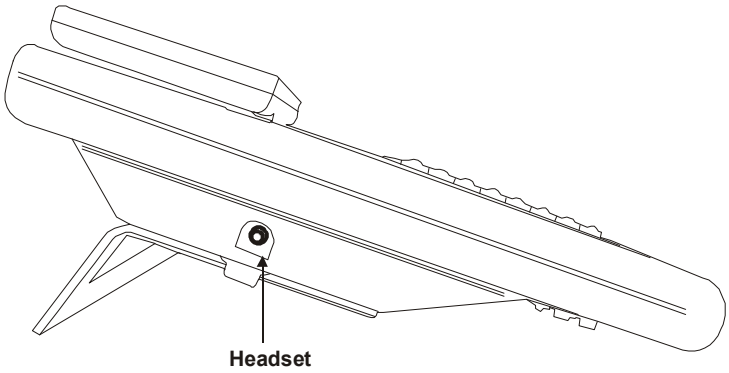
Foot Stand at
30 degree angle



Foot Stand at
50 degree angle

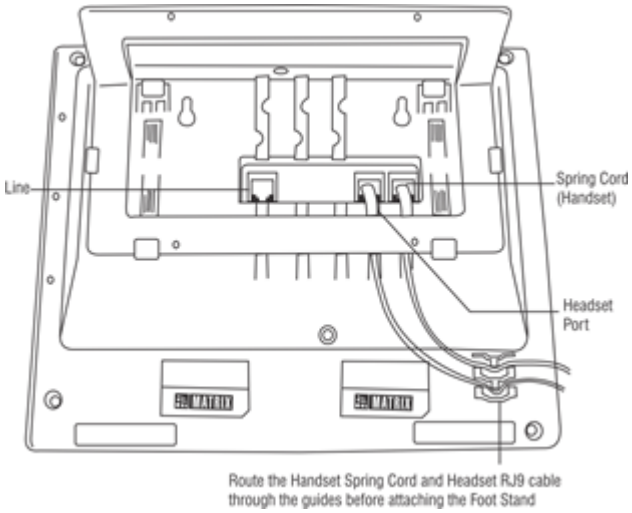
- If you attach the Foot Stand at 50°, the phone will be placed in an almost upright position on your desk.
- Decide which of these positions would work for you best and accordingly attach the Foot Stand.
- Connect the handset of the EON48 to the phone body using the spring cord.

- To use a Headset (not supplied with the phone), plug any standard stereo headset with 2.5mm single connector into the headset jack with the symbol on the left side panel of the phone.

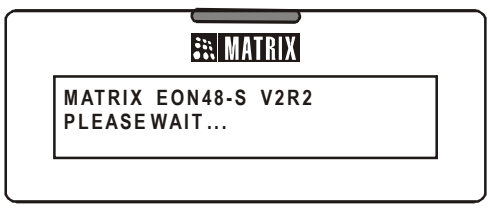


You may also plug in a stereo headset with an RJ9 connector into the headset port at the bottom of the phone, marked with the symbol .

- Plug one end of the RJ11 cable supplied with the phone into the RJ11 connector of the phone labeled as '**LINE**' and the other end into the wall jack/DKP Port.



- When the system is powered ON, the EON will reset. The EON communicates with the system. The handshaking lasts for 5-6 seconds. The EON model, version and revision number, along with the message *'Please wait'…* appear on the LCD display.



- After successful handshaking and reset cycle, if the DKP Parameters have been configured, the LCD display of the EON will show the extension number and the extension name in one line and the day, date and time and the time zone in the other line.

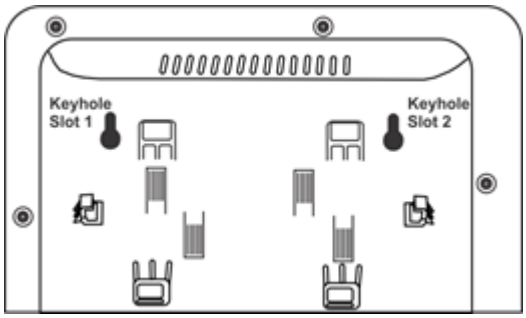
- You may adjust the LCD for brightness, contrast and backlight. Refer the topic, “Digital Key Phone-Operation” for instructions.

Installing EON310

- Unpack the box and verify the package contents.
- You can mount the phone on a wall or on desk.

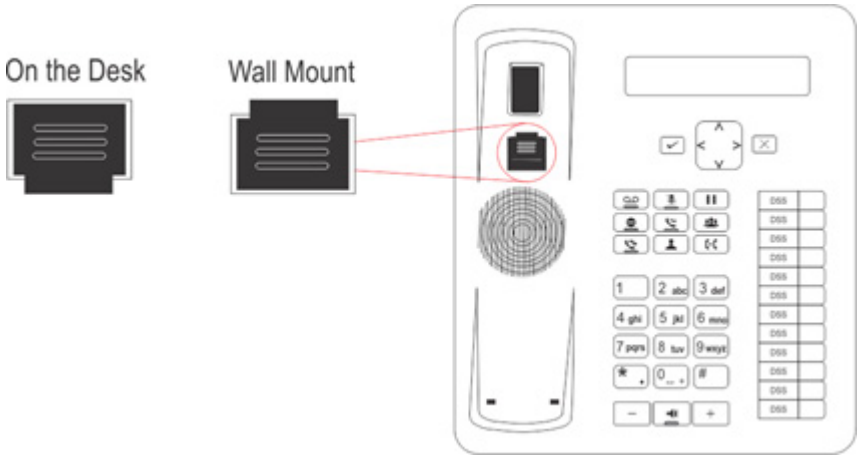
Mount the phone on a Wall

- Use the mounting template to drill holes of appropriate size and distance.
- Fix the screw grips in the holes you drilled.
- Fix two screws in the holes on the wall, ensuring that they are aligned with the Keyhole Slots 1 and 2. The screws should protrude from the wall to fit into the keyhole slots.



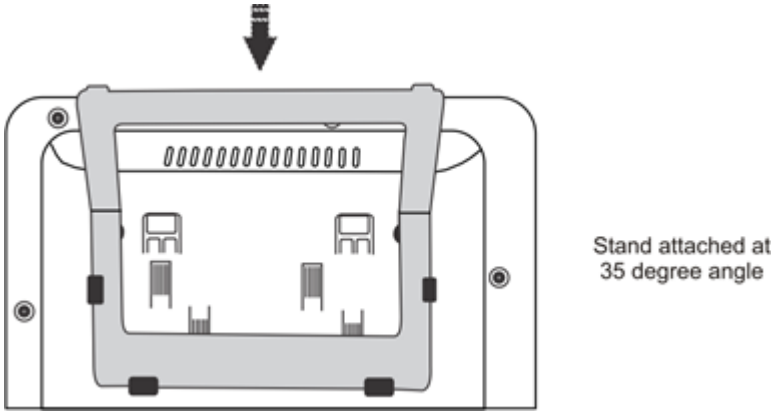
- Now, mount the phone with the screws fitting into the keyhole slots.

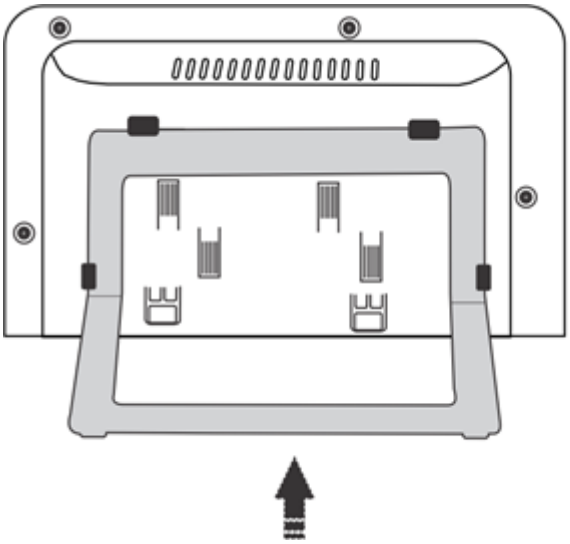
- Reverse the handset wall mount tab to make sure that the handset remains intact when you mount the phone. Push the handset wall mount tab upwards to remove it from the slot. Rotate it 180 degrees clockwise and push it downwards into the slot.



Mount the phone on the Desk

- You can attach the Foot Stand in the following ways—at an angle of **35° Angle** or at **50° Angle**.

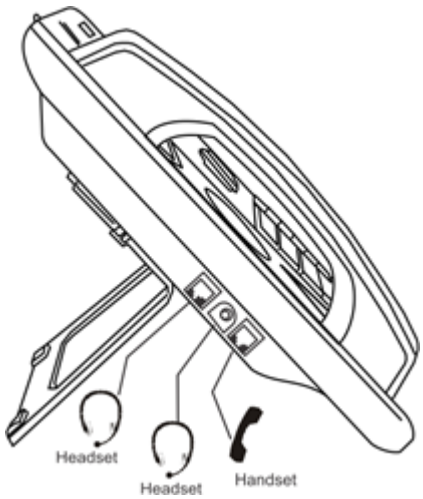




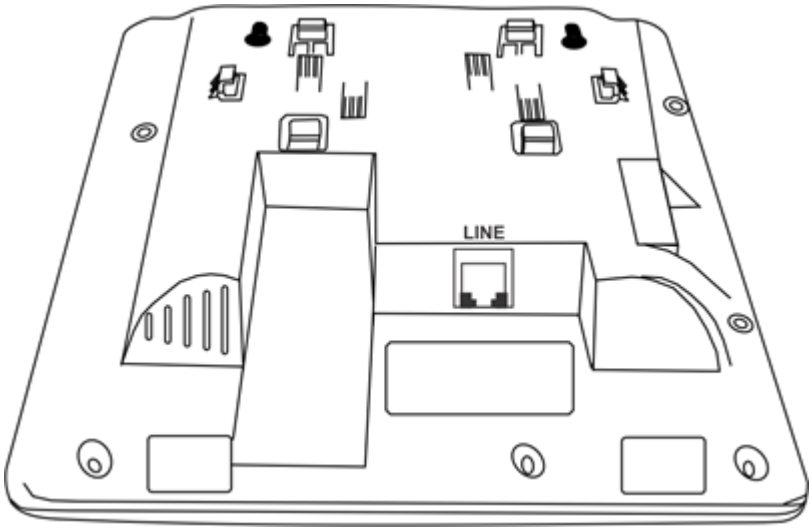
Stand attached at
50 degree angle

- Decide which of these positions would work for you best and accordingly attach the Foot Stand.
- Connect the handset of the EON310 to the phone body using the spring cord.
- To use a Headset (not supplied with the phone), plug any standard stereo headset with 3.5mm single connector into the headset jack with the symbol on the left side panel of the phone.

  You may also plug in a stereo headset with an RJ9 connector into the headset port marked with the symbol , on the left side panel of the phone.



- Plug one end of the RJ11 cable supplied with the phone into the RJ11 connector of the phone labeled as '**LINE**' and the other end into the wall jack/DKP Port.



- When the system is powered ON, the EON will get reset and the message "Welcome to Matrix. Booting"…appears on the LCD display.

- The EON communicates with the system. The handshaking lasts for 5-6 seconds. The EON model, version and revision number, along with the message "Please Wait"…appears on the LCD display.

- After successful handshaking and reset cycle, the extension number, day, date and time will appear on the LCD of the phone. If you have already assigned extension number and name, in the DKP Parameters, these will appear on the LCD.

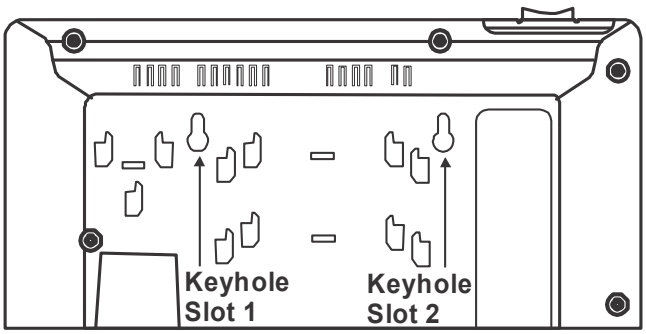
- You may adjust the LCD for brightness, contrast and backlight. Refer the topic, “Digital Key Phone-Operation”.

Installing EON510

- Unpack the box and verify the package contents.
- You can mount the phone on a wall or on desk.

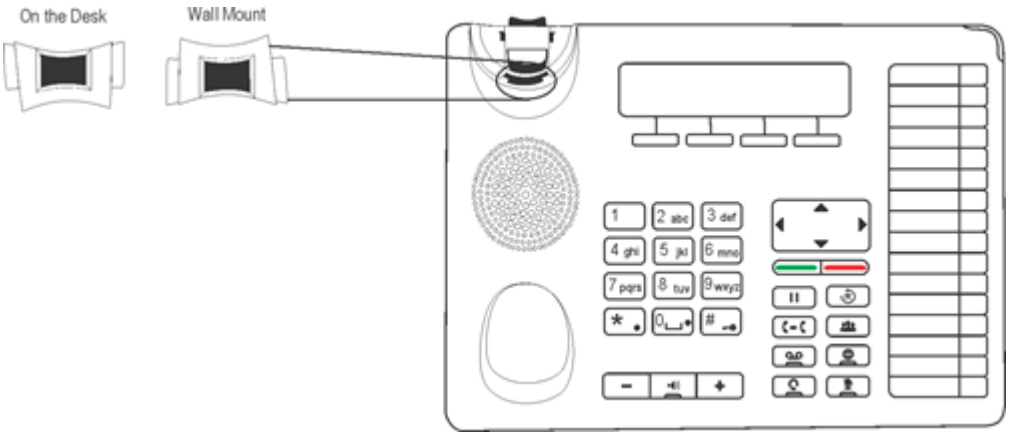
Mount the phone on a Wall

- Use the mounting template to drill holes of appropriate size and distance.
- Fix the screw grips in the holes you drilled.
- Fix two screws in the holes on the wall, ensuring that they are aligned with the Keyhole Slots 1 and 2. The screws should protrude from the wall to fit into the keyhole slots.



- Now, mount the phone with the screws fitting into the keyhole slots.

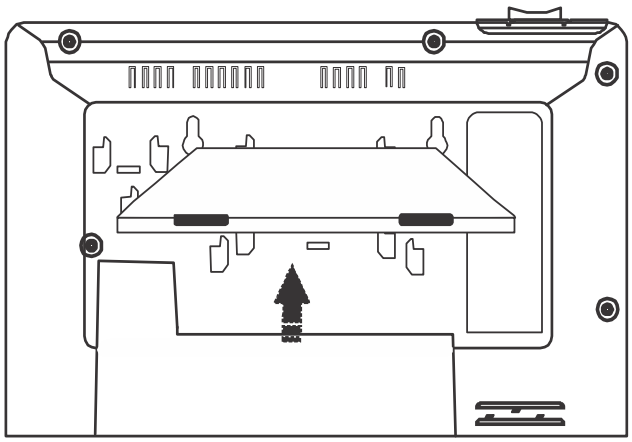
- Reverse the handset wall mount tab to make sure that the handset remains intact when you mount the phone. Push the handset wall mount tab upwards to remove it from the slot. Rotate it 180 degrees clockwise and push it downwards into the slot.



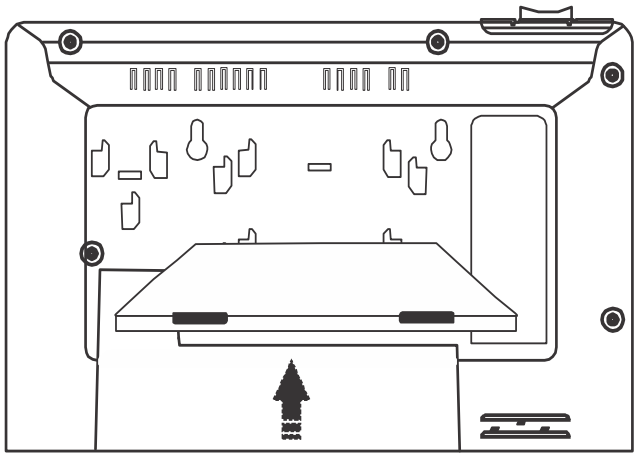
If you are unable to remove the wall mount tab, you may use a tool like a minus screw driver to remove it.

Mount the phone on the Desk

- You can attach the Foot Stand in the following ways—at an angle of **45° Angle** or at **55° Angle**.

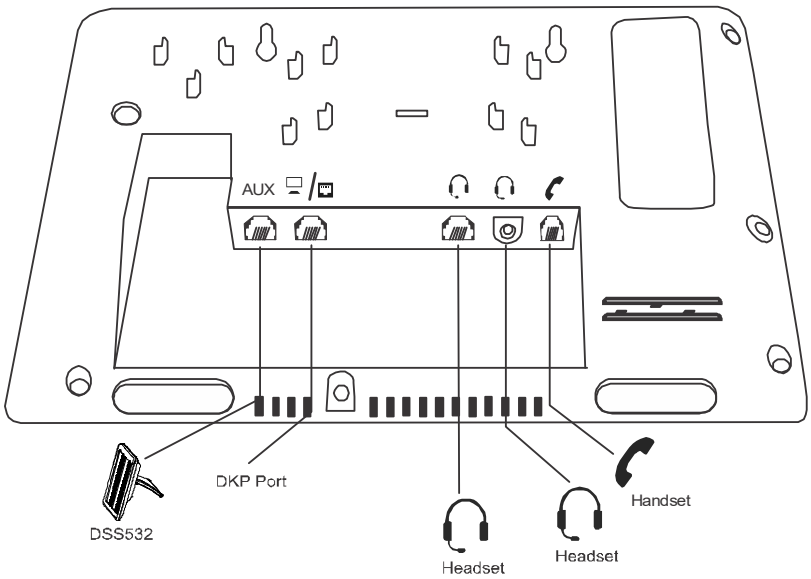


Stand attached at
45 degree angle



Stand attached at
55 degree angle

- Decide which of these positions would work for you best and accordingly attach the Foot Stand.



- Connect the handset of the EON510 to the phone body using the spring cord.
- To use a Headset (not supplied with the phone), plug any standard stereo headset with 3.5mm single connector into the headset jack with the symbol on the left side panel of the phone.

  You may also plug in a stereo headset with an RJ9 connector into the headset port marked with the symbol , on the left side panel of the phone.

- Plug one end of the RJ11 cable supplied with the phone into the RJ11 connector of the phone marked with the symbol and the other end into the wall jack/DKP Port.

- To connect DSS532 with the phone, plug one end of the RJ11 cable into the AUX Port of the phone and the other end into the IN Port of the DSS532. For installation, see “Installing DSS532 with EON510”.
- When the system is powered ON, the EON will get reset.and the message 'Phone is Booting; Please wait…’ appear on the LCD display.

- The EON communicates with the system. The handshaking lasts for 5-6 seconds. The message 'Loading Firmware Version-Revision; Please wait…' appear on the LCD display.

- After successful handshaking and reset cycle, the extension number, day, date and time will appear on the LCD of the phone. If you have already assigned extension number and name, in the DKP Parameters, these will appear, as illustrated below.

You may adjust the LCD for brightness, contrast and backlight. Refer the topic, “Digital Key Phone-Operation”.

Installing DSS Consoles

Installing DSS64

Once you have installed EON48/310 with SARVAM UCS, installing the DSS Consoles can be done in a few simple steps, very much similar to those involved in the installation of EON.

1. Unpack the box and verify the package contents[58].
2. Place the DSS Console next to the DKP to which it is to be attached.
3. Decide which DKP Ports on the DKP Card are to be assigned to the DSS Consoles. You may select any free (unused) port on the card for DSS Consoles. It is not necessary for the DSS Console ports to be in a sequence with the DKP ports to which they are attached.

   For example: you have connected DKP1 to Port 1 on the first RJ45 connector of the DKP8 card. You want to attach two DSS Consoles to DKP1. The two DSS Consoles may be connected to any port on the second connector of the card, not necessarily to Port 2 and Port 3 on the first connector.
4. The wire-pairs from the DKP Ports designated for DSS Consoles should be terminated on the bottom of the Krone Connector (of 'Station Lines' on the MDF).
5. The wire-pairs of the DSS Consoles should be terminated into the top of the Krone Connector (of 'Station Lines' on the MDF). Refer the topic “The Main Distribution Frame (MDF)” for illustration.

   You can connect maximum two DSS64 with a single EON48/310.

58. *See “Packing List” of Appendix topic.*

6. The system automatically detects the DSS Console you connect and it will be will appear under **Unassigned DSS64** in “DSS Status”. You must first assign these DSS Consoles to the respective DKP Ports and thereafter you will be able to configure the DSS Keys.

7. To assign the DSS Consoles, see “DSS Status” and to configure the DSS Keys, see “Programming DSS Console Keys”.

Installing DSS532 with EON510

The instructions for installing DSS532 with EON510 or SPARSH VP510 are same. For detailed instructions, refer to “Installing DSS532 with SPARSH VP510”.

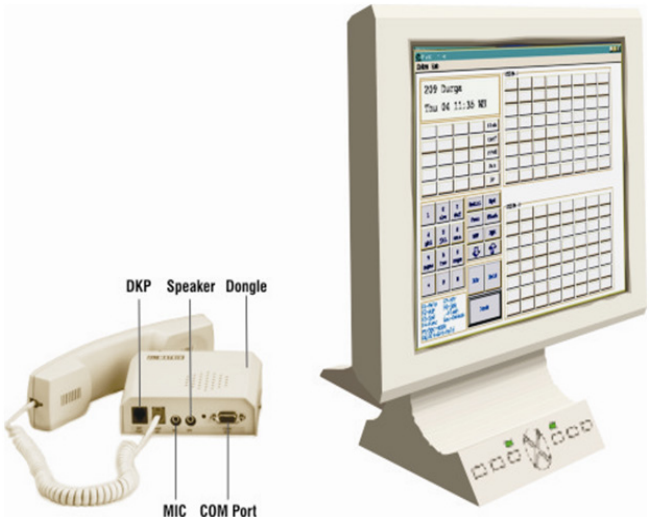
Installing EONSOFT

To install EONSOFT, you must have a computer with Windows as the operating system. The EONSOFT is compatible with the following Operating Systems of Windows:

- Windows 98
- Windows XP
- Windows NT
- Windows 2003
- Windows Vista
- Windows 2007

1. Unpack the box and verify the package contents[59].

2. Connect the Handset to the dongle in the handset jack. If using a headset, connect the microphone and the speaker connectors into the dongle.



3. Connect one end of the Communication cable to the COM port of the dongle. Connect the other end of the communication cable into the COM port of the computer.

4. Connect a wire-pair of a DKP port to the RJ11 port marked 'DKP' on the dongle.

---

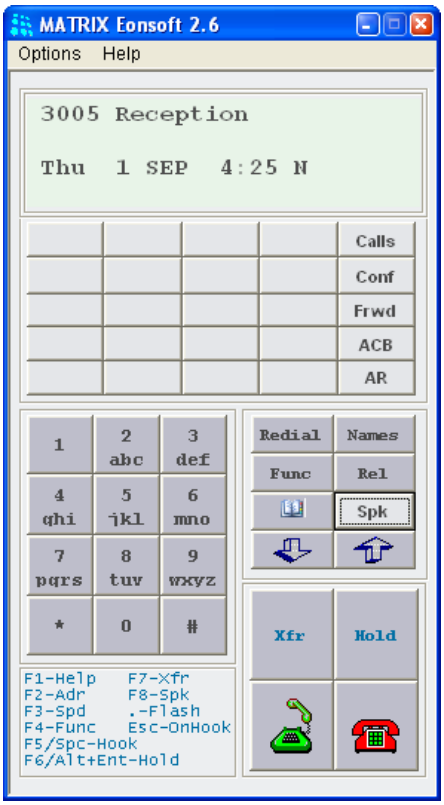
59. See “EONSOFT” under ‘Packing List’ of Appendix topic.

5. Switch ON the computer. The computer must have Windows Operating System installed on it.

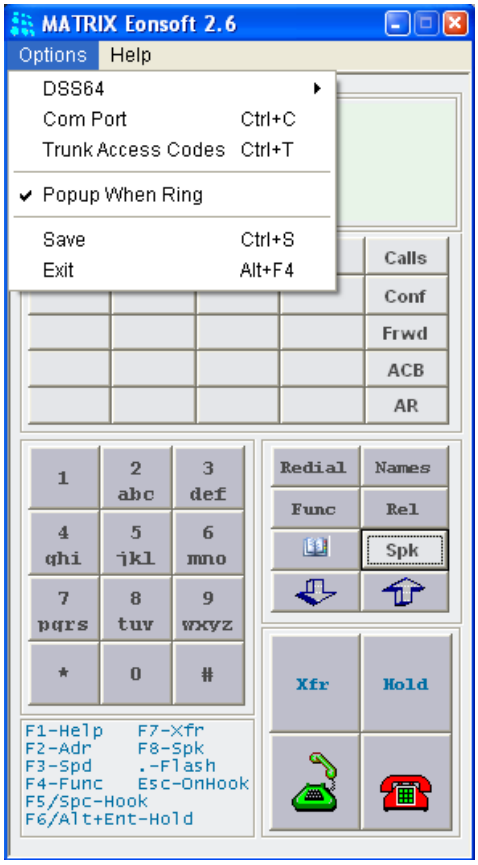
6. Copy the EONSOFT Application Software provided by the Support Team onto your PC and install the application.

7. After the program has been installed and run, a shortcut will be automatically created and appear on your desktop.

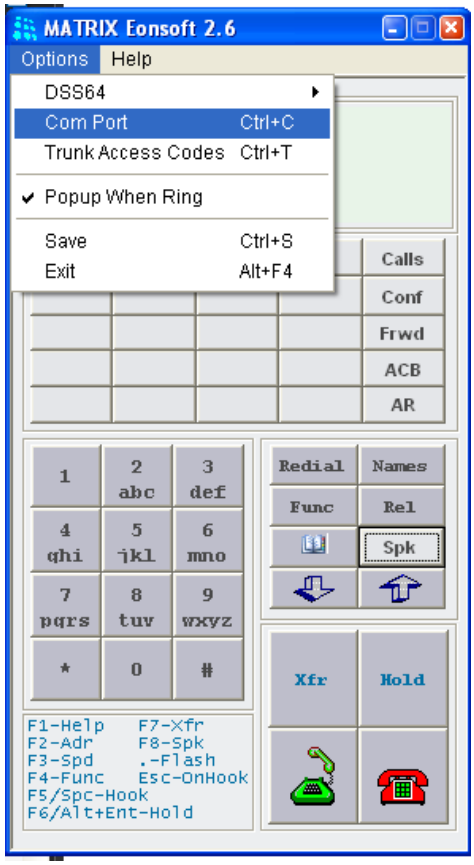
8. Click the shortcut to open the program. The EONSOFT window will open:



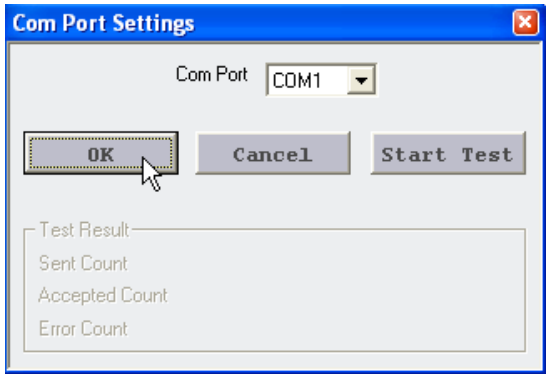
9. Click **Options** at the top left of the window. A drop down menu will appear.



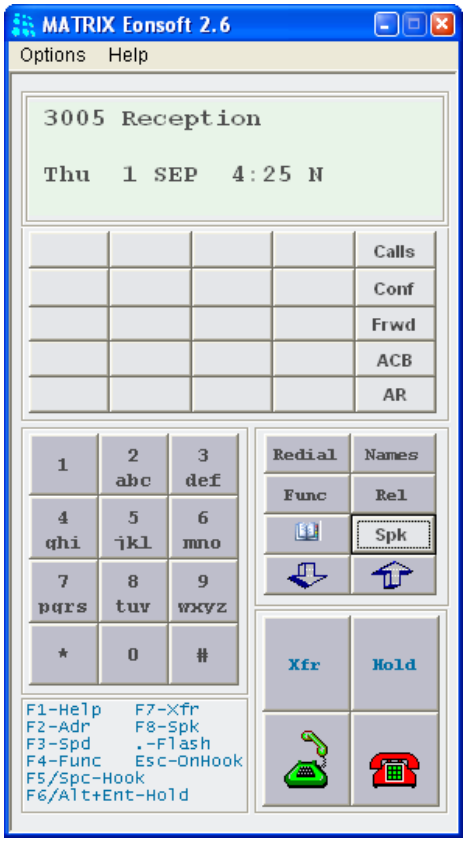
10. Click the option **COM Port**.



11. Select the COM Port to which the communication cable is connected.



12. EONSOFT is now connected. If you have already configured the DKP parameters like Access Code and Name for the port to which EONSOFT is connected, these will appear.



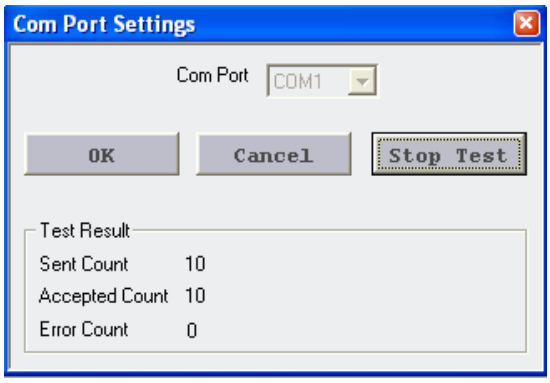
- *If this window does not appear after you have selected the COM Port Option, test the COM Port for data transfer.*
- *If the wrong COM port has been selected, a dialog box will pop up on your screen with the message: "COMx is invalid or busy, please select another COM Port". Select the correct COM Port.*

Test the functioning of the COM Port of the PC and the communication cable, before you install the EONSOFT.

Testing the COM Port

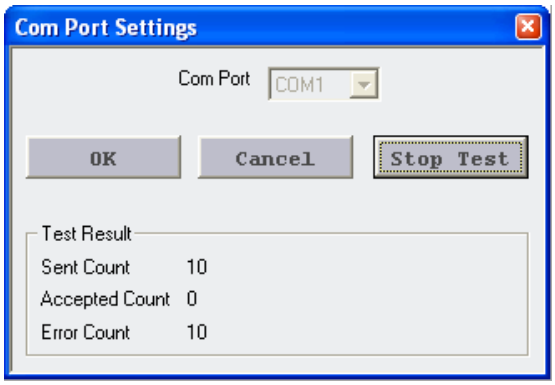
- From the drop down menu of **Options**, select the **COM Port** to which you have connected the communication cable.
- The COM Port Settings dialog box will open.
- Connect the communication cable to the COM Port of the PC.
- Short pin2 and pin3 of the DB-9 connector at the free end of the cable.
- Click the button labeled **Start Test** in the COM Port Settings dialog box.
- After clicking this button, observe the Test Result section on the dialog box.

- The **Error Count** value shows zero, if both the communication cable and the COM port are working.



The above screen shows that the COM Port/communication cable is working.

- If the **Error Count** shows a value other than zero, it means that either the communication cable or the COM port of the PC is faulty.



Above screen shows the faulty COM Port/Communication Cable.

- Remove the communication cable from the COM Port of the PC.
- Short pin2 and pin3 of the communication port of the computer and click 'Start Test' in the COM Port Settings dialog box.
- Now, if the error count is zero, please check the Communication Cable.
- If the error count is not a zero, the COM Port of the PC is faulty. Try another communication port.

The CO Card

The CO Card provides the interface to connect the system with the Two-Wire Analog Trunk lines from the CO Network. The CO Card supports the different standards and features of CO Networks across the world.

The CO Card is available in the following configurations. CO interface is also available in combination with SLT ports on a single card.

CO Cards for ETERNITY MENX

Card Name	Configuration and Application
ETERNITY ME Card CO16	16-port card to connect 16 Two-wire Trunk lines from the CO network
ETERNITY ME Card CO8	8-port card to connect 8 Two-wire Trunk lines from the CO network
ETERNITY ME Card CO8+SLT24	Combination card, with 8 CO ports to connect 8 CO analog trunk lines and 24 SLT ports to connect 24 Single Line Telephones This Card supports Power Fail Transfer. To know more, see “Power Fail Transfer”.

Choose a CO Card with the configuration that meets your requirement for CO trunk ports, keeping in mind the maximum CO Trunk Port capacity of the system you are installing.

The maximum CO Trunk Ports supported are 128.

Connectors

The CO Card has RJ45 connectors, with 4 CO ports on each connector. A multi-pair, MDF cable is supplied for each connector on the card.

LED

The CO Cards have a single tri-color LED to indicate:

- the health of the card during the Reset Cycle.
- the status of a selected Trunk port during normal functioning of the system.

You can assign the LED to any CO port on the card which you want to monitor[60].

Installing the CO Card

For CO connectivity, you must install at least one of the above mentioned CO Cards in the system.

1. Take all the necessary precautions prescribed for handling the cards and electronic equipment. Make sure that power supply is turned off before you begin the installation of the card. Put on an electrostatic-discharge preventive wrist strap/belt and use a grounding mat.

60. *To assign the LED to a selected port for monitoring its functioning, you must enter SE mode and dial the SE Command 7902-Slot-LED Number-Port, where Slot is the number of the universal slot in which the card is installed and Port is the port on the card to which the LED is to be assigned to monitor its functioning. LED Number is the number of the LED on the card, which will monitor the port.*

2. Unpack the CO card and check the package contents.

3. Select any free (empty) slot from the Universal Slots. Unscrew and remove the filler bracket of the empty slot. Preserve the filler bracket for future use!

4. Insert the CO Card into the guide rails of the free slot you selected for the card. The connectors on the card should make perfect contact with those of the slot on the backplane motherboard.

5. Press down the lever on the card mounting brackets to secure the card in its slot. Fix the mounting bracket in place with the two screws provided.

If installing more than one CO Card, it is not necessary to insert the other cards in subsequent slots. Any card can be inserted in any of the Universal Slots.

6. Use the cables supplied for each connector on the CO Card to connect the Trunk Lines with the Main Distribution Frame.

   Refer the illustrations below for the pinout details of the connectors on each card.

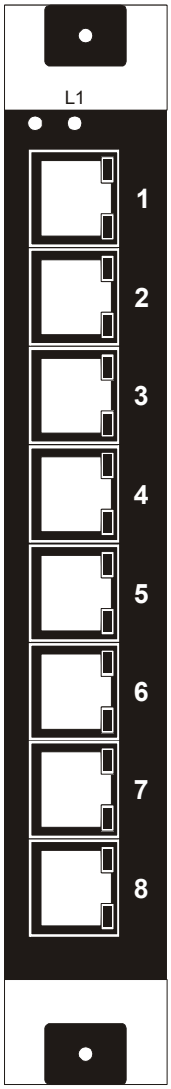
ETERNITY ME Card CO16

Connector	Color	Connection	H/w Port Offset
RJ45-1	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	CO CO CO CO	01 02 03 04
RJ45-2	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	CO CO CO CO	05 06 07 08
RJ45-3	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	CO CO CO CO	09 10 11 12
RJ45-4	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	CO CO CO CO	13 14 15 16

ETERNITY ME Card CO8

Connector	Color	Connection	H/w Port Offset
RJ45-1	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	CO CO CO CO	01 02 03 04
RJ45-2	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	CO CO CO CO	05 06 07 08
RJ45-3	Unused		
RJ45-4	Unused		

ETERNITY ME Card CO8+SLT24



Connector	Color	Connection	H/w Port Offset
RJ45-1	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	01 02 03 04
RJ45-2	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	05 06 07 08
RJ45-3	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	09 10 11 12
RJ45-4	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	13 14 15 16
RJ45-5	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	17 18 19 20
RJ45-6	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	21 22 23 24
RJ45-7	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	CO CO CO CO	01 02 03 04
RJ45-8	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	CO CO CO CO	05 06 07 08

7. Plug in the RJ45 end of the Trunk Card cables into the respective connectors.

8. Terminate the free end of the CO Card cable into the punch down blocks of the Krone modules designated for 'Trunk Lines' on “The Main Distribution Frame (MDF)”.

   Trunk cables from the system are to be connected with the Trunk Lines from the PSTN/CO terminated on the MDF. Each wire-pair form the CO Port must be terminated on the bottom of the Krone Connector, while the wire-pair of the trunk line from the CO Network to be connected to this port must be terminated on the top of the Krone Connector.

   Refer the topics “The Main Distribution Frame (MDF)” and “Terminating Trunk and Extension Cables on the MDF”.

9. Repeat these steps to install other CO Cards, if applicable.

10. If you have completed all other installation tasks, power ON the system.

LED Pattern of the CO Card

Stage	LED Color	Cadence
Auto Upgradation[a]		
Card waiting for application	RED	ON-200ms-OFF 200ms
Card is up, loaded with new application	GREEN	ON-200ms-OFF 200ms
Initialization		
	RED	ON 500ms-OFF 500ms
	GREEN	ON 500ms - OFF 500ms
	ORANGE	ON 500ms - OFF 500ms
Stand-by task	ORANGE	1 sec GREEN - 1Sec Orange
Errors		
Flash Failure	None	None
RAM Failure	None	None

a. Done by the boot loader application.

Status of Selected CO Port

PORT Status	LED Color	LED Cadence
Selected CO's data are transmitted to CPU Card	RED	Toggle[a] on each event
Selected CO's data are received from CPU Card	RED	Toggle[b] on each request from Master

a. The current LED state will remain the same until the next event is received from the application on the CO Port. For example, if the current LED state is Green/Red ON, on the next event, the LED will be turned OFF. It will remain OFF until the next event occurs. When the next event is received it will be turned Green/Red ON again. This process continues.

b. Same as above note.

Jumpers on the Main Board

Jumper Number	Position	Function
J1	AB (default)	Normal Operation
	BC	For uploading software using COM Port
J2 & J4	AB	NA
	BC (default)	Normal Operation/ Debug

The BRI Card

The BRI Card provides the interface to connect system with ISDN BRI Lines. The BRI lines may be from a public ISDN exchange, a private ISDN exchange.

BRI Cards of ETERNITY MENX

Card Name	Configuration and Application
ETERNITY ME Card BRI8	8-Port card to connect 8 ISDN BRI Lines or ISDN Compatible Devices
ETERNITY ME Card BRI4	4-Port card to connect 4 ISDN BRI Lines or ISDN Compatible Devices

The maximum number of BRI lines supported are 32.

Connectors

The BRI cards have RJ45 connectors. The ETERNITY ME BRI8 card has 8 RJ45 connectors for 8 BRI ports.

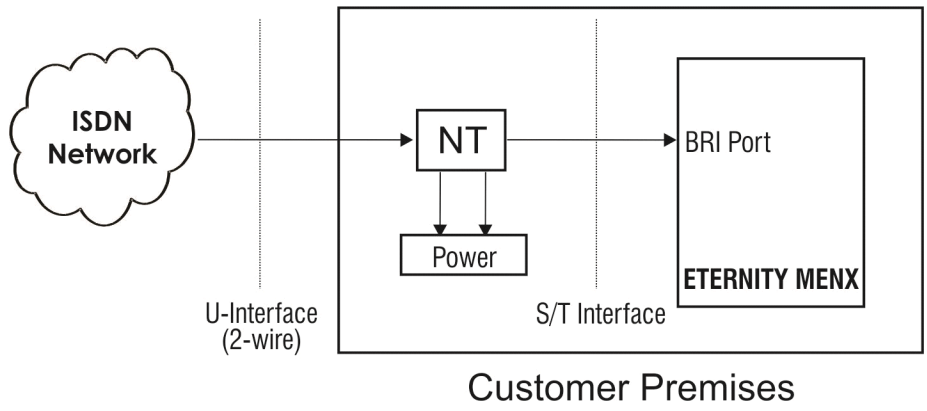
The ETERNITY ME BRI4 card has 4 RJ45 connectors for 4 BRI ports. A separate cable is supplied for each connector.

LEDs

The ETERNITY ME BRI8 and ETERNITY ME BRI4 has 4 LEDs.

ISDN BRI - Installation Scenarios

Most ISDN Service Providers also provide the NT1 device along with the BRI line. The BRI Line from the ISDN central office is terminated on the NT1 on the Customer's Premises, as illustrated below.



Where,

- U Interface = between the NT1 equipment and the ISDN central office.
- S/T Interface = between the ISDN user equipment, in this case, ETERNITY MENX and the Network Interface Equipment (NT1).

The BRI line is terminated on the NT1. The S/T interface of the NT1 is connected to BRI port of the ETERNITY MENX.

TE and NT Modes

In this illustration, the BRI line from ISDN Service Provider is directly connected to BRI port of the ETERNITY MENX via the NT1 device. Here, the ETERNITY MENX is the Terminal Equipment, so the BRI Port must be programmed to work in the TE mode.

When an ISDN Phone is to be connected to the BRI port of ETERNITY MENX, the BRI port must be programmed to work in NT mode.

When a BRI port of another ISDN System is to be connected to the BRI port of the ETERNITY MENX, in such a configuration, you may configure

- the BRI port of the other ISDN System in the TE mode and the BRI Port of the ETERNITY MENX in the NT mode.

  OR

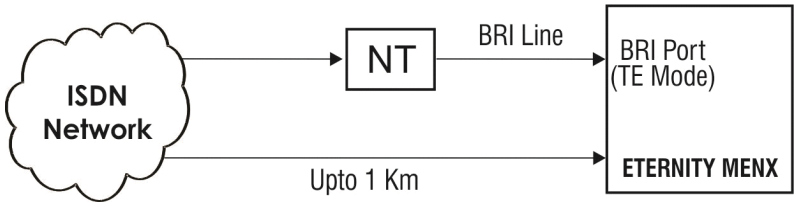
- the BRI port of the other ISDN System in the NT mode and the BRI Port of the ETERNITY MENX in the TE mode

Also refer the topic “Configuring BRI Trunks” to know more.

Types of BRI Configuration

There are two types of configurations in BRI: Point-to-Point Configuration and Point-to-Multipoint Configuration. Each of these is discussed below.

Point-to-Point Configuration



The maximum distance between the NT (Network Termination, NT1 or NT2) and a single Terminal Equipment, in this case ETERNITY MENX, can be up to 1 kilometer.

Point-to-Multipoint Configuration

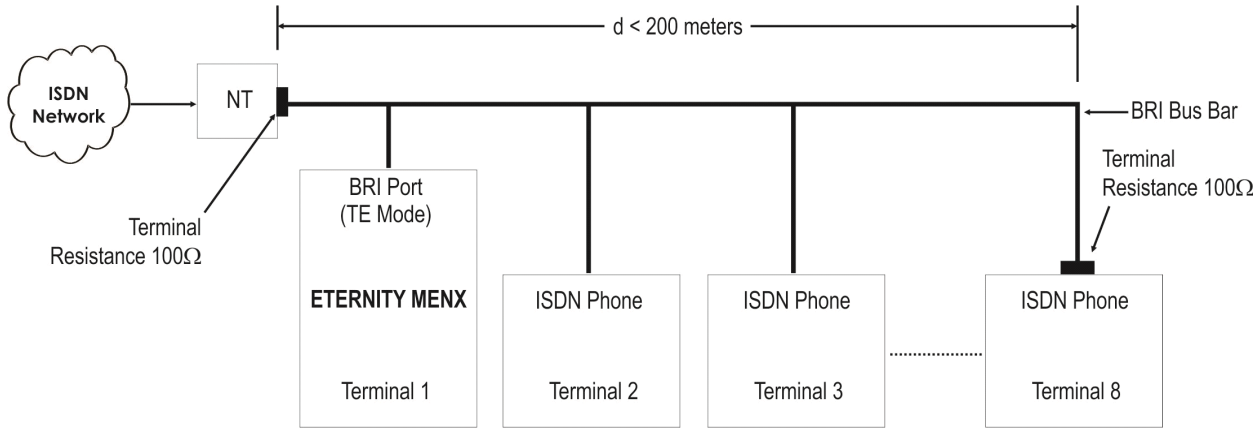
A maximum of 8 ISDN equipment can be connected on a single BRI Bus line in a Point-to-Multipoint configuration.

Further, two configurations are possible in a Point-to-Multipoint configuration:

a. Short Passive Bus Configuration

b. Extended Passive Bus Configuration

Short Passive Bus Configuration

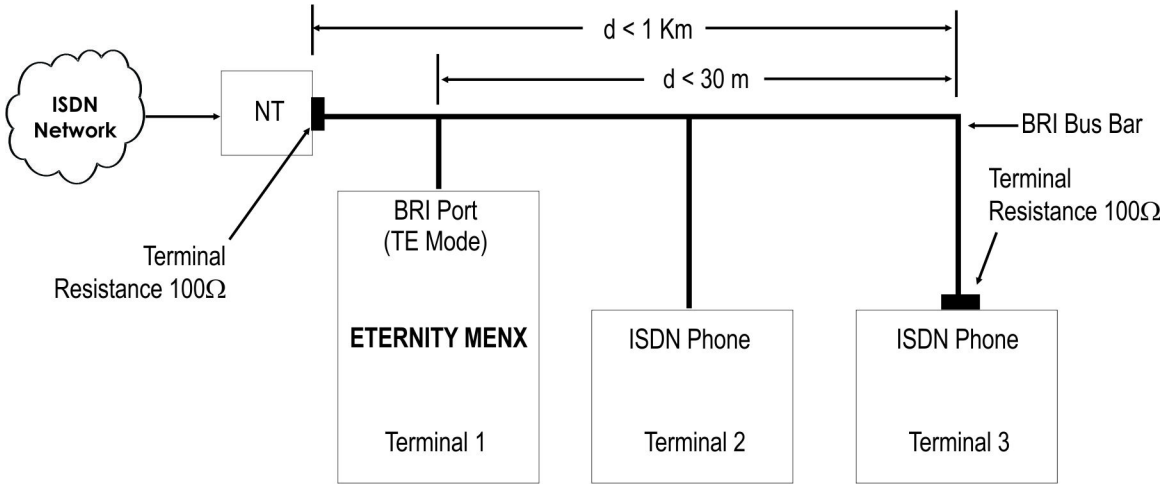


Where,
TE = Terminal Equipment or ISDN device (End user device)
NT = Network Termination provided by the ISDN Service Provider
d = distance from NT to the last TE equipment.

In a Short Passive Bus Configuration,

- A maximum of 8 TEs or ISDN devices can be connected to a single NT on a bus up to 200 meters from the NT.
- 100Ω Terminal Resistance is required to be inserted at the NT side as well as the last TE Equipment as shown in the figure.
- Using this configuration, any subscriber from ETERNITY MENX can access a BRI line and can make outgoing calls. At the same time, another subscriber from ETERNITY MENX or any ISDN phone shown in the figure can make outgoing call from the same BRI. In the same way, incoming calls are possible on the same BRI.
- Only two simultaneous speech paths can be established, as BRI supports 2 voice channels only.
- This configuration is useful on the smaller premises, where a single BRI line and multiple ISDN devices are used.

Extended Passive Bus Configuration



Where,
TE = Terminal equipment of any ISDN Equipment
NT = Network Termination provided by Service Provider
TR Terminal Resistance 100Ω
d = distance from NT to the last TE Equipment
d1 = the total distance from first TE equipment and the last TE equipment.

In an Extended Passive Bus Configuration,

- You can connect only 3 Terminal Equipment or ISDN devices. These devices are grouped together at one end of the bus, with may extend to a distance of up to 1 kilometer from the NT.
- However, all the 3 Terminal Equipment/ISDN devices must be located within a range of 30 meters, as shown in the figure.
- Using this configuration, any subscriber from ETERNITY MENX can access the BRI line and make outgoing calls. At the same time, another subscriber from the ETERNITY MENX or any ISDN phone shown in the figure can make outgoing calls from the same BRI. In the same way, incoming calls are possible on the same BRI.
- Only two simultaneous speech paths can be established, as BRI supports 2 voice channels only.
- This configuration is useful on large premises where a limited number of ISDN devices (maximum 3) are to be used within a range of 30 meters.

Installing the BRI Card

1. Take all the necessary precautions prescribed for handling the cards and electronic equipment: turn off power supply, always wear an electrostatic-discharge preventive wrist strap/belt and use a grounding mat.
2. Unpack the BRI Card and check the package contents.
3. Select any free (empty) slot from the Universal Slots. Unscrew and remove the filler bracket of the empty slot. Do not discard the filler bracket! Preserve it for future use!

Setting Orientation Type of BRI Port

4. The BRI Ports can be configured for different applications and can be interfaced directly with the BRI Network with Terminal Equipment like an ISDN Phone, with an ISDN-System.

   To connect the BRI Port to the public network, BRI Port must configured in the TE mode.

   To connect ISDN phones, an ISDN System or any ISDN equipment, the BRI Port must be configured in the NT mode.

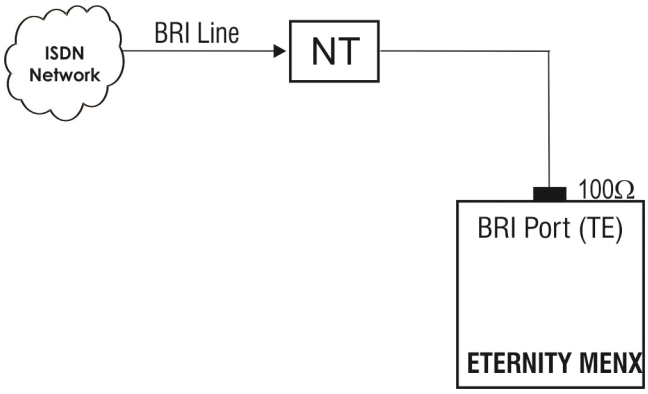
   By default, BRI Ports are configured in the TE mode.

   To set Orientation Type of the BRI Port, under **Configuration**, open the **BRI Configuration** link, and under **BRI Parameters**, set the **Orientation Type**.

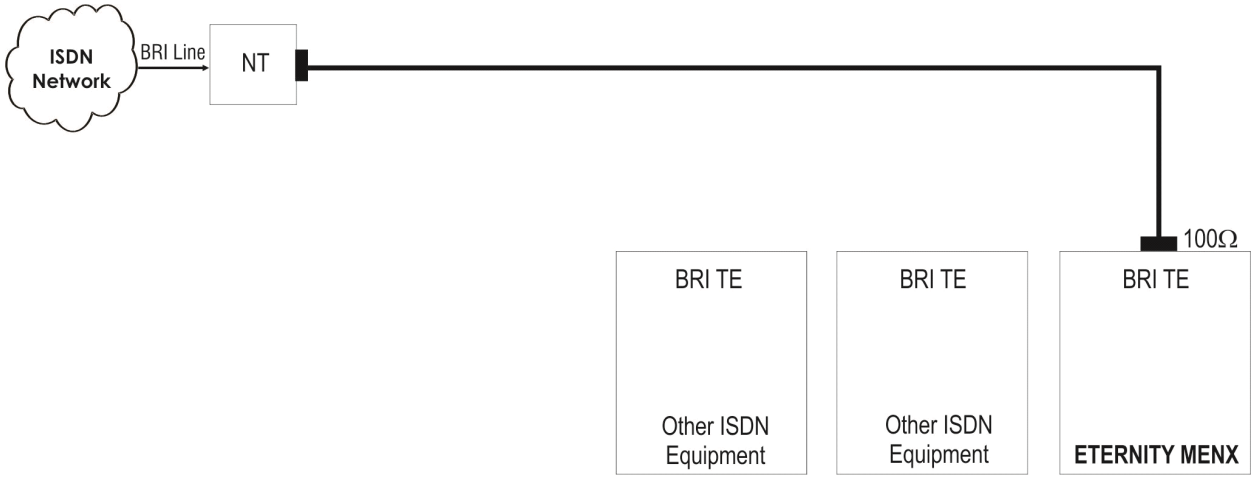
Inserting Termination Resistance

5. Termination of 100Ω should be inserted in the following cases:

   - When the BRI port is configured in the TE mode and connected in a Point-to-Point configuration as shown below.



   - When the BRI port is configured in the TE mode in a Point-to-Multipoint configuration as shown below. 100Ω Termination is required on the last Terminal connected on the S0 bus to terminate calls properly.



In a Point-to-Multipoint configuration, 100Ω termination can be provided on either of the following:
- Last TE equipment
- Last point of the bus bar where the last TE equipment is connected.

- When BRI port is configured in the NT mode.

- *If the S0 bus itself supports Terminating resistors, Termination Resistance need not be inserted when*
  - *BRI Port is configured as TE and connected in a Point-to-Point Configuration as illustrated above.*
  - *BRI Port is configured as NT.*

- *Termination need not be inserted if the BRI port of ETERNITY MENX (configured in TE mode) is connected as any terminal other than the last terminal on the S0 bus (in a Multi-point configuration).*

Termination in TE Equipment (BRI Port)

6. To set the 100Ω termination on the BRI port set the Jumpers on the BRI Module (daughter-board) to the position described below:

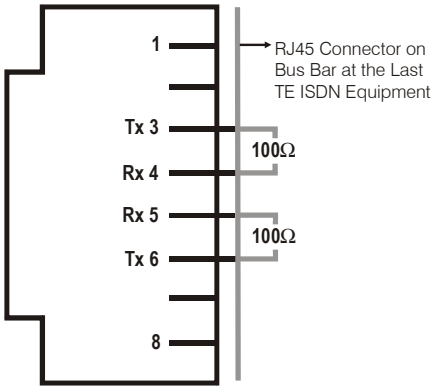
Function	**Module 2 (M2)**				**Module 3 (M3)**			
	BRI Port 1		BRI Port 2		BRI Port 3		BRI Port 4	
	Jumper Position		Jumper Position		Jumper Position		Jumper Position	
	J6	J8	J7	J9	J6	J8	J7	J9
To insert 100Ω termination	AB	AB	AB	AB	AB	AB	AB	AB
To remove 100Ω termination	BC	BC	BC	BC	BC	BC	BC	BC

Function	**Module 4 (M4)**				**Module 5 (M5)**			
	BRI Port 5		BRI Port 6		BRI Port 7		BRI Port 8	
	Jumper Position		Jumper Position		Jumper Position		Jumper Position	
	J6	J8	J7	J9	J6	J8	J7	J9
To insert 100Ω termination	AB	AB	AB	AB	AB	AB	AB	AB
To remove 100Ω termination	BC	BC	BC	BC	BC	BC	BC	BC

By default, Termination Resistance of 100Ω is set on the BRI port (the Jumpers are in AB position).

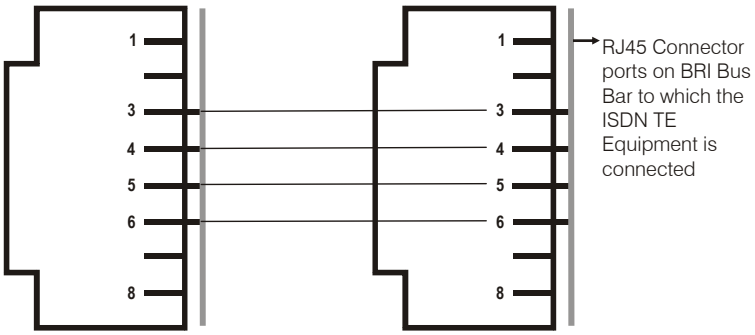
Termination in the Bus Bar

7. 100Ω termination resister can be connected between TX and RX, between pin number 3-4 and 5-6 in the RJ45 connector as illustrated below.



As shown in the application diagrams for Point-to-Multipoint connectivity, each ISDN TE device is connected in a Bus Bar, which may be Short Passive Bus Bar configuration or an Extended Passive Bus Bar configuration.

Illustrated below is the connection diagram of two ports connected with each other on the same BRI bus bar.



- The above figure shows the connection details of two ports on the BRI Bus Bar. Similarly, you can connect 8 ports on the Bus Bar, keeping in mind the Termination Resister for the NT and the Last TE on the Bus bar.
- Pin number 3, 4, 5 and 6 of the RJ45 connector are used for connectivity.
- Pin number 3 and 6 are used for Transmit (Tx) and pin number 4 and 5 are used for Receive (Rx) from the ISDN TE side.
- Pin number 3 and 6 are used for Receive (Rx) and pin number 4 and 5 are used for Transmit (Tx) from the NT side.

Feeding Power to the Terminal

8. When the BRI Port of the ETERNITY MENX is used as BRI-NT, you can feed power to the terminal equipment connected to the BRI-NT Port from the ETERNITY MENX.

   To do this,

   - Enable Feed Power on the BRI Port. For instructions see Power Feed under “Configuring BRI Trunks”.

- By default, the Jumpers are set in AB position to feed power through Tx and Rx wires (Phantom Power). If you want to feed power through a separate pair of wires, you may change the position of the Jumpers on the BRI module as mentioned in the table below.

Function	**Module 2 (M2)**				**Module 3 (M3)**			
	BRI Port 1		BRI Port 2		BRI Port 3		BRI Port 4	
	Jumper Position		Jumper Position		Jumper Position		Jumper Position	
	J4	J5	J2	J3	J4	J5	J2	J3
To feed power on Tx and Rx wires (Phantom Power)	AB	AB	AB	AB	AB	AB	AB	AB
To feed power on separate pair of wires	BC	BC	BC	BC	BC	BC	BC	BC

Function	**Module 4 (M4)**				**Module 5 (M5)**			
	BRI Port 5		BRI Port 6		BRI Port 7		BRI Port 8	
	Jumper Position		Jumper Position		Jumper Position		Jumper Position	
	J4	J5	J2	J3	J4	J5	J2	J3
To feed power on Tx and Rx wires (Phantom Power)	AB	AB	AB	AB	AB	AB	AB	AB
To feed power on separate pair of wires	BC	BC	BC	BC	BC	BC	BC	BC

- *The maximum power that can be fed to a single BRI port is 50mA.*
- *From signaling point of view, a maximum of 8 terminal equipment can be connected on the BRI port configured in the NT mode.*
- *The number of ISDN Terminals that can be connected on the BRI port configured in the NT mode depends on the power consumed by the ISDN terminals.*

9. Insert the BRI Card into the guide rails of the free slot you selected for the card. The connectors on the card should make perfect contact with those of the slot on the backplane motherboard.

Press down the lever on the card mounting brackets to secure the card in its slot. Fix the mounting bracket in place with the two screws provided.

If installing more than one BRI Card, it is not necessary to insert the other cards in subsequent slots. Any card can be inserted in any of the Universal Slots. Remember to set the Orientation Type, Termination Resistance and Power Feed, as required.

10. Use the straight cables supplied for each connector on the BRI card to connect the BRI Ports to the NT1 device supplied by your ISDN service provider. Refer the configuration and pinout details given below for guidance.

Configuration details of the U interface (RJ-45) on NT1

Pin Number	Pin Details
4	Tx
5	Rx

Configuration details of the S/T interface (RJ-45) on NT1

Pin Number	Pin Details
3	Rx1
4	Tx1
5	Tx2
6	Rx2

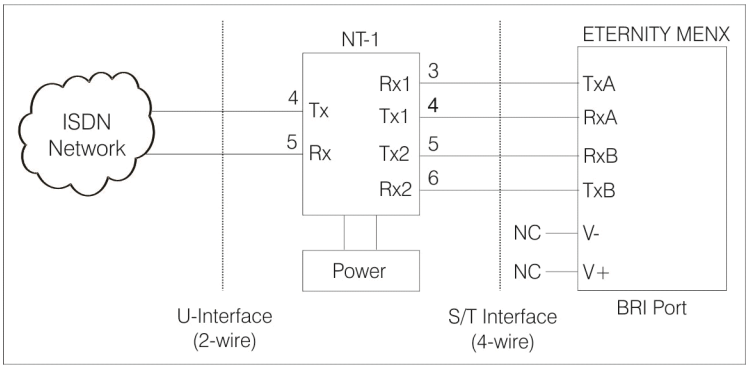
Pinout and Cable Details of BRI4 Port in TE Mode

Pin	Color	Connection
1	Orange-White	Not connected
2	Orange	Not connected
3	Green-White	TxA
4	Blue	RxA
5	Blue-White	RxB
6	Green	TxB
7	Brown-White	V-
8	Brown	V+

Pinout and Cable Details of BRI4 Port in NT Mode

Pin	Color	Connection
1	Orange-White	Not connected
2	Orange	Not connected
3	Green-White	RxA
4	Blue	TxA
5	Blue-White	TxB
6	Green	RxB
7	Brown-White	V-
8	Brown	V+

The following diagram shows how to connect a BRI Line to the BRI port in the TE mode.



11. If you have completed all other installation tasks, you may turn ON the system and observe the Reset Cycle and the LED pattern of the BRI Card.

LED Pattern of the BRI Card

- The BRI8 Card has 8 LEDs: L1[61], L2, L3, L4, L5[62], L6, L7 and L8. These display the status of each port.
- The BRI4 Card has 4 LEDs: L1[63], L2, L3 and L4. These display the status of each port.

  The LEDs show the Status of the Ports as summarized in the table below:

Port Status	LED Color	LED Cadence
Port is not active	RED	Continuously ON
Port is active	GREEN	Continuously ON

61. *This LED keeps blinking. It displays the system heart bits, at the rate of 100ms. It will remain OFF for 100ms and will show the status of port 1 for the next 100ms.*
62. *This LED keeps blinking. It displays the system heart bits, at the rate of 100ms. It will remain OFF for 100ms and will show the status of port 1 for the next 100ms.*
63. *This LED keeps blinking. It displays the system heart bits, at the rate of 100ms. It will remain OFF for 100ms and will show the status of port 1 for the next 100ms.*

The T1E1PRI Card

The ETERNITY ME T1E1PRI Card provides the interface to connect the system to ISDN Network.

When connected to T1 carrier lines, the Card supports the following signaling types:

- PRI
- Robbed Bit Signaling
- Q-Signaling (QSIG)
- E&M

When connected to E1 carrier lines, the Card supports the following signaling types:

- PRI
- Channel Associated Signaling (CAS)
- Q-Signaling (QSIG)
- E&M

The T1E1PRI Card is available in the following configurations:

T1E1PRI Card for ETERNITY MENX

Card Name	Configuration and Application
ETERNITY ME Card T1E1PRI Dual	2-Port card with QSIG support to connect 2 ISDN T1/E1 PRI Lines or ISDN Compatible Devices
ETERNITY ME Card T1E1PRI Single	1-Port card with QSIG support to connect 1 ISDN T1/E1 PRI Line or ISDN Compatible Device

The maximum number of PRI Lines supported are 24.

Connectors

The T1E1PRI card has an RJ45 Connector for each port. The ETERNITY ME T1E1PRI Dual card has 2 RJ45 Connectors for the two ports, while the ETERNITY ME T1E1PRI Single card has a single RJ45 Connector.

A cable with RJ45 plugs on both ends is supplied for each connector.

LEDs

The ETERNITY ME T1E1PRI Dual Card has four LEDs: L1, L2, L3 and L4.
The ETERNITY ME T1E1PRI Single Card has two LEDs L1 and L2.

Installing the T1E1PRI Card

1. Before installing the card, take the necessary precautions prescribed for handling the cards. Always wear an electrostatic-discharge preventive wrist strap and use a grounding mat. Make sure the power supply is turned off.

2. Unpack the T1E1PRI Card and check the package contents.

3. Select any free (empty) slot from the Universal Slots. Unscrew and remove the filler bracket of the empty slot. Do not discard the filler bracket.

Setting Line Termination Resistor

4. The default positions of SW3, SW4, SW6, SW7 switches should be as follows:

Pin1	Pin2	Pin3	Pin4	Termination Resistance (Ω)
OFF	OFF	OFF	ON	0 (Default)

Do Not Change the positions of any of these switches!

5. By default, termination resistance of PRI port is set as 120Ω, which is for E1 connectivity.

   - To use the PRI Port for T1 connectivity, termination resistance must be changed to 100Ω.

   - Use DIP Switch SW5 to change the Termination Resistance of PRI Port 1. Set the Pins of SW5 as shown below:

Pin-1	Pin-2	Pin-3	Pin-4	Resistance
OFF	OFF	ON	OFF	120Ω (for E1)
OFF	ON	OFF	OFF	100Ω (for T1)

   - If using the ETERNITY ME T1E1PRI Dual Card, use DIP Switch SW2 to change the Termination Resistance of PRI Port 2. Set the Pins of SW2 as shown below:

Pin-1	Pin-2	Pin-3	Pin-4	Resistance
OFF	OFF	ON	OFF	120Ω (for E1)
OFF	ON	OFF	OFF	100Ω (for T1)

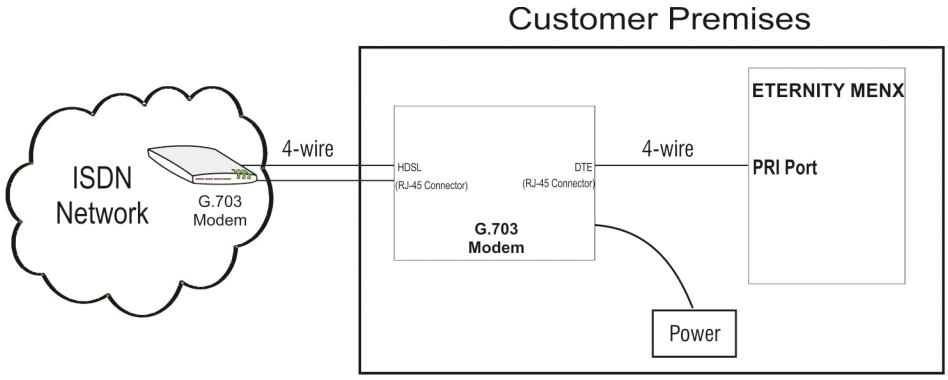
6. Insert the T1E1PRI Card into the guide rails of the free slot you selected for the card. Make sure that the connectors on the card make perfect contact with those of the slot on the backplane motherboard.

7. Now, press down the levers on the card mounting brackets to secure the card in its slot. Fix the card in place with the two screws provided.

Connecting ISDN T1/E1 PRI Lines

8. Use the cable supplied with the T1E1PRI Card to connect the system to the T1/E1 PRI network interface equipment (modem), which is usually supplied by your ISDN Service Provider along with the PRI line.

The diagram below illustrates this.



- Most Service Providers insist on connecting an ISDN modem at both the ends of the PRI line—one at the Local Exchange and other at the Customer's Premises.
- At the Customer's Premises, the PRI line is terminated on the HDSL interface of the modem.
- The DTE interface of the modem is to be connected to the PRI port (RJ-45 connector on the ETERNITY ME T1E1PRI Card).

9. Plug in one end of the RJ45 cable supplied with the card into the card's connector. Plug the other end of the RJ45 cable into the Network Termination Unit.

10. Refer the following pin details for connecting the Network Termination Unit with the system.

Pin details of HDSL Interface of the G.703 Modem. (HDSL Network Termination Unit)

Pin Number	Pin Details
1	Line A
2	Line A
3	Not used
4	Line B
5	Line B
6	Not used
7	Not used
8	Not used

Pin details of DTE Interface of G.703 Modem. (HDSL Network Interface Unit)

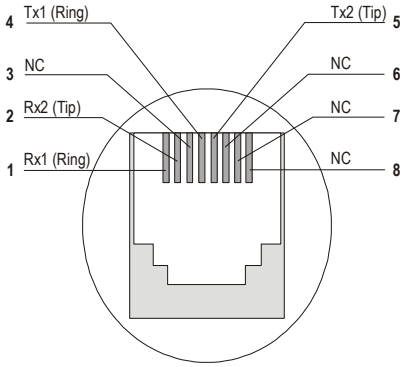
Pin Number	Pin Details
1	TX1 (Tip)
2	TX2 (Ring)
3	Not used
4	RX1 (Ring)

Pin Number	Pin Details
5	RX2 (Tip)
6	Not used
7	Not used
8	Not used

Most of the HDSL Network Termination Unit manufacturers use these connectors. But you are advised to read the installation guide of the HDSL Network Termination Unit being used by you.

Pin details of ETERNITY MENX T1E1PR1 Port

The T1E1PRI Port of the system terminates in an 8-pin RJ45, female connector and is wired according to the table below.



The cable wires may have to be crossed depending on the pinout of the DTE Interface of the modem.

11. Repeat the same steps to install another card. It is not necessary to install the other T1E1PRI Cards in a sequence. Any card can be installed in any of the slots.

12. If you have completed all other installation tasks. Power the system. After the Reset Cycle, observe the LED patterns of the T1E1PRI Card.

LED Patterns

The ETERNITY ME T1E1PRI Dual Card has four LEDs: L1, L2, L3 and L4.

- L1 and L2 are assigned to PRI Port 1 (PRI#1)
- L3 and L4 are assigned to PRI Port 2 (PR1#2)
- L1[64] shows Card Heart Bit as well as status of Port 1.

The ETERNITY ME T1E1PRI Single Card has two LEDs: L1 and L2.

Given below are the LED Patterns defined for indicating port states in the signaling types supported by the ETERNITY MENX.

64. This LED keeps blinking. It displays the system heart bits, at the rate of one second. It will remain OFF for one second and will show the status of port 1 for the next one second.

2. Port Active Mode

Signaling Type: E1-PRI

LED1/LED3 Pattern:

Port Status	Color	Cadence
Layer 1 established successfully	GREEN	Continuous ON
CRC4 Alarm	GREEN	100ms ON-100 ms OFF
BFA Alarm	RED	500ms ON-500 ms OFF
LOS Alarm	RED	Continuous ON

LED2/LED4 Pattern:

Port Status	Color	Cadence
Layer 1 established successfully	GREEN	Continuous ON
RAI Alarm	RED	500ms ON-500 ms OFF
AIS or LOS Alarm	RED	Continuous ON

Signaling Type: E1-CAS

LED1/LED3 Pattern:

Port Status	Color	Cadence
Layer 1 established successfully	GREEN	Continuous ON
CRC4 Alarm	GREEN	100ms ON-100 ms OFF
MFA Alarm	RED	100ms ON-100 ms OFF
BFA Alarm	RED	500ms ON-500ms OFF
LOS Alarm	RED	Continuous ON

LED2/LED4 Pattern:

Port Status	Color	Cadence
Layer 1 established successfully	GREEN	Continuous ON
Y-Bit Alarm	GREEN	100ms ON-100 ms OFF
AIS16 Alarm	RED	100ms ON-100 ms OFF
RAI Alarm	RED	500ms ON-500 ms OFF
AIS or LOS Alarm	RED	Continuous ON

Signaling Type: T1-RBS or T1-PRI

LED1/LED3 Pattern:

Port Status	Color	Cadence
No Alarm	GREEN	Continuous ON
BFA Alarm or MFA Alarm	RED	500ms ON-500 ms OFF
AIS Alarm	RED	100ms ON-100 ms OFF
LOS Alarm	RED	Continuous ON

LED2/LED4 Pattern:

Port Status	Color	Cadence
Layer 1 established successfully	GREEN	Continuous ON
RAI or LOS Alarm	RED	Continuous ON

2. Port Maintenance Mode

LED1/LED3 Pattern:

Port Status	Color	Cadence
Maintenance Mode	RED -GREEN	500 ms RED-500 ms GREEN

LED2/LED4 Pattern:

Port Status	Color	Cadence
Near end loop back wait before activate	RED	100ms ON-100 ms OFF
Near end loop active	RED	Continuous ON
Near end loop back wait before deactivate	RED	500ms ON-500 ms OFF
Far end loop back wait after activate	GREEN	100ms ON-100 ms OFF
Far end loop active	GREEN	Continuous ON
Far end loop back wait after deactivate	GREEN	500ms ON-500 ms OFF

3. Port Disable Mode

LED1/LED3 Pattern:

Port Status	Color	Cadence
Port Disable	RED	Continuous ON

LED2/LED4 Pattern:

Port Status	Color	Cadence
Port Disabled	OFF	OFF

Jumpers on the Main Board

Jumper Number	Position	Function
J5[a]	AB	Dual T1E1
	BC	Single T1E1
J6	AB	JTAG Mode
	BC (default)	Embedded ICE
J7	AB	Internal Boot
	BC (default)	External Boot
J8 & J12	AB (default)	Debug
	BC	UART
J9, J10 & J11	BC (default)	Normal
	AB	Boot

a. In T1E1 PRI Dual Card, default jumper position will be AB. In T1E1 PRI Single Card, default jumper position will be BC.

The E1FO Card

The ETERNITY LE E1FO Card provides the interface to connect the ETERNITY to the ISDN PRI Network. For E1 carrier lines the card supports the signaling types PRI, Q-Sig and CAS. For T1 carrier lines the card supports the signaling types PRI, Q-Sig and Robbed Bit Signaling. The T1/E1 ports can be set in Terminal or Network mode.

The E1FO Card supports Copper and Fiber Optic (FO) interfaces. At a time, either the Copper interface or the FO interface can be used. The FO interface supports only Single-mode (Mono mode) fiber connectivity and will work within a range of 30km.

E1 connectivity is supported over, the Copper interface as well as the Fiber Optic interface. The T1 connectivity is supported over the Copper interface only.

The E1FO Card is available in the following configurations:

E1FO Card for ETERNITY MENX

Card Name	Configuration and Application
ETERNITY ME Card E1FO Dual	2-Port card to connect 2 ISDN T1/E1 Lines.
ETERNITY ME Card E1FO Single	1-Port card to connect 1 ISDN T1/E1 Line.

Connectors

The E1FO Dual card has two RJ45 Connectors and two Fiber Optic connectors. The E1FO Single card has one RJ45 Connector and one Fiber Optic connector.

LEDs

The ETERNITY ME E1FO Dual Card has four LEDs: L1, L2, L3 and L4.
The ETERNITY ME E1FO Single Card has two LEDs L1 and L2.

Installing the E1FO Card

1. Before installing the card, take the necessary precautions prescribed for handling the cards. Always wear an electrostatic-discharge preventive wrist strap and use a grounding mat. Make sure the power supply is turned off.

2. Unpack the E1FO Card and check the package contents.

3. Select any free (empty) slot from the Universal Slots. Unscrew and remove the filler bracket of the empty slot. Do not discard the filler bracket.

Setting Line Termination Resistor

4. The default positions of SW3, SW4, SW6, SW7 switches should be as follows:

Pin1	Pin2	Pin3	Pin4	Termination Resistance (Ω)
OFF	OFF	OFF	ON	0 (Default)

Do Not Change the positions of any of these switches!

5. By default, termination resistance of PRI port is set as 120Ω, which is for E1 connectivity.

   - To use the PRI Port for T1 connectivity, termination resistance must be changed to 100Ω.

   - Use DIP Switch SW5 to change the Termination Resistance of PRI Port 1. Set the Pins of SW5 as shown below:

Pin-1	Pin-2	Pin-3	Pin-4	Resistance
OFF	OFF	ON	OFF	120Ω (for E1)
OFF	ON	OFF	OFF	100Ω (for T1)

   - If using the ETERNITY ME E1FO Dual Card, use DIP Switch SW2 to change the Termination Resistance of PRI Port 2. Set the Pins of SW2 as shown below:

Pin-1	Pin-2	Pin-3	Pin-4	Resistance
OFF	OFF	ON	OFF	120Ω (for E1)
OFF	ON	OFF	OFF	100Ω (for T1)

6. Insert the E1FO Card into the guide rails of the free slot you selected for the card. Make sure that the connectors on the card make perfect contact with those of the slot on the backplane motherboard.

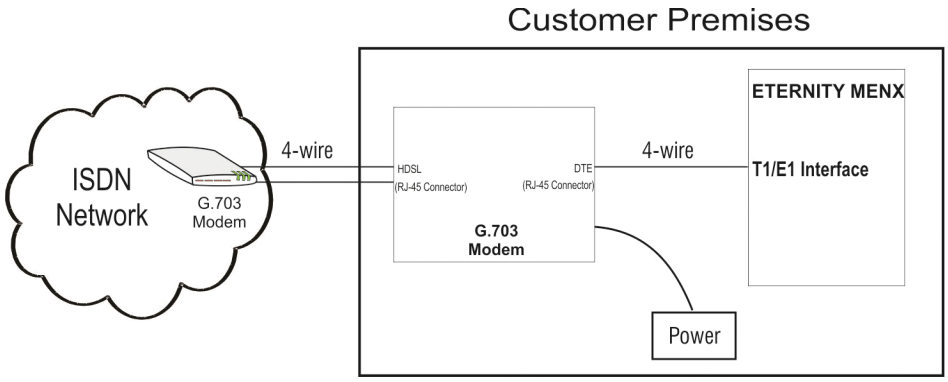
7. Now, press down the levers on the card mounting brackets to secure the card in its slot. Fix the card in place with the two screws provided.

Connecting the T1/E1 Lines

Copper Interface Connectivity

8. Use the cable to connect the T1/E1 Port to the T1/E1 network interface equipment (modem), which is usually supplied by your ISDN Service Provider along with the PRI line.

The diagram below illustrates this.



- Most Service Providers insist on connecting an ISDN modem at both the ends of the PRI line—one at the Local Exchange and other at the Customer's Premises.
- At the Customer's Premises, the PRI line is terminated on the HDSL interface of the modem.
- The DTE interface of the modem is to be connected to the PRI port (RJ45 connector on the Matrix ETERNITY ME CARD E1FOPRI Single).

9. Plug in one end of the RJ45 cable into the card's connector. Plug the other end of the RJ45 cable into the Network Termination Unit.

10. Refer the following pin details for connecting the Network Termination Unit with the system.

Pin details of HDSL Interface of the G.703 Modem. (HDSL Network Termination Unit)

Pin Number	Pin Details
1	Line A
2	Line A
3	Not used
4	Line B
5	Line B
6	Not used
7	Not used
8	Not used

Pin details of DTE Interface of G.703 Modem. (HDSL Network Interface Unit)

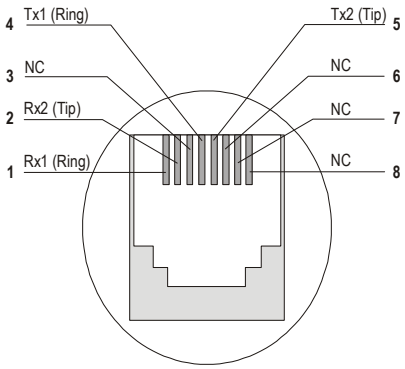
Pin Number	Pin Details
1	TX1 (Tip)
2	TX2 (Ring)
3	Not used
4	RX1 (Ring)

Pin Number	Pin Details
5	RX2 (Tip)
6	Not used
7	Not used
8	Not used

Most of the HDSL Network Termination Unit manufacturers use these connectors. But you are advised to read the installation guide of the HDSL Network Termination Unit being used by you.

Pin details of the T1/E1 PRI Interface

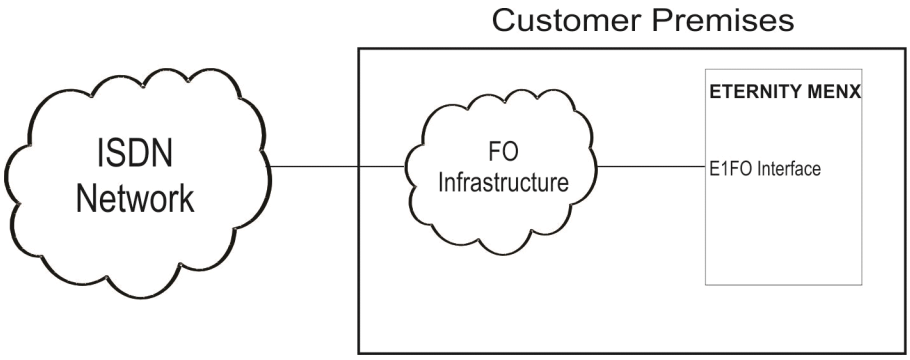
The T1/E1 Interface of the ETERNITY ME CARD E1FOPRI Single terminates in an 8-pin RJ45, female connector and is wired according to the table below.



The cable wires may have to be crossed depending on the pinout of the DTE Interface of the modem.

Fiber Optic Interface Connectivity

- *Fiber Optic interface supports E1 connectivity only.*
- *The Fiber Option (FO) interface supports only Single-mode (Mono mode) fiber connectivity and will work within a range of 30km.*



11. If your service provider provides Fiber Optic connectivity or you have an existing Fiber Optic infrastructure, you can connect the FO line to the E1FO interface of the E1FO Card.

    To use Fiber Optic interface, make sure you select **Line Coding Mechanism** as **NRZ (Fiber Optic)**. For details, see “Configuring E1 Trunks”.

12. Repeat the same steps to install another card. It is not necessary to install the other E1FO Cards in a sequence. Any card can be installed in any of the slots.

13. If you have completed all other installation tasks. Power the system. After the Reset Cycle, observe the LED patterns of the E1FO Card.

LED Patterns

The ETERNITY ME E1FO Dual Card has four LEDs: L1, L2, L3 and L4.

- L1 and L2 are assigned to PRI Port 1 (PRI#1)
- L3 and L4 are assigned to PRI Port 2 (PR1#2)
- L1[65] shows Card Heart Bit as well as status of Port 1.

The ETERNITY ME E1FO Single Card has two LEDs: L1 and L2.

Given below are the LED Patterns defined for indicating port states in the signaling types supported by the ETERNITY MENX.

1. Port Active Mode

Signaling Type: E1-PRI

LED1/LED3 Pattern:

Port Status	Color	Cadence
Layer 1 established successfully	GREEN	Continuous ON
CRC4 Alarm	GREEN	100ms ON-100 ms OFF
BFA Alarm	RED	500ms ON-500 ms OFF
LOS Alarm	RED	Continuous ON

LED2/LED4 Pattern:

Port Status	Color	Cadence
Layer 1 established successfully	GREEN	Continuous ON
RAI Alarm	RED	500ms ON-500 ms OFF
AIS or LOS Alarm	RED	Continuous ON

65. *This LED keeps blinking. It displays the system heart bits, at the rate of one second. It will remain OFF for one second and will show the status of port 1 for the next one second.*

Signaling Type: E1-CAS

LED1/LED3 Pattern:

Port Status	Color	Cadence
Layer 1 established successfully	GREEN	Continuous ON
CRC4 Alarm	GREEN	100ms ON-100 ms OFF
MFA Alarm	RED	100ms ON-100 ms OFF
BFA Alarm	RED	500ms ON-500ms OFF
LOS Alarm	RED	Continuous ON

LED2/LED4 Pattern:

Port Status	Color	Cadence
Layer 1 established successfully	GREEN	Continuous ON
Y-Bit Alarm	GREEN	100ms ON-100 ms OFF
AIS16 Alarm	RED	100ms ON-100 ms OFF
RAI Alarm	RED	500ms ON-500 ms OFF
AIS or LOS Alarm	RED	Continuous ON

Signaling Type: T1-RBS or T1-PRI

LED1/LED3 Pattern:

Port Status	Color	Cadence
No Alarm	GREEN	Continuous ON
BFA Alarm or MFA Alarm	RED	500ms ON-500 ms OFF
AIS Alarm	RED	100ms ON-100 ms OFF
LOS Alarm	RED	Continuous ON

LED2/LED4 Pattern:

Port Status	Color	Cadence
Layer 1 established successfully	GREEN	Continuous ON
RAI or LOS Alarm	RED	Continuous ON

2. Port Maintenance Mode

LED1/LED3 Pattern:

Port Status	Color	Cadence
Maintenance Mode	RED -GREEN	500 ms RED-500 ms GREEN

LED2/LED4 Pattern:

Port Status	Color	Cadence
Near end loop back wait before activate	RED	100ms ON-100 ms OFF
Near end loop active	RED	Continuous ON
Near end loop back wait before deactivate	RED	500ms ON-500 ms OFF
Far end loop back wait after activate	GREEN	100ms ON-100 ms OFF
Far end loop active	GREEN	Continuous ON
Far end loop back wait after deactivate	GREEN	500ms ON-500 ms OFF

3. Port Disable Mode

LED1/LED3 Pattern:

Port Status	Color	Cadence
Port Disable	RED	Continuous ON

LED2/LED4 Pattern:

Port Status	Color	Cadence
Port Disabled	OFF	OFF

The E&M Card

The E&M Card of the system provides the interface for analog trunking to connect various communication equipment telephone switches, Routers, Leased Lines, etc. using Tie-Lines.

The E&M Card is required for the following applications:

- Power Line Carrier Communication (PLCC) Networks, where several systems are connected with each other through E&M tie lines. Refer "PLCC-An Introduction" to know more.
- "Closed User Group (CUG)", where several systems are connected with each other through E&M tie lines[66].
- System expansion, where two systems are connected with each other with E&M tie lines.
- Connecting remote systems over E&M tie lines.

Refer the topic "E&M Connectivity" to know more.

An E&M Port can be programmed to behave as a Trunk Interface, a Subscriber (Station) Interface or both, as a Tie Line with the dual personality of a Trunk and a Subscriber.

The E&M Card supports

- E&M Interface - Types IV and V
- Speech Interface - Two-wire and four-wire.
- E&M Trunk Seizure Type[67]: Immediate, Immediate + Wink, Immediate with Ack, Immediate with Ack+Wink, Seizure Pulse, Seizure Pulse + Wink, Express, and Compander Control Signal.
- Address Signaling: Pulse dial (Pulse 10PPS, Pulse 20PPS) and Tone Dial (DTMF).

The E&M Card is available in the following configurations:

E&M Card for ETERNITY MENX

Card Name	Configuration and Application
ETERNITY ME Card E&M8	8-port card to connect 8 E&M Tie Lines
ETERNITY ME Card E&M4	4-port card to connect 4 E&M Tie Lines

The maximum number of E&M ports supported are 64.

Connectors

The E&M Card has RJ45 Connectors. A separate MDF cable is supplied for each connector.

66. *The Systems in a "Closed User Group (CUG)" can be connected over ISDN T1E1PRI Lines as well. Refer the topic Closed User Groups to know more.*
67. *This is the line protocol that defines how the equipment seizes the E&M trunk. Also referred to as Start Dial Supervision Signaling Protocol.*

LEDs

The ETERNITY ME Card E&M8 has eight tri-color LEDs. The ETERNITY ME Card E&M4 has 4 LEDs, to indicate the functioning of the ports.

Installing the E&M Card

An E&M port can be programmed to take on the function of:

- **a Station** - works like an extension interface, receiving incoming calls.

  OR

- **a Trunk** - works like a trunk interface when any of the extensions of the system makes an outgoing call through it.

  OR

- **a Tie Line** - takes on a dual personality: functioning as both as an extension and a trunk. The E&M port works like an extension interface for incoming calls. It works like a trunk interface when any extension makes an outgoing call through it.

  This dual function is used in systems that are used as Transit Exchanges as in a PLCC Network. Read "PLCC-An Introduction" to know more.

You cannot connect a trunk line or an SLT / DKP to an E&M port.

1. Have the necessary wiring for the E&M Analog trunk in place. Take the necessary safety precautions before you begin handling the card; switch off power supply and always wear an antistatic wrist strap and use a grounding mat.
2. Unpack the E&M Card and check the package contents.
3. The E&M Card supports E&M Interface Type IV and Type V connection. To select the appropriate Interface Type out of the two, you need to change the Jumper Settings.

   Refer the table below to select the desired Interface Type and Speech Interface.

Jumper Number	Position	Function
J1 and J2	AB	Type IV E&M Interface
	BC	Type V E&M Interface

   - By default all the E&M Ports are set to support Type-IV.
   - To select the Type-V connection for the E&M Port, set Jumpers J1 and J2 (located on the E&M module) in BC Position.

4. Select the speech interface - 2-wire speech or 4-wire speech - as required, by changing the jumper settings. Refer the table below.

Jumper Number	Position	Function
J3 and J4	AB	4-wire speech interface
	BC	2-wire speech interface

- By default all the E&M Ports are set to support 2-wire Speech Interface.
- To select 2-wire speech interface for the E&M Port, set Jumpers J3 and J4 (given on E&M module) to BC Position.
- To select 4-wire speech interface for the E&M Port, set Jumpers J3 and J4 on E&M module to AB Position.

Keep Jumper number J5 in BC position and Jumper number J6 in BC/Open position.

5. Now, select a free slot for the E&M Card. Unscrew and remove the filler bracket by pushing up the levers on the bracket. Preserve the filler bracket for future use.

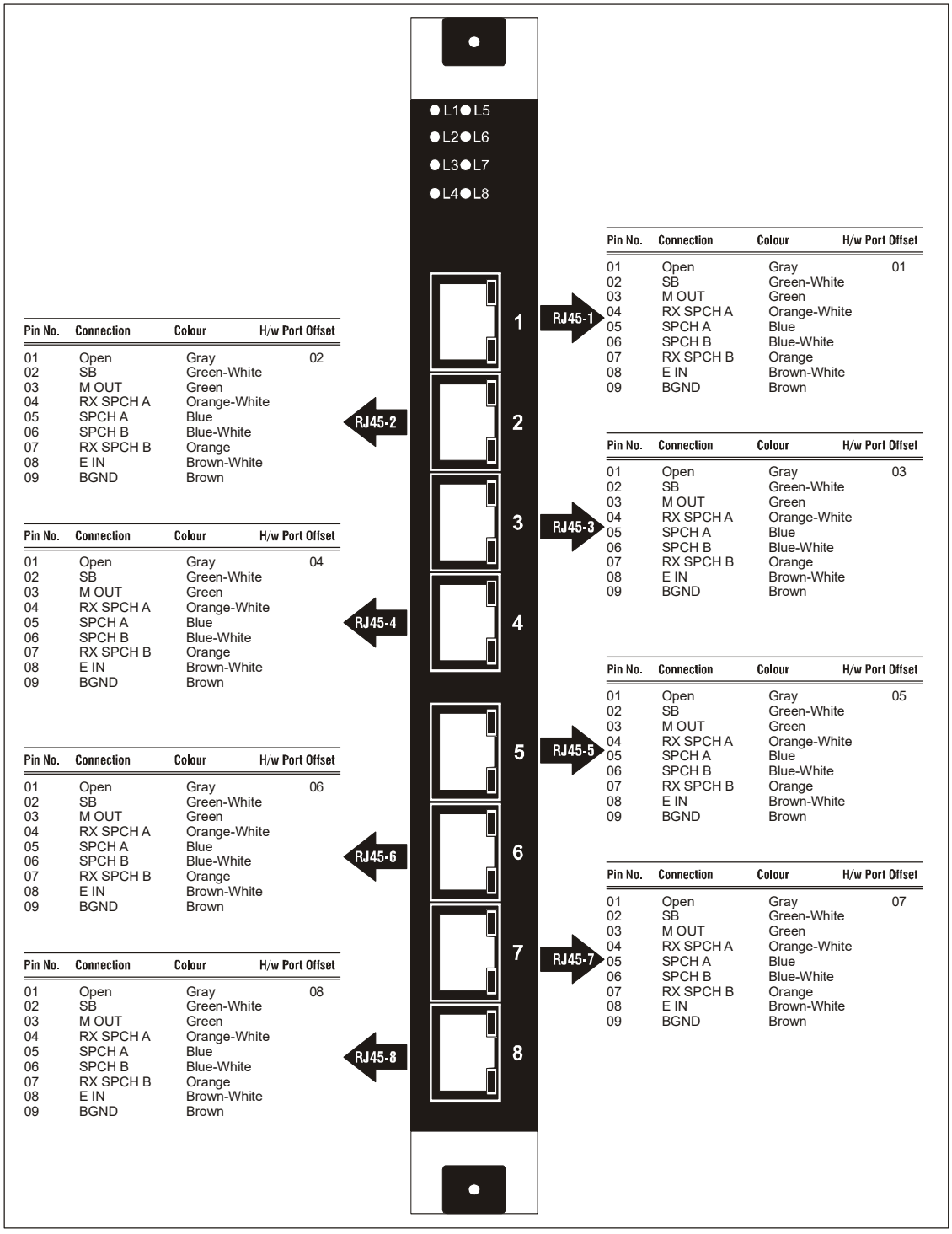
6. Insert the E&M Card into guide rails of the empty slot. Make sure the connectors on the card make perfect contact with those on the backplane motherboard. Secure the card by pressing down the levers and fix the bracket with the screws provided with the card.

7. Connect the cables supplied with the E&M Card into the RJ45 connectors on the E&M Card.

8. Connect the free ends of the cables into the E&M Ports of the other System/PBX/Router/Tie Line equipment by appropriate crossing of the wires.

   Refer the following pin-out details for the E&M Card and for each Interface and Speech Interface Type.

Pinout details of ETERNITY ME Card E&M8 - RJ45 Connectors



Pin No.	Connection	Colour	H/w Port Offset
01	Open	Gray	01
02	SB	Green-White	
03	M OUT	Green	
04	RX SPCH A	Orange-White	
05	SPCH A	Blue	
06	SPCH B	Blue-White	
07	RX SPCH B	Orange	
08	E IN	Brown-White	
09	BGND	Brown	

Pin No.	Connection	Colour	H/w Port Offset
01	Open	Gray	02
02	SB	Green-White	
03	M OUT	Green	
04	RX SPCH A	Orange-White	
05	SPCH A	Blue	
06	SPCH B	Blue-White	
07	RX SPCH B	Orange	
08	E IN	Brown-White	
09	BGND	Brown	

Pin No.	Connection	Colour	H/w Port Offset
01	Open	Gray	03
02	SB	Green-White	
03	M OUT	Green	
04	RX SPCH A	Orange-White	
05	SPCH A	Blue	
06	SPCH B	Blue-White	
07	RX SPCH B	Orange	
08	E IN	Brown-White	
09	BGND	Brown	

Pin No.	Connection	Colour	H/w Port Offset
01	Open	Gray	04
02	SB	Green-White	
03	M OUT	Green	
04	RX SPCH A	Orange-White	
05	SPCH A	Blue	
06	SPCH B	Blue-White	
07	RX SPCH B	Orange	
08	E IN	Brown-White	
09	BGND	Brown	

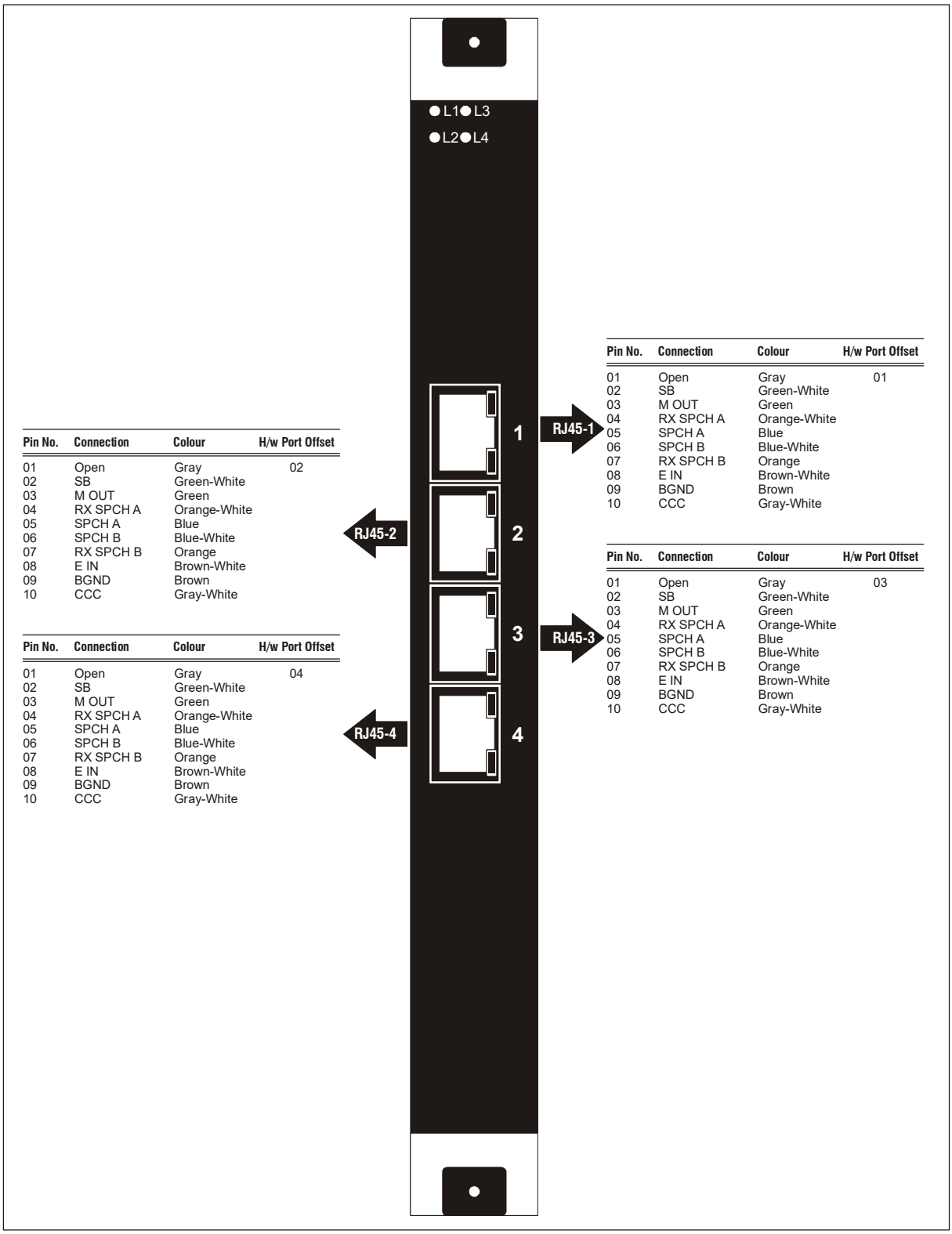
Pin No.	Connection	Colour	H/w Port Offset
01	Open	Gray	05
02	SB	Green-White	
03	M OUT	Green	
04	RX SPCH A	Orange-White	
05	SPCH A	Blue	
06	SPCH B	Blue-White	
07	RX SPCH B	Orange	
08	E IN	Brown-White	
09	BGND	Brown	

Pin No.	Connection	Colour	H/w Port Offset
01	Open	Gray	06
02	SB	Green-White	
03	M OUT	Green	
04	RX SPCH A	Orange-White	
05	SPCH A	Blue	
06	SPCH B	Blue-White	
07	RX SPCH B	Orange	
08	E IN	Brown-White	
09	BGND	Brown	

Pin No.	Connection	Colour	H/w Port Offset
01	Open	Gray	07
02	SB	Green-White	
03	M OUT	Green	
04	RX SPCH A	Orange-White	
05	SPCH A	Blue	
06	SPCH B	Blue-White	
07	RX SPCH B	Orange	
08	E IN	Brown-White	
09	BGND	Brown	

Pin No.	Connection	Colour	H/w Port Offset
01	Open	Gray	08
02	SB	Green-White	
03	M OUT	Green	
04	RX SPCH A	Orange-White	
05	SPCH A	Blue	
06	SPCH B	Blue-White	
07	RX SPCH B	Orange	
08	E IN	Brown-White	
09	BGND	Brown	

Pinout details of ETERNITY ME Card E&M4 - RJ45 Connectors



Pin No.	Connection	Colour	H/w Port Offset
01	Open	Gray	01
02	SB	Green-White	
03	M OUT	Green	
04	RX SPCH A	Orange-White	
05	SPCH A	Blue	
06	SPCH B	Blue-White	
07	RX SPCH B	Orange	
08	E IN	Brown-White	
09	BGND	Brown	
10	CCC	Gray-White	

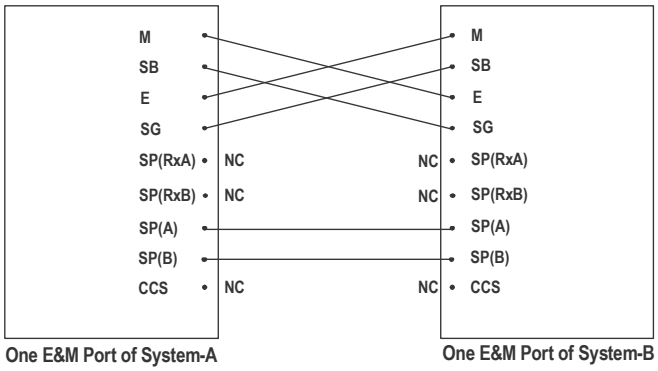
Pin No.	Connection	Colour	H/w Port Offset
01	Open	Gray	02
02	SB	Green-White	
03	M OUT	Green	
04	RX SPCH A	Orange-White	
05	SPCH A	Blue	
06	SPCH B	Blue-White	
07	RX SPCH B	Orange	
08	E IN	Brown-White	
09	BGND	Brown	
10	CCC	Gray-White	

Pin No.	Connection	Colour	H/w Port Offset
01	Open	Gray	03
02	SB	Green-White	
03	M OUT	Green	
04	RX SPCH A	Orange-White	
05	SPCH A	Blue	
06	SPCH B	Blue-White	
07	RX SPCH B	Orange	
08	E IN	Brown-White	
09	BGND	Brown	
10	CCC	Gray-White	

Pin No.	Connection	Colour	H/w Port Offset
01	Open	Gray	04
02	SB	Green-White	
03	M OUT	Green	
04	RX SPCH A	Orange-White	
05	SPCH A	Blue	
06	SPCH B	Blue-White	
07	RX SPCH B	Orange	
08	E IN	Brown-White	
09	BGND	Brown	
10	CCC	Gray-White	

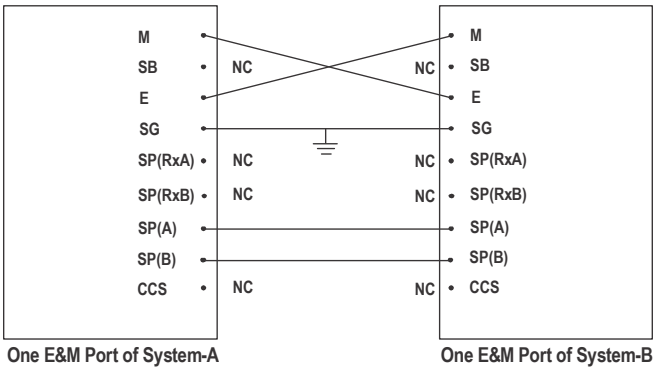
2 Wire, Type IV E&M Connection

2 Wire / Type IV E&M Connection



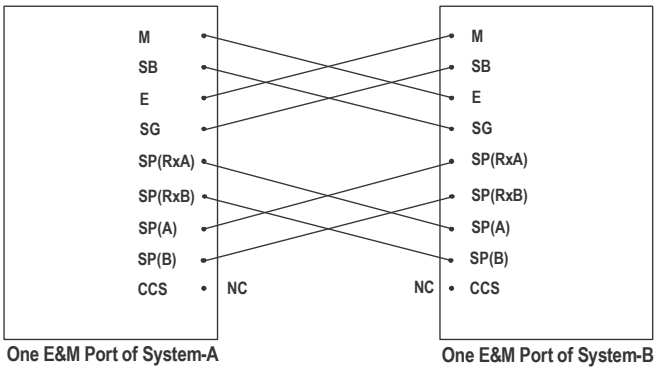
2 Wire, Type V E&M Connection

2 Wire / Type V E&M Connection

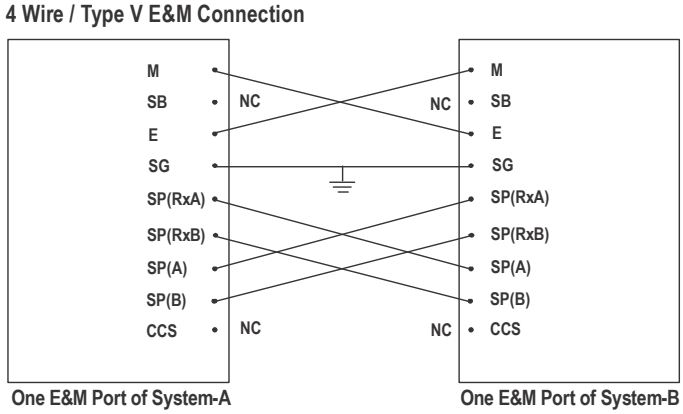


4 Wire, Type IV E&M Connection

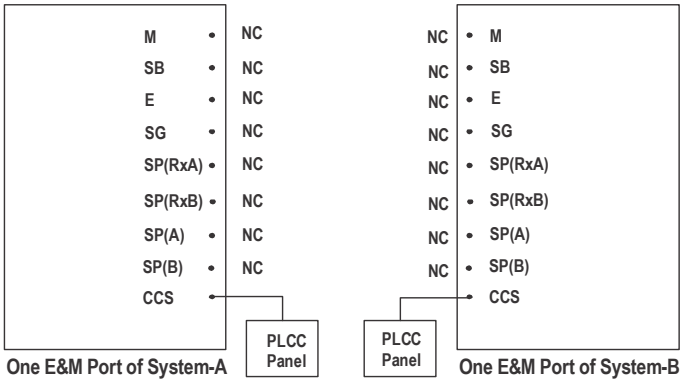
4 Wire / Type IV E&M Connection



4 Wire, Type V E&M Connection



9. If you are connecting two PLCC EPAX in a Power Line Carrier Communication Network Compander Control Signal (CCS) Connection should be made as illustrated in the block diagram below for any of the four combinations of E&M and Speech Interfaces illustrated in the previous step.



Compander Control Signal (CCS) is a special type of signal used by Power Line Carrier Communication Networks to improve quality of speech transmission. The PLCC network expects this signal from the system when speech is established and the E&M Card supports this facility. The system sends CCS signal to the PLCC panel.

- *When the E&M port is used as an Endpoint; the system sends a CCS to the PLCC panel while making an outgoing call through the E&M port or when a call is received at the E&M port.*
- *When the E&M port is used for Transit Exchange; the system sends a CCS to the PLCC panel while there is a Transit call through the E&M port.*

10. If you have completed all installation tasks, power ON the system, observe the Reset Cycle and the LED pattern of the E&M Card.

LED Pattern of the E&M Card

Stage	LED Color	LED Cadence
Initialization		
At Power ON	RED	L1 to L8 glow one after the other in a sequence, ON 100ms -OFF. L1 ON 100ms-OFF, L2 100ms ON-OFF, L3 100ms ON-OFF, L4 100ms ON-OFF……L8 100ms ON-OFF
	GREEN	L1 to L8 glow one after the other in a sequence, ON 100ms -OFF. L1 ON 100ms-OFF, L2 100ms ON-OFF, L3 100ms ON-OFF, L4 100ms ON-OFF……L8 100ms ON-OFF
Stand-By[a]	GREEN, ORANGE	L1 toggles GREEN ON 1 sec, ORANGE ON 1 sec
Normal (Port Event)		
M-Wire High	GREEN	LED of the Port continuously ON
M-Wire Low		LED of the Port continuously OFF
E-Wire High	RED	LED of the Port continuously ON
M-Wire Low		LED of the Port continuously OFF
E-Wire and M-Wire High	ORANGE	LED of the Port continuously ON
Errors		
Controller RAM failure	ORANGE	All LEDs Toggle at 1 sec
External RAM failure	ORANGE	All LEDs Toggle at 2 sec
Eprom failure	ORANGE	All LEDs Toggle at 3 sec
Invalid Slot detected	ORANGE	All LEDs Toggle at 6 sec

a. Waiting to be detected by Master Card.

Jumpers on the Main Board

Jumper Number	Position	Function
J1	AB (default)	Normal Operation
	BC	For uploading software using COM Port
J4 & J5	AB (default)	Normal Operation
	BC	For uploading software using COM Port

The Mobile Card

The Mobile Card interfaces the system with 2G/3G/CDMA/4G networks. It routes calls made and received over mobile networks, like a mobile handset.

The Mobile Cards are available in CDMA, 2G, 3G and 4G variants.

The Mobile Card does not support GPRS features, Fax and Data services, network supported services, except CLIR and USSD.

For compatibility and use of Matrix GSM products (2G/3G/4G) in Russia and Iran Province connect with Matrix Sales or Technical Support Team.

The Mobile Card for ETERNITY MENX

Card Name	Configuration and Application
ETERNITY ME Card GSM8	8-port card to connect to 8 GSM networks (8 SIM Cards can be installed). To know more, refer to "ETERNITY ME Card GSM8/GSM8 3G without SIM Hot-swap". For Hardware Design V3R2, CPLD V3R2 and PCB Version Revision V3R1 This version onwards SIM Hot Swap is supported, that is the SIM card can be removed and inserted in the SIM Slots without turning off the system. To know more, refer to "ETERNITY ME Card GSM8/GSM8 3G/GSM8 4G with SIM Hot-swap".
ETERNITY ME Card GSM8 3G	8-port card to connect to 8 GSM networks with 3G support (8 SIM Cards can be installed). "ETERNITY ME Card GSM8/GSM8 3G without SIM Hot-swap". For Hardware Design V3R2, CPLD V3R2 and PCB Version Revision V3R1 This version onwards SIM Hot Swap is supported, that is the SIM card can be removed and inserted in the SIM Slots without turning off the system. To know more, refer to "ETERNITY ME Card GSM8/GSM8 3G/GSM8 4G with SIM Hot-swap".
ETERNITY ME CARD GSM8 4G	8-port card to connect to 8 GSM networks with 4G support (8 SIM Cards can be installed) with SIM Hot Swap. To know more, refer to "ETERNITY ME Card GSM8/GSM8 3G/GSM8 4G with SIM Hot-swap".
ETERNITY ME CARD CDMA2	2-port card to connect to 2 CDMA networks (2 RUIM Cards can be installed) Cards with Version V4R1 cannot be downgraded to earlier versions.

If you are installing CDMA Mobile Card in your system, it is recommended to avoid using features that support DTMF Detection.

The features where the caller is asked to dial digits and the system has to detect it, for example, DID, Voice Mail Auto Attendant etc might not work efficiently.

Just like mobile handsets, each Mobile Port has a unique IMEI (International Mobile Equipment Identity) number, pasted on the mobile engine.

The maximum Mobile ports supported are 64.
SIM cards from different service providers can be used.

Antenna

There is a single rooftop (RT) antenna for four GSM ports. A splitter connects all the four ports on the card into a single antenna. An antenna cable is also provided, giving you the flexibility to move the antenna to another position (in case of weak signal).

Personal Identification Number (PIN)[68]

The SIM cards can be protected from unauthorized use by programming a Personal Identification Number (PIN) on the SIM. If the wrong SIM PIN is entered thrice in a row, by a user, the SIM card suspects the user and asks for the Personal Unlock Keyword (PUK).

LEDs

There is a tri-color LED for each mobile port on the card to indicate the functioning of the card and the status of the ports.

Installing the Mobile Card

To be able to connect the system to 2G/3G/CDMA/4G networks, you must have one of the above mobile cards installed in the system.

1. To install the Mobile Card,

   - If using a 2G/3G/CDMA/4G card, get the SIM Card from the 2G/3G/CDMA/4G service provider of your choice ready. Use SIM PIN protection[69], if required.

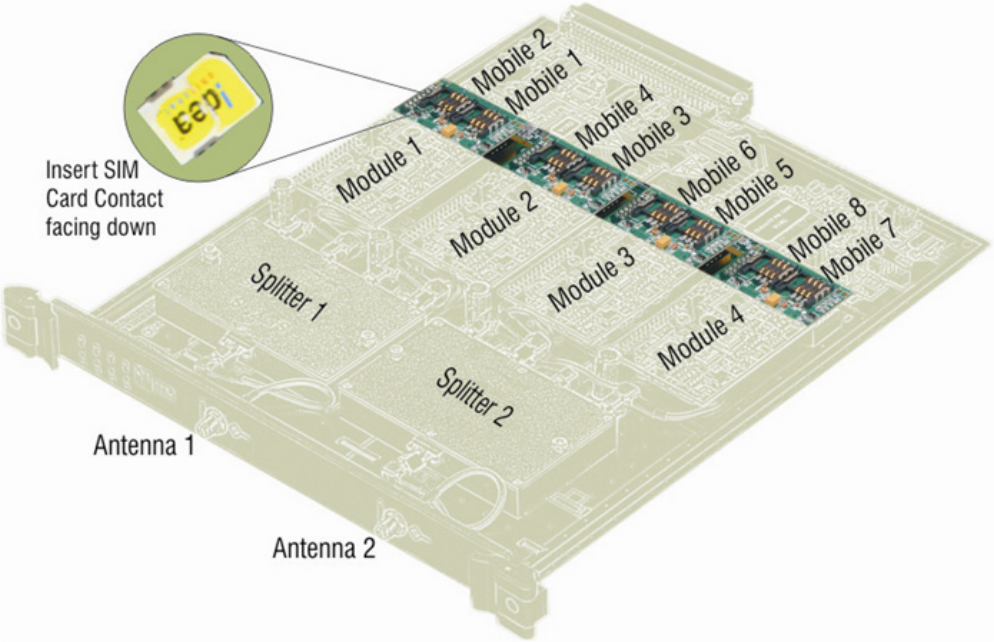
Disable Call Waiting in the SIM, else it may result in call disconnection.

2. Make sure that the system is installed at a location where sufficient network coverage is available. The power supply should be turned off, and you must be wearing an electrostatic discharge preventive wrist strap and must have a grounding mat, before you begin handling the card.

3. Unpack the Mobile Card and verify the package contents.

68. Not applicable if you are installing CDMA Mobile Card in your system.
69. SIM PIN Protection is not applicable if you are installing CDMA Mobile Card.

ETERNITY ME Card GSM8/GSM8 3G without SIM Hot-swap



Enabling PIN Protection on SIM[70]

4. For the 2G/3G/CDMA Card, enable SIM PIN before installing the SIM card in the system.
   - insert the SIM into a mobile handset first.
   - enable PIN Protection from the mobile handset.
   - change the SIM PIN to 1234 (this is the default PIN for all SIM cards used in the system). Changing the SIM PIN to '1234' enables you to change the SIM PIN from the Jeeves later (Refer SIM PIN under “Configuring Mobile Trunks” for instructions).
   - remove the SIM from the mobile handset.

If you do not want to use PIN protection, insert the SIM in the mobile handset and disable PIN protection. Remove the SIM Card from the mobile handset.

5. Insert the SIM card (PIN changed to 1234), with its connector side down into the SIM holder on the Mobile card. You can insert multiple SIM cards of the same GSM service provider or of different service providers.

6. Insert the Mobile Card into the guide rails of the Universal Slot you have selected for this card. Make sure that the card is inserted deep enough to make perfect contact with the connectors in the backplane. Now, press down the levers on the card mount bracket to secure the card in its slot.

7. Connect the antenna provided with the card on the splitter connector on the front panel of the card. You may also use the antenna cable to place the antenna at another position.

8. Repeat Steps 1-7 to insert another Mobile Card.

9. If you have completed all installations tasks, power the system.

10. Wait for the system to register with the Mobile network. By default, the Mobile ports are set to select and register with the Mobile networks automatically. Now, observe the LED Patterns of the Mobile Ports.

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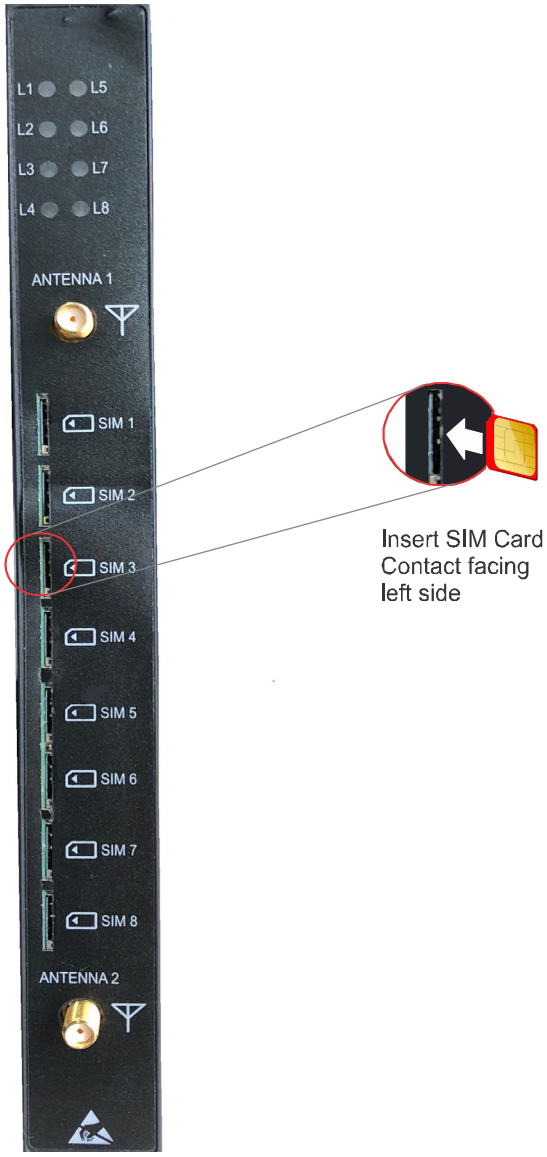
70. Disable PIN Protection on RUIM if you are installing CDMA Mobile Card in your system.

- *At every power up of the system, it takes about 3 minutes for the Mobile ports to get registered with the network. Once registration with the GSM network is completed, the mobile port can be used.*

- *Each time the Mobile Port sends a request, such as Registration Request, the system waits for the duration of the Network Response Timer. This Timer signifies the time for which the Mobile Port waits for a response from the Mobile network. It is fixed for 150 seconds for all Mobile ports.*

ETERNITY ME Card GSM8/GSM8 3G/GSM8 4G with SIM Hot-swap



Enabling PIN Protection on SIM

4. For the 2G/3G/4G Card, enable SIM PIN before installing the SIM card in the system.
   - insert the SIM into a mobile handset first.
   - enable PIN Protection from the mobile handset.
   - change the SIM PIN to 1234 (this is the default PIN for all SIM cards used in the system). Changing the SIM PIN to '1234' enables you to change the SIM PIN from the Jeeves later (Refer SIM PIN under "Configuring Mobile Trunks" for instructions).
   - remove the SIM from the mobile handset.

If you do not want to use PIN protection, insert the SIM in the mobile handset and disable PIN protection. Remove the SIM Card from the mobile handset.

5. Insert the SIM with its contact side facing left into the SIM slot located on the fascia of ETERNITY ME Card.

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6. Push the SIM backwards into the slot until you hear a click and the SIM is locked in place.

7. To unlock the SIM, push the protruded portion of the SIM backwards again and release it.

The Mobile cards with SIM Hot - swap are designed keeping in mind the Standard Nano SIM size. In case, you face any issues due to the SIM size, contact your respective Service Provider for assistance.

8. Repeat the same steps to insert another SIM Card. You can insert multiple SIM cards of the same GSM service provider or of different service providers.

9. Insert the Mobile Card into the guide rails of the Universal Slot you have selected for this card. Make sure that the card is inserted deep enough to make perfect contact with the connectors in the backplane. Now, press down the levers on the card mount bracket to secure the card in its slot.

10. Connect the antenna provided with the card on the splitter connector on the front panel of the card. You may also use the antenna cable to place the antenna at another position.

11. Repeat Steps 1-7 to insert another Mobile Card.

12. If you have completed all installations tasks, power the system.

13. Wait for the system to register with the Mobile network. By default, the Mobile ports are set to select and register with the Mobile networks automatically. Now, observe the LED Patterns of the Mobile Ports.

- *At every power up of the system, it takes about 3 minutes for the Mobile ports to get registered with the network. Once registration with the GSM network is completed, the mobile port can be used.*
- *Each time the Mobile Port sends a request, such as Registration Request, the system waits for the duration of the Network Response Timer. This Timer signifies the time for which the Mobile Port waits for a response from the Mobile network. It is fixed for 150 seconds for all Mobile ports.*

LED Pattern of Mobile Ports

The number of tri-color LEDs on the Mobile card corresponds with the number of mobile ports on the card.

At Power On: All LEDs will blink 1 second ON and 1 second OFF in the color sequence: Red-Green-Orange until the Reset cycle is complete.

In the Stand-by state: All LEDs will glow Orange for a second and turn Green for a second, repeatedly.

During normal functioning: The LEDs will various events on the Mobile port in the color and cadence described in the table below:

Event	Color	Cadence in msec (1 cadence is of 3000 msec)
Port disabled	-	LED OFF

Event	Color	Cadence in msec (1 cadence is of 3000 msec)
Port idle	-	LED OFF
Port Active (All States other than Ring and Speech)	Red	Continuous ON
Ring Event	Green	400ms ON-200ms OFF400ms ON-200ms OFF
Speech	Green	Continuous ON
GSM initialization	Orange	200ms ON-200ms OFF-200ms ON-200ms OFF-200ms ON-200ms OFF-200ms ON-200ms OFF-200ms ON-1200ms OFF (5 blinks)
PUK required	Orange	200ms ON-200ms OFF-200ms ON-200ms OFF-200ms ON-200ms OFF-200ms ON-1600ms OFF-
SIM PIN faulty	Orange	200ms ON-200ms OFF-200ms ON-200ms OFF-200ms ON-2000ms OFF (3 blinks)
SIM Absent	Orange	200ms ON-200ms OFF-200ms ON-2400ms OFF (2 blinks)
Network Link Down (Absence of GSM Network)	Orange	200ms ON-2800ms OFF

The Magneto Card

The Magneto Card is used for connecting the system to Magneto Telephones[71], which are widely used by the defense establishments as field phones in front lines, and by other establishments such as railroad companies (signaling emergencies, crossings, etc.), electric utilities, pipeline companies, who need to have their networks at places that are too remote to be serviced by public telephone networks.

The Magneto Card lands calls from magneto field telephones on the extensions of the system and places calls from the extensions of the system on magneto telephones.

Magneto Card for ETERNITY MENX

Card Name	Configuration and Application
ETERNITY ME Card Magneto8	8-port card to connect 8 Magneto Phones

The maximum number of magneto ports supported are 16.

Connectors

The Magneto Card has RJ45 connectors. A multi-pair cable is provided for each connector.

LED

The ETERNITY Magneto8 has 8 LEDs for each magneto port supported by the card.

The LEDs indicate the health of the cards during the Reset Cycle and the status of the ports during the normal functioning of the system.

Installing the Magneto Card

1. Have the necessary wiring for the Magneto Ports in place.

   You may install an MDF to connect the Magneto Ports with the Field Telephone wires.

   OR

   You may connect the wires from the Magneto Field Telephones directly to the Magneto Port.

   You are advised to use a separate set of Krone Modules for connecting the Magneto phones to the Magneto ports of the system

2. Prepare for the card installation by switching off power supply and wearing an electrostatic discharge preventive wrist strap and use a grounding mat.

3. Unpack the Magneto Card and check the package contents.

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71. A magneto telephone is a local battery telephone set, in which signaling current is provided by a magneto hand generator, usually a magneto. The hand generator, commonly referred to as 'crank', is located on the right hand side of the telephone set and is turned to produce energy to ring other phones or to signal the CO. The magneto, also called the generator, is used to convert the mechanical motion via the crank to produce sufficient energy to ring other phones or to signal the CO.

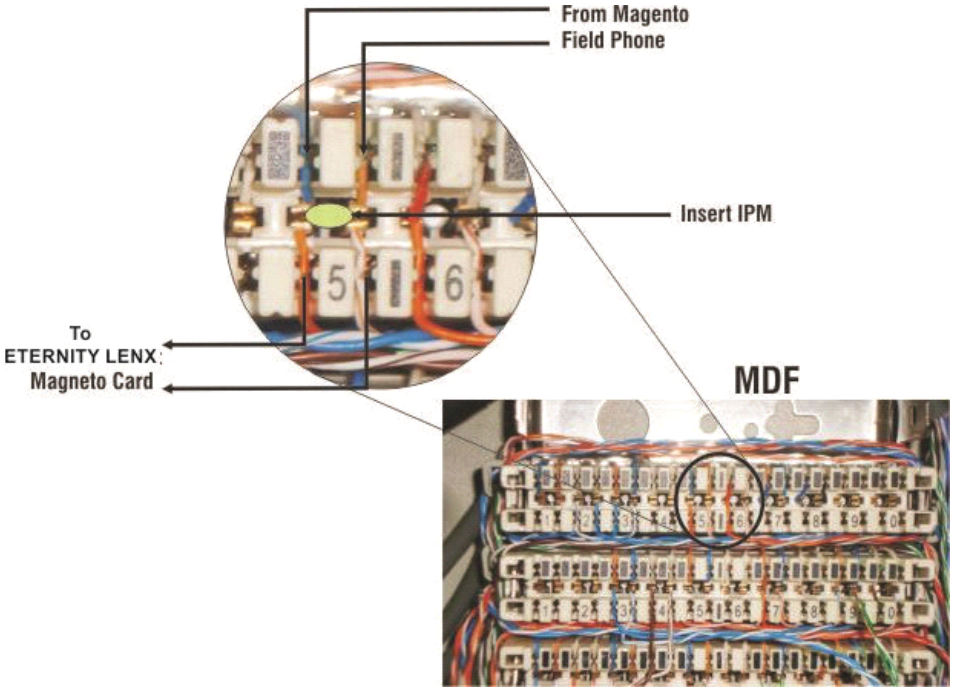
4. Select any universal slot to insert the card. Unscrew the filler bracket and remove it by pushing up the levers on the bracket.

5. Insert the Magneto card into the guide rails of the free slot. The card's connectors must make perfect contact with the connectors on the backplane motherboard. Press down the levers of the mounting bracket to secure the card in its slot and fix the two screws provided with the card on the mounting bracket.

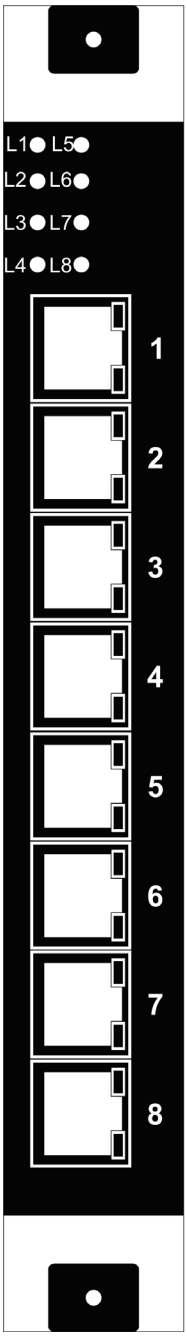
6. Now, plug in the cables supplied with the Magneto Card into the connectors on the card. Terminate the free ends of the cables into the MDF, if applicable.

   Refer to the following block diagram for terminating the cables from the Magneto Card and the wires from the Magneto Field Telephones.

Connecting Magneto Telephones to the Magneto Card



7. Connect the pairs of wires from the Magneto Field Phones to the appropriate pairs emerging from the Magneto Card on the MDF. Refer the cable diagram below.



Connector	Color	Connection	H/w Port Offset
RJ45-1 (Blue)	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	Magneto — — —	01 — — —
RJ45-2 (Orange)	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	Magneto — — —	01 — — —
RJ45-3 (Green)	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	Magneto — — —	01 — — —
RJ45-4 (Brown)	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	Magneto — — —	01 — — —
RJ45-5 (Blue)	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	Magneto — — —	01 — — —
RJ45-6 (Orange)	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	Magneto — — —	01 — — —
RJ45-7 (Green)	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	Magneto — — —	01 — — —
RJ45-8 (Brown)	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	Magneto — — —	01 — — —

8. Repeat the same steps to install the other Magneto Cards.

9. If you do not have any other Card to insert and have completed the installation procedures, power on the system and observe the Reset Cycle and the LED Pattern of the Magneto Card.

LED Pattern of the Magneto Card

Stage	LED Color	LED Cadence
Auto Upgradation / Auto Downgradation		
Card waiting for application	RED	L1 turned ON for 200ms, turned OFF simultaneously.
Card is up, loaded with new application	GREEN	L1 turned ON for 200ms, turned OFF simultaneously.
Initialization		
	RED	L1 to L8 turned ON 500ms-OFF 500ms
	GREEN	L1 to L8 turned ON 500ms-OFF 500ms
	ORANGE	L1 to L8 turned ON 500ms-OFF 500ms
Stand-By[a]	ORANGE, GREEN,	L1 toggles 1 sec Orange and RED
Normal (Port Event)		
Ring (incoming/outgoing call)	RED	LED of the Port continuously ON
Port Disabled		LED of the Port continuously OFF
OFF-Hook (in Speech) For Outgoing call from Magneto: When called extension answers the call For Incoming call to Magneto: When stopping Ring on Magneto using MRE key or #	GREEN	LED of the Port continuously ON
Port Idle		LED of the Port continuously OFF
Errors		
Invalid Card Configuration Jumper	ORANGE	All LEDs Flash (250ms ON-250ms OFF) twice, OFF 3 sec.
Invalid Slot detection[b]	ORANGE	LEDs Flash (250ms ON-250ms OFF) 6 times, OFF 3 sec.

a. Waiting to be detected by CPU Card.
b. Waiting to be detected by CPU Card.

Jumpers on the Main Board

Jumper Number	Position	Function
J1 & J2	AB	NA
	BC (default)	Normal Operation
J3, J5 & J6	AB (default)	Normal Operation
	BC	For uploading software using COM Port

The Radio Card

The Radio Interface Card (RIC) adds the Two-way Radio functionality in the system. In Two-way radio, the speech can be transmitted as well as received by the radio devices such as Radio Phone, Radio Repeater. Such devices are called Radio Transceivers. The Two-way radio works on High Frequency (HF), Very High Frequency (VHF) or Ultra High Frequency (UHF).

Radio Cards for ETERNITY MENX

Card Name	Configuration and Application
ETERNITY ME Card Radio8	8-port card to connect 8 Radio devices.
ETERNITY ME Card Radio4	4-port card to connect 8 Radio devices.

The maximum number of radio ported supported are 16.

Connectors

The Radio Card has RJ45 connectors. A multi-pair cable is provided for each connector.

LED

The Radio Card has one tri color LED.

Installing the Radio Card

1. Have the necessary wiring for the Radio Ports in place.

   You may install an MDF to connect the Radio Ports with the Radio device wires.

   OR

   You may connect the wires from the Radio device directly to the Radio Port.

   You are advised to use a separate set of Krone Modules for connecting the Radio devices to the Radio ports of the system.

2. Prepare for the card installation by switching off power supply and wearing an electrostatic discharge preventive wrist strap and use a grounding mat.

3. Unpack the Radio Card and check the package contents.

4. Select any universal slot to insert the card. Unscrew the filler bracket and remove it by pushing up the levers on the bracket.

5. Insert the Radio Card into the guide rails of the free slot. The card's connectors must make perfect contact with the connectors on the backplane motherboard. Press down the levers of the mounting bracket to secure the card in its slot and fix the two screws provided with the card on the mounting bracket.

6. Now, plug in the cables supplied with the Radio Card into the connectors on the card. Terminate the free ends of the cables into the MDF, if applicable.

7. Connect the pairs of wires from the Radio devices to the appropriate pairs emerging from the Radio Card of the system on the MDF. For more details, see “The Main Distribution Frame (MDF)”.

Refer to the Pin-out details given below.

ETERNITY ME Card Radio8

Connector	Color	Pin Number	Signaling	H/w Port Offset
RJ45-1 to RJ45-8	Orange & White	1	PTT	01 to 08
	Orange	2	PTT_RTN	
	Green & White	3	Rx-	
	Blue	4	Tx+	
	Blue & White	5	Tx-	
	Green	6	Rx+	
	Brown & White	7	Unused	
	Brown	8	Unused	

ETERNITY ME Card Radio4

Connector	Color	Pin Number	Signaling	H/w Port Offset
RJ45-1 to RJ45-4	Orange & White	1	PTT	01 to 04
	Orange	2	PTT_RTN	
	Green & White	3	Rx-	
	Blue	4	Tx+	
	Blue & White	5	Tx-	
	Green	6	Rx+	
	Brown & White	7	Unused	
	Brown	8	Unused	

8. Repeat the same steps to install the other Radio Cards.

9. If you do not have any other Card to insert and have completed the installation procedures, power on the system. Observe the Reset Cycle and the LED Pattern of the Radio Card.

LED Pattern

Stage	**LED Color**	**LED Cadence**
Auto Upgradation / Auto Downgradation		
Card waiting for application	RED	200ms ON - 200ms OFF

Stage	LED Color	LED Cadence
Card is up, loaded with new application	GREEN	200ms ON - 200ms OFF
Initialization		
	RED	500ms ON - 500ms OFF
	GREEN	500ms ON - 500ms OFF
	ORANGE	500ms ON - 500ms OFF
Stand-By	ORANGE, GREEN	Toggles 1 sec Orange 1 sec Green
Port Status		
Selected Port's data are transmitted to master card	RED	Toggle on each event
Selected Port's data are received from master card	GREEN	Toggle on each event from master

Jumpers on the Main Board

Jumper Number	Position	Function
J3	AB (default)	Normal Operation
	BC	For uploading software using COM Port
J4 and J5	AB (default)	Normal Operation
	BC	For uploading software using COM Port

In PCB-P-200-80-01-01, for 4-wire speech the Jumpers J1 and J2 on the modules must always be set in AB position.

In PCB-P-200-80-01-02 onwards only 4-wire speech is possible therefore there are no Jumpers on the modules.

The Data Card

The Data Card supports four Ethernet 10/100mbps interfaces. Ethernet data coming to ports can be mapped to 2Mb streams. Each data port can be mapped to one 2Mb - E1 stream, that is 30 channels. The remaining channels of E1 can be used for voice applications. The Data Card has a 4-port Ethernet switch on board, which can aggregate multiple data streams to PCM streams of the system.

The Data Card can be installed in any of the Universal Slots of the system.

Ports and Connectors

The Data Card has an RJ45 Connector for each port. Use the cables supplied with the card for connectivity.

You can connect the cables from the LAN switch to these connectors.

LEDs

The Data Card has one dual color LED.

Installing the Data Card

1. Unpack the Data Card and check the package contents. It is recommended that you switch off the power supply, before you begin the installation of the card. Always wear an electrostatic discharge prevention wrist strap/belt and use a grounding mat.

2. Select any free (empty) slot from the Universal Slots. Unscrew and remove the filler bracket of the empty slot. Do not discard the filler bracket! Preserve it for future use!

3. Insert the Data Card into the guide rails of the free slot you selected for the card. The connectors on the card should make perfect contact with those of the slot on the backplane motherboard.

   Press down the lever on the card mounting brackets to secure the card in its slot. Fix the mounting bracket in place with the two screws provided.

4. Use the cable supplied with the card to connect the Data Ports to the Ethernet Network (Switch/PC).

5. If you have completed all other installation tasks, you may turn ON the system and observe the Reset Cycle and the LED pattern of the Data Card.

   LED Pattern of the Data Card

Status	Color	Cadence
Waiting for Master to start Up-gradation procedure (waits for 10 seconds)	Red	Blinking 200ms ON - 200ms OFF
Up-gradation in progress	Green	Blinking 200ms ON - 200ms OFF
Application run error after up-gradation procedure	Red	Continuous ON
Auto upgradation completed successfully and in process to start new application	Green	Continuous ON

Status	Color	Cadence
Main application is being executed	Red	Blinking 1 sec ON - 1 sec OFF

Jumpers

Jumper Number	Position	Function
J9	AB (default)	External Boot - Normal
	BC	Internal Boot

SIP Extensions

SARVAM UCS supports up to 2000 SIP/UC Users. The SIP/UC Users function in the same way as DKP/SLT extensions of the system. SIP/UC Users can make and receive calls to any extension user of the system and to external numbers over various telecom networks like CO, Mobile, ISDN PRI, BRI, and VoIP[72].

You may register any SIP-enabled device — a Matrix UC Client, an IP-phone, a Soft phone, Analog Phone Adapter — as the SIP User of the system.

The Matrix UC Clients also offer UC functionalities in addition to the SIP functionalities.

The SIP Users register with the CPU Card of the system. Five free SIP Users are provided by default. You may register any of the SIP-enabled devices except the Matrix UC Clients with these free SIP Users. For registering the Matrix UC Clients, you must purchase the Matrix VARTA User License. If you require additional SIP Extensions you must purchase the IP Subscribers License.

The system supports two NX DBM VOCODER64 Modules. You must purchase the module separately. Each NX DBM VOCODER64 module supports a maximum of 64 VOCODER channels. The Vocoder channels are required for — VoIP to Non-VoIP calls, VoIP to VMS calls and VoIP to VoIP calls — where transcoding is required.

The system provides 4 pre-activated VOCODER channels by default. To use these channels make sure you have installed atleast one NX DBM VOCODER64 module. If you require more channels, you can purchase the channel licenses according to your requirement.

For more information on Licenses — Matrix VARTA User License, IP Subscribers License and VOCODER Channel License, see “License Management”.

You may connect any Standard Phone or Extended IP Phones of Matrix as SIP Users.

Matrix VARTA WIN200, VARTA ADR100 and VARTA AMP100 can be registered as SIP Users, also offering the support for UC functionalities.

You may also connect/register[73] the following as SIP Extensions of the system:

- Connect SPARSH VP248, the Extended IP Phone. For instructions, see “Connecting SPARSH VP248 as Extended SIP Extension”.
- Connect SPARSH VP310, the Extended IP Phone. For instructions, see “Connecting SPARSH VP310 as Extended SIP Extension”.
- Connect SPARSH VP330, the Touch Screen Extended IP Phone. For instruction,“Connecting SPARSH VP330 as Extended SIP Extension”.
- Connect SPARSH VP510, the Premium IP Phone. For instruction, “Connecting SPARSH VP510 as Extended SIP Extension”.
- Connect SPARSH VP210, the Entry Level IP Phone. For instruction, “Connecting SPARSH VP210 as Extended SIP Extension”.
- Connect Extended SPARSH VP710, the Smart Video IP Phone. For instruction,“Connecting Extended SPARSH VP710 as Extended SIP Extension”.

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72. Calls between VoIP, Public and Private Networks may be subject to Regulation in your country. You may have to configure your system to allow or restrict call traffic between networks to comply with the telecom regulations of your country. To know more, read “Logical Partition”.

73. SARVAM UCS supports only IPv4 Addresses for registering SPARSH VP248, SPARSH VP310, SPARASH VP330, Extended SPARSH VP710, Mobile UC Clients as well as VARTA WIN200.

You can register following UC Clients as SIP Users of the system:

- Matrix VARTA WIN200, Unified Communication Client for Windows. For instruction, refer to the *MATRIX VARTA WIN200* User Guide.
- Matrix Mobile UC Clients, as given below:
  - Matrix VARTA AMP100, the Mobile UC Client for iPhones. For instruction, refer to the *Matrix VARTA AMP100* User Guide.
  - Matrix VARTA ADR100, the Mobile UC Client for Android Smartphones/Tablets. For instruction, refer to the *Matrix VARTA ADR100* User Guide.

Refer to “SARVAM UCS Features Supported in Terminals” to know the features supported in each client.

The SIP Users may be registered over ***WAN*** or over ***LAN*** according to your preference and your IP network installation scenario. Extended SIP Phones and UC Clients can be registered with SARVAM UCS, for details refer to the specific topic.

You can register the same SIP User from three different locations.

If you register the Extended IP Phone outside the Region/Country selected for SARVAM UCS, the time and Time Zone dependant features, such as Alarms, Reminders, Time Zone Display, of the phone at each location will operate according to the Real Time Clock of SARVAM UCS. Also, Access Codes and Emergency Numbers will work according to the Region/Country selected for SARVAM UCS.

- Connect the Extended IP Phone, or any Standard IP Phone to the LAN Switch.

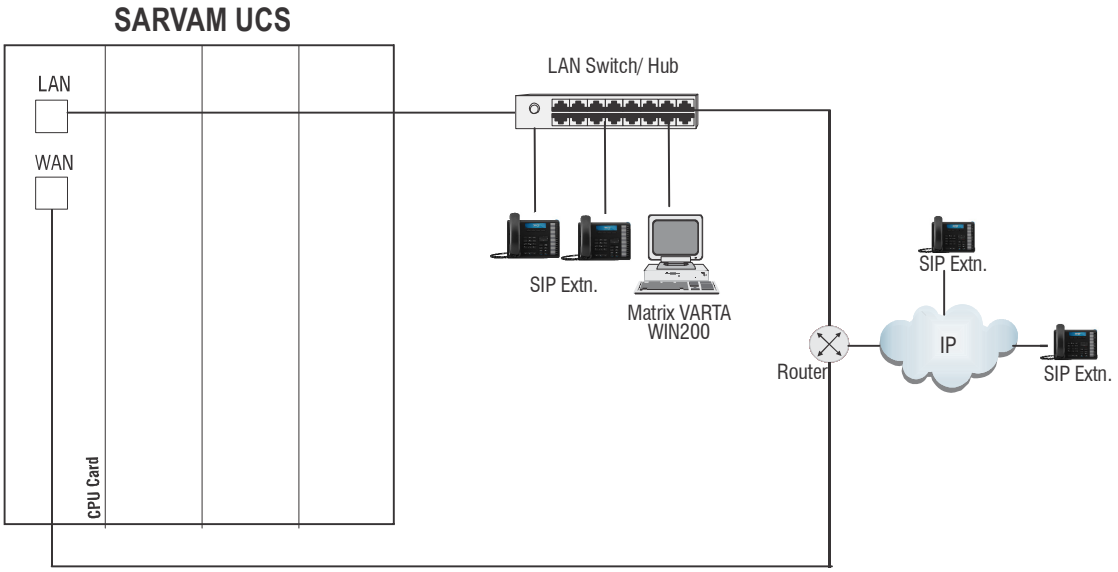
- Register any SIP device (Extended IP phone/ Soft clients or Standard IP phone) on the public network as SIP Extension.

- When you register the Matrix Extended IP Phone with the system, the WAN/LAN port is used for Auto Configuration as well for Registration of the Extended IP Phones.

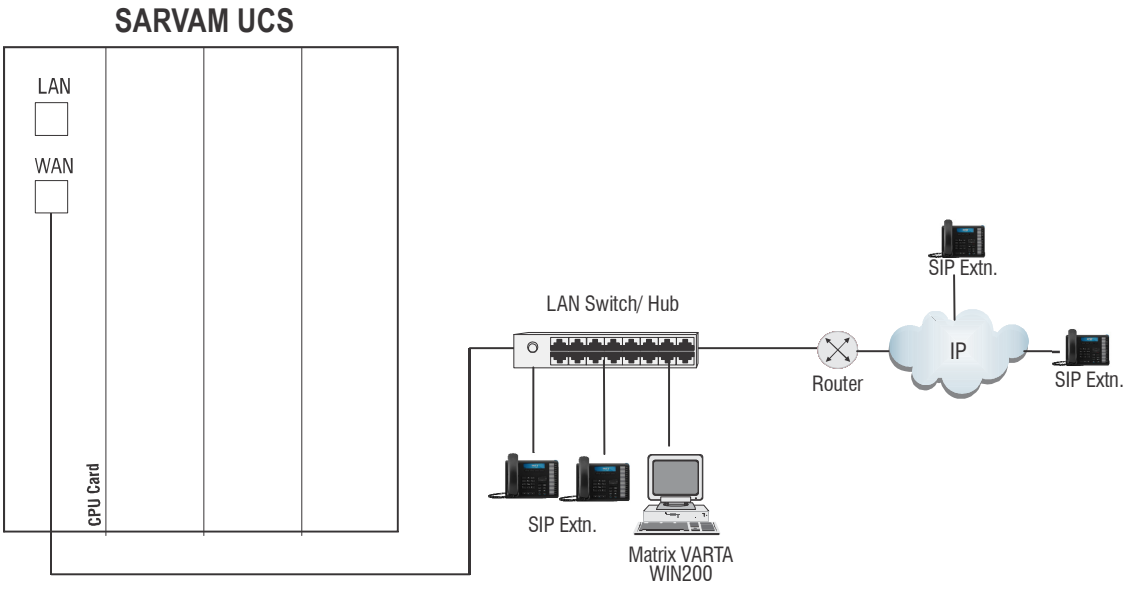
- When you register a SIP device other than the Matrix Extended IP Phone on the public network as SIP Extension, do the following:
  - In this SIP device configure the following:
    - the Registrar Server Address of SARVAM UCS
    - the Registrar Server Port
    - the SIP ID
    - Authentication ID and Password.
  - Configure Port Forwarding for the WAN Port of SARVAM UCS on the Router.

If the SARVAM UCS is connected to a ***Public Network***,

- Connect the Matrix VARTA WIN200, Extended IP Phone, or any Standard SIP device to the LAN Switch.
- Register any SIP device (Matrix VARTA UC Clients, Extended IP phone or Standard SIP phone) on the public network as SIP extension.



If the SARVAM UCS is connected to a ***Private Network (Behind the NAT)***,



- Connect Matrix VARTA WIN200, Extended IP Phones or Standard SIP Phones to the LAN Switch
- You may also register any SIP device (Matrix VARTA UC Clients, Extended IP Phone or Standard SIP phone) on the public network as SIP Extension.

  When you register the Matrix Extended IP Phone with SARVAM UCS, configure **Port Forwarding** for the **WAN port of the CPU Card** on the Router. The WAN Port is used for Auto Configuration of the Extended IP Phones.

Connecting SPARSH VP248 as Extended SIP Extension

You are recommended to complete the following steps before connecting the Matrix Extended IP Phone to SARVAM UCS:

- Decide the location of the Extended IP Phone, whether within the same network or outside, according to your installation scenario.

*If you want to use the **DHCP Server** for assigning IP Address to the Extended IP Phone, select **DHCP option 224** and **Data Type** as **'String'** and program the LAN or WAN IP Address /Domain Name of SARVAM UCS and SPARSH Port in the format **"IP_Address:Port"** in your DHCP Server as per your installation scenario.*

- Log in to *Jeeves*. For instructions, read the topic "Configuring SARVAM UCS".

- Assign SIP User ID (will work as an extension number) to the Extended IP Phone. For instructions on assigning SIP ID, see "Configuring SIP Extensions".

  For the SIP User ID you assigned to the Extended IP Phone, you must configure the necessary parameters in SARVAM UCS so that Extended IP Phone can register as a SIP Extension. For instructions, see the topic "Configuring SIP Extension Settings as per the Extended Phone Type" under *Configuring SIP Extensions*.

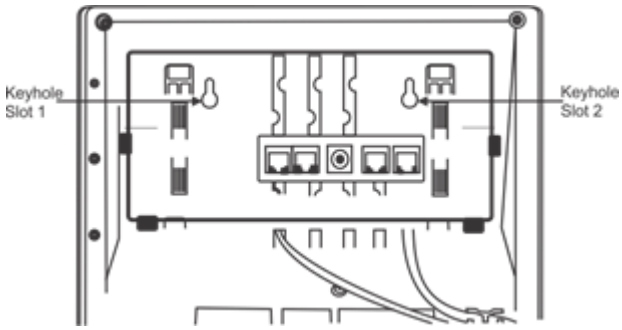
Now, follow the steps described below to install the Extended IP Phone. The instructions are common for all models of the SPARSH VP248. For the purpose of illustration, the premium model, SPARSH VP248P, has been used.

1. Unpack the SPARSH VP248 box and verify package contents.

2. Mount the phone on a desk or wall at a location convenient to you.

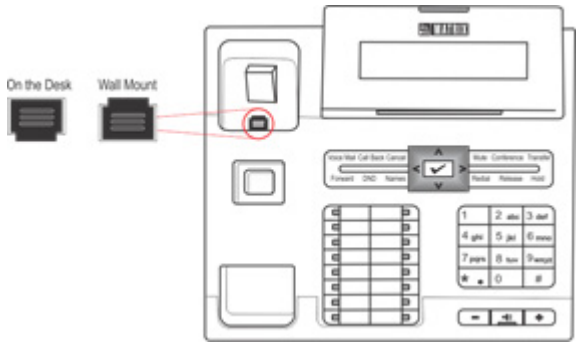
   - When mounting the phone on the wall,

     - Use the mounting template to drill holes of appropriate size and distance. Fix the screw grips in the holes you drilled.
     - Fix two screws in the holes on the wall, ensuring that they are aligned with the Keyhole Slots 1 and 2.
     - Use wall plugs, if required, to fix the screws. Leave the screw heads protruding from the wall to fit into the Keyholes.



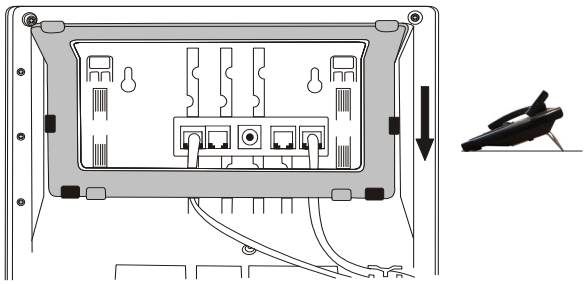
   - Now, mount the phone on the wall, with the screws fitting into the Keyhole slots.

- Reverse the handset wall mount tab to make sure the handset remains intact when you mount the phone. Push the handset wall mount tab upwards to remove it from the slot. Rotate it 180 degrees clockwise and push it downwards into the slot.

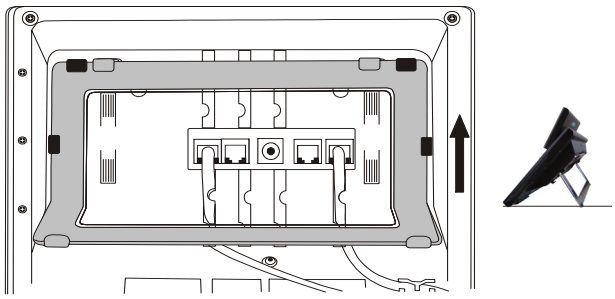


- When you mount the phone on a desk,
  - You can attach the Foot Stand in two ways as illustrated in the following.

Foot Stand attached at 30° Angle



Foot Stand attached at 50° Angle

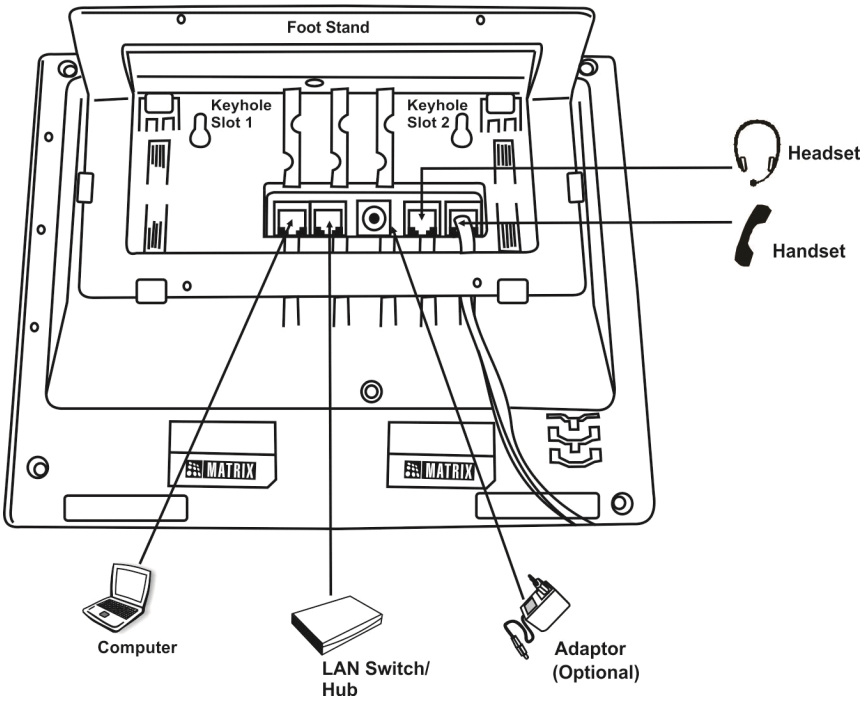


If you attach the Foot Stand at 50°, the phone will be placed in an almost upright position on your desk.

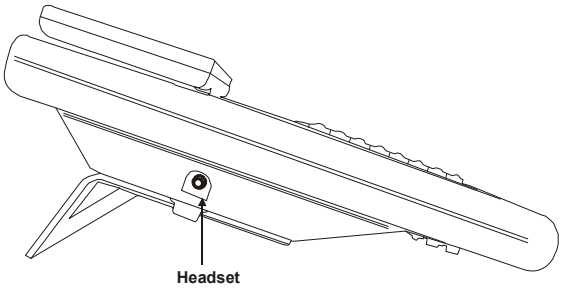
  - Decide which of these positions would work for you best and accordingly attach the Foot Stand.

3. Connect the Handset to the Phone body.
   - Plug the long straightened end of the phone cord into the handset jack at the bottom of the phone marked with the handset symbol.

- Plug the other (short straight) end of the phone cord into the jack at the bottom of the handset.

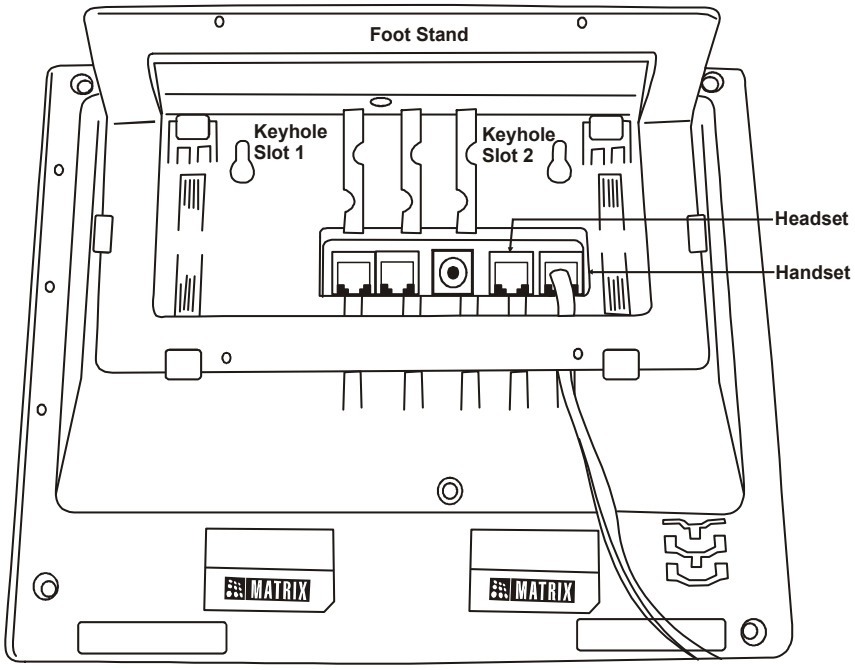


4. If you want to use a Headset (not supplied) with your phone, you may plug a headset with a 2.5 mm single connector into the headset jack headset jack with the symbol on the left side panel of the phone, as illustrated in the figure below.



OR

You may plug a headset with an RJ9 connector into the headset port at the bottom of the phone, marked with the symbol , as illustrated in the figure below.



5. Connect the LAN Port of SPARSH VP248 to the LAN Switch/Hub or a Router, according to your installation scenario.

6. To connect your phone to a computer on your desk, use an Ethernet cable (not supplied with this phone) to connect the PC Port of the phone to the LAN Port of the computer.

7. It is a PoE enabled phone and can be powered over Ethernet by connecting it to a PoE enabled LAN Switch (IEEE 802.3af Compliant). In this case you need not connect the Power Adapter.

   If you do not want to use PoE, plug in the connector of the Adapter into the power jack (DC Jack) at the bottom of the phone. Plug in the Power Adapter into a power outlet.

If both the power options, that is, PoE as well as Power Adapter are available to the phone, then the phone will derive power from the PoE enabled LAN Switch.

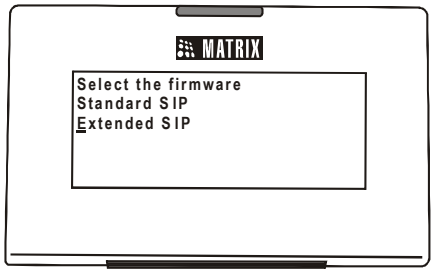
The IP phone should be used with Matrix original power adapter (5V/2A) only. The use of any third-party power adapter may cause damage to the phone.

8. Switch ON power supply.

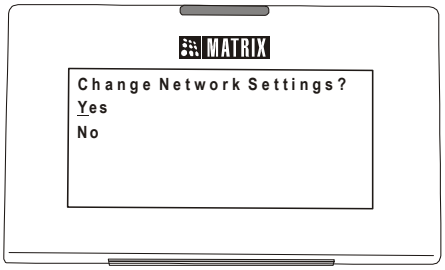
   When you power the phone, the boot process will be initiated in the following sequence.

   - All keys with LED, including the Speaker key, and the Ringer LED, will glow.
   - The LCD display will light up and booting message appears.
   - As soon as the ‘Loading...’ message appears on the phone display, press **#** key.

- Select the firmware **Extended - IP Phone**. Move the cursor by pressing the DOWN navigation key **V**.
- When the cursor is placed under the Extended IP Phone, press Enter key.



- The phone will start loading the Extended IP Phone Firmware. It will display current firmware being loaded.
- After loading the firmware, the phone will prompt you to change Network settings.



- Wait for a few seconds.

If you want to change the Network Settings or Server Settings, press the Enter key. Detailed instructions for changing the Network Settings of the phone are provided at the end of this topic. See "Network Settings"at the end of this topic.

- The phone makes DHCP Discovery and fetches its IP Address and Server Address from the DHCP Server.

  On getting the IP Address and Server Address, the phone initiates Auto Configuration to download the configuration files from SARVAM UCS.
- As the phone downloads the configuration files, the file names will appear one by one.
- On successful download of all configuration files, the phone attempts to register with SARVAM UCS.
- On successful registration, the phone will display the current day, date and time, the extension number and name assigned to the Extended IP Phone.

Network Settings

You can change the network settings of the Extended IP Phone by accessing the Local Menu of the phone. To move the cursor and scroll through the menu and submenu options, use the following touch sense navigation keys on your phone.

- The Enter key ✓ to make a selection or to complete an action.
- The Up key ∧ to move up the Menu.

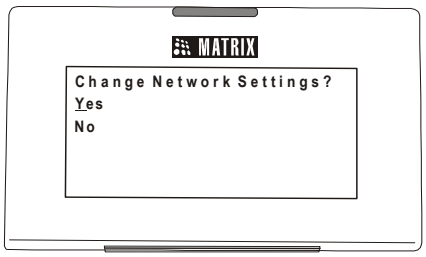
- The Down key ⋁ move down the Menu.
- The Forward key > move the cursor one character.
- The Back key < to move the cursor one character and to return from the submenu to the main menu.

The cursor is a non-blinking underscore that appears under the first letter of the first option in the menu. To make a selection in the menu, you must move the cursor in the desired direction using the Up, Down, Forward and Back key. When the cursor is at the desired position, press Enter key to make a selection.

Accessing Network Settings

You can access the Network Settings of the Extended IP Phone in any of the following stages:

1. During start-up, when the phone prompts you to change the network settings after loading the firmware.

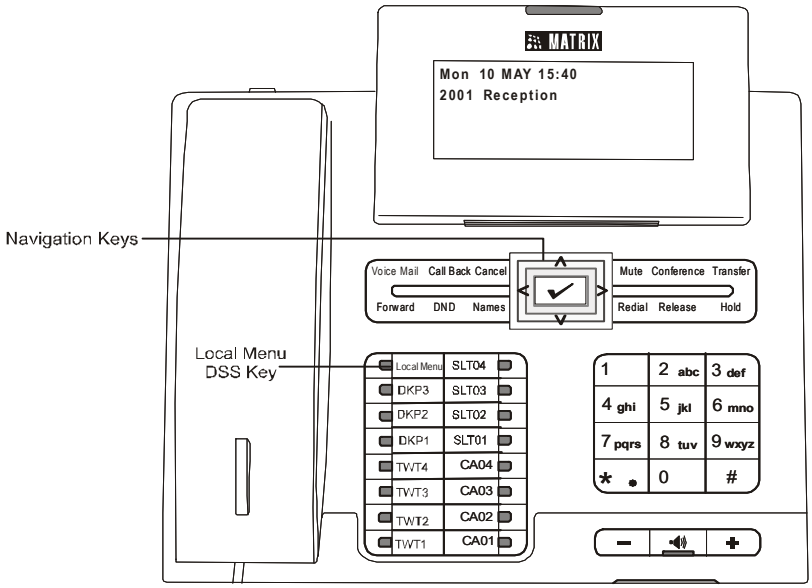


   You must press the Enter Key to select Yes and access network settings.

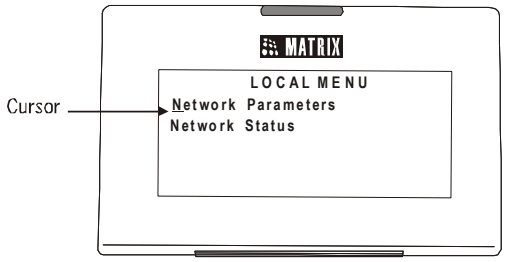
2. When the phone is making Network discovery, downloading configuration files, attempting registration.

   You must press the Enter Key ✓ to access network settings,

3. When the phone is in idle state. You must press the DSS key assigned to ‘Local Menu’.



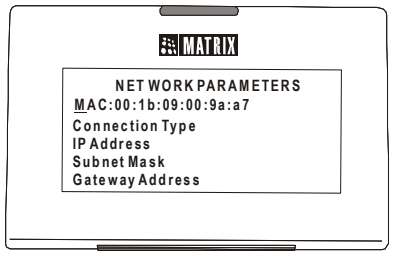
4. When you press the Local Menu DSS Key (in idle state) or when you press the Enter key during any process, the Local Menu appears on your phone display.



You can configure Network Parameters and view Network status from the Local Menu.

Configuring Network Parameters

- In the Local Menu of the phone, select Network Parameters by pressing the Enter Key.
- The Network Parameters submenu appears.



- Use the Down/Up key to reach the desired network parameter and press Enter key to select and change the settings.
- You can configure all network parameters described below, except the MAC Address.

- *To enter a dot in the editable fields — IP Address, Subnet Mask, Gateway Address, DNS Address, Server Address — press * (Star) key.*
- *If you want to clear a single digit/character, move the cursor to the right of the digit/character you wish to clear. Now press the Cancel key. The digit to the left of the cursor will be deleted. If the cursor is to the extreme left and you press the Cancel key, you will go to the previous menu.*

Connection Type

- Select the Connection Type as DHCP, PPPoE or Static according to the IP Addressing scheme of your network.

  If you select DHCP or PPPoE, the phone will be assigned IP Address, Subnet Mask and Gateway Address, DNS Address Server Address, automatically by the DHCP/PPPoE server.

  For PPPoE Connection Type, you must configure the PPPoE User ID and Password provided by the Internet Service Provider.

  If you select Static, you must assign the IP Address, Subnet Mask and Gateway Address to the phone.

IP Address

- If you select Static as Connection Type, enter the static IP Address to be assigned to the phone.

  Enter the desired Static IP Address by pressing the digit keys.

  To enter the dot/period in the IP Address, press the Star '*' key.

Subnet Mask

- If you select Static as Connection Type, enter the Subnet Mask to be applied on the phone by pressing the digit keys.

  To enter the dot/period in the IP Address, press the Star '*' key.

Gateway Address

- If you select Static as Connection Type, enter the Gateway Address here. This is the IP Address of the LAN Port of the Router.

DNS Server

- If you select Static as Connection Type, select the DNS Server option **Static** and configure the DNS Address.

- If you select DHCP or PPPoE as Connection Type and your Internet Service Provider provides DNS Address, select the DNS Server option **Automatic**. However, if your Internet Service Provider does not provide DNS Address, select **Static** and configure the DNS Address.

DNS Address

- If you select DNS Server as Static, enter the DNS Address here.

  To enter dot/period in the IP Address, press the Star '*' key.

DNS Domain Name

- If you select DNS Server as Static, enter the DNS Domain Name here. DNS Domain Name is optional.

PPPoE User ID

- If you have selected PPPoE as Connection Type, you must enter the User ID provided to you by your Internet Service Provider.

PPPoE Password

- This is the password provided by your Internet Service Provider for the PPPoE User ID. If you have selected PPPoE as Connection Type, you must enter the password provided by your Internet Service provider here.

PPPoE Service Name

- If your Internet Service Provider has provided a Service Name, enter the Service Name here. If your Internet Service Provider has not provided a Service Name, do not configure this parameter.

Server Address

- The system works as the Auto Configuration Server for the phone. Enter the LAN or WAN IP Address/ Domain Name of SARVAM UCS here. Default: blank. The phone sends the request for configuration files to this Server Address.

  If you have selected DHCP as Connection Type, the phone will get the Server Address and Port automatically from the DHCP Server. For this, use ***DHCP option 224*** and ***Data Type*** as '***String***' to provide Server Address and Port from the DHCP Server.

  For PPPoE and Static Connection Types, you need to enter the Server Address.

Server Port

- Enter the SPARSH Port of SARVAM UCS here. The phone sends the request for configuration files to this port.

  Valid range of the port is: 80 or 1025–65535. Default: 80.

VLAN Setting

If your phone is connected to a virtual LAN, you need to configure VLAN Settings.

To enable the VLAN switch to correctly route packets generated by the phone and the computers (on the LAN) to each other, the packets must be tagged with a VLAN header.

The VLAN header consists of the VLAN ID (12-bit) and Class of Service (CoS, 3-bit) for prioritization of traffic[74].

The meaning of CoS bits with respect to traffic type is as follows:

CoS	Traffic Type
0	Best Effort
1	Background
2	Spare
3	Excellent Effort
4	Controlled Load
5	Video
6	Voice
7	Network Control

- Select **Phone VLAN/COS** to add VLAN header to the packets generated by the phone, and add VLAN header to the packets relayed from the PC to its LAN port (packets generated by the PC connected to its PC port).

  - To configure Phone VLAN/COS, select **Enable?**. The VLAN ID will be tagged on all packets generated by the phone (SIP, RTP, DNS, ARP, etc.). Default: Disabled.

74. The IEEE 802.1P standard allows Layer2 switches to prioritize the traffic, thus providing Quality of Service (QoS), that is, better handling of data that pass over a network, thereby resulting in greater reliability and quality. Quality of Service (QoS) on Layer2 is referred to as Class of Service (CoS) which is defined by IEEE 802.1P.

    - Select **VLAN ID** and enter the VLAN ID that you have assigned to the VLAN in which the IP Phones are connected. Valid range: 0-4094. Default: 1.
    - Select **SIP CoS** and define the CoS (priority) bits in all SIP packets. Valid range: 0-7. Default: 3
    - Select **RTP CoS** and define the CoS (priority) bits in all RTP packets. Valid range: 0-7. Default: 6.
- Select **PC/VLAN CoS** to add VLAN header to all packets entering the PC Port and leaving the LAN port of the phone. Default: Disabled.
    - To configure PC VLAN/COS, select **Enable?**.
    - Select **VLAN ID** and enter the same ID as you have assigned to the VLAN in which the computers are connected. Valid range: 0-4094. Default: 1.
    - Select **CoS** and define the Layer 2 CoS (priority) bits. Valid range: 0-7. Default: 0.

PCAP

To capture packets sent and received from and by the phone for monitoring and troubleshooting, you can enable PCAP on the phone. The phone captures up to 1 MB of packets. For more information and for instructions on how to use PCAP Trace on the phone, see “Using PCAP Trace for Matrix SPARSH VP248 Extended IP Phone”, under *PCAP Trace*.

When you change the Network Settings, the phone will restart.

MAC Cloning

If you require cloning of the MAC Address, configure the following:

- Select **Enable?**.
- In **Enter Clone MAC Address**, enter the address you wish to clone.

802.1x Authentication

If you want to restrict unauthorized clients from connecting to your LAN, you need to enable 802.1x Authentication. Using EAP MD5 protocol the PC connected to the LAN port of the IP Phone is first authenticated and then it gets connected to LAN.

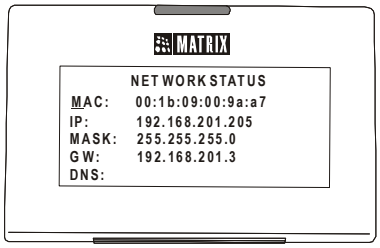
You need to configure the following 802.1x Authentication parameters:

- Select **Enable?**.
- Enter the 802.1x Authentication **Identity** provided by you network administrator.
- Enter the 802.1x Authentication **MD5 Password** associated with identity provided by your network administrator.

Viewing Network Status

- In the Local Menu of the phone, place the cursor on Network Status and press the Enter key.

- The Network Status submenu appears.



Use the Down/Up key to view the status of the various network parameters. The status of the following parameters appear on your display as you scroll.

- **MAC:** This is the MAC Address of the phone.
- **IP:** The current IP Address assigned to the phone.
- **MASK:** The current Subnet mask assigned to the phone.
- **GW:** The current Gateway IP Address assigned to the phone.
- **DNS:** The Domain Name Server address assigned to the phone.
- **S. ADD:** The LAN or WAN IP Address / Domain Name of the SARVAM UCS.
- **S. PORT:** The SPARSH Port SARVAM UCS.
- **DOMAIN:** The Domain Name assigned to the phone.
- **802.1x Authentication:** The 802.1x authentication status is displayed—Success, Failure, Authenticating or Disabled.
- **FIRM:** The version of the current Firmware of the phone.
- **UBOOT**: The UBOOT release date.
- **KERNEL**: The KERNEL release date.

Connecting SPARSH VP310 as Extended SIP Extension

You are recommended to complete the following steps before connecting the Matrix Extended IP Phone to SARVAM UCS:

- Decide the location of the Extended IP Phone, whether within the same network or outside, according to your installation scenario.

*If you want to use the **DHCP Server** for assigning IP Address to the Extended IP Phone, select **DHCP option 224** and **Data Type** as '**String**' and program the LAN or WAN IP Address /Domain Name of SARVAM UCS and SPARSH Port in the format "**IP_Address:Port**" in your DHCP Server as per your installation scenario.*

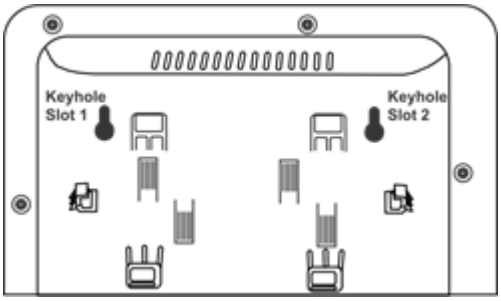
- Log in to *Jeeves*. For instructions, read the topic "Configuring SARVAM UCS".

- Assign an extension number (**SIP ID**) to the Extended IP Phone. For instructions on assigning SIP ID, see "Configuring SIP Extensions".

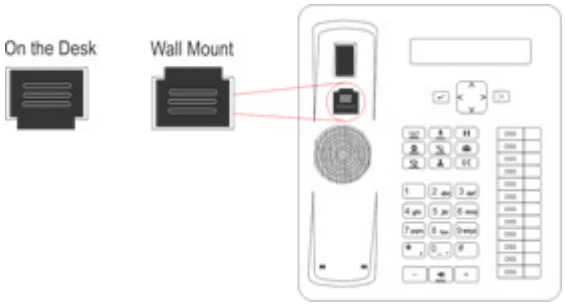
  For the SIP extension number you assigned to the Extended IP Phone, you must configure the necessary parameters in SARVAM UCS so that Extended IP Phone can register as a SIP Extension. For instructions, see the topic "Configuring SIP Extension Settings as per the Extended Phone Type" under *Configuring SIP Extensions*.

Now, follow the steps described below to install the Extended IP Phone.

1. Unpack the SPARSH VP310 box and verify package contents.
2. You can mount the phone on a wall or on the desk.
3. When you mount SPARSH VP310 on a wall,
   - Use the mounting template to drill holes of appropriate size and distance.
   - Fix the screw grips in the holes you drilled.
   - Fix two screws in the holes on the wall, ensuring that they are aligned with the Keyhole Slots 1 and 2 of SPARSH VP310. The screws should protrude from the wall to fit into the Keyhole Slots.

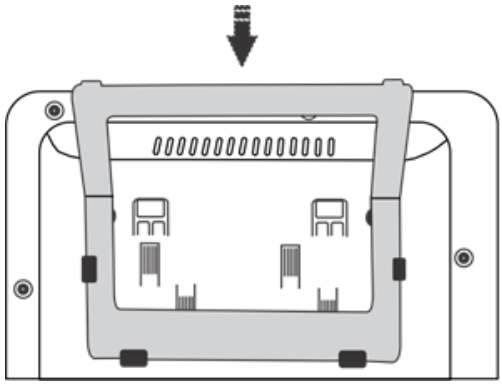


   - Now, mount the phone with the screws fitting into the Keyhole Slot.
   - Reverse the handset wall mount tab to make sure the handset remains intact when you mount the phone. Push the handset wall mount tab upwards to remove it from the slot. Rotate it 180 degrees clockwise and push it downwards into the slot.



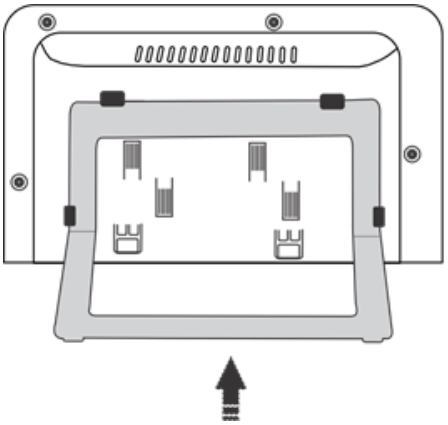
- When you mount the phone on a desk,
  - You can attach the Foot Stand in two ways as illustrated in the following.

Foot Stand attached at 35° Angle



Stand attached at
35 degree angle

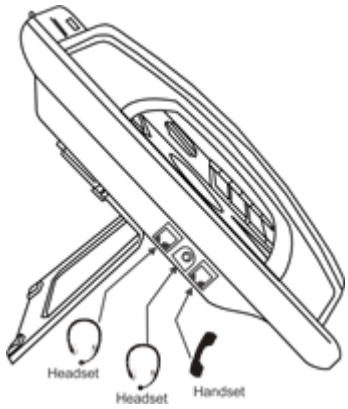
Foot Stand attached at 50° Angle



Stand attached at
50 degree angle

If you attach the Foot Stand at 50°, the phone will be placed in an almost upright position on your desk.

- Decide which of these positions would work for you best and accordingly attach the Foot Stand.



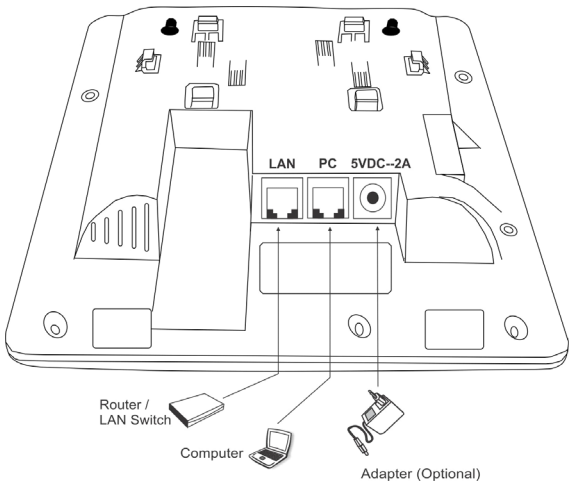
4. Connect the Handset to the Phone body.

- Plug the long straightened end of the phone cord into the handset jack on the left side panel of the phone marked with the handset symbol .
- Plug the other (short straight) end of the phone cord into the jack at the bottom of the handset.

5. If you want to use a Headset (not supplied) with your phone, you may plug a headset with a 3.5 mm single connector into the headset jack headset jack with the symbol on the left side panel of the phone, as illustrated in the figure above.

   OR

   You may also plug in a headset with RJ9 connector into the headset port on the left side panel of the phone, marked with the symbol .



6. Connect the LAN Port of SPARSH VP310 to the LAN Switch/Hub or a Router, according to your installation scenario.

7. To connect your phone to a computer on your desk, use an Ethernet cable (not supplied with this phone) to connect the PC Port of the phone to the LAN Port of the computer.

8. It is a PoE enabled phone and can be powered over Ethernet by connecting it to a PoE enabled LAN Switch (IEEE 802.3af Compliant).In this case you need not connect the Power Adapter.

   If you do not want to use PoE, plug in the connector of the Adapter into the power jack (DC Jack) with the label 5VDC-2A at the bottom of the phone. Plug in the Power Adapter into a power outlet.

If both the power options, that is, PoE as well as Power Adapter are available to the phone, then the phone will derive power from the PoE enabled LAN Switch.

The IP phone should be used with Matrix original power adapter (5V/2A) only. The use of any third-party power adapter may cause damage to the phone.

9. Switch ON power supply.

   When you power the phone, the boot process will be initiated in the following sequence.

   - All keys with LED, including the Speaker key, and the Ringer LED, will glow.
   - The LCD display will light up and the booting message appears.
   - Then the ‘Loading...’ message appears on the phone display.

- The phone will start loading the Extended IP Phone Firmware. It will display current firmware being loaded.
- After loading the firmware, the phone will prompt you to change Network settings.

- Wait for a few seconds.

If you want to change the Network Settings or Server Settings, press the Enter key. Detailed instructions for changing the Network Settings of the phone are provided at the end of this topic. See "Network Settings" at the end of this topic.

- The phone makes DHCP Discovery and fetches its IP Address and Server Address from the DHCP Server.

  On getting the IP Address and Server Address, the phone initiates Auto Configuration to download the configuration files from SARVAM UCS.
- As the phone downloads the configuration files, the file names will appear one by one.
- On successful download of all configuration files, the phone attempts to register with SARVAM UCS.
- On successful registration, the phone will display the current day, date and time, the extension number and name assigned to the Extended IP Phone.

Network Settings

You can change the network settings of the Extended IP Phone. Press the Down key ∨ when the phone is in idle state. To move the cursor and scroll through the menu and submenu options, use the following navigation keys on your phone.

- The Enter key ✓ to make a selection or to complete an action.
- The Up key ∧ to move up the Menu.
- The Down key ∨ move down the Menu.
- The Forward key > move the cursor one character.
- The Back key < to move the cursor one character and to return from the submenu to the main menu.
- The Cancel key ✕ to exit a menu.

The cursor is a non-blinking underscore that appears under the first letter of the first option in the menu. To make a selection in the menu, you must move the cursor in the desired direction using the Up, Down, Forward and Back key. When the cursor is at the desired position, press Enter key to make a selection.

Accessing Network Settings

You can access the Network Settings of the Extended IP Phone in any of the following stages:

1. During start-up, when the phone prompts you to change the network settings after loading the firmware.

   You must press the Enter key ✓ to select Yes and access network settings.

2. When the phone is making Network discovery, downloading configuration files, attempting registration.

   You must press the Down key ⋁ to access network settings.

3. When the phone is in idle state. You must press the Down key ⋁ to access the Network Settings.

Configuring Network Parameters

- When the phone is in idle state. You must press the Down key ⋁ to access the Network Settings.
- Press Enter key to select Network Parameters.
- The Network Parameters submenu appears.
- Use the Down/Up key to reach the desired network parameter and press Enter key to select and change the settings.
- You can configure all network parameters described below, except the MAC Address.

- *To enter a dot in the editable fields — IP Address, Subnet Mask, Gateway Address, DNS Address, Server Address — press * (Star) key.*
- *If you want to clear a single digit/character, move the cursor to the right of the digit/character you wish to clear. Now press the Cancel key. The digit to the left of the cursor will be deleted. If the cursor is to the extreme left and you press the Cancel key, you will go to the previous menu.*

Connection Type

- Select the Connection Type as DHCP, PPPoE or Static according to the IP Addressing scheme of your network.

  If you select DHCP or PPPoE, the phone will be assigned IP Address, Subnet Mask and Gateway Address, DNS Address, Server Address, automatically by the DHCP/PPPoE server.

  For PPPoE Connection Type, you must configure the PPPoE User ID and Password provided by the Internet Service Provider.

  If you select Static, you must assign the IP Address, Subnet Mask and Gateway Address to the phone.

IP Address

- If you select Static as Connection Type, enter the static IP Address to be assigned to the phone. Enter the desired Static IP Address by pressing the digit keys.

  To enter the dot/period in the IP Address, press the Star '*' key.

Subnet Mask

- If you select Static as Connection Type, enter the Subnet Mask to be applied on the phone by pressing the digit keys.

  To enter the dot/period in the IP Address, press the Star '*' key.

Gateway Address

- If you select Static as Connection Type, enter the Gateway Address here. This is the IP Address of the LAN Port of the Router.

DNS Server

- If you select Static as Connection Type, select the DNS Server option **Static** and configure the DNS Address.

- If you select DHCP or PPPoE as Connection Type and your Internet Service Provider provides DNS Address, select the DNS Server option **Automatic**. However, if your Internet Service Provider does not provide DNS Address, select **Static** and configure the DNS Address.

DNS Address

- If you select DNS Server as Static, enter the DNS Address here.
  To enter dot/period in the IP Address, press the Star '*' key.

DNS Domain Name

- If you select DNS Server as Static, enter the DNS Domain Name here. DNS Domain Name is optional.

PPPoE User ID

- If you have selected PPPoE as Connection Type, you must enter the User ID provided to you by your Internet Service Provider.

PPPoE Password

- This is the password provided by your Internet Service Provider for the PPPoE User ID. If you have selected PPPoE as Connection Type, you must enter the password provided by your Internet Service provider here.

PPPoE Service Name

- If your Internet Service Provider has provided a Service Name, enter the Service Name here. If your Internet Service Provider has not provided a Service Name, do not configure this parameter.

Primary Server Address

- The system works as the Auto Configuration Server for the phone. Enter the LAN or WAN IP Address/ Domain Name of SARVAM UCS here. Default: blank. The phone sends the request for configuration files to this Server Address.

If you have selected DHCP as Connection Type, the phone will get the Primary Server Address and Port automatically from the DHCP Server. For this, use ***DHCP option 224*** and ***Data Type*** as '***String***' to provide Server Address and Port from the DHCP Server.

For PPPoE and Static Connection Types, you need to enter the Server Address.

Primary Server Port

- Enter the SPARSH Port of SARVAM UCS here. The phone sends the request for configuration files to this port.

  Valid range of the port is: 80 or 1025–65535. Default: 80.

Secondary Server Address

- If required, you can also configure the Secondary Sever Address as a fallback option. If the registration with the Primary Server fails the phone will send the registration and configuration requests to the Secondary Server Address.Speech-cut or unclear speech may be observed during on-going mature calls.

Secondary Server Port

- Enter the Secondary Server Port. The phone sends the request for configuration files to this port if the Primary Server fails.

  Valid range of the port is: 80 or 1025–65535. Default: 80.

VLAN Setting

If your phone is connected to a virtual LAN, you need to configure VLAN Settings.

To enable the VLAN switch to correctly route packets generated by the phone and the computers (on the LAN) to each other, the packets must be tagged with a VLAN header.

The VLAN header consists of the VLAN ID (12-bit) and Class of Service (CoS, 3-bit) for prioritization of traffic[75].

The meaning of CoS bits with respect to traffic type is as follows:

CoS	Traffic Type
0	Best Effort
1	Background
2	Spare
3	Excellent Effort
4	Controlled Load
5	Video
6	Voice
7	Network Control

75. The IEEE 802.1P standard allows Layer2 switches to prioritize the traffic, thus providing Quality of Service (QoS), that is, better handling of data that pass over a network, thereby resulting in greater reliability and quality. Quality of Service (QoS) on Layer2 is referred to as Class of Service (CoS) which is defined by IEEE 802.1P.

- Select **Phone VLAN/COS** to add VLAN header to the packets generated by the phone, and add VLAN header to the packets relayed from the PC to its LAN port (packets generated by the PC connected to its PC port).

    - To configure Phone VLAN/COS, select **Enable?**. The VLAN ID will be tagged on all packets generated by the phone (SIP, RTP, DNS, ARP, etc.). Default: Disabled.

    - Select **VLAN ID** and enter the VLAN ID that you have assigned to the VLAN in which the IP Phones are connected. Valid range: 0-4094. Default: 1.

    - Select **SIP CoS** and define the CoS (priority) bits in all SIP packets. Valid range: 0-7. Default: 3

    - Select **RTP CoS** and define the CoS (priority) bits in all RTP packets. Valid range: 0-7. Default: 6.

- Select **PC/VLAN CoS** to add VLAN header to all packets entering the PC Port and leaving the LAN port of the phone. Default: Disabled.

    - To configure PC VLAN/COS, select **Enable?**.

    - Select **VLAN ID** and enter the same ID as you have assigned to the VLAN in which the computers are connected. Valid range: 0-4094. Default: 1.

    - Select **CoS** and define the Layer 2 CoS (priority) bits. Valid range: 0-7. Default: 0.

PCAP

To capture packets sent and received from and by the phone for monitoring and troubleshooting, you can enable PCAP on the phone. The phone captures up to 1 MB of packets. For more information and for instructions on how to use PCAP Trace on the phone, see “Using PCAP Trace for Matrix SPARSH VP310 Matrix Extended IP Phone”, under *PCAP Trace*.

MAC Cloning

If you require cloning of the MAC Address, configure the following:

- Select **Enable?** and press the Enter key. Select Yes to enable.

- In **Enter Clone MAC Address**, enter the address you wish to clone.

802.1x Authentication

If you want to restrict unauthorized clients from connecting to your LAN, you need to enable 802.1x Authentication. Using EAP MD5 protocol the PC connected to the LAN port of the IP Phone is first authenticated and then it gets connected to LAN.

You need to configure the following 802.1x Authentication parameters:

- Select **Enable?** and press the Enter key. Select Yes to enable.

- Enter the 802.1x Authentication **Identity** provided by you network administrator.

- Enter the 802.1x Authentication **MD5 Password** associated with identity provided by your network administrator.

When you change the Network Settings, the phone will restart.

Viewing Network Status

- When the phone is in idle state. You must press the Down key ⋁ to access the Network Settings.
- Again press Down key ⋁ to select Network Status and press the Enter key ✓.

  Use the Down/Up key to view the status of the various network parameters. The status of the following parameters appear on your display as you scroll.

  - **MAC:** This is the MAC Address of the phone.
  - **IP:** The current IP Address assigned to the phone.
  - **MASK:** The current Subnet mask assigned to the phone.
  - **GW:** The current Gateway IP Address assigned to the phone.
  - **DNS:** The Domain Name Server address assigned to the phone.
  - **Active Server:** This displays the Server that is active — Primary, Secondary — with which the phone is currently registered.
  - **S. ADD:** This displays the IP address of the Active Server. It may be the LAN or WAN IP Address / Domain Name of the SARVAM UCS or the Secondary Server IP Address (if configured) or any Fallback Server.
  - **S. PORT:** This displays the port of the Active Server. It may be the SPARSH Port of SARVAM UCS or the Secondary Server Port (if configured) or the Fallback Server Port.
  - **DOMAIN:** The Domain Name assigned to the phone.
  - **802.1x Authentication:** The 802.1x authentication status is displayed—Success, Failure, Authenticating or Disabled.
  - **FIRM:** The version of the current Firmware of the phone.
  - **UBOOT**: The UBOOT release date.
  - **KERNEL**: The KERNEL release date.

Connecting SPARSH VP330 as Extended SIP Extension

You are recommended to complete the following steps before connecting the Matrix SPARSH VP330 to SARVAM UCS:

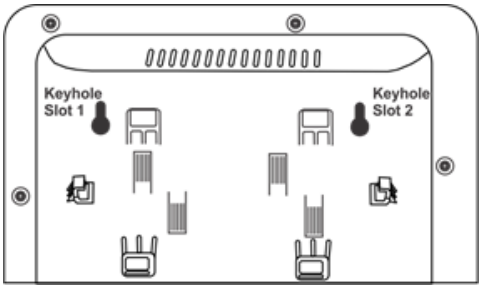
- Decide the location where you want to place SPARSH VP330 within your LAN.
- By Default, in SPARSH VP330, the Connection Type selected is DHCP.
- If you want to use the **DHCP Server** for assigning IP Address to the Extended IP Phone, select **DHCP option 224** and **Data Type** as '**String**' and program the LAN or WAN IP Address /Domain Name of

SARVAM UCS and SPARSH Port in the format "**IP_Address:Port"** in your LAN DHCP Server as per your installation scenario.

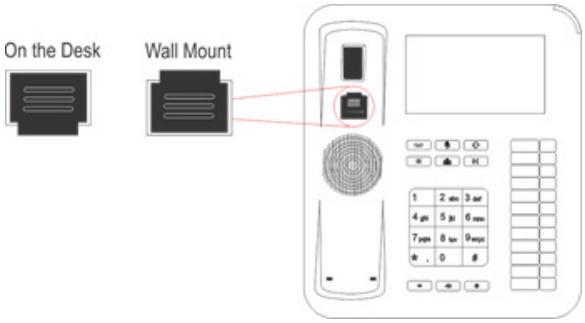
- Log in to *Jeeves*. For instructions, read the topic "Configuring SARVAM UCS".
- You must configure the necessary parameters in SARVAM UCS so that SPARSH VP330 can register as a SIP Extension. For instructions, see "Configuring Matrix SPARSH VP330".

Now, follow the steps described below to install SPARSH VP330.

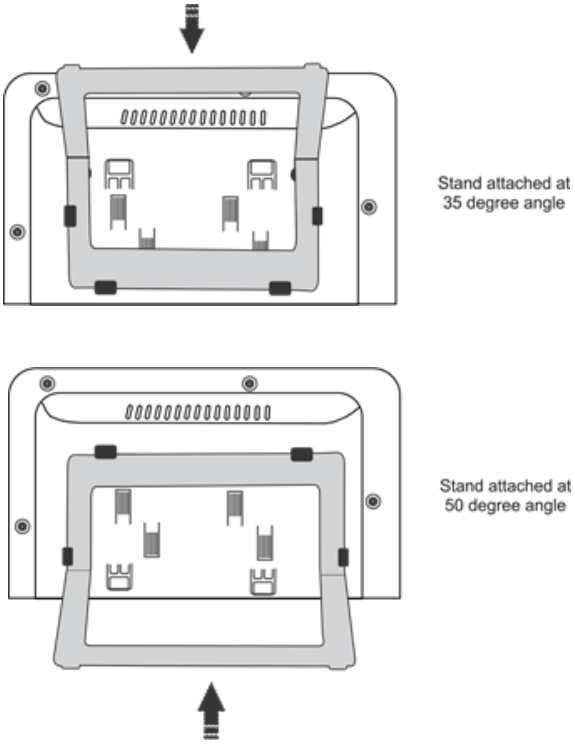
1. Unpack the SPARSH VP330 box and verify package contents.
2. Mount the phone on a desk or wall at a location convenient to you.
    - When mounting the phone on the wall,
        - Use the mounting template to drill holes of appropriate size and distance. Fix the screw grips in the holes you drilled.
        - Fix two screws in the holes on the wall, ensuring that they are aligned with the Keyhole Slots 1 and 2.
        - Use wall plugs, if required, to fix the screws. Leave the screw heads protruding from the wall to fit into the Keyholes.



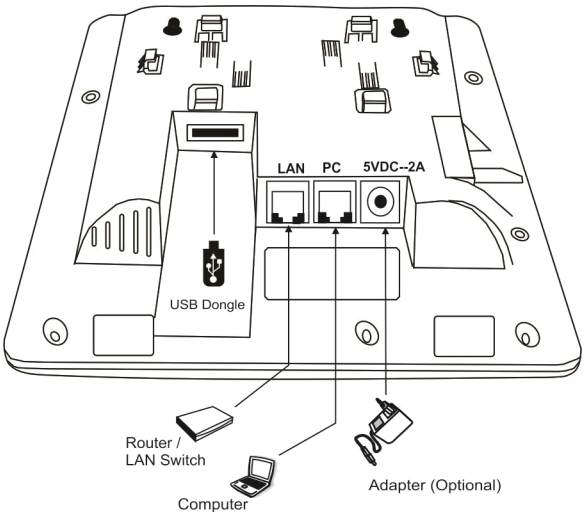
        - Now, mount the phone on the wall, with the screws fitting into the Keyhole slots.
        - Reverse the handset wall mount tab to make sure the handset remains intact when you mount the phone. Push the handset wall mount tab upwards to remove it from the slot. Rotate it 180 degrees clockwise and push it downwards into the slot.



3. When you mount the phone on a desk, you can attach the Foot Stand in two ways at **35° Angle** or at **50° Angle**.



- If you attach the Foot Stand at 50°, the phone will be placed in an almost upright position on your desk.
- Decide which of these positions would work for you best and accordingly attach the Foot Stand.



4. Connect the Handset to the Phone body.

- Plug the long straightened end of the phone cord into the handset jack on the left side panel of the phone marked with the handset symbol.

- Plug the other (short straight) end of the phone cord into the jack at the bottom of the handset.

5. If you want to use a Headset (not supplied) with your phone, you may plug a headset with a 3.5 mm single connector into the headset jack headset jack with the symbol on the left side panel of the phone.

   OR

   You may plug a headset with an RJ9 connector into the headset port on the side panel of the phone, marked with the symbol .

6. Connect the LAN Port of SPARSH VP330 to the IP Network — A Router or LAN Switch — using the Ethernet Cable.

   OR
   Connect the Wi-Fi USB Adapter into the USB port of the phone.

You can purchase the Wi-Fi USB Adapter from Matrix as an optional peripheral device to support wireless network.

7. To connect your phone to a computer on your desk, use an Ethernet cable (not supplied with this phone) to connect the PC Port at the bottom of the phone to the LAN Port of the computer.

8. It is a PoE enabled phone and can be powered over Ethernet by connecting it to a PoE enabled LAN Switch (IEEE 802.3af Compliant). In this case you need not connect the Power Adapter.

   If you do not want to use PoE, plug in the connector of the Adapter into the power jack (DC Jack) with the label 5VDC-2A at the bottom of the phone. Plug in the Power Adapter into a power outlet.

If both the power options, that is, PoE as well as Power Adapter are available to the phone, then the phone will derive power from the PoE enabled LAN Switch.

The IP phone should be used with Matrix original power adapter (5V/2A) only. The use of any third-party power adapter may cause damage to the phone.

9. Switch ON power supply.

   When you power the phone, the boot process will be initiated in the following sequence.

   - All keys with LED, including the Speaker key, and the Ringer LED, will glow.
   - The LCD display will light up and booting message appears.
   - While loading the application then the loading message appears on the phone display.
   - The phone makes DHCP Discovery and fetches its IP Address and Server Address from the DHCP Server.

If you want to change the Network Settings/Server Settings or want to use Wi-Fi for connectivity, press Settings .

Refer to the SPARSH VP330 User Guide, for detailed instructions:

- *To change the Network Settings of the phone and configure the network parameters.*
- *To use Wi-Fi for connectivity and configure its parameters.*

- On getting the IP Address and Server Address, the phone initiates Auto Configuration to download the configuration files from SARVAM UCS.
- On successful download of all configuration files, the phone attempts to register with SARVAM UCS.
- On successful registration, the Home screen appears.

The phone will register successfully, only if the SIP Extension parameters in SARVAM UCS have been correctly configured as per your installation scenario.

Refer to the ***SPARSH VP330 User Guide*** to know more.

Connecting SPARSH VP510 as Extended SIP Extension

You are recommended to complete the following steps before connecting the Matrix Extended IP Phone to the system when used with SARVAM UCS application:

- Decide the location of the Extended IP Phone, whether within the same network or outside, according to your installation scenario.

*If you want to use the **DHCP Server** for assigning IP Address to the Extended IP Phone, select **DHCP option 224** and **Data Type** as '**String**' and program the LAN or WAN Port IP Address /Domain Name and SPARSH Port in the format "**IP_Address:Port**" in your DHCP Server as per your installation scenario.*

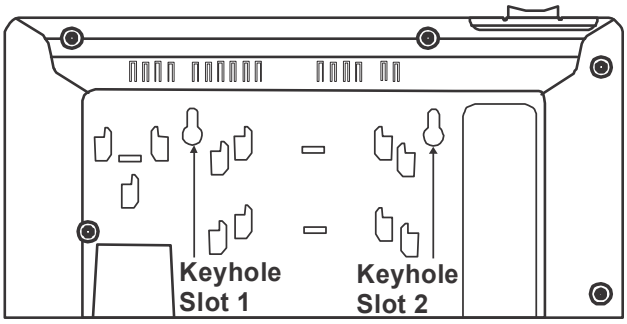
- Log in to *Jeeves*. For instructions, read the topic "Configuring SARVAM UCS".
- Assign an extension number (**SIP ID**) to the Extended IP Phone. For instructions on assigning SIP ID, see "Configuring SIP Extensions".

  For the SIP extension number you assigned to the Extended IP Phone, you must configure the necessary parameters in SARVAM UCS so that Extended IP Phone can register as a SIP Extension. For instructions, see the topic "Configuring SIP Extension Settings as per the Extended Phone Type" under *Configuring SIP Extensions*.

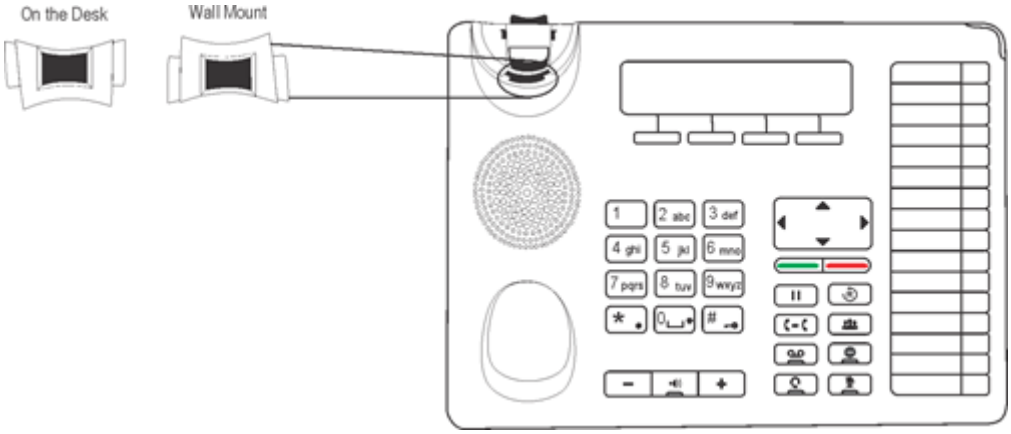
Now, follow the steps described below to install the Extended IP Phone:

1. Unpack the SPARSH VP510 box and verify package contents.
2. You can mount the phone on a wall or on the desk.
3. When you mount SPARSH VP510 on a wall,

- Use the mounting template to drill holed of appropriate size and distance.
- Fix the screw grips in the holes you drilled.
- Fix two screws in the holes on the wall, ensuring that they are aligned with the Keyhole Slots 1 and 2 of SPARSH VP510. The screws should protrude from the wall to fit into the Keyhole Slots.



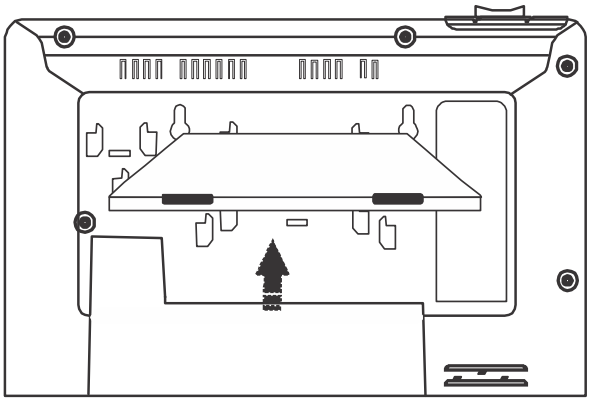
- Now, mount the phone with the screws into the Keyhole Slots.
- Reverse the handset wall mount tab to make sure that the handset remains intact when you mount the phone. Push the handset wall mount tab upwards to remove it from the slot. Rotate it 180 degrees clockwise and push it downwards into the slot.



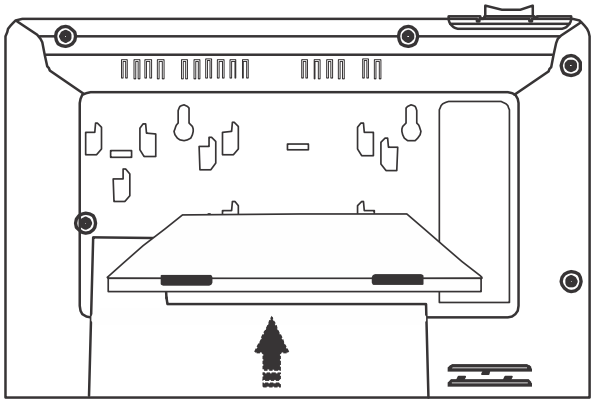
If you are unable to remove the wall mount tab, you may use a tool like a minus screw-driver to remove it.

- When you mount the phone on a desk,

- You can attach the Foot Stand in the following ways — at an angle of 45 degrees or 55 degrees



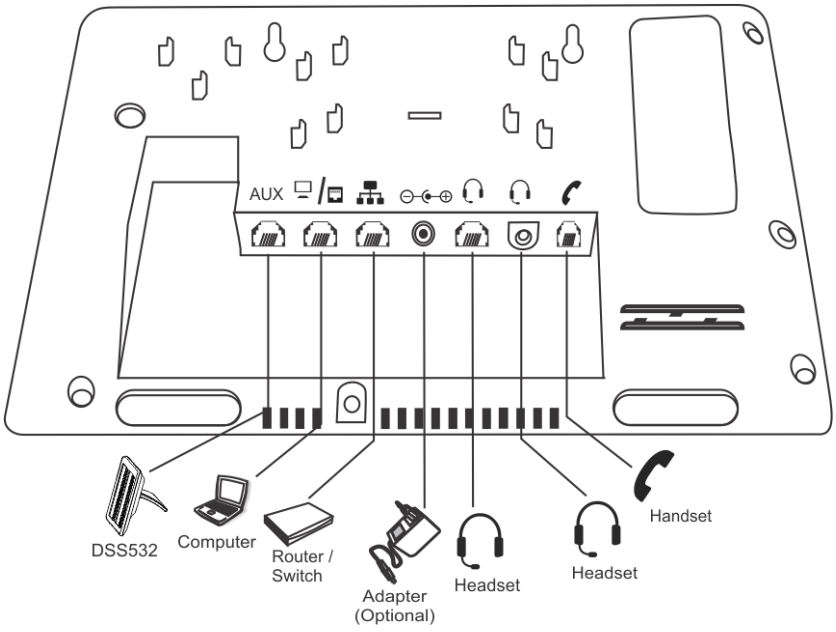
Stand attached at
45 degree angle



Stand attached at
55 degree angle

- Decide which of these positions would work for you best and accordingly attach the Foot Stand.

Refer to the diagram below for connectivity.



4. Connect the Handset.

   - Plug the long straightened end of the Spring Cord into the handset jack at the bottom of the phone, marked with the handset symbol .
   - Plug the other (short straight) end of the Spring Cord into the jack at the bottom of the handset.

5. Connect the Headset (not supplied by Matrix).

   - To use a Headset (not supplied with the phone), plug any standard stereo headset with 3.5mm single connector into the headset audio jack at the bottom of the phone, marked with the symbol .

     OR

     You may also plug in a headset with an RJ9 connector into the headset port at the bottom of the phone, marked with the symbol .

6. Connect to the IP Network.

   - Plug one end of the Ethernet Cable into the LAN Port at the bottom of the phone, marked with the symbol and the other end to the IP Network — A Router or LAN Switch.

7. Connect a PC to the Phone.

   - Plug one end of the Ethernet Cable into the PC Port at the bottom of the phone, marked with the symbol and the other end into the LAN Port of your PC/LAN Switch.

8. Connect DSS532 with the Phone.

   - To connect DSS532 with the phone, plug one end of the RJ11 cable into the AUX Port of the phone and the other end into the IN Port of the DSS532. For installation, see “Installing DSS532 with SPARSH VP510”.

9. Connect the Power Supply.

   - It is a PoE enabled phone and can be powered over Ethernet by connecting it to a PoE enabled LAN Switch (IEEE 802.3af Compliant).

     If you do not want to use PoE, plug in the connector of the Adapter into the power jack (DC Jack) at the bottom of the phone, marked with the symbol . Plug in the Power Adapter into a power outlet.

If both the power options, that is, PoE as well as Power Adapter are available to the phone, then the phone will derive power from the PoE enabled LAN Switch.

The IP phone should be used with Matrix original power adapter (5V/2A) only. The use of any third-party power adapter may cause damage to the phone.

10. Switch ON power supply.

    When you power the phone, the boot process will be initiated in the following sequence.

    - All keys with LED, including the Speaker key, and the Ringer LED, will glow.
    - The LCD display will light up and the booting message appears.
    - Then the 'Loading...' message appears on the phone display.
    - The phone will start loading the Extended IP Phone Firmware. It will display current firmware being loaded.
    - After loading the firmware, the phone will prompt you to change Network settings.
    - Wait for a few seconds.

If you want to change the Network Settings or Server Settings, press Yes key. Detailed instructions for changing the Network Settings of the phone are provided at the end of this topic. See "Network Settings" at the end of this topic.

- The phone makes DHCP Discovery and fetches its IP Address and Server Address from the DHCP Server.

  On getting the IP Address and Server Address, the phone initiates Auto Configuration to download the configuration files from SARVAM UCS.

- As the phone downloads the configuration files, the file names will appear one by one.
- On successful download of all configuration files, the phone attempts to register with SARVAM UCS.
- On successful registration, the phone will display the current day, date and time, the extension number and name assigned to the Extended IP Phone.

Network Settings

You can change the network settings of the Extended IP Phone. Press the Down key ⋁ when the phone is in idle state.

To navigate the menu,

- Press the Menu Key when the phone is idle.
- Scroll by pressing the Up/Down Navigation Key to reach the desired Menu option.
- Press the Select / OK Key to select the desired Menu option.
- Scroll by pressing the Up/Down Navigation Key to reach the desired sub-menu option.
- Press the Select / OK Key to select the desired sub-menu option.

To exit menu,

- Press Cancel Key.
  or
  Go ON-Hook.

The cursor is a non-blinking underscore that appears under the first letter of the first option in the menu. To make a selection in the menu, you must move the cursor in the desired direction using the Up, Down, Forward and Back key. When the cursor is at the desired position, press Enter key to make a selection.

Accessing Network Settings

You can access the Network Settings of the Extended IP Phone in any of the following stages:

1. During start-up, when the phone prompts you to change the network settings after loading the firmware.

   You must press **Yes** key and access network settings.

2. When the phone is making Network discovery, downloading configuration files, attempting registration.

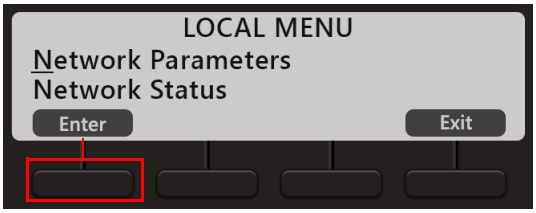
   You must press the **Menu** key to access network settings.

3. When the phone is in idle state, press the Down key ⋁ .

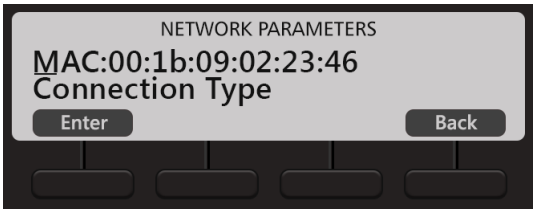
You can configure Network Parameters and view Network status from the Local Menu.

Configuring Network Parameters

- In the Local Menu of the phone, select Network Parameters by pressing the Enter Key.



- The Network Parameters submenu appears.



- Use the Down/Up key to reach the desired network parameter and press Enter key to select. Change the settings as per your requirements.
- Press **Save** key, to save the changes you make.
- You can configure all network parameters described below, except the MAC Address.

- *To enter a dot in the editable fields — IP Address, Subnet Mask, Gateway Address, DNS Address, Server Address — press * (Star) key.*
- *If you want to clear a single digit/character, move the cursor to the right of the digit/character you wish to clear. Now press the Delete key. The digit to the left of the cursor will be deleted.*

Before you start configuring the Network Parameters, get acquainted with following context keys:

Context Keys	Description
Enter/OK	To select a particular parameter
Save	To save the changes
Back	To move a step backwards without saving the changes
Delete	To delete previous characters from the cursor position
2Ab/123	2Ab - Alphanumeric Mode 123 - Numeric Mode

Connection Type

- Select the Connection Type as DHCP, PPPoE or Static according to the IP Addressing scheme of your network.

  If you select DHCP or PPPoE, the phone will be assigned IP Address, Subnet Mask and Gateway Address, DNS Address, Server Address, automatically by the DHCP/PPPoE server.

  For PPPoE Connection Type, you must configure the PPPoE User ID and Password provided by the Internet Service Provider.

  If you select Static, you must assign the IP Address, Subnet Mask and Gateway Address to the phone.

IP Address

- If you select Static as Connection Type, enter the static IP Address to be assigned to the phone.

  Enter the desired Static IP Address by pressing the digit keys.

To enter the dot/period in the IP Address, press the Star '*' key.

Subnet Mask

- If you select Static as Connection Type, enter the Subnet Mask to be applied on the phone by pressing the digit keys.

  To enter the dot/period in the IP Address, press the Star '*' key.

Gateway Address

- If you select Static as Connection Type, enter the Gateway Address here. This is the IP Address of the LAN Port of the Router.

DNS Server

- If you select Static as Connection Type, select the DNS Server option **Static** and configure the DNS Address.

- If you select DHCP or PPPoE as Connection Type and your Internet Service Provider provides DNS Address, select the DNS Server option **Automatic**. However, if your Internet Service Provider does not provide DNS Address, select **Static** and configure the DNS Address.

DNS Address

- If you select DNS Server as Static, enter the DNS Address here.

  To enter dot/period in the IP Address, press the Star '*' key.

DNS Domain Name

- If you select DNS Server as Static, enter the DNS Domain Name here. DNS Domain Name is optional.

PPPoE User ID

- If you have selected PPPoE as Connection Type, you must enter the User ID provided to you by your Internet Service Provider.

PPPoE Password

- This is the password provided by your Internet Service Provider for the PPPoE User ID. If you have selected PPPoE as Connection Type, you must enter the password provided by your Internet Service provider here.

PPPoE Service Name

- If your Internet Service Provider has provided a Service Name, enter the Service Name here. If your Internet Service Provider has not provided a Service Name, do not configure this parameter.

Primary Server Address

- The system works as the Auto Configuration Server for the phone. Enter the LAN or WAN IP Address/ Domain Name of SARVAM UCS here. Default: blank. The phone sends the request for configuration files to this Server Address.

  If you have selected DHCP as Connection Type, the phone will get the Primary Server Address and Port automatically from the DHCP Server. For this, use ***DHCP option 224*** and ***Data Type*** as '***String***' to provide Server Address and Port from the DHCP Server.

For PPPoE and Static Connection Types, you need to enter the Server Address.

Primary Server Port

- Enter the SPARSH Port of SARVAM UCS here. The phone sends the request for configuration files to this port.

  Valid range of the port is: 80 or 1025–65535. Default: 80.

Secondary Server Address

- If required, you can also configure the Secondary Sever Address as a fallback option. If the registration with the Primary Server fails the phone will send the registration and configuration requests to the Secondary Server Address. Speech-cut or unclear speech may be observed during on-going mature calls.

Secondary Server Port

- Enter the Secondary Server Port. The phone sends the request for configuration files to this port if the Primary Server fails.

  Valid range of the port is: 80 or 1025–65535. Default: 80.

VLAN Setting

If your phone is connected to a virtual LAN, you need to configure VLAN Settings.

To enable the VLAN switch to correctly route packets generated by the phone and the computers (on the LAN) to each other, the packets must be tagged with a VLAN header.

The VLAN header consists of the VLAN ID (12-bit) and Class of Service (CoS, 3-bit) for prioritization of traffic[76]. The meaning of CoS bits with respect to traffic type is as follows:

CoS	Traffic Type
0	Best Effort
1	Background
2	Spare
3	Excellent Effort
4	Controlled Load
5	Video
6	Voice
7	Network Control

- Select **Phone VLAN/COS** to add VLAN header to the packets generated by the phone, and add VLAN header to the packets relayed from the PC to its LAN port (packets generated by the PC connected to its PC port).

  - To configure Phone VLAN/COS, select **Enable?**. The VLAN ID will be tagged on all packets generated by the phone (SIP, RTP, DNS, ARP, etc.). Default: Disabled.

76. The IEEE 802.1P standard allows Layer2 switches to prioritize the traffic, thus providing Quality of Service (QoS), that is, better handling of data that pass over a network, thereby resulting in greater reliability and quality. Quality of Service (QoS) on Layer2 is referred to as Class of Service (CoS) which is defined by IEEE 802.1P.

- Select **VLAN ID** and enter the VLAN ID that you have assigned to the VLAN in which the IP Phones are connected. Valid range: 0-4094. Default: 1.
- Select **SIP CoS** and define the CoS (priority) bits in all SIP packets. Valid range: 0-7. Default: 3
- Select **RTP CoS** and define the CoS (priority) bits in all RTP packets. Valid range: 0-7. Default: 6.

- Select **PC/VLAN CoS** to add VLAN header to all packets entering the PC Port and leaving the LAN port of the phone. Default: Disabled.
  - To configure PC VLAN/COS, select **Enable?**.
  - Select **VLAN ID** and enter the same ID as you have assigned to the VLAN in which the computers are connected. Valid range: 0-4094. Default: 1.
  - Select **CoS** and define the Layer 2 CoS (priority) bits. Valid range: 0-7. Default: 0.

PCAP

To capture packets sent and received from and by the phone for monitoring and troubleshooting, you can enable PCAP on the phone. The phone captures up to 1 MB of packets. For more information and instructions on how to use PCAP Trace on the phone, refer to the *EON510_SPARSH VP510 User Guide*.

MAC Cloning

If you require cloning of the MAC Address, configure the following:

- Select **Enable?** and press the Enter key. Select Yes to enable.
- In **Enter Clone MAC Address**, enter the address you wish to clone.

802.1x Authentication

If you want to restrict unauthorized clients from connecting to your LAN, you need to enable 802.1x Authentication. Using EAP MD5 protocol the PC connected to the LAN port of the IP Phone is first authenticated and then it gets connected to LAN.

You need to configure the following 802.1x Authentication parameters:

- Select **Enable?** and press the Enter key. Select Yes to enable.
- Enter the 802.1x Authentication **Identity** provided by you network administrator.
- Enter the 802.1x Authentication **MD5 Password** associated with identity provided by your network administrator.

When you change the Network Settings, the phone will restart.

Viewing Network Status

- When the phone is in idle state. You must press the Down key ⋁ to access the Network Settings.
- Again press Down key ⋁ to select Network Status and press the Enter key.

Use the Down/Up key to view the status of the various network parameters. The status of the following parameters appear on your display as you scroll.

- **MAC:** This is the MAC Address of the phone.
- **IP:** The current IP Address assigned to the phone.
- **MASK:** The current Subnet mask assigned to the phone.
- **GW:** The current Gateway IP Address assigned to the phone.
- **DNS:** The Domain Name Server address assigned to the phone.
- **Active Server:** This displays the Server that is active — Primary, Secondary — with which the phone is currently registered.
- **S. ADD:** This displays the IP address of the Active Server. It may be the LAN or WAN IP Address / Domain Name of the SARVAM UCS or the Secondary Server IP Address (if configured) or any Fallback Server.
- **S. PORT:** This displays the port of the Active Server. It may be the SPARSH Port of SARVAM UCS or the Secondary Server Port (if configured) or the Fallback Server Port.
- **DOMAIN:** The Domain Name assigned to the phone.
- **802.1x Authentication:** The 802.1x authentication status is displayed—Success, Failure, Authenticating or Disabled.
- **FIRM:** The version of the current Firmware of the phone.
- **UBOOT**: The UBOOT release date.
- **KERNEL**: The KERNEL release date.

Refer to the ***EON510_SPARSH VP510 User Guide*** to know more.

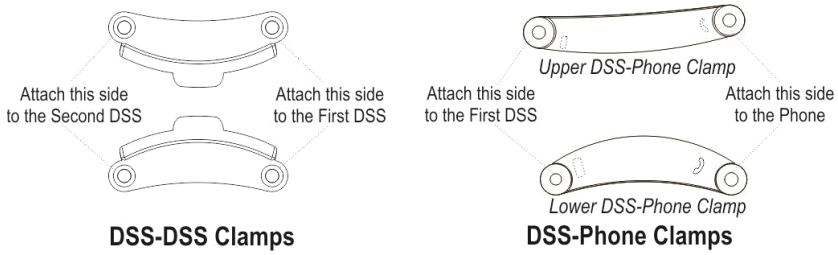
Installing DSS532 with SPARSH VP510

Once you have installed SPARSH VP510 with SARVAM UCS, you can install the DSS532 by following the steps given below:

1. Unpack the box and verify the package contents[77].

77. *See "Packing List" of Appendix topic.*

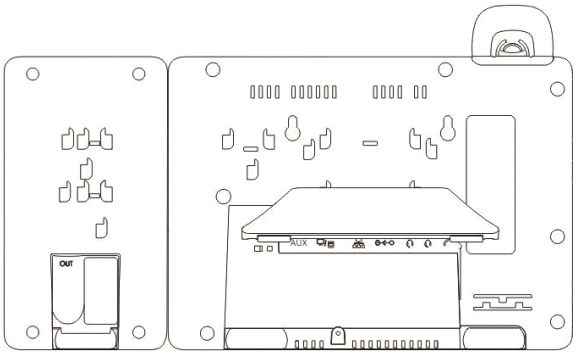
2. Four clamps are provided with the phone — 2 DSS-Phone Clamps and 2 DSS-DSS Clamp.



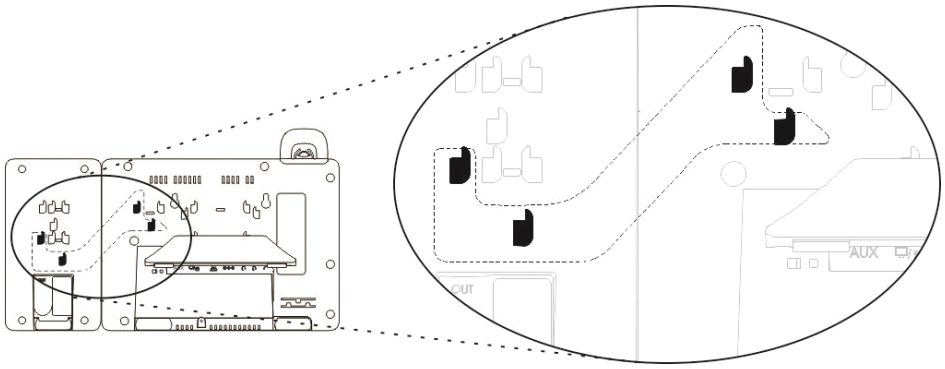
Connecting the First DSS532

Connecting the Extender

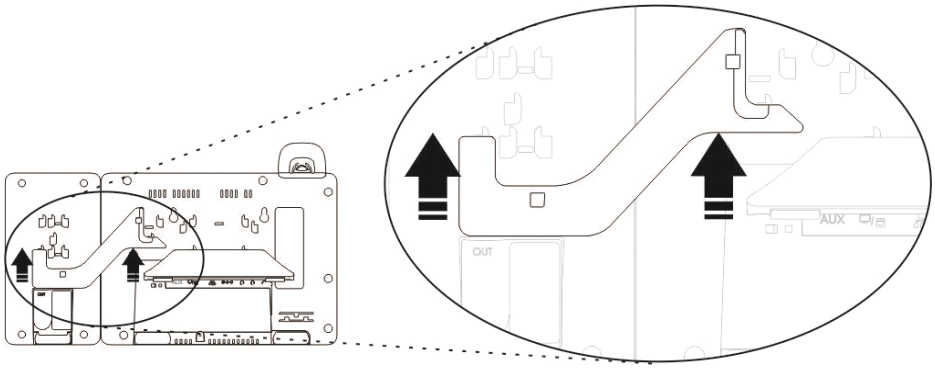
3. Turn the phone upside down on the table and place the inverted DSS532 adjacent to it.



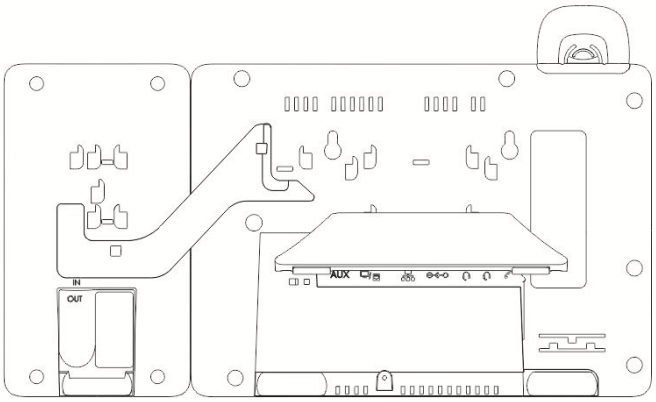
4. To attach the DSS532 with the phone, place the DSS Extender as illustrated below.



5. Insert the hooks on the Extender into the slots provided on the phone and the DSS532.

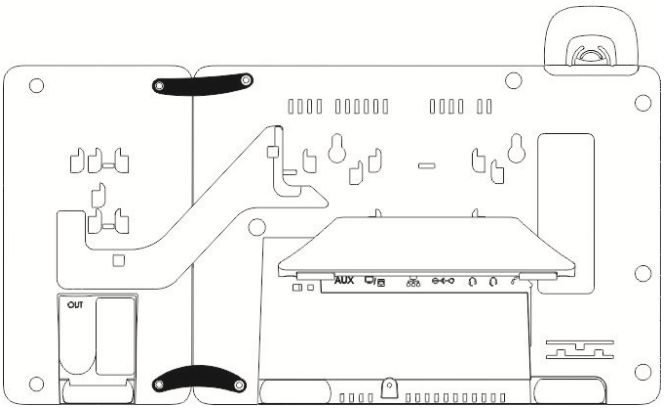


6. Firmly slide the DSS Extender upwards to lock them in place.



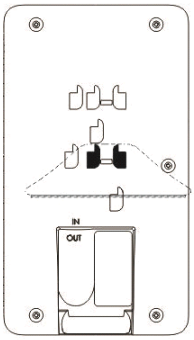
Attaching the Clamps

7. Now attach the clamps. To do so,
   - Remove the screws to attach the clamps.
   - Place the DSS-Phone Clamps between the DSS532 and the phone.
   - Insert the screws back to fix the clamps.

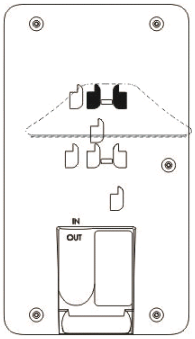


Attaching the Footstand

8. You can mount the DSS532 with the phone on the desk at two angles — **45 degrees** or **55 degrees** by attaching the Foot Stand.



Stand attached at 45 degree angle

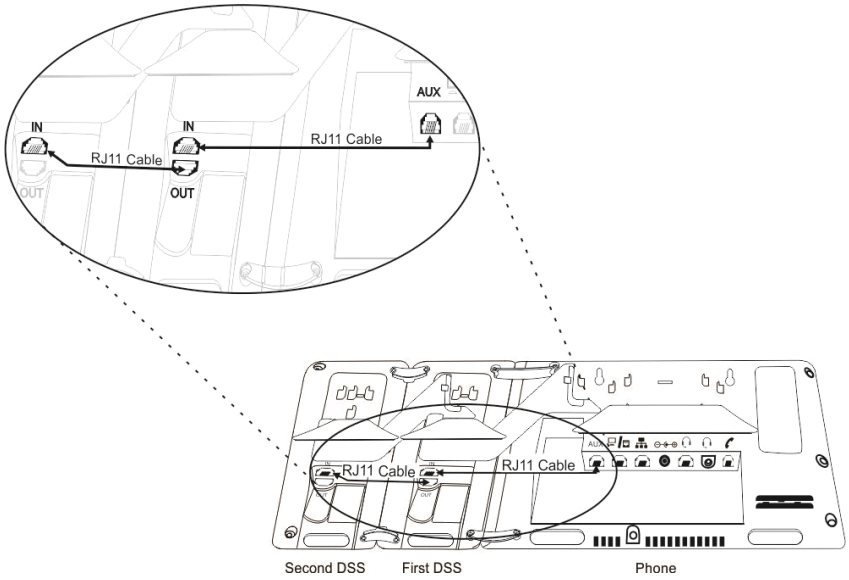


Stand attached at 55 degree angle

Make sure both, the DSS532 and phone are mounted at the same angle.

Connecting the Cables

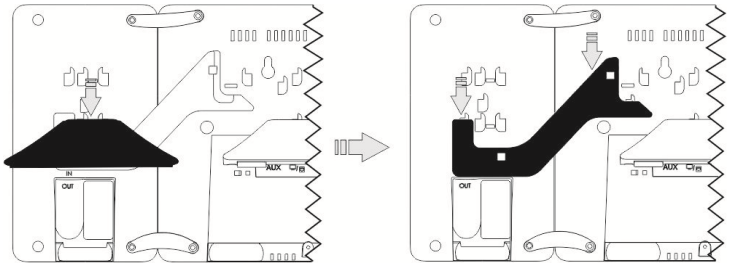
9. To connect the DSS532 with phone, plug one end of RJ11 Cable into **Auxiliary(AUX) Port** of the phone and the other end into the **IN Port** of the DSS532.



Connecting Multiple DSS532

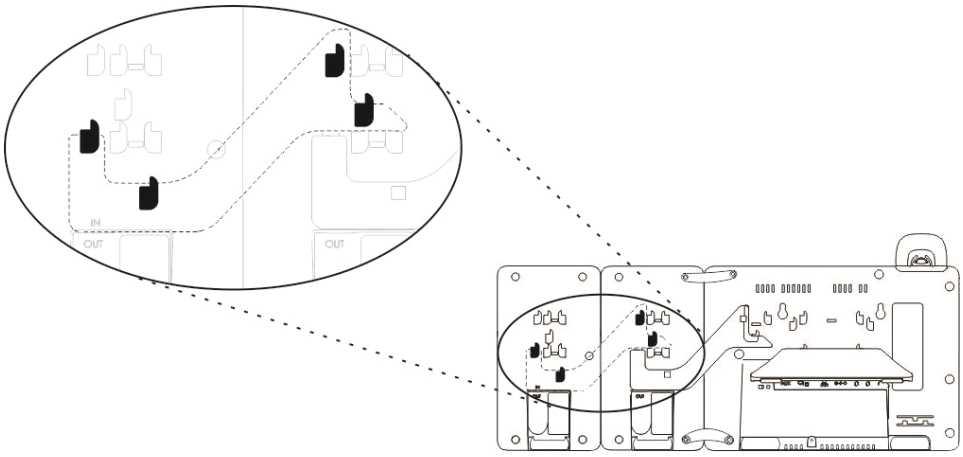
Remove the Foot Stand

10. Remove the Foot Stand of attached DSS532. To do so,
    - Firmly slide the Foot Stand of the attached DSS532 downward to unlock.
    - Now, slide down the attached DSS Extender in downward direction.

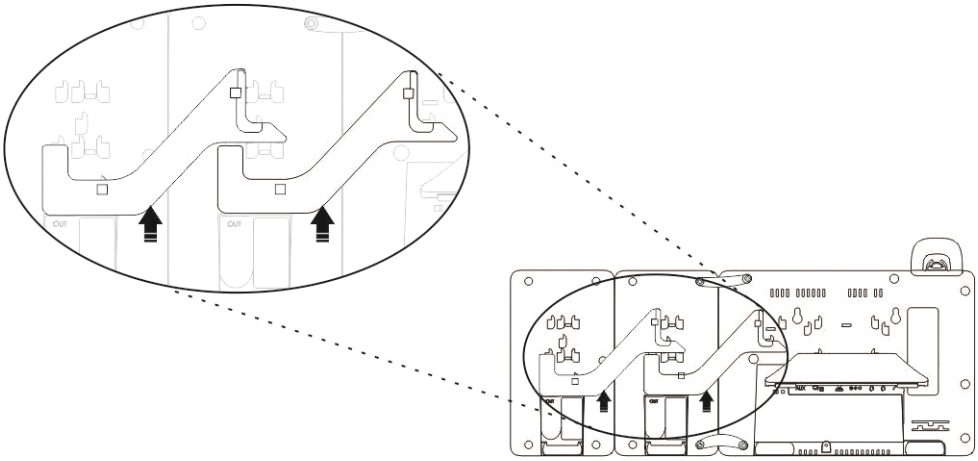


Attach the second DSS Extender

11. To attach the second DSS Extender,
    - Place another inverted DSS532 adjacent to the existing assembly.
    - Place the DSS Extender as illustrated in the diagram below.
    - Insert the hooks on the Extender into the slots provided on both the DSS532.

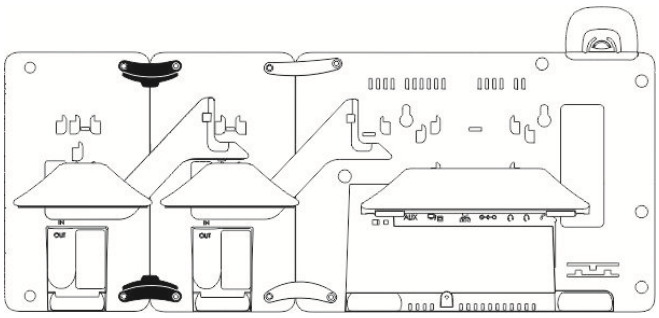


12. Firmly slide both the DSS Extenders upward consecutively (attach the second extender first followed by the existing one attached to the phone) and lock them in place.



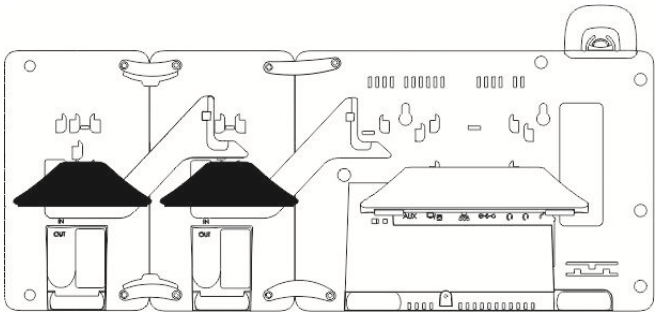
Attach the Clamps

13. Attach the DSS-DSS Clamps between both the DSS532.



Attach the Foot Stand

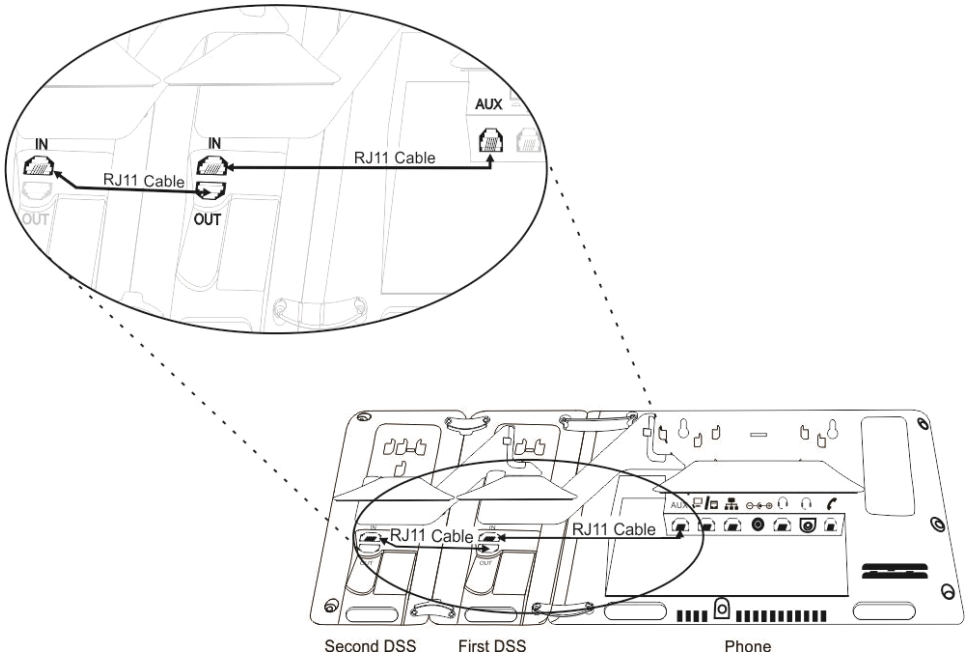
14. Attach the Foot Stand of both the DSS532.



Make sure both, the DSS532 and the phone are mounted at the same angle.

Connect the second DSS532 to the existing assembly

15. Plug one end of the RJ11 Cable into the OUT Port of the existing DSS532 (already connected with the phone) and the other end into the IN Port of the second DSS532.



You can install a maximum of four DSS532 with a phone.

16. After you have connected the DSS532 with the phone, you can configure the DSS Keys. For instructions, see “Programming DSS Console Keys”.

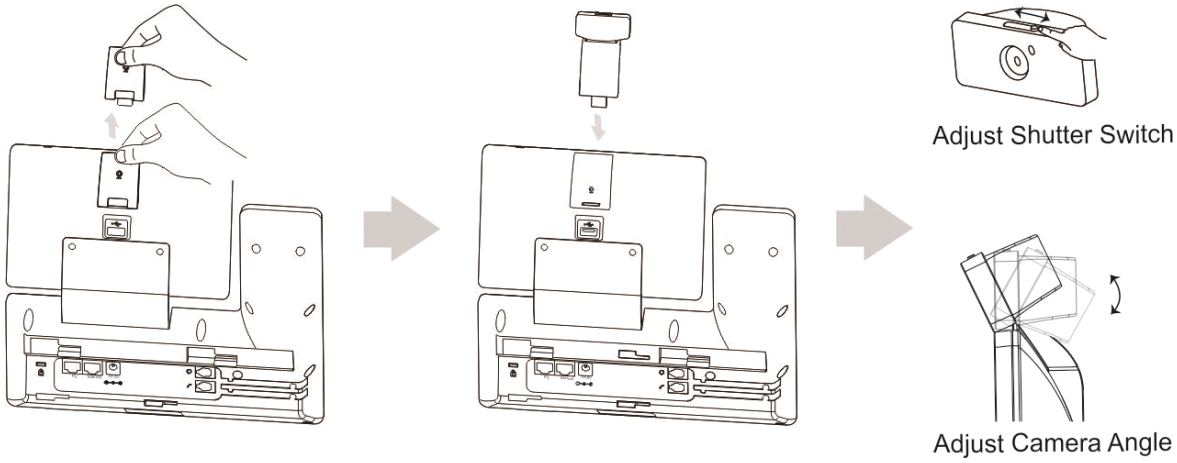
Connecting Extended SPARSH VP710 as Extended SIP Extension

You are recommended to complete the following steps before connecting the Matrix Extended SPARSH VP710 to SARVAM UCS:

- Decide the location where you want to place Matrix Extended SPARSH VP710 within your LAN.
- Log in to *Jeeves*. For instructions, read the topic “Configuring SARVAM UCS”.
- You must configure the necessary parameters in SARVAM UCS so that Extended SPARSH VP710 can register as a SIP Extension. For instructions, see “Configuring Matrix Extended SPARSH VP710”.

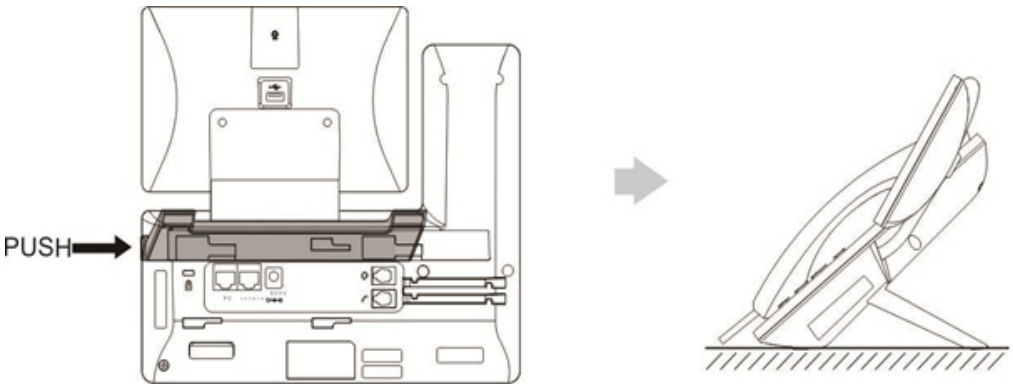
Now, follow the steps described below to install Extended SPARSH VP710.

1. **Inserting the camera**

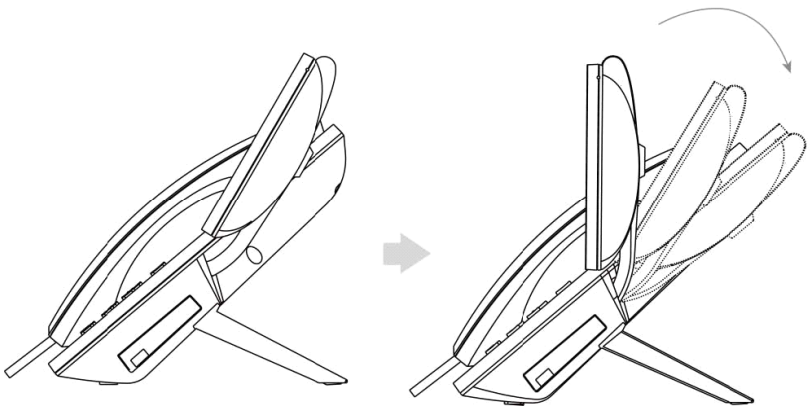


It is recommended to use only the Matrix original Camera, supplied with the IP Phone for video calling. The use of any third-party camera may cause damage to the phone. Damages to the phone caused by using third-party camera is not covered by Matrix warranty.

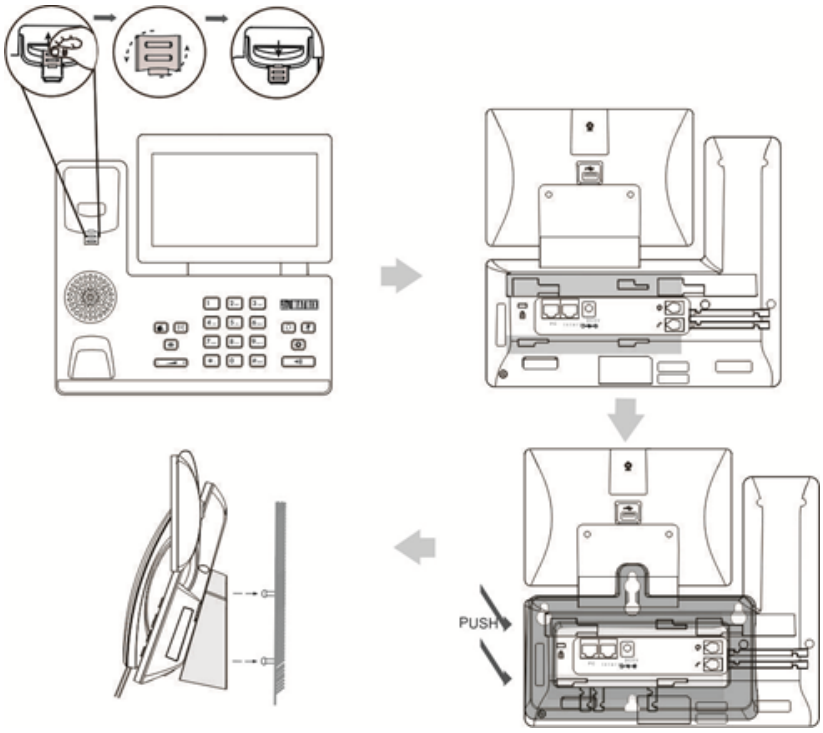
2. **Attaching the stand**



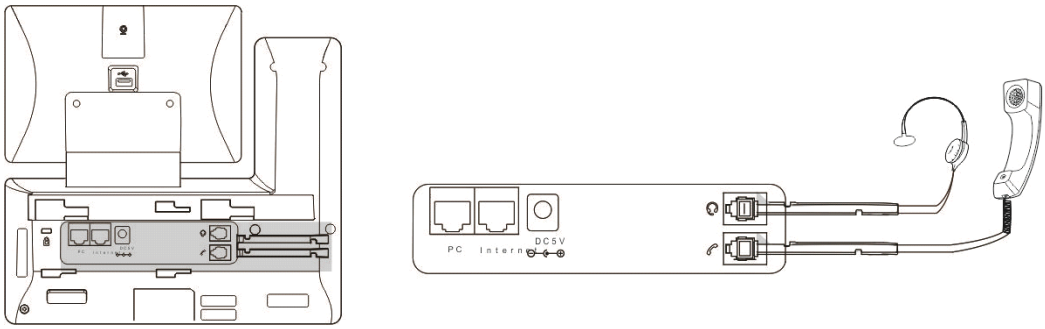
3. **Adjusting the angle of the touch screen**.



4. **Attaching the optional wall mounting bracket**

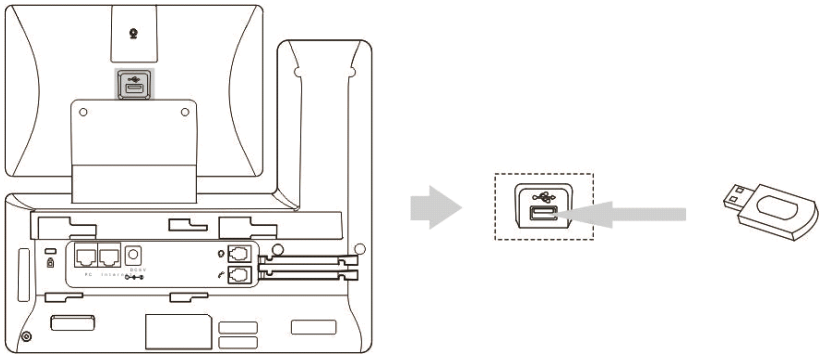


5. **Connect the handset and optional headset.**



A headset is not included in the packaging contents. Contact your dealer/reseller for more information.

6. **Connect the optional USB Flash drive.**



7. **Connect the network and power.**

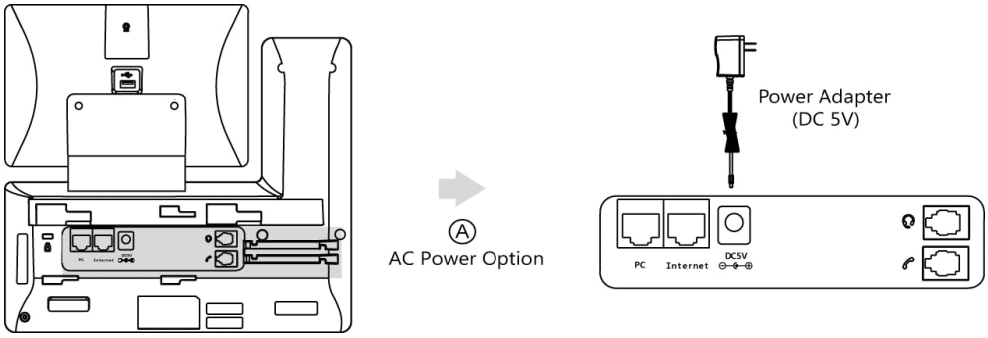
There are two options to connect the power and the network.

- AC power
- Power over Ethernet (PoE)

AC Power

To connect the AC power:

- Connect the DC plug on the power adapter to the DC5V port on the phone and connect the other end of the power adapter into an electrical power outlet.

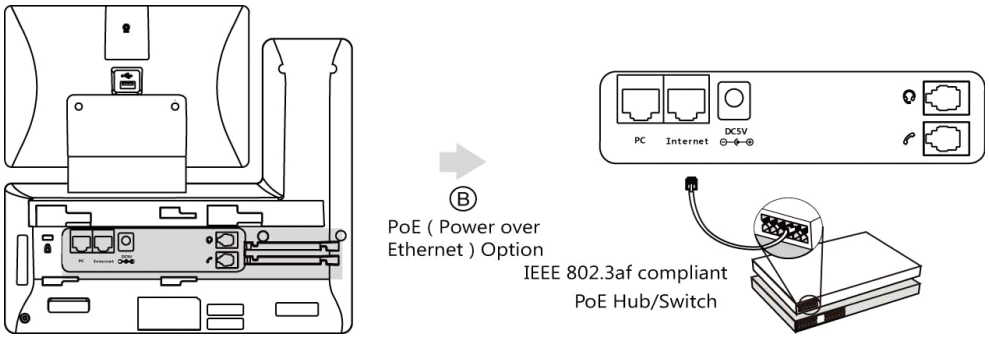


Power over Ethernet (PoE)

With the included or a regular Ethernet cable, the IP Phone can be powered from a PoE-compliant switch or hub.

To connect the PoE:

- Connect the Ethernet cable between the Internet port on the phone and an available port on the in-line power switch/hub.



If in-line power switch/hub is provided, you don't need to connect the phone to the power adapter. Make sure the switch/hub is PoE-compliant.

Do not unplug or remove power while the phone is updating firmware.

After the IP Phone is assembled and connected to the power supply, it automatically begins the initialization process.

During this process, the IP Phone displays the start up screen "Welcome Initializing…please wait".

Once the IP Phone is initialized, it displays two different phone modes:

- Standard SIP
- Extended SIP

- Select Extended SIP, to operate the IP Phone in the extended mode. As soon as you select this mode, the booting process initiates again and the start up screen displays "Welcome Initializing…please wait". After the IP Phone is initialized, it attempts to contact a DHCP Server in your network to obtain valid IPv4 network settings (example: IP address, Subnet Mask, Gateway address, DNS address).You need to configure the basic network parameters of the IP Phone manually, if these are not provided by the DHCP Server or if your network does not support DHCP.

  Refer to the *EXTENDED SPARSH VP710 User Guide*, for detailed instructions:

  - To change the Network Settings of the phone and configure the network parameters.
  - To use Wi-Fi for connectivity and configure its parameters.

- On getting the IP Address and Server Address, the phone initiates Auto Configuration (when DHCP is selected) to download the configuration files from SARVAM UCS.
- On successful download of all configuration files, the phone attempts to register with SARVAM UCS.
- On successful registration, the Home screen appears.

The phone will register successfully, only if the SIP Extension parameters in SARVAM UCS have been correctly configured as per your installation scenario.

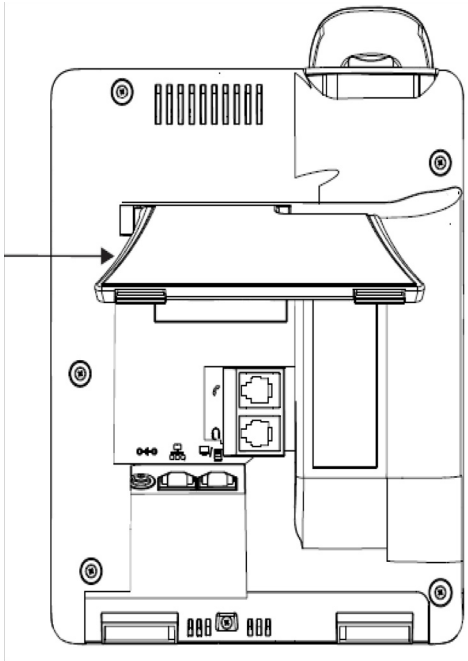
Connecting SPARSH VP210 as Extended SIP Extension

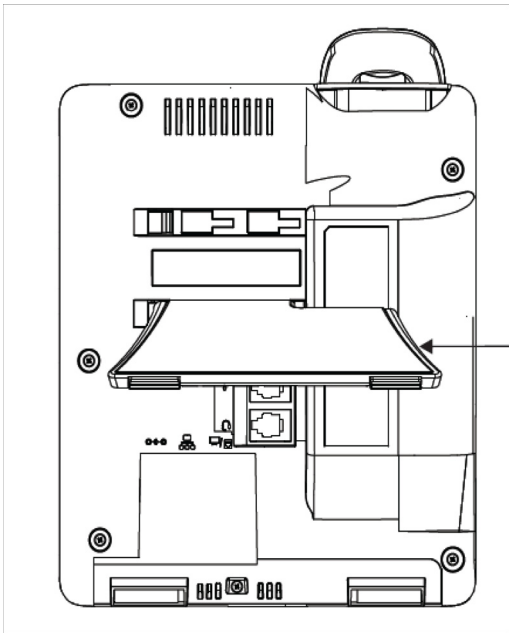
You are recommended to complete the following steps before connecting the Matrix SPARSH VP210 to SARVAM UCS:

- Decide the location where you want to place SPARSH VP210 within your LAN.
- By Default, in SPARSH VP210, the Connection Type selected is DHCP.
- If you want to use the **DHCP Server** for assigning IP Address to the Extended IP Phone, select **DHCP option 224** and **Data Type** as '**String**' and program the LAN or WAN IP Address /Domain Name of SARVAM UCS and SPARSH Port in the format **"IP_Address:Port"** in your LAN DHCP Server as per your installation scenario.
- Log in to *Jeeves*. For instructions, read the topic "Configuring SARVAM UCS".
- You must configure the necessary parameters in SARVAM UCS so that SPARSH VP210 can register as a SIP Extension. For instructions, see "Configuring Matrix SPARSH VP210".

Now, follow the steps described below to install SPARSH VP210.

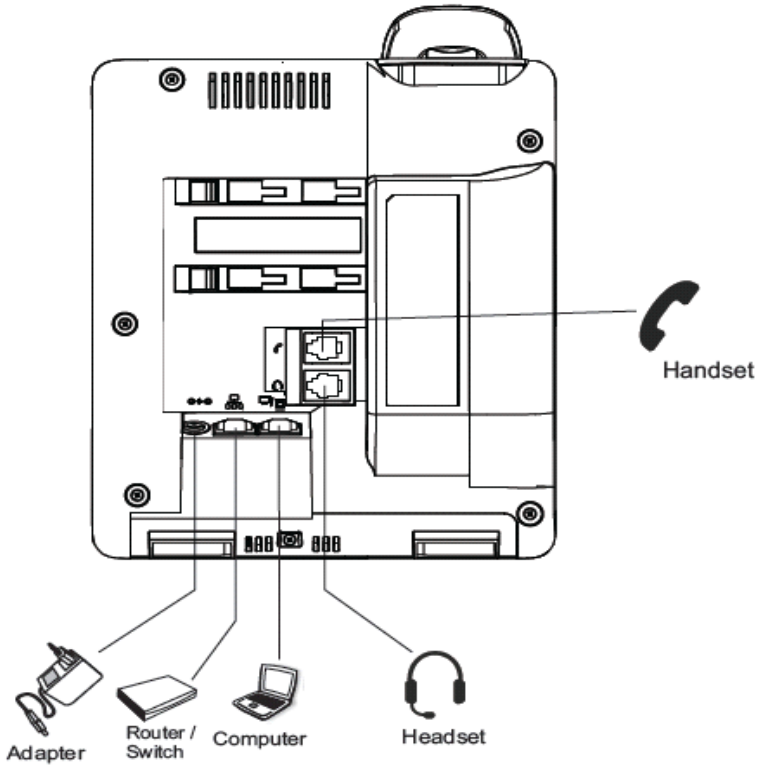
1. Unpack the SPARSH VP210 box and verify package contents.
2. When you mount the phone on a desk, you can attach the Foot Stand in two ways at **45° Angle** or at **55° Angle**.





- Decide which of these positions would work for you best and accordingly attach the Foot Stand.

  Refer to the diagram below for connectivity.



3. Connect the Handset to the Phone body.

- Plug the long straightened end of the Spring Cord into the handset jack at the bottom of the phone, marked with the handset symbol .
- Plug the other (short straight) end of the Spring Cord into the jack at the bottom of the handset.

4. If you want to use a Headset (not supplied) with your phone, You may plug in a headset with an RJ9 connector into the headset port at the bottom of the phone, marked with the symbol .
5. To connect the LAN, Port , plug one end of the Ethernet Cable into the LAN Port at the bottom of the phone marked with the symbol and the other end to the IP Network — A Router or LAN Switch.
6. To connect your phone to a computer on your desk, plug one end of the Ethernet Cable (not supplied with this phone) into the PC Port at the bottom of the phone, marked with the symbol and the other end into the LAN Port of your PC/LAN Switch.
7. It is a PoE enabled phone and can be powered over Ethernet by connecting it to a PoE enabled LAN Switch (IEEE 802.3af Compliant). In this case you need not connect the Power Adapter.

   If you do not want to use PoE, plug in the connector of the Adapter into the power jack (DC Jack) at the bottom of the phone, marked with the symbol . Plug in the Power Adapter into a power outlet.

If both the power options, that is, PoE as well as Power Adapter are available to the phone, then the phone will derive power from the PoE enabled LAN Switch.

The IP phone should be used with Matrix original power adapter (5V/0.6A) only. The use of any third-party power adapter may cause damage to the phone.

8. Switch ON power supply.

   When you power the phone, the boot process will be initiated in the following sequence.

   - The LCD display will light up and booting message appears.
   - While loading the application then the loading message appears on the phone display.
   - The phone makes DHCP Discovery and fetches its IP Address and Server Address from the DHCP Server.

If you want to change the Server Settings, press Settings.

Refer to the SPARSH VP210 (Extended) User Guide, for detailed instructions, to change the Network Settings of the phone and configure the network parameters.

- On getting the IP Address and Server Address, the phone initiates Auto Configuration to download the configuration files from SARVAM UCS.
- On successful download of all configuration files, the phone attempts to register with SARVAM UCS.
- On successful registration, the Home screen appears.

The phone will register successfully, only if the SIP Extension parameters in SARVAM UCS have been correctly configured as per your installation scenario.

Refer to the ***SPARSH VP210 (Extended) User Guide*** to know more.

Starting Up ETERNITY MENX

Power ON

- If you have completed all the installation tasks, switch on power supply. Keep the MCB Switch ON of your PS48V card installed in the system and power the FCBC.
- Observe the Reset Cycle.

Reset Cycle

- Reset Cycle (Power-ON Self Test) takes about 2 minutes to finish.
- All the LEDs of the system, the cards and the keys of the DKP/SIP devices attached to the System are turned on.

Interpreting LEDs

The functioning of the LEDs of the system and the various cards and their meaning are summarized at the end of the installation instructions for each Card Type.

Refer to the LED Patterns described for each Card Type to verify if the system is operating properly and locate faults, where they occur.

When the reset cycle is successful, the default Extension "Access Codes" loaded by the system and the date and time of the "Real Time Clock (RTC)" of the system will appear on the LCD display of the DKPs/ IP Phones you have connected with the system.

CHAPTER 7

Installing ETERNITY GENX

- *The Matrix ETERNITY GENX is to be installed by persons trained and experienced in telecom wiring.*
- *The person installing the ETERNITY GENX must be familiar with trunks, physical wiring of the MDF on both the exchange (System) side and the line side (CO).*
- *When installing any equipment, make sure that you take all the necessary precautions for handling electronic and electrical appliances. Follow proper procedures for static electricity, while handling the system and its cards to prevent damage to the system and harm to yourself.*
- *Use a grounding mat and wear an anti-static strap/belt. Read the do's and don'ts listed in "Protecting the System and Yourself".*
- *If you have complied with the requirements and instructions described in "Before You Start", you may now begin the installation of your ETERNITY GENX.*

Hardware Version V1R2 and earlier

The Matrix ETERNITY GENX is shipped factory fitted with the Power Supply Card, the CPU Card in their respective fixed slots (refer the section "Know Your SARVAM UCS"). VoIP and VMS are in-skin to the CPU Card. Hence, separate VMS and VoIP Cards are not required.

ETERNITY GENX has a total of 12 Universal slots. The Slave Cards - BRI, T1E1PRI, GSM/CDMA, DKP, CO, SLT, E&M, Radio, Data, Magneto, E1FO - are shipped separately as per the order placed by individual customers. These Cards are installed in any of the Universal slots.

If you upgrade the system firmware to V1R3 and later, the Expansion Slots license will be applicable for the universal slots. No universal slots will be functional by default. You must purchase the SARVAM EXP4 SME license to activate the universal slots as required.

For details, see "Expansion Slots" under "License Management".

Hardware Version V1R3 and later

The Matrix ETERNITY GENX is shipped factory fitted with the Power Supply Card, the CPU Card in their respective fixed slots (refer the section "Know Your SARVAM UCS"). VoIP and VMS are in-skin to the CPU Card. Hence, separate VMS and VoIP Cards are not required.

ETERNITY GENX has a total of 12 Universal slots. The Slave Cards - BRI, T1E1PRI, GSM/CDMA, DKP, CO, SLT, E&M, Radio, Data, Magneto, E1FO - are shipped separately as per the order placed by individual customers. These Cards are installed in any of the Universal slots.

If you have upgraded the system firmware to V1R3 and later in the old ETERNITY GENX system, the Expansion Slots license will be applicable for the universal slots. No universal slots will be functional by default. You must purchase the SARVAM EXP4 SME license to activate the universal slots as required.

If you have purchased the new ETERNITY GENX system with the firmware V1R3, the Expansion Slots license will be applicable for the universal slots. The first four universal slots after the power supply card will be functional by default. If you require more functional universal slots, you must purchase the SARVAM EXP4 SME license.

Each SARVAM EXP4 SME license will provide the activation for next four universal slots in sequence.

For details, see “Expansion Slots” under “License Management”.

Illustrated below is the position of the fixed and universal slots in ETERNITY GENX.

ETERNITY GENX

The extreme left slot is reserved for the Power Supply Card and the extreme right slot is reserved for the CPU Card.

Follow the installation instructions for Cards described here also when you expand the system (add more Cards) or remove or swap Cards for maintenance and repair.

1. Unpack the box. Check the package contents (see “Packing List”). Contact your Dealer/Distributor if any of the items is missing, faulty or damaged. Do not discard the packaging material.

Mounting the System

2. Decide where to mount the ETERNITY GENX - on a table or wall - taking into consideration the mechanical dimensions and the weight. If mounting the system on a wall, you may refer the mechanical dimensions and the Mounting Template for drilling holes at appropriate places on the wall. Make sure the system orientation is horizontal.

3. When installing the system in a rack, allow adequate space between the system and other units for air circulation. Make sure the system orientation is horizontal.

4. Mount the system at the selected site. Make sure that the system is placed such that you have full access to the front and back panels. The holes in the panels are provided for ventilation; Make sure that these are not blocked, to prevent overheating.

Connecting Input Power Supply

5. Ensure that a proper electrical earth and telecom earth are in place.

6. Check the voltage at the power point from where the supply is to be given to the system. It should be as per the specifications. Earth the system properly. (Refer “How to Make the Telecom Earth”).

Inserting Cards

7. Make sure that the ETERNITY GENX power is off and the power cord is unplugged.

8. Select a free slot from the universal slots.

9. Unscrew and remove the filler bracket that covers the card-slot opening of the slot you intend to use.

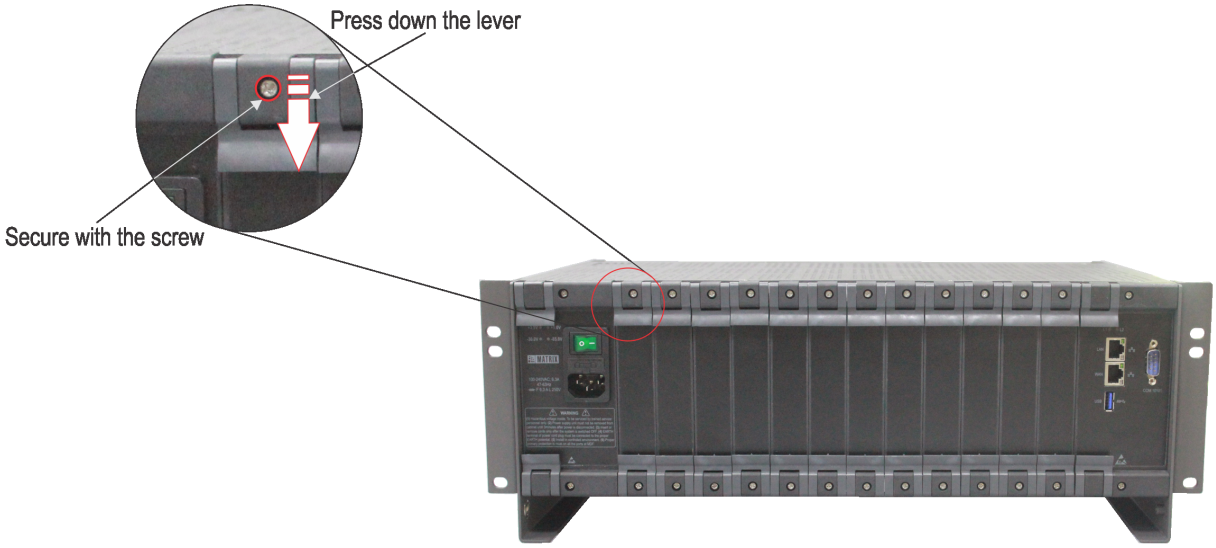
10. Hold the card with the connectors facing you. Do not grab the card from both ends.

11. Slide the card into the slot, along the guide rails provided for each slot at the top and bottom planes.

12. Ensure that the cards are inserted deep enough for all the connector pins on the cards make complete contact with those of the motherboard on the backplane.

Do not force the card into the slot. Doing so can damage the card or the slot connector.

13. When the card is firmly seated in the connector, push down the levers on the card mounting bracket and secure the card with the screw provided.



14. Tighten the screws on either side of the bracket.

15. Following the above steps, install each card into the universal slots.

    Detailed installing instructions are provided for each card - DKP, SLT, CO, ISDN BRI, ISDN T1E1PRI, GSM/CDMA, E&M, E1FO, Magneto, Radio - later in this section. Refer to them when installing each card type.

16. To remove a card:

    - Switch off power supply, unplug the power cord.
    - Disconnect any cables connected to the card.
    - Remove the screws from the card-mounting bracket.
    - Lift the levers on the mounting bracket to release the card.
    - The card will emerge out of the slot.
    - Grasp the card by its mounting bracket, and ease it out of its slot.

- *If you are removing the card permanently or for a certain period of time, install a filler bracket over the empty card opening in the chassis.*

- *Installing filler brackets over empty card-slot openings is necessary to protect the system from dust, dirt, insects and damage.*

17. Using the cables supplied with each card, and terminate the cables in the Main Distribution Frame (SLT, DKP, CO, and E&M lines), the NT1 device (ISDN BRI lines), ISDN Modem (ISDN PRI Lines), as applicable.

    Lead the cables neatly and tangle-free into the MDF.

18. After you have completed inserting and connecting the cards, power ON the system and observe the Reset cycle and the LED pattern of each card, where applicable.

The Power Supply Card

Three types of Power Supply Cards are supported by the system: ETERNITY GE CARD PSUNI (250 W) and ETERNITY GE CARD PS48VDC (250W) and ETERNITY GE CARD PSBB.

- **ETERNITY GE CARD PSUNI (250W)** with 100-240VAC, 47-63Hz Mains as Input AC Voltage Power Supply and output 250W.

  This card is designed on the SMPS scheme. As this card does not have any provision for battery backup, it is recommended that a UPS be connected to keep the system powered during outages.

  This card has four Green LEDs, a Mains Switch, and a Socket assembly for connecting the mains cord.

- **ETERNITY GE CARD PS48VDC (250W)** with 48VDC as Input DC Power Supply Voltage and output as 250W. A Float cum Boost Charger (FCBC) is required to feed 48VDC power to the card. The FCBC works on input AC mains.

  The card has four Green LEDs (for four different voltages), one Red LED (for input reverse indication, an MCB Switch, a power ON/OFF Switch, and a 3-way termination block for connecting the power cord.

  Both, the PSUNI card and the PS48V Card provide DC output voltages as: +3.5V, +5.0V, -30V and -85V. These are indicated by LEDs.

- **ETERNITY GE CARD PS Battery Backup (PSBB)** with 100-240VAC, 47-63Hz Mains as Input AC Voltage Power Supply.

  This card is designed on the SMPS scheme and has a provision for Battery Backup. The card is supplied with a Battery cable to connect a 24VDC external battery. When the system is functioning using the AC power supply the battery will also be charged by the card. This card has no LEDs.

Installing the Power Supply Card

The Power Supply Card is located in a fixed slot. No other card can be inserted in this slot.

The Power Supply Card is delivered factory fitted, when you buy the system. However, if you want to remove the card for the purpose of maintenance or replace it with a new one, please follow the instructions below:

1. Unpack the Power Supply Card and verify the package contents.

   If already installed, switch OFF power supply, unplug the power cord. Remove the screws securing the card. Lift the levers on the mounting bracket to release the card. As the card emerges from the slot, ease it out of the slot.

AC Power Supply Card must be removed from the platform three minutes after the power supply is switched off.

Make sure you do not place the Power Supply Card on any conductive surface.

2. Insert the Power Supply card into the guide rails of the first slot on the extreme left, designated for the Power Supply Card. Make sure that the card is inserted deep enough to make perfect contact with the connectors on the motherboard at the backplane.

3. Now, press down the levers on the card mounting bracket to secure the card in its slot.

4. Secure the card in the slot by screwing the bracket on both ends.

5. If installing the PSUNI card, connect the three-pin power cord into the socket of the PSUNI card and plug in the cord into the mains supply.

   You may connect the PSUNI Card to a UPS to keep the system live during power outages.

   Select a UPS considering the typical power consumption of the system. See “Power Consumption Table”.

6. If installing the **Eternity GE CARD PS48VDC (250W)**, connect the Float cum Boost Charger (FCBC) or AC-DC Power Supply with 48VDC Output. Terminate the power cord from the FCBC output into the 3-way termination block on the PS48V card.

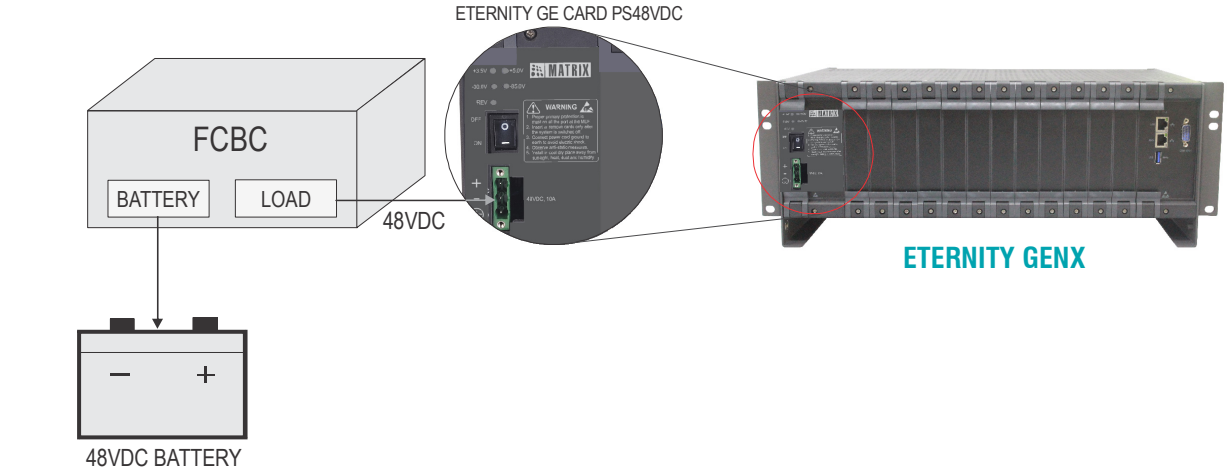
   Polarity is critical. Ensure that the wires are connected with the correct polarity. Follow the standard color codes used by FCBC manufacturers:

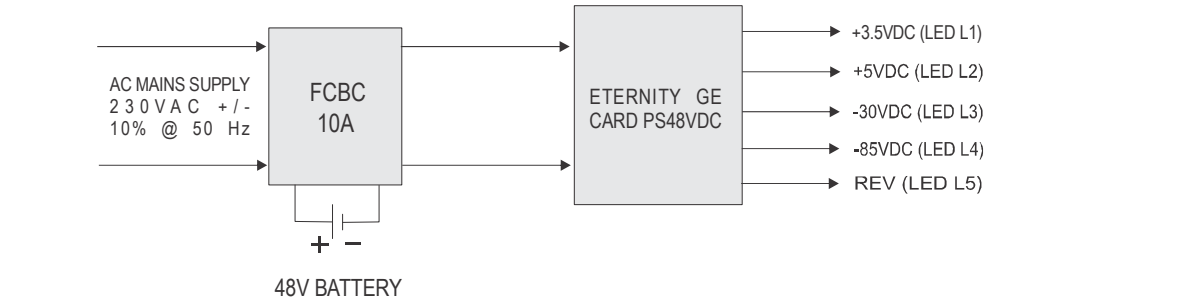
Color	Signal	Symbol
Red	+48VDC	+
Black	GND	-
Green	Earth	

   It is recommended that you measure the voltage before connecting the power cable to the power supply card. Ensure that the earth is connected.

Connecting an FCBC to the System

As PS48VDC Card works with FCBC, the input voltage working range has hysterisis.The power supply turns ON at Voltage greater than 45VDC and it remains ON till 40VDC during battery discharge condition. Moreover, the input voltage for the card must be from 40VDC to 60VDC.





7. Connect the Battery Backup[78]. Battery backup time depends upon the total load. The total load is the sum of system's load and load of active extensions.

8. If installing **Eternity GE CARD PSBB**, **c**onnect the Battery Backup. Terminate the power cord from the Battery Backup output into the 3-way termination block on the PS Battery Backup card. The Battery cable supplied with the card has a connector on one end and the other end is free. Insert the cable connector into the connector on the card as shown below. At the other free end connect the Battery.

   Polarity is critical. Ensure that the wires are connected with the correct polarity. Follow the standard color codes used by Battery manufacturers:

Color	Signal
Red	+24VDC
Black	GND

78. *When the batteries are drained, the FCBC goes into the charge mode and begins to charge the batteries at higher current. When the batteries reach a preset voltage level (typically set to 56.0 volts), the FCBC goes to float mode. In the float mode the FCBC keeps charging the battery but at lower current. The FCBC monitors the voltage level of the batteries. As soon as the battery voltage goes below preset voltage (typically set to 50.4 volts), FCBC goes from float mode to charge mode. The change over from mains to battery and vice-versa is automatic. The advantage of using an FCBC is that batteries get charged faster, since the batteries are charged with higher current initially.*

Color	Signal
Green	Earth

It is recommended that you measure the voltage before connecting the power cable to the power supply card. Ensure that the earth is connected

The following table displays the charging time of the external battery

Type of Battery	**Charging Time (Hours)**
14Ah	13
27Ah	25

The following table displays the maximum number of off-hook ports supported

Condition	**Number of off-hook ports**
Continuous run condition	55 off-hook ports
Temporary overload	85 off-hook ports

9. Switch on power supply, after completing all other installation tasks.

Power Consumption Table

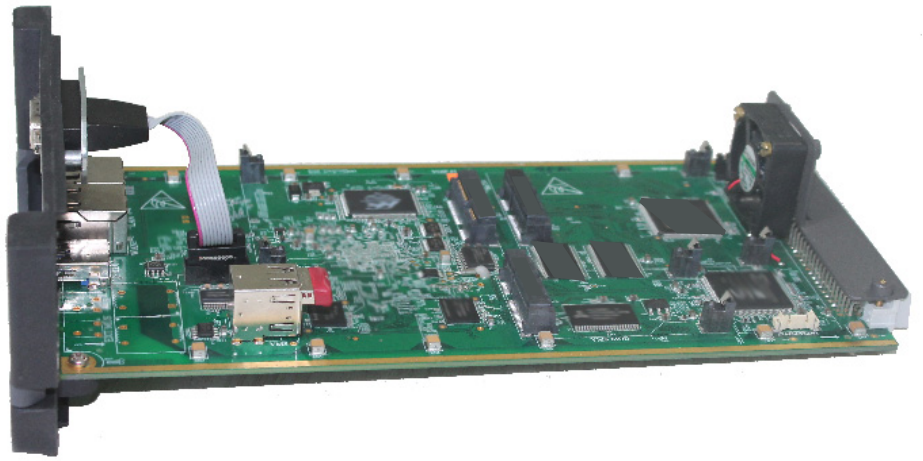
Platform	Number of Ports	Power Consumption[a]	Battery Backup[b]
		Off - Hook 30%	30%
ETERNITY GENX (System)	240	227 W (AC)	5.5hr(b)
		183 W (DC)	

a. Power Consumption is in Watts, if you want it in BTU per Hour then use the following relation:
1Watt=3.412 BTU/Hr
b. This Backup detail shown is considered with respect to 48V | 25Ah Battery.

The Power Supply unit can prove hazardous due to high Voltage Power it carries. Make sure you get it serviced by a trained service person only.

The CPU Card

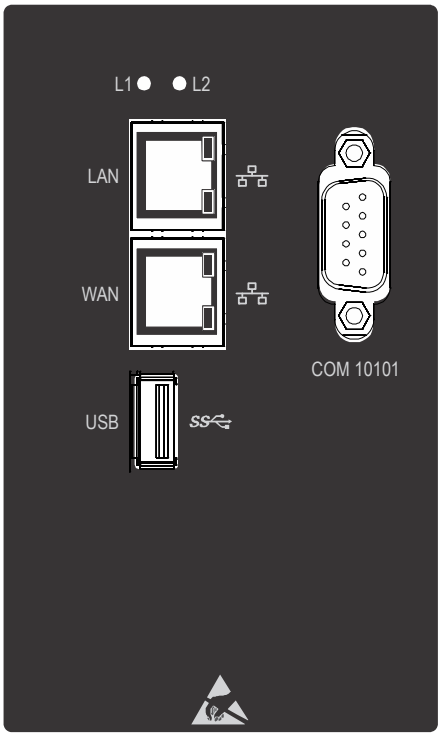
The ETERNITY GENX CPU Card hosts the SARVAM UCS Application. It supports two VOCODER modules and one VMS module. Both the modules — NX DBM VOCODER64 and NX DBM VMS64 are optional and can be purchased separately.



This card hosts Communication Manager, Feature Server, VoIP Server, VMS Server and other important servers and modules which controls all other slave cards (SLT, DKP, CO+SLT, DKP+SLT, E&M, BRI, T1E1,E1FO, GSM/ CDMA etc.). All the configuration and programming information is stored on this card.

The CPU Card occupies a fixed slot, the first slot on the extreme right, with a unique arrangement of connectors. So no other card can be inserted in the slot of the CPU Card.

The CPU Card has a WAN Port, LAN Port, USB Port and COM Port on the front panel. It also has an Internal USB Port with a factory fitted pendrive.



Ports and Connectors:

Port	Connector	Description
LAN	RJ45	Used for connecting the Ethernet cable into LAN Port to connect to a PC or a LAN Switch.
WAN	RJ45	Used for connecting the Ethernet cable into WAN Port to connect to a Broadband Router/Modem.
USB	USB to COM Converter (Optional)	The External USB can be used as COM Port by connecting the USB to COM Converter. The USB be COM Port can be used to: • set up and run software applications — PMS and CAS. • capture System Activity Log, System Fault log and Hotel Motel Activity logs. • generate SMDR reports.
COM	DB-9	Used to: • set up and run software applications — PMS and CAS. • capture System Activity Log, System Fault log and Hotel Motel Activity logs. • generate SMDR reports.

If you buy a spare CPU Card separately, the default pendrive will not be provided along with it.

LAN Interface

The LAN Port is provided to connect:

- the system to a PC or a LAN. This port is used for operating the web-based programming software Jeeves.
- the CPU Card to the Local Area Network to register SIP extensions through the LAN Port.
- set up and run software applications such as PMS and CAS on any PC on the LAN.
- generate Station Message Detail Record (SMDR) Reports on any PC on the LAN.
- capture "System Activity Log", "System Fault Log" and Hotel Motel Activity Log.

WAN Interface

The WAN Port is provided to connect:

- a LAN Switch/Hub/Router/Modem.
- the CPU Card to the public network over a Router/Modem. Any user on the public network can be registered as SIP Extension through the WAN Port.
- set up and run software applications such as PMS and CAS on any PC on the LAN.
- generate Station Message Detail Record (SMDR) Reports on any PC on the LAN.
- capture "System Activity Log", "System Fault Log" and Hotel Motel Activity Log.

VoIP Interface

The CPU Card supports two NX DBM VOCODER64 modules. You must purchase the module separately for VoIP functionality.

VOCODER Channels

The system supports two NX DBM VOCODER64 Modules. Each module supports 64 VOCODER Channels[79]. You must purchase the modules separately. The system provides 4 pre-activated VOCODER channels by default which can be used after installing NX DBM VOCODER64 module. If you require more channels, you can purchase the licenses accordingly. Matrix provides two licenses — SARVAM VOCODER CHNL4 and SARVAM VOCODER CHNL16.

If you require more than 64 VOCODER channels, you can install another NX DBM VOCODER64 Module.

A call made from a SIP Extension or SIP Trunk to another SIP Extension or SIP Trunk will consume two VOCODER channels, whereas a call made from a SLT or DKP extension to a SIP Extension or SIP Trunk will consume one VOCODER channel. Thus, the number of speech paths available to make simultaneous calls will depend not only on the number of VOCODER channels, but also on the number of channels consumed by such SIP-to-SIP and Analog/Digital extension to SIP Trunk/SIP Extension calls.

VMS Interface

The system supports a full-fledged, 'in-skin' Voice Mail System module to provide mailbox facility to all its extensions users. The Voice Mail System also forms the basis of other features like Conversation Recording and Call Taping.

Each Mailbox has the capacity of storing 15,000 voice messages. The maximum size of each Mailbox is 60,000 minutes. By default, the size of each Mailbox is set to 5 minutes. The maximum Message Length for each Mailbox is 9,999 seconds. By default, the Maximum Message Length for each Mailbox is set to 15 seconds.

79. The number of VOCODER channels that will be supported would be as per the license you purchase.

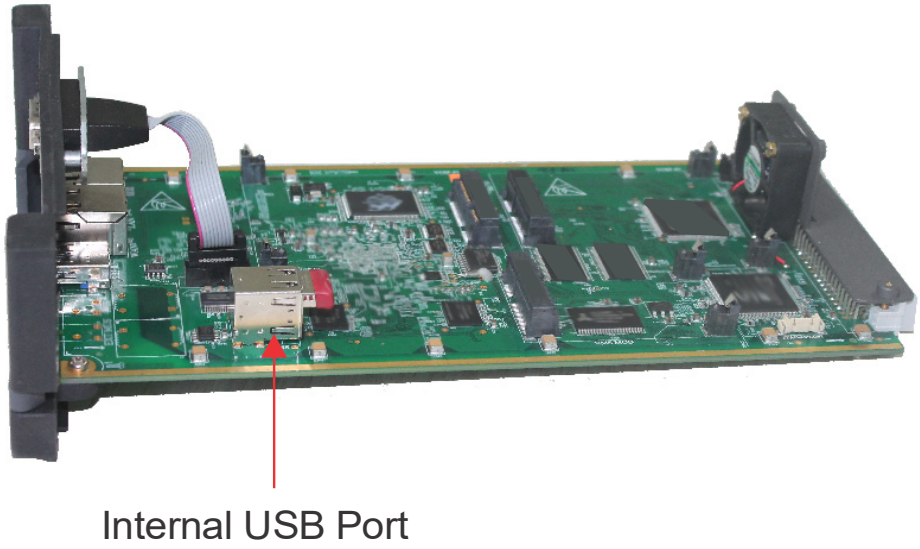
The NX DBM VMS64 Module is an optional module. It must be purchased separately. The factory fitted Pen Drive provided also contains the VMS configuration files, and voice messages for prompts and greetings along with the SARVAM UCS SME Application. The Pen Drive is also the storage device for mailbox messages.

If required, you may use a Pen Drive of upto 64GB by replacing the factory fitted pendrive with a new one.

The system supports a maximum of 64 channels out of which 4 channels are provided by default. If you require more channels, you can purchase the licenses accordingly. Matrix provides two licenses — SARVAM VMS CHNL4 and SARVAM VMS CHNL16.

Internal USB Port

The CPU Card has an Internal USB Port with a pendrive inserted into it.



The pendrive supports FAT32 file format. It contains the SARVAM UCS Application, VMS greetings, messages, mail boxes, Matrix Extended IP phone firmwares and SMS Server firmware.

Do not remove the pendrive.

When you select the SARVAM UCS SME Application, the system fetches the application from the pendrive.

External USB Port (Device Port) 3.0

The CPU Card has an External USB Port on the fascia. This can be used as a COM Port by connecting the USB to COM Converter.

The USB to COM Converter will not be provided by Matrix.

The following USB to COM Converters are supported:

- Prolific PL2303 by BAFO
- CH341 by Winchiphead

If you use any other USB to COM Converter, Matrix does not guarantee it's proper functioning.

The USB to COM Port has a DB-9 connector.

The port allows you to connect a PC to the system, so that you can install and operate the following features:

- set up and run software applications such as PMS and CAS on any PC on the LAN.
- generate Station Message Detail Record (SMDR) Reports on any PC on the LAN.
- capture "System Activity Log" and "System Fault Log", Hotel Motel Activity Log.

Communication Port (COM Port)

There is a asynchronous, serial, full-duplex RS-232C Communication (COM) Port, labeled as COM10101. The COM Port has a DB-9 connector.

The COM port allows you to connect a PC to the system, so that you can install and operate the following features:

- set up and run software applications such as PMS and CAS on any PC on the LAN.
- generate Station Message Detail Record (SMDR) Reports on any PC on the LAN.
- capture "System Activity Log" and "System Fault Log", Hotel Motel Activity Log.

LED

The CPU Card has two dual color (Green and Red) LEDs.

- LED 1 - L1 works as a Heart Bit of CPU Card. In Normal Condition, L1 will be turned ON Green for 1 sec and OFF for 1 sec.

- LED 2 - L2 indicates the Layer Application status. In Normal condition, L2 will be turned ON Orange and will blink very fast.
- Both L1 and L2 also indicate Application status.

  Case 1: L1 is steady GREEN/OFF and LED2 is OFF. This means the Application is hung and there is some problem in the Application code.

  Case 2: LED1 is steady GREEN/OFF and LED2 is GREEN/RED/ORANGE. This means the Layer is hung and there is some problem with the Layer code.

Jumpers

The position and function of the Jumpers on the CPU Card are:

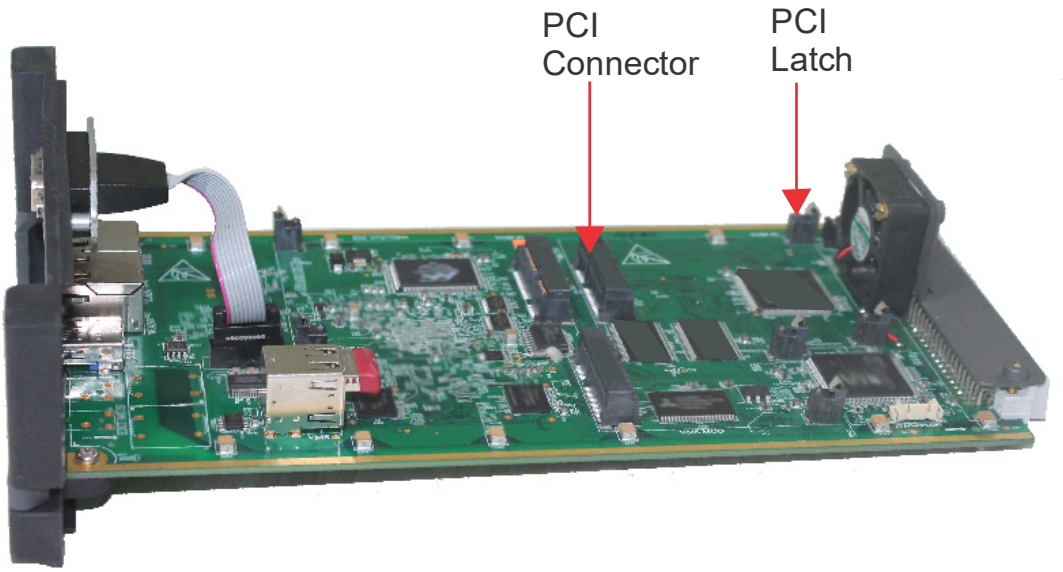
Jumper Number	Position	Function
J1	AB	Default SE Password.
	BC (default)	Normal.

Installing the VOCODER Module

To install,

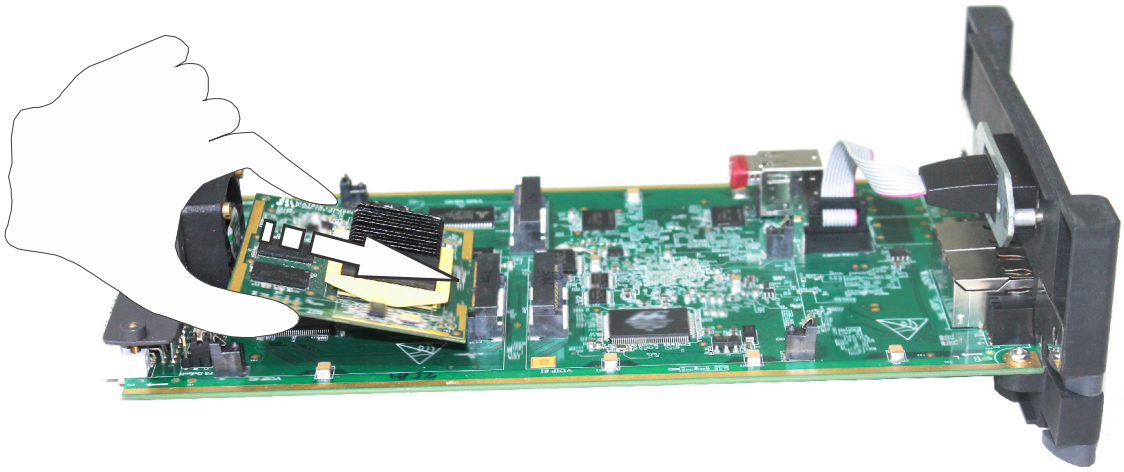
- Unpack the NX DBM VOCODER64 Module.

- If the CPU Card is already installed, switch off power supply, unplug the power cord. Remove the screws securing the card. Lift the levers on the mounting bracket to release the card. As the card emerges from the slot, ease it out of the slot.
- Place the card carefully on a table with some packing underneath it. Avoid any physical contact with the PCB part of the card as this could cause Electrostatic discharge (ESD) and may damage the hardware.
- The NX DBM VOCODER64 Module is to be mounted adjacent to the fan on the CPU board.

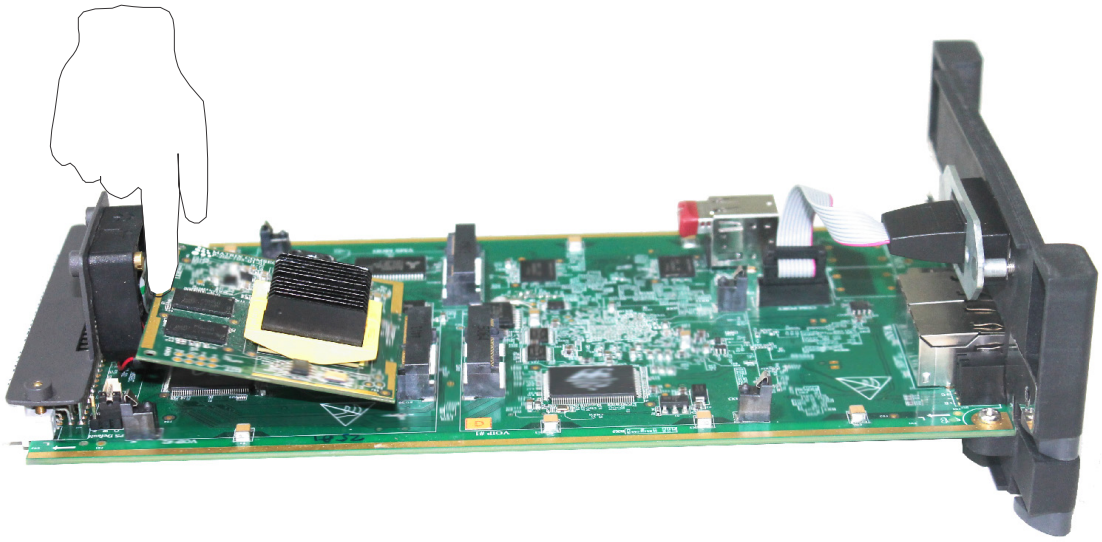


- Locate the PCI Connector and PCI Latch on the CPU board.

- Carefully hold the NX DBM VOCODER64 Module from the edges. Make sure you do not touch the PCB area.



- Insert the NX DBM VOCODER64 Module into the PCI Connector socket.

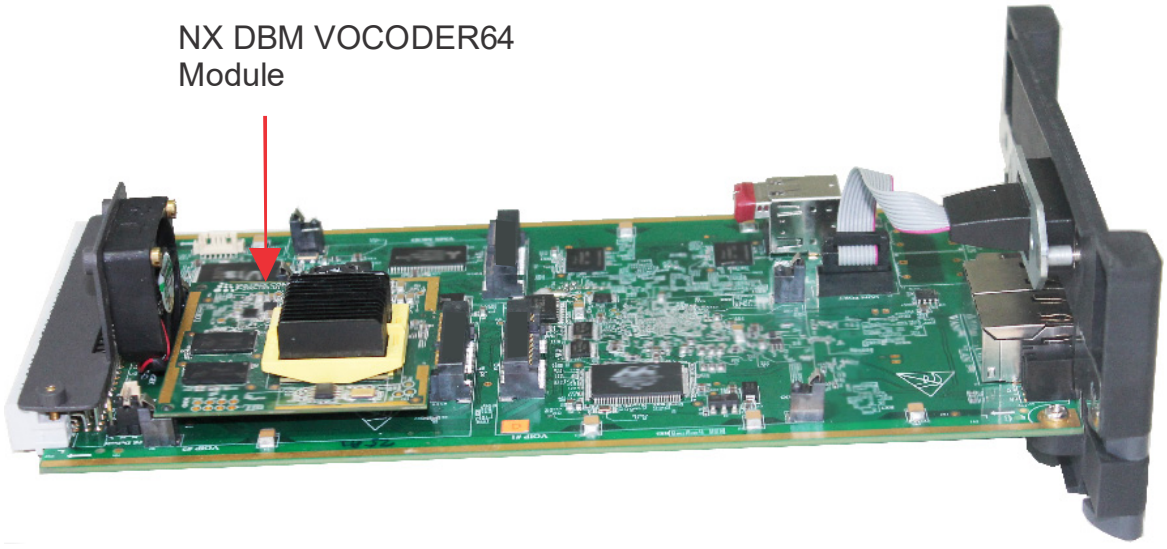


- Press the Module with a finger to fix the latches perfectly into the mounting holes. Make sure you do not touch the PCB area of the module except the yellow line provided for grounding at the front end of the module.

  Do not apply excessive pressure. Follow the same steps if you wish to install another module.

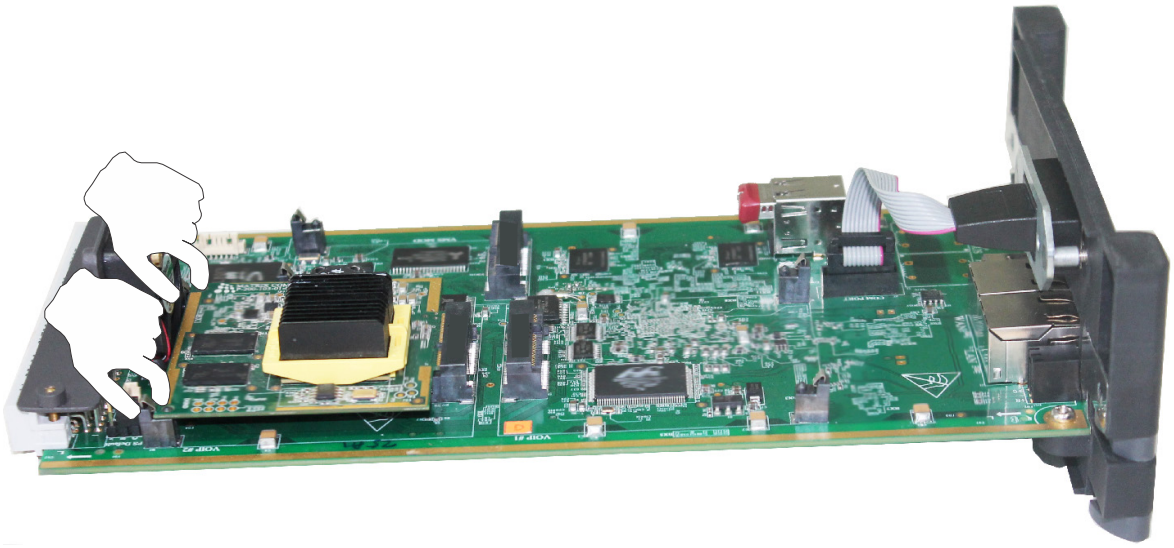
Removing the VOCODER Module

- Locate the VOCODER Module you want to remove from the CPU Card.

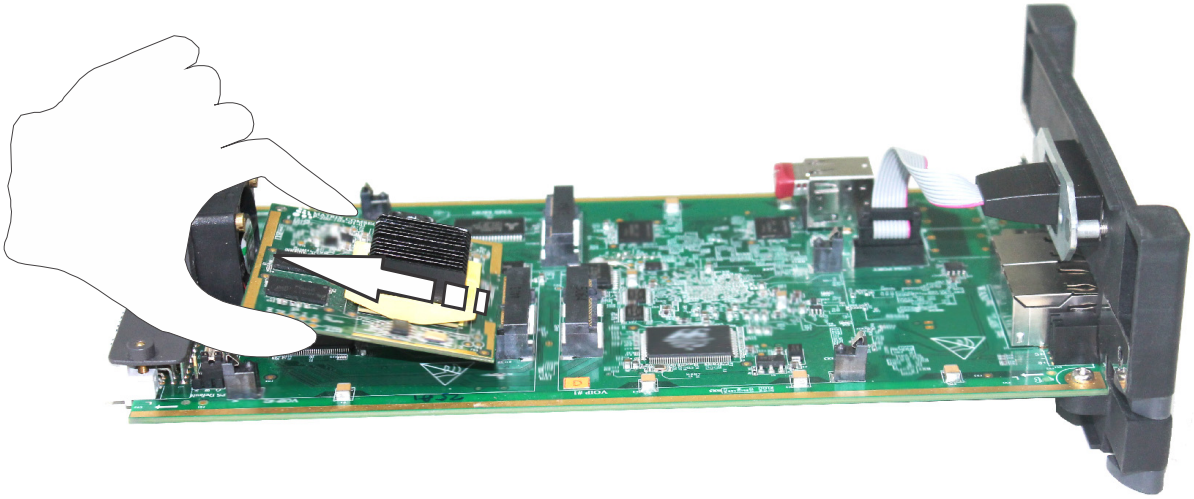


- Press both the latches together.

Make sure you support the base of the latches from behind with your forefinger.



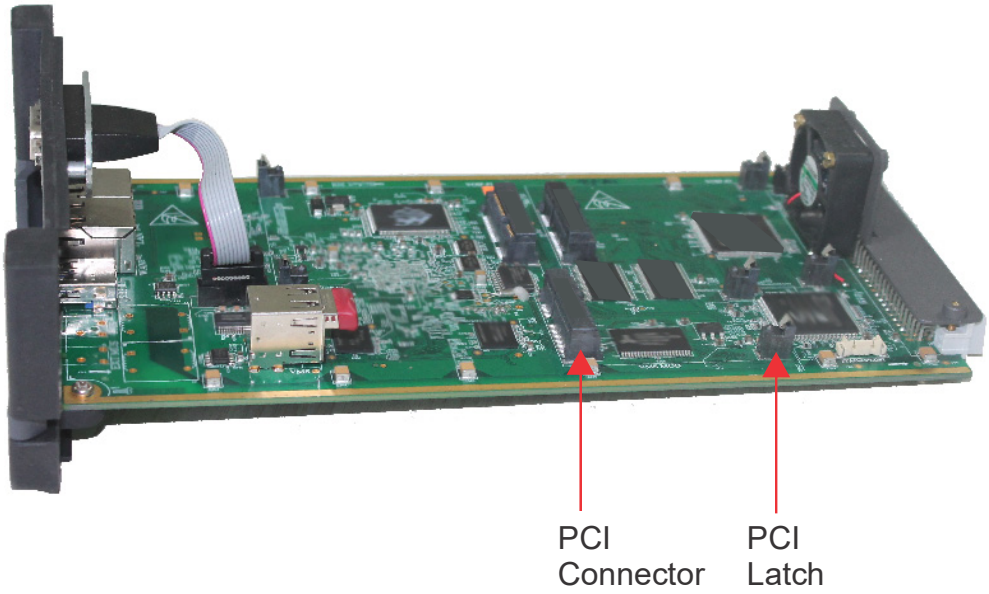
- Firmly hold the module and ease it out of the PCI connector carefully.



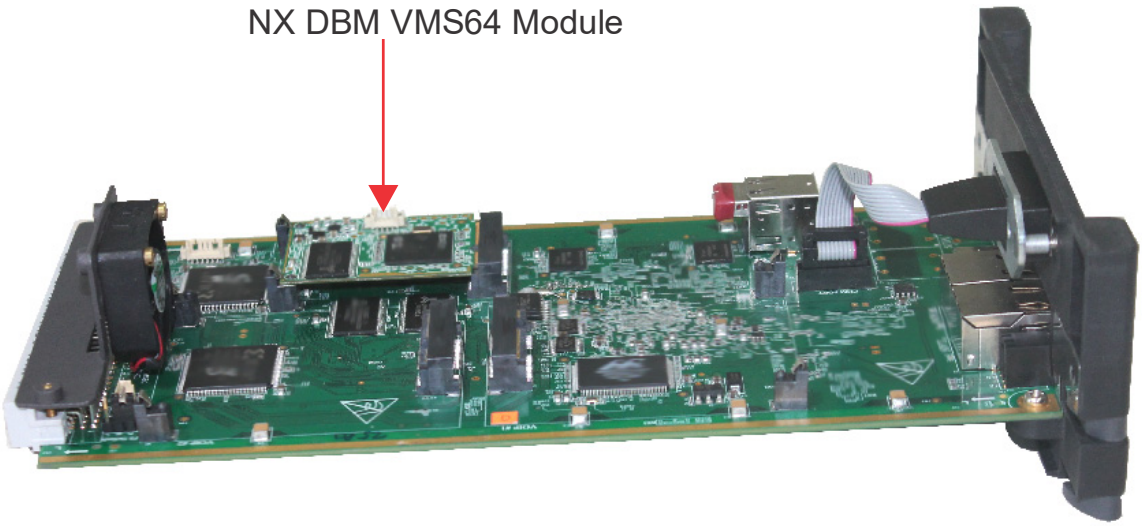
Installing the VMS Module

To install,

- Locate the PCI connector for NX DBM VMS64 Module on the CPU card.



- Follow the same steps as described in installing the NX DBM VOCODER64 Module. See “Installing the VOCODER Module”.



- The pendrive which is provided to you by default contains VMS data and VMS firmware. You will be able to use the VMS features once you activate the VMS License.

  If you want to store more voice mail messages or greetings then you will need more space to store the same. You can replace this default pendrive with a new one having more space.

  To do so, you need to format your new pendrive with FAT32 file format and then copy all the contents of the factory fitted pendrive into the new pendrive.

Make sure you switch-off the system to replace the pendrive. The system will not detect the new pendrive if you do not restart the system after replacement.

- If you have no other modules to install, insert the card back into the ETERNITY GENX.
- Connect a computer to the LAN/WAN Port of the system with the ethernet cable supplied for the port.
- Open a Web browser on the computer to access the embedded Web server, Jeeves.
- Activate the VMS License Voucher. See “License Management” for instructions.
- Configure VMS. For detailed instructions, see “Configuring Voice Mail System”.

For removing the VMS Module, follow the same steps as described for removing the VOCODER Module. See “Removing the VOCODER Module”.

The Single Line Telephone Card

The Single Line Telephone (SLT) Card provides the interface to connect as extension phones, any standard, two-wire, analog single line telephone instrument - rotary, pulse-tone, cordless, feature phones with or without Calling Line Identification.

The SLT Card is available in the following configurations. SLT interface also is available in combination with Two-wire trunk and digital key phone interfaces on a single card.

SLT Cards for ETERNITY GENX

Card Name	Configuration and Application
ETERNITY GE CARD SLT20	20-port card to connect 20 Single Line Telephones
ETERNITY GE CARD SLT16	16-port card to connect 16 Single Line Telephones
ETERNITY GE CARD SLT8	8-port card to connect 8 Single Line Telephones
ETERNITY GE CARD DKP4+SLT16	Combination card, with 4-ports to connect to 4 Digital Key Phones and 16 ports to connect 16 Single Line Telephones
ETERNITY GE CARD CO4+SLT16	Combination card with 4 ports to connect 4 Two-wire Trunk lines, and 16 ports to connect 16 Single Line Telephones
ETERNITY GE CARD CO4+DKP2+SLT12	Combination card, with 4 ports to connect 4 Two-wire Trunk lines, 2 ports to connect 2 Digital Key Phones and 12 ports to connect 12 Single Line Telephones. This Card supports Power Fail Transfer. To know more, see “Power Fail Transfer”.
ETERNITY GE CARD CO4+DKP2+SLT8	Combination card, with 4 ports to connect 4 Two-wire Trunk lines, 2 ports to connect 2 Digital Key Phones and 8 ports to connect 8 Single Line Telephones. This Card supports Power Fail Transfer. To know more, see “Power Fail Transfer”.
ETERNITY GE CARD CO2+DKP2+SLT16	Combination card, with 2 ports to connect 2 Two-wire Trunk lines, 2 ports to connect 2 Digital Key Phones, and 16 ports to connect 16 Single Line Telephones

The maximum number of SLT ports supported are 240. However, the maximum number of simultaneous off-hook SLT ports supported are 120.

Connectors

The SLT Cards have RJ45 connectors, with each connector having 4 SLT ports. A multi-pair, MDF cable is supplied for each connector.

LEDs

The Cards SLT8 and SLT16 have 2 LEDs, while SLT20 has no LED:

- The LEDs indicate the health of the card during the Reset Cycle.

- the status of any one of the extension ports during normal functioning of the system.

You can monitor any of the SLT Extension ports by assigning the LED to that port[80].

LED Pattern of SLT Card

LED 2 (L2)

PORT Status	LED Color	LED Cadence
Commands from Application to SLT Port.	GREEN	Toggle[a] at each command
Events to Application from SLT Port.	RED	Toggle[b] at each event

a. The current LED state will remain the same until the next command is received from the application on the SLT Port. For example, if the current LED state is Green/Red ON, on the next command received, the LED will be turned OFF. It will remain OFF until the next command is received. When the next command is received it will be turned Green/Red ON again. This process continues.

b. Same as above note.

Installing Single Line Telephones

To be able to connect Single Line Telephones as Extensions to your system, you must install at least one of the aforementioned SLT Cards in the System.

1. Decide the number of SLT extensions required and arrange for as many telephone instruments.

   You may use any standard telephone instrument like a rotary phone, a pulse-tone switchable push-button phone, a feature phone or a cordless phone.

Use SLTs equipped with a 'Flash' key, as several of the features and facilities of the system require you to press Flash. If any of the SLTs you have selected does not have a Flash key, tap the Hook switch of the phone to dial Flash.

2. Unpack the SLT Card and check the package contents. Ensure that the power supply is switched off, before you begin the installation of the card. Always wear an electrostatic discharge prevention wrist strap/ belt and use a grounding mat.

3. Unscrew and remove the filler card mount bracket of any of the free (empty) Universal Slots. Do not discard the filler bracket! You may require it at a later stage.

4. Insert the SLT Card into the guide rails of the free slot you selected for the card.

   Make sure that the connectors on the card make perfect contact with those on the motherboard on the backplane.

5. Press down the levers on the mounting bracket to secure the card in its slot. Now, secure the mounting bracket with the two screws provided.

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*80. To do this, enter SE mode, enter the programming mode from any extension connected to the system, by dialing 1#91-SE Password. Dial the command 7902-Slot-LED Number-Port, where Slot is the number of the universal slot in which the card is installed and Port is the port on the card to which the LED is to be assigned to monitor its functioning. LED Number is the number of the LED on the card, which will monitor the port. Exit programming mode by dialing '**00**'.*

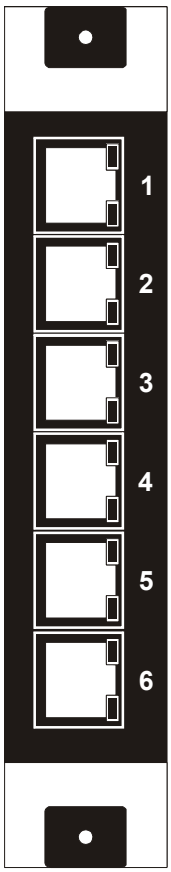
If you are installing more than one SLT Card, you can install the second card in any other free slot. It is not necessary to install the second/third card in the subsequent slots.

6. Use the cables supplied with the SLT Card to connect the SLT wires with the Main Distribution Frame.

   For each connector on the SLT Card, there is a separate 4-pair cable with an RJ45 jack on one end and free at the other end.

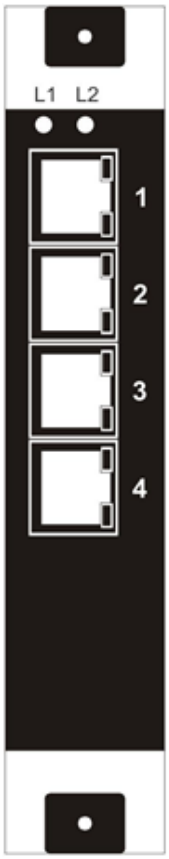
   Refer the illustrations below for pin out details of each connector.

ETERNITY GE CARD SLT20



Connector	Color	Connection	H/w Port Offset
RJ45-1	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	01 02 03 04
RJ45-2	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	05 06 07 08
RJ45-3	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	09 10 11 12
RJ45-4	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	13 14 15 16
RJ45-5	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT - -	17 18 - -
RJ45-6	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT - -	19 20 - -

ETERNITY GE CARD SLT16

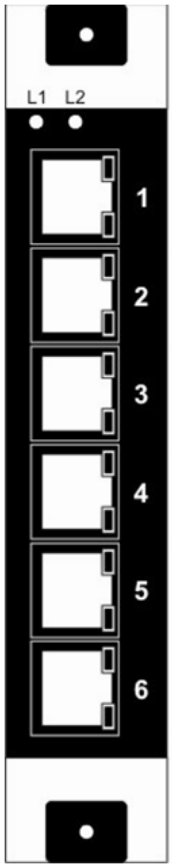


Connector	Color	Connection	H/w Port Offset
RJ45-1	Blue - (Blue & White)	SLT	01
	Orange - (Orange & White)	SLT	02
	Green - (Green & White)	SLT	03
	Brown - (Brown & White)	SLT	04
RJ45-2	Blue - (Blue & White)	SLT	05
	Orange - (Orange & White)	SLT	06
	Green - (Green & White)	SLT	07
	Brown - (Brown & White)	SLT	08
RJ45-3	Blue - (Blue & White)	SLT	09
	Orange - (Orange & White)	SLT	10
	Green - (Green & White)	SLT	11
	Brown - (Brown & White)	SLT	12
RJ45-4	Blue - (Blue & White)	SLT	13
	Orange - (Orange & White)	SLT	14
	Green - (Green & White)	SLT	15
	Brown - (Brown & White)	SLT	16

ETERNITY GE CARD SLT8

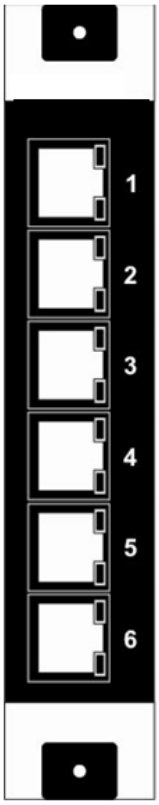
Connector	Color	Connection	H/w Port Offset
RJ45-1	Blue - (Blue & White)	SLT	01
	Orange - (Orange & White)	SLT	02
	Green - (Green & White)	SLT	03
	Brown - (Brown & White)	SLT	04
RJ45-2	Blue - (Blue & White)	SLT	05
	Orange - (Orange & White)	SLT	06
	Green - (Green & White)	SLT	07
	Brown - (Brown & White)	SLT	08

ETERNITY GE CARD DKP4+SLT16



Connector	Color	Connection	H/w Port Offset
RJ45-1	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	01 02 03 04
RJ45-2	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	05 06 07 08
RJ45-3	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	09 10 11 12
RJ45-4	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	13 14 15 16
RJ45-5	Blue - (Blue & White) Orange - (Orange & White)	DKP DKP	01 02
RJ45-6	Blue - (Blue & White) Orange - (Orange & White)	DKP DKP	03 04

ETERNITY GE CARD CO2+DKP2+SLT16

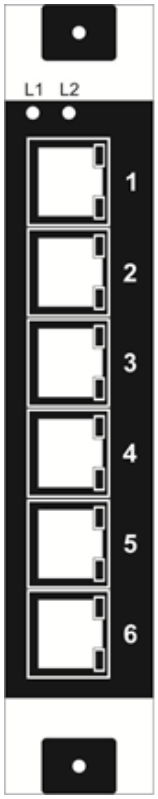


Connector	Color	Connection	H/w Port Offset
RJ45-1	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	01 02 03 04
RJ45- 2	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	05 06 07 08
RJ45-3	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	09 10 11 12
RJ45-4	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	13 14 15 16
RJ45-5	Blue - (Blue & White) Orange - (Orange & White)	DKP DKP	01 02
RJ45-6	Blue - (Blue & White) Orange - (Orange & White)	CO CO	01 02

ETERNITY GE CARD CO4+DKP2+SLT12

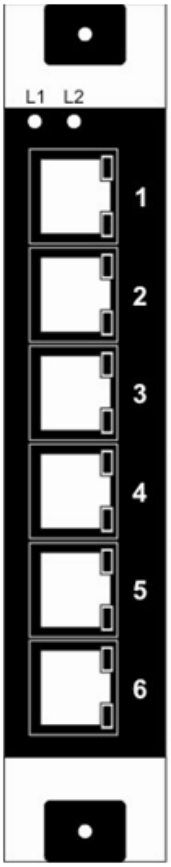
Connector	Color	Connection	H/w Port Offset
RJ45-1	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	01 02 03 04
RJ45-2	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	05 06 07 08
RJ45-3	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	09 10 11 12
RJ45-4	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	CO CO CO CO	01 02 03 04
RJ45-5	Blue - (Blue & White) Orange - (Orange & White)	DKP DKP	01 02
RJ45-6	Unused		

ETERNITY GE CARD CO4+DKP2+SLT8



Connector	Color	Connection	H/w Port Offset
RJ45-1	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	01 02 03 04
RJ45-2	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	05 06 07 08
RJ45-3	Unused		
RJ45-4	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	CO CO CO CO	01 02 03 04
RJ45-5	Blue - (Blue & White) Orange - (Orange & White)	DKP DKP	01 02
RJ45-6	Unused		

ETERNITY GE CARD CO4+SLT16



Connector	Color	Connection	H/w Port Offset
RJ45-1	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	01 02 03 04
RJ45-2	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	05 06 07 08
RJ45-3	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	09 10 11 12
RJ45-4	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	13 14 15 16
RJ45-5	Blue - (Blue & White) Orange - (Orange & White)	TWT TWT	01 02
RJ45-6	Blue - (Blue & White) Orange - (Orange & White)	TWT TWT	03 04

7. Plug in the RJ45 end of the MDF cables supplied with the card into the respective connectors. Refer to the pinout details of the connectors of each SLT Card type illustrated above.

8. Terminate the open end of the cables into the punch down blocks of the Krone modules designated for 'Station Lines' in the “The Main Distribution Frame (MDF)”.

   Each wire-pair from the SLT Port must be terminated to the bottom of the Krone Connector, while the wire-pair of the extension line to be connected to this port must be terminated on the top of the Krone connector. Refer the topic “The Main Distribution Frame (MDF)” for illustration.

9. Repeat the same steps to install another SLT Card.

Connecting SLT instruments

10. Connect the SLT instruments you have arranged for. Plug in the SLTs into the wall socket/outlets.

- *For the purpose of testing, you may connect one or two Single Line Telephone instruments by plugging in the phone cables into the RJ45 connectors on the card.*
- *When you plug the RJ11 connector of SLT into an RJ45 connector on the SLT Card, the SLT will be connected on the first port on the connector.*

The Intercom Line Card[81]

For the Building Intercom application, the system supports the Intercom Line Card (ILC).

You can connect any standard, two-wire, analog single line telephone instrument - rotary, pulse-tone, cordless, feature phones with or without Calling Line Identification to the Intercom Line Card.

ILC Cards for ETERNITY GENX

Card Name	Configuration and Application
ETERNITY GE CARD ILC20	20-port card to connect 20 Single Line Telephones

The maximum number of intercom ports supported are 240.

Connectors

The ILC Cards have RJ45 connectors, with each connector having 4 ports. A multi-pair, MDF cable is supplied for each connector.

LEDs

The Card ILC 20 has no LED:

- The LEDs indicate the health of the card during the Reset Cycle.
- the status of any one of the extension ports during normal functioning of the system.

Installing Single Line Telephones

To be able to connect intercom telephones to the system, you must install at least one of the aforementioned intercom line cards in the system.

1. Decide the number of intercom extensions required and arrange for as many telephone instruments.

2. Ensure that the extension wiring is completed according to your requirements. The extension cables from the wall jack are terminated in the Main Distribution Frame and the telephones are connected to the wall jacks.

3. Always wear an electrostatic discharge prevention wrist strap/belt and use a grounding mat to prevent damage to the components of the card.

4. Unpack the ILC Card and check the package contents. Switch off power supply before you install the card.

5. Unscrew and remove the filler card mount bracket of any of the free (empty) Universal Slots. Keep the filler bracket for future use.

6. Insert the ILC Card into the guide rails of the free slot you selected for the card.

81. This card has been phased-out. if required, only support will be provided for the same.

Make sure that the connectors on the card make perfect contact with those on the motherboard on the backplane.

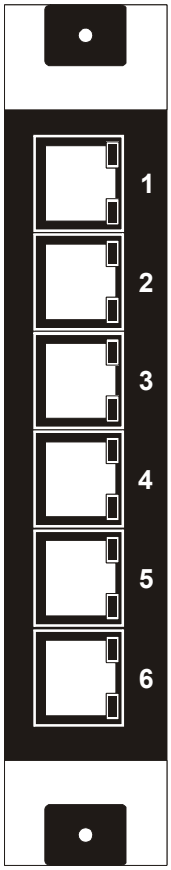
7. Press down the levers on the mounting bracket to secure the card in its slot. Now, secure the mounting bracket with the two screws provided.

8. Repeat these steps to install another card.

9. Now, use the cables supplied with the ILC Card to connect the card to the Main Distribution Frame to which the intercom phones are connected.

   For each connector on the card, there is a separate 4-pair cable with an RJ45 jack on one end and free at the other end. Refer the illustrations of the pinout of the intercom cards to connect the wires.

ETERNITY GE CARD ILC20



Connector	Color	Connection	H/w Port Offset
RJ45-1	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	01 02 03 04
RJ45-2	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	05 06 07 08
RJ45-3	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	09 10 11 12
RJ45-4	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	13 14 15 16
RJ45-5	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT - -	17 18 - -
RJ45-6	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT - -	19 20 - -

10. If you have completed all other installation tasks, power ON the system, observe the Reset Cycle.

The Digital Key Phone Card

The Digital Key Phone (DKP) Card provides the interface to connect the proprietary digital key phones of the EON series, the proprietary PC-based phone EONSOFT, the Direct Station Selection (DSS) Consoles, with the system.

The DKP Card is available in the following configurations:

DKP Cards for ETERNITY GENX

Card Name	Configuration and Application
ETERNITY GE CARD DKP16	16-port card to connect 16 DKP/DSS Consoles
ETERNITY GE CARD DKP8	8-port card to connect 8 DKP/DSS Consoles
ETERNITY GE CARD DKP4+SLT16	Combination card, with 4-ports to connect to 4 Digital Key Phones and 16 ports to connect 16 Single Line Telephones
ETERNITY GE CARD CO2+DKP2+SLT16	Combination card, with 2 ports to connect 2 Two-wire Trunk lines, 2 ports to connect 2 Digital Key Phones, and 16 ports to connect 16 Single Line Telephones
ETERNITY GE CARD CO4+DKP2+SLT12	Combination card, with 4 ports to connect 4 Two-wire Trunk lines, 2 ports to connect 2 Digital Key Phones and 12 ports to connect 12 Single Line Telephones. This Card supports Power Fail Transfer. To know more, see “Power Fail Transfer”.
ETERNITY GE CARD CO4+DKP2+SLT8	Combination card, with 4 ports to connect 4 Two-wire Trunk lines, 2 ports to connect 2 Digital Key Phones and 8 ports to connect 8 Single Line Telephones. This Card supports Power Fail Transfer. To know more, see “Power Fail Transfer”.

To connect the proprietary digital key phones with the system, you must have at least one of the above mentioned DKP Cards installed in the system.

The maximum number of DKP Ports supported by the system are 96.

Connectors

The DKP Cards have RJ45 connectors, with each connector having 4 DKP ports. A multi-pair MDF cable is supplied for each connector on the card.

LEDs

The DKP16 and DKP8 Cards have two dual color LEDs:

- LED1 indicates the health of the card during the Reset Cycle.
- LED2 monitors the status of any one of the extension ports during normal functioning of the system.

  LED2 can be assigned to any DKP port to monitor the status of that port[82].

Installing the Digital Key Phone Card

Decide the number of DKP extensions and DSS Consoles required and arrange for as many EON, EONSOFT and DSS Consoles.

Decide the locations of the DKP extensions and make sure that the necessary wiring for the DKP extensions, from the wall jack to the MDF, is done.

1. Unpack the DKP Card and check the package contents[83]. Before handling the card, make sure that power supply is switched off and you are wearing an antistatic-wrist strap/belt and have a grounding mat.

2. Unscrew and remove the filler card mount bracket of any of the free (empty) Universal Slots. Do not discard the filler bracket, keep for future use to cover empty slots.

3. Insert the DKP Card into the guide rails of the free slot you have selected for the card. All the pins on the connector of the card should make perfect contact with those on the connector of the slot on the backplane motherboard.

4. Press down the levers on the mounting bracket to secure the card in its slot. Now, fix the card in its slot with the two screws provided.

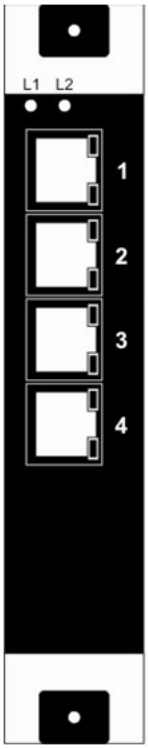
If you are installing more than one DKP Card, it is not necessary to install the next card in the subsequent slot.

5. Using the MDF Cables supplied with the DKP Card connect the DKP ports to the Main Distribution Frame.

   Refer the connector pin details for each DKP Card type given in the following.

82. *You can do this from the SE mode, by dialing the SE Command 7902-Slot-LED Number-Port, where Slot is the number of the universal slot in which the card is installed and Port is the port on the card to which the LED is to be assigned to monitor its functioning. LED Number is the number of the LED on the card, which will monitor the port.*
83. *See "ETERNITY GENX Cards" under 'Packing List' of Appendix topic.*

ETERNITY GE CARD DKP16

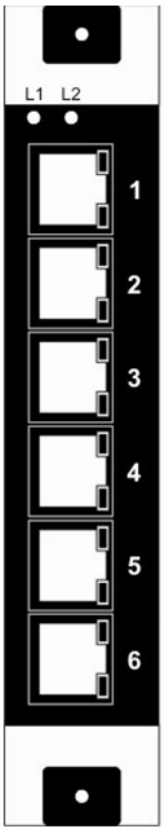


Connector	Color	Connection	H/w Port Offset
RJ45-1	Blue - (Blue & White)	DKP	01
	Orange - (Orange & White)	DKP	02
	Green - (Green & White)	DKP	03
	Brown - (Brown & White)	DKP	04
RJ45-2	Blue - (Blue & White)	DKP	05
	Orange - (Orange & White)	DKP	06
	Green - (Green & White)	DKP	07
	Brown - (Brown & White)	DKP	08
RJ45-3	Blue - (Blue & White)	DKP	09
	Orange - (Orange & White)	DKP	10
	Green - (Green & White)	DKP	11
	Brown - (Brown & White)	DKP	12
RJ45-4	Blue - (Blue & White)	DKP	13
	Orange - (Orange & White)	DKP	14
	Green - (Green & White)	DKP	15
	Brown - (Brown & White)	DKP	16

ETERNITY GE CARD DKP8

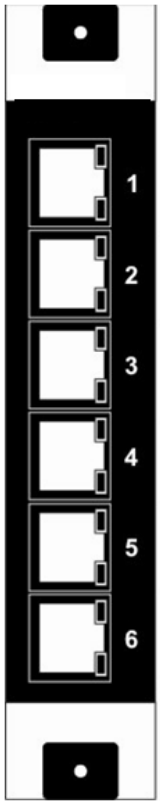
Connector	Color	Connection	H/w Port Offset
RJ45-1	Blue - (Blue & White)	DKP	01
	Orange - (Orange & White)	DKP	02
	Green - (Green & White)	DKP	03
	Brown - (Brown & White)	DKP	04
RJ45-2	Blue - (Blue & White)	DKP	05
	Orange - (Orange & White)	DKP	06
	Green - (Green & White)	DKP	07
	Brown - (Brown & White)	DKP	08

ETERNITY GE CARD DKP4+SLT16



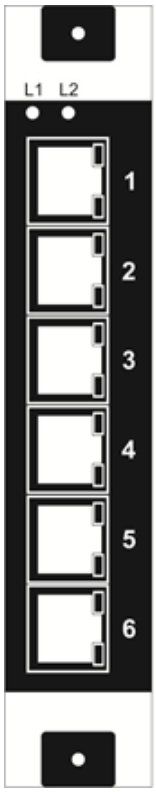
Connector	Color	Connection	H/w Port Offset
RJ45-1	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	01 02 03 04
RJ45-2	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	05 06 07 08
RJ45-3	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	09 10 11 12
RJ45-4	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	13 14 15 16
RJ45-5	Blue - (Blue & White) Orange - (Orange & White)	DKP DKP	01 02
RJ45-6	Blue - (Blue & White) Orange - (Orange & White)	DKP DKP	03 04

ETERNITY GE CARD CO2+DKP2+SLT16



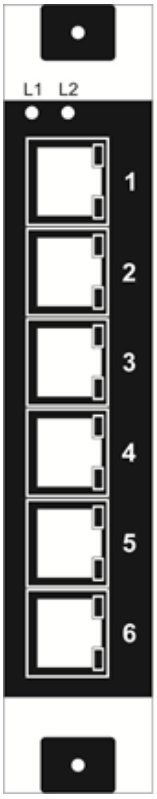
Connector	Color	Connection	H/w Port Offset
RJ45-1	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	01 02 03 04
RJ45- 2	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	05 06 07 08
RJ45-3	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	09 10 11 12
RJ45-4	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	13 14 15 16
RJ45-5	Blue - (Blue & White) Orange - (Orange & White)	DKP DKP	01 02
RJ45-6	Blue - (Blue & White) Orange - (Orange & White)	CO CO	01 02

ETERNITY GE CARD CO4+DKP2+SLT12



Connector	Color	Connection	H/w Port Offset
RJ45-1	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	01 02 03 04
RJ45-2	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	05 06 07 08
RJ45-3	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	09 10 11 12
RJ45-4	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	CO CO CO CO	01 02 03 04
RJ45-5	Blue - (Blue & White) Orange - (Orange & White)	DKP DKP	01 02
RJ45-6	Unused		

ETERNITY GE CARD CO4+DKP2+SLT8



Connector	Color	Connection	H/w Port Offset
RJ45-1	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	01 02 03 04
RJ45-2	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	05 06 07 08
RJ45-3	Unused		
RJ45-4	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	CO CO CO CO	01 02 03 04
RJ45-5	Blue - (Blue & White) Orange - (Orange & White)	DKP DKP	01 02
RJ45-6	Unused		

6. Plug in the RJ45 end of the MDF cables provided with the DKP Card into the respective connectors.

7. Terminate the free end of the cables into the punch down blocks of the Krone modules designated for 'Station Lines' in the Main Distribution Frame (MDF).

   Each wire-pair from the DKP Port must be terminated to the bottom of the Krone Connector, while the wire-pair of the extension line to be connected to this port must be terminated on the top of the Krone connector. Refer the topic “The Main Distribution Frame (MDF)” for illustration.

8. Connect the Digital Key Phones to the wall jacks at their respective locations. Detailed installations instructions for EON and EONSOFT are provided separately.

If you have completed all installation tasks, power on the system and observe the Reset Cycle and the LED Pattern of the DKP Card.

LED Pattern DKP Card

LED 2 (L2)

PORT Status	LED Color	LED Cadence
Commands from Application to DKP Port.	GREEN	Toggle[a]
Events to Application from DKP Port.	RED	

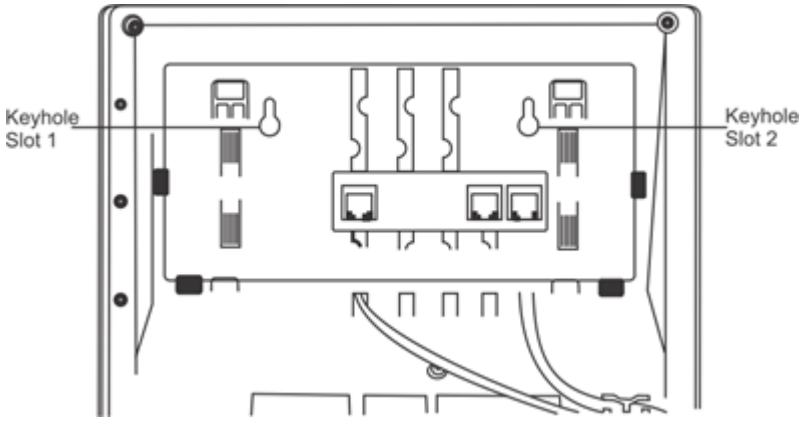
a. The current LED state will remain the same until the next command is received from the application on the DKP Port. For example, if the current LED state is Green/Red ON, on the next command received, the LED will be turned OFF. It will remain OFF until the next command is received. When the next command is received it will be turned Green/Red ON again. This process continues.

Installing EON48

- Unpack the box and verify the package contents.
- You can mount the phone on a wall or on desk.

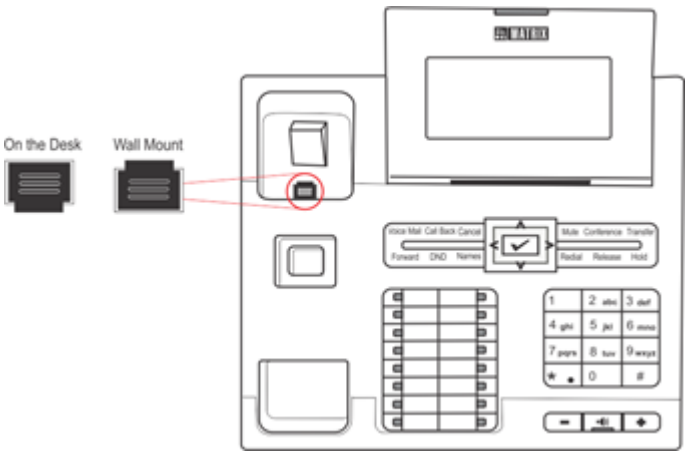
Mount the phone on a Wall

- Use the mounting template to drill holes of appropriate size and distance.
- Fix the screw grips in the holes you drilled.
- Fix two screws in the holes on the wall, ensuring that they are aligned with the Keyhole Slots 1 and 2. The screws should protrude from the wall to fit into the keyhole slots.



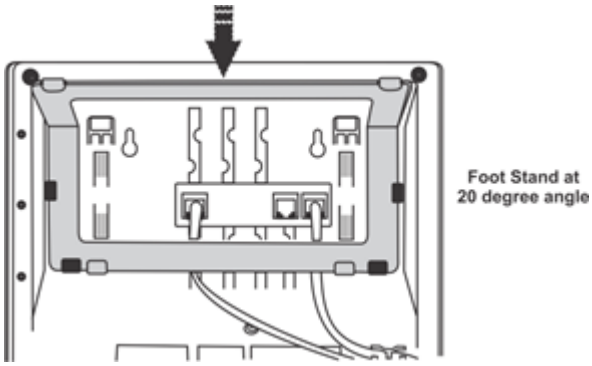
- Now, mount the phone with the screws fitting into the keyhole slots.

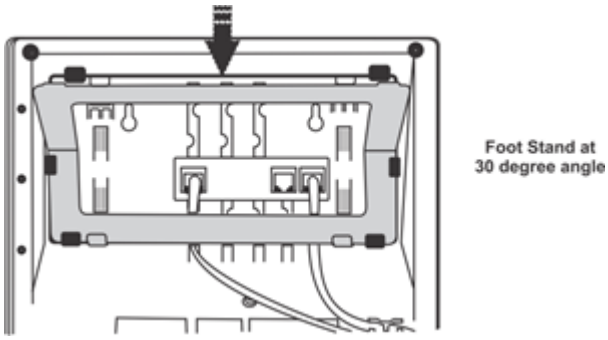
- Reverse the handset wall mount tab to make sure that the handset remains intact when you mount the phone. Push the handset wall mount tab upwards to remove it from the slot. Rotate it 180 degrees clockwise and push it downwards into the slot.

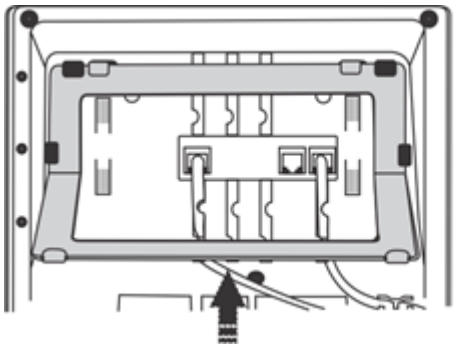


Mount the phone on the Desk

- You can attach the Foot Stand in the following ways—at an angle of **20° Angle** or at **30° Angle** or at **50° Angle**.

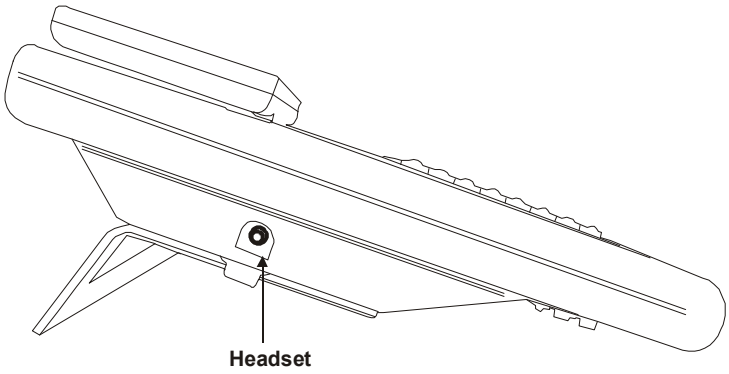






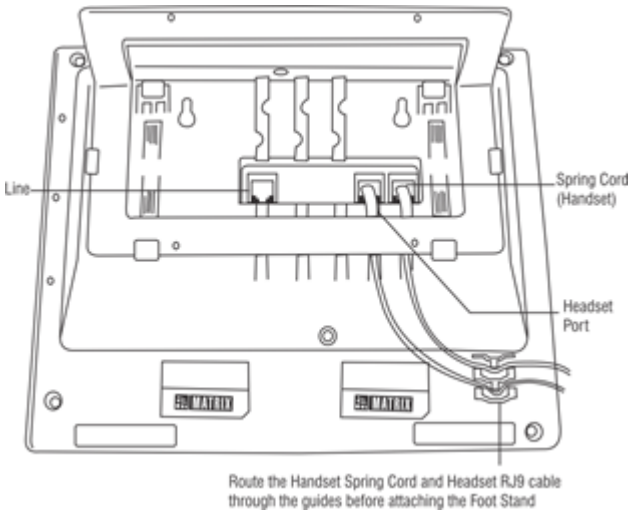
Foot Stand at
50 degree angle

- If you attach the Foot Stand at 50°, the phone will be placed in an almost upright position on your desk.
- Decide which of these positions would work for you best and accordingly attach the Foot Stand.
- Connect the handset of the EON48 to the phone body using the spring cord.
- To use a Headset (not supplied with the phone), plug any standard stereo headset with 2.5mm single connector into the headset jack with the symbol on the left side panel of the phone.

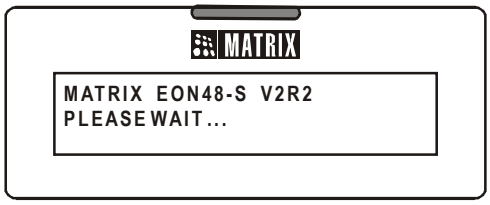


You may also plug in a stereo headset with an RJ9 connector into the headset port at the bottom of the phone, marked with the symbol .

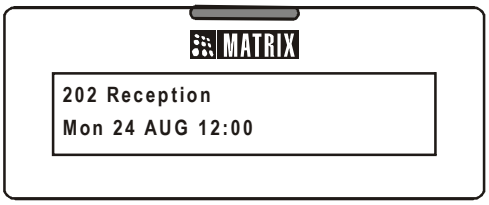
- Plug one end of the RJ11 cable supplied with the phone into the RJ11 connector of the phone labeled as '**LINE**' and the other end into the wall jack/DKP Port.



- When the system is powered ON, the EON will reset. The EON communicates with the system. The handshaking lasts for 5-6 seconds. The EON model, version and revision number, along with the message *'Please wait'…* appear on the LCD display.



- After successful handshaking and reset cycle, if the DKP Parameters have been configured, the LCD display of the EON will show the extension number and the extension name in one line and the day, date and time and the time zone in the other line.



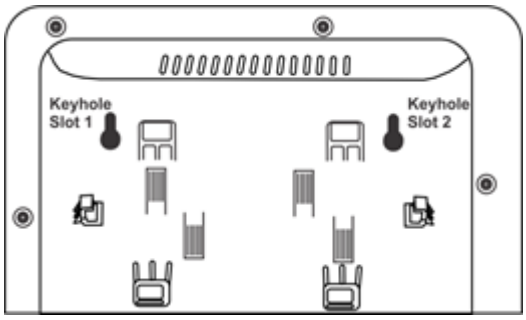
- You may adjust the LCD for brightness, contrast and backlight. Refer the topic, “Digital Key Phone-Operation” for instructions.

Installing EON310

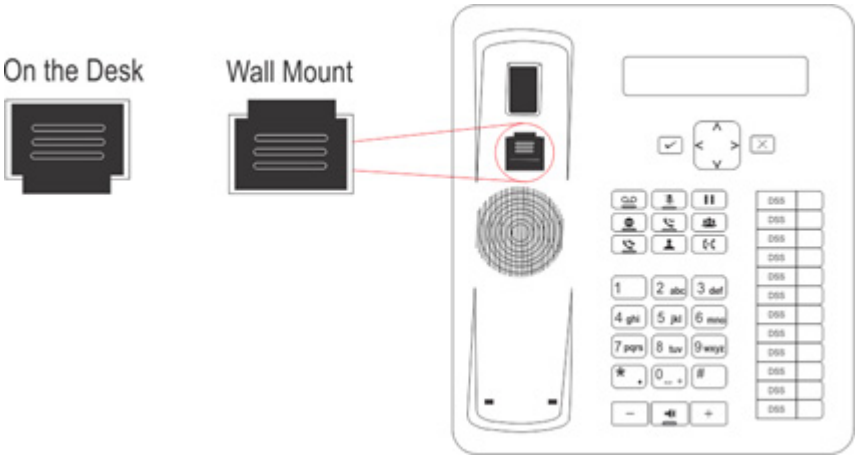
- Unpack the box and verify the package contents.
- You can mount the phone on a wall or on desk.

Mount the phone on a Wall

- Use the mounting template to drill holes of appropriate size and distance.
- Fix the screw grips in the holes you drilled.
- Fix two screws in the holes on the wall, ensuring that they are aligned with the Keyhole Slots 1 and 2. The screws should protrude from the wall to fit into the keyhole slots.

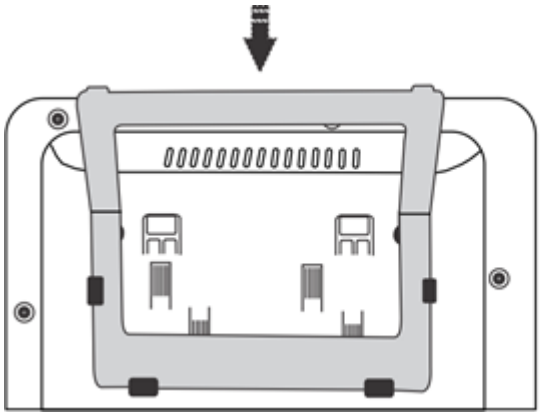


- Now, mount the phone with the screws fitting into the keyhole slots.
- Reverse the handset wall mount tab to make sure that the handset remains intact when you mount the phone. Push the handset wall mount tab upwards to remove it from the slot. Rotate it 180 degrees clockwise and push it downwards into the slot.

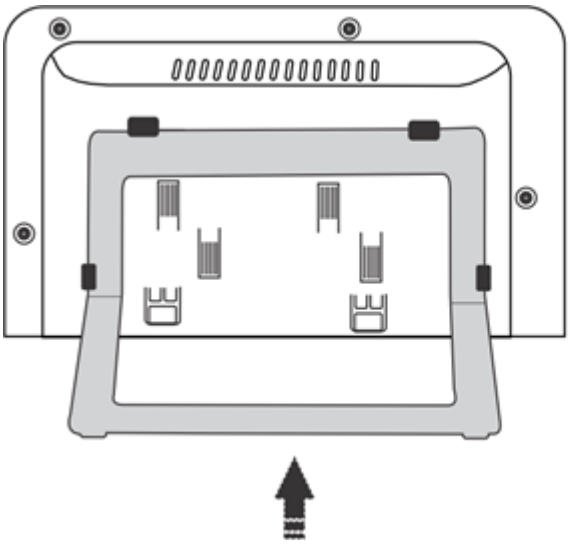


Mount the phone on the Desk

- You can attach the Foot Stand in the following ways—at an angle of **35° Angle** or at **50° Angle**.



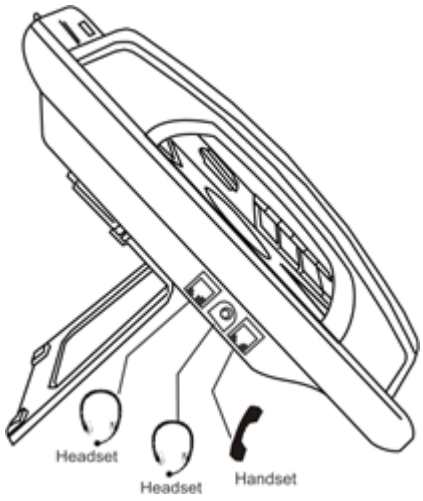
Stand attached at
35 degree angle



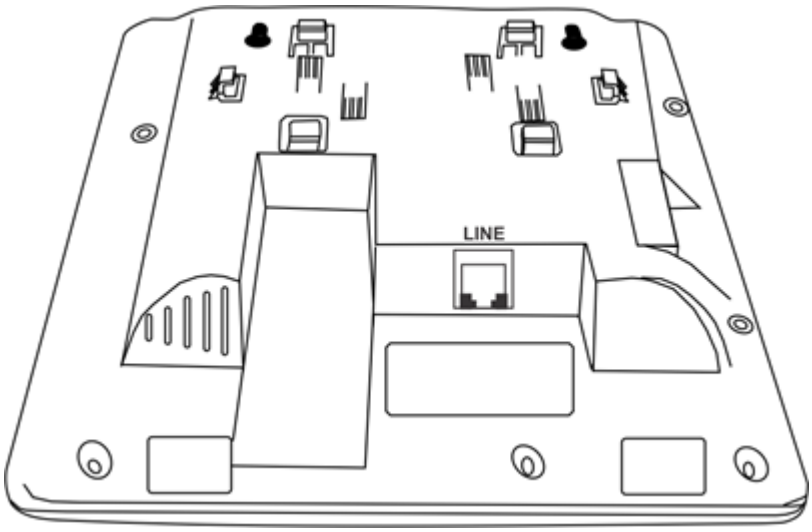
Stand attached at
50 degree angle

- Decide which of these positions would work for you best and accordingly attach the Foot Stand.
- Connect the handset of the EON310 to the phone body using the spring cord.
- To use a Headset (not supplied with the phone), plug any standard stereo headset with 3.5mm single connector into the headset jack with the symbol on the left side panel of the phone.

  You may also plug in a stereo headset with an RJ9 connector into the headset port marked with the symbol , on the left side panel of the phone.



- Plug one end of the RJ11 cable supplied with the phone into the RJ11 connector of the phone labeled as '**LINE**' and the other end into the wall jack/DKP Port.



- When the system is powered ON, the EON will get reset and the message "Welcome to Matrix. Booting"…appears on the LCD display.

- The EON communicates with the system. The handshaking lasts for 5-6 seconds. The EON model, version and revision number, along with the message “Please Wait”…appears on the LCD display.

- After successful handshaking and reset cycle, the extension number, day, date and time will appear on the LCD of the phone. If you have already assigned extension number and name, in the DKP Parameters, these will appear on the LCD.

- You may adjust the LCD for brightness, contrast and backlight. Refer the topic, “Digital Key Phone-Operation”.

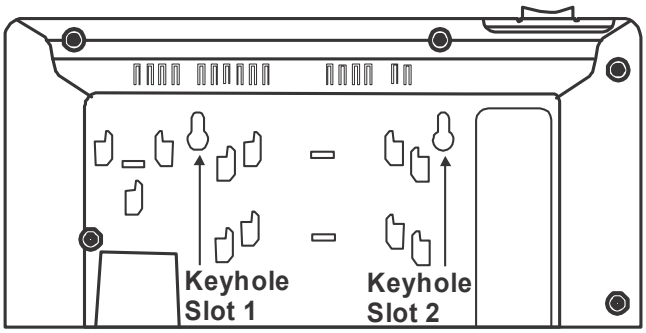
Installing EON510

- Unpack the box and verify the package contents.
- You can mount the phone on a wall or on desk.

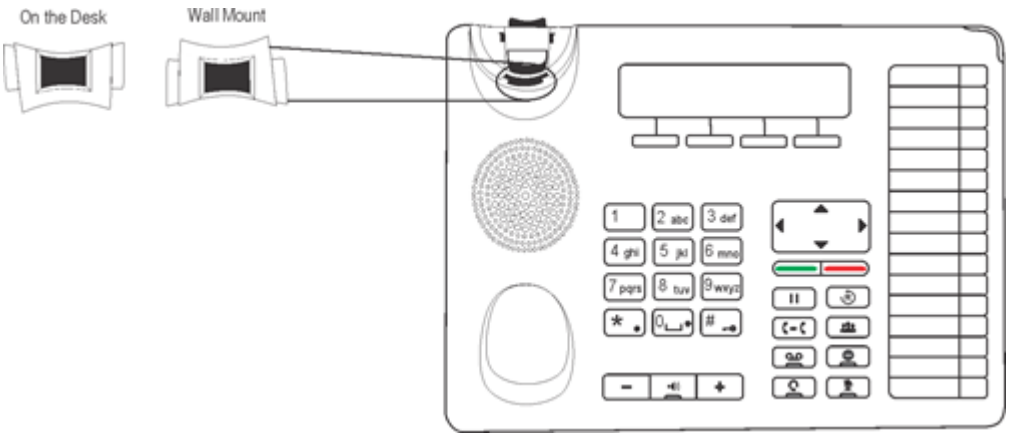
Mount the phone on a Wall

- Use the mounting template to drill holes of appropriate size and distance.
- Fix the screw grips in the holes you drilled.

- Fix two screws in the holes on the wall, ensuring that they are aligned with the Keyhole Slots 1 and 2. The screws should protrude from the wall to fit into the keyhole slots.



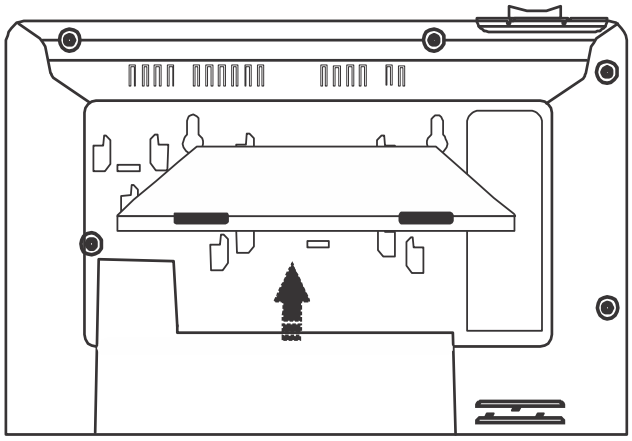
- Now, mount the phone with the screws fitting into the keyhole slots.
- Reverse the handset wall mount tab to make sure that the handset remains intact when you mount the phone. Push the handset wall mount tab upwards to remove it from the slot. Rotate it 180 degrees clockwise and push it downwards into the slot.



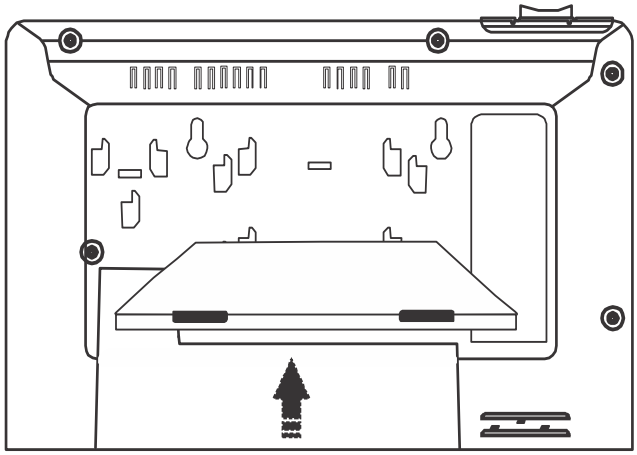
If you are unable to remove the wall mount tab, you may use a tool like a minus screw driver to remove it.

Mount the phone on the Desk

- You can attach the Foot Stand in the following ways—at an angle of **45° Angle** or at **55° Angle**.

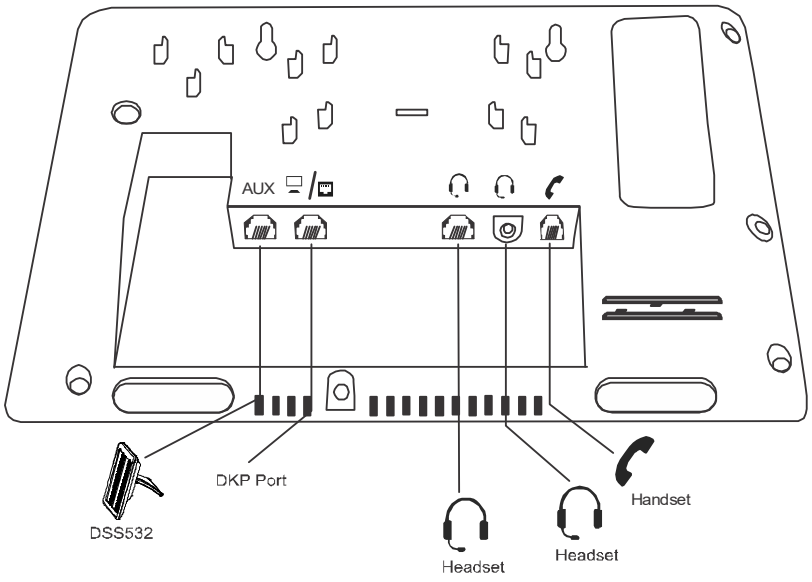


Stand attached at
45 degree angle



Stand attached at
55 degree angle

- Decide which of these positions would work for you best and accordingly attach the Foot Stand.



- Connect the handset of the EON510 to the phone body using the spring cord.
- To use a Headset (not supplied with the phone), plug any standard stereo headset with 3.5mm single connector into the headset jack with the symbol on the left side panel of the phone.

  You may also plug in a stereo headset with an RJ9 connector into the headset port marked with the symbol , on the left side panel of the phone.
- Plug one end of the RJ11 cable supplied with the phone into the RJ11 connector of the phone marked with the symbol and the other end into the wall jack/DKP Port.
- To connect DSS532 with the phone, plug one end of the RJ11 cable into the AUX Port of the phone and the other end into the IN Port of the DSS532. For installation, see “Installing DSS532 with EON510”.
- When the system is powered ON, the EON will get reset.and the message 'Phone is Booting; Please wait…’ appear on the LCD display.

- The EON communicates with the system. The handshaking lasts for 5-6 seconds. The message 'Loading Firmware Version-Revision; Please wait…’ appear on the LCD display.

- After successful handshaking and reset cycle, the extension number, day, date and time will appear on the LCD of the phone. If you have already assigned extension number and name, in the DKP Parameters, these will appear, as illustrated below.

You may adjust the LCD for brightness, contrast and backlight. Refer the topic, “Digital Key Phone-Operation”.

Installing DSS Consoles

Installing DSS64

Once you have installed EON48/310 with SARVAM UCS, installing the DSS Consoles can be done in a few simple steps, very much similar to those involved in the installation of EON.

1. Unpack the box and verify the package contents[84].

2. Place the DSS Console next to the DKP to which it is to be attached.

3. Decide which DKP Ports on the DKP Card are to be assigned to the DSS Consoles. You may select any free (unused) port on the card for DSS Consoles. It is not necessary for the DSS Console ports to be in a sequence with the DKP ports to which they are attached.

   For example: you have connected DKP1 to Port 1 on the first RJ45 connector of the DKP8 card. You want to attach two DSS Consoles to DKP1. The two DSS Consoles may be connected to any port on the second connector of the card, not necessarily to Port 2 and Port 3 on the first connector.

4. The wire-pairs from the DKP Ports designated for DSS Consoles should be terminated on the bottom of the Krone Connector (of 'Station Lines' on the MDF).

5. The wire-pairs of the DSS Consoles should be terminated into the top of the Krone Connector (of 'Station Lines' on the MDF). Refer the topic “The Main Distribution Frame (MDF)” for illustration.

   You can connect maximum two DSS64 with a single EON48/310.

6. The system automatically detects the DSS Console you connect and it will be will appear under **Unassigned DSS64** in “DSS Status”. You must first assign these DSS Consoles to the respective DKP Ports and thereafter you will be able to configure the DSS Keys.

7. To assign the DSS Consoles, see “DSS Status” and to configure the DSS Keys, see “Programming DSS Console Keys”.

Installing DSS532 with EON510

The instructions for installting DSS532 with EON510 or SPARSH VP510 are same. For detailed instructions, refer to “Installing DSS532 with SPARSH VP510”.

Installing EONSOFT

To install EONSOFT, you must have a computer with Windows as the operating system. The EONSOFT is compatible with the following Operating Systems of Windows:

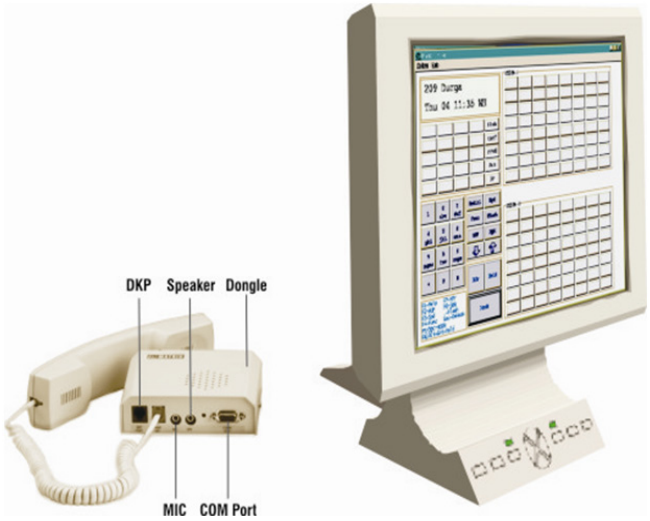
- Windows 98
- Windows XP
- Windows NT
- Windows 2003
- Windows Vista
- Windows 2007

---

84. See “Packing List” of Appendix topic.

1. Unpack the box and verify the package contents[85].

2. Connect the Handset to the dongle in the handset jack. If using a headset, connect the microphone and the speaker connectors into the dongle.



3. Connect one end of the Communication cable to the COM port of the dongle. Connect the other end of the communication cable into the COM port of the computer.

4. Connect a wire-pair of a DKP port to the RJ11 port marked 'DKP' on the dongle.

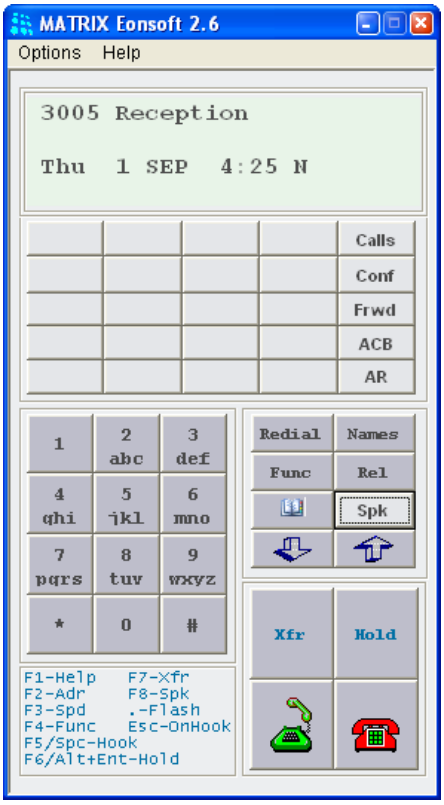
5. Switch ON the computer. The computer must have Windows Operating System installed on it.

6. Copy the EONSOFT Application Software provided by the Support Team onto your PC and install the application.

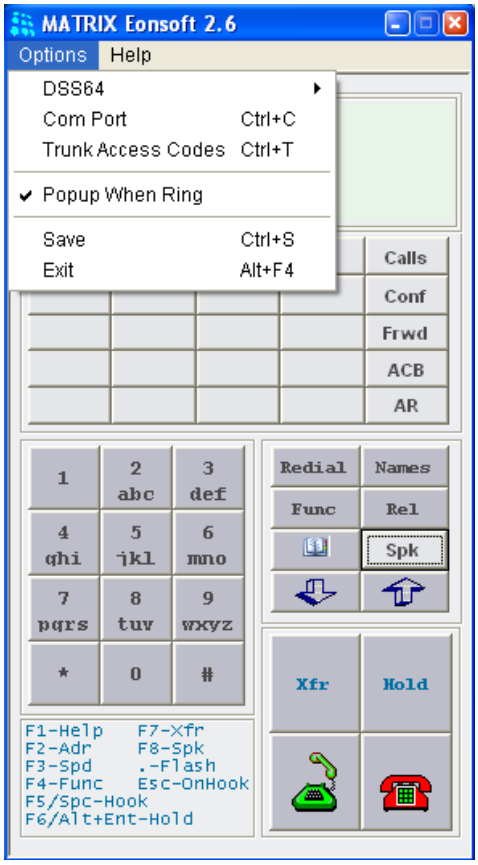
7. After the program has been installed and run, a shortcut will be automatically created and appear on your desktop.

85. *See "EONSOFT" under 'Packing List' of Appendix topic.*

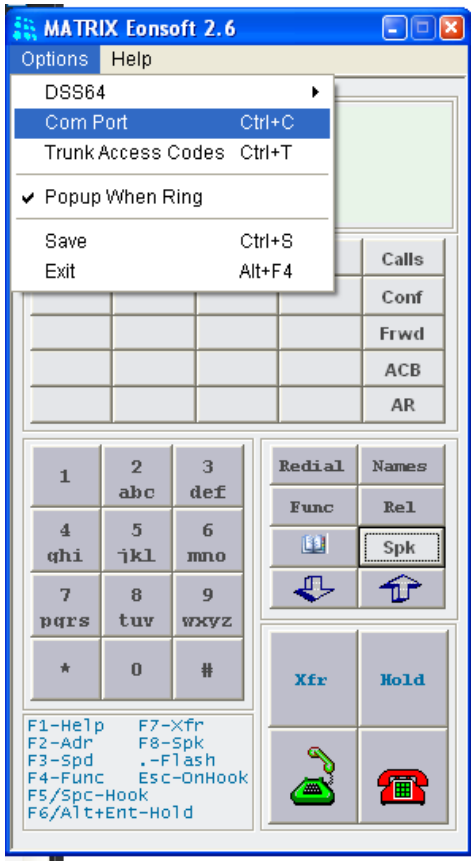
8. Click the shortcut to open the program. The EONSOFT window will open:



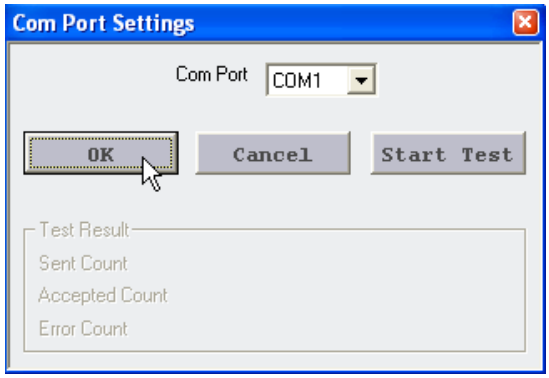
9. Click **Options** at the top left of the window. A drop down menu will appear.



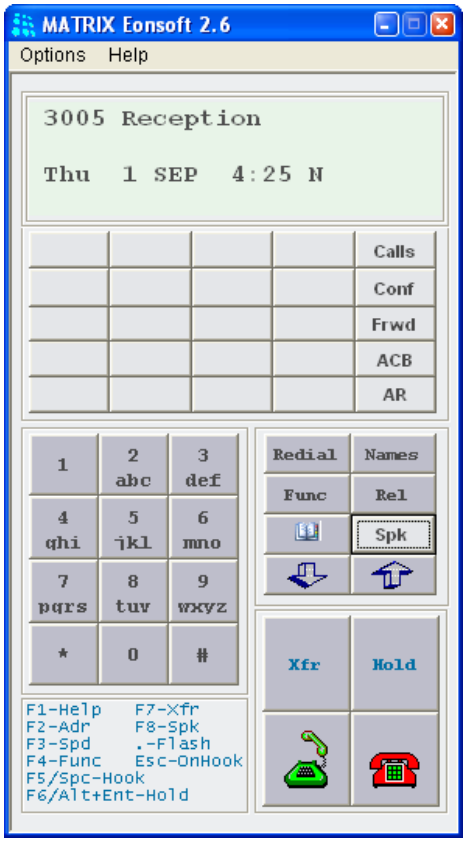
10. Click the option **COM Port**.



11. Select the COM Port to which the communication cable is connected.



12. EONSOFT is now connected. If you have already configured the DKP parameters like Access Code and Name for the port to which EONSOFT is connected, these will appear.



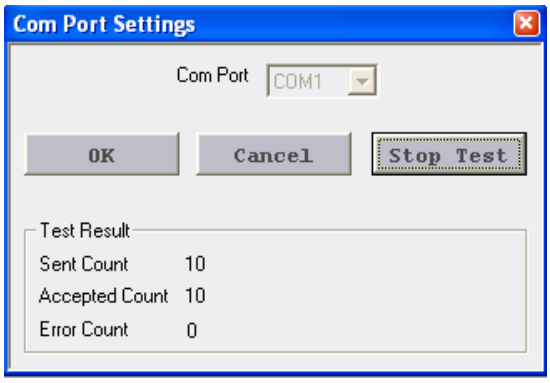
- *If this window does not appear after you have selected the COM Port Option, test the COM Port for data transfer.*
- *If the wrong COM port has been selected, a dialog box will pop up on your screen with the message: "COMx is invalid or busy, please select another COM Port". Select the correct COM Port.*

Test the functioning of the COM Port of the PC and the communication cable, before you install the EONSOFT.

Testing the COM Port

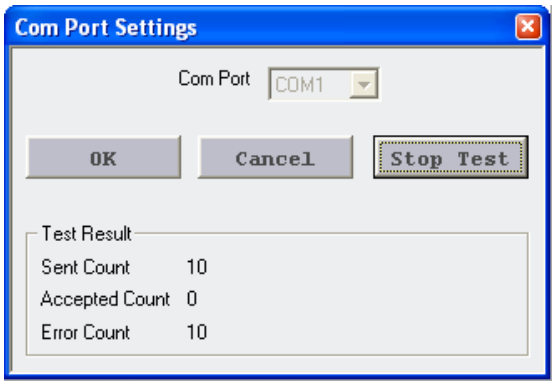
- From the drop down menu of **Options**, select the **COM Port** to which you have connected the communication cable.
- The COM Port Settings dialog box will open.
- Connect the communication cable to the COM Port of the PC.
- Short pin2 and pin3 of the DB-9 connector at the free end of the cable.
- Click the button labeled **Start Test** in the COM Port Settings dialog box.
- After clicking this button, observe the Test Result section on the dialog box.

- The **Error Count** value shows zero, if both the communication cable and the COM port are working.



The above screen shows that the COM Port/communication cable is working.

- If the **Error Count** shows a value other than zero, it means that either the communication cable or the COM port of the PC is faulty.



Above screen shows the faulty COM Port/Communication Cable.

- Remove the communication cable from the COM Port of the PC.
- Short pin2 and pin3 of the communication port of the computer and click 'Start Test' in the COM Port Settings dialog box.
- Now, if the error count is zero, please check the Communication Cable.
- If the error count is not a zero, the COM Port of the PC is faulty. Try another communication port.

The CO Card

The CO Card provides the interface to connect the system with the Two-Wire Analog Trunk lines from the CO Network. The CO Card supports the different standards and features of CO Networks across the world.

The CO Card is available in the following configurations. CO interface is also available in combination with SLT and DKP ports on a single card.

CO Cards for ETERNITY GENX

Card Name	Configuration and Application
ETERNITY GE CARD CO16	16-port card to connect 16 Two-wire Trunk lines from the CO network
ETERNITY GE CARD CO8	8-port card to connect 8 Two-wire Trunk lines from the CO network
ETERNITY GE CARD CO4+SLT16	Combination card, with 4 CO ports to connect 4 CO analog trunk lines and 16 SLT ports to connect 16 Single Line Telephones
ETERNITY GE CARD CO2+DKP2+SLT16	Combination card, with 2 CO ports to connect 2 Two-wire Trunk lines, 2 DKP ports to connect 2 Digital Key Phones, and 16 SLT ports to connect 16 Single Line Telephones
ETERNITY GE CARD CO4+DKP2+SLT12	Combination card, with 4 ports to connect 4 Two-wire Trunk lines, 2 ports to connect 2 Digital Key Phones and 12 ports to connect 12 Single Line Telephones. This Card supports Power Fail Transfer. To know more, see “Power Fail Transfer”.
ETERNITY GE CARD CO4+DKP2+SLT8	Combination card, with 4 ports to connect 4 Two-wire Trunk lines, 2 ports to connect 2 Digital Key Phones and 8 ports to connect 8 Single Line Telephones. This Card supports Power Fail Transfer. To know more, see “Power Fail Transfer”.

The maximum CO Trunk Ports supported are 64.

Connectors

The CO Card has RJ45 connectors, with 4 CO ports on each connector. A multi-pair, MDF cable is supplied for each connector on the card.

LED

The CO16 and CO8 Cards have two LEDs to indicate:

- the health of the card during the Reset Cycle
- the status of a selected Trunk port during normal functioning of the system

You can assign the LED to any CO port on the card which you want to monitor[86].

86. *To assign the LED to a selected port for monitoring its functioning, you must enter SE mode and dial the SE Command 7902-Slot--LED Number-Port, where Slot is the number of the universal slot in which the card is installed and Port is the port on the card to which the LED is to be assigned to monitor its functioning. LED Number is the number of the LED on the card, which will monitor the port.*

Among the combination cards CO2+DKP2+SLT16, CO2+DKP2+SLT8, CO4+CO16, CO4+DKP2+SLT8 have no LEDs. The combination card CO8+SLT8 has two LEDs.

LED Pattern of CO Card

LED 2 (L2)

PORT Status	LED Color	LED Cadence
Commands from Application to CO Port.	GREEN	Toggle[a]
Events to Application from CO Port.	RED	

a. The current LED state will remain the same until the next command is received from the application on the CO Port. For example, if the current LED state is Green/Red ON, on the next command received, the LED will be turned OFF. It will remain OFF until the next command is received. When the next command is received it will be turned Green/Red ON again. This process continues.

Installing the CO Card

For CO connectivity, you must install at least one of the above mentioned CO Cards in the system.

1. Take all the necessary precautions prescribed for handling the cards and electronic equipment. Make sure that power supply is turned off before you begin the installation of the card. Put on an electrostatic-discharge preventive wrist strap/belt and use a grounding mat.

2. Unpack the CO card and check the package contents.

3. Select any free (empty) slot from the Universal Slots. Unscrew and remove the filler bracket of the empty slot. Preserve the filler bracket for future use!

4. Insert the CO Card into the guide rails of the free slot you selected for the card. The connectors on the card should make perfect contact with those of the slot on the backplane motherboard.

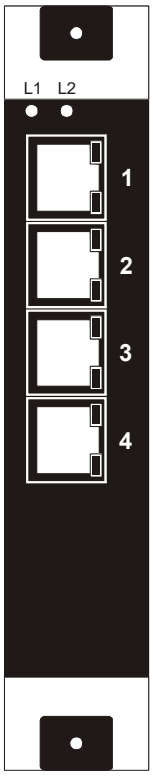
5. Press down the lever on the card mounting brackets to secure the card in its slot. Fix the mounting bracket in place with the two screws provided.

If installing more than one CO Card, it is not necessary to insert the other cards in subsequent slots. Any card can be inserted in any of the Universal Slots.

6. Use the cables supplied for each connector on the CO Card to connect the Trunk Lines with the Main Distribution Frame.

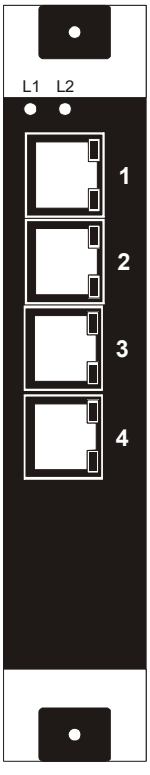
   Refer the illustrations below for the pinout details of the connectors on each card.

ETERNITY GE CARD CO16



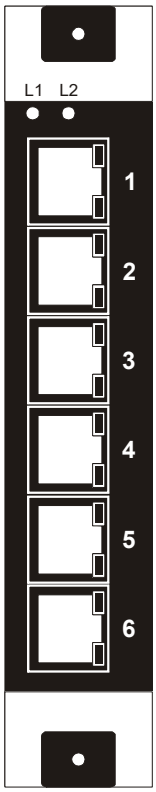
Connector	Color	Connection	H/w Port Offset
RJ45-1	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	CO CO CO CO	01 02 03 04
RJ45-2	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	CO CO CO CO	05 06 07 08
RJ45-3	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	CO CO CO CO	09 10 11 12
RJ45-4	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	CO CO CO CO	13 14 15 16

ETERNITY GE CARD CO8



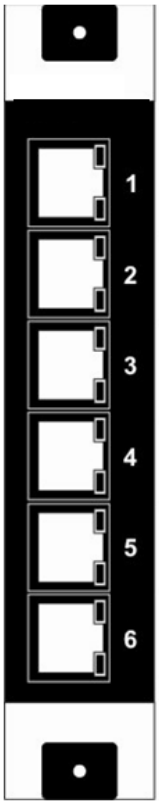
Connector	Color	Connection	H/w Port Offset
RJ45-1	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	CO CO CO CO	01 02 03 04
RJ45-2	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	CO CO CO CO	05 06 07 08
RJ45-3	Unused		
RJ45-4	Unused		

ETERNITY GE CARD CO4+SLT16



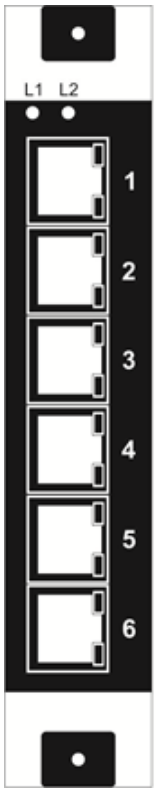
Connector	Color	Connection	H/w Port Offset
RJ45-1	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	01 02 03 04
RJ45-2	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	05 06 07 08
RJ45-3	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	09 10 11 12
RJ45-4	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	13 14 15 16
RJ45-5	Blue - (Blue & White) Orange - (Orange & White)	CO CO	01 02
RJ45-6	Blue - (Blue & White) Orange - (Orange & White)	CO CO	03 04

ETERNITY GE CARD CO2+DKP2+SLT16



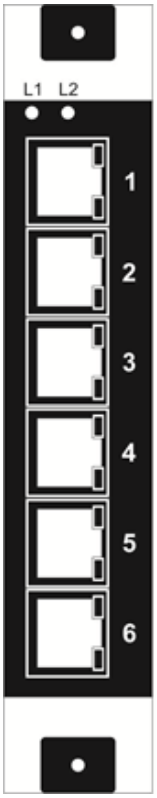
Connector	Color	Connection	H/w Port Offset
RJ45-1	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	01 02 03 04
RJ45-2	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	05 06 07 08
RJ45-3	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	09 10 11 12
RJ45-4	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	13 14 15 16
RJ45-5	Blue - (Blue & White) Orange - (Orange & White)	DKP DKP	01 02
RJ45-6	Blue - (Blue & White) Orange - (Orange & White)	CO CO	01 02

ETERNITY GE CARD CO4+DKP2+SLT12



Connector	Color	Connection	H/w Port Offset
RJ45-1	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	01 02 03 04
RJ45-2	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	05 06 07 08
RJ45-3	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	09 10 11 12
RJ45-4	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	CO CO CO CO	01 02 03 04
RJ45-5	Blue - (Blue & White) Orange - (Orange & White)	DKP DKP	01 02
RJ45-6	Unused		

ETERNITY GE CARD CO4+DKP2+SLT8



Connector	Color	Connection	H/w Port Offset
RJ45-1	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	01 02 03 04
RJ45-2	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	05 06 07 08
RJ45-3	Unused		
RJ45-4	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	CO CO CO CO	01 02 03 04
RJ45-5	Blue - (Blue & White) Orange - (Orange & White)	DKP DKP	01 02
RJ45-6	Unused		

7. Plug in the RJ45 end of the Trunk Card cables into the respective connectors. Refer to the connector diagrams illustrated above for each CO Card type.

8. Terminate the free end of the CO Card cable into the punch down blocks of the Krone modules designated for 'Trunk Lines' on “The Main Distribution Frame (MDF)”.

   Trunk cables from the system are to be connected with the Trunk Lines from the PSTN/CO terminated on the MDF. Each wire-pair form the CO Port must be terminated on the bottom of the Krone Connector, while the wire-pair of the trunk line from the CO Network to be connected to this port must be terminated on the top of the Krone Connector.

   Refer the topics “The Main Distribution Frame (MDF)” and “Terminating Trunk and Extension Cables on the MDF”.

9. Repeat these steps to install other CO Cards, if applicable.

The BRI Card

The BRI Card provides the interface to connect system with ISDN BRI Lines. The BRI lines may be from a public ISDN exchange, a private ISDN exchange.

BRI Cards of ETERNITY GENX

Card Name	Configuration and Application
ETERNITY GE CARD BRI4	4-Port card to connect 4 ISDN BRI Lines or ISDN Compatible Devices

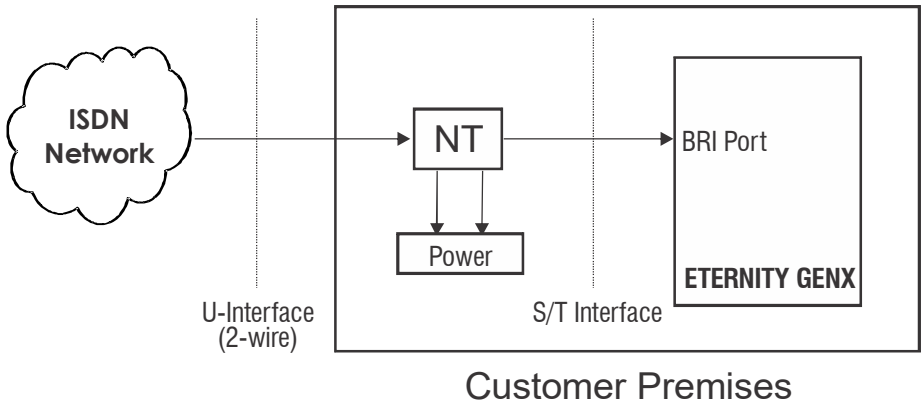
The maximum number of BRI lines supported are 32.

Connectors

The BRI Card has 4 RJ45 Connectors. A separate cable is supplied for each connector.

ISDN BRI - Installation Scenarios

Most ISDN Service Providers also provide the NT1 device along with the BRI line. The BRI Line from the ISDN central office is terminated on the NT1 on the Customer's Premises, as illustrated below.



Where,

- U Interface = between the NT1 equipment and the ISDN central office.
- S/T Interface = between the ISDN user equipment, in this case, ETERNITY GENX and the Network Interface Equipment (NT1).

The BRI line is terminated on the NT1. The S/T interface of the NT1 is connected to BRI port of the ETERNITY GENX.

TE and NT Modes

In this illustration, the BRI line from ISDN Service Provider is directly connected to BRI port of the ETERNITY GENX via the NT1 device. Here, the ETERNITY GENX is the Terminal Equipment, so the BRI Port must be programmed to work in the TE mode.

When an ISDN Phone is to be connected to the BRI port of ETERNITY GENX, the BRI port must be programmed to work in NT mode.

When a BRI port of another ISDN System is to be connected to the BRI port of the ETERNITY GENX, in such a configuration, you may configure

- the BRI port of the other ISDN System in the TE mode and the BRI Port of the ETERNITY GENX in the NT mode.

  OR

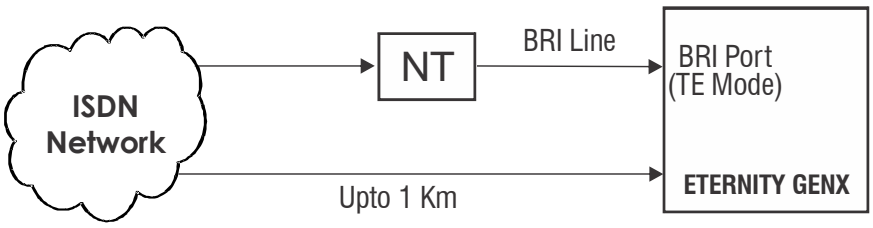
- the BRI port of the other ISDN System in the NT mode and the BRI Port of the ETERNITY GENX in the TE mode

Also refer the topic “Configuring BRI Trunks” to know more.

Types of BRI Configuration

There are two types of configurations in BRI: Point-to-Point Configuration and Point-to-Multipoint Configuration. Each of these is discussed below.

Point-to-Point Configuration



The maximum distance between the NT (Network Termination, NT1 or NT2) and a single Terminal Equipment, in this case ETERNITY GENX, can be up to 1 kilometer.

Point-to-Multipoint Configuration

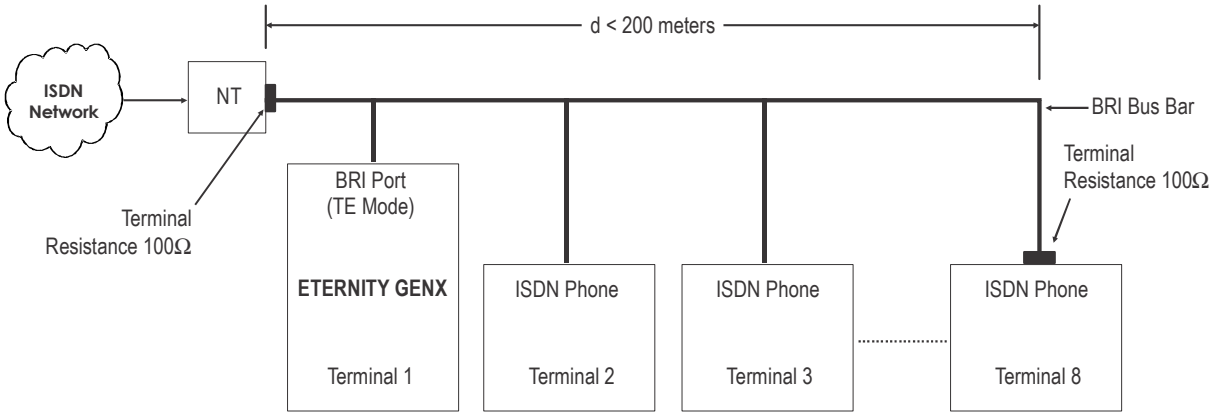
A maximum of 8 ISDN equipment can be connected on a single BRI Bus line in a Point-to-Multipoint configuration.

Further, two configurations are possible in a Point-to-Multipoint configuration:

a. Short Passive Bus Configuration

b. Extended Passive Bus Configuration

Short Passive Bus Configuration

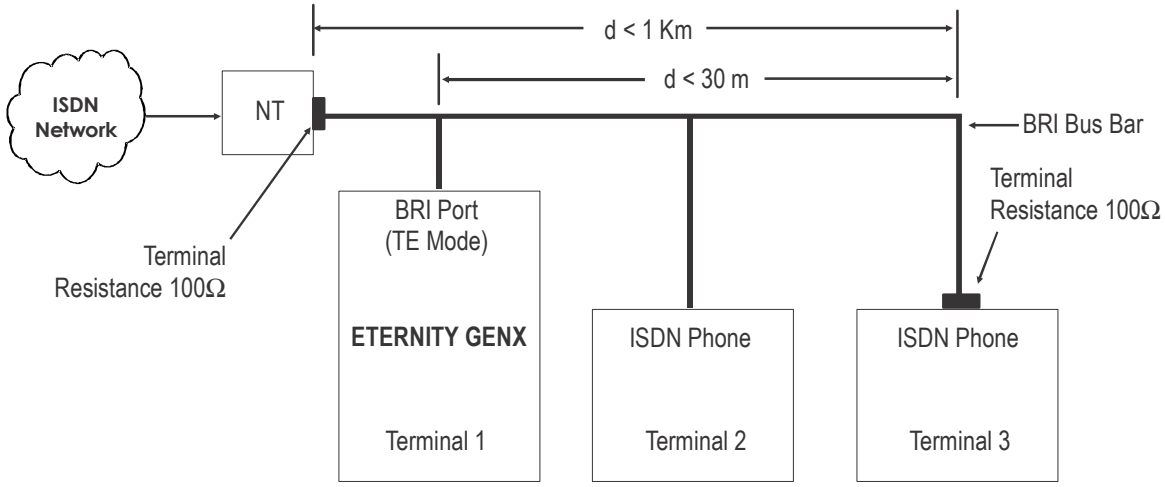


Where,
TE = Terminal Equipment or ISDN device (End user device)
NT = Network Termination provided by the ISDN Service Provider
d = distance from NT to the last TE equipment.

In a Short Passive Bus Configuration,

- A maximum of 8 TEs or ISDN devices can be connected to a single NT on a bus up to 200 meters from the NT.
- 100Ω Terminal Resistance is required to be inserted at the NT side as well as the last TE Equipment as shown in the figure.
- Using this configuration, any subscriber from ETERNITY GENX can access a BRI line and can make outgoing calls. At the same time, another subscriber from ETERNITY GENX or any ISDN phone shown in the figure can make outgoing call from the same BRI. In the same way, incoming calls are possible on the same BRI.
- Only two simultaneous speech paths can be established, as BRI supports 2 voice channels only.
- This configuration is useful on the smaller premises, where a single BRI line and multiple ISDN devices are used.

Extended Passive Bus Configuration



Where,
TE = Terminal equipment of any ISDN Equipment
NT = Network Termination provided by Service Provider
TR Terminal Resistance 100Ω
d = distance from NT to the last TE Equipment
d1 = the total distance from first TE equipment and the last TE equipment.

In an Extended Passive Bus Configuration,

- You can connect only 3 Terminal Equipment or ISDN devices. These devices are grouped together at one end of the bus, with may extend to a distance of up to 1 kilometer from the NT.
- However, all the 3 Terminal Equipment/ISDN devices must be located within a range of 30 meters, as shown in the figure.
- Using this configuration, any subscriber from ETERNITY GENX can access the BRI line and make outgoing calls. At the same time, another subscriber from the ETERNITY GENX or any ISDN phone shown in the figure can make outgoing calls from the same BRI. In the same way, incoming calls are possible on the same BRI.
- Only two simultaneous speech paths can be established, as BRI supports 2 voice channels only.
- This configuration is useful on large premises where a limited number of ISDN devices (maximum 3) are to be used within a range of 30 meters.

Installing the BRI Card

1. Take all the necessary precautions prescribed for handling the cards and electronic equipment: turn off power supply, always wear an electrostatic-discharge preventive wrist strap/belt and use a grounding mat.
2. Unpack the BRI Card and check the package contents.
3. Select any free (empty) slot from the Universal Slots. Unscrew and remove the filler bracket of the empty slot. Do not discard the filler bracket! Preserve it for future use!

Setting Orientation Type of BRI Port

4. The BRI Ports can be configured for different applications and can be interfaced directly with the BRI Network with Terminal Equipment like an ISDN Phone, with an ISDN-System.

   To connect the BRI Port to the public network, BRI Port must configured in the TE mode.

   To connect ISDN phones, an ISDN System or any ISDN equipment, the BRI Port must be configured in the NT mode.

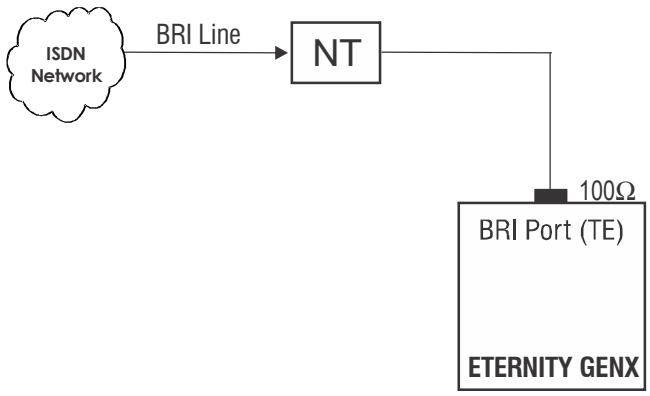
   By default, BRI Ports are configured in the TE mode.

   To set Orientation Type of the BRI Port, under **Configuration**, open the **BRI Configuration** link, and under **BRI Parameters**, set the **Orientation Type**.

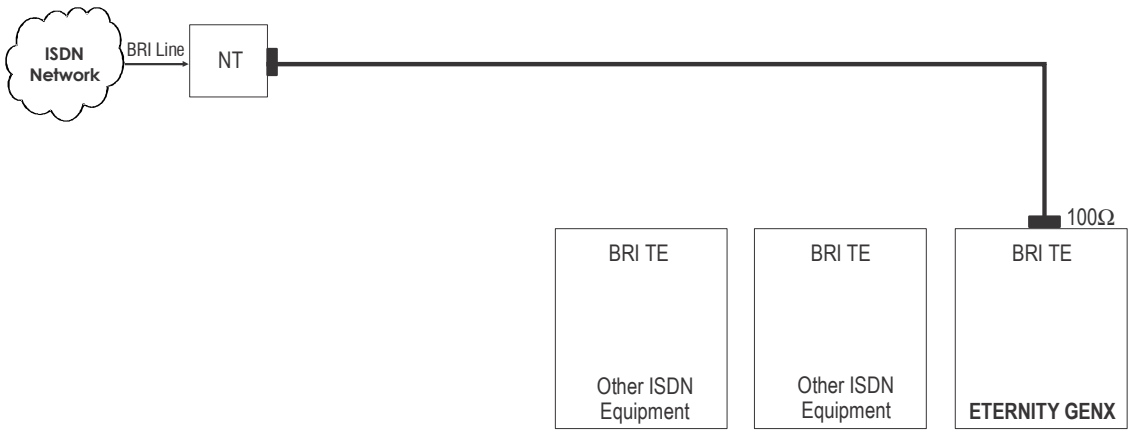
Inserting Termination Resistance

5. Termination of 100Ω should be inserted in the following cases:

   - When the BRI port is configured in the TE mode and connected in a Point-to-Point configuration as shown below.



   - When the BRI port is configured in the TE mode in a Point-to-Multipoint configuration as shown below. 100Ω Termination is required on the last Terminal connected on the S0 bus to terminate calls properly.



In a Point-to-Multipoint configuration, 100Ω termination can be provided on either of the following:

- Last TE equipment
- Last point of the bus bar where the last TE equipment is connected.

- When BRI port is configured in the NT mode.

- *If the S0 bus itself supports Terminating resistors, Termination Resistance need not be inserted when*
  - *BRI Port is configured as TE and connected in a Point-to-Point Configuration as illustrated above.*
  - *BRI Port is configured as NT.*

- *Termination need not be inserted if the BRI port of ETERNITY GENX (configured in TE mode) is connected as any terminal other than the last terminal on the S0 bus (in a Multi-point configuration).*

Termination in TE Equipment (BRI Port)

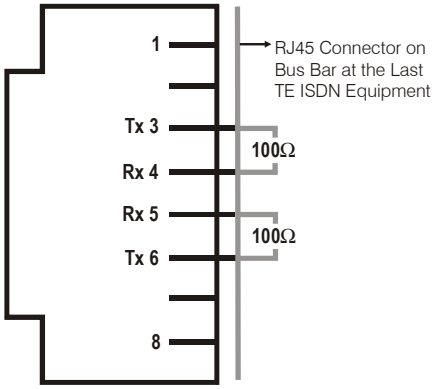
6. To set the 100Ω termination on the BRI port set the Jumpers on the BRI module of each port as given in the table below.

Function	Module 1 (M1)				Module 2 (M2)			
	BRI Port 1		BRI Port 2		BRI Port 3		BRI Port 4	
	Jumper Position		Jumper Position		Jumper Position		Jumper Position	
	J6	J8	J7	J9	J6	J8	J7	J9
To insert 100Ω termination	AB	AB	AB	AB	AB	AB	AB	AB
To remove 100Ω termination	BC	BC	BC	BC	BC	BC	BC	BC

By default, Termination Resistance of 100Ω is set on the BRI port (the Jumpers are in AB position)

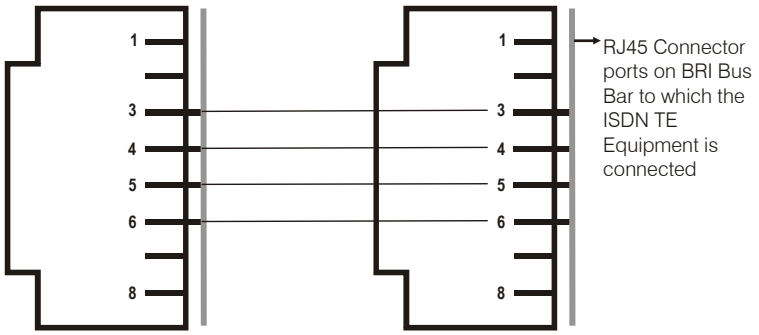
Termination in the Bus Bar

7. 100Ω termination resister can be connected between TX and RX, between pin number 3-4 and 5-6 in the RJ45 connector as illustrated below.



As shown in the application diagrams for Point-to-Multipoint connectivity, each ISDN TE device is connected in a Bus Bar, which may be Short Passive Bus Bar configuration or an Extended Passive Bus Bar configuration.

Illustrated below is the connection diagram of two ports connected with each other on the same BRI bus bar.



- The above figure shows the connection details of two ports on the BRI Bus Bar. Similarly, you can connect 8 ports on the Bus Bar, keeping in mind the Termination Resister for the NT and the Last TE on the Bus bar.
- Pin number 3, 4, 5 and 6 of the RJ45 connector are used for connectivity.
- Pin number 3 and 6 are used for Transmit (Tx) and pin number 4 and 5 are used for Receive (Rx) from the ISDN TE side.
- Pin number 3 and 6 are used for Receive (Rx) and pin number 4 and 5 are used for Transmit (Tx) from the NT side.

Feeding Power to the Terminal

8. When the BRI Port of the ETERNITY GENX is used as BRI-NT, you can feed power to the terminal equipment connected to the BRI-NT Port from the ETERNITY GENX.

   To do this,

   - Enable Feed Power on the BRI Port. For instructions see Power Feed under “Configuring BRI Trunks”.
   - By default, the Jumpers are set in AB position to feed power through Tx and Rx wires (Phantom Power). If you want to feed power through a separate pair of wires, you may change the position of the Jumpers on the BRI module as mentioned in the table below.

Function	**Module 1 (M1)**				**Module 2 (M2)**			
	BRI Port 1		BRI Port 2		BRI Port 3		BRI Port 4	
	Jumper Position		Jumper Position		Jumper Position		Jumper Position	
	J4	J5	J2	J3	J4	J5	J2	J3
To feed power on Tx and Rx wires (Phantom Power)	AB	AB	AB	AB	AB	AB	AB	AB
To feed power on separate pair of wires	BC	BC	BC	BC	BC	BC	BC	BC

- *The maximum power that can be fed to a single BRI port is 50mA.*
- *From signaling point of view, a maximum of 8 terminal equipment can be connected on the BRI port configured in the NT mode.*
- *The number of ISDN Terminals that can be connected on the BRI port configured in the NT mode depends on the power consumed by the ISDN terminals.*

9. Insert the BRI Card into the guide rails of the free slot you selected for the card. The connectors on the card should make perfect contact with those of the slot on the backplane motherboard.

Press down the lever on the card mounting brackets to secure the card in its slot. Fix the mounting bracket in place with the two screws provided.

If installing more than one BRI Card, it is not necessary to insert the other cards in subsequent slots. Any card can be inserted in any of the Universal Slots. Remember to set the Orientation Type, Termination Resistance and Power Feed, as required.

10. Use the straight cables supplied for each connector on the BRI card to connect the BRI Ports to the NT1 device supplied by your ISDN service provider. Refer the configuration and pinout details given below for guidance.

Configuration details of the U interface (RJ-45) at NT1

Pin Number	Pin Details
4	Tx
5	Rx

Configuration details of the S/T interface (RJ-45) on NT1

Pin Number	Pin Details
3	Rx1
4	Tx1
5	Tx2
6	Rx2

Pinout and Cable Details of BRI4 Port in TE Mode

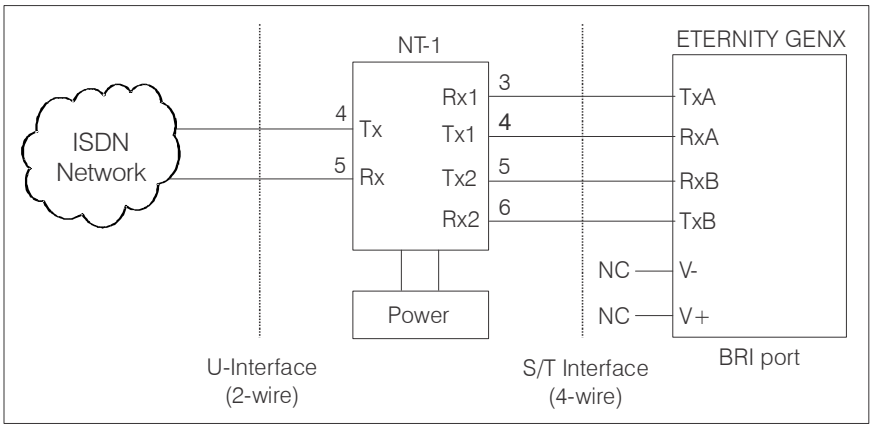
Pin	Color	Connection
1	Orange-White	Not connected
2	Orange	Not connected
3	Green-White	TxA
4	Blue	RxA
5	Blue-White	RxB
6	Green	TxB

Pin	Color	Connection
7	Brown-White	V-
8	Brown	V+

Pinout and Cable Details of BRI4 Port in NT Mode

Pin	Color	Connection
1	Orange-White	Not connected
2	Orange	Not connected
3	Green-White	RxA
4	Blue	TxA
5	Blue-White	TxB
6	Green	RxB
7	Brown-White	V-
8	Brown	V+

The following diagram shows how to connect a BRI Line to the BRI port in the TE mode.



11. If you have completed all other installation tasks, you may turn ON the system and observe the Reset Cycle and the LED pattern of the BRI Card.

LED Pattern of the BRI Card

The BRI4 Card has an LED for each port: L1, L2, L3, L4.

The LEDs show the Status of the Ports as summarized in the table below:

Port Status	LED Color	LED Cadence
Port is not active	RED	Continuously ON
Port is active	GREEN	Continuously ON

The T1E1PRI Card

The ETERNITY GE CARD T1E1PRI Single provides the interface to connect the system to ISDN PRI Network.

When connected to T1 carrier lines, the Card supports the following signaling types:

- PRI
- Robbed Bit Signaling
- Q-Signaling (QSIG)
- E&M

When connected to E1 carrier lines, the Card supports the following signaling types:

- PRI
- Channel Associated Signaling (CAS)
- Q-Signaling (QSIG)
- E&M

T1E1PRI Card for ETERNITY GENX

Card Name	Configuration and Application
ETERNITY GE CARD T1E1PRI Single	1-Port card with QSIG support to connect 1 ISDN T1/E1 PRI Line or ISDN Compatible Device

The maximum number of PRI Lines supported are 8.

Connectors

The T1E1PRI Card has an RJ45 Connector. A cable with RJ45 plugs on both ends is supplied for the connector.

LEDs

The ETERNITY GE CARD T1E1PRI Single has 2 LEDs - L1 and L2 - for indicating the port states.

Installing the T1E1PRI Card

1. Before installing the card, take the necessary precautions prescribed for handling the cards. Always wear an electrostatic-discharge preventive wrist strap and use a grounding mat. Make sure the power supply is turned off.

2. Unpack the T1E1PRI Card and check the package contents.

3. Select any free (empty) slot from the Universal Slots. Unscrew and remove the filler bracket of the empty slot. Do not discard the filler bracket.

Setting Line Termination Resistor

4. To set the Line Termination Resistor for the PRI Port for T1 or E1 Connectivity, you must change the position of the jumper J5. Refer the table below.

Jumper	Position	Meaning
J5	BC	To set termination resistance of 120Ω for E1 connectivity
J5	AB	To set termination resistance of 100Ω for T1 connectivity

   - By default J5 is set to BC position to provide 120Ω termination resistance for E1 connectivity.
   - To set 100Ω termination resistance for T1 connectivity, set jumper J5 to AB position.

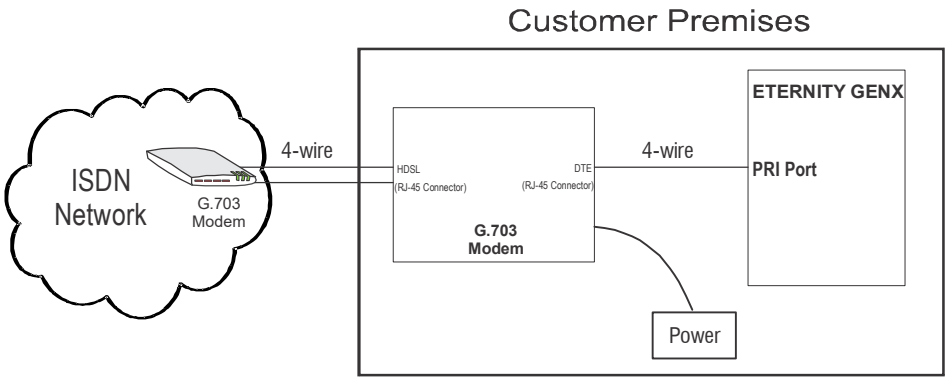
5. Insert the T1E1PRI Card into the guide rails of the free slot you selected for the card. Make sure that the connectors on the card make perfect contact with those of the slot on the backplane motherboard.

6. Now, press down the levers on the card mounting brackets to secure the card in its slot. Fix the card in place with the two screws provided.

Connecting ISDN T1/E1 PRI Lines

7. Use the cable supplied with the T1E1PRI Card to connect the system to the T1/E1 PRI network interface equipment (modem), which is usually supplied by your ISDN Service Provider along with the PRI line.

   The diagram below illustrates this.



   - Most Service Providers insist on connecting an ISDN modem at both the ends of the PRI line—one at the Local Exchange and other at the Customer's Premises.
   - At the Customer's Premises, the PRI line is terminated on the HDSL interface of the modem.
   - The DTE interface of the modem is to be connected to the PRI port (RJ-45 connector on the ETERNITY GE T1E1PRI Card).

8. Plug in one end of the RJ45 cable supplied with the card into the card's connector. Plug the other end of the RJ45 cable into the Network Termination Unit.

9. Refer the following pin details for connecting the Network Termination Unit with the system.

Pin details of HDSL Interface of the G.703 Modem. (HDSL Network Termination Unit)

Pin Number	Pin Details
1	Line A
2	Line A
3	Not used
4	Line B
5	Line B
6	Not used
7	Not used
8	Not used

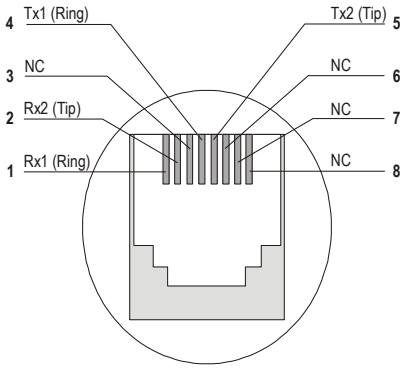
Pin details of DTE Interface of G.703 Modem. (HDSL Network Interface Unit)

Pin Number	Pin Details
1	TX1 (Tip)
2	TX2 (Ring)
3	Not used
4	RX1 (Ring)
5	RX2 (Tip)
6	Not used
7	Not used
8	Not used

Most of the HDSL Network Termination Unit manufacturers use these connectors. But you are advised to read the installation guide of the HDSL Network Termination Unit being used by you.

Pin details of ETERNITY GENX T1E1PR1 Port

The T1E1PRI Port of the system terminates in an 8-pin RJ45, female connector and is wired according to the table below.



The cable wires may have to be crossed depending on the pinout of the DTE Interface of the modem.

10. Repeat the same steps to install another card. It is not necessary to install the other T1E1PRI Cards in a sequence. Any card can be installed in any of the slots.

11. If you have completed all other installation tasks. Power the system. After the Reset Cycle, observe the LED patterns of the T1E1PRI Card.

LED Patterns

The ETERNITY GE CARD T1E1PRI has 2 LEDs: L1 and L2. Given below are the LED Patterns defined for each port state in the different signaling types supported by the system.

1. Port Active Mode

Signaling Type: E1-PRI

LED1 Pattern:

Port Status	Color	Cadence
Layer 1 established successfully	GREEN	Continuous ON
CRC4 Alarm	GREEN	100ms ON-100 ms OFF
BFA Alarm	RED	500ms ON-500 ms OFF
LOS Alarm	RED	Continuous ON

LED2 Pattern:

Port Status	Color	Cadence
Layer 1 established successfully	GREEN	Continuous ON
RAI Alarm	RED	500ms ON-500 ms OFF
AIS or LOS Alarm	RED	Continuous ON

Signaling Type: E1-CAS

LED1 Pattern:

Port Status	Color	Cadence
Layer 1 established successfully	GREEN	Continuous ON
CRC4 Alarm	GREEN	100ms ON-100 ms OFF
MFA Alarm	RED	100ms ON-100 ms OFF
BFA Alarm	RED	500ms ON-500 ms OFF
LOS Alarm	RED	Continuous ON

LED2 Pattern:

Port Status	Color	Cadence
Layer 1 established successfully	GREEN	Continuous ON
Y-Bit Alarm	GREEN	100ms ON-100 ms OFF
AIS16 Alarm	RED	100ms ON-100 ms OFF
RAI Alarm	RED	500ms ON-500 ms OFF
AIS or LOS Alarm	RED	Continuous ON

Signaling Type: T1-RBS or T1-PRI

LED1 Pattern:

Port Status	Color	Cadence
No Alarm	GREEN	Continuous ON
TFA Alarm or MFA Alarm	RED	500ms ON-500 ms OFF
AIS Alarm	RED	100ms ON-100 ms OFF
LOS Alarm	RED	Continuous ON

LED2 Pattern:

Port Status	Color	Cadence
Layer 1 established successfully	GREEN	Continuous ON
RAI or LOS Alarm	RED	Continuous ON

2. Port Maintenance Mode

LED1 Pattern:

Port Status	Color	Cadence
Maintenance Mode	RED -GREEN	500 ms RED-500 ms GREEN

LED2 Pattern:

Port Status	Color	Cadence
Near end loop back wait Before activate	RED	100ms ON-100 ms OFF
Near end loop active	RED	Continuous ON
Near end loop back wait Before deactivate	RED	500ms ON-500 ms OFF
Far end loop back wait after activate	GREEN	100ms ON-100 ms OFF
Far end loop active	GREEN	Continuous ON
Far end loop back wait after deactivate	GREEN	500ms ON-500 ms OFF

3. Port Disable Mode

LED1 Pattern:

Port Status	Color	Cadence
Port Disable	RED	Continuous ON

LED2 Pattern:

Port Status	Color	Cadence
Port Disabled	OFF	OFF

The E1FO Card

The ETERNITY GE CARD E1FO provides the interface to connect the system to the ISDN PRI Network. For E1 carrier lines the card supports the signaling types PRI, Q-Sig and CAS. For T1 carrier lines the card supports the signaling types PRI, Q-Sig and Robbed Bit Signaling. The T1/E1 ports can be set in Terminal or Network mode.

The E1FO Card supports Copper and Fiber Optic (FO) interfaces. At a time, either the Copper interface or the FO interface can be used. The FO interface supports only Single-mode (Mono mode) fiber connectivity and will work within a range of 30km.

E1 connectivity is supported over, the Copper interface as well as the Fiber Optic interface. The T1 connectivity is supported over the Copper interface only.

E1FO Card for ETERNITY GENX

Card Name	Configuration and Application
ETERNITY GE CARD E1FO Single	1-Port card to connect 1 ISDN T1/E1 Line on copper interface or E1 line on Fiber Optic (FO) interface.

Connectors

The E1FO Card has one RJ45 Connector and one Fiber Optic connector.

LEDs

The ETERNITY GE E1FO Card has 2 LEDs - L1 and L2 - for indicating the port states.

Installing the E1FO Card

1. Before installing the card, take the necessary precautions prescribed for handling the cards. Always wear an electrostatic-discharge preventive wrist strap and use a grounding mat. Make sure the power supply is turned off.

2. Unpack the E1FO Card and check the package contents.

3. Select any free (empty) slot from the Universal Slots. Unscrew and remove the filler bracket of the empty slot. Do not discard the filler bracket.

Setting Line Termination Resistor

4. To set the Line Termination Resistor for the Port for T1 or E1 Connectivity, you must change the position of the jumper J5. Refer the table below.

Jumper	Position	Meaning
J5	BC	To set termination resistance of 120Ω for E1 connectivity
J5	AB	To set termination resistance of 100Ω for T1 connectivity

- By default J5 is set to BC position to provide 120Ω termination resistance for E1 connectivity.

- To set 100Ω termination resistance for T1 connectivity, set jumper J5 to AB position.

5. Insert the E1FO Card into the guide rails of the free slot you selected for the card. Make sure that the connectors on the card make perfect contact with those of the slot on the backplane motherboard.

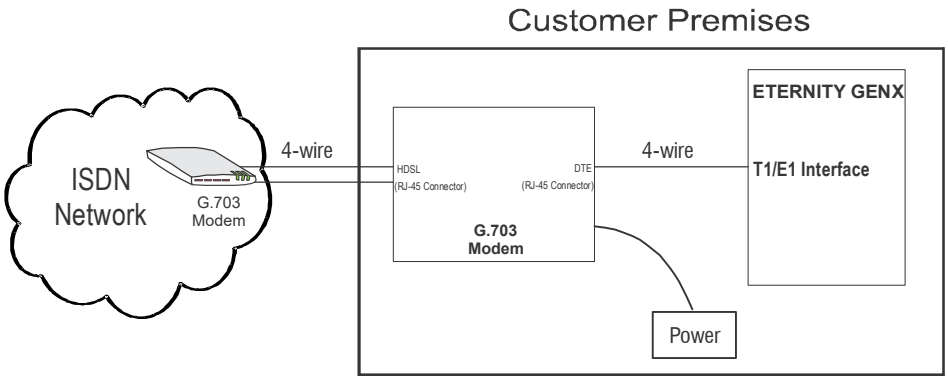
6. Now, press down the levers on the card mounting brackets to secure the card in its slot. Fix the card in place with the two screws provided.

Connecting the T1/E1 Lines

Copper Interface Connectivity

7. Use the cable to connect the T1/E1 Port to the T1/E1 network interface equipment (modem), which is usually supplied by your ISDN Service Provider along with the PRI line.

   The diagram below illustrates this.



- Most Service Providers insist on connecting an ISDN modem at both the ends of the PRI line—one at the Local Exchange and other at the Customer's Premises.

- At the Customer's Premises, the PRI line is terminated on the HDSL interface of the modem.

- The DTE interface of the modem is to be connected to the PRI port (RJ45 connector on the Matrix ETERNITY GE CARD E1FOPRI Single).

8. Plug in one end of the RJ45 cable into the card's connector. Plug the other end of the RJ45 cable into the Network Termination Unit.

9. Refer the following pin details for connecting the Network Termination Unit with the system.

   Pin details of HDSL Interface of the G.703 Modem. (HDSL Network Termination Unit)

Pin Number	Pin Details
1	Line A
2	Line A
3	Not used
4	Line B
5	Line B

Pin Number	Pin Details
6	Not used
7	Not used
8	Not used

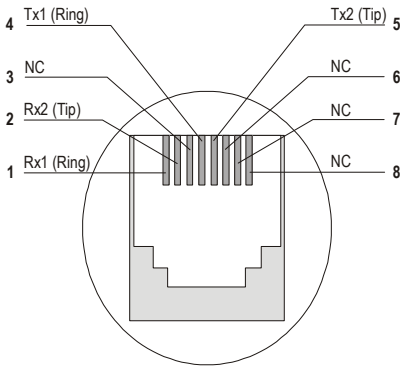
Pin details of DTE Interface of G.703 Modem. (HDSL Network Interface Unit)

Pin Number	Pin Details
1	TX1 (Tip)
2	TX2 (Ring)
3	Not used
4	RX1 (Ring)
5	RX2 (Tip)
6	Not used
7	Not used
8	Not used

Most of the HDSL Network Termination Unit manufacturers use these connectors. But you are advised to read the installation guide of the HDSL Network Termination Unit being used by you.

Pin details of the T1/E1 PRI Interface

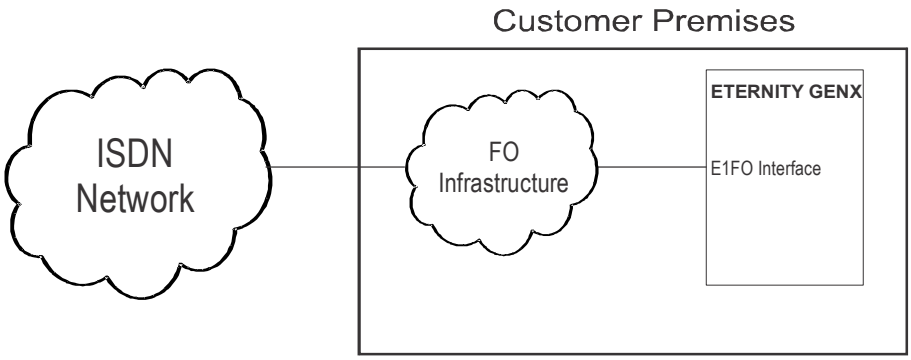
The T1/E1 Interface of the ETERNITY GE CARD E1FOPRI Single terminates in an 8-pin RJ45, female connector and is wired according to the table below.



The cable wires may have to be crossed depending on the pinout of the DTE Interface of the modem.

Fiber Optic Interface Connectivity

- *Fiber Optic interface supports E1 connectivity only.*
- *The Fiber Option (FO) interface supports only Single-mode (Mono mode) fiber connectivity and will work within a range of 30km.*



10. If your service provider provides Fiber Optic connectivity or you have an existing Fiber Optic infrastructure, you can connect the FO line to the E1FO interface of the E1FO Card.

    To use Fiber Optic interface, make sure you select **Line Coding Mechanism** as **NRZ (Fiber Optic)**. For details, see “Configuring E1 Trunks”.

11. Repeat the same steps to install another card. It is not necessary to install the other E1FO Cards in a sequence. Any card can be installed in any of the slots.

12. If you have completed all other installation tasks. Power the system. After the Reset Cycle, observe the LED patterns of the E1FO Card.

LED Patterns

The ETERNITY GE E1FO Card has 2 LEDs: L1 and L2. Given below are the LED Patterns defined for each port state in the different signaling types supported.

1. Port Active Mode

Signaling Type: E1-PRI

LED1 Pattern:

Port Status	Color	Cadence
Layer 1 established successfully	GREEN	Continuous ON
CRC4 Alarm	GREEN	100ms ON-100 ms OFF
BFA Alarm	RED	500ms ON-500 ms OFF
LOS Alarm	RED	Continuous ON

LED2 Pattern:

Port Status	Color	Cadence
Layer 1 established successfully	GREEN	Continuous ON
RAI Alarm	RED	500ms ON-500 ms OFF
AIS or LOS Alarm	RED	Continuous ON

Signaling Type: E1-CAS

LED1 Pattern:

Port Status	Color	Cadence
Layer 1 established successfully	GREEN	Continuous ON
CRC4 Alarm	GREEN	100ms ON-100 ms OFF
MFA Alarm	RED	100ms ON-100 ms OFF
BFA Alarm	RED	500ms ON-500 ms OFF
LOS Alarm	RED	Continuous ON

LED2 Pattern:

Port Status	Color	Cadence
Layer 1 established successfully	GREEN	Continuous ON
Y-Bit Alarm	GREEN	100ms ON-100 ms OFF
AIS16 Alarm	RED	100ms ON-100 ms OFF
RAI Alarm	RED	500ms ON-500 ms OFF
AIS or LOS Alarm	RED	Continuous ON

Signaling Type: T1-RBS or T1-PRI

LED1 Pattern:

Port Status	Color	Cadence
No Alarm	GREEN	Continuous ON
TFA Alarm or MFA Alarm	RED	500ms ON-500 ms OFF
AIS Alarm	RED	100ms ON-100 ms OFF
LOS Alarm	RED	Continuous ON

LED2 Pattern:

Port Status	Color	Cadence
Layer 1 established successfully	GREEN	Continuous ON

Port Status	Color	Cadence
RAI or LOS Alarm	RED	Continuous ON

2. Port Maintenance Mode

LED1 Pattern:

Port Status	Color	Cadence
Maintenance Mode	RED -GREEN	500 ms RED-500 ms GREEN

LED2 Pattern:

Port Status	Color	Cadence
Near end loop back wait Before activate	RED	100ms ON-100 ms OFF
Near end loop active	RED	Continuous ON
Near end loop back wait Before deactivate	RED	500ms ON-500 ms OFF
Far end loop back wait after activate	GREEN	100ms ON-100 ms OFF
Far end loop active	GREEN	Continuous ON
Far end loop back wait after deactivate	GREEN	500ms ON-500 ms OFF

3. Port Disable Mode

LED1 Pattern:

Port Status	Color	Cadence
Port Disable	RED	Continuous ON

LED2 Pattern:

Port Status	Color	Cadence
Port Disabled	OFF	OFF

The E&M Card

The E&M Card of the system provides the interface for analog trunking to connect various communication equipment telephone switches, Routers, Leased Lines, etc. using Tie-Lines.

The E&M Card is required for the following applications:

- Power Line Carrier Communication (PLCC) Networks, where several systems are connected with each other through E&M tie lines. Refer "PLCC-An Introduction" to know more.
- "Closed User Group (CUG)", where several systems are connected with each other through E&M tie lines[87].
- System expansion, where two systems are connected with each other with E&M tie lines.
- Connecting remote systems over E&M tie lines.

Refer the topic "E&M Connectivity" to know more.

An E&M Port can be programmed to behave as a Trunk Interface, a Subscriber (Station) Interface or both, as a Tie Line with the dual personality of a Trunk and a Subscriber.

The E&M Card supports

- E&M Interface - Types IV and V
- Speech Interface - Two-wire and four-wire.
- E&M Trunk Seizure Type[88]: Immediate, Immediate + Wink, Immediate with Ack, Immediate with Ack+Wink, Seizure Pulse, Seizure Pulse + Wink, Express, and Compander Control Signal.
- Address Signaling: Pulse dial (Pulse 10PPS, Pulse 20PPS) and Tone Dial (DTMF).

E&M Card for ETERNITY GENX

Card Name	Configuration and Application
ETERNITY GE CARD E&M4	4-port card to connect 4 E&M Tie Lines

The maximum number of E&M ports supported are 32.

Connectors

The E&M Card has RJ45 Connectors. A separate MDF cable is supplied for each connector.

LEDs

The ETERNITY GE E&M4 Card has 4 LEDs to indicate the functioning of the ports.

87. *The Systems in a "Closed User Group (CUG)" can be connected over ISDN T1E1PRI Lines as well. Refer the topic Closed User Groups to know more.*
88. *This is the line protocol that defines how the equipment seizes the E&M trunk. Also referred to as Start Dial Supervision Signaling Protocol.*

Installing the E&M Card

An E&M port can be programmed to take on the function of:

- **a Station** - works like an extension interface, receiving incoming calls.

  OR

- **a Trunk** - works like a trunk interface when any of the extensions of the system makes an outgoing call through it.

  OR

- **a Tie Line** - takes on a dual personality: functioning as both as an extension and a trunk. The E&M port works like an extension interface for incoming calls. It works like a trunk interface when any extension makes an outgoing call through it.

  This dual function is used in systems that are used as Transit Exchanges as in a PLCC Network. Read "PLCC-An Introduction" to know more.

You cannot connect a trunk line or an SLT / DKP to an E&M port.

1. Have the necessary wiring for the E&M Analog trunk in place. Take the necessary safety precautions before you begin handling the card; switch off power supply and always wear an antistatic wrist strap and use a grounding mat.

2. Unpack the E&M Card and check the package contents.

3. The E&M Card supports E&M Interface Type IV and Type V connection. To select the appropriate Interface Type out of the two, you need to change the Jumper Settings.

   Refer the table below to select the desired Interface Type and Speech Interface.

Jumper Number	Position	Function
J1 and J2	AB	Type IV E&M Interface
	BC	Type V E&M Interface

   - By default all the E&M Ports are set to support Type-IV.

   - To select the Type-V connection for the E&M Port, set Jumpers J1 and J2 (located on the E&M module) in BC Position.

4. Select the speech interface - 2-wire speech or 4-wire speech - as required, by changing the jumper settings. Refer the table below.

Jumper Number	Position	Function
J3 and J4	AB	4-wire speech interface
	BC	2-wire speech interface

   - By default all the E&M Ports are set to support 2-wire Speech Interface.

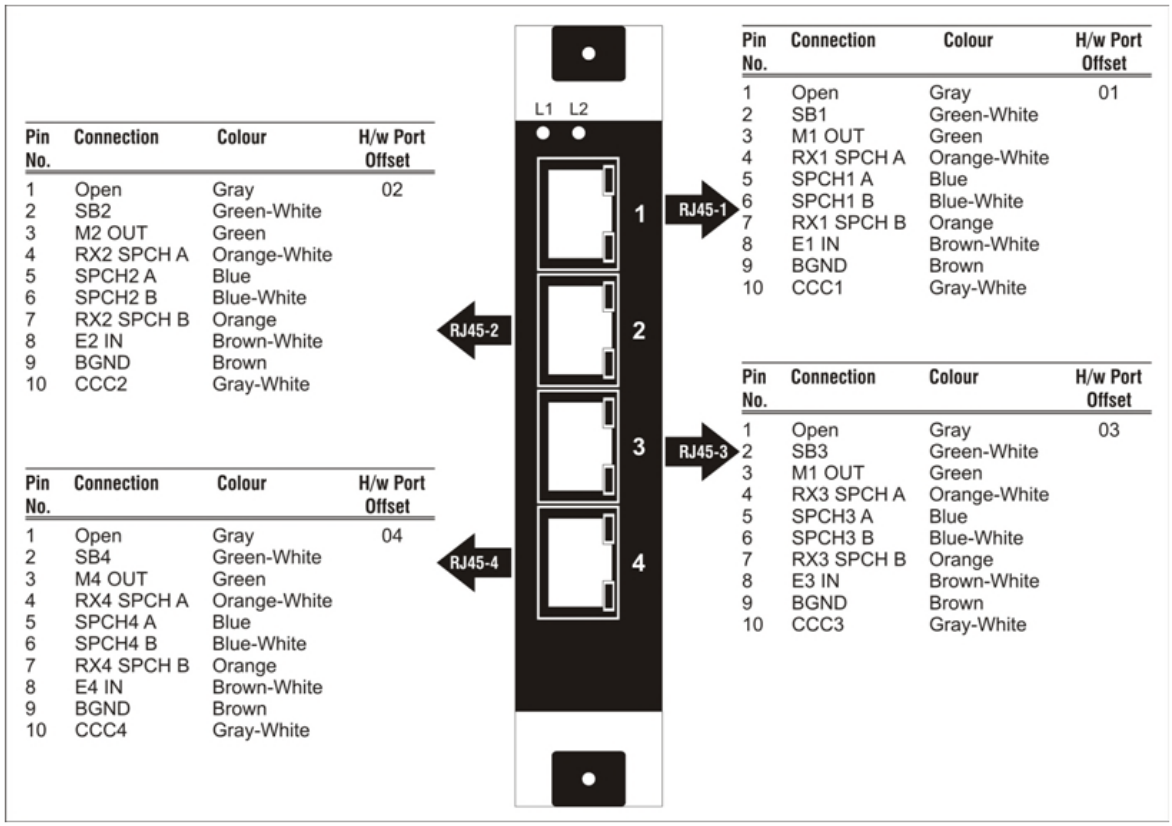
- To select 2-wire speech interface for the E&M Port, set Jumpers J3 and J4 (given on E&M module) to BC Position.
- To select 4-wire speech interface for the E&M Port, set Jumpers J3 and J4 on E&M module to AB Position.

Keep Jumper number J5 in BC position and Jumper number J6 in BC/Open position.

5. Now, select a free slot for the E&M Card. Unscrew and remove the filler bracket by pushing up the levers on the bracket. Preserve the filler bracket for future use.
6. Insert the E&M Card into guide rails of the empty slot. Make sure the connectors on the card make perfect contact with those on the backplane motherboard. Secure the card by pressing down the levers and fix the bracket with the screws provided with the card.
7. Connect the cables supplied with the E&M Card into the RJ45 connectors on the E&M Card.
8. Connect the free ends of the cables into the E&M Ports of the other System/PBX/Router/Tie Line equipment by appropriate crossing of the wires.

   Refer the following pin-out details for the E&M Card and for each Interface and Speech Interface Type.

Pinout details of ETERNITY GE CARD E&M4



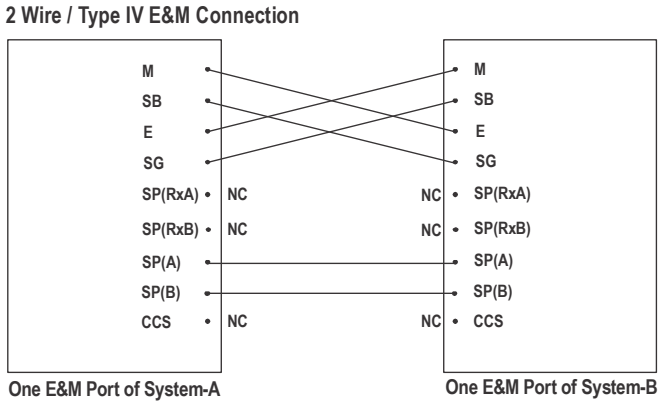
Pin No.	Connection	Colour	H/w Port Offset
1	Open	Gray	02
2	SB2	Green-White	
3	M2 OUT	Green	
4	RX2 SPCH A	Orange-White	
5	SPCH2 A	Blue	
6	SPCH2 B	Blue-White	
7	RX2 SPCH B	Orange	
8	E2 IN	Brown-White	
9	BGND	Brown	
10	CCC2	Gray-White	

Pin No.	Connection	Colour	H/w Port Offset
1	Open	Gray	04
2	SB4	Green-White	
3	M4 OUT	Green	
4	RX4 SPCH A	Orange-White	
5	SPCH4 A	Blue	
6	SPCH4 B	Blue-White	
7	RX4 SPCH B	Orange	
8	E4 IN	Brown-White	
9	BGND	Brown	
10	CCC4	Gray-White	

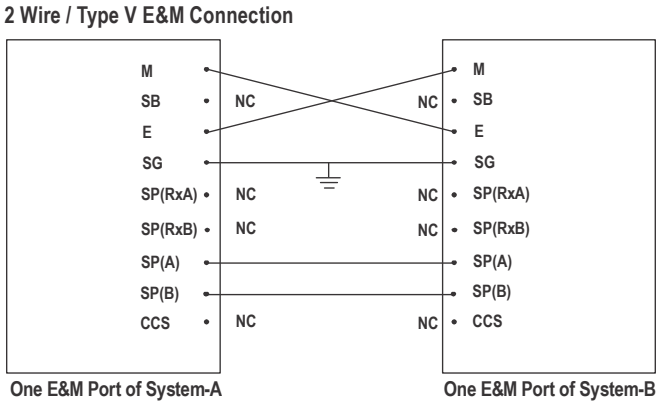
Pin No.	Connection	Colour	H/w Port Offset
1	Open	Gray	01
2	SB1	Green-White	
3	M1 OUT	Green	
4	RX1 SPCH A	Orange-White	
5	SPCH1 A	Blue	
6	SPCH1 B	Blue-White	
7	RX1 SPCH B	Orange	
8	E1 IN	Brown-White	
9	BGND	Brown	
10	CCC1	Gray-White	

Pin No.	Connection	Colour	H/w Port Offset
1	Open	Gray	03
2	SB3	Green-White	
3	M1 OUT	Green	
4	RX3 SPCH A	Orange-White	
5	SPCH3 A	Blue	
6	SPCH3 B	Blue-White	
7	RX3 SPCH B	Orange	
8	E3 IN	Brown-White	
9	BGND	Brown	
10	CCC3	Gray-White	

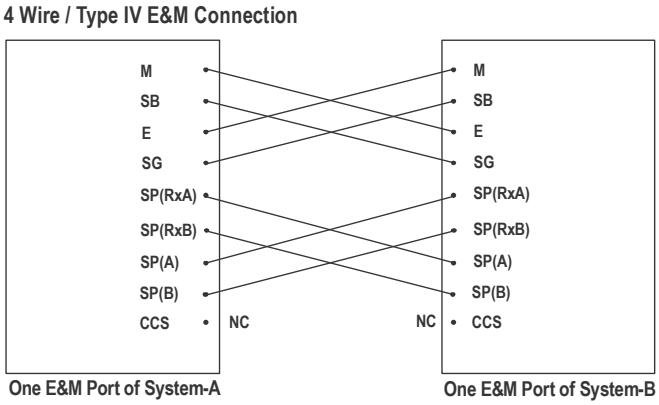
2 Wire, Type IV E&M Connection



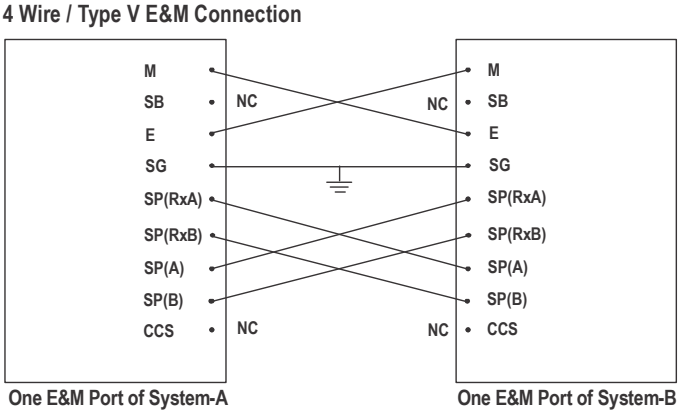
2 Wire, Type V E&M Connection



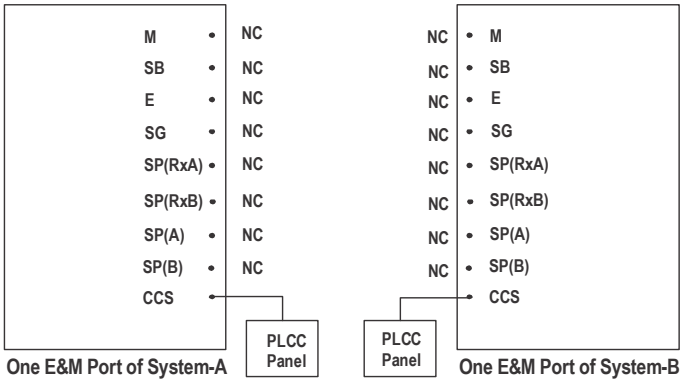
4 Wire, Type IV E&M Connection



4 Wire, Type V E&M Connection



9. If you are connecting two PLCC EPAX in a Power Line Carrier Communication Network Compander Control Signal (CCS) Connection should be made as illustrated in the block diagram below for any of the four combinations of E&M and Speech Interfaces illustrated in the previous step.



Compander Control Signal (CCS) is a special type of signal used by Power Line Carrier Communication Networks to improve quality of speech transmission. The PLCC network expects this signal from the system when speech is established and the E&M Card supports this facility. The system sends CCS signal to the PLCC panel.

- *When the E&M port is used as an Endpoint; the system sends a CCS to the PLCC panel while making an outgoing call through the E&M port or when a call is received at the E&M port.*
- *When the E&M port is used for Transit Exchange; the system sends a CCS to the PLCC panel while there is a Transit call through the E&M port.*

10. If you have completed all installation tasks, power ON the system, observe the Reset Cycle and the LED pattern of the E&M Card.

LED Pattern of E&M Card

Stage	LED Color	LED Cadence
At Power ON		LED OFF
After 30-60 seconds		LED OFF

Stage	LED Color	LED Cadence
After 60-90 seconds	RED	L1, L2, L3, L4 ON 500ms - L1, L2, L3, L4 OFF
	GREEN	L1, L2, L3, L4 ON 500ms - L1, L2, L3, L4 OFF
After 65-95 seconds	RED	L1, L2 L3, L4 ON 500ms - L1, L2, L3, L4 OFF
	GREEN	L1, L2 L3, L4 ON 500ms - L1, L2, L3, L4 OFF
Normal (Port Event)		
M-Wire High	Green	LED of the Port continuously ON
M-Wire Low		LED of the Port continuously OFF
E-Wire High	Red	LED of the Port continuously ON
E-Wire Low		LED of the Port continuously OFF
E-Wire and M-Wire High	Orange	LED of the Port continuously ON

The Mobile Card

The Mobile Card interfaces the system with GSM/3G/CDMA/LTE networks. It routes calls made and received over mobile networks, like a mobile handset.

The Mobile Cards are available in CDMA, 2G, 3G and 4G variants.

The Mobile Card does not support GPRS features, Fax and Data services, network supported services, except CLIR and USSD.

For compatibility and use of Matrix GSM products (2G/3G/4G) in Russia and Iran Province connect with Matrix Sales or Technical Support Team.

The Mobile Card for ETERNITY GENX

Card Name	Configuration and Application
ETERNITY GE CARD GSM4	4-port card to connect to 4 GSM networks (4 SIM Cards can be installed). To know more, refer to “ETERNITY GE CARD GSM4/GSM4 3G/CDMA2 without SIM Hot-swap”. For Hardware Design V2R2, CPLD V2R2 and PCB V2R1 This version onwards SIM Hot Swap is supported, that is the SIM card can be removed and inserted in the SIM Slots without turning off the system. To know more, refer to “ETERNITY GE Card GSM4/GSM4 3G/GSM4 4G with SIM Hot-swap”.
ETERNITY GE CARD GSM4 3G	4-port card to connect to 4 GSM networks with 3G support (4 SIM Cards can be installed). For Hardware Design V2R2, CPLD V2R2 and PCB V2R1 This version onwards SIM Hot Swap is supported, that is the SIM card can be removed and inserted in the SIM Slots without turning off the system. To know more, refer to “ETERNITY GE Card GSM4/GSM4 3G/GSM4 4G with SIM Hot-swap”.
ETERNITY GE CARD GSM4 4G	4-port card to connect to 4 GSM networks with 4G support (4 SIM Cards can be installed). To know more, refer to “ETERNITY GE Card GSM4/GSM4 3G/GSM4 4G with SIM Hot-swap”.
ETERNITY GE CARD CDMA2	2-port card to connect to 2 CDMA networks (2 RUIM Cards can be installed). Cards with Version V4R1 cannot be downgraded to earlier versions.

If you are installing CDMA Mobile Card in your system, it is recommended to avoid using features that support DTMF Detection.
The features where the caller is asked to dial digits and the system has to detect it, for example, DID, Voice Mail Auto Attendant etc might not work efficiently.

Just like mobile handsets, each Mobile Port has a unique IMEI (International Mobile Equipment Identity) number, pasted on the mobile engine.

The maximum Mobile ports supported are 40.
SIM cards from different service providers can be used.

Antenna

ETERNITY GE CARD GSM4 has a single antenna for the four ports. A splitter connects all the four ports on the card into a single antenna. An antenna cable is also provided, giving you the flexibility to move the antenna to another position (in case of weak signal).

Personal Identification Number (PIN)[89]

The SIM cards can be protected from unauthorized use by programming a Personal Identification Number (PIN) on the SIM. If the wrong SIM PIN is entered thrice in a row, by a user, the SIM card suspects the user and asks for the Personal Unlock Keyword (PUK).

Installing the Mobile Card

To be able to connect the system to GSM/3G/4G/CDMA networks, you must have one of the above mobile cards installed in the system.

1. To install the Mobile Card,

   - If using a GSM/3G/4G/CDMA card, get the SIM Card from the GSM/3G/4G/CDMA service provider of your choice ready. Use SIM PIN protection[90], if required.

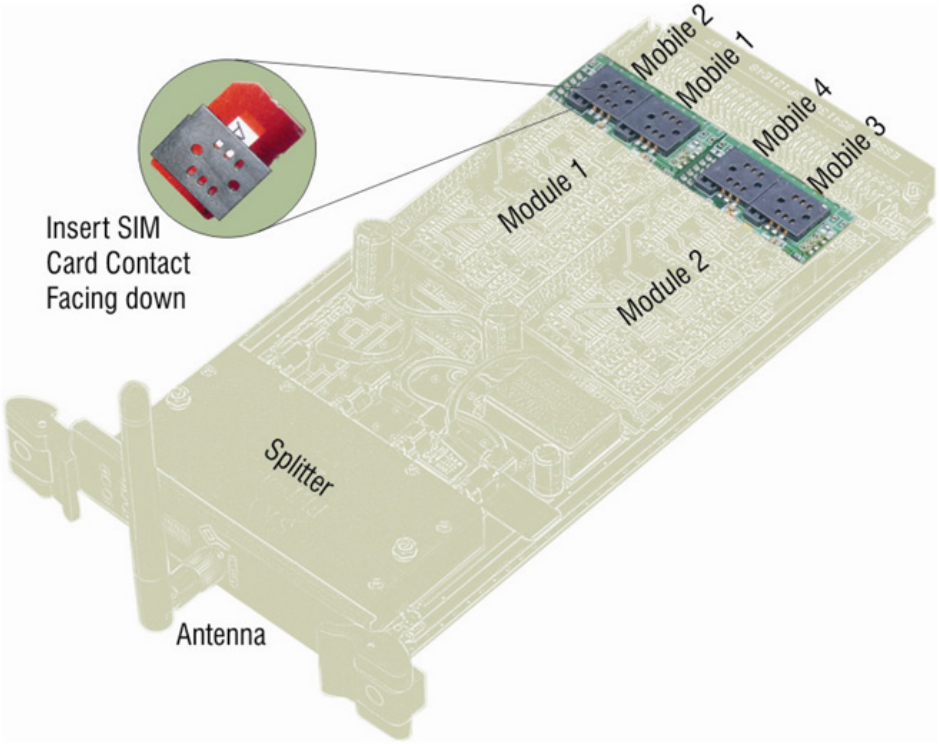
Disable Call Waiting in the SIM, else it may result in call disconnection.

2. Make sure that the system is installed at a location where sufficient network coverage is available. The power supply should be turned off, and you must be wearing an electrostatic discharge preventive wrist strap and must have a grounding mat, before you begin handling the card.

3. Unpack the Mobile Card and verify the package contents.

89. Not applicable if you are installing CDMA Mobile Card in your system.
90. SIM PIN Protection is not applicable if you are installing CDMA Mobile Card.

ETERNITY GE CARD GSM4/GSM4 3G/CDMA2[91] without SIM Hot-swap



Enabling PIN Protection on SIM[92]

4. For the 2G/3G/CDMA Card, enable SIM PIN before installing the SIM card in the system.
   - insert the SIM into a mobile handset first.
   - enable PIN Protection from the mobile handset.
   - change the SIM PIN to 1234 (this is the default PIN for all SIM cards used in the system). Changing the SIM PIN to '1234' enables you to change the SIM PIN from the Jeeves later (Refer SIM PIN under “Configuring Mobile Trunks” for instructions).
   - remove the SIM from the mobile handset.

If you do not want to use PIN protection, insert the SIM in the mobile handset and disable PIN protection. Remove the SIM Card from the mobile handset.

5. Insert the SIM card (PIN changed to 1234), with its connector side down into the SIM holder on the Mobile card. You can insert multiple SIM cards of the same GSM service provider or of different service providers.

6. Insert the Mobile Card into the guide rails of the Universal Slot you have selected for this card. Make sure that the card is inserted deep enough to make perfect contact with the connectors in the backplane. Now, press down the levers on the card mount bracket to secure the card in its slot.

7. Connect the antenna provided with the card on the splitter connector on the front panel of the card. You may also use the antenna cable to place the antenna at another position.

8. Repeat Steps 1-7 to insert another Mobile Card.

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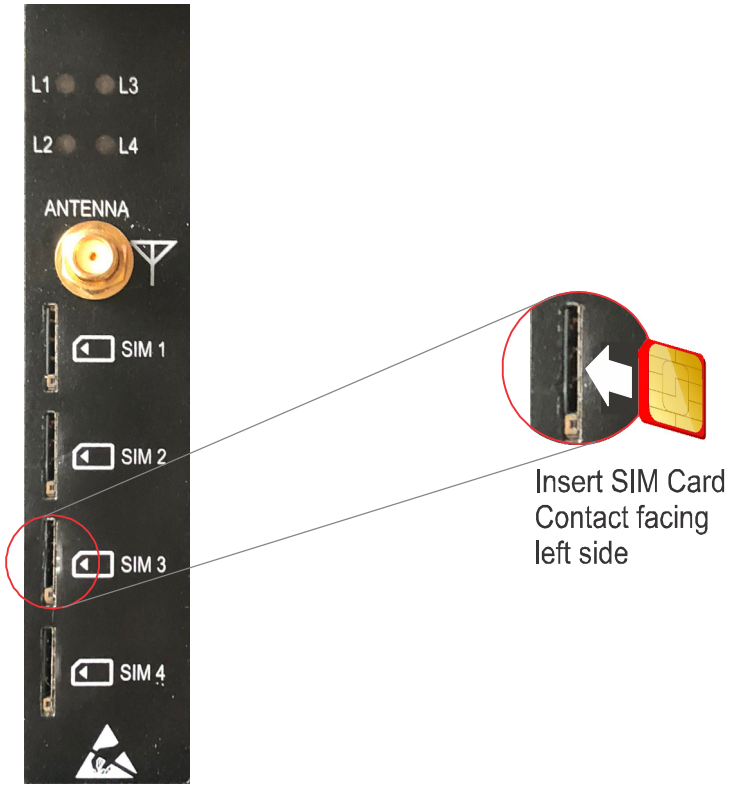
91. Insert RUIM Card instead of SIM Card if you are installing CDMA Mobile Card. Insert the RUIM Cards in Mobile 1 and 2.
92. Disable PIN Protection on RUIM if you are installing CDMA Mobile Card in your system.

9. If you have completed all installations tasks, power the system.

10. Wait for the system to register with the Mobile network. By default, the Mobile ports are set to select and register with the Mobile networks automatically. Now, observe the LED Patterns of the Mobile Ports.

- *At every power up of the system, it takes about 3 minutes for the Mobile ports to get registered with the network. Once registration with the GSM network is completed, the mobile port can be used.*
- *Each time the Mobile Port sends a request, such as Registration Request, the system waits for the duration of the Network Response Timer. This Timer signifies the time for which the Mobile Port waits for a response from the Mobile network. It is fixed for 150 seconds for all Mobile ports.*

ETERNITY GE Card GSM4/GSM4 3G/GSM4 4G with SIM Hot-swap



Enabling PIN Protection on SIM

4. For the 2G/3G/4G Card, enable SIM PIN before installing the SIM card in the system.
   - insert the SIM into a mobile handset first.
   - enable PIN Protection from the mobile handset.
   - change the SIM PIN to 1234 (this is the default PIN for all SIM cards used in the system). Changing the SIM PIN to '1234' enables you to change the SIM PIN from the Jeeves later (Refer SIM PIN under “Configuring Mobile Trunks” for instructions).
   - remove the SIM from the mobile handset.

If you do not want to use PIN protection, insert the SIM in the mobile handset and disable PIN protection. Remove the SIM Card from the mobile handset.

5. Insert the SIM with its contact side facing left into the SIM slot located on the fascia of ETERNITY GE Card.

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6. Push the SIM backwards into the slot until you hear a click and the SIM is locked in place.

7. To unlock the SIM, push the protruded portion of the SIM backwards again and release it.

The Mobile cards with SIM Hot - swap are designed keeping in mind the Standard Nano SIM size. In case, you face any issues due to the SIM size, contact your respective Service Provider for assistance.

8. Repeat the same steps to insert another SIM Card. You can insert multiple SIM cards of the same GSM service provider or of different service providers.

9. Insert the Mobile Card into the guide rails of the Universal Slot you have selected for this card. Make sure that the card is inserted deep enough to make perfect contact with the connectors in the backplane. Now, press down the levers on the card mount bracket to secure the card in its slot.

10. Connect the antenna provided with the card on the splitter connector on the front panel of the card. You may also use the antenna cable to place the antenna at another position.

11. Repeat Steps 1-7 to insert another Mobile Card.

12. If you have completed all installations tasks, power the system.

13. Wait for the system to register with the Mobile network. By default, the Mobile ports are set to select and register with the Mobile networks automatically. Now, observe the LED Patterns of the Mobile Ports.

- *At every power up of the system, it takes about 3 minutes for the Mobile ports to get registered with the network. Once registration with the GSM network is completed, the mobile port can be used.*
- *Each time the Mobile Port sends a request, such as Registration Request, the system waits for the duration of the Network Response Timer. This Timer signifies the time for which the Mobile Port waits for a response from the Mobile network. It is fixed for 150 seconds for all Mobile ports.*

LED Pattern of Mobile Ports

The number of LEDs on the Mobile Card corresponds with the number of mobile ports on the card.

After the Reset cycle is complete, during normal functioning, the LEDs color and cadence is described in the table below for various events on the Mobile port:

Event	Color	Cadence in msec (1 cadence is of 3000 msec)
Port idle	-	LED OFF
Port Active (All States other than Ring and Speech)	Red	Continuous ON
Ring Event	Green	400ms ON-200ms OFF400ms ON-200ms OFF
Speech	Green	Continuous ON
GSM initialization	Orange	200ms ON-200ms OFF-200ms ON-200ms OFF-200ms ON-200ms OFF-200ms ON-200ms OFF-200ms ON-1200ms OFF (5 blinks)
PUK required	Orange	200ms ON-200ms OFF-200ms ON-200ms OFF-200ms ON-200ms OFF-200ms ON-1600ms OFF-

Event	Color	Cadence in msec (1 cadence is of 3000 msec)
SIM PIN faulty	Orange	200ms ON-200ms OFF-200ms ON-200ms OFF-200ms ON-2000ms OFF (3 blinks)
SIM Absent	Orange	200ms ON-200ms OFF-200ms ON-2400ms OFF (2 blinks)
Network Link Down (Absence of GSM Network)	Orange	200ms ON-2800ms OFF

The Magneto Card

The Magneto Card is used for connecting the system to Magneto Telephones[93], which are widely used by the defense establishments as field phones in front lines, and by other establishments such as railroad companies (signaling emergencies, crossings, etc.), electric utilities, pipeline companies, who need to have their networks at places that are too remote to be serviced by public telephone networks.

The Magneto Card lands calls from magneto field telephones on the extensions of the system and places calls from the extensions of the system on magneto telephones.

Magneto Card for ETERNITY GENX

Card Name	Configuration and Application
ETERNITY GE CARD MAGNETO4	4-port card to connect 4 Magneto Phones

The maximum number of magneto ports supported are 16.

Connectors

The Magneto Card has RJ45 connectors. A multi-pair cable is provided for each connector.

LED

The ETERNITY GE CARD MAGNETO4 does not support any LED.

Installing the Magneto Card

1. Have the necessary wiring for the Magneto Ports in place.

   You may install an MDF to connect the Magneto Ports with the Field Telephone wires.

   OR

   You may connect the wires from the Magneto Field Telephones directly to the Magneto Port.

   You are advised to use a separate set of Krone Modules for connecting the Magneto phones to the Magneto ports of the system

2. Prepare for the card installation by switching off power supply and wearing an electrostatic discharge preventive wrist strap and use a grounding mat.

3. Unpack the Magneto Card and check the package contents.

4. Select any universal slot to insert the card. Unscrew the filler bracket and remove it by pushing up the levers on the bracket.

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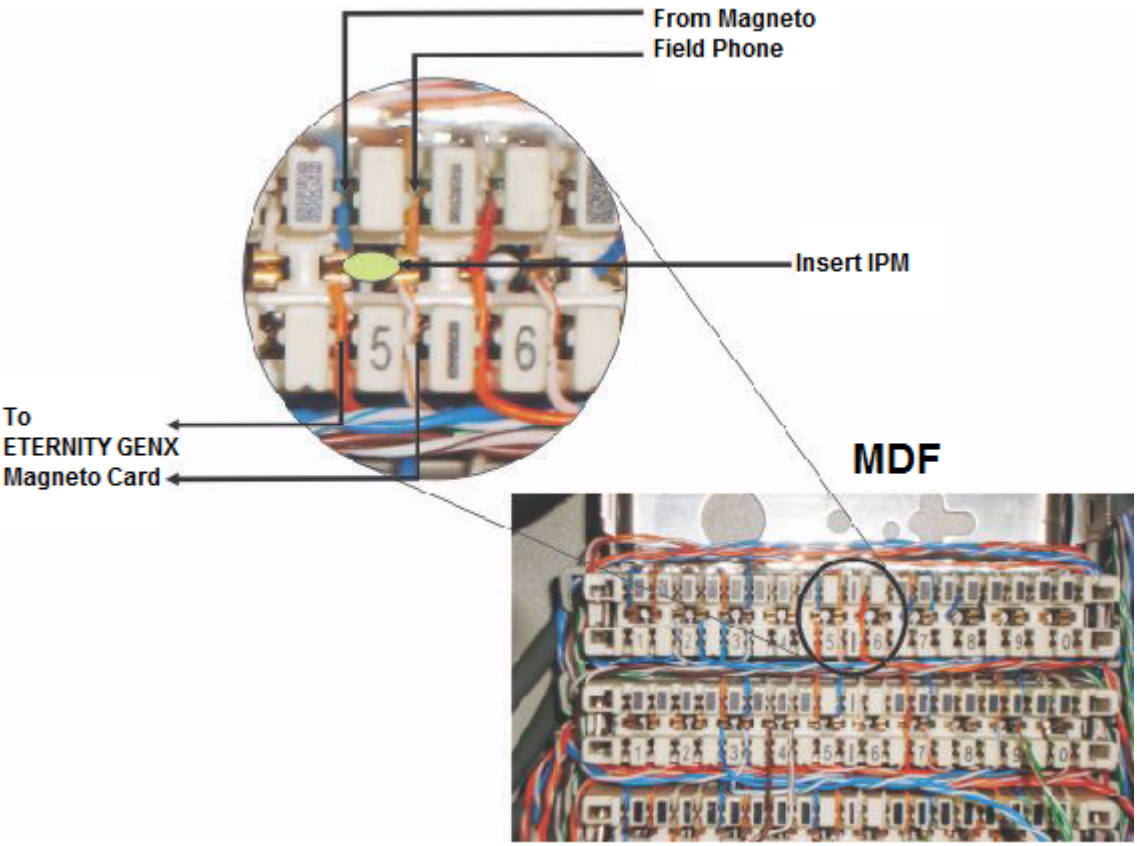
93. A magneto telephone is a local battery telephone set, in which signaling current is provided by a magneto hand generator, usually a magneto. The hand generator, commonly referred to as 'crank', is located on the right hand side of the telephone set and is turned to produce energy to ring other phones or to signal the CO. The magneto, also called the generator, is used to convert the mechanical motion via the crank to produce sufficient energy to ring other phones or to signal the CO.

5. Insert the Magneto card into the guide rails of the free slot. The card's connectors must make perfect contact with the connectors on the backplane motherboard. Press down the levers of the mounting bracket to secure the card in its slot and fix the two screws provided with the card on the mounting bracket.

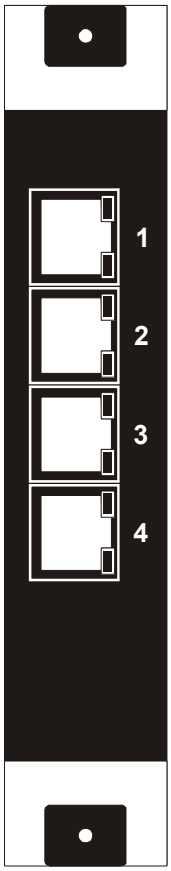
6. Now, plug in the cables supplied with the Magneto Card into the connectors on the card. Terminate the free ends of the cables into the MDF, if applicable.

   Refer to the following block diagram for terminating the cables from the Magneto Card and the wires from the Magneto Field Telephones.

Connecting Magneto Telephones to the Magneto Card



7. Connect the pairs of wires from the Magneto Field Phones to the appropriate pairs emerging from the ETERNITY GE CARD MAGNETO4 on the MDF. Refer the cable diagram below.



Connector	Color	Connection	H/w Port Offset
RJ45-1 (Blue)	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	Magneto - - -	01 - - -
RJ45-1 (Orange)	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	Magneto - - -	02 - - -
RJ45-1 (Green)	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	Magneto - - -	03 - - -
RJ45-1 (Brown)	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	Magneto - - -	04 - - -

8. Repeat the same steps to install the other Magneto Cards.

9. If you do not have any other Card to insert and have completed the installation procedures, power on the system.

The Radio Card

The Radio Interface Card (RIC) adds the Two-way Radio functionality in the system. In Two-way radio, the speech can be transmitted as well as received by the radio devices such as Radio Phone, Radio Repeater. Such devices are called Radio Transceivers. The Two-way radio works on High Frequency (HF), Very High Frequency (VHF) or Ultra High Frequency (UHF).

Radio Cards for ETERNITY GENX

Card Name	Configuration and Application
ETERNITY GE CARD RADIO4	4-port card to connect 4 Radio devices

The maximum number of radio ported supported are 16.

Connectors

The Radio Card has RJ45 connectors. A multi-pair cable is provided for each connector.

LED

The ETERNITY GE CARD RADIO4 does not support any LED.

Installing the Radio Card

1. Have the necessary wiring for the Radio Ports in place.

   You may install an MDF to connect the Radio Ports with the Radio device wires.

   OR

   You may connect the wires from the Radio device directly to the Radio Port.

   You are advised to use a separate set of Krone Modules for connecting the Radio devices to the Radio ports of the system.

2. Prepare for the card installation by switching off power supply and wearing an electrostatic discharge preventive wrist strap and use a grounding mat.

3. Unpack the Radio Card and check the package contents.

4. Select any universal slot to insert the card. Unscrew the filler bracket and remove it by pushing up the levers on the bracket.

5. Insert the Radio Card into the guide rails of the free slot. The card's connectors must make perfect contact with the connectors on the backplane motherboard. Press down the levers of the mounting bracket to secure the card in its slot and fix the two screws provided with the card on the mounting bracket.

6. Now, plug in the cables supplied with the Radio Card into the connectors on the card. Terminate the free ends of the cables into the MDF, if applicable.

7. Connect the pairs of wires from the Radio devices to the appropriate pairs emerging from the Radio Card of the system on the MDF. For more details, see “The Main Distribution Frame (MDF)”.

Refer to the Pin-out details given below.

ETERNITY GENX Card Radio4

Connector	Color	Pin Number	Signaling	H/w Port Offset
RJ45-1 to RJ45-4	Orange & White	1	PTT	01 to 04
	Orange	2	PTT_RTN	
	Green & White	3	Rx-	
	Blue	4	Tx+	
	Blue & White	5	Tx-	
	Green	6	Rx+	
	Brown & White	7	Unused	
	Brown	8	Unused	

8. Repeat the same steps to install the other Radio Cards.

9. If you do not have any other Card to insert and have completed the installation procedures, power on the system.

The Data Card

The Data Card supports four Ethernet 10/100mbps interfaces. Ethernet data coming to ports can be mapped to 2Mb streams. Each data port can be mapped to one 2Mb - E1 stream, that is 30 channels. The remaining channels of E1 can be used for voice applications. The Data Card has a 4-port Ethernet switch on board, which can aggregate multiple data streams to PCM streams of the system.

The Data Card can be installed in any of the Universal Slots of the system.

Ports and Connectors

The Data Card has an RJ45 Connector for each port. Use the cables supplied with the card for connectivity.

You can connect the cables from the LAN switch to these connectors.

LEDs

The Data Card does not have any LED.

Installing the Data Card

1. Unpack the Data Card and check the package contents. It is recommended that you switch off the power supply, before you begin the installation of the card. Always wear an electrostatic discharge prevention wrist strap/belt and use a grounding mat.

2. Select any free (empty) slot from the Universal Slots. Unscrew and remove the filler bracket of the empty slot. Do not discard the filler bracket! Preserve it for future use!

3. Insert the Data Card into the guide rails of the free slot you selected for the card. The connectors on the card should make perfect contact with those of the slot on the backplane motherboard.

   Press down the lever on the card mounting brackets to secure the card in its slot. Fix the mounting bracket in place with the two screws provided.

4. Use the cable supplied with the card to connect the Data Ports to the Ethernet Network (Switch/PC).

5. If you have completed all other installation tasks, you may turn ON the system.

Jumpers

Jumper Number	Position	Function
J7	AB (default)	External Boot - Normal
	BC	Internal Boot

SIP Extensions

SARVAM UCS supports up to 999 SIP/UC Users. The SIP/UC Users function in the same way as DKP/SLT extensions of the system. SIP/UC Users can make and receive calls to any extension user of the system and to external numbers over various telecom networks like CO, Mobile, ISDN PRI, BRI, and VoIP[94].

You may register any SIP-enabled device — a Matrix UC Client, an IP-phone, a Soft phone, Analog Phone Adapter — as the SIP User of the system.

The Matrix UC Clients also offer UC functionalities in addition to the SIP functionalities.

The SIP Users register with the CPU Card of the system. Five free SIP Users are provided by default. You may register any of the SIP-enabled devices except the Matrix UC Clients with these free SIP Users. For registering the Matrix UC Clients, you must purchase the Matrix VARTA User License. If you require additional SIP Extensions you must purchase the IP Subscribers License.

The system supports two NX DBM VOCODER64 Modules. You must purchase the module separately. Each NX DBM VOCODER64 module supports a maximum of 64 VOCODER channels. The Vocoder channels are required for — VoIP to Non-VoIP calls, VoIP to VMS calls and VoIP to VoIP calls — where transcoding is required.

The system provides 4 pre-activated VOCODER channels by default. To use these channels make sure you have installed atleast one NX DBM VOCODER64 module. If you require more channels, you can purchase the channel licenses according to your requirement.

For more information on Licenses — Matrix VARTA User License, IP Subscribers License and VOCODER Channel License, see “License Management”.

You may connect any Standard Phone or Extended IP Phones of Matrix as SIP Users.

Matrix VARTA WIN200, VARTA ADR100 and VARTA AMP100 can be registered as SIP Users, also offering the support for UC functionalities.

You may also connect/register[95] the following as SIP Extensions of the system:

- Connect SPARSH VP248, the Extended IP Phone. For instructions, see “Connecting SPARSH VP248 as Extended SIP Extension”.
- Connect SPARSH VP310, the Extended IP Phone. For instructions, see “Connecting SPARSH VP310 as Extended SIP Extension”.
- Connect SPARSH VP330, the Touch Screen Extended IP Phone. For instruction,“Connecting SPARSH VP330 as Extended SIP Extension”.
- Connect SPARSH VP510, the Premium IP Phone. For instruction, “Connecting SPARSH VP510 as Extended SIP Extension”.
- Connect SPARSH VP210, the Entry Level IP Phone. For instruction, “Connecting SPARSH VP210 as Extended SIP Extension”.
- Connect Extended SPARSH VP710, the Smart Video IP Phone. For instruction,“Connecting Extended SPARSH VP710 as Extended SIP Extension”.

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94. Calls between VoIP, Public and Private Networks may be subject to Regulation in your country. You may have to configure your system to allow or restrict call traffic between networks to comply with the telecom regulations of your country. To know more, read “Logical Partition”.

95. SARVAM UCS supports only IPv4 Addresses for registering SPARSH VP248, SPARSH VP310, SPARASH VP330, Extended SPARSH VP710, Mobile UC Clients as well as VARTA WIN200.

You can register following UC Clients as SIP Users of the system:

- Matrix VARTA WIN200, Unified Communication Client for Windows. For instruction, refer to the *MATRIX VARTA WIN200* User Guide.
- Matrix Mobile UC Clients, as given below:
  - Matrix VARTA AMP100, the Mobile UC Client for iPhones. For instruction, refer to the *Matrix VARTA AMP100* User Guide.
  - Matrix VARTA ADR100, the Mobile UC Client for Android Smartphones/Tablets. For instruction, refer to the *Matrix VARTA ADR100* User Guide.

Refer to “SARVAM UCS Features Supported in Terminals” to know the features supported in each client.

The SIP Users may be registered over ***WAN*** or over ***LAN*** according to your preference and your IP network installation scenario. Extended SIP Phones and UC Clients can be registered with SARVAM UCS,for details refer to the specific topic.

You can register the same SIP User from three different locations.

If you register the Extended IP Phone outside the Region/Country selected for SARVAM UCS, the time and Time Zone dependant features, such as Alarms, Reminders, Time Zone Display, of the phone at each location will operate according to the Real Time Clock of SARVAM UCS. Also, Access Codes and Emergency Numbers will work according to the Region/Country selected for SARVAM UCS.

- Connect the Extended IP Phone, or any Standard IP Phone to the LAN Switch.

- Register any SIP device (Extended IP phone/ Soft clients or Standard IP phone) on the public network as SIP Extension.

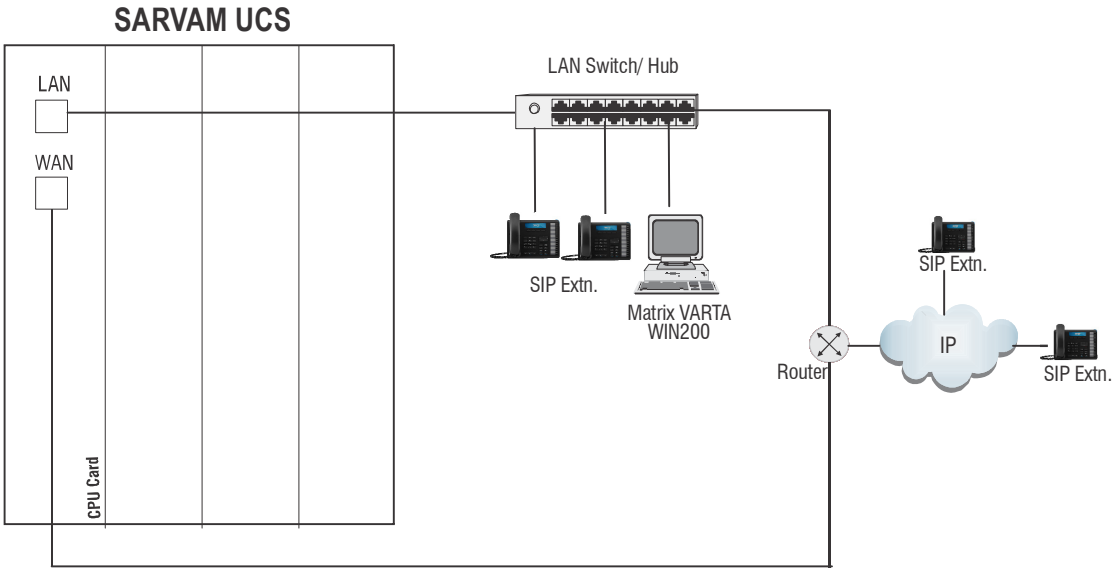
- When you register the Matrix Extended IP Phone with the system, the WAN/LAN port is used for Auto Configuration as well for Registration of the Extended IP Phones.

- When you register a SIP device other than the Matrix Extended IP Phone on the public network as SIP Extension, do the following:
  - In this SIP device configure the following:
    - the Registrar Server Address of SARVAM UCS
    - the Registrar Server Port
    - the SIP ID
    - Authentication ID and Password.

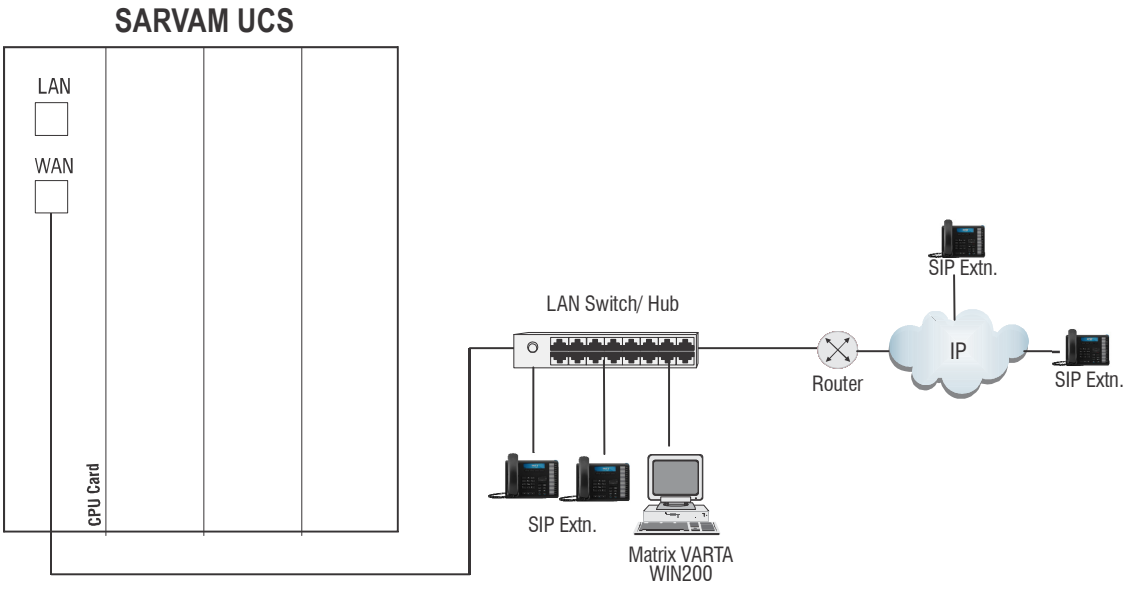
  - Configure Port Forwarding for the WAN Port of SARVAM UCS on the Router.

If the SARVAM UCS is connected to a ***Public Network***,

- Connect the Matrix VARTA WIN200, Extended IP Phone, or any Standard SIP device to the LAN Switch.
- Register any SIP device (Matrix VARTA UC Clients, Extended IP phone or Standard SIP phone) on the public network as SIP extension.



If the SARVAM UCS is connected to a ***Private Network (Behind the NAT)***,



- Connect Matrix VARTA WIN200, Extended IP Phones or Standard SIP Phones to the LAN Switch
- You may also register any SIP device (Matrix VARTA UC Clients, Extended IP Phone or Standard SIP phone) on the public network as SIP Extension.

  When you register the Matrix Extended IP Phone with SARVAM UCS, configure **Port Forwarding** for the **WAN port of the CPU Card** on the Router. The WAN Port is used for Auto Configuration of the Extended IP Phones.

Connecting SPARSH VP248 as Extended SIP Extension

You are recommended to complete the following steps before connecting the Matrix Extended IP Phone to SARVAM UCS:

- Decide the location of the Extended IP Phone, whether within the same network or outside, according to your installation scenario.

*If you want to use the **DHCP Server** for assigning IP Address to the Extended IP Phone, select **DHCP option 224** and **Data Type** as **'String'** and program the LAN or WAN IP Address /Domain Name of SARVAM UCS and SPARSH Port in the format **"IP_Address:Port"** in your DHCP Server as per your installation scenario.*

- Log in to *Jeeves*. For instructions, read the topic "Configuring SARVAM UCS".

- Assign SIP User ID (will work as an extension number) to the Extended IP Phone. For instructions on assigning SIP ID, see "Configuring SIP Extensions".

  For the SIP User ID you assigned to the Extended IP Phone, you must configure the necessary parameters in SARVAM UCS so that Extended IP Phone can register as a SIP Extension. For instructions, see the topic "Configuring SIP Extension Settings as per the Extended Phone Type" under *Configuring SIP Extensions*.

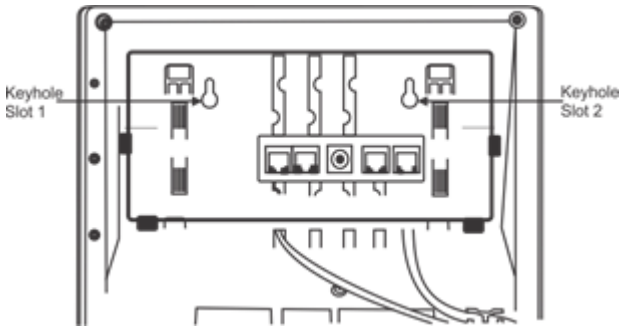
Now, follow the steps described below to install the Extended IP Phone. The instructions are common for all models of the SPARSH VP248. For the purpose of illustration, the premium model, SPARSH VP248P, has been used.

1. Unpack the SPARSH VP248 box and verify package contents.

2. Mount the phone on a desk or wall at a location convenient to you.

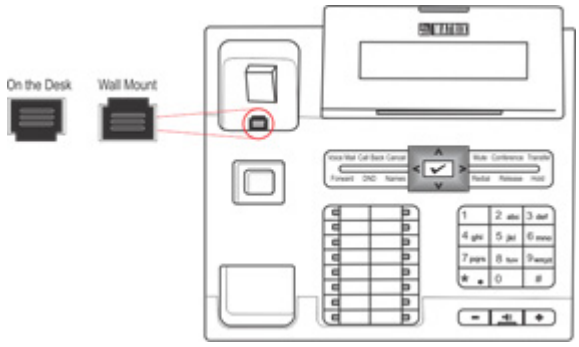
   - When mounting the phone on the wall,

     - Use the mounting template to drill holes of appropriate size and distance. Fix the screw grips in the holes you drilled.
     - Fix two screws in the holes on the wall, ensuring that they are aligned with the Keyhole Slots 1 and 2.
     - Use wall plugs, if required, to fix the screws. Leave the screw heads protruding from the wall to fit into the Keyholes.



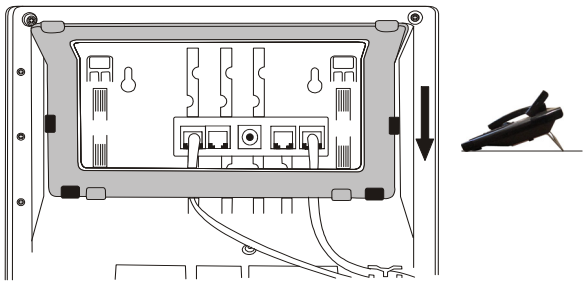
   - Now, mount the phone on the wall, with the screws fitting into the Keyhole slots.

- Reverse the handset wall mount tab to make sure the handset remains intact when you mount the phone. Push the handset wall mount tab upwards to remove it from the slot. Rotate it 180 degrees clockwise and push it downwards into the slot.

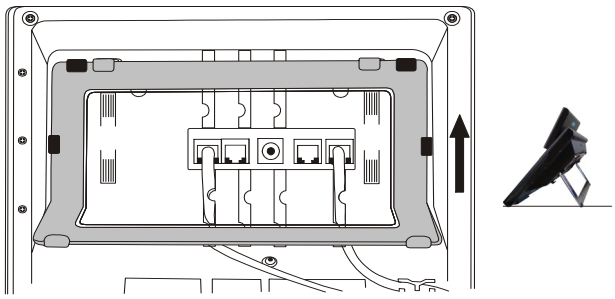


- When you mount the phone on a desk,
  - You can attach the Foot Stand in two ways as illustrated in the following.

Foot Stand attached at 30° Angle



Foot Stand attached at 50° Angle

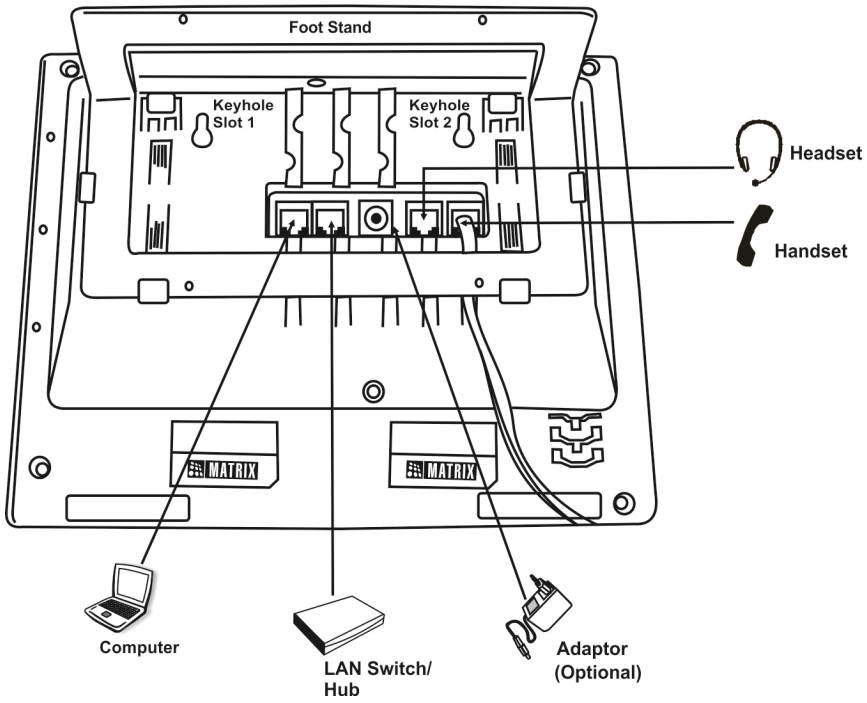


If you attach the Foot Stand at 50°, the phone will be placed in an almost upright position on your desk.

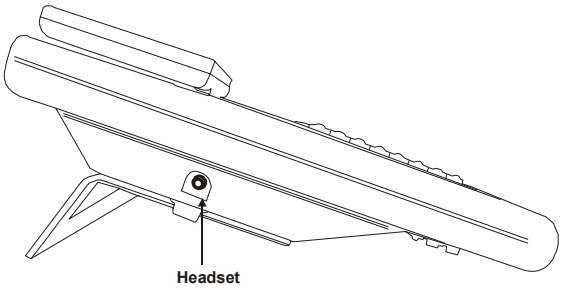
  - Decide which of these positions would work for you best and accordingly attach the Foot Stand.

3. Connect the Handset to the Phone body.
   - Plug the long straightened end of the phone cord into the handset jack at the bottom of the phone marked with the handset symbol.

- Plug the other (short straight) end of the phone cord into the jack at the bottom of the handset.

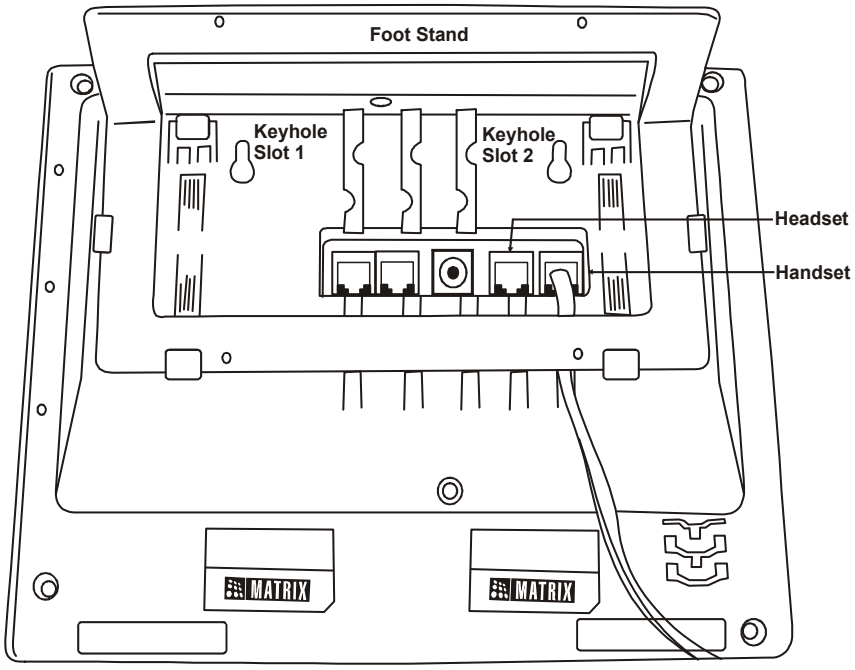


4. If you want to use a Headset (not supplied) with your phone, you may plug a headset with a 2.5 mm single connector into the headset jack headset jack with the symbol on the left side panel of the phone, as illustrated in the figure below.



OR

You may plug a headset with an RJ9 connector into the headset port at the bottom of the phone, marked with the symbol , as illustrated in the figure below.



Connect the LAN Port of SPARSH VP248 to the LAN Switch/Hub or a Router, according to your installation scenario.

5. To connect your phone to a computer on your desk, use an Ethernet cable (not supplied with this phone) to connect the PC Port of the phone to the LAN Port of the computer.

6. It is a PoE enabled phone and can be powered over Ethernet by connecting it to a PoE enabled LAN Switch (IEEE 802.3af Compliant). In this case you need not connect the Power Adapter.

   If you do not want to use PoE, plug in the connector of the Adapter into the power jack (DC Jack) at the bottom of the phone. Plug in the Power Adapter into a power outlet.

If both the power options, that is, PoE as well as Power Adapter are available to the phone, then the phone will derive power from the PoE enabled LAN Switch.

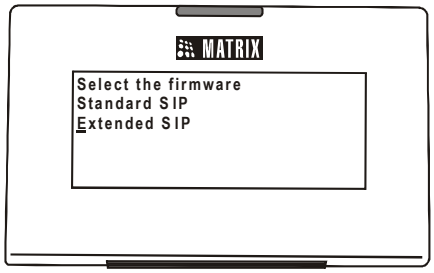
The IP phone should be used with Matrix original power adapter (5V/2A) only. The use of any third-party power adapter may cause damage to the phone.

7. Switch ON power supply.

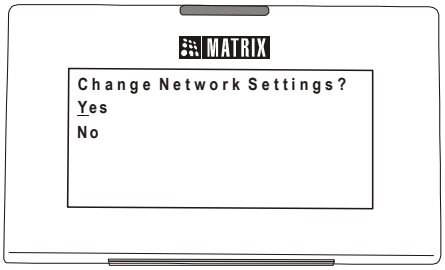
   When you power the phone, the boot process will be initiated in the following sequence.

   - All keys with LED, including the Speaker key, and the Ringer LED, will glow.
   - The LCD display will light up and booting message appears.
   - As soon as the ‘Loading...’ message appears on the phone display, press **#** key.

- Select the firmware **Extended - IP Phone**. Move the cursor by pressing the DOWN navigation key **V**.
- When the cursor is placed under the Extended IP Phone, press Enter key.



- The phone will start loading the Extended IP Phone Firmware. It will display current firmware being loaded.
- After loading the firmware, the phone will prompt you to change Network settings.



- Wait for a few seconds.

If you want to change the Network Settings or Server Settings, press the Enter key. Detailed instructions for changing the Network Settings of the phone are provided at the end of this topic. See "Network Settings"at the end of this topic.

- The phone makes DHCP Discovery and fetches its IP Address and Server Address from the DHCP Server.

  On getting the IP Address and Server Address, the phone initiates Auto Configuration to download the configuration files from SARVAM UCS.

- As the phone downloads the configuration files, the file names will appear one by one.
- On successful download of all configuration files, the phone attempts to register with SARVAM UCS.
- On successful registration, the phone will display the current day, date and time, the extension number and name assigned to the Extended IP Phone.

Network Settings

You can change the network settings of the Extended IP Phone by accessing the Local Menu of the phone. To move the cursor and scroll through the menu and submenu options, use the following touch sense navigation keys on your phone.

- The Enter key ✓ to make a selection or to complete an action.
- The Up key ^ to move up the Menu.

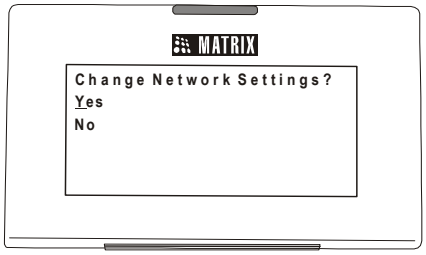
- The Down key ⋁ move down the Menu.
- The Forward key > move the cursor one character.
- The Back key < to move the cursor one character and to return from the submenu to the main menu.

The cursor is a non-blinking underscore that appears under the first letter of the first option in the menu. To make a selection in the menu, you must move the cursor in the desired direction using the Up, Down, Forward and Back key. When the cursor is at the desired position, press Enter key to make a selection.

Accessing Network Settings

You can access the Network Settings of the Extended IP Phone in any of the following stages:

1. During start-up, when the phone prompts you to change the network settings after loading the firmware.

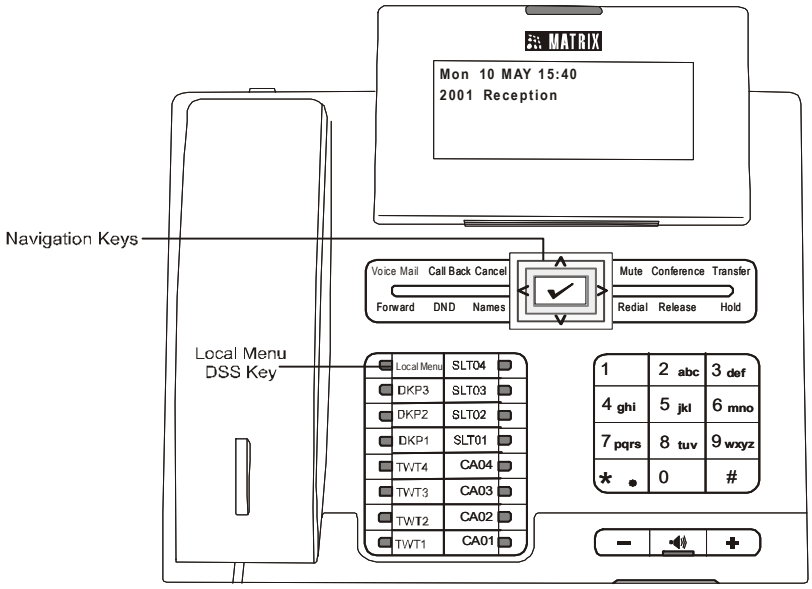


   You must press the Enter Key to select Yes and access network settings.

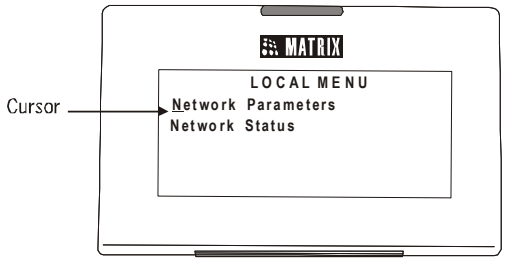
2. When the phone is making Network discovery, downloading configuration files, attempting registration.

   You must press the Enter Key ✓ to access network settings,

3. When the phone is in idle state. You must press the DSS key assigned to ‘Local Menu’.



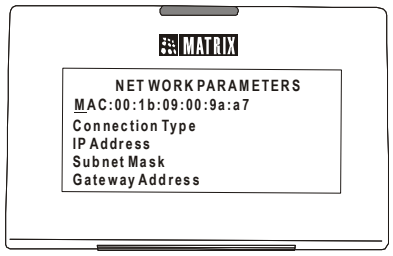
4. When you press the Local Menu DSS Key (in idle state) or when you press the Enter key during any process, the Local Menu appears on your phone display.



You can configure Network Parameters and view Network status from the Local Menu.

Configuring Network Parameters

- In the Local Menu of the phone, select Network Parameters by pressing the Enter Key.
- The Network Parameters submenu appears.



- Use the Down/Up key to reach the desired network parameter and press Enter key to select and change the settings.
- You can configure all network parameters described below, except the MAC Address.

- *To enter a dot in the editable fields — IP Address, Subnet Mask, Gateway Address, DNS Address, Server Address — press * (Star) key.*
- *If you want to clear a single digit/character, move the cursor to the right of the digit/character you wish to clear. Now press the Cancel key. The digit to the left of the cursor will be deleted. If the cursor is to the extreme left and you press the Cancel key, you will go to the previous menu.*

Connection Type

- Select the Connection Type as DHCP, PPPoE or Static according to the IP Addressing scheme of your network.

  If you select DHCP or PPPoE, the phone will be assigned IP Address, Subnet Mask and Gateway Address, DNS Address Server Address, automatically by the DHCP/PPPoE server.

  For PPPoE Connection Type, you must configure the PPPoE User ID and Password provided by the Internet Service Provider.

  If you select Static, you must assign the IP Address, Subnet Mask and Gateway Address to the phone.

IP Address

- If you select Static as Connection Type, enter the static IP Address to be assigned to the phone.

  Enter the desired Static IP Address by pressing the digit keys.

  To enter the dot/period in the IP Address, press the Star '*' key.

Subnet Mask

- If you select Static as Connection Type, enter the Subnet Mask to be applied on the phone by pressing the digit keys.

  To enter the dot/period in the IP Address, press the Star '*' key.

Gateway Address

- If you select Static as Connection Type, enter the Gateway Address here. This is the IP Address of the LAN Port of the Router.

DNS Server

- If you select Static as Connection Type, select the DNS Server option **Static** and configure the DNS Address.

- If you select DHCP or PPPoE as Connection Type and your Internet Service Provider provides DNS Address, select the DNS Server option **Automatic**. However, if your Internet Service Provider does not provide DNS Address, select **Static** and configure the DNS Address.

DNS Address

- If you select DNS Server as Static, enter the DNS Address here.

  To enter dot/period in the IP Address, press the Star '*' key.

DNS Domain Name

- If you select DNS Server as Static, enter the DNS Domain Name here. DNS Domain Name is optional.

PPPoE User ID

- If you have selected PPPoE as Connection Type, you must enter the User ID provided to you by your Internet Service Provider.

PPPoE Password

- This is the password provided by your Internet Service Provider for the PPPoE User ID. If you have selected PPPoE as Connection Type, you must enter the password provided by your Internet Service provider here.

PPPoE Service Name

- If your Internet Service Provider has provided a Service Name, enter the Service Name here. If your Internet Service Provider has not provided a Service Name, do not configure this parameter.

Server Address

- The system works as the Auto Configuration Server for the phone. Enter the LAN or WAN IP Address/ Domain Name of SARVAM UCS here. Default: blank. The phone sends the request for configuration files to this Server Address.

  If you have selected DHCP as Connection Type, the phone will get the Server Address and Port automatically from the DHCP Server. For this, use ***DHCP option 224*** and ***Data Type*** as '***String***' to provide Server Address and Port from the DHCP Server.

  For PPPoE and Static Connection Types, you need to enter the Server Address.

Server Port

- Enter the SPARSH Port of SARVAM UCS here. The phone sends the request for configuration files to this port.

  Valid range of the port is: 80 or 1025–65535. Default: 80.

VLAN Setting

If your phone is connected to a virtual LAN, you need to configure VLAN Settings.

To enable the VLAN switch to correctly route packets generated by the phone and the computers (on the LAN) to each other, the packets must be tagged with a VLAN header.

The VLAN header consists of the VLAN ID (12-bit) and Class of Service (CoS, 3-bit) for prioritization of traffic[96].

The meaning of CoS bits with respect to traffic type is as follows:

CoS	Traffic Type
0	Best Effort
1	Background
2	Spare
3	Excellent Effort
4	Controlled Load
5	Video
6	Voice
7	Network Control

- Select **Phone VLAN/COS** to add VLAN header to the packets generated by the phone, and add VLAN header to the packets relayed from the PC to its LAN port (packets generated by the PC connected to its PC port).

  - To configure Phone VLAN/COS, select **Enable?**. The VLAN ID will be tagged on all packets generated by the phone (SIP, RTP, DNS, ARP, etc.). Default: Disabled.

96. The IEEE 802.1P standard allows Layer2 switches to prioritize the traffic, thus providing Quality of Service (QoS), that is, better handling of data that pass over a network, thereby resulting in greater reliability and quality. Quality of Service (QoS) on Layer2 is referred to as Class of Service (CoS) which is defined by IEEE 802.1P.

    - Select **VLAN ID** and enter the VLAN ID that you have assigned to the VLAN in which the IP Phones are connected. Valid range: 0-4094. Default: 1.
    - Select **SIP CoS** and define the CoS (priority) bits in all SIP packets. Valid range: 0-7. Default: 3
    - Select **RTP CoS** and define the CoS (priority) bits in all RTP packets. Valid range: 0-7. Default: 6.
- Select **PC/VLAN CoS** to add VLAN header to all packets entering the PC Port and leaving the LAN port of the phone. Default: Disabled.
    - To configure PC VLAN/COS, select **Enable?**.
    - Select **VLAN ID** and enter the same ID as you have assigned to the VLAN in which the computers are connected. Valid range: 0-4094. Default: 1.
    - Select **CoS** and define the Layer 2 CoS (priority) bits. Valid range: 0-7. Default: 0.

PCAP

To capture packets sent and received from and by the phone for monitoring and troubleshooting, you can enable PCAP on the phone. The phone captures up to 1 MB of packets. For more information and for instructions on how to use PCAP Trace on the phone, see “Using PCAP Trace for Matrix SPARSH VP248 Extended IP Phone”, under *PCAP Trace*.

When you change the Network Settings, the phone will restart.

MAC Cloning

If you require cloning of the MAC Address, configure the following:

- Select **Enable?**.
- In **Enter Clone MAC Address**, enter the address you wish to clone.

802.1x Authentication

If you want to restrict unauthorized clients from connecting to your LAN, you need to enable 802.1x Authentication. Using EAP MD5 protocol the PC connected to the LAN port of the IP Phone is first authenticated and then it gets connected to LAN.

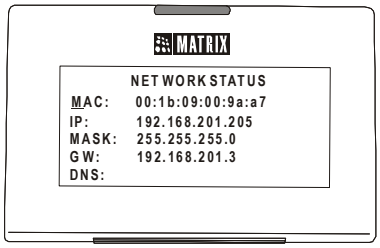
You need to configure the following 802.1x Authentication parameters:

- Select **Enable?**.
- Enter the 802.1x Authentication **Identity** provided by you network administrator.
- Enter the 802.1x Authentication **MD5 Password** associated with identity provided by your network administrator.

Viewing Network Status

- In the Local Menu of the phone, place the cursor on Network Status and press the Enter key.

- The Network Status submenu appears.



Use the Down/Up key to view the status of the various network parameters. The status of the following parameters appear on your display as you scroll.

- **MAC:** This is the MAC Address of the phone.
- **IP:** The current IP Address assigned to the phone.
- **MASK:** The current Subnet mask assigned to the phone.
- **GW:** The current Gateway IP Address assigned to the phone.
- **DNS:** The Domain Name Server address assigned to the phone.
- **S. ADD:** The LAN or WAN IP Address / Domain Name of the SARVAM UCS.
- **S. PORT:** The SPARSH Port SARVAM UCS.
- **DOMAIN:** The Domain Name assigned to the phone.
- **802.1x Authentication:** The 802.1x authentication status is displayed—Success, Failure, Authenticating or Disabled.
- **FIRM:** The version of the current Firmware of the phone.
- **UBOOT**: The UBOOT release date.
- **KERNEL**: The KERNEL release date.

Connecting SPARSH VP310 as Extended SIP Extension

You are recommended to complete the following steps before connecting the Matrix Extended IP Phone to SARVAM UCS:

- Decide the location of the Extended IP Phone, whether within the same network or outside, according to your installation scenario.

*If you want to use the **DHCP Server** for assigning IP Address to the Extended IP Phone, select **DHCP option 224** and **Data Type** as '**String**' and program the LAN or WAN IP Address /Domain Name of SARVAM UCS and SPARSH Port in the format "**IP_Address:Port"** in your DHCP Server as per your installation scenario.*

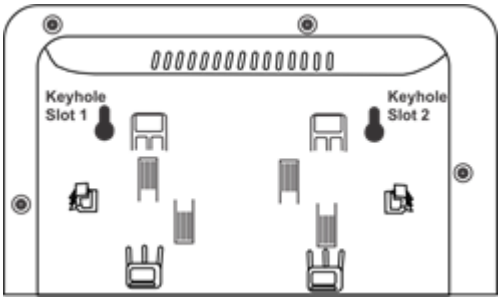
- Log in to *Jeeves*. For instructions, read the topic "Configuring SARVAM UCS".

- Assign an extension number (**SIP ID**) to the Extended IP Phone. For instructions on assigning SIP ID, see "Configuring SIP Extensions".

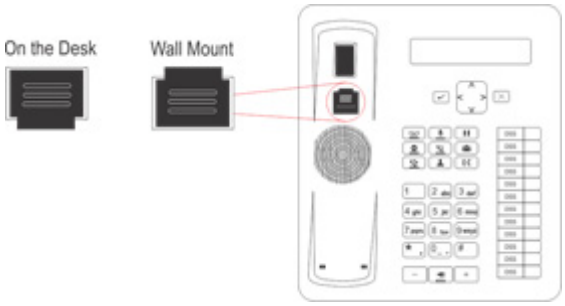
  For the SIP extension number you assigned to the Extended IP Phone, you must configure the necessary parameters in SARVAM UCS so that Extended IP Phone can register as a SIP Extension. For instructions, see the topic "Configuring SIP Extension Settings as per the Extended Phone Type" under *Configuring SIP Extensions*.

Now, follow the steps described below to install the Extended IP Phone.

1. Unpack the SPARSH VP310 box and verify package contents.
2. You can mount the phone on a wall or on the desk.
3. When you mount SPARSH VP310 on a wall,
   - Use the mounting template to drill holes of appropriate size and distance.
   - Fix the screw grips in the holes you drilled.
   - Fix two screws in the holes on the wall, ensuring that they are aligned with the Keyhole Slots 1 and 2 of SPARSH VP310. The screws should protrude from the wall to fit into the Keyhole Slots.

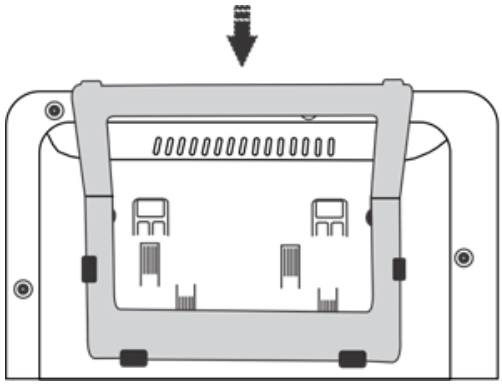


   - Now, mount the phone with the screws fitting into the Keyhole Slot.
   - Reverse the handset wall mount tab to make sure the handset remains intact when you mount the phone. Push the handset wall mount tab upwards to remove it from the slot. Rotate it 180 degrees clockwise and push it downwards into the slot.



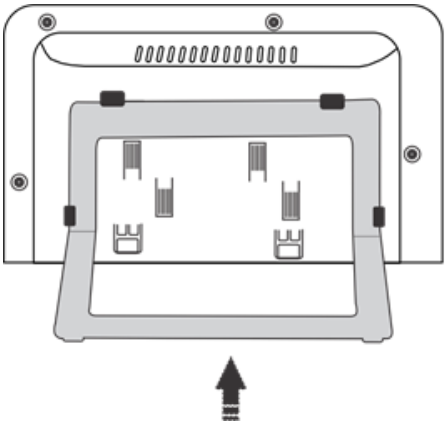
- When you mount the phone on a desk,
  - You can attach the Foot Stand in two ways as illustrated in the following.

Foot Stand attached at 35° Angle



Stand attached at
35 degree angle

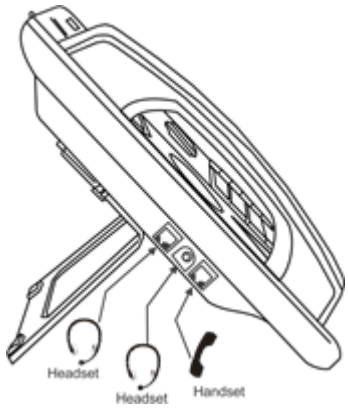
Foot Stand attached at 50° Angle



Stand attached at
50 degree angle

If you attach the Foot Stand at 50°, the phone will be placed in an almost upright position on your desk.

- Decide which of these positions would work for you best and accordingly attach the Foot Stand.



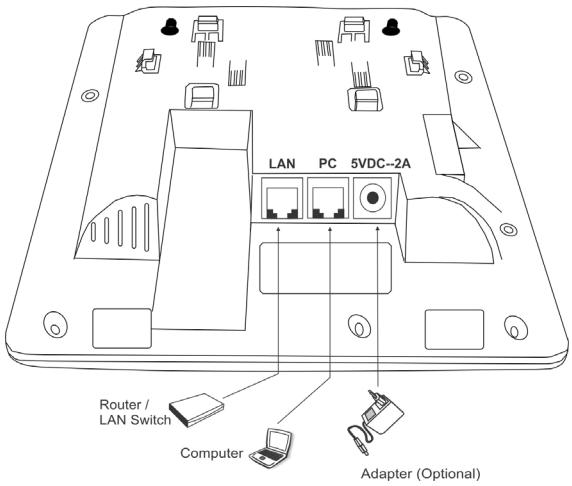
4. Connect the Handset to the Phone body.

- Plug the long straightened end of the phone cord into the handset jack on the left side panel of the phone marked with the handset symbol .
- Plug the other (short straight) end of the phone cord into the jack at the bottom of the handset.

5. If you want to use a Headset (not supplied) with your phone, you may plug a headset with a 3.5 mm single connector into the headset jack headset jack with the symbol on the left side panel of the phone, as illustrated in the figure above.

   OR

   You may also plug in a headset with RJ9 connector into the headset port on the left side panel of the phone, marked with the symbol .



6. Connect the LAN Port of SPARSH VP310 to the LAN Switch/Hub or a Router, according to your installation scenario.

7. To connect your phone to a computer on your desk, use an Ethernet cable (not supplied with this phone) to connect the PC Port of the phone to the LAN Port of the computer.

8. It is a PoE enabled phone and can be powered over Ethernet by connecting it to a PoE enabled LAN Switch (IEEE 802.3af Compliant). In this case you need not connect the Power Adapter.

   If you do not want to use PoE, plug in the connector of the Adapter into the power jack (DC Jack) with the label 5VDC-2A at the bottom of the phone. Plug in the Power Adapter into a power outlet.

If both the power options, that is, PoE as well as Power Adapter are available to the phone, then the phone will derive power from the PoE enabled LAN Switch.

The IP phone should be used with Matrix original power adapter (5V/2A) only. The use of any third-party power adapter may cause damage to the phone.

9. Switch ON power supply.

   When you power the phone, the boot process will be initiated in the following sequence.

   - All keys with LED, including the Speaker key, and the Ringer LED, will glow.
   - The LCD display will light up and the booting message appears.
   - Then the ‘Loading...’ message appears on the phone display.

- The phone will start loading the Extended IP Phone Firmware. It will display current firmware being loaded.
- After loading the firmware, the phone will prompt you to change Network settings.

- Wait for a few seconds.

If you want to change the Network Settings or Server Settings, press the Enter key. Detailed instructions for changing the Network Settings of the phone are provided at the end of this topic. See "Network Settings" at the end of this topic.

- The phone makes DHCP Discovery and fetches its IP Address and Server Address from the DHCP Server.

  On getting the IP Address and Server Address, the phone initiates Auto Configuration to download the configuration files from SARVAM UCS.
- As the phone downloads the configuration files, the file names will appear one by one.
- On successful download of all configuration files, the phone attempts to register with SARVAM UCS.
- On successful registration, the phone will display the current day, date and time, the extension number and name assigned to the Extended IP Phone.

Network Settings

You can change the network settings of the Extended IP Phone. Press the Down key ⋁ when the phone is in idle state. To move the cursor and scroll through the menu and submenu options, use the following navigation keys on your phone.

- The Enter key ✓ to make a selection or to complete an action.
- The Up key ⋀ to move up the Menu.
- The Down key ⋁ move down the Menu.
- The Forward key > move the cursor one character.
- The Back key < to move the cursor one character and to return from the submenu to the main menu.
- The Cancel key ✕ to exit a menu.

The cursor is a non-blinking underscore that appears under the first letter of the first option in the menu. To make a selection in the menu, you must move the cursor in the desired direction using the Up, Down, Forward and Back key. When the cursor is at the desired position, press Enter key to make a selection.

Accessing Network Settings

You can access the Network Settings of the Extended IP Phone in any of the following stages:

1. During start-up, when the phone prompts you to change the network settings after loading the firmware.

   You must press the Enter key ✓ to select Yes and access network settings.

2. When the phone is making Network discovery, downloading configuration files, attempting registration.

   You must press the Down key ⋁ to access network settings.

3. When the phone is in idle state. You must press the Down key ⋁ to access the Network Settings.

Configuring Network Parameters

- When the phone is in idle state. You must press the Down key ⋁ to access the Network Settings.
- Press Enter key to select Network Parameters.
- The Network Parameters submenu appears.
- Use the Down/Up key to reach the desired network parameter and press Enter key to select and change the settings.
- You can configure all network parameters described below, except the MAC Address.

- *To enter a dot in the editable fields — IP Address, Subnet Mask, Gateway Address, DNS Address, Server Address — press * (Star) key.*
- *If you want to clear a single digit/character, move the cursor to the right of the digit/character you wish to clear. Now press the Cancel key. The digit to the left of the cursor will be deleted. If the cursor is to the extreme left and you press the Cancel key, you will go to the previous menu.*

Connection Type

- Select the Connection Type as DHCP, PPPoE or Static according to the IP Addressing scheme of your network.

  If you select DHCP or PPPoE, the phone will be assigned IP Address, Subnet Mask and Gateway Address, DNS Address, Server Address, automatically by the DHCP/PPPoE server.

  For PPPoE Connection Type, you must configure the PPPoE User ID and Password provided by the Internet Service Provider.

  If you select Static, you must assign the IP Address, Subnet Mask and Gateway Address to the phone.

IP Address

- If you select Static as Connection Type, enter the static IP Address to be assigned to the phone. Enter the desired Static IP Address by pressing the digit keys.

  To enter the dot/period in the IP Address, press the Star '*' key.

Subnet Mask

- If you select Static as Connection Type, enter the Subnet Mask to be applied on the phone by pressing the digit keys.

  To enter the dot/period in the IP Address, press the Star '*' key.

Gateway Address

- If you select Static as Connection Type, enter the Gateway Address here. This is the IP Address of the LAN Port of the Router.

DNS Server

- If you select Static as Connection Type, select the DNS Server option **Static** and configure the DNS Address.

- If you select DHCP or PPPoE as Connection Type and your Internet Service Provider provides DNS Address, select the DNS Server option **Automatic**. However, if your Internet Service Provider does not provide DNS Address, select **Static** and configure the DNS Address.

DNS Address

- If you select DNS Server as Static, enter the DNS Address here.
  To enter dot/period in the IP Address, press the Star '*' key.

DNS Domain Name

- If you select DNS Server as Static, enter the DNS Domain Name here. DNS Domain Name is optional.

PPPoE User ID

- If you have selected PPPoE as Connection Type, you must enter the User ID provided to you by your Internet Service Provider.

PPPoE Password

- This is the password provided by your Internet Service Provider for the PPPoE User ID. If you have selected PPPoE as Connection Type, you must enter the password provided by your Internet Service provider here.

PPPoE Service Name

- If your Internet Service Provider has provided a Service Name, enter the Service Name here. If your Internet Service Provider has not provided a Service Name, do not configure this parameter.

Primary Server Address

- The system works as the Auto Configuration Server for the phone. Enter the LAN or WAN IP Address/ Domain Name of SARVAM UCS here. Default: blank. The phone sends the request for configuration files to this Server Address.

If you have selected DHCP as Connection Type, the phone will get the Primary Server Address and Port automatically from the DHCP Server. For this, use ***DHCP option 224*** and ***Data Type*** as '***String***' to provide Server Address and Port from the DHCP Server.

For PPPoE and Static Connection Types, you need to enter the Server Address.

Primary Server Port

- Enter the SPARSH Port of SARVAM UCS here. The phone sends the request for configuration files to this port.

  Valid range of the port is: 80 or 1025–65535. Default: 80.

Secondary Server Address

- If required, you can also configure the Secondary Sever Address as a fallback option. If the registration with the Primary Server fails the phone will send the registration and configuration requests to the Secondary Server Address. Speech-cut or unclear speech may be observed during on-going mature calls.

Secondary Server Port

- Enter the Secondary Server Port. The phone sends the request for configuration files to this port if the Primary Server fails.

  Valid range of the port is: 80 or 1025–65535. Default: 80.

VLAN Setting

If your phone is connected to a virtual LAN, you need to configure VLAN Settings.

To enable the VLAN switch to correctly route packets generated by the phone and the computers (on the LAN) to each other, the packets must be tagged with a VLAN header.

The VLAN header consists of the VLAN ID (12-bit) and Class of Service (CoS, 3-bit) for prioritization of traffic[97].

The meaning of CoS bits with respect to traffic type is as follows:

CoS	Traffic Type
0	Best Effort
1	Background
2	Spare
3	Excellent Effort
4	Controlled Load
5	Video
6	Voice
7	Network Control

97. The IEEE 802.1P standard allows Layer2 switches to prioritize the traffic, thus providing Quality of Service (QoS), that is, better handling of data that pass over a network, thereby resulting in greater reliability and quality. Quality of Service (QoS) on Layer2 is referred to as Class of Service (CoS) which is defined by IEEE 802.1P.

- Select **Phone VLAN/COS** to add VLAN header to the packets generated by the phone, and add VLAN header to the packets relayed from the PC to its LAN port (packets generated by the PC connected to its PC port).
    - To configure Phone VLAN/COS, select **Enable?**. The VLAN ID will be tagged on all packets generated by the phone (SIP, RTP, DNS, ARP, etc.). Default: Disabled.
    - Select **VLAN ID** and enter the VLAN ID that you have assigned to the VLAN in which the IP Phones are connected. Valid range: 0-4094. Default: 1.
    - Select **SIP CoS** and define the CoS (priority) bits in all SIP packets. Valid range: 0-7. Default: 3
    - Select **RTP CoS** and define the CoS (priority) bits in all RTP packets. Valid range: 0-7. Default: 6.
- Select **PC/VLAN CoS** to add VLAN header to all packets entering the PC Port and leaving the LAN port of the phone. Default: Disabled.
    - To configure PC VLAN/COS, select **Enable?**.
    - Select **VLAN ID** and enter the same ID as you have assigned to the VLAN in which the computers are connected. Valid range: 0-4094. Default: 1.
    - Select **CoS** and define the Layer 2 CoS (priority) bits. Valid range: 0-7. Default: 0.

PCAP

To capture packets sent and received from and by the phone for monitoring and troubleshooting, you can enable PCAP on the phone. The phone captures up to 1 MB of packets. For more information and for instructions on how to use PCAP Trace on the phone, see “Using PCAP Trace for Matrix SPARSH VP310 Matrix Extended IP Phone”, under *PCAP Trace*.

MAC Cloning

If you require cloning of the MAC Address, configure the following:

- Select **Enable?** and press the Enter key. Select Yes to enable.
- In **Enter Clone MAC Address**, enter the address you wish to clone.

802.1x Authentication

If you want to restrict unauthorized clients from connecting to your LAN, you need to enable 802.1x Authentication. Using EAP MD5 protocol the PC connected to the LAN port of the IP Phone is first authenticated and then it gets connected to LAN.

You need to configure the following 802.1x Authentication parameters:

- Select **Enable?** and press the Enter key. Select Yes to enable.
- Enter the 802.1x Authentication **Identity** provided by you network administrator.
- Enter the 802.1x Authentication **MD5 Password** associated with identity provided by your network administrator.

When you change the Network Settings, the phone will restart.

Viewing Network Status

- When the phone is in idle state. You must press the Down key ⋁ to access the Network Settings.
- Again press Down key ⋁ to select Network Status and press the Enter key ✓.

  Use the Down/Up key to view the status of the various network parameters. The status of the following parameters appear on your display as you scroll.

  - **MAC:** This is the MAC Address of the phone.
  - **IP:** The current IP Address assigned to the phone.
  - **MASK:** The current Subnet mask assigned to the phone.
  - **GW:** The current Gateway IP Address assigned to the phone.
  - **DNS:** The Domain Name Server address assigned to the phone.
  - **Active Server:** This displays the Server that is active — Primary, Secondary — with which the phone is currently registered.
  - **S. ADD:** This displays the IP address of the Active Server. It may be the LAN or WAN IP Address / Domain Name of the SARVAM UCS or the Secondary Server IP Address (if configured) or any Fallback Server.
  - **S. PORT:** This displays the port of the Active Server. It may be the SPARSH Port of SARVAM UCS or the Secondary Server Port (if configured) or the Fallback Server Port.
  - **DOMAIN:** The Domain Name assigned to the phone.
  - **802.1x Authentication:** The 802.1x authentication status is displayed—Success, Failure, Authenticating or Disabled.
  - **FIRM:** The version of the current Firmware of the phone.
  - **UBOOT**: The UBOOT release date.
  - **KERNEL**: The KERNEL release date.

Connecting SPARSH VP330 as Extended SIP Extension

You are recommended to complete the following steps before connecting the Matrix SPARSH VP330 to SARVAM UCS:

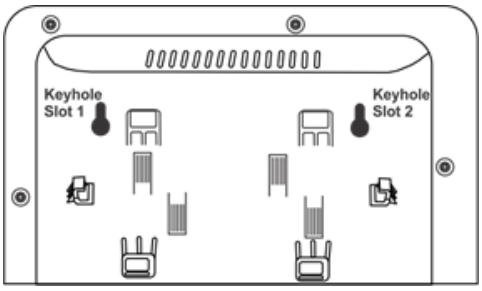
- Decide the location where you want to place SPARSH VP330 within your LAN.
- By Default, in SPARSH VP330, the Connection Type selected is DHCP.
- If you want to use the **DHCP Server** for assigning IP Address to the Extended IP Phone, select **DHCP option 224** and **Data Type** as '**String**' and program the LAN or WAN IP Address /Domain Name of

SARVAM UCS and SPARSH Port in the format "**IP_Address:Port"** in your LAN DHCP Server as per your installation scenario.

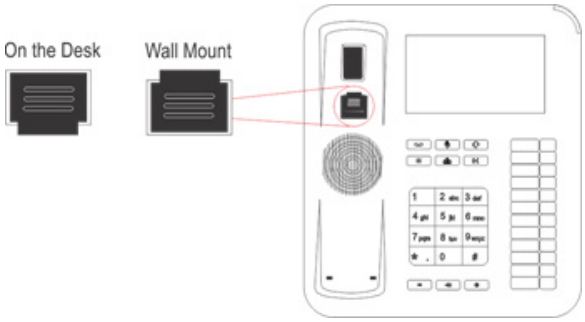
- Log in to *Jeeves*. For instructions, read the topic "Configuring SARVAM UCS".
- You must configure the necessary parameters in SARVAM UCS so that SPARSH VP330 can register as a SIP Extension. For instructions, see "Configuring Matrix SPARSH VP330".

Now, follow the steps described below to install SPARSH VP330.

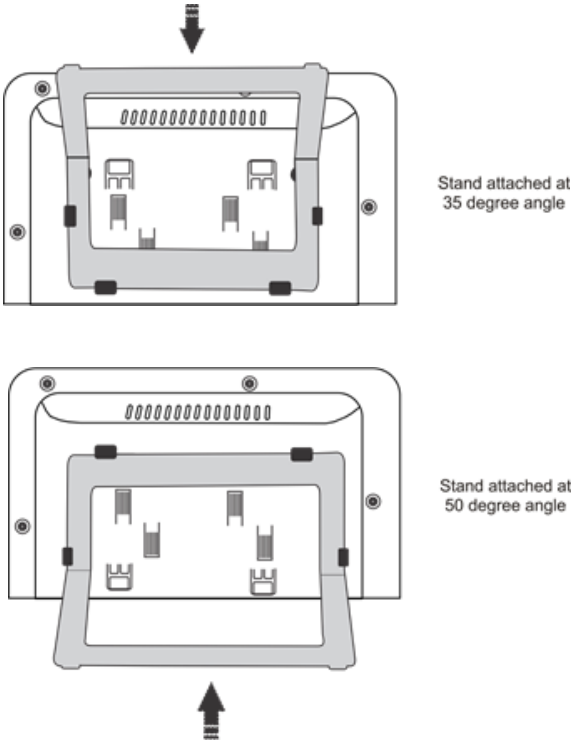
1. Unpack the SPARSH VP330 box and verify package contents.
2. Mount the phone on a desk or wall at a location convenient to you.
    - When mounting the phone on the wall,
        - Use the mounting template to drill holes of appropriate size and distance. Fix the screw grips in the holes you drilled.
        - Fix two screws in the holes on the wall, ensuring that they are aligned with the Keyhole Slots 1 and 2.
        - Use wall plugs, if required, to fix the screws. Leave the screw heads protruding from the wall to fit into the Keyholes.



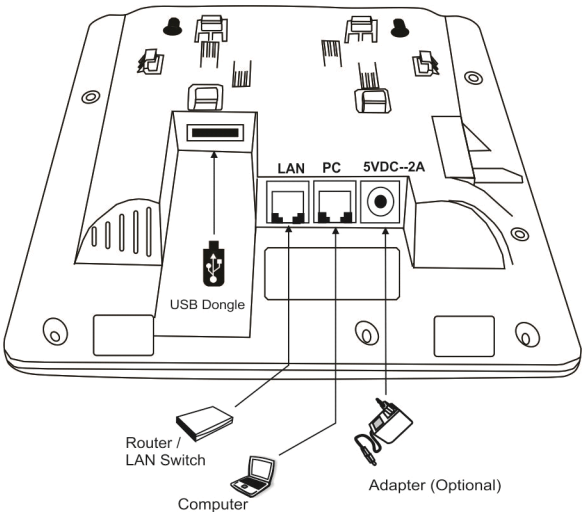
- Now, mount the phone on the wall, with the screws fitting into the Keyhole slots.
- Reverse the handset wall mount tab to make sure the handset remains intact when you mount the phone. Push the handset wall mount tab upwards to remove it from the slot. Rotate it 180 degrees clockwise and push it downwards into the slot.



3. When you mount the phone on a desk, you can attach the Foot Stand in two ways at **35° Angle** or at **50° Angle**.



- If you attach the Foot Stand at 50°, the phone will be placed in an almost upright position on your desk.
- Decide which of these positions would work for you best and accordingly attach the Foot Stand.



4. Connect the Handset to the Phone body.

- Plug the long straightened end of the phone cord into the handset jack on the left side panel of the phone marked with the handset symbol.

- Plug the other (short straight) end of the phone cord into the jack at the bottom of the handset.

5. If you want to use a Headset (not supplied) with your phone, you may plug a headset with a 3.5 mm single connector into the headset jack headset jack with the symbol on the left side panel of the phone.

   OR

   You may plug a headset with an RJ9 connector into the headset port on the side panel of the phone, marked with the symbol .

6. Connect the LAN Port of SPARSH VP330 to the IP Network — A Router or LAN Switch — using the Ethernet Cable.

   OR
   Connect the Wi-Fi USB Adapter into the USB port of the phone.

You can purchase the Wi-Fi USB Adapter from Matrix as an optional peripheral device to support wireless network.

7. To connect your phone to a computer on your desk, use an Ethernet cable (not supplied with this phone) to connect the PC Port at the bottom of the phone to the LAN Port of the computer.

8. It is a PoE enabled phone and can be powered over Ethernet by connecting it to a PoE enabled LAN Switch (IEEE 802.3af Compliant). In this case you need not connect the Power Adapter.

   If you do not want to use PoE, plug in the connector of the Adapter into the power jack (DC Jack) with the label 5VDC-2A at the bottom of the phone. Plug in the Power Adapter into a power outlet.

If both the power options, that is, PoE as well as Power Adapter are available to the phone, then the phone will derive power from the PoE enabled LAN Switch.

The IP phone should be used with Matrix original power adapter (5V/2A) only. The use of any third-party power adapter may cause damage to the phone.

9. Switch ON power supply.

   When you power the phone, the boot process will be initiated in the following sequence.

   - All keys with LED, including the Speaker key, and the Ringer LED, will glow.
   - The LCD display will light up and booting message appears.
   - While loading the application then the loading message appears on the phone display,
   - The phone makes DHCP Discovery and fetches its IP Address and Server Address from the DHCP Server.

If you want to change the Network Settings/Server Settings or want to use Wi-Fi for connectivity, press Settings .

Refer to the SPARSH VP330 User Guide, for detailed instructions:

- *To change the Network Settings of the phone and configure the network parameters.*
- *To use Wi-Fi for connectivity and configure its parameters.*

- On getting the IP Address and Server Address, the phone initiates Auto Configuration to download the configuration files from SARVAM UCS.
- On successful download of all configuration files, the phone attempts to register with SARVAM UCS.
- On successful registration, the Home screen appears.

The phone will register successfully, only if the SIP Extension parameters in SARVAM UCS have been correctly configured as per your installation scenario.

Refer to the ***SPARSH VP330 User Guide*** to know more.

Connecting SPARSH VP510 as Extended SIP Extension

You are recommended to complete the following steps before connecting the Matrix Extended IP Phone to the system when used with SARVAM UCS application:

- Decide the location of the Extended IP Phone, whether within the same network or outside, according to your installation scenario.

*If you want to use the **DHCP Server** for assigning IP Address to the Extended IP Phone, select **DHCP option 224** and **Data Type** as '**String**' and program the LAN or WAN Port IP Address /Domain Name and SPARSH Port in the format "**IP_Address:Port**" in your DHCP Server as per your installation scenario.*
•Log in to *Jeeves*. For instructions, read the topic "Configuring SARVAM UCS".

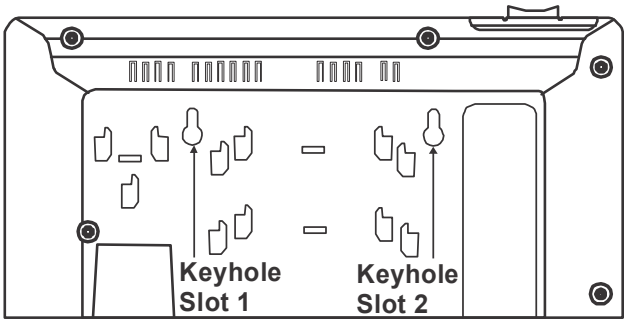
- Assign an extension number (**SIP ID**) to the Extended IP Phone. For instructions on assigning SIP ID, see "Configuring SIP Extensions".

  For the SIP extension number you assigned to the Extended IP Phone, you must configure the necessary parameters in SARVAM UCS so that Extended IP Phone can register as a SIP Extension. For instructions, see the topic "Configuring SIP Extension Settings as per the Extended Phone Type" under *Configuring SIP Extensions*.

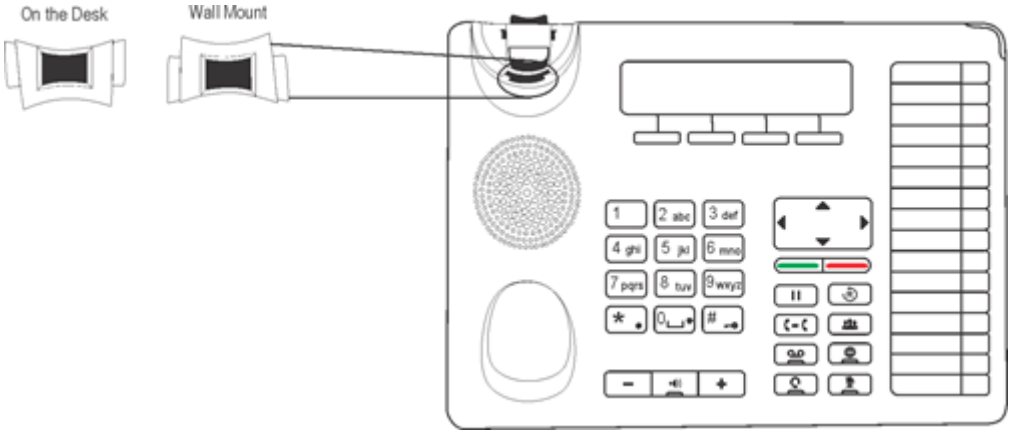
Now, follow the steps described below to install the Extended IP Phone:

1. Unpack the SPARSH VP510 box and verify package contents.
2. You can mount the phone on a wall or on the desk.
3. When you mount SPARSH VP510 on a wall,

- Use the mounting template to drill holed of appropriate size and distance.
- Fix the screw grips in the holes you drilled.
- Fix two screws in the holes on the wall, ensuring that they are aligned with the Keyhole Slots 1 and 2 of SPARSH VP510. The screws should protrude from the wall to fit into the Keyhole Slots.



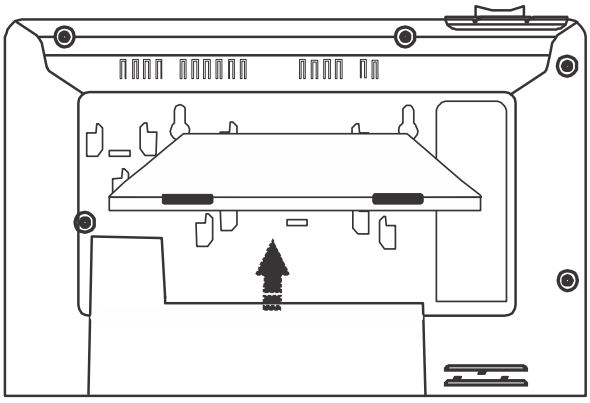
- Now, mount the phone with the screws into the Keyhole Slots.
- Reverse the handset wall mount tab to make sure that the handset remains intact when you mount the phone. Push the handset wall mount tab upwards to remove it from the slot. Rotate it 180 degrees clockwise and push it downwards into the slot.



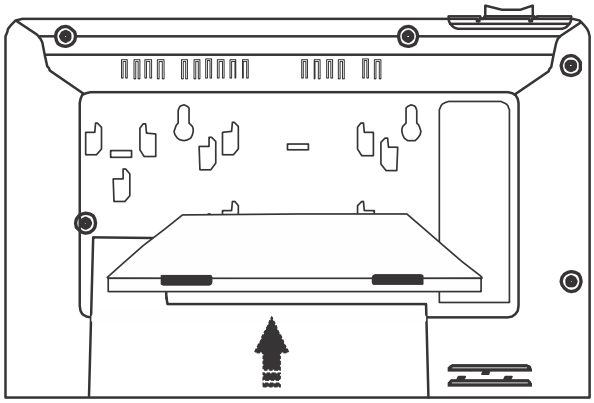
If you are unable to remove the wall mount tab, you may use a tool like a minus screw-driver to remove it.

- When you mount the phone on a desk,

- You can attach the Foot Stand in the following ways — at an angle of 45 degrees or 55 degrees



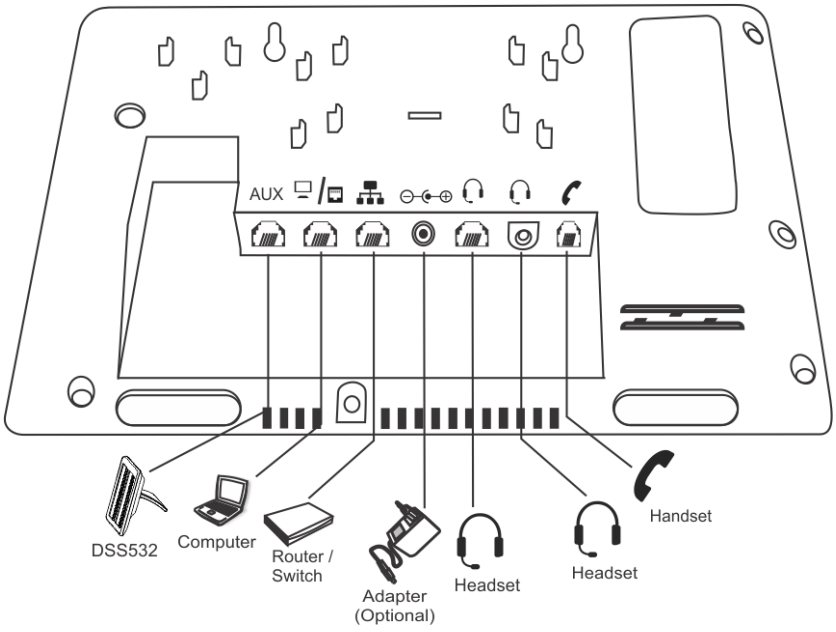
Stand attached at
45 degree angle



Stand attached at
55 degree angle

- Decide which of these positions would work for you best and accordingly attach the Foot Stand.

Refer to the diagram below for connectivity.



4. Connect the Handset.

   - Plug the long straightened end of the Spring Cord into the handset jack at the bottom of the phone, marked with the handset symbol .
   - Plug the other (short straight) end of the Spring Cord into the jack at the bottom of the handset.

5. Connect the Headset (not supplied by Matrix).

   - To use a Headset (not supplied with the phone), plug any standard stereo headset with 3.5mm single connector into the headset audio jack at the bottom of the phone, marked with the symbol .

     OR

     You may also plug in a headset with an RJ9 connector into the headset port at the bottom of the phone, marked with the symbol .

6. Connect to the IP Network.

   - Plug one end of the Ethernet Cable into the LAN Port at the bottom of the phone, marked with the symbol and the other end to the IP Network — A Router or LAN Switch.

7. Connect a PC to the Phone.

   - Plug one end of the Ethernet Cable into the PC Port at the bottom of the phone, marked with the symbol and the other end into the LAN Port of your PC/LAN Switch.

8. Connect DSS532 with the Phone.

   - To connect DSS532 with the phone, plug one end of the RJ11 cable into the AUX Port of the phone and the other end into the IN Port of the DSS532. For installation, see “Installing DSS532 with SPARSH VP510”.

9. Connect the Power Supply.

   - It is a PoE enabled phone and can be powered over Ethernet by connecting it to a PoE enabled LAN Switch (IEEE 802.3af Compliant).

     If you do not want to use PoE, plug in the connector of the Adapter into the power jack (DC Jack) at the bottom of the phone, marked with the symbol . Plug in the Power Adapter into a power outlet.

If both the power options, that is, PoE as well as Power Adapter are available to the phone, then the phone will derive power from the PoE enabled LAN Switch.

The IP phone should be used with Matrix original power adapter (5V/2A) only. The use of any third-party power adapter may cause damage to the phone.

10. Switch ON power supply.

    When you power the phone, the boot process will be initiated in the following sequence.

    - All keys with LED, including the Speaker key, and the Ringer LED, will glow.
    - The LCD display will light up and the booting message appears.
    - Then the 'Loading...' message appears on the phone display.
    - The phone will start loading the Extended IP Phone Firmware. It will display current firmware being loaded.
    - After loading the firmware, the phone will prompt you to change Network settings.
    - Wait for a few seconds.

If you want to change the Network Settings or Server Settings, press Yes key. Detailed instructions for changing the Network Settings of the phone are provided at the end of this topic. See "Network Settings" at the end of this topic.

- The phone makes DHCP Discovery and fetches its IP Address and Server Address from the DHCP Server.

  On getting the IP Address and Server Address, the phone initiates Auto Configuration to download the configuration files from SARVAM UCS.

- As the phone downloads the configuration files, the file names will appear one by one.
- On successful download of all configuration files, the phone attempts to register with SARVAM UCS.
- On successful registration, the phone will display the current day, date and time, the extension number and name assigned to the Extended IP Phone.

Network Settings

You can change the network settings of the Extended IP Phone. Press the Down key ⋁ when the phone is in idle state.

To navigate the menu,

- Press the Menu Key when the phone is idle.
- Scroll by pressing the Up/Down Navigation Key to reach the desired Menu option.
- Press the Select / OK Key to select the desired Menu option.
- Scroll by pressing the Up/Down Navigation Key to reach the desired sub-menu option.
- Press the Select / OK Key to select the desired sub-menu option.

To exit menu,

- Press Cancel Key.
  or
  Go ON-Hook.

The cursor is a non-blinking underscore that appears under the first letter of the first option in the menu. To make a selection in the menu, you must move the cursor in the desired direction using the Up, Down, Forward and Back key. When the cursor is at the desired position, press Enter key to make a selection.

Accessing Network Settings

You can access the Network Settings of the Extended IP Phone in any of the following stages:

1. During start-up, when the phone prompts you to change the network settings after loading the firmware.

   You must press **Yes** key and access network settings.

2. When the phone is making Network discovery, downloading configuration files, attempting registration.

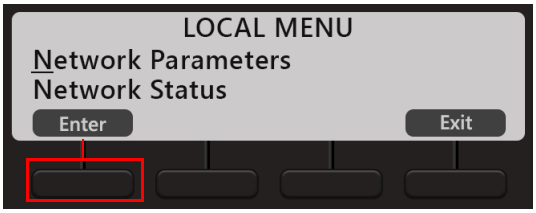
   You must press the **Menu** key to access network settings.

3. When the phone is in idle state, press the Down key ⋁ .

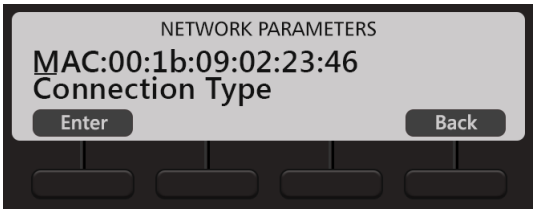
You can configure Network Parameters and view Network status from the Local Menu.

Configuring Network Parameters

- In the Local Menu of the phone, select Network Parameters by pressing the Enter Key.



- The Network Parameters submenu appears.



- Use the Down/Up key to reach the desired network parameter and press Enter key to select. Change the settings as per your requirements.
- Press **Save** key, to save the changes you make.
- You can configure all network parameters described below, except the MAC Address.

- *To enter a dot in the editable fields — IP Address, Subnet Mask, Gateway Address, DNS Address, Server Address — press * (Star) key.*
- *If you want to clear a single digit/character, move the cursor to the right of the digit/character you wish to clear. Now press the Delete key. The digit to the left of the cursor will be deleted.*

Before you start configuring the Network Parameters, get acquainted with following context keys:

Context Keys	Description
Enter/OK	To select a particular parameter
Save	To save the changes
Back	To move a step backwards without saving the changes
Delete	To delete previous characters from the cursor position
2Ab/123	2Ab - Alphanumeric Mode 123 - Numeric Mode

Connection Type

- Select the Connection Type as DHCP, PPPoE or Static according to the IP Addressing scheme of your network.

  If you select DHCP or PPPoE, the phone will be assigned IP Address, Subnet Mask and Gateway Address, DNS Address, Server Address, automatically by the DHCP/PPPoE server.

  For PPPoE Connection Type, you must configure the PPPoE User ID and Password provided by the Internet Service Provider.

  If you select Static, you must assign the IP Address, Subnet Mask and Gateway Address to the phone.

IP Address

- If you select Static as Connection Type, enter the static IP Address to be assigned to the phone.

  Enter the desired Static IP Address by pressing the digit keys.

To enter the dot/period in the IP Address, press the Star '*' key.

Subnet Mask

- If you select Static as Connection Type, enter the Subnet Mask to be applied on the phone by pressing the digit keys.

  To enter the dot/period in the IP Address, press the Star '*' key.

Gateway Address

- If you select Static as Connection Type, enter the Gateway Address here. This is the IP Address of the LAN Port of the Router.

DNS Server

- If you select Static as Connection Type, select the DNS Server option **Static** and configure the DNS Address.

- If you select DHCP or PPPoE as Connection Type and your Internet Service Provider provides DNS Address, select the DNS Server option **Automatic**. However, if your Internet Service Provider does not provide DNS Address, select **Static** and configure the DNS Address.

DNS Address

- If you select DNS Server as Static, enter the DNS Address here.

  To enter dot/period in the IP Address, press the Star '*' key.

DNS Domain Name

- If you select DNS Server as Static, enter the DNS Domain Name here. DNS Domain Name is optional.

PPPoE User ID

- If you have selected PPPoE as Connection Type, you must enter the User ID provided to you by your Internet Service Provider.

PPPoE Password

- This is the password provided by your Internet Service Provider for the PPPoE User ID. If you have selected PPPoE as Connection Type, you must enter the password provided by your Internet Service provider here.

PPPoE Service Name

- If your Internet Service Provider has provided a Service Name, enter the Service Name here. If your Internet Service Provider has not provided a Service Name, do not configure this parameter.

Primary Server Address

- The system works as the Auto Configuration Server for the phone. Enter the LAN or WAN IP Address/ Domain Name of SARVAM UCS here. Default: blank. The phone sends the request for configuration files to this Server Address.

  If you have selected DHCP as Connection Type, the phone will get the Primary Server Address and Port automatically from the DHCP Server. For this, use ***DHCP option 224*** and ***Data Type*** as '***String***' to provide Server Address and Port from the DHCP Server.

For PPPoE and Static Connection Types, you need to enter the Server Address.

Primary Server Port

- Enter the SPARSH Port of SARVAM UCS here. The phone sends the request for configuration files to this port.

  Valid range of the port is: 80 or 1025–65535. Default: 80.

Secondary Server Address

- If required, you can also configure the Secondary Sever Address as a fallback option. If the registration with the Primary Server fails the phone will send the registration and configuration requests to the Secondary Server Address. Speech-cut or unclear speech may be observed during on-going mature calls.

Secondary Server Port

- Enter the Secondary Server Port. The phone sends the request for configuration files to this port if the Primary Server fails.

  Valid range of the port is: 80 or 1025–65535. Default: 80.

VLAN Setting

If your phone is connected to a virtual LAN, you need to configure VLAN Settings.

To enable the VLAN switch to correctly route packets generated by the phone and the computers (on the LAN) to each other, the packets must be tagged with a VLAN header.

The VLAN header consists of the VLAN ID (12-bit) and Class of Service (CoS, 3-bit) for prioritization of traffic[98]. The meaning of CoS bits with respect to traffic type is as follows:

CoS	Traffic Type
0	Best Effort
1	Background
2	Spare
3	Excellent Effort
4	Controlled Load
5	Video
6	Voice
7	Network Control

- Select **Phone VLAN/COS** to add VLAN header to the packets generated by the phone, and add VLAN header to the packets relayed from the PC to its LAN port (packets generated by the PC connected to its PC port).
  - To configure Phone VLAN/COS, select **Enable?**. The VLAN ID will be tagged on all packets generated by the phone (SIP, RTP, DNS, ARP, etc.). Default: Disabled.

98. The IEEE 802.1P standard allows Layer2 switches to prioritize the traffic, thus providing Quality of Service (QoS), that is, better handling of data that pass over a network, thereby resulting in greater reliability and quality. Quality of Service (QoS) on Layer2 is referred to as Class of Service (CoS) which is defined by IEEE 802.1P.

- Select **VLAN ID** and enter the VLAN ID that you have assigned to the VLAN in which the IP Phones are connected. Valid range: 0-4094. Default: 1.
- Select **SIP CoS** and define the CoS (priority) bits in all SIP packets. Valid range: 0-7. Default: 3
- Select **RTP CoS** and define the CoS (priority) bits in all RTP packets. Valid range: 0-7. Default: 6.

- Select **PC/VLAN CoS** to add VLAN header to all packets entering the PC Port and leaving the LAN port of the phone. Default: Disabled.
  - To configure PC VLAN/COS, select **Enable?**.
  - Select **VLAN ID** and enter the same ID as you have assigned to the VLAN in which the computers are connected. Valid range: 0-4094. Default: 1.
  - Select **CoS** and define the Layer 2 CoS (priority) bits. Valid range: 0-7. Default: 0.

PCAP

To capture packets sent and received from and by the phone for monitoring and troubleshooting, you can enable PCAP on the phone. The phone captures up to 1 MB of packets. For more information and instructions on how to use PCAP Trace on the phone, refer to the *EON510_SPARSH VP510 User Guide*.

MAC Cloning

If you require cloning of the MAC Address, configure the following:

- Select **Enable?** and press the Enter key. Select Yes to enable.
- In **Enter Clone MAC Address**, enter the address you wish to clone.

802.1x Authentication

If you want to restrict unauthorized clients from connecting to your LAN, you need to enable 802.1x Authentication. Using EAP MD5 protocol the PC connected to the LAN port of the IP Phone is first authenticated and then it gets connected to LAN.

You need to configure the following 802.1x Authentication parameters:

- Select **Enable?** and press the Enter key. Select Yes to enable.
- Enter the 802.1x Authentication **Identity** provided by you network administrator.
- Enter the 802.1x Authentication **MD5 Password** associated with identity provided by your network administrator.

When you change the Network Settings, the phone will restart.

Viewing Network Status

- When the phone is in idle state. You must press the Down key ⋁ to access the Network Settings.
- Again press Down key ⋁ to select Network Status and press the Enter key.

Use the Down/Up key to view the status of the various network parameters. The status of the following parameters appear on your display as you scroll.

- **MAC:** This is the MAC Address of the phone.
- **IP:** The current IP Address assigned to the phone.
- **MASK:** The current Subnet mask assigned to the phone.
- **GW:** The current Gateway IP Address assigned to the phone.
- **DNS:** The Domain Name Server address assigned to the phone.
- **Active Server:** This displays the Server that is active — Primary, Secondary — with which the phone is currently registered.
- **S. ADD:** This displays the IP address of the Active Server. It may be the LAN or WAN IP Address / Domain Name of the SARVAM UCS or the Secondary Server IP Address (if configured) or any Fallback Server.
- **S. PORT:** This displays the port of the Active Server. It may be the SPARSH Port of SARVAM UCS or the Secondary Server Port (if configured) or the Fallback Server Port.
- **DOMAIN:** The Domain Name assigned to the phone.
- **802.1x Authentication:** The 802.1x authentication status is displayed—Success, Failure, Authenticating or Disabled.
- **FIRM:** The version of the current Firmware of the phone.
- **UBOOT**: The UBOOT release date.
- **KERNEL**: The KERNEL release date.

Refer to the ***EON510_SPARSH VP510 User Guide*** to know more.

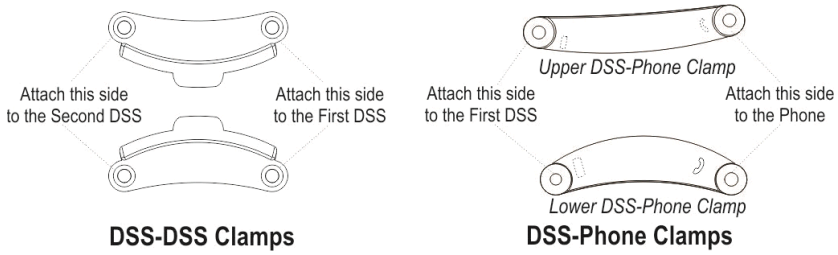
Installing DSS532 with SPARSH VP510

Once you have installed SPARSH VP510 with SARVAM UCS, you can install the DSS532 by following the steps given below:

1. Unpack the box and verify the package contents[99].

99. See "Packing List" of Appendix topic.

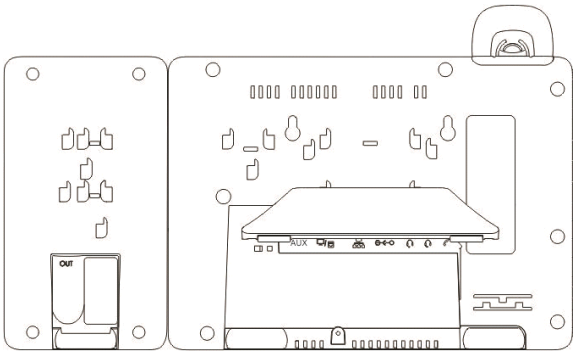
2. Four clamps are provided with the phone — 2 DSS-Phone Clamps and 2 DSS-DSS Clamp.



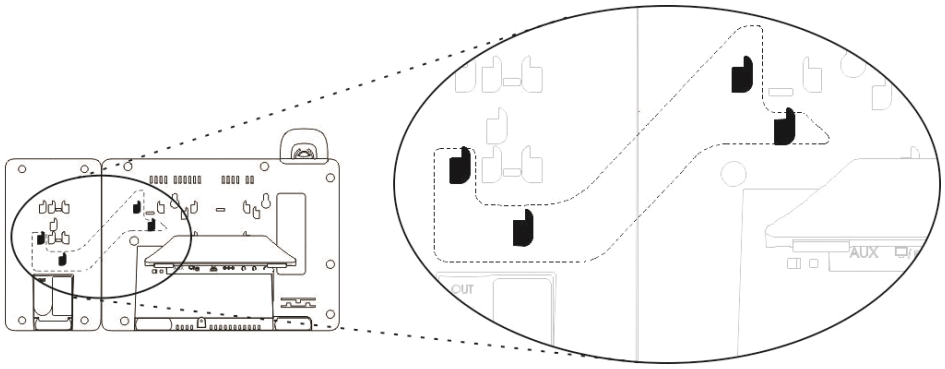
Connecting the First DSS532

Connecting the Extender

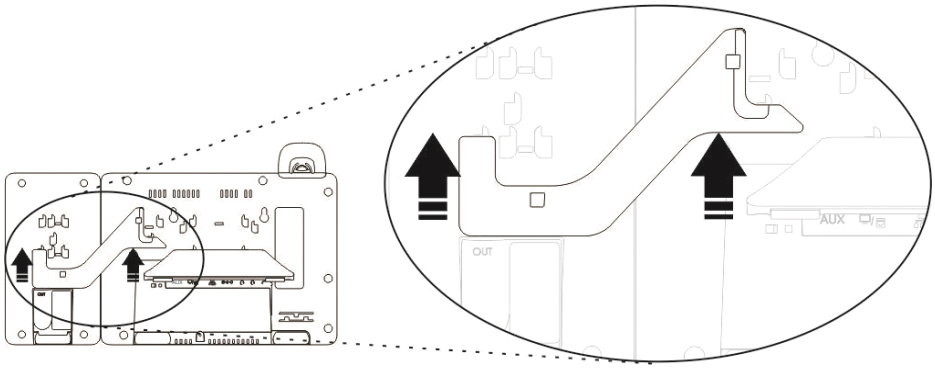
3. Turn the phone upside down on the table and place the inverted DSS532 adjacent to it.



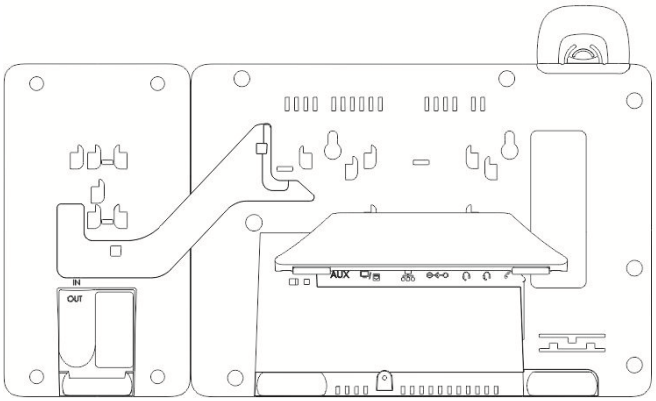
4. To attach the DSS532 with the phone, place the DSS Extender as illustrated below.



5. Insert the hooks on the Extender into the slots provided on the phone and the DSS532.

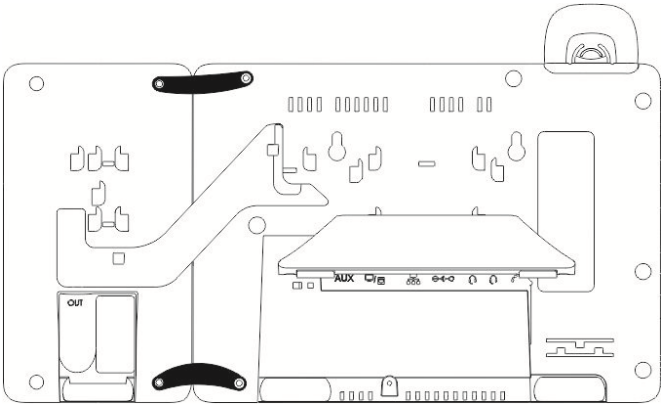


6. Firmly slide the DSS Extender upwards to lock them in place.



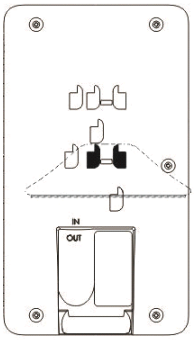
Attaching the Clamps

7. Now attach the clamps. To do so,
   - Remove the screws to attach the clamps.
   - Place the DSS-Phone Clamps between the DSS532 and the phone.
   - Insert the screws back to fix the clamps.

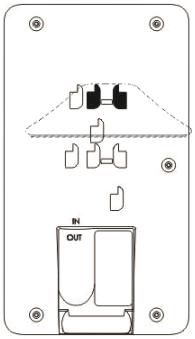


Attaching the Footstand

8. You can mount the DSS532 with the phone on the desk at two angles — **45 degrees** or **55 degrees** by attaching the Foot Stand.



Stand attached at 45 degree angle

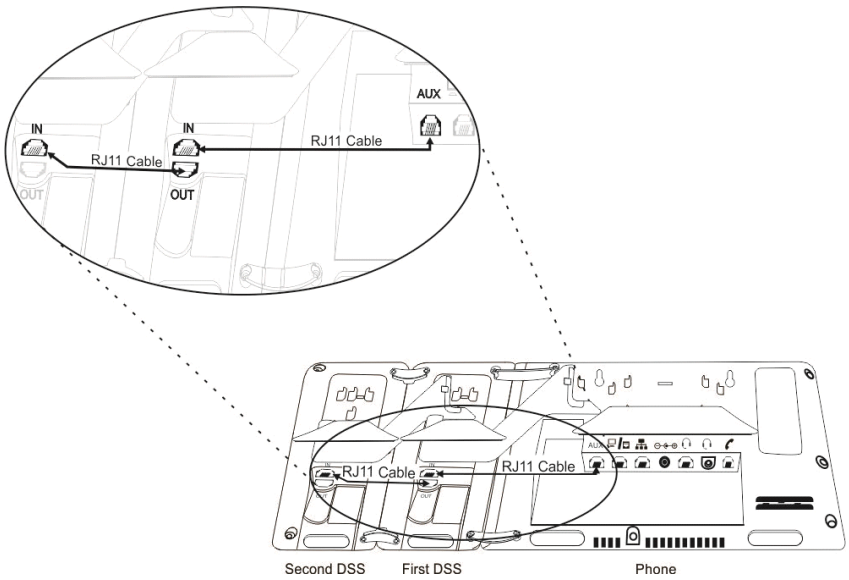


Stand attached at 55 degree angle

Make sure both, the DSS532 and phone are mounted at the same angle.

Connecting the Cables

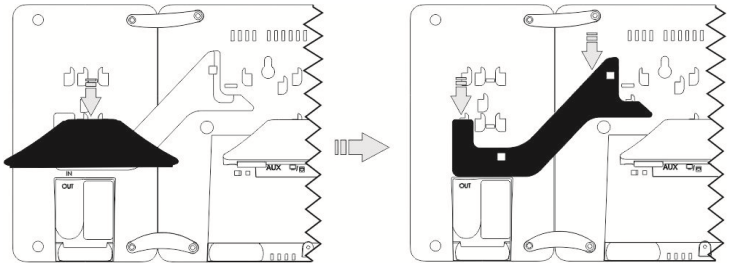
9. To connect the DSS532 with phone, plug one end of RJ11 Cable into **Auxiliary(AUX) Port** of the phone and the other end into the **IN Port** of the DSS532.



Connecting Multiple DSS532

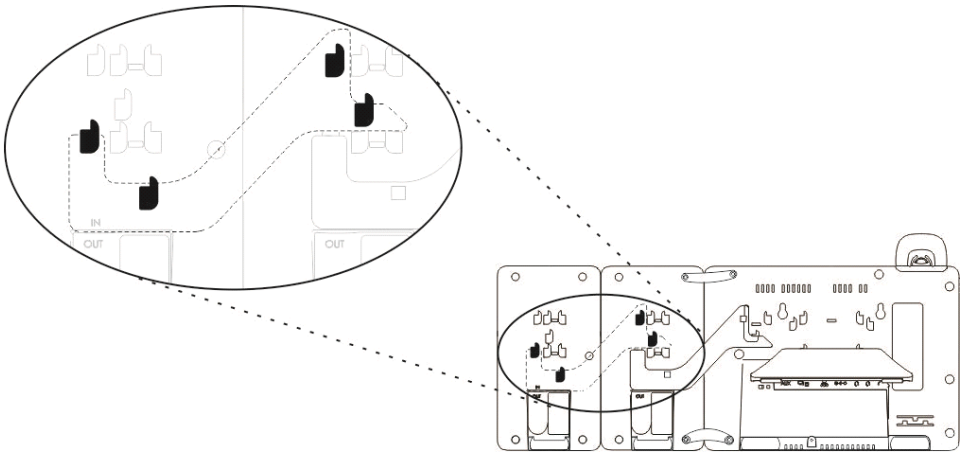
Remove the Foot Stand

10. Remove the Foot Stand of attached DSS532. To do so,
    - Firmly slide the Foot Stand of the attached DSS532 downward to unlock.
    - Now, slide down the attached DSS Extender in downward direction.

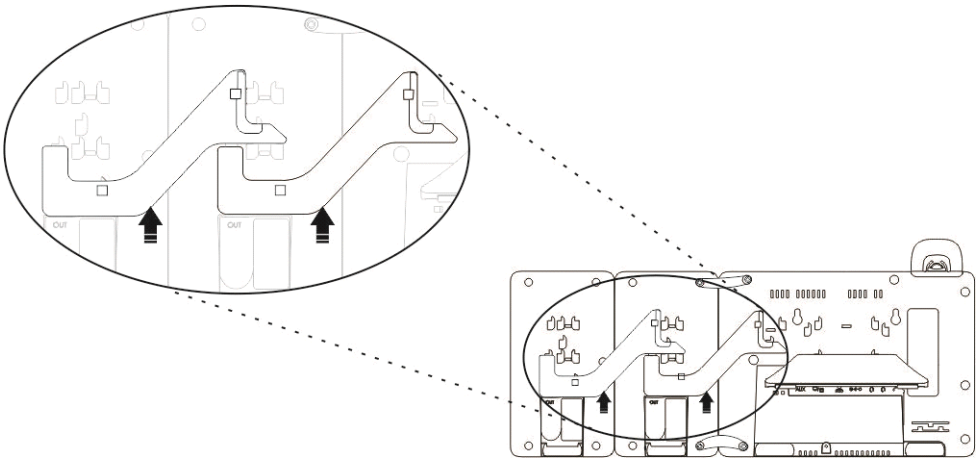


Attach the second DSS Extender

11. To attach the second DSS Extender,
    - Place another inverted DSS532 adjacent to the existing assembly.
    - Place the DSS Extender as illustrated in the diagram below.
    - Insert the hooks on the Extender into the slots provided on both the DSS532.

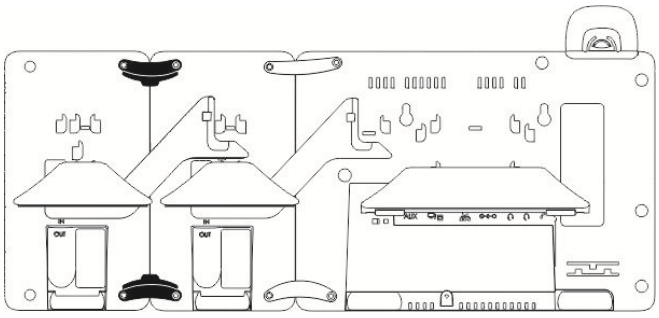


12. Firmly slide both the DSS Extenders upward consecutively (attach the second extender first followed by the existing one attached to the phone) and lock them in place.



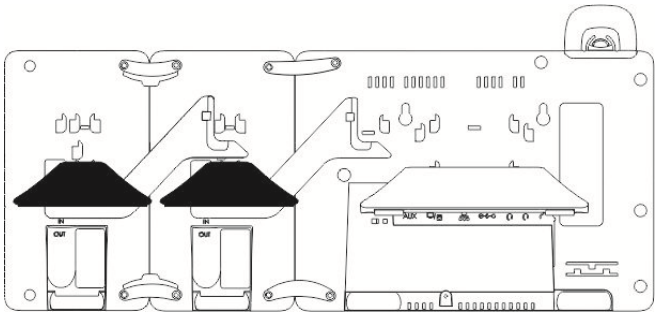
Attach the Clamps

13. Attach the DSS-DSS Clamps between both the DSS532.



Attach the Foot Stand

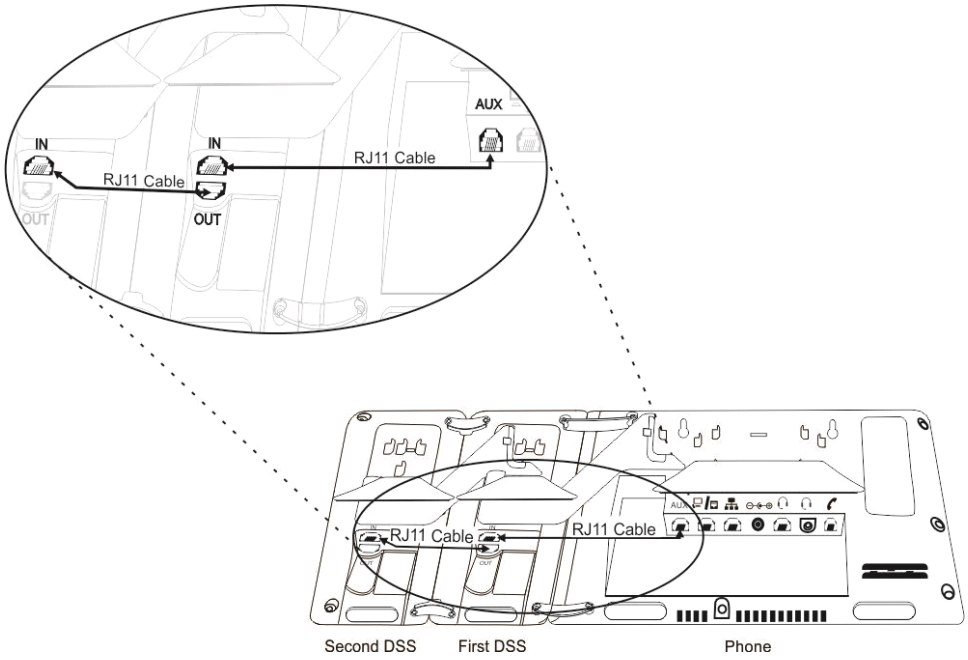
14. Attach the Foot Stand of both the DSS532.



Make sure both, the DSS532 and the phone are mounted at the same angle.

Connect the second DSS532 to the existing assembly

15. Plug one end of the RJ11 Cable into the OUT Port of the existing DSS532 (already connected with the phone) and the other end into the IN Port of the second DSS532.



You can install a maximum of four DSS532 with a phone.

16. After you have connected the DSS532 with the phone, you can configure the DSS Keys. For instructions, see “Programming DSS Console Keys”.

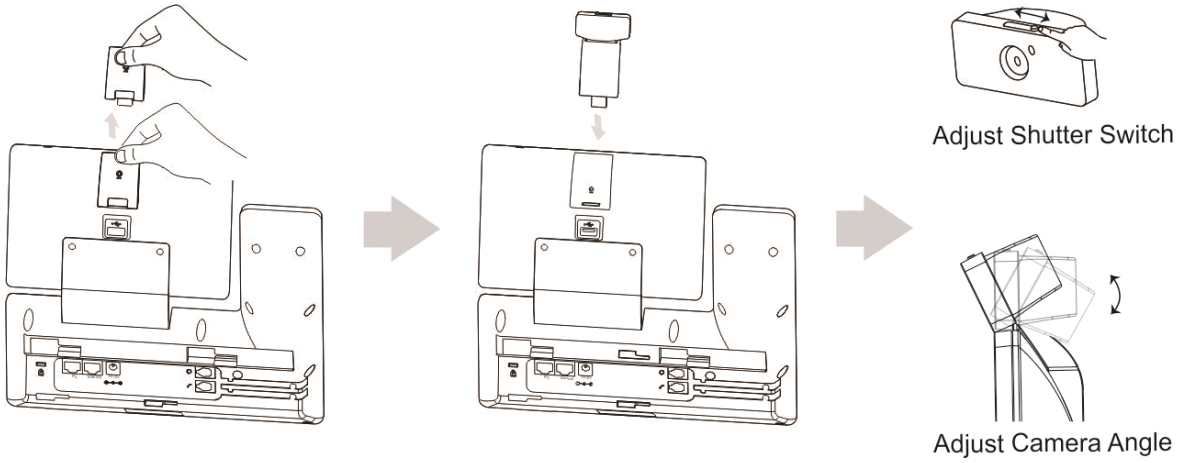
Connecting Extended SPARSH VP710 as Extended SIP Extension

You are recommended to complete the following steps before connecting the Matrix Extended SPARSH VP710 to SARVAM UCS:

- Decide the location where you want to place Matrix Extended SPARSH VP710 within your LAN.
- Log in to *Jeeves*. For instructions, read the topic “Configuring SARVAM UCS”.
- You must configure the necessary parameters in SARVAM UCS so that Extended SPARSH VP710 can register as a SIP Extension. For instructions, see “Configuring Matrix Extended SPARSH VP710”.

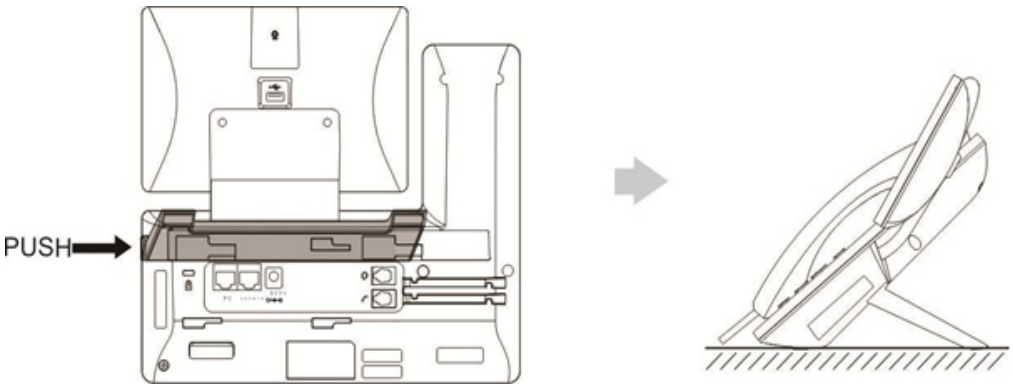
Now, follow the steps described below to install Extended SPARSH VP710.

1. **Inserting the camera**

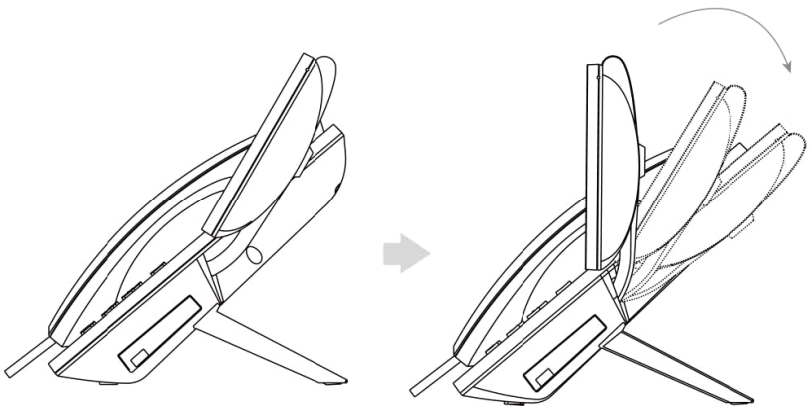


It is recommended to use only the Matrix original Camera, supplied with the IP Phone for video calling. The use of any third-party camera may cause damage to the phone. Damages to the phone caused by using third-party camera is not covered by Matrix warranty.

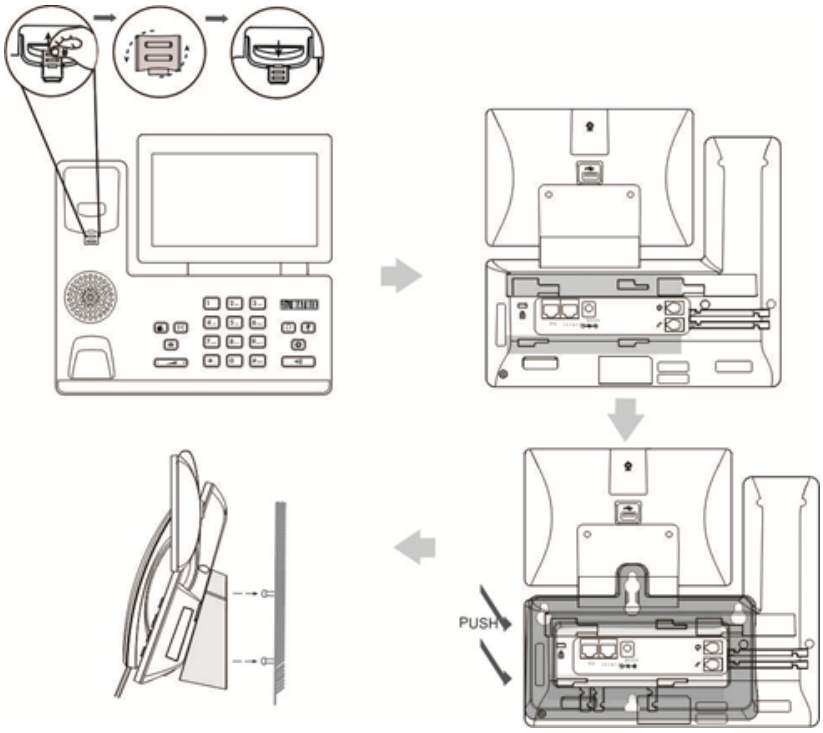
2. **Attaching the stand**



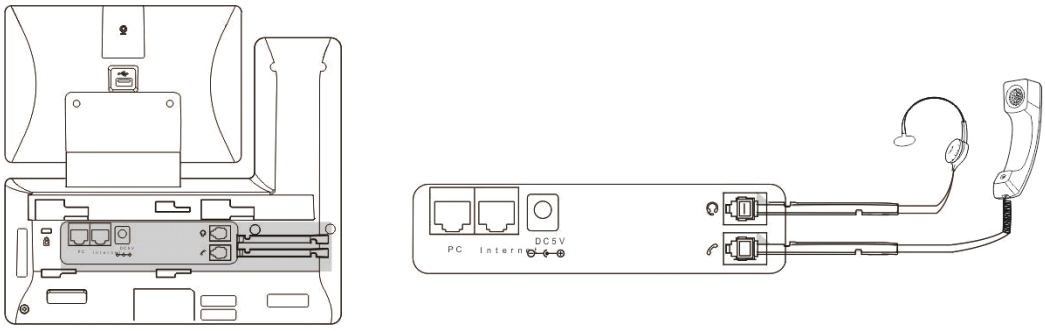
3. **Adjusting the angle of the touch screen**.



4. **Attaching the optional wall mounting bracket**

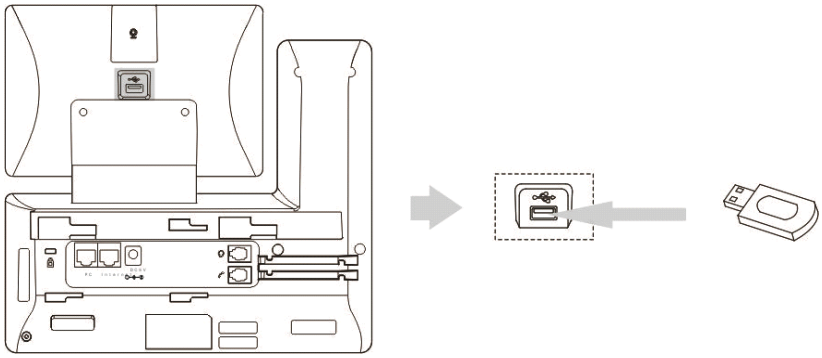


5. **Connect the handset and optional headset.**



A headset is not included in the packaging contents. Contact your dealer/reseller for more information.

6. **Connect the optional USB Flash drive.**



7. **Connect the network and power.**

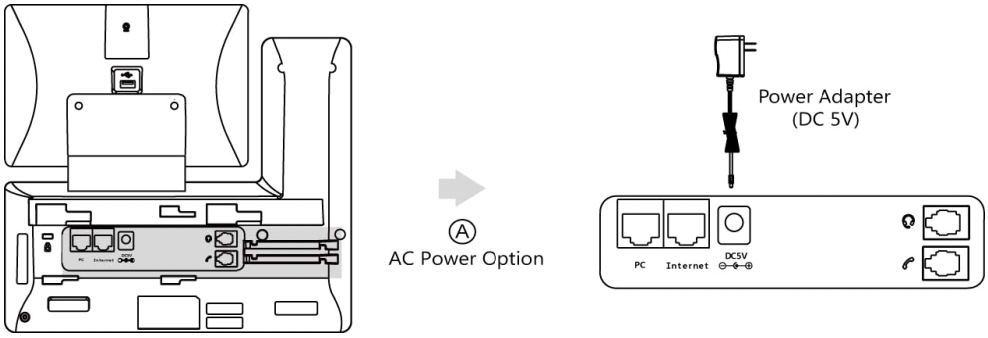
There are two options to connect the power and the network.

- AC power
- Power over Ethernet (PoE)

AC Power

To connect the AC power:

- Connect the DC plug on the power adapter to the DC5V port on the phone and connect the other end of the power adapter into an electrical power outlet.

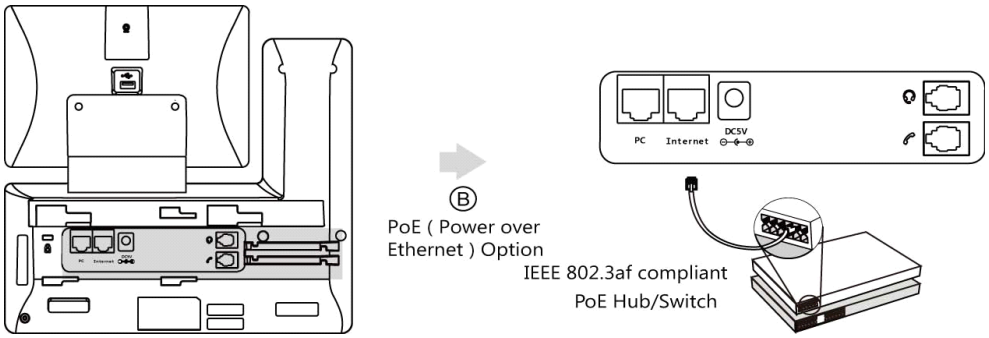


Power over Ethernet (PoE)

With the included or a regular Ethernet cable, the IP Phone can be powered from a PoE-compliant switch or hub.

To connect the PoE:

- Connect the Ethernet cable between the Internet port on the phone and an available port on the in-line power switch/hub.



If in-line power switch/hub is provided, you don't need to connect the phone to the power adapter. Make sure the switch/hub is PoE-compliant.

Do not unplug or remove power while the phone is updating firmware.

After the IP Phone is assembled and connected to the power supply, it automatically begins the initialization process.

During this process, the IP Phone displays the start up screen "Welcome Initializing…please wait".

Once the IP Phone is initialized, it displays two different phone modes:

- Standard SIP
- Extended SIP

- Select Extended SIP, to operate the IP Phone in the extended mode. As soon as you select this mode, the booting process initiates again and the start up screen displays "Welcome Initializing…please wait". After the IP Phone is initialized, it attempts to contact a DHCP Server in your network to obtain valid IPv4 network settings (example: IP address, Subnet Mask, Gateway address, DNS address).You need to configure the basic network parameters of the IP Phone manually, if these are not provided by the DHCP Server or if your network does not support DHCP.

  Refer to the *EXTENDED SPARSH VP710 User Guide*, for detailed instructions:

  - To change the Network Settings of the phone and configure the network parameters.
  - To use Wi-Fi for connectivity and configure its parameters.

- On getting the IP Address and Server Address, the phone initiates Auto Configuration (when DHCP is selected) to download the configuration files from SARVAM UCS.
- On successful download of all configuration files, the phone attempts to register with SARVAM UCS.
- On successful registration, the Home screen appears.

The phone will register successfully, only if the SIP Extension parameters in SARVAM UCS have been correctly configured as per your installation scenario.

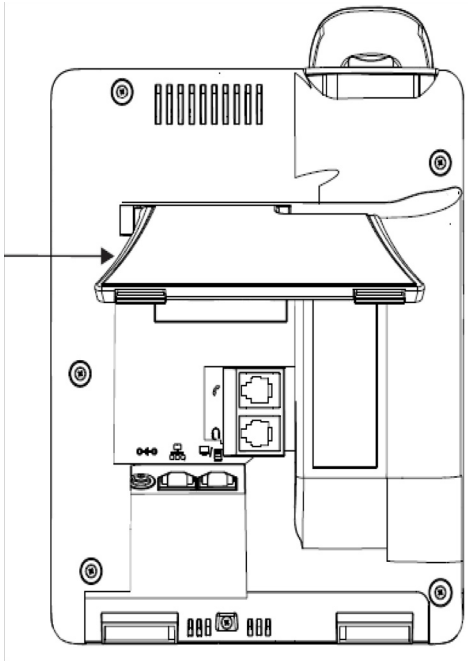
Connecting SPARSH VP210 as Extended SIP Extension

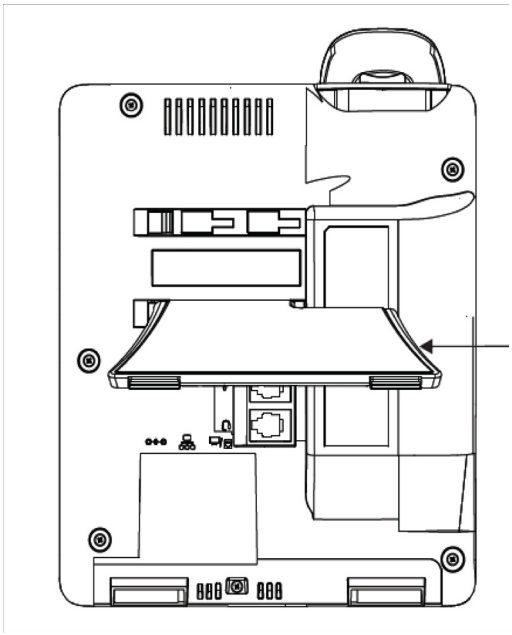
You are recommended to complete the following steps before connecting the Matrix SPARSH VP210 to SARVAM UCS:

- Decide the location where you want to place SPARSH VP210 within your LAN.
- By Default, in SPARSH VP210, the Connection Type selected is DHCP.
- If you want to use the **DHCP Server** for assigning IP Address to the Extended IP Phone, select **DHCP option 224** and **Data Type** as '**String**' and program the LAN or WAN IP Address /Domain Name of SARVAM UCS and SPARSH Port in the format **"IP_Address:Port"** in your LAN DHCP Server as per your installation scenario.
- Log in to *Jeeves*. For instructions, read the topic "Configuring SARVAM UCS".
- You must configure the necessary parameters in SARVAM UCS so that SPARSH VP210 can register as a SIP Extension. For instructions, see "Configuring Matrix SPARSH VP210".

Now, follow the steps described below to install SPARSH VP210.

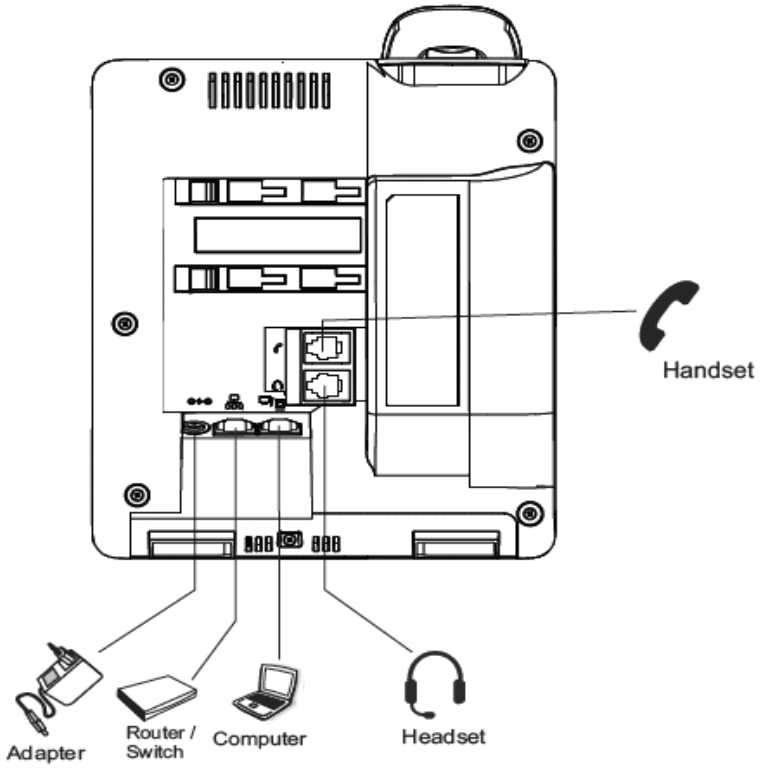
1. Unpack the SPARSH VP210 box and verify package contents.
2. When you mount the phone on a desk, you can attach the Foot Stand in two ways at **45° Angle** or at **55° Angle**.





- Decide which of these positions would work for you best and accordingly attach the Foot Stand.

  Refer to the diagram below for connectivity.



3. Connect the Handset to the Phone body.

- Plug the long straightened end of the Spring Cord into the handset jack at the bottom of the phone, marked with the handset symbol .
- Plug the other (short straight) end of the Spring Cord into the jack at the bottom of the handset.

4. If you want to use a Headset (not supplied) with your phone, You may plug in a headset with an RJ9 connector into the headset port at the bottom of the phone, marked with the symbol .
5. To connect the LAN, Port , plug one end of the Ethernet Cable into the LAN Port at the bottom of the phone marked with the symbol and the other end to the IP Network — A Router or LAN Switch.
6. To connect your phone to a computer on your desk, plug one end of the Ethernet Cable (not supplied with this phone) into the PC Port at the bottom of the phone, marked with the symbol and the other end into the LAN Port of your PC/LAN Switch.
7. It is a PoE enabled phone and can be powered over Ethernet by connecting it to a PoE enabled LAN Switch (IEEE 802.3af Compliant). In this case you need not connect the Power Adapter.

   If you do not want to use PoE, plug in the connector of the Adapter into the power jack (DC Jack) at the bottom of the phone, marked with the symbol . Plug in the Power Adapter into a power outlet.

If both the power options, that is, PoE as well as Power Adapter are available to the phone, then the phone will derive power from the PoE enabled LAN Switch.

The IP phone should be used with Matrix original power adapter (5V/0.6A) only. The use of any third-party power adapter may cause damage to the phone.

8. Switch ON power supply.

   When you power the phone, the boot process will be initiated in the following sequence.

   - The LCD display will light up and booting message appears.
   - While loading the application then the loading message appears on the phone display.
   - The phone makes DHCP Discovery and fetches its IP Address and Server Address from the DHCP Server.

If you want to change the Server Settings, press Settings.

Refer to the SPARSH VP210 (Extended) User Guide, for detailed instructions, to change the Network Settings of the phone and configure the network parameters.

- On getting the IP Address and Server Address, the phone initiates Auto Configuration to download the configuration files from SARVAM UCS.
- On successful download of all configuration files, the phone attempts to register with SARVAM UCS.
- On successful registration, the Home screen appears.

The phone will register successfully, only if the SIP Extension parameters in SARVAM UCS have been correctly configured as per your installation scenario.

Refer to the ***SPARSH VP210 (Extended) User Guide*** to know more.

Starting Up ETERNITY GENX

Power ON

1. If you have completed all the installation tasks, switch on power supply.

   - For PSUNI Card installed in the system, connect the three-prong plug of the power cord from the system into the AC outlet, and switch on power supply.

   - For PS48V Card installed in the system, keep the MCB Switch ON and power the FCBC.

2. Observe the Reset Cycle.

Reset Cycle

- Reset Cycle (Power-ON Self Test) takes about 2 minutes to finish.

- All the LEDs of the system, the cards and the keys of the DKP/SIP devices attached to the System are turned on.

Interpreting LEDs

The functioning of the LEDs of the system and the various cards and their meaning are summarized at the end of the installation instructions for each Card Type.

Refer to the LED Patterns described for each Card Type to verify if the system is operating properly and locate faults, where they occur.

When the reset cycle is successful, the default Extension “Access Codes” loaded by the system and the date and time of the “Real Time Clock (RTC)” of the system will appear on the LCD display of the DKPs/ IP Phones you have connected with the system.

CHAPTER 8

Installing ETERNITY PENX

- *The Matrix ETERNITY PENX is to be installed by persons trained and experienced in telecom wiring.*
- *The person installing the ETERNITY PENX must be familiar with trunks, physical wiring of the MDF on both the exchange (System) side and the line side (CO).*
- *When installing any equipment, make sure that you take all the necessary precautions for handling electronic and electrical appliances. Follow proper procedures for static electricity, while handling the system and its cards to prevent damage to the system and harm to yourself.*
- *Use a grounding mat and wear an anti-static strap/belt. Read the do's and don'ts listed in "Protecting the System and Yourself".*
- *If you have complied with the requirements and instructions described in "Before You Start", you may now begin the installation of your ETERNITY PENX.*

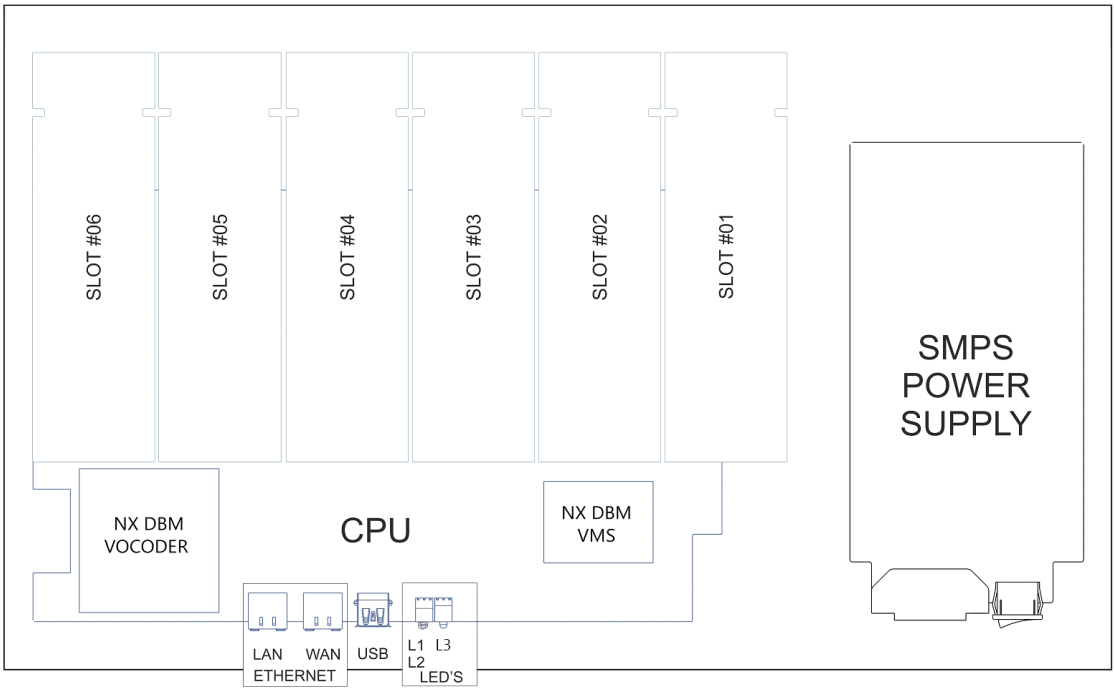
The Matrix ETERNITY PENX is shipped factory fitted with the Power supply unit and the CPU Card (refer the section "Hardware Overview").

The cards – SLT, DKP, CO,T1E1PRI, GSM – and the modules – VoIP and VMS - are shipped separately as per the order placed by individual customers.

Illustrated below are the positions of the universal slots.

ETERNITY PENX

The Power Supply unit and the CPU are in-built, and fixed on the bottom plane of the ETERNITY PENX.



Six Universal slots are located on the CPU. The connectors of the slots are located on the CPU.

Cards are mounted on the CPU and secured on the three studs on the CPU, with the screws provided.

ETERNITY PENX has – LAN Port, WAN Port, External USB, Internal USB, 3-PIN Power Plug Connector and ON/OFF Switch along with three LEDs.

Instructions are provided in the following for installation of the cards. These instructions are to be followed also when you expand the system (add more cards) or remove cards for maintenance and repair.

1. Have all the necessary wiring ready. Read the topic *Main Distribution Frame* for guidance on how to set up the MDF and connect the system with the MDF, and install Primary Protection against heavy voltages.

2. Unpack the box. Check the package contents (see Packaging List). Contact your Dealer/Distributor if any of the items is missing, faulty or damaged. Do not discard the packaging material.

Mounting the System

3. You can mount the system on a table, a wall. or a rack.

   If you have decided to mount the ETERNITY PENX on a wall, use the Mounting Template for drilling the holes at appropriate distances on the wall.

4. Mount the system at the selected site. Make sure that the system is placed such that you have full access to the front and back panels. The holes in the side panels are provided for ventilation; Make sure that these are not blocked, to prevent overheating.

When installing the system in a rack allow adequate space between the system and other units for air circulation.

Connecting Input Power Supply

5. Ensure that a proper electrical earth and telecom earth are in place.

6. Check the voltage at the power point from where the supply is to be given to the system. It should be as per the specifications. Earth the system properly.

Inserting Cards

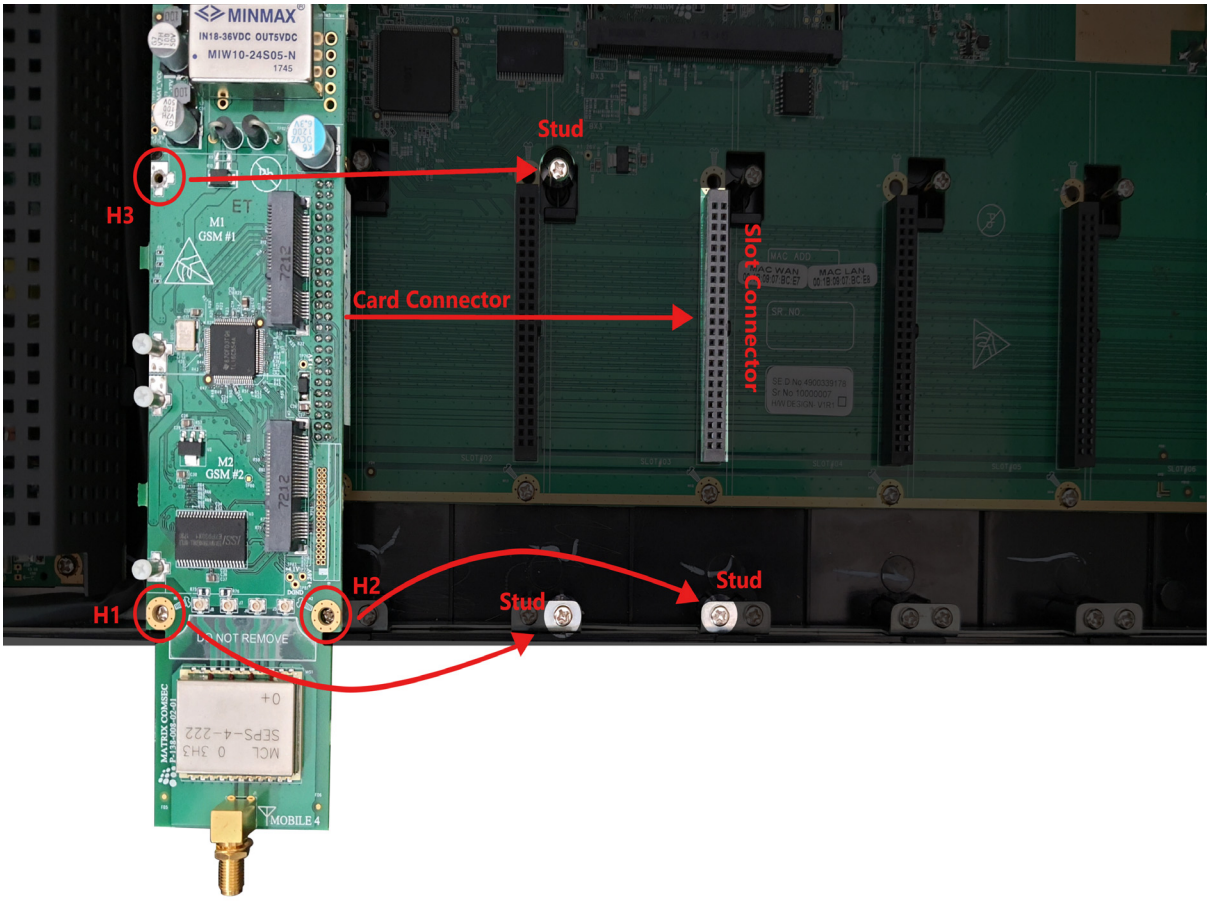
7. Make sure the power supply is turned off and the power cord is unplugged, before you begin inserting the cards.

8. Unpack the card and check the package contents.

9. Unscrew the top cover of the ETERNITY PENX and remove it by sliding it out. Keep the cover and the screws aside.

10. Select a free slot from the universal slots.

11. Grasp the card by its sides or corners. Fit the card into connectors of the selected slot. Ensure that the card is seated perfectly for all the connector pins on the cards make complete contact with those on the CPU (motherboard) on the bottom plane.



12. When the card is firmly seated in the connector, use the three screws provided with the card to secure it on the studs labeled as H1, H2 and H3.

13. Following the above steps, install each card into the universal slots.

    Detailed installing instructions are provided for each card — DKP, SLT, CO, T1/E1/PRI, GSM and modules — VoIP and VMS — later in this section. Refer to them when installing each card type.

14. Using the cables supplied with the cards, connect the card interfaces with the MDF (for SLT, DKP and CO), the ISDN modem (for T1/E1 PRI lines), the IP network, a Computer as applicable for each card.

15. Lead the cables out of the enclosure through any of the two cable outlets on either side of the enclosure.

    Lead the cables neatly and tangle-free into the MDF.

16. When you have completed inserting the cards and connecting the cables, replace the top cover by sliding it in place. Secure the cover with the two screws you removed.

Since the connectors of the cards will not be visible after the cover has been replaced, you are advised to label the cables appropriately to facilitate identification.

17. To remove a card:
    - Switch off power supply, unplug the power cord.

- Disconnect any cables connected to the card.
- Remove the screws from the studs H1, H2 and H3.
- Grasp the card by corners; gently rock the card to ease it out of its slot connectors.

The Power Supply Unit

The Power Supply in the ETERNITY PENX is an in-built universal SMPS, with 100-240VAC, 47-63Hz Mains as Input AC Voltage Power Supply.

There is no provision for battery backup. So, you are recommended to provide Battery Backup of 24 VDC, 7-10A.H. by connecting a UPS to keep the system powered during outages.

Typical power consumption[100] of ETERNITY PENX is 40 W.

The Power Supply Unit is factory-fitted. It must be removed and refitted by trained technicians only, and only for the purpose of fault repair or replacement.

The CPU

The CPU of ETERNITY PENX manages the entire system, controls all other cards (SLT, DKP, CO, DKP+SLT, CO+SLT, DKP+CO T1E1PRI Single, GSM, etc.). All configuration and programming information is stored on this card.

The CPU is factory-fitted on the bottom plane of the ETERNITY PENX.

The CPU may be removed and reinstalled solely for the purpose of fault repair or replacement, and by trained technicians only.

Ports and Connectors:

Port	Connector	Description
LAN	RJ45	Used for connecting the Ethernet cable into LAN Port to connect to a PC or a LAN Switch.
WAN	RJ45	Used for connecting the Ethernet cable into WAN Port to connect to a Broadband Router/Modem.
USB	USB to COM Converter	The External USB can be used as COM Port by connecting the USB to COM Converter. The USB to COM Port can be used to: • set up and run software applications — PMS and CAS. • capture System Activity Log, System Fault log and Hotel Motel Activity logs. • generate SMDR reports.
Power	3-PIN Power Cord	Used to connect Power Adapter
ON/OFF	Switch	Used to Switch ON/OFF Power.

100. Considering 30% SLT off-hook when 48 SLTs are connected.

Jumpers

Jumper J6 on the CPU card of ETERNITY PENX are used to Reset the SE Password. Refer the table below:

Jumper Number	Position	Function
J6	AB BC (default)	Reset SE Password. Normal.

ETERNITY PENX does not support Redundancy and Hot Swap.

Providing Battery Backup

Provide Battery Backup of 24 VDC, 7-10A.H. by connecting the system with a UPS.

Connecting ETERNITY PENX to the Local Area Network

Plug one end of the Ethernet cable supplied with the system into the LAN port of the ETERNITY PENX and plug the other end into the LAN switch.

With the ETERNITY PENX connected to a LAN, you can:

- access the web-based programming tool Jeeves from any PC on the LAN.
- set up and run software applications such as PMS and CAS on any PC on the LAN.
- generate Station Message Detail Record (SMDR) Reports on any PC on the LAN.

When you connect the ETERNITY PENX to a Ethernet PC, you need to make sure that

- *The IP Address of the LAN Port of the ETERNITY PENX and the Ethernet Port of the PC are not the same.*
- *The LAN Port of ETERNITY PENX and the Ethernet Port of the PC are in the same Subnet.*
- For instructions to change the IP address and Subnet Mask, refer *"Changing IP Address and Subnet Mask of the WAN Port Only"* at the end of this topic.

Connect the ETERNITY PENX to a Standalone PC

You can connect ETERNITY PENX to a standalone PC through the LAN/WAN Port, depending on the type of application you want to use.

1. Connect the LAN/WAN Port of ETERNITY with the Ethernet Port of the stand-alone PC using the Ethernet cable supplied with ETERNITY.

   If you connect to the PC via the LAN/WAN port you can:

   - Use the web-based programming Tool Jeeves
   - Configure LAN/WAN using FTP
   - Take debug/report on ETERNITY PENX Syslog
   - Capture SMDR reports, SMDR Online and SMDR Posting

- Capture System Activity Log and System Fault Log, Hotel Motel Activity Log
- Run PMS Interface or CAS Interface in the Hotel Application[101]

When you connect the ETERNITY PENX to a standalone PC, you need to make sure that

- *The IP Address of the LAN/WAN Port of the ETERNITY PENX and the Ethernet Port of the PC are not the same.*
- *The LAN/WAN Port of ETERNITY PENX and the Ethernet Port of the PC are in the same Subnet.*
- *For instructions to change the IP address and Subnet Mask, refer "Changing IP Address and Subnet Mask of the WAN Port Only" at the end of this topic.*

Changing IP Address and Subnet Mask of the WAN Port Only

The default IP Address of the LAN Port of the ETERNITY PENX is 192.168.2.100, WAN Port of the ETERNITY PENX is 192.168.1.100 and the Subnet Mask is 255.255.255.0.

- Ascertain the IP Address and Subnet of the standalone PC.

  If the system is connected to a LAN PC, ask the LAN Administrator to assign an IP Address and a Subnet Mask to the ETERNITY PENX[102].

- Replace the cover of the ETERNITY PENX and switch ON the system.
- Change the IP Address and the Subnet Mask of the WAN Port by dialing the following commands from an extension of the ETERNITY PENX.

To change IP Address of the WAN Port:

- Dial ***2110-IP Address***
  For example: to change the IP Address to 192.168.50.10 dial ***2110-192168050010.***
  You get confirmation tone.
- Dial ***2111-Subnet Mask***
  For example: to change the Subnet Mask to 255.255.255.0 dial ***2111-255255255000.***
  You get confirmation tone.
- Dial ***00*** to exit from the programming mode.

- *If there is a DHCP server on the LAN to which the LAN Port of the ETERNITY PENX is connected, there is no need to change the IP Address or Subnet Mask, as these will be provided automatically by the DHCP server.*
- *You must only enable the DHCP flag of the LAN/WAN Port of ETERNITY PENX.*
- To enable DHCP flag
  - Dial ***1#91-SE Password***

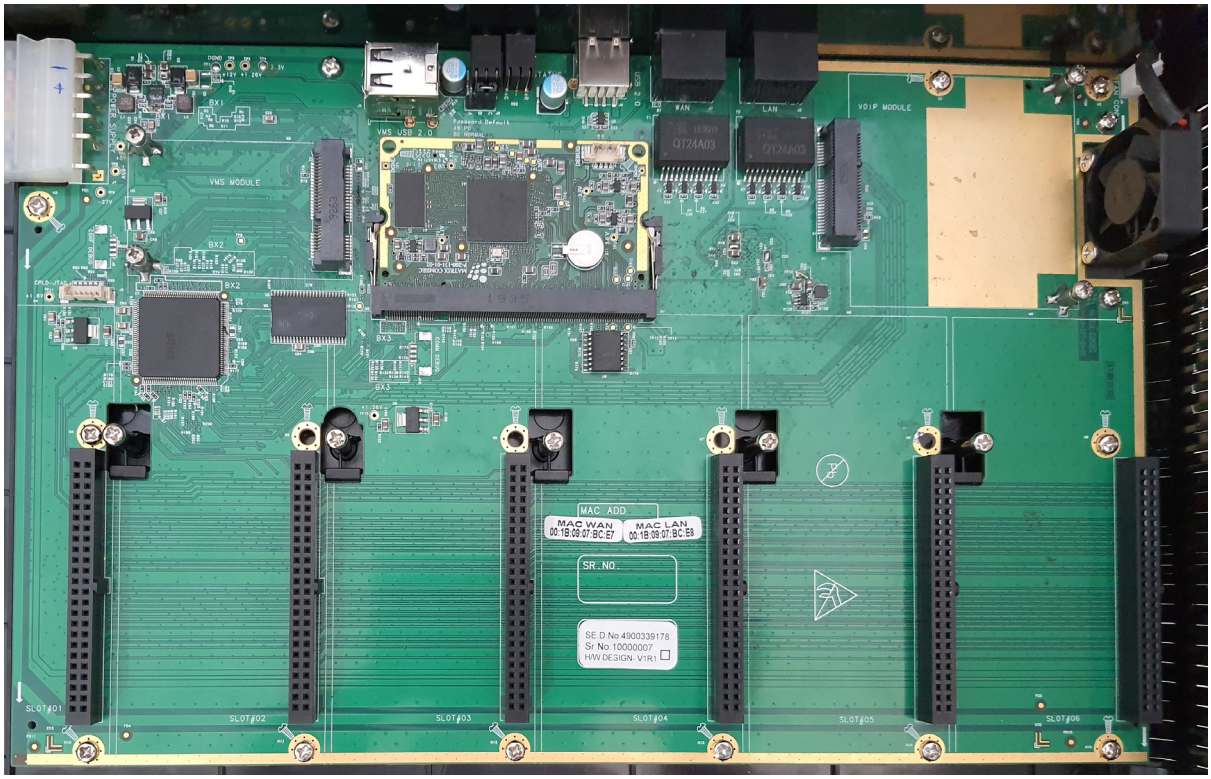
101. Refer the ETERNITY Hospitality System Manual to know more.
102. This will not be necessary, if there is a Dynamic Host Configuration Protocol (DHCP) server on the LAN.

- You get programming tone.
- Dial ***2117-1***
- You get confirmation tone.

- Dial ***00*** to exit from the programming mode.

The CPU Card

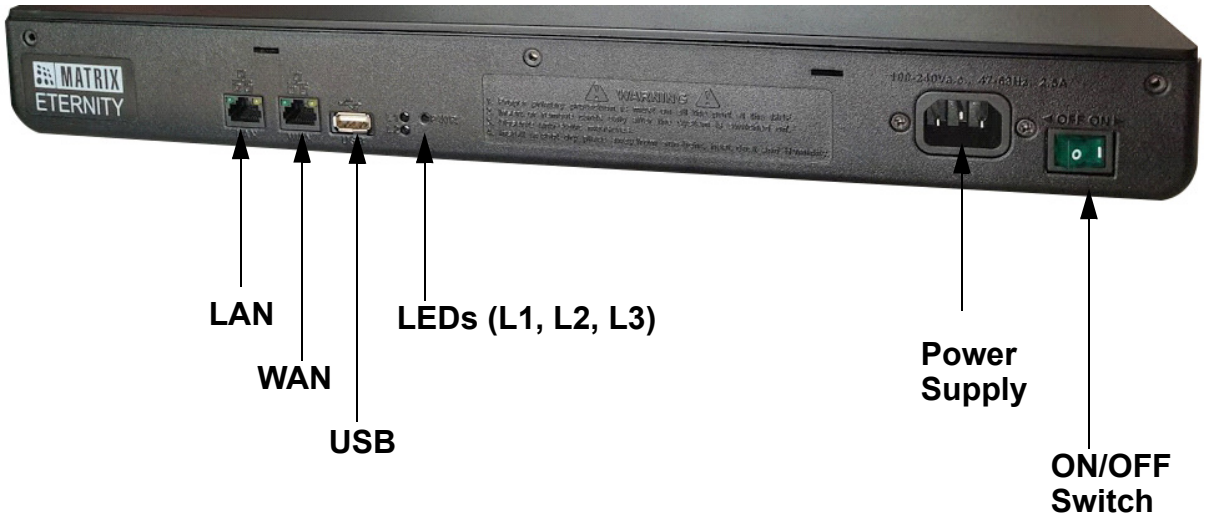
The ETERNITY PENX CPU Card hosts the SARVAM UCS SMB Application. It supports one VOCODER modules and one VMS module. Both the modules — NX DBM VOCODER64 and NX DBM VMS64 are optional and can be purchased separately.



This card hosts Communication Manager, Feature Server, VoIP Server, VMS Server and other important servers and modules which controls all other slave cards (SLT, DKP, CO+SLT, DKP+SLT, T1E1, GSM etc.). All the configuration and programming information is stored on this card.

The CPU Card has a WAN Port, LAN Port and USB Port on the front panel. It also has an Internal USB Port with a factory fitted pendrive.

Ports and Connectors:



Port	Connector	Description
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Power	3-PIN Power Cord	Used to connect Power Adapter
ON/OFF	Switch	Used to Switch ON/OFF Power.

If you buy a spare CPU Card separately, the default pendrive will not be provided along with it.

LAN Interface

The LAN Port is provided to connect:

- the system to a PC or a LAN. This port is used for operating the web-based programming software Jeeves.
- the CPU Card to the Local Area Network to register SIP extensions through the LAN Port.
- set up and run software applications such as PMS and CAS on any PC on the LAN.
- generate Station Message Detail Record (SMDR) Reports on any PC on the LAN.
- capture *"System Activity Log"*, *"System Fault Log"* and Hotel Motel Activity Log.

WAN Interface

The WAN Port is provided to connect:

- a LAN Switch/Hub/Router/Modem.
- the CPU Card to the public network over a Router/Modem. Any user on the public network can be registered as SIP Extension through the WAN Port.
- set up and run software applications such as PMS and CAS on any PC on the LAN.
- generate Station Message Detail Record (SMDR) Reports on any PC on the LAN.
- capture *"System Activity Log"*, *"System Fault Log"* and Hotel Motel Activity Log.

VoIP Interface

The CPU Card supports one NX DBM VOCODER64 modules. You must purchase the module separately for VoIP functionality.

VOCODER Channels

The system supports one NX DBM VOCODER64 Module. This module supports 64 VOCODER Channels[103]. You must purchase the module separately. The system provides 4 pre-activated VOCODER channels by default which can be used after installing NX DBM VOCODER64 module. If you require more channels, you can purchase the licenses accordingly. Matrix provides two licenses — SARVAM VOCODER CHNL4 and SARVAM VOCODER CHNL16.

If you require more than 64 VOCODER channels, you can install another NX DBM VOCODER64 Module.

A call made from a SIP Extension or SIP Trunk to another SIP Extension or SIP Trunk will consume two VOCODER channels, whereas a call made from a SLT or DKP extension to a SIP Extension or SIP Trunk will consume one VOCODER channel. Thus, the number of speech paths available to make simultaneous calls will depend not only on the number of VOCODER channels, but also on the number of channels consumed by such SIP-to-SIP and Analog/Digital extension to SIP Trunk/SIP Extension calls.

VMS Interface

The system supports a full-fledged, 'in-skin' Voice Mail System module to provide mailbox facility to all its extensions users. The Voice Mail System also forms the basis of other features like Conversation Recording and Call Taping.

Each Mailbox has the capacity of storing 15,000 voice messages. The maximum size of each Mailbox is 60,000 minutes. By default, the size of each Mailbox is set to 5 minutes. The maximum Message Length for each Mailbox is 9,999 seconds. By default, the Maximum Message Length for each Mailbox is set to 15 seconds.
The NX DBM VMS64 Module is an optional module. It must be purchased separately. The factory fitted Pen Drive provided also contains the VMS configuration files, and voice messages for prompts and greetings along with the SARVAM UCS SMB Application. The Pen Drive is also the storage device for mailbox messages.

If required, you may use a Pen Drive of upto 64GB by replacing the factory fitted pendrive with a new one.

The system supports a maximum of 16 channels out of which 4 channels are provided by default. If you require more channels, you can purchase the licenses accordingly. Matrix provides two licenses — SARVAM VMS CHNL4 and SARVAM VMS CHNL16.

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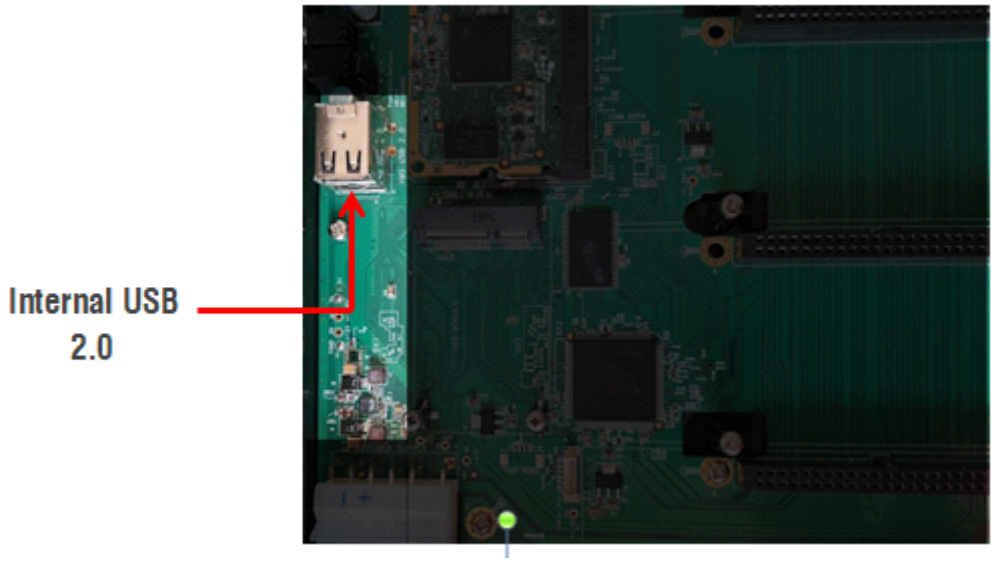
103. The number of VOCODER channels that will be supported would be as per the license you purchase.

Internal USB Port

The CPU Card has an Internal USB Port with a pendrive inserted into it.



The pendrive supports FAT32 file format. It contains the SARVAM UCS SMB Application, VMS greetings, messages, mail boxes, Matrix Extended IP phone firmwares and SMS Server firmware.

Do not remove the pendrive.

When you select the SARVAM UCS SMB Application, the system fetches the application from the pendrive.

External USB Port (Device Port) 2.0

The CPU Card has an External USB Port on the fascia. This can be used as a COM Port by connecting the USB to COM Converter.

The USB to COM Converter will not be provided by Matrix.

The following USB to COM Converters are supported:

- Prolific PL2303 by BAFO
- CH341 by Winchiphead

If you use any other USB to COM Converter, Matrix does not guarantee it's proper functioning.

The USB to COM Port has a DB-9 connector.

The port allows you to connect a PC to the system, so that you can install and operate the following features:

- set up and run software applications such as PMS and CAS on any PC on the LAN.
- generate Station Message Detail Record (SMDR) Reports on any PC on the LAN.

- capture *“System Activity Log”* and *“System Fault Log”*, Hotel Motel Activity Log.

LED

The System has three dual color (Green and Red) LEDs.

LED 1 and 2 denote the health of the system and LED 3 is for Power Status.

- LED 1 - L1 works as a Heart Bit of CPU Card. In Normal Condition, L1 will be turned ON Green for 1 sec and OFF for 1 sec.
- LED 2 - L2 indicates the Layer Application status. In Normal condition, L2 will be turned ON Red and will blink very fast.
- Both L1 and L2 also indicate Application status.

  Case 1: L1 is steady GREEN/OFF and LED2 is OFF. This means the Application is hung and there is some problem in the Application code.

  Case 2: LED1 is steady GREEN/OFF and LED2 is GREEN/RED/ORANGE. This means the Layer is hung and there is some problem with the Layer code.
- LED 3 - L3 indicates Power ON/OFF Status.
  - At Power ON, Power LED will turn ON (Continuous Green).
  - System LED (STS) will display following error/events/status.

LED Status	Color	Comment
1sec On - 1sec Off (Continuous)	Green	ETERNITY PENX started successfully.
5sec On - 5sec Off (Continuous)	Green	Software Mismatch (uboot checksum did not match)
10sec On - 10sec Off (Continuous)	Green	Flash Lock due to License
100msec ON - 100msec OFF - 100msec ON - 100msec OFF - 100msec ON - 5000msec OFF	Green	Recovery Mode[a]

a.If you get Recovery Mode LED status, contact Matrix Support Team.

- Mobile Ports take about 3 minutes to get registered with the network.

Jumpers

The position and function of the Jumpers on the CPU Card are:

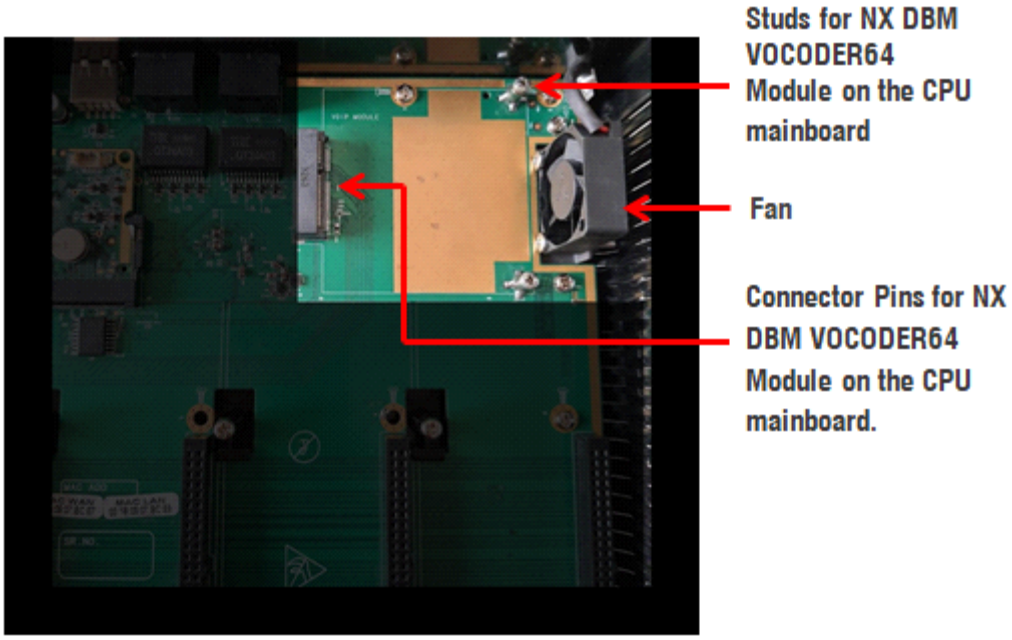
Jumper Number	Position	Function
J6	AB BC (default)	Reset SE Password. Normal.

Installing the VOCODER Module

To install,

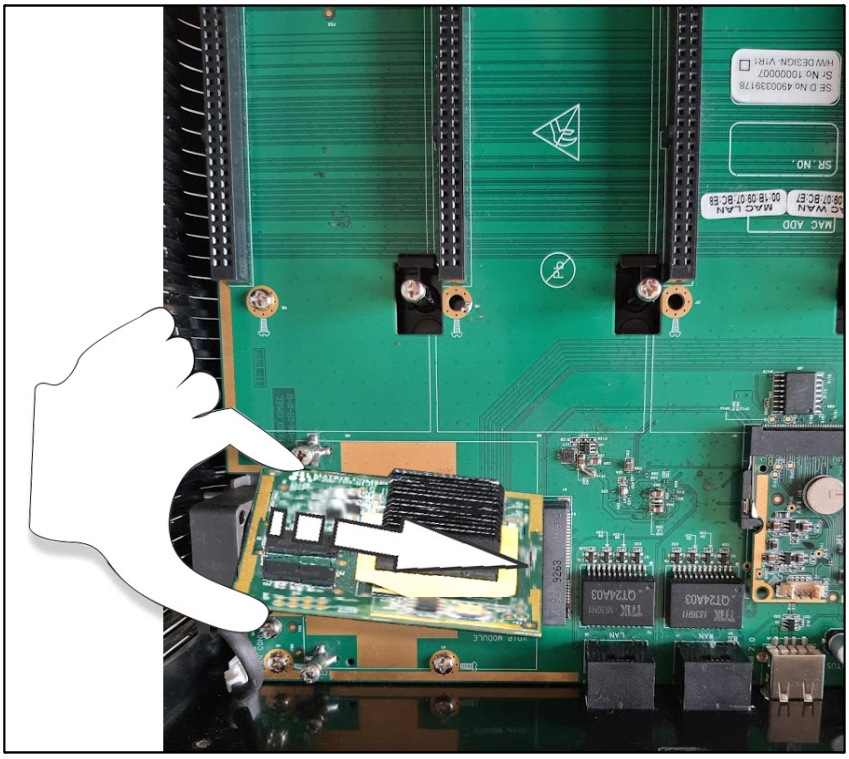
- Unpack the NX DBM VOCODER64 Module.

- Unscrew the top cover of the ETERNITY PENX and remove it by sliding it out. Keep the cover and the screws aside.
- The NX DBM VOCODER64 Module is to be mounted adjacent to the fan on the CPU board.
- Locate the studs on the CPU mainboard and unscrew them to install the NX DBM VOCODER64 Module.



- Carefully hold the NX DBM VOCODER64 Module from the edges. Make sure you do not touch the PCB area.

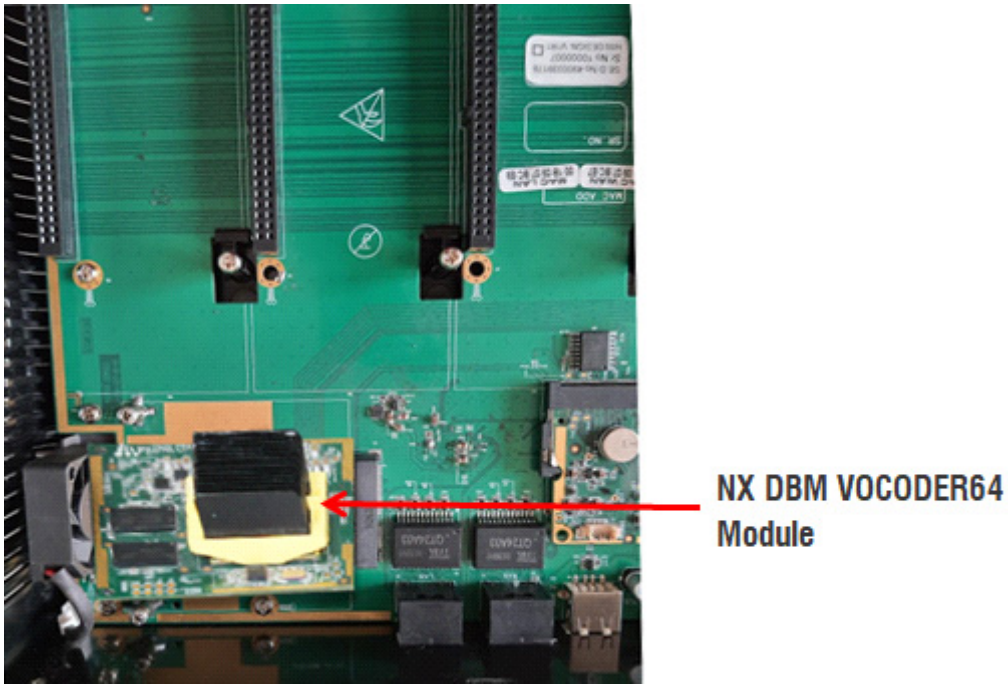
- Gently seat the NX DBM VOCODER64 Module on the studs on the CPU mainboard. The connector pins on the module must make complete contact with those on the CPU mainboard. Do not apply pressure.



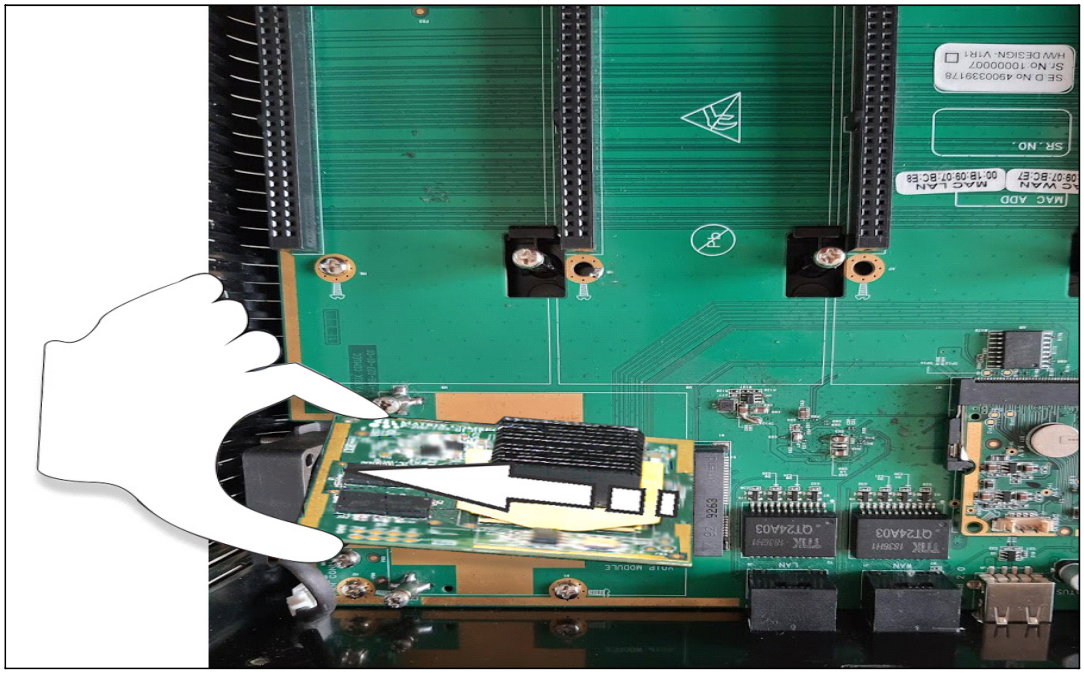
- When the module is seated firmly on the studs on the CPU mainboard, secure the module with the screws on the studs.
- Do not apply excessive pressure. Follow the same steps if you wish to install VMS module, see *"Installing the VMS Module"*.
- If you have no other modules to install, replace the top cover and secure the cover with the screws.

Removing the VOCODER Module

- Locate the VOCODER Module you want to remove from the CPU Card.



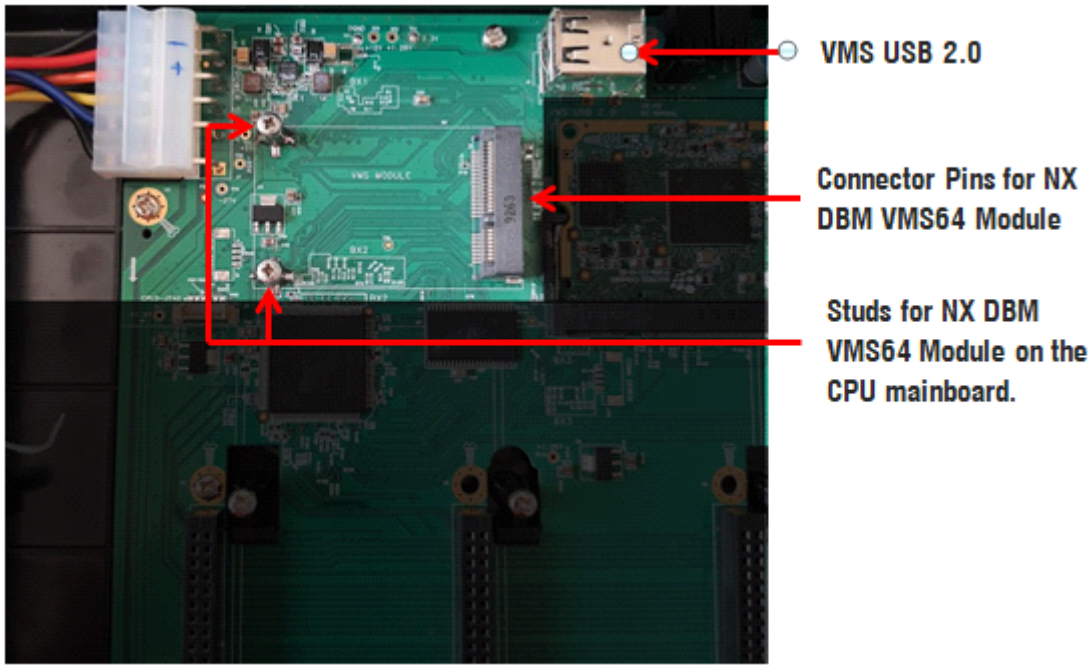
- Remove the screws, firmly hold the module and ease it out of the PCI connector carefully.



Installing the VMS Module

To install,

- Locate the PCI connector for NX DBM VMS64 Module on the CPU card.



- Follow the same steps as described in installing the NX DBM VOCODER64 Module. See *"Installing the VOCODER Module"*.
- The pendrive which is provided to you by default contains VMS data and VMS firmware. You will be able to use the VMS features once you activate the VMS License.

  If you want to store more voice mail messages or greetings then you will need more space to store the same. You can replace this default pendrive with a new one having more space.

  To do so, you need to format your new pendrive with FAT32 file format and then copy all the contents of the factory fitted pendrive into the new pendrive.

Make sure you switch-off the system to replace the pendrive. The system will not detect the new pendrive if you do not restart the system after replacement.

- If you have no other modules to install, replace the top cover and secure the cover with the screws.
- Connect a computer to the LAN/WAN Port of the system with the Ethernet cable supplied for the port.
- Open a Web browser on the computer to access the embedded Web server, Jeeves.
- Activate the VMS License Voucher. See *"License Management"* for instructions.
- Configure VMS. For detailed instructions, see *"Configuring Voice Mail System"*.

The Single Line Telephone Card

The Single Line Telephone (SLT) Card provides the interface to connect as extension phones, any standard, two-wire, analog single line telephone instrument-rotary, pulse-tone, cordless, feature phones with or without Calling Line Identification.

The SLT Card is available in the following configurations for ETERNITY PENX. The SLT interface is available in combination with Digital Key Phone ports and Two-wire Trunk ports on a single card.

SLT Cards for ETERNITY PENX

Card Name	Configuration and Application
ETERNITY PE Card SLT8	8-port card to connect 8 Single Line Telephones
ETERNITY PE Card SLT4	4-port card to connect 4 Single Line Telephones
ETERNITY PE Card DKP2+SLT6	Combination card, with 2 ports to connect to 2 Digital Key Phones and 6 Single Line Telephones
ETERNITY PE Card CO4+SLT4	Combination card, with 4 ports to connect to 4 Two-wire Analog trunk lines and 4 Single Line Telephones
ETERNITY PE Card CO2+DKP2+SLT4	Combination card, with 2 ports to connect to 2 Two-wire Analog trunk lines, 2 Digital Key Phones, and 4 Single Line Telephones
ETERNITY PE Card CO2+SLT6	Combination card with 2 ports to connect 2 Two-wire Analog trunk lines, and 6 ports to connect 6 Single Line Telephones

Choose an SLT Card with the configuration that meets your requirement for SLT ports. Also consider the maximum SLT Port capacity of the system you are installing.

The maximum number of SLT ports supported by ETERNITY PENX are 48 SLT Ports.

Connectors

The SLT Cards have RJ45 connectors. A multi-pair, cable is supplied for each connector on the card.

Installing Single Line Telephones

To be able to connect Single Line Telephones as Extensions to your ETERNITY PENX, you must install at least one of the aforementioned SLT cards in the System.

1. Decide the number of SLT extensions required and arrange for as many telephone instruments.

   You may use any standard telephone instrument like a rotary phone, a pulse-tone switchable push-button phone, a feature phone or a cordless phone.

Use SLTs equipped with a 'Flash' key, as several of the features and facilities of the ETERNITY require you to press Flash. If any of the SLTs you have selected does not have a Flash key, tap the Hook switch of the phone to dial Flash.

2. Unpack the SLT card and check the package contents.

3. Make sure that the power supply is switched off, before you begin the installation of the card. Always wear an electrostatic discharge prevention wrist strap/belt and use a grounding mat.

4. Unscrew the top cover of the ETERNITY PENX and slide it out. Keep the cover and the screws aside.

5. Select any of the free slots from the universal slots.

6. Grasping the card by its sides or corners fit it onto the connectors of the selected slot. The card should be seated such that its connector pins make perfect contact with those on the CPU (motherboard) on the bottom plane.

7. Secure the card on the studs labeled H1, H2 and H3 with the three screws provided.

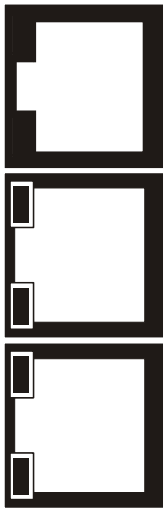
8. Repeat the same steps to install another SLT card. It is not necessary to install the other SLT cards in subsequent slots. You may install the other SLT cards in any of the universal slots.

9. Now, use the cables supplied with the SLT card to connect the SLT wires with the Main Distribution Frame.

   For each connector on the SLT Card, there is a separate cable with an RJ45 plug on one end and free at the other end.

   Refer to the pin-out details of the connectors on each card type to connect the wire-pairs.

ETERNITY PE Card SLT4

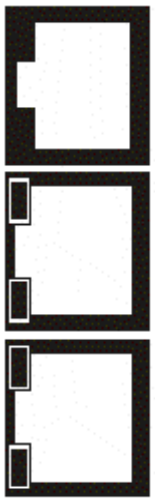


Connector	Color	Connection	H/w port Offset
RJ45-1	Blue - (Blue & White)	SLT	01
	Orange - (Orange & White)	SLT	02
	Green - (Green & White)	SLT	03
	Brown - (Brown & White)	SLT	04
RJ45-2			
RJ45-3			

ETERNITY PE Card SLT8

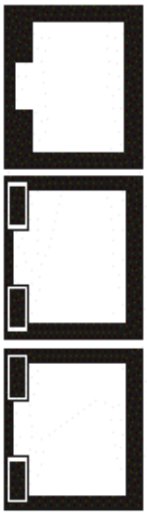
Connector	Color	Connection	H/w port Offset
RJ45-1	Blue - (Blue & White)	SLT	01
	Orange - (Orange & White)	SLT	02
	Green - (Green & White)	SLT	03
	Brown - (Brown & White)	SLT	04
RJ45-2	Blue - (Blue & White)	SLT	05
	Orange - (Orange & White)	SLT	06
RJ45-3	Blue - (Blue & White)	SLT	07
	Orange - (Orange & White)	SLT	08

ETERNITY PE DKP2+SLT6



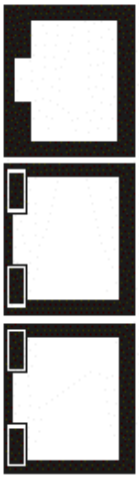
Connector	Color	Connection	H/w port Offset
RJ45-1	Blue - (Blue & White)	SLT	01
	Orange - (Orange & White)	SLT	02
	Green - (Green & White)	SLT	03
	Brown - (Brown & White)	SLT	04
RJ45-2	Blue - (Blue & White)	SLT	05
	Orange - (Orange & White)	SLT	06
RJ45-3	Blue - (Blue & White)	DKP	01
	Orange - (Orange & White)	DKP	02

ETERNITY PE Card CO4+SLT4



Connector	Color	Connection	H/w port Offset
RJ45-1	Blue - (Blue & White)	SLT	01
	Orange - (Orange & White)	SLT	02
	Green - (Green & White)	SLT	03
	Brown - (Brown & White)	SLT	04
RJ45-2	Blue - (Blue & White)	CO	01
	Orange - (Orange & White)	CO	02
RJ45-3	Blue - (Blue & White)	CO	03
	Orange - (Orange & White)	CO	04

ETERNITY PE Card CO2+DKP2+SLT4



Connector	Color	Connection	H/w port Offset
RJ45-1	Blue - (Blue & White)	SLT	01
	Orange - (Orange & White)	SLT	02
	Green - (Green & White)	SLT	03
	Brown - (Brown & White)	SLT	04
RJ45-2	Blue - (Blue & White)	CO	01
	Orange - (Orange & White)	CO	02
RJ45-3	Blue - (Blue & White)	DKP	01
	Orange - (Orange & White)	DKP	02

10. Plug in the RJ45 end of the SLT cables into the respective connectors.

11. Lead the cables out of the enclosure through any of the two cable outlets on either side of the enclosure.

12. Terminate the free end of the cables into the punch down blocks of the Krone modules designated for 'Station Lines' in the Main Distribution Frame (MDF).

    Each wire-pair from the ETERNITY PENX SLT Port must be terminated to the bottom of the Krone Connector, while the wire-pair of the extension line to be connected to this port must be terminated on the top of the Krone connector. Refer the topic *"The Main Distribution Frame (MDF)"* for illustration.

13. If you have completed installing all cards, replace the top cover by sliding it in place. Secure the cover with the two screws you removed.

Connecting SLT instruments

14. Connect the SLT instruments you have arranged for. Plug in the SLTs into the wall socket/outlets.

- *For the purpose of testing, you may connect one or two Single Line Telephone instruments by plugging in the phone cables into the RJ45 connectors on the card.*
- *When you plug the RJ11 connector of SLT into an RJ45 connector on the SLT card, the SLT will be connected to the first port on the connector.*

The Digital Key Phone Card

The Digital Key Phone (DKP) Card provides the interface to connect the proprietary digital key phones of the EON series, the proprietary PC-based phone EONSOFT, the Direct Station Selection (DSS) Consoles, with ETERNITY PENX.

The DKP Card is available in the following configurations for ETERNITY PENX. The DKP interface is available in combination with SLT ports and Two-wire Trunk ports on a single card.

DKP Cards for ETERNITY PENX

Card Name	Configuration and Application
ETERNITY PE Card DKP8	8-port card to connect 8 DKP/DSS Consoles
ETERNITY PE Card DKP2+SLT6	Combination card, with 2 ports to connect to 2 Digital Key Phones and 6 Single Line Telephones
ETERNITY PE Card CO2+DKP2+SLT4	Combination card, with 2 ports to connect to 2 Two-wire Analog trunk lines, 2 ports to connect 2 DKP/DSS Consoles, and 4 ports to connect 4 Single Line Telephones

To connect the proprietary digital key phones with ETERNITY PENX, you must have at least one of the above mentioned DKP Cards installed in the system.

Select a DKP Card with the configuration that meets your requirement for DKP Ports. Also consider the maximum DKP Port capacity of the system you are installing.

The maximum number of DKP ports supported by ETERNITY PENX are 16 DKP ports

Installing the Digital Key Phone Card

Decide the number of DKP extensions and DSS Consoles required and arrange for as many EON, EONSOFT and DSS Consoles.

Decide the locations of the DKP extensions and make sure that the necessary wiring for the DKP extensions, from the wall jack to the MDF, is done.

1. Unpack the DKP card and check the package contents.
2. Make sure that the power supply is switched off, before you begin the installation of the card. Always wear an electrostatic discharge prevention wrist strap/belt and use a grounding mat.
3. Unscrew the top cover of the ETERNITY PENX and slide it out. Keep the cover and the screws aside.
4. Select any of the free slots from the universal slots.
5. Grasping the card by its sides or corners fit it onto the connectors of the selected slot. The card should be seated such that its connector pins make perfect contact with those on the CPU (motherboard) on the bottom plane.
6. Secure the card on the studs labeled H1, H2 and H3 with the three screws provided.

7. Repeat the same steps to install another DKP card. It is not necessary to install the other DKP cards in subsequent slots. You may install the other DKP cards in any of the universal slots.

8. Now, use the cables supplied with the DKP card to connect the DKP wires with the Main Distribution Frame.

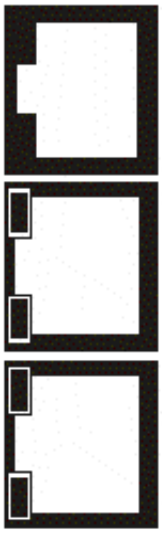
   For each connector on the DKP Card, there is a separate cable with an RJ45 plug on one end and free at the other end.

   Refer the connector pin details for each DKP Card type given in the following.

ETERNITY PE DKP8

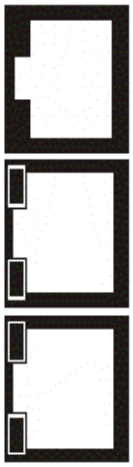
Connector	Color	Connection	H/w port Offset
RJ45-1	Blue - (Blue & White)	DKP	01
	Orange - (Orange & White)	DKP	02
	Green - (Green & White)	DKP	03
	Brown - (Brown & White)	DKP	04
RJ45-2	Blue - (Blue & White)	DKP	05
	Orange - (Orange & White)	DKP	06
RJ45-3	Blue - (Blue & White)	DKP	07
	Orange - (Orange & White)	DKP	08

ETERNITY PE Card DKP2+SLT6



Connector	Color	Connection	H/w port Offset
RJ45-1	Blue - (Blue & White)	SLT	01
	Orange - (Orange & White)	SLT	02
	Green - (Green & White)	SLT	03
	Brown - (Brown & White)	SLT	04
RJ45-2	Blue - (Blue & White)	SLT	05
	Orange - (Orange & White)	SLT	06
RJ45-3	Blue - (Blue & White)	DKP	01
	Orange - (Orange & White)	DKP	02

ETERNITY PE Card CO2+DKP2+SLT4



Connector	Color	Connection	H/w port Offset
RJ45-1	Blue - (Blue & White)	SLT	01
	Orange - (Orange & White)	SLT	02
	Green - (Green & White)	SLT	03
	Brown - (Brown & White)	SLT	04
RJ45-2	Blue - (Blue & White)	CO	01
	Orange - (Orange & White)	CO	02
RJ45-3	Blue - (Blue & White)	DKP	01
	Orange - (Orange & White)	DKP	02

- Plug in the RJ45 end of the MDF cables provided with the DKP card into the respective connectors.
- Terminate the free end of the cables into the punch down blocks of the Krone modules designated for 'Station Lines' in the Main Distribution Frame (MDF).

  Each wire-pair from the ETERNITY PENX DKP Port must be terminated to the bottom of the Krone Connector, while the wire-pair of the extension line to be connected to this port must be terminated on the top of the Krone connector. Refer the topic *"The Main Distribution Frame (MDF)"* for illustration.

9. Connect the Digital Key Phones to the wall jacks at their respective locations. Detailed installations instructions for EON and EONSOFT are provided separately.

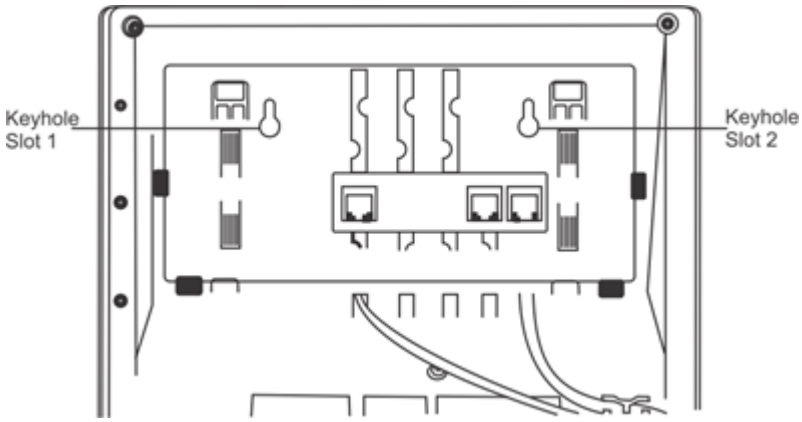
If you have completed all installation tasks, power on the system and observe the Reset Cycle and the LED Pattern of the DKP Card.

Installing EON48

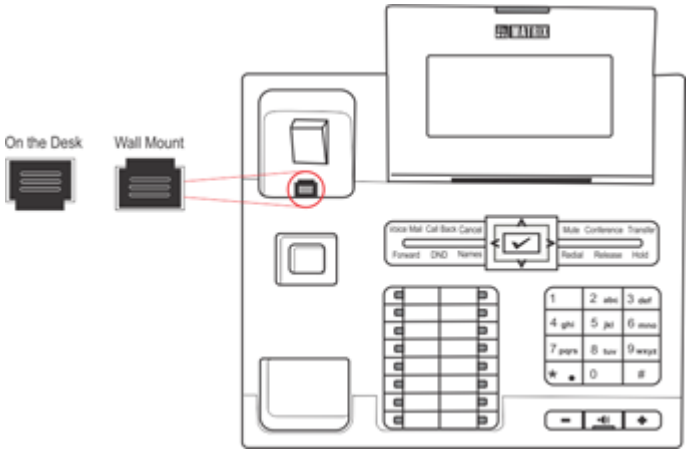
- Unpack the box and verify the package contents.
- You can mount the phone on a wall or on desk.

Mount the phone on a Wall

- Use the mounting template to drill holes of appropriate size and distance.
- Fix the screw grips in the holes you drilled.
- Fix two screws in the holes on the wall, ensuring that they are aligned with the Keyhole Slots 1 and 2. The screws should protrude from the wall to fit into the keyhole slots.

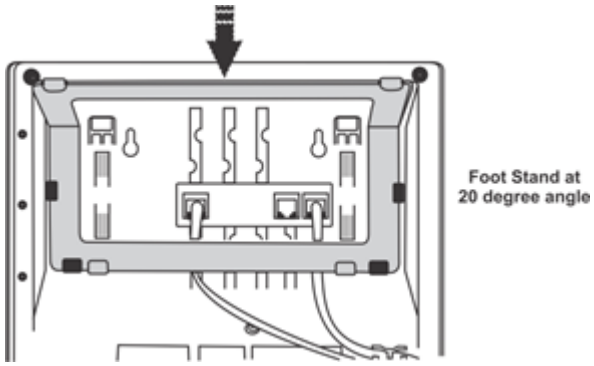


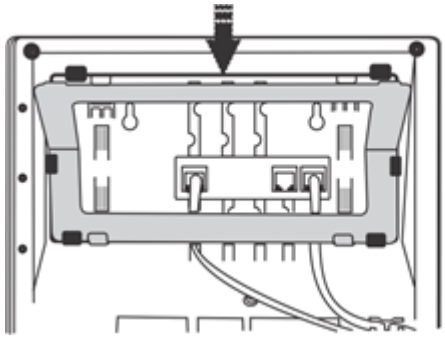
- Now, mount the phone with the screws fitting into the keyhole slots.
- Reverse the handset wall mount tab to make sure that the handset remains intact when you mount the phone. Push the handset wall mount tab upwards to remove it from the slot. Rotate it 180 degrees clockwise and push it downwards into the slot.



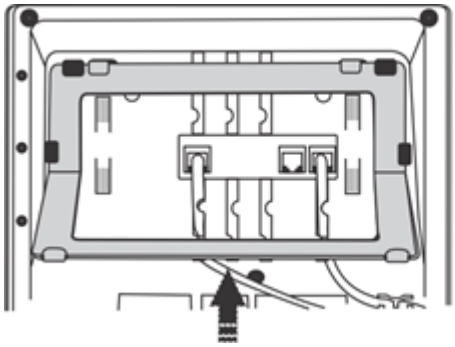
Mount the phone on the Desk

- You can attach the Foot Stand in the following ways—at an angle of **20° Angle** or at **30° Angle** or at **50° Angle**.



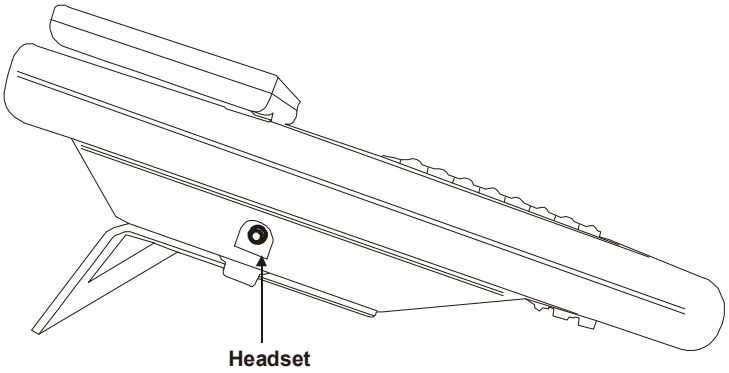


Foot Stand at
30 degree angle



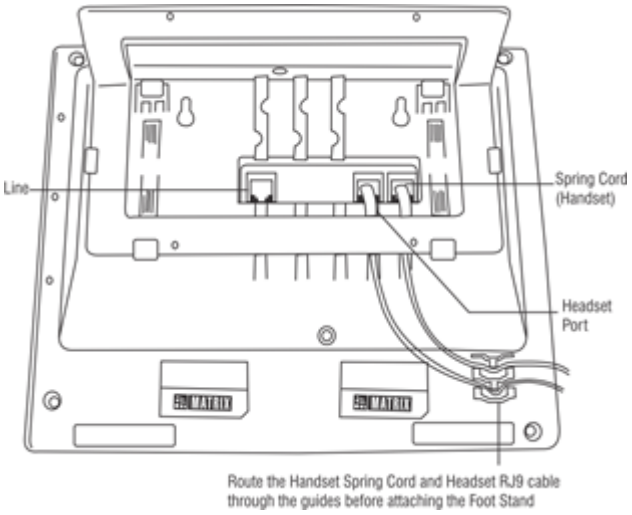
Foot Stand at
50 degree angle

- If you attach the Foot Stand at 50°, the phone will be placed in an almost upright position on your desk.
- Decide which of these positions would work for you best and accordingly attach the Foot Stand.
- Connect the handset of the EON48 to the phone body using the spring cord.
- To use a Headset (not supplied with the phone), plug any standard stereo headset with 2.5mm single connector into the headset jack with the symbol on the left side panel of the phone.

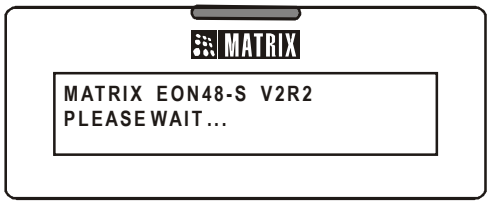


You may also plug in a stereo headset with an RJ9 connector into the headset port at the bottom of the phone, marked with the symbol .

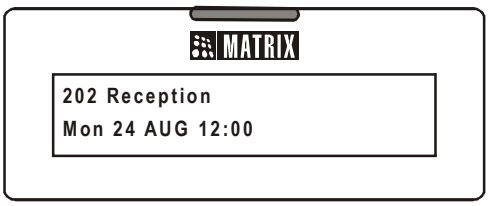
- Plug one end of the RJ11 cable supplied with the phone into the RJ11 connector of the phone labeled as '**LINE**' and the other end into the wall jack/DKP Port.



- When the ETERNITY PENX is powered ON, the EON will reset. The EON communicates with the ETERNITY. The handshaking lasts for 5-6 seconds. The EON model, version and revision number, along with the message *'Please wait'…* appear on the LCD display.



- After successful handshaking and reset cycle, if the DKP Parameters have been configured, the LCD display of the EON will show the extension number and the extension name in one line and the day, date and time and the time zone in the other line.



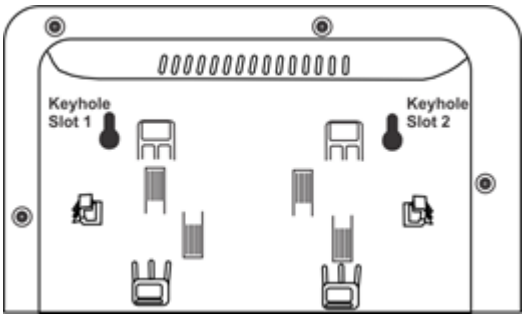
- You may adjust the LCD for brightness, contrast and backlight. Refer the topic, *"Digital Key Phone-Operation"* for instructions.

Installing EON310

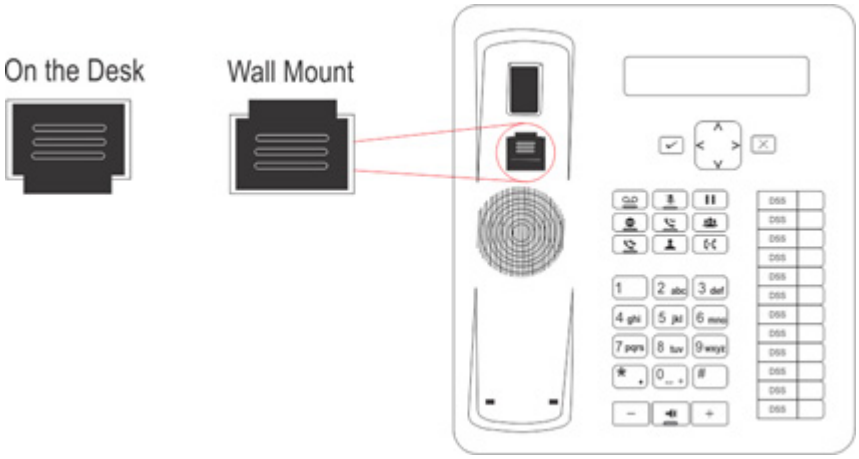
- Unpack the box and verify the package contents.
- You can mount the phone on a wall or on desk.

Mount the phone on a Wall

- Use the mounting template to drill holes of appropriate size and distance.
- Fix the screw grips in the holes you drilled.
- Fix two screws in the holes on the wall, ensuring that they are aligned with the Keyhole Slots 1 and 2 . The screws should protrude from the wall to fit into the keyhole slots.

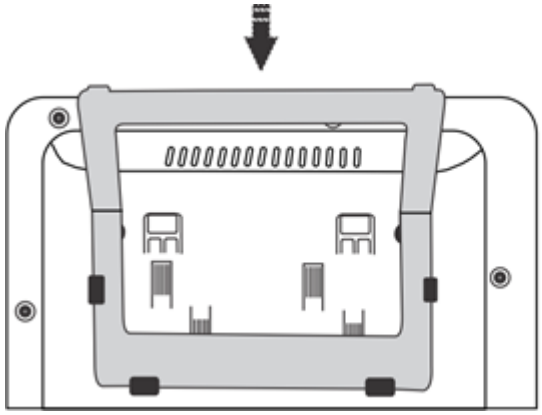


- Now, mount the phone with the screws fitting into the keyhole slots.
- Reverse the handset wall mount tab to make sure that the handset remains intact when you mount the phone. Push the handset wall mount tab upwards to remove it from the slot. Rotate it 180 degrees clockwise and push it downwards into the slot.

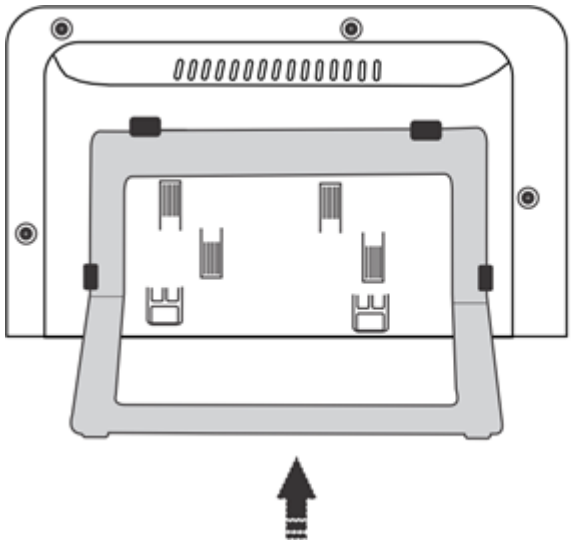


Mount the phone on the Desk

- You can attach the Foot Stand in the following ways—at an angle of **35° Angle** or at **50° Angle**.



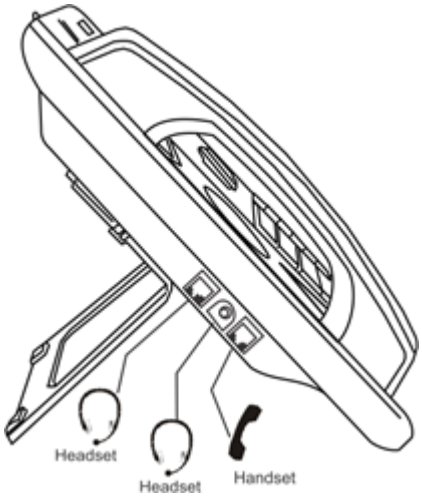
Stand attached at
35 degree angle



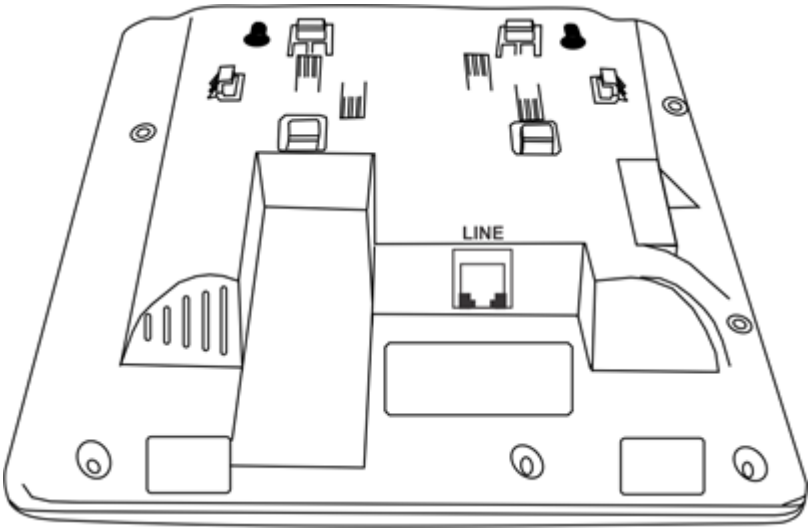
Stand attached at
50 degree angle

- Decide which of these positions would work for you best and accordingly attach the Foot Stand.
- Connect the handset of the EON310 to the phone body using the spring cord.
- To use a Headset (not supplied with the phone), plug any standard stereo headset with 3.5mm single connector into the headset jack with the symbol on the left side panel of the phone.

  You may also plug in a stereo headset with an RJ9 connector into the headset port marked with the symbol , on the left side panel of the phone.



- Plug one end of the RJ11 cable supplied with the phone into the RJ11 connector of the phone labeled as '**LINE**' and the other end into the wall jack/DKP Port.



- When the ETERNITY PENX is powered ON, the EON will get reset and the message "Welcome to Matrix. Booting" appears on the LCD display.

- The EON communicates with the ETERNITY PENX. The handshaking lasts for 5-6 seconds. The EON model, version and revision number, along with the message “Please Wait” appears on the LCD display.

- After successful handshaking and reset cycle, the extension number, day, date and time will appear on the LCD of the phone. If you have already assigned extension number and name, in the DKP Parameters, these will appear on the LCD.

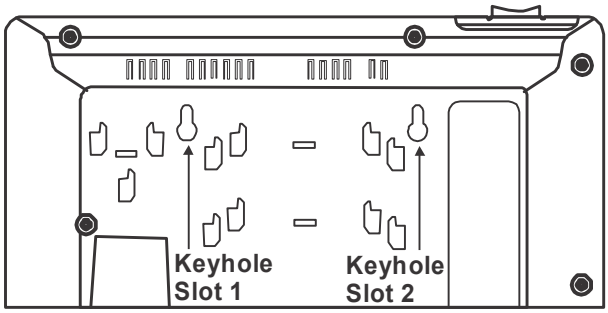
- You may adjust the LCD for brightness, contrast and backlight. Refer the topic,*“Digital Key Phone-Operation”*.

Installing EON510

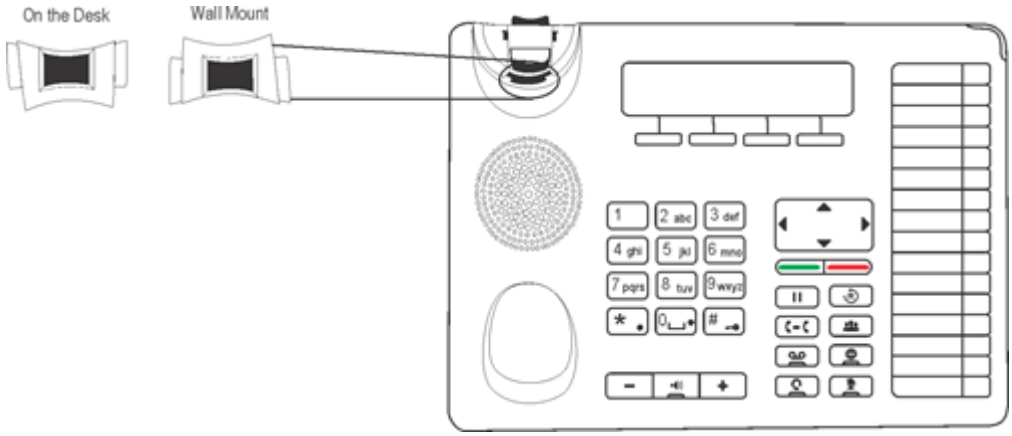
- Unpack the box and verify the package contents.
- You can mount the phone on a wall or on desk.

Mount the phone on a Wall

- Use the mounting template to drill holes of appropriate size and distance.
- Fix the screw grips in the holes you drilled.
- Fix two screws in the holes on the wall, ensuring that they are aligned with the Keyhole Slots 1 and 2 . The screws should protrude from the wall to fit into the keyhole slots.



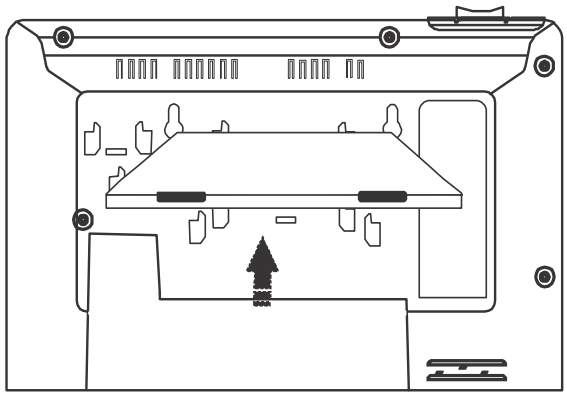
- Now, mount the phone with the screws fitting into the keyhole slots.
- Reverse the handset wall mount tab to make sure that the handset remains intact when you mount the phone. Push the handset wall mount tab upwards to remove it from the slot. Rotate it 180 degrees clockwise and push it downwards into the slot.

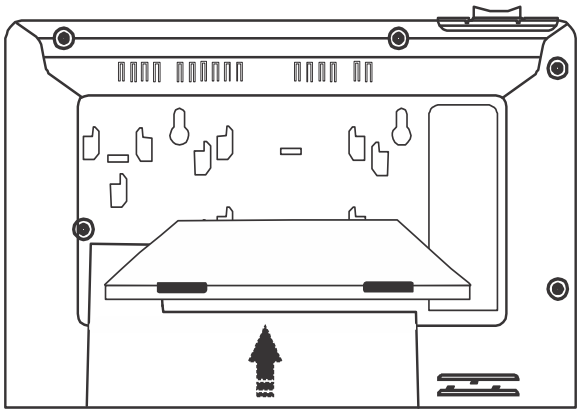


If you are unable to remove the wall mount tab, you may use a tool like a minus screw driver to remove it.

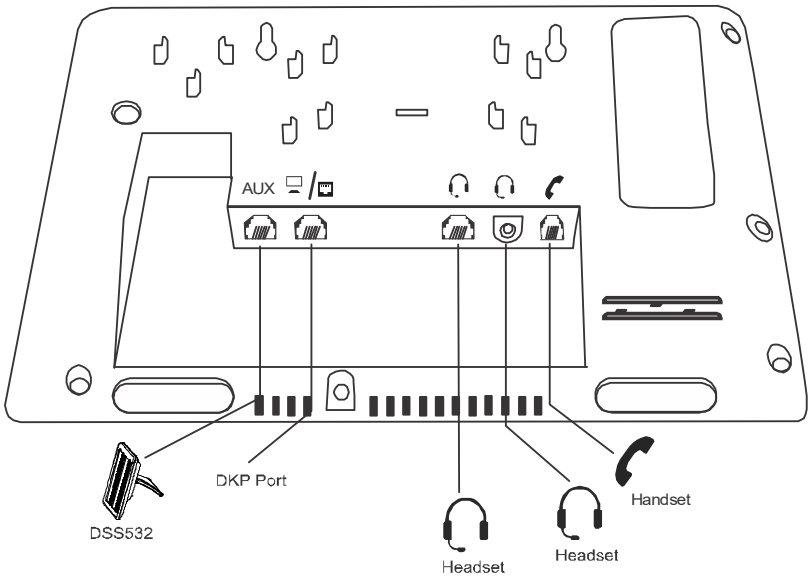
Mount the phone on the Desk

- You can attach the Foot Stand in the following ways—at an angle of **45° Angle** or at **55° Angle**.





- Decide which of these positions would work for you best and accordingly attach the Foot Stand.



- Plug the long straightened end of the Spring Cord into the handset jack at the bottom of the phone, marked with the handset symbol .
- Plug the other (short straight) end of the Spring Cord into the jack at the bottom of the handset.
- To use a Headset (not supplied with the phone), plug any standard stereo headset with 3.5mm single connector into the headset jack with the symbol on the left side panel of the phone.
  You may also plug in a stereo headset with an RJ9 connector into the headset port marked with the symbol , on the left side panel of the phone.
- Plug one end of the RJ11 cable supplied with the phone into the RJ11 connector of the phone marked with the symbol and the other end into the wall jack/DKP Port.

- To connect DSS532 with the phone, plug one end of the RJ11 cable into the AUX Port of the phone and the other end into the IN Port of the DSS532. For installation, see *"Installing DSS532 with EON510"*.

- When the ETERNITY PENX is powered ON, the EON will get reset.and the message 'Phone is Booting; Please wait…' appear on the LCD display.

- The EON communicates with the ETERNITY PENX. The handshaking lasts for 5-6 seconds. The message 'Loading Firmware Version-Revision; Please wait…' appear on the LCD display.

- After successful handshaking and reset cycle, the extension number, day, date and time will appear on the LCD of the phone. If you have already assigned extension number and name, in the DKP Parameters, these will appear, as illustrated below.

  You may adjust the LCD for brightness, contrast and backlight. Refer the topic,*"The Digital Key Phone Card"*.

Installing DSS Consoles

Installing DSS64

Once you have installed EON with ETERNITY PENX, installing DSS Consoles can be done in a few simple steps, very much similar to those involved in the installation of EON.

1. Unpack the box and verify the package contents[104].

2. Place the DSS Console next to the DKP, to which it is to be attached.

   You can install two DSS consoles to a DKP. Refer *"Direct Station Selection Console"* for possible combinations for installing the models of DSS Consoles.

---

104. See "Packing List" under Appendix topic.

3. Decide which DKP Ports on the DKP Card are to be assigned to the DSS Consoles. You may select any free (unused) port on the card for DSS Consoles. It is not necessary for the DSS Console ports to be in a sequence with the DKP ports to which they are attached.

   For example: you have connected DKP1 to Port 1 on the first RJ45 connector of the DKP8 card. You want to attach two DSS Consoles to DKP1. The two DSS Consoles may be connected to any port on the second connector of the card, not necessarily to Port 2 and Port 3 on the first connector.

4. The wire-pairs from the DKP Ports designated for DSS Consoles should be terminated on the bottom of the Krone Connector (of 'Station Lines' on the MDF).

5. The wire-pairs of the DSS Consoles should be terminated into the top of the Krone Connector (of 'Station Lines' on the MDF). Refer the topic *"The Main Distribution Frame (MDF)"* for illustration.

   You can connect maximum two DSS64 with a single EON48/310.

6. The system automatically detects the DSS Console you connect and it will be will appear under **Unassigned DSS64** in *"DSS Status"*. You must first assign these DSS Consoles to the respective DKP Ports and thereafter you will be able to configure the DSS Keys.

7. To assign the DSS Consoles, see *"DSS Status"* and to configure the DSS Keys, see *"Programming DSS Console Keys"*.

Installing DSS532 with EON510

The instructions for installing DSS532 with EON510 or SPARSH VP510 are the same. For detailed instructions, refer to *"Installing DSS532 with SPARSH VP510"*.

Installing EONSOFT

To install EONSOFT,

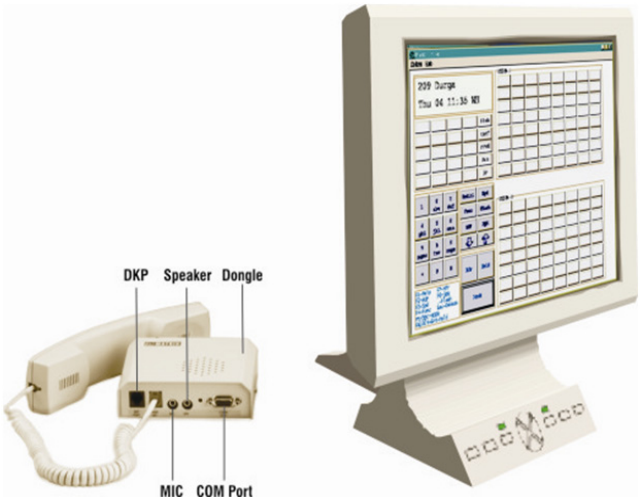
- Contact Matrix Support Team for the EONSOFT Installation Software Application.

- You must have a computer with Windows as the operating system. The EONSOFT is compatible with the following Operating Systems of Windows:
  - Windows 98
  - Windows XP
  - Windows NT
  - Windows 2003
  - Windows Vista
  - Windows 2007

1. Unpack the box and verify the package contents[105].

2. Connect the Handset to the dongle in the handset jack. If using a headset, connect the microphone and the speaker connectors into the dongle.

105. See "EONSOFT" under 'Packing List' of Appendix topic.



3. Connect one end of the Communication cable to the COM port of the dongle. Connect the other end of the communication cable into the COM port of the computer.

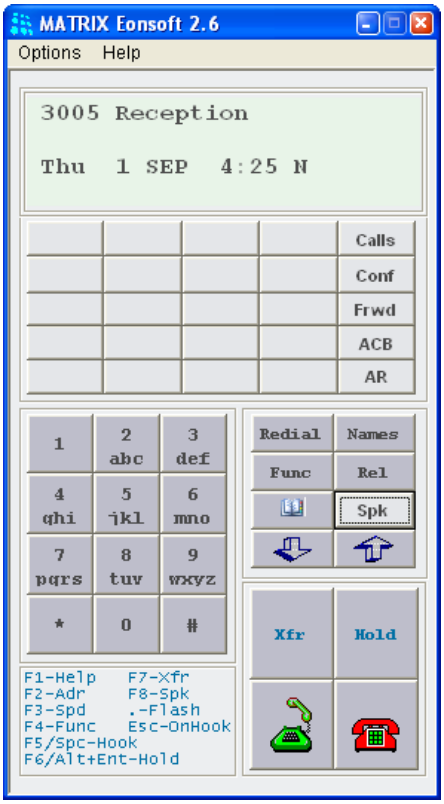
4. Connect a wire-pair of a DKP port of the ETERNITY PENX to the RJ11 port marked 'DKP' on the dongle.

5. Switch ON the computer. The computer must have Windows Operating System installed on it. If you do not have of the operating systems mentioned above, install any compatible Windows Operating System.

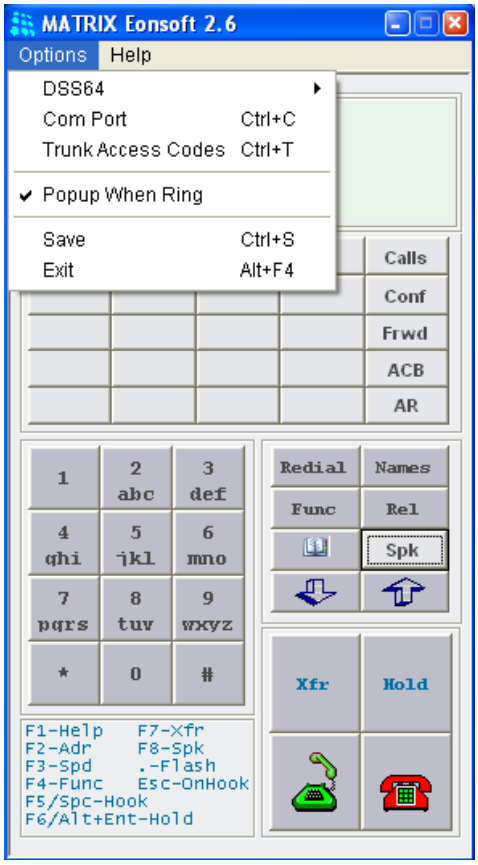
6. Copy the EONSOFT Application Software provided by the Support Team onto your PC and install the application.

7. After the program has been installed and run, a shortcut will be automatically created and appear on your desktop.

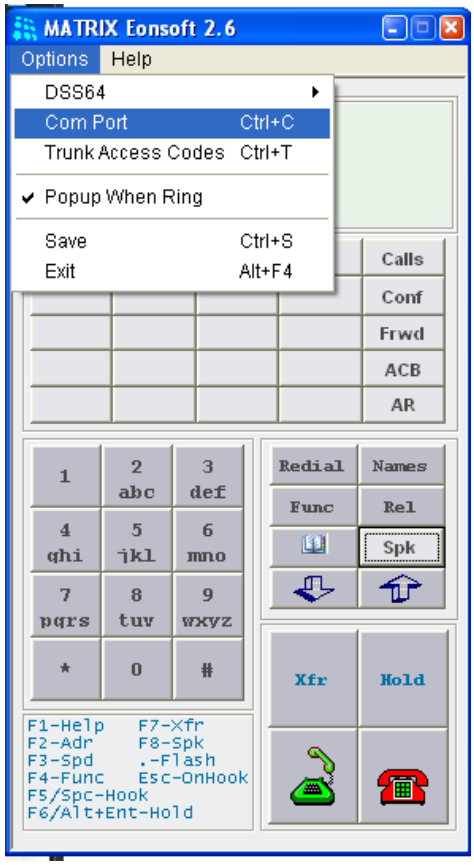
8. Click the shortcut to open the program. The EONSOFT window will open:



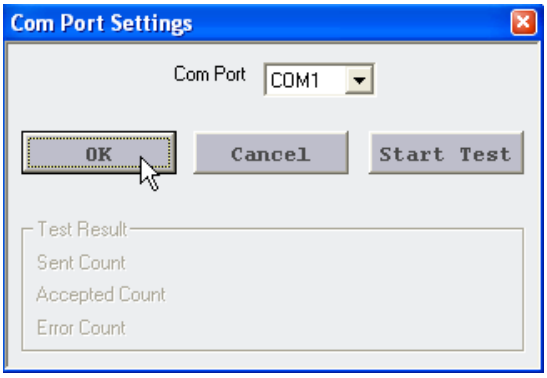
9. Click **Options** at the top left of the window. A drop down menu will appear.



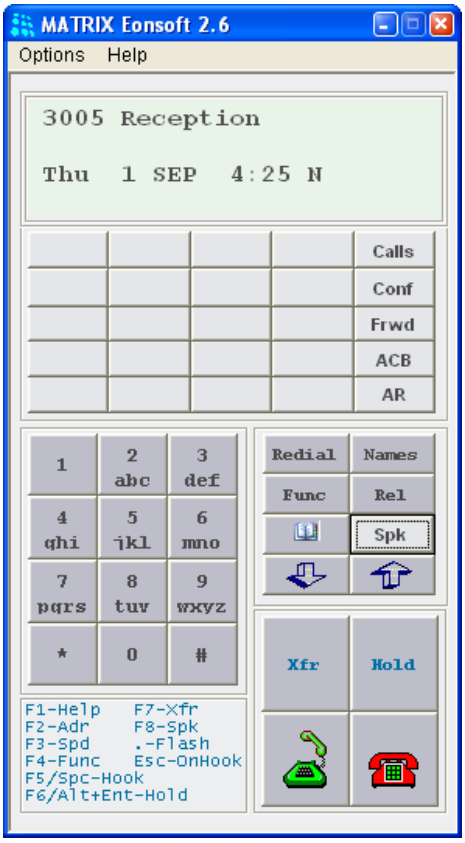
10. Click the option **COM Port**.



11. Select the COM Port to which the communication cable is connected.



12. EONSOFT is now connected. If you have already configured the DKP parameters like Access Code and Name for the port to which EONSOFT is connected, these will appear.



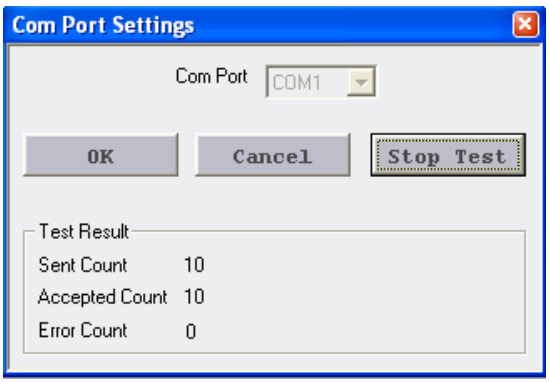
- *If this window does not appear after you have selected the COM Port Option, test the COM Port for data transfer.*
- *If the wrong COM port has been selected, a dialog box will pop up on your screen with the message: "COMx is invalid or busy, please select another COM Port". Select the correct COM Port.*

Test the functioning of the COM Port of the PC and the communication cable, before you install the EONSOFT.

Testing the COM Port

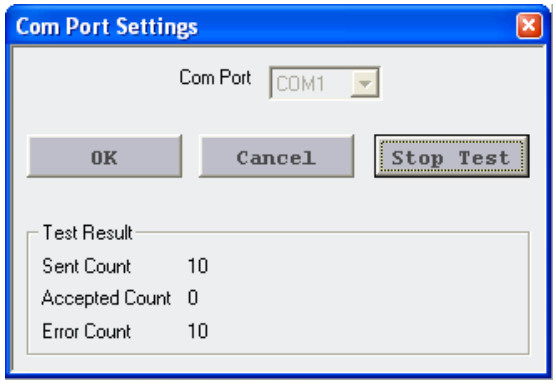
- From the drop down menu of 'Options', select the 'COM Port' to which you have connected the communication cable.
- The COM Port Settings dialog box will open.
- Connect the communication cable to the COM Port of the PC.
- Short pin2 and pin3 of the DB-9 connector at the free end of the cable.
- Click the button labeled 'Start Test' in the COM Port Settings dialog box.
- After clicking this button, observe the Test Result section in the dialog box.

- The **Error Count** value shows zero as value, if both the communication cable and the COM port are working.



The above screen shows that the COM Port/communication cable is working.

- If the **Error Count** shows a value other than zero, it means that either the communication cable or the COM port of the PC is faulty.



The above screen shows the faulty COM Port/Communication Cable.

- Remove the communication cable from the COM Port of the PC.
- Short pin2 and pin3 of the communication port of the computer and click 'Start Test' in the COM Port Settings dialog box.
- Now, if the error count is zero, please check the Communication Cable.
- If the error count is not a zero, the COM Port of the PC is faulty. Try another communication port.

The CO Card

The CO Card provides the interface to connect the ETERNITY PENX with the Two-Wire Analog Trunk lines from the CO Network. The CO Card supports the different standards and features of CO Networks across the world.

The CO Card is available in the following configurations for the variants of ETERNITY PENX. CO interface is also available in combination with SLT ports and with Digital Key Phone ports on a single card.

CO Cards for ETERNITY PENX

Card Name	Configuration and Application
ETERNITY PE Card CO8	8-port card to connect 8 Two-wire analog Trunk lines from the CO network
ETERNITY PE Card CO4+SLT4	Combination card, with 4 CO ports to connect 4 Two-wire analog Trunk lines from the CO network, and 4 SLT ports to connect 4 Single Line Telephones
ETERNITY PE Card CO2+DKP2+SLT4	Combination card, with 2 ports to connect to 2 Two-wire Analog trunk lines, 2 ports to connect 2 DKP/DSS Consoles, and 4 ports to connect 4 Single Line Telephones
ETERNITY PE Card CO2+SLT6	Combination card with 2 ports to connect 2 Two-wire Analog trunk lines, and 6 ports to connect 6 Single Line Telephones

Choose a CO Card with the configuration that meets your requirement for CO trunk ports, keeping in mind the maximum CO Trunk Port capacity of the system you are installing.

The maximum number of CO ports supported by ETERNITY PENX are 16 CO ports.

Connectors

The CO Card has RJ45 connectors, with 4 CO ports on each connector. A multi-pair, MDF cable is supplied for each connector on the card.

Installing the CO Card

For CO connectivity, you must install at least one of the above mentioned CO Cards in the system.

1. Take all the necessary precautions prescribed for handling the cards and electronic equipment. Make sure that power supply is turned off, and the power cord is unplugged before you begin the installation of the card. Put on an electrostatic-discharge preventive wrist strap/belt and use a grounding mat.
2. Unpack the CO card and check the package contents.
3. Unscrew and remove the top cover of the ETERNITY PENX, and keep it aside with the screws.
4. Select any of the free slots from the universal slots.
5. Seat the card onto the connectors of the selected slot. The connector pins of the card should make perfect contact with those on the CPU (motherboard) on the bottom plane.
6. Secure the card on the studs labeled H1, H2 and H3 with the three screws provided.

7. Repeat the same steps to install another CO card. You may install the other CO cards in any of the universal slots, but not necessarily in a sequence.

8. Now, use the cables supplied with the CO card to connect the CO ports with the Main Distribution Frame.

   For each connector on the CO Card, there is a separate cable with an RJ45 plug on one end and free at the other end.

9. Plug in the RJ45 end of the CO cables into the respective connectors. Refer to the pinouts of the connectors illustrated below for each CO Card type.

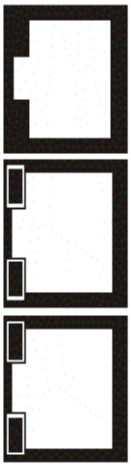
ETERNITY PE Card CO8

Connector	Color	Connection	H/w port Offset
RJ45-1	Blue - (Blue & White)'	CO	01
	Orange - (Orange & White)	CO	02
	Green - (Green & White)	CO	03
	Brown - (Brown & White)	CO	04
RJ45-2	Blue - (Blue & White)	CO	05
	Orange - (Orange & White)	CO	06
RJ45-3	Blue - (Blue & White)	CO	07
	Orange - (Orange & White)	CO	08

ETERNITY PE Card CO4+SLT4

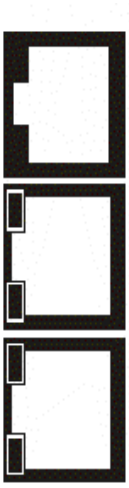
Connector	Color	Connection	H/w port Offset
RJ45-1	Blue - (Blue & White)	SLT	01
	Orange - (Orange & White)	SLT	02
	Green - (Green & White)	SLT	03
	Brown - (Brown & White)	SLT	04
RJ45-2	Blue - (Blue & White)	CO	01
	Orange - (Orange & White)	CO	02
RJ45-3	Blue - (Blue & White)	CO	03
	Orange - (Orange & White)	CO	04

ETERNITY PE Card CO2+SLT6



Connector	Color	Connection	H/w port Offset
RJ45-1	Blue - (Blue & White)	SLT	01
	Orange - (Orange & White)	SLT	02
	Green - (Green & White)	SLT	03
	Brown - (Brown & White)	SLT	04
RJ45-2	Blue - (Blue & White)	SLT	05
	Orange - (Orange & White)	SLT	06
RJ45-3	Blue - (Blue & White)	CO	01
	Orange - (Orange & White)	CO	02

ETERNITY PE Card CO2+DKP2+SLT4



Connector	Color	Connection	H/w port Offset
RJ45-1	Blue - (Blue & White)	SLT	01
	Orange - (Orange & White)	SLT	02
	Green - (Green & White)	SLT	03
	Brown - (Brown & White)	SLT	04
RJ45-2	Blue - (Blue & White)	CO	01
	Orange - (Orange & White)	CO	02
RJ45-3	Blue - (Blue & White)	DKP	01
	Orange - (Orange & White)	DKP	02

10. Lead the cables out of the enclosure through any of the two cable outlets on either side of the enclosure.

11. Terminate the free end of the cables into the punch down blocks of the Krone modules designated for 'Trunk Lines' in the Main Distribution Frame (MDF).

    Each wire-pair from the ETERNITY PENX CO Port must be terminated to the bottom of the Krone Connector, while the wire-pair of the trunk line from the CO Network to be connected to this port must be terminated on the top of the Krone connector.

    Refer the topics *"The Main Distribution Frame (MDF)"* and *"Terminating Trunk and Extension Cables on the MDF"*.

12. If you have completed installing all cards, replace the top cover by sliding it in place. Secure the cover with the two screws you removed.

The T1E1PRI Card

The ETERNITY PENX T1E1PRI Card provides the interface to connect ETERNITY PENX to ISDN Network.

When connected to T1 carrier lines, the Card supports the following signaling types:

- PRI
- Robbed Bit Signaling
- Q-Signaling (QSIG)

When connected to E1 carrier lines, the card supports the following signaling types:

- PRI
- Channel Associated Signaling (CAS)
- Q-Signaling (QSIG)

The T1E1PRI Card is available in the following configuration for ETERNITY PENX:

T1E1PRI Card for ETERNITY PENX

Card Name	Configuration and Application
ETERNITY PE Card T1E1PRI Single	1-port card with QSIG support to connect 1 ISDN T1/E1 PRI Line or ISDN Compatible Device

The maximum number of ISDN PRI lines supported by Eternity PENX are 2 T1/E1 PRI Lines.

Connectors

The T1E1PRI card has an RJ45 Connector. A cable with RJ45 plugs on both ends is supplied with the card.

Installing the T1E1PRI Card

1. Before installing the card, take the necessary precautions prescribed for handling the cards. Always wear an electrostatic-discharge preventive wrist strap and use a grounding mat. Make sure the power supply is turned off.

2. Unpack the T1E1PRI card and check the package contents.

3. Unscrew and remove the top cover of the ETERNITY PENX (if not opened already). Keep the cover and the screws aside.

4. Select any free slot from the Universal Slots.

5. Grasp the card by its sides or corners. Fit the card's connectors into the connectors of the selected slot. Ensure that the card's connector pins make perfect contact with those on the CPU on the bottom plane.

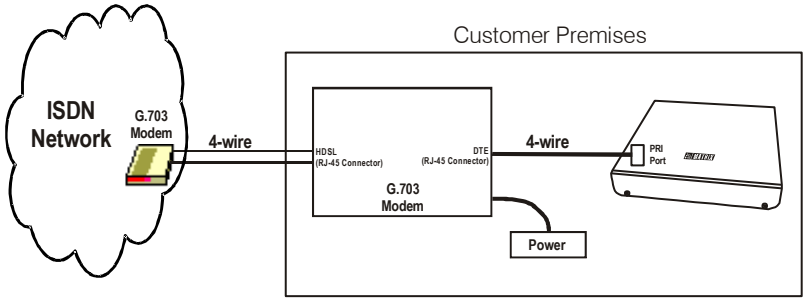
6. Secure the card on the studs labeled H1, H2 and H3 with the three screws provided.

7. Repeat the same steps to install another T1E1PRI card. You may install the other T1E1PRI cards in any of the universal slots, but not necessarily in a sequence. Any card can be inserted in any of the universal slots.

Connecting ISDN T1/E1 PRI Lines

8. Use the cable supplied with the T1E1PRI Card to connect the ETERNITY PENX to the T1/E1 PRI network interface equipment (modem), which is usually supplied by your ISDN Service Provider along with the PRI line.

   The block diagram illustrates this.



   - Most Service Providers insist on connecting an ISDN modem at both the ends of the PRI line, one at the Local Exchange and other at the Customer's Premises.
   - At the Customer's Premises, the PRI line is terminated on the HDSL interface of the modem.
   - The DTE interface of the modem is to be connected to the PRI port (RJ45 connector on the Matrix ETERNITY PENX T1E1PRI Single Card).

9. Plug in one end of the RJ45 cable supplied with the card into the card's connector. Lead the cable out of the enclosure through any of the two cable outlets on either side of the enclosure.

10. Plug the other end of the RJ45 cable into the Network Termination Unit.

11. Refer the following pin details for connecting the Network Termination Unit with the ETERNITY PENX.

    Pin details of HDSL Interface of the G.703 Modem. (HDSL Network Termination Unit)

Pin Number	Pin Details
1	Line A
2	Line A
3	Not used
4	Line B
5	Line B
6	Not used
7	Not used
8	Not used

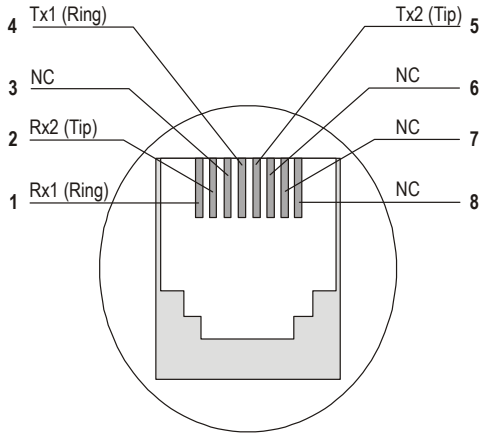
Pin details of DTE Interface of G.703 Modem. (HDSL Network Interface Unit)

Pin Number	Pin Details
1	TX1 (Tip)
2	TX2 (Ring)
3	Not used
4	RX1 (Ring)
5	RX2 (Tip)
6	Not used
7	Not used
8	Not used

Most of the HDSL Network Termination Unit manufacturers use these connectors. But you are advised to read the installation guide of the HDSL Network Termination Unit being used by you.

Pin details of ETERNITY T1E1PR1 Port

The T1E1PRI Port of the ETERNITY PENX terminates in an 8-pin RJ45, female connector and is wired according to the figure below.



The cable wires may have to be crossed depending on the pin-out of the DTE Interface of the modem.

Setting Line Termination Resistor

12. Termination Resistance of the PRI Port for T1 or E1 Connectivity is set by changing the position of the Switch (S1) as given in the table below:

Switch (S1)	Meaning
ON	To set termination resistance of 100Ω for T1 connectivity
OFF	To set termination resistance of 120Ω for E1 connectivity

13. Repeat the same steps to connect another PRI card, if installed.

14. If you have no other card to install, replace the top cover, by sliding it in place. Secure the cover with the two screws you removed.

15. If you have completed all other installation tasks. Power the system and observe the Reset Cycle.

The Mobile Card

The Mobile Card interfaces the system with GSM/3G/LTE networks. It routes calls made and received over mobile networks, like a mobile handset.

The card does not support GPRS features, Fax and Data services, network supported services, except CLIR and USSD.

For compatibility and use of Matrix GSM products (2G/3G/4G) in Russia and Iran Province connect with Matrix Sales or Technical Support Team.

The Mobile card is available in the following configuration for ETERNITY PENX.

The Mobile Card for ETERNITY PENX

Card Name	Configuration and Application
ETERNITY PE Card GSM4	4-port card to connect to the GSM network (4 SIM Cards can be installed)
ETERNITY PE Card GSM4 3G	4-port card to connect to 3G network (4 SIM Cards can be installed)
ETERNITY PE Card GSM4 4G	4-port card to connect to 4G network (4 SIM Cards can be installed)

Just like mobile handsets, each Mobile Port has a unique IMEI (International Mobile Equipment Identity) number, pasted on the mobile engine.

The maximum number of Mobile ports trunks supported by ETERNITY PENX are 8 GSM Ports.

SIM of different service providers can be used.

Antenna

For all four mobile ports, there are four antennas with male connectors on the card. Antenna cables are also provided.

Personal Identification Number (PIN)

The SIM cards can be protected from unauthorized use by programming a Personal Identification Number (PIN) on the SIM. If the wrong SIM PIN is entered thrice in a row, by a user, the SIM Card suspects the user and asks for the Personal Unlock Keyword (PUK).

Installing the Mobile Card

1. Before you install the Mobile Card, make sure that

   - the ETERNITY is installed at a location where sufficient network coverage is available.
   - the power supply is turned off.

- you are wearing an electrostatic discharge preventive wrist strap and have a grounding mat, before you begin handling the card.

2. Get the SIM Card from the GSM service provider of your choice ready. Use SIM PIN protection, if required.

Disable Call Waiting in the SIM.

3. Unpack the Mobile card and verify the package contents.

SIM PIN Protection

4. Enable SIM PIN to protect your SIM card. Before installing the SIM card in the system, insert the SIM into a mobile handset first. From the mobile handset,

   - enable PIN Protection.

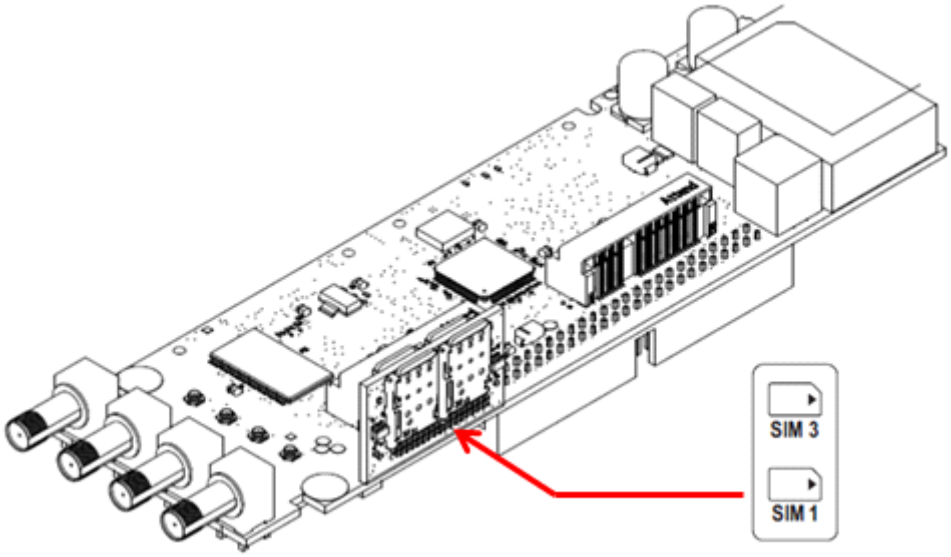
   - change the SIM PIN to 1234 (this is the default PIN for all SIM cards used in the system). Changing the SIM PIN to '1234' enables you to change the SIM PIN from the ETERNITY later. (Refer SIM PIN under *"Configuring Mobile Trunks"* for instructions).

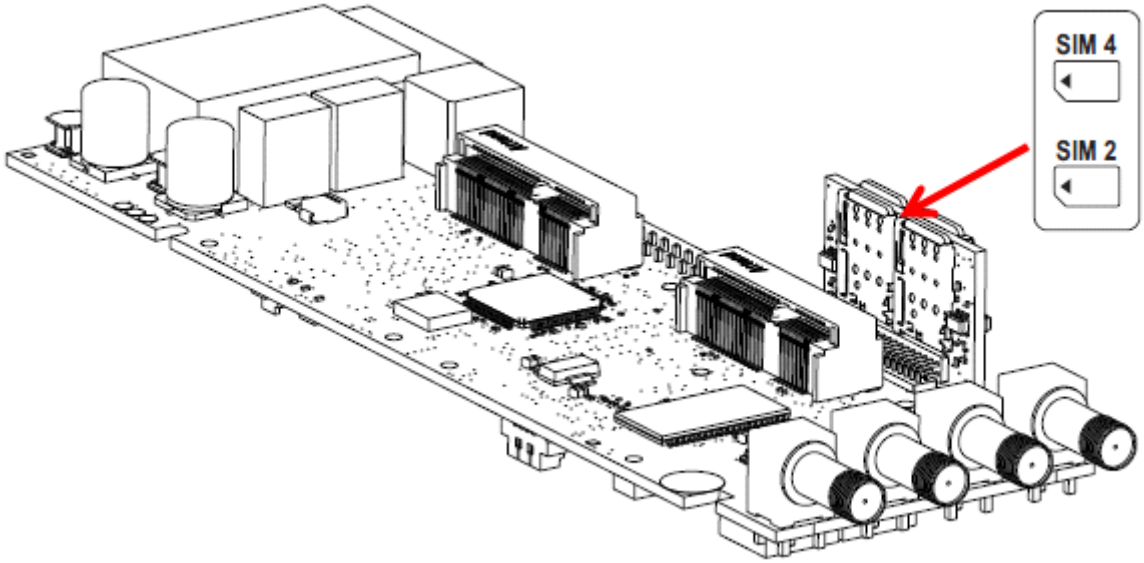
   - remove the SIM from the mobile handset.

If you do not want to use PIN protection, insert the SIM in the mobile handset and disable PIN protection. Remove the SIM Card from the mobile handset.

5. Now, insert the SIM card (PIN changed to 1234), with its connector side down into the SIM holder on the Mobile card. You can insert multiple SIM cards of the same GSM service provider or of different service providers.

ETERNITY PE Card GSM4





6. Remove the top cover of the ETERNITY PENX, if not opened already. Keep the cover and the screws aside.

7. Select any of the free universal slots. Grasping the card by its sides or corners fit it onto the connectors of the selected slot.

   The card should be seated such that its connector pins make perfect contact with those on the CPU (motherboard) on the bottom plane.

8. Secure the card on the studs labeled H1, H2 and H3 with the three screws provided.

9. Screw one end of the Antenna cable onto the Antenna Male Connector on the card.

10. Lead the antenna cable out of the enclosure through any of the two cable outlets on either side of the enclosure. Now, place the antenna at an appropriate location.

11. Repeat the same steps to install another Mobile card, in another free slot. It is not necessary to install the Mobile cards in subsequent slots.

12. If you have completed installing all cards, replace the top cover by sliding it in place. Secure the cover with the two screws you removed.

- *At every power up of the system, it takes about 3 minutes for the Mobile ports to get registered with the network. Once registration with the GSM network is completed, the mobile port can be used.*

- *Each time the Mobile Port sends a request, such as a Registration Request, the system waits for the duration of the Network Response Timer. This Timer signifies the time for which the Mobile Port waits for a response from the GSM network. It is fixed for 150 seconds for all Mobile ports.*

SIP Extensions

SARVAM UCS supports up to 100 SIP/UC Users. The SIP/UC Users function in the same way as DKP/SLT extensions of the system. SIP/UC Users can make and receive calls to any extension user of the system and to external numbers over various telecom networks like CO, Mobile, ISDN PRI and VoIP[106].

You may register any SIP-enabled device — a Matrix UC Client, an IP-phone, a Soft phone, Analog Phone Adapter — as the SIP User of the system.

The Matrix UC Clients also offer UC functionalities in addition to the SIP functionalities.

The SIP Users register with the CPU Card of the system. Five free SIP Users are provided by default. You may register any of the SIP-enabled devices except the Matrix UC Clients with these free SIP Users. For registering the Matrix UC Clients, you must purchase the Matrix VARTA User License. If you require additional SIP Extensions you must purchase the IP Subscribers License.

The system supports one NX DBM VOCODER64 Modules. You must purchase the module separately. NX DBM VOCODER64 module supports a maximum of 64 VOCODER channels. The Vocoder channels are required for — VoIP to Non-VoIP calls, VoIP to VMS calls and VoIP to VoIP calls — where transcoding is required.

The system provides 4 pre-activated VOCODER channels by default. To use these channels make sure you have installed the NX DBM VOCODER64 module.

For more information on Licenses — Matrix VARTA User License, IP Subscribers License and VOCODER Channel License, see *"License Management"*.

You may connect any Standard Phone or Extended IP Phones of Matrix as SIP Users.

Matrix VARTA WIN200, VARTA ADR100 and VARTA AMP100 can be registered as SIP Users, also offering the support for UC functionalities.

You may also connect/register[107] the following as SIP Extensions of the system:

- Connect SPARSH VP248, the Extended IP Phone. For instructions, see "Connecting SPARSH VP248 as Extended SIP Extension".
- Connect SPARSH VP310, the Extended IP Phone. For instructions, see "Connecting SPARSH VP310 as Extended SIP Extension".
- Connect SPARSH VP330, the Touch Screen Extended IP Phone. For instruction,"Connecting SPARSH VP330 as Extended SIP Extension".
- Connect SPARSH VP510, the Premium IP Phone. For instruction, "Connecting SPARSH VP510 as Extended SIP Extension".
- Connect SPARSH VP210, the Entry Level IP Phone. For instruction, "Connecting SPARSH VP210 as Extended SIP Extension".
- Connect Extended SPARSH VP710, the Smart Video IP Phone. For instruction,"Connecting Extended SPARSH VP710 as Extended SIP Extension".

You can register following UC Clients as SIP Users of the system:

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106. Calls between VoIP, Public and Private Networks may be subject to Regulation in your country. You may have to configure your system to allow or restrict call traffic between networks to comply with the telecom regulations of your country. To know more, read "Logical Partition".

107. SARVAM UCS supports only IPv4 Addresses for registering SPARSH VP248, SPARSH VP310, SPARASH VP330, Extended SPARSH VP710, Mobile UC Clients as well as VARTA WIN200.

- Matrix VARTA WIN200, Unified Communication Client for Windows. For instruction, refer to the *MATRIX VARTA WIN200* User Guide.
- Matrix Mobile UC Clients, as given below:
  - Matrix VARTA AMP100, the Mobile UC Client for iPhones. For instruction, refer to the *Matrix VARTA AMP100* User Guide.
  - Matrix VARTA ADR100, the Mobile UC Client for Android Smartphones/Tablets. For instruction, refer to the *Matrix VARTA ADR100* User Guide.

Refer to *"SARVAM UCS Features Supported in Terminals"* to know the features supported in each client.

The SIP Users may be registered over ***WAN*** or over ***LAN*** according to your preference and your IP network installation scenario. Extended SIP Phones and UC Clients can be registered with SARVAM UCS, for details refer to the specific topic.

You can register the same SIP User from three different locations.

If you register the Extended IP Phone outside the Region/Country selected for SARVAM UCS, the time and Time Zone dependent features, such as Alarms, Reminders, Time Zone Display, of the phone at each location will operate according to the Real Time Clock of SARVAM UCS. Also, Access Codes and Emergency Numbers will work according to the Region/Country selected for SARVAM UCS.

- Connect the Extended IP Phone, or any Standard IP Phone to the LAN Switch.

- Register any SIP device (Extended IP phone/ Soft clients or Standard IP phone) on the public network as SIP Extension.

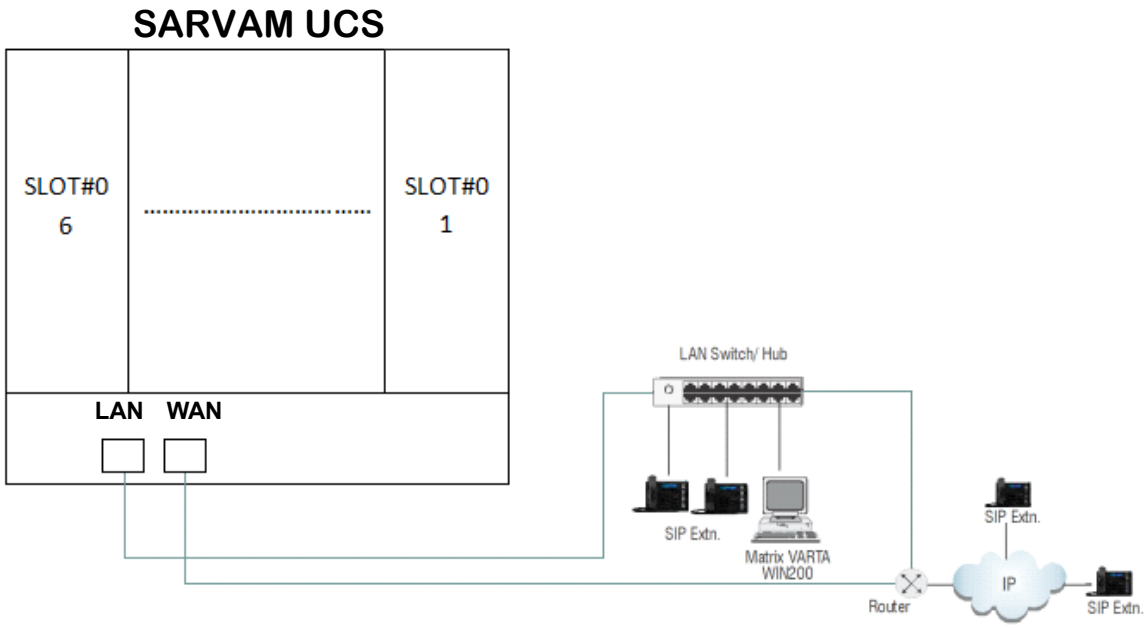
- When you register the Matrix Extended IP Phone with the system, the WAN/LAN port is used for Auto Configuration as well for Registration of the Extended IP Phones.

- When you register a SIP device other than the Matrix Extended IP Phone on the public network as SIP Extension, do the following:
  - In this SIP device configure the following:
    - the Registrar Server Address of SARVAM UCS SMB
    - the Registrar Server Port
    - the SIP ID
    - Authentication ID and Password.

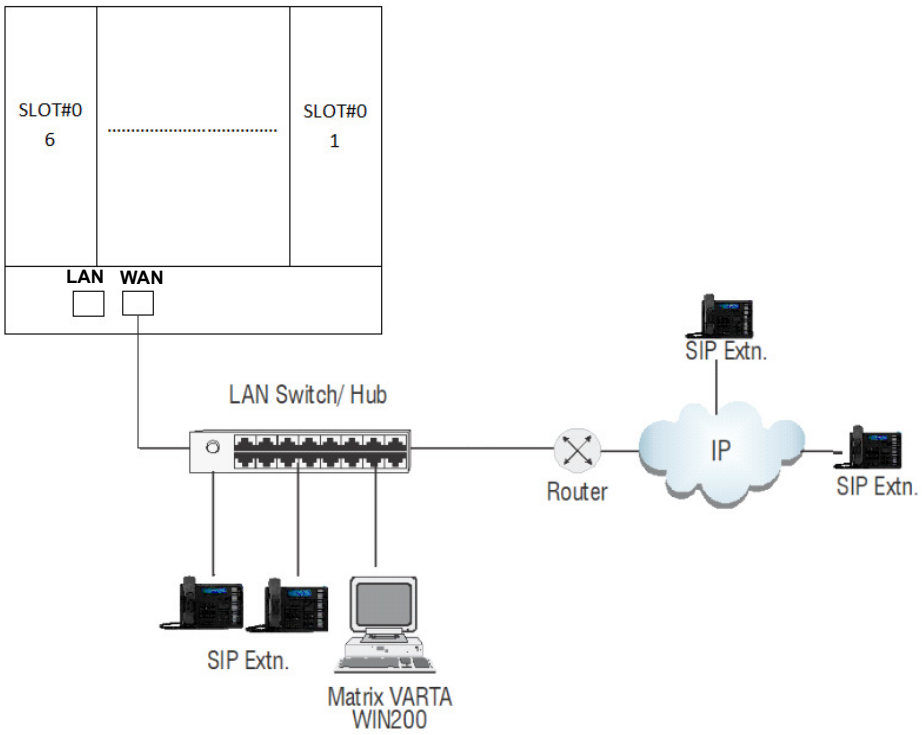
  - Configure Port Forwarding for the WAN Port of SARVAM UCS SMB on the Router.

If the SARVAM UCS is connected to a ***Public Network***,

- Connect the Matrix VARTA WIN200, Extended IP Phone, or any Standard SIP device to the LAN Switch.
- Register any SIP device (Matrix VARTA UC Clients, Extended IP phone or Standard SIP phone) on the public network as SIP extension.



If the SARVAM UCS is connected to a ***Private Network (Behind the NAT)***,



- Connect Matrix VARTA WIN200, Extended IP Phones or Standard SIP Phones to the LAN Switch

- You may also register any SIP device (Matrix VARTA UC Clients, Extended IP Phone or Standard SIP phone) on the public network as SIP Extension.

  When you register the Matrix Extended IP Phone with SARVAM UCS, configure **Port Forwarding** for the **WAN port of the CPU Card** on the Router. The WAN Port is used for Auto Configuration of the Extended IP Phones.

Connecting SPARSH VP248 as Extended SIP Extension

You are recommended to complete the following steps before connecting the Matrix Extended IP Phone to SARVAM UCS:

- Decide the location of the Extended IP Phone, whether within the same network or outside, according to your installation scenario.

*If you want to use the **DHCP Server** for assigning IP Address to the Extended IP Phone, select **DHCP option 224** and **Data Type** as **'String'** and program the LAN or WAN IP Address /Domain Name of SARVAM UCS and SPARSH Port in the format **"IP_Address:Port"** in your DHCP Server as per your installation scenario.*

- Log in to *Jeeves*. For instructions, read the topic *"Configuring SARVAM UCS"*.

- Assign SIP User ID (will work as an extension number) to the Extended IP Phone. For instructions on assigning SIP ID, see *"Configuring SIP Extensions"*.

  For the SIP User ID you assigned to the Extended IP Phone, you must configure the necessary parameters in SARVAM UCS so that Extended IP Phone can register as a SIP Extension. For instructions, see the topic *"Configuring SIP Extension Settings as per the Extended Phone Type"* under *Configuring SIP Extensions*.

Now, follow the steps described below to install the Extended IP Phone. The instructions are common for all models of the SPARSH VP248. For the purpose of illustration, the premium model, SPARSH VP248P, has been used.

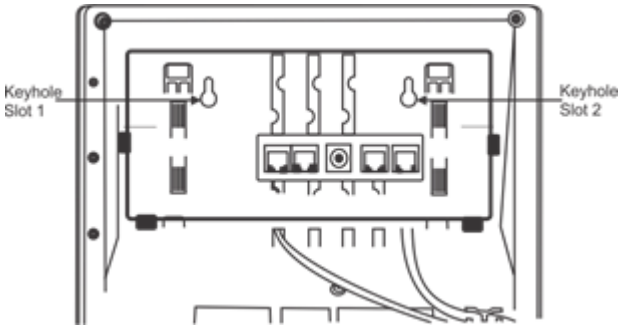
1. Unpack the SPARSH VP248 box and verify package contents.

2. Mount the phone on a desk or wall at a location convenient to you.

   - When mounting the phone on the wall,

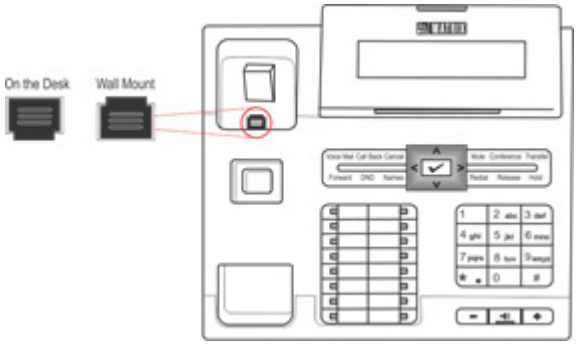
     - Use the mounting template to drill holes of appropriate size and distance. Fix the screw grips in the holes you drilled.
     - Fix two screws in the holes on the wall, ensuring that they are aligned with the Keyhole Slots 1 and 2.

- Use wall plugs, if required, to fix the screws. Leave the screw heads protruding from the wall to fit into the Keyholes.



- Now, mount the phone on the wall, with the screws fitting into the Keyhole slots.

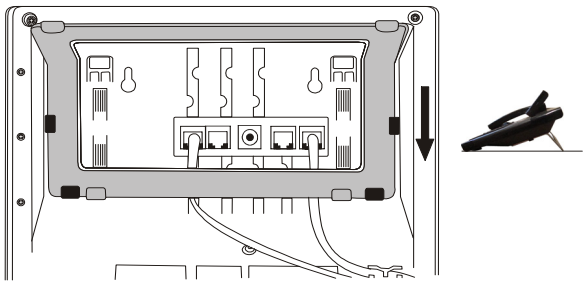
- Reverse the handset wall mount tab to make sure the handset remains intact when you mount the phone. Push the handset wall mount tab upwards to remove it from the slot. Rotate it 180 degrees clockwise and push it downwards into the slot.



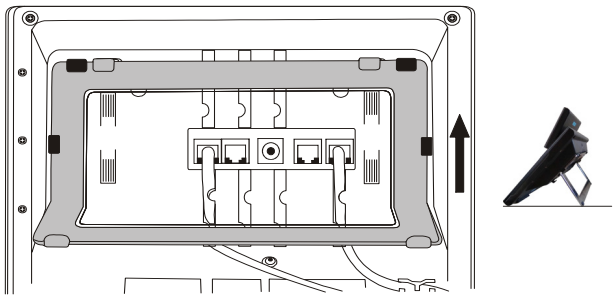
- When you mount the phone on a desk,

  - You can attach the Foot Stand in two ways as illustrated in the following.

  Foot Stand attached at 30° Angle



Foot Stand attached at 50° Angle



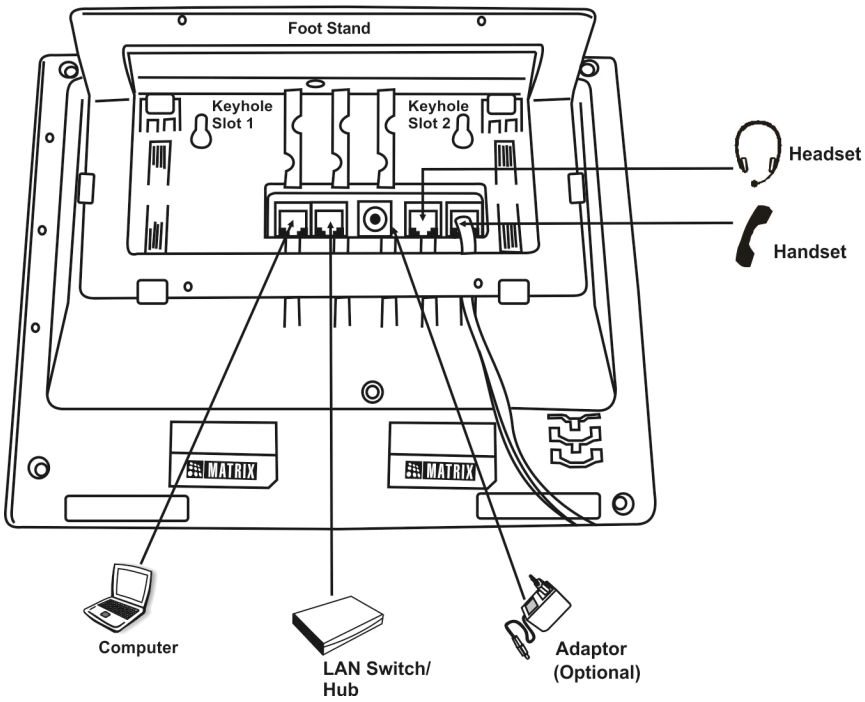
If you attach the Foot Stand at 50°, the phone will be placed in an almost upright position on your desk.

- Decide which of these positions would work for you best and accordingly attach the Foot Stand.

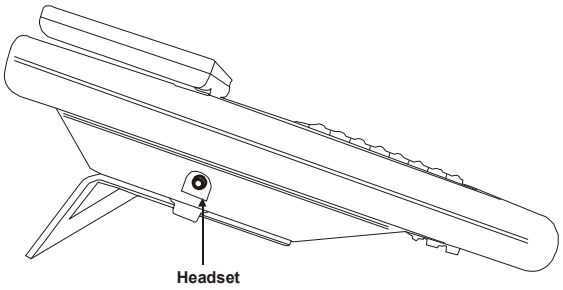
3. Connect the Handset to the Phone body.

- Plug the long straightened end of the phone cord into the handset jack at the bottom of the phone marked with the handset symbol.

- Plug the other (short straight) end of the phone cord into the jack at the bottom of the handset.

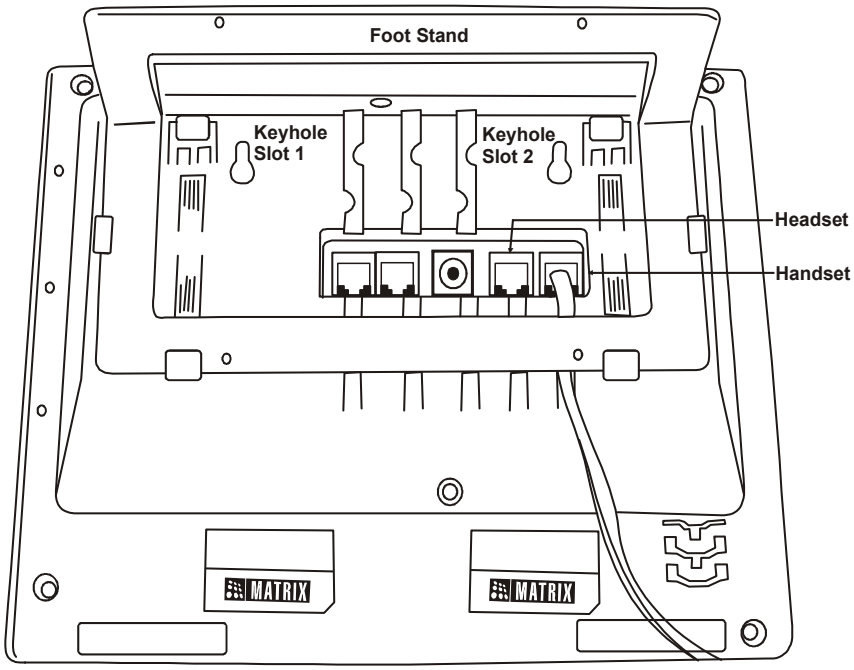


4. If you want to use a Headset (not supplied) with your phone, you may plug a headset with a 2.5 mm single connector into the headset jack headset jack with the symbol on the left side panel of the phone, as illustrated in the figure below.



OR

You may plug a headset with an RJ9 connector into the headset port at the bottom of the phone, marked with the symbol , as illustrated in the figure below.



Connect the LAN Port of SPARSH VP248 to the LAN Switch/Hub or a Router, according to your installation scenario.

5. To connect your phone to a computer on your desk, use an Ethernet cable (not supplied with this phone) to connect the PC Port of the phone to the LAN Port of the computer.

6. It is a PoE enabled phone and can be powered over Ethernet by connecting it to a PoE enabled LAN Switch (IEEE 802.3af Compliant). In this case you need not connect the Power Adapter.

   If you do not want to use PoE, plug in the connector of the Adapter into the power jack (DC Jack) at the bottom of the phone. Plug in the Power Adapter into a power outlet.

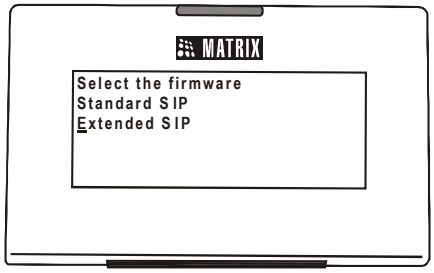
If both the power options, that is, PoE as well as Power Adapter are available to the phone, then the phone will derive power from the PoE enabled LAN Switch.

The IP phone should be used with Matrix original power adapter (5V/2A) only. The use of any third-party power adapter may cause damage to the phone.

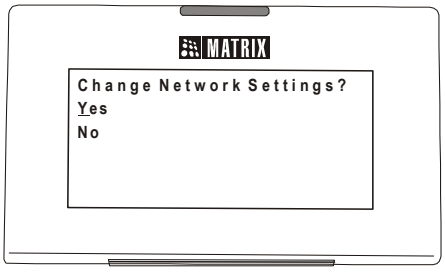
7. Switch ON power supply.

   When you power the phone, the boot process will be initiated in the following sequence.

   - All keys with LED, including the Speaker key, and the Ringer LED, will glow.
   - The LCD display will light up and booting message appears.
   - As soon as the 'Loading...' message appears on the phone display, press **#** key.
   - Select the firmware **Extended - IP Phone**. Move the cursor by pressing the DOWN navigation key **V**.
   - When the cursor is placed under the Extended IP Phone, press Enter key.



- The phone will start loading the Extended IP Phone Firmware. It will display current firmware being loaded.
- After loading the firmware, the phone will prompt you to change Network settings.



- Wait for a few seconds.

If you want to change the Network Settings or Server Settings, press the Enter key. Detailed instructions for changing the Network Settings of the phone are provided at the end of this topic. See "Network Settings" at the end of this topic.

- The phone makes DHCP Discovery and fetches its IP Address and Server Address from the DHCP Server.

On getting the IP Address and Server Address, the phone initiates Auto Configuration to download the configuration files from SARVAM UCS.

- As the phone downloads the configuration files, the file names will appear one by one.
- On successful download of all configuration files, the phone attempts to register with SARVAM UCS.
- On successful registration, the phone will display the current day, date and time, the extension number and name assigned to the Extended IP Phone.

Network Settings

You can change the network settings of the Extended IP Phone by accessing the Local Menu of the phone. To move the cursor and scroll through the menu and submenu options, use the following touch sense navigation keys on your phone.

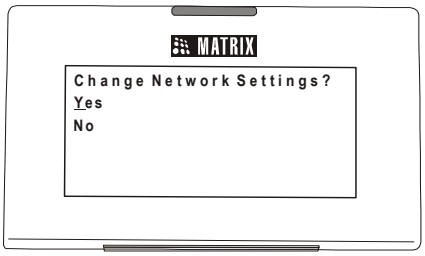
- The Enter key ✓ to make a selection or to complete an action.
- The Up key ∧ to move up the Menu.
- The Down key ∨ move down the Menu.
- The Forward key > move the cursor one character.
- The Back key < to move the cursor one character and to return from the submenu to the main menu.

The cursor is a non-blinking underscore that appears under the first letter of the first option in the menu. To make a selection in the menu, you must move the cursor in the desired direction using the Up, Down, Forward and Back key. When the cursor is at the desired position, press Enter key to make a selection.

Accessing Network Settings

You can access the Network Settings of the Extended IP Phone in any of the following stages:

1. During start-up, when the phone prompts you to change the network settings after loading the firmware.

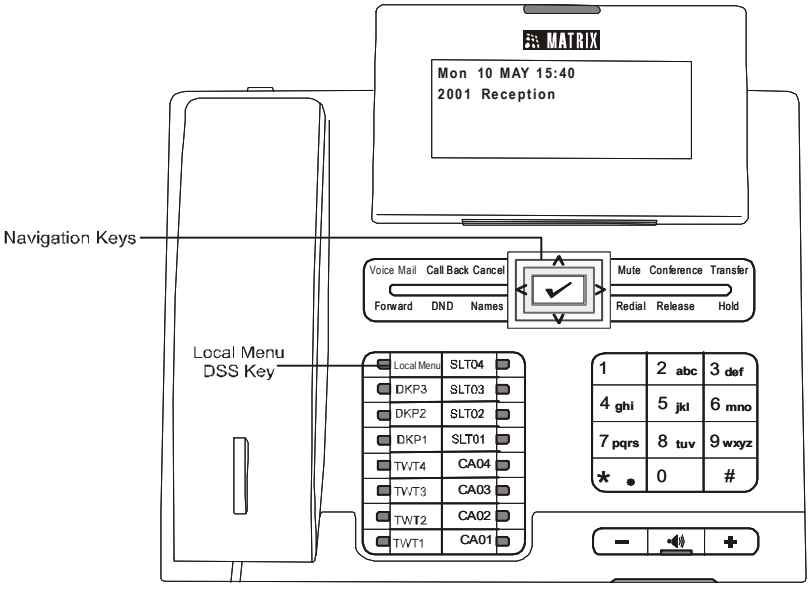


   You must press the Enter Key to select Yes and access network settings.

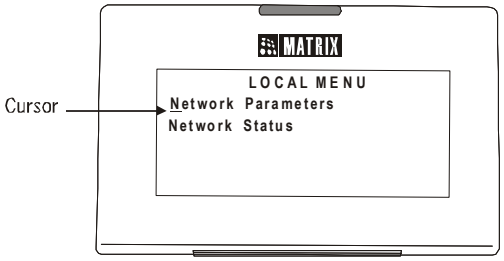
2. When the phone is making Network discovery, downloading configuration files, attempting registration.

   You must press the Enter Key ✓ to access network settings,

3. When the phone is in idle state. You must press the DSS key assigned to ‘Local Menu’.



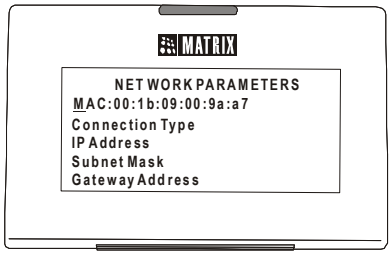
4. When you press the Local Menu DSS Key (in idle state) or when you press the Enter key during any process, the Local Menu appears on your phone display.



You can configure Network Parameters and view Network status from the Local Menu.

Configuring Network Parameters

- In the Local Menu of the phone, select Network Parameters by pressing the Enter Key.
- The Network Parameters submenu appears.



- Use the Down/Up key to reach the desired network parameter and press Enter key to select and change the settings.
- You can configure all network parameters described below, except the MAC Address.

- *To enter a dot in the editable fields — IP Address, Subnet Mask, Gateway Address, DNS Address, Server Address — press * (Star) key.*
- *If you want to clear a single digit/character, move the cursor to the right of the digit/character you wish to clear. Now press the Cancel key. The digit to the left of the cursor will be deleted. If the cursor is to the extreme left and you press the Cancel key, you will go to the previous menu.*

Connection Type

- Select the Connection Type as DHCP, PPPoE or Static according to the IP Addressing scheme of your network.

  If you select DHCP or PPPoE, the phone will be assigned IP Address, Subnet Mask and Gateway Address, DNS Address Server Address, automatically by the DHCP/PPPoE server.

  For PPPoE Connection Type, you must configure the PPPoE User ID and Password provided by the Internet Service Provider.

  If you select Static, you must assign the IP Address, Subnet Mask and Gateway Address to the phone.

IP Address

- If you select Static as Connection Type, enter the static IP Address to be assigned to the phone.

  Enter the desired Static IP Address by pressing the digit keys.

  To enter the dot/period in the IP Address, press the Star '*' key.

Subnet Mask

- If you select Static as Connection Type, enter the Subnet Mask to be applied on the phone by pressing the digit keys.

  To enter the dot/period in the IP Address, press the Star '*' key.

Gateway Address

- If you select Static as Connection Type, enter the Gateway Address here. This is the IP Address of the LAN Port of the Router.

DNS Server

- If you select Static as Connection Type, select the DNS Server option **Static** and configure the DNS Address.

- If you select DHCP or PPPoE as Connection Type and your Internet Service Provider provides DNS Address, select the DNS Server option **Automatic**. However, if your Internet Service Provider does not provide DNS Address, select **Static** and configure the DNS Address.

DNS Address

- If you select DNS Server as Static, enter the DNS Address here.

  To enter dot/period in the IP Address, press the Star '*' key.

DNS Domain Name

- If you select DNS Server as Static, enter the DNS Domain Name here. DNS Domain Name is optional.

PPPoE User ID

- If you have selected PPPoE as Connection Type, you must enter the User ID provided to you by your Internet Service Provider.

PPPoE Password

- This is the password provided by your Internet Service Provider for the PPPoE User ID. If you have selected PPPoE as Connection Type, you must enter the password provided by your Internet Service provider here.

PPPoE Service Name

- If your Internet Service Provider has provided a Service Name, enter the Service Name here. If your Internet Service Provider has not provided a Service Name, do not configure this parameter.

Server Address

- The system works as the Auto Configuration Server for the phone. Enter the LAN or WAN IP Address/ Domain Name of SARVAM UCS here. Default: blank. The phone sends the request for configuration files to this Server Address.

  If you have selected DHCP as Connection Type, the phone will get the Server Address and Port automatically from the DHCP Server. For this, use ***DHCP option 224*** and ***Data Type*** as '***String***' to provide Server Address and Port from the DHCP Server.

  For PPPoE and Static Connection Types, you need to enter the Server Address.

Server Port

- Enter the SPARSH Port of SARVAM UCS here. The phone sends the request for configuration files to this port.

  Valid range of the port is: 80 or 1025–65535. Default: 80.

VLAN Setting

If your phone is connected to a virtual LAN, you need to configure VLAN Settings.

To enable the VLAN switch to correctly route packets generated by the phone and the computers (on the LAN) to each other, the packets must be tagged with a VLAN header.

The VLAN header consists of the VLAN ID (12-bit) and Class of Service (CoS, 3-bit) for prioritization of traffic[108].

The meaning of CoS bits with respect to traffic type is as follows:

CoS	Traffic Type
0	Best Effort
1	Background

108.The IEEE 802.1P standard allows Layer2 switches to prioritize the traffic, thus providing Quality of Service (QoS), that is, better handling of data that pass over a network, thereby resulting in greater reliability and quality. Quality of Service (QoS) on Layer2 is referred to as Class of Service (CoS) which is defined by IEEE 802.1P.

CoS	Traffic Type
2	Spare
3	Excellent Effort
4	Controlled Load
5	Video
6	Voice
7	Network Control

- Select **Phone VLAN/COS** to add VLAN header to the packets generated by the phone, and add VLAN header to the packets relayed from the PC to its LAN port (packets generated by the PC connected to its PC port).
    - To configure Phone VLAN/COS, select **Enable?**. The VLAN ID will be tagged on all packets generated by the phone (SIP, RTP, DNS, ARP, etc.). Default: Disabled.
    - Select **VLAN ID** and enter the VLAN ID that you have assigned to the VLAN in which the IP Phones are connected. Valid range: 0-4094. Default: 1.
    - Select **SIP CoS** and define the CoS (priority) bits in all SIP packets. Valid range: 0-7. Default: 3
    - Select **RTP CoS** and define the CoS (priority) bits in all RTP packets. Valid range: 0-7. Default: 6.
- Select **PC/VLAN CoS** to add VLAN header to all packets entering the PC Port and leaving the LAN port of the phone. Default: Disabled.
    - To configure PC VLAN/COS, select **Enable?**.
    - Select **VLAN ID** and enter the same ID as you have assigned to the VLAN in which the computers are connected. Valid range: 0-4094. Default: 1.
    - Select **CoS** and define the Layer 2 CoS (priority) bits. Valid range: 0-7. Default: 0.

PCAP

To capture packets sent and received from and by the phone for monitoring and troubleshooting, you can enable PCAP on the phone. The phone captures up to 1 MB of packets. For more information and for instructions on how to use PCAP Trace on the phone, see *"Using PCAP Trace for Matrix SPARSH VP248 Extended IP Phone"*, under *PCAP Trace*.

When you change the Network Settings, the phone will restart.

MAC Cloning

If you require cloning of the MAC Address, configure the following:

- Select **Enable?**.
- In **Enter Clone MAC Address**, enter the address you wish to clone.

802.1x Authentication

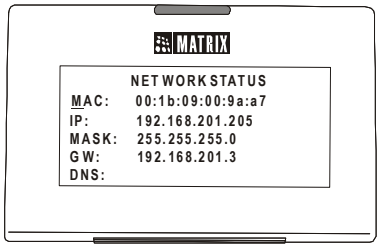
If you want to restrict unauthorized clients from connecting to your LAN, you need to enable 802.1x Authentication. Using EAP MD5 protocol the PC connected to the LAN port of the IP Phone is first authenticated and then it gets connected to LAN.

You need to configure the following 802.1x Authentication parameters:

- Select **Enable?**.
- Enter the 802.1x Authentication **Identity** provided by you network administrator.
- Enter the 802.1x Authentication **MD5 Password** associated with identity provided by your network administrator.

Viewing Network Status

- In the Local Menu of the phone, place the cursor on Network Status and press the Enter key.
- The Network Status submenu appears.



Use the Down/Up key to view the status of the various network parameters. The status of the following parameters appear on your display as you scroll.

- **MAC:** This is the MAC Address of the phone.
- **IP:** The current IP Address assigned to the phone.
- **MASK:** The current Subnet mask assigned to the phone.
- **GW:** The current Gateway IP Address assigned to the phone.
- **DNS:** The Domain Name Server address assigned to the phone.
- **S. ADD:** The LAN or WAN IP Address / Domain Name of the SARVAM UCS.
- **S. PORT:** The SPARSH Port SARVAM UCS.
- **DOMAIN:** The Domain Name assigned to the phone.
- **802.1x Authentication:** The 802.1x authentication status is displayed—Success, Failure, Authenticating or Disabled.
- **FIRM:** The version of the current Firmware of the phone.
- **UBOOT**: The UBOOT release date.
- **KERNEL**: The KERNEL release date.

Connecting SPARSH VP310 as Extended SIP Extension

You are recommended to complete the following steps before connecting the Matrix Extended IP Phone to SARVAM UCS:

- Decide the location of the Extended IP Phone, whether within the same network or outside, according to your installation scenario.

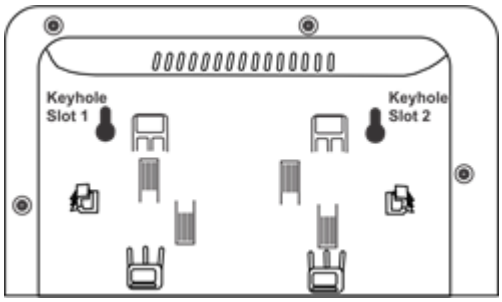
*If you want to use the **DHCP Server** for assigning IP Address to the Extended IP Phone, select **DHCP option 224** and **Data Type** as '**String**' and program the LAN or WAN IP Address /Domain Name of SARVAM UCS and SPARSH Port in the format "**IP_Address:Port"** in your DHCP Server as per your installation scenario.*

- Log in to *Jeeves*. For instructions, read the topic *"Configuring SARVAM UCS"*.
- Assign an extension number (**SIP ID**) to the Extended IP Phone. For instructions on assigning SIP ID, see *"Configuring SIP Extensions"*.

  For the SIP extension number you assigned to the Extended IP Phone, you must configure the necessary parameters in SARVAM UCS so that Extended IP Phone can register as a SIP Extension. For instructions, see the topic *"Configuring SIP Extension Settings as per the Extended Phone Type"* under *Configuring SIP Extensions*.

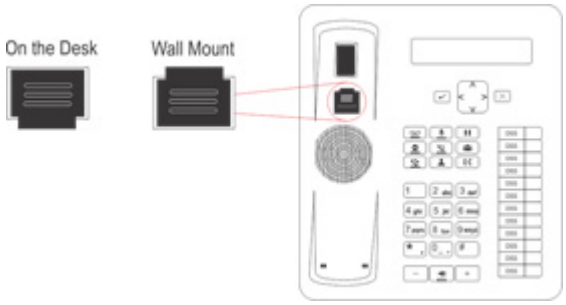
Now, follow the steps described below to install the Extended IP Phone.

1. Unpack the SPARSH VP310 box and verify package contents.
2. You can mount the phone on a wall or on the desk.
3. When you mount SPARSH VP310 on a wall,
   - Use the mounting template to drill holes of appropriate size and distance.
   - Fix the screw grips in the holes you drilled.
   - Fix two screws in the holes on the wall, ensuring that they are aligned with the Keyhole Slots 1 and 2 of SPARSH VP310. The screws should protrude from the wall to fit into the Keyhole Slots.



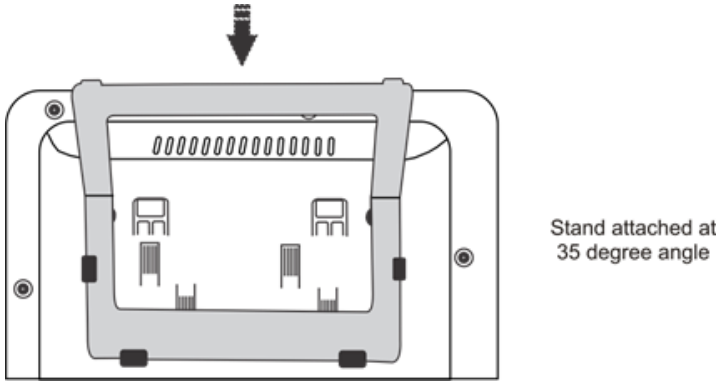
   - Now, mount the phone with the screws fitting into the Keyhole Slot.

- Reverse the handset wall mount tab to make sure the handset remains intact when you mount the phone. Push the handset wall mount tab upwards to remove it from the slot. Rotate it 180 degrees clockwise and push it downwards into the slot.

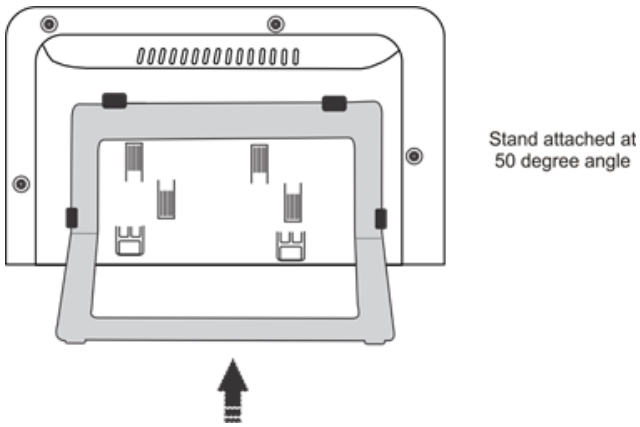


- When you mount the phone on a desk,
  - You can attach the Foot Stand in two ways as illustrated in the following.

    Foot Stand attached at 35° Angle

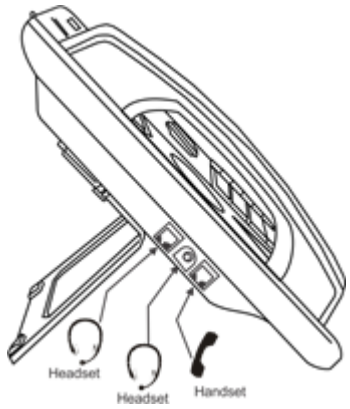


    Foot Stand attached at 50° Angle



    If you attach the Foot Stand at 50°, the phone will be placed in an almost upright position on your desk.

  - Decide which of these positions would work for you best and accordingly attach the Foot Stand.



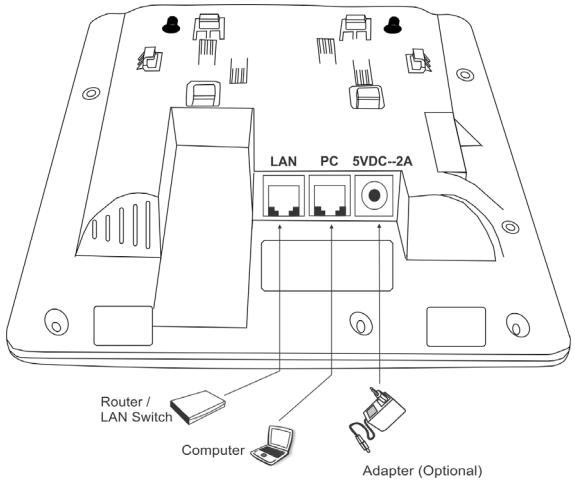
4. Connect the Handset to the Phone body.

   - Plug the long straightened end of the phone cord into the handset jack on the left side panel of the phone marked with the handset symbol .
   - Plug the other (short straight) end of the phone cord into the jack at the bottom of the handset.

5. If you want to use a Headset (not supplied) with your phone, you may plug a headset with a 3.5 mm single connector into the headset jack headset jack with the symbol on the left side panel of the phone, as illustrated in the figure above.

   OR

   You may also plug in a headset with RJ9 connector into the headset port on the left side panel of the phone, marked with the symbol .



6. Connect the LAN Port of SPARSH VP310 to the LAN Switch/Hub or a Router, according to your installation scenario.

7. To connect your phone to a computer on your desk, use an Ethernet cable (not supplied with this phone) to connect the PC Port of the phone to the LAN Port of the computer.

8. It is a PoE enabled phone and can be powered over Ethernet by connecting it to a PoE enabled LAN Switch (IEEE 802.3af Compliant). In this case you need not connect the Power Adapter.

   If you do not want to use PoE, plug in the connector of the Adapter into the power jack (DC Jack) with the label 5VDC-2A at the bottom of the phone. Plug in the Power Adapter into a power outlet.

If both the power options, that is, PoE as well as Power Adapter are available to the phone, then the phone will derive power from the PoE enabled LAN Switch.

The IP phone should be used with Matrix original power adapter (5V/2A) only. The use of any third-party power adapter may cause damage to the phone.

9. Switch ON power supply.

   When you power the phone, the boot process will be initiated in the following sequence.

   - All keys with LED, including the Speaker key, and the Ringer LED, will glow.
   - The LCD display will light up and the booting message appears.
   - Then the 'Loading...' message appears on the phone display.

- The phone will start loading the Extended IP Phone Firmware. It will display current firmware being loaded.
- After loading the firmware, the phone will prompt you to change Network settings.

- Wait for a few seconds.

If you want to change the Network Settings or Server Settings, press the Enter key. Detailed instructions for changing the Network Settings of the phone are provided at the end of this topic. See "Network Settings" at the end of this topic.

- The phone makes DHCP Discovery and fetches its IP Address and Server Address from the DHCP Server.

  On getting the IP Address and Server Address, the phone initiates Auto Configuration to download the configuration files from SARVAM UCS.

- As the phone downloads the configuration files, the file names will appear one by one.
- On successful download of all configuration files, the phone attempts to register with SARVAM UCS.
- On successful registration, the phone will display the current day, date and time, the extension number and name assigned to the Extended IP Phone.

Network Settings

You can change the network settings of the Extended IP Phone. Press the Down key ⋁ when the phone is in idle state. To move the cursor and scroll through the menu and submenu options, use the following navigation keys on your phone.

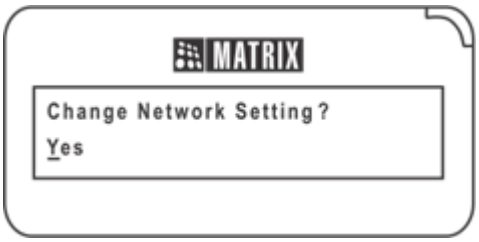
- The Enter key ✓ to make a selection or to complete an action.
- The Up key ⋀ to move up the Menu.
- The Down key ⋁ move down the Menu.
- The Forward key > move the cursor one character.
- The Back key < to move the cursor one character and to return from the submenu to the main menu.
- The Cancel key ✕ to exit a menu.

The cursor is a non-blinking underscore that appears under the first letter of the first option in the menu. To make a selection in the menu, you must move the cursor in the desired direction using the Up, Down, Forward and Back key. When the cursor is at the desired position, press Enter key to make a selection.

Accessing Network Settings

You can access the Network Settings of the Extended IP Phone in any of the following stages:

1. During start-up, when the phone prompts you to change the network settings after loading the firmware.



   You must press the Enter key ✓ to select Yes and access network settings.

2. When the phone is making Network discovery, downloading configuration files, attempting registration.

   You must press the Down key ⋁ to access network settings.

3. When the phone is in idle state. You must press the Down key ⋁ to access the Network Settings.

Configuring Network Parameters

- When the phone is in idle state. You must press the Down key ⋁ to access the Network Settings.
- Press Enter key to select Network Parameters.
- The Network Parameters submenu appears.
- Use the Down/Up key to reach the desired network parameter and press Enter key to select and change the settings.
- You can configure all network parameters described below, except the MAC Address.

- *To enter a dot in the editable fields — IP Address, Subnet Mask, Gateway Address, DNS Address, Server Address — press * (Star) key.*
- *If you want to clear a single digit/character, move the cursor to the right of the digit/character you wish to clear. Now press the Cancel key. The digit to the left of the cursor will be deleted. If the cursor is to the extreme left and you press the Cancel key, you will go to the previous menu.*

Connection Type

- Select the Connection Type as DHCP, PPPoE or Static according to the IP Addressing scheme of your network.

  If you select DHCP or PPPoE, the phone will be assigned IP Address, Subnet Mask and Gateway Address, DNS Address, Server Address, automatically by the DHCP/PPPoE server.

  For PPPoE Connection Type, you must configure the PPPoE User ID and Password provided by the Internet Service Provider.

  If you select Static, you must assign the IP Address, Subnet Mask and Gateway Address to the phone.

IP Address

- If you select Static as Connection Type, enter the static IP Address to be assigned to the phone. Enter the desired Static IP Address by pressing the digit keys.

  To enter the dot/period in the IP Address, press the Star '*' key.

Subnet Mask

- If you select Static as Connection Type, enter the Subnet Mask to be applied on the phone by pressing the digit keys.

  To enter the dot/period in the IP Address, press the Star '*' key.

Gateway Address

- If you select Static as Connection Type, enter the Gateway Address here. This is the IP Address of the LAN Port of the Router.

DNS Server

- If you select Static as Connection Type, select the DNS Server option **Static** and configure the DNS Address.

- If you select DHCP or PPPoE as Connection Type and your Internet Service Provider provides DNS Address, select the DNS Server option **Automatic**. However, if your Internet Service Provider does not provide DNS Address, select **Static** and configure the DNS Address.

DNS Address

- If you select DNS Server as Static, enter the DNS Address here.
  To enter dot/period in the IP Address, press the Star '*' key.

DNS Domain Name

- If you select DNS Server as Static, enter the DNS Domain Name here. DNS Domain Name is optional.

PPPoE User ID

- If you have selected PPPoE as Connection Type, you must enter the User ID provided to you by your Internet Service Provider.

PPPoE Password

- This is the password provided by your Internet Service Provider for the PPPoE User ID. If you have selected PPPoE as Connection Type, you must enter the password provided by your Internet Service provider here.

PPPoE Service Name

- If your Internet Service Provider has provided a Service Name, enter the Service Name here. If your Internet Service Provider has not provided a Service Name, do not configure this parameter.

Primary Server Address

- The system works as the Auto Configuration Server for the phone. Enter the LAN or WAN IP Address/ Domain Name of SARVAM UCS here. Default: blank. The phone sends the request for configuration files to this Server Address.

  If you have selected DHCP as Connection Type, the phone will get the Primary Server Address and Port automatically from the DHCP Server. For this, use ***DHCP option 224*** and ***Data Type*** as '***String***' to provide Server Address and Port from the DHCP Server.

  For PPPoE and Static Connection Types, you need to enter the Server Address.

Primary Server Port

- Enter the SPARSH Port of SARVAM UCS here. The phone sends the request for configuration files to this port.

  Valid range of the port is: 80 or 1025–65535. Default: 80.

Secondary Server Address

- If required, you can also configure the Secondary Sever Address as a fallback option. If the registration with the Primary Server fails the phone will send the registration and configuration requests to the Secondary Server Address. Speech-cut or unclear speech may be observed during on-going mature calls.

Secondary Server Port

- Enter the Secondary Server Port. The phone sends the request for configuration files to this port if the Primary Server fails.

  Valid range of the port is: 80 or 1025–65535. Default: 80.

VLAN Setting

If your phone is connected to a virtual LAN, you need to configure VLAN Settings.

To enable the VLAN switch to correctly route packets generated by the phone and the computers (on the LAN) to each other, the packets must be tagged with a VLAN header.

The VLAN header consists of the VLAN ID (12-bit) and Class of Service (CoS, 3-bit) for prioritization of traffic[109].

The meaning of CoS bits with respect to traffic type is as follows:

CoS	Traffic Type
0	Best Effort
1	Background
2	Spare
3	Excellent Effort
4	Controlled Load
5	Video
6	Voice
7	Network Control

- Select **Phone VLAN/COS** to add VLAN header to the packets generated by the phone, and add VLAN header to the packets relayed from the PC to its LAN port (packets generated by the PC connected to its PC port).
    - To configure Phone VLAN/COS, select **Enable?**. The VLAN ID will be tagged on all packets generated by the phone (SIP, RTP, DNS, ARP, etc.). Default: Disabled.
    - Select **VLAN ID** and enter the VLAN ID that you have assigned to the VLAN in which the IP Phones are connected. Valid range: 0-4094. Default: 1.
    - Select **SIP CoS** and define the CoS (priority) bits in all SIP packets. Valid range: 0-7. Default: 3
    - Select **RTP CoS** and define the CoS (priority) bits in all RTP packets. Valid range: 0-7. Default: 6.
- Select **PC/VLAN CoS** to add VLAN header to all packets entering the PC Port and leaving the LAN port of the phone. Default: Disabled.
    - To configure PC VLAN/COS, select **Enable?**.
    - Select **VLAN ID** and enter the same ID as you have assigned to the VLAN in which the computers are connected. Valid range: 0-4094. Default: 1.
    - Select **CoS** and define the Layer 2 CoS (priority) bits. Valid range: 0-7. Default: 0.

PCAP

To capture packets sent and received from and by the phone for monitoring and troubleshooting, you can enable PCAP on the phone. The phone captures up to 1 MB of packets. For more information and for instructions on how to use PCAP Trace on the phone, see *"Using PCAP Trace for Matrix SPARSH VP310 Matrix Extended IP Phone"*, under *PCAP Trace*.

MAC Cloning

If you require cloning of the MAC Address, configure the following:

109.The IEEE 802.1P standard allows Layer2 switches to prioritize the traffic, thus providing Quality of Service (QoS), that is, better handling of data that pass over a network, thereby resulting in greater reliability and quality. Quality of Service (QoS) on Layer2 is referred to as Class of Service (CoS) which is defined by IEEE 802.1P.

- Select **Enable?** and press the Enter key. Select Yes to enable.
- In **Enter Clone MAC Address**, enter the address you wish to clone.

802.1x Authentication

If you want to restrict unauthorized clients from connecting to your LAN, you need to enable 802.1x Authentication. Using EAP MD5 protocol the PC connected to the LAN port of the IP Phone is first authenticated and then it gets connected to LAN.

You need to configure the following 802.1x Authentication parameters:

- Select **Enable?** and press the Enter key. Select Yes to enable.
- Enter the 802.1x Authentication **Identity** provided by you network administrator.
- Enter the 802.1x Authentication **MD5 Password** associated with identity provided by your network administrator.

When you change the Network Settings, the phone will restart.

Viewing Network Status

- When the phone is in idle state. You must press the Down key ⋁ to access the Network Settings.
- Again press Down key ⋁ to select Network Status and press the Enter key ✓.

  Use the Down/Up key to view the status of the various network parameters. The status of the following parameters appear on your display as you scroll.

  - **MAC:** This is the MAC Address of the phone.
  - **IP:** The current IP Address assigned to the phone.
  - **MASK:** The current Subnet mask assigned to the phone.
  - **GW:** The current Gateway IP Address assigned to the phone.
  - **DNS:** The Domain Name Server address assigned to the phone.
  - **Active Server:** This displays the Server that is active — Primary, Secondary — with which the phone is currently registered.
  - **S. ADD:** This displays the IP address of the Active Server. It may be the LAN or WAN IP Address / Domain Name of the SARVAM UCS or the Secondary Server IP Address (if configured) or any Fallback Server.
  - **S. PORT:** This displays the port of the Active Server. It may be the SPARSH Port of SARVAM UCS or the Secondary Server Port (if configured) or the Fallback Server Port.
  - **DOMAIN:** The Domain Name assigned to the phone.
  - **802.1x Authentication:** The 802.1x authentication status is displayed—Success, Failure, Authenticating or Disabled.

- **FIRM:** The version of the current Firmware of the phone.
- **UBOOT**: The UBOOT release date.
- **KERNEL**: The KERNEL release date.

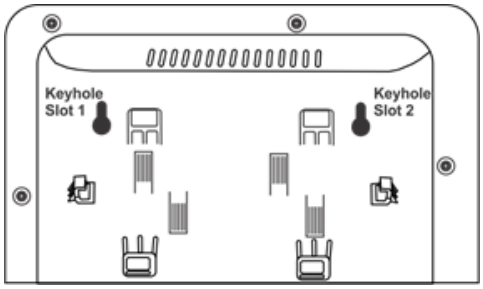
Connecting SPARSH VP330 as Extended SIP Extension

You are recommended to complete the following steps before connecting the Matrix SPARSH VP330 to SARVAM UCS:

- Decide the location where you want to place SPARSH VP330 within your LAN.
- By Default, in SPARSH VP330, the Connection Type selected is DHCP.
- If you want to use the **DHCP Server** for assigning IP Address to the Extended IP Phone, select **DHCP option 224** and **Data Type** as '**String**' and program the LAN or WAN IP Address /Domain Name of SARVAM UCS and SPARSH Port in the format **"IP_Address:Port"** in your LAN DHCP Server as per your installation scenario.
- Log in to *Jeeves*. For instructions, read the topic "Configuring SARVAM UCS".
- You must configure the necessary parameters in SARVAM UCS so that SPARSH VP330 can register as a SIP Extension. For instructions, see "Configuring Matrix SPARSH VP330".

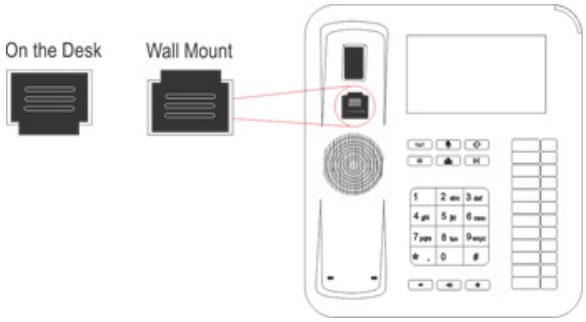
Now, follow the steps described below to install SPARSH VP330.

1. Unpack the SPARSH VP330 box and verify package contents.
2. Mount the phone on a desk or wall at a location convenient to you.
    - When mounting the phone on the wall,
        - Use the mounting template to drill holes of appropriate size and distance. Fix the screw grips in the holes you drilled.
        - Fix two screws in the holes on the wall, ensuring that they are aligned with the Keyhole Slots 1 and 2.
        - Use wall plugs, if required, to fix the screws. Leave the screw heads protruding from the wall to fit into the Keyholes.

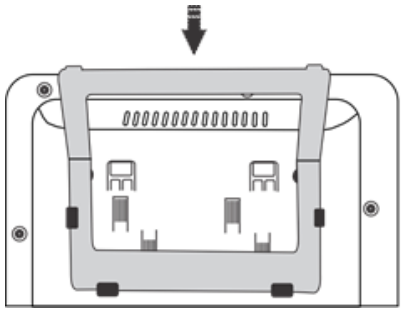


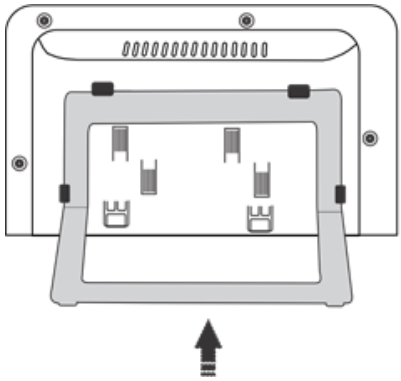
    - Now, mount the phone on the wall, with the screws fitting into the Keyhole slots.

- Reverse the handset wall mount tab to make sure the handset remains intact when you mount the phone. Push the handset wall mount tab upwards to remove it from the slot. Rotate it 180 degrees clockwise and push it downwards into the slot.



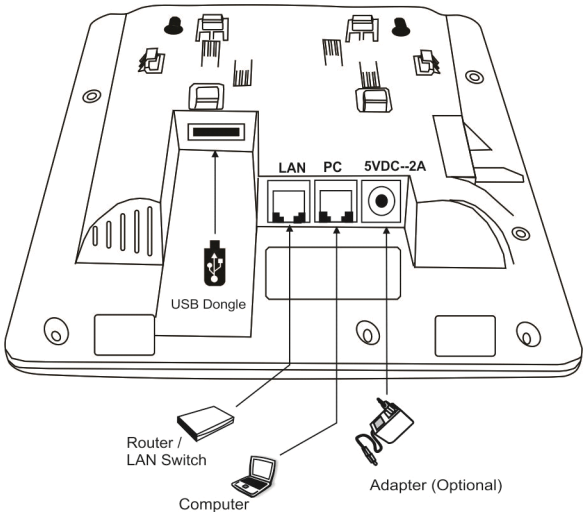
3. When you mount the phone on a desk, you can attach the Foot Stand in two ways at **35° Angle** or at **50° Angle**.





- If you attach the Foot Stand at 50°, the phone will be placed in an almost upright position on your desk.

- Decide which of these positions would work for you best and accordingly attach the Foot Stand.



4. Connect the Handset to the Phone body.

   - Plug the long straightened end of the phone cord into the handset jack on the left side panel of the phone marked with the handset symbol.
   - Plug the other (short straight) end of the phone cord into the jack at the bottom of the handset.

5. If you want to use a Headset (not supplied) with your phone, you may plug a headset with a 3.5 mm single connector into the headset jack headset jack with the symbol on the left side panel of the phone.

   OR

   You may plug a headset with an RJ9 connector into the headset port on the side panel of the phone, marked with the symbol .

6. Connect the LAN Port of SPARSH VP330 to the IP Network — A Router or LAN Switch — using the Ethernet Cable.

   OR
   Connect the Wi-Fi USB Adapter into the USB port of the phone.

You can purchase the Wi-Fi USB Adapter from Matrix as an optional peripheral device to support wireless network.

7. To connect your phone to a computer on your desk, use an Ethernet cable (not supplied with this phone) to connect the PC Port at the bottom of the phone to the LAN Port of the computer.

8. It is a PoE enabled phone and can be powered over Ethernet by connecting it to a PoE enabled LAN Switch (IEEE 802.3af Compliant). In this case you need not connect the Power Adapter.

   If you do not want to use PoE, plug in the connector of the Adapter into the power jack (DC Jack) with the label 5VDC-2A at the bottom of the phone. Plug in the Power Adapter into a power outlet.

If both the power options, that is, PoE as well as Power Adapter are available to the phone, then the phone will derive power from the PoE enabled LAN Switch.

The IP phone should be used with Matrix original power adapter (5V/2A) only. The use of any third-party power adapter may cause damage to the phone.

9. Switch ON power supply.

   When you power the phone, the boot process will be initiated in the following sequence.

   - All keys with LED, including the Speaker key, and the Ringer LED, will glow.
   - The LCD display will light up and booting message appears.
   - While loading the application then the loading message appears on the phone display,
   - The phone makes DHCP Discovery and fetches its IP Address and Server Address from the DHCP Server.

If you want to change the Network Settings/Server Settings or want to use Wi-Fi for connectivity, press Settings *.*

Refer to the SPARSH VP330 User Guide, for detailed instructions:

- *To change the Network Settings of the phone and configure the network parameters.*
- *To use Wi-Fi for connectivity and configure its parameters.*

- On getting the IP Address and Server Address, the phone initiates Auto Configuration to download the configuration files from SARVAM UCS.
- On successful download of all configuration files, the phone attempts to register with SARVAM UCS.
- On successful registration, the Home screen appears.

The phone will register successfully, only if the SIP Extension parameters in SARVAM UCS have been correctly configured as per your installation scenario.

Refer to the ***SPARSH VP330 User Guide*** to know more.

Connecting SPARSH VP510 as Extended SIP Extension

You are recommended to complete the following steps before connecting the Matrix Extended IP Phone to the system when used with SARVAM UCS application:

- Decide the location of the Extended IP Phone, whether within the same network or outside, according to your installation scenario.

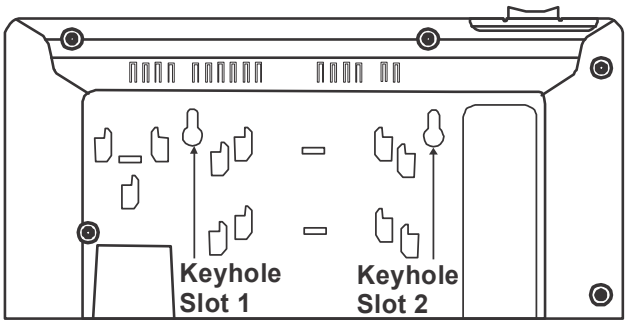
*If you want to use the **DHCP Server** for assigning IP Address to the Extended IP Phone, select **DHCP option 224** and **Data Type** as '**String**' and program the LAN or WAN Port IP Address /Domain Name and SPARSH Port in the format "**IP_Address:Port**" in your DHCP Server as per your installation scenario.*
•Log in to *Jeeves*. For instructions, read the topic "Configuring SARVAM UCS".

- Assign an extension number (**SIP ID**) to the Extended IP Phone. For instructions on assigning SIP ID, see "Configuring SIP Extensions".

  For the SIP extension number you assigned to the Extended IP Phone, you must configure the necessary parameters in SARVAM UCS so that Extended IP Phone can register as a SIP Extension. For instructions, see the topic "Configuring SIP Extension Settings as per the Extended Phone Type" under *Configuring SIP Extensions*.

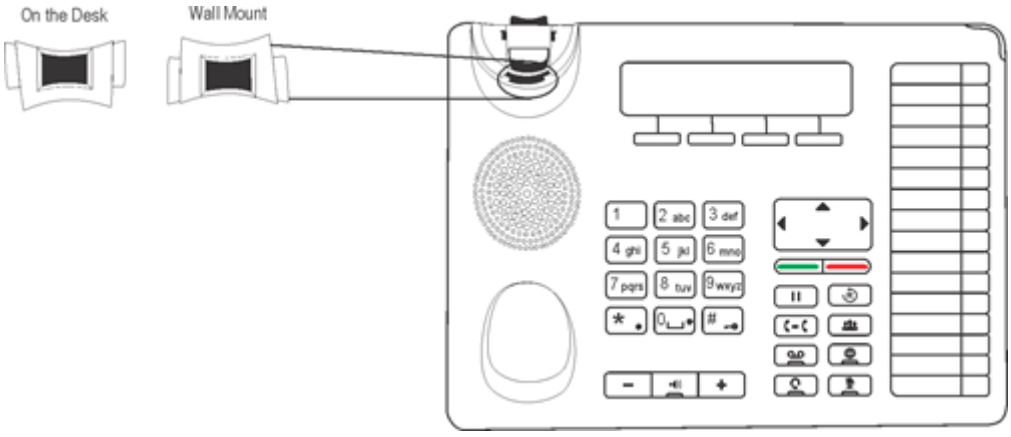
Now, follow the steps described below to install the Extended IP Phone:

1. Unpack the SPARSH VP510 box and verify package contents.
2. You can mount the phone on a wall or on the desk.
3. When you mount SPARSH VP510 on a wall,
   - Use the mounting template to drill holed of appropriate size and distance.
   - Fix the screw grips in the holes you drilled.
   - Fix two screws in the holes on the wall, ensuring that they are aligned with the Keyhole Slots 1 and 2 of SPARSH VP510. The screws should protrude from the wall to fit into the Keyhole Slots.



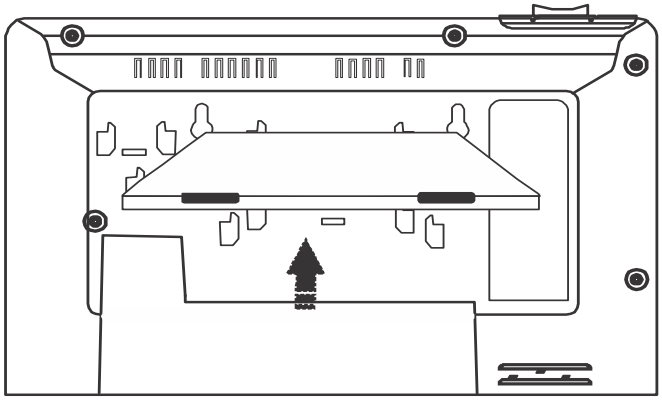
   - Now, mount the phone with the screws into the Keyhole Slots.

- Reverse the handset wall mount tab to make sure that the handset remains intact when you mount the phone. Push the handset wall mount tab upwards to remove it from the slot. Rotate it 180 degrees clockwise and push it downwards into the slot.

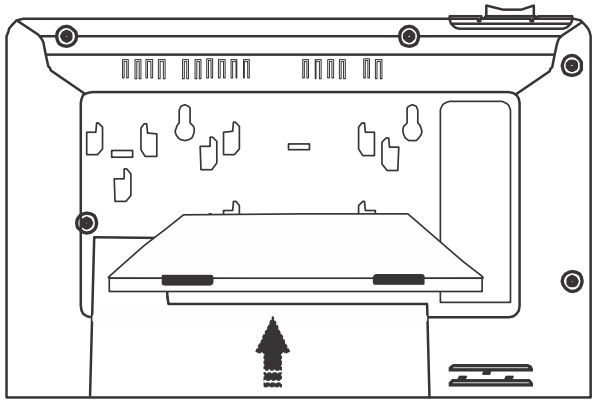


If you are unable to remove the wall mount tab, you may use a tool like a minus screw-driver to remove it.

- When you mount the phone on a desk,
  - You can attach the Foot Stand in the following ways — at an angle of 45 degrees or 55 degrees



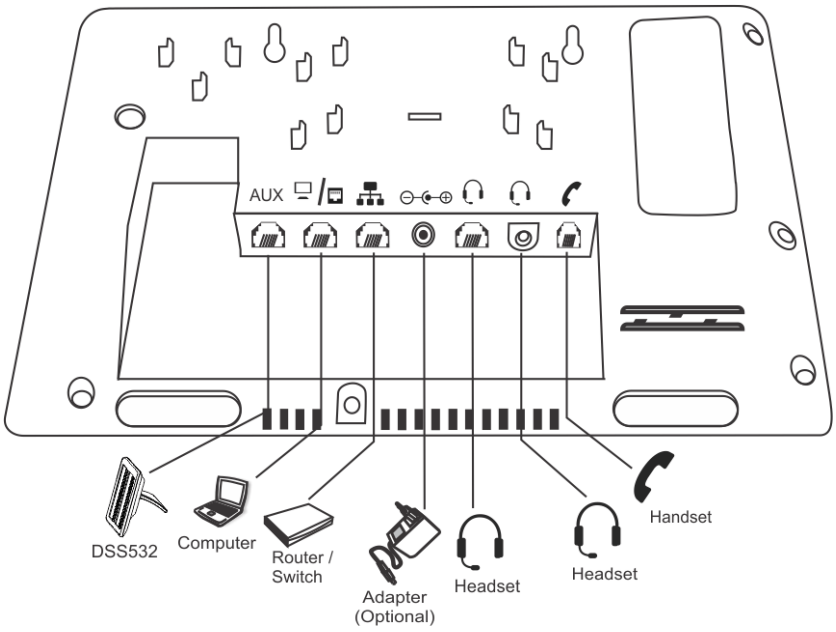
Stand attached at 45 degree angle



Stand attached at 55 degree angle

  - Decide which of these positions would work for you best and accordingly attach the Foot Stand.

Refer to the diagram below for connectivity.



4. Connect the Handset.
   - Plug the long straightened end of the Spring Cord into the handset jack at the bottom of the phone, marked with the handset symbol .
   - Plug the other (short straight) end of the Spring Cord into the jack at the bottom of the handset.
5. Connect the Headset (not supplied by Matrix).
   - To use a Headset (not supplied with the phone), plug any standard stereo headset with 3.5mm single connector into the headset audio jack at the bottom of the phone, marked with the symbol .

     OR

     You may also plug in a headset with an RJ9 connector into the headset port at the bottom of the phone, marked with the symbol .
6. Connect to the IP Network.
   - Plug one end of the Ethernet Cable into the LAN Port at the bottom of the phone, marked with the symbol and the other end to the IP Network — A Router or LAN Switch.
7. Connect a PC to the Phone.
   - Plug one end of the Ethernet Cable into the PC Port at the bottom of the phone, marked with the symbol and the other end into the LAN Port of your PC/LAN Switch.

8. Connect DSS532 with the Phone.

   - To connect DSS532 with the phone, plug one end of the RJ11 cable into the AUX Port of the phone and the other end into the IN Port of the DSS532. For installation, see “Installing DSS532 with SPARSH VP510”.

9. Connect the Power Supply.

   - It is a PoE enabled phone and can be powered over Ethernet by connecting it to a PoE enabled LAN Switch (IEEE 802.3af Compliant).

     If you do not want to use PoE, plug in the connector of the Adapter into the power jack (DC Jack) at the bottom of the phone, marked with the symbol . Plug in the Power Adapter into a power outlet.

If both the power options, that is, PoE as well as Power Adapter are available to the phone, then the phone will derive power from the PoE enabled LAN Switch.

The IP phone should be used with Matrix original power adapter (5V/2A) only. The use of any third-party power adapter may cause damage to the phone.

10. Switch ON power supply.

    When you power the phone, the boot process will be initiated in the following sequence.

    - All keys with LED, including the Speaker key, and the Ringer LED, will glow.
    - The LCD display will light up and the booting message appears.
    - Then the ‘Loading...’ message appears on the phone display.
    - The phone will start loading the Extended IP Phone Firmware. It will display current firmware being loaded.
    - After loading the firmware, the phone will prompt you to change Network settings.
    - Wait for a few seconds.

If you want to change the Network Settings or Server Settings, press Yes key. Detailed instructions for changing the Network Settings of the phone are provided at the end of this topic. See “Network Settings” at the end of this topic.

- The phone makes DHCP Discovery and fetches its IP Address and Server Address from the DHCP Server.

  On getting the IP Address and Server Address, the phone initiates Auto Configuration to download the configuration files from SARVAM UCS.

- As the phone downloads the configuration files, the file names will appear one by one.
- On successful download of all configuration files, the phone attempts to register with SARVAM UCS.

- On successful registration, the phone will display the current day, date and time, the extension number and name assigned to the Extended IP Phone.

Network Settings

You can change the network settings of the Extended IP Phone. Press the Down key ⋁ when the phone is in idle state.

To navigate the menu,

- Press the Menu Key when the phone is idle.
- Scroll by pressing the Up/Down Navigation Key to reach the desired Menu option.
- Press the Select / OK Key to select the desired Menu option.
- Scroll by pressing the Up/Down Navigation Key to reach the desired sub-menu option.
- Press the Select / OK Key to select the desired sub-menu option.

To exit menu,

- Press Cancel Key.
  or
  Go ON-Hook.

The cursor is a non-blinking underscore that appears under the first letter of the first option in the menu. To make a selection in the menu, you must move the cursor in the desired direction using the Up, Down, Forward and Back key. When the cursor is at the desired position, press Enter key to make a selection.

Accessing Network Settings

You can access the Network Settings of the Extended IP Phone in any of the following stages:

1. During start-up, when the phone prompts you to change the network settings after loading the firmware.

   You must press **Yes** key and access network settings.

2. When the phone is making Network discovery, downloading configuration files, attempting registration.

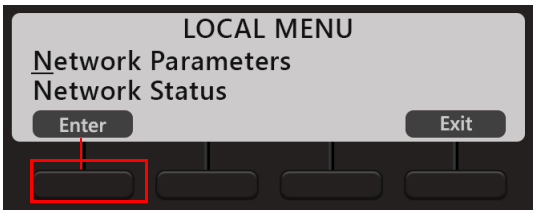
   You must press the **Menu** key to access network settings.

3. When the phone is in idle state, press the Down key ⋁ .

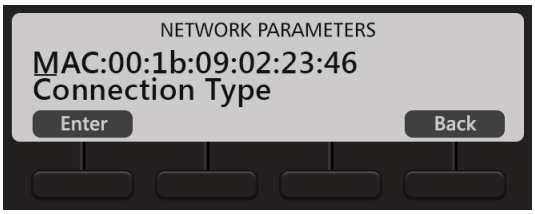
You can configure Network Parameters and view Network status from the Local Menu.

Configuring Network Parameters

- In the Local Menu of the phone, select Network Parameters by pressing the Enter Key.



- The Network Parameters submenu appears.



- Use the Down/Up key to reach the desired network parameter and press Enter key to select. Change the settings as per your requirements.
- Press **Save** key, to save the changes you make.
- You can configure all network parameters described below, except the MAC Address.

- *To enter a dot in the editable fields — IP Address, Subnet Mask, Gateway Address, DNS Address, Server Address — press * (Star) key.*
- *If you want to clear a single digit/character, move the cursor to the right of the digit/character you wish to clear. Now press the Delete key. The digit to the left of the cursor will be deleted.*

Before you start configuring the Network Parameters, get acquainted with following context keys:

Context Keys	Description
Enter/OK	To select a particular parameter
Save	To save the changes
Back	To move a step backwards without saving the changes
Delete	To delete previous characters from the cursor position
2Ab/123	2Ab - Alphanumeric Mode 123 - Numeric Mode

Connection Type

- Select the Connection Type as DHCP, PPPoE or Static according to the IP Addressing scheme of your network.

If you select DHCP or PPPoE, the phone will be assigned IP Address, Subnet Mask and Gateway Address, DNS Address, Server Address, automatically by the DHCP/PPPoE server.

For PPPoE Connection Type, you must configure the PPPoE User ID and Password provided by the Internet Service Provider.

If you select Static, you must assign the IP Address, Subnet Mask and Gateway Address to the phone.

IP Address

- If you select Static as Connection Type, enter the static IP Address to be assigned to the phone.

  Enter the desired Static IP Address by pressing the digit keys.

  To enter the dot/period in the IP Address, press the Star '*' key.

Subnet Mask

- If you select Static as Connection Type, enter the Subnet Mask to be applied on the phone by pressing the digit keys.

  To enter the dot/period in the IP Address, press the Star '*' key.

Gateway Address

- If you select Static as Connection Type, enter the Gateway Address here. This is the IP Address of the LAN Port of the Router.

DNS Server

- If you select Static as Connection Type, select the DNS Server option **Static** and configure the DNS Address.

- If you select DHCP or PPPoE as Connection Type and your Internet Service Provider provides DNS Address, select the DNS Server option **Automatic**. However, if your Internet Service Provider does not provide DNS Address, select **Static** and configure the DNS Address.

DNS Address

- If you select DNS Server as Static, enter the DNS Address here.

  To enter dot/period in the IP Address, press the Star '*' key.

DNS Domain Name

- If you select DNS Server as Static, enter the DNS Domain Name here. DNS Domain Name is optional.

PPPoE User ID

- If you have selected PPPoE as Connection Type, you must enter the User ID provided to you by your Internet Service Provider.

PPPoE Password

- This is the password provided by your Internet Service Provider for the PPPoE User ID. If you have selected PPPoE as Connection Type, you must enter the password provided by your Internet Service provider here.

PPPoE Service Name

- If your Internet Service Provider has provided a Service Name, enter the Service Name here. If your Internet Service Provider has not provided a Service Name, do not configure this parameter.

Primary Server Address

- The system works as the Auto Configuration Server for the phone. Enter the LAN or WAN IP Address/ Domain Name of SARVAM UCS here. Default: blank. The phone sends the request for configuration files to this Server Address.

  If you have selected DHCP as Connection Type, the phone will get the Primary Server Address and Port automatically from the DHCP Server. For this, use ***DHCP option 224*** and ***Data Type*** as '***String***' to provide Server Address and Port from the DHCP Server.
  For PPPoE and Static Connection Types, you need to enter the Server Address.

Primary Server Port

- Enter the SPARSH Port of SARVAM UCS here. The phone sends the request for configuration files to this port.

  Valid range of the port is: 80 or 1025–65535. Default: 80.

Secondary Server Address

- If required, you can also configure the Secondary Sever Address as a fallback option. If the registration with the Primary Server fails the phone will send the registration and configuration requests to the Secondary Server Address. Speech-cut or unclear speech may be observed during on-going mature calls.

Secondary Server Port

- Enter the Secondary Server Port. The phone sends the request for configuration files to this port if the Primary Server fails.

  Valid range of the port is: 80 or 1025–65535. Default: 80.

VLAN Setting

If your phone is connected to a virtual LAN, you need to configure VLAN Settings.

To enable the VLAN switch to correctly route packets generated by the phone and the computers (on the LAN) to each other, the packets must be tagged with a VLAN header.

The VLAN header consists of the VLAN ID (12-bit) and Class of Service (CoS, 3-bit) for prioritization of traffic[110]. The meaning of CoS bits with respect to traffic type is as follows:

CoS	Traffic Type
0	Best Effort
1	Background
2	Spare

110.The IEEE 802.1P standard allows Layer2 switches to prioritize the traffic, thus providing Quality of Service (QoS), that is, better handling of data that pass over a network, thereby resulting in greater reliability and quality. Quality of Service (QoS) on Layer2 is referred to as Class of Service (CoS) which is defined by IEEE 802.1P.

CoS	Traffic Type
3	Excellent Effort
4	Controlled Load
5	Video
6	Voice
7	Network Control

- Select **Phone VLAN/COS** to add VLAN header to the packets generated by the phone, and add VLAN header to the packets relayed from the PC to its LAN port (packets generated by the PC connected to its PC port).
  - To configure Phone VLAN/COS, select **Enable?**. The VLAN ID will be tagged on all packets generated by the phone (SIP, RTP, DNS, ARP, etc.). Default: Disabled.
  - Select **VLAN ID** and enter the VLAN ID that you have assigned to the VLAN in which the IP Phones are connected. Valid range: 0-4094. Default: 1.
  - Select **SIP CoS** and define the CoS (priority) bits in all SIP packets. Valid range: 0-7. Default: 3
  - Select **RTP CoS** and define the CoS (priority) bits in all RTP packets. Valid range: 0-7. Default: 6.
- Select **PC/VLAN CoS** to add VLAN header to all packets entering the PC Port and leaving the LAN port of the phone. Default: Disabled.
  - To configure PC VLAN/COS, select **Enable?**.
  - Select **VLAN ID** and enter the same ID as you have assigned to the VLAN in which the computers are connected. Valid range: 0-4094. Default: 1.
  - Select **CoS** and define the Layer 2 CoS (priority) bits. Valid range: 0-7. Default: 0.

PCAP

To capture packets sent and received from and by the phone for monitoring and troubleshooting, you can enable PCAP on the phone. The phone captures up to 1 MB of packets. For more information and instructions on how to use PCAP Trace on the phone, refer to the *EON510_SPARSH VP510 User Guide*.

MAC Cloning

If you require cloning of the MAC Address, configure the following:

- Select **Enable?** and press the Enter key. Select Yes to enable.
- In **Enter Clone MAC Address**, enter the address you wish to clone.

802.1x Authentication

If you want to restrict unauthorized clients from connecting to your LAN, you need to enable 802.1x Authentication. Using EAP MD5 protocol the PC connected to the LAN port of the IP Phone is first authenticated and then it gets connected to LAN.

You need to configure the following 802.1x Authentication parameters:

- Select **Enable?** and press the Enter key. Select Yes to enable.

- Enter the 802.1x Authentication **Identity** provided by you network administrator.
- Enter the 802.1x Authentication **MD5 Password** associated with identity provided by your network administrator.

When you change the Network Settings, the phone will restart.

Viewing Network Status

- When the phone is in idle state. You must press the Down key ⋁ to access the Network Settings.
- Again press Down key ⋁ to select Network Status and press the Enter key.

  Use the Down/Up key to view the status of the various network parameters. The status of the following parameters appear on your display as you scroll.

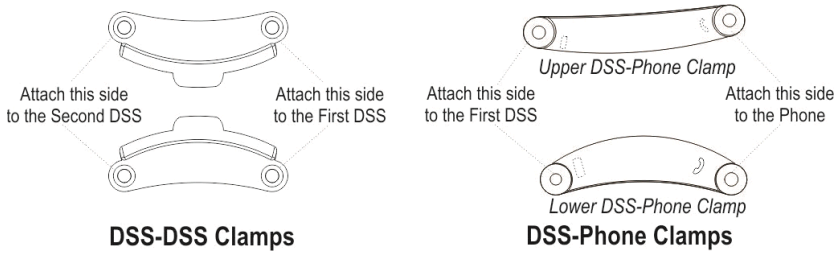
  - **MAC:** This is the MAC Address of the phone.
  - **IP:** The current IP Address assigned to the phone.
  - **MASK:** The current Subnet mask assigned to the phone.
  - **GW:** The current Gateway IP Address assigned to the phone.
  - **DNS:** The Domain Name Server address assigned to the phone.
  - **Active Server:** This displays the Server that is active — Primary, Secondary — with which the phone is currently registered.
  - **S. ADD:** This displays the IP address of the Active Server. It may be the LAN or WAN IP Address / Domain Name of the SARVAM UCS or the Secondary Server IP Address (if configured) or any Fallback Server.
  - **S. PORT:** This displays the port of the Active Server. It may be the SPARSH Port of SARVAM UCS or the Secondary Server Port (if configured) or the Fallback Server Port.
  - **DOMAIN:** The Domain Name assigned to the phone.
  - **802.1x Authentication:** The 802.1x authentication status is displayed—Success, Failure, Authenticating or Disabled.
  - **FIRM:** The version of the current Firmware of the phone.
  - **UBOOT**: The UBOOT release date.
  - **KERNEL**: The KERNEL release date.

Refer to the ***EON510_SPARSH VP510 User Guide*** to know more.

Installing DSS532 with SPARSH VP510

Once you have installed SPARSH VP510 with SARVAM UCS, you can install the DSS532 by following the steps given below:

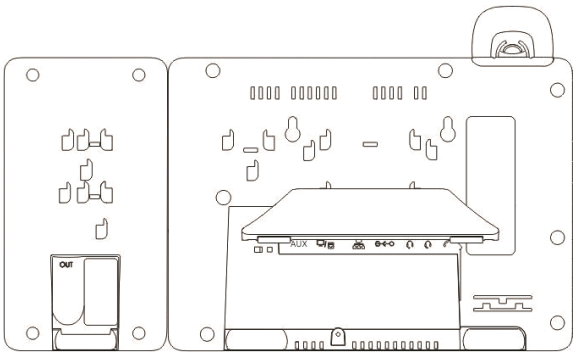
1. Unpack the box and verify the package contents[111].
2. Four clamps are provided with the phone — 2 DSS-Phone Clamps and 2 DSS-DSS Clamp.



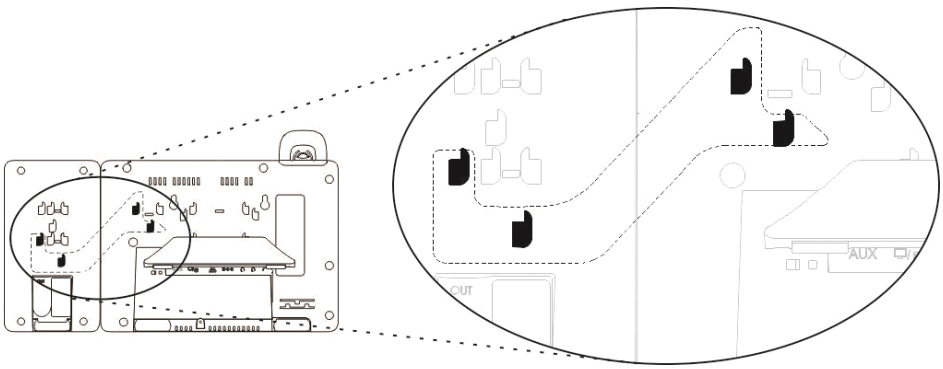
Connecting the First DSS532

Connecting the Extender

3. Turn the phone upside down on the table and place the inverted DSS532 adjacent to it.

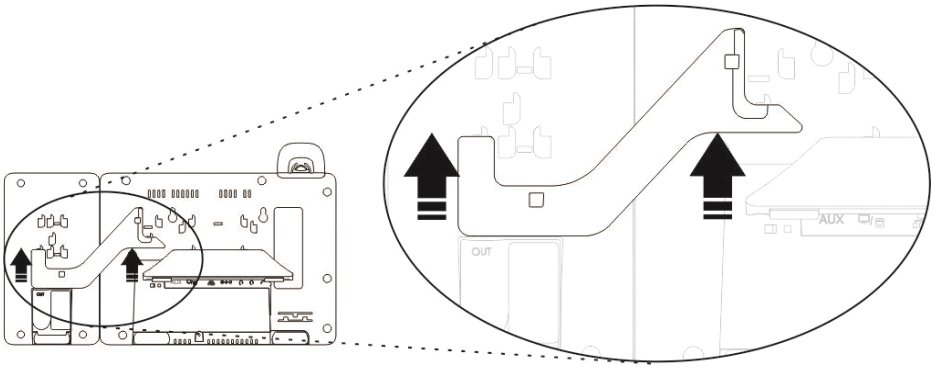


4. To attach the DSS532 with the phone, place the DSS Extender as illustrated below.

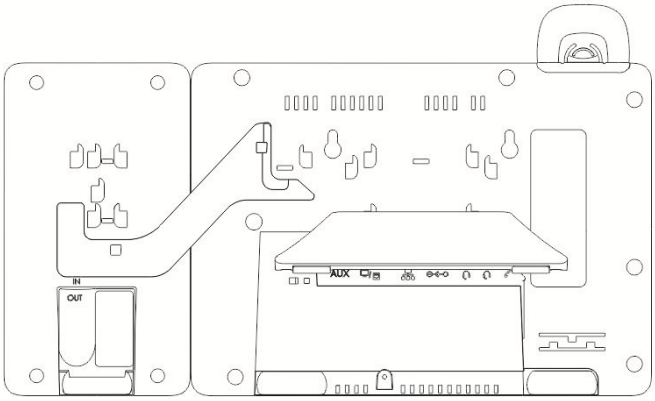


111. See *"Packing List" of Appendix topic.*

5. Insert the hooks on the Extender into the slots provided on the phone and the DSS532.

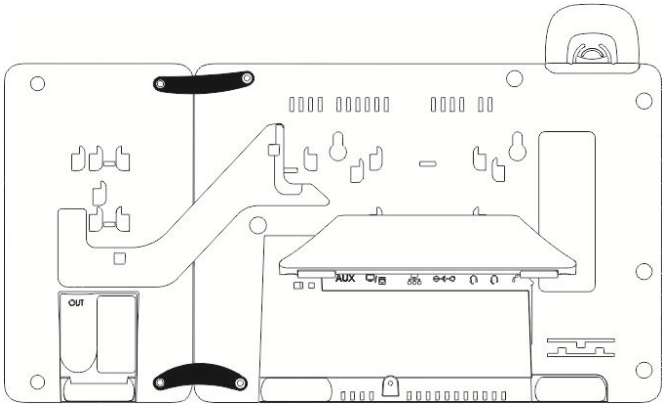


6. Firmly slide the DSS Extender upwards to lock them in place.



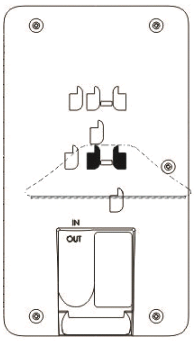
Attaching the Clamps

7. Now attach the clamps. To do so,
   - Remove the screws to attach the clamps.
   - Place the DSS-Phone Clamps between the DSS532 and the phone.
   - Insert the screws back to fix the clamps.

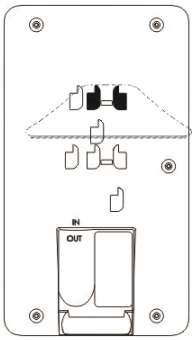


Attaching the Footstand

8. You can mount the DSS532 with the phone on the desk at two angles — **45 degrees** or **55 degrees** by attaching the Foot Stand.



Stand attached at 45 degree angle

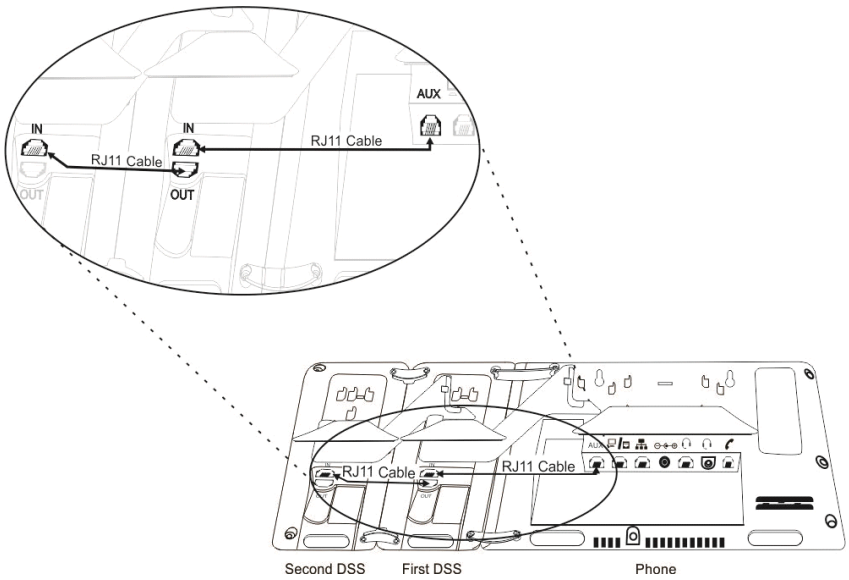


Stand attached at 55 degree angle

Make sure both, the DSS532 and phone are mounted at the same angle.

Connecting the Cables

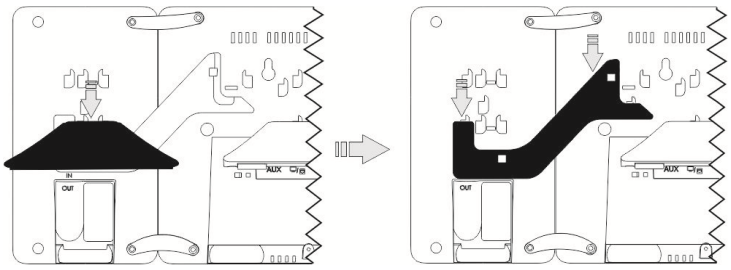
9. To connect the DSS532 with phone, plug one end of RJ11 Cable into **Auxiliary(AUX) Port** of the phone and the other end into the **IN Port** of the DSS532.



Connecting Multiple DSS532

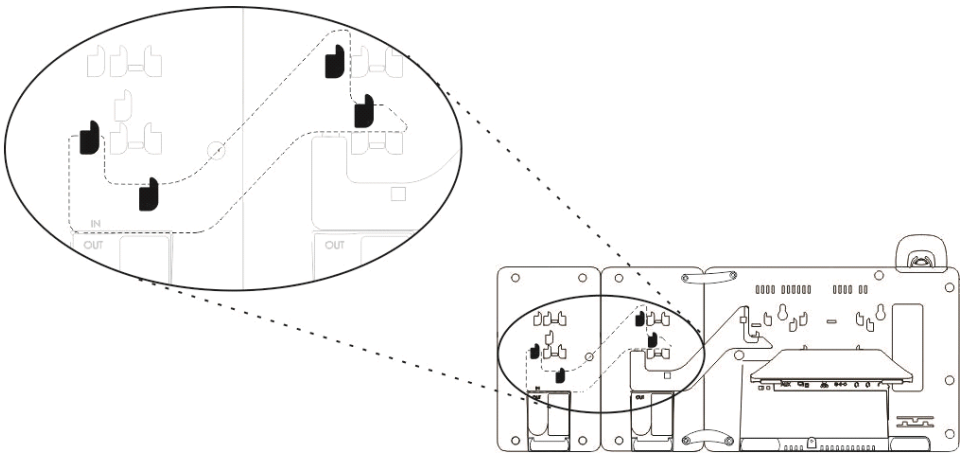
Remove the Foot Stand

10. Remove the Foot Stand of attached DSS532. To do so,
    - Firmly slide the Foot Stand of the attached DSS532 downward to unlock.
    - Now, slide down the attached DSS Extender in downward direction.

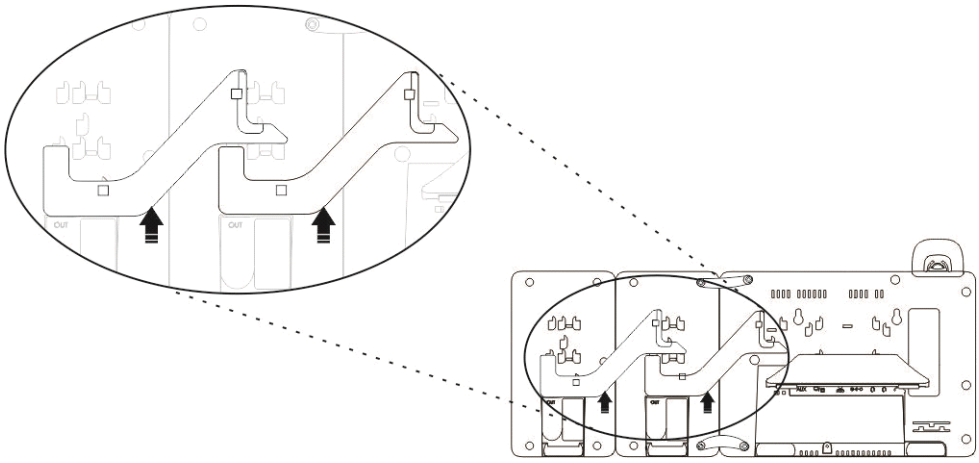


Attach the second DSS Extender

11. To attach the second DSS Extender,
    - Place another inverted DSS532 adjacent to the existing assembly.
    - Place the DSS Extender as illustrated in the diagram below.
    - Insert the hooks on the Extender into the slots provided on both the DSS532.

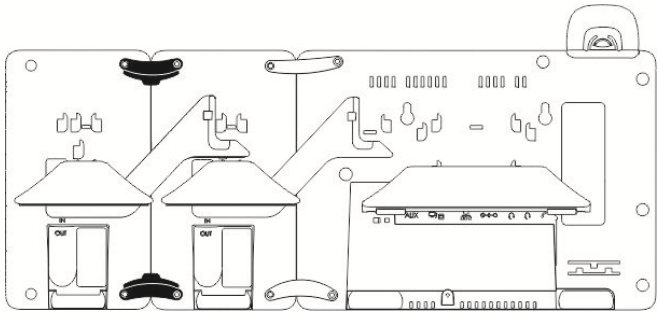


12. Firmly slide both the DSS Extenders upward consecutively (attach the second extender first followed by the existing one attached to the phone) and lock them in place.



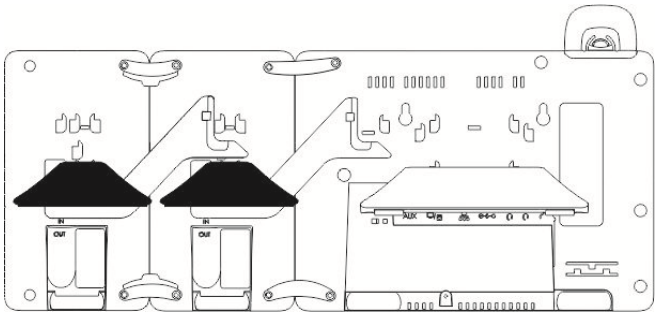
Attach the Clamps

13. Attach the DSS-DSS Clamps between both the DSS532.



Attach the Foot Stand

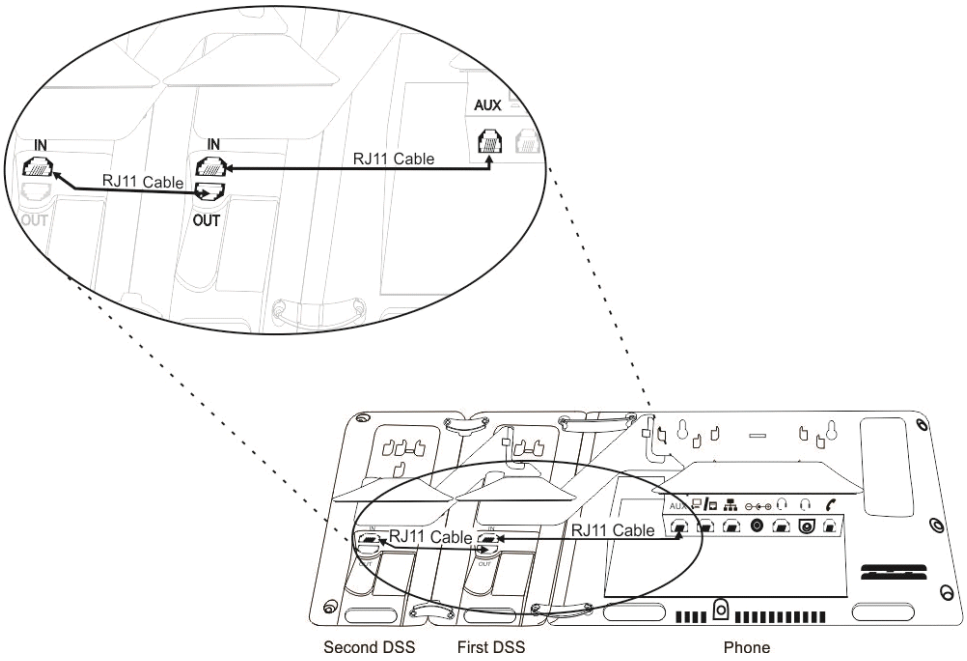
14. Attach the Foot Stand of both the DSS532.



Make sure both, the DSS532 and the phone are mounted at the same angle.

Connect the second DSS532 to the existing assembly

15. Plug one end of the RJ11 Cable into the OUT Port of the existing DSS532 (already connected with the phone) and the other end into the IN Port of the second DSS532.



You can install a maximum of four DSS532 with a phone.

16. After you have connected the DSS532 with the phone, you can configure the DSS Keys. For instructions, see “Programming DSS Console Keys”.

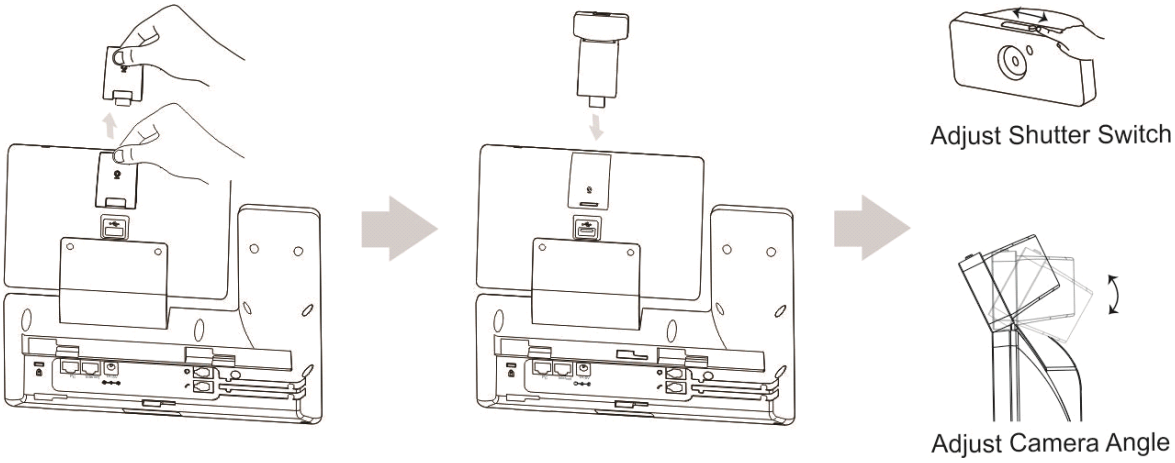
Connecting Extended SPARSH VP710 as Extended SIP Extension

You are recommended to complete the following steps before connecting the Matrix Extended SPARSH VP710 to SARVAM UCS:

- Decide the location where you want to place Matrix Extended SPARSH VP710 within your LAN.
- Log in to *Jeeves*. For instructions, read the topic “Configuring SARVAM UCS”.
- You must configure the necessary parameters in SARVAM UCS so that Extended SPARSH VP710 can register as a SIP Extension. For instructions, see “Configuring Matrix Extended SPARSH VP710”.

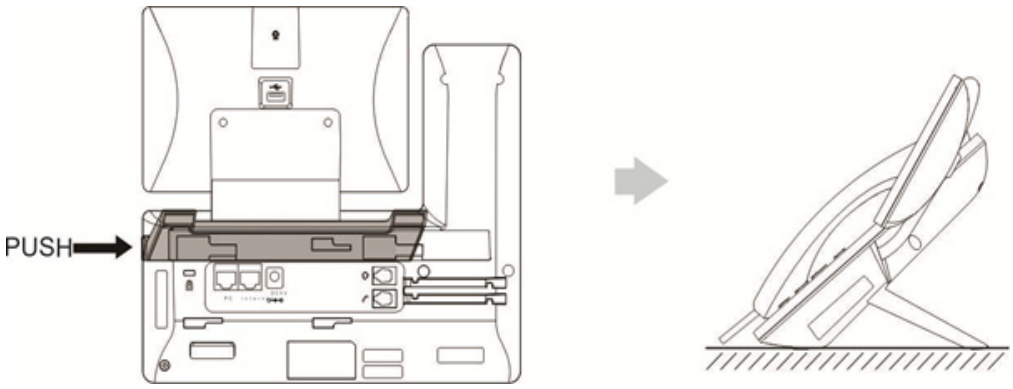
Now, follow the steps described below to install Extended SPARSH VP710.

1. **Inserting the camera**

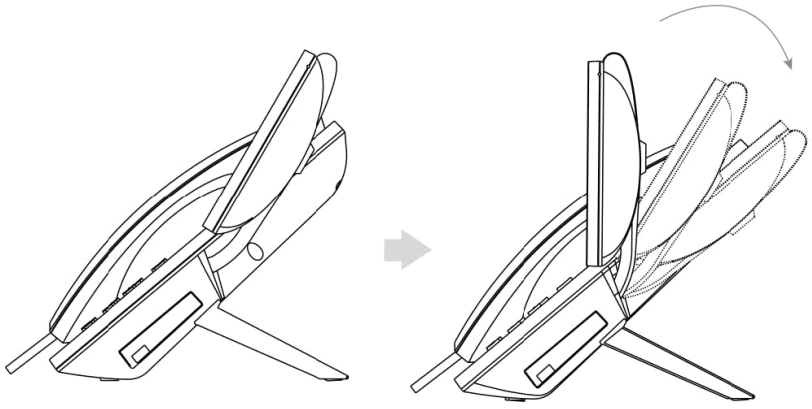


It is recommended to use only the Matrix original Camera, supplied with the IP Phone for video calling. The use of any third-party camera may cause damage to the phone. Damages to the phone caused by using third-party camera is not covered by Matrix warranty.

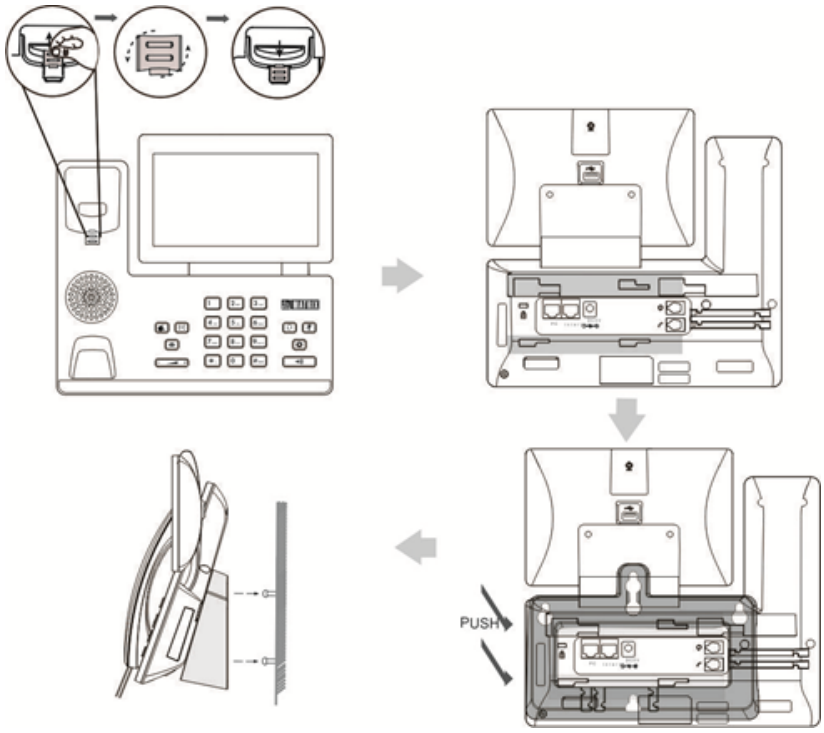
2. **Attaching the stand**



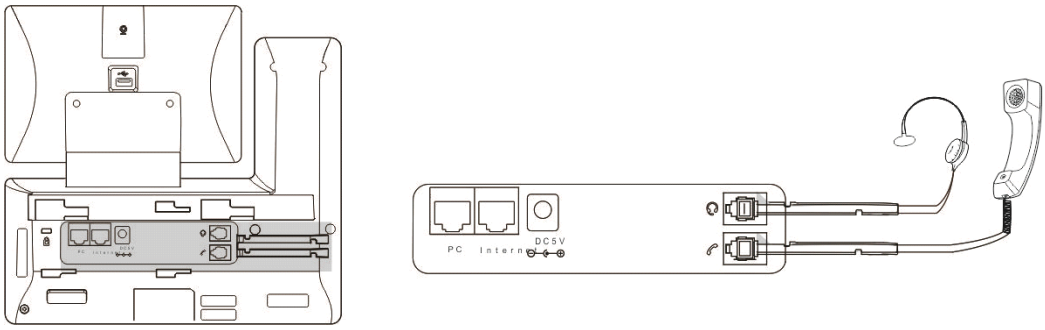
3. **Adjusting the angle of the touch screen**.



4. **Attaching the optional wall mounting bracket**

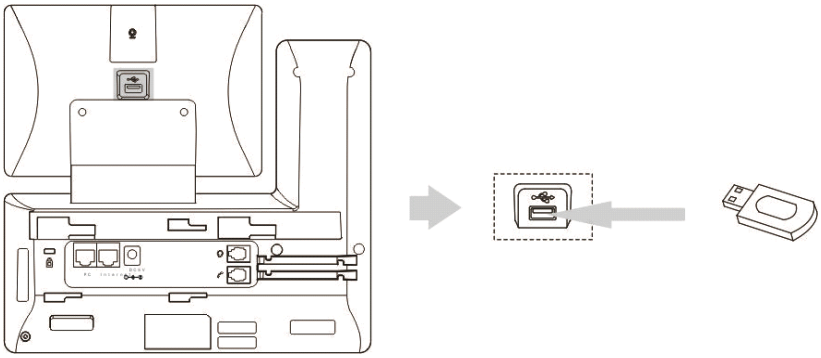


5. **Connect the handset and optional headset.**



A headset is not included in the packaging contents. Contact your dealer/reseller for more information.

6. **Connect the optional USB Flash drive.**



7. **Connect the network and power.**

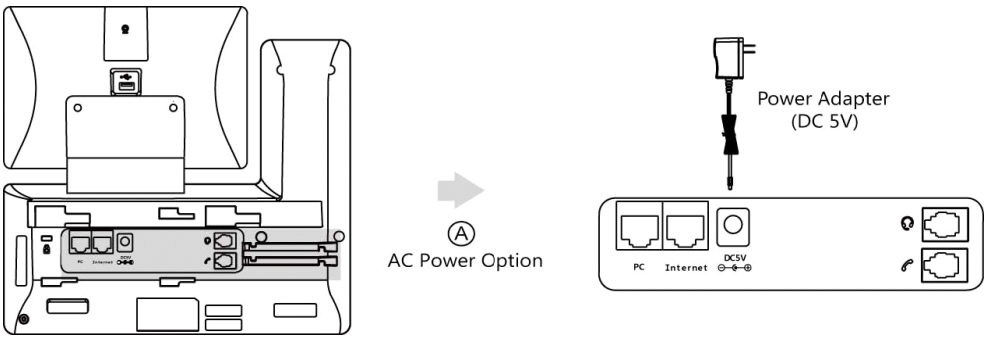
There are two options to connect the power and the network.

- AC power
- Power over Ethernet (PoE)

AC Power

To connect the AC power:

- Connect the DC plug on the power adapter to the DC5V port on the phone and connect the other end of the power adapter into an electrical power outlet.

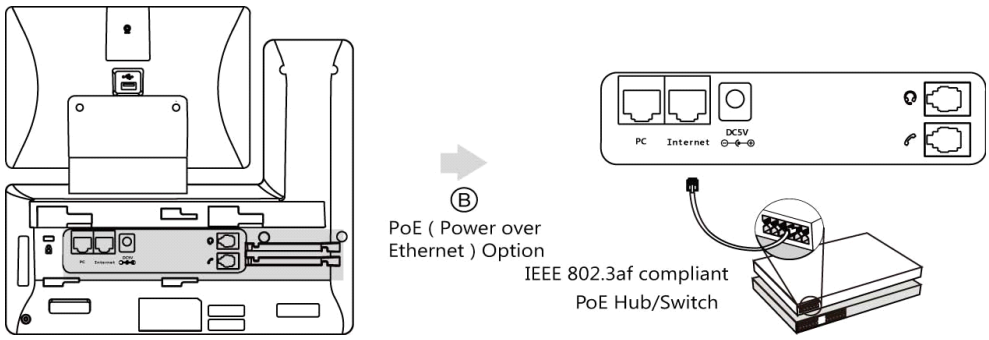


Power over Ethernet (PoE)

With the included or a regular Ethernet cable, the IP Phone can be powered from a PoE-compliant switch or hub.

To connect the PoE:

- Connect the Ethernet cable between the Internet port on the phone and an available port on the in-line power switch/hub.



If in-line power switch/hub is provided, you don't need to connect the phone to the power adapter. Make sure the switch/hub is PoE-compliant.

Do not unplug or remove power while the phone is updating firmware.

After the IP Phone is assembled and connected to the power supply, it automatically begins the initialization process.

During this process, the IP Phone displays the start up screen "Welcome Initializing…please wait".

Once the IP Phone is initialized, it displays two different phone modes:

- Standard SIP
- Extended SIP

- Select Extended SIP, to operate the IP Phone in the extended mode. As soon as you select this mode, the booting process initiates again and the start up screen displays "Welcome Initializing…please wait". After the IP Phone is initialized, it attempts to contact a DHCP Server in your network to obtain valid IPv4 network settings (example: IP address, Subnet Mask, Gateway address, DNS address).You need to configure the basic network parameters of the IP Phone manually, if these are not provided by the DHCP Server or if your network does not support DHCP.

  Refer to the *EXTENDED SPARSH VP710 User Guide*, for detailed instructions:

  - To change the Network Settings of the phone and configure the network parameters.
  - To use Wi-Fi for connectivity and configure its parameters.

- On getting the IP Address and Server Address, the phone initiates Auto Configuration (when DHCP is selected) to download the configuration files from SARVAM UCS.
- On successful download of all configuration files, the phone attempts to register with SARVAM UCS.
- On successful registration, the Home screen appears.

The phone will register successfully, only if the SIP Extension parameters in SARVAM UCS have been correctly configured as per your installation scenario.

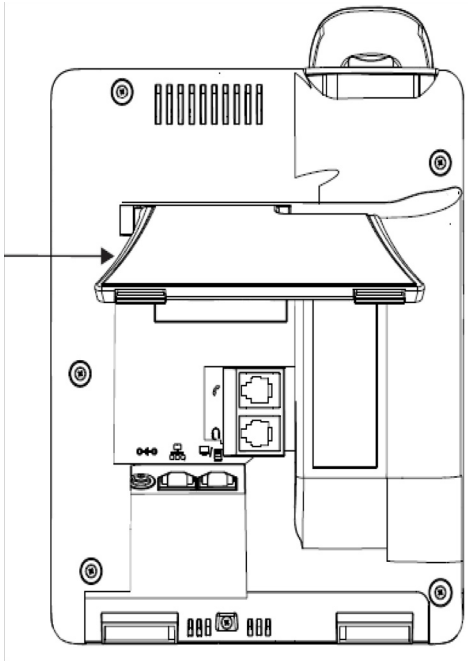
Connecting SPARSH VP210 as Extended SIP Extension

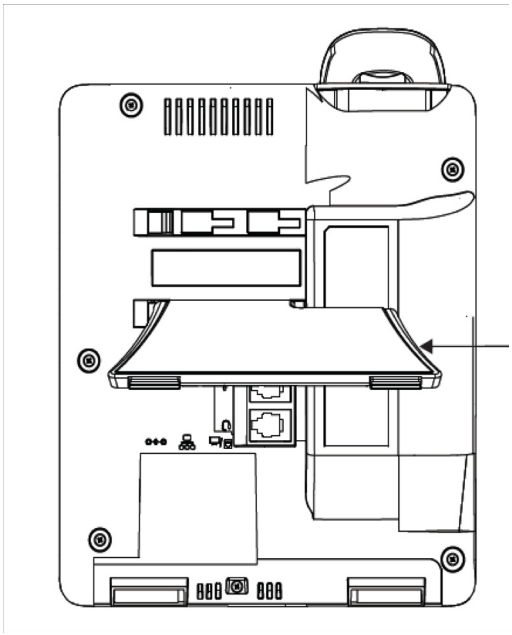
You are recommended to complete the following steps before connecting the Matrix SPARSH VP210 to SARVAM UCS:

- Decide the location where you want to place SPARSH VP210 within your LAN.
- By Default, in SPARSH VP210, the Connection Type selected is DHCP.
- If you want to use the **DHCP Server** for assigning IP Address to the Extended IP Phone, select **DHCP option 224** and **Data Type** as '**String**' and program the LAN or WAN IP Address /Domain Name of SARVAM UCS and SPARSH Port in the format **"IP_Address:Port"** in your LAN DHCP Server as per your installation scenario.
- Log in to *Jeeves*. For instructions, read the topic "Configuring SARVAM UCS".
- You must configure the necessary parameters in SARVAM UCS so that SPARSH VP210 can register as a SIP Extension. For instructions, see "Configuring Matrix SPARSH VP210".

Now, follow the steps described below to install SPARSH VP210.

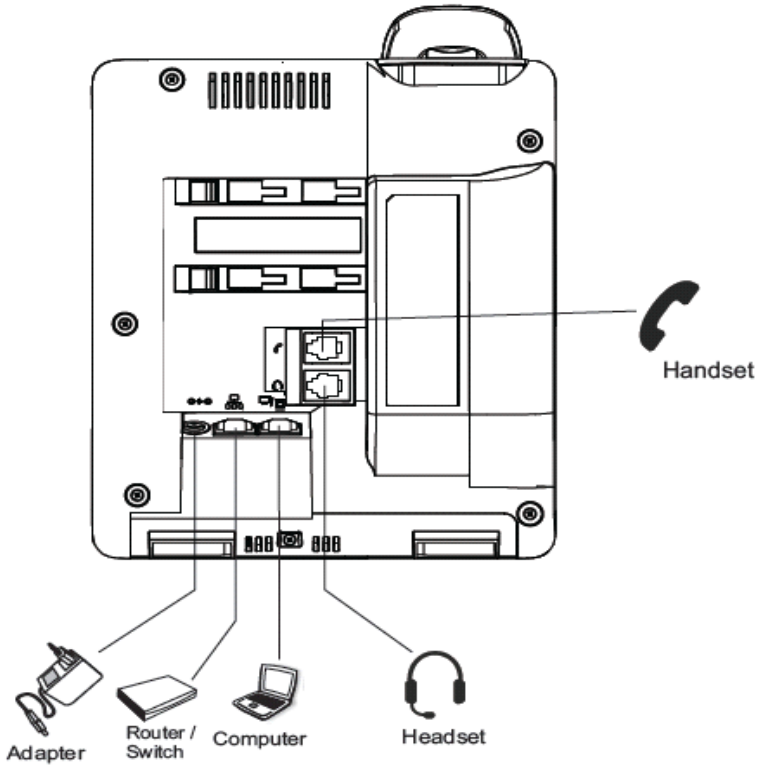
1. Unpack the SPARSH VP210 box and verify package contents.
2. When you mount the phone on a desk, you can attach the Foot Stand in two ways at **45° Angle** or at **55° Angle**.





- Decide which of these positions would work for you best and accordingly attach the Foot Stand.

  Refer to the diagram below for connectivity.



3. Connect the Handset to the Phone body.

- Plug the long straightened end of the Spring Cord into the handset jack at the bottom of the phone, marked with the handset symbol .
- Plug the other (short straight) end of the Spring Cord into the jack at the bottom of the handset.

4. If you want to use a Headset (not supplied) with your phone, You may plug in a headset with an RJ9 connector into the headset port at the bottom of the phone, marked with the symbol .
5. To connect the LAN, Port , plug one end of the Ethernet Cable into the LAN Port at the bottom of the phone marked with the symbol and the other end to the IP Network — A Router or LAN Switch.
6. To connect your phone to a computer on your desk, plug one end of the Ethernet Cable (not supplied with this phone) into the PC Port at the bottom of the phone, marked with the symbol and the other end into the LAN Port of your PC/LAN Switch.
7. It is a PoE enabled phone and can be powered over Ethernet by connecting it to a PoE enabled LAN Switch (IEEE 802.3af Compliant). In this case you need not connect the Power Adapter.

   If you do not want to use PoE, plug in the connector of the Adapter into the power jack (DC Jack) at the bottom of the phone, marked with the symbol . Plug in the Power Adapter into a power outlet.

If both the power options, that is, PoE as well as Power Adapter are available to the phone, then the phone will derive power from the PoE enabled LAN Switch.

The IP phone should be used with Matrix original power adapter (5V/0.6A) only. The use of any third-party power adapter may cause damage to the phone.

8. Switch ON power supply.

   When you power the phone, the boot process will be initiated in the following sequence.

   - The LCD display will light up and booting message appears.
   - While loading the application then the loading message appears on the phone display.
   - The phone makes DHCP Discovery and fetches its IP Address and Server Address from the DHCP Server.

If you want to change the Server Settings, press Settings.

Refer to the SPARSH VP210 (Extended) User Guide, for detailed instructions, to change the Network Settings of the phone and configure the network parameters.

- On getting the IP Address and Server Address, the phone initiates Auto Configuration to download the configuration files from SARVAM UCS.
- On successful download of all configuration files, the phone attempts to register with SARVAM UCS.
- On successful registration, the Home screen appears.

The phone will register successfully, only if the SIP Extension parameters in SARVAM UCS have been correctly configured as per your installation scenario.

Refer to the ***SPARSH VP210 (Extended) User Guide*** to know more.

Starting Up ETERNITY PENX

Power ON

1. If you have completed all the installation tasks, connect the three-prong plug of the power cord from the PENX into the AC outlet, and switch on the power supply.

2. Observe the Reset Cycle.

Reset Cycle

- Reset Cycle (Power-ON Self Test) takes about 2 minutes to finish.

- All the LEDs of the system and the keys of the DKP/SIP devices attached to the System are turned on.

  When the Reset Cycle is successful, the default Extension Access Codes loaded by the system and the date and time of the Real Time Clock of the system will appear on the LCD display of the Digital Key Phones you have connected with the system.

CHAPTER 9

Configuring ETERNITY GENX

ETERNITY GENX provides a Graphic User Interface (GUI), Jeeves, the proprietary web-based configuration software of Matrix. The built-in web server Jeeves allows you to select the Application you want to run on the ETERNITY GENX Platform.

The accessibility to the Web-based GUI is secured by a password. This password cannot be used to configure the system using commands.

To be able to access Jeeves,

- the LAN/WAN Port of the system must be connected with a stand-alone PC or in a LAN. The default IP Address of the LAN Port is 192.168.2.100, WAN Port is 192.168.1.100 and the Subnet Mask is 255.255.255.0.
- a web-browser, either Internet Explorer 7 or later or Mozilla Firefox 3.5.1 or later, must be installed on the PC.

If the computer for accessing Jeeves is connected in a LAN Switch and the WAN Port of the system is connected behind a NAT router, make sure that both the LAN and WAN connections are in different Subnet Masks.

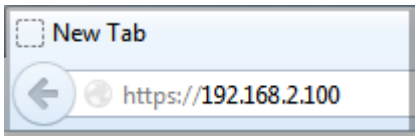
To login,

- Open the browser (Internet Explorer/Mozilla Firefox) on the PC (Standalone or LAN PC) to which the system is connected.
- Make sure the IP Address of the computer and the LAN Port of the system do not conflict, and that both are in the same Subnet.

  The default IP Address of the LAN Port is: **192.168.2.100**

  The default Subnet Mask of the LAN Port is: **255.255.255.000**

  Change the Subnet of the computer, if necessary.
- In the address bar of the browser, enter **https://192.168.2.100**.

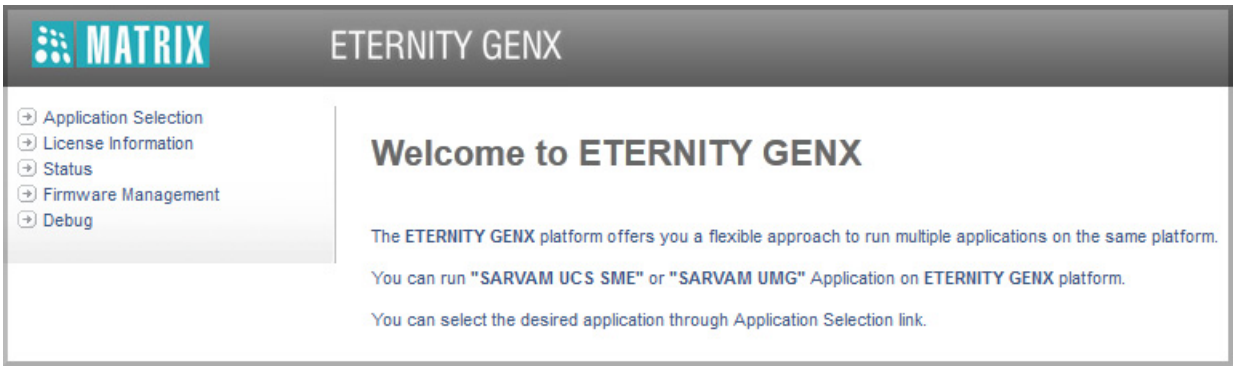


*If you enter the IP Address **192.168.2.100** directly, you will be redirected to the HTTPS protocol for secure access. Click the **https://192.168.2.100** link provided.*

- The **Login** page will open.
- In **Login Password**, enter ***1234***, the default Password.

- Click the **Login** button.
- On successful login, the **Home** page of Jeeves opens.

  The left navigation bar displays the links — **Application Selection, License Information, Status, Firmware Management** and **Debug**.



Application Selection enables you to select the application you wish to run on the ETERNITY GENX platform. You may select — SARVAM UCS SME or SARVAM UMG.

License Information displays the License key along with the License details of the applications.

Status displays the system details and the status of all the ports.

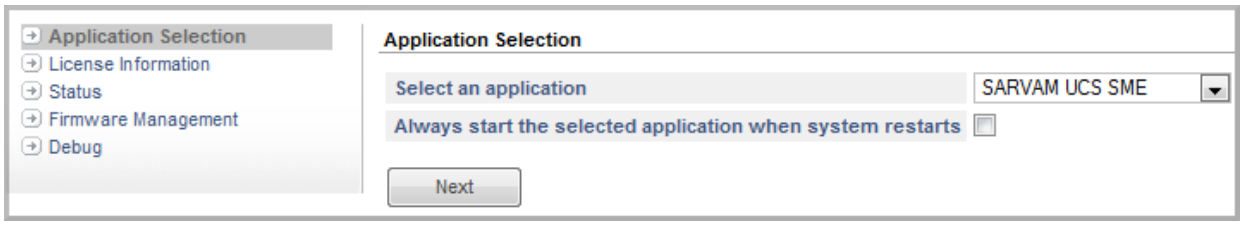
Firmware Management enables you to manage the upgradation of the system software with a click of a button.

Debug allows you to enable and configure the debug settings.

Application Selection

Through **Application Selection**, you can select the application you wish to run on the ETERNITY GENX Platform.

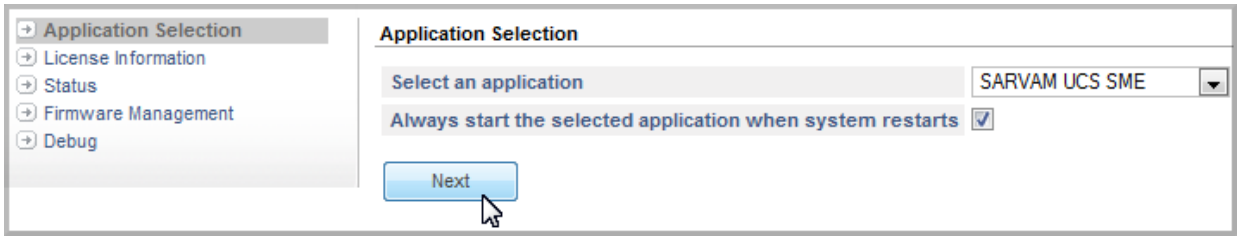
- In **Select an Application**, you may select the required application — SARVAM UCS SME or SARVAM UMG. Default: SARVAM UCS SME.



- Select the **Always start the selected application when system restarts** check box, if you want the selected application to start whenever the system restarts.

  Keep the check box disabled only if you want to select the application to be run on the ETERNITY GENX platform everytime the system restarts. Default: Disabled.

*To run the selected application directly whenever the system restarts, **Enable** the **Always start the selected application when system restarts** check box.*



- Now, click on the **Next** button to proceed to the SARVAM UCS SME Configuration.

When you click the Next button, you will be redirected to the SARVAM UCS SME Application. To return back to this page, see "Switch Application" in "System Details".

Once you are redirected to the SARVAM UCS SME Application,

- If you have purchased the SARVAM UCS SME License, you must activate it. For detailed instructions, refer "How to activate your License".

- If you have not purchased the license and you wish to use the features on trial basis, you can use the Demo Provision. Demo Provision enables you to use the SARVAM UCS SME application, free of cost for a period of 60 days.

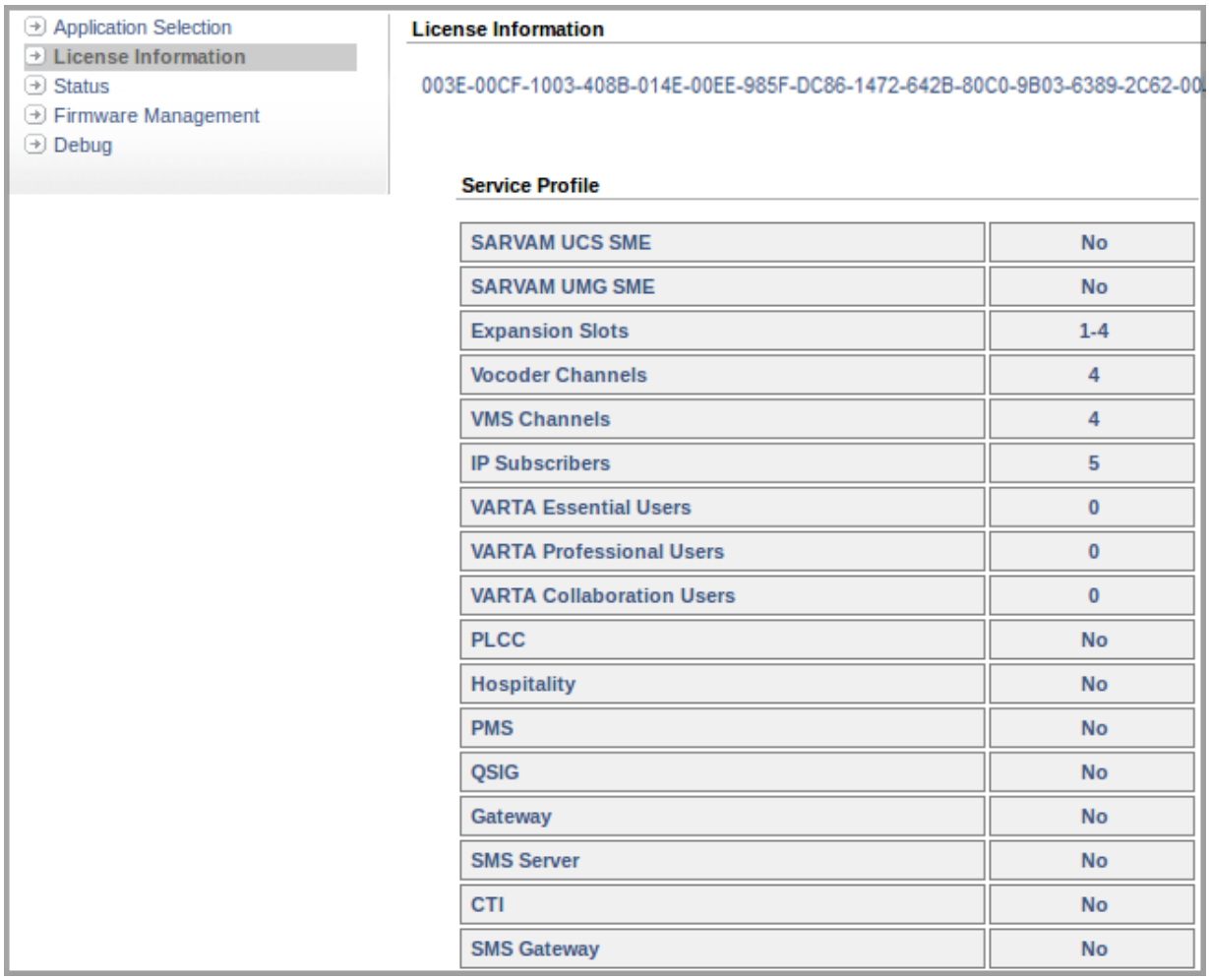
  During the Demo Provision you can access and use all the features and functionalities[112] supported by the application. For detailed instructions, refer "Pre-activated Licenses".

112. *The number of Vocoder channels that will be supported in demo period will be as per the license you purchase.*

- If you do not have the license for the SARVAM UCS SME Application and you do not start the Demo Period, the system will disconnect all the connected calls[113] from any port after 60 seconds.

License Information

License Information displays the License key along with the License details of the applications. Whenever you want to activate license for any software application, you can check the current license key from here.



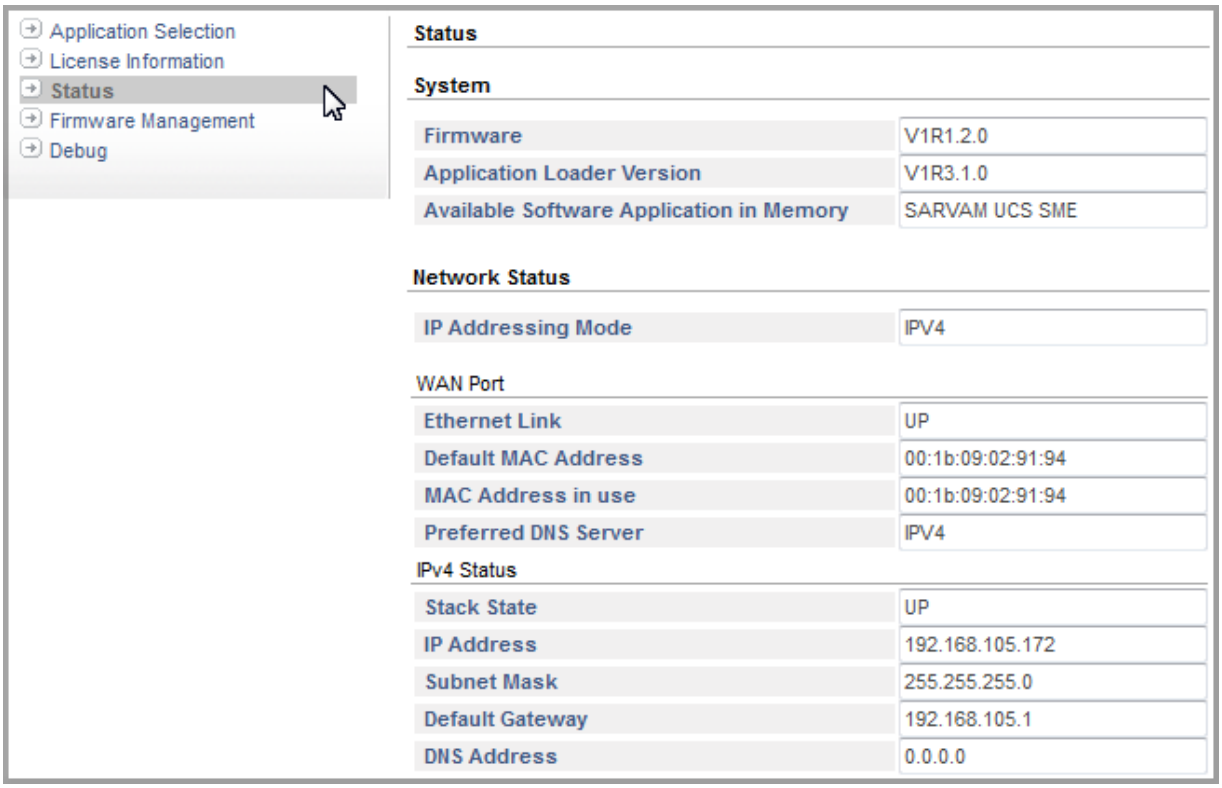
To activate the license, refer topic “How to activate your License”.

To know more regarding the licenses, refer topic “License Management”.

113. *Connected calls means where speech is connected between the calling party port and the called party port even if the called party port is not matured.*

Status

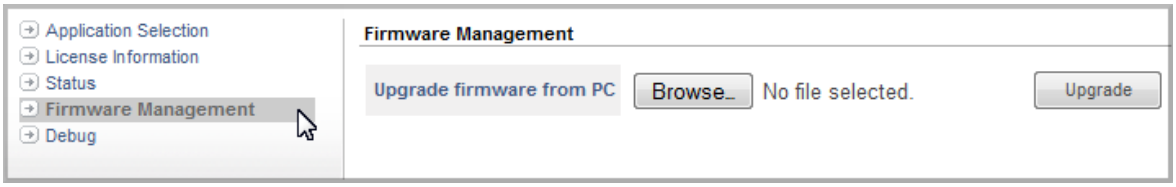
Status displays the system details, that is, Product name and the Available Software Application, Network status, WAN Port and LAN Port status.



Firmware Management

To upgrade firmware of ETERNITY GENX,

- In **Upgrade Firmware from PC**, click the **Browse** button to reach the location on the local disk where the firmware files are stored in your PC. Make sure that the file is a zip file with **.zip** extension.



- After selecting the required firmware zip file from the PC, click the **Upgrade** button.

  The system starts the upgradation process. After successful upgradation and validating the file, the system restarts with the upgraded firmware.

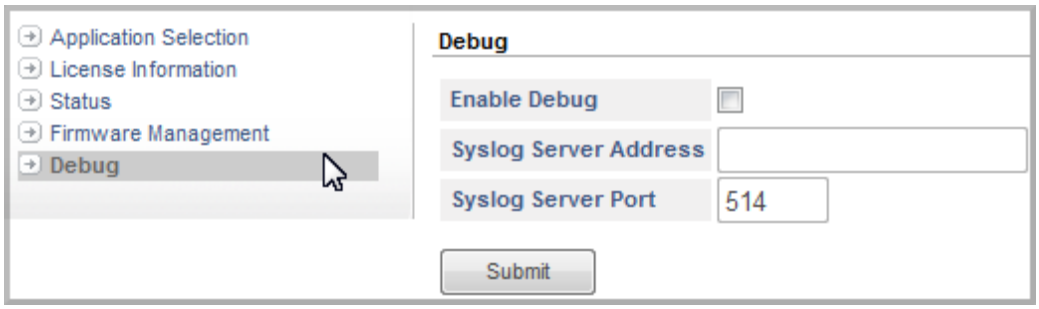
If you select a file other than zip file, an Alert Message is displayed when you click the Upgrade button.

Debug

ETERNITY GENX supports Syslog Client for debugging. Debug messages are sent to the remote Syslog Server.

- Select the **Enable Debug** check box to enable system debug. Default: Disabled.

  You will be able to configure the Debug Settings only after you enable this check box.



- In **Syslog Server IP Address**, enter the remote Syslog Server IP Address. Default: Blank.

  In **Syslog Server Port**, enter the port number. The range of the server port is 514, 1024 to 65535. Default: 514.

CHAPTER 10

Configuring SARVAM UCS

The SARVAM UCS can be configured using the following tools:

- A web-based graphic user interface (GUI), called Jeeves.
- A telephone
- Serial Communication Port

Each of these is explained in detail in this chapter

Due to security concerns, the default system settings have been changed. If you have purchased a new system with Firmware later than V1R6.7, the new default settings will be applied automatically. Refer to “Modified default parameter values for Firmwares later than V1R6.7”. With these default setting the incoming calls will be placed on the system but outgoing calls (except calls between extensions) will not be routed. For configuring the parameters to route outgoing calls refer to “Outgoing Call Routing”.

If you are upgrading the system, refer to “After updating Firmware later than V1R6.7” and “Modified default parameter values for Firmwares later than V1R6.7”.

Configuring using Web-based GUI: *Jeeves*

SARVAM UCS provides a Graphic User Interface (GUI), Jeeves, the proprietary web-based configuration software of Matrix. It is a Web server built into the SARVAM UCS.

The default IP Address of the LAN Port is 192.168.2.100, WAN Port is 192.168.1.100 and the Subnet Mask is 255.255.255.0.

Jeeves allows system configuration at three levels: System Engineer, System Administrator and the Front Desk User.

A distinct set of features and facilities can be configured at each of these levels. The accessibility to each level is secured by a password. These passwords cannot be used to configure the system using commands.

These passwords can be changed using Jeeves only.

It is possible to configure the SARVAM UCS from any location using Jeeves. You can use Jeeves to configure the system On-site (where it is installed) and Off-site, from a remote location.

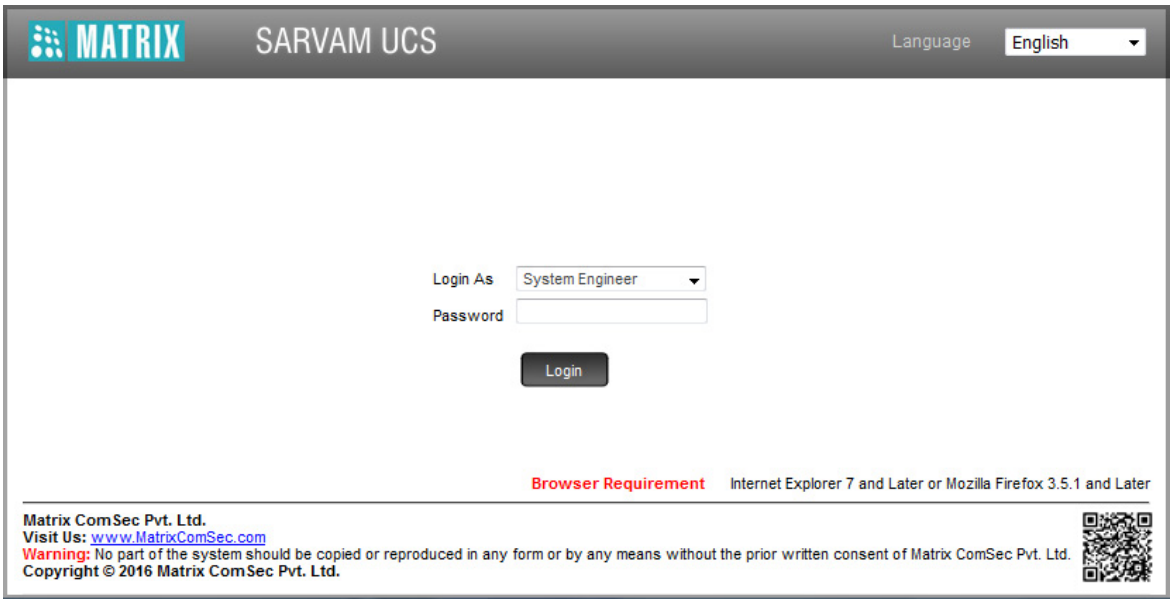
1. System Engineer Mode

At the System Engineer level, the entire system configuration, all configurable features and facilities of the SARVAM UCS can be changed to match user requirements and preferences.

To be able to do this, the System Engineer must enter the System Engineer (SE) configuration mode, by logging into Jeeves as System Engineer.

Only the System Engineer, who installs, configures and maintains the is allowed access to this mode. Hence access to the SE mode is protected by means of a password, referred throughout this document as the SE Password.

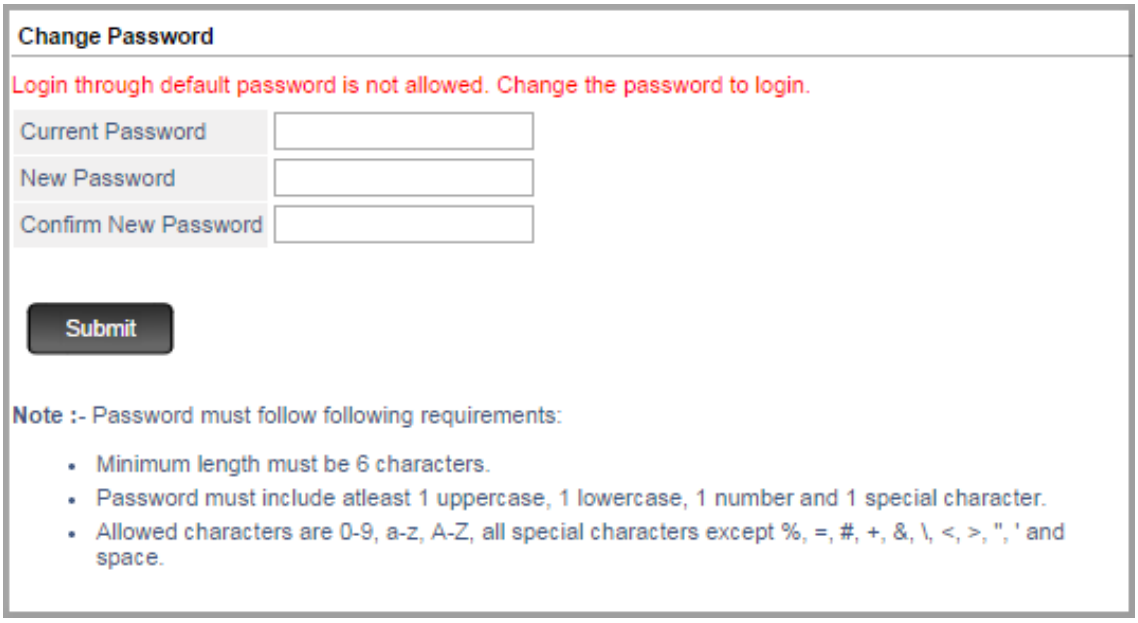
To login into the SE mode, on the login page in **Login as** select **System Engineer.**



- In **Password**, enter the default SE password, 1234.
- Click **Login**.

Before you start configuring the system, if you wish to view or download the SARVAM UCS Quick Start/ SARVAM UCS Hospitality Manual or other related documents, you can scan the QR Code present in the login page of Jeeves.

- You are prompted to change the default password.



- In **Current Password**, enter the default SE password.
- Enter the **New Password**. The new password must be:
  - a minimum of 6 characters to a maximum of 12 characters.
  - all ASCII characters (except Percentage **%**, Hash **#,** Equal to **=,** Plus **+**, And **&**, Backslash **\,** Less than **<,** Greater than **>,** Apostrophe **',** Double Quote **"** and **Space**) are allowed.
  - include atleast one upper-case, one lower-case, one number and one special character.
- In **Confirm New Password**, re-enter the new password to confirm.
- Click **Submit**. You will be re-directed to the Login page again.
- Now, in **Login as** select **System Engineer** and in **Password** enter the new password.

  You will be prompted to change the default **SE Extension Password.**

- Enter the **New Password**. The new password can be a minimum of 4 digits to a maximum of 12 digits. The valid digits are from 0 to 9.

You cannot set 1234 as the New SE Extension Password as it is the default SE Extension Password.

- In **Confirm New Password**, re-enter the new password to confirm.

- Click **Submit** to save your new password. The Home page will open.

You can configure only the basic network and debug parameters from your telephone using the default SE Extension Password, 1234. However, if you want to configure other parameters from your telephone, you must change the default SE Extension Password. You can change the default SE Extension Password using Jeeves only.

To configure these parameters using the default SE Extension Password, you must connect a SLT or DKP extension to the system. To know the detailed list of parameters, which can be configured using the default SE Extension Password, see "Basic Network" and "System Debug".

As this password is meant for restricting access to the SE mode, we strongly recommend you to:

- *Keep the password secret.*
- *Select a complex password that cannot be easily guessed.*
- *Change the password regularly. See "System Security".*
- *Not use the **"Remember Password**" property of your Web Browser.*

- *Each login session into the SE Mode is set to 60 minutes by default. So, the login session will expire at the end of 60 minutes. The duration of the login session can be extended or shortened according to your preference by changing the settings for the 'Web Configuration Timer'. Refer the topic "Changing Login Session Time Out of Jeeves".*

- *It is possible for four users to simultaneously log into the System Engineer Mode of Jeeves. It is also possible to log out all of these users at once or log out any of these users selectively. Refer the topic "Logging Out Users from Jeeves".*

Quick Installation Wizards

For the ease of installation, as well as to simplify and speed up the process of setting up the SARVAM UCS, the Jeeves offers two Wizards, namely:

- **Quick Installation Wizard - Standard PBX:** This wizard helps the Installer/System Engineer to quickly set-up the SARVAM UCS for the standard UC (Enterprise) Application.

  Using this Wizard, the Installer/System Engineer can configure as much as 80 percent of the system configuration, covering all the parameters necessary for the functioning of the system. For advanced configuration of features and facilities, the Installer/System Engineer must use the "Using Configuration" mode.

  Detailed information on this Wizard and instructions for using it for system configuration are given later in this chapter. Refer the topic "Using Quick Installation Wizard - Standard PBX".

- **Quick Installation Wizard - Hotel:** This is a special wizard for the Hospitality Application of the SARVAM UCS. The Hotel Installation Wizard simplifies the configuration process and helps the Installer/System Engineer to configure the system for the special telephone and guest/patient management facilities and features for hotels and hospitals.

  You may use this Wizard if you have deployed the SARVAM UCS as a Hospitality Application. The Hotel Installation Wizard is recommended to be used when configuring the SARVAM UCS for the first time.

  Refer the separate *SARVAM UCS Hospitality System Manual* to know more.

Configuration

While the Quick Installation Wizard provides a fast-track way for system configuration, detailed and advanced configuration of the system can be done only from the links under Configuration of Jeeves.

The Configuration, as the title itself suggests, enables the Installer to change the values of all configurable parameters of the system, including those not covered by the Wizard.

Refer the topic "Using Configuration" for more information and instructions.

As many as four System Engineers can simultaneously login into SE Mode and configure the system via the Configuration and the Quick Installation Wizard-Standard PBX. However, it is recommended that multiple login be avoided when using the Quick Installation Wizard and the use of the Wizard be restricted to a single person only.

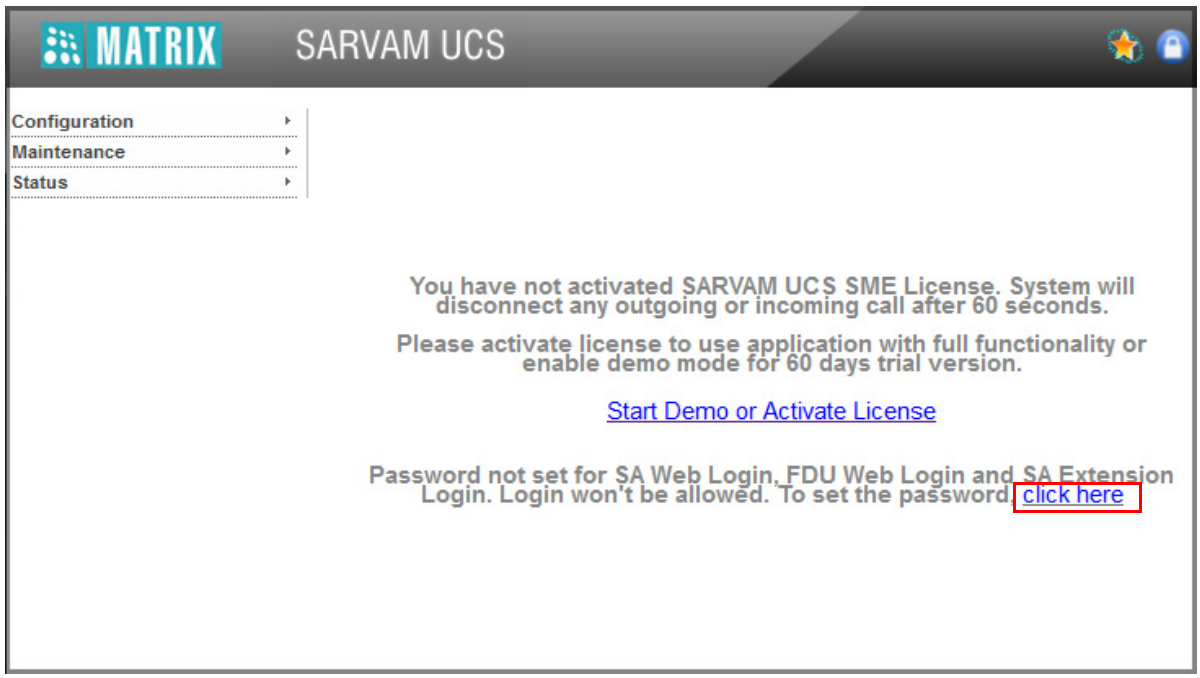
2. System Administrator Mode

At the System Administrator level, the settings of the features for the Extension users can be changed, and various system activity logs and reports such as fault log reports, Station Message Detail Recording reports, and Hotel reports, can be captured and printed.

For this, the System Administrator, who is usually the operator or receptionist, must login as System Administrator into Jeeves.

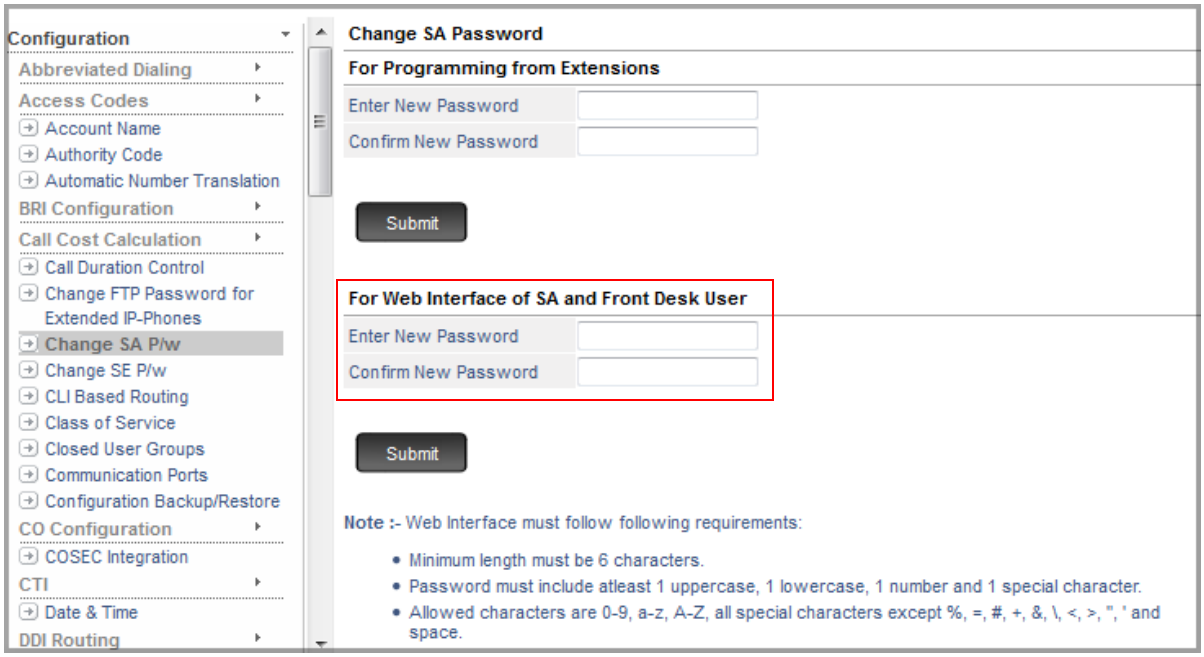
To login into the SA mode,

- On the Login page, in **Login As** select **System Engineer**. To know more, refer to "1. System Engineer Mode".
- Click **Login**. The Home page opens.



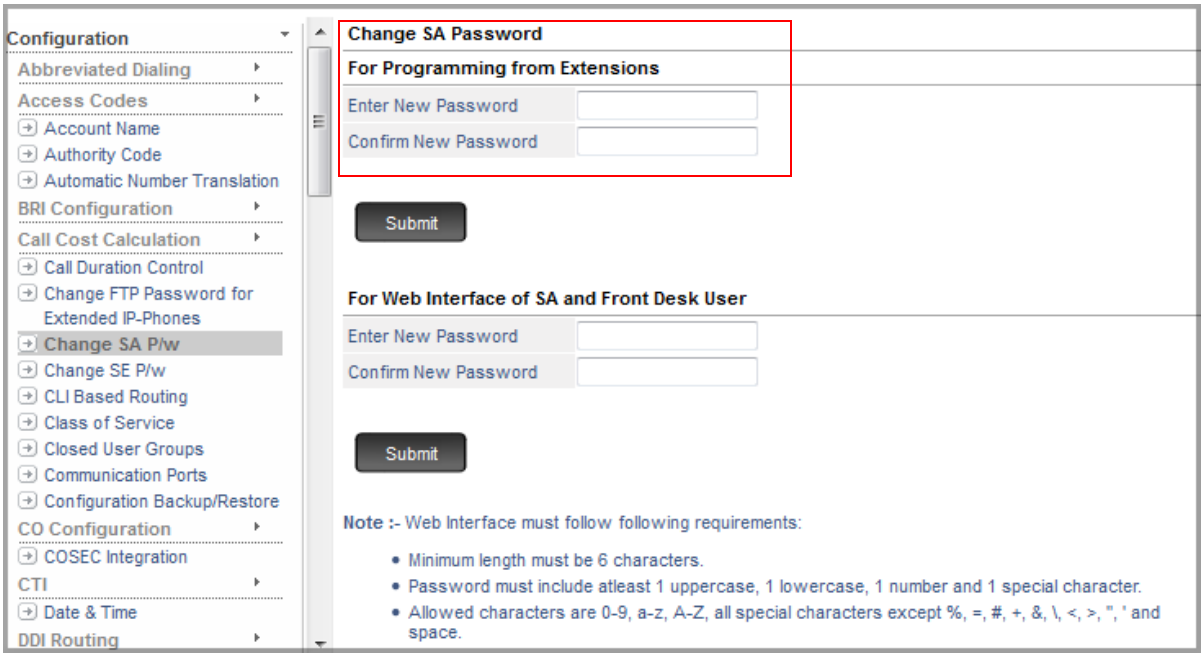
- To set the password for **SA Web Login, FDU Web Login** and **SA Extension Login**, click on the link.

For Accessing Jeeves



- Under **For Web Interface of SA User and Front Desk User**,
  - **Enter New Password**. All ASCII characters (except Percentage **%**, Hash **#,** Equal to **=,** Plus **+**, And **&**, Backslash **\,** Less than **<,** Greater than **>,** Apostrophe **' ,** Double Quote **"** and **Space**) are allowed. The new password must be:
    - a minimum of 6 characters to a maximum of 12 characters.
    - include atleast one upper-case, one lower-case, one number and one special character.
  - In **Confirm New Password**, re-enter the new password to confirm.
  - Click **Submit** to save your new password.

For Programming from Extensions



- Under **For Programming from Extensions**,
  - **Enter New Password**. The new password can be a minimum of 4 digits to a maximum of 12 digits. The valid digits are from 0 to 9.
  - In **Confirm New Password**, re-enter the new password to confirm.
  - Click **Submit** to save your new SA password.
- Log out of SE mode to enter SA mode.

The SA password is a code for preventing unauthorized access to the SA mode. As this password is meant for restricting access to the SA mode, we strongly recommend you to:

- *Keep the password secret.*
- *Select a complex password that cannot be easily guessed.*
- *Change the password regularly. See "System Security".*
- *Not to use the **"Remember Password"** property of your Web Browser.*

You can log into the SA mode through Jeeves or from extensions only after you have set the password from SE mode. The password can be set using Jeeves only and it must be as per the specifications given below:

- *It must be a minimum of 6 characters and a maximum of 12 characters.*
- *It must include at least one upper-case, one lower-case, one number and one special character.*
- *All ASCII characters (except Percentage **%**, Hash **#,** Equal to **=,** Plus **+**, And **&**, Backslash **\,** Less than **<,** Greater than **>,** Apostrophe **' ,** Double Quote **"** and **Space**) are allowed.*

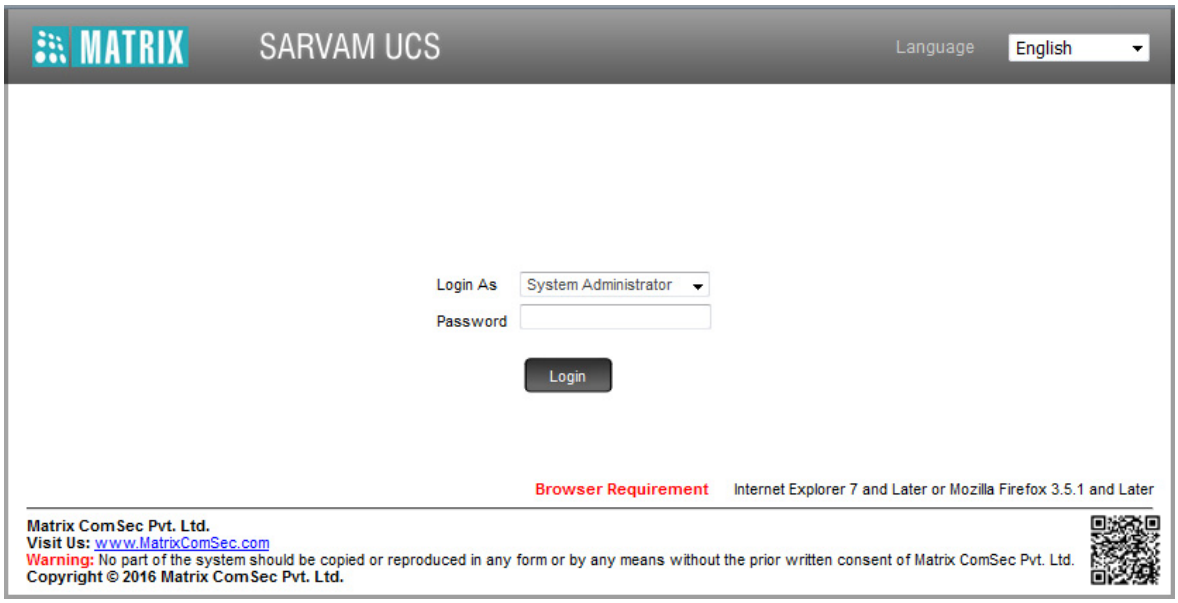
To provide additional security,

- *the password will be valid for 90 days only and you will not be able to login with the existing password after 90 days. You will be prompted to change the password.*
- *if you enter a wrong password for five times consecutively within 10 minutes, the system will block the source IP Address for 10 minutes. This activity will be logged in the "System Activity Log".*

Once the SA password is set, you can log into the SA mode through Jeeves.

To enter the SA mode,

- On the **Login** page, in **Login as** select **System Administrator**.



- In **Password**, enter the new SA password.
- Click **Login**.

3. Front Desk User Mode

The Front Desk Mode is relevant only for the Hospitality Application of the SARVAM UCS.

This mode is meant for the personnel at the Front Desk of the Hotels/Motels, allowing them access to and operation of the hospitality features of SARVAM UCS, for example: Check-In/Out of guests, Changing Room Occupancy and Clean Status, setting Call Budgets for guests, setting Wake-up calls, Reminders, Do Not Disturb for guests, printing Call Reports and Hotel reports, and several others.

The Front Desk User mode too is password protected. Refer the *SARVAM UCS Hospitality System Manual* to know more.

- *If you select the language on the 'Welcome' page or on any of the Login pages, it is valid for the session only. The default language will be applied on next login.*

- *When you select the "Configuring 'Region'" for the country in which SARVAM UCS is being installed, the system will load the country-specific default settings and automatically select the local language of the country, if the local language is among the above listed languages. The default local language set on selecting the Region will be applied for every login session, unless you select another language as the default local language.*

- *The default local language set on selecting the Region can also be changed from the "System Parameters" page of Jeeves.*

The system can be configured both, 'on site' and from a remote location (refer "Remote Programming"), and from multiple extensions simultaneously, without its functioning being affected.

With the default setting in the system, outgoing calls (except calls between SIP extensions) will not be routed. For details, refer "Outgoing Call Routing".

Configuring using a Telephone

The SARVAM UCS can be configured by dialing the relevant command strings from a telephone. The telephone may be any Single Line Telephone (SLT), the Matrix proprietary Digital Key Phone (DKP), EON, or the Matrix Extended IP Phone connected as an extension of the SARVAM UCS.

For the ease of operation, you may use EON. Using EON has the advantage as,

- you can view the command strings that you have keyed in on the LCD display of EON.
- you will get prompts and confirmatory messages on the phone's LCD display, in addition to confirmation tone played to you.
- you can dial alphanumeric command strings.

All these facilities will not be available to you on an SLT.

System Configuration using a Telephone can be done at two levels: System Engineer Level and System Administrator Level.

A distinct set of features and facilities can programmed at each of these levels.

It is possible to configure the SARVAM UCS from any location using a Telephone. You can use a Telephone connected as an extension of the SARVAM UCS to configure the system On-site (where it is installed). You can also configure the system using a Telephone Off-site, from a Remote location, using the "Direct Inward System Access (DISA)" feature of the SARVAM UCS. You can access both the System Engineer mode as well as the System Administrator mode from the remote location. Refer the topic 'Remote Configuration using a Telephone' later in this chapter for instructions.

Make sure you change the default SE Extension password, before configuring SARVAM UCS from your telephone. You can only configure the basic network and debug parameters using this default SE Extension Password. The default SE Extension password can be changed only from Jeeves. To know more, refer to "Configuring SARVAM UCS".

1. System Engineer Mode

At the System Engineer level, the entire system configuration, all programmable features and facilities of the SARVAM UCS can be configured by dialing command strings referred to in this document as SE Commands.

The System Engineer has Full Programming Access when configuring the system using a Telephone. In other words, the System Engineer can configure the entire system, nearly all the programmable parameters, using a telephone.

Refer the topic *"Using Configuration"* for more information and instructions.

SE Commands

These are number strings to be dialed by the System Engineer on entering the SE mode via a telephone. SE Commands are unique to the feature/facility being programmed. Hence SE Commands for configuring a particular feature/facility are presented in description of that feature/facility in this manual.

For a complete List of all SE Commands, refer *"System Commands"* and for SE Commands for PENX, refer *"System Commands for PENX"*

SE Password for programming from extensions

The SE password is a code used to restrict unauthorized access to the SE Mode. The password can be a minimum of 4 digits to a maximum of 12 digits. The valid digits are from 0 to 9. The default SE Password is 1234. To avoid unauthorized access, we recommend you to change the password. Make sure it is strong and is kept confidential.

Refer the topic "System Security" for instructions on how to change the SE password. In case the System Engineer forgets the password, it can be restored to the default password and changed again. Read the section "Default Settings" for instructions on restoring the default SE password.

There is no restriction on the number of persons who can simultaneously enter SE mode from a Telephone and configure the system.

Entering the SE mode using a Telephone

The telephone that you use to enter the SE programming mode may be a DKP or an SLT connected as extension phone of the SARVAM UCS.

To enter the SE mode via a DKP/SLT (for all countries except New Zealand):

- Dial ***1#91-SE Password***
- You get programming tone to indicate entry into the SE mode.
- Now, dial the desired SE Command to configure.
- You get confirmation tone and message on the LCD if using EON.

To enter the SE mode via a DKP/SLT (for New Zealand):

- Dial ****1#91-SE Password***
- You get programming tone.
- Dial the desired SE Command to configure.
- You get confirmation tone and message on the LCD if using EON.

To exit the SE mode:

- Dial ***00.***
- You will hear the dial tone of the SARVAM UCS.

- *If the SE Password you entered is incorrect, you will be played an Error Tone and an Error Message on EON.*
- *The system accepts and executes the command immediately, but it takes approximately 2 minutes to save a command. So, it is advisable that you do not turn OFF the system for 2 to 3 minutes after entering the last command.*

2. System Administrator Mode

Make sure you set the SA Extension password, before configuring SARVAM UCS from your telephone. The SA Extension password can be set only from Jeeves. To know more, refer to "2. System Administrator Mode".

At the System Administrator level, you can set/cancel features settings for extensions, capture and print various system activity logs and reports.

The Operator or Receptionist, who usually administer the system, must enter the System Administrator (SA) mode and then after dial command strings referred to as SA Commands from a telephone.

SA Commands

SA commands consist of a prefix string 1072, followed by the Command string. For example: the SA command for setting Do Not Disturb for an extension is ***1072-001-extension number-1***, where 1072 is the prefix string and 001 is the command string.

The Prefix string in the SA Command (1072) can be changed by the installer/System Engineer. However, the command strings of the SA Command (001 in the above example) cannot be changed.

The command for entering SA mode is also non-programmable. Refer "Access Codes" in the chapter Features and Facilities.

To know how to use or change feature settings with SA Commands, please refer the description of individual features under "Features and Facilities".

For a complete list of SA Commands, refer "SA Commands".

SA Password for programming from extensions

The access to SA mode may be protected by means of an SA password.

The SA password is code for preventing unauthorized access to the SA mode. The password can be a minimum of 4 digits to a maximum of 12 digits. The valid digits are from 0 to 9. To avoid unauthorized access, we recommend you to change the password. Make sure it is strong and is kept confidential. It can be changed and reset by the System Engineer.

Refer the topic "System Security" for instructions on how to change and reset the SA Password using SE commands.

- *When the SARVAM UCS is used in the Standard Application, you can enter SA mode only from extensions which have the features 'SA Mode' and/or 'SA Extension' enabled in their Class of Service.*
- *When the feature 'SA Extension' is enabled in the Class of Service of an extension, the extension will always be in SA mode. You do not need to enter SA mode by dialing the SA password. You can enter the SA mode by dialing the SA command prefix string.*
- *When the feature 'SA Mode' is enabled in the Class of Service of an extension, dialing of the SA Password is required to enter the SA mode. SA Commands can be dialed only after successfully entering the SA Mode.*
- *There is no restriction on the number of persons who can simultaneously enter and operate from the SA mode using a telephone.*

Entering SA Mode using a Telephone

You may use a DKP or an SLT extension of the SARVAM UCS to enter the SA Mode.

To enter SA mode via an extension phone (for all countries except New Zealand),

- Dial ***1#92-SA Password***[114]
- You get a confirmation Tone of the SARVAM UCS to indicate successful entry into the SA mode.
- Dial SA Command strings: ***1072-Feature Access Code.***
- You get a confirmatory tone and text message on the phone display (if using EON).
- Replace handset to exit SA mode.

OR

- Dial ***1#92***

To enter SA mode from an extension phone (for New Zealand),

- Dial ****1#92-SA Password***[115]
- You get a confirmation Tone.
- Dial SA Command strings: ***1072-Feature Access Code.***
- You get a confirmation tone and text message on the phone display (if using EON).
- Replace handset to exit SA mode.
  OR
- Dial ****1#92***

You can exit from the SA mode automatically or manually. To exit the SA mode automatically, you must configure the SA Mode Timer. On the expiry of the set time, the system disconnects the extension phone from the SA mode. This Timer is loaded automatically every time a new SA command is issued.

To program the SA Mode Timer:

- Dial ***1072-016-SA Mode Timer***
  Where,
  SA Mode Timer = 000 to 255 minutes.

- You can also exit the SA mode before the SA Mode Timer expires by dialing this command. If the SA Mode Timer is set to 000 minutes, you can exit the SA mode only by dialing the command.

You may change the SA Mode Timer from the SA mode of Jeeves also.

Changing Login Session Time Out of Jeeves

As mentioned earlier, each login session of Jeeves has been set to 60 minutes by default. The Login session will expire at the end of 60 minutes. The duration of the Login session can be changed as per your preference by changing the settings of the 'Web Configuration Time Out Timer'.

To do this:

- Dial ***1#91-SE Password*** to enter SE mode from a DKP/SLT.
- Dial ***2118-Time***
  Where,
  Time is from 001 to 255 minutes.
  For example, to set log out time to 45 minutes, dial ***2118-045.***
- Press 'Enter' key to save setting, if using EON.

---

114. If the password is less than 12 digits, you must dial # after the new password to indicate end of dialing. When you dial this command, the system will check if the facility 'SA Mode' is enabled in the Class of Service allowed to the extension from which you have dialed this command. If the SA mode is not allowed, an Error Tone will be played. The Error Tone will be played also when the SA password is entered incorrectly.*
If the facility 'SA Extension' is enabled in the Class of Service of the extension you are using, you can skip this step and directly dial the SA Command (1072-<Command String>).

115. When you dial this command, the system will check if the facility 'SA Mode' is enabled in the Class of Service allowed to the extension from which you have dialed this command. If the SA mode is not allowed, an Error Tone will be played. The Error Tone will be played also when the SA password is entered incorrectly.
If the facility 'SA Extension' is enabled in the Class of Service of the extension you are using, you can skip this step and directly dial the SA Command (1072-<Command String>).

Logging Out Users from Jeeves

It is possible for four users to simultaneously log in as System Engineers and use Jeeves. It is also possible to log out all these users at once. To do this,

- Dial ***1#91-SE Password*** to enter SE mode from a DKP/SLT.
- Dial ***2188*** to log out all users.
- Press 'Enter' key to save setting, if using EON.

Using Quick Installation Wizard - Standard PBX

The Quick Installation Wizard-Standard PBX helps you with the basic configuration of the system in easy steps. It is designed to break down the complexities of programming and can cover as much as 80% of your basic installation requirements.

The advantage of using the Wizard is that it speeds up system configuration, as you configure parameters that are specific to the SARVAM UCS and show only those Trunks and Extensions that are connected to the system for configuration. For instance, if you are installing only 12 out of the 64 CO trunks supported by SARVAM UCS,12 DKP extensions and 240 SLT extensions, the Wizard will show only these many trunks and extension ports for configuration. Thus, the Wizard makes the entire system configuration very focused.

Further, the Wizard can be used for system configuration for first time installation as well as any time post-installation to make modifications in the system configuration.

Structure of the Wizard

The Wizard stores its own set of configuration files and once the installation is completed, it updates the system configuration files with the data you entered. However, the reverse does not work. The changes you make in the system configurations under Configuration will not be reflected in the Quick Installation Wizard.

The Quick Installation Wizard - Standard PBX offers the configuration of the following parameters:

1. Region
2. Pre-requisites
3. Extension Numbers in Range
4. Extension Numbering Plan
5. Trunks
6. Day-Night time
7. Number Patterns
8. Operator
9. Extensions
10. Least Cost Routing (LCR)
11. Call Pickup Group
12. CO Trunks
13. BRI Trunks
14. T1E1 PRI Trunks
15. Mobile Trunks
16. VoIP Network
17. SIP Trunks
18. Emergency Numbers

Help Text

There is a help text to explain the parameters on each screen. You may use this help text to guide you in entering the information.

The Buttons

The Wizard has the following buttons:

- **Next:** Clicking this button will cause the existing values of the parameters in a page to be submitted and takes you to the next page of the Wizard.

- **Submit:** Clicking this button will cause the parameters configured in the page to be submitted, but will not take you to the next page. The 'Submit' button is to be used when there are multiple pages for configuring a facility, for example, configuring extensions, CO trunks. Instead of navigating all the pages, you can access the desired page, make changes and submit them. Similarly, if you want to make any changes post-installation, instead of navigating through all pages of the Wizard and clicking **Done** to effect the changes, you can reach the desired page, make the necessary changes and submit the changes.

  To navigate further, you must click the **Next** button.

- **Skip:** Clicking this button will take you to the next page, without making any changes to the current page. The Skip button is to be used when the Wizard is used post-installation to reach a page which is to be modified, without changing the existing configuration settings on other pages.

- **Undo:** Clicking this button refreshes the page with the parameters configured in the system. You may use this button if you are not sure about the values you entered or have entered incorrect values and wish to start all over again. This button is available only on select pages of the Wizard.

- **Help:** Clicking this button on a page opens a new window, containing Help Text for that page. The window can be maximized.

- **Default:** Clicking this button will populate all the fields of the page with the default values. Use this button when you want to assign default values to parameters.

- **Clear:** Clicking this button will cause the values of all the fields on the page to be cleared. Use this button to make corrections or start entering the values all over again.

- **Exit:** Clicking this button will exit the Jeeves amidst of an activity.

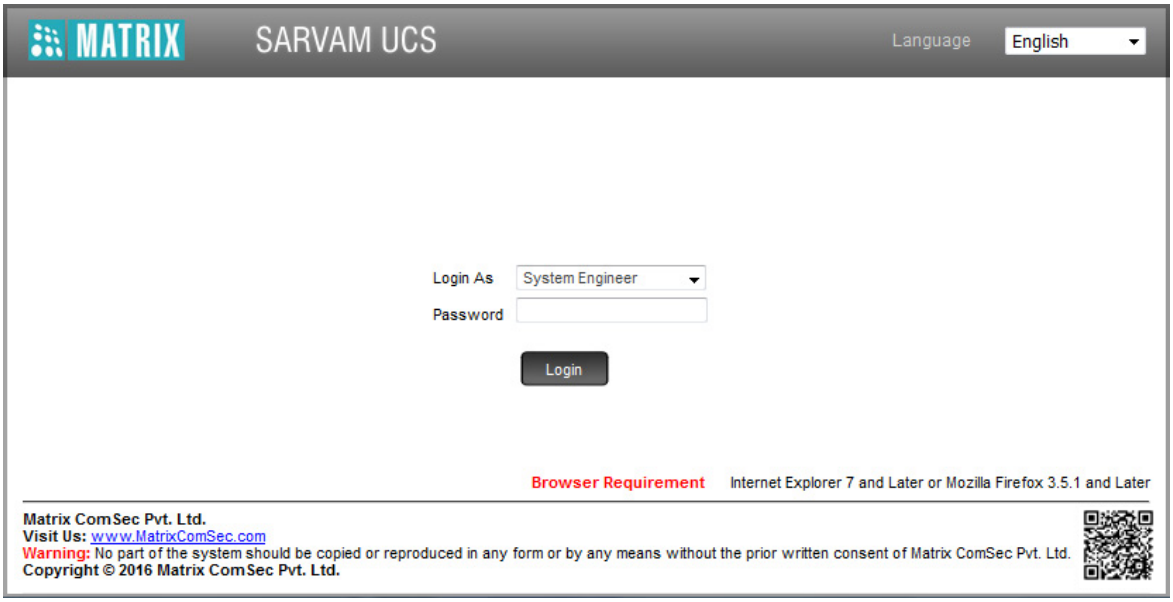
The changes you make in the system configuration under Configuration will not be reflected in the Quick Installation Wizard.

You are advised to either use the Wizard only during installation and to make modifications post installation OR use the Wizard only for basic configuration (first time installation) and for making subsequent changes click the desired link under Configuration.

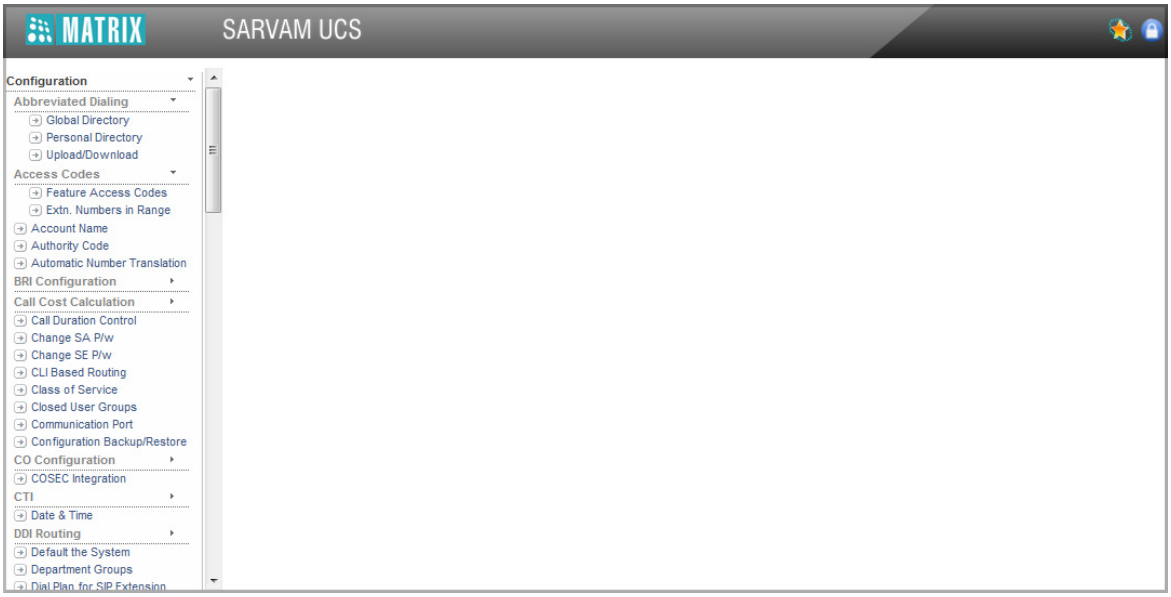
Entering the Wizard

- Open Jeeves (see “Configuring SARVAM UCS” for instructions). The **Login** page will open.

- In **Login as** select **System Engineer**.



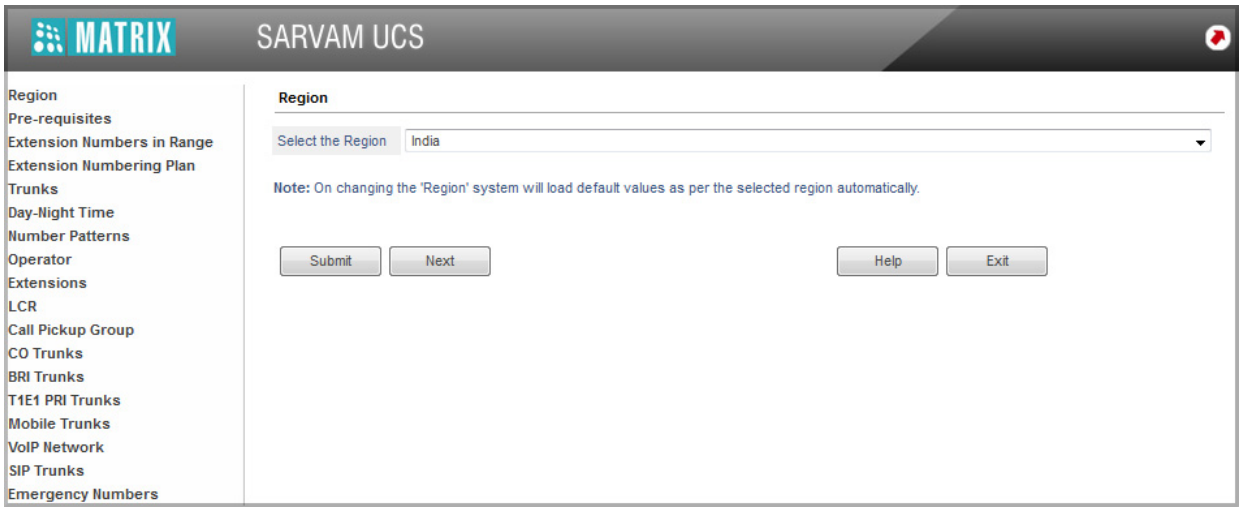
- In **Password** enter the SE password.
- Click **Login**.
- On successful login, the **Home** page opens.



- Click the **Wizard icon** on top right corner of the browser.

- To configure the Basic Settings, select the **Use Quick Installation Wizard-Standard PBX**.

- The **Quick Installation Wizard** opens.



Jeeves allows 4 persons to simultaneously log in as System Engineer. When you are logged into the Wizard, it is recommended that no other person logs into Jeeves as System Engineer.

Configuring Parameters

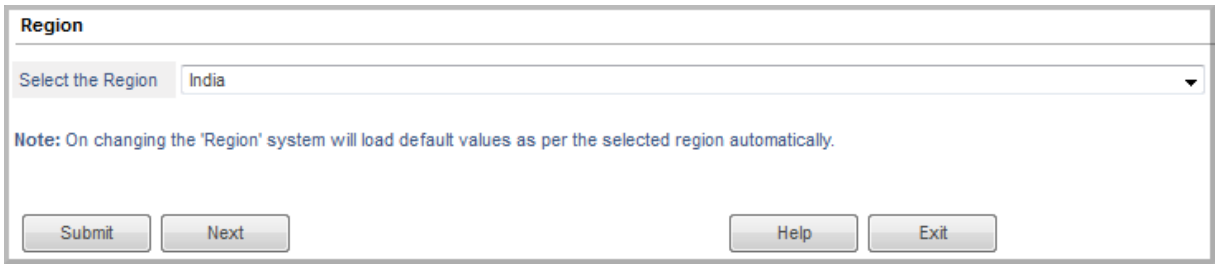
The Wizard gives you two options for configuring the system:

- **Selective Configuration** - This option allows you to choose what you want to configure and click the related links on the left navigation bar of the Wizard page. You are recommended to use this option for making the desired changes post-installation.

- **Navigate Wizard** - In this option, the Wizard will lead you logically, step-by-step through the configuration of the desired parameters. Click the **Next** button to navigate through the Wizard. You are recommended to use this option when installing the system for the first time.

Region

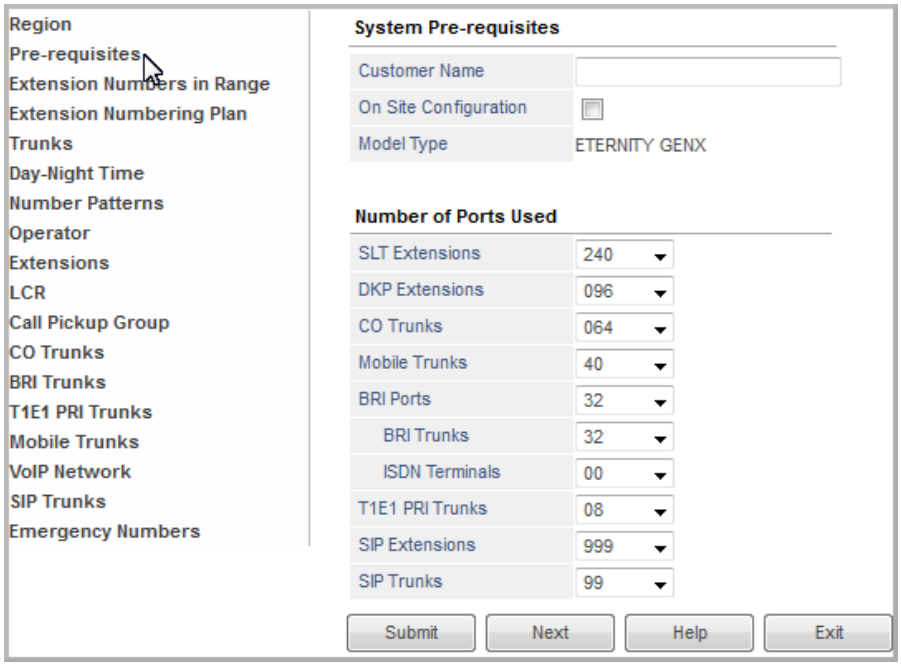
- In **Select the Region**, select the country where the system is installed, and then click **Next**.



Selecting the **Region** will instruct the system to load the default values of features/facilities according to the country specific requirements.

The default **Region** is India.

System Pre-requisites



- Enter the "Customer Name". You may enter the name (and address, if desired) of the organization/ enterprise. For example: Prudent Investment, 701 Sunshine Boulevard, Bannerghatta, Bangalore. The Customer Name can be a maximum of 80 characters.

- If you enable the **On Site configuration** check box, the configuration GUI of SARVAM UCS, Jeeves, will show the pages for only those trunks and extension port types that are on board in the system, that is, detected by the system at Power-On.

- The **Model Type** displays the name of the model that you are configuring — SARVAM UCS SME or SARVAM UCS ENT.

- Select the number of ports to be used for each Port Type (CO, DKP, SLT, Mobile, T1E1 PRI, BRI, SIP Trunks, SIP Extensions) from the respective boxes. For example, if you want 8 CO Trunks, 24 DKP extensions, and 128 SLT extensions to be used, select the same numbers from the box.

- Click **Submit** to save the changes.

  If you have selected the System Pre-requisites, navigate to the next page of the Wizard by clicking the **Next** button.

- *You can set the Wizard to display only those trunk and extension port types which are present in the system. This can be done by enabling the 'On Site Configuration' check box.*

- *When the **On Site Configuration** check box is enabled, the Wizard will display the only those ports that are present in the system (detected by the system at Power-ON) on this page.*

  For example, the system detects SLT20 Card at power-on, so the maximum number of SLT ports in the box will be 20. If you want only 18 SLTs to be used, select 18 in the box.

  The Wizard will now consider that there are only 18 SLTs in the system and will modify the relevant pages accordingly.

  The fields for port types which are not available on-board (detected at Power-On) are displayed as non-editable.

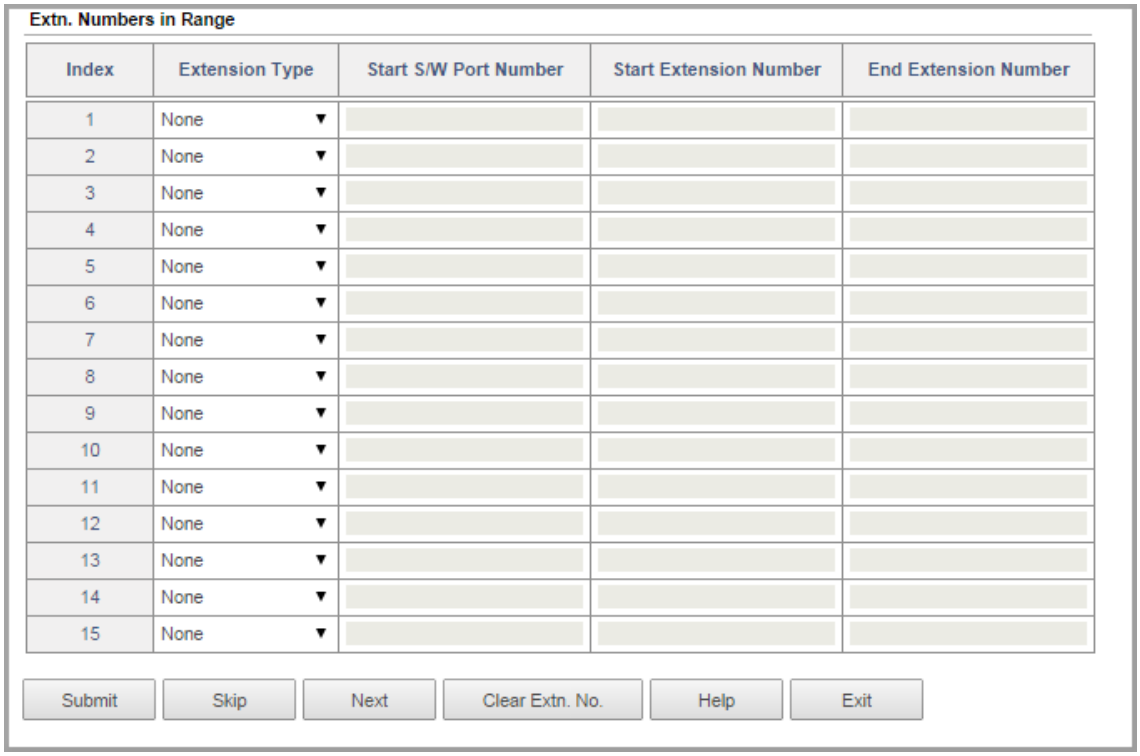
- *To enable the **On site Configuration** check box you must login as System Engineer.*

- *The Wizard does not provide for the port types **Magneto, Radio, Data, ILC** and **E&M** as these are less commonly used. If your system has Magneto Card and/or E&M Card installed, you must configure the related trunk parameters under Configuration.*

*It is recommended that you enable the **On Site Configuration** check box when you are configuring the system at the installation site.*

Extension Numbers in Range

You can assign Access Codes to a range of Extensions — SLT, DKP, ISDN, SIP. If you are assigning a range of Extension Numbers (Access Codes) to the desired ports, and a match is found for the same extension numbers, the system will clear these extension numbers from the existing database. The new extension numbers will be assigned according to the given range.



- Against each Index configure the following:
  - **Extension Type:** Select the Extension Type. You can select SLT,DKP, SIP, ISDN Terminal.
  - **Start S/W Port Number:** Enter the Software Port Number from which you want the system to start assigning the desired extension numbers.
  - Define the range of Station Access Codes/Extension Numbers (Flexible Number/Access Code) that you wish to assign to the Extension Type you selected in **Start Extension Number** and **End Extension Number**.

    For the given range of extension numbers, the system will assign extension numbers from the software port number specified in Start S/W Port Number for this particular entry. The range of extension numbers will be assigned in ascending order of the Software Port Number.

    For example:
    Extension Type you selected is SLT
    Start Software Port Number is 1
    Start Extension Number is 2001
    End Extension Number is 2100

    The system will assign Extension Number 2001 to Software Port Number 1, 2002 to Software Port Number 2 and so on. The system will assign the last Extension Number 2100 to Software Port Number 100.

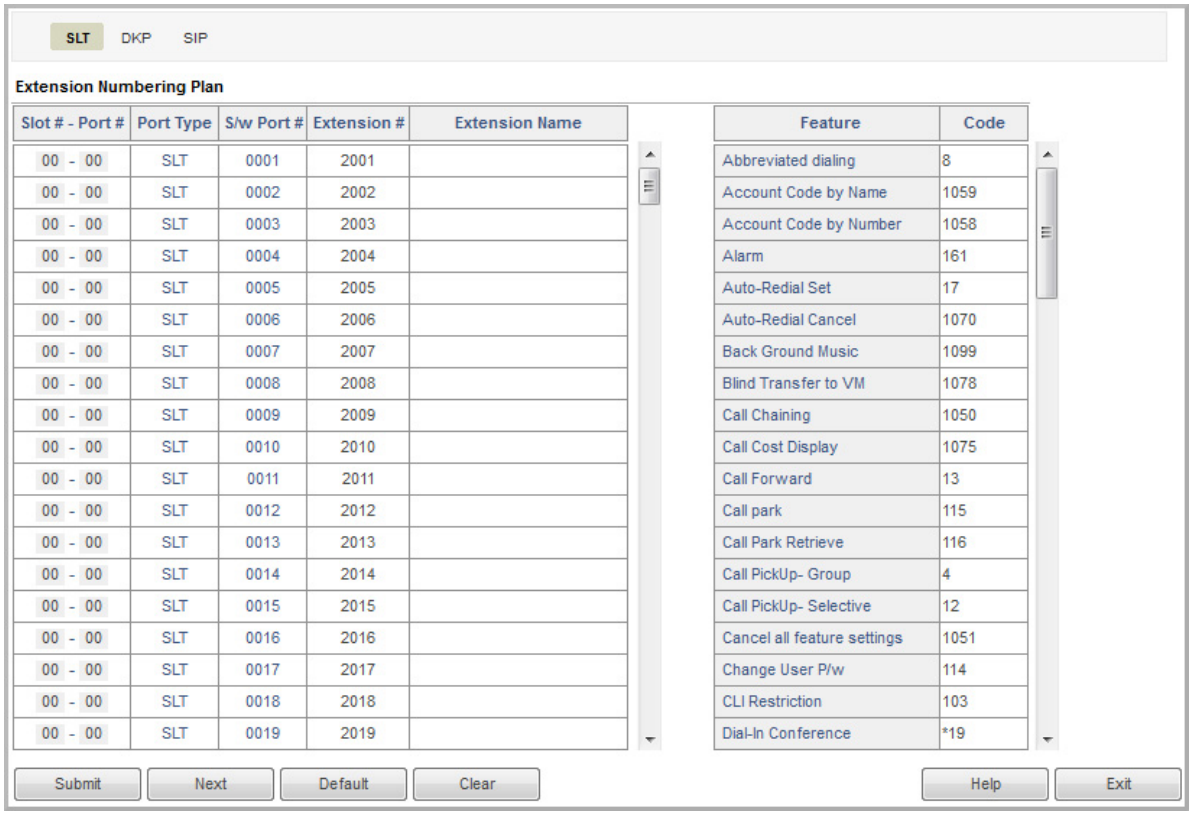
*If you want to assign access codes starting with **#** or * to a range of extensions, make sure both **Start** and **End extension numbers** begin with **#** or *.*

- Click **Submit**.

- If you do not want to assign access codes to range of extensions, click **Skip**.
- To assign the extension numbers to a range all over again, click the **Clear Extn. No.** button on this page.

Extension Numbering Plan

- Click the respective tab to assign extension numbers and names for the SLT, DKP and SIP Extension as desired on this page.



You can also configure Access Codes from this page.

The desired Extension Number can be 6 digits long. The digits 0 to 9, # and * are allowed.

The desired Extension Name may be the name of the person who will use the extension. The name can be a maximum of 18 characters.

When you change the extension numbers, make sure that they do not clash with any other Feature Access Code in the dialing phase. To know more, refer the topic “Access Codes”.

The Wizard pops up an alert about clashing extension numbers and prompts you to resolve the conflicting numbers.

- If you click the SIP tab, in **Authentication Password**, enter the password to be used by the system to authenticate the SIP messages received from the SIP Extension. To avoid unauthorized access, we recommend you to change the password regularly. Make sure it is strong and is kept confidential. The password must:
  - be of minimum 6 characters and can be a maximum of 12 characters.
  - include atleast one upper-case, one lower-case, one number and one special character.

- all ASCII characters (except Percentage **%**, Hash **#,** Equal to **=,** Plus **+**, And **&**, Backslash **\,** Less than **<,** Greater than **>,** Apostrophe **',** Double Quote **"** and **Space**) are allowed.
  Default: Blank.

The Wizard will display only those ports available in the system and the number of ports you have defined earlier in the System Pre-requisites page. The Wizard automatically detects the Hardware Slot and Port Offset of the ports and assigns them to Software Ports. The Wizard also assigns the default extension numbers to the ports, but leaves the extension names blank.

The default extension numbers for the above port types are:
2001 to 2240 for SLTs 001 to 240.
3001 to 3128 for DKPs 001 to 96.
Blank for SIP Extensions 001 to 999.
Blank for ISDN Terminals 01 to 64.

The Hardware Slot and Port Offset is not applicable for SIP Trunks.

- If you have finished assigning numbers and names, click **Submit**.
- To assign the extension numbers and names all over again, click the **Clear** button on this page.
- To default all extension numbers, click the **Default** button on this page.

Access Codes

- Change the Feature Access Codes, if required. Feature Access Codes may consist of single digits or a sequence of a maximum of 6 digits. Digits 0 to 9,# and * are allowed. The default Access Codes that appear on the screen are country-dependant. Also, refer "Access Codes" to know more.
- Click **Submit** and click **Next** to move to the next page.

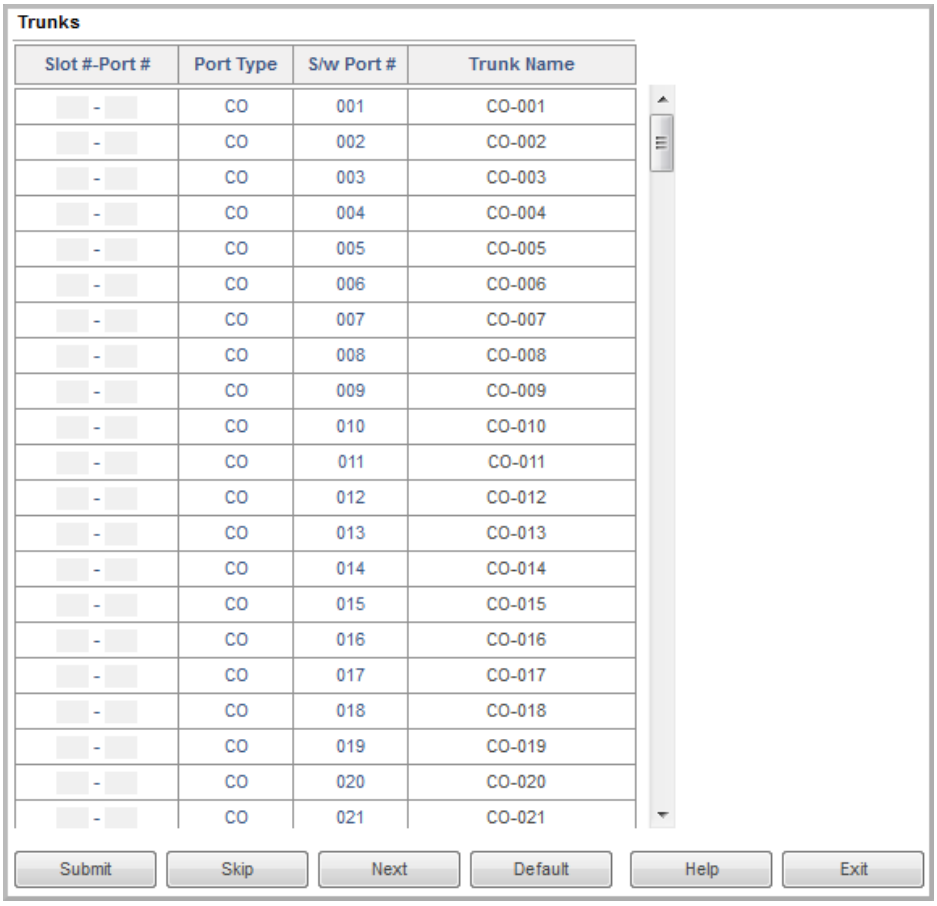
*If you assign the same Access Code to more than one feature, the Wizard will pop up a **Total Conflict** message and ask you to resolve the conflicting codes. It will not allow you to submit until you have resolved the conflict. If you assign an Access Code which has the same first digit as another Access Code, the Wizard will pop up an alert about the clashing numbers. You must resolve the clashing numbers by clicking **OK**.*

You may choose to resolve the clashing number by clicking 'Cancel' or you may ignore the alert and continue programming by clicking 'OK'.

Trunks

- Configure names for Trunk ports for easy identification of the trunks. The name may consist of a maximum of 18 alphanumeric characters.

  The Wizard automatically detects the Hardware Slot and Port Offset of the trunk ports and assigns them to Software Ports. The Wizard also assigns the default trunk port names along with their respective software port numbers. For example, if a Two-wire Trunk is assigned software port number 001; the name will be displayed as CO -001. Thus CO trunks are named as CO-001 to CO-128, Mobile Ports as MOB-001 to 064, BRI Ports as BRI-001 to BRI-032, and so on.



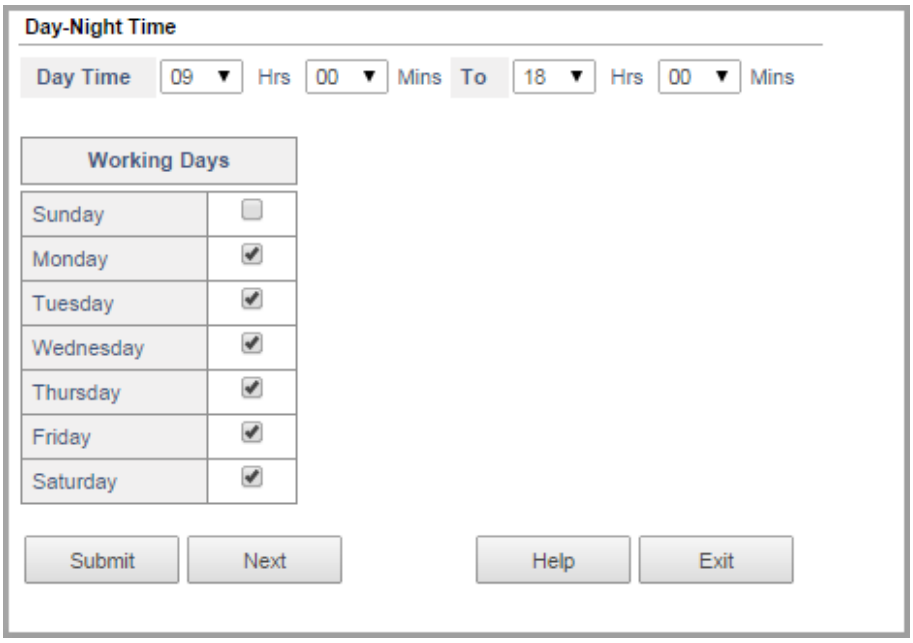
The Wizard will display only those Trunk port types available in the system and the number of Trunk ports you have defined for each trunk port type earlier on the System Pre-requisites page.

The Hardware Slot and Port Offset is not applicable for SIP Trunks.

- Click **Submit** and click **Next** to navigate the page further.

Day-Night Time

- Ask your customer about their working days and working hours (24 hours format) and program the parameters accordingly. The Wizard considers the working hours you have selected as Daytime and the remaining hours as Night Time (non-working hours). The default working hours are from 09:00 to 18:00. The default Working Days are Monday to Saturday.



- *Day-Night Time is assigned to Trunks and Extensions, so that they behave differently according to the Time of the day. For example, the customer may want the system to route trunk calls to the security personnel when the office is closed, or deny certain extensions access to outgoing long distance calls during non-working hours and days, or to play a different greeting message to callers on holidays.*

- *This parameter is based on Time Table 1*[116]*, which is assigned to Trunks and Extensions by default.*

- *The Wizard simplifies the assignment of Time Tables to trunks and extensions, requiring you to configure only the working hours and working days, instead of prompting you to define the non-working hours and break hours. The Wizard automatically applies the working hours and days you have programmed to time table-dependent facilities and features such as Class of Service, Toll Control, Outgoing Trunk Access, etc.*

- *Skip this page if you feel that the requirements of your customer are not served by this parameter*[117]*. Configure the Time Tables and related features like Class of Service, Toll Control, Outgoing Trunk Access, etc. under Configuration.*

- *If you want to change the working hours and days at a later stage while navigating the Wizard, you may use the **Back** button of your browser to return to this page and make the changes.*

- Click the **Next** button to navigate the Wizard further.

116. *A Time Table is a schedule of the three time zones (working hours, break hours, non-working hours) for a week. There are 8 different Time Table templates to select from. Different Time Tables can be assigned to different trunks and extensions. Refer the section "Time Tables" to know more.*
117. *For instance, the working hours are not the same throughout the week.*

Number Patterns

This parameter is related to the "Toll Control" feature, which allows you to define a particular calling permission for each extension, referred to as **Call Privilege**. A Call Privilege allows the extension to call certain areas and restricts it from calling others. The extension can also be restricted from the dialing of specific telephone numbers. The SARVAM UCS supports different types of Call Privileges, these are: No Calls, All Calls, Local Calls, Regional Calls, National Calls and International Calls.

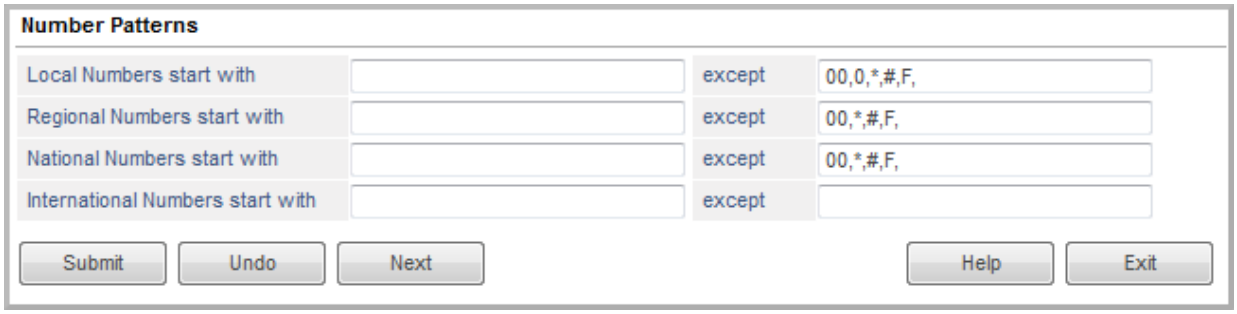
- On this page, you are required to define the number strings which the system should consider as **Local Numbers, Regional Numbers, National Numbers** and **International Numbers**.

  In **Numbers starting with**, you may enter only the first digit of the number string, or a part of the string, or the complete number string.

  In **except**, enter the number strings which you want to restrict from being dialed out.

  Each number string you enter must not exceed 16 characters. Separate number strings with comma.

  Do not provide <space> between commas and numbers.



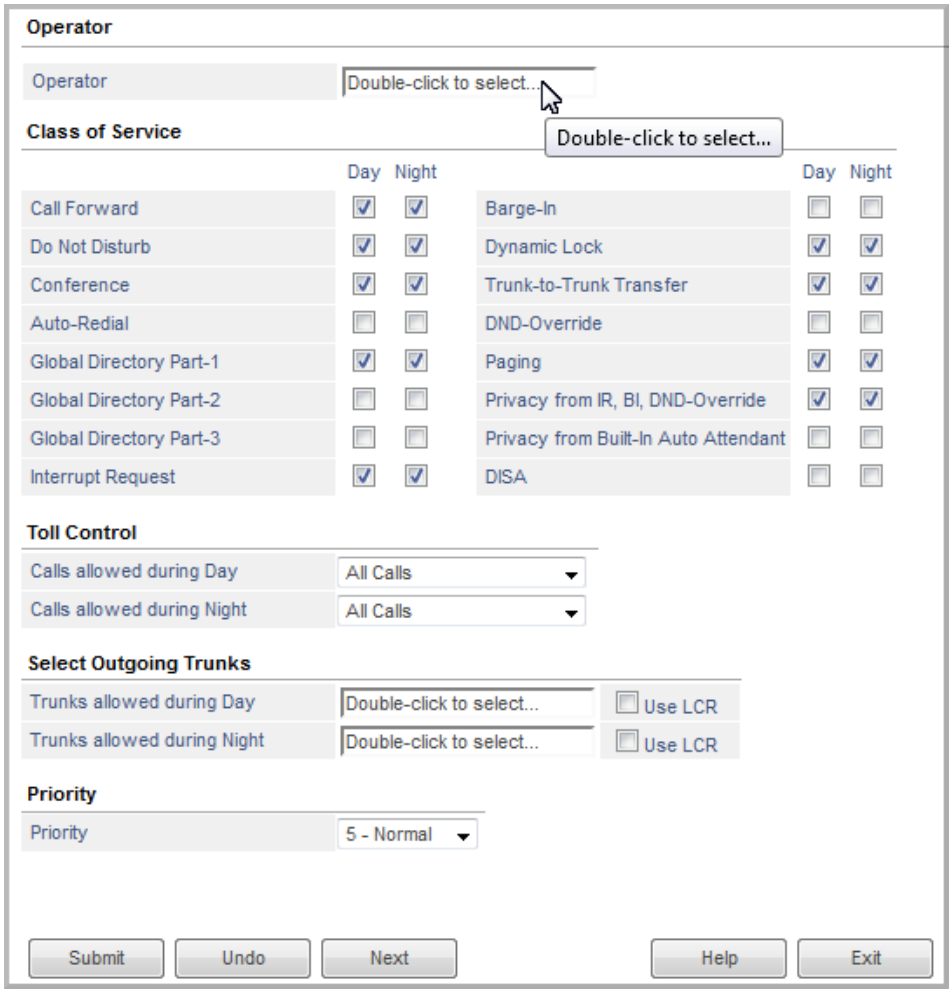
- Click the **Next** button to navigate the Wizard further.

- *The Call Privilege Type **No Calls** and **All Calls** does not require any number pattern programming.*
- *The Call Privilege Type **Limited Calls** allows the dialing of only specific telephone numbers. It can be programmed only under Configuration. To know more, refer the feature description for "Toll Control".*

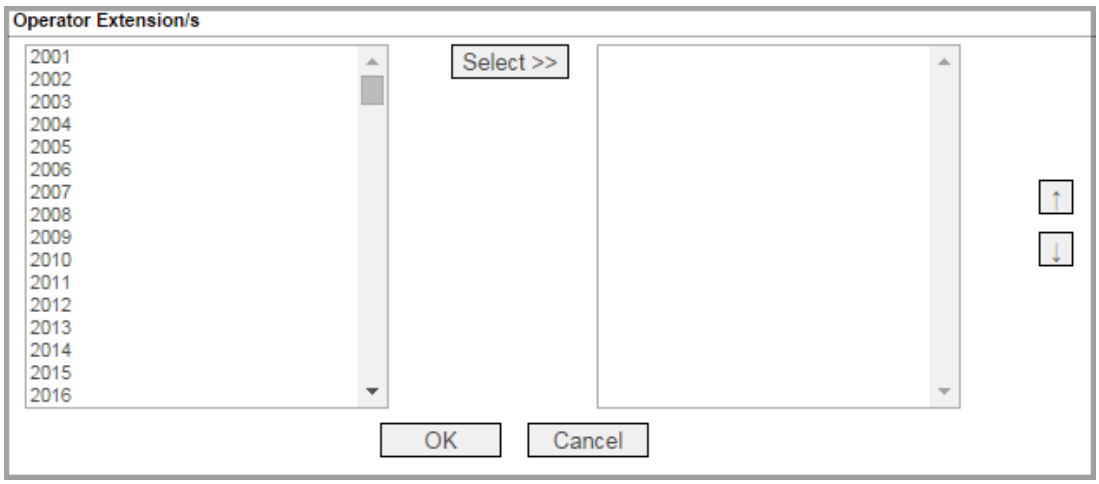
Operator

- Select the extensions which are to be used as Operator.

When you double click the **Operator** field, a multiple selection box opens.



The box on the left displays the extension names and numbers you configured or the default access codes. Place your cursor on the desired extension and click the **Select>>** button. The selected extensions will appear in the box on the right.



It is possible to change the sequence of extensions on the right side box using the Up/Down Arrow.

To remove/delete any extension number from the right box, select the extension number and press the delete key from the keyboard.

A maximum of 16 extensions can be selected as Operator extension.

You will be shown an alert if you configure more than 16 extensions. You may select and delete the excess extensions or choose to ignore the alert by clicking 'OK' or closing the alert dialog box. Regardless of this, the system will consider only 16 extensions. You can change the sequence of the extensions on the right hand side using the Up and Down arrows.

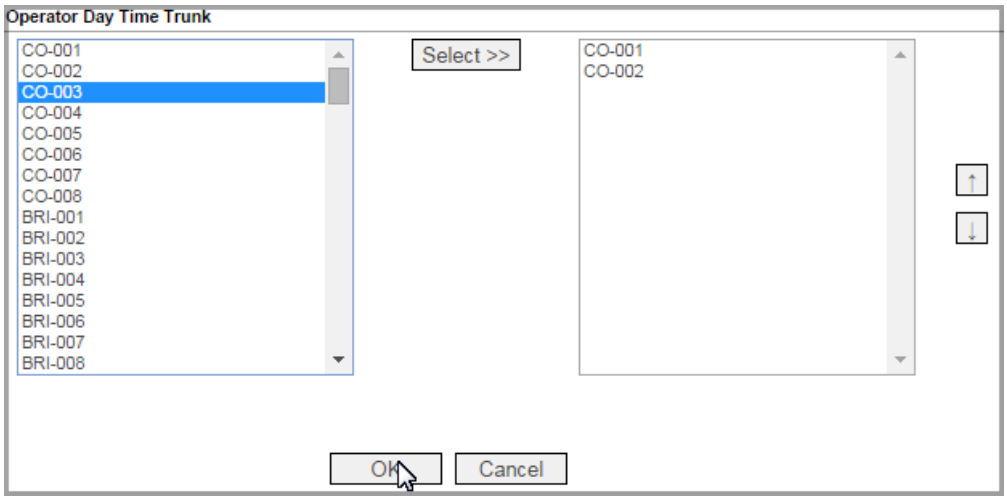
When you click **OK**, all the extensions you selected will appear sequentially, separated by comma in the order you selected the extensions.

- Define the Class of Service for the Operator extension. Select the features to be allowed to the Operator extension during the day and at night by selecting the check boxes of the features listed in the table.

- Configure the Toll Control for the Operator extension for day time and night time. Select the type of calls to be allowed during the day, and the type of calls to be allowed during the night. By default, all types of calls are allowed during the day and at night.

- Select the Outgoing Trunks for the Day Time (the trunks through which calls are to be routed during the day). If you want the system to use Least Cost Routing, select the check box.

  When you double click this field, the multiple selection box opens. Select the outgoing trunks from the left side box.



As you can see, the same trunk types are arranged sequentially, regardless of their hardware port location.

If you select trunks of the same type in sequential order, for example: CO-001, CO-002, CO-003, BRI-001, BRI-002 and MOB-002, the same trunk type will be grouped in one OG Trunk Bundle: CO-001, CO-002, CO-003 will be OGTB #1, BRI-001 and BRI-002 will be grouped as OGTB#2, and MOB-002 as OGTB#3.

These are default trunk names. If you have changed the trunk names (see Naming of Trunks), the name will appear here, instead of the default trunk names.

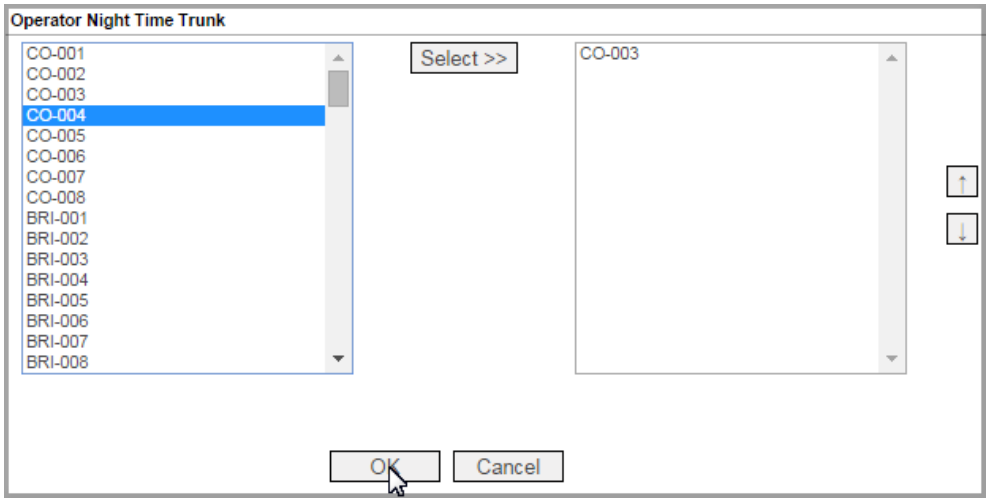
If you select the trunks of the same type in a non-sequential order, for example, CO-001, MOB-001, BRI-001 and CO-002, four OGTBs will be formed with CO-001 as member of OGTB#1, MOB-001 as member of OGTB#2, BRI-001 in OGTB#3 and CO-002 in OGTB#4. So, despite two same trunk types being selected (CO-001 and CO-002) they are grouped in separate OGTBs.

A maximum of 8 OGTB are allowed. If you exceed the number, the Wizard will show an alert indicating that the system is short of resources.

It is possible to change the sequence of the trunks on the right side box using the Up/Down Arrow.

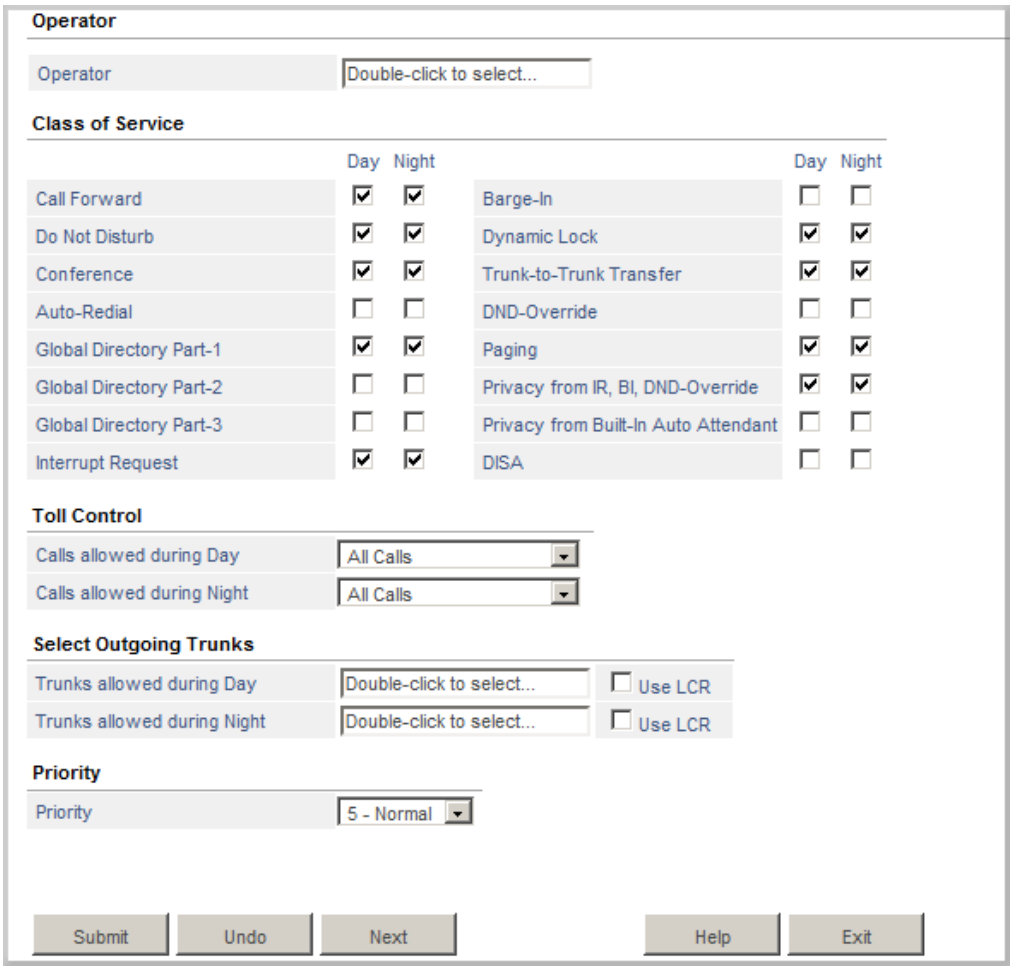
To remove/delete any trunk number from the right box, select the trunk number and press the delete key from the keyboard.

- Select the Outgoing Trunks for the Night Time (the trunks through which calls are to be routed during the night). If you want the system to use Least Cost Routing, select the check box.



Follow the same instructions for selecting the trunks from the multiple selection box as described in the previous step.

- Set the "Priority"[118] for the Operator extension, by selecting the desired Priority Level from 1-9 from the box. By default the priority level for the Operator extension is set to level 5.



- Click **Next** to navigate to the next page.

For the parameters of the Operator extension, the Wizard uses the following resources:
- *Class of Service Groups 18 to 19.*
- *Outgoing Trunk Bundle Groups 18 and 19*
- *Outgoing Trunk Bundles 69 to 76.*

When configuring the system from the links under Configuration, do not modify the settings of these parameters. It will affect the settings made by the Wizard.

Extensions

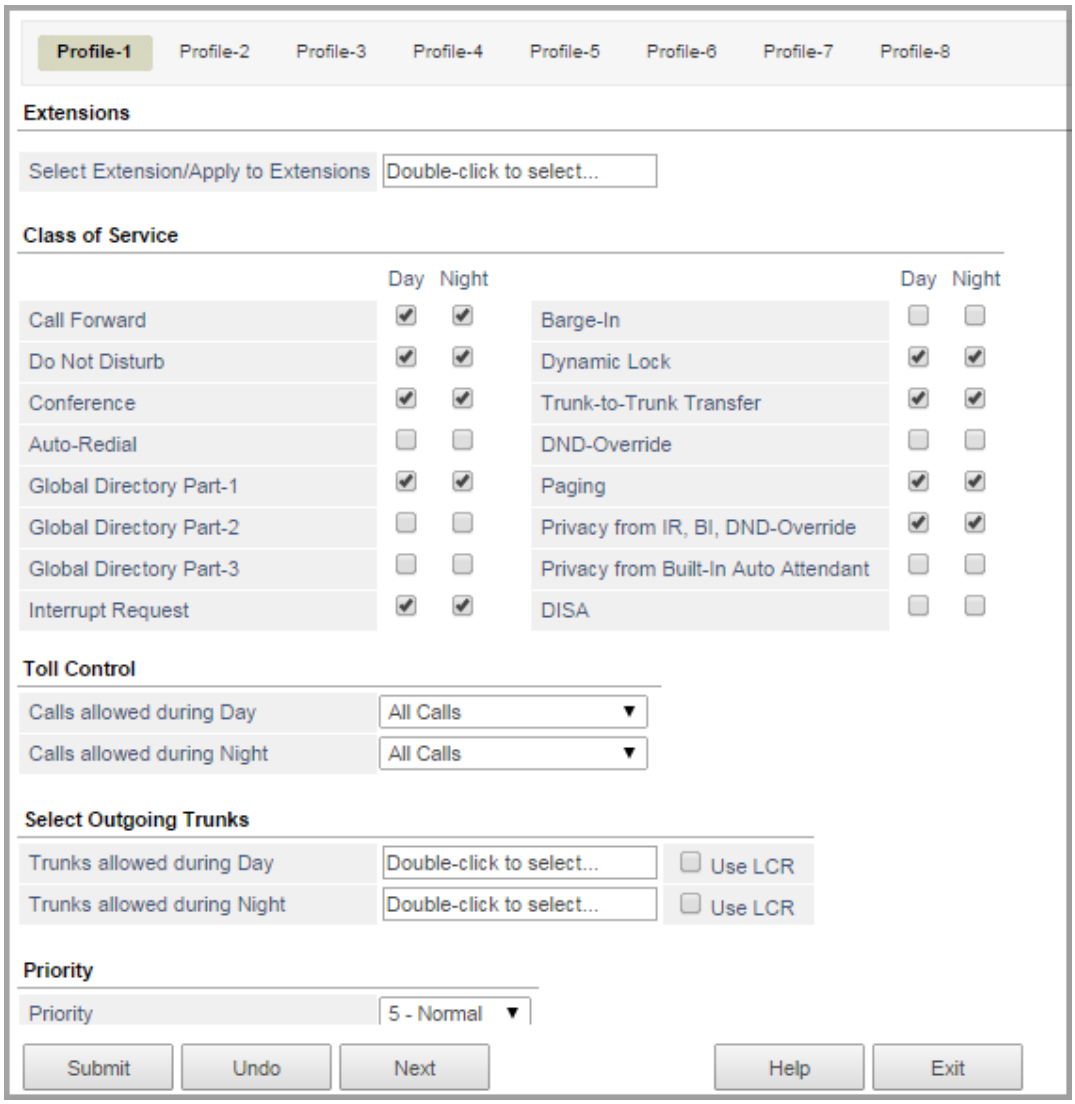
- Create a User Profile for extensions. A User Profile consists of Class of Service (COS), Toll Control and Trunk Access to be assigned to an extension during day time and night time.

118. *Each extension of the SARVAM UCS is assigned a Priority Level starting from 1, 2, 3, 4...to 9. With 1 being the lowest priority and 9 being the highest priority. The calls from an extension with higher priority have preference in call landing. When an extension with higher priority calls another with lower priority, a triple ring is placed on the called extension, and the call will land first on the extension when there are multiple incoming calls on the extension with lower priority.*

The Wizard makes configuration of the extensions easy with **User Profiles**. Instead of configuring each extension individually, you can group together extensions that are to be allowed the same COS group, Toll Control, Trunk Access (Outgoing Trunk Bundle and Outgoing Trunk Bundle Group) and Priority in a single User Profile.

User Profiles meet the requirements of organizations that desire to assign a different set of features to their personnel according to their position in the organization, like senior managers, field executives, administrative assistants etc. In such cases, each of these groups of users can be assigned a different User Profile.

As many as 8 different User Profiles can be programmed using the Wizard.



You can name each User Profile such that it reflects the extension user group to which it is assigned. For example, you may rename User Profile-1 created for managers as 'Manager', User Profile-2 created for field executives as 'Executive', User Profile-3 created for administrative assistants as 'Admin'.

- To change the label of the User Profile, click the 'Pencil' icon.
- A prompt will pop up, asking you if you want to rename the User Profile Name. Enter the desired name in the field. You can enter a maximum of 18 characters as name. Click 'OK'.

- The name you entered will appear in place of 'User Profile' number.
- To assign the User Profile to the desired extensions, double click the box.

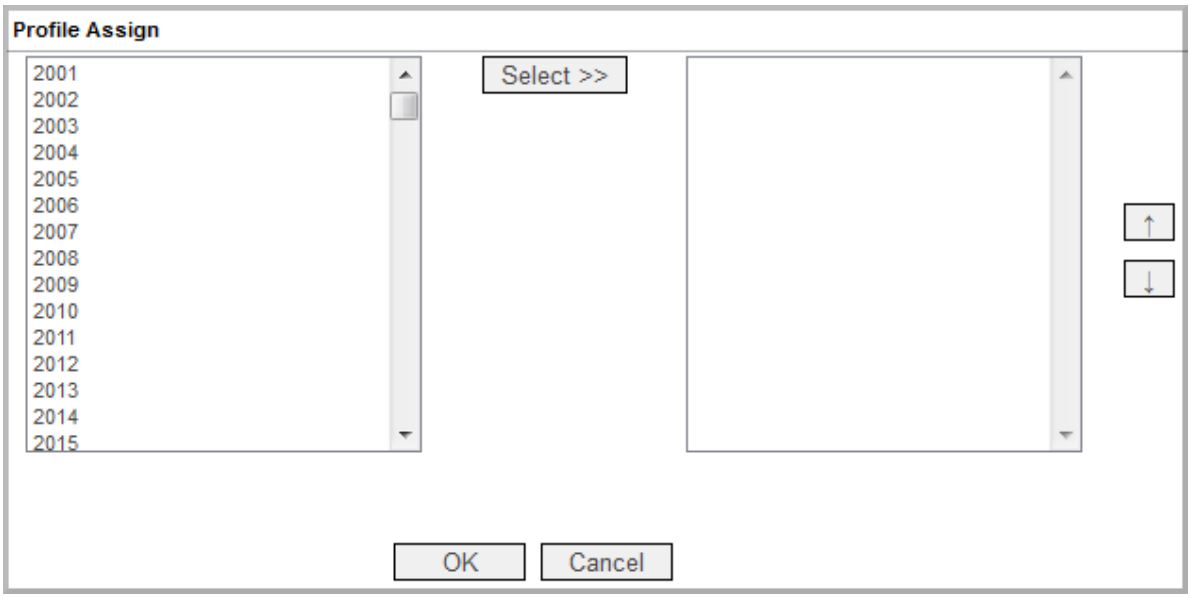
  The Wizard will display the configured extension numbers on the left box. Place your cursor on the desired extension numbers and click **Select >>**. The selected extension numbers will appear on the right box.

You can select the extensions for the User Profile after configuring the other parameters of the User Profile.

- Define the Class of Service for the User Profile for day time and night time by selecting the check boxes of the features you want to allow in the Class of Service.
- Define the Toll Control for the User Profile for the day time and night time by selecting the desired Toll Control level in the box.

  The Toll Control levels on this page are based on the number lists you have programmed earlier on the **Number Patterns** page of the Wizard.
- Select the Outgoing Trunks for the Day Time (the trunks through which calls are to be routed during the day). If you want to the system to use Least Cost Routing, select the check box.

  When you double click the field, the multiple selection box opens. Select the outgoing trunks from the left box.



As you can see, the same trunk types are arranged sequentially, regardless of their hardware port location.

If you select trunks of the same type in sequential order, for example, CO-001, CO-002, CO-003, BRI-001, BRI-002 and MOB-002, the same trunk type will be grouped in one OG Trunk Bundle: CO-001, CO-002, CO-003 will be OGTB #1, BRI-001 and BRI-002 will be grouped as OGTB#2, and MOB-002 as OGTB#3.

If you select the trunks of the same type in a non-sequential order, such as: CO-001, MOB-001, BRI-001 and CO-002, four OGTBs will be formed with CO-001 as member of OGTB#1, MOB-001 as member of OGTB#2, BRI-001 in OGTB#3 and CO-002 in OGTB#4. So, despite two same trunk types being selected (CO001 and CO002) they are grouped in separate OGTBs.

A maximum of 8 OGTB are allowed. If you exceed the number, the Wizard will show an alert indicating that the system is short of resources.

It is possible to change the sequence of trunks on the right side box using the Up/Down Arrow. You can also delete a particular trunk from the right box.

- Select the Outgoing Trunks for the Night Time (the trunks through which calls are to be routed during the night). If you want to the system to use Least Cost Routing, select the check box.

  Follow the same instructions for selecting the trunks from the box as described in the previous step.

- Set the "Priority"[119] for the extension, by selecting the desired Priority Level from 1-9 in the box. By default the priority level for the extension is set to level 5.

  You may set a different Priority Level in each User Profile, depending on the requirement of the extension users, whose extension the User Profile is to be assigned. For example, you may set a higher Priority level to the User Profile to be assigned to Managers.

- Click **Submit** to save the changes in the User Profile.

- To program another User Profile, click the label.

- Rename the label, if required, by clicking the 'Pencil' icon. Repeat all the steps described above to define Class of Service, Toll Control, Trunk Access and Priority.

- Click **Submit** to save changes. Repeat the same steps to program another User Profile.

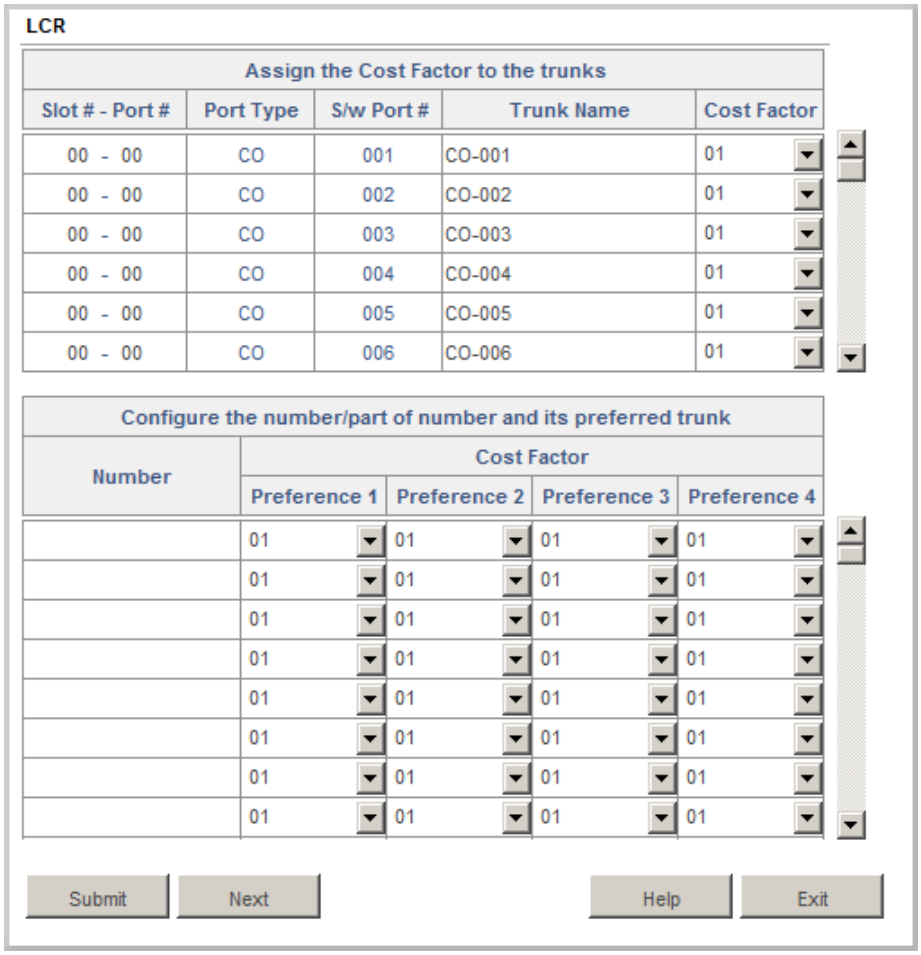
- When you have finished programming the desired number of User Profiles, click **Next** button to navigate the Wizard further.

- *The User Profiles 1-8 use the following resources in the system:*
  - *Station Basic Feature Template number 2 to 9*
  - *Class of Service Groups 2 to 17.*
  - *Outgoing Trunk Bundle Groups 2 to 17*
  - *Outgoing Trunk Bundles 5 to 68.*

- *Do not use them when configuring the system from Configuration.*

119. *Each extension of the SARVAM UCS is assigned a Priority Level starting from 1, 2, 3, 4...to 9. With 1 being the lowest priority and 9 being the highest priority. The calls from an extension with higher priority has preference in call landing. When an extension with higher priority calls another with lower priority, a triple ring is placed on the called extension, and the call will land first on the extension when there are multiple incoming calls on the extension with lower priority. Refer the feature description for "Priority" to know more.*

Least Cost Routing (LCR)



- Assign Cost Factor to the trunks. This parameter is of relevance only if 'Least Cost Routing' feature is applied on the CO Trunk port.
- On this page, the Wizard displays the number of trunk ports you selected in the **System Pre-Requisites** page.
- Cost Factor is a number assigned to each trunk for identification. This number also serves as a preference number for the trunk. The Cost Factor can be from 1 to 99. Trunks having the same preference must be assigned the same Cost Factor. Different trunk types can also be assigned the same Cost Factor. These trunks are used for routing calls.
- Now configure the number of or part of the number and the preferred trunk to route that number. The number can be a maximum of 16 digits.
- Select the cost factor trunk applicable for each number in the order of preference. Select the foremost preferred Cost Factor trunk in Preference 1, the second most preferred Cost Factor trunk in Preference 2, the third preferred Cost Factor trunk in Preference 3 and the least preferred Cost Factor trunk in Preference 4.
- Click **Next** to navigate the Wizard further.

SARVAM UCS supports four different types of LCR, but the Wizard supports only Number based LCR. To know more, refer "Configuring LCR".

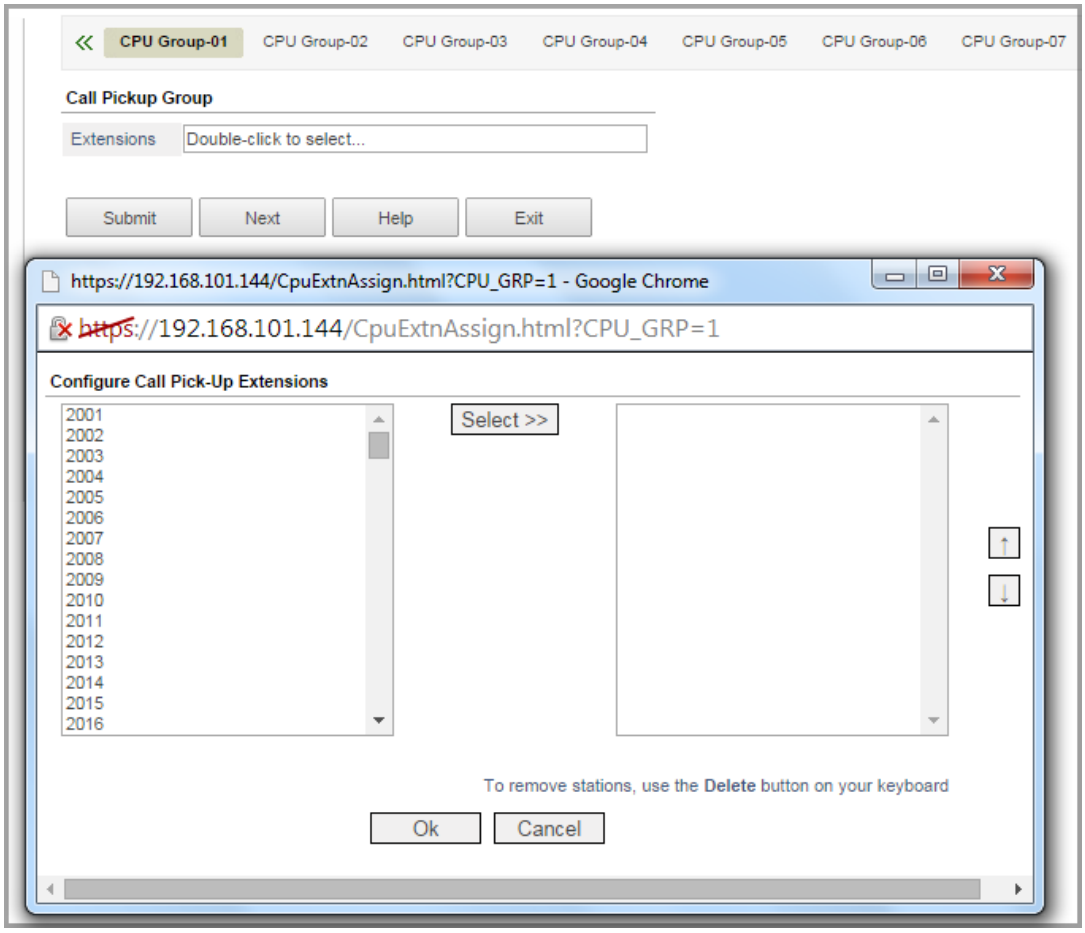
Call Pickup Group

- Assign extensions to Call Pick-Up (CPU) groups.

  Call Pick-Up allows extension users to answer (internal and trunk) calls ringing on other extensions from their own extension; without physically going to the ringing extensions. For this extensions must be assigned to CPU Groups.

- While you can create as many as 01 to 99 CPU Groups, and assign extensions to these groups, the Wizard allows you to create and assign extensions to 01 to 16 CPU Groups. By default, all extensions are assigned to CPU Group number 99.

- To create a CPU group, click the CPU group number, for example CPU 01, and double click to select the extensions to be assigned to this group from the box.

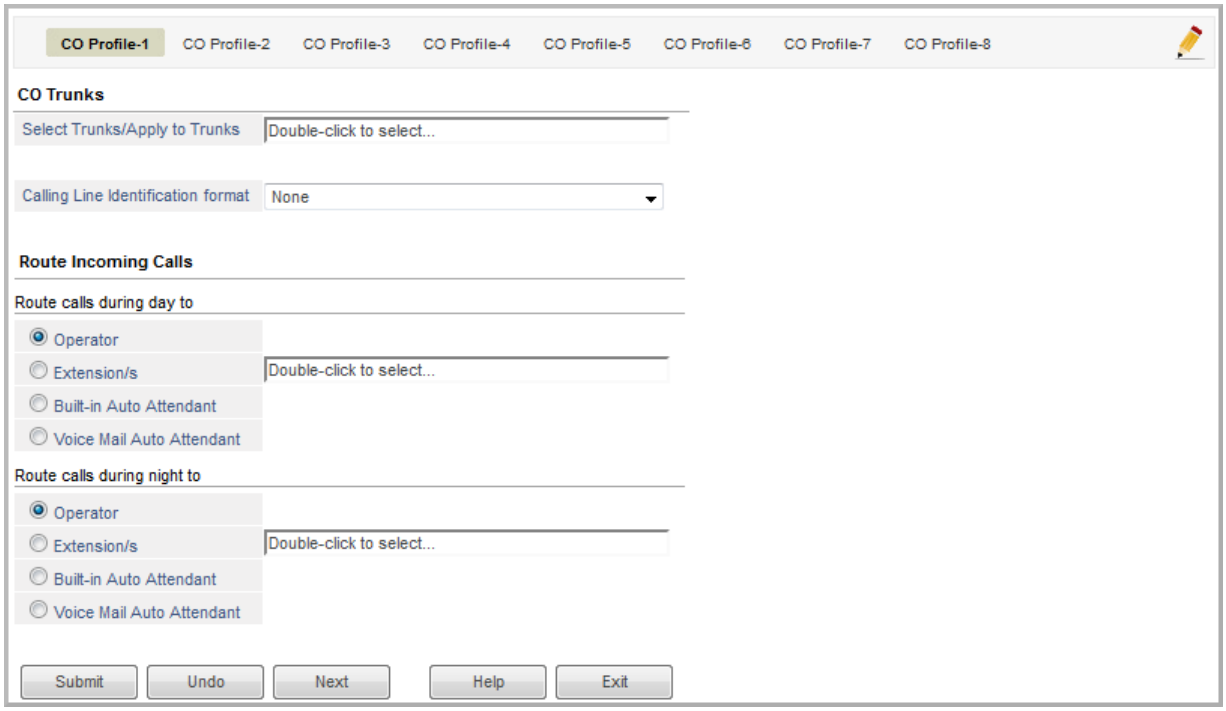


- Click **Submit** to save the settings of the CPU group you created. Now, repeat the same steps to create another group.

- If you have finished programming CPU groups, click **Next** to navigate to the next page of the Wizard.

To know more about this feature, refer the topic "Call Pick Up".

CO Trunks



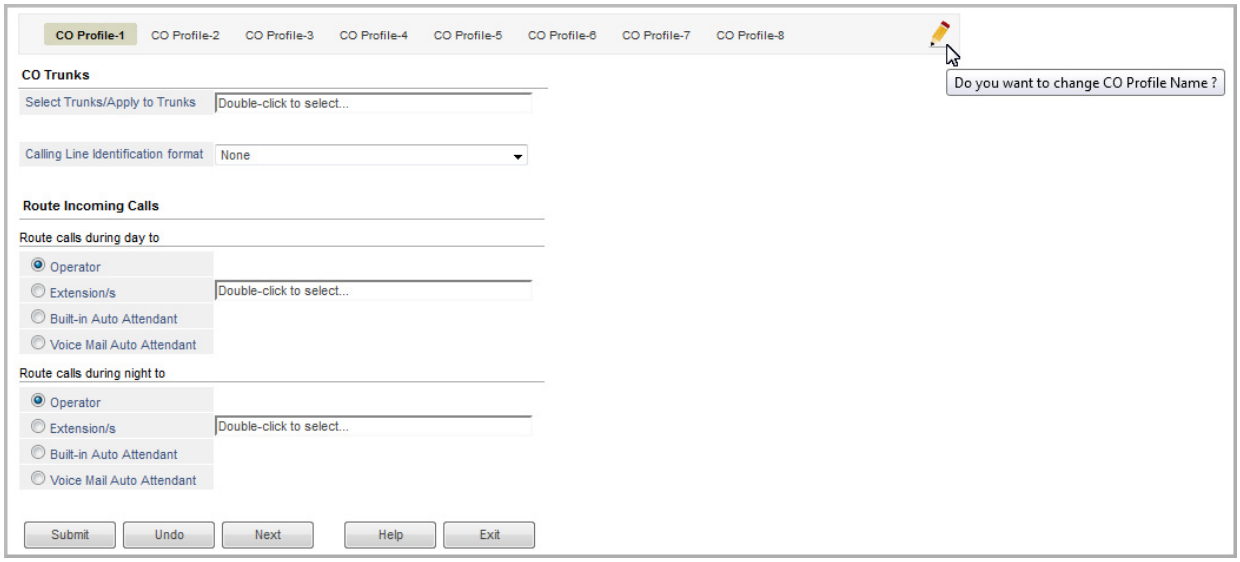
- Configure features of the Two-wire Trunks (CO) connected to the SARVAM UCS on this page.

  The Wizard makes configuration of the CO Trunks easy with **CO Profiles**. Instead of configuring each CO Trunk individually, you can group together trunks that are to be assigned the same features - Calling Line Identification, Trunk Landing Group - in a single CO Profile.

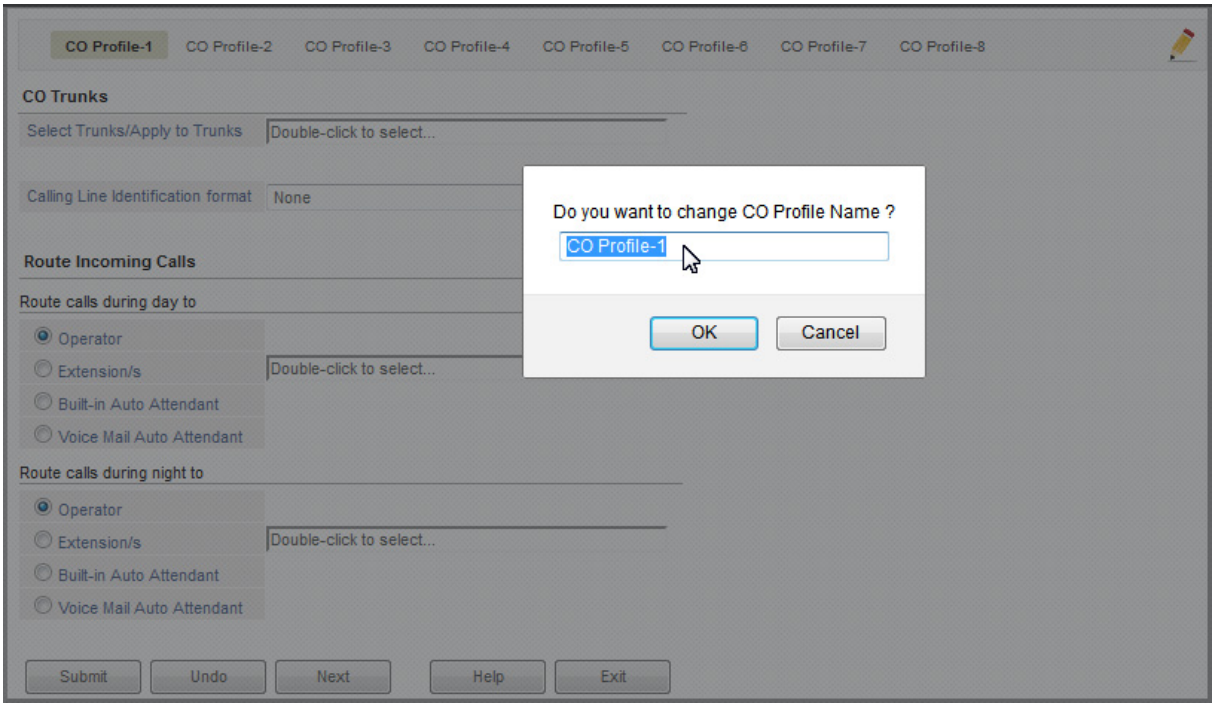
  You can configure as many as 8 different CO Profiles using the Wizard.

  It is also possible to name each CO Profile by the service provider. For example, you may rename CO Profile-1 created as 'BSNL', CO Profile-2 as 'Reliance' and so on.

- To change the label of the CO Profile, click the **Pencil** icon.

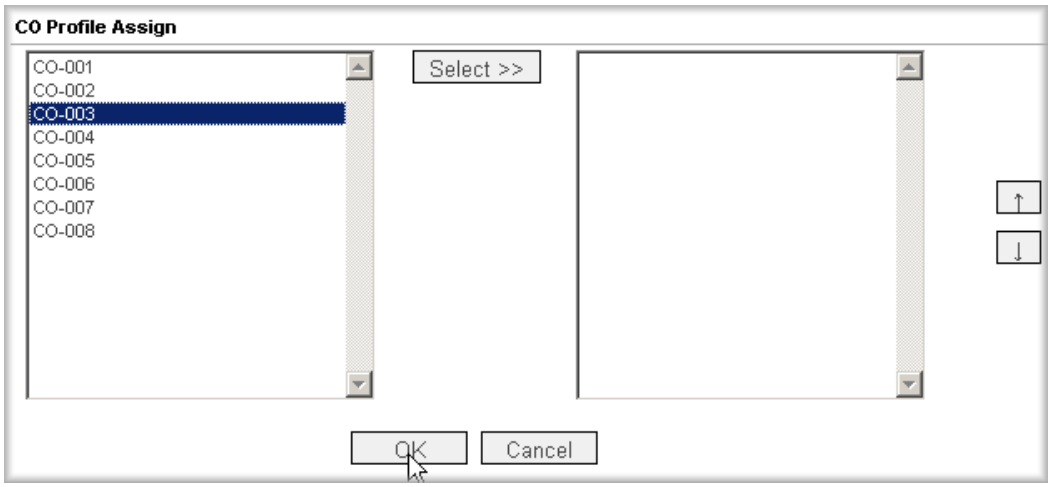


- A prompt will pop up, asking you if you want to rename the CO Profile Name. Enter the desired name in the field. You can enter a maximum of 18 characters as name. Click **OK**.



- The name you entered will appear in place of **CO Profile** number.

- To assign the CO Profile to the desired trunks, double click the field Select Trunks/Apply to Trunks.



The left side box will display the number of CO trunks (along their names) you configured in the **System Pre-requisites** page. Place your cursor on the desired trunks and click **Select >>**. The selected CO trunks will appear on the right side box.

You can select the trunks for the CO Profile after configuring the other parameters of the CO Profile.

- Define the Calling Line Identification Format for the CO Profile by selecting the desired option in the box. You are advised to consult the service provider in this regard.

- In the field **Route Calls during day to**, select the landing destination for calls on trunks in the CO Profile during day time. You may select any option: Operator extension, other extensions, Built-In Auto Attendant or the Voice Mail Auto Attendant (if available) from the box.

- If you select **Built-In Auto Attendant**, the default Voice Modules 02 to 13 containing the default Voice Messages (Morning, Afternoon and Evening Greetings, Built-In Auto Attendant Greeting and Guidance Messages) will be applied. Refer "Voice Message Applications" to know more.

- If you select **Voice Mail Auto Attendant**, the voice messages recorded and stored in Voice Mail module will be played.

- If you select **Extension/s** as the landing destination, select the extensions in the corresponding field. Double click the field, the multiple selection box will open. Select the extensions from the left box.

  A maximum of 16 extensions can be selected. If you exceed the number, the Wizard will prompt you to make the selection again. You can delete the excess extensions from those you selected. You can also change the order of the selected landing extensions in the right side box using the Up/Down Arrow.

  The selected extension numbers will appear in the field, separated by comma.

- In the field **Route Calls during Night to**, select the landing destination for calls on trunks in the CO Profile during night time. Follow the same instruction as in the previous step.

- Click **Submit** to save your settings in the CO Profile. Repeat these steps to create another CO Profile.

- If you have finished creating CO profiles and assigning them to trunks, click **Next** to navigate further.

- If there is any trunk you have not assigned a CO Profile, the Wizard will pop-up an alert informing you about the trunk you have not programmed.

  You must complete configuring the parameters for this CO trunk and only then will the wizard allow you to configure other parameters.

The Wizard uses the following resources for this page:
- *Trunk Feature Templates 34 to 41.*
- *Routing Groups 34 to 49.*
- *CO Hardware Templates 02 to 09.*

Do not make any modifications to them when configuring the system from Configuration.

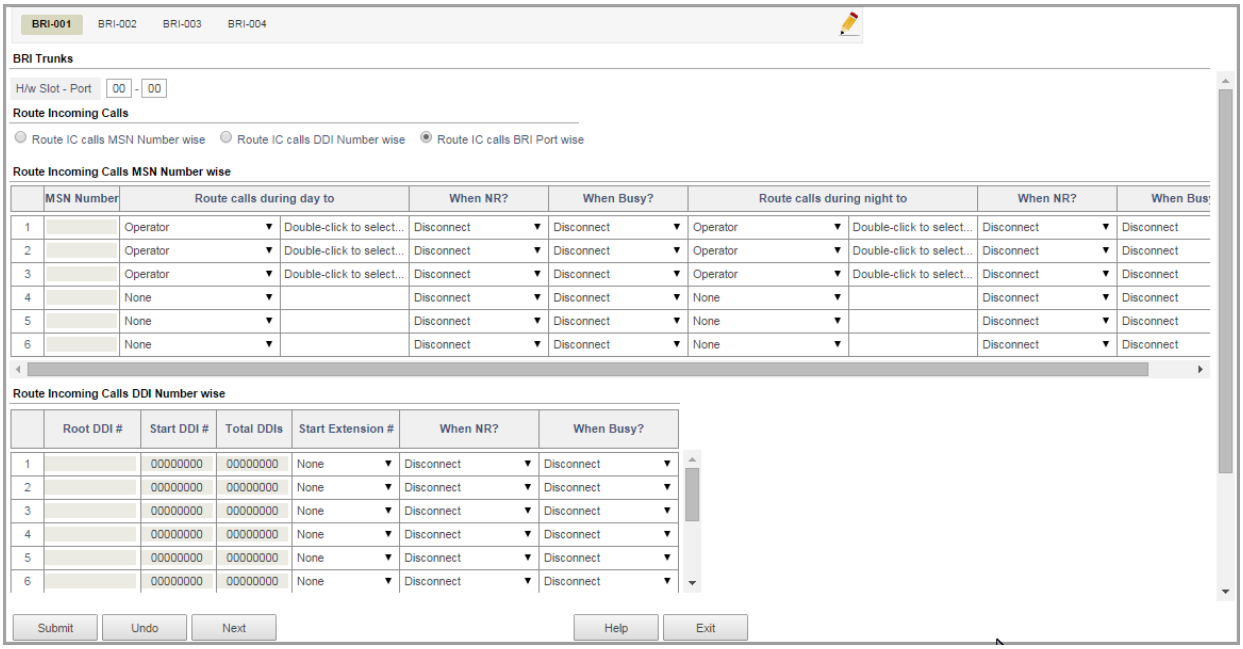
BRI Trunks

- *You will reach this page only if your system has detected the presence of BRI Trunks or if you have specified the number of BRI Trunks to be used in the **System Pre-requisites** page.*

- *The SARVAM UCS supports a maximum of 32 BRI Trunks.*

- *Using the Wizard, you can configure only the first four BRI Trunks in your system. You can configure the remaining BRI Trunks from Configuration.*

- The Wizard will display the name of first BRI Trunk.



The name will be displayed only if you have programmed names in the **Trunks** page of the Wizard. If you have not named the trunk on that page, you may change the label; select the desired BRI Trunk tab and click the **Pencil** icon.

A prompt will pop up, asking you if you want to rename the BRI Trunk. Enter the desired name in the field. The name you entered will appear on the screen.

If required, you may change the Hardware slot and port number of the BRI trunk displayed here.

- Select the mode for routing incoming calls. Incoming calls may be routed according to MSN Numbers, DDI Numbers or Port-wise.

  - **Route Incoming Calls MSN Number wise:** Select this option if you want to route incoming calls according to MSN numbers[120]. You can program maximum 6 MSN numbers.

  - **Route Incoming Calls DDI Number wise:** Select this option if you want to route incoming calls according to DDI Numbers[121]. Select this option only if your extensions are arranged sequentially. If the extensions are not arranged sequentially (for example, DDI numbers 2630551 to 2630559 are to be routed to extensions 3001 to 3009) you are recommended to program this parameter from the Full Programming Access mode. You can program maximum 12 DDI numbers.

  - **Route Incoming Calls Port wise:** Select this option if you want to route all incoming calls on the BRI trunk port to extensions, irrespective of dialed MSN/DDI number.

- If you select MSN Number-wise routing, configure the following parameters:

  - **MSN Number:** Enter the MSN Number provided by your Service Provider. This number is used to route the incoming calls and sent to the ISDN Exchange when making an outgoing call. You can enter up to 6 MSN numbers using the Wizard.

  - **Route calls during day to:** Select where you want to the route the calls during day time. You can route the calls to the Operator or to a group of extensions or to Built-In Auto Attendant or to the Voice Mail Auto Attendant[122].

    If you click the option **Built-In Auto Attendant**, the default Voice Modules 02 to 13 containing the default Voice Messages (Morning, Afternoon and Evening Greetings, Built-In Auto Attendant Greeting and Guidance Messages) will be applied by the Wizard. Refer “Voice Message Applications” to know more.

    If you select **Voice Mail Auto Attendant**, the voice messages recorded and stored in Voice Mail module will be played.

    If you select **Extensions**, double click the empty field, the multiple selection box will open. Select the extensions from the left side box.

    A maximum of 16 extensions can be selected. If you exceed the number, the Wizard will prompt you to make the selection again. You can delete the excess extensions from those you selected. You can also change the order of the selected landing extensions in the right side box using drag and drop option.

    The selected extension numbers will appear in the field, separated by comma.

  - **When NR?:** Select an option for what action the system should take when there is no response (NR), that is, the incoming call is not answered within the DDI Ring Timer (default: 45 sec.) by the destination. By default, it is set to 'Disconnect'.

  - **When Busy?:** Select an option for what action the system should take when the landing destination is busy. By default, it is set to 'Disconnect'.

---

120. MSN numbers is a set of numbers with no logical connection between the numbers themselves. For example, MSN numbers for a BRI connection could be 2630555, 2634872, 2635098, etc. Up to 8 MSN numbers are provided on a single BRI connection.

121. DDI numbers are a set of numbers arranged sequentially, for example DDI numbers for a BRI connection could be 2630551 to 2630559.

122. The in-built Auto Attendant offers 9 simultaneous calls and Voice Mail's Auto Attendant offers 64 simultaneous calls.

- **Route calls during night to:** Select where you want to the route the calls during night time. You can route the calls to the Operator or to a group of extensions or to Built-In Auto Attendant or to the Voice Mail Auto Attendant.

  If you select **Extensions**, you must now select the extensions by clicking the field provided for it. A maximum of 16 extensions can be selected. Follow the same procedure described above for selection of extensions for calls during day time.

- **When NR?:** Select an option for what action the system should take when there is no response (NR), that is, the incoming call is not answered within the DDI Ring Timer (default: 45 sec.) by the destination. By default, it is set to 'Disconnect'.

- **When Busy?:** Select an option for what action the system should take when the landing destination is busy. By default, it is set to 'Disconnect'.

- *The Wizard allows you to select the desired landing extension (Operator/ extensions/ Built-In Auto Attendant/Voice Mail's Auto Attendant) only for the first 3 MSN numbers. The remaining 3 MSN numbers can be routed to a single extension only.*

- *So, if you want to route incoming calls to a particular extension during the day time and night time, enter the MSN numbers in MSN number 4 to MSN number 6.*

- Configure the same parameters for MSN number 2 to 6, as applicable.

- If you select DDI Number-wise routing, configure the following parameters for each DDI#:

  - **Root DDI #:** Enter the Root DDI number. This number is used to send the DDI number to the ISDN Exchange. This number is also sent to the Exchange when an outgoing call is made by an extension which is not assigned DDI number. This is the same as the Pilot Number.

  - **Start DDI #:** Enter the first DDI number provided by your Service Provider. More often than not, the Start DDI# is the same as the Root DDI#.

  - **Total DDIs:** Enter the total DDI numbers provided by your Service Provider.

  - **Start Extension#:** Enter the number of the first extension from which the DDI number assignment is to be done.

    For example, your Service Provider has given you DDI numbers 2630550 to 2630560. These are to be assigned to extension numbers 2001 to 2010. In this case 2630550 will be the Root DDI# as well as Start DDI#. As 10 DDI numbers are provided to you, the Total DDI# would be 10, and the Start Extension# would be 2001.

  - **When NR?:** Select an option for what action the system should take when there is no response (NR), that is, the incoming call is not answered within the DDI Ring Timer (default: 45 sec.) by the destination. By default, it is set to 'Disconnect'.

  - **When Busy?:** Select an option for what action the system should take when the landing destination is busy, during DDI Ring Timer (default: 45 sec.). By default, it is set to 'Disconnect'.

  - Now, configure the same parameters for DDI#2 to #12.

- If you select BRI port-wise routing, configure the following parameters:

- **Route Calls during Day to:** Select the landing destination for calls on BRI trunk during day time. You may select any option: Operator extension, other extensions, Built-In Auto Attendant or the Auto Attendant of the Voice Mail (if available).

  If you select **Built-In Auto Attendant**, the default Voice Modules 02 to 13 containing the default Voice Messages (Morning, Afternoon and Evening Greetings, Built-In Auto Attendant Greeting and Guidance Messages) will be applied by the Wizard. Refer "Voice Message Applications" to know more.

  If you select **Voice Mail Auto Attendant**, the voice messages recorded and stored in Voice Mail module will be played.

  If you select **Extensions** as landing destination, you must select the extensions. Double click the field and select the extensions from the left box.

  A maximum of 16 extensions can be selected. If you exceed the number, the Wizard will prompt you to make the selection again. You can delete the excess extensions from those you selected. You can also change the order of the selected landing extensions in the right side box using the Up/Down Arrow.

  The selected extension numbers will appear in the field, separated by comma.

- **Route Calls during Night to:** Select the landing destination for calls on BRI trunk during night time. Follow the same procedure for selecting extensions as described above.

- Click **Submit** to save changes. Repeat the same steps to program the other BRI Trunks.

- If you have finished configuring the BRI Trunks, click **Next** to navigate further.

- If there is a BRI trunk you have not configured, the Wizard will pop-up an alert informing you about the trunk you have not programmed.

  You must complete configuring the parameters for this trunk and only then will the wizard allow you to configure other parameters.

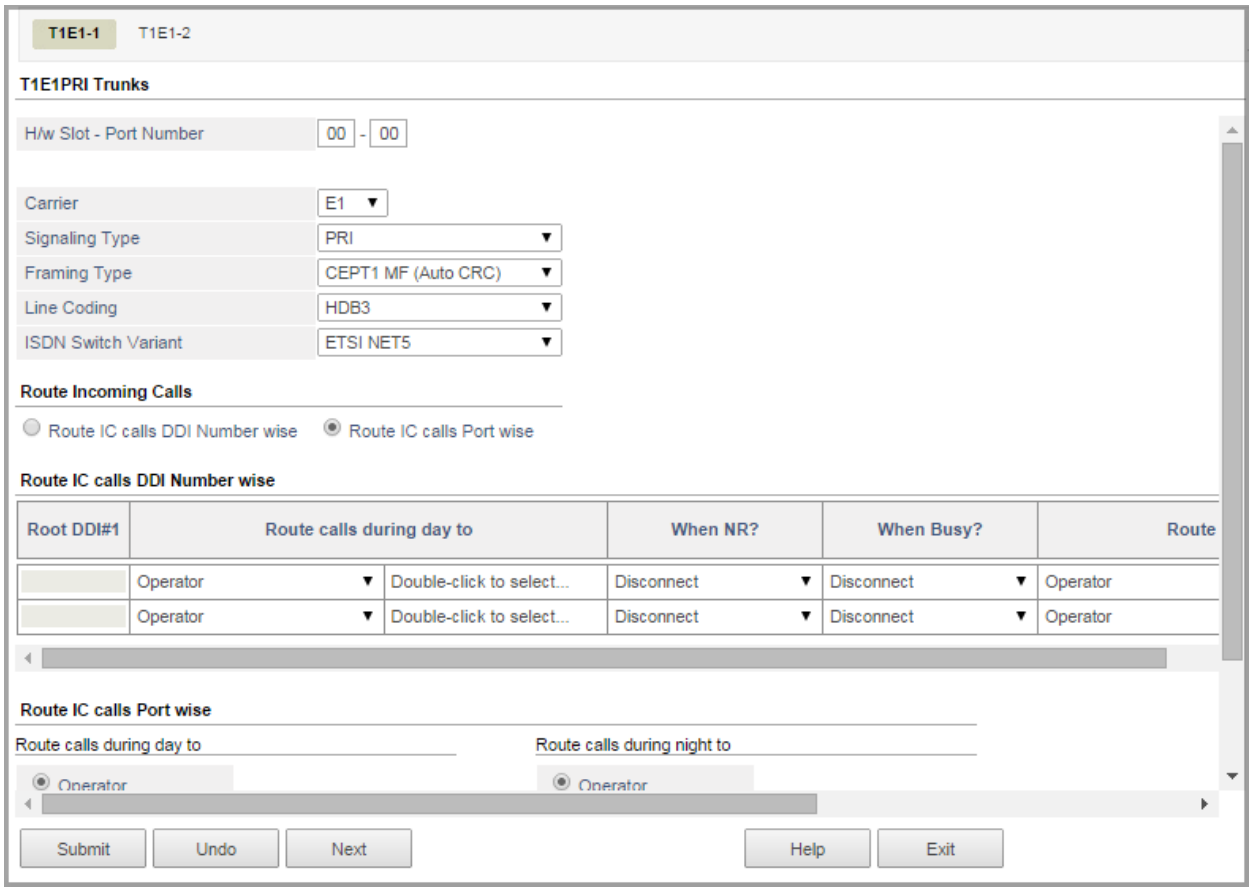
For the BRI parameters, the Wizard uses the following resources:
- *Entries 01 to 48 of the Incoming Call (IC) Reference Table.*
- *Entries 01 to 48 of the Outgoing Reference Table.*
- *Routing Groups 01 to 25*
- *Trunk Feature Template 01 to 25.*

When programming the system from Configuration, do not modify the settings of these resources. It will affect the settings made by the Wizard.

T1E1 PRI Trunks

- *The SARVAM UCS supports a maximum of 8 T1E1PRI Trunks.*

- *However, using the Wizard you can configure only the first 2 PRI Trunks in the system. You may configure the remaining Trunks, if applicable, from Configuration.*



- Change the label of the T1E1PRI Trunk, if required. Click the **Pencil** icon. A prompt will pop-up. Enter the desired name in the field of the prompt and click **OK**. The name you programmed will appear.

  You may enter the name of the Service Provider to make the identification of the Trunk easy.

- The Wizard will display the Hardware Slot-Port Number of the first T1E1 PRI Trunk the system has detected in this field.

- Select the carrier type as E1 or T1. By default **E1** is selected.

- Select the Signaling Type as applicable: from PRI, RBS, QSIG, and E&M. These are the signaling types supported by the SARVAM UCS. By default **PRI** selected.

- Select the Framing Mode, as applicable. By default **CEPT1 MF (Auto CRC)** is selected as Framing Mode.

- Select the Line Coding as applicable. By default **HDB3** is selected.

- Select the ISDN Switch variant. By default **ETSI NET5** is selected as the variant.

- Select the mode for routing incoming calls. Incoming calls may be routed according to DDI Numbers or Port-wise.

  - **Route Incoming Calls Port wise:** Select this option if you want to route all incoming calls on the T1E1PRI trunk port to groups of extensions, without identifying the DDI number.

- **Route Incoming Calls DDI Number wise:** Select this option if you want to route incoming calls according to DDI numbers. Select this option only if your extension numbers are arranged sequentially. If the extensions are not arranged sequentially you are recommended to program this parameter from the Full Programming Access mode.

  If your installation uses multiple DDI blocks on the same T1E1PRI connection, you can configure only 2 such DDI blocks using the Wizard. You can configure more DDI blocks on a single T1E1PRI connection from Configuration.

- If you select DDI Number-wise routing, configure the following parameters:

  - **Root DDI #1:** enter the Root DDI # provided by your Service Provider. The Root DDI # is the main number assigned to the T1E1PRI trunk. It is also known as MSN Number, Pilot Number or Main Number[123].

    If your exchange requires area code to be sent with the DDI number, program the root number with area code. For example: Root DDI # is 2630555 and area code where the PBX is installed is 265, enter 2652630555.

  - **Route calls during day to:** Select where you want to the route the calls during day time. You can route the calls to the Operator or to a group of extensions or to Built-In Auto Attendant or to the Voice Mail Auto Attendant[124].

    If you select **Built-In Auto Attendant**, the default Voice Modules 02 to 13 containing the default Voice Messages (Morning, Afternoon and Evening Greetings, Built-In Auto Attendant Greeting and Guidance Messages) will be applied by the Wizard. Refer “Voice Message Applications” to know more.

    If you select **Voice Mail Auto Attendant**, the voice messages recorded and stored in Voice Mail module will be played.

    If you select **Extensions**, then select the extensions by clicking the empty field. A multiple selection box will open. Select the extensions from the left box.

    A maximum of 16 extensions can be selected. If you exceed the number, the Wizard will prompt you to make the selection again. You can delete the excess extensions from those you selected. You can also change the order of the selected landing extensions in the right box using drag and drop option.

    The selected extension numbers will appear in the field, separated by comma.

    - **When NR?:** Select an option for what action the system should take when there is no response (the incoming call is not answered within the *DDI Ring Timer,* set by default to 45 seconds) by the destination. By default, it is set to 'Disconnect'.

    - **When Busy?:** Select an option for what action the system should take when the landing destination is busy (for the duration of the *DDI Ring Timer;* default: 45 seconds). By default, it is set to 'Disconnect'.

---

123. This number will be used to prepare the DDI number to the Exchange (Reverse DDI) when an Outgoing call is made from the SARVAM UCS by the Extension assigned DDI Number. Also, this number will be sent to the Exchange without any modification when an Outgoing call will be made by an extension which is not assigned DDI number.

124. The Built-In Attendant offers 5 simultaneous calls, whereas Voice Mail's Auto Attendant offers 16 simultaneous calls.

- **Route calls during night to:** Select where you want to the route the calls during night time from the options: Operator or to a group of extensions or to Built-In Auto Attendant or to the Voice Mail Auto Attendant.

  If you select **Extensions**, you must now select the extensions by clicking the field provided for it. A maximum of 16 extensions can be selected. Follow the same procedure described above for selection of extensions for calls during day time.

  - **When NR?:** same as described above for calls during day time.
  - **When Busy?:** same as described above for calls during day time.

- **Start DDI#:** Enter the first DDI number given by the Service Provider. The Start DDI Number may be the Root DDI number, as seen in most of the cases.
- **Total DDIs:** Enter the Total DDI numbers. Ask your Service Provider.
- **Start Extension#:** Enter the number of the first extension from which the DDI assignment is to be done.

  For example, your Service Provider has given you DDI numbers 2630551 to 2630560. These numbers are to be routed to extensions 2001 to 2010. In this case, the Root DDI number as well as the Start DDI# will be 2630550. As 10 DDI numbers are used, enter Total DDI numbers =10 and Start Extension # = 2001.

- **When NR?:** same as described above for calls during day time.
- **When Busy?:** same as described above for calls during day time.

Now, configure the same parameters for Root DDI#2

- If you select Port-wise routing, configure the following parameters:

  - **Route Calls during Day to:** Select the landing destination for calls on T1E1PRI trunks during day time. You may select any option: Operator extension, other extensions, Built-In Auto Attendant or the Auto Attendant of the Voice Mail (if available).

    If you select **Built-In Auto Attendant**, the default Voice Modules 02 to 13 containing the default Voice Messages (Morning, Afternoon and Evening Greetings, Built-In Auto Attendant Greeting and Guidance Messages) will be applied by the Wizard. Refer “Voice Message Applications” to know more.

    If you select **Voice Mail Auto Attendant**, the voice messages recorded and stored in Voice Mail module will be played.

    If you select **Extensions** as landing destination, you must select the extensions. Double click the field and select the extensions from the left box.

    A maximum of 16 extensions can be selected. If you exceed the number, the Wizard will prompt you to make the selection again. You can delete the excess extensions from those you selected. You can also change the order of the selected landing extensions in the right side box using the Up/Down Arrow.

    The selected extension numbers will appear in the field, separated by comma.

- **Route Calls during Night to:** Select the landing destination for calls on T1E1PRI trunks during night time. Follow the same procedure for selecting extensions as described above.

- Click **Submit** to save changes. Repeat the same steps to program the second T1E1PRI Trunk.

- If you have finished configuring the T1E1 PRI Trunks, click **Next** to navigate further.

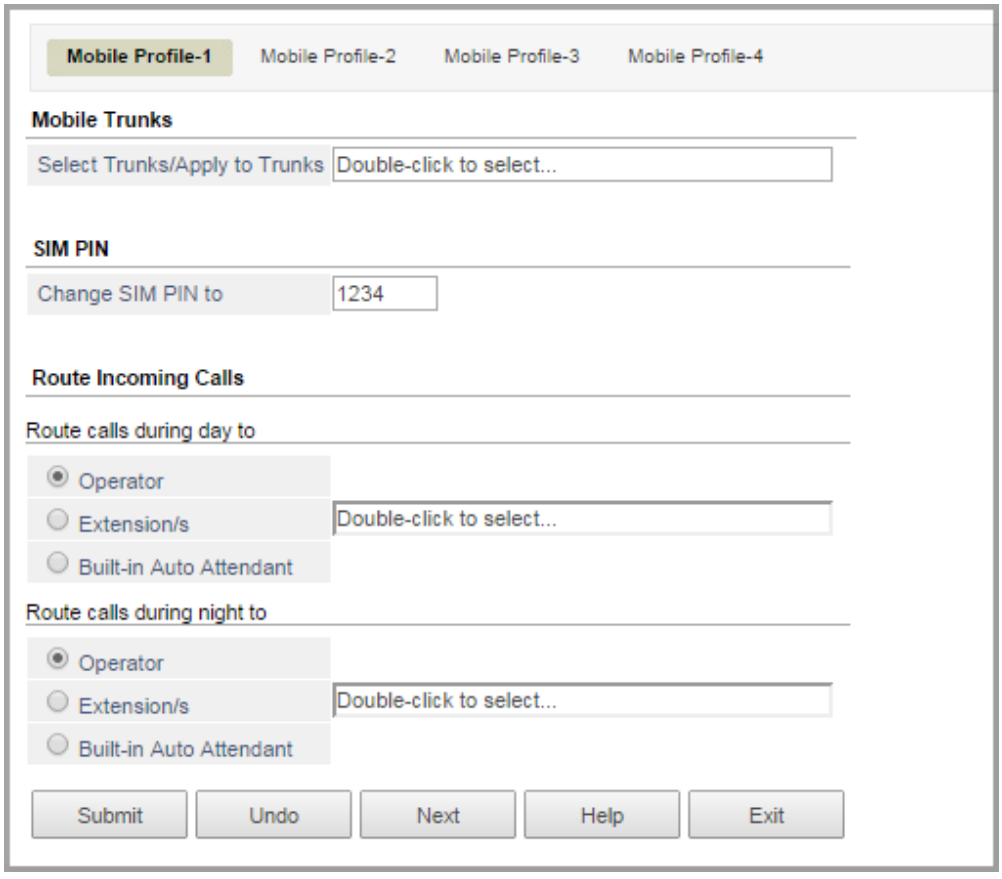
Mobile Trunks

- *You will reach this page only if your system has detected the presence of the Mobile Card in the system (as the **On-Site Configuration** check box is enabled) or if you have specified the number of Mobile Ports Used earlier in the **System Pre-requisites** page of the Wizard.*

- *SARVAM UCS supports up to 40 Mobile Ports.*

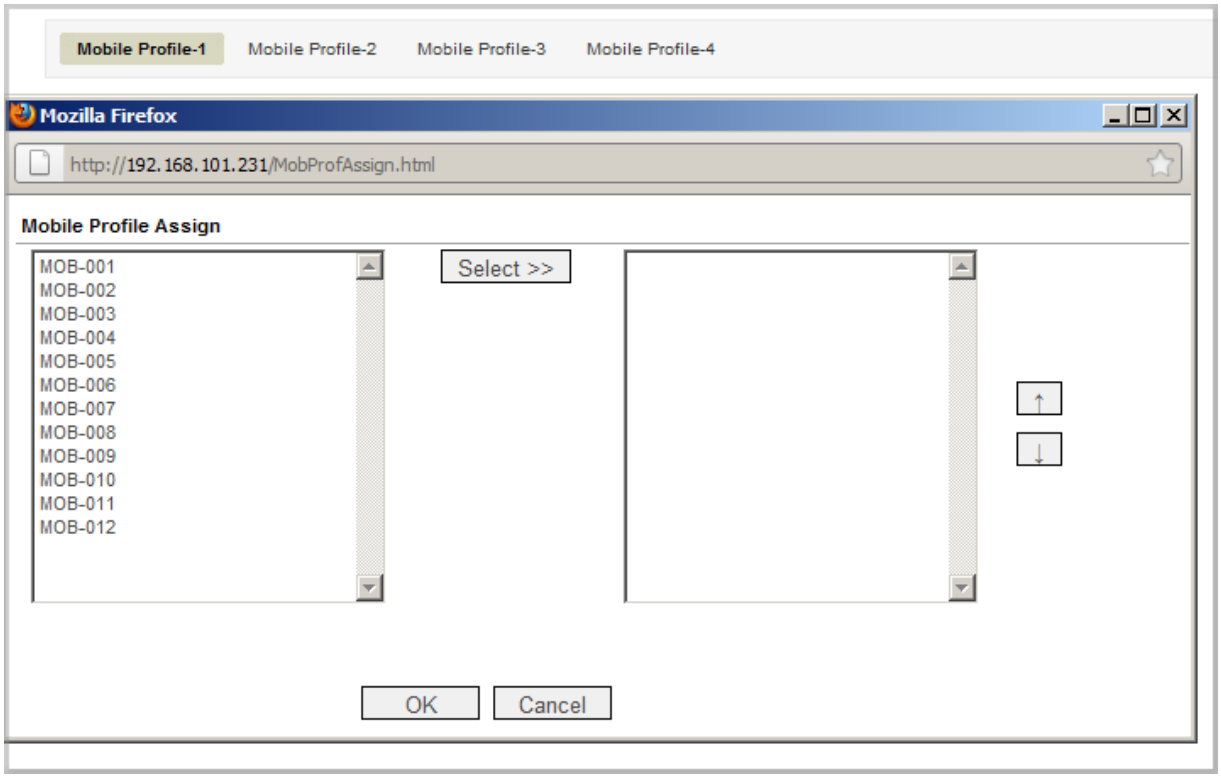
- *The Wizard makes configuration of the Mobile Trunks easy with **Mobile Profiles**. Instead of configuring each Mobile Trunk individually, you can group together trunks that are to be assigned the same features in a single Mobile Profile.*

You can configure as many as 4 different Mobile Profiles using the Wizard.



- Change the label of the Mobile Profile, if required. Select the desired Mobile Trunk tab and click the **Pencil** icon. A prompt will pop-up. Enter the desired name in the field of the prompt and click **OK**. The name you programmed will appear. For example rename Mobile Profile 1 as 'Vodafone'. Naming the profile with the Service Provider's name makes identification of the Trunk on which this profile is applied easy.

- Select the mobile trunks on which this Mobile Profile is to be applied. It is also possible to configure all the other parameters and then select the mobile trunks.



To select the desired mobile trunks, double click the box. The Wizard will display the number of Mobile Trunks you specified in the **System Pre-requisites** page on the left side box. All the Mobile Trunks appear in this box and are arranged sequentially in the increasing order of their software port number.

Place your cursor on the desired trunks and click **Select>>** button. The selected Mobile Trunks will appear on the right side box. It is also possible to select a range of trunks at a time pressing the 'SHIFT and 'Down' arrow keys.

- Change the SIM PIN for the mobile profile.

  Recall that you have changed the SIM PIN of the SIM Card using a Handset before installing it in the system to the default **1234**. Now, you may change it to a desired SIM PIN.

  Since you are changing the SIM PIN for a Mobile Profile, this SIM PIN will be applied for all the Mobile Trunks you selected for this Mobile Profile.

You must click 'Submit' after you enter the new PIN. Wait for 5 seconds, and then refresh this page to view the new SIM PIN.

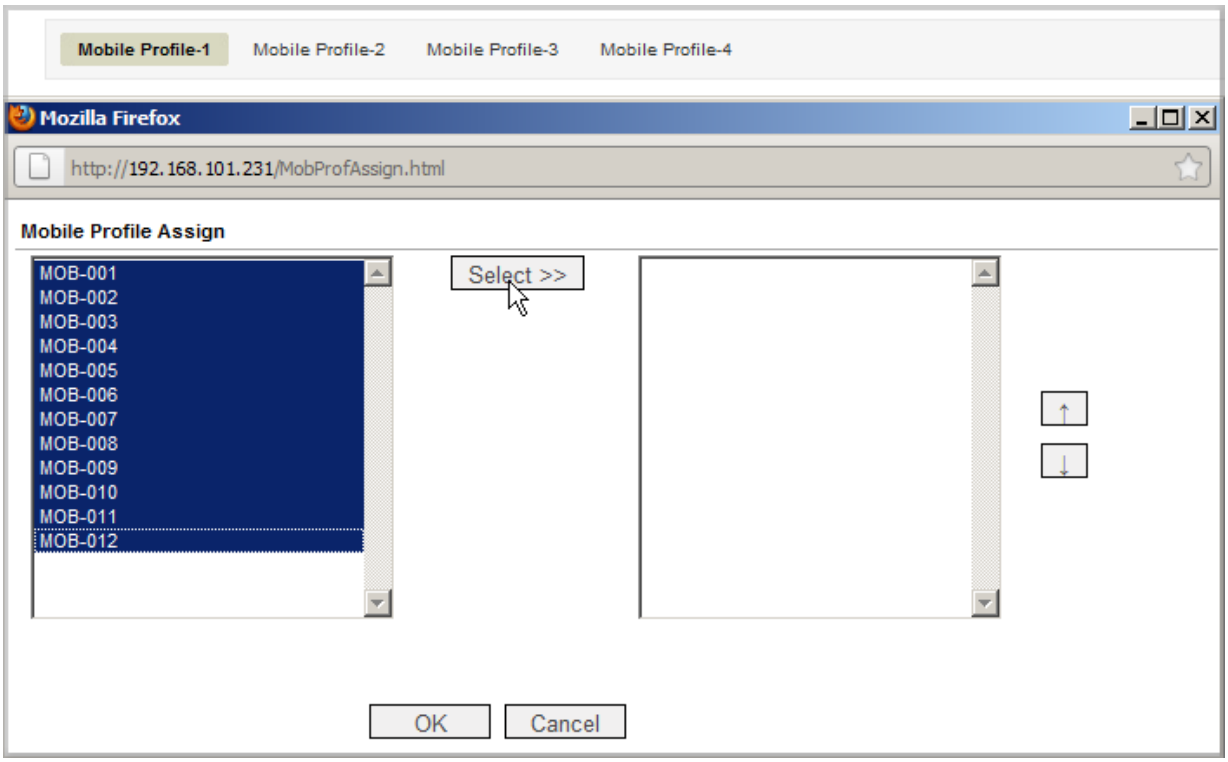
- Select the extensions to which the incoming calls on the Mobile Trunks should be routed.

- In **Route Calls during Day to**, select the landing destination for calls on Mobile trunks during day time. You may select any option: Operator extension, other extensions, Built-In Auto Attendant or the Auto Attendant of the Voice Mail (if available).

If you select **Built-In Auto Attendant**, the default Voice Modules 02 to 13 containing the default Voice Messages (Morning, Afternoon and Evening Greetings, Built-In Auto Attendant Greeting and Guidance Messages) will be applied by the Wizard. Refer "Voice Message Applications" to know more.

If you select **Voice Mail Auto Attendant**, the voice messages recorded and stored in Voice Mail module will be played.

If you select **Extensions** as the landing destination, select the extensions in the corresponding field. Double click this field, the multiple selection box will open. Select the extensions from the left side box.



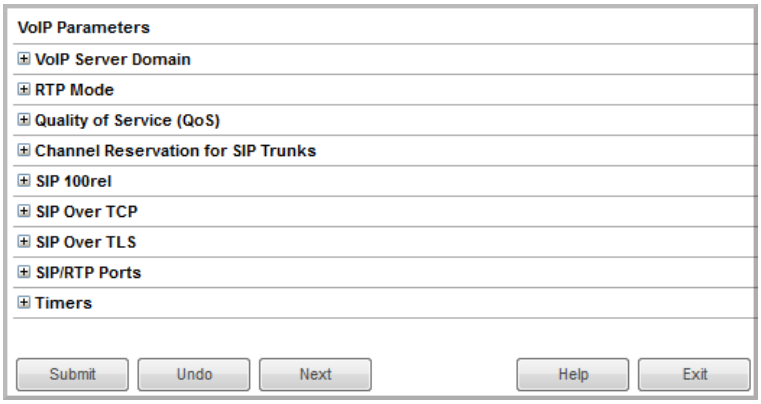
A maximum of 16 extensions can be selected. If you exceed the number, the Wizard will prompt you to make the selection again. You can delete the excess extensions from those you selected. You can also change the order of the selected landing extensions in the right side box using the Up/Down Arrow.

The selected extension numbers will appear in the field, separated by a comma.

- In **Route Calls during Night to**, select the landing destination for calls on Mobile trunks during night time. Follow the same instruction as in the previous step to select extensions as the landing destination.
- Click **Submit** to save your settings in the Mobile Profile. Repeat these steps to configure another Mobile Profile.
- If you have finished configuring the Mobile Profiles, click **Next** to navigate further.
- If there is any mobile trunk you have not assigned a Mobile Profile, the Wizard will pop-up an alert informing you about the trunk you have not programmed. You must complete configuring the parameters for this trunk and only then will the wizard allow you to configure other parameters.

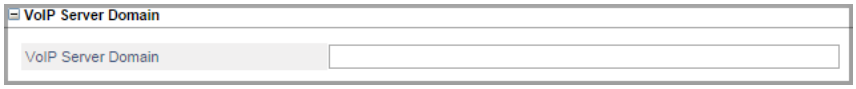
VoIP Network

- To configure the VoIP Parameters, click **VoIP Network**. The VoIP Parameters page opens.



VoIP Server Domain

Click **VoIP Server Domain** to expand.



- **VoIP Server Domain:** This parameter is of relevance only if you are configuring SIP Extensions on the system.

  The system is capable of maintaining a domain for registering SIP/UC clients (any SIP-enabled device) as SIP Extensions.

  Configure the Server Domain if you want SIP/UC clients to register with the Registrar Server of the system using the domain handled by the system[125]. The Domain Name can be a maximum of 40 characters. Default: Blank.

If you configure Server Domain for registration of SIP clients, you must also map the Domain Name and the IP Address of the WAN Port to the DNS Server in the network.

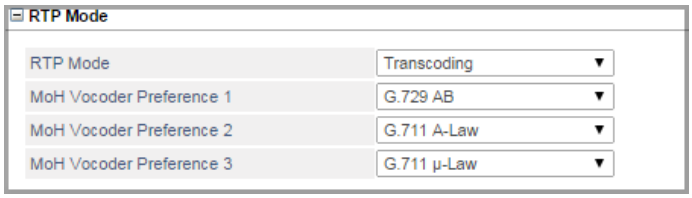
125. *SIP clients can be registered with the CPU either using the domain handled by the Vocoder module or using the WAN or LAN Port IP Address.*

     If domain is programmed, the CPU will listen for the SIP message which is redirected to the programmed domain only. It will also listen for SIP messages on the WAN IP address and LAN IP address.

     But if domain is not programmed, the CPU will listen for SIP messages only on the WAN IP Address and LAN IP address.

RTP Mode

Click **RTP Mode** to expand.



- **RTP Mode:** Select the desired RTP mode using which you want SARVAM UCS to route SIP to SIP calls. You can select from the following:

  - **Transcoding:** When this option is selected, RTP packets will be routed through the Vocoder module and Vocoder channels[126] will be used for SIP to SIP calls. SIP/UC Users (Standard/Extended SIP Devices and Matrix VARTA UC Clients) will be able to access all the features of SARVAM UCS. This option uses two Vocoder channels for SIP to SIP calls. Thus the maximum number of SIP to SIP calls per Vocoder Module is equal to the number of Vocoder channels divide by 2.

  - **RTP Relay:** When this option is selected, RTP packets will be routed through the Vocoder module but no Vocoder channel will be used for SIP to SIP calls. The system will use Vocoder channels to route SIP to TDM calls and vice versa. System will use Vocoder channels for some features and Standard SIP Clients will be able to use limited features of SARVAM UCS. Refer to "SARVAM UCS Features supported with RTP/Direct RTP" for more details.

  - **Direct RTP:** When this option is selected, no Vocoder channel will be used for SIP to SIP calls and RTP packets will be sent to and fro directly between SIP end points. If transfer of RTP packets is not possible between SIP end points, system will use RTP Relay as the fallback option. The system will use Vocoder channels to route SIP to TDM calls and vice versa. System will use Vocoder channels for some features and Standard SIP Clients will be able to use limited features of SARVAM UCS. Refer to "SARVAM UCS Features supported with RTP/Direct RTP" for more details.

  - *Direct RTP is not supported on SIP Trunks and Matrix VARTA UC Clients. System will use RTP Relay for SIP Trunks and Matrix VARTA UC Clients.*

  - *If RTP Relay or Direct RTP is selected,*
    - *Maximum 550 Audio calls[127] or 55 Video calls are supported in SARVAM UCS.*
    - *Certain voice messages will not be played. See "Voice Message Applications" for more details.*

  - *If you change the RTP Mode, the system will release all ongoing calls.*

- **MoH Vocoder Preference:** If RTP Mode is set as RTP Relay or Direct RTP, the system will play the MoH through the Vocoder module when the system holds any SIP end point. The MoH is played as per the configured MoH Vocoder Preference.

  Select the Vocoders in the order of their preference, for **MoH Vocoder Preference 1**, **MoH Vocoder Preference 2** and **MoH Vocoder Preference 3**.

  If required, you can customize the MoH to be played as per your requirement. For detailed instructions, see "Uploading Custom MoH".

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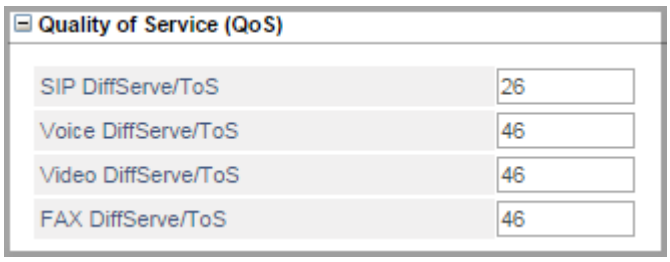
126. *The number of Vocoder channels that will be supported would be as per the license you purchase.*
127. *For PENX, maximum 64 Audio Calls or 16 Video Calls are supported in SARVAM SMB.*

Quality of Service (QoS)

Click **Quality of Service (QoS)** to expand.

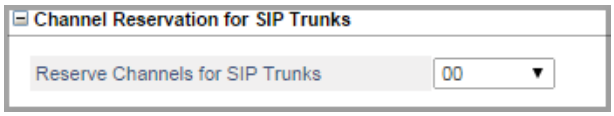


- **Quality of Service (QoS):** QoS refers to priority of IP packets on network layer. It can be programmed for both signaling (SIP) and media (Voice, Video and Fax). Configure the following types QoS:
    - **SIP DiffServe/ToS:** The system sends all the SIP signaling messages with this QoS setting. This field defines the priority bits for SIP messages. The Valid *DiffServe* range is from 00-63, default: 26
    - **Voice DiffServe/ToS:** The system sends all the Voice packets with this QoS setting. This field defines the priority bits for Voice packet. The Valid *DiffServe* range is from 00-63, default: 46
    - **Video DiffServe/ToS:** The system sends all the Video packets with this QoS setting. This field defines the priority bits for Video packet. The Valid *DiffServe* range is from 00-63, default: 46
    - **Fax DiffServe/ToS:** The system sends all the Fax packets with this QoS setting. This field defines the priority bits for Fax packet. The Valid *DiffServe* range is from 00-63, default: 46

QoS parameters are applicable for all packets (SIP/ Media) leaving both LAN and WAN port as well as TCP connection.

Channel Reservation for SIP Trunks

Click **Channel Reservation for SIP Trunks** to expand.



- **Channels Reserved for SIP Trunks:** The system supports up to 128 voice channels, which can be used by SIP Extensions and SIP trunks.

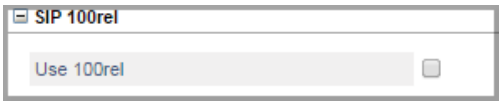
  It may happen that SIP Extension users use up most of the channels of the Vocoder module, leaving too few or none for making/receiving SIP Trunk calls.

  This can be avoided by reserving some voice channels exclusively for SIP trunk calls.

  Specify the minimum number of voice channels you want to reserve for SIP Trunk calls. By default, no channel is reserved.

SIP 100rel

Click **SIP 100rel** to expand.



- **SIP 100rel:** This parameter is to be configured if you want to support reliable transmission of (SIP) provisional responses. Enable 100rel by selecting the check box, if you want CPU to use 100rel for reliable transmission of SIP provisional responses and to use PRACK (Provisional Acknowledgement). Default: Disabled.

SIP Over TCP

Click **SIP Over TCP** to expand.

- **SIP Over TCP:** SARVAM UCS supports transporting of SIP messages over User Datagram Protocol (UDP), Transfer Control Protocol (TCP) as well as Transport Layer Security connection (TLS). Despite the advantages that SIP over TCP and SIP over TLS offer, it is more common to use UDP to transport SIP messages.

  By default, SIP Over TCP is enabled. To be able to send SIP messages over TCP, you must configure 'TCP' as the 'Default Transport for Outgoing Messages' on the 'SIP Trunk Parameters' page.

  If you do not want to transport SIP messages using TCP, clear the SIP Over TCP check box.

SIP Over TLS

Click **SIP Over TLS** to expand.

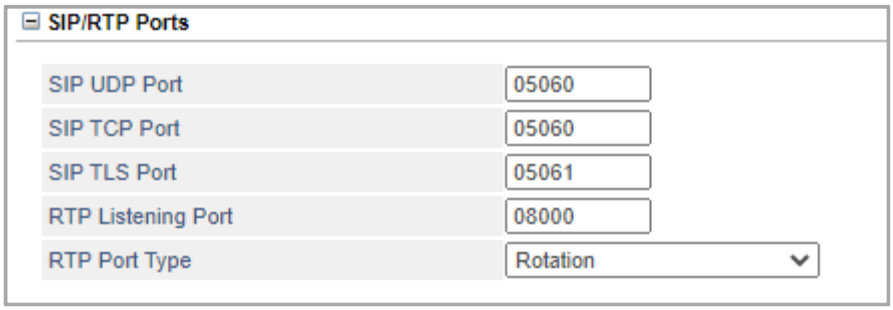
- **SIP Over TLS:** SARVAM UCS supports transporting of SIP messages over Transport Layer Security. TLS offers secure SIP signaling.

  By default, SIP Over TLS is enabled. To be able to send SIP messages over TLS, you must configure 'TLS' as the 'Default Transport for Outgoing Messages' on the 'SIP Trunk Parameters' page.

  If you do not want SIP messages to be transported using TLS, clear the SIP Over TLS check box.

SIP/RTP Ports

Click **SIP/RTP Ports** to expand.



- **SIP UDP Port:** This port defines the port on which SARVAM UCS listens for SIP messages transported over UDP. This port is also used as the source port for sending SIP messages to the remote peer. The valid range for this port is 1025-65535. Default: 05060.

- **SIP TCP Port:** This port defines the port on which SARVAM UCS listens for SIP messages transported over TCP. This port is also used as the source port for sending SIP messages to the remote peer. The valid range for this port is 1025-65535. The default SIP TCP Port is 05060.

- **SIP TLS Port:** This port defines the port on which SARVAM UCS listens for SIP messages transported over TLS. This port is also used as the source port for sending SIP messages to the remote peer. The valid range for this port is 1025-65535. The default SIP TLS Port is 05061.

- **RTP Listening Port:** This port defines the port on which SARVAM UCS listens for RTP Packets. This port is also used as the source port for sending RTP packets to the remote peer. The valid range for this port is 1025-56734. The default RTP Listening Port is 08000.

Valid Port Range for MENX/LENX: 1025 - 56734.
Valid Port Range for PENX: 1025 - 64510.

- **RTP Port Type:** Select the desired Port Type — Rotation, First Free — from the drop-down.The default RTP Port Type is Rotation.

  If you select Rotation, for each new session the system checks and allocates the next available RPT port in a sequential manner. Make sure all the RTP Ports are forwarded.

  If you select First Free, for each new session the system allocates the first available RTP Port regardless of its sequence.That is, the system will start checking from the starting RTP port for every new session and allocate the first free port available.In this case the audio and video ports need to forwarded separately. However, for a video call both audio and video ports will be consumed. For a video call maximum 400 ports can be forwarded.

  For LENX/MENX and GENX
  RTP Listening port = configured RTP Listening port
  Ending number of audio port = RTP Listening port + (number of audio calls * 4)
  Range of audio port = RTP Listening port to Ending number of audio port
  Starting number of video port = RTP Listening Port + 4400
  Ending number of video port = The total number of video port starting range + (number of video calls * 4)
  Range of Video port = Starting number of video port to Ending number of video port

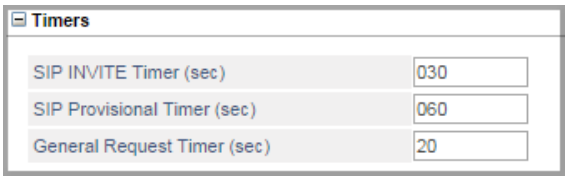
Let us understand this with the help of an example:
Number of audio calls to be made 100 and number of video calls to be made 50
RTP Listening port = 12000
Ending number of audio port = 12000 + (100*4) = 12400
Range of audio port = Range 12000 to 12400
Starting number of video port = 12000 + 4400 = 16400
Ending number of video port = 16400 + (50 * 4) = 16600
Range of Video port = 16400 to 16600

For PENX
RTP Listening port = configured RTP Listening port
Ending number of audio port = RTP Listening port + (number of audio calls * 4)
Range of audio port = RTP Listening port to Ending number of audio port
Starting number of video port = RTP Listening Port + 512
Ending number of video port = The total number of video port starting range + (number of video calls * 4)
Range of Video port = Starting number of video port to Ending number of video port

Let us understand this with the help of an example:
Number of audio calls to be made 50 and number of video calls to be made 8
RTP Listening port = 12000
Ending number of audio port = 12000 + (50*4) = 12200
Range of audio port = Range 12000 to 12200
Starting number of video port = 12000 + 512 = 12512
Ending number of video port = 12512 + (8 * 4) = 12544
Range of Video port = 12512 to 12544

Timers

Click **Timers** to expand.



- **SIP Invite Timer (sec):** This is the time in seconds that the Vocoder module waits for a response from the called party after ending INVITE message. This timer starts after sending INVITE message to the called party and stops on receipt of the provisional response or the final response or when the user disconnects the call. On expiry of the timer, the call process is terminated by the SARVAM UCS and an error tone is played to the user. The range of the SIP Invite Timer is 010-180 seconds. Default: 30 seconds.

- **SIP Provisional Timer (sec):** This is the time in seconds that the Vocoder module waits for the final response after receiving the provisional response from the called party. This timer starts on the receipt of the provisional response from the called party and stops on receipt of the final response from the called party or when the user disconnects the call. On expiry of the timer, the SARVAM UCS terminates the call process and plays error tone to the user. The range of SIP provisional Timer is 010-180. Default: 60 seconds.

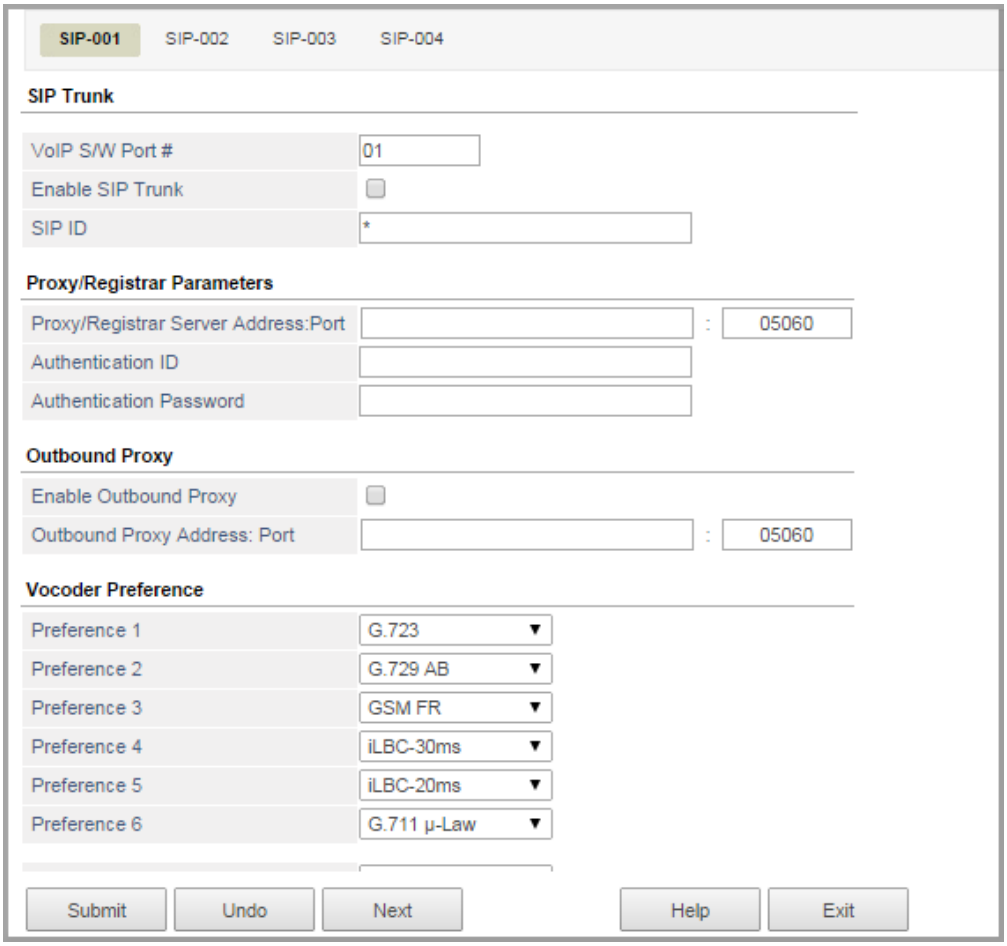
- **General Request Timer (sec):** The time in seconds for which the Vocoder module waits for response for a transaction request. This timer starts on the initiation of a transaction. This timer stops on receipt of a response for the request. On expiry of the timer, the Vocoder module clears the transaction. This timer is

used for Registration request, etc. The range of the General Request Timer is 10-60 seconds. Default: 20 seconds.

- Click **Submit** to save.
- Click 'Next' to navigate to further.

SIP Trunks

- *You will reach this page only if you have specified **Number of SIP Trunks Used** on the **System Pre-requisites** page.*
- *SIP Trunks are to be configured only if you are using Internet Telephony Service Providers for VoIP calls.*
- *The Wizard however, allows you to configure only 4 SIP Trunks (even if you have specified more than 4 SIP Trunks being used on the **System Pre-Requisites** page). You can program the remaining SIP Trunks from Configuration.*



- You may change the label of the SIP Trunk (rename 'SIP Trunk 1'), if required. You may use the name of the ITSP to make identification of the SIP Trunk easy.

  Click the **Pencil** icon. A prompt will pop up, asking you if you want to change the name of this SIP Trunk.

Enter the desired name in the blank field. Click **OK**. The new name will appear instead of the SIP Trunk number.

- Select the **Enable SIP Trunk** check box to enable the SIP Trunk.
- Enter the **SIP ID** provided by the ITSP. This is the ID which callers will use to call this SIP Trunk. The SIP ID may be a number or text.
- Enter the **Proxy/Registrar Server Address** and the **Registrar Server Port** provided by the ITSP. The Registrar Server Address can be an IP Address or domain. The Registrar Server Listening Port ranges from 1025 to 65535. The default Registrar Server Port is 5060.
- Enter the **Authentication ID** (User ID) and **Password** provided by the ITSP.
- Select the **Enable Outbound Proxy** check box, if the ITSP who provided this SIP Trunk has a SIP outbound server to handle voice calls. By default Outbound Proxy is disabled.
- If you have enabled Outbound Proxy, enter the **Outbound Proxy Server Address** and the **Server Port** provided by the ITSP. This can be an IP Address or Domain name.
- Set the desired **Vocoder Preference** for this SIP Trunk. Vocoders are the various Voice Codecs used to compress the data in RTP packets for optimum use of bandwidth and for ensuring voice quality. You can set 7 Vocoders options in the order of preference for this SIP account.
- Select the **DTMF Option**. The DTMF option you select will determine how the DTMF digits will be sent over the IP Network, when a DTMF digit is pressed. The system supports three DTMF options: RTP (RFC 2833), SIP Info, and InBand. Select the appropriate option. By default RTP (RFC 2833) is selected.
- **RFC2833 Payload Type:** If you have selected RTP (RFC 2833) as the DTMF Type, you must configure the value of RFC2833 Payload Type. The RTP packets will be tagged as DTMF as per the set value. The value of RFC2833 Payload Type can be set from 96 to 124.
- Select the appropriate **Fax Type**. The system supports T.38 (UDPTL), T.39 (RTP) and Pass-Through.
- Select the **Source Port IP Address**.
  - Select **Use Ethernet Port IP Address**, if the WAN port of the system is connected directly to the public internet.
  - Select **Use IP Address Fetched Using STUN**, if the WAN port of the system is located behind a NAT router other than Symmetric.
  - Select **Use Router's Public IP Address**, if the WAN port of the system is located behind a NAT Router (any type).
- In **No. of Simultaneous Calls**, select the number of simultaneous calls you want to allow on this SIP Trunk.

  The number of simultaneous SIP calls depends on the number of simultaneous calls allowed by the ITSP with whom you have subscribed this SIP Trunk and the number of simultaneous calls supported by SARVAM UCS.

- In **Route Calls during Day to**, select the landing destination for calls on SIP trunks during day time. You may select any option: Operator extension, other extensions, Built-In Auto Attendant or the Auto Attendant of the Voice Mail (if available).

  If you select **Built-In Auto Attendant**, the default Voice Modules 02 to 13 containing the default Voice Messages (Morning, Afternoon and Evening Greetings, Built-In Auto Attendant Greeting and Guidance Messages) will be applied by the Wizard. Refer "Voice Message Applications" to know more.

  If you select **Voice Mail Auto Attendant**, the voice messages recorded and stored in Voice Mail module will be played.

  If you select **Extensions** as the landing destination, select the extensions in the corresponding field. Double click the field, the multiple selection box will open. Select the extensions from the left box.

  A maximum of 16 extensions can be selected. If you exceed the number, the Wizard will prompt you to make the selection again. You can delete the excess extensions from those you selected. You can also change the order of the selected landing extensions in the right side box using Up/Down Arrow.

  The selected extension numbers will appear in the field, separated by comma.

- In **Route Calls during Night to**, select the landing destination for calls on SIP trunks during night time. Follow the same instruction as in the previous step to select extensions as the landing destination.

- Click **Submit** to save your settings in the SIP Trunk. Repeat these steps to configure another SIP Trunk.

- If you have finished configuring the SIP Trunks, click **Next** to navigate further.

- If there is any SIP Trunk you have not configured, the Wizard will pop-up an alert informing you about the trunk you have not programmed.

  You must complete configuring the parameters for this SIP Trunk and only then will the wizard allow you to configure other parameters.

For the SIP Trunks, the Wizard uses the following resources:
- *Routing Groups 67-74.*
- *Trunk Feature Template 46-49.*
- *SIP Hardware Template 2-5.*
- *Do not change the these resources when using Configuration.*

Emergency Numbers

- *This page displays the default Emergency Numbers according to the **Region** selected for the system. In other words, it displays the Emergency Numbers specific to your country where the SARVAM UCS is installed.*

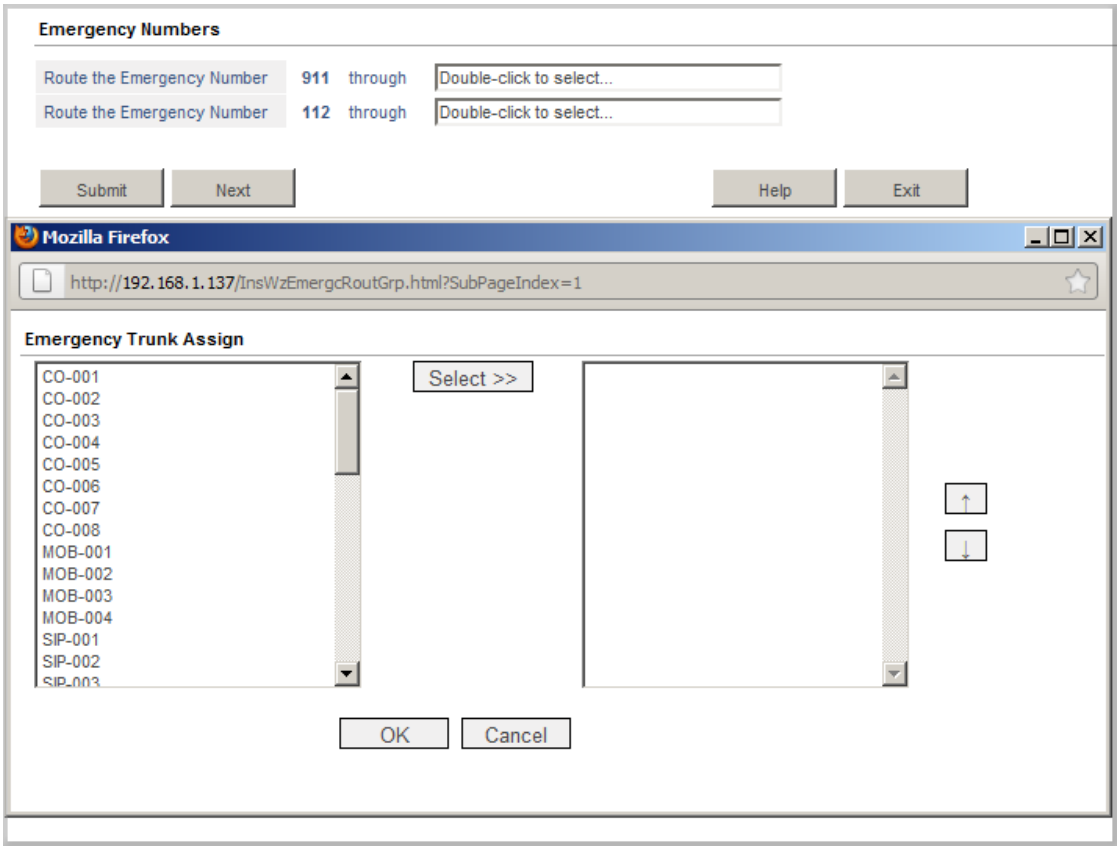
- *If there are no Emergency Numbers defined as per the selected region, the system displays the message ' No Emergency Number programmed.'*

- *The Emergency Numbers on this page are non-editable. All you need to do is to select the Outgoing Trunk Bundle Group (OGTB) for each Emergency Number. For example, '112' is the default Emergency Number for the mobile network. So, you may select the Mobile Trunk for dialing this number.*

- *Make sure that the trunks configured by default for each Emergency number route the Emergency call to the correct network.*

- For each default Emergency number, select the trunks to be used to route this call. Double click the **Through** field. The box will open, displaying the trunk types present in the system in the left box.
  All the Trunk types present in the system are arranged by their type, in the sequence of their software port, irrespective of their hardware location.

- Double click the desired trunk type and click **Select>>**. The selected trunk type will appear on the box on the right.



If you select trunks of the same type in sequential order, like: CO-001, CO-002, CO-003, BRI-001, BRI-002 and MOB-002, the same trunk type will be grouped in one OG Trunk Bundle; CO-001, CO-002, CO-003 will be OGTB #1, BRI-001 and BRI-002 will be grouped as OGTB#2, and MOB-002 as OGTB#3.

If you select the trunks of the same type in a non-sequential order, for example: CO-001, MOB-001, BRI-001 and CO-002, four OGTBs will be formed with CO-001 as member of OGTB#1, MOB-001 as member of OGTB#2, BRI-001 in OGTB#3 and CO-002 in OGTB#4. So, despite two same trunk types being selected (CO-001 and CO-002) they are grouped in separate OGTBs.

It is possible to change the sequence of trunks on the right side box using the Up/Down Arrow. You can also delete a particular trunk from the right box.

A maximum of 8 OGTB are allowed. If you exceed the number, the Wizard will show an alert indicating that the system is short of resources.

- Click **OK**. All the trunks you selected for a particular Emergency Number (in the right-hand side box) will appear on the 'Through' field, sequentially separated by comma in the order of selection made.

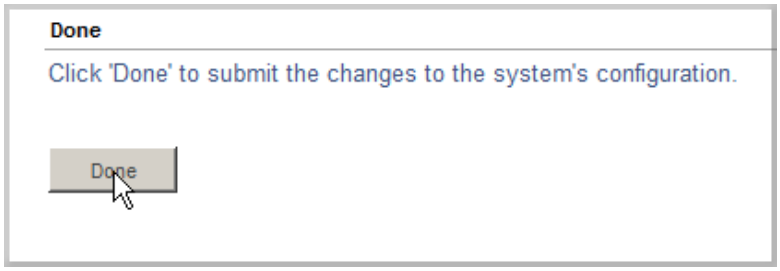
- Click **Submit** to save the values entered on this page.
- Click **Next** to navigate further.

You may program the Emergency Numbers of your country from Configuration.

Disclaimer:

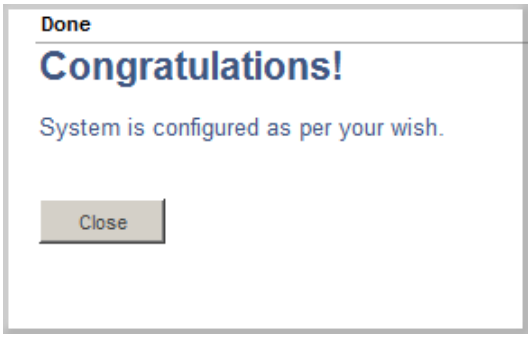
- *Matrix Comsec will not be responsible for incorrect programming of Emergency Numbers.*

Done



You have now reached the last screen of the Wizard. Click **Done** to submit the configuration you have done using the Wizard.

After completing the validation of the configuration, the Wizard will inform you about successful configuration of the system with the message: “Congratulations! System is configured as per your wish".



Exiting the Wizard

Each page of the Wizard also has an **Exit** button, which you may click to exit the Wizard.

However, the changes you made till the page you exited from will not be applied to the system configuration files.

The Standard PBX Wizard has been designed keeping a very broad user base in mind and is limited to the configuration of the parameters most commonly required by a broad user group. So, it does not cover the configuration of all the features and facilities of the system. You can configure other features “Configuring SARVAM UCS”.

Using Configuration

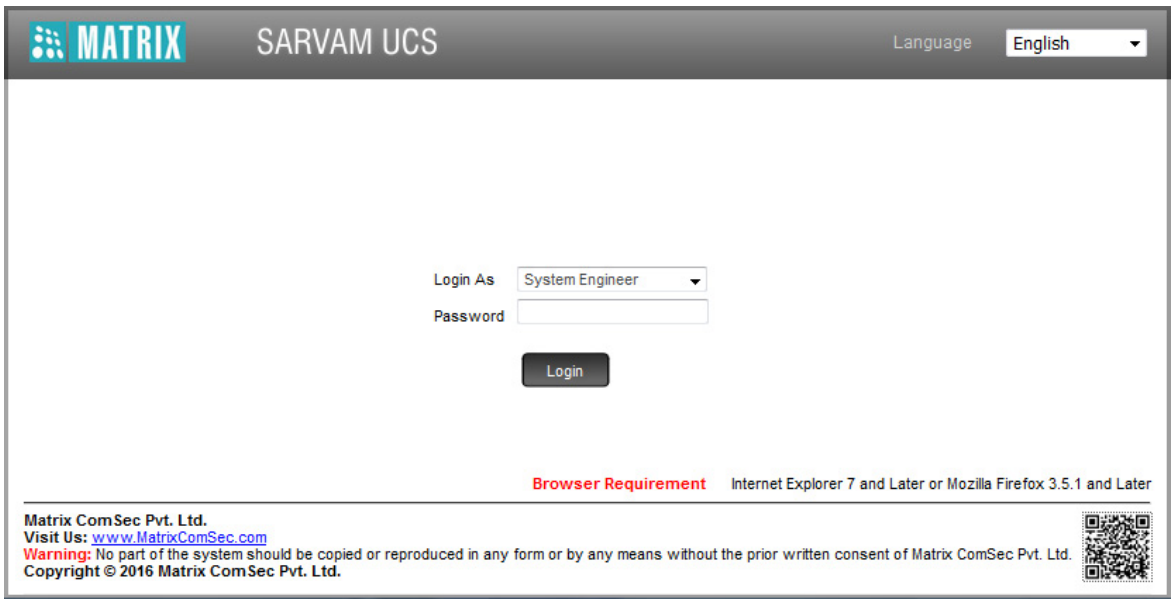
The Configuration allows configuration of all programmable parameters of the system.

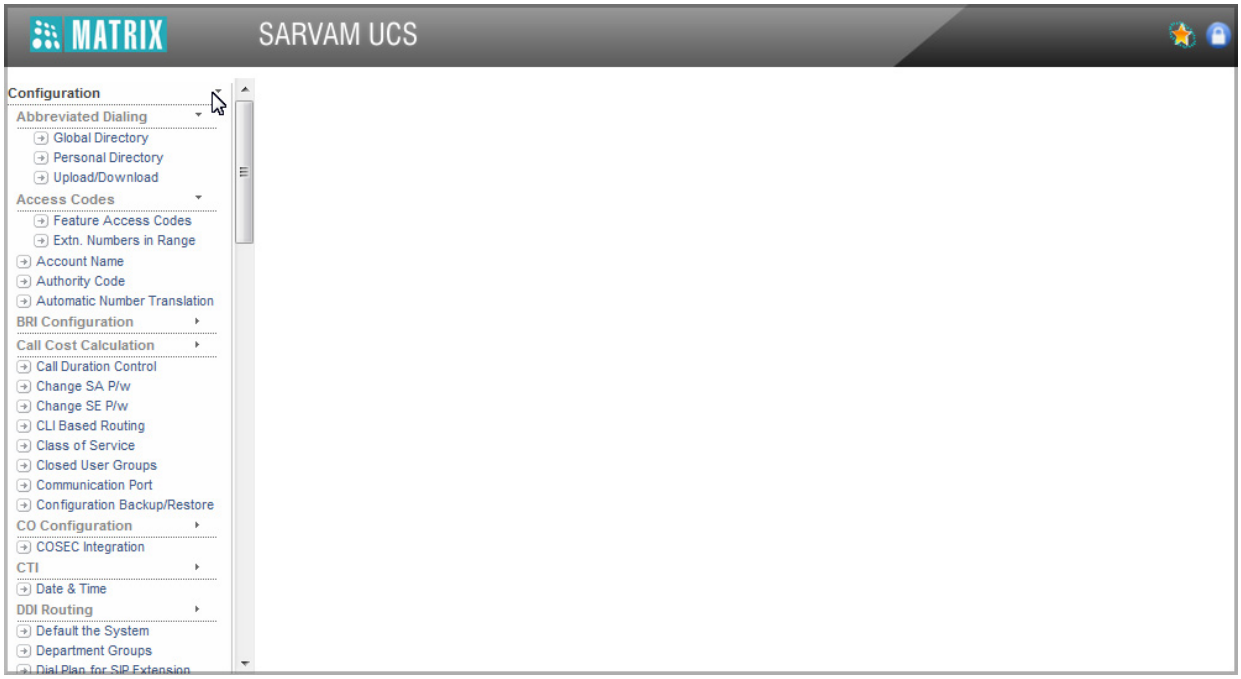
You can program all the parameters using Jeeves or an extension Telephone (DKP or SLT) of the SARVAM UCS.

You are recommended to use Jeeves. If you choose to use a telephone, you are recommended to use a DKP for ease of operation.

If you use an extension telephone to configure the system then make sure you change the default SE Extension Password.You can only configure the basic network and debug parameters using this default SE Extension Password.

To configure the Configuration parameters, you must log into Jeeves via System Engineer Login and click the **Configuration** link.





Also, refer the topic “Configuring SARVAM UCS”.

To program the parameters using a Telephone, enter into SE mode from a DKP/SLT by dialing ***1#91*** followed by the SE password.

For detailed instructions, refer the topic “Entering the SE mode using a Telephone”.

Avoid modifications or use of the following system resources, when configuring the system from the Configuration.

- *Station Basic Feature Template number 2 to 10.*
- *Class of Service Groups 2 to 19.*
- *Outgoing Trunk Bundle Groups 2 to 27*
- *Outgoing Trunk Bundles 5 to 108.*

The system uses these resources for the Quick Installation Wizard-Standard PBX. Changes made in them will not be updated in the Wizard.

Configuring System Pre-requisites

System Pre-requisites for SARVAM UCS

This displays the System resources (number of trunk and extension ports supported) of SARVAM UCS. It is quite common for users to utilize the system resources below its capacity, especially when they begin using a new system.

To make the task of configuring for such users easier, SARVAM UCS allows you to specify the number of trunks and extension ports you want to configure. Accordingly all the relevant pages of Jeeves will show only as many trunk and extension ports that you have specified, instead of showing all the ports supported by SARVAM UCS.

To be able to do this, you must configure the **System Pre-requisites** using Jeeves or a Telephone.

On-site Configuration

SARVAM UCS makes configuration even more focused by making it possible to configure only those trunk and extension ports which are actually present in the system.

This can be done by enabling the ***On Site Configuration*** flag.

When **On Site Configuration** flag is enabled, SARVAM UCS will detect all the different trunk and extension port types present in the system (at Power-ON). Accordingly, all the relevant pages of Jeeves will show only those ports detected by the system for configuration.

The system will detect the presence of ports at each Power ON/Reset. Whenever a new card is found, the range of ports is updated and displayed on Jeeves. You can then define the number of ports to be used.

By default, the On-site Configuration flag is disabled.

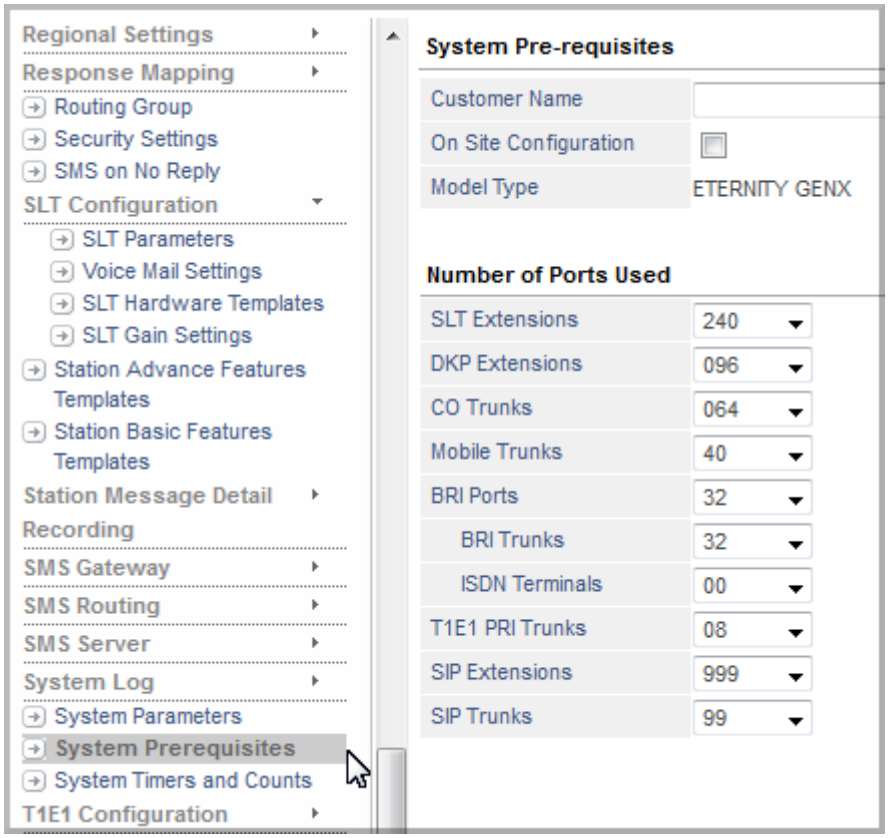
When the On Site Configuration flag is enabled, the Quick Installation Wizard - Standard PBX will also display only the ports and port types that are on-board. Enable this flag if you want the Wizard to display only the port types present in the system.

It is recommended that you enable the On-Site Configuration flag when you are configuring the system at the installation site.

Defining System Pre-requisites using Jeeves

- Log into Jeeves as System Engineer.

- Under **Configuration**, click **System Pre-requisites** to open the page.



- Configure the following parameters:

  - **Customer Name:** You can assign the name of the enterprise/organization that is using SARVAM UCS as the Customer Name. The Customer Name may contain up to 80 characters. You may enter the address of organization/enterprise along with the name.

    The Customer Name you assign will appear on the various System Reports generated and printed by the SARVAM UCS.

  - **On Site Configuration:** Click the **On-site Configuration** check box to enable.

    Click **Submit** at the bottom of the page.

    The fields for **Number of Ports Used**, will be populated with exactly the number the system has detected.

    The fields for port types which are not available on-board are displayed as non-editable fields.

    For example, the system has detected SLT20 Card at power-on, so the maximum number of SLT ports will be 20. If you want only 18 SLTs to be used, select 18.

  - *If the system has detected BRI ports, the fields **BRI Trunks** and **ISDN Terminals** will be editable.*

  - *By default, the number of BRI Trunks will be equal to the Number of BRI ports used.*

  - *The number of ISDN terminals used by default will be zero. Only when the System Engineer changes the value of the Number of BRI Trunks to be used will the number of ISDN Terminals change to the*

number of available BRI ports x 8. In other words, the number of ISDN terminals will be: Number of BRI Ports used minus the Number of BRI Trunks used multiplied by 8.

- **Model Type**: This displays the name of the model that you are configuring — ETERNITY LENX, ETERNITY MENX, ETERNITY GENX or ETERNITY PENX.

- **Number of Ports Used:** Define the number of ports to be used for each Port Type (CO, DKP, SLT, Mobile, T1E1 PRI, BRI, SIP) in the respective boxes.

  For example, if you want 8 CO Trunks, 24 DKP extensions, and 128 SLT extensions to be used, select the same numbers in the respective boxes.

- Click **Submit** at the bottom of the page to save changes.

Configuring ‘Region’

The SARVAM UCS is a versatile system that can operate anywhere in the world, meeting the diverse customer requirements worldwide.

To speed up the process of system configuration, SARVAM UCS is supplied with factory-set values for the system and feature settings, referred to as “Default Settings”. These factory-set values are loaded when the system is installed and are sufficient for getting the system into operation. However, users may alter or customize the Default Settings to match their exact requirement.

SARVAM UCS provides Default Settings to match country/region-specific requirements of users around the world. The system is designed to work efficiently in any country with these default settings.

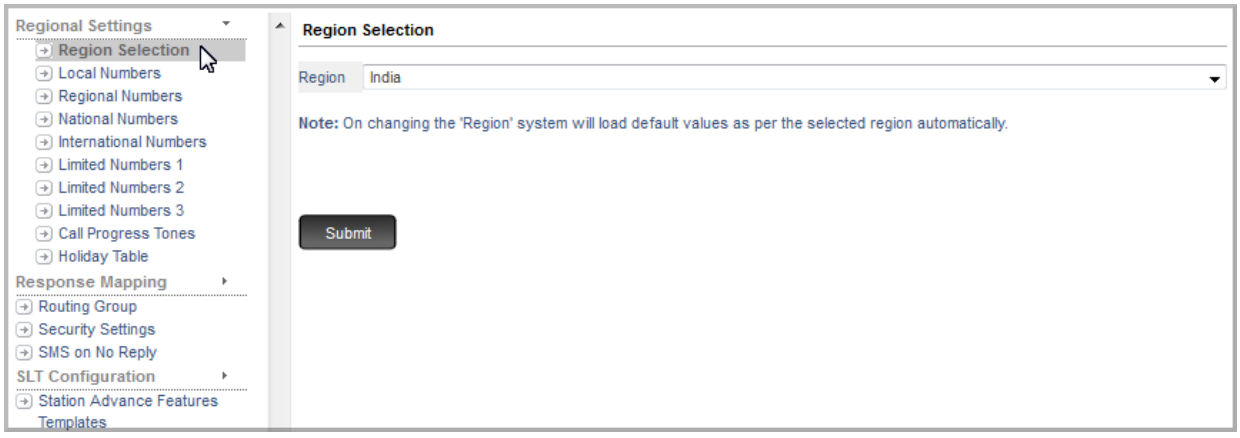
To load the country-specific Default Settings, users must select the Region that is the country in which the system is installed.

Certain countries are divided into various regions. If you select only a different region in the same country the DST and Date and Time Settings only will be change as per the selected region. The other parameters are country-specific.

India is selected as the default Region. So, if you are installing SARVAM UCS in a country other than India, change the Region.

Changing Region using Jeeves

- Log into Jeeves as System Engineer.
- Under **Configuration**, click **Regional Settings**.
- Click **Region Selection** to open the page.



- In the Region list, select the country where the system is installed.
- Click **Submit** at the bottom of the page to save changes.

Changing Region using a Telephone

- Enter SE mode from a DKP/SLT.

  To select Region, dial:

  - ***5301-Region Code***

    Where,

    Region Code is the code for the country in which SARVAM UCS is installed.

    Refer the following table and select the appropriate Region Code.

Region Code	Country Name
001	Afghanistan
002	Algeria
003	Antigua and Barbuda
004	Argentina
005	Australia (Perth)*
006	Australia (Adelaide)*
007	Australia (Brisbane, Canberra, Melbourne, Sydney)*
008	Austria
009	Bahamas
010	Bahrain
011	Bangladesh
012	Belarus
013	Belgium
014	Bhutan
015	Bolivia
016	Bosnia and Herzegovina
017	Botswana
018	Brunei
019	Brazil (Fernando De Noronha)*
020	Brazil (Brasilia, Rio de Janeiro, Sao Paulo)*
021	Brazil (Manaus)*
022	Brazil (Acre)*
023	Bulgaria
024	Cambodia
025	Cameroon
026	Canada (St. John's)*
027	Canada (Halifax)*
028	Canada (Montreal, Ottawa, Toronto)*
029	Canada (Winnipeg)*
030	Canada (Calgary)*
031	Canada (Vancouver)*
032	Chile
033	China
034	Colombia
035	Costa Rica
036	Croatia
037	Cuba
038	Cyprus

Region Code	Country Name
039	Czech Republic
040	Denmark
041	Egypt
042	Fiji
043	Finland
044	France
045	Germany
046	Greece
047	Guyana
048	Hong Kong
049	Hungary
050	India
051	Indonesia
052	Iran
053	Iraq
054	Ireland
055	Israel
056	Italy
057	Japan
058	Jordan
059	Kazakhstan
060	Kenya
061	Korea - North
062	Korea - South
063	Kuwait
064	Kyrgyzstan
065	Lebanon
066	Libya
067	Malaysia
068	Maldives
069	Mauritius
070	Mexico (Mexico City)*
071	Mexico (Chihuahua)*
072	Mexico (Tijuana)*
073	Mongolia
074	Mozambique
075	Myanmar
076	Namibia
077	Nepal
078	Netherlands
079	New Zealand
080	Nigeria
081	Norway
082	Oman
083	Pakistan
084	Paraguay
085	Peru

Region Code	Country Name
086	Philippines
087	Poland
088	Portugal
089	Qatar
090	Romania
091	Russia (Moscow, St. Petersburg)*
092	Russia (Novosibirsk)*
093	Russia (Vladivostok)*
094	Singapore
095	Slovakia
096	South Africa
097	Spain
098	Sri Lanka
099	Sudan
100	Sweden
101	Switzerland
102	Syria
103	Taiwan
104	Tajikistan
105	Thailand
106	Turkey
107	Uganda
108	Ukraine
109	United Arab Emirates
110	United Kingdom
111	United States (Atlanta, Augusta, Boston, Charlotte, Columbus, Detroit, Indiapolis, Miami, NY, Philadelphia, Washington)*
112	United States (Chicago, Dallas, Des Moines, Memphis, Minneapolis, New Orleans, Oklahoma, Omaha, St. Louis)*
113	United States (Albuquerque, Boise, Cheyenne, Denver, Salt Lake City)*
114	United States (Las Vegas, Los Angeles, Phoenix, San Francisco, Seattle)*
115	United States (Juneau)*
116	United States (Hawaii)*
117	Uzbekistan
118	Venezuela
119	Vietnam
120	Yemen
121	Yugoslavia
122	Zambia
123	Zimbabwe
124	Saudi Arabia
125	Cote d'Ivoire

** These countries are divided into several regions hence only Date and Time Settings and DST will change as per the region you select. Other parameters will be country-specific.*

By default, the region code is 050 (India).

For example, if SARVAM UCS is installed in the Philippines, dial ***5301-086***

- Exit SE mode.

Configuring Network Parameters

SARVAM UCS has a LAN and WAN Port.The network parameters must be configured according to your network scenario.

SARVAM UCS may be installed typically, in a Public IP Network or in a Private network, behind a NAT Router.

When the SARVAM UCS is installed in a Public IP Network,

- the WAN Port of the card is connected to a Broadband Router/Modem.
- Public IP is assigned to the WAN Port.
- the LAN Port of the card is connected to a LAN Switch/Hub to which SIP devices are connected.

When the SARVAM UCS is installed in a Private Network, behind a NAT Router,

- the WAN Port of the card is connected to the LAN Switch/Hub.
- Private IP is assigned to the WAN Port.
- SIP devices within the LAN can get registered with the card.

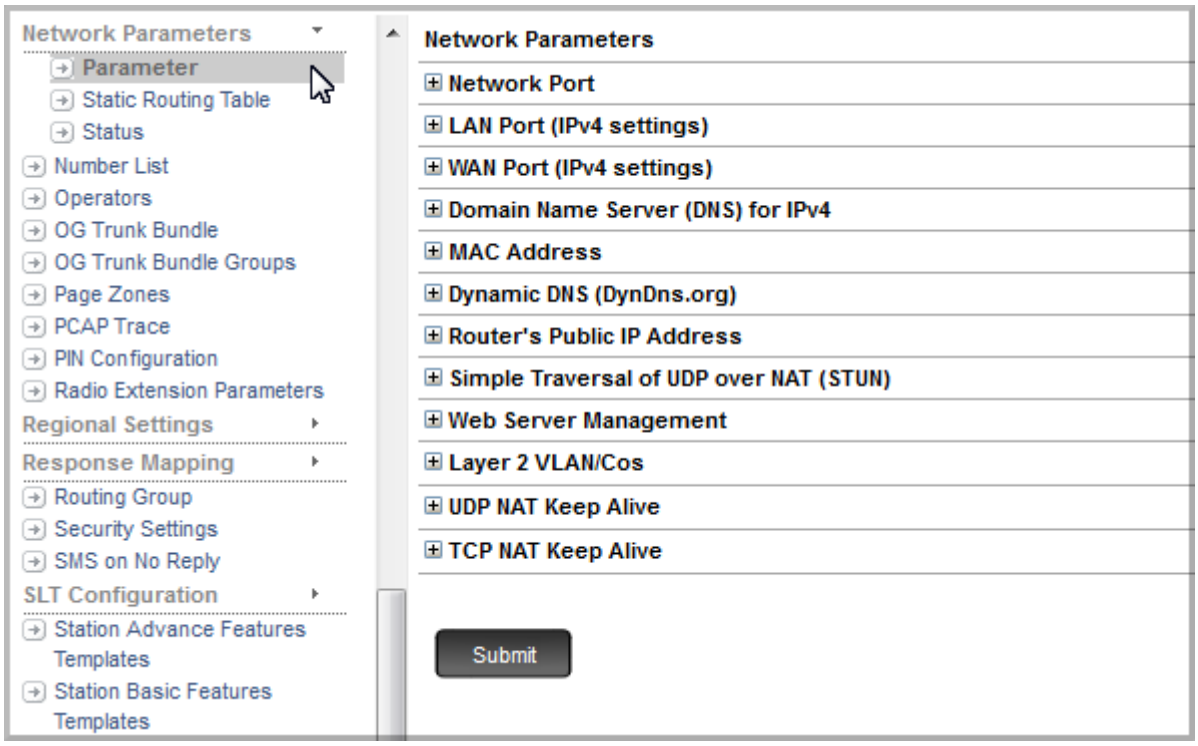
When your system is installed in a Private Network, you may have to change the IP Address and Subnet Mask of the WAN Port, before connecting it to the LAN Switch/Hub. However, this will not be necessary, if there is a DHCP server on the LAN which will automatically assign an IP Address that does not conflict with any other device on the LAN.

If you need to change the IP Address and the Subnet Mask of the LAN Port and the WAN Port of SARVAM UCS, you may do so by dialing System Commands from the telephone connected to the SLT Port. It is also possible to view the current IP Address and Subnet Mask of the LAN Port and the WAN Port by dialing System Commands from the telephone connected to the SLT Port. For instructions, see "System Commands".

Depending on your installation scenario, configure the Network Parameters using Jeeves or dialing commands from a Telephone.

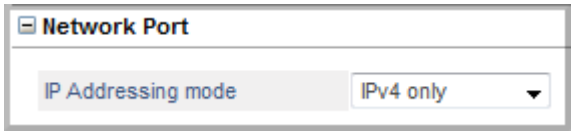
Configuring Network Parameters using Jeeves

- Login as System Engineer.
- Under **Configuration**, click **Network Parameters**.
- Click **Parameters** to open the page.



Network Port

Click **Network Port** to expand.



- **IP Addressing mode:** Select the IP version you want the system to use. You may select — IPv4 only or IPv4 and IPv6. Default: IPv4 only.

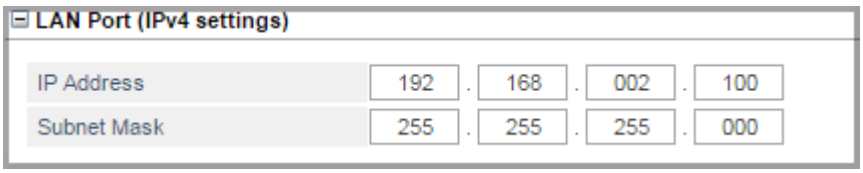
  If you select IPv4 only, you can configure the IPv4 parameters only.

  If you select IPv4 and IPv6, you can configure both IPv4 and IPv6 parameters.

- **Preferred DNS Server:** If you select IPv4 and IPv6 as the IP Addressing mode, you must select the Preferred DNS Server — IPv4 or IPv6. Default: IPv4

LAN Port (IPv4 settings)

Click **LAN Port (IPv4 settings)** to expand.

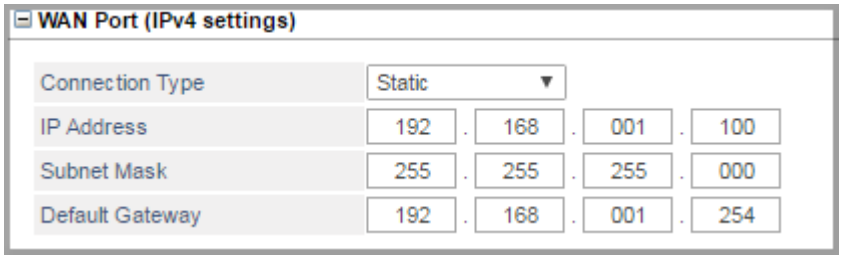


- **IP Address:** Enter the IP Address to be assigned to the LAN Port. The default IP Address is 192.168.002.100. You can assign only Static IP to the LAN Port.

- **Subnet Mask:** Enter the Subnet Mask to be assigned to the LAN Port. The default Subnet Mask is 255.255.255.0

WAN Port (IPv4 settings)

Click **WAN Port (IPv4 settings)** to expand.



- **Connection Type:** Select the appropriate Connection Type for the WAN port, according to the IP Addressing scheme of your installation scenario. Consult your System Administrator also in this regard. Default: Static.

  - **Static:** Select this option if the connection type is Static. When you select this option, you must:
    - assign an IP Address to the WAN Port.
    - change the Subnet mask of the WAN Port as appropriate.
    - configure the Router's LAN Interface IP Address as the Gateway IP Address.
    - Configure the DNS Address/Domain Name provided by your ISP or ask your LAN Administrator for the DNS Address and Domain Name.

  - **DHCP:** Select this option if the connection type DHCP. As the DHCP Server will automatically assign IP Address, Subnet Mask, Gateway Address to the WAN Port, you need not configure any of these.

  - **PPPoE:** Select this option if the connection type is PPPoE. As the PPPoE server will automatically assign the IP Address, Subnet Mask and Gateway Address to the WAN Port, you need not change any of these. You must program the User ID, Password and PPPoE Service Name as provided by your ISP. Program the Service Name only if it has been provided. You must set DNS address.

    Configure the following PPPoE parameters:

    - **User ID:** Enter the User ID provided by the Internet Service Provider. The User ID may be a maximum of 64 characters.

    - **Password:** Enter the User Password provided by the Internet Service Provider. The password may be a maximum of 64 characters.

    - **Service Name:** Enter the PPPoE Service Name, if provided by your Internet Service Provider. The Service Name may consist of a maximum of 64 characters. If Service Name is not required, leave this field blank.

- **IP Address:** If you have selected 'Static' as the Connection Type, the default IP is 192.168.001.100.

If you configure the PPPoE User ID, Password and Service Name using commands, you can configure a maximum of 16 characters only.

If you have selected DHCP or PPPoE as the Connection Type, the IP Address will be assigned by the DHCP/PPPoE server.

- **Subnet Mask:** You must enter the Subnet Mask, only if you have selected 'Static' as the Connection Type.

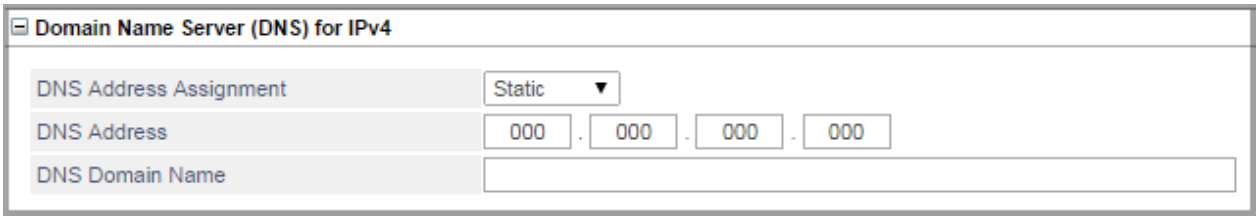
  If you have selected DHCP/PPPoE as the Connection Type, the Subnet Mask will be assigned by the DHCP/PPPoE server.

- **Default Gateway:** You must enter the Gateway IP Address, only if you have selected 'Static' as the Connection Type.

  If you have selected DHCP/PPPoE as the Connection Type, the Gateway IP Address will be assigned by the DHCP/PPPoE server.

Domain Name Server (DNS) for IPv4

Click **Domain Name Server (DNS) for IPv4** to expand.



- Configure the following DNS Connection settings for the WAN Port:

  - **DNS Address Assignment:** If you have selected 'Static' as your network Connection Type (IP Addressing), you can select only 'Static' as the DNS Address Assignment.

    If you have selected DHCP as your network Connection Type, and the DHCP server provides DNS Address, set the DNS Address Assignment to 'Auto'. If the DHCP server does not provide DNS Address, set DNS Address Assignment as 'Static' and configure the DNS Server Address provided by your ISP.

    If you have selected PPPoE as your network Connection Type, and the PPPoE server provides DNS Address, set the DNS Address Assignment as 'Auto'. If the PPPoE server does not provide DNS Address, set the DNS Address Assignment as 'Static' and configure the DNS Server Address provided by your ISP.

  - **DNS Address:** This field will be editable only if you have selected DNS Address Assignment as 'Static'. Enter the DNS Address here.

    If you have selected DNS Address Assignment as 'Auto', the DNS Address will be assigned by the DHCP/PPPoE server.

  - **DNS Domain Name:** Configure the DNS Domain Name if provided by your ITSP/LAN Administrator. Otherwise, keep it blank. The Domain Name may be a maximum of 40 characters. Default: Blank.

LAN Port (IPv6 settings)

Click **LAN Port (IPv6 settings)** to expand.

- **IPv6 Addressing using:** You can select — Complete Address or Prefix. Default: Complete Address.

  If you select Complete Address,

  - Configure the **IPv6 Address** and the **Prefix Length**. The IP Address configured will be considered as the complete IPv6 address.

    For example: 2001:0:3238:DFE1:63::FEFB

    The Prefix Length is a decimal value that indicates how many of the high-order contiguous bits of the address comprise the prefix (the network portion of the address).

    Valid Range of the IPv6 Address is A to F, a to f, 0 to 9,:(colon). It can be a maximum of 39 characters. Default: Blank.

    The Prefix Length range is from 1 to 128 bits. Default: Blank.

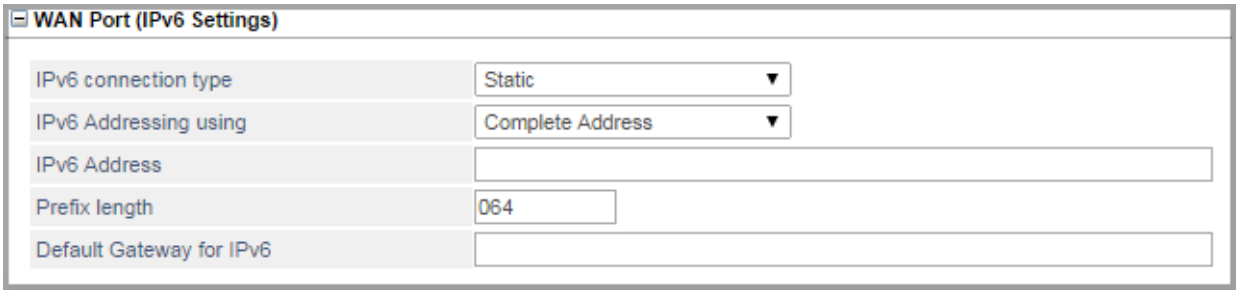
  If you select Prefix,

  - Configure the **IPv6 Prefix**. The system will consider the configured value as 64 bit Prefix of the IPv6 Address. Then the system will generate the complete IPv6 Address from it. Default: Blank.

    Valid characters 0 to 9, a to f, A to F and : (colon). It can be a maximum of 21 characters.

WAN Port (IPv6 Settings)

Click **WAN Port (IPv6 Settings)** to expand.



- **IPv6 Connection Type:** Select the appropriate Connection Type for the WAN port, according to the IP Addressing scheme of your installation scenario.

  You can select — Static, Statefull DHCPv6, Stateless Auto-Configuration, PPPoE. Default: Static.

  - **Static:** Select this option if the connection type is Static.

- **IPv6 Addressing using:** You can select — Complete Address or Prefix.

  If you select Complete Address,

  - Configure the **IPv6 Address** and the **Prefix Length**. The IP Address configured will be considered as the complete IPv6 Address.

    The Prefix Length is a decimal value that indicates how many of the high-order contiguous bits of the address comprise the prefix (the network portion of the address).

    Valid Range of the IPv6 Address is A to F, a to f, 0 to 9,:(colon). It can be a maximum of 39 characters. Default: Blank.

    The Prefix Length range is from 1 to 128 bits. Default: Blank.

  If you select Prefix,

  - Configure the **IPv6 Prefix**. The system will consider the configured value as 64 bit Prefix of the IPv6 Address. Then the system will generate the complete IPv6 Address from it. Default: Blank.

    Valid characters 0 to 9, a to f, A to F and : (colon). It can be a maximum of 21 characters.

- **Default Gateway:** Configure the Gateway IP Address for the WAN Port. It can be a maximum of 39 characters.

- **PPPoE:** Select this option if the connection type is PPPoE. As the PPPoE server will automatically assign the IP Address, Subnet Mask and Gateway Address to the WAN Port, you need not change any of these.

  Configure the following PPPoE parameters:

  - **IPv6 Scope Preference:** IPv6 includes support of Global as well as Non-Global Addresses(Unique). Select the scope of preference — Global or Unique. Default: Global.

  - **User ID:** Enter the User ID provided by the Internet Service Provider. The User ID may be a maximum of 64 characters.

  - **Password:** Enter the User Password provided by the Internet Service Provider. The password may be a maximum of 64 characters.

  - **Service Name:** Enter the PPPoE Service Name, if provided by your Internet Service Provider. The Service Name may consist of a maximum of 64 characters. If Service Name is not required, leave this blank.

  - **Prefix Length:** Configure the Prefix Length. Valid Range: 1 to 128 bits. Default: 064.

    The Prefix Length is a decimal value that indicates how many of the high-order contiguous bits of the address comprise the prefix (the network portion of the address).

    The Prefix Length range is from 1 to 128 bits. Default: Blank.

- **Statefull DHCPv6:** Select this option as the connection type, if your network uses DHCP to obtain various necessary parameters from DHCP Servers so the DHCP clients can operate in an Internet Protocol (IP)

network. Statefull DHCP is centrally managed on a DHCP server(s); and the DHCP clients use Statefull DHCP to obtain an IP address(es) and other useful configuration information from the DHCP server(s).

- **Prefix Length:** Configure the Prefix Length. Valid Range: 1 to 128 bits. Default: 064.

  The Prefix Length is a decimal value that indicates how many of the high-order contiguous bits of the address comprise the prefix (the network portion of the address).

  The Prefix Length range is from 1 to 128 bits. Default: Blank.

- **Stateless Auto-Configuration:** Select this option as the connection type, if your network uses DHCP to obtain various necessary parameters from DHCP Servers so the DHCP clients can operate in an Internet Protocol (IP) network. DHCPv6 for stateless configuration parameters allows a stateless or statefull DHCPv6 client to export configuration parameters (DHCPv6 options) to a local DHCPv6 server pool. The local DHCPv6 server can then provide the imported configuration parameters to other DHCPv6 clients.

  - **IPv6 Scope Preference:** IPv6 includes support of Global as well as Non-Global Addresses. Select the scope of preference — Global or Unique. Default: Global.

  - **Prefix Length:** Configure the Prefix Length. Valid Range: 1 to 128 bits. Default: 064.

    The Prefix Length is a decimal value that indicates how many of the high-order contiguous bits of the address comprise the prefix (the network portion of the address).

    The Prefix Length range is from 1 to 128 bits. Default: Blank.

Domain Name Server (DNS) for IPv6

Click **Domain Name Server (DNS) for IPv6** to expand.

- Configure the following DNS Connection settings for the WAN Port:

  - **DNS Address Assignment:** If you selected 'Static' as your IPv6 connection type, you can select only 'Static' as the DNS Address Assignment.

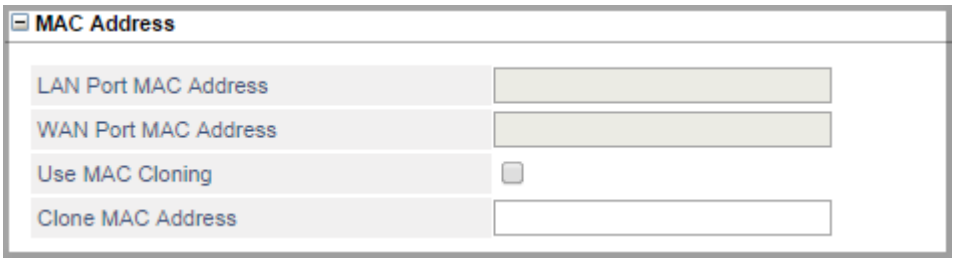
    If you have selected PPPoE as your IPv6 connection type, and the PPPoE server provides DNS Address, set the DNS Address Assignment as 'Auto'. If the PPPoE server does not provide DNS Address, set the DNS Address Assignment as 'Static' and configure the DNS Server Address provided by your ISP.

    If you have selected Statefull DHCP or Stateless Auto-Configuration as your network Connection Type, and the a server provides DNS Address, set the DNS Address Assignment as 'Auto'. If the server does not provide DNS Address, set the DNS Address Assignment as 'Static' and configure the DNS Server Address provided by your ISP

  - **DNS Address:** This field will be editable only if you have selected DNS Address Assignment as 'Static'. Enter the DNS Address here. The DNS Address can be a maximum of 39 characters.
    If you selected DNS Address Assignment as 'Auto', the DNS Address will be assigned by the Server.

MAC Address

Click **MAC Address** to expand.



- **LAN Port MAC Address:** This non-editable field displays the MAC Address of the LAN port.
- **WAN Port MAC Address:** This non-editable field displays the MAC Address of the WAN port.
- **Use MAC Cloning:** MAC Cloning is required when you want the WAN Port to use a MAC Address other than its own unique MAC Address as source MAC Address.

  When MAC Address Cloning is disabled, the WAN Port will use its unique MAC Address as the source MAC Address on all Ethernet Frames. When MAC Cloning is enabled, the WAN Port will use the cloned MAC Address on all Ethernet frames.

  Select the check box to enable cloning of the MAC Address of the WAN Port. Default: Disabled.
- **Clone MAC Address:** If you have enabled MAC Cloning, enter the MAC Address to be cloned here.

Dynamic DNS (DynDns.org)

Click **Dynamic DNS (DynDns.org)** to expand.

- **Dynamic DNS (DynDNS.org):** This parameter is applicable only when you are going to configure the SIP Extensions.

  When the WAN port is assigned dynamic IP Address using DHCP or PPPoE, SIP-enabled devices registered with the system as SIP Extensions need to change their configuration whenever a new IP Address is assigned to the WAN port. Dynamic DNS resolves this.

  SARVAM UCS supports Dynamic DNS Server client of the Service Provider Dynamic DNS.org.

  If you want to use the DNS Service of DynDNS.org, configure these parameters:

- **Enable Dynamic DNS:** If you have taken the services of DynDNS.org, you must enable this check box. Default: Disabled.

- **Update IP Address at Power ON:** When your WAN port is registered with the DynDNS.org, the Dynamic DNS server stores the mapping between hostname and IP Address, which can be updated periodically. However, if the system frequently sends IP Address update request to the DDNS server, the server is likely to block the hostname in its database and terminate the DDNS services provided to you.

  So, if you restart the SARVAM UCS frequently, there is a great chance that DDNS server will block the hostname configured in the system. This will in-turn affect the ability of the system to receive the calls using DDNS host name since the entry (mapping between host name and IP Address) in the DNS server will be deleted during those scenarios.

  The system allows you a control over whether the system should update the IP Address in the DDNS server at each Power ON or not. It also allows you to update the IP Address at any time, as required.

  If you want to update the IP Address in the DDNS Server at each Power ON, select the check box. Default: Disabled.

- **User ID:** Enter the User ID created by you with DynDNS.org here. A maximum of 40 characters, including all ASCII characters are allowed. Default: Blank.

- **Password:** Enter the Password created by you for your User ID with DynDNS.org here. The password may be not more than 24 characters long. Default: Blank.

- **Host Name:** Enter the Host Name created by you with DynDNS.org here. A maximum of 40 characters. All ASCII characters except < > and “ (double quote) are allowed.Default: Blank.

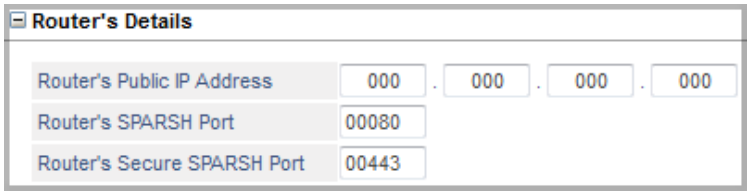
- **Retry Trials:** This count defines the number of attempts that the system should make to send the IP Address Update Request to the Dynamic DNS Server. The Retry Count may be set from 1 to 9. By default the count is set to 1. Default: 5.

- **Update IP Address Now?:** Click the **Click to Update IP Address** button**,** if you want to update the IP Address in the DDNS server at any point of time. Default: Disabled.

  This option is useful if you have not enabled the option Update IP Address during each Power ON. You can update the IP Address in the DDNS server whenever required.

Router's Public IP Address

Click **Router's Public IP Address** to expand.



- **Router's Public IP Address:** This parameter is of relevance if the WAN port of the system is located behind a NAT Router and SIP Messages are to be forwarded to the public internet.

Router's Public IP Address specifies the fixed IP Address of your NAT router required for NAT Traversal in SIP messages.

- **Router's SPARSH Port:** Enter the Router's Port mapped with the SPARSH Port of the system. This allows auto configuration of the external VARTA clients (VARTA ADR100/ VARTA AMP100) using the Auto Sign-In Email. If you want the VARTA clients to auto configure with the system, you must enter the Router's SPARSH port value as the Server Port along with the Server Address in the VARTA ADR100/ VARTA AMP100.

  Valid range: 80 or any value ranging from 1025 to 65535. Default: 80.

- **Router's Secure SPARSH Port:** Enter the Router's Port mapped with the SPARSH Secure Port of the system. This allows auto configuration of the external VARTA clients (VARTA ADR100/ VARTA AMP100) using the Auto Sign-In Email. If you want the VARTA clients to auto configure with the system using a secure protocol, you must enter the Router's Secure SPARSH port value as the Server Port along with the Server Address in the VARTA ADR100/ VARTA AMP100.

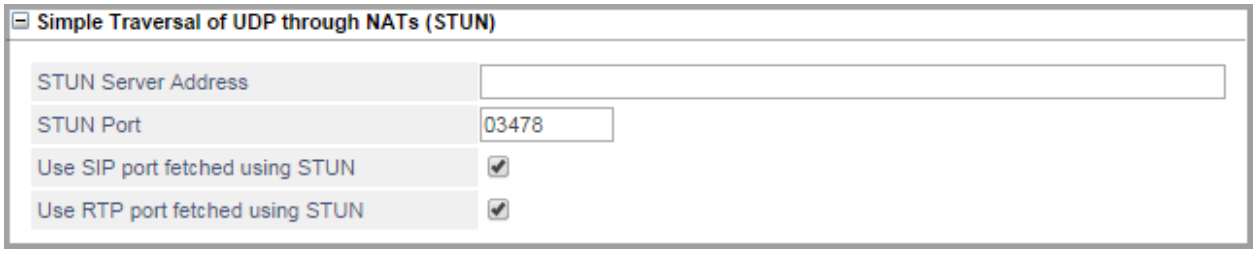
  Valid range: 443 or any value ranging from 1025 to 65535. Default: 443.

Make sure you select the Use Router/STUN's IP Address option as the Registrar Server Address, when you configure SIP Extensions.

You can also use STUN as an alternative to the Router's Public IP Address as NAT Traversal mechanism. Ask your Network Administrator about the NAT Traversal mechanism that suits best for your voice network and program this parameter.

Simple Traversal of UDP through NATs (STUN)

Click **Simple Traversal of UDP through NATs (STUN)** to expand.



- **Simple Traversal of UDPs through NATs (STUN):**This parameter is to be configured only if the WAN port is located behind a NAT Router and SIP Messages need to be forwarded to the public internet.

  Simple Traversal of UDP through NAT (STUN) specifies the mechanism required for NAT traversal in SIP messages. The STUN Server facilitates traversing through most NATs, except symmetric NATs. If your router has symmetric NAT, do not program this parameter. If your router as asymmetric NAT, configure the following STUN parameters:

  - **STUN Server Address:** Enter the STUN Server Address, a maximum of 40 characters. Default: Blank.

  - **STUN Port:** Enter the Listening Port of the STUN Server. The valid range for this field is from 1025-65535. The default STUN Port is 03478.

- **Use SIP Port fetched using STUN:** By default, this check box is selected (enabled), to allow SIP Port Number to be fetched using STUN in the SIP message. Clear this check box, if you are using Port-Forwarding in the Router for SIP messages.

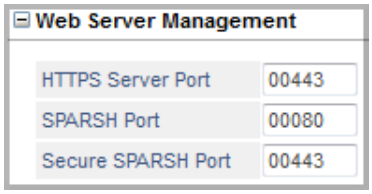
- **Use RTP Port fetched using STUN:** By default, this check box is selected (enabled), to allow RTP Port Number to be fetched using STUN in the SIP message. Clear this check box, if you are using Port-Forwarding in the Router for SIP messages.

You also need to select the 'Use IP Address fetched using STUN'' check box in Network Parameters and select the same as the Registrar Server Address, when you configure SIP Extensions.

Since STUN does not work with symmetric NAT, as an alternative to STUN you can use the Router's Public IP Address as NAT Traversal mechanism. Ask your Network Administrator about the NAT Traversal mechanism that suits best for your voice network and program this parameter.

Web Server Management

Click **Web Server Management** to expand.



- **HTTPS Server Port**: Enter the HTTPS Port number. Valid range: 443 or any value ranging from 1025 to 65535. Default: 443.

- **SPARSH Port:** Enter the SPARSH port number. The system will listen for the configuration request of the Extended IP Phones/ VARTA clients/ Standard SIP Phones on this port. If you want any Extended IP Phones/ VARTA clients/ Standard SIP IP Phones to auto configure with the system, you must configure the SPARSH port value as the Server Port along with the Server Address in the Extended IP Phones/ VARTA clients/ Standard SIP IP Phones.

  Valid range: 80 or any value ranging from 1025 to 65535. Default: 80.

When the system is set to default, the SPARSH Port value will not be set to default.

- **Secure SPARSH Port:** Enter the Secure SPARSH port number if you want to auto configure the VARTA AMP100/VARTA ADR100/ VARTA WIN200 with the server using a secure protocol. The system will listen for the configuration request from the VARTA clients on this port. You must also configure the Secure SPARSH port value as the Server Port along with the Server Address in the VARTA clients.

  Valid range: 443 or any value ranging from 1025 to 65535. Default: 443.

- *If you want the SARVAM UCS and VARTA AMP100/ VARTA ADR100/ VARTA WIN200 to communicate using secure protocol, make sure you have enabled the option Secure Connection with Server in the clients.*

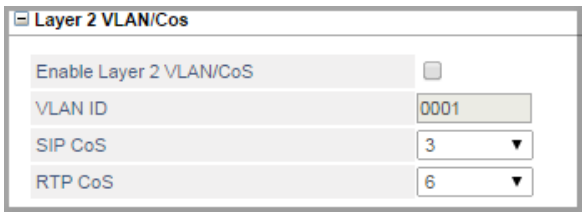
- *When the system is set to default, the port values of HTTPS Port, SPARSH Port and Secure SPARSH Port are not set to default.*

- *We recommend you to connect SARVAM UCS behind a router/firewall to avoid attacks such as ping flood, DoS etc.*

- *If SARVAM UCS is connected on the public internet, the system is prone to attacks, hence to ensure system security we recommend you to change the default ports.*

Layer 2 VLAN/Cos

Click **Layer 2 VLAN/Cos** to expand.



- **Layer 2 VLAN/CoS:** This parameter is to be configured if the WAN port is to be connected in VLAN network.

  This parameter enables the SARVAM UCS to add VLAN header to the packets generated by it. The VLAN header consists of the VLAN ID (12-bit) and Class of Service (CoS, 3-bit) for prioritization of traffic[128].

  VLAN Tag is applied on all packets generated by system (SIP, RTP, DNS, ARP, etc.), whereas CoS bits are applied only for SIP and RTP packets generated by system.
  The corresponding meaning of CoS bits with respect to traffic type is as follows:

COS	Traffic Type
0	Best Effort
1	Background
2	Spare
3	Excellent Effort
4	Controlled Load
5	Video
6	Voice
7	Network Control

- **Enable Layer 2 VLAN/CoS:** Select this check box, if you want all packets generated by the system (SIP, RTP, DNS, ARP, etc.) to be tagged with VLAN ID as configured. The CoS bits as configured for SIP and RTP packets will be included in the VLAN header. Default: Disabled.

- **VLAN ID:** Consult your network administrator and configure the VLAN ID. The valid range for this is from 0 - 4094. Default: 1.

---

128. The IEEE 802.1P standard allows Layer2 switches to prioritize the traffic, thus providing Quality of Service (QoS), better handling of data that pass over a network, thereby resulting in greater reliability and quality. Quality of Service (QoS) on Layer2 is referred to as Class of Service (CoS) which is defined by IEEE 802.1P.

- **SIP CoS:** Define the CoS (priority) bits in all SIP packets. The range of CoS bits is from 0 to 7. Default: 3.

- **RTP CoS:** Define the CoS (priority) bits in all RTP packets. The range of CoS bits is from 0 to 7. Default: 6.

UDP NAT Keep Alive

Click **UDP NAT Keep Alive** to expand.

- **UDP NAT Keep Alive:** This parameter is to be configured when the WAN port is connected behind a NAT router[129] and SIP messages are transported over UDP. UDP NAT Keep Alive messages must be sent to refresh the UDP binding in the NAT router.
  - **Enable UDP NAT Keep Alive:** Enable this flag to send UDP NAT Keep Alive messages periodically to refresh the binding in the NAT router. Default: Disabled.

  - **Interval (sec):** Select Time period after which the WAN Port should send UDP NAT Keep Alive messages. This time period should be less than the UDP Binding Timer of the router. The valid range is 001-999 seconds. Default: 180 seconds.
  - **Type of Message:** Select the type of message type to be sent when UDP NAT Keep Alive is enabled. Select either REGISTER or NOTIFY. Default: NOTIFY.

TCP NAT Keep Alive

Click **TCP NAT Keep Alive** to expand.

- **TCP NAT Keep Alive:** This parameter is to configured when the WAN Port is connected behind a NAT router and SIP messages are transported over TCP. TCP NAT Keep Alive messages must be sent to refresh the TCP binding in the NAT router.

  - **Enable TCP NAT Keep Alive:** Enable this flag to send TCP NAT Keep Alive messages periodically to refresh the binding in the NAT router. Default: Disabled.

  - **Interval:** Select Time period after which the WAN Port should send TCP NAT Keep Alive messages. This time period should be less than the Binding Timer of the router. The valid range is 0001-9999 seconds. Default: 120 seconds.

Viewing Network Parameters

You can view the status of WAN Port in Jeeves. To do this,

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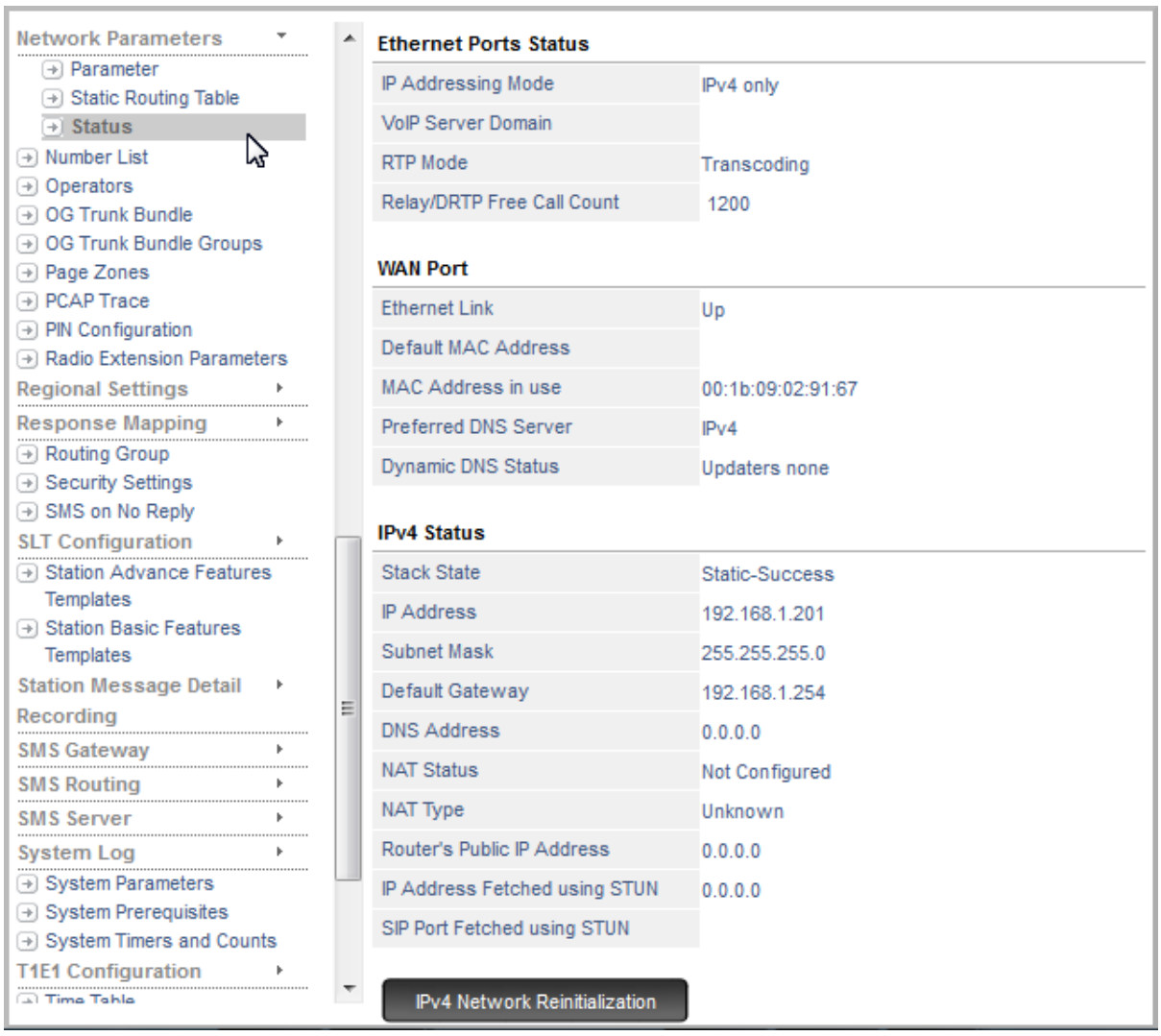
129. Network Address Traversal (NAT) allows multiple hosts in the network to share the single public routable IP address. Means all the hosts in the private network shall be identified by single public IP address in the global IP cloud.

- Login as System Engineer.
- Under **Configuration**, click **Network Parameters**.
- Click **Status** to open the page.



It displays the statuses of the various IPv6 and IPv4 parameters, Maintenance Port Parameters and Redundancy Configuration parameters.

When the system receives the IP Address from the DHCP/ PPPoE Server, the system performs DAD (Duplicate Address Detection). If DAD fails due to conflict in IP Address, the respective network parameters (WAN or LAN) need to be re-initialized.

To re-initialize the **WAN** parameters, click the **IPv4 Network Reinitialization** or **IPv6 Network Reinitialization** button under WAN. Similarly, to re-initialize the **LAN** parameters, click the **IPv4 Network Reinitialization** or **IPv6 Network Reinitialization** button.

Configuring WAN Port using a Telephone

- Enter SE mode from a DKP/SLT.

To select Connection Type, dial:

- ***2116-Network Connection Type***
  Where,
  Connection Type is
  1 for Static
  2 for DHCP
  3 for PPPoE
  Default: Static

To program PPPoE User ID, dial:

- ***2117 - PPPoE User ID***
  Where,
  PPPoE User ID may consist of a maximum of 16 ASCII characters.
  Default: Blank.

To program PPPoE Password, dial:

- ***2123- PPPoE Password***
  Where,
  PPPoE Password may consist of a maximum of 16 ASCII characters.
  Default: Blank.

To program PPPoE Service Name, dial:

- ***2124-PPPoE Service Name***
  Where,
  PPPoE Service Name may consist of a maximum of 16 ASCII characters.
  Default: Blank.

To assign IP Address to WAN Port, dial:

- ***2110-IP Address***
  Where,
  IP Address is 15 digits maximum. 000 to 255 for the first 3 Octets and 001 to 254 for the 4th Octet. Use zeros as fillers and dial the digits in a continuous sequence. Do not dial '.' in the IP Address.
  For example: to change the IP Address to 192.168.50.10 dial ***2110-192168050010.***

To assign Network Mask to the WAN Port, dial:

- ***2111-Network Mask***
  For example: to change the Subnet Mask to 255.255.255.0 dial ***2111-255255255000.***

To program the Gateway Address, dial:

- ***2112-Gateway Address***
  Where,
  Gateway Address is the Router's LAN Interface IP Address. Gateway IP Address may be a maximum of 15 digits max. Follow the same instructions as assigning IP Address to the WAN Port.

To select DNS Connection Type, dial:

- ***2115-DNS Connection Type***
  Where,
  DNS Address Assignment Type is
  1 for Static
  2 for Automatic
  Default: Static

To enable/disable Dynamic DNS, dial:

- ***2125-Code***
  Where,
  Code is
  0 for Disable
  1 for Enable
  Default: Disable.

To program Dynamic DNS User ID, dial:

- ***2126-DDNS User ID***
  Where,
  DDNS User ID may be a maximum of 40 ASCII characters.

To program Dynamic DNS User Password, dial:

- ***2127-DDNS User Password***
  Where,
  DynDNS User Password may consist of a maximum of 24 characters.

To program Dynamic DNS Host Name, dial:

- ***2128 - DDNS Host Name***
  Where,
  Host Name may consist of a maximum of 40 ASCII characters.

To program Dynamic DNS - Retry Trial Count, dial:

- ***2129 - DDNS Retry Count***
  Where,
  Retry Trial Count is from 1 to 9.
  Default: 5.

To Update Dynamic DNS IP Address binding (‘Update IP Address Now?’ flag), dial

- ***2130***

To program Router's Public IP Address, dial:

- ***2132-IP Address***
  Default: Blank.

To program STUN Server Address, dial:

- ***2133-STUN Server Address***.
  Where,
  STUN Server Address may be a maximum of 40 ASCII Characters.
  Default: Blank.

To program STUN Server Port, dial:

- ***2134- Port***
  Where,
  STUN Server Port range is from 1025 to 65535.
  Default: 3478.

To set STUN Query Interval, dial:

- ***2135 - Interval***
  Where,
  Interval is from 0001 to 9999 minutes.
  Default: 120 min.

To select IP Addressing Mode, dial:

- **2145 - IP Addressing Mode**
  Where,
  IP Addressing Mode is
  1 for IPv4 Only
  2 for IPv4 and IPv6
  3 for IPv6 Only

To select IPv6 Connection Type, dial:

- ***2146 - IPv6 Connection Type***
  Where,
  Connection Type is
  1 for Statefull DHCP
  2 for Stateless Auto-configuration
  3 for PPPoE
  4 for Static
  Default: Static

- Exit SE mode.

Viewing WAN Status from a Telephone

- Enter SE mode from a DKP (only).

  To view IP Address of the WAN Port, dial:

  - ***2150***

  To view Network Mask of the WAN Port, dial:

  - ***2151***

  To view Gateway Address of the WAN, dial:

  - ***2152***

  To view current STUN query status, dial:

  - ***2159***

  To view Router's Public IP Address fetched by STUN, dial:

  - ***2160***

  To view Dynamic DNS Status, dial:

  - ***2161***

  To view WAN link Status, dial:

  - ***2162***

  To view the configured IPv6 Address, dial:

  - **2164**

- Exit SE mode.

Configuring Maintenance Port Parameters

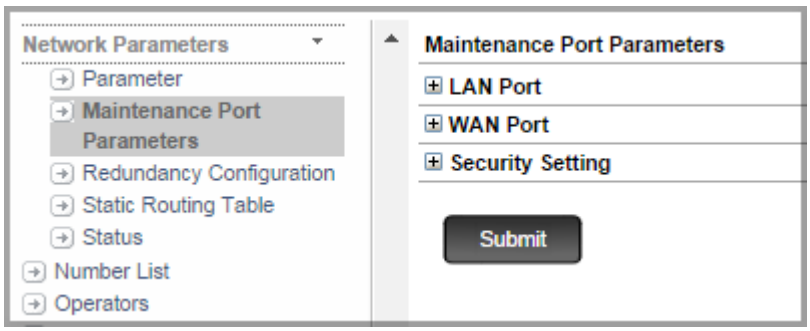
These Parameters are applicable only for ETERNITY LENX/MENX system.

You need to configure the Maintenance Port parameters to:

- transfer the configuration files from the Active card to the Redundant (Standby) card and vise-versa.
- access jeeves of the Redundant (Standby) card.
- restrict access to the facilities — Web Jeeves, Remote Login.

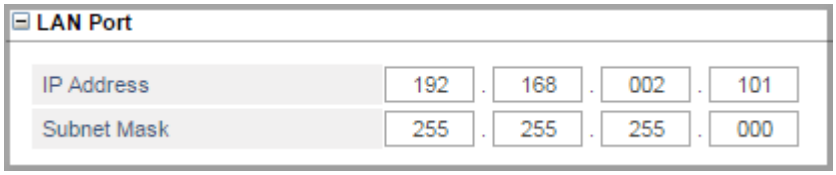
Configuring Maintenance Port Parameters using Jeeves

- Login as System Engineer.
- Under **Configuration**, click **Network Parameters**.
- Click **Maintenance Port Parameters** to open the page.



LAN Port

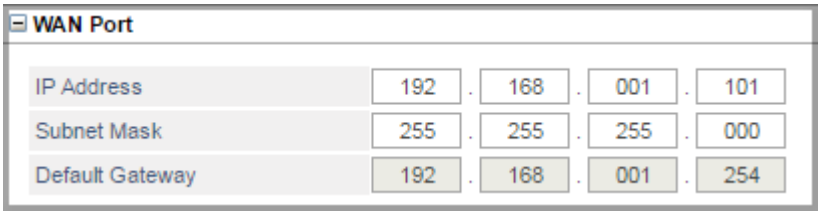
Click **LAN Port** to expand.



- **IP Address:** Enter the IP Address to be assigned to the LAN Port of the Active Card. The default IP Address is 192.168.002.101. You can assign only Static IP to the LAN Port.
- **Subnet Mask:** Enter the Subnet Mask to be assigned to the LAN Port of the Active Card. The default Subnet Mask is 255.255.255.0

WAN Port

Click **WAN Port** to expand.



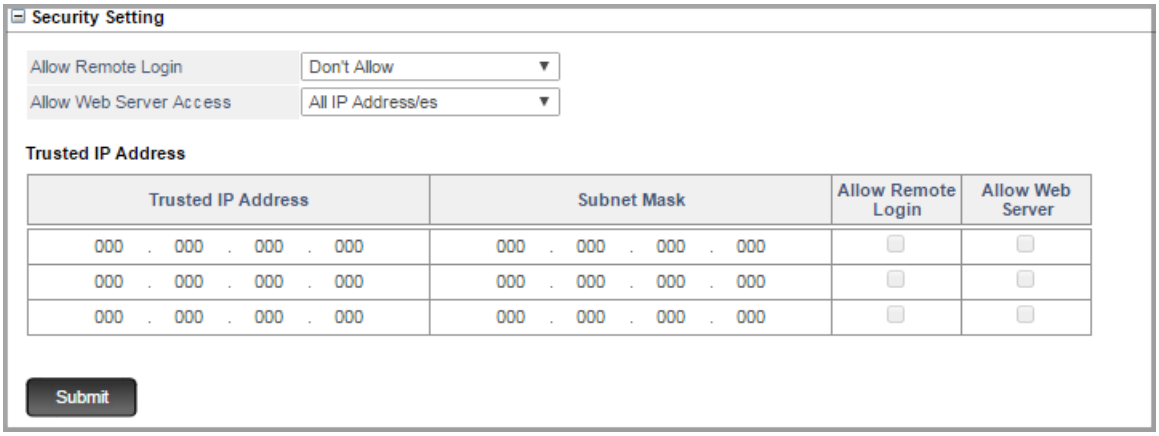
- **IP Address:** Enter the IP Address to be assigned to the WAN Port of the Active Card. The default IP Address is 192.168.001.101. You can assign only Static IP to the WAN Port.
- **Subnet Mask:** Enter the Subnet Mask to be assigned to the WAN Port of the Active Card. The default Subnet Mask is 255.255.255.0
- **Default Gateway:** It displays the Gateway IP Address you configured on the Network Parameters page. This is un-editable.

Make sure the LAN and WAN IP Addresses are in different Subnets and these do not conflict with the LAN and WAN IP addresses configured on the Network Parameters page.

Security Settings

You can restrict unauthorized access to the system using Remote Login or Web. To know more, refer to “Security Settings”.

Click **Security Settings** to expand.



- In **Allow Remote Login**, you can select Don’t Allow, All IP Address/es or Only Trusted IP Address/es option. By default, **Don’t Allow** is selected.

  If you want to allow access to generate the password for the Remote Login from all IP Addresses, select **All IP Address/es** option.

  If you want to allow access to generate the password for the Remote Login from specific IP Addresses only:
  - select **Only Trusted IP Address/es** option.

- configure the IP Address/es and their respective Subnet Mask in the **Trusted IP Address** table.
- enable the **Allow Remote Login** check box in the **Trusted IP Address** table.

- In **Allow Web Server** access, you can select Don’t Allow, All IP Address/es or Only Trusted IP Address/es option. By default, All IP Address/es is selected.

  If you do not want to allow access to the Web Server, select **Don’t Allow** option.

  If you want to allow access to the Web Server from specific IP Addresses only:
  - select **Only Trusted IP Address/es** option.
  - configure the IP Address/es and their respective Subnet Mask in the **Trusted IP Address** table.
  - enable the **Allow Web Server** check box in the **Trusted IP Address** table.

Click **Submit**.

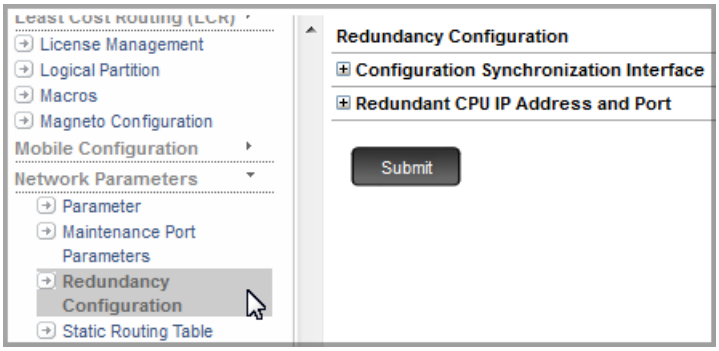
Configuring Redundancy Parameters

These Parameters are applicable only for ETERNITY LENX/MENX system.

The Redundancy parameters need to be configured to initiate the Redundancy process, that is, to transfer all configuration and call information from the CPU Card 1 (Active Card) to CPU Card 2 (Standby Card). To know about redundancy, refer to “Redundancy”.

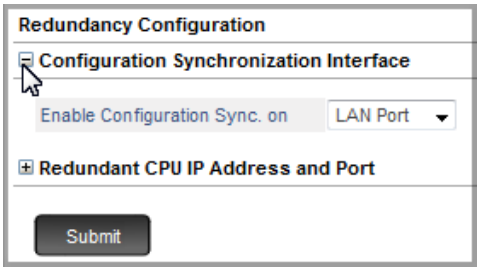
Configuring Redundancy Parameters using Jeeves

- Login as System Engineer.
- Under **Configuration**, click **Network Parameters**.
- Click **Redundancy Configuration** to open the page.



Configuration Synchronization Interface

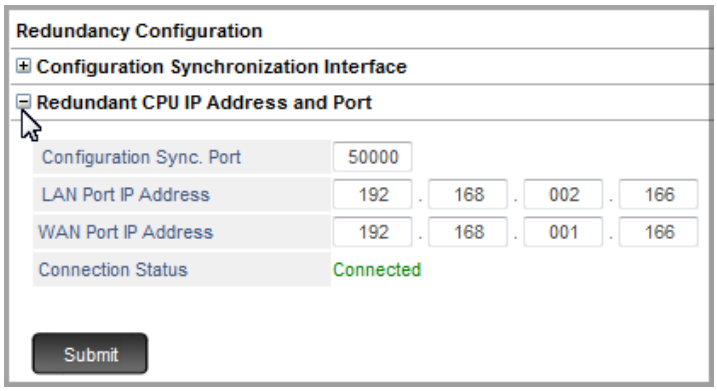
- Click **Configuration Synchronization Interface** to expand.



- In **Enable Configuration Sync. on**, you can select LAN Port or WAN Port. The connection between the CPU Card 1 (Active Card) and CPU Card 2 (Standby Card) will be established through this port. By default, LAN Port is selected.

Redundant CPU IP Address and Port

- Click **Redundant CPU IP Address and Port** to expand.



- In **Configuration Sync. Port,** enter the port number of CPU Card 2 (Standby Card). The CPU Card 1 (Active Card) will communicate with the CPU Card 2 (Standby Card) through this port.

Make sure the same port is configured in the CPU Card 2 (Standby Card).

- In **LAN Port IP Address**, enter the LAN Port IP Address of the CPU Card 2 (Standby Card). If you have selected LAN Port as the **Enable Configuration Sync. on** option, the CPU Card 1 (Active Card) will use this IP to communicate with the CPU Card 2 (Standby Card). The default IP Address is 192.168.002.101. You can assign only Static IP to the LAN Port.

- In **WAN Port IP Address**, enter the WAN IP Address of the CPU Card 2 (Standby Card). If you have selected WAN Port as the **Enable Configuration Sync. on** option, the CPU Card 1 (Active Card) will use this IP to communicate with the CPU Card 2 (Standby Card). The default IP Address is 192.168.001.101. You can assign only Static IP to the WAN Port.

- The **Connection Status** will display the connectivity status between CPU Card 1 (Active Card) and CPU Card 2 (Standby Card). It will display — Connected or Not Connected accordingly.

Click **Submit**.

Make sure you have configured these parameters for the other CPU Card also by accessing it's respective IP Address.

Configuring Extensions

The SARVAM UCS supports the following types of extension ports:

- **SIP Extensions:** Any SIP-enabled device like a Matrix VARTA UC Client, an IP-phone, a Softphone or a Wi-Fi mobile handset can be registered with the SARVAM UCS and function as the 'SIP Extension' of the SARVAM UCS.

  SIP Extensions function like any normal DKP/SLT extension of the SARVAM UCS, allowing you to make and receive calls to any extension user of the SARVAM UCS as well as any external numbers over PSTN, GSM, VoIP and E&M lines, depending on the "Logical Partition" configured in the System.

  SARVAM UCS supports 999 SIP Extensions. SIP Extensions are a licensed feature. To know more, refer the topic *"License Management"*.

- **DKP Extension Ports:** The proprietary Digital Key Phone of Matrix is connected to these ports. SARVAM UCS supports up to 96[130] DKP extensions. The number of DKP ports available to you depends on the number and configuration of the DKP Cards installed in the system.

- **ISDN Terminals:** These are ISDN phones connected to the BRI Ports of the SARVAM UCS. ISDN Terminals can be connected only in a Point-to-Multipoint BRI configuration (Short or Extended Passive Bus Configuration). Refer the topic *Installing BRI Card* under Installation instructions for your model of SARVAM UCS to know more.

  A maximum of 8 ISDN Terminals (phones) can be connected on a single BRI Bus line in a Point-to-Multipoint configuration. In a Short Passive Bus Configuration, you can connect up to 8 ISDN Terminals while in the Extended Passive Bus Configuration you can connect up to 3 ISDN Terminals.

  Depending on the number of BRI ports available to you and the type of Point-to-Multipoint Configuration (Short or Extended Passive Bus), a maximum of 64 ISDN Terminals can be connected to the SARVAM UCS.

- **SLT Extension Ports:** Single Line Telephones (SLT) is connected to these ports. SARVAM UCS supports up to 240[131] SLT extensions. The number of SLT ports available to you depends on the number and configuration of the SLT Cards installed in the system.

- **Radio Extensions/Ports:** Radio devices are connected to these ports and function as extensions of the SARVAM UCS. A maximum of 16 Radio Ports are supported.

- **Magneto Ports:** Magneto telephones are connected to these ports and function as extensions of the SARVAM UCS. A maximum of 16 Magneto Ports are supported.

- **E&M Ports functioning as Stations:** An E&M port of SARVAM UCS can be programmed to take on the function of a Subscriber (Station), to work like an extension interface, receiving incoming calls.

  Presuming that you have connected the extensions successfully, you may now configure the extensions using Jeeves or a Telephone.

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130. *The maximum number of DKP ports supported in ETERNITY MENX may vary according to the type of Power Supply (whether PSUNI or DC) being used. See Technical Specifications provided in the Appendix.*
131. *The maximum number of concurrent off-hook SLT ports supported may vary according to the type of Power Supply (whether PSUNI or DC) being used. See Technical Specifications provided in the Appendix.*

- **Virtual Extensions**. For more information see “Virtual Extension”.

Templates for Configuring Extensions

To make the task of configuring of extensions easy, the SARVAM UCS offers Templates, namely:

- SIP Hardware Template - for SIP Extensions (and SIP Trunks) only. See “SIP Hardware Template” under *Configuring Trunks*.
- SLT Hardware Template - for SLT extensions.
- Station Basic Feature Template - for DKP, SLT, SIP Extensions, ISDN Terminals, and E&M Ports functioning as Stations, Magneto ports, Radio ports and Virtual Extensions.
- Station Advanced Feature Template - for DKP, SLT, SIP Extensions, ISDN Terminals, and E&M Ports functioning as Stations, Radio ports, Magneto ports, and Virtual Extensions.

You can use these templates to program extensions which are to be assigned the same set of features at one go, saving you the effort for painstaking configuration of each extension.

The features in these templates are loaded with default values that fulfill the requirements of a very broad user base. The Templates may be customized as per user requirements and applied to the extensions.

Before you start the configuration of the extensions, please read the description of the templates and how to customize the templates according to user requirements.

SLT Hardware Template

An SLT Hardware Template is a set of features that define the behavior of the SLT hardware port, and needless to say, it is assigned to SLT Ports only.

The SLT Hardware Template allows you to configure according to user requirements, a common set parameters (features) like Caller ID Presentation (DTMF, FSK), Digit Pad Count, Ring Type, AC Impedance, Answer Signaling type, etc. to be assigned to all SLT Hardware Ports.

Each of these hardware parameters are briefly described below.

SLT Hardware Template Parameters

1. **CLIP Type:** The SARVAM UCS provides a facility to detect the calling number and present it to the SLT. This is known as Calling Line Identification and Presentation (CLIP). For this feature to work, the telephone instrument connected to the SLT port must support CLIP.

   The SARVAM UCS supports 3 signaling protocols for CLI on the SLT port: DTMF, FSK-V.23, and FSK-BellCore. Select the appropriate signaling protocol.

   If you want to disable CLI on the SLT port, select 'None'.

   By default, DTMF protocol is set as CLIP Type.

2. **Digit Pad Count:** Certain SLT instruments that support CLI require a minimum number of digits in the calling party's number to be able identify and display it. The Digit Pad Count signifies the number of zeroes

to be added with the Calling party's number before displaying it on the called party's instrument. This count entirely depends on the instrument connected to it. By default, Digit Pad Count is set as 0.

3. **Ring Type:** The SLIC used with SLT port allows you to change the Ring type: Sinusoidal, Trapezoidal, Low Sinusoidal, Low Trapezoidal. This is helpful in cases when telephone instruments, which expect sinusoidal type of ringing current, are connected to the SLT port. By default the Ring Type for all SLT ports is Trapezoidal.

4. **SLT Gain Settings Template:** You can increase or decrease the level of Incoming Speech (Receive Gain) and Outgoing Speech (Transmit Gain) on the SLT port by changing the Rx Gain and Tx Gain to the desired level. Different levels can be set for each port type in the SLT Gain Settings Template. By default, SLT Gain Template 1 is assigned to all the SLT Hardware Templates. If you want to assign a different Template, you must customize the SLT Gain Settings Template first and then assign the number of the SLT Gain Settings Template in this Template. To customize the SLT Gain Settings, see “Gain Settings”.

5. **AC Impedance:** The SLIC used with each SLT port provides a facility to adjust the AC impedance of the SLT port with the communication equipment connected to it.

   Generally, most telephone instruments that are connected have nominal characteristics with AC impedance of 600Ω. However, the SARVAM UCS allows you to connect instruments with AC impedance other than 600Ω.

6. **Flash Timer (msec):** In Pulse Dialing, codes are dialed in pulses. A Flash key is generally used to dial this code. Flash is breaking the loop current for 70ms to 900ms. Flash Timer defines the time period which should be considered as Flash, if the loop current breaks.

   The range of the timer is from 70 to 900 msecs. By default, the Flash Timer is set to 101-600 msec. Program the Flash Timer as per user requirement.

If the Flash Timer range is configured as 70-100 msec, Pulse dialing for that particular SLT Phone will not work.

7. **Answer Signaling:** An Answer Signal is a signal generated on the SLT port to indicate that the called party (remote party) has answered the call and the call is now mature.

   Answer Signaling on the SLT port is particularly useful when there is a PCO machine or any Billing equipment connected to the SLT port. With Answer Signaling enabled on an SLT port, during an outgoing call is made from that SLT port to any other port - CO/Mobile/SIP/T1E1/BRI - when the called party (remote party) answers, the Public Network provides an Answer Signal to the trunk port to indicate call maturity.

   This information can be passed on to the PCO machine billing equipment in the form of Answer Signaling. On detecting Answer Signaling the PCO machine billing equipment can start billing.

   Answer Signaling is generated in the form of **Polarity Reversal** or **Battery Reversal**, whereby the Battery polarity of the SLT port gets reversed. For example, if the battery polarity of the SLT port is +ve for TIP and -ve for RING in speech condition, then on call maturity, TIP becomes -ve and Ring becomes +ve.

   To generate Answer Signaling on the SLT Port, select Polarity Reversal. Select **None** if Answer Signaling is not be generated on the SLT port.

   By default None is selected.

8. **Disconnect Signaling:** A Disconnect Signal is the signal generated on the SLT port to indicate that the called party (remote party has disconnected the call.

   Disconnect Signaling on the SLT port is useful when there is a PCO machine or any Billing equipment connected to the SLT port. With Disconnect Signaling enabled on an SLT port, during an outgoing call is made from that SLT port to any other port - CO/Mobile/SIP/T1E1/BRI - when the called party (remote party) disconnects (goes ON Hook), the Public Network provides a Disconnect Signal to trunk port indicate call disconnection. This signal can be generated on the on the SLT port to indicate to the PCO machine/ Billing equipment connected to this port to consider the call as disconnected and stop billing. Thus, Disconnect Signaling on the SLT port helps prevent excessive billing.

   SARVAM UCS supports two types of Disconnect Signals on the SLT Port:

   - **Polarity Reversal:** Call Disconnection is signaled in the form of Polarity Reversal. The Battery polarity of the SLT port will be reversed. For example, if the battery polarity of the SLT port is '+ve' for TIP and '-ve' for RING in speech condition then on disconnection on other port, TIP will become '-ve' and Ring '+ve'. When call is disconnected, user will get Error tone.

   - **Open Loop:** Call Disconnection is signaled in the form of Open Loop Disconnect Pulse, whereby the Battery voltage on the SLT port is removed for the duration of the Open Loop Disconnect Timer programmed for that SLT port and will be restored on the expiry of this Timer. However, the Polarity of Battery Voltage on the SLT port is not changed. When call is disconnected, the SLT extension user gets an Error tone.

     To generate Disconnect Signaling on the SLT Port, select Polarity Reversal or Open Loop as appropriate. Select **None** if Disconnect Signaling is not be generated on the SLT port.

     By default None is selected.

9. **Open Loop Disconnect Timer (msec):** This parameter is applicable only if the option Open Loop Disconnect is selected as Disconnect Signaling type on the SLT port.

   Open Loop Disconnect Timer is the time period for which the system will remove Battery Voltage on the SLT port and restore Battery Voltage on the expiry of the Timer to signal Call Disconnection.

   The range of the timer is from 68 to 952 msecs. By default, the Timer is set to 478 msec.

10. **Loop Current (mA):** The SLT Port provides Loop Current to the telephone instrument connected to the SLT port to drive the telephone instrument.

    The Loop Current is to be increased/decreased according to the length of the telephony wiring cable between the wall jack (into which the SLT telephone instrument is plugged) and the MDF (into which the cables from the SLT port are terminated).

    The longer the Loop Length of the SLT port, the greater the likelihood of current dissipation, affecting speech quality of the telephone instrument connected to the SLT port.

    The system supports Loop Current of 25, 30, 35 and 40 mA. By default the Loop Current for SLT ports is set to 25mA and the SLT port, which is sufficient to support Loop Length of 1 kilometer.

    You may change the Loop Current according to the Loop Length of the SLT.

11. **Loop Length:** The Loop Length is the distance between the Central Office and the telephone instrument connected to the SLT port. Select the **Loop Length** as **Upto 5 Km** or **Above 5 Km** according to your installation scenario.

12. **Minimum Current for OFF-Hook Detection:** The system detects OFF-Hook state of an SLT instrument and gives dial tone on the basis of the current drawn by it from the SLT port. However, all types and brands of SLT instruments may not uniformly draw the same minimum current; some may draw lesser and some may draw more, making OFF-Hook detection difficult for the system. To resolve this, SARVAM UCS provides for programmable values for threshold current for OFF-Hook detection: 10mA, 12mA, 14mA, 16mA and 18mA.

    By default, the value of the Minimum Current for OFF-Hook detection is set to 12 mA. Change this value according to the current drawn by your SLT instrument.

    When an SLT instrument draws current equal to or greater than the programmed threshold value of current for off-hook detection, the system will consider the SLT instrument as OFF-Hook and will offer dial tone to the SLT.

13. **ON-Hook Detection Current (mA):** SARVAM UCS detects ON-Hook state of an SLT instrument to route calls on the basis of the current drawn by it from the SLT port. However, as all types and brands of SLT instruments may not uniformly draw the same current, ON-Hook detection becomes difficult for the system.

    To resolve this, SARVAM UCS provides for programmable values for threshold current for ON-Hook Detection: 10mA, 12mA, 14mA, 16mA and 18mA.

    By default, the value of the ON-Hook Detection Current is set to 10 mA.

    When an SLT instrument draws current equal to or lower than the programmed threshold value of current for ON-Hook detection, SARVAM UCS will consider the SLT instrument as ON-Hook.

    SLT instruments also vary in the level of current drawn during the normal 'idle' state and when Flash is dialed[132] (the simulated idle state). So, when the Flash key of an SLT instrument is pressed, and if the instrument draws a higher current than the threshold defined for the 'idle' state, the system will not be able to detect Flash (ON-Hook state).

    Consider this when changing the value of ON-Hook Detection Current. Define the value considering the current drawn by your SLT instrument in idle state, as well as when Flash key is pressed.

14. **Low Power Mode:** SARVAM UCS supports Low Power Mode option. That is, in On-Hook state the SLT instrument will consume less power. Select the Low Power Mode check box to enable this mode.

    SARVAM UCS also supports Thermal Shutdown. If the thermal temperature of the SLIC increases above the threshold level, the SLT instrument will stop functioning. This port number appears red in SLT Parameters. You must restart the card or the system to resume functioning of the phone.

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132. Dialing 'Flash' either with the 'Flash Key' or by pressing the Hook-switch causes the phone to go in ON-Hook state briefly for 600-800 milliseconds. Thus ON-Hook state is simulated briefly. The SLT may draw a higher current when 'Flash' is dialed.

Customizing SLT Hardware Templates

There are 50 SLT Hardware Templates that can be customized and assigned to the SLT ports. These templates contain the default values of the above-listed parameters.

The default parameter values of the SLT Hardware Templates are country specific and are loaded in each template according to the Country selected as the **Region**.

For example, when India is selected as the Region, the default value of the CLIP Type in the SLT Hardware Templates is DTMF, whereas it is FSK-Bellcore when the Region is selected as US or Canada and FSK-V.23 when UK is selected as Region. Similarly, the default values of AC Impedance on the SLT Hardware Templates will vary according to the Region selected; 600 ohms for Region India, 900Ω for Region the Philippines, and 350Ω +(1000Ω || 0.21µF) for Region UK.

By default SLT Hardware Template Number 01 is assigned to all the SLT ports. This template has default values fulfilling the common requirements of a very broad user base.

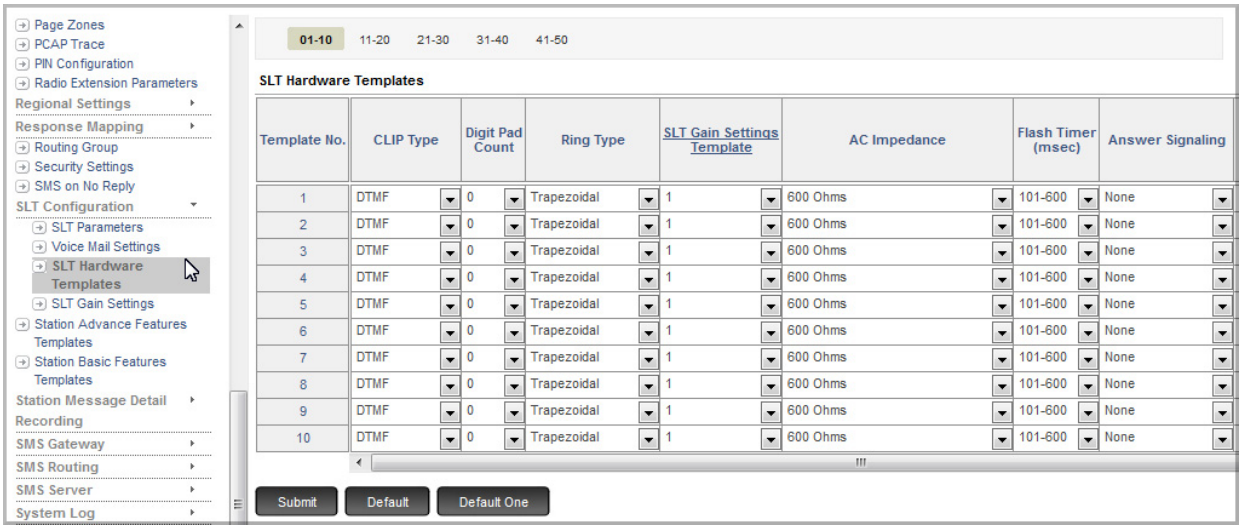
If the default SLT Hardware Template 01 fulfills the user and country requirements, retain Template 01.

If you want to change the values of certain SLT Hardware Parameters, but apply the same parameter values to all SLT ports, simply customize the desired parameters in Template 01.

However, if different hardware parameters are to be applied to different SLTs, then you can customize different the SLT Hardware Templates using Jeeves or a Telephone.

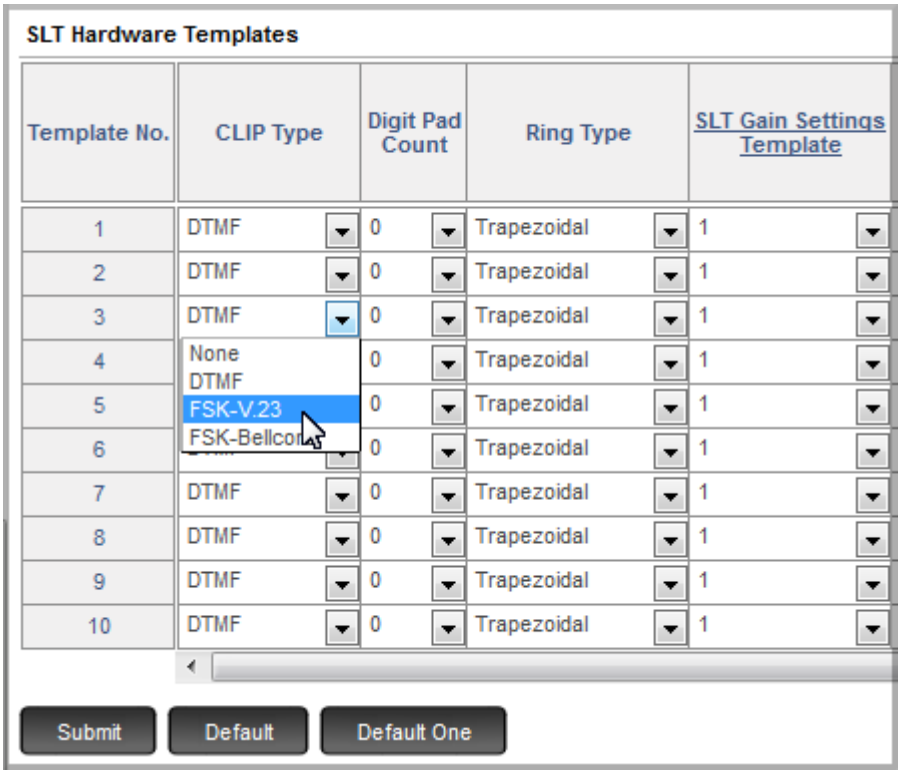
Customizing SLT Hardware Template using Jeeves

- Log into Jeeves as System Engineer.
- Under **Configuration**, click **SLT Configuration**.
- Click **SLT Hardware Template** to open the page.

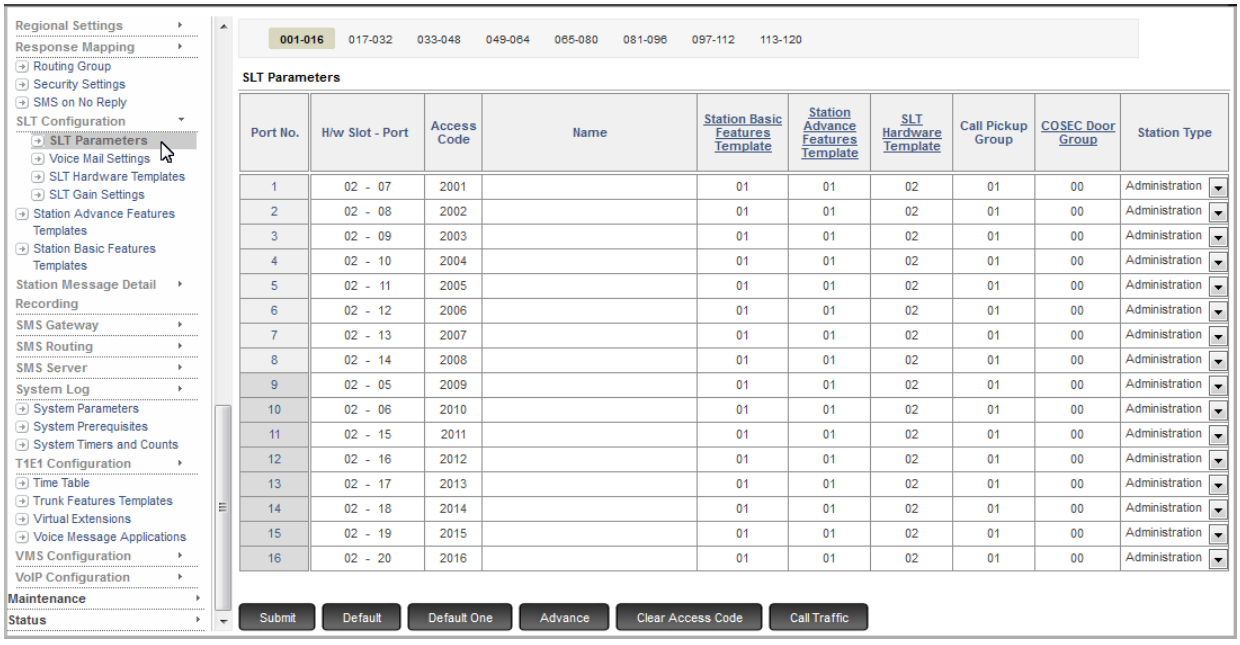


- Select a Template number you wish to customize, for example Template 03.

- Change the values of the SLT Hardware Template parameters as desired.



- Click **Submit** at the bottom of the page to save your changes.
- Now, apply this SLT Hardware Template 03 on the SLT ports. To do this,
- Click **SLT Parameters** to open the page.



- Go to the SLT software ports to which this Template is to be assigned, for example SLT-001, 002, and 003.

- Enter the number of the Template you customized, 02, in the field **SLT Hardware Template** of each of these SLT ports.

SLT Parameters

Port No.	H/w Slot - Port	Access Code	Name	Station Basic Features Template	Station Advance Features Template	SLT Hardware Template
1	02 - 07	2001		01	01	03
2	02 - 08	2002		01	01	03
3	02 - 09	2003		01	01	02
4	02 - 10	2004		01	01	02
5	02 - 11	2005		01	01	02
6	02 - 12	2006		01	01	02
7	02 - 13	2007		01	01	02
8	02 - 14	2008		01	01	02

- Click **Submit** at the bottom of the page to save your setting.
- Repeat the same steps to customize another template.

Customizing SLT Hardware Template using a Telephone

- Enter SE mode from a DKP/SLT.

  To change the default values of a SLT Hardware Parameter in a Template, dial:

  - ***5702-1-Template Number-Parameter Number-Code*** to program the value of a parameter in a single template.
  - ***5702-2-Template Number-Template Number-Parameter Number-Code*** to program the same value for the parameter in a range of templates.
  - ***5702-*-Parameter Number-Code*** to program the same value for the parameter in all templates.
    Where,
    Template Number is the number of the SLT Hardware Template from 01 to 50.
    Parameter Number is the number of the SLT Hardware Template Parameter from 01 to 13.
    Code is the value for each parameter from 0 to 14.
    Refer the table below for parameter numbers and meaning of codes.

    For example, to change the CLIP Type in Template 02 from default the DTMF to FSK-Bell, dial ***5702-1-02-01-3***
    Where,
    02 is the template number
    01 is the parameter number for CLIP Type
    3 is the code for FSK-Bell

Default values of SLT Hardware Templates

Parameter No.	01	02	03		06	07	08
Template No.	CLIP Type	Digit Pad Count	Ring Type	SLT Gain Settings Template	AC Impedance	Flash Timer (msec)	Answer Signaling
01	DTMF	0	Trapezoidal	1	600W	101-600	None
02 50	Same as Template Number 01						
Parameters Values							
0	None	0-9					None
1	DTMF		Low Sinusoidal	1	600W	70-100	Polarity Reversal
2	FSK-V.23		LowTrapezoidal	2	900W	101-200	
3	FSK-Bellcore		Sinusoidal	3	350W+(1000W || 0.21µF)	101-300	
4			Trapezoidal	4	220W+(820W || 120nF)	101-400	
5					270Ω+(750Ω || 150 nF)	101-500	
6						101-600	
7						101-700	
8						101-800	
9						101-900	
10							
11							
12							
13							
14							

Parameter No.	09	10	11	17	12	13
Template No.	Disconnect Signaling	Open Loop Disconnect Timer (msec)	Loop Current (mA)	Loop Length	Minimum Current for Off-Hook Detection (mA)	On-Hook Detection Current (mA) or lower
01	None	476	25	Upto 5 Km (16404 ft)	12	10
02 50	Same as Template Number 01					
Parameters Values						
0	None					
1	Polarity Reversal	68	25	Upto 5 Km (16404 ft)	10	10
2	Open Loop	136	30	Above 5 Km (16404 ft)	12	12
3		204	35		14	14
4		272	40		16	16
5		340			18	18
6		408				
7		476				
8		544				
9		612				
10		680				
11		748				
12		816				
13		884				
14		952				

- *The default values of the SLT Hardware Templates are for the default Region India. The default values will differ according to the Region you have selected for the system.*

To default SLT Hardware Template, dial:

- ***5701-1-Template Number*** to default a single template.
- ***5701-2-Template Number-Template Number*** to default a range of templates.
- ***5701-**** to default all templates.

When the SE command to set the AC impedance for the SLT is issued, the settings will be effective when the SLT extension (for which AC Impedance is programmed) goes ON-Hook. SARVAM UCS will restart, if the SLT is OFF-Hook.

To apply the Customized SLT Hardware Template to SLT port, dial:

- ***5703-1-SLT-Template Number*** to assign a hardware template to a single SLT port.
- ***5703-2-SLT-SLT-Template Number*** to assign a hardware template to a range of SLT ports.
- ***5703-*-Template Number*** to assign a hardware template to all SLT ports.
  Where,
  SLT is the Software Port number of the SLT port from 001 to 512.
  Template Number is the number of the customized SLT Hardware Template, from 01 to 50. Default: 01.
  For example, to assign Template 02 to SLT ports 003 to 010, dial ***5703-2-003-010-02***

- Exit SE mode.

Station Basic Feature Template

The Station Basic Feature Template is a set of general features that completely define the basic behavior of an extension. Instead of programming each extension individually, the Station Basic Feature makes it possible to group together extensions that are to be assigned the same set of features, prepare a Station Basic Feature Template with the common set of features and apply it on these extensions.

The SARVAM UCS offers 50 such Station Basic Feature Templates. A Station Basic Feature Template is assigned to all the types of extensions, namely SLT, DKP, ISDN Terminals, E&M[133], T1E1PRI and BRI[134], Radio Extensions, Virtual Extensions.

These templates have commonly used values, but can be customized as per the requirement and applied on the extensions.

Station Basic Feature Template Parameters

The Station Basic Feature Template contains the following features:

- **Time Table:** A Time Table is a schedule of the three Time Zones, namely: Working Hours, Break Hours, Non-Working hours for a week.

  Certain features of the SARVAM UCS like Operator, Class of Service, Toll Control, Outgoing Trunk Access, among others, require the extension to behave differently in each Time Zone[135].

  So, a Time Table is assigned to extensions defining the Time Zones for the entire week, so that the system can execute the Time Zone-dependent features and facilities according to the Time Table.

  There are 8 different Time Table templates to select from. By default, the Time Table 1 is assigned to all Station Basic Feature Templates.

  You may also customize the default Time Table 1 OR customize and assign a different Time Table to the Station Basic Feature Template. Please refer the topic “Time Tables” for more details.

- **Operator:** Define the Operator for the extensions on which the template is applied.

  The system supports multiple Operators. In each Time Zone one of the 20 Operators can be programmed.

  Operator can be a single extension or a group of extensions, so that call management is more efficient. For instance certain extensions may be assigned Operator 1, certain others Operator 2 and the rest may be assigned Operator 3.

  Operator 1 is the default in the Station Basic Feature Template. If you want to assign different extensions to different Operators, you must program a separate Station Basic Feature Template with a different Operator for each extension group.

  Refer the topic “Configuring 'Operator'” to know more.

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133. Applicable only when Orientation type is 'Station'.

134. Applicable only when the T1E1PRI and BRI ports are in NT mode.

135. For example, incoming calls are to be routed to the security personnel extension, instead of the Operator when the office is closed (non-working hours), or certain features in the Class of Service are to be allowed only during working hours, or access to outgoing long distance calls are to be denied during non-working hours, or the extension must play a different greeting message to the callers during break hours and holidays (non-working hours).

- **Class of Service:** Class of Service (COS) defines the set features of the system that the extension is to be allowed access to.

  Not all extensions may require the same set of features. Some extensions may require voice mail, while another group of extensions may need the ability to forward calls to a cell phone, and still others may have no need to make calls outside the office.

  Similarly, certain features may be required during working hours, but not during break or non-working hours.

  It is possible to assign a different Class of Service to different extensions according to their feature requirements as well as according to the Time Zones.

  By default COS group 01 is assigned to the Station Basic Feature Templates for all Time Zones. If you want to assign a different COS for each Time Zone, you must customize the COS group first and then assign the number of the COS group in the Template.

  Refer the topic “Class of Service (COS)” to know more and for instructions on how to enable or disable a feature in a COS group.

- **Call Budget:** This flag is for enabling the Call Budget feature. The Call Budget feature will allot a 'budget' limit for outgoing calls made by extensions on which the template is applied. Refer “Call Budget on Extension” for more details.

- **Toll Control:** This Toll Control Level allows you to define the Call Privilege (calling permission) to be allowed to extensions according to the time of the day, during working hours (WH), break hours (BH) and non-working hours (NH). For each Time Zone, you may define the calling permission to be allowed to extensions by selecting the Type of Call Privilege.

  With the default Toll Control Levels assigned, calling will not be possible. Make sure you configure the parameters as per your requirements.

  Toll Control Level 0 (WH): This Toll Control Level allows you to define the Call Privilege (calling permission) allowed to an extension during Working Hours.

  - **Call Privilege:** Define the type of calling permission to be allowed to the extension during the Working Hours. The call privilege options are: No Calls, Local Calls, Regional Calls, National Calls, International Calls, All Calls, Limited Calls. Default: No Calls.

  - **Allowed List:** For the type of Call Privilege you select, define the list of numbers to be allowed during Working Hours.

  - **Denied List:** For the type of Call Privilege you select, define the list of numbers to be denied during the Working Hours.

  Toll Control Level 0 (BH): This Toll Control Level allows you to define the Call Privilege (calling permission) allowed to an extension during Break Hours.

  - **Call Privilege:** Define the type of calling permission to be allowed to the extension during the Break Hours. The call privilege options are: No Calls, Local Calls, Regional Calls, National Calls, International Calls, All Calls, Limited Calls. Default: No Calls.

- **Allowed List:** For the type of Call Privilege you select, define the list of numbers to be allowed during Break Hours.

- **Denied List:** For the type of Call Privilege you select, define the list of numbers to be denied during the Break Hours.

Toll Control Level 0 (NH): This Toll Control Level allows you to define the Call Privilege (calling permission) allowed to an extension during Non-Working Hours.

- **Call Privilege:** Define the type of calling permission to be allowed to the extension during the Non-Working Hours. The call privilege options are: No Calls, Local Calls, Regional Calls, National Calls, International Calls, All Calls, Limited Calls. Default: No Calls.

- **Allowed List:** For the type of Call Privilege you select, define the list of numbers to be allowed during Non-Working Hours.

- **Denied List:** For the type of Call Privilege you select, define the list of numbers to be denied during the Non-Working Hours.

- **Toll Control Level 1:** This Toll Control Level allows you to define the Call Privilege (calling permission) to be allowed to an extension, regardless of Time Zone. Default: No Calls.

- **Toll Control Level 2:** This Toll Control Level allows you to define the Call Privilege (calling permission) to be allowed to an extension, regardless of Time Zone. Default: No Calls.

- **Toll Control Level 3:** This Toll Control Level allows you to define the Call Privilege (calling permission) to be allowed to an extension, regardless of Time Zone. Default: No Calls.

- **OG-Trunk Bundle Group (WH):** This is the Outgoing Trunk Bundle Group to be allotted to the extension for Working Hours. The extension will be allowed to make outgoing calls through the trunks in this group.

  With the default **OG-Trunk Bundle Group** assigned, calling will not be possible. Make sure you configure the parameters as per your requirements.

- **OG-Trunk Bundle Group (BH):** This is the Outgoing Trunk Bundle Group to be allotted to the extension for Break Hours.

  With the default **OG-Trunk Bundle Group** assigned, calling will not be possible. Make sure you configure the parameters as per your requirements.

- **OG-Trunk Bundle Group (NH):** This is the Outgoing Trunk Bundle Group to be allotted to the extension for Non-Working Hours.

  With the default **OG-Trunk Bundle Group** assigned, calling will not be possible. Make sure you configure the parameters as per your requirements.

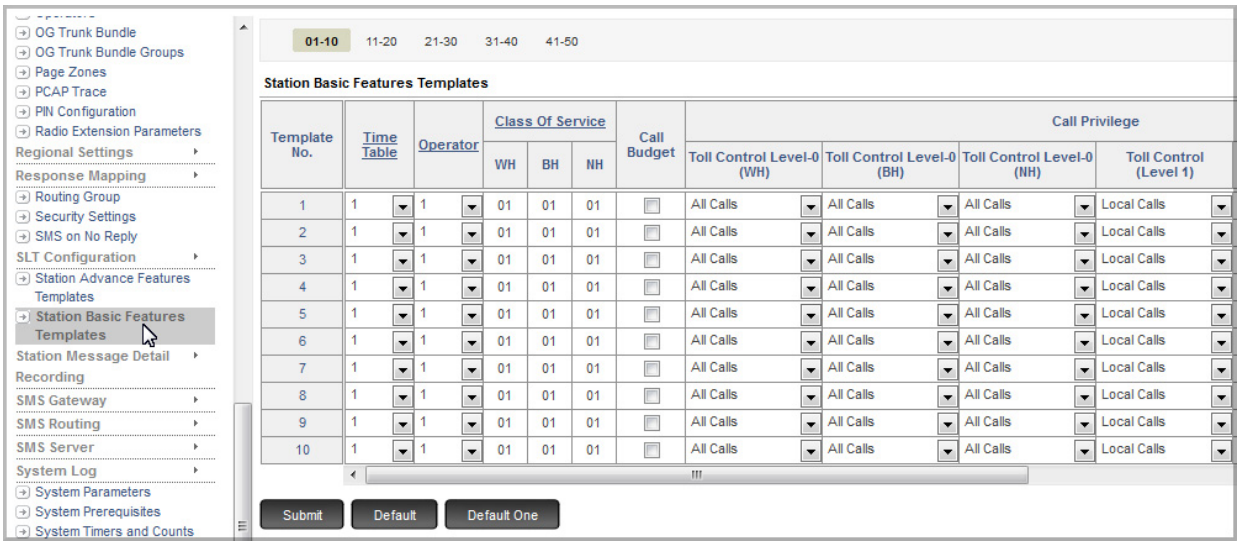
  Refer the topic “OG Trunk Bundle Group” for more details.

- **Store Outgoing Calls:** You can enable or disable the storage of call details - Station Message Detail Records - of Outgoing Calls landing on the extensions on which the template is applied. Select the check box to enable and clear it to disable. Refer the topic “Station Message Detail Recording-Storage” for more details.

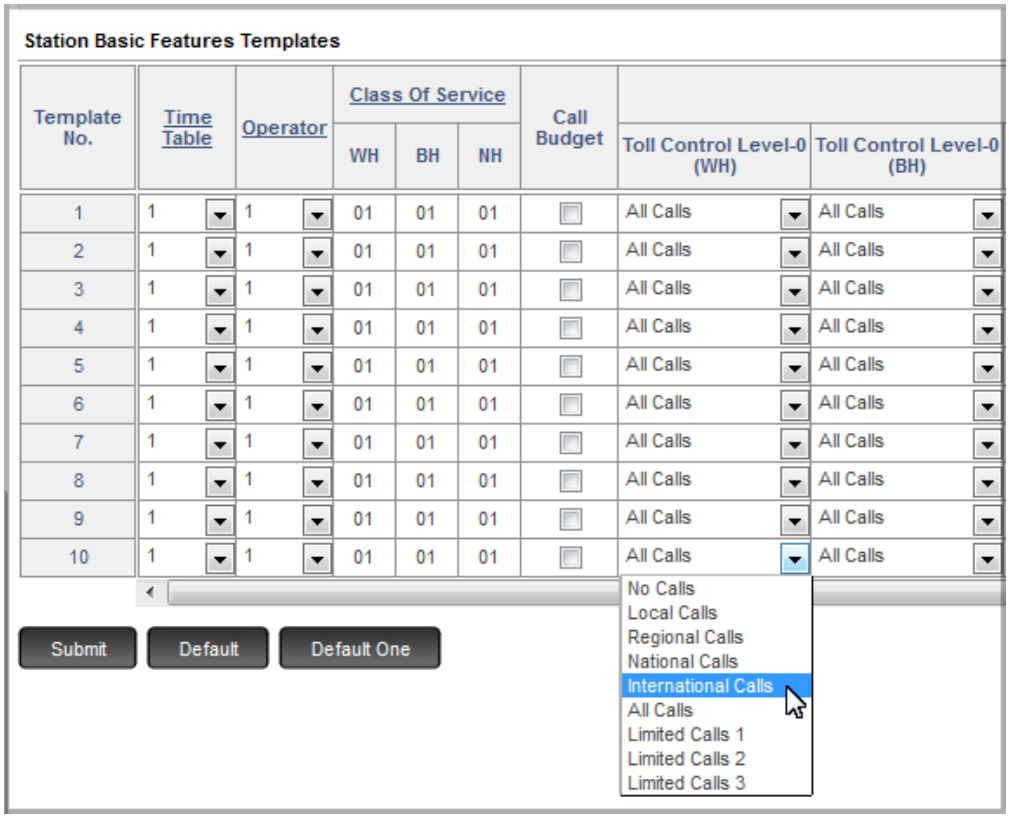
- **Store Incoming Calls:** You can enable or disable the storage of call details - Station Message Detail Records - of Incoming Calls landing on the extensions on which the template is applied. Select the check box to enable and clear it to disable. Refer the topic “Station Message Detail Recording-Storage” for more details.

Customizing Station Basic Feature Template using Jeeves

- Log into Jeeves as System Engineer.
- Under **Configuration**, click **Station Basic Feature Template** to open the page.



- Select a Template number you wish to customize, for example Template 10.



- Change the values of the Station Basic Feature Template parameters as desired.
- Click **Submit** at the bottom of the page to save your changes.
- Now, apply this Template 10 on the SLT/DKP/ISDN Terminals/SIP Extensions.

To apply the customized template on SLT ports,

- Under **Configuration**, click **SLT Configuration**.
- Click **SLT Parameters** to open the page.
- Go to the SLT software ports to which this Template is to be assigned, for example SLT-003 and 004.

- Enter the number of the Template you customized, 10, in the field **Station Basic Feature Template** of each of these SLT ports.

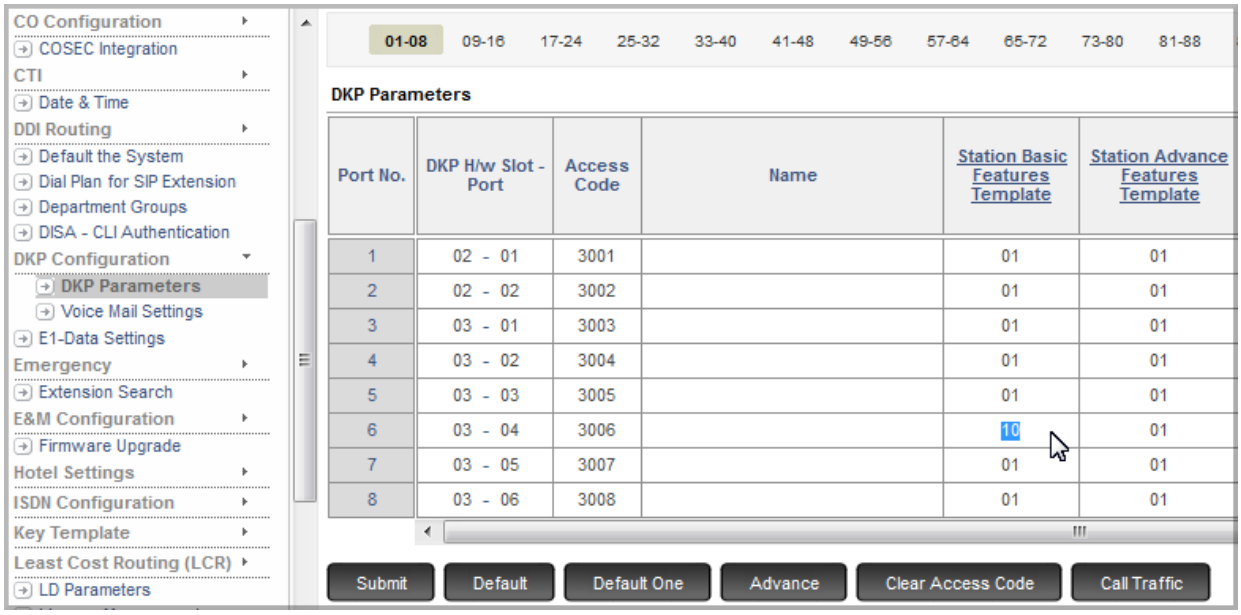
SLT Parameters

Port No.	H/w Slot - Port	Access Code	Name	Station Basic Features Template	Station Advance Features Template
1	02 - 07	2001		01	01
2	02 - 08	2002		01	01
3	02 - 09	2003		01	10
4	02 - 10	2004		01	10
5	02 - 11	2005		01	01
6	02 - 12	2006		01	01
7	02 - 13	2007		01	01

- Click **Submit** at the bottom of the page to save your setting.

To apply the customized template on DKP ports,

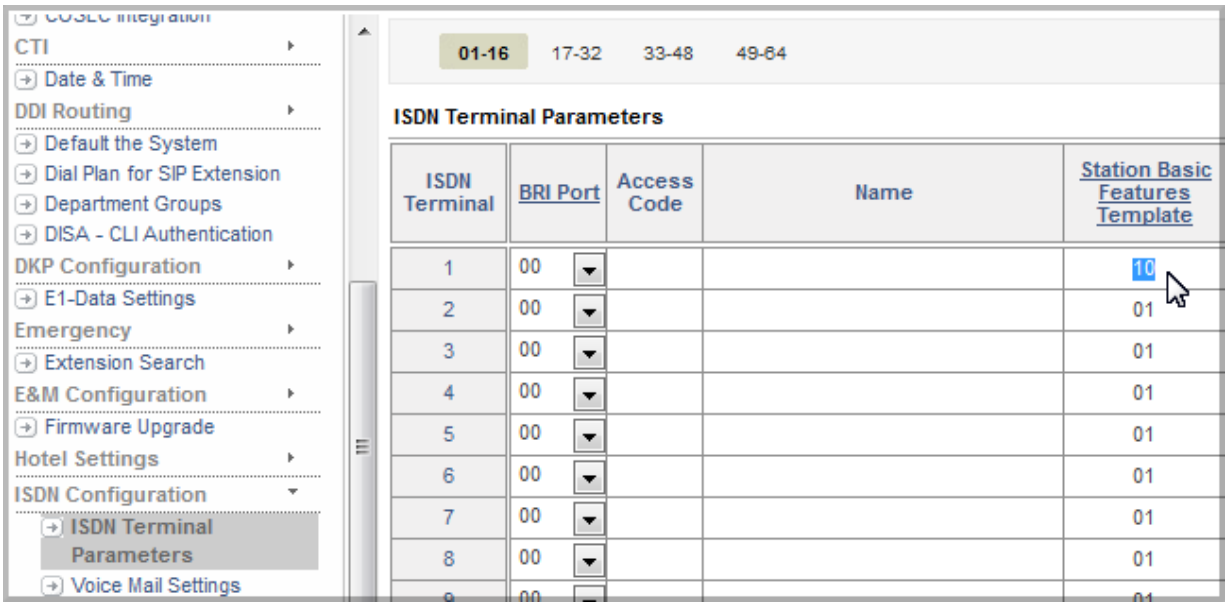
- Under **Configuration**, click **DKP Configuration**.
- Click **DKP Parameters** to open the page.
- Go to the DKP software ports to which this Template is to be assigned, for example DKP-005 to 008.
- Enter the number of the Template you customized, 10, in the field **Station Basic Feature Template** of each of these DKP software ports.



- Click **Submit** at the bottom of the page to save your setting.

To apply the customized template on ISDN Terminal ports,

- Under **Configuration**, click the link **ISDN Terminal Parameters** to open the page.
- Go to the ISDN Terminal software ports to which this Template is to be assigned, for example ISDN-01.
- Enter the number of the Template you customized, 10, in the field **Station Basic Feature Template** of each of the ISDN Terminal software port.

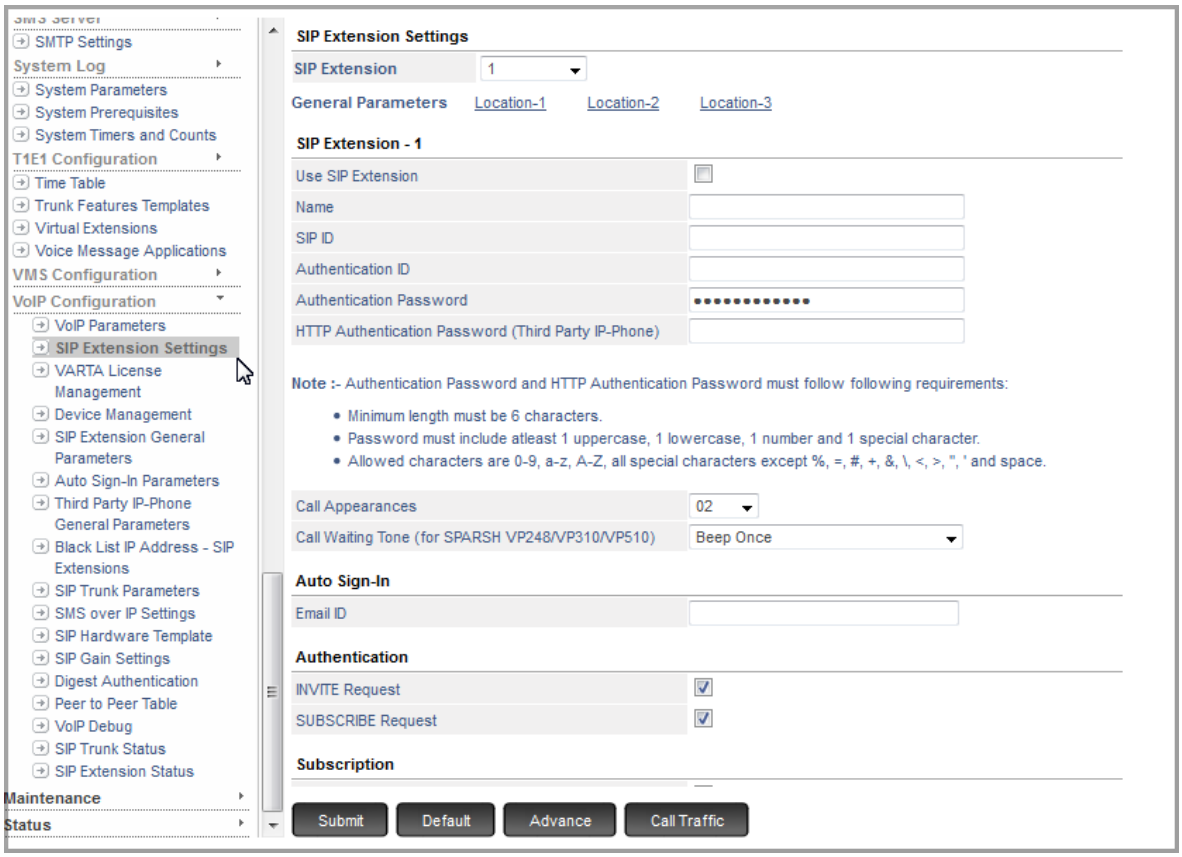


- Click **Submit** at the bottom of the page to save your setting.

To apply the customized template on SIP Extensions,

- Under **Configuration**, click **VoIP Configuration**.
- Click **SIP Extension Settings**.

- Go to the SIP Extensions, for example SIP Extension 1, to which this Template is to be assigned, and enter the Template number.



- Click **Submit** to save your setting.

- Repeat the same steps to customize another template and apply it on the extension ports.

Customizing Station Basic Feature Template using a Telephone

- Enter SE mode from a DKP/SLT.

  To program a feature in a Station Basic Feature Template, dial:
  - ***5502-1-Template Number-Feature Number-Code*** to program the value of a feature in a single template.
  - ***5502-2-Template Number-Template Number-Feature Number-Code*** to program the same value for a feature in a range of templates.
  - ***5502-*-Feature Number-Code*** to program the same value for a feature in all templates.

  Where,
  Template Number is the number of the Station Basic Feature Template, from 01 to 50.
  Feature Number is the number of the Feature in the Template, from 01 to 39.
  Code is the parameter value of the Feature.

  For example, you want to customize Template number 10, by enabling Call Budget feature and select 'Local Calls' as the Toll Control when Call Budget Consumed.
  Dial ***5502-1-10-06-1*** to enable Call Budget feature (number 06) on Template 10.
  Dial ***5502-1-10-19-1*** to select Local Calls (1) as Toll Control when Call Budget Consumed (19) in Template 10.

Refer the following Table for the Feature Number and Codes for the Station Basic Feature Templates.

Parameter No.	01	02	03	04	05	06	07	08	09	10	11	12	13	14	15	16	17
Template No.	Time Table	Operator	Class Of Service			Call Budget	Toll Control - Level 0 (WH)			Toll Control - Level 0 (BH)			Toll Control - Level 0 (NH)			Toll Control (Level 1)	Toll Control (Level 2)
			WH	BH	NH		Call Privilege	Allowed List	Denied List	Call Privilege	Allowed List	Denied List	Call Privilege	Allowed List	Denied List		
01	1	01	01	01	01	X	No Calls	01	01	No Calls	01	01	No Calls	01	01	No Calls	No Calls
02-50	Same as Template# 01																
Parameter Values																	
0	0 or 1-8	01-20	01-16	01 - 16	01 -16	X		01-16	01-16		01-16	01-16		01-16	01-16		
1						O	No Calls			No Calls			No Calls			No Calls	No Calls
2							Local Calls			Local Calls			Local Calls			Local Calls	Local Calls
3							Regional Calls			Regional Calls			Regional Calls			Regional Calls	Regional Calls
4							National Calls			National Calls			National Calls			National Calls	National Calls
5							International Calls			International Calls			International Calls			International Calls	International Calls
6							All Calls			All Calls			All Calls			All Calls	All Calls
7							Limited Calls 1			Limited Calls 1			Limited Calls 1			Limited Calls 1	Limited Calls 1
8							Limited Calls 2			Limited Calls 2			Limited Calls 2			Limited Calls 2	Limited Calls 2
9							Limited Calls 3			Limited Calls 3			Limited Calls 3			Limited Calls 3	Limited Calls 3

Parameter No.	18	19	20	21	22	23	24	25	26	27	28	29	30	31	32	33	34	35	36	37	38	39
Template No.	Toll Control (Level 3)	Toll Control - Call Budget Consumed	OG Trunk Bundle Group - WH						OG Trunk Bundle Group - BH						OG Trunk Bundle Group - NH						Store Outgoing Calls	Store Incoming Calls
			TAC -1	TAC-2	TAC -3	TAC -4	TAC-5	TAC-6	TAC-1	TAC-2	TAC-3	TAC-4	TAC-5	TAC -6	TAC-1	TAC-2	TAC-3	TAC-4	TAC-5	TAC -6		
01	No Calls	No Calls	01	01	01	01	01	01	01	01	01	01	01	01	01	01	01	01	01	01	√	√
02-50	Same as Template# 01																					
Parameter Values																						
0			01-32																		X	X
1	No Calls	No Calls																			√	√
2	Local Calls	Local Calls																				
3	Regional Calls	Regional Calls																				
4	National Calls	National Calls																				
5	International Calls	International Calls																				
6	All Calls	All Calls																				
7	Limited Calls 1	Limited Calls 1																				
8	Limited Calls 2	Limited Calls 2																				
9	Limited Calls 3	Limited Calls 3																				

To default a Station Basic Feature Template, dial:

- ***5501-1-Template Number*** to default the values of a single template.
- ***5501-2-Template Number-Template Number*** to default the values of a range of templates.
- ***5501-**** to default the values of all templates

To assign a Station Basic Feature Template to SLT, dial:

- ***5503-1-SLT-Template Number*** to assign the template to a single SLT port.
- ***5503-2-SLT-SLT-Template Number*** to assign the same template number to a range of SLT ports.
- ***5503-*-Template Number*** to assign the same template number to all SLT ports.

  Where,

  SLT is the number of the Software port of the SLT, from 001 to 512.

  Template Number is the number of the Station Basic Feature Template, from 01 to 50.

  Default: Template 01 is assigned to all SLT ports.

  For example, to assign Station Basic Feature Template number 10 to the SLT software ports 004 to 010, dial ***5503-2-004-010-10.***

To assign a Station Basic Feature Template to a DKP, dial:

- ***5504-1-DKP-Template Number*** to assign a template to a single DKP.
- ***5504-2-DKP-DKP-Template Number*** to assign the same template to a range of DKPs.
- ***5504-*-Template Number*** to assign the same template to all DKPs.
  Where,
  DKP is the number of the Software port of the DKP, from 001 to 128.
  Template Number is the number of the Station Basic Feature Template, from 01 to 50.
  Default: Template 01 is assigned to all DKP ports.

  For example, to assign Station Basic Feature Template 10 to DKP software ports DKP-005 to 010, dial ***5504-2-005-010-10***

To assign a Station Basic Feature Template to an ISDN Terminal port, dial:
- ***5507-1-ISDN-Template Number*** to assign a template to a single ISDN Terminal port.
- ***5507-2-ISDN-ISDN-Template Number*** to assign the same template to a range of ISDN terminal ports.
- ***5507-*-Template Number*** to assign the same template to all ISDN terminal ports.
  Where,
  ISDN Terminal is the number of the ISDN Terminal Software port, from 01 to 64.
  Template Number is the number of the Station Basic Feature Template, from 01 to 50.
  Default: Template 01 is assigned to all ISDN Terminals.

  For example, to assign Station Basic Feature Template number 10 to the ISDN Terminal Software ports 01 to 04, dial ***5507-2-01-04-10***

To assign a Template to an E&M (Station) port, dial:
- ***5505-1-E&M-Template Number*** to assign a template to a single E&M port.
- ***5505-2-E&M-E&M-Template Number*** to assign the same template to a range of E&M ports.
- ***5505-*-Template Number*** to assign the same template to all E&M ports.
  Where,
  E&M is the number of the E&M Software port, from 001 to 128.
  Template Number is the number of the Station Basic Feature Template, from 01 to 50.
  Default: Template 01 is assigned to all E&M (Station) ports

To assign a Station Basic Feature Template to a T1E1PRI port, dial:
- ***5506-1-T1E1PRI-Template Number*** to assign a template to a single T1E1 port.
- ***5506-2-T1E1PRI-T1E1PRI-Template Number*** to assign the same template to a range of T1E1 ports.
- ***5506-*-Template Number*** to assign the same template to all T1E1 ports.
  Where,
  T1E1PRI is the number of the T1E1PRI Software port, from 01 to 08.
  Template Number is the number of the Station Basic Feature Template, from 01 to 50.
  Default: Template 01 is assigned to all T1E1 ports.

To assign a Station Basic Feature Template to a BRI port, dial:
- ***5509-1-BRI-Template Number*** to assign a template to a single BRI port.
- ***5509-2-BRI-BRI-Template Number*** to assign the same template to a range of BRI ports.
- ***5509-*-Template Number*** to assign the same template to all BRI ports
  Where,
  BRI is the number of the BRI Software port, from 01 to 32.
  Template is the number of the Station Basic Feature Template, from 01 to 50.
  Default: Template 01 is assigned to all BRI ports.

To assign a Station Basic Feature Template to a Virtual Extension, dial:
- ***5513-1-Virtual Extension-Template Number*** to assign template to a single Virtual extension.
- ***5513-2-Virtual Extension-Virtual Extension-Template Number*** to assign the same template to a range of Virtual extensions.

- ***5513-*-Template Number*** to assign the same template to all the Virtual extensions.
  Where,
  Virtual Extension is from 01 to 64.
  Station Basic Feature Template is from 01 to 50.
  By default, Template 01 is assigned.

To assign a Station Basic Feature Template to a Magneto Port, dial:

- ***5511-1-Magneto Port-Template Number*** to assign template to a single Magneto Port.
- ***5511-2-Magneto Port-Magneto Port-Template Number*** to assign the same template to a range of Magneto Port.
- ***5511-*-Template Number*** to assign the same template to all the Magneto Port.
  Where,
  Magneto Port is from 001 to 128.
  Station Basic Feature Template is from 01 to 50.
  By default, Template 01 is assigned.

- Exit SE mode.

Station Advanced Feature Template

The Station Advanced Feature Template is a set of advanced extension features, to be applied on extensions to support features like CLIP, Call Duration Control, Storage of Internal SMDR, Call Taping, Alarm Notification, Floor Service, Walk-In Class of Service, etc.

Instead of programming each extension individually, the Station Advanced Feature makes it possible to group together extensions that are to be assigned the same set of features, prepare a Station Advanced Feature Template with the common set of features and apply it on these extensions.

The SARVAM UCS offers 50 such Station Advanced Feature Templates. A Station Advanced Feature Template is assigned to all the types of extensions viz. SLT, DKP, ISDN Terminals, E&M[136], T1E1PRI and BRI[137], Radio Virtual and SIP.

These templates have commonly used values, but can be customized per the requirement and applied on the extensions.

Station Advanced Feature Template Parameters

The Advanced Feature Template comprises the following features on extensions:

- **Caller ID Presentation while Transfer:** This parameter is related to the CLIP feature. It allows you to select whether you want the CLI of the ‘Held Party’ or the CLI of the ‘Transferring Party’ to be displayed to the transfer destination extension while the call is being transferred. Refer the feature description for “Calling Line Identification and Presentation (CLIP)” to know more.

- **Call Forward No-Reply Timer:** This parameter is related to the Call-Forward-No Reply/Preset Call Forward No reply feature. The Timer is the duration for which the system will wait for an extension to answer an incoming call, before forwarding the call it to the programmed destination phone number as Call Forward-No Reply. By default the Timer is set to 30 seconds. Refer the feature description for “Call Forward” and “Preset Call Forward” to know more.

136. *Applicable only when Orientation type is 'Station'.*
137. *Applicable only when the T1E1PRI and BRI ports are in NT mode.*

- **Preset Call Forward:** This parameter is relevant when users do not want to set/cancel Call Forward from their extensions. You can enable the desired type of Preset Call Forward — When Busy, When No Reply or When Busy or No Reply and the calls will be forwarded to the selected destination— Voicemail, Extension or Department Group. You can set Preset Call Forward for each time zone—Working Hours, Break Hours and Non-working Hours. Refer the feature description of "Preset Call Forward" to know more.

- **DDI IC Routing:** This flag is relevant only if you are using DDI IC Routing. As this flag is enabled by default, hence all the extensions configured in the IC Reference Table will behave as DDI Extensions. If you do not want any of these extensions to function as a DDI extensions, disable this flag in the feature template applied on that extension. By default, the flag is enabled.

  Refer the topics "Direct Dialing-In (DDI)" to know more.

- **Send DDI Number as CLI?:** This flag is relevant only if the extension is programmed as a DDI Extensions (the DDI IC Routing is enabled). You can choose whether to the Calling Line Identification (CLI) of the DDI extension should be sent for outgoing calls made from that extension. By default the flag is enabled.

- **Internal Calls Storage:** This parameter is related to the storage of Station Message Detail Records of internal calls, made between extensions of the SARVAM UCS. You can select the type of internal calls to be stored by the system: i) Calls made from the extension, ii) Calls made to the extension, iii) Calls made from as well as calls made to the extension, and iv) no internal calls. By default, Calls made from as well as calls made to the extension are stored. Select desired type of internal call storage. Refer "Station Message Detail Recording-Storage" to know more.

- **Walk-Out Mode:** This parameter is related to the feature Walk-In Class of Service. SARVAM UCS offers two types of Walk-In: i) One-Call per Walk-In, whereby the user is automatically logged out after a call. ii) Walk-In until Logout, whereby the user remains logged on until s/he manually walks out or a second user walks into the same extension.

  You must select the Walk-Out mode for the extension. If One-Call per Walk-In is to be supported on the extension, select 'One Call' as Walk-Out mode.

  If Walk-In until Logout is to be supported on the extension, select 'Multiple Calls' as the Walk-Out mode.

  To know more about this feature, refer "Walk-In Class of Service".

- **CDC Table:** This parameter is to be programmed if you have enabled the "Call Duration Control (CDC)" feature on the extension. The system will check the Call Duration Control (CDC) Table applied to the extension to implement this feature on the extension. So, you must first program the CDC Table and enter the number of the CDC Table you have programmed in this field.

  You can program 8 different CDC tables. By default, CDC Table No. 1 is assigned to all extensions. If CDC is to be applied on extensions of the SARVAM UCS, simply program the default CDC Table No. 1.

  To do this, click the link CDC Table to open the page. Program the CDC Table parameters and 'Submit' to save your settings. Now return to the Station Advanced Feature Template and enter the number of the CDC Table you programmed in the CDC Table field of the template.

  Refer the feature description "Call Duration Control (CDC)" to know more and for instructions on creating CDC Tables.

- **Forced Account Code:** This flag used to enable or disable the feature Forced Account Code on the SLT/ DKP/ISDN Terminal extensions. When this flag is enabled, the system will allow the extension user to dial

an external number only after entering the Account Code. Refer the feature description for “Account Codes” to know more.

- **Department Billing Group:** This parameter enables you to know the total cost of the calls made by a particular group of extensions. This parameter is used as a one of the filters for printing SMDR Reports namely, “Print outgoing calls department group wise”. To be able to use this filter, you must assign the extensions to a Department Bill Group. You can create as many as 99 different Department Bill Groups. Enter the number of the Bill Group you want to assign the extensions to in this field.

- **Floor Service Group:** This parameter is related to the Floor Service feature. Floor Service can be floor-wise or centralized. Floor Service requires you to program Routing Groups as landing destinations for extension calls.

  Program the Floor Service (Routing) Group first and enter this Floor Service (Routing) Group number in this field. There are 96 different Routing Groups to be programmed as Floor Service Groups. By default, no Routing Group is assigned to Floor Service in the Template ('00').

  To know more about this feature, refer the feature description for “Floor Service”.

  Calls from the extension will land on the Floor Service (Routing) Group you have assigned in this field.

- **Alarm Notification Type:** This parameter is related to the Alarm feature of the SARVAM UCS. You can select any from the following options for Alarm Notification to extension users for Alarm calls:

  - **Voice Message:** Extension users will be played a message recorded in the Voice Module, when they answer the alarm call. By default, this option is selected.

  - **Music-on-Hold:** Extension users will be played music-on-hold when they answer the alarm call.

  - **Voice Mail:** The Extension users will be played the message recorded in the VMS, when they answer the alarm call. You must have the VMS Module installed in your system if you want to select this option.

  - **Routing Group:** Extension users will be connected to the extensions programmed in the 'Alarm Notification Group'. For this you must have programmed a Routing Group.

    If you select this option, you can also connect external Messaging Devices to play real-time updated information like date, time, greetings, weather information, specific event announcements, etc., when then extension users answer the alarm calls. The messaging devices must be connected to any of the SLT ports and included in the Routing Group for alarm notification.

- **Alarm Notification Routing Group:** Program this parameter if you have selected 'Routing Group' as the Alarm Notification Type. Enter the number of the Routing Group you have programmed for Alarm Calls.

  By default Routing Group 31 is assigned as the Alarm Notification Routing Group. If the same Routing Group is to be assigned to all extensions, click the link 'Alarm Notification Routing Group' to open the Routing Groups page. Select members (extensions) in this routing group. Save your changes by clicking 'Submit' button.

  You can program a different Routing Group repeating these steps. Make sure to enter the number of the Routing Group you programmed in this field.

  Refer the feature description for “Alarms” to know more.

- **Help Desk:** Enable this flag if you want to define the extension as a "Help Desk". When this flag is enabled, Auto Call Back will be automatically set whenever this extension is found busy.

- **GPAX - Charge Internal Calls:** This parameter is related to the GPAX application. If the extension is programmed as a GPAX user, enable this flag for billing internal calls made by the extension. When this flag is enabled, the system will record all calls made from the extension in the Station Message Detail Record-Outgoing buffer for "GPAX Billing". If the flag is disabled the calls will not be billed and will be recorded in the Station Message Detail Record - Internal buffer as an internal call.

- **Call Taping:** If you want to use the "Call Taping" feature on the extension, you must program the related parameters described below.

  - **Tape Calls coming without CLI:** Enable this flag if you want incoming calls without CLI to be taped. By default, the flag is disabled. The system will not tape incoming calls without CLI.

  - **Number List-Incoming Calls:** Assign a Number List containing numbers of Incoming Calls that must be taped. You must first program the Number List. By default, Number List 09 is assigned.

  - **Number List-Outgoing Calls:** Assign a Number List containing numbers of Outgoing Calls that must be taped. You must first program the Number List. By default, Number List 10 is assigned also for outgoing calls.

    If Number list 10 is already used for another application, prepare a different number list and assign it to the template.

  - **Call Taping for Internal Calls:** Enable this flag if you want to allow Call Taping of internal calls made and received by the extension.

- **Caller Category:** Program this parameter only:

  a. if the extensions are connected to T1 Trunks using RBS protocol.

  b. If you want to define extensions of the systems networked over E&M lines using MFCR2 Signaling as either 'Priority Subscriber' or 'Ordinary Subscriber'.

  Select the appropriate extension category from the following options:
  - Ordinary Subscriber (default)
  - Priority Subscriber
  - Maintenance Equipment
  - Operator
  - Pay Phone
  - Data Transmission
  - Interception Operator

- **Allow External Call Forward for:** This parameter defines the types of calls for which the External Call Forwarding is to be applied. This parameter is relevant for the features "Call Forward" and "Mobility Extension". You may select from the following options:
  - Internal Calls Only
  - Trunk Calls Only
  - Internal + Trunk Calls.

- **Intercept Destination for DND:** This parameter defines the destination extension to route incoming calls landing on the DND set extension. You may select the desired destination for all three time zones —

Working Hours (WH), Break Hours (BH) and Non-working Hours (NH). The **Destination** can be a Voicemail or any extension (SLT, DKP, SIP). If you select the **Destination** as an extension, enter the desired Port Number (**Port No.**). For details, see “Do Not Disturb (DND)”.

- **Route Global Directory Calls using**: This parameter decides the OGTBG to be used to route the Global Directory number. You may select either **OGTBG configured in the Global Dir.** or any other **OGTBG from 01 to 32**.

- **SMS for OG Call - No Reply:** This parameter decides whether to send an SMS or not to the called party for the calls that have not been answered. For more details, see “SMS on No Reply”.

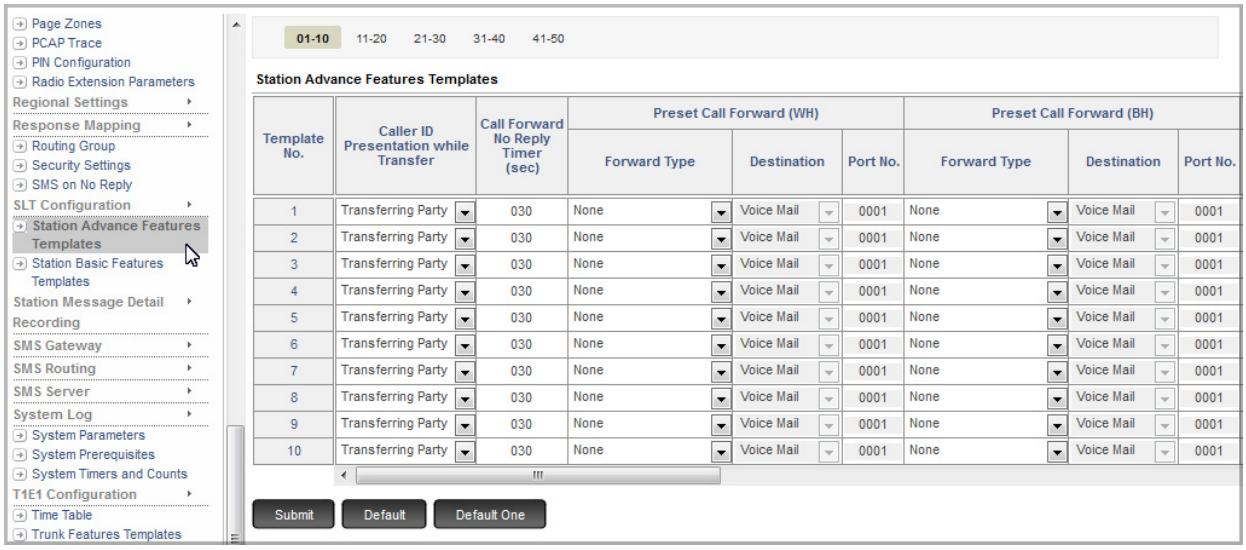
- **Ringer LED:** This parameter decides whether the Ringer LED should glow or not on the desired extensions for incoming calls and as missed call notification. By default, the LED will continue to blink until the missed call log is read or the call is answered/disconnected or the silent CSF Key[138] is pressed. Disable this check box if you do not want the Ringer LED to glow.

- *The Ringer LED for incoming calls is supported on the phones — EON310, EON510, SPARSH VP310 and SPARSH VP510.*
- *The Ringer LED for missed call notification is supported on the phones EON510 and SPARSH VP510 only.*

Customizing Station Advanced Feature Template using Jeeves

- Log into Jeeves as System Engineer.

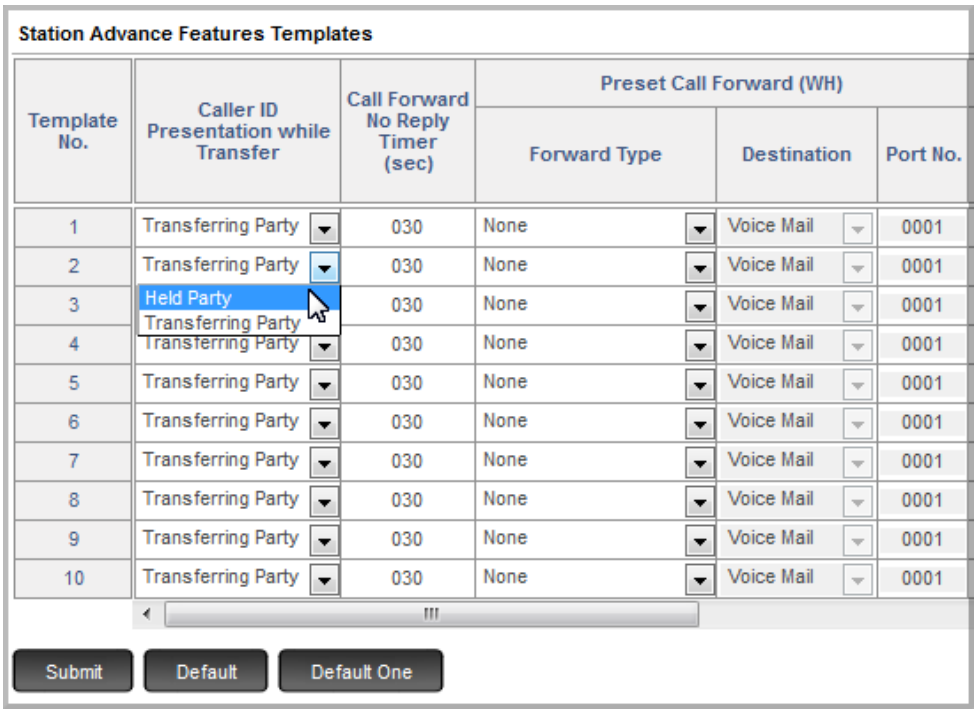
- Under **Configuration**, click **Station Advanced Feature Template** to open the page.



- Select a Template number you wish to customize, for example Template 02.

138. *Supported on EON510 and SPARSH VP510 only.*

- Change the values of the Station Advanced Feature Template parameters as desired.



- Click **Submit** at the bottom of the page to save your changes.
- Now, apply this Template 02 on the SLT/DKP/ISDN Terminal ports, and Virtual Extensions.

To apply the customized template on SLT ports,

- Under **Configuration**, click **SLT Configuration**.
- Click **SLT Parameters** to open the page.
- Go to the SLT software ports to which this Template is to be assigned, for example SLT-003 and 004.

- Enter the number of the Template you customized, 02, in the field **Station Advanced Feature Template** of each of these SLT ports.

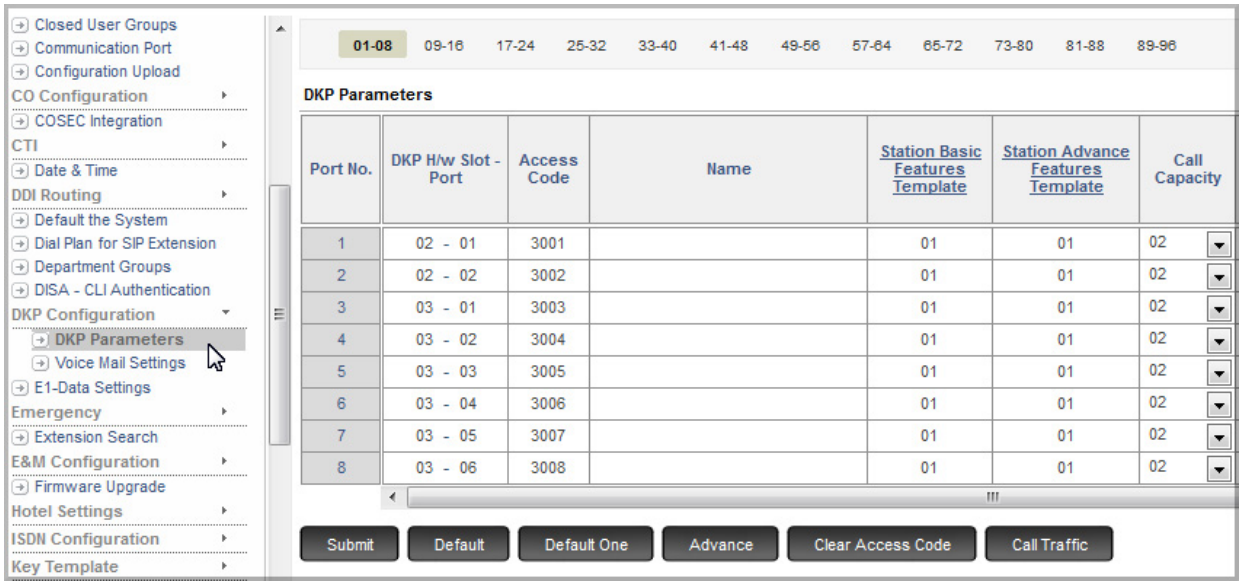
SLT Parameters

Port No.	H/w Slot - Port	Access Code	Name	Station Basic Features Template	Station Advance Features Template
1	02 - 07	2001		01	01
2	02 - 08	2002		01	01
3	02 - 09	2003		01	02
4	02 - 10	2004		01	02
5	02 - 11	2005		01	01
6	02 - 12	2006		01	01
7	02 - 13	2007		01	01
8	02 - 14	2008		01	01

- Click **Submit** at the bottom of the page to save your setting.

To apply the customized template on DKP ports,

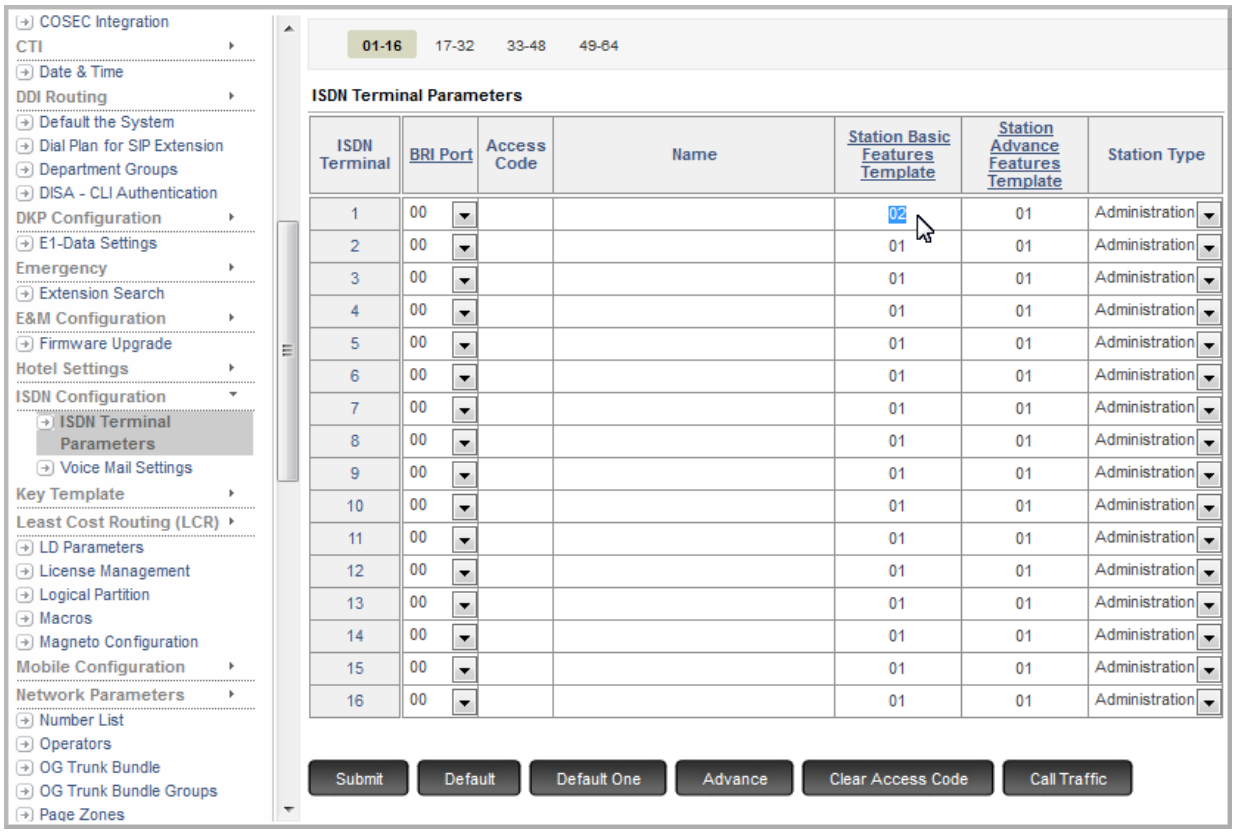
- Under **Configuration**, click the link **DKP Configuration**.
- Click **DKP Parameters** to open the page.
- Go to the DKP software ports to which this Template is to be assigned, for example DKP-005 to 008.Enter the number of the Template you customized, 02, in the field **Station Advanced Feature Template** of each of these DKP software ports.



- Click **Submit** at the bottom of the page to save your setting.

To apply the customized template on ISDN Terminal ports,

- Under **Configuration**, click **ISDN Terminal Parameters** to open the page.
- Go to the ISDN Terminal software ports to which this Template is to be assigned, for example ISDN-01.
- Enter the number of the Template you customized, 02, in the field **Station Basic Feature Template** of each of this ISDN Terminal software port.



- Click **Submit** at the bottom of the page to save your setting.
- Repeat the same steps to customize another template and apply it on other extension ports.

Customizing Station Advanced Feature Template using a Telephone

- Enter SE mode from a DKP/SLT.

  To program a feature in a Station Advanced Feature Template, dial:
  - ***5602-1-Template Number-Feature Number-Code*** to program the value of a feature in a single template.
  - ***5602-2-Template Number-Template Number-Feature Number-Code*** to program the same value for a feature in a range of templates.
  - ***5602-*-Feature Number-Code*** to program the same value for a feature in all templates.

  Where,
  Template Number is the number of the Station Advanced Feature Template, from 01 to 50.
  Feature Number is the number of the Feature in the Template, from 01 to 20.

  For example, you want to customize Template number 02, by changing the default settings of the following features:

- *Call Forward No Reply Timer* to set to 45 seconds, dial ***5602-1-02-02-045*** (Template 02, Feature Number 02 for No Reply Timer, Code 045 for 45 seconds).

- *Forced Account Code Flag* to be enabled, ***5602-1-02-08-1*** (Template 02, Feature Number 08 for Forced Account Code, Code 1 for enable flag).

Refer the following Table for the Feature Number and Codes for the Station Advanced Feature Templates.

Parameter No.	01	02	03	04	05	06	07	08	09
Template No.	Caller ID Presentation while Transfer	Call Forward No Reply Timer (sec)	DDI IC Routing	Send DDI Number as CLI?	Internal Calls Storage	Walk Out Mode	CDC Table	Force Account Code	Department Billing Group
01	Held Party	30	√	√	Store all Calls	One Call	1	X	00
02 - 50	**same as Template No. 1**								

Parameter Values

code										
	0		001 - 255	X	X	Don't Store any Call		1 to 8	X	00-99
	1	Held Party		√	√	Store Call if made from this station	One Call		√	
	2	Transferring Party				Store call if made to this station	Multiple Calls			
	3					Store all the calls made to/from this station				

Parameter No.	10	11	12	13	14	15	16	17	18	19	20
Template No.	Floor Service Group	Alarm Notification Type	Alarm Notification Routing Group	Help Desk	GPAX - Charge Internal Calls	Call Taping: Tape calls coming without CLI	Call Taping: Number List - Incoming Calls	Call Taping: Number List - Outgoing Calls	Call Taping: Call Tapping for Internal Calls	Caller Category	Allow External Call Forward For
01	00	Voice Message	32	X	X	X	32	32	X	Ordinary Subscriber	Trunk Calls
02 - 50											

Parameter Values

code												
	0	01 to 95		01-95	X	X	X	01 to 32	01 to 32	X		
	1		Music on Hold		√	√	√			√	Ordinary Subscriber	Internal Calls
	2		Voice Message								Priority Subscriber	Trunk Calls
	3		External Music								Maintenance Equipment	Internal + Trunk Calls
	4		Routing								Operator	
	5										Pay phone	
	6										Data Transmission	
	7										Interception	

To default a Station Advanced Feature Template, dial:

- ***5601-1-Template Number*** to default the values of a single template.
- ***5601-2-Template Number-Template Number*** to default the values of a range of templates.
- ***5601-**** to default the values of all templates

To assign a Station Advanced Feature Template to SLT, dial:

- ***5603-1-SLT-Template Number*** to assign the template to a single SLT port.
- ***5603-2-SLT-SLT-Template Number*** to assign the same template number to a range of SLT ports.
- ***5603-*-Template Number*** to assign the same template number to all SLT ports.
  Where,
  SLT is the number of the Software port of the SLT, from 001 to 512.
  Template Number is the number of the Station Advanced Feature Template, from 01 to 50.
  Default: Template 01 is assigned to all SLT ports.

For example, to assign Station Advanced Feature Template number 02 to the SLT software ports 004 to 010, dial ***5603-2-004-010-02***

To assign a Station Advanced Feature Template to a DKP, dial:

- ***5604-1-DKP-Template Number*** to assign a template to a single DKP.
- ***5604-2-DKP-DKP-Template Number*** to assign the same template to a range of DKPs.
- ***5604-*-Template Number*** to assign the same template to all DKPs.
  Where,
  DKP is the number of the Software port of the DKP, from 001 to 128.
  Template Number is the number of the Station Advanced Feature Template, from 01 to 50.
  Default: Template 01 is assigned to all DKP ports.

  For example, to assign Station Advanced Feature Template 02 to DKP software ports DKP-005 to 010, dial ***5604-2-005-010-02***

To assign a Station Advanced Feature Template to an ISDN Terminal port, dial:

- ***5607-1-ISDN-Template Number*** to assign a template to a single ISDN Terminal port.
- ***5607-2-ISDN-ISDN-Template Number*** to assign the same template to a range of ISDN terminal ports.
- ***5607-*-Template Number*** to assign the same template to all ISDN terminal ports.
  Where,
  ISDN is the number of the ISDN Terminal Software port, from 01 to 64.
  Template Number is the number of ~Template number is the number of the Station Advanced Feature Template, from 01 to 50.
  Default: Template 01 is assigned to all SIP extensions.

To assign a Station Advanced Feature Template to a Virtual Extension, dial:

- **5613 - 1 - Virtual Extension - Station Advance Feature Template** to assign the template to a single extension.
- ***5613-2-Virtual Extension-Virtual Extension*-Station Advance Feature Template** to assign the same template to a range of extensions.
- ***5613-*- Station Advance Feature Template*** to assign the same template to all extensions.
  Where,
  Virtual Extension is from 01 to 64.
  Station Advance Feature Template is the number of the Station Advanced Feature Template, from 01 to 50.
  Default: Template 01 is assigned to all Virtual Extensions.

To assign a Station Advance Feature Template to a Magneto Port, dial:

- ***5611-1-Magneto-Station Advanced Feature Template Number*** to assign a template to a single Magneto port.
- ***5611-2-Magneto-Magneto- Station Advanced Feature Template Number*** to assign the same template to a range of Magneto ports.
- ***5611-*- Station Advanced Feature Template Number*** to assign the same template to all Magneto ports.
  Where,
  Magneto is the Software Port number of the Magneto port from 001 to 128.
  Station Advanced Feature Template is from 01 to 50.
  Default: Station Advanced Feature Template Number 01.

- Exit SE mode.

Configuring SIP Extensions

SARVAM UCS supports 999 SIP extensions. You can:

- Connect SPARSH VP248, the Extended IP Phone for SARVAM UCS supplied by Matrix.
- Connect SPARSH VP330, the Touch Screen Extended IP Phone for SARVAM UCS supplied by Matrix.
- Connect SPARSH VP310, the Executive IP Phone for SARVAM UCS supplied by Matrix.
- Connect SPARSH VP510, the Premium IP Phone for SARVAM UCS supplied by Matrix.
- Connect Extended SPARSH VP710, the Smart Video IP Phone for SARVAM UCS supplied by Matrix.
- Connect SPARSH VP210, the Entry Level IP Phone for SARVAM UCS supplied by Matrix.
- Register the Matrix VARTA ADR100 and VARTA AMP100 UC Clients for Mobile with SARVAM UCS.
- Register the MATRIX VARTA WIN200 Desktop UC Client with SARVAM UCS.
- Connect Matrix Standard SIP Phones.
- Connect any other Standard SIP phone or SIP enabled device, such as an IP phone, a Soft phone, an Analog phone adapter.

To know more about Extended IP Phones, Mobile and Desktop UC Clients, see “Extended IP Phone/VARTA UC Client - Operation”.

SARVAM UCS supports interoperability with the standard IP Phones. For a list of IP phones on which various features of SARVAM UCS have been tested, see “SARVAM UCS Features tested on IP Phones of different Brands” in the Appendix.

SIP Extensions function like any normal DKP/SLT extension of the SARVAM UCS. SIP Extensions can make and receive calls to any extension user of the SARVAM UCS as well as to external numbers over PSTN, GSM, VoIP and E&M lines, depending on the “Logical Partition” configured in the System.

- *SIP Extensions (IP Subscribers) are a licensed feature. Decide the number of IP Subscribers you will require and buy the license. Refer the topic “License Management” to know more.*
- *You can register a SIP Extension at three different locations as a single SIP Extension for Call Forking.*

Configuring SIP Extension using Jeeves

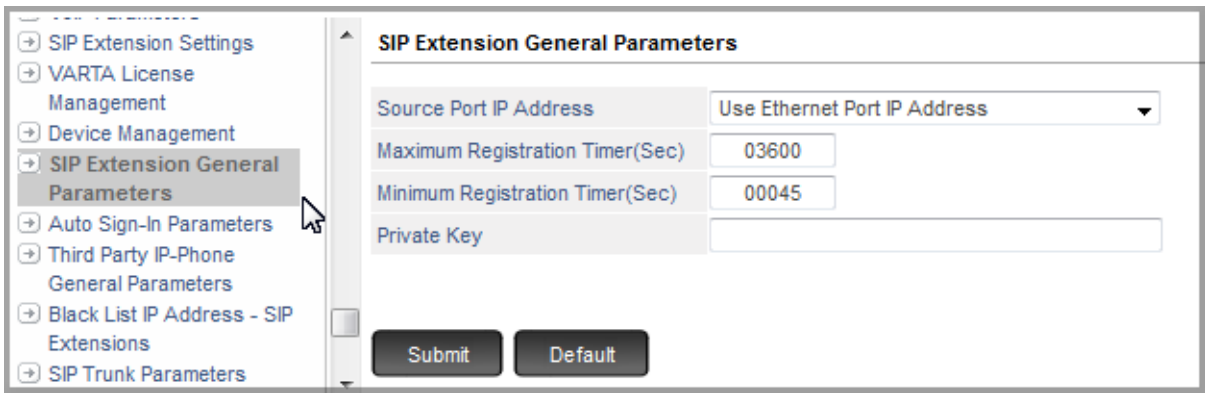
You need to configure the following parameters for SIP Extensions[139]:

- **SIP Extension General Parameters**, see “Configuring SIP Extension General Parameters”.
- **SIP Extension Settings**, you can configure the SIP extensions either one by one or by using bulk configuration. See “Configuring SIP Extension Settings as per the Extended Phone Type” to know how to configure extensions one-by-one and “Configuring SIP Extensions using Bulk Configuration” to know how to configure extensions using bulk configuration.
- **Voice Mail Settings,** if you want to provide mailbox to the SIP extensions. See “Extension Voice Mail Settings”.

139. *SARVAM UCS supports only IPv4 Addresses for registering SPARSH VP248, SPARSH VP310, SPARASH VP330, Extended SPARSH VP710, Mobile UC Clients as well as VARTA WIN200.*

Configuring SIP Extension General Parameters

- Log into Jeeves as System Engineer.
- Under **Configuration**, click **VoIP Configuration**.
- Click **SIP Extension General Parameters** link.



- Set the following SIP Extension General Parameters, as required.
  - As the **Source Port IP Address,** select the NAT Traversal mechanism for SIP messages from the following options:
    - ***Use Ethernet Port IP Address:*** Select this option, if your system is not located behind a NAT Router.
    - ***Use IP Address fetched using STUN:*** Select this parameter, if your system is located behind a NAT Router, and you have set '*Use IP Address fetched using STUN*' as the NAT Traversal mechanism in the "Configuring VoIP Parameters".
    - ***Use Router's Public IP Address:*** Select this option, if your system is located behind a NAT Router, and you have set '*Router's Public IP Address*' as the NAT Traversal mechanism in the "Configuring Network Parameters".
  - You may set the **Maximum Registration Timer (sec)** as required. This is the Maximum Expiry Timer, which the system will accept in the REGISTER request received. If the value of Maximum Expiry Timer received in the REGISTER request is greater than the value you have set here, the system will send the value you have set in the SIP message. The same timer is used for handling SUBSCRIBE requests. The valid range of this timer is from 10 to 99999 seconds. By default it is set to 3600 seconds.
  - You may set the **Minimum Registration Timer (sec)**, as required. This is the Minimum Expiry Timer, which the User Agent should send in its REGISTER request. If the expiry value in the REGISTER message is less than this value, the request will be rejected. The valid range of this timer is from 10 to 99999 seconds. By default, it is set to 45 seconds.

The Timers will be applicable only after System Restart.

- In the **Private Key field**, enter the MD5 authentication key of SARVAM UCS should use to encrypt/ decrypt the SIP messages. The Private Key may consist of a maximum of 24 characters. By default, the field is blank.

- Click **Submit** to save your settings.
- You can restore the default values of any one or all the parameters on this page by clicking the **Default One** and the **Default** button respectively.

Configuring SIP Extension Settings as per the Extended Phone Type

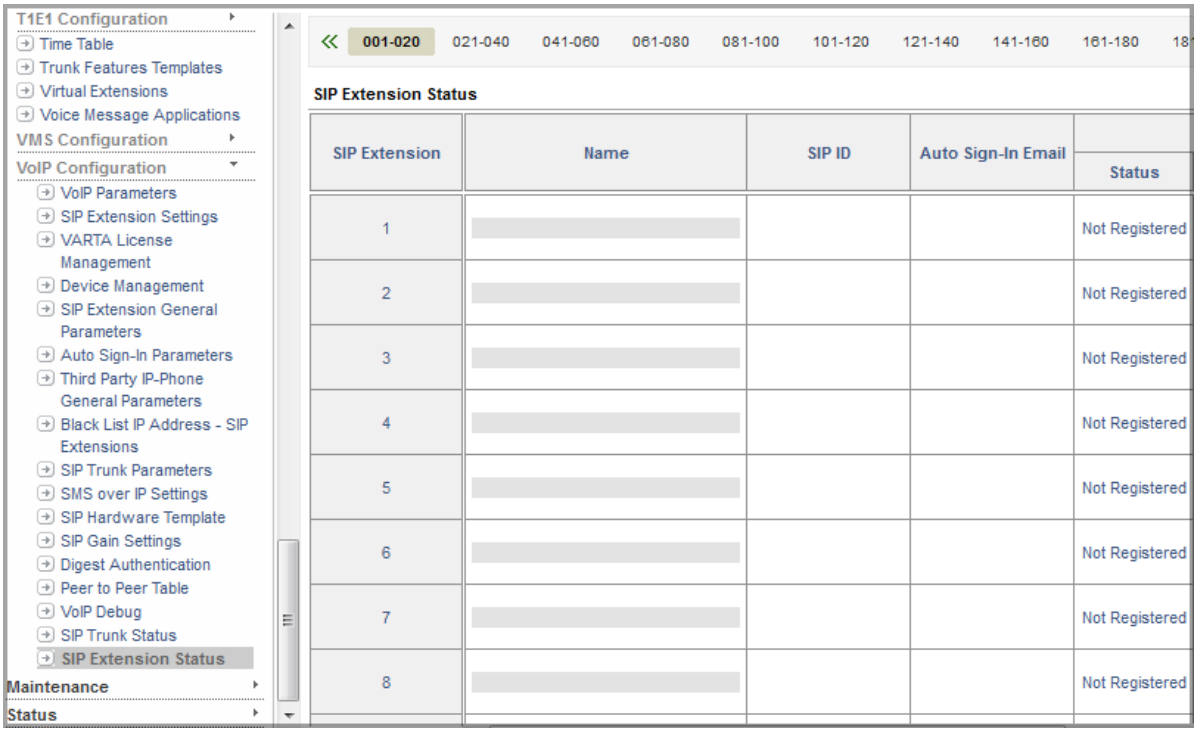
- If you have registered the Matrix Mobile UC Clients as SIP Extensions, for configuration instructions see “Configuring Matrix VARTA ADR100/AMP100 UC Clients”.
- If you have registered the MATRIX VARTA WIN200 Desktop UC Client as SIP Extensions, for configuration instructions see “Configuring Matrix VARTA WIN200 UC Client”.
- If you have connected the Matrix SPARSH VP248 as SIP Extensions, for configuration instructions see “Configuring Matrix SPARSH VP248”.
- If you have connected the Matrix SPARSH VP330 as SIP Extensions, for configuration instructions see “Configuring Matrix SPARSH VP330”.
- If you have connected the Matrix SPARSH VP310 as SIP Extensions, for configuration instructions see “Configuring Matrix SPARSH VP310”.
- If you have connected the Matrix SPARSH VP510 as SIP Extensions, for configuration instructions see “Configuring Matrix SPARSH VP510”.
- If you have connected the Matrix Extended SPARSH VP710 as SIP Extensions, for configuration instructions see “Configuring Matrix Extended SPARSH VP710”.
- If you have connected the Matrix SPARSH VP210 as SIP Extensions, for configuration instructions see “Configuring Matrix SPARSH VP210”.
- If you have connected Standard SIP Phones or SIP enabled devices as SIP Extensions, for configuration instructions see “Configuring Standard SIP Phones”.

Viewing SIP Extension Status

You can view the Status of SIP Extension using Jeeves. To do this,

- Log into Jeeves as System Engineer.
- Under **Configuration**, click **VoIP Configuration**.

- Click **SIP Extension Status**.



- The SIP Extension Status page will open and display the following for each SIP Extension,
    - SIP Extension number
    - Name of the SIP extension
    - SIP ID assigned to the SIP Extension
    - Status of Auto Sign-In Email
    - REGISTRATION status; whether the SIP Extension is registered or not.
    - Contact 1
    - Contact 2
    - Contact 3
- You may Log out of Jeeves.

*You can also view the SIP Extension Status from the **Status** link. To view, click the SIP Extension link under Status.*

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Configuring Matrix SPARSH VP248

SPARSH VP248[140], the proprietary SIP-based IP Phone for SARVAM UCS, supplied by Matrix, is a feature-rich phone, providing voice communication over IP network. It functions like EON48, the proprietary digital key phone of Matrix. To know the list of features supported, refer to “SARVAM UCS Features Supported in Terminals”.

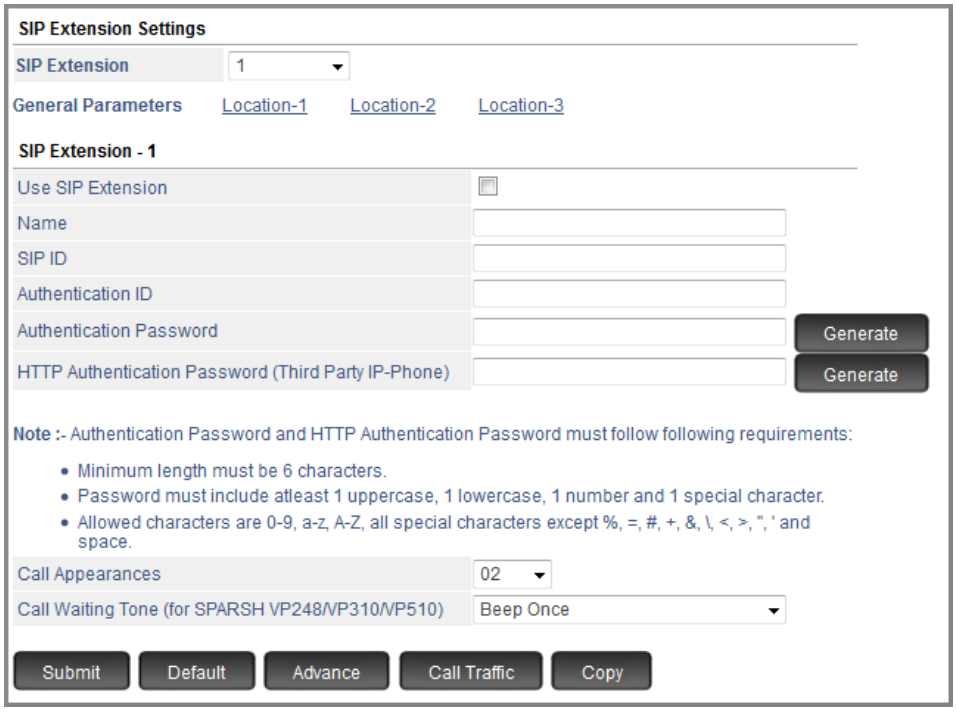
For instructions on how to use SPARSH VP248, refer to the common *EON48_310_SPARSH VP248_310 User Guide.*

To be able to use SPARSH VP248 - Extended IP Phone, you must configure the following:

- SIP Extension General Parameters, see “Configuring SIP Extension General Parameters”.
- SIP Extension Settings, see “Configuring SIP Extension Settings using Jeeves”
- Extended IP Phone Settings, see “Configuring Matrix Extended Phone Settings using Jeeves”
- Voice Mail Settings, if you want to provide mailbox facility to the extension. See “Extension Voice Mail Settings”.

Configuring SIP Extension Settings using Jeeves

- Log into Jeeves as System Engineer.
- Under **Configuration**, click **VoIP Configuration**.
- Click **SIP Extension Settings**.



The page of SIP Extension 001 opens.

- You may select the **SIP Extension** number you want to configure.

140. SARVAM UCS supports only IPv4 Addresses for registering SPARSH VP248.

The parameters of the SIP Extension number you selected will appear on this page.

For SARVAM UCS upto 999 SIP Extensions can be registered with the system.

- Select the **Use SIP Extension** check box to enable the SIP extension. Default:disabled.

- In the **Name** field, enter a name for the SIP Extension, which may be the name of the person who will use the SIP Extension or the name of a Department. The name you enter here will be displayed as the Caller ID of the SIP Extension on the remote user's phone, when the SIP Extension user makes calls.

  The name may consist of a maximum of 18 alphanumeric characters.

- Enter the **SIP ID** for the extension. The SIP ID is necessary for registering the SIP Extension with the Registrar of the system. It is the number with which you can call the SIP Extension. Any extension user of the SARVAM UCS can call a SIP Extension by dialing the SIP ID assigned to the SIP extension. SIP ID of each SIP Extension must be a unique number string of a maximum of 6 digits. Any combination of digits from 0 to 9 and the characters * and # are allowed.You cannot assign the same SIP ID to more than one extension.

  To assign SIP IDs according to your preference and requirement to a range of SIP Extensions, see “Assigning Access Codes to a Range of Extensions”.

  By default, the SIP IDs are Blank.

The SIP ID will be set to default value (blank), when you restore the default settings of the system.

- In **Authentication ID**, enter the number which you want the system to use for user authentication of the SIP messages received from the SIP Extension. You cannot keep this field blank and it must be unique for each SIP extension. The number may be a string of maximum 6 alphanumeric characters. All ASCII characters except < > and “ (double quote) are allowed. Default: Blank.

Make sure the User ID configured in “Digest Authentication” does not conflict with the Authentication ID configured above.

- In **Authentication Password**, enter the password manually or click **Generate** to automatically generate a unique password. This password will be used by the system to authenticate the SIP messages received from the SIP Extension. To avoid unauthorized access, we recommend you to change the password regularly. Make sure it is strong and is kept confidential. When you enter the password manually, the password must:

  - be of minimum 6 characters and can be a maximum of 12 characters.
  - include atleast one upper-case, one lower-case, one number and one special character.
  - all ASCII characters (except Percentage **%**, Hash **#,** Equal to **=,** Plus **+**, And **&**, Backslash **\,** Less than **<,** Greater than **>,** Apostrophe **’,** Double Quote **"** and **Space**) are allowed.
    Default: Blank.

  To provide additional security, when the Authentication fails 10 times consecutively due to wrong Authentication ID / Authentication Password, the system will blacklist the IP Address and Port for registration of this SIP Extension. However, you can remove the IP Address from the Blacklist IP Address list. See “Black List IP Address - SIP Extensions” for more details. This activity will be logged in the “System Activity Log” as well as “Simple Network Management Protocol (SNMP)”.

Make sure you note down or copy the Authentication Password in a confidential file.

- In **Call Appearances**, define the maximum number[141] of simultaneous incoming calls that the SIP Extension user should be allowed to receive. You can set up to 10 call appearances for a SIP Extension. Default: 2.

  When Call Appearance is set to 2, the SIP Extension can receive 2 calls at a time.

- During an on-going conversation, if there is a second incoming call, the system plays beeps to indicate the second incoming call. You can set the frequency of the **Call Waiting Tone (for SPARSH VP248/VP310/ VP510)** beeps as per your requirement. You can select from the following options:
  - Off
  - Beep Once
  - Beep until Answered

  Default: Beep Once

  However, when an ongoing call is being taped or recorded, the call waiting tone for any new incoming call will not be played.

- Under **Auto Sign-In,** enter the **Email ID** of the extension user you wish to store. The Email ID can be a maximum of 64 characters. The Email ID is used for various server features.

Auto Sign-In parameters — Send Configuration Mail and Mail Status, *are applicable only for Mobile Clients — VARTA ADR100, VARTA AMP100 applications.*

*The **Send Configuration Mail** button will appear only after you have enabled the SIP Extension and configured the SIP ID, Authentication ID and Password.*

- Under **Authentication**, enable Authentication of any or all of the following SIP Message Options by selecting the respective check boxes:
  - **INVITE Request**
  - **SUBSCRIBE Request**

  By default, the SIP Message Options INVITE and SUBSCRIBE are enabled.

Make sure that the Authentication ID for the SIP Extension has been programmed.

- For secure conversations over SIP, enable **SRTP Mode**. The SARVAM UCS supports the following options:

  - **Disable:** SARVAM UCS uses normal RTP for transporting the speech packets.

  - **Optional:** SARVAM UCS uses SRTP for transporting the speech packets. If the remote user does not support SRTP, SARVAM UCS will use normal RTP for transporting the speech packets.

    - If you select this option, you must configure the **SRTP Media Type**. You may select **AVP** or **SAVP**. By default, AVP is selected as the SRTP Media Type.

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141. For the calls that are routed through the CPU, the number of Vocoder channels that will be supported would be as per the license you purchase.

- **Forced:** SARVAM UCS uses only SRTP (SAVP) for transporting the speech packets. If the remote user does not support SRTP, SARVAM UCS will reject incoming calls from and drop outgoing calls made to such users.

  By default, SRTP Mode is Disabled.

- Assign a **SIP Hardware Template** to the SIP Extension. Default: 01. The "SIP Hardware Template" contains voice quality related features such as Voice Codec selection, Tx and Rx Gains, Echo Cancellation, Jitter Buffer and Fax-over-IP options and related parameters

  There are 32 different templates to choose from. Each template can also be altered to suit your requirement and preferences. By default, Template number 01 assigned to all SIP Extensions as well as to SIP Trunks.

  Check if the values in this template fulfill requirements of the SIP Extension. If Template 01 fulfills the feature requirements, retain Template 01.

  If a different set of SIP hardware features are to be allowed to this SIP Extensions, prepare another template and assign it to this extension. To do this,

  - Under **VoIP Configuration**, click the **SIP Hardware Template** link.
  - Select a Template number, for example 02.
  - Customize Template number 02 and click **Submit**.
  - Return to the **SIP Extension Settings** page.
  - Select the number of the Template you customized, Template 02 in the **SIP Hardware Template** field.

  Also see the topic "SIP Hardware Template" to know more about customizing the templates and applying on the SIP Extensions.

- Assign a **Station Basic Feature Template** to the SIP Extension. Default: The "Station Basic Feature Template" has a set of features like Time Table, Class of Service, Toll Control, Operator, Storage of Incoming and Outgoing Calls, Outgoing Trunk Bundle groups. There are 50 different templates to choose from. Each template can also be altered to suit your requirement and preferences.

  If the default Station Basic Feature Template 01 fulfills the feature requirements of the SIP Extension ("Class of Service (COS)", "Toll Control", "OG Trunk Bundle Group", etc.) retain this template, you may also customize this template. If you want to assign a different set of features to this SIP Extension, prepare a different Station Basic Feature Template and apply it to this extension. To do this,

  - Under **Configuration**, click the **Station Basic Feature Template** link.
  - Select a Template number, for example 05.
  - Customize Template number 05 and click **Submit**.
  - Return to the **SIP Extension Settings** page.

- Select the number of the Template you customized, Template 05, in the **Station Basic Feature Template** field.

- Click **Submit** to save changes.

Also, see the topic "Station Basic Feature Template" to know more about customizing the templates and applying on extensions.

- Assign a **Station Advanced Feature Template** to the SIP Extension. Default: Template 01. The "Station Advanced Feature Template" has a set of advanced features for extensions such as Alarm Notification settings, Routing of Incoming Auto Attendant Calls, Call Duration Control, Floor Service, etc. There are 50 different templates to choose from. Each template can also be altered to suit your requirement and preferences.

  Check if the default template fulfills the feature requirements of the SIP Extension by clicking the **Station Advanced Feature Template** link.

  You may retain this template and customize it further, or customize another template if a different set of features are to be allowed to this SIP Extension. To customize/prepare another template,

  - Under **Configuration**, click the **Station Advanced Feature Template** link.

  - Select the Template number, for example 02, and customize this template.

  - Click **Submit** and return to the 'SIP Extension Settings' page.

  - In the **Station Advanced Feature Template** field, select the number of the template you customized.

  - Click **Submit** to save changes.

  Also see the topic "Station Advanced Feature Template" for instructions on customizing these templates and applying them on the extensions.

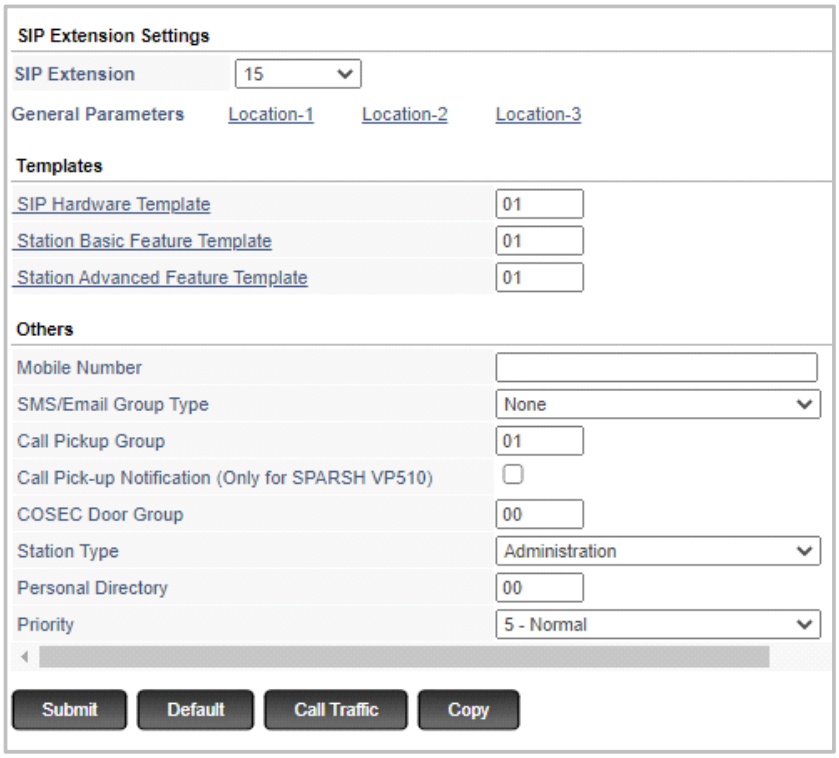
- In **Voice Mail Settings**, click the Voice Mail Settings link. The respective Extension Voice Mail Settings window will open. You may edit the parameters. For details, see "Extension Voice Mail Settings".

The Voice Mail Settings link will be visible only if you have configured the respective SIP ID.

Click **Close** to close the window.

Advanced Configuration Parameters

- If you want to provide other features like Personal Directory, Priority, or assign a Station Type to the SIP Extension, click the **Advanced** button at the bottom of the page.



- Enter the **Mobile Number** of the extension user you wish to store. The Number can be a maximum of 16 digits.

- You can assign the extension user to a Group. Select the desired **SMS/Email Group Type** from the list.The system clubs together extension users assigned the same Group. Default: None. For details, see “SMS/Email Group”.

- Assign the SIP Extension to a **Call Pick-up Group**, if required.

  Call Pick Up allows the SIP Extension to 'pick up' (answer) calls ringing on any other extension, by dialing a feature code, without physically going to the ringing extension. It also allows incoming calls for the SIP Extension to be answered by the other extensions assigned the same Call Pick-Up group.

  For this to work, both the ringing extension and the extension picking up the call must be in the same 'Call Pick Up Group'. Refer “Call Pick Up” for instructions on how to create groups. You can create as many as 99 groups numbered from 01 to 99.

  Enter the number of the Call Pick-Up Group you created for this SIP Extension in this field.

- Select the **Call Pick-up Notification (Only for SPARSH VP510)** check box, if you want the Call Pick-up Notification to be displayed on the phone LCD. The Call Pick-up Notification will be displayed for internal as well as external calls. The notification will be displayed on SPARSH VP510 only if the ringing extension is in the same Call Pick-up Group. The notification will be displayed only when SPARSH VP510 is in idle or dial state. The notification will display the name/number of the Caller along with the name/number of the Called. For example, if the name and number of the Caller is ABC, 2001 and the name and number of

Called is XYZ, 1001, the LCD notification will be displayed as ABC -> XYZ. Along with the LCD display the phone back-light will also be lit.

For this to work, make sure you have enabled Call Pick-up in COS as well as assigned a Call Pick-up Group to the extension. Refer to "Call Pick Up" and "Class of Service (COS)". Call Pick-up Notifications will be displayed for DKP, SLT as well as SIP Extensions and for calls landing through CO, SIP as well as T1E1 Trunks. For details of the Notification, refer to the EON510_SPARSH VP510 V2 User Guide.

- You must assign the extension user to a **COSEC Door Group** for COSEC Integration. The users in the same group must be assigned the same group. You can create as many as 50 groups numbered from 00 to 50. Users who are assigned COSEC Door Group '00' are not a part of any group. See "COSEC Integration" for more information.

- If this is an Operator extension and you want the system to play beeps during a conference to the participants, to indicate the presence or absence of the Operator, select the **Station Type** as **Assistant**.

  If you are using the system in the *Hotel Mode*, select the **Station Type** for the SIP Extension as **Administration/Assistant** or **Guest**. The system will consider the options Administrator and Assistant as same.

- You may assign a **Personal Directory** number to the SIP Extension. Default: 00.

  A Personal Directory is a list of 25 frequently dialed numbers, each of which are stored by Index number (location code), Name and Trunk Access Codes ("Out Going Trunk Bundle Group Index"). The Personal Directory is necessary for using the features "Abbreviated Dialing" and "Dial By Name".

  When a Personal Directory is assigned to a SIP Extension, make sure you also configure this directory. The Personal Directory can be programmed by the SIP Extension users and by the System Engineer. Refer the topic "Abbreviated Dialing" for instructions on programming the Personal Directory.
  If Personal Directory is not to be assigned, enter 00 in this field.

- Select a **Priority** Level for the SIP Extension from 1 to 9. Default; 5-Normal.

  Each extension of the SARVAM UCS is assigned a Priority Level starting from 1, 2, 3... to 9, with '1' being lowest Priority and '9' being highest Priority. Whenever an extension (phone) with higher priority calls an extension with lower priority, a triple ring is placed on the called extension. To know more, read the feature description "Priority".

  If this SIP extension is assigned to Operator, you may want to set a higher priority for this extension.
- Click **Submit** to save your SIP Extension Settings.

Configuring Matrix Extended Phone Settings using Jeeves

You can register three Matrix Extended IP Phones/UC Clients at three different locations as a single SIP Extension. You can connect/register the same or different types of Extended Phones/UC Clients —SPARSH VP248, SPARSH VP310, SPARSH VP510, SPARSH VP210, SPARSH VP330, Matrix VARTA UC Clients or MATRIX VARTA WIN200 Desktop UC Client — at each location. In this case we assume that SPARSH VP248 is connected at Location 1, 2 and 3.

If you want to use more than one SPARSH VP248 Extended IP Phones as a SIP Extension, configure their settings at **Location 1**, **Location 2** and **Location 3**.

If you have connected SPARSH VP310 at any of the locations, refer to "Configuring Matrix SPARSH VP310".

If you have connected SPARSH VP330 at any of the locations, refer to “Configuring Matrix SPARSH VP330”.

If you have connected SPARSH VP510 at any of the locations, refer to “Configuring Matrix SPARSH VP510”.

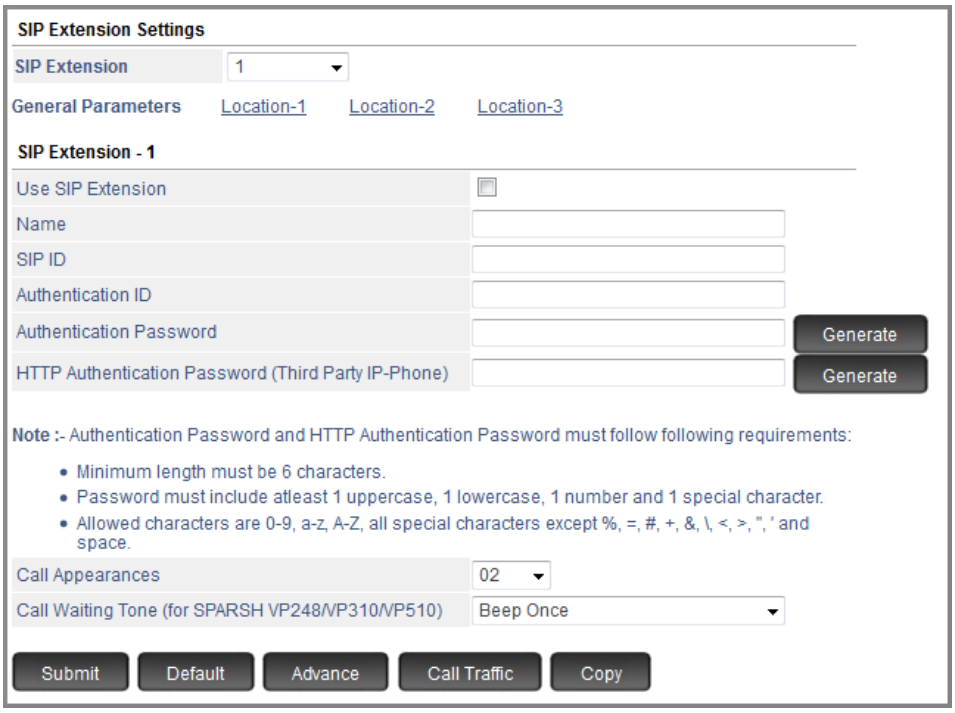
If you have connected Extended SPARSH VP710 at any of the locations, refer to “Configuring Matrix Extended SPARSH VP710”.

If you have connected Extended SPARSH VP210 at any of the locations, refer to “Configuring Matrix SPARSH VP210”.

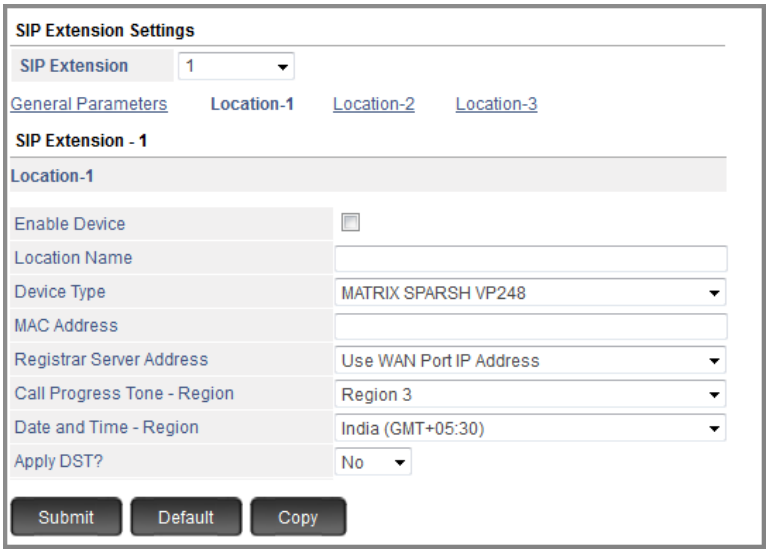
If you have registered Matrix VARTA ADR100 and VARTA AMP100 Mobile UC Clients in any of the locations, refer to “Configuring Matrix VARTA ADR100/AMP100 UC Clients”.

If you have registered MATRIX VARTA WIN200 Desktop UC Client in any of the locations, refer to “Configuring Matrix VARTA WIN200 UC Client”.

- Click **Location 1**.



- The settings of the phone at **Location 1** appear.



- Select the **Enable Device** check box. Default: Disabled.

- Enter the **Location Name** for the phone to identify the phone. Location name may be the place where the phone is located (e.g.: Head office, branch, residence). The Location Name may consist of 18 characters (maximum). Default: Blank.

- Select **MATRIX SPARSH VP248** as the **Device Type** at this location.

- Enter the **MAC Address**[142] of the SPARSH VP248 connected at this location in hexadecimal format: 00:1b:09:XX:XX:XX. Default: blank.

  SARVAM UCS validates the Extended Phone on the basis of the MAC Address, and provides configuration on validation.

  As SARVAM UCS allows registration of the SIP Extension from three different locations, it identifies the SIP Extension in each location by the programmed MAC address.

- Select the appropriate **Registrar Server Address** to register the SPARSH VP248 with the SIP Registrar of SARVAM UCS, according to your installation scenario:

  - If the SPARSH VP248 is connected on the WAN network, select **Use WAN Port IP Address** as Registrar Server IP Address.

  - If the SPARSH VP248 is connected on the LAN network, select **Use LAN Port IP Address** as Registrar Server IP Address.

  - If the SPARSH VP248 is connected in the Global Network and SARVAM UCS is located behind a Router, or behind a NAT Router and STUN is programmed, select **Use Router/STUN's IP Address** as Registrar Server IP Address.

142. *MAC address is the address of the electronic hardware devices such as a computer, which is hard-coded into the device during manufacture and cannot be modified. No two devices can have similar MAC address and thus it uniquely identifies your phone. MAC address is assigned as per the IANA standard. The MAC Address of the phone will be used as source MAC address on all Ethernet frames.*

Make sure you configure either the **Router's Public IP Address** or **Simple Traversal of UDPs through NATs (STUN)** in Network Parameters. For details, see "Configuring Network Parameters".

- If Dynamic DNS is configured in the Network Parameters, select **Use Dynamic DNS Host Name** as Registrar Server IP Address.

  By default, Use WAN Port IP Address is selected as the Registrar Server IP Address.

- To set the call progress tone generation standards of the country where the SPARSH VP248 is installed, select the **Call Progress Tone - Region**. Default: Region 1.

  See "Call Progress Tones" to know more.

- To display the Date and Time of the country where the SPARSH VP248 is installed, select the **Date and Time - Region**. Default: India.

- If you want to enable Daylight Saving Time (DST) on the phone, set **Apply DST?** to Yes. Default: No.

  The Daylight Saving Time convention followed in the country/region you selected will be automatically applied. The SPARSH VP248 will change its date and time settings according to the DST convention of the selected country/region.

- Select the **CO CLIP Pattern** for the SPARSH VP248. This is the type of Calling Line Presentation on the phone for incoming calls from trunks. You can select any of these options:
  - **Name Only** (only the name of the caller will be displayed).
  - **Number Only** (only the number of the caller will be displayed).
  - **Number + Name** (both the name and the number of the caller will be displayed).

  Default: Number + Name.

- Select the **Language** for the SPARSH VP248. Default: English.

  SARVAM UCS provides language support for English, French, German, Spanish, Portuguese, and Italian on the SPARSH VP248. When you select any of these languages, all the prompts and command strings will appear in the selected language.

SIP Extension users can change the language by accessing and navigating through the phone menu.

The SA can change the Language by logging into the SA Jeeves.

- Select a **Ringer Mode** for the phone from the four options:
  - Ring immediately (it rings immediately as a fresh calls lands on the phone).
  - Ring if idle (rings only if the phone is idle).
  - Ring after a delay (if the call is still not answered).
  - Silent.

  Default: Ring Immediate.

- If you selected *Ring after a delay* as Ringer Mode, set the **Ring Delay Timer (sec)**, if required, to the desired value.
  The Ring Delay Timer is the time in seconds the system waits on receiving a call before ringing on the phone. The range of this timer is 0 to 99 seconds. Default: 10 seconds.

- If you want to enable *Ringer Auto Acknowledge* mode, set the **Acknowledge Timer** (sec) to the desired value.

  The Ringer Auto Acknowledge mode determines when to stop the ring on the phone. There are two options for Ringer Auto Acknowledge:
  - Stop only when the call is answered.
  - Stop after a delay.

  To stop the ring on the phone after a delay, the Acknowledge Timer must be configured. The range of this timer is 01 to 99 seconds. Default: 00 seconds.

  To stop the ring only when the Call is answered or manually acknowledged, the Acknowledge Timer must be set to '00'. By default, Ring Auto Acknowledge is turned OFF.

- To assign the Ring Destination for the SPARSH VP248, select the desired destination for **Play Ring on**. You may choose
  - **Speakerphone:** The ring will be played on the Speakerphone.
  - **Headset:** The ring will be played on the Headset.

  Default: Speakerphone.

  When you select the Headset as the destination, make sure that you set the flag '*Headset Connected?'* to Yes, connect a Headset to the SPARSH VP248.

- Select the desired **Ring Tune** according to your/SPARSH VP248 user's preference. Default: 1.

- Set the **Ringer Volume** to the desired level, from 0 to 7, according to your preference. Default: 4.

- To increase/decrease the volume of outgoing speech (Transmit Gain) on the handset of the SPARSH VP248, set the **Handset Transmit Volume Level** to the desired level, from 0 to 7. Default: 4.

- To increase/decrease the volume of incoming speech (Receive Gain) on the handset of the SPARSH VP248, set the **Handset Receive Volume Level** to the desired level, from 0 to 7. Default: 4.

- To increase/decrease the volume of outgoing speech (Transmit Gain) on the headset of the SPARSH VP248, set the **Headset Transmit Volume Level** to the desired level, from 0 to 7. Default: 4.

- To increase/decrease the volume of outgoing speech (Receive Gain) on the headset of the SPARSH VP248, set the **Headset Receive Volume Level** to the desired level, from 0 to 7. Default: 4.

- To change the Transmit Gain of the Speakerphone MIC Volume, set **Speaker Transmit Volume Level** to the desired level, from 0 to 7. Default: 4.

- To change the Receive Gain of the Speakerphone MIC Volume, set **Speaker Receive Volume Level** to the desired level, from 0 to 7. Default: 4.

- To use a Headset with the SPARSH VP248, select the **Headset Connected?** check box**.** Default: Disabled.

  Make sure that you connect a Headset to the SPARSH VP248, if you enable this option.

- Select the **Auto Answer** check box to enable this feature on the SPARSH VP248. Default: Disabled.

When you set the “Auto Answer” feature on the SPARSH VP248, the phone goes OFF-Hook automatically after a preset period of time, without the extension user having to pick up the handset or press the speaker or headset key. When you enable Auto Answer, you must configure the Auto Answer Timer.

- If you enabled Auto Answer on the phone, set the **Auto Answer Timer (sec)** to the desired value.

  This timer defines the time in seconds that the SPARSH VP248 should wait before going OFF-Hook to auto answer a call. The range of this timer is 1 to 9 seconds. Default: 1 second.

- Adjust the Backlight brightness of the phone’s LCD display, by setting the **LCD Backlight Level** to the desired value, from 1 to 4. Default: 3.

- Set the **Back Light Off Timer (sec)** to the desired value, if required, from 000 to 999 seconds. Default: 10 seconds.

- Set the **LCD Contrast Level** to a level from 1 to 4 that is comfortable to you. Default: 3.

DSS Key Settings

- You can select the desired key template — Operator, Executive1, Executive2, Executive3, Hotel Attendant, Guest or any other template you added. See “Customizing Extended IP Phone Templates using Jeeves” for more details.

  OR

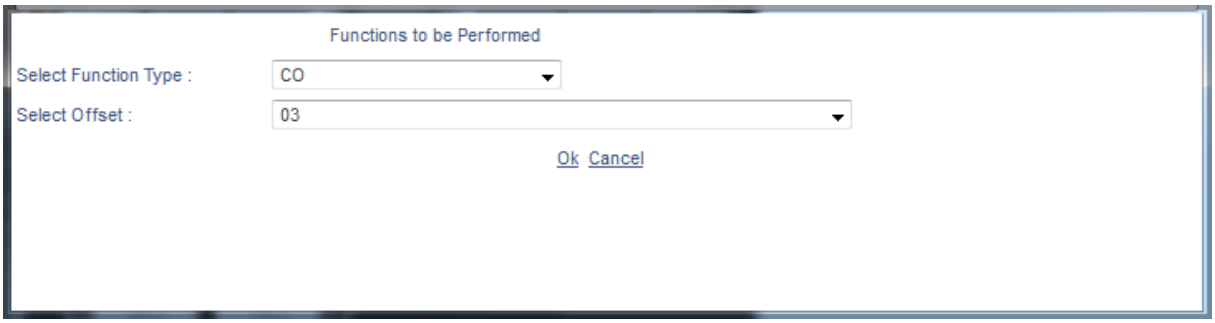
- You can personalize the key map of the SPARSH VP248 for this location. To do so,

  - Select **Personalized** as the **Key Template** option.

  - Click **Key Settings**.

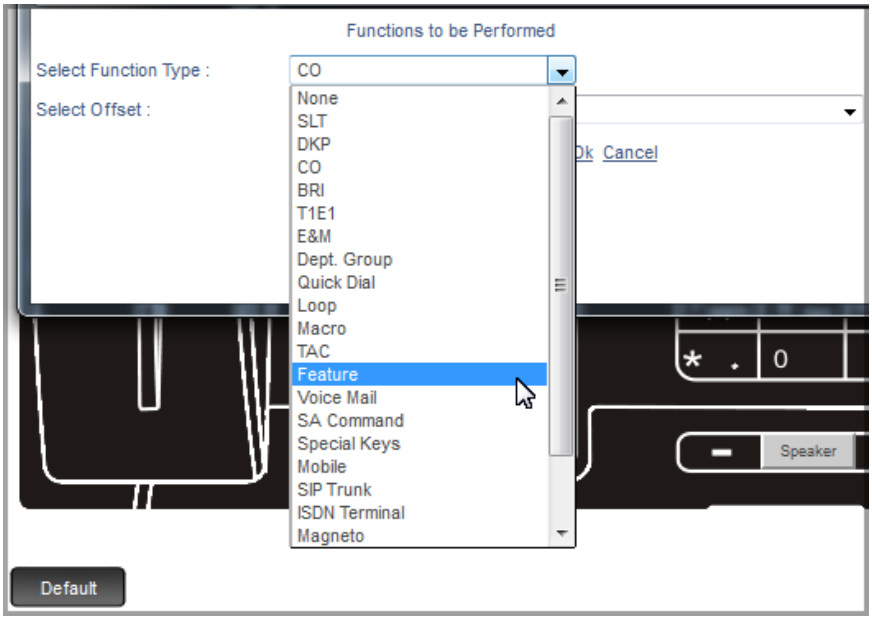
  - The key map of the Extended Phone opens in a new window on your screen.

- Click the key you want to configure. For example, **CO3**.
  The **Functions to be Performed** by the key opens in a new window.



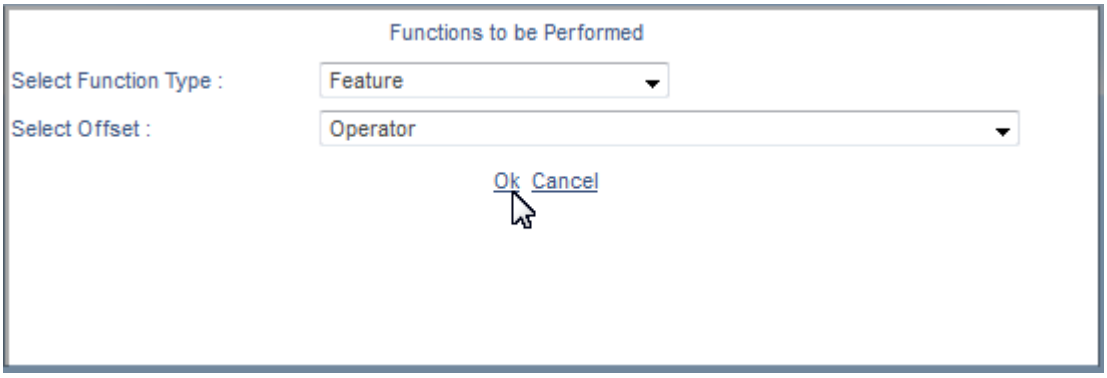
- In the **Select Function Type** list, select the function to be performed by the key. For example, you want to use the key to call the **Operator**.

The Operator function is a Feature, so select the option **FEATURE** from the **Select Function Type** list box.



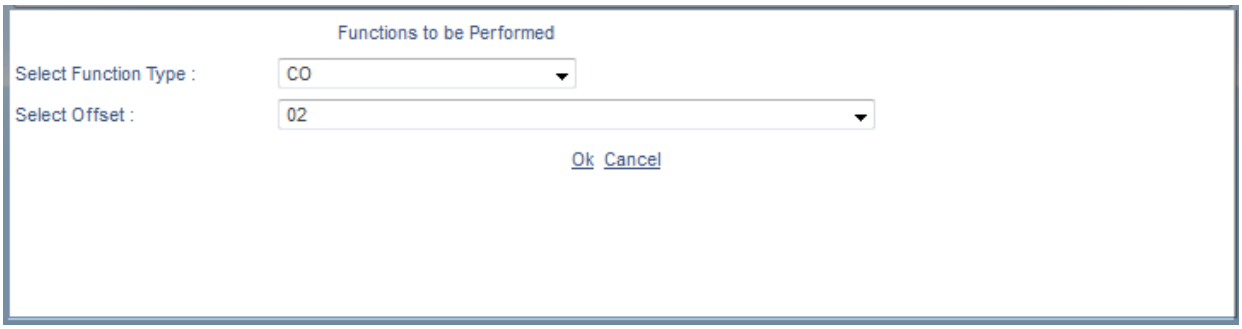
From the **Select Offset** drop down list, all the features that can be assigned to keys are listed.

- Select **Operator** from the list of features in the **Select Offset** box.
- Click **OK**.

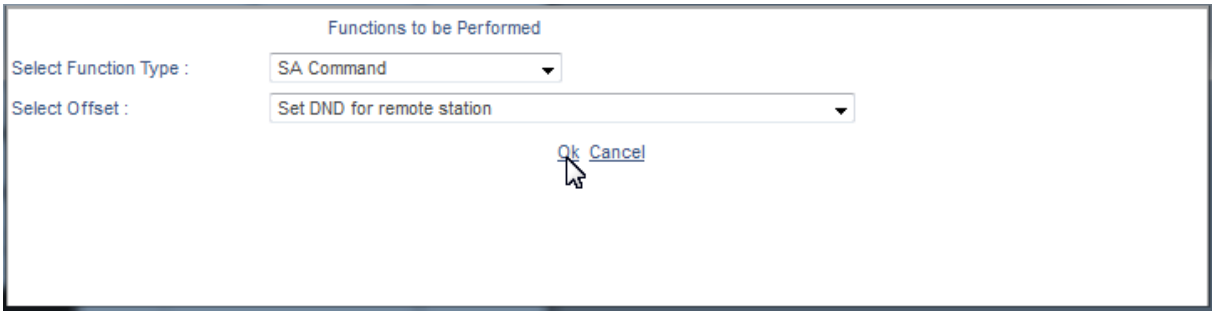


The *Operator* feature appears on the key label.

- To take a second example, if you want to assign **Remote DND** to the key currently assigned **CO 2** key, click the key.



- In the **Select Function Type** list box, select the option **SA Command**, as Remote DND is a System Administrator (SA) Command.



- In the **Select Offset** box, select the option **Set DND for remote station**.
- Click **OK**. The box closes. Remote DND feature will appear in abbreviated form as *R-DND* on the key label.

- Follow the same instructions to assign features to other DSS keys. Selecting the appropriate Function Type and the Offset for each feature/function.

  If you want assign a feature, select **FEATURE** as function type, and select the desired feature as Offset.

  If you want to use the key to call a DKP or a SIP extension, select **DKP** or **SIP Extension** as Function Type and select the number of the extension as Offset.

To assign direct access to a mobile trunk, select **MOBILE** as Function Type and the desired port number **1** or **2** as Offset.

To assign direct access to a SIP Trunk, select **SIP** as Function Type and the desired trunk number from **1** to **4** as Offset.

Click **OK**, each time you select a Function Type and Offset in the dialog box.

You can reinstate default key assignment any time, by clicking the **Default** button at the bottom of the window.

- When you complete assigning functions to keys, close the window.

- If you assign/re-assign functions to the following keys, the Phone will restart:
  - Speaker
  - Headset
  - Ringer
  - Acknowledge
  - Local Menu

If you have upgraded your SPARSH VP248 to an Extended IP Phone with firmware V5Rx, the key labels listed in the table below will have the following functions:

Key Label	Function assumed with Firmware V5Rx
Fwd Busy	Voice Mail
Fwd NR	Auto Call Back
DND	Forward
Reject	Release

RTP Port

- Define **RTP Port**:

  - **RTP Listening Port:** This is the port on which the SPARSH VP248 listens for RTP messages over UDP. This port is also used as the source port for sending RTP packets. This port is also used as the source port for sending RTP packets to the remote peer. The valid range for this port is 1025-65278. Default: 8000.

Quality of Service

- Set the **SIP Quality of Service (QoS)** for SIP signaling as:

  - **SIP DiffServe/ToS**. Valid range is 00 to 63. Default: 26.
    OR
  - **RTP DiffServe/ToS**. Valid range is 00 to 63. Default: 46.

NAT Keep Alive

- If the SPARSH VP248 is connected behind a NAT router, configure **NAT Keep Alive**.

- Select the check box **Enable NAT Keep Alive** to send Keep Alive messages periodically to refresh the binding in the NAT router. Default: Disabled.

- Define as **Interval (sec)**, the time period, from 001 to 999 seconds, after which the phone should send Keep Alive message. Default: 120 seconds.

  The time period you define should be less than the binding timer of the router.

Timers

- Set the following **Timers** to the desired value, where required:

  - **SIP INVITE Timer (sec):** This is the time in seconds that the phone waits for a response from the called party after ending INVITE message. This timer starts after sending INVITE message to the called party and stops on receipt of the provisional response or the final response or when the user disconnects the call. On expiry of the timer, the phone terminates the call process and gives an error tone to the user. The range of the SIP INVITE TIMER is 10-180 seconds. Default: 30 seconds.

  - **SIP Provisional Timer (sec)**: This is the time in seconds that the phone waits for final response after receiving the provisional response from the called party. This timer starts on the receipt of the provisional response from the called party and stops on receipt of the final response from the called party or when the user disconnects the call. On expiry of the timer, the IP phone terminates the call process and gives error tone to the user. The range of SIP Provisional Timer is 10-180 seconds. Default: 60 seconds.

  - **General Request Timer (sec):** This is the time in seconds for which the phone waits for response of a transaction request. This timer starts on initiating a transaction. This timer stops on receipt of a response for the request. On expiry of the timer, the phone clears the transaction. This timer is used for Registration request, etc. The range of the General Request Timer is 10-60 seconds. Default: 20 seconds.

Debug

- To debug using Syslog Client supported by the SPARSH VP248, configure Debug parameters:

  - Select the **Enable Debug?** check box. Default: disabled.

    When the Debug flag is enabled, the phone will send the debug messages to the Syslog Server IP address. Debug report can be viewed on the Syslog Server or any other application which can capture the Syslog messages/debug sent by the phone.

  - Enter the IP Address and port of the remote Syslog Server and as **Syslog Server Address and Server Port**.

    The address of the Listening Port of the Syslog Server is from 1025-65535;514. Default: 514. Syslog uses the UDP as transport protocol and listens on the port 514 (the default listening port).

  - You may select the **Debug Level** from the following options, by selecting the respective check box:
    - SIP
    - System
    - Hardware
    - Call
    - Network
    - VoPP

You may select any or all of these debug levels. The Syslog Client will send only the debug messages for the selected level to the remote server on the IP network. For example, if the debug log of 'Call's is required, you can select this option, and disable all others.

- Click **Submit** to save settings.

- If you have completed the configuration of the SPARSH VP248 Settings at Location 1, follow the same steps as described above to configure the SPARSH VP248 at Location 2 and Location 3.

  However, if you want to replicate the configuration of SPARSH VP248 Settings same as Location 1 to Location 2 and Location 3, you can use the **Copy** button present at the bottom of the page. To know more, refer to “Copy Parameter Values”.

If you wish to Copy only the configurations from a location to all other locations where the same IP Phone is connected, make sure you only clear the Device Type check box.

If you wish to copy the Device Type as well as all the configurations from a location to all the SIP Extensions, make sure all the check boxes are selected.

The SA can change the Language by logging into the SA Jeeves.

When you change any of the parameters listed below in the SIP Extension at Location 1, 2, 3, the phone will restart automatically, if registered:

- Use SIP Extension
- SIP ID
- Authentication ID
- Authentication Password
- Registrar Server IP Address
- MAC Address
- Enable Device
- Device Type
- Key Map in the Key Template assigned to phone
- Call Progress Tone
- Date and Time
- Apply DST?
- QoS
- RTP Ports
- NAT Keep Alive
- SIP Timers
- The SE Password of SARVAM UCS is changed
- Specific parameters in VoIP Parameters are changed
- Specific parameters Network Port parameters are changed
- You restart the System
- Set the System to Default

Configuring Matrix SPARSH VP310

SPARSH VP310, the Executive IP Phone is engineered to offer a contemporary design with crystal-clear audio and feature-rich capabilities at economical price. It functions like SPARSH VP248,the proprietary SIP-based IP Phone for SARVAM UCS, of Matrix. To know the list of features supported, refer to “SARVAM UCS Features Supported in Terminals”.

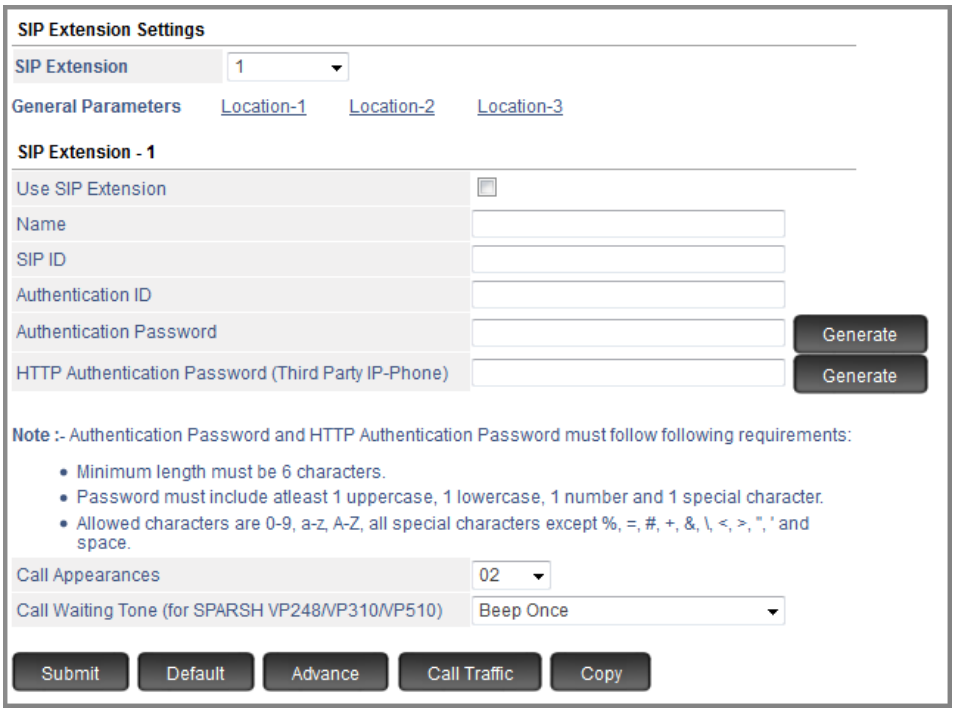
For instructions on how to use SPARSH VP310, refer to the common *EON48_310_SPARSH VP248_310 User Guide.*

To be able to use SPARSH VP310[143] - Extended IP Phone, you must configure the following:

- SIP Extension General Parameters, see “Configuring SIP Extension General Parameters”.
- SIP Extension Settings, see “Configuring SIP Extension Settings using Jeeves”
- Extended IP Phone Settings, see “Configuring Matrix Extended Phone Settings using Jeeves”
- Voice Mail Settings, if you want to provide mailbox facility to the extension. See “Extension Voice Mail Settings”.

Configuring SIP Extension Settings using Jeeves

- Log into Jeeves as System Engineer.
- Under **Configuration**, click **VoIP Configuration**.
- Click **SIP Extension Settings**.



The page of SIP Extension 001 opens.

143. *SARVAM UCS supports only IPv4 Addresses for registering SPARSH VP310.*

- You may select the **SIP Extension** number you want to configure.

  The parameters of the SIP Extension number you selected will appear on this page.

For SARVAM UCS upto 999 SIP Extensions can be registered with the system.

- Select the **Use SIP Extension** check box to enable the SIP extension. Default: disabled.

- In the **Name** field, enter a name for the SIP Extension, which may be the name of the person who will use the SIP Extension or the name of a Department. The name you enter here will be displayed as the Caller ID of the SIP Extension on the remote user's phone, when the SIP Extension user makes calls.

  The name may consist of a maximum of 18 alphanumeric characters.

- Enter the **SIP ID** for the extension. The SIP ID is necessary for registering the SIP Extension with the Registrar of the system. It is the number with which you can call the SIP Extension. Any extension user of the SARVAM UCS can call a SIP Extension by dialing the SIP ID assigned to the SIP extension. SIP ID of each SIP Extension must be a unique number string of a maximum of 6 digits. Any combination of digits from 0 to 9 and the characters * and # are allowed. You cannot assign the same SIP ID to more than one extension.

  To assign SIP IDs according to your preference and requirement to a range of SIP Extensions, see “Assigning Access Codes to a Range of Extensions”.

  By default, the SIP IDs are Blank.

The SIP ID will be set to default value (blank), when you restore the default settings of the system.

- In **Authentication ID**, enter the number which you want the system to use for user authentication of the SIP messages received from the SIP Extension. You cannot keep this field blank and it must be unique for each SIP extension. The number may be a string of maximum 6 alphanumeric characters. All ASCII characters except < > and “ (double quote) are allowed. Default: Blank.

Make sure the User ID configured in “Digest Authentication” does not conflict with the Authentication ID configured above.

- In **Authentication Password**, enter the password manually or click **Generate** to automatically generate a unique password. This password will be used by the system to authenticate the SIP messages received from the SIP Extension. To avoid unauthorized access, we recommend you to change the password regularly. Make sure it is strong and is kept confidential. When you enter the password manually, the password must:

  - be of minimum 6 characters and can be a maximum of 12 characters.
  - include atleast one upper-case, one lower-case, one number and one special character.
  - all ASCII characters (except Percentage **%**, Hash **#,** Equal to **=,** Plus **+**, And **&**, Backslash **\,** Less than **<,** Greater than **>,** Apostrophe **’,** Double Quote **"** and **Space**) are allowed.
    Default: Blank.

  To provide additional security, when the Authentication fails 10 times consecutively due to wrong Authentication ID / Authentication Password, the system will blacklist the IP Address and Port for registration of this SIP Extension. However, you can remove the IP Address from the Blacklist IP Address

list. See “Black List IP Address - SIP Extensions” for more details. This activity will be logged in the “System Activity Log” as well as “Simple Network Management Protocol (SNMP)”.

Make sure you note down or copy the Authentication Password in a confidential file.

- In **Call Appearances**, define the maximum number[144] of simultaneous incoming calls that the SIP Extension user should be allowed to receive. You can set up to 10 call appearances for a SIP Extension. Default: 2.

  When Call Appearance is set to 2, the SIP Extension can receive 2 calls at a time.

- During an on-going conversation, if there is a second incoming call, the system plays beeps to indicate the second incoming call. You can set the frequency of the **Call Waiting Tone (for SPARSH VP248/VP310/ VP510)** beeps as per your requirement. You can select from the following options:
  - Off
  - Beep Once
  - Beep until Answered

  Default: Beep Once.

  However, when an ongoing call is being taped or recorded, the call waiting tone for any new incoming call will not be played.

- Under **Auto Sign-In,** enter the **Email ID** of the extension user you wish to store. The Email ID can be a maximum of 64 characters. The Email ID is used for various server features.

Auto Sign-In parameters — Send Configuration Mail and Mail Status, *are applicable only for Mobile Clients — VARTA ADR100, VARTA AMP100 applications.*

*The **Send Configuration Mail** button will appear only after you have enabled the SIP Extension and configured the SIP ID, Authentication ID and Password.*

- Under **Authentication**, enable Authentication of any or all of the following SIP Message Options by selecting the respective check boxes:
  - **INVITE Request**
  - **SUBSCRIBE Request**

  By default, the SIP Message Options INVITE and SUBSCRIBE are enabled.

Make sure that the Authentication ID for the SIP Extension has been programmed.

- For secure conversations over SIP, enable **SRTP Mode**. The SARVAM UCS supports the following options:
  - **Disable:** SARVAM UCS uses normal RTP for transporting the speech packets.

  - **Optional:** SARVAM UCS uses SRTP for transporting the speech packets. If the remote user does not support SRTP, SARVAM UCS will use normal RTP for transporting the speech packets.

    - If you select this option, you must configure the **SRTP Media Type**. You may select **AVP** or **SAVP**. By default, AVP is selected as the SRTP Media Type.

144. *For the calls that are routed through the CPU, the number of Vocoder channels that will be supported would be as per the license you purchase.*

- **Forced:** SARVAM UCS uses only SRTP (SAVP) for transporting the speech packets. If the remote user does not support SRTP, SARVAM UCS will reject incoming calls from and drop outgoing calls made to such users.

  By default, SRTP Mode is Disabled.

- Assign a **SIP Hardware Template** to the SIP Extension. Default: 01. The "SIP Hardware Template" contains voice quality related features such as Voice Codec selection, Tx and Rx Gains, Echo Cancellation, Jitter Buffer and Fax-over-IP options and related parameters

  There are 32 different templates to choose from. Each template can also be altered to suit your requirement and preferences. By default, Template number 01 assigned to all SIP Extensions as well as to SIP Trunks.

  Check if the values in this template fulfill requirements of the SIP Extension. If Template 01 fulfills the feature requirements, retain Template 01.

  If a different set of SIP hardware features are to be allowed to this SIP Extensions, prepare another template and assign it to this extension. To do this,

  - Under **VoIP Configuration**, click the **SIP Hardware Template** link.
  - Select a Template number, for example 02.
  - Customize Template number 02 and click **Submit**.
  - Return to the **SIP Extension Settings** page.
  - Select the number of the Template you customized, Template 02 in the **SIP Hardware Template** field.

  Also see the topic "SIP Hardware Template" to know more about customizing the templates and applying on the SIP Extensions.

- Assign a **Station Basic Feature Template** to the SIP Extension. Default: The "Station Basic Feature Template" has a set of features like Time Table, Class of Service, Toll Control, Operator, Storage of Incoming and Outgoing Calls, Outgoing Trunk Bundle groups. There are 50 different templates to choose from. Each template can also be altered to suit your requirement and preferences.

  If the default Station Basic Feature Template 01 fulfills the feature requirements of the SIP Extension ("Class of Service (COS)", "Toll Control", "OG Trunk Bundle Group", etc.) retain this template, you may also customize this template. If you want to assign a different set of features to this SIP Extension, prepare a different Station Basic Feature Template and apply it to this extension. To do this,

  - Under **Configuration**, click the **Station Basic Feature Template** link.
  - Select a Template number, for example 05.
  - Customize Template number 05 and click **Submit**.
  - Return to the **SIP Extension Settings** page.

- Select the number of the Template you customized, Template 05, in the **Station Basic Feature Template** field.

- Click **Submit** to save changes.

Also, see the topic "Station Basic Feature Template" to know more about customizing the templates and applying on extensions.

- Assign a **Station Advanced Feature Template** to the SIP Extension. Default: Template 01. The "Station Advanced Feature Template" has a set of advanced features for extensions such as Alarm Notification settings, Routing of Incoming Auto Attendant Calls, Call Duration Control, Floor Service, etc. There are 50 different templates to choose from. Each template can also be altered to suit your requirement and preferences.

  Check if the default template fulfills the feature requirements of the SIP Extension by clicking the **Station Advanced Feature Template** link.

  You may retain this template and customize it further, or customize another template if a different set of features are to be allowed to this SIP Extension. To customize/prepare another template,

  - Under **Configuration**, click the **Station Advanced Feature Template** link.

  - Select the Template number, for example 02, and customize this template.

  - Click **Submit** and return to the 'SIP Extension Settings' page.

  - In the **Station Advanced Feature Template** field, select the number of the template you customized.

  - Click **Submit** to save changes.

  Also see the topic "Station Advanced Feature Template" for instructions on customizing these templates and applying them on the extensions.

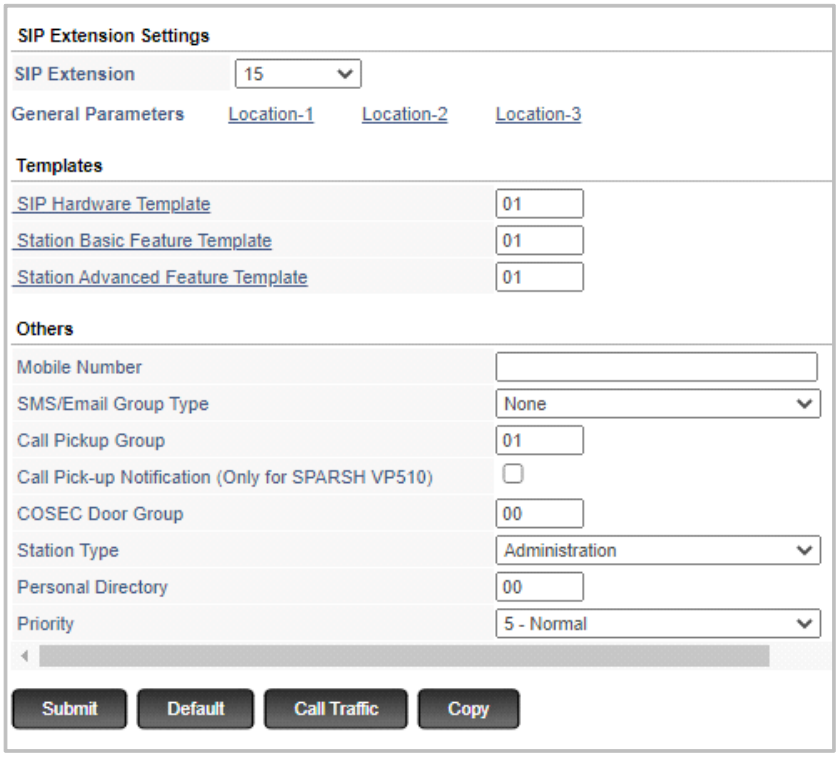
- In **Voice Mail Settings**, click the Voice Mail Settings link. The respective Extension Voice Mail Settings window will open. You may edit the parameters. For details, see "Extension Voice Mail Settings".

The Voice Mail Settings link will be visible only if you have configured the respective SIP ID.

Click **Close** to close the window.

Advanced Configuration Parameters

- If you want to provide other features like Personal Directory, Priority, or assign a Station Type to the SIP Extension, click the **Advanced** button at the bottom of the page.



- Enter the **Mobile Number** of the extension user you wish to store. The Number can be a maximum of 16 digits.

- You can assign the extension user to a Group. Select the desired **SMS/Email Group Type** from the list.The system clubs together extension users assigned the same Group. Default: None. For details, see “SMS/Email Group”.

- Assign the SIP Extension to a **Call Pick-up Group**, if required. Default: 01

  Call Pick Up allows the SIP Extension user to 'pick up' (answer) calls ringing on any other extension, by dialing a feature code, without physically going to the ringing extension. It also allows incoming calls for the SIP Extension to be answered by the other extensions assigned the same Call Pick-Up group.

  For this to work, both the ringing extension and the extension picking up the call must be in the same 'Call Pick Up Group'. Refer “Call Pick Up” for instructions on how to create groups. You can create as many as 99 groups numbered from 01 to 99.

  Enter the number of the Call Pick-Up Group you created for this SIP Extension in this field.

- Select the **Call Pick-up Notification (Only for SPARSH VP510)** check box, if you want the Call Pick-up Notification to be displayed on the phone LCD. The Call Pick-up Notification will be displayed for internal as well as external calls. The notification will be displayed on SPARSH VP510 only if the ringing extension is in the same Call Pick-up Group. The notification will be displayed only when SPARSH VP510 is in idle or dial state. The notification will display the name/number of the Caller along with the name/number of the Called. For example, if the name and number of the Caller is ABC, 2001 and the name and number of

Called is XYZ, 1001, the LCD notification will be displayed as ABC -> XYZ. Along with the LCD display the phone back-light will also be lit.

For this to work, make sure you have enabled Call Pick-up in COS as well as assigned a Call Pick-up Group to the extension. Refer to "Call Pick Up" and "Class of Service (COS)". Call Pick-up Notifications will be displayed for DKP, SLT as well as SIP Extensions and for calls landing through CO, SIP as well as T1E1 Trunks. For details of the Notification, refer to the EON510_SPARSH VP510 V2 User Guide.

- You must assign the extension user to a **COSEC Door Group** for COSEC Integration. The users in the same group must be assigned the same group. You can create as many as 50 groups numbered from 00 to 50. Users who are assigned COSEC Door Group '00' are not a part of any group. See "COSEC Integration" for more information.

- If this is an Operator extension and you want the system to play beeps during a conference to the participants, to indicate the presence or absence of the Operator, select the **Station Type** as **Assistant**.

  If you are using the system in the *Hotel Mode*, select the **Station Type** for the SIP Extension as **Administration/Assistant** or **Guest**. The system will consider the options Administrator and Assistant as same.

- You may assign a **Personal Directory** number to the SIP Extension. Default: 00.

  A Personal Directory is a list of 25 frequently dialed numbers, each of which are stored by Index number (location code), Name and Trunk Access Codes ("Out Going Trunk Bundle Group Index"). The Personal Directory is necessary for using the features "Abbreviated Dialing" and "Dial By Name".

  When a Personal Directory is assigned to a SIP Extension, make sure you also configure this directory. The Personal Directory can be programmed by the SIP Extension users and by the System Engineer. Refer the topic "Abbreviated Dialing" for instructions on programming the Personal Directory.
  If Personal Directory is not to be assigned, enter 00 in this field.

- Select a **Priority** Level for the SIP Extension from 1 to 9. Default; 5-Normal.

  Each extension of the SARVAM UCS is assigned a Priority Level starting from 1, 2, 3... to 9, with '1' being lowest Priority and '9' being highest Priority. Whenever an extension (phone) with higher priority calls an extension with lower priority, a triple ring is placed on the called extension. To know more, read the feature description "Priority".

  If this SIP extension is assigned to Operator, you may want to set a higher priority for this extension.
- Click **Submit** to save your SIP Extension Settings.

Configuring Matrix Extended Phone Settings using Jeeves

You can register three Matrix Extended IP Phones/Soft Clients at three different locations as a single SIP Extension. You can connect/register the same or different types of Extended Phones—SPARSH VP248, SPARSH VP310, SPARSH VP510, SPARSH VP330, SPARSH VP210, Matrix VARTA ADR100 Mobile UC Client, VARTA AMP100 Mobile UC Client or MATRIX VARTA WIN200 Desktop UC Client — at each location. In this case we assume that SPARSH VP310 is connected at Location 1, 2 and 3.

If you want to use more than one SPARSH VP310 Extended IP Phones as a SIP Extension, configure their settings at **Location 1**, **Location 2** and **Location 3**.

If you have connected SPARSH VP248 at any of the locations, refer to “Configuring Matrix Extended Phone Settings using Jeeves”.

If you have connected SPARSH VP330 at any of the locations, refer to “Configuring Matrix SPARSH VP330”.

If you have connected SPARSH VP510 at any of the locations, refer to “Configuring Matrix SPARSH VP510”.

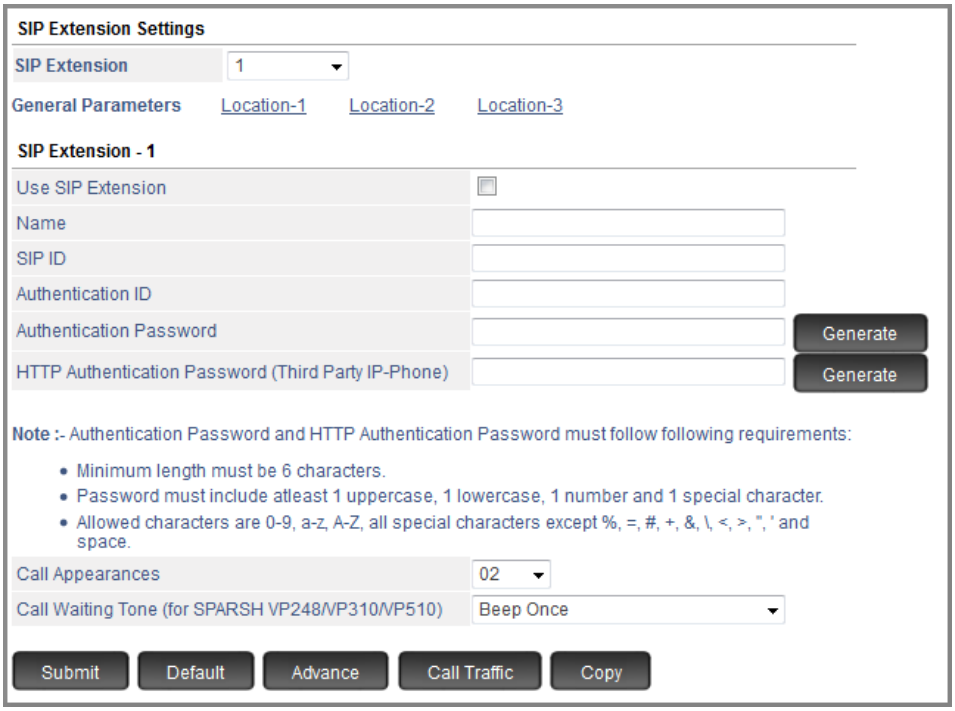
If you have connected Extended SPARSH VP710 at any of the locations, refer to “Configuring Matrix Extended SPARSH VP710”.

If you have connected Extended SPARSH VP210 at any of the locations, refer to “Configuring Matrix SPARSH VP210”.

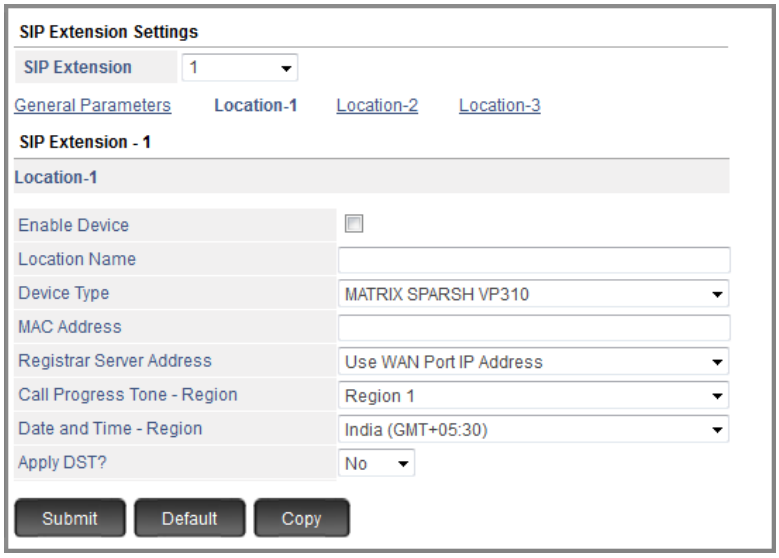
If you have registered Matrix VARTA Mobile UC Clients in any of the locations, refer to “Configuring Matrix VARTA ADR100/AMP100 UC Clients”.

If you have registered MATRIX VARTA WIN200 Desktop UC Client in any of the locations, refer to “Configuring Matrix VARTA WIN200 UC Client”.

- Click **Location 1**.



- The settings of the phone at **Location 1** appear.



- Select the **Enable Device** check box. Default: Disabled.

- Enter the **Location Name** for the phone to identify the phone. Location name may be the place where the phone is located (e.g.: Head office, branch, residence). The Location Name may consist of 18 characters (maximum). Default: Blank.

- Select **MATRIX SPARSH VP310** as the **Device Type** at this location.

- Enter the **MAC Address**[145] of the SPARSH VP310 connected at this location in hexadecimal format: 00:1b:09:XX:XX:XX. Default: blank.
  SARVAM UCS validates the Extended Phone on the basis of the MAC Address, and provides configuration on validation.

  As SARVAM UCS allows registration of the SIP Extension from three different locations, it identifies the SIP Extension in each location by the programmed MAC address.

- Select the appropriate **Registrar Server Address** to register the SPARSH VP310 with the SIP Registrar of SARVAM UCS, according to your installation scenario:

  - If the SPARSH VP310 is connected on the WAN network, select **Use WAN Port IP Address** as Registrar Server IP Address.

  - If the SPARSH VP310 is connected on the LAN network, select **Use LAN Port IP Address** as Registrar Server IP Address.

  - If the SPARSH VP310 is connected in the Global Network and SARVAM UCS is located behind a Router, or behind a NAT Router and STUN is programmed, select **Use Router/STUN's IP Address** as Registrar Server IP Address.

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145. MAC address is the address of the electronic hardware devices such as a computer, which is hard-coded into the device during manufacture and cannot be modified. No two devices can have similar MAC address and thus it uniquely identifies your phone. MAC address is assigned as per the IANA standard. The MAC Address of the phone will be used as source MAC address on all Ethernet frames.

Make sure you configure either the **Router's Public IP Address** or **Simple Traversal of UDPs through NATs (STUN)** in Network Parameters. For details , see "Configuring Network Parameters".

- If Dynamic DNS is configured in the Network Parameters, select **Use Dynamic DNS Host Name** as Registrar Server IP Address.

  By default, Use WAN Port IP Address is selected as the Registrar Server IP Address.

- To set the call progress tone generation standards of the country where the SPARSH VP310 is installed, select the **Call Progress Tone - Region**. Default: Region 1.

  See "Call Progress Tones" to know more.

- To display the Date and Time of the country where the SPARSH VP310 is installed, select the **Date and Time - Region**. Default: India.

- If you want to enable Daylight Saving Time (DST) on the phone, set **Apply DST?** to either **Manual** or **Scheduled** as per your requirement.

  When you select **Scheduled** as the DST option, the Real Time Clock of SARVAM UCS is advanced and set backward automatically according to the DST convention of the country/region where the IP Phone is installed.

  Scheduled DST Adjustment is useful in countries/regions where DST Time is fixed, such as in Europe, USA and Canada, without yearly variations.

  SARVAM UCS supports 18 DST Types for Scheduled DST Adjustment. To know more, refer to "Daylight Saving Time (DST)". To know more about Scheduled DST assigned for the respective region, refer to *Time Zone* in "Default Settings".

  When you select **Manual** as the DST option, the Real Time Clock of SARVAM UCS is advanced manually and set backward automatically according to the DST convention of the country/region where the IP Phone is installed.

  Manual DST Adjustment is to be used in regions/countries that have no fixed DST Convention and where yearly variations in DST practices are likely.

  When DST option is set as 'Manual', you must set the DST Start and the DST End time, that is, the time at which the clock is to be advanced and the time at which the clock is to be delayed. To do so,

  - In **Time Offset,** enter the time you wish to forward or backward the DST start time with.

  - In **DST Type**, select the desired option: **Date-Month Wise** OR **Day-Month Wise**.

    If you select '**Date-Month Wise**' in **DST Type**, you should now select the desired options in each of the following to specify the **DST Start** details.

    - **Date**: Select the date on which DST begins (1-31).
    - **Month**: Select the month when DST begins (January-December).
    - **Time (Hours)**: Select the time at which DST will begin to change.The time mode is of 24 hours ranging from 00 to 23 hours.
    - **Time (Minutes):** Select the time at which DST will begin to change.The time mode is of 60 minutes ranging from 00 to 59 minutes.

Similarly, in the **DST End** configure the desired DST End Time details.

If you select **'Day-Month Wise'** in **DST Type**, you should now select the desired options in each of the following to specify the **DST Start** details.

- **Ordinal**: Select the Ordinal number of the day of the month, that is, the 1st, 2nd, 3rd, 4th, 5th day, when DST begins.
- **Day:** Select the day of the month - Sunday, Monday, Tuesday, Wednesday, Thursday, Friday, Saturday- when DST begins.
- **Month**: Select the month when DST begins (January-December).
- **Time (Hours)**: Select the time at which DST must begin to change.The duration is of 24 hours ranging from 00 to 23.
- **Time (Minutes):** Select the time at which DST must begin to change.The duration is of 60 minutes ranging from 00 to 59.

Similarly, in **DST End** configure the DST End Time details.

Once the DST Ends, the time of the IP Phone is set back to the Standard time automatically.

When the DST of a particular country starts or ends on the Last Sunday or any other day, for example, the last Tuesday, last Friday of the month, always set the Ordinal Number as '5th'.

- If you want to enable Daylight Saving Time (DST) on the phone, set **Apply DST?** to Yes. Default: No. The Daylight Saving Time convention followed in the country/region you selected will be automatically applied. The SPARSH VP310 will change its date and time settings according to the DST convention of the selected country/region.

- Select the **CO CLIP Pattern** for the SPARSH VP310. This is the type of Calling Line Presentation on the phone for incoming calls from trunks. You can select any of these options:
  - **Name Only** (only the name of the caller will be displayed).
  - **Number Only** (only the number of the caller will be displayed).
  - **Number + Name** (both the name and the number of the caller will be displayed).

  Default: Number + Name.

- Select the **Language** for the SPARSH VP310. Default: English.

  SARVAM UCS provides language support for English, French, German, Spanish, Portuguese, and Italian on the SPARSH VP310. When you select any of these languages, all the prompts and command strings will appear in the selected language.

SIP Extension users can change the language by accessing and navigating through the phone menu.

The SA can change the Language by logging into the SA Jeeves.

- Select a **Ringer Mode** for the phone from the four options:
  - Ring immediately (it rings immediately as a fresh calls lands on the phone).
  - Ring if idle (rings only if the phone is idle).
  - Ring after a delay (if the call is still not answered).
  - Silent.

  Default: Ring Immediate.

- If you selected *Ring after a delay* as Ringer Mode, set the **Ring Delay Timer (sec)**, if required, to the desired value.

  The Ring Delay Timer is the time in seconds the system waits on receiving a call before ringing on the phone. The range of this timer is 0 to 99 seconds. Default: 10 seconds.

- If you want to enable *Ringer Auto Acknowledge* mode, set the **Acknowledge Timer** (sec) to the desired value.

  The Ringer Auto Acknowledge mode determines when to stop the ring on the phone. There are two options for Ringer Auto Acknowledge:
  - Stop only when the call is answered.
  - Stop after a delay.

  To stop the ring on the phone after a delay, the Acknowledge Timer must be configured. The range of this timer is 01 to 99 seconds. Default: 00 seconds.

  To stop the ring only when the Call is answered or manually acknowledged, the Acknowledge Timer must be set to '00'. By default, Ring Auto Acknowledge is turned OFF.

- To assign the Ring Destination for the SPARSH VP310, select the desired destination for **Play Ring on**. You may choose
  - **Speakerphone:** The ring will be played on the Speakerphone.
  - **Headset:** The ring will be played on the Headset.

  Default: Speakerphone.

  When you select the Headset as the destination, make sure that you set the flag '*Headset Connected?'* to Yes, connect a Headset to the SPARSH VP310.

- Set the **Ringer Volume** to the desired level, from 0 to 7, according to your preference. Default: 4.

- To increase/decrease the volume of outgoing speech (Transmit Gain) on the handset of the SPARSH VP310, set the **Handset Transmit Volume Level** to the desired level, from 0 to 7. Default: 4.

- To increase/decrease the volume of incoming speech (Receive Gain) on the handset of the SPARSH VP310, set the **Handset Receive Volume Level** to the desired level, from 0 to 7. Default: 4.

- To increase/decrease the volume of outgoing speech (Transmit Gain) on the headset of the SPARSH VP310, set the **Headset Transmit Volume Level** to the desired level, from 0 to 7. Default: 4.

- To increase/decrease the volume of outgoing speech (Receive Gain) on the headset of the SPARSH VP310, set the **Headset Receive Volume Level** to the desired level, from 0 to 7. Default: 4.

- To change the Transmit Gain of the Speakerphone MIC Volume, set **Speaker Transmit Volume Level** to the desired level, from 0 to 7. Default: 4.

- To change the Receive Gain of the Speakerphone MIC Volume, set **Speaker Receive Volume Level** to the desired level, from 0 to 7. Default: 4.

- To use a Headset with the SPARSH VP310, select the **Headset Connected?** check box**.** Default: Disabled.

  Make sure that you connect a Headset to the SPARSH VP310, if you enable this option.

- Select the **Auto Answer** check box to enable this feature on the SPARSH VP310. Default: Disabled.

  When you set the "Auto Answer" feature on the SPARSH VP310, the phone goes OFF-Hook automatically after a preset period of time, without the extension user having to pick up the handset or press the speaker or headset key. When you enable Auto Answer, you must configure the Auto Answer Timer.

- If you enabled Auto Answer on the phone, set the **Auto Answer Timer (sec)** to the desired value.

  This timer defines the time in seconds that the SPARSH VP310 should wait before going OFF-Hook to auto answer a call. The range of this timer is 1 to 9 seconds. Default: 1 second.

- Adjust the Backlight brightness of the phone's LCD display, by setting the **LCD Backlight Level** to the desired value, from 1 to 4. Default: 3.

- Set the **Back Light Off Timer (sec)** to the desired value, if required, from 000 to 999 seconds. Default: 10 seconds.

- Set the **LCD Contrast Level** to a level from 1 to 4 that is comfortable to you. Default: 3.

DSS Key Settings

- You can select the desired key template — Operator, Executive1, Executive2, Executive3, Hotel Attendant, Guest or any other template you added. See "Customizing Extended IP Phone Templates using Jeeves" for more details.

  OR

- You can personalize the key map of the SPARSH VP310 for this location. To do so,

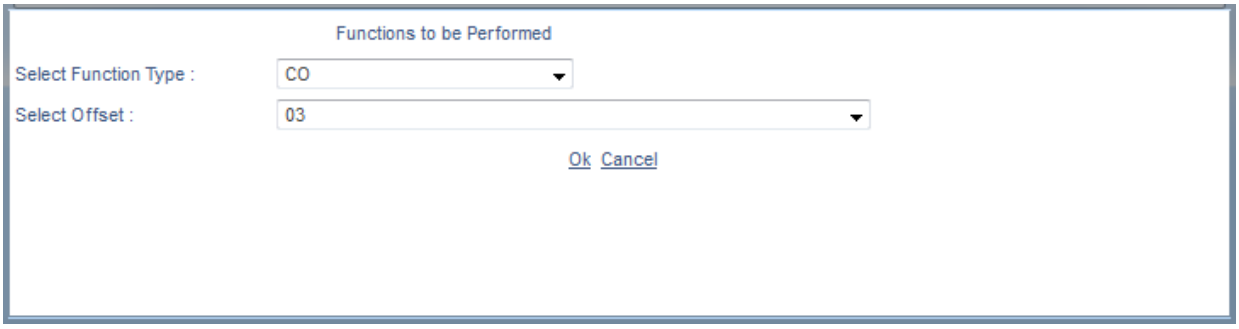
  - Select **Personalized** as the **Key Template** option.

  - Click **Key Settings**.

  - The key map of the Extended Phone opens in a new window on your screen.

- Click the key you want to configure. For example, **CO3**.

  The **Functions to be Performed** by the key opens in a new window.

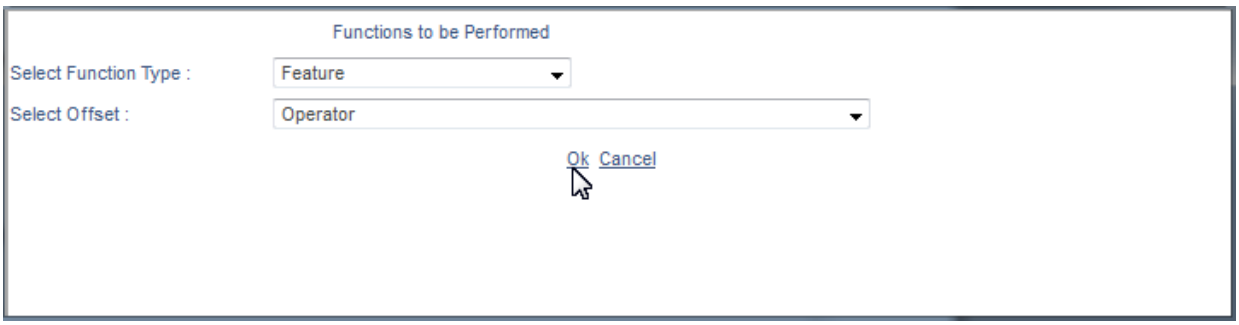


- In the **Select Function Type** list, select the function to be performed by the key. For example, you want to use the key to call the **Operator**.

  The Operator function is a Feature, so select the option **FEATURE** from the **Select Function Type** list box.

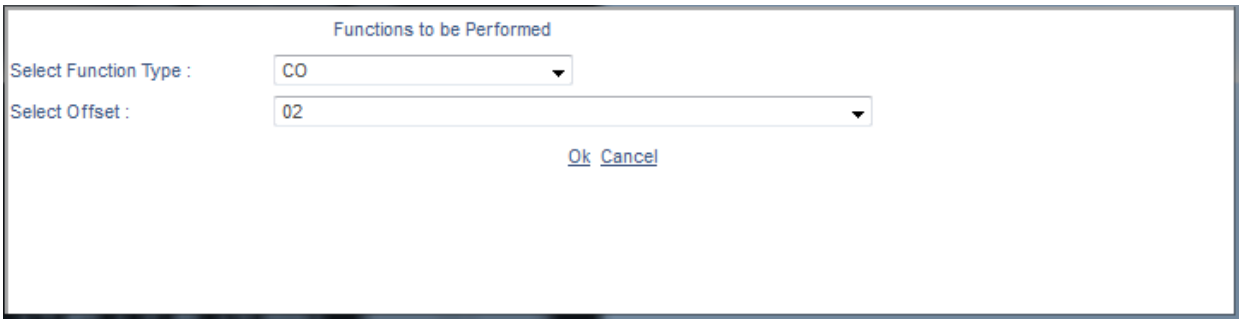
  From the **Select Offset** drop down list, all the features that can be assigned to keys are listed.

- Select **Operator** from the list of features in the **Select Offset** box.
- Click **OK**.

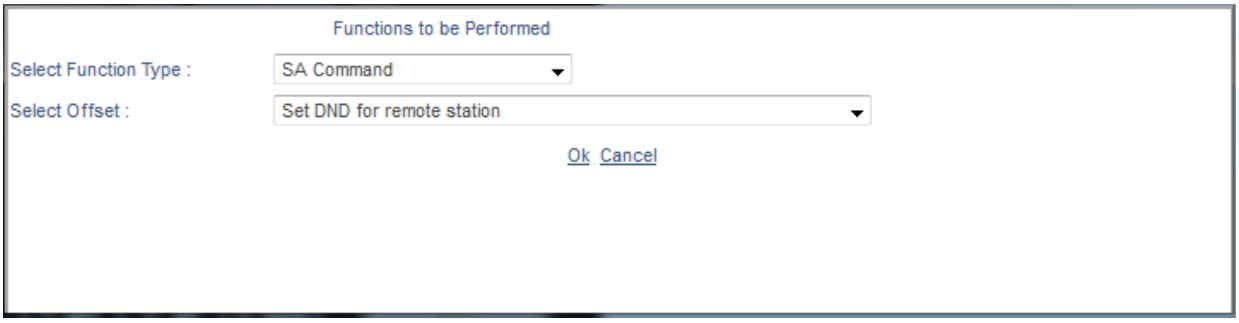


The *Operator* feature appears on the key label.

- To take a second example, if you want to assign **Remote DND** to the key currently assigned **CO 2** key, click the key.



- In the **Select Function Type** list box, select the option **SA Command**, as Remote DND is a System Administrator (SA) Command.



- In the **Select Offset** box, select the option **Set DND for remote station**.

- Click **OK**. The box closes. Remote DND feature will appear in abbreviated form as *R-DND* on the key label.

- Follow the same instructions to assign features to other DSS keys. Selecting the appropriate Function Type and the Offset for each feature/function.

  If you want assign a feature, select **FEATURE** as function type, and select the desired feature as Offset.
  If you want to use the key to call a DKP or a SIP extension, select **DKP** or **SIP Extension** as Function Type and select the number of the extension as Offset.

  To assign direct access to a Mobile Trunk, select **MOBILE** as Function Type and the desired port number **01** or any other trunk number as Offset.

  To assign direct access to a SIP Trunk, select **SIP** as Function Type and the desired trunk number from **01** or any other trunk number as Offset.

  Click **OK**, each time you select a Function Type and Offset in the dialog box.

  You can reinstate default key assignment any time, by clicking the **Default** button at the bottom of the window.

- When you complete assigning functions to keys, close the window.

- If you assign/re-assign functions to the following keys, the Phone will restart:
  - Speaker
  - Headset
  - Ringer Acknowledge
  - Local Menu

Transport Mode and SRTP

- Select the protocol to be used to transport the SIP messages. You can select the **Transport Mode** as **TCP** or **TLS**.

If you select TCP, make sure the SIP Over TCP check box is selected in VoIP Parameters.

If you select TLS, make sure the SIP Over TLS check box is selected in VoIP Parameters.

- For secure conversations over SIP, select the **Enable SRTP**? check box. The SIP messages will be transported over SRTP only.

If you select this check box, make sure you have selected SRTP Mode as Forced or Optional in the General Parameters under SIP Extension Settings.

RTP Port

- Define **RTP Port**:

  - **RTP Listening Port:** This is the port on which the SPARSH VP310 listens for RTP messages over UDP. This port is also used as the source port for sending RTP packets. This port is also used as the source port for sending RTP packets to the remote peer. The valid range for this port is 1025-65278. Default: 8000.

Quality of Service

- Set the **SIP Quality of Service (QoS)** for SIP signaling as:

  - **SIP DiffServe/ToS**. Valid range is 00 to 63. Default: 26.
    OR
  - **RTP DiffServe/ToS**. Valid range is 00 to 63. Default: 46.

NAT Keep Alive

- If the SPARSH VP310 is connected behind a NAT router, configure **NAT Keep Alive**.

  - Select the check box **Enable NAT Keep Alive** to send Keep Alive messages periodically to refresh the binding in the NAT router. Default: Disabled.

  - Define as **Interval (sec)**, the time period, from 001 to 999 seconds, after which the phone should send Keep Alive message. Default: 120 seconds.

    The time period you define should be less than the binding timer of the router.

Timers

- Set the following **Timers** to the desired value, where required:

- **SIP INVITE Timer (sec):** This is the time in seconds that the phone waits for a response from the called party after ending INVITE message. This timer starts after sending INVITE message to the called party and stops on receipt of the provisional response or the final response or when the user disconnects the call. On expiry of the timer, the phone terminates the call process and gives an error tone to the user. The range of the SIP INVITE TIMER is 10-180 seconds. Default: 30 seconds.

- **SIP Provisional Timer (sec)**: This is the time in seconds that the phone waits for final response after receiving the provisional response from the called party. This timer starts on the receipt of the provisional response from the called party and stops on receipt of the final response from the called party or when the user disconnects the call. On expiry of the timer, the IP phone terminates the call process and gives error tone to the user. The range of SIP Provisional Timer is 10-180 seconds. Default: 60 seconds.

- **General Request Timer (sec):** This is the time in seconds for which the phone waits for response of a transaction request. This timer starts on initiating a transaction. This timer stops on receipt of a response for the request. On expiry of the timer, the phone clears the transaction. This timer is used for Registration request, etc. The range of the General Request Timer is 10-60 seconds. Default: 20 seconds.

Debug

- To debug using Syslog Client supported by the SPARSH VP310, configure Debug parameters:

  - Select the **Enable Debug?** check box. Default: disabled.

    When the Debug flag is enabled, the phone will send the debug messages to the Syslog Server IP address. Debug report can be viewed on the Syslog Server or any other application which can capture the Syslog messages/debug sent by the phone.

  - Enter the IP Address and port of the remote Syslog Server and as **Syslog Server Address and Server Port**.

    The address of the Listening Port of the Syslog Server is from 1025-65535;514. Default: 514. Syslog uses the UDP as transport protocol and listens on the port 514 (the default listening port).

  - You may select the **Debug Level** from the following options, by selecting the respective check box:
    - SIP
    - System
    - Hardware
    - Call
    - Network
    - VoPP

    You may select any or all of these debug levels. The Syslog Client will send only the debug messages for the selected level to the remote server on the IP network. For example, if the debug log of 'Call's is required, you can select this option, and disable all others.

- Click **Submit** to save settings.

- If you have completed the configuration of the SPARSH VP310 Settings at Location 1, follow the same steps as described above to configure the SPARSH VP310 at Location 2 and Location 3.

However, if you want to replicate the configuration of the SPARSH VP310 Settings same as Location 1 to Location 2 and Location 3, you can use the **Copy** button present at the bottom of the page. To know more, refer to “Copy Parameter Values”.

If you wish to Copy only the configurations from a location to all other locations where the same IP Phone is connected, make sure you only clear the Device Type check box.

If you wish to copy the Device Type as well as all the configurations from a location to all the SIP Extensions, make sure all the check boxes are selected.

When you change any of the parameters listed below in the SIP Extension at Location 1, 2, 3, the phone will restart automatically, if registered:

- Use SIP Extension
- SIP ID
- Authentication ID
- Authentication Password
- Registrar Server IP Address
- MAC Address
- Enable Device
- Device Type
- Key Map in the Key Template assigned to phone
- Call Progress Tone
- Date and Time
- Apply DST?
- Transport Mode and SRTP
- QoS
- RTP Ports
- NAT Keep Alive
- SIP Timers
- The SE Password of SARVAM UCS is changed
- Specific parameters in VoIP Parameters are changed
- Specific parameters in Network Port parameters are changed
- You restart the System
- Set the System to Default

Configuring Matrix SPARSH VP330

SPARSH VP330 is the proprietary Extended IP Phone with graphical touch-screen user interface, supplied by Matrix. The feature-rich SIP based phone support most of the features and functions of the proprietary digital key phones of SARVAM UCS.To know the list of features supported, refer to “SARVAM UCS Features Supported in Terminals”.

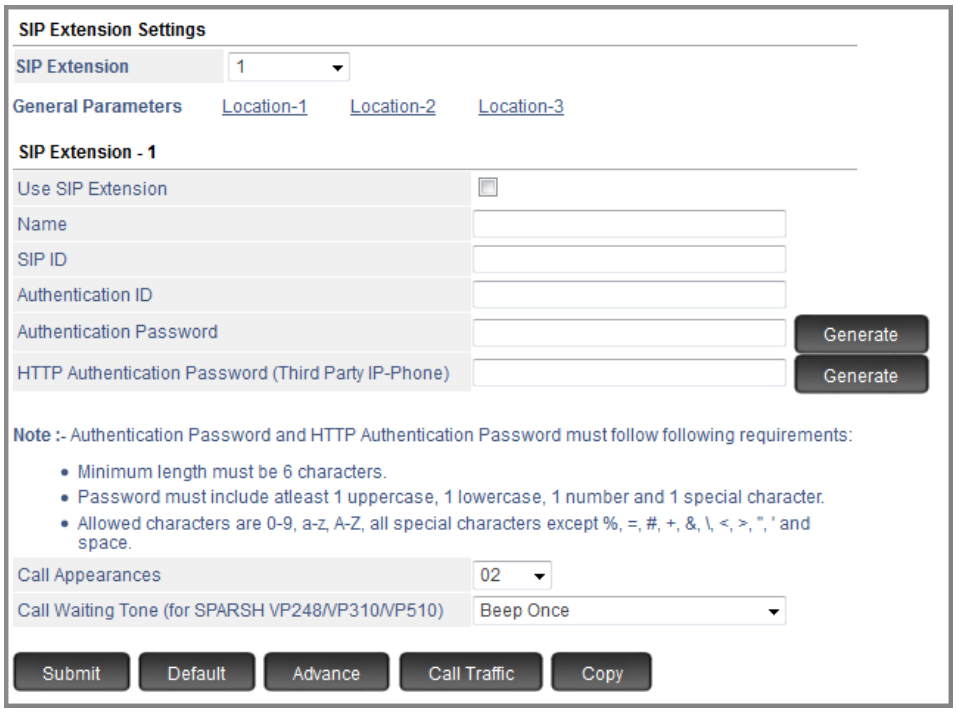
For detailed product information and operation instructions, refer to the *SPARSH VP330 User Guide.*

To be able to use SPARSH VP330[146] as SIP Extensions, you must configure the following:

- SIP Extension General Parameters, see “Configuring SIP Extension General Parameters”.
- SIP Extension Settings, see “Configuring SIP Extension Settings using Jeeves”
- Extended Phone Settings, see “Configuring Matrix Extended Phone Settings using Jeeves”
- Voice Mail Settings, if you want to provide mailbox facility to the extension. See “Extension Voice Mail Settings”.

Configuring SIP Extension Settings using Jeeves

- Log into Jeeves as System Engineer.
- Under **Configuration**, click **VoIP Configuration**.
- Click **SIP Extension Settings**.



The page of SIP Extension 001 opens.

- You may select the **SIP Extension** number you want to configure.

146. *SARVAM UCS supports only IPv4 Addresses for registering SPARSH VP330.*

The parameters of the SIP Extension number you selected will appear on this page.

For SARVAM UCS upto 999 SIP Extensions can be registered with the system.

- Select the **Use SIP Extension** check box to enable the SIP extension. Default: disabled.

- In the **Name** field, enter a name for the SIP Extension, which may be the name of the person who will use the SIP Extension or the name of a Department. The name you enter here will be displayed as the Caller ID of the SIP Extension on the remote user's phone, when the SIP Extension user makes calls.

  The name may consist of a maximum of 18 alphanumeric characters.

- Enter the **SIP ID** for the extension. The SIP ID is necessary for registering the SIP Extension with the Registrar of the system. It is the number with which you can call the SIP Extension. Any extension user of the SARVAM UCS can call a SIP Extension by dialing the SIP ID assigned to the SIP extension. SIP ID of each SIP Extension must be a unique number string of a maximum of 6 digits. Any combination of digits from 0 to 9 and the characters * and # are allowed.

  You cannot assign the same SIP ID to more than one extension.

  To assign SIP IDs according to your preference and requirement to a range of SIP Extensions, see “Assigning Access Codes to a Range of Extensions”.

  By default, the SIP IDs are Blank.

The SIP ID will be set to default value (blank), when you restore the default settings of the system.

- In **Authentication ID**, enter the number which you want the system to use for user authentication of the SIP messages received from the SIP Extension. You cannot keep this field blank and it must be unique for each SIP extension. The number may be a string of maximum 6 digits. All ASCII characters except < > and “ (double quote) are allowed. Default: Blank.

Make sure the User ID configured in “Digest Authentication” does not conflict with the Authentication ID configured above.

- In **Authentication Password**, enter the password manually or click **Generate** to automatically generate a unique password. This password will be used by the system to authenticate the SIP messages received from the SIP Extension. To avoid unauthorized access, we recommend you to change the password regularly. Make sure it is strong and is kept confidential. When you enter the password manually, the password must:

  - be of minimum 6 characters and can be a maximum of 12 characters.
  - include atleast one upper-case, one lower-case, one number and one special character.
  - all ASCII characters (except Percentage **%**, Hash **#,** Equal to **=,** Plus **+**, And **&**, Backslash **\,** Less than **<,** Greater than **>,** Apostrophe **’,** Double Quote **"** and **Space**) are allowed.
    Default: Blank.

  To provide additional security, when the Authentication fails 10 times consecutively due to wrong Authentication ID / Authentication Password, the system will blacklist the IP Address and Port for registration of this SIP Extension. However, you can remove the IP Address from the Blacklist IP Address list. See “Black List IP Address - SIP Extensions” for more details. This activity will be logged in the “System Activity Log” as well as “Simple Network Management Protocol (SNMP)”.

Make sure you note down or copy the Authentication Password in a confidential file.

•In **Call Appearances**, define the maximum number[147] of simultaneous incoming calls that the SIP Extension user should be allowed to receive. You can set up to 10 call appearances for a SIP Extension. Default: 2.

When Call Appearance is set to 2, the SIP Extension can receive 2 calls at a time.

- Under **Auto Sign-In,** enter the **Email ID** of the extension user you wish to store. The Email ID can be a maximum of 64 characters. The Email ID is used for various server features.

Auto Sign-In parameters — Send Configuration Mail and Mail Status, *are applicable only for Mobile Clients — VARTA ADR100, VARTA AMP100 applications.*

*The **Send Configuration Mail** button will appear only after you have enabled the SIP Extension and configured the SIP ID, Authentication ID and Password.*

- Under **Authentication**, enable Authentication of any or all of the following SIP Message Options by selecting the respective check boxes:
  - **INVITE Request**
  - **SUBSCRIBE Request**

  By default, the SIP Message Options INVITE and SUBSCRIBE are enabled.

Make sure that the Authentication ID for the SIP Extension has been programmed.

- For secure conversations over SIP, enable **SRTP Mode**. The SARVAM UCS supports the following options:

  - **Disable:** SARVAM UCS uses normal RTP for transporting the speech packets.

  - **Optional:** SARVAM UCS uses SRTP for transporting the speech packets. If the remote user does not support SRTP, SARVAM UCS will use normal RTP for transporting the speech packets.

    - If you select this option, you must configure the **SRTP Media Type**. You may select **AVP** or **SAVP**. By default, AVP is selected as the SRTP Media Type.

  - **Forced:** SARVAM UCS uses only SRTP (SAVP) for transporting the speech packets. If the remote user does not support SRTP, SARVAM UCS will reject incoming calls from and drop outgoing calls made to such users.

    By default, SRTP Mode is Disabled.

- Assign a **SIP Hardware Template** to the SIP Extension. Default: 01. The “SIP Hardware Template” contains voice quality related features such as Voice Codec selection, Tx and Rx Gains, Echo Cancellation, Jitter Buffer and Fax-over-IP options and related parameters

  There are 32 different templates to choose from. Each template can also be altered to suit your requirement and preferences. By default, Template number 01 assigned to all SIP Extensions as well as to SIP Trunks.

147. *For the calls that are routed through the CPU, the number of Vocoder channels that will be supported would be as per the license you purchase.*

Check if the values in this template fulfill requirements of the SIP Extension. If Template 01 fulfills the feature requirements, retain Template 01.

If a different set of SIP hardware features are to be allowed to this SIP Extensions, prepare another template and assign it to this extension. To do this,

- Under **VoIP Configuration**, click the **SIP Hardware Template** link.
- Select a Template number, for example 02.
- Customize Template number 02 and click **Submit**.
- Return to the **SIP Extension Settings** page.
- Select the number of the Template you customized, Template 02 in the **SIP Hardware Template** field.

Also see the topic "SIP Hardware Template" to know more about customizing the templates and applying on the SIP Extensions.

- Assign a **Station Basic Feature Template** to the SIP Extension. Default: The "Station Basic Feature Template" has a set of features like Time Table, Class of Service, Toll Control, Operator, Storage of Incoming and Outgoing Calls, Outgoing Trunk Bundle groups. There are 50 different templates to choose from. Each template can also be altered to suit your requirement and preferences.

  If the default Station Basic Feature Template 01 fulfills the feature requirements of the SIP Extension ("Class of Service (COS)", "Toll Control", "OG Trunk Bundle Group", etc.) retain this template, you may also customize this template. If you want to assign a different set of features to this SIP Extension, prepare a different Station Basic Feature Template and apply it to this extension. To do this,

  - Under **Configuration**, click the **Station Basic Feature Template** link.
  - Select a Template number, for example 05.
  - Customize Template number 05 and click **Submit**.
  - Return to the **SIP Extension Settings** page.
  - Select the number of the Template you customized, Template 05, in the **Station Basic Feature Template** field.
  - Click **Submit** to save changes.

  Also, see the topic "Station Basic Feature Template" to know more about customizing the templates and applying on extensions.

- Assign a **Station Advanced Feature Template** to the SIP Extension. Default: Template 01. The "Station Advanced Feature Template" has a set of advanced features for extensions such as Alarm Notification settings, Routing of Incoming Auto Attendant Calls, Call Duration Control, Floor Service, etc. There are 50 different templates to choose from. Each template can also be altered to suit your requirement and preferences.

Check if the default template fulfills the feature requirements of the SIP Extension by clicking the **Station Advanced Feature Template** link.

You may retain this template and customize it further, or customize another template if a different set of features are to be allowed to this SIP Extension. To customize/prepare another template,

- Under **Configuration**, click the **Station Advanced Feature Template** link.
- Select the Template number, for example 02, and customize this template.
- Click **Submit** and return to the 'SIP Extension Settings' page.
- In the **Station Advanced Feature Template** field, select the number of the template you customized.
- Click **Submit** to save changes.

Also see the topic "Station Advanced Feature Template" for instructions on customizing these templates and applying them on the extensions.

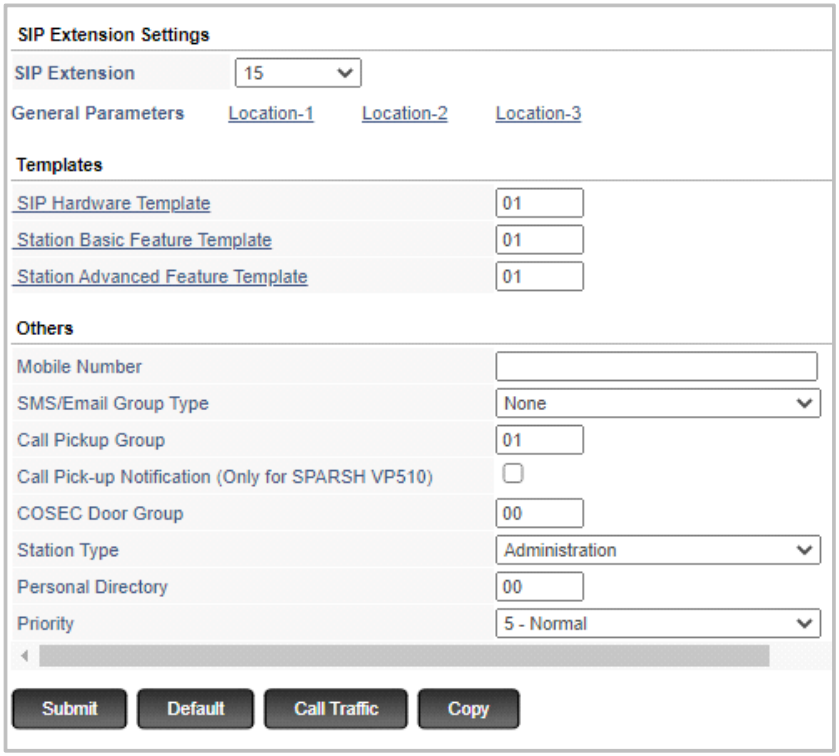
- In **Voice Mail Settings**, click the Voice Mail Settings link. The respective Extension Voice Mail Settings window will open. You may edit the parameters. For details, see "Extension Voice Mail Settings".

The Voice Mail Settings link will be visible only if you have configured the respective SIP ID.

Click **Close** to close the window.

Advanced Configuration Parameters

- If you want to provide other features like Personal Directory, Priority, or assign a Station Type to the SIP Extension, click the **Advanced** button at the bottom of the page.



- Enter the **Mobile Number** of the extension user you wish to store. The Number can be a maximum of 16 digits.

- You can assign the extension user to a Group. Select the desired **SMS/Email Group Type** from the list.The system clubs together extension users assigned the same Group. Default: None. For details, see “SMS/Email Group”.

- Assign the SIP Extension to a **Call Pick-up Group**, if required.

  Call Pick Up allows the SIP Extension user to 'pick up' (answer) calls ringing on any other extension, by dialing a feature code, without physically going to the ringing extension. It also allows incoming calls for the SIP Extension to be answered by the other extensions assigned the same Call Pick-Up group.
  For this to work, both the ringing extension and the extension picking up the call must be in the same 'Call Pick Up Group'. Refer “Call Pick Up” for instructions on how to create groups. You can create as many as 99 groups numbered from 01 to 99.

  Enter the number of the Call Pick-Up Group you created for this SIP Extension in this field.

- Select the **Call Pick-up Notification (Only for SPARSH VP510)** check box, if you want the Call Pick-up Notification to be displayed on the phone LCD. The Call Pick-up Notification will be displayed for internal as well as external calls. The notification will be displayed on SPARSH VP510 only if the ringing extension is in the same Call Pick-up Group. The notification will be displayed only when SPARSH VP510 is in idle or dial state. The notification will display the name/number of the Caller along with the name/number of the Called. For example, if the name and number of the Caller is ABC, 2001 and the name and number of Called is XYZ, 1001, the LCD notification will be displayed as ABC -> XYZ. Along with the LCD display the phone back-light will also be lit.

  For this to work, make sure you have enabled Call Pick-up in COS as well as assigned a Call Pick-up Group to the extension. Refer to “Call Pick Up” and “Class of Service (COS)”. Call Pick-up Notifications will be displayed for DKP, SLT as well as SIP Extensions and for calls landing through CO, SIP as well as T1E1 Trunks. For details of the Notification, refer to the EON510_SPARSH VP510 V2 User Guide.

- You must assign the extension user to a **COSEC Door Group** for COSEC Integration. The users in the same group must be assigned the same group. You can create as many as 50 groups numbered from 00 to 50. Users who are assigned COSEC Door Group ‘00’ are not a part of any group. See “COSEC Integration” for more information.

- If this is an Operator extension and you want the system to play beeps during a conference to the participants, to indicate the presence or absence of the Operator, select the **Station Type** as **Assistant**.

  If you are using the system in the *Hotel Mode*, select the **Station Type** for the SIP Extension as **Administration/Assistant** or **Guest**. The system will consider the options Administrator and Assistant as same.

- You may assign a **Personal Directory** number to the SIP Extension. Default: 00.

  A Personal Directory is a list of 25 frequently dialed numbers, each of which are stored by Index number (location code), Name and Trunk Access Codes ("Out Going Trunk Bundle Group Index"). The Personal Directory is necessary for using the features “Abbreviated Dialing” and “Dial By Name”.

  When a Personal Directory is assigned to a SIP Extension, make sure you also configure this directory. The Personal Directory can be programmed by the SIP Extension users and by the System Engineer. Refer the topic “Abbreviated Dialing” for instructions on programming the Personal Directory.

If Personal Directory is not to be assigned, enter 00 in this field.

- Select a **Priority** Level for the SIP Extension from 1 to 9. Default; 5-Normal.

  Each extension of the SARVAM UCS is assigned a Priority Level starting from 1, 2, 3... to 9, with '1' being lowest Priority and '9' being highest Priority. Whenever an extension (phone) with higher priority calls an extension with lower priority, a triple ring is placed on the called extension. To know more, read the feature description “Priority”.

  If this SIP extension is assigned to Operator, you may want to set a higher priority for this extension.

- Click **Submit** to save your SIP Extension Settings.

Configuring Matrix Extended Phone Settings using Jeeves

You can register three Matrix Extended Phones/Soft Clients at three different locations as a single SIP Extension. You can connect the same or different types of Extended Phones/Soft Clients —SPARSH VP248, SPARSH VP310, SPARSH VP510, SPARSH VP330,Extended SPARSH VP710, SPARSH VP210, Matrix VARTA ADR100 Mobile UC Client, Matrix VARTA AMP100 Mobile UC Client or MATRIX VARTA WIN200 Desktop UC Client — at each location. In this case we assume that SPARSH VP330 is connected at Location 1, 2 and 3.

If you have connected SPARSH VP248 refer to “Configuring Matrix Extended Phone Settings using Jeeves” in *Configuring Matrix SPARSH VP248 as SIP Extensions*.

If you have connected SPARSH VP310 at any of the locations, refer to “Configuring Matrix SPARSH VP310”.

If you have connected SPARSH VP510 at any of the locations, refer to “Configuring Matrix SPARSH VP510”.

If you have connected Extended SPARSH VP710 at any of the locations, refer to “Configuring Matrix Extended SPARSH VP710”.

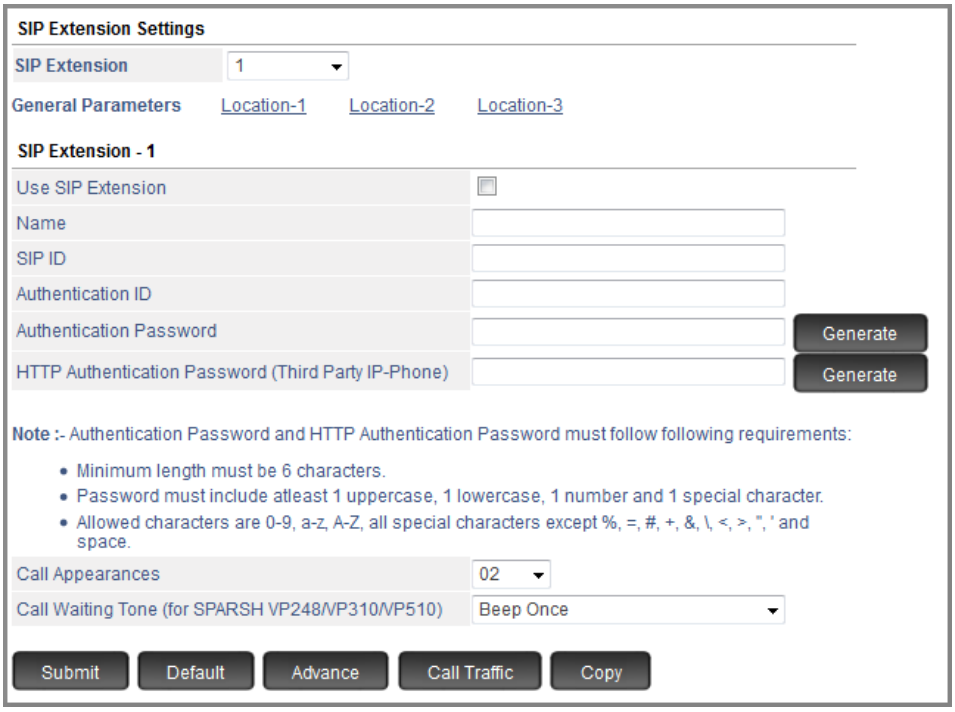
If you have connected Extended SPARSH VP210 at any of the locations, refer to “Configuring Matrix SPARSH VP210”.

If you have registered Matrix VARTA Mobile UC Clients in any of the locations, refer to “Configuring Matrix VARTA ADR100/AMP100 UC Clients”.

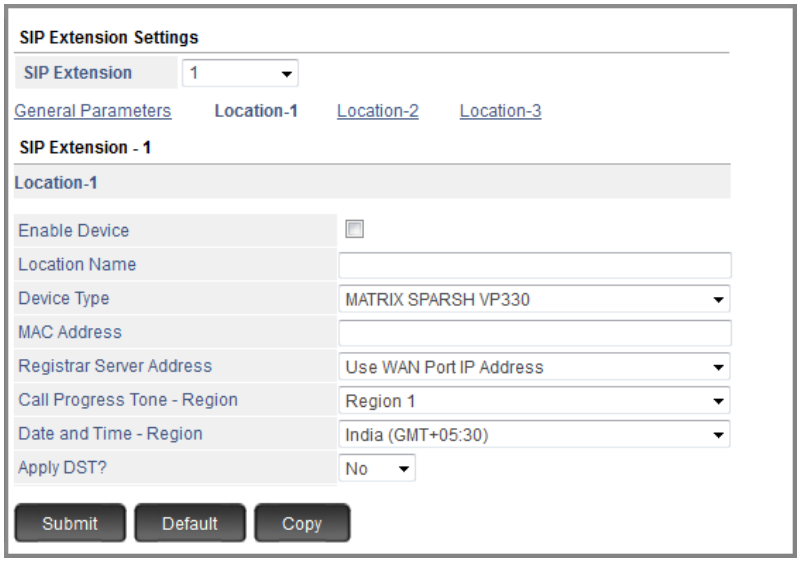
If you have registered MATRIX VARTA WIN200 Desktop UC Client in any of the locations, refer to “Configuring Matrix VARTA WIN200 UC Client”.

If you want to use more than one SPARSH VP330 Phones as a SIP Extension, configure their settings at **Location 1**, **Location 2** and **Location 3**.

- Click **Location 1**.



- The settings of the phone at **Location 1** appear.



- Select the **Enable Device** check box. Default: Disabled.
- Enter the **Location Name** for the phone to identify the phone. Location name may be the place where the phone is located (e.g.: Head office, branch, residence). The Location Name may consist of 18 characters (maximum). Default: Blank.
- Select **MATRIX SPARSH VP330** as the **Device Type** at this location.
- Enter the **MAC Address**[148] of the SPARSH VP330 connected at this location in hexadecimal format: 00:1b:09:XX:XX:XX. Default: blank.

SARVAM UCS validates the SPARSH VP330 on the basis of the MAC Address, and provides configuration on validation.

As SARVAM UCS allows registration of the SIP Extension from three different locations, it identifies the SIP Extension in each location by the programmed MAC address.

- Select the appropriate **Registrar Server Address** to register the SPARSH VP330 with the SIP Registrar of SARVAM UCS, according to your installation scenario:

  - If the SPARSH VP330 is connected on the WAN network, select **Use WAN Port IP Address** as Registrar Server IP Address.

  - If the SPARSH VP330 is connected on the LAN network, select **Use LAN Port IP Address** as Registrar Server IP Address.

  - If the SPARSH VP330 is connected in the Global Network and SARVAM UCS is located behind a Router, or behind a NAT Router and STUN is programmed, select **Use Router/STUN's IP Address** as Registrar Server IP Address.

    Make sure you configure either the **Router's Public IP Address** or **Simple Traversal of UDPs through NATs (STUN)** in Network Parameters. For details, see "Configuring Network Parameters".

  - If Dynamic DNS is configured in the Network Parameters, select **Use Dynamic DNS Host Name** as Registrar Server IP Address.

    By default, Use WAN Port IP Address is selected as the Registrar Server IP Address.

- To set the call progress tone generation standards of the country where the SPARSH VP330 is installed, select the **Call Progress Tone - Region**. Default: Region 1.

  See "Call Progress Tones" to know more.

- To display the Date and Time of the country where the SPARSH VP330 is installed, select the **Date and Time - Region**. Default: India.

- If you want to enable Daylight Saving Time (DST) on the phone, set **Apply DST?** to either **Manual** or **Scheduled** as per your requirement.

  When you select **Scheduled** as the DST option, the Real Time Clock of SARVAM UCS is advanced and set backward automatically according to the DST convention of the country/region where the IP Phone is installed.

  Scheduled DST Adjustment is useful in countries/regions where DST Time is fixed, such as in Europe, USA and Canada, without yearly variations.

  SARVAM UCS supports 18 DST Types for Scheduled DST Adjustment. To know more, refer to "Daylight Saving Time (DST)". To know more about Scheduled DST assigned for the respective region, refer to *Time Zone* in "Default Settings".

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148. *MAC address is the address of the electronic hardware devices such as a computer, which is hard-coded into the device during manufacture and cannot be modified. No two devices can have similar MAC address and thus it uniquely identifies your phone. MAC address is assigned as per the IANA standard. The MAC Address of the phone will be used as source MAC address on all Ethernet frames.*

When you select **Manual** as the DST option, the Real Time Clock of SARVAM UCS is advanced manually and set backward automatically according to the DST convention of the country/region where the IP Phone is installed.

Manual DST Adjustment is to be used in regions/countries that have no fixed DST Convention and where yearly variations in DST practices are likely.

When DST option is set as 'Manual', you must set the DST Start and the DST End time, that is, the time at which the clock is to be advanced and the time at which the clock is to be delayed. To do so,

- In **Time Offset,** enter the time you wish to forward or backward the DST start time with.

- In **DST Type**, select the desired option: **Date-Month Wise** OR **Day-Month Wise**.

  If you select '**Date-Month Wise'** in **DST Type**, you should now select the desired options in each of the following to specify the **DST Start** details.

  - **Date**: Select the date on which DST begins (1-31).
  - **Month**: Select the month when DST begins (January-December).
  - **Time (Hours)**: Select the time at which DST will begin to change.The time mode is of 24 hours ranging from 00 to 23 hours.
  - **Time (Minutes):** Select the time at which DST will begin to change.The time mode is of 60 minutes ranging from 00 to 59 minutes.

  Similarly, in the **DST End** configure the desired DST End Time details.

  If you select **'Day-Month Wise'** in **DST Type**, you should now select the desired options in each of the following to specify the **DST Start** details.

  - **Ordinal**: Select the Ordinal number of the day of the month, that is, the 1st, 2nd, 3rd, 4th, 5th day, when DST begins.
  - **Day:** Select the day of the month - Sunday, Monday, Tuesday, Wednesday, Thursday, Friday, Saturday- when DST begins.
  - **Month**: Select the month when DST begins (January-December).
  - **Time (Hours)**: Select the time at which DST must begin to change.The duration is of 24 hours ranging from 00 to 23.
  - **Time (Minutes):** Select the time at which DST must begin to change.The duration is of 60 minutes ranging from 00 to 59.

  Similarly, in **DST End** configure the DST End Time details.

  Once the DST Ends, the time of the IP Phone is set back to the Standard time automatically.

When the DST of a particular country starts or ends on the Last Sunday or any other day, for example, the last Tuesday, last Friday of the month, always set the Ordinal Number as '5th'.

- If you want to enable Daylight Saving Time (DST) on the phone, set **Apply DST?** to Yes. Default: No. The Daylight Saving Time convention followed in the country/region you selected will be automatically applied. The IP phone will change its date and time settings according to the DST convention of the selected country/region.

- Select the **Language** for the SPARSH VP330. Default: English.

SARVAM UCS provides language support for English, French, German, Spanish, Portuguese, and Italian on the SPARSH VP330. When you select any of these languages, all the prompts and command strings will appear in the selected language.

SIP Extension users can change the language by accessing and navigating through the phone menu.

The SA can change the Language by logging into the SA Jeeves.

DSS Key Settings

- You can select the desired key template — Operator, Executive1, Executive2, Executive3, Hotel Attendant, Guest or any other template you added. See "Customizing Extended IP Phone Templates using Jeeves" for more details.

  OR

- You can personalize the key map of the SPARSH VP330 for this location. To do so,

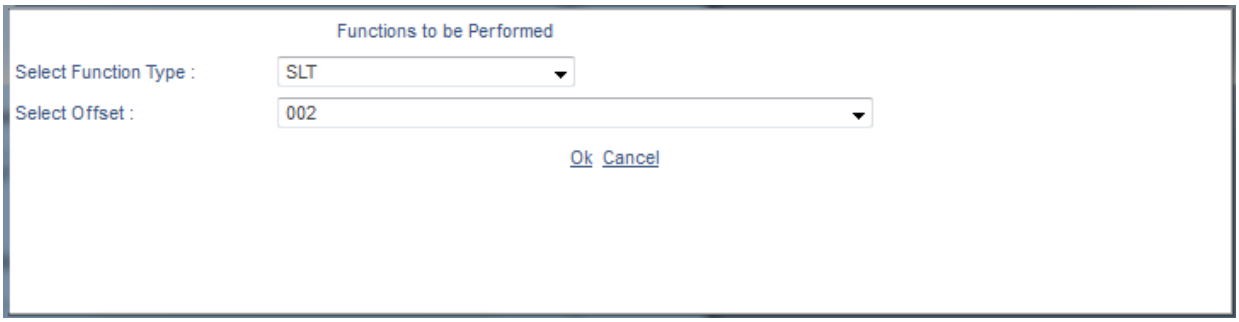
  - Select **Personalized** as the **Key Template** option.

  - Click **Key Settings**.

- The key map of the Extended Phone opens in a new window on your screen.

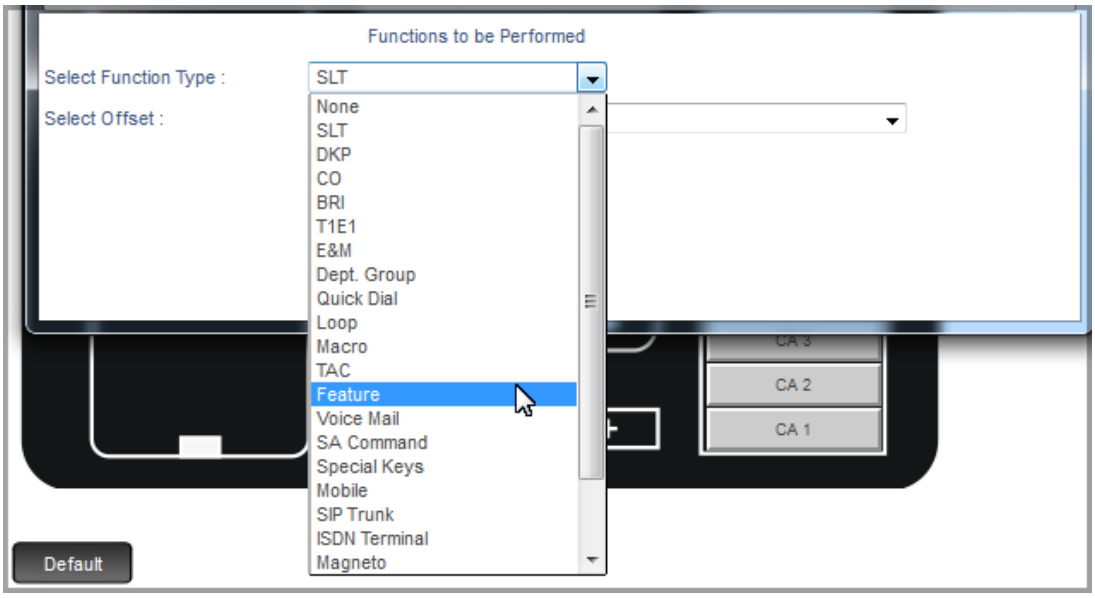
- Click the key you want to configure. For example, **SLT2**.

  The **Functions to be Performed** by the key opens in a new window.



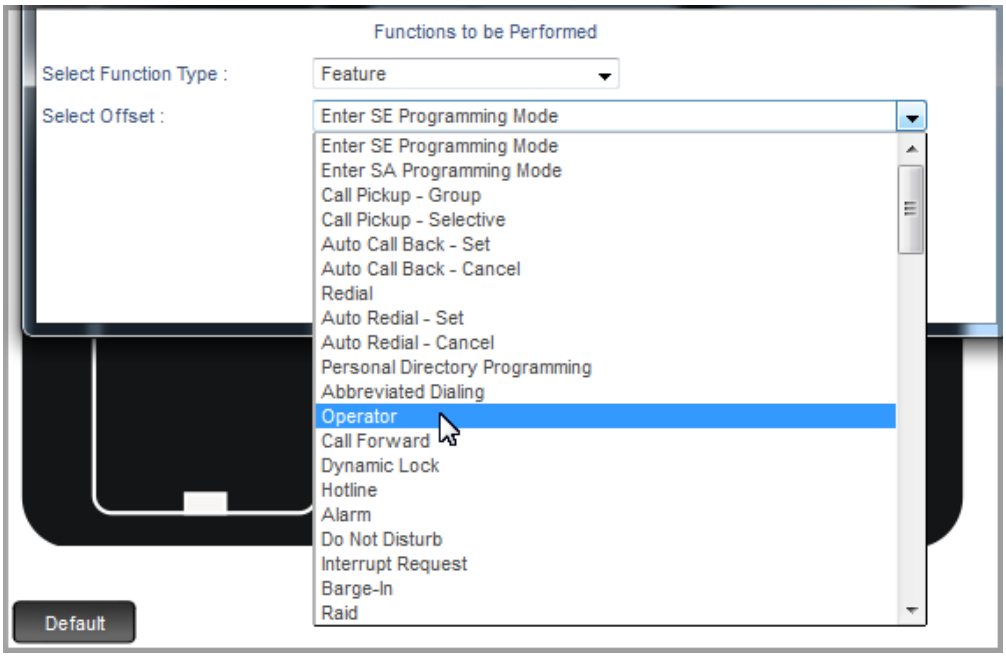
- In the **Select Function Type** list, select the function to be performed by the key. For example, you want to use the key to call the **Operator**.

The Operator function is a Feature, so select the option **FEATURE** from the **Select Function Type** list box.

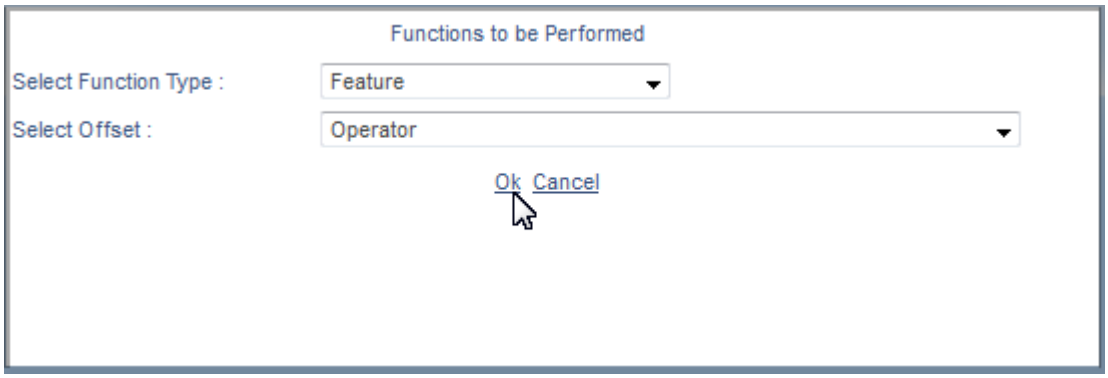


From the **Select Offset** drop down list, all the features that can be assigned to keys are listed.

- Select **Operator** from the list of features in the **Select Offset** box.



- Click **OK**.



The *Operator* feature appears on the key label.

- Follow the same instructions to assign features to other DSS keys. Selecting the appropriate Function Type and the Offset for each feature/function.

  If you want assign a feature, select **FEATURE** as function type, and select the desired feature as Offset.

  If you want to use the key to call a DKP or a SIP extension, select **DKP** or **SIP Extension** as Function Type and select the number of the extension as Offset.

  To assign direct access to a mobile trunk, select **MOBILE** as Function Type and the desired port number **1** or **2** as Offset.

To assign direct access to a SIP Trunk, select **SIP** as Function Type and the desired trunk number from **1** to **4** as Offset.

Click **OK**, each time you select a Function Type and Offset in the dialog box.

You can reinstate default key assignment any time, by clicking the **Default** button at the bottom of the window.

- When you complete assigning functions to keys, close the window.

The phone will enter the Auto Configuration mode, when you assign/re-assign certain features in the key maps. To know more, refer to the SPARSH VP330 User Guide.

Even if you assign keys for the following feature in the Key Templates, these features will not function:

Function Type	Offset
Macro	
SA Command	
Special Keys	Digit Pause
	Digit A
	Digit B
	Digit C
	Digit D
	Enter
	Local Menu
Feature	Enter SE Programming Mode
	Enter SA Programming Mode
	Personal Directory Programming
	Abbreviated Dialing
	Change Room Clean Status
	Guest Access Code
	Minibar Details
	Emergency Conference
	Self Ring Test
	SA Command Prefix
	PMS - User Defined Fields
	Department Group Call Forward

Transport Mode and SRTP

- Select the protocol to be used to transport the SIP messages. You can select the **Transport Mode** as **TCP** or **TLS**[149].

149. SPARSH VP330 supports TLS Version V1.0 only. To configure the TLS version, refer "Advance Options" in Security Settings.

If you select TCP, make sure the SIP Over TCP check box is selected in VoIP Parameters.
If you select TLS, make sure the SIP Over TLS check box is selected in VoIP Parameters.

- For secure conversations over SIP, select the **Enable SRTP?** check box. The SIP messages will be transported over SRTP only.

RTP Port

- Define **RTP Port**:

  - **RTP Listening Port:** This is the port on which the phone listens for RTP messages over UDP. This port is also used as the source port for sending RTP packets. This port is also used as the source port for sending RTP packets to the remote peer. The valid range for this port is 1025-65278. Default: 8000.

Quality of Service

- Set the **SIP Quality of Service (QoS)** for SIP signaling as:

  - **SIP DiffServe/ToS**. Valid range is 00 to 63. Default: 26.
    OR
  - **RTP DiffServe/ToS**. Valid range is 00 to 63. Default: 46.

NAT Keep Alive

- If the SPARSH VP330 is connected behind a NAT router, configure **NAT Keep Alive**.

  - Select the check box **Enable NAT Keep Alive** to send Keep Alive messages periodically to refresh the binding in the NAT router. Default: Disabled.

  - Define as **Interval (sec)**, the time period, from 001 to 999 seconds, after which the phone should send Keep Alive message. Default: 120 seconds.

    The time period you define should be less than the binding timer of the router.

Timers

- Set the following **Timers** to the desired value, where required:

  - **SIP INVITE Timer (sec):** This is the time in seconds that the phone waits for a response from the called party after ending INVITE message. This timer starts after sending INVITE message to the called party and stops on receipt of the provisional response or the final response or when the user disconnects the call. On expiry of the timer, the phone terminates the call process and gives an error tone to the user. The range of the SIP INVITE TIMER is 10-180 seconds. Default: 30 seconds.

  - **SIP Provisional Timer (sec)**: This is the time in seconds that the phone waits for final response after receiving the provisional response from the called party. This timer starts on the receipt of the provisional response from the called party and stops on receipt of the final response from the called party or when the user disconnects the call. On expiry of the timer, the IP phone terminates the call process and gives error tone to the user. The range of SIP Provisional Timer is 10-180 seconds. Default: 60 seconds.

  - **General Request Timer (sec):** This is the time in seconds for which the phone waits for response of a transaction request. This timer starts on initiating a transaction. This timer stops on receipt of a

response for the request. On expiry of the timer, the phone clears the transaction. This timer is used for Registration request, etc. The range of the General Request Timer is 10-60 seconds. Default: 20 seconds.

- Click **Submit** to save settings.

- If you have completed the configuration of the SPARSH VP330 Phone Settings at Location 1, follow the same steps as described above to configure the SPARSH VP330 Phone at Location 2 and Location 3.

  However, if you want to replicate the configuration of the SPARSH VP330 Settings same as Location 1 to Location 2 and Location 3, you can use the **Copy** button present at the bottom of the page. To know more, refer to “Copy Parameter Values”.

If you wish to Copy only the configurations from a location to all other locations where the same IP Phone is connected, make sure you only clear the Device Type check box.

If you wish to copy the Device Type as well as all the configurations from a location to all the SIP Extensions, make sure all the check boxes are selected.

When you change any of the parameters listed below in the SIP Extension at Location 1, 2, 3, the phone will go in Auto Configuration mode automatically, if registered:

- Use SIP Extension
- SIP ID
- Name
- Authentication ID
- Authentication Password
- Registrar Server IP Address
- MAC Address
- Enable Device
- Device Type
- Key Map in the Key Template assigned to phone
- Language
- Call Progress Tone
- Date and Time
- Apply DST?
- Transport Mode and SRTP
- QoS
- RTP Ports
- NAT Keep Alive
- SIP Timers
- Class of Service
- Trunk Access Code
- Emergency Numbers

The SIP Extension registered at Location 1, 2, 3, will also restart, if:

- The SE Password of SARVAM UCS is changed
- Specific parameters in Network Port parameters are changed
- Specific parameters in VoIP Parameters are changed
- You restart the System
- Set the System to Default

Configuring Matrix SPARSH VP510

SPARSH VP510, the Premium IP Phone is engineered to offer a contemporary design with crystal-clear audio and feature-rich capabilities at economical price. To know the list of features supported, refer to “SARVAM UCS Features Supported in Terminals”.

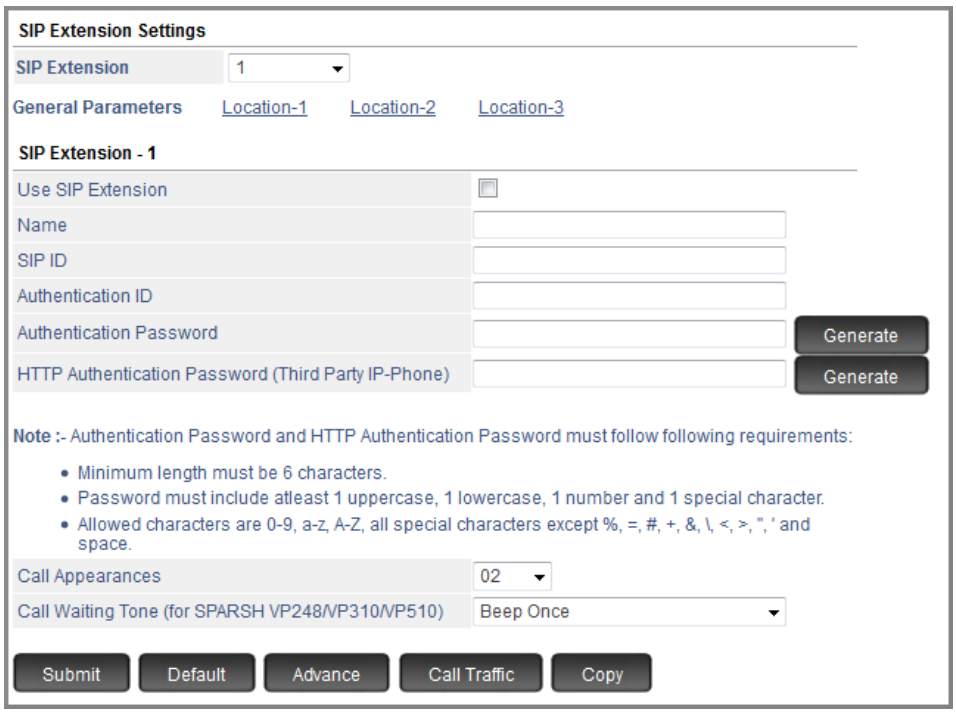
For instructions on how to use SPARSH VP510, refer to the *EON510_SPARSH VP510 User Guide.*

To be able to use SPARSH VP510 - Extended IP Phone, you must configure the following:

- SIP Extension General Parameters, see “Configuring SIP Extension General Parameters”.
- SIP Extension Settings, see “Configuring SIP Extension Settings using Jeeves”
- Extended Phone Settings, see “Configuring Matrix Extended Phone Settings using Jeeves”
- Voice Mail Settings, if you want to provide mailbox facility to the extension. See “Extension Voice Mail Settings”.

Configuring SIP Extension Settings using Jeeves

- Log into Jeeves as System Engineer.
- Under **Configuration**, click **VoIP Configuration**.
- Click **SIP Extension Settings**.



The page of SIP Extension 001 opens.

- You may select the **SIP Extension** number you want to configure.
  The parameters of the SIP Extension number you selected will appear on this page.

For SARVAM UCS upto 999 SIP Extensions can be registered with the system.

- Select the **Use SIP Extension** check box to enable the SIP extension. Default: disabled.

- In the **Name** field, enter a name for the SIP Extension, which may be the name of the person who will use the SIP Extension or the name of a Department. The name you enter here will be displayed as the Caller ID of the SIP Extension on the remote user's phone, when the SIP Extension user makes calls.

  The name may consist of a maximum of 18 alphanumeric characters.

- Enter the **SIP ID** for the extension. The SIP ID is necessary for registering the SIP Extension with the Registrar of the system. It is the number with which you can call the SIP Extension. Any extension user of the SARVAM UCS can call a SIP Extension by dialing the SIP ID assigned to the SIP extension. SIP ID of each SIP Extension must be a unique number string of a maximum of 6 digits. Any combination of digits from 0 to 9 and the characters * and # are allowed.

  You cannot assign the same SIP ID to more than one extension.

  To assign SIP IDs according to your preference and requirement to a range of SIP Extensions, see "Assigning Access Codes to a Range of Extensions".

  By default, the SIP IDs are Blank.

The SIP ID will be set to default value (blank), when you restore the default settings of the system.

- In **Authentication ID**, enter the number which you want the system to use for user authentication of the SIP messages received from the SIP Extension. You cannot keep this field blank and it must be unique for each SIP extension. The number may be a string of maximum 6 alphanumeric characters. All ASCII characters except < > and " (double quote) are allowed. Default: Blank.

Make sure the User ID configured in "Digest Authentication" does not conflict with the Authentication ID configured above.

- In **Authentication Password**, enter the password manually or click **Generate** to automatically generate a unique password. This password will be used by the system to authenticate the SIP messages received from the SIP Extension. To avoid unauthorized access, we recommend you to change the password regularly. Make sure it is strong and is kept confidential. When you enter the password manually, the password must:

  - be of minimum 6 characters and can be a maximum of 12 characters.
  - include atleast one upper-case, one lower-case, one number and one special character.
  - all ASCII characters (except Percentage **%**, Hash **#,** Equal to **=,** Plus **+**, And **&**, Backslash **\,** Less than **<,** Greater than **>,** Apostrophe **',** Double Quote **"** and **Space**) are allowed.
    Default: Blank.

    To provide additional security, when the Authentication fails 10 times consecutively due to wrong Authentication ID / Authentication Password, the system will blacklist the IP Address and Port for registration of this SIP Extension. However, you can remove the IP Address from the Blacklist IP Address list. See "Black List IP Address - SIP Extensions" for more details. This activity will be logged in the "System Activity Log" as well as "Simple Network Management Protocol (SNMP)".

Make sure you note down or copy the Authentication Password in a confidential file.

- In **Call Appearances**, define the maximum number[150] of simultaneous incoming calls that the SIP Extension user should be allowed to receive. You can set up to 10 call appearances for a SIP Extension. Default: 2.

  When Call Appearance is set to 2, the SIP Extension can receive 2 calls at a time.

- During an on-going conversation, if there is a second incoming call, the system plays beeps to indicate the second incoming call. You can set the frequency of the **Call Waiting Tone (for SPARSH VP248/VP310/ VP510)** beeps as per your requirement. You can select from the following options:
  - Off
  - Beep Once
  - Beep until Answered

  Default: Beep Once

  However, when an ongoing call is being taped or recorded, the call waiting tone for any new incoming call will not be played.

- Select the **Allow Standard SIP Registration** check box to allow the Standard SIP Extensions to get registered with the system. Default: Disabled.

- Under **Auto Sign-In,** enter the **Email ID** of the extension user you wish to store. The Email ID can be a maximum of 64 characters. The Email ID is used for various server features.

Auto Sign-In parameters — Send Configuration Mail and Mail Status, *are applicable only for Mobile Clients — VARTA ADR100, VARTA AMP100 applications.*

*The **Send Configuration Mail** button will appear only after you have enabled the SIP Extension and configured the SIP ID, Authentication ID and Password.*

- Under **Authentication**, enable Authentication of any or all of the following SIP Message Options by selecting the respective check boxes:
  - **INVITE Request**
  - **SUBSCRIBE Request**

  By default, the SIP Message Options INVITE and SUBSCRIBE are enabled.

Make sure that the Authentication ID for the SIP Extension has been programmed.

- For secure conversations over SIP, enable **SRTP Mode**. The SARVAM UCS supports the following options:

  - **Disable:** SARVAM UCS uses normal RTP for transporting the speech packets.

  - **Optional:** SARVAM UCS uses SRTP for transporting the speech packets. If the remote user does not support SRTP, SARVAM UCS will use normal RTP for transporting the speech packets.

    - If you select this option, you must configure the **SRTP Media Type**. You may select **AVP** or **SAVP**.

150. For the calls that are routed through the CPU, the number of Vocoder channels that will be supported would be as per the license you purchase.

By default, AVP is selected as the SRTP Media Type.

- **Forced:** SARVAM UCS uses only SRTP (SAVP) for transporting the speech packets. If the remote user does not support SRTP, SARVAM UCS will reject incoming calls from and drop outgoing calls made to such users.

  By default, SRTP Mode is Disabled.

- Assign a **SIP Hardware Template** to the SIP Extension. Default: 01. The "SIP Hardware Template" contains voice quality related features such as Voice Codec selection, Tx and Rx Gains, Echo Cancellation, Jitter Buffer and Fax-over-IP options and related parameters.

  There are 32 different templates to choose from. Each template can also be altered to suit your requirement and preferences. By default, Template number 01 assigned to all SIP Extensions as well as to SIP Trunks.Check if the values in this template fulfill requirements of the SIP Extension. If Template 01 fulfills the feature requirements, retain Template 01.

  If a different set of SIP hardware features are to be allowed to this SIP Extensions, prepare another template and assign it to this extension. To do this,

  - Under **VoIP Configuration**, click the **SIP Hardware Template** link.
  - Select a Template number, for example 02.
  - Customize Template number 02 and click **Submit**.
  - Return to the **SIP Extension Settings** page.
  - Select the number of the Template you customized, Template 02 in the **SIP Hardware Template** field.

  Also see the topic "SIP Hardware Template" to know more about customizing the templates and applying on the SIP Extensions.

- Assign a **Station Basic Feature Template** to the SIP Extension. Default: The "Station Basic Feature Template" has a set of features like Time Table, Class of Service, Toll Control, Operator, Storage of Incoming and Outgoing Calls, Outgoing Trunk Bundle groups. There are 50 different templates to choose from. Each template can also be altered to suit your requirement and preferences.

  If the default Station Basic Feature Template 01 fulfills the feature requirements of the SIP Extension ("Class of Service (COS)", "Toll Control", "OG Trunk Bundle Group", etc.) retain this template, you may also customize this template. If you want to assign a different set of features to this SIP Extension, prepare a different Station Basic Feature Template and apply it to this extension. To do this,

  - Under **Configuration**, click the **Station Basic Feature Template** link.
  - Select a Template number, for example 05.
  - Customize Template number 05 and click **Submit**.
  - Return to the **SIP Extension Settings** page.

- Select the number of the Template you customized, Template 05, in the **Station Basic Feature Template** field.

- Click **Submit** to save changes.

Also, see the topic "Station Basic Feature Template" to know more about customizing the templates and applying on extensions.

- Assign a **Station Advanced Feature Template** to the SIP Extension. Default: Template 01. The "Station Advanced Feature Template" has a set of advanced features for extensions such as Alarm Notification settings, Routing of Incoming Auto Attendant Calls, Call Duration Control, Floor Service, etc. There are 50 different templates to choose from. Each template can also be altered to suit your requirement and preferences.

  Check if the default template fulfills the feature requirements of the SIP Extension by clicking the **Station Advanced Feature Template** link.

  You may retain this template and customize it further, or customize another template if a different set of features are to be allowed to this SIP Extension. To customize/prepare another template,

  - Under **Configuration**, click the **Station Advanced Feature Template** link.

  - Select the Template number, for example 02, and customize this template.

  - Click **Submit** and return to the 'SIP Extension Settings' page.

  - In the **Station Advanced Feature Template** field, select the number of the template you customized.

  - Click **Submit** to save changes.

  Also see the topic "Station Advanced Feature Template" for instructions on customizing these templates and applying them on the extensions.

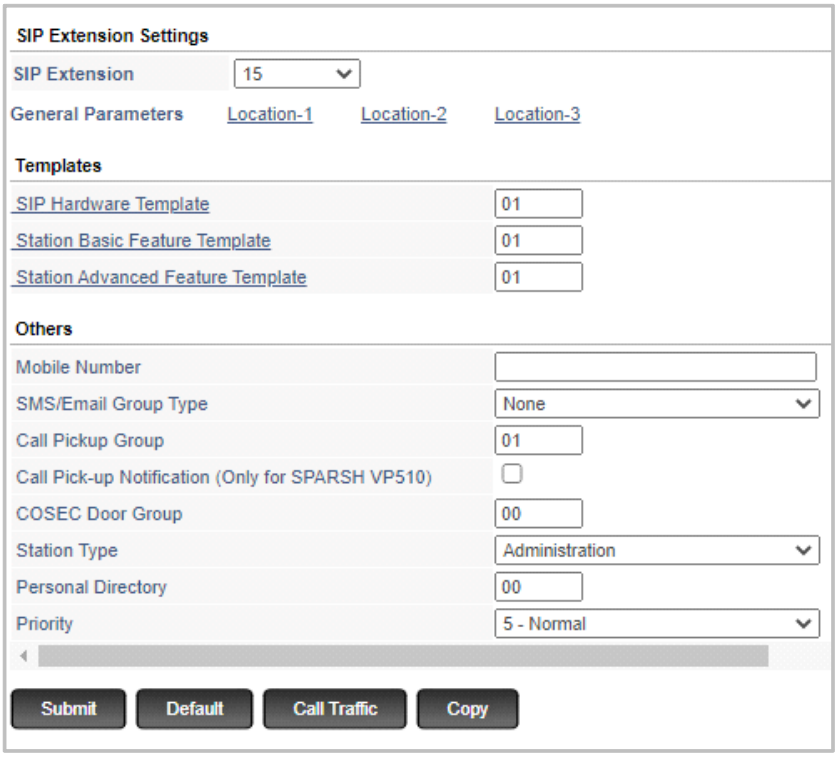
- In **Voice Mail Settings**, click the Voice Mail Settings link. The respective Extension Voice Mail Settings window will open. You may edit the parameters. For details, see "Extension Voice Mail Settings".

The Voice Mail Settings link will be visible only if you have configured the respective SIP ID.

- Click **Close** to close the window.

Advanced Configuration Parameters

- If you want to provide other features like Personal Directory, Priority, or assign a Station Type to the SIP Extension, click the **Advanced** button at the bottom of the page.



- Enter the **Mobile Number** of the extension user you wish to store. The Number can be a maximum of 16 digits.

- You can assign the extension user to a Group. Select the desired **SMS/Email Group Type** from the list.The system clubs together extension users assigned the same Group. Default: None. For details, see “SMS/Email Group”.

- Assign the SIP Extension to a **Call Pick-up Group**, if required. Default: 01

  Call Pick Up allows the SIP Extension user to 'pick up' (answer) calls ringing on any other extension, by dialing a feature code, without physically going to the ringing extension. It also allows incoming calls for the SIP Extension to be answered by the other extensions assigned the same Call Pick-Up group.

  For this to work, both the ringing extension and the extension picking up the call must be in the same 'Call Pick Up Group'. Refer “Call Pick Up” for instructions on how to create groups. You can create as many as 99 groups numbered from 01 to 99.

  Enter the number of the Call Pick-Up Group you created for this SIP Extension in this field.

- Select the **Call Pick-up Notification (Only for SPARSH VP510)** check box, if you want the Call Pick-up Notification to be displayed on the phone LCD. The Call Pick-up Notification will be displayed for internal as well as external calls. The notification will be displayed on SPARSH VP510 only if the ringing extension is in the same Call Pick-up Group. The notification will be displayed only when SPARSH VP510 is in idle or dial state. The notification will display the name/number of the Caller along with the name/number of the Called. For example, if the name and number of the Caller is ABC, 2001 and the name and number of

Called is XYZ, 1001, the LCD notification will be displayed as ABC -> XYZ. Along with the LCD display the phone back-light will also be lit.

For this to work, make sure you have enabled Call Pick-up in COS as well as assigned a Call Pick-up Group to the extension. Refer to "Call Pick Up" and "Class of Service (COS)". Call Pick-up Notifications will be displayed for DKP, SLT as well as SIP Extensions and for calls landing through CO, SIP as well as T1E1 Trunks. For details of the Notification, refer to the EON510_SPARSH VP510 V2 User Guide.

- You must assign the extension user to a **COSEC Door Group** for COSEC Integration. The users in the same group must be assigned the same group. You can create as many as 50 groups numbered from 00 to 50. Users who are assigned COSEC Door Group '00' are not a part of any group. See "COSEC Integration" for more information.

- If this is an Operator extension and you want the system to play beeps during a conference to the participants, to indicate the presence or absence of the Operator, select the **Station Type** as **Assistant**.

  If you are using the system in the *Hotel Mode*, select the **Station Type** for the SIP Extension as **Administration/Assistant** or **Guest**. The system will consider the options Administrator and Assistant as same.

- You may assign a **Personal Directory** number to the SIP Extension. Default: 00.

  A Personal Directory is a list of 25 frequently dialed numbers, each of which are stored by Index number (location code), Name and Trunk Access Codes ("Out Going Trunk Bundle Group Index"). The Personal Directory is necessary for using the features "Abbreviated Dialing" and "Dial By Name".

  When a Personal Directory is assigned to a SIP Extension, make sure you also configure this directory. The Personal Directory can be programmed by the SIP Extension users and by the System Engineer. Refer the topic "Abbreviated Dialing" for instructions on programming the Personal Directory.
  If Personal Directory is not to be assigned, enter 00 in this field.

- Select a **Priority** Level for the SIP Extension from 1 to 9. Default; 5-Normal.

  Each extension of the SARVAM UCS is assigned a Priority Level starting from 1, 2, 3... to 9, with '1' being lowest Priority and '9' being highest Priority. Whenever an extension (phone) with higher priority calls an extension with lower priority, a triple ring is placed on the called extension. To know more, read the feature description "Priority".

  If this SIP extension is assigned to Operator, you may want to set a higher priority for this extension.

- Click **Submit** to save your SIP Extension Settings.

Configuring Matrix Extended Phone Settings using Jeeves

You can register three Matrix Extended IP Phones/Soft Clients at three different locations as a single SIP Extension. You can connect/register the same or different types of Extended Phones/Soft Clients —SPARSH VP248, SPARSH VP310, SPARSH VP330, SPARSH VP510, SPARSH VP210, Matrix Extended SPARSH VP710, Matrix VARTA ADR100 Mobile UC Client, VARTA AMP100 Mobile UC Client or MATRIX VARTA WIN200 Desktop UC Client— at each location. In this case we assume that SPARSH VP510 is connected at Location 1, 2 and 3.

If you have connected SPARSH VP248 at any of the locations, refer to "Configuring Matrix Extended Phone Settings using Jeeves".

If you have connected SPARSH VP310 at any of the locations, refer to “Configuring Matrix SPARSH VP310”.

If you have connected SPARSH VP330 at any of the locations, refer to “Configuring Matrix SPARSH VP330”.

If you have connected Extended SPARSH VP710 at any of the locations, refer to “Configuring Matrix Extended SPARSH VP710”.

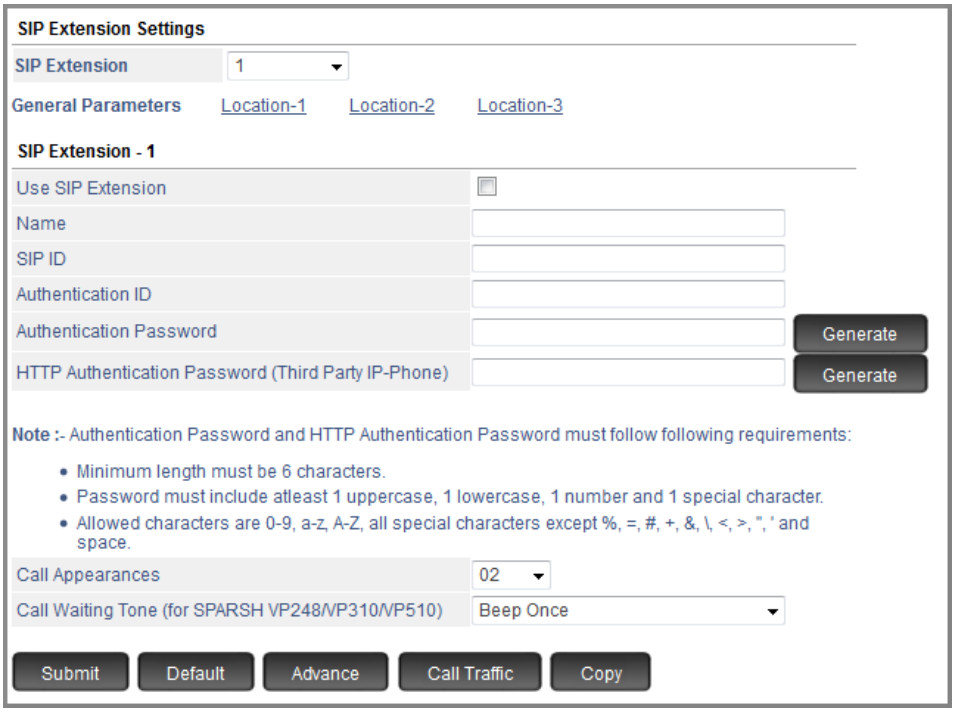
If you have connected Extended SPARSH VP210 at any of the locations, refer to “Configuring Matrix SPARSH VP210”.

If you have registered Matrix VARTA ADR100 and VARTA AMP100 Mobile UC Clients in any of the locations, refer to “Configuring Matrix VARTA ADR100/AMP100 UC Clients”.

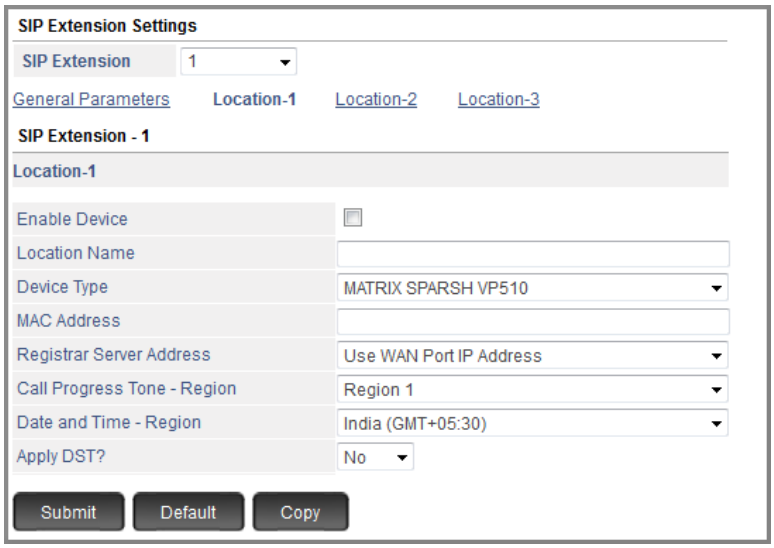
If you have registered MATRIX VARTA WIN200 Desktop UC Client in any of the locations, refer to “Configuring Matrix VARTA WIN200 UC Client”.

If you want to use more than one SPARSH VP510 Extended IP Phones as a SIP Extension, configure their settings at **Location 1**, **Location 2** and **Location 3**.

- Click **Location 1**.



- The settings of the phone at **Location 1** appear.



- Select the **Enable Device** check box. Default: Disabled.

- Enter the **Location Name** for the phone to identify the phone. Location name may be the place where the phone is located (e.g.: Head office, branch, residence). The Location Name may consist of 18 characters (maximum). Default: Blank.

- Select **MATRIX SPARSH VP510** as the **Device Type** at this location.

- Enter the **MAC Address**[151] of the SPARSH VP510 connected at this location in hexadecimal format: 00:1b:09:XX:XX:XX. Default: Blank.

  SARVAM UCS validates the Extended Phone on the basis of the MAC Address, and provides configuration on validation.

  As SARVAM UCS allows registration of the SIP Extension from three different locations, it identifies the SIP Extension in each location by the programmed MAC address.

- Select the appropriate **Registrar Server Address** to register the SPARSH VP510 with the SIP Registrar of SARVAM UCS, according to your installation scenario:

  - If the SPARSH VP510 is connected on the WAN network, select **Use WAN Port IP Address** as Registrar Server IP Address.

  - If the SPARSH VP510 is connected on the LAN network, select **Use LAN Port IP Address** as Registrar Server IP Address.

  - If the SPARSH VP510 is connected in the Global Network and SARVAM UCS is located behind a Router, or behind a NAT Router and STUN is programmed, select **Use Router/STUN's IP Address** as Registrar Server IP Address.

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151. MAC address is the address of the electronic hardware devices such as a computer, which is hard-coded into the device during manufacture and cannot be modified. No two devices can have similar MAC address and thus it uniquely identifies your phone. MAC address is assigned as per the IANA standard. The MAC Address of the phone will be used as source MAC address on all Ethernet frames.

Make sure you configure either the **Router's Public IP Address** or **Simple Traversal of UDPs through NATs (STUN)** in Network Parameters. For details, see "Configuring Network Parameters".

- If Dynamic DNS is configured in the Network Parameters, select **Use Dynamic DNS Host Name** as Registrar Server IP Address.

  By default, Use WAN Port IP Address is selected as the Registrar Server IP Address.

- To set the call progress tone generation standards of the country where the SPARSH VP510 is installed, select the **Call Progress Tone - Region**. Default: Region 1.

  See "Call Progress Tones" to know more.

- To display the Date and Time of the country where the SPARSH VP510 is installed, select the **Date and Time - Region**. Default: India.

- If you want to enable Daylight Saving Time (DST) on the phone, set **Apply DST?** to either **Manual** or **Scheduled** as per your requirement.

  When you select **Scheduled** as the DST option, the Real Time Clock of SARVAM UCS is advanced and set backward automatically according to the DST convention of the country/region where the IP Phone is installed.

  Scheduled DST Adjustment is useful in countries/regions where DST Time is fixed, such as in Europe, USA and Canada, without yearly variations.

  SARVAM UCS supports 18 DST Types for Scheduled DST Adjustment. To know more, refer to "Daylight Saving Time (DST)". To know more about Scheduled DST assigned for the respective region, refer to *Time Zone* in "Default Settings".

  When you select **Manual** as the DST option, the Real Time Clock of SARVAM UCS is advanced manually and set backward automatically according to the DST convention of the country/region where the IP Phone is installed.

  Manual DST Adjustment is to be used in regions/countries that have no fixed DST Convention and where yearly variations in DST practices are likely.

  When DST option is set as 'Manual', you must set the DST Start and the DST End time, that is, the time at which the clock is to be advanced and the time at which the clock is to be delayed. To do so,

  - In **Time Offset,** enter the time you wish to forward or backward the DST start time with.

  - In **DST Type**, select the desired option: **Date-Month Wise** OR **Day-Month Wise**.

    If you select '**Date-Month Wise**' in **DST Type**, you should now select the desired options in each of the following to specify the **DST Start** details.

    - **Date**: Select the date on which DST begins (1-31).
    - **Month**: Select the month when DST begins (January-December).
    - **Time (Hours)**: Select the time at which DST will begin to change.The time mode is of 24 hours ranging from 00 to 23 hours.
    - **Time (Minutes):** Select the time at which DST will begin to change.The time mode is of 60 minutes ranging from 00 to 59 minutes.

Similarly, in the **DST End** configure the desired DST End Time details.

If you select **'Day-Month Wise'** in **DST Type**, you should now select the desired options in each of the following to specify the **DST Start** details.

- **Ordinal**: Select the Ordinal number of the day of the month, that is, the 1st, 2nd, 3rd, 4th, 5th day, when DST begins.
- **Day:** Select the day of the month - Sunday, Monday, Tuesday, Wednesday, Thursday, Friday, Saturday- when DST begins.
- **Month**: Select the month when DST begins (January-December).
- **Time (Hours)**: Select the time at which DST must begin to change.The duration is of 24 hours ranging from 00 to 23.
- **Time (Minutes):** Select the time at which DST must begin to change.The duration is of 60 minutes ranging from 00 to 59.

Similarly, in **DST End** configure the DST End Time details.

Once the DST Ends, the time of the IP Phone is set back to the Standard time automatically.

When the DST of a particular country starts or ends on the Last Sunday or any other day, for example, the last Tuesday, last Friday of the month, always set the Ordinal Number as '5th'.

- If you want to enable Daylight Saving Time (DST) on the phone, set **Apply DST?** to Yes. Default: No. The Daylight Saving Time convention followed in the country/region you selected will be automatically applied. The SPARSH VP510 will change its date and time settings according to the DST convention of the selected country/region.

- Select the **CO CLIP Pattern** for the SPARSH VP510. This is the type of Calling Line Presentation on the phone for incoming calls from trunks. You can select any of these options:
  - **Name Only** (only the name of the caller will be displayed).
  - **Number Only** (only the number of the caller will be displayed).
  - **Number + Name** (both the name and the number of the caller will be displayed).

  Default: Number + Name.

- Select the **Language** for the SPARSH VP510. Default: English.

  SARVAM UCS provides language support for English, French, German, Spanish, Portuguese, and Italian on the SPARSH VP510. When you select any of these languages, all the prompts and command strings will appear in the selected language.

SIP Extension users can change the language by accessing and navigating through the phone menu.

The SA can change the Language by logging into the SA Jeeves.

- Select a **Ringer Mode** for the phone from the four options:
  - Ring immediately (it rings immediately as a fresh calls lands on the phone).
  - Ring if idle (rings only if the phone is idle).
  - Ring after a delay (if the call is still not answered).
  - Silent.

  Default: Ring Immediate.

- If you selected *Ring after a delay* as Ringer Mode, set the **Ring Delay Timer (sec)**, if required, to the desired value.

  The Ring Delay Timer is the time in seconds the system waits on receiving a call before ringing on the phone. The range of this timer is 0 to 99 seconds. Default: 10 seconds.

- If you want to enable *Ringer Auto Acknowledge* mode, set the **Acknowledge Timer** (sec) to the desired value.

  The Ringer Auto Acknowledge mode determines when to stop the ring on the phone. There are two options for Ringer Auto Acknowledge:
  - Stop only when the call is answered.
  - Stop after a delay.

  To stop the ring on the phone after a delay, the Acknowledge Timer must be configured. The range of this timer is 01 to 99 seconds. Default: 00 seconds.

  To stop the ring only when the Call is answered or manually acknowledged, the Acknowledge Timer must be set to '00'. By default, Ring Auto Acknowledge is turned OFF.

- To assign the Ring Destination for the SPARSH VP510, select the desired destination for **Play Ring on**. You may choose
  - **Speakerphone:** The ring will be played on the Speakerphone.
  - **Headset:** The ring will be played on the Headset.

  Default: Speakerphone.

  When you select the Headset as the destination, make sure that you set the flag '*Headset Connected?'* to Yes, connect a Headset to the SPARSH VP510.

- Set the **Ringer Volume** to the desired level, from 0 to 7, according to your preference. Default: 4.

You can also set the Ringer Tune. For detailed instructions, refer to the EON510_SPARSH VP510 User Guide.

- To increase/decrease the volume of outgoing speech (Transmit Gain) on the handset of the SPARSH VP510, set the **Handset Transmit Volume Level** to the desired level, from 0 to 7. Default: 4.

- To increase/decrease the volume of incoming speech (Receive Gain) on the handset of the SPARSH VP510, set the **Handset Receive Volume Level** to the desired level, from 0 to 7. Default: 4.

- To increase/decrease the volume of outgoing speech (Transmit Gain) on the headset of the SPARSH VP510, set the **Headset Transmit Volume Level** to the desired level, from 0 to 7. Default: 4.

- To increase/decrease the volume of outgoing speech (Receive Gain) on the headset of the SPARSH VP510, set the **Headset Receive Volume Level** to the desired level, from 0 to 7. Default: 4.

- To change the Transmit Gain of the Speakerphone MIC Volume, set **Speaker Transmit Volume Level** to the desired level, from 0 to 7. Default: 4.

- To change the Receive Gain of the Speakerphone MIC Volume, set **Speaker Receive Volume Level** to the desired level, from 0 to 7. Default: 4.

- To increase the volume of the incoming speech on the handset, select the **Handset High Gain Mode** check box. This is useful for individuals with hearing aids. Default: Disabled.
- To use a Headset with the SPARSH VP510, select the **Headset Connected?** check box**.** Default: Disabled.

  Make sure that you connect a Headset to the SPARSH VP510, if you enable this option.
- Select the **Auto Answer** check box to enable this feature on the SPARSH VP510. Default: Disabled.

  When you set the "Auto Answer" feature on the SPARSH VP510, the phone goes OFF-Hook automatically after a preset period of time, without the extension user having to pick up the handset or press the speaker or headset key. When you enable Auto Answer, you must configure the Auto Answer Timer.
- If you enabled Auto Answer on the phone, set the **Auto Answer Timer (sec)** to the desired value.

  This timer defines the time in seconds that the SPARSH VP510 should wait before going OFF-Hook to auto answer a call. The range of this timer is 1 to 9 seconds. Default: 1 second.
- Adjust the Backlight brightness of the phone's LCD display, by setting the **LCD Backlight Level** to the desired value, from 1 to 4. Default: 3.
- Set the **Back Light Off Timer (sec)** to the desired value, if required, from 000 to 999 seconds. Default: 10 seconds.
- Set the **LCD Contrast Level** to a level from 1 to 4 that is comfortable to you. Default: 3.

DSS Key Settings

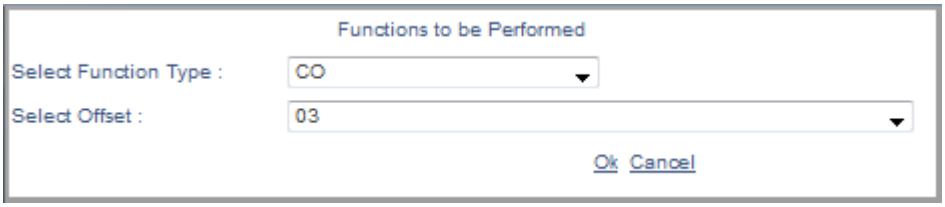
- You can select the desired key template — Operator, Executive1, Executive2, Executive3, Hotel Attendant, Guest or any other template you added. See "Customizing Extended IP Phone Templates using Jeeves" for more details.

  OR
- You can personalize the key map of the SPARSH VP510 for this location. To do so,
  - Select **Personalized** as the **Key Template** option.
  - Click **Submit**.
  - Click **Key Settings**.

  - The key map of the Extended Phone opens in a new window on your screen.

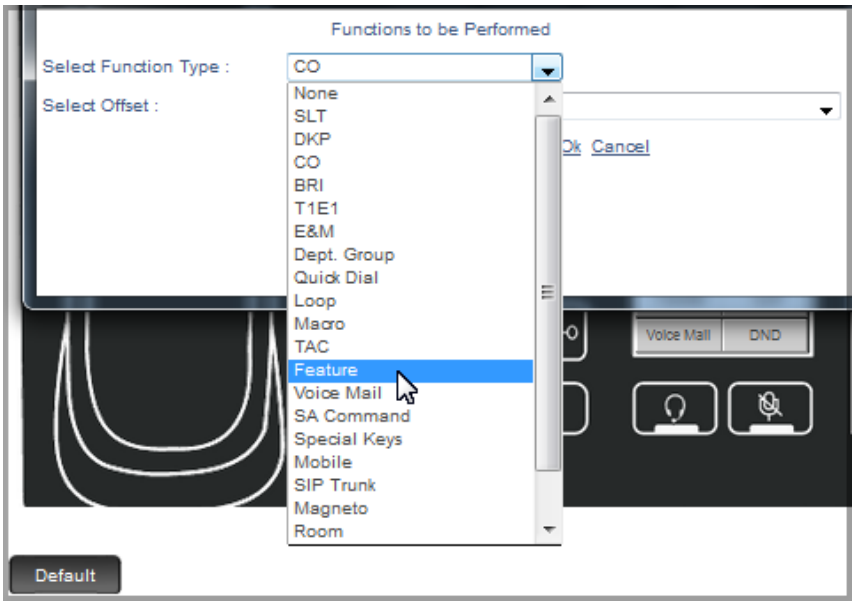
- Click the key you want to configure. For example, **CO3**.

  The **Functions to be Performed** by the key opens in a new window.



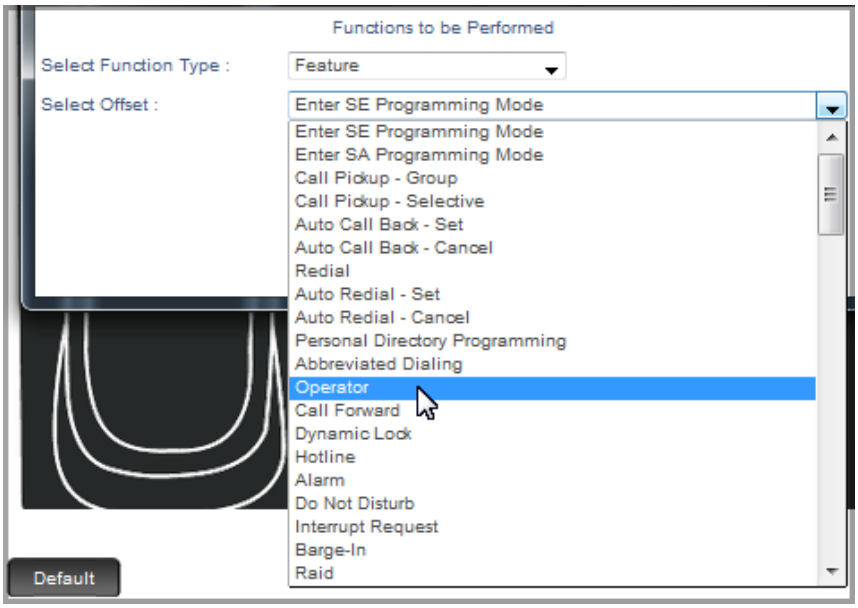
- In the **Select Function Type** list, select the function to be performed by the key. For example, you want to use the key to call the **Operator**.

The Operator function is a Feature, so select the option **FEATURE** from the **Select Function Type** list box.

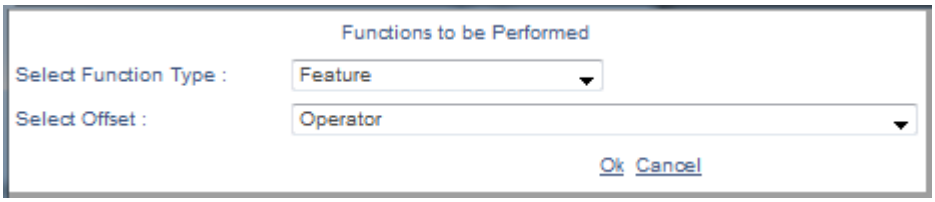


From the **Select Offset** drop down list, all the features that can be assigned to keys are listed.

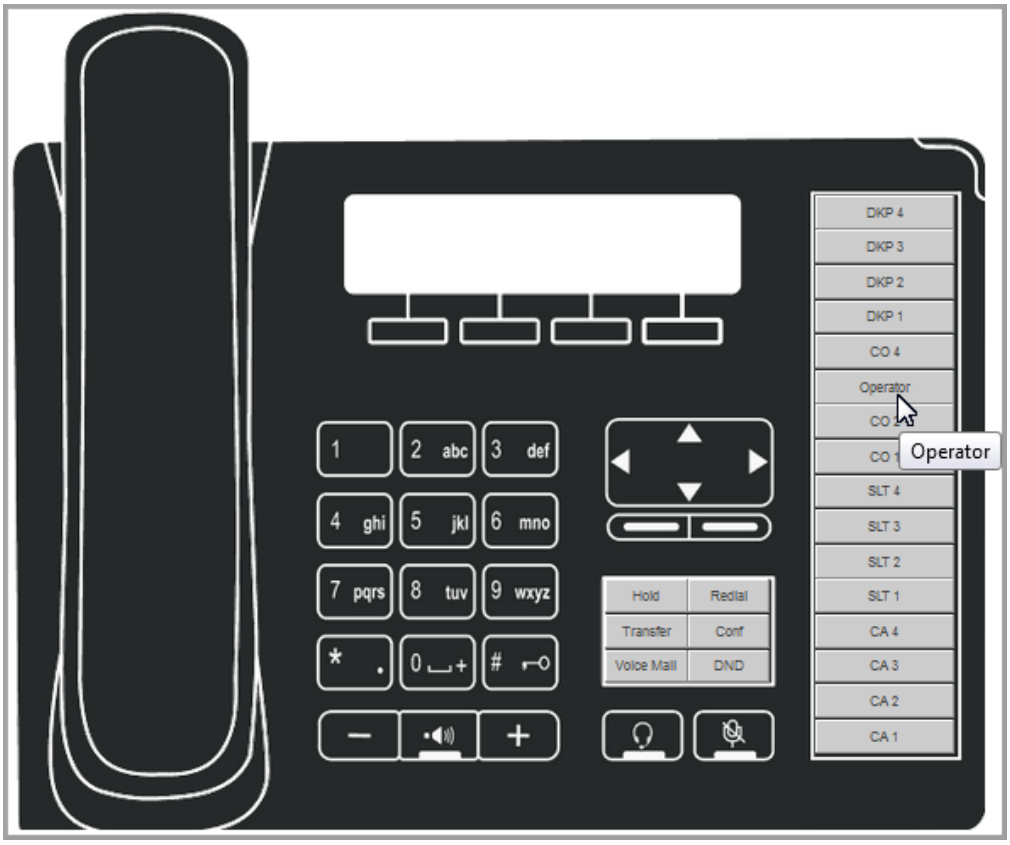
- Select **Operator** from the list of features in the **Select Offset** box.



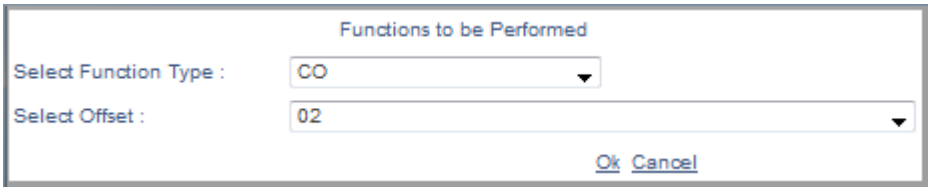
- Click **OK**.



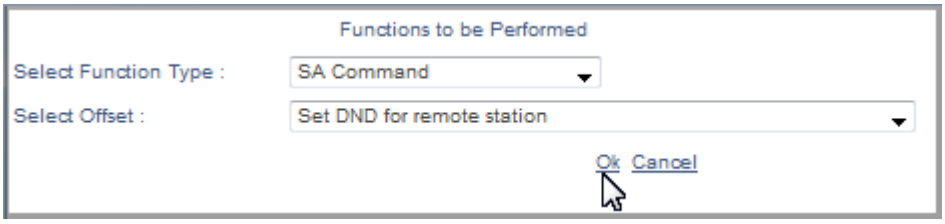
The *Operator* feature appears on the key label.



- To take a second example, if you want to assign **Remote DND** to the key currently assigned **CO 2** key, click the key.

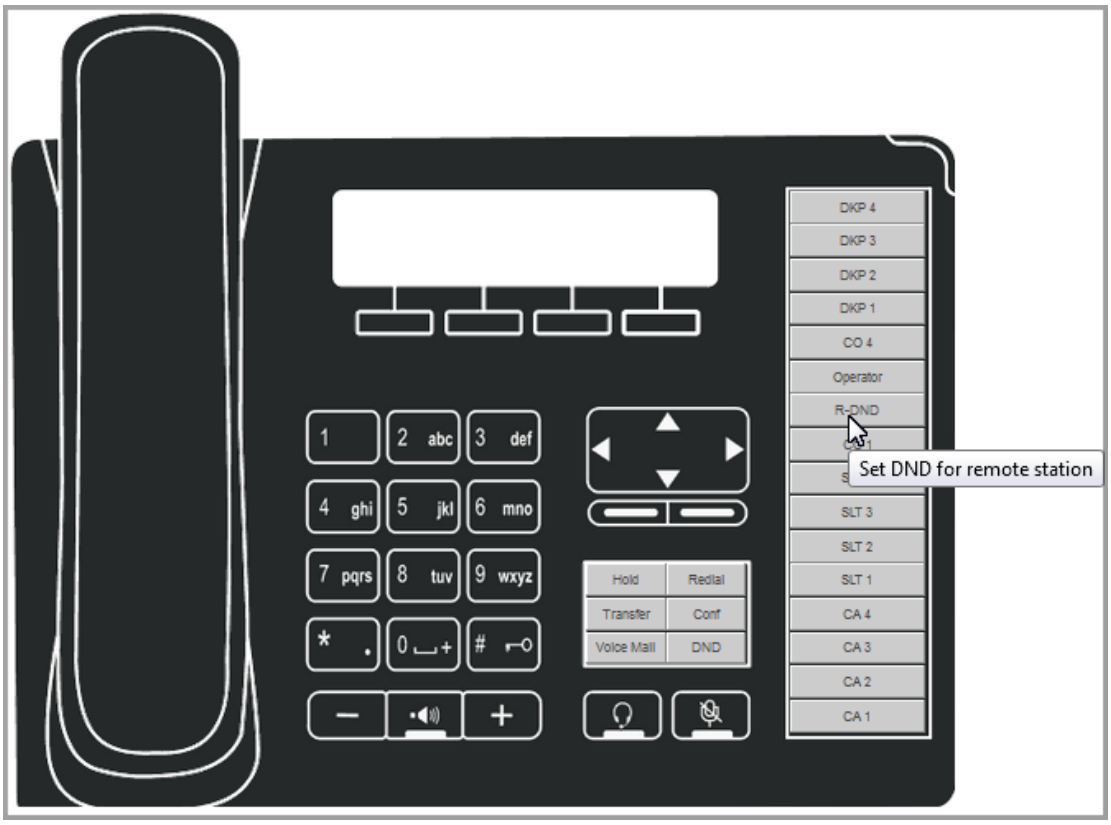


- In the **Select Function Type** list box, select the option **SA Command**, as Remote DND is a System Administrator (SA) Command.



- In the **Select Offset** box, select the option **Set DND for remote station**.

- Click **OK**. The box closes. Remote DND feature will appear in abbreviated form as *R-DND* on the key label.



- Follow the same instructions to assign features to other DSS keys. Selecting the appropriate Function Type and the Offset for each feature/function.

  If you want assign a feature, select **FEATURE** as function type, and select the desired feature as Offset.

  If you want to use the key to call a DKP or a SIP extension, select **DKP** or **SIP Extension** as Function Type and select the number of the extension as Offset.

  To assign direct access to a Mobile Trunk, select **MOBILE** as Function Type and the desired port number **01** or any other trunk number as Offset.

  To assign direct access to a SIP Trunk, select **SIP** as Function Type and the desired trunk number from **01** or any other trunk number as Offset.

  Click **OK**, each time you select a Function Type and Offset in the dialog box.

  You can reinstate default key assignment any time, by clicking the **Default** button at the bottom of the window.

- When you complete assigning functions to keys, close the window.

- If you assign/re-assign functions to the following keys, the Phone will restart:
  - Speaker
  - Headset
  - Ringer Acknowledge

- Local Menu

- You can also connect a DSS Console (DSS532) with SPARSH VP510. For instructions:

  - to install the DSS532 with SPARSH VP510, see “Installing DSS532 with SPARSH VP510”.
  - to configure the DSS keys of the Console, see “Configuring DSS Console Keys connected to SPARSH VP510”.

Transport Mode and SRTP

- Select the protocol to be used to transport the SIP messages. You can select the **Transport Mode** as **TCP** or **TLS**.

If you select TCP, make sure the SIP Over TCP check box is selected in VoIP Parameters.

If you select TLS, make sure the SIP Over TLS check box is selected in VoIP Parameters.

- For secure conversations over SIP, select the **Enable SRTP?** check box. The SIP messages will be transported over SRTP only.

If you select this check box, make sure you have selected SRTP Mode as Forced or Optional in the General Parameters under SIP Extension Settings.

RTP Port

- Define the RTP Port:

  - **RTP Listening Port:** This is the port on which the SPARSH VP510 listens for RTP messages over UDP. This port is also used as the source port for sending RTP packets. This port is also used as the source port for sending RTP packets to the remote peer. The valid range for this port is 1025-65278. Default: 8000.

Quality of Service

- Set the **SIP Quality of Service (QoS)** for SIP signaling as:

  - **SIP DiffServe/ToS**. Valid range is 00 to 63. Default: 26.
    OR
  - **RTP DiffServe/ToS**. Valid range is 00 to 63. Default: 46.

NAT Keep Alive

- If the SPARSH VP510 is connected behind a NAT router, configure **NAT Keep Alive**.

  - Select the check box **Enable NAT Keep Alive** to send Keep Alive messages periodically to refresh the binding in the NAT router. Default: Disabled.

  - Define as **Interval (sec)**, the time period, from 001 to 999 seconds, after which the phone should send Keep Alive message. Default: 120 seconds.

    The time period you define should be less than the binding timer of the router.

Timers

- Set the following **Timers** to the desired value, where required:

  - **SIP INVITE Timer (sec):** This is the time in seconds that the phone waits for a response from the called party after ending INVITE message. This timer starts after sending INVITE message to the called party and stops on receipt of the provisional response or the final response or when the user disconnects the call. On expiry of the timer, the phone terminates the call process and gives an error tone to the user. The range of the SIP INVITE TIMER is 10-180 seconds. Default: 30 seconds.

  - **SIP Provisional Timer (sec)**: This is the time in seconds that the phone waits for final response after receiving the provisional response from the called party. This timer starts on the receipt of the provisional response from the called party and stops on receipt of the final response from the called party or when the user disconnects the call. On expiry of the timer, the IP phone terminates the call process and gives error tone to the user. The range of SIP Provisional Timer is 10-180 seconds. Default: 60 seconds.

  - **General Request Timer (sec):** This is the time in seconds for which the phone waits for response of a transaction request. This timer starts on initiating a transaction. This timer stops on receipt of a response for the request. On expiry of the timer, the phone clears the transaction. This timer is used for Registration request, etc. The range of the General Request Timer is 10-60 seconds. Default: 20 seconds.

Debug

- To debug using Syslog Client supported by the SPARSH VP510, configure Debug parameters:

  - Select the **Enable Debug?** check box. Default: disabled.

    When the Debug flag is enabled, the phone will send the debug messages to the Syslog Server IP address. Debug report can be viewed on the Syslog Server or any other application which can capture the Syslog messages/debug sent by the phone.

  - Enter the IP Address and port of the remote Syslog Server and as **Syslog Server Address and Server Port**.

    The address of the Listening Port of the Syslog Server is from 1025-65535;514. Default: 514. Syslog uses the UDP as transport protocol and listens on the port 514 (the default listening port).

  - You may select the **Debug Level** from the following options, by selecting the respective check box:
    - SIP
    - System
    - Hardware
    - Call
    - Network
    - VoPP

    You may select any or all of these debug levels. The Syslog Client will send only the debug messages for the selected level to the remote server on the IP network. For example, if the debug log of 'Call's is required, you can select this option, and disable all others.

- Click **Submit** to save settings.

- If you have completed the configuration of the SPARSH VP510 Settings at Location 1, follow the same steps as described above to configure the SPARSH VP510 at Location 2 and Location 3.

  However, if you want to replicate the configuration of SPARSH VP510 Settings same as Location 1 to Location 2 and Location 3, you can use the **Copy** button present at the bottom of the page. To know more, refer to “Copy Parameter Values”.

If you wish to Copy only the configurations from a location to all other locations where the same IP Phone is connected, make sure you clear the Device Type check box.

If you wish to copy the Device Type as well as all the configurations from a location to all the SIP Extensions, make sure all the check boxes are selected.

When you change any of the parameters listed below in the SIP Extension at Location 1, 2, 3, the phone will restart automatically, if registered:

- Use SIP Extension
- SIP ID
- Authentication ID
- Authentication Password
- Registrar Server IP Address
- MAC Address
- Enable Device
- Device Type
- Key Map in the Key Template assigned to phone
- Call Progress Tone
- Date and Time
- Apply DST?
- Transport Mode and SRTP
- QoS
- RTP Ports
- NAT Keep Alive
- SIP Timers
- The SE Password of SARVAM UCS is changed
- Specific parameters in VoIP Parameters are changed
- Specific parameters in Network Port parameters are changed
- You restart the System
- Set the System to Default

Configuring Matrix SPARSH VP210

SPARSH VP210 is the proprietary Entry Level IP Phone by Matrix which is engineered to offer a contemporary design with clear audio and feature-rich capabilities at economical price.To know the list of features supported, refer to “SARVAM UCS Features Supported in Terminals”.

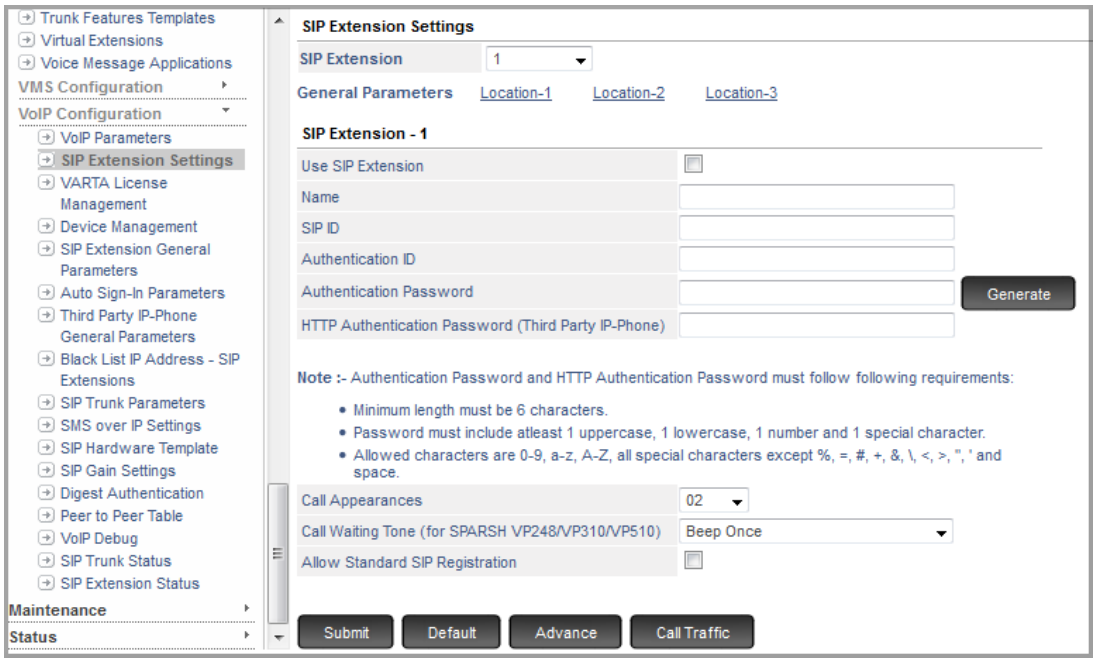
For detailed product information and operation instructions, refer to the *SPARSH VP210 (Extended) User Guide.*

To be able to use SPARSH VP210 as SIP Extensions, you must configure the following:

- SIP Extension General Parameters, see “Configuring SIP Extension General Parameters”.
- SIP Extension Settings, see “Configuring SIP Extension Settings using Jeeves”
- Extended Phone Settings, see “Configuring Matrix Extended Phone Settings using Jeeves”
- Voice Mail Settings, if you want to provide mailbox facility to the extension. See “Extension Voice Mail Settings”.

Configuring SIP Extension Settings using Jeeves

- Log into Jeeves as System Engineer.
- Under **Configuration**, click **VoIP Configuration**.
- Click **SIP Extension Settings**.



  The page of SIP Extension 001 opens.

- You may select the **SIP Extension** number you want to configure.

  The parameters of the SIP Extension number you selected will appear on this page.

For SARVAM UCS upto 999 SIP Extensions can be registered with the system.

- Select the **Use SIP Extension** check box to enable the SIP extension. Default: disabled.

- In the **Name** field, enter a name for the SIP Extension, which may be the name of the person who will use the SIP Extension or the name of a Department. The name you enter here will be displayed as the Caller ID of the SIP Extension on the remote user's phone, when the SIP Extension user makes calls.

  The name may consist of a maximum of 18 alphanumeric characters.

- Enter the **SIP ID** for the extension. The SIP ID is necessary for registering the SIP Extension with the Registrar of the system. It is the number with which you can call the SIP Extension. Any extension user of the SARVAM UCS can call a SIP Extension by dialing the SIP ID assigned to the SIP extension. SIP ID of each SIP Extension must be a unique number string of a maximum of 6 digits. Any combination of digits from 0 to 9 and the characters * and # are allowed.

  You cannot assign the same SIP ID to more than one extension.

  To assign SIP IDs according to your preference and requirement to a range of SIP Extensions, see "Assigning Access Codes to a Range of Extensions".

  By default, the SIP IDs are Blank.

The SIP ID will be set to default value (blank), when you restore the default settings of the system.

- In **Authentication ID**, enter the number which you want the system to use for user authentication of the SIP messages received from the SIP Extension. You cannot keep this field blank and it must be unique for each SIP extension. The number may be a string of maximum 6 digits. All ASCII characters except < > and " (double quote) are allowed. Default: Blank.

Make sure the User ID configured in "Digest Authentication" does not conflict with the Authentication ID configured above.

- In **Authentication Password**, enter the password manually or click **Generate** to automatically generate a unique password. This password will be used by the system to authenticate the SIP messages received from the SIP Extension. To avoid unauthorized access, we recommend you to change the password regularly. Make sure it is strong and is kept confidential. When you enter the password manually, the password must:

  - be of minimum 6 characters and can be a maximum of 12 characters.
  - include atleast one upper-case, one lower-case, one number and one special character.
  - all ASCII characters (except Percentage **%**, Hash **#,** Equal to **=,** Plus **+**, And **&**, Backslash **\,** Less than **<,** Greater than **>,** Apostrophe **',** Double Quote **"** and **Space**) and digits 0 to 9 are allowed. Default: Blank.

  To provide additional security, when the Authentication fails 10 times consecutively due to wrong Authentication ID / Authentication Password, the system will blacklist the IP Address and Port for registration of this SIP Extension. However, you can remove the IP Address from the Blacklist IP Address list. See "Black List IP Address - SIP Extensions" for more details. This activity will be logged in the "System Activity Log" as well as "Simple Network Management Protocol (SNMP)".

- In **Call Appearances**, define the maximum number[152] of simultaneous incoming calls that the SIP Extension user should be allowed to receive. You can set up to 10 call appearances for a SIP Extension. Default: 2.

  When Call Appearance is set to 2, the SIP Extension can receive 2 calls at a time.

- Under **Auto Sign-In,** enter the **Email ID** of the extension user you wish to store. The Email ID can be a maximum of 64 characters. The Email ID is used for various server features.

Auto Sign-In parameters — Send Configuration Mail and Mail Status, *are applicable only for Mobile Clients — VARTA ADR100, VARTA AMP100 applications.*

*The **Send Configuration Mail** button will appear only after you have enabled the SIP Extension and configured the SIP ID, Authentication ID and Password.*

- Under **Authentication**, enable Authentication of any or all of the following SIP Message Options by selecting the respective check boxes:
  - **INVITE Request**
  - **SUBSCRIBE Request**

  By default, the SIP Message Options INVITE and SUBSCRIBE are enabled.

Make sure that the Authentication ID for the SIP Extension has been programmed.

- For secure conversations over SIP, enable **SRTP Mode**. The SARVAM UCS supports the following options:

  - **Disable:** SARVAM UCS uses normal RTP for transporting the speech packets.

  - **Optional:** SARVAM UCS uses SRTP for transporting the speech packets. If the remote user does not support SRTP, SARVAM UCS will use normal RTP for transporting the speech packets.

    - If you select this option, you must configure the **SRTP Media Type**. You may select **AVP** or **SAVP**. By default, AVP is selected as the SRTP Media Type.

  - **Forced:** SARVAM UCS uses only SRTP (SAVP) for transporting the speech packets. If the remote user does not support SRTP, SARVAM UCS will reject incoming calls from and drop outgoing calls made to such users.

    By default, SRTP Mode is Disabled.

- Assign a **SIP Hardware Template** to the SIP Extension. Default: 01. The “SIP Hardware Template” contains voice quality related features such as Voice Codec selection, Tx and Rx Gains, Echo Cancellation, Jitter Buffer and Fax-over-IP options and related parameters

  There are 32 different templates to choose from. Each template can also be altered to suit your requirement and preferences. By default, Template number 01 assigned to all SIP Extensions as well as to SIP Trunks.

  Check if the values in this template fulfill requirements of the SIP Extension. If Template 01 fulfills the feature requirements, retain Template 01.

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152. For the calls that are routed through the CPU, the number of Vocoder channels that will be supported would be as per the license you purchase.

If a different set of SIP hardware features are to be allowed to this SIP Extensions, prepare another template and assign it to this extension. To do this,

- Under **VoIP Configuration**, click the **SIP Hardware Template** link.
- Select a Template number, for example 02.
- Customize Template number 02 and click **Submit**.
- Return to the **SIP Extension Settings** page.
- Select the number of the Template you customized, Template 02 in the **SIP Hardware Template** field.

Also see the topic "SIP Hardware Template" to know more about customizing the templates and applying on the SIP Extensions.

- Assign a **Station Basic Feature Template** to the SIP Extension. Default: The "Station Basic Feature Template" has a set of features like Time Table, Class of Service, Toll Control, Operator, Storage of Incoming and Outgoing Calls, Outgoing Trunk Bundle groups. There are 50 different templates to choose from. Each template can also be altered to suit your requirement and preferences.

  If the default Station Basic Feature Template 01 fulfills the feature requirements of the SIP Extension ("Class of Service (COS)", "Toll Control", "OG Trunk Bundle Group", etc.) retain this template, you may also customize this template. If you want to assign a different set of features to this SIP Extension, prepare a different Station Basic Feature Template and apply it to this extension. To do this,

  - Under **Configuration**, click the **Station Basic Feature Template** link.
  - Select a Template number, for example 05.
  - Customize Template number 05 and click **Submit**.
  - Return to the **SIP Extension Settings** page.
  - Select the number of the Template you customized, Template 05, in the **Station Basic Feature Template** field.
  - Click **Submit** to save changes.

  Also, see the topic "Station Basic Feature Template" to know more about customizing the templates and applying on extensions.

- Assign a **Station Advanced Feature Template** to the SIP Extension. Default: Template 01. The "Station Advanced Feature Template" has a set of advanced features for extensions such as Alarm Notification settings, Routing of Incoming Auto Attendant Calls, Call Duration Control, Floor Service, etc. There are 50 different templates to choose from. Each template can also be altered to suit your requirement and preferences.

  Check if the default template fulfills the feature requirements of the SIP Extension by clicking the **Station Advanced Feature Template** link.

You may retain this template and customize it further, or customize another template if a different set of features are to be allowed to this SIP Extension. To customize/prepare another template,

- Under **Configuration**, click the **Station Advanced Feature Template** link.
- Select the Template number, for example 02, and customize this template.
- Click **Submit** and return to the 'SIP Extension Settings' page.
- In the **Station Advanced Feature Template** field, select the number of the template you customized.
- Click **Submit** to save changes.

Also see the topic "Station Advanced Feature Template" for instructions on customizing these templates and applying them on the extensions.

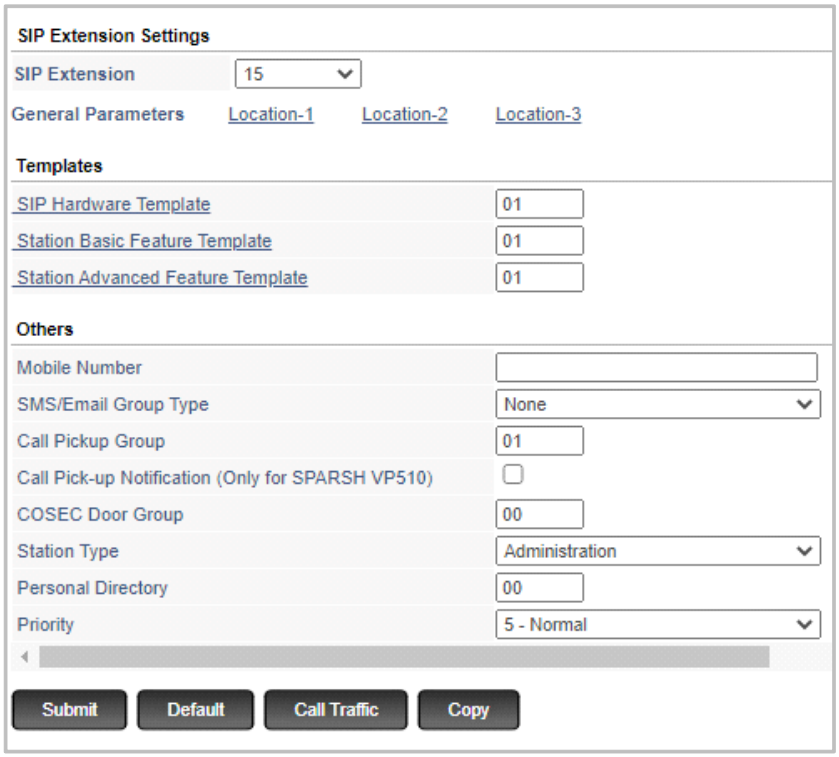
- In **Voice Mail Settings**, click the Voice Mail Settings link. The respective Extension Voice Mail Settings window will open. You may edit the parameters. For details, see "Extension Voice Mail Settings".

The Voice Mail Settings link will be visible only if you have configured the respective SIP ID.

Click **Close** to close the window.

Advanced Configuration Parameters

- If you want to provide other features like Personal Directory, Priority, or assign a Station Type to the SIP Extension, click the **Advanced** button at the bottom of the page.



- Enter the **Mobile Number** of the extension user you wish to store. The Number can be a maximum of 16 digits.

- You can assign the extension user to a Group. Select the desired **SMS/Email Group Type** from the list.The system clubs together extension users assigned the same Group. Default: None. For details, see “SMS/Email Group”.

- Assign the SIP Extension to a **Call Pick-up Group**, if required.

  Call Pick Up allows the SIP Extension user to 'pick up' (answer) calls ringing on any other extension, by dialing a feature code, without physically going to the ringing extension. It also allows incoming calls for the SIP Extension to be answered by the other extensions assigned the same Call Pick-Up group.

  For this to work, both the ringing extension and the extension picking up the call must be in the same 'Call Pick Up Group'. Refer “Call Pick Up” for instructions on how to create groups. You can create as many as 99 groups numbered from 01 to 99.

  Enter the number of the Call Pick-Up Group you created for this SIP Extension in this field.

- Select the **Call Pick-up Notification (Only for SPARSH VP510)** check box, if you want the Call Pick-up Notification to be displayed on the phone LCD. The Call Pick-up Notification will be displayed for internal as well as external calls. The notification will be displayed on SPARSH VP510 only if the ringing extension is in the same Call Pick-up Group. The notification will be displayed only when SPARSH VP510 is in idle or dial state. The notification will display the name/number of the Caller along with the name/number of the Called. For example, if the name and number of the Caller is ABC, 2001 and the name and number of

Called is XYZ, 1001, the LCD notification will be displayed as ABC -> XYZ. Along with the LCD display the phone back-light will also be lit.

For this to work, make sure you have enabled Call Pick-up in COS as well as assigned a Call Pick-up Group to the extension. Refer to "Call Pick Up" and "Class of Service (COS)". Call Pick-up Notifications will be displayed for DKP, SLT as well as SIP Extensions and for calls landing through CO, SIP as well as T1E1 Trunks. For details of the Notification, refer to the EON510_SPARSH VP510 V2 User Guide.

- You must assign the extension user to a **COSEC Door Group** for COSEC Integration. The users in the same group must be assigned the same group. You can create as many as 50 groups numbered from 00 to 50. Users who are assigned COSEC Door Group '00' are not a part of any group. See "COSEC Integration" for more information.

- If this is an Operator extension and you want the system to play beeps during a conference to the participants, to indicate the presence or absence of the Operator, select the **Station Type** as **Assistant**.

  If you are using the system in the *Hotel Mode*, select the **Station Type** for the SIP Extension as **Administration/Assistant** or **Guest**. The system will consider the options Administrator and Assistant as same.

- You may assign a **Personal Directory** number to the SIP Extension. Default: 00.

  A Personal Directory is a list of 25 frequently dialed numbers, each of which are stored by Index number (location code), Name and Trunk Access Codes ("Out Going Trunk Bundle Group Index"). The Personal Directory is necessary for using the features "Abbreviated Dialing" and "Dial By Name".

  When a Personal Directory is assigned to a SIP Extension, make sure you also configure this directory. The Personal Directory can be programmed by the SIP Extension users and by the System Engineer. Refer the topic "Abbreviated Dialing" for instructions on programming the Personal Directory.
  If Personal Directory is not to be assigned, enter 00 in this field.

- Select a **Priority** Level for the SIP Extension from 1 to 9. Default; 5-Normal.

  Each extension of the SARVAM UCS is assigned a Priority Level starting from 1, 2, 3... to 9, with '1' being lowest Priority and '9' being highest Priority. Whenever an extension (phone) with higher priority calls an extension with lower priority, a triple ring is placed on the called extension. To know more, read the feature description "Priority".

  If this SIP extension is assigned to Operator, you may want to set a higher priority for this extension.

- Click **Submit** to save your SIP Extension Settings.

Configuring Matrix Extended Phone Settings using Jeeves

You can register three Matrix Extended Phones/Soft Clients at three different locations as a single SIP Extension. You can connect the same or different types of Extended Phones/Soft Clients —SPARSH VP210, SPARSH VP248, SPARSH VP310, SPARSH VP510, SPARSH VP330,Extended SPARSH VP710, Matrix VARTA ADR100 Mobile UC Client, Matrix VARTA AMP100 Mobile UC Client or MATRIX VARTA WIN200 Desktop UC Client — at each location. In this case we assume that SPARSH VP210 is connected at Location 1, 2 and 3.

If you have connected SPARSH VP248 at any of the locations, refer to "Configuring Matrix Extended Phone Settings using Jeeves".

If you have connected SPARSH VP310 at any of the locations, refer to “Configuring Matrix SPARSH VP310”.

If you have connected SPARSH VP330 at any of the locations, refer to “Configuring Matrix SPARSH VP330”.

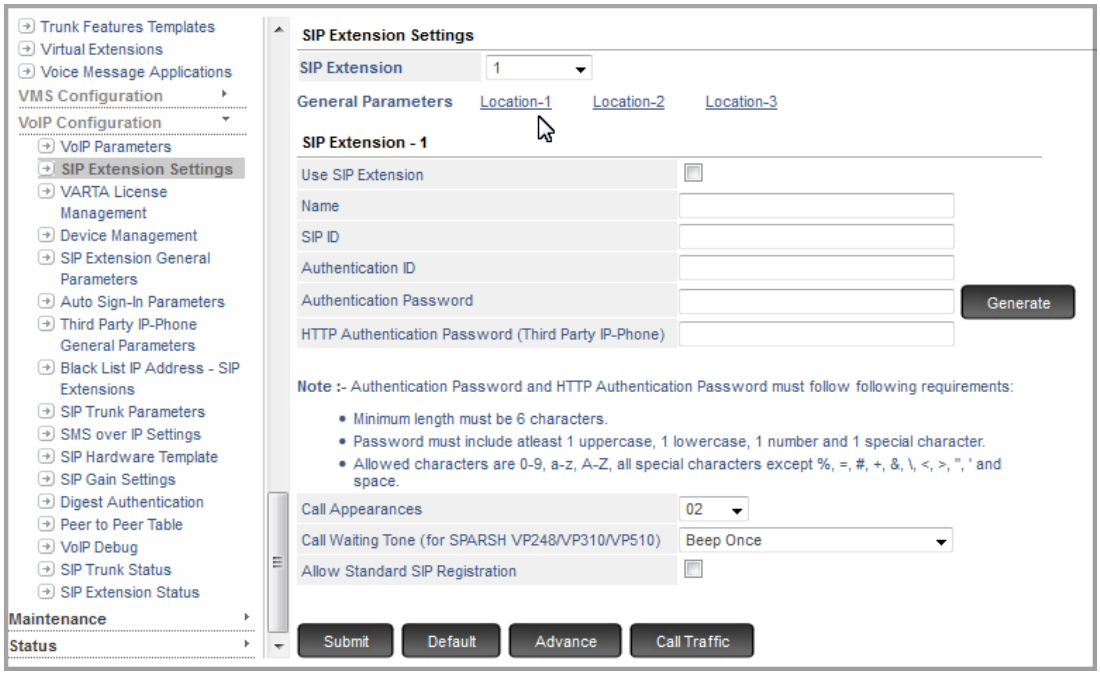
If you have connected Extended SPARSH VP710 at any of the locations, refer to “Configuring Matrix Extended SPARSH VP710”.

If you have registered Matrix VARTA Mobile UC Clients in any of the locations, refer to “Configuring Matrix VARTA ADR100/AMP100 UC Clients”.

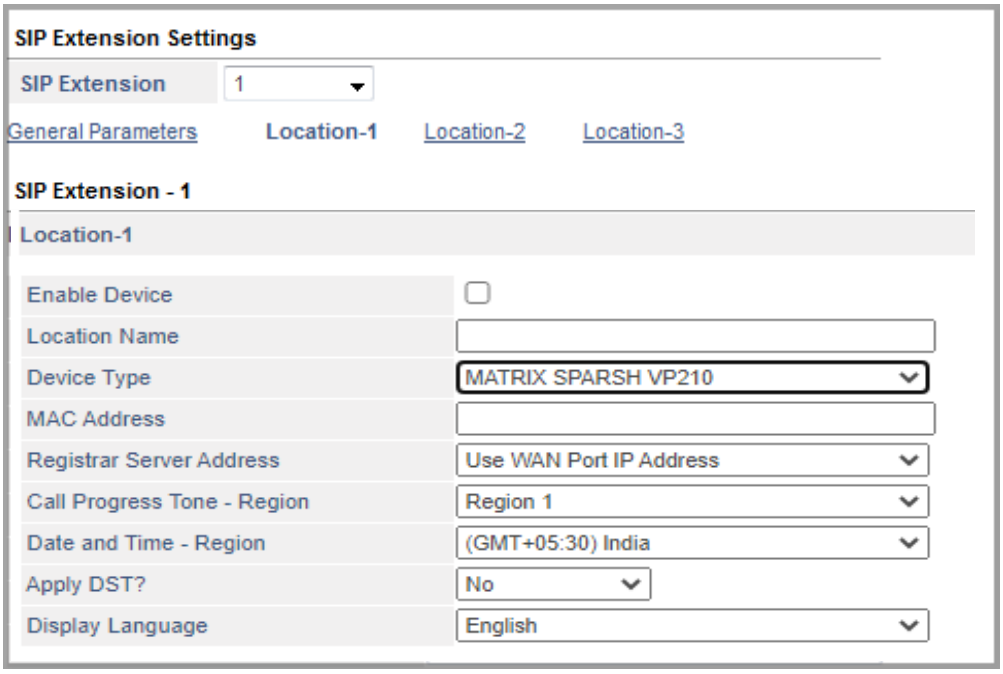
If you have registered MATRIX VARTA WIN200 Desktop UC Client in any of the locations, refer to “Configuring Matrix VARTA WIN200 UC Client”.

If you want to use more than one SPARSH VP210 Phones as a SIP Extension, configure their settings at **Location 1**, **Location 2** and **Location 3**.

- Click **Location 1**.



- The settings of the phone at **Location 1** appear.



- Select the **Enable Device** check box. Default: Disabled.

- Enter the **Location Name** for the phone to identify the phone. Location name may be the place where the phone is located (e.g.: Head office, branch, residence). The Location Name may consist of 18 characters (maximum). Default: Blank.

- Select **MATRIX SPARSH VP210** as the **Device Type** at this location.

- Enter the **MAC Address**[153] of the SPARSH VP210 connected at this location in hexadecimal format: 00:1b:09:XX:XX:XX. Default: blank.

  SARVAM UCS validates the SPARSH VP210 on the basis of the MAC Address, and provides configuration on validation.

  As SARVAM UCS allows registration of the SIP Extension from three different locations, it identifies the SIP Extension in each location by the programmed MAC address.

- Select the appropriate **Registrar Server Address** to register the SPARSH VP210 with the SIP Registrar of SARVAM UCS, according to your installation scenario:

  - If the SPARSH VP210 is connected on the WAN network, select **Use WAN Port IP Address** as Registrar Server IP Address.

  - If the SPARSH VP210 is connected on the LAN network, select **Use LAN Port IP Address** as Registrar Server IP Address.

153. *MAC address is the address of the electronic hardware devices such as a computer, which is hard-coded into the device during manufacture and cannot be modified. No two devices can have similar MAC address and thus it uniquely identifies your phone. MAC address is assigned as per the IANA standard. The MAC Address of the phone will be used as source MAC address on all Ethernet frames.*

- If the SPARSH VP210 is connected in the Global Network and SARVAM UCS is located behind a Router, or behind a NAT Router and STUN is programmed, select **Use Router/STUN's IP Address** as Registrar Server IP Address.

  Make sure you configure either the **Router's Public IP Address** or **Simple Traversal of UDPs through NATs (STUN)** in Network Parameters. For details, see "Configuring Network Parameters".

- If Dynamic DNS is configured in the Network Parameters, select **Use Dynamic DNS Host Name** as Registrar Server IP Address.

  By default, Use WAN Port IP Address is selected as the Registrar Server IP Address.

- To set the call progress tone generation standards of the country where the SPARSH VP210 is installed, select the **Call Progress Tone - Region**. Default: Region 1.

  See "Call Progress Tones" to know more.

- To display the Date and Time of the country where the SPARSH VP210 is installed, select the **Date and Time - Region**. Default: India.

- If you want to enable Daylight Saving Time (DST) on the phone, set **Apply DST?** to Yes. Default: No. The Daylight Saving Time convention followed in the country/region you selected will be automatically applied. The IP phone will change its date and time settings according to the DST convention of the selected country/region.

- Select the **Display Language** for the SPARSH VP210. Default: English.

  SARVAM UCS provides language support for English, French, German, Spanish, Portuguese, and Italian on the SPARSH VP210. When you select any of these languages, all the prompts and command strings will appear in the selected language.

SIP Extension users can change the language by accessing and navigating through the phone menu.

The SA can change the Language by logging into the SA Jeeves.

DSS Key Settings

- You can select the desired key template — Operator, Executive1, Executive2, Executive3, Hotel Attendant, Guest or any other template you added. See "Customizing Extended IP Phone Templates using Jeeves" for more details.

  OR

- You can personalize the key map of the SPARSH VP210 for this location. To do so,

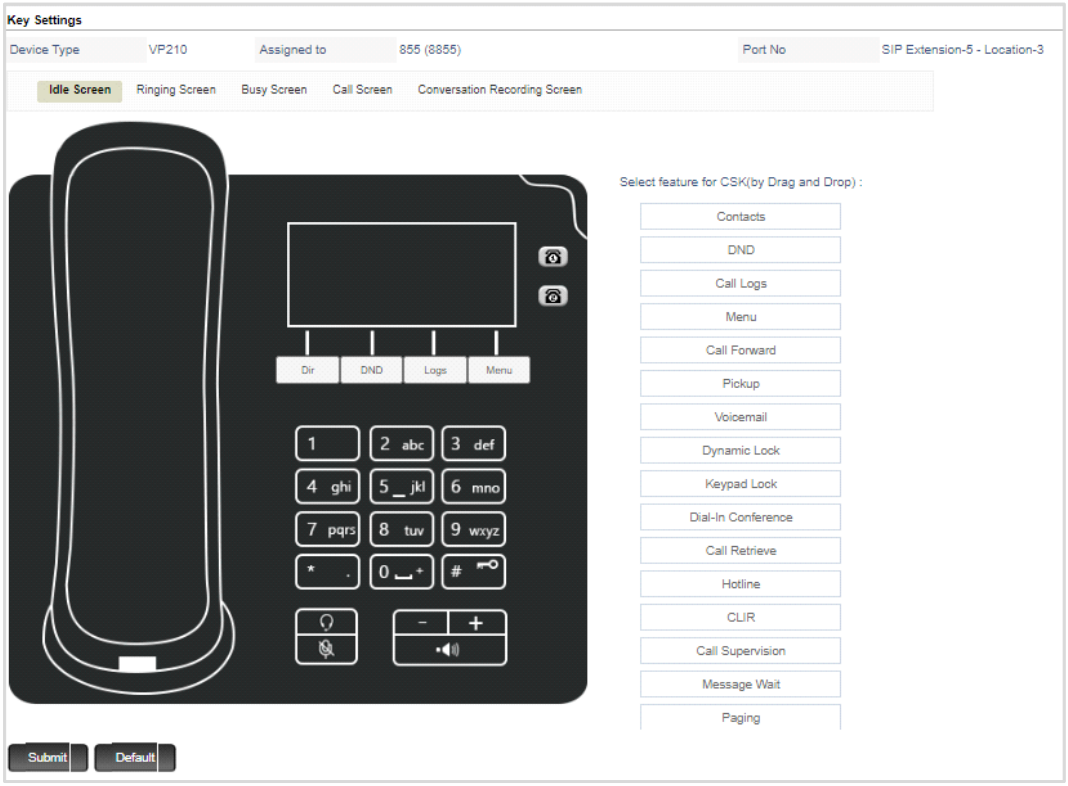
  - Select **Personalized** as the **Key Template** option.

  - Click **Submit**.

- Click **Key Settings**.

- The key map of the Extended Phone opens in a new window on your screen.

- Click **Idle Screen**.
- Each Context key, 1 to 4 can be assigned features.
- The feature assignment cum priority list appears on the right. You can change the feature assignments/ priorities as per your preference.
- To set the priority, drag and drop the features in the order of your preference. This will have two implications — the Context Key will be assigned the desired feature as well it will set the priority.
- For example, if you wish to assign DND feature to Context Key 2, then drag and drop the DND feature at position 2 as below. Also make sure that the priority of Menu Key is kept as either of the four Context Keys.



- Click **Submit**.
- The key map will refresh and DND appears as Context Key 2.

Menu must be assigned to one of the first four Context Keys.

- Similarly, you can click **Ringing Screen, Busy Screen**, **Call Screen** or **Conversation Recording Screen** and can set the feature priorities as per your preference.

  To assign features/set feature priorities for other Context Keys, follow the same instructions. You can reinstate default key assignment any time, by clicking the **Default** button at the bottom of the window.

- When you complete assigning functions/priorities to the keys, close the window.

Transport Mode and SRTP

- Select the protocol to be used to transport the SIP messages. You can select the **Transport Mode** as **TCP** or **TLS**.

If you select TCP, make sure the SIP Over TCP check box is selected in VoIP Parameters.
If you select TLS, make sure the SIP Over TLS check box is selected in VoIP Parameters.

- For secure conversations over SIP, select the **Enable SRTP?** check box. The SIP messages will be transported over SRTP only.

RTP Port

- Define **RTP Port**:

    - **RTP Listening Port:** This is the port on which the phone listens for RTP messages over UDP. This port is also used as the source port for sending RTP packets. This port is also used as the source port for sending RTP packets to the remote peer. The valid range for this port is 1025-65278. Default: 8000.

Quality of Service

- Set the **SIP Quality of Service (QoS)** for SIP signaling as:

    - **SIP DiffServe/ToS**. Valid range is 00 to 63. Default: 26.
      OR
    - **RTP DiffServe/ToS**. Valid range is 00 to 63. Default: 46.

NAT Keep Alive

- If the SPARSH VP210 is connected behind a NAT router, configure **NAT Keep Alive**.

    - Select the check box **Enable NAT Keep Alive** to send Keep Alive messages periodically to refresh the binding in the NAT router. Default: Disabled.

    - Define as **Interval (sec)**, the time period, from 001 to 999 seconds, after which the phone should send Keep Alive message. Default: 120 seconds.

      The time period you define should be less than the binding timer of the router.

Timers

- Set the following **Timers** to the desired value, where required:

    - **SIP INVITE Timer (sec):** This is the time in seconds that the phone waits for a response from the called party after ending INVITE message. This timer starts after sending INVITE message to the called party and stops on receipt of the provisional response or the final response or when the user disconnects the call. On expiry of the timer, the phone terminates the call process and gives an error tone to the user. The range of the SIP INVITE TIMER is 10-180 seconds. Default: 30 seconds.

    - **SIP Provisional Timer (sec)**: This is the time in seconds that the phone waits for final response after receiving the provisional response from the called party. This timer starts on the receipt of the provisional response from the called party and stops on receipt of the final response from the called party or when the user disconnects the call. On expiry of the timer, the IP phone terminates the call process and gives error tone to the user. The range of SIP Provisional Timer is 10-180 seconds. Default: 60 seconds.

    - **General Request Timer (sec):** This is the time in seconds for which the phone waits for response of a transaction request. This timer starts on initiating a transaction. This timer stops on receipt of a response for the request. On expiry of the timer, the phone clears the transaction. This timer is used for Registration request, etc. The range of the General Request Timer is 10-60 seconds. Default: 20 seconds.

Debug

- To debug using Syslog Client supported by the SPARSH VP210, configure Debug parameters:
  - Select the **Enable Debug?** check box. Default: disabled.

    When the Debug flag is enabled, the phone will send the debug messages to the Syslog Server IP address. Debug report can be viewed on the Syslog Server or any other application which can capture the Syslog messages/debug sent by the phone.

  - Enter the IP Address and port of the remote Syslog Server and as **Syslog Server Address and Server Port**.

    The address of the Listening Port of the Syslog Server is from 1025-65535;514. Default: 514. Syslog uses the UDP as transport protocol and listens on the port 514 (the default listening port).

  - You may select the **Debug Level** from the following options, by selecting the respective check box:
    - Debug Level 1
    - Debug Level 2
    - Debug Level 3

    You may select any or all of these debug levels. The Syslog Client will send only the debug messages for the selected level to the remote server on the IP network.

- Click **Submit** to save settings.

- If you have completed the configuration of the SPARSH VP210 Phone Settings at Location 1, follow the same steps as described above to configure the SPARSH VP210 Phone at Location 2 and Location 3.

  However, if you want to replicate the configuration of SPARSH VP210 Settings same as Location 1 to Location 2 and Location 3, you can use the **Copy** button present at the bottom of the page. To know more, refer to “Copy Parameter Values”.

If you wish to Copy only the configurations from a location to all other locations where the same IP Phone is connected, make sure you clear the Device Type check box.

If you wish to copy the Device Type as well as all the configurations from a location to all the SIP Extensions, make sure all the check boxes are selected.

When you change any of the parameters listed below in the SIP Extension at Location 1, 2, 3, the phone will go in Auto Configuration mode automatically, if registered:

- Use SIP Extension
- SIP ID
- Name
- Authentication ID
- Authentication Password
- Registrar Server IP Address
- MAC Address
- Enable Device
- Device Type
- Key Map in the Key Template assigned to phone
- Language
- Call Progress Tone
- Date and Time

- Apply DST?
- Transport Mode and SRTP
- QoS
- RTP Ports
- NAT Keep Alive
- SIP Timers
- Class of Service
- Trunk Access Code
- Emergency Numbers

The SIP Extension registered at Location 1, 2, 3, will also restart, if:

- The SE Password of SARVAM UCS is changed
- Specific parameters in Network Port parameters are changed
- Specific parameters in VoIP Parameters are changed
- You restart the System
- Set the System to Default

Configuring Matrix Extended SPARSH VP710

Extended SPARSH VP710[154], the Smart Video IP Phone is engineered to offer a contemporary design with crystal-clear audio and feature-rich capabilities at economical price.This IP Phone is an integration of SPARSH VP710, android based deskphone with VARTA ADR100 application. To know the list of features supported, refer to "SARVAM UCS Features Supported in Terminals".

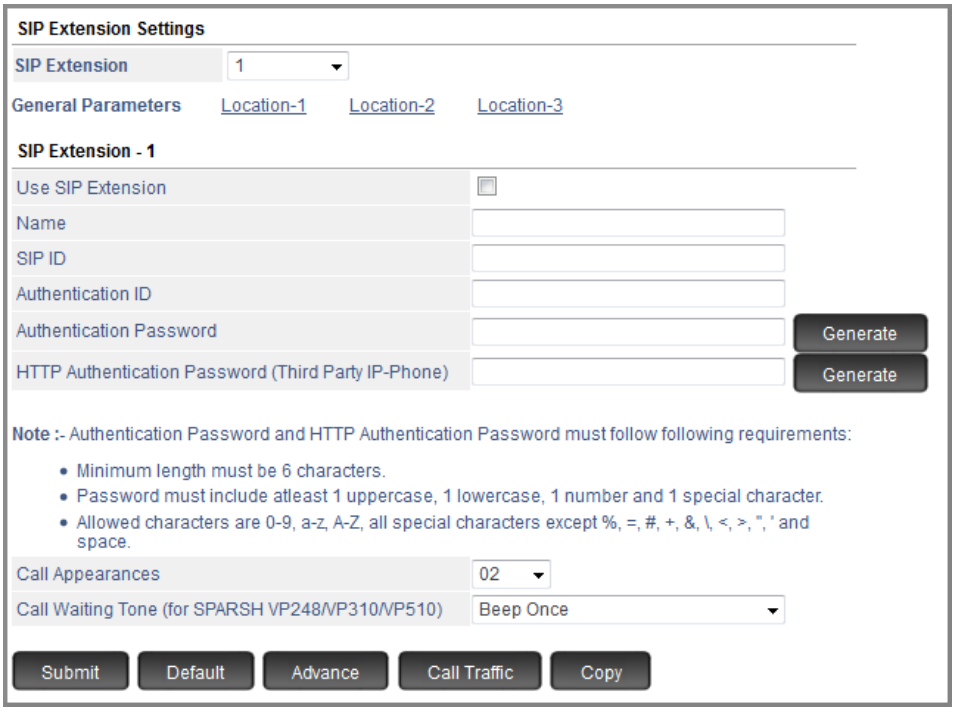
For instructions on how to use Extended SPARSH VP710, refer to the *EXTENDED SPARSH VP710 User Guide.*

To be able to use Extended SPARSH VP710[155], you must configure the following:

- SIP Extension General Parameters, see "Configuring SIP Extension General Parameters".
- SIP Extension Settings, see "Configuring SIP Extension using Jeeves".
- Extended Phone Settings, see "Configuring Matrix Extended Phone Settings using Jeeves".
- Voice Mail Settings, if you want to provide mailbox facility to the extension. See "Extension Voice Mail Settings".

Configuring SIP Extension Settings using Jeeves

- Log into Jeeves as System Engineer.
- Under **Configuration**, click **VoIP Configuration**.
- Click **SIP Extension Settings**.



The page of SIP Extension 001 opens.

- You may select the **SIP Extension** number you want to configure.

154. Check for availability.
155. SARVAM UCS supports only IPv4 Addresses for registering Extended SPARSH VP710.

The parameters of the SIP Extension number you select will appear on this page.

For SARVAM UCS upto 999 SIP Extensions can be registered with the system.

- Select the **Use SIP Extension** check box to enable the SIP extension. By default, it is disabled.

- In the **Name** field, enter a name for the SIP Extension, which may be the name of the person who will use the SIP Extension or the name of a Department. The name you enter here will be displayed as the Caller ID of the SIP Extension on the remote user's phone, when the SIP Extension user makes calls.

  The name may consist of a maximum of 18 alphanumeric characters.

- Enter the **SIP ID** for the extension. The SIP ID is necessary for registering the SIP Extension with the Registrar of the system. It is the number with which you can call the SIP Extension. Any extension user of the SARVAM UCS can call a SIP Extension by dialing the SIP ID assigned to the SIP extension. SIP ID of each SIP Extension must be a unique number string of a maximum of 6 digits. Any combination of digits from 0 to 9 and the characters * and # are allowed.

  You cannot assign the same SIP ID to more than one extension.

  To assign SIP IDs according to your preference and requirement to a range of SIP Extensions, see “Assigning Access Codes to a Range of Extensions”.

  By default, the SIP IDs are Blank.

The SIP ID will be set to default value (blank), when you restore the default settings of the system.

- In **Authentication ID**, enter the number which you want the system to use for user authentication of the SIP messages received from the SIP Extension. You cannot keep this field blank and it must be unique for each SIP extension. The number may be a string of maximum 6 alphanumeric characters. All ASCII characters except < > and “ (double quote) are allowed. Default: Blank.

Make sure the User ID configured in “Digest Authentication” does not conflict with the Authentication ID configured above.

- In **Authentication Password**, enter the password manually or click **Generate** to automatically generate a unique password. This password will be used by the system to authenticate the SIP messages received from the SIP Extension. To avoid unauthorized access, we recommend you to change the password regularly. Make sure it is strong and is kept confidential. When you enter the password manually, the password must:

  - be of minimum 6 characters and can be a maximum of 12 characters.
  - include atleast one upper-case, one lower-case, one number and one special character.
  - all ASCII characters (except Percentage **%**, Hash **#,** Equal to **=,** Plus **+**, And **&**, Backslash **\,** Less than **<,** Greater than **>,** Apostrophe **’,** Double Quote **"** and **Space**) and digits 0 to 9 are allowed. Default: Blank.

    To provide additional security, when the Authentication fails 10 times consecutively due to wrong Authentication ID / Authentication Password, the system will blacklist the IP Address and Port for registration of this SIP Extension. However, you can remove the IP Address from the Blacklist IP Address list. See “Black List IP Address - SIP Extensions” for more details. This activity will be logged in the “System Activity Log” as well as “Simple Network Management Protocol (SNMP)”.

Make sure you note down or copy the Authentication Password in a confidential file.

- In **Call Appearances**, define the maximum number[156] of simultaneous incoming calls that the SIP Extension user should be allowed to receive. You can set up to 10 call appearances for a SIP Extension. Default: 2.

  When Call Appearance is set to 2, the SIP Extension can receive 2 calls at a time.

- Under **Auto Sign-In,** enter the **Email ID** of the extension user you wish to store. The Email ID can be a maximum of 64 characters. The Email ID is used for various server features.

Auto Sign-In parameters — Send Configuration Mail and Mail Status, *are applicable only for Mobile Clients — VARTA ADR100, VARTA AMP100 applications.*

*The **Send Configuration Mail** button will appear only after you have enabled the SIP Extension and configured the SIP ID, Authentication ID and Password.*

- Under **Authentication**, enable Authentication of any or all of the following SIP Message Options by selecting the respective check boxes:
  - **INVITE Request**
  - **SUBSCRIBE Request**

  By default, the SIP Message Options INVITE and SUBSCRIBE are enabled.

Make sure that the Authentication ID for the SIP Extension has been programmed.

- For secure conversations over SIP, enable **SRTP Mode**. The SARVAM UCS supports the following options:

  - **Disable:** SARVAM UCS uses normal RTP for transporting the speech packets.

  - **Optional:** SARVAM UCS uses SRTP for transporting the speech packets. If the remote user does not support SRTP, SARVAM UCS will use normal RTP for transporting the speech packets.

    - If you select this option, you must configure the **SRTP Media Type**. You may select **AVP** or **SAVP**. By default, AVP is selected as the SRTP Media Type.

  - **Forced:** SARVAM UCS uses only SRTP (SAVP) for transporting the speech packets. If the remote user does not support SRTP, SARVAM UCS will reject incoming calls from and drop outgoing calls made to such users.

    By default, SRTP Mode is Disabled.

- Assign a **SIP Hardware Template** to the SIP Extension. Default: Template 01. The “SIP Hardware Template” contains voice quality related features such as Voice Codec selection, Tx and Rx Gains, Echo Cancellation, Jitter Buffer and Fax-over-IP options and related parameters.

  There are 32 different templates to choose from. Each template can also be altered to suit your requirement and preferences. By default, Template number 01 assigned to all SIP Extensions as well as to

156. *For the calls that are routed through the CPU, the number of Vocoder channels that will be supported would be as per the license you purchase.*

SIP Trunks. Check if the values in this template fulfill requirements of the SIP Extension. If Template 01 fulfills the feature requirements, retain Template 01.

If a different set of SIP hardware features are to be allowed to this SIP Extensions, prepare another template and assign it to this extension. To do this,

- Under **VoIP Configuration**, click the **SIP Hardware Template** link.
- Select a Template number, for example 02.
- Customize Template number 02 and click **Submit**.
- Return to the **SIP Extension Settings** page.
- Select the number of the Template you customized, Template 02 in the **SIP Hardware Template** field.

Also see the topic "SIP Hardware Template" to know more about customizing the templates and applying on the SIP Extensions.

- Assign a **Station Basic Feature Template** to the SIP Extension. Default: Template 01. The "Station Basic Feature Template" has a set of features like Time Table, Class of Service, Toll Control, Operator, Storage of Incoming and Outgoing Calls, Outgoing Trunk Bundle groups. There are 50 different templates to choose from. Each template can also be altered to suit your requirement and preferences.

  If the default Station Basic Feature Template 01 fulfills the feature requirements of the SIP Extension ("Class of Service (COS)", "Toll Control", "OG Trunk Bundle Group", etc.) retain this template, you may also customize this template. If you want to assign a different set of features to this SIP Extension, prepare a different Station Basic Feature Template and apply it to this extension. To do this,

  - Under **Configuration**, click the **Station Basic Feature Template** link.
  - Select a Template number, for example 05.
  - Customize Template number 05 and click **Submit**.
  - Return to the **SIP Extension Settings** page.
  - Select the number of the Template you customized, Template 05, in the **Station Basic Feature Template** field.
  - Click **Submit** to save changes.

  Also, see the topic "Station Basic Feature Template" to know more about customizing the templates and applying on extensions.

- Assign a **Station Advanced Feature Template** to the SIP Extension. Default: Template 01. The "Station Advanced Feature Template" has a set of advanced features for extensions such as Alarm Notification settings, Routing of Incoming Auto Attendant Calls, Call Duration Control, Floor Service, etc. There are 50 different templates to choose from. Each template can also be altered to suit your requirement and preferences.

  Check if the default template fulfills the feature requirements of the SIP Extension by clicking the **Station Advanced Feature Template** link.

You may retain this template and customize it further, or customize another template if a different set of features are to be allowed to this SIP Extension. To customize/prepare another template,

- Under **Configuration**, click the **Station Advanced Feature Template** link.
- Select the Template number, for example 02, and customize this template.
- Click **Submit** and return to the 'SIP Extension Settings' page.
- In the **Station Advanced Feature Template** field, select the number of the template you customized.
- Click **Submit** to save changes.

Also see the topic "Station Advanced Feature Template" for instructions on customizing these templates and applying them on the extensions.

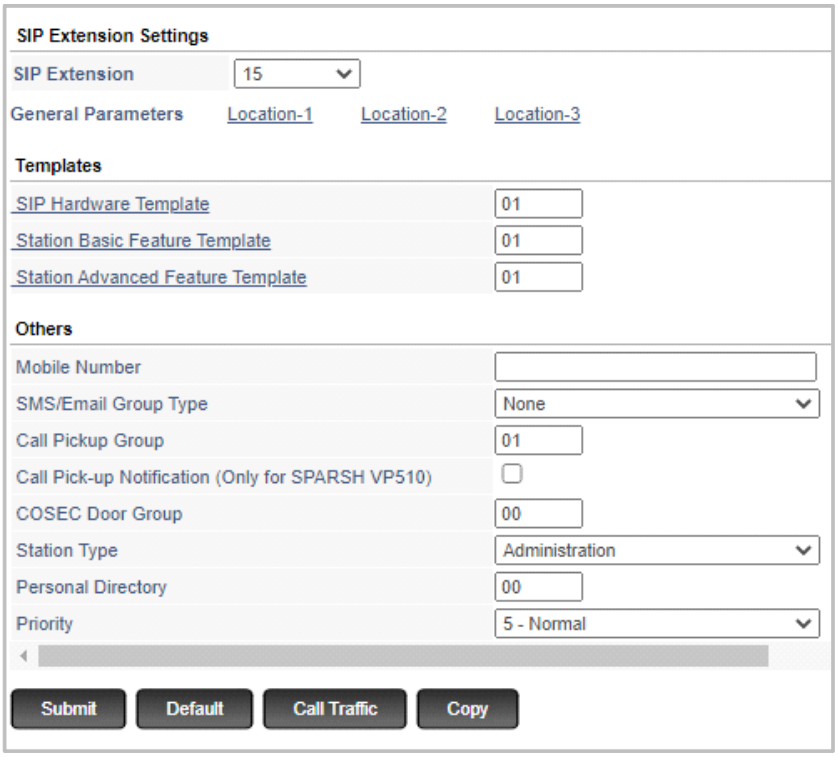
- In **Voice Mail Settings**, click the Voice Mail Settings link. The respective Extension Voice Mail Settings window will open. You may edit the parameters. For details, see "Extension Voice Mail Settings".

The Voice Mail Settings link will be visible only if you have configured the respective SIP ID.

Click **Close** to close the window.

Advanced Configuration Parameters

- If you want to provide other features like Personal Directory, Priority, or assign a Station Type to the SIP Extension, click the **Advanced** button at the bottom of the page.



- Enter the **Mobile Number** of the extension user you wish to store. The Number can be a maximum of 16 digits.

- You can assign the extension user to a Group. Select the desired **SMS/Email Group Type** from the list.The system clubs together extension users assigned the same Group. Default: None. For details, see "SMS/Email Group".

- Assign the SIP Extension to a **Call Pick-up Group**, if required. Default: 01

  Call Pick Up allows the SIP Extension user to 'pick up' (answer) calls ringing on any other extension, by dialing a feature code, without physically going to the ringing extension. It also allows incoming calls for the SIP Extension to be answered by the other extensions assigned the same Call Pick-Up group.

  For this to work, both the ringing extension and the extension picking up the call must be in the same 'Call Pick Up Group'. Refer "Call Pick Up" for instructions on how to create groups. You can create as many as 99 groups numbered from 01 to 99.

  Enter the number of the Call Pick-Up Group you created for this SIP Extension in this field.

- Select the **Call Pick-up Notification (Only for SPARSH VP510)** check box, if you want the Call Pick-up Notification to be displayed on the phone LCD. The Call Pick-up Notification will be displayed for internal as well as external calls. The notification will be displayed on SPARSH VP510 only if the ringing extension is in the same Call Pick-up Group. The notification will be displayed only when SPARSH VP510 is in idle or dial state. The notification will display the name/number of the Caller along with the name/number of the Called. For example, if the name and number of the Caller is ABC, 2001 and the name and number of Called is XYZ, 1001, the LCD notification will be displayed as ABC -> XYZ. Along with the LCD display the phone back-light will also be lit.

  For this to work, make sure you have enabled Call Pick-up in COS as well as assigned a Call Pick-up Group to the extension. Refer to "Call Pick Up" and "Class of Service (COS)". Call Pick-up Notifications will be displayed for DKP, SLT as well as SIP Extensions and for calls landing through CO, SIP as well as T1E1 Trunks. For details of the Notification, refer to the EON510_SPARSH VP510 V2 User Guide.

- You must assign the extension user to a **COSEC Door Group** for COSEC Integration. The users in the same group must be assigned the same group. You can create as many as 50 groups numbered from 00 to 50. Users who are assigned COSEC Door Group '00' are not a part of any group. See "COSEC Integration" for more information.

- If this is an Operator extension and you want the system to play beeps during a conference to the participants, to indicate the presence or absence of the Operator, select the **Station Type** as **Assistant**.

  If you are using the system in the *Hotel Mode*, select the **Station Type** for the SIP Extension as **Administration/Assistant** or **Guest**. The system will consider the options Administrator and Assistant as same.

- You may assign a **Personal Directory** number to the SIP Extension. Default: 00.

  A Personal Directory is a list of 25 frequently dialed numbers, each of which are stored by Index number (location code), Name and Trunk Access Codes ("Out Going Trunk Bundle Group Index"). The Personal Directory is necessary for using the features "Abbreviated Dialing" and "Dial By Name".

  When a Personal Directory is assigned to a SIP Extension, make sure you also configure this directory. The Personal Directory can be programmed by the SIP Extension users and by the System Engineer. Refer the topic "Abbreviated Dialing" for instructions on programming the Personal Directory.
  If Personal Directory is not to be assigned, enter 00 in this field.

- Select a **Priority** Level for the SIP Extension from 1 to 9. Default; 5-Normal.

  Each extension of the SARVAM UCS is assigned a Priority Level starting from 1, 2, 3... to 9, with '1' being lowest Priority and '9' being highest Priority. Whenever an extension (phone) with higher priority calls an extension with lower priority, a triple ring is placed on the called extension. To know more, read the feature description “Priority”.

  If this SIP extension is assigned to Operator, you may want to set a higher priority for this extension.

- Click **Submit** to save your SIP Extension Settings.

Configuring Matrix Extended Phone Settings using Jeeves

You can register three Matrix Extended IP Phones/Soft Clients at three different locations as a single SIP Extension. You can connect/register the same or different types of Extended Phones/Soft Clients —SPARSH VP248, SPARSH VP310, SPARSH VP330, SPARSH VP510, SPARSH VP210, Extended SPARSH VP710, Matrix VARTA ADR100 Mobile UC Client, VARTA AMP100 Mobile UC Client or MATRIX VARTA WIN200 Desktop UC Client— at each location. In this case we assume that Extended SPARSH VP710 is connected at Location 1, 2 and 3.
If you want to use more than one Extended SPARSH VP710 IP Phone as a SIP Extension, configure their settings at **Location 1**, **Location 2** and **Location 3**.

If you have connected SPARSH VP248 at any of the locations, refer to “Configuring Matrix SPARSH VP248”.

If you have connected SPARSH VP310 at any of the locations, refer to “Configuring Matrix SPARSH VP310”.

If you have connected SPARSH VP330 at any of the locations, refer to “Configuring Matrix SPARSH VP330”.

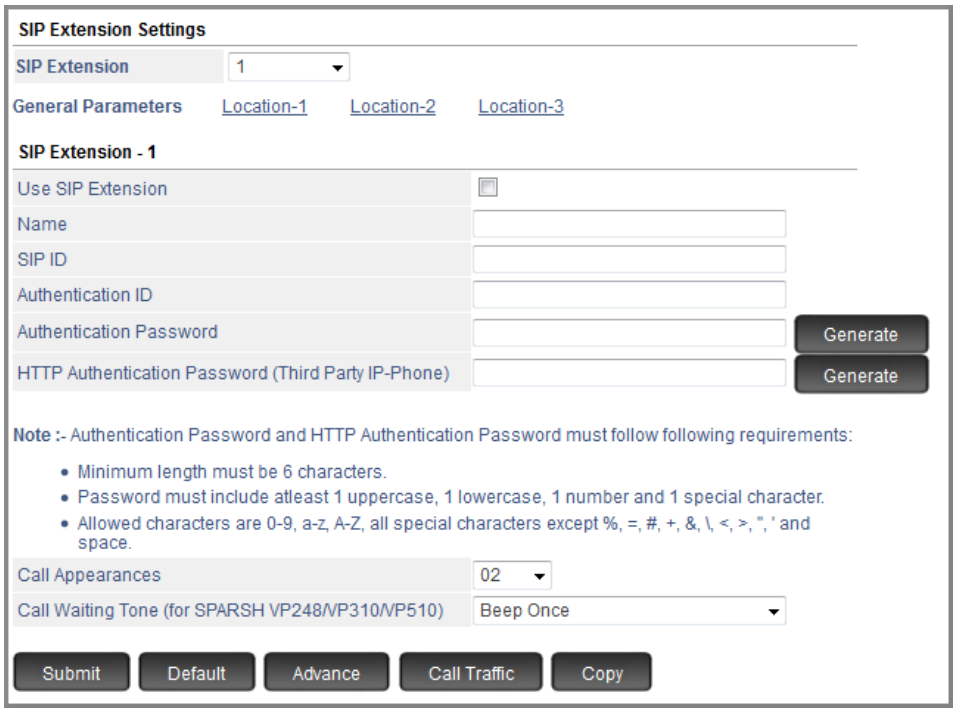
If you have connected SPARSH VP510 at any of the locations, refer to “Configuring Matrix SPARSH VP510”.

If you have connected Extended SPARSH VP210 at any of the locations, refer to “Configuring Matrix SPARSH VP210”.

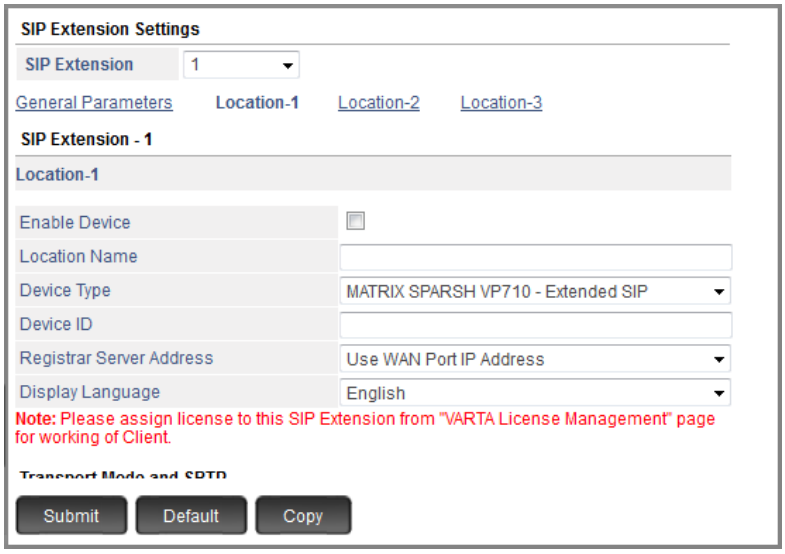
If you have registered Matrix VARTA ADR100 and VARTA AMP100 Mobile UC Clients in any of the locations, refer to “Configuring Matrix VARTA ADR100/AMP100 UC Clients”.

If you have registered MATRIX VARTA WIN200 Desktop UC Client in any of the locations, refer to “Configuring Matrix VARTA WIN200 UC Client”.

- Click **Location 1**.



- The settings of the phone at **Location 1** appear.



- Select the **Enable Device** check box. Default: Disabled.

- Enter the **Location Name** for the phone to identify the phone. Location name may be the place where the phone is located (e.g.: Head office, branch, residence). The Location Name may consist of 18 characters (maximum). Default: Blank.

- Select **MATRIX SPARSH VP710 - Extended SIP** as the **Device Type** at this location.

- Enter the **MAC Address**[157] of the Extended SPARSH VP710 connected at this location in hexadecimal format: 00:1b:09:XX:XX:XX. Default: Blank.

SARVAM UCS validates the Extended Phone on the basis of the MAC Address, and provides configuration on validation.

As SARVAM UCS allows registration of the SIP Extension from three different locations, it identifies the SIP Extension in each location by the programmed MAC address.

- Select the appropriate **Registrar Server Address** to register the Extended SPARSH VP710 with the SIP Registrar of SARVAM UCS, according to your installation scenario:
  - If the Extended SPARSH VP710 is connected on the WAN network, select **Use WAN Port IP Address** as Registrar Server IP Address.
  - If the Extended SPARSH VP710 is connected on the LAN network, select **Use LAN Port IP Address** as Registrar Server IP Address.
  - If the Extended SPARSH VP710 is connected in the Global Network and SARVAM UCS is located behind a Router, or behind a NAT Router and STUN is programmed, select **Use Router/STUN's IP Address** as Registrar Server IP Address.

    Make sure you configure either the **Router's Public IP Address** or **Simple Traversal of UDPs through NATs (STUN)** in Network Parameters. For details, see "Configuring Network Parameters".
  - If Dynamic DNS is configured in the Network Parameters, select **Use Dynamic DNS Host Name** as Registrar Server IP Address.

    By default, Use WAN Port IP Address is selected as the Registrar Server IP Address.
- Select the **Language** for the Extended SPARSH VP710. Default: English.

  SARVAM UCS provides language support for English, French, German, Spanish, Portuguese, and Italian on the Extended SPARSH VP710. When you select any of these languages, all the prompts and command strings will appear in the selected language.

SIP Extension users can change the language by accessing and navigating through the phone menu.

The SA can change the Language by logging into the SA Jeeves.

Transport Mode and SRTP

- Select the protocol to be used to transport the SIP messages. You can select the **Transport Mode** as **TCP** or **TLS**.

If you select TCP, make sure the SIP Over TCP check box is selected in VoIP Parameters.

If you select TLS, make sure the SIP Over TLS check box is selected in VoIP Parameters.

- For secure conversations over SIP, select the **Enable SRTP?** check box. The SIP messages will be transported over SRTP only.

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157. *MAC address is the address of the electronic hardware devices such as a computer, which is hard-coded into the device during manufacture and cannot be modified. No two devices can have similar MAC address and thus it uniquely identifies your phone. MAC address is assigned as per the IANA standard. The MAC Address of the phone will be used as source MAC address on all Ethernet frames.*

If you select this check box, make sure you have selected SRTP Mode as Forced or Optional in the General Parameters under SIP Extension Settings.

SMS Over IP

- If you want Extended SPARSH VP710 users to send SMS to any extension user as well as receive IM from any extension user, select the Enable SMS Over IP check box. For detailed information, see "SMS over IP"

RTP Port

- Define the RTP Port:
  RTP Listening Port: This is the port on which the Extended SPARSH VP710 listens for RTP messages over UDP. This port is also used as the source port for sending RTP packets. This port is also used as the source port for sending RTP packets to the remote peer. The valid range for this port is 1025-65278. Default: 8000.

Quality of Service

- Set the **SIP Quality of Service (QoS)** for SIP signaling as:

  - **SIP DiffServe/ToS.** Valid range is 00 to 63. Default: 26.
    OR
  - **Voice DiffServe/ToS**. Valid range is 00 to 63. Default: 46.
    OR
    Video DiffServe/ToS. Valid range is 00 to 63. Default: 46.

NAT Keep Alive

- If Extended SPARSH VP710 is connected behind a NAT router, configure **NAT Keep Alive**.

  - Select the check box **Enable NAT Keep Alive** to send Keep Alive messages periodically to refresh the binding in the NAT router. Default: Disabled.

  - Define as **Interval (sec)**, the time period, from 001 to 999 seconds, after which the phone should send Keep Alive message. Default: 120 seconds.

    The time period you define should be less than the binding timer of the router.

Timers

- Set the following **Timers** to the desired value, where required:

  - **SIP INVITE Timer (sec):** This is the time in seconds that the phone waits for a response from the called party after ending INVITE message. This timer starts after sending INVITE message to the called party and stops on receipt of the provisional response or the final response or when the user disconnects the call. On expiry of the timer, the phone terminates the call process and gives an error tone to the user. The range of the SIP INVITE TIMER is 10-180 seconds. Default: 30 seconds.

  - **SIP Provisional Timer (sec)**: This is the time in seconds that the phone waits for final response after receiving the provisional response from the called party. This timer starts on the receipt of the provisional response from the called party and stops on receipt of the final response from the called party or when the user disconnects the call. On expiry of the timer, the IP phone terminates the call

process and gives error tone to the user. The range of SIP Provisional Timer is 10-180 seconds. Default: 60 seconds.

- **General Request Timer (sec):** This is the time in seconds for which the phone waits for response of a transaction request. This timer starts on initiating a transaction. This timer stops on receipt of a response for the request. On expiry of the timer, the phone clears the transaction. This timer is used for Registration request, etc. The range of the General Request Timer is 10-60 seconds. Default: 20 seconds.

- Click **Submit** to save settings.

- If you have completed the configuration of the Extended SPARSH VP710 Settings at Location 1, follow the same steps as described above to configure the Extended SPARSH VP710 at Location 2 and Location 3.

  However, if you want to replicate the configuration of Extended SPARSH VP710 Settings same as Location 1 to Location 2 and Location 3, you can use the **Copy** button present at the bottom of the page. To know more, refer to “Copy Parameter Values”.

If you wish to Copy only the configurations from a location to all other locations where the same IP Phone is connected, make sure you clear the Device Type check box.

If you wish to copy the Device Type as well as all the configurations from a location to all the SIP Extensions, make sure all the check boxes are selected.

When you change any of the parameters listed below in the SIP Extension at Location 1, 2, 3, the phone will go in Auto Configuration mode automatically, if registered:

- Use SIP Extension
- SIP ID
- Authentication ID
- Authentication Password
- Registrar Server IP Address
- MAC Address
- Enable Device
- Device Type
- Transport Mode and SRTP
- QoS
- RTP Ports
- NAT Keep Alive
- SIP Timers
- The SE Password of SARVAM UCS is changed
- Specific parameters in VoIP Parameters are changed
- Specific parameters in Network Port parameters are changed
- You restart the System
- Set the System to Default

Configuring Matrix VARTA WIN200 UC Client

MATRIX VARTA WIN200, is a SIP (Session Initiation Protocol) based Unified Communication Desktop Client running on Windows OS, delivering full-array of the System features to the user on-the-go along with an added advantage of video calling. Through tight integration with the enterprise features of the System, UC Client provides advance call capabilities including Conferencing, Corporate Directory Access (Global Directory), Call Logs and Conversation Recording with one-touch access. Other than these you can take the advantage of using premium features like Presence subscription and notification, Corporate Voicemail access to enhance your overall mobile experience.

To use MATRIX VARTA WIN200 Desktop UC Client, make sure you have:

- Purchased and activated the VARTA Essential, VARTA Professional or VARTA Collaboration license. For more details, see “License Management”.
- Assigned the desired license to the SIP Extension. For more details, see “VARTA License Management”.

To know the list of featured supported, refer to “SARVAM UCS Features Supported in Terminals”.

For detailed product information and operation instructions, refer to the *MATRIX VARTA WIN200 User Guide.*

Configuring MATRIX VARTA WIN200 using Jeeves

To be able to register and use the Desktop Client[158], you must configure the following:

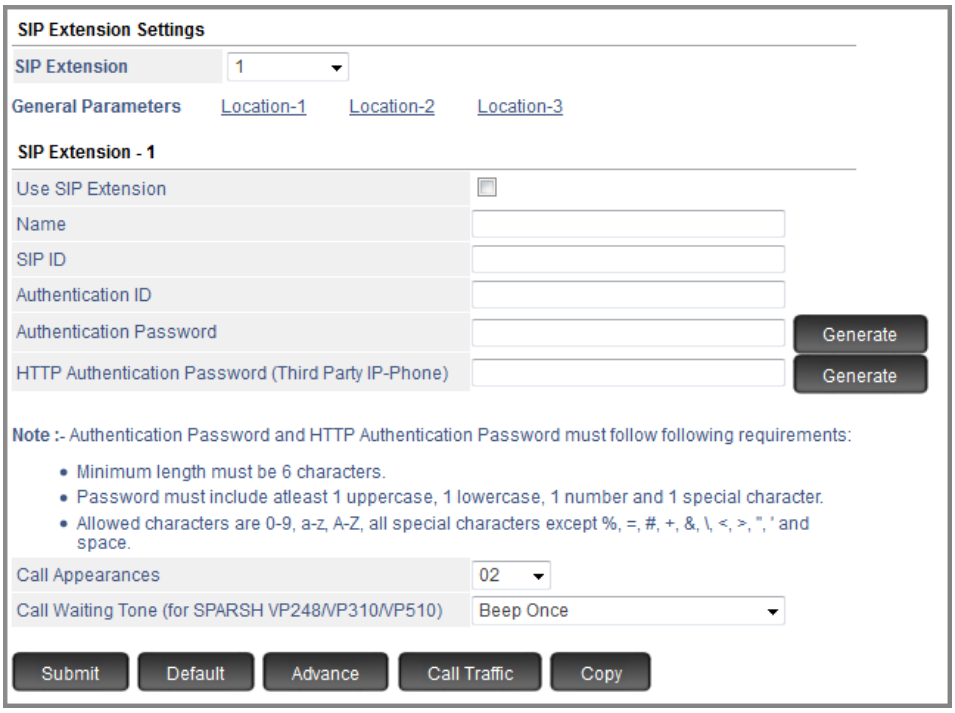
- SIP Extension General Parameters, see “Configuring SIP Extension General Parameters”.
- SIP Extension Settings, see “Configuring SIP Extension using Jeeves”.
- Extended Phone Settings, see “Configuring Matrix Extended Phone Settings using Jeeves”.
- Voice Mail Settings, if you want to provide Voice Mail to the extension. See “Extension Voice Mail Settings”.

Configuring SIP Extension Settings using Jeeves

- Log into Jeeves as System Engineer.
- Under **Configuration**, click **VoIP Configuration**.

158. SARVAM UCS supports only IPv4 Addresses for registering VARTA WIN200.

- Click **SIP Extension Settings**.



The page of SIP Extension 001 opens.

- You may select the **SIP Extension** number you want to configure.

  The parameters of the SIP Extension number you selected will appear on this page.

For SARVAM UCS upto 999 SIP Extensions can be registered with the system.

- Select the **Use SIP Extension** check box to enable the SIP extension. Default: disabled.

- In **Name**, enter a name for the SIP Extension, which may be the name of the person who will use the SIP Extension or the name of a Department. The name you enter here will be displayed as the Caller ID of the SIP Extension on the remote user's phone, when the SIP Extension user makes calls.

  The name may consist of a maximum of 18 alphanumeric characters.

- Enter the **SIP ID** for the extension. The SIP ID is necessary for registering the SIP Extension with the Registrar of the system. It is the number with which you can call the SIP Extension. Any extension user of the SARVAM UCS can call a SIP Extension by dialing the SIP ID assigned to the SIP extension. SIP ID of each SIP Extension must be a unique number string of a maximum of 6 digits. Any combination of digits from 0 to 9 and the characters * and # are allowed. You cannot assign the same SIP ID to more than one extension.

  To assign SIP IDs according to your preference and requirement to a range of SIP Extensions, see “Assigning Access Codes to a Range of Extensions”.

  By default, the SIP IDs are Blank.

The SIP ID will be set to default value (blank), when you restore the default settings of the system.

- In **Authentication ID**, enter the number which you want the system to use for user authentication of the SIP messages received from the SIP Extension. You cannot keep this field blank and it must be unique for each SIP extension. The number may be a string of maximum 6 digits. All ASCII characters except < > and " (double quote) are allowed. Default: Blank.

Make sure the User ID configured in "Digest Authentication" does not conflict with the Authentication ID configured above.

- In **Authentication Password**, enter the password manually or click **Generate** to automatically generate a unique password. This password will be used by the system to authenticate the SIP messages received from the SIP Extension. To avoid unauthorized access, we recommend you to change the password regularly. Make sure it is strong and is kept confidential. When you enter the password manually, the password must:

  - be of minimum 6 characters and can be a maximum of 12 characters.
  - include atleast one upper-case, one lower-case, one number and one special character.
  - all ASCII characters (except Percentage **%**, Hash **#,** Equal to **=,** Plus **+**, And **&**, Backslash **\,** Less than **<,** Greater than **>,** Apostrophe **',** Double Quote **"** and **Space**) are allowed.
    Default: Blank.

  To provide additional security, when the Authentication fails 10 times consecutively due to wrong Authentication ID / Authentication Password, the system will blacklist the IP Address and Port for registration of this SIP Extension. However, you can remove the IP Address from the Blacklist IP Address list. See "Black List IP Address - SIP Extensions" for more details. This activity will be logged in the "System Activity Log" as well as "Simple Network Management Protocol (SNMP)".

Make sure you note down or copy the Authentication Password in a confidential file.

- In **Call Appearances**, define the maximum number[159] of simultaneous incoming calls that the SIP Extension user should be allowed to receive. You can set up to 10 call appearances for a SIP Extension. Default: 2.

  When Call Appearance is set to 2, the SIP Extension can receive 2 calls at a time.

- Under **Auto Sign-In,** enter the **Email ID** of the extension user you wish to store. The Email ID can be a maximum of 64 characters. The Email ID is used for various server features.

***Auto Sign-In parameters — Send Configuration Mail and Mail Status,** are applicable only for Mobile Clients — VARTA ADR100, VARTA AMP100 applications.*

*The **Send Configuration Mail** button will appear only after you have enabled the SIP Extension and configured the SIP ID, Authentication ID and Password.*

- Under **Authentication**, enable Authentication of any or all of the following SIP Message Options by selecting the respective check boxes:
  - INVITE Request
  - SUBSCRIBE Request

159. *For the calls that are routed through the CPU, the number of Vocoder channels that will be supported would be as per the license you purchase.*

By default, the SIP Message Options INVITE and SUBSCRIBE are enabled.

Make sure that the Authentication ID for the SIP Extension has been programmed.

- For secure conversations over SIP, enable **SRTP Mode**. The SARVAM UCS supports the following options:
  - **Disable:** SARVAM UCS uses normal RTP for transporting the speech packets.
  - **Optional:** SARVAM UCS uses SRTP for transporting the speech packets. If the remote user does not support SRTP, SARVAM UCS will use normal RTP for transporting the speech packets.
    - If you select this option, you must configure the **SRTP Media Type**. You may select **AVP** or **SAVP**. By default, AVP is selected as the SRTP Media Type.
  - **Forced:** SARVAM UCS uses only SRTP (SAVP) for transporting the speech packets. If the remote user does not support SRTP, SARVAM UCS will reject incoming calls from and drop outgoing calls made to such users.

    By default, SRTP Mode is Disabled.
- Assign a **SIP Hardware Template** to the SIP Extension. Default: 01. The "SIP Hardware Template" contains voice quality related features such as Voice Codec selection, Tx and Rx Gains, Echo Cancellation, Jitter Buffer and Fax-over-IP options and related parameters

  There are 32 different templates to choose from. Each template can also be altered to suit your requirement and preferences. By default, Template number 01 assigned to all SIP Extensions as well as to SIP Trunks.

  Check if the values in this template fulfill requirements of the SIP Extension. If Template 01 fulfills the feature requirements, retain Template 01.

  If a different set of SIP hardware features are to be allowed to this SIP Extensions, prepare another template and assign it to this extension. To do this,
  - Under **VoIP Configuration**, click the **SIP Hardware Template** link.
  - Select a Template number, for example 02.
  - Customize Template number 02 and click **Submit**.
  - Return to the **SIP Extension Settings** page.
  - Select the number of the Template you customized, Template 02 in the **SIP Hardware Template** field.

  Also see the topic "SIP Hardware Template" to know more about customizing the templates and applying on the SIP Extensions.
- Assign a **Station Basic Feature Template** to the SIP Extension. Default: The "Station Basic Feature Template" has a set of features like Time Table, Class of Service, Toll Control, Operator, Storage of Incoming and Outgoing Calls, Outgoing Trunk Bundle groups. There are 50 different templates to choose from. Each template can also be altered to suit your requirement and preferences.

If the default Station Basic Feature Template 01 fulfills the feature requirements of the SIP Extension ("Class of Service (COS)", "Toll Control", "OG Trunk Bundle Group", etc.) retain this template, you may also customize this template. If you want to assign a different set of features to this SIP Extension, prepare a different Station Basic Feature Template and apply it to this extension. To do this,

- Under **Configuration**, click the **Station Basic Feature Template** link.
- Select a Template number, for example 05.
- Customize Template number 05 and click **Submit**.
- Return to the **SIP Extension Settings** page.
- Select the number of the Template you customized, Template 05, in the **Station Basic Feature Template** field.
- Click **Submit** to save changes.

Also, see the topic "Station Basic Feature Template" to know more about customizing the templates and applying on extensions.

- Assign a **Station Advanced Feature Template** to the SIP Extension. Default: Template 01. The "Station Advanced Feature Template" has a set of advanced features for extensions such as Alarm Notification settings, Routing of Incoming Auto Attendant Calls, Call Duration Control, Floor Service, etc. There are 50 different templates to choose from. Each template can also be altered to suit your requirement and preferences.

  Check if the default template fulfills the feature requirements of the SIP Extension by clicking the **Station Advanced Feature Template** link.

  You may retain this template and customize it further, or customize another template if a different set of features are to be allowed to this SIP Extension. To customize/prepare another template,

  - Under **Configuration**, click the **Station Advanced Feature Template** link.
  - Select the Template number, for example 02, and customize this template.
  - Click **Submit** and return to the 'SIP Extension Settings' page.
  - In the **Station Advanced Feature Template** field, select the number of the template you customized.
  - Click **Submit** to save changes.

  Also see the topic "Station Advanced Feature Template" for instructions on customizing these templates and applying them on the extensions.

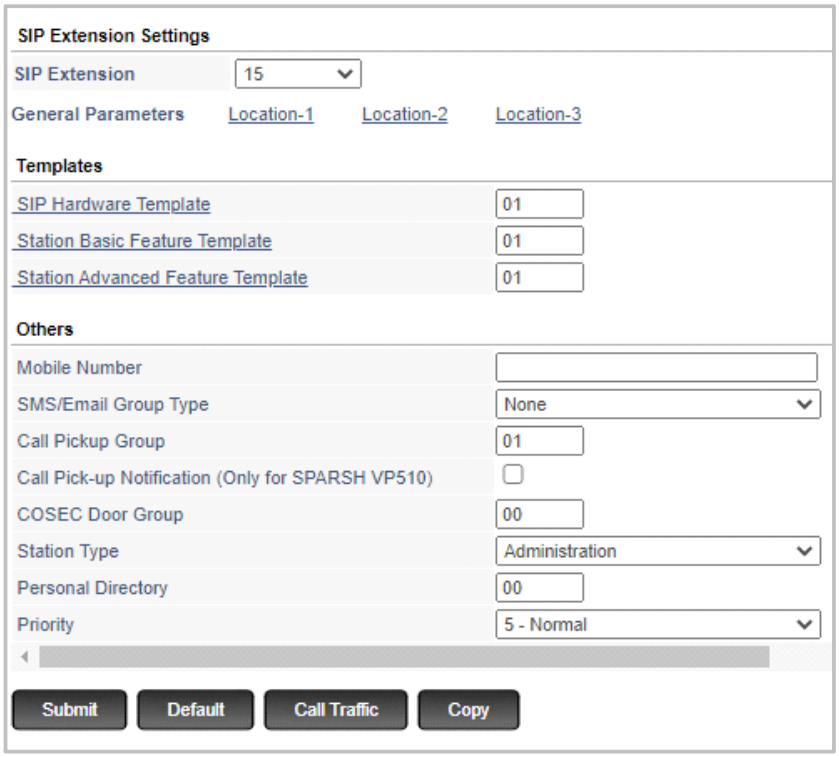
- In **Voice Mail Settings**, click the Voice Mail Settings link. The respective Extension Voice Mail Settings window will open. You may edit the parameters. For details, see "Extension Voice Mail Settings".

The Voice Mail Settings link will be visible only if you have configured the respective SIP ID.

Click **Close** to close the window.

Advanced Configuration Parameters

- If you want to provide other features like Personal Directory, Priority, or assign a Station Type to the SIP Extension, click the **Advanced** button at the bottom of the page.



- Enter the **Mobile Number** of the extension user you wish to store. The Number can be a maximum of 16 digits.

- You can assign the extension user to a Group. Select the desired **SMS/Email Group Type** from the list.The system clubs together extension users assigned the same Group. Default: None. For details, see “SMS/Email Group”.

- Assign the SIP Extension to a **Call Pick-up Group**, if required.

  Call Pick Up allows the SIP Extension to 'pick up' (answer) calls ringing on any other extension, by using the respective icon from Client GUI, without physically going to the ringing extension. It also allows incoming calls for the SIP Extension to be answered by the other extensions assigned the same Call Pick-Up group.

  For this to work, both the ringing extension and the extension picking up the call must be in the same 'Call Pick Up Group'. Refer “Call Pick Up” for instructions on how to create groups. You can create as many as 99 groups numbered from 01 to 99.

  Enter the number of the Call Pick-Up Group you created for this SIP Extension in this field.

- Select the **Call Pick-up Notification (Only for SPARSH VP510)** check box, if you want the Call Pick-up Notification to be displayed on the phone LCD. The Call Pick-up Notification will be displayed for internal as well as external calls. The notification will be displayed on SPARSH VP510 only if the ringing extension is in the same Call Pick-up Group. The notification will be displayed only when SPARSH VP510 is in idle or dial state. The notification will display the name/number of the Caller along with the name/number of the Called. For example, if the name and number of the Caller is ABC, 2001 and the name and number of

Called is XYZ, 1001, the LCD notification will be displayed as ABC -> XYZ. Along with the LCD display the phone back-light will also be lit.

For this to work, make sure you have enabled Call Pick-up in COS as well as assigned a Call Pick-up Group to the extension. Refer to "Call Pick Up" and "Class of Service (COS)". Call Pick-up Notifications will be displayed for DKP, SLT as well as SIP Extensions and for calls landing through CO, SIP as well as T1E1 Trunks. For details of the Notification, refer to the EON510_SPARSH VP510 V2 User Guide.

- You must assign the extension user to a **COSEC Door Group** for COSEC Integration. The users in the same group must be assigned the same group. You can create as many as 50 groups numbered from 00 to 50. Users who are assigned COSEC Door Group '00' are not a part of any group. See "COSEC Integration" for more information.

- If this is an Operator extension and you want the system to play beeps during a conference to the participants, to indicate the presence or absence of the Operator, select the **Station Type** as **Assistant**.

  If you are using the system in the *Hotel Mode*, select the **Station Type** for the SIP Extension as **Administration/Assistant** or **Guest**. The system will consider the options Administrator and Assistant as same.

- You may assign a **Personal Directory** number to the SIP Extension. Default: 00.

  A Personal Directory is a list of 25 frequently dialed numbers, each of which are stored by Index number (location code), Name and Trunk Access Codes ("Out Going Trunk Bundle Group Index"). The Personal Directory is necessary for using the features "Abbreviated Dialing" and "Dial By Name".

  When a Personal Directory is assigned to a SIP Extension, make sure you also configure this directory. The Personal Directory can be programmed by the SIP Extension users and by the System Engineer. Refer the topic "Abbreviated Dialing" for instructions on programming the Personal Directory.
  If Personal Directory is not to be assigned, enter 00 in this field.

- Select a **Priority** Level for the SIP Extension from 1 to 9. Default; 5-Normal.

  Each extension of the SARVAM UCS is assigned a Priority Level starting from 1, 2, 3... to 9, with '1' being lowest Priority and '9' being highest Priority. Whenever an extension (phone) with higher priority calls an extension with lower priority, a triple ring is placed on the called extension. To know more, read the feature description "Priority".

  If this SIP extension is assigned to Operator, you may want to set a higher priority for this extension.

- Click **Submit** to save your SIP Extension Settings.

Configuring Matrix Extended Phone Settings using Jeeves

You can register three Matrix Extended Phones/Soft Clients at three different locations as a single SIP Extension. You can connect the same or different types of Extended Phones/Soft Clients — SPARSH VP248, SPARSH VP330, SPARSH VP310, SPARSH VP510, SPARSH VP210, Matrix VARTA ADR100 Mobile UC Client, Matrix VARTA AMP100 Mobile UC Client or MATRIX VARTA WIN200 Desktop UC Client — at each location. In this case we assume that MATRIX VARTA WIN200 Desktop UC Client are registered at Location 1, 2 and 3.

If you have connected SPARSH VP248 refer to "Configuring Matrix Extended Phone Settings using Jeeves" in *Configuring Matrix SPARSH VP248 as SIP Extensions*.

If you have connected SPARSH VP310 at any of the locations, refer to “Configuring Matrix SPARSH VP310”.

If you have connected SPARSH VP330 at any of the locations, refer to “Configuring Matrix SPARSH VP330”.

If you have connected SPARSH VP510 at any of the locations, refer to “Configuring Matrix SPARSH VP510”.

If you have connected Extended SPARSH VP710 at any of the locations, refer to “Configuring Matrix Extended SPARSH VP710”.

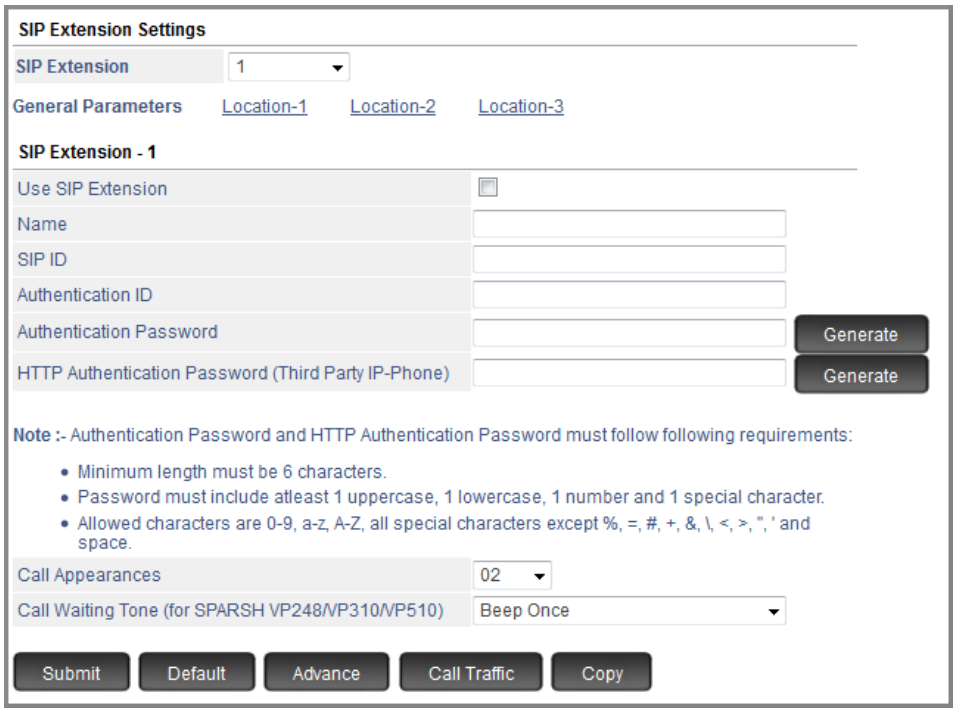
If you have connected Extended SPARSH VP210 at any of the locations, refer to “Configuring Matrix SPARSH VP210”.

If you have registered Matrix VARTA ADR100 and VARTA AMP100 Mobile UC Clients in any of the locations, refer to “Configuring Matrix VARTA ADR100/AMP100 UC Clients”.

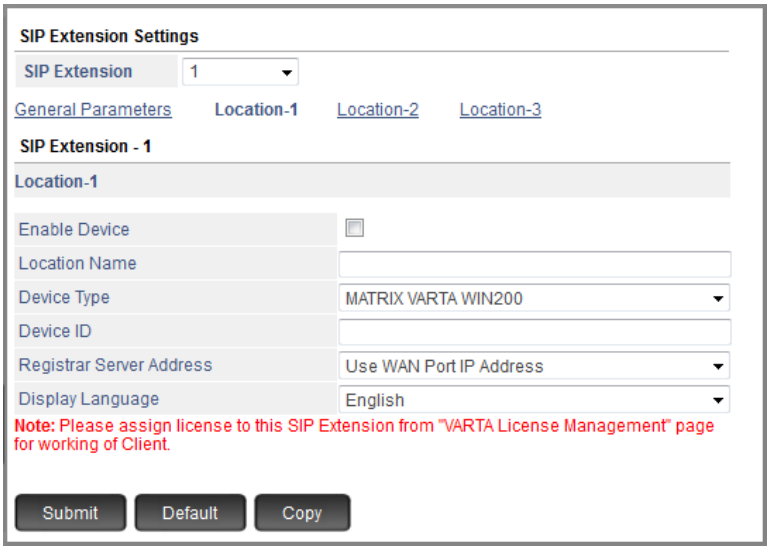
If you want to use more than one UC Clients as a SIP Extension, configure their settings as **Location 1**, **Location 2** and **Location 3**.

*If you want to use the IM functionality in the UC Client, you must configure it at **Location-1** only.*

- Click **Location 1**.



- The settings of the phone at **Location 1** appear.



- Select the **Enable Device** check box. Default: Disabled.

- Enter the **Location Name** for the phone to identify the phone. Location name may be the place where the phone is located (e.g.: Head office, branch, residence). The Location Name may consist of 18 characters (maximum). Default: Blank.

- Select **MATRIX VARTA WIN200** as the **Device Type** at this location. Make sure you assign the desired license to this SIP extension. For details, see “VARTA License Management”.

- In **MAC Address** enter the Device ID here. Default: blank.

  SARVAM UCS validates the desktop/PC on which you have installed the UC Client on the basis of the Device ID, and provides configuration on validation.

  As SARVAM UCS allows registration of the SIP Extension from three different locations, it identifies the SIP Extension in each location by the programmed Device ID.

- Select the appropriate **Registrar Server Address** to register the UC Client with the SIP Registrar of SARVAM UCS, according to your installation scenario:

  - If you want the UC Client to register using the WAN network, select **Use WAN Port IP Address** as Registrar Server Address.

  - If you want the UC Client to register using the LAN network, select **Use LAN Port IP Address** as Registrar Server Address.

  - If the UC Client is registered in the Public (Global) Network and SARVAM UCS is located behind a Router, or behind a NAT Router and STUN is programmed, select **Use Router/STUN's IP Address** as the Registrar Server Address.

    Make sure you configure either the **Router’s Public IP Address** or **Simple Traversal of UDPs through NATs (STUN)** in Network Parameters. For details, see “Configuring Network Parameters”.

- If Dynamic DNS is configured in the Network Parameters, select **Use Dynamic DNS Host Name** as Registrar Server Address.

  By default, Use WAN Port IP Address is selected as the Registrar Server IP Address.

Transport Mode and SRTP

- Select the protocol to be used to transport the SIP messages. You can select the **Transport Mode** as **TCP** or **TLS**.

If you select TCP, make sure the SIP Over TCP check box is selected in VoIP Parameters.

If you select TLS, make sure the SIP Over TLS check box is selected in VoIP Parameters.

- For secure conversations over SIP, select the **Enable SRTP?** check box. The SIP messages will be transported over SRTP only.

  The UC Client supports RTP Relay. For detailed description, see “Configuring VoIP Parameters”.

SMS Over IP

- If you want UC Client users to send SMS to any extension user as well as receive IM from any extension user, select the **Enable SMS Over IP** check box. For detailed information, see “SMS over IP”.

RTP Port

- Define the **RTP Port**. This is the port on which the client listens for RTP packets. This port is also used as the source port for sending RTP packets to the remote peer. The valid range for this port is 1025-65278. Default: 8000.

Quality of Service

- Set the **SIP Quality of Service (QoS)** for SIP signaling:

  - **SIP DiffServe/ToS**: The system sends all the SIP signaling messages with this QoS setting. This field defines the priority bits for SIP messages. The Valid *SIP DiffServe/ToS* range is from 00-63, default: 26.

  - **Voice DiffServe/ToS:** The system sends all the Voice packets with this QoS setting. This field defines the priority bits for Voice packet. It also improves the voice quality.The Valid *Voice DiffServe/ToS* range is from 00-63, default: 46.

  - **Video DiffServe/ToS:** The system sends all the Video packets with this QoS setting. This field defines the priority bits for Video packet. It also improves the video quality.The Valid *Video DiffServe/ToS* range is from 00-63, default: 46.

Configure any decimal value as per your requirement from the table mentioned below:

Traffic Type	DSCP Value (dec)
Best Effort	0
Background	8
Excellent Effort	40
AudioVideo	40
Voice	56
Control	56

NAT Keep Alive

- If the UC Client is connected behind a NAT router, configure **NAT Keep Alive**.
    - Select the check box **Enable NAT Keep Alive** to send Keep Alive messages periodically to refresh the binding in the NAT router. Default: Disabled.
    - Define as **Interval (sec)**, the time period, from 001 to 999 seconds, after which the phone should send Keep Alive message. Default: 120 seconds. The time period you define should be less than the binding timer of the router.

Timers

- Set the following **Timers** to the desired value, where required:
    - **SIP INVITE Timer (sec):** This is the time in seconds that the client waits for a response from the called party after ending INVITE message. This timer starts after sending INVITE message to the called party and stops on receipt of the provisional response or the final response or when the user disconnects the call. On expiry of the timer, the client terminates the call process and gives an error tone to the user. The range of the SIP INVITE TIMER is 10-180 seconds. Default: 30 seconds.
    - **SIP Provisional Timer (sec)**: This is the time in seconds that the client waits for final response after receiving the provisional response from the called party. This timer starts on the receipt of the provisional response from the called party and stops on receipt of the final response from the called party or when the user disconnects the call. On expiry of the timer, the client terminates the call process and gives error tone to the user. The range of SIP Provisional Timer is 10-180 seconds. Default: 60 seconds.
    - **General Request Timer (sec):** This is the time in seconds for which the client waits for response of a transaction request. This timer starts on initiating a transaction. This timer stops on receipt of a response for the request. On expiry of the timer, the phone clears the transaction. This timer is used for Registration request, etc. The range of the General Request Timer is 10-60 seconds. Default: 20 seconds.
- Click **Submit** to save settings.
- If you have completed the configuration of the MATRIX VARTA WIN200 at Location 1, follow the same steps as described above to configure the UC Client at Location 2 and Location 3.

  However, if you want to replicate the configuration of the UC Client Settings same as Location 1 to Location 2 and Location 3, you can use the **Copy** button present at the bottom of the page. To know more, refer to “Copy Parameter Values”.

If you wish to Copy only the configurations from a location to all other locations where the same IP Phone is connected, make sure you clear the Device Type check box.

If you wish to copy the Device Type as well as all the configurations from a location to all the SIP Extensions, make sure all the check boxes are selected.

When you change any of the parameters listed below in the SIP Extension at Location 1, 2, 3, the phone will go in Auto Configuration mode automatically, if registered:

- Use SIP Extension
- SIP ID
- Name
- Authentication ID
- Authentication Password
- Registrar Server Address
- MAC Address/IMEI/ESN Number
- Enable Device
- Device Type
- Transport Mode
- Enable SRTP
- QoS
- SIP/RTP Ports
- RTP Listening Port
- SMS over IP
- NAT Keep Alive
- SIP Timers
- Class of Service
- Trunk Access Code
- The SE Password of SARVAM UCS is changed
- Specific parameters in Network Port parameters are changed
- Specific parameters in VoIP Parameters are changed
- You restart the System
- Set the System to Default

Configuring Matrix VARTA ADR100/AMP100 UC Clients

Matrix offers the following Mobile UC Clients[160]:

- MATRIX VARTA ADR100 for Android Smartphones/Tablets
- MATRIX VARTA AMP100 for iPhones

To use MATRIX VARTA UC Clients for Mobile, make sure you have:

- Purchased and activated the VARTA Essential, VARTA Professional or VARTA Collaboration license. For more details, see “License Management”.
- Assigned the desired license to the SIP Extension. For more details, see “VARTA License Management”.

To know the list of featured supported, refer to “SARVAM UCS Features Supported in Terminals”.

For the VARTA Mobile Clients the system supports,

- automatic configuration and registration though Auto-Sign-In Configuration Mail. For this you must:

  - Configure the **Auto Sign-In Parameters**. For details, refer “Configuring Auto-Sign-In Parameters”
  - Configure the **General Parameters** in **SIP Extensions Settings**. For details, refer “Configuring SIP Extension Settings using Jeeves”
  - Make sure you send the **Auto Sign-In Configuration Mail**. For details, refer “Configuring SIP Extension Settings using Jeeves”

    You can also view the status of Auto Sign-In Email in “Viewing SIP Extension Status”

- manual configuration and registration, follow the instructions in “Configuring Mobile UC Clients using Jeeves”

MATRIX VARTA ADR100

MATRIX VARTA ADR100 is a proprietary SIP (Session Initiation Protocol) based UC Client Application running on Android Phones/Tablets, delivering full-array of Matrix SARVAM UCS features to the user on-the-go along with an added advantage of UC features. Through tight integration with the enterprise mobility features of the SARVAM UCS, VARTA ADR100 provides advance call capabilities including Conferencing, Corporate Directory Access (Global Directory), Call Logs and Conversation Recording with one-touch access. Other than these you can take the advantage of using premium features like Video Calling, IM, IM to SMS, Presence notification and corporate Voicemail System access to enhance your overall mobile experience.

Mobile workers can use any Wi-Fi or cellular data networks to stay connected with business communications while working from office, home or travelling to any location. An innovative and easy to understand user interface delivers all productivity features at fingertips that enhance speed of communication and collaboration with office users and customers.

For detailed product information and operation instructions, refer to the *MATRIX VARTA ADR100 User Guide for Mobile/Tablet.*

160. SARVAM UCS supports only IPv4 Addresses for registering Mobile UC Clients.

MATRIX VARTA AMP100

MATRIX VARTA AMP100 is a proprietary SIP (Session Initiation Protocol) based UC Client Application running on iPhones, delivering full-array of Matrix SARVAM UCS features to the user on-the-go along with an added advantage of UC features. Through tight integration with the enterprise mobility features of the SARVAM UCS, VARTA AMP100 provides advanced call capabilities including Conferencing, Corporate Directory Access (Global Directory), Call Logs and Conversation Recording with one-touch access. Other than these, you can take the advantage of using premium features like Video Calling, IM, IM to SMS, Presence notification/subscription and corporate Voicemail access to enhance your overall mobile experience.

Mobile workers can use any Wi-Fi or cellular data networks to stay connected with business communications while working from office, home or traveling to any location. An innovative and easy to understand user interface delivers all productivity features at fingertips that enhances speed of communication and collaboration with office users and customers.

For detailed product information and operation instructions, refer to the *MATRIX VARTA AMP100 User Guide.*

Configuring Mobile UC Clients using Jeeves

To be able to register and use the Mobile Handsets/Tablets as Matrix VARTA Mobile UC Clients, you must configure the following:

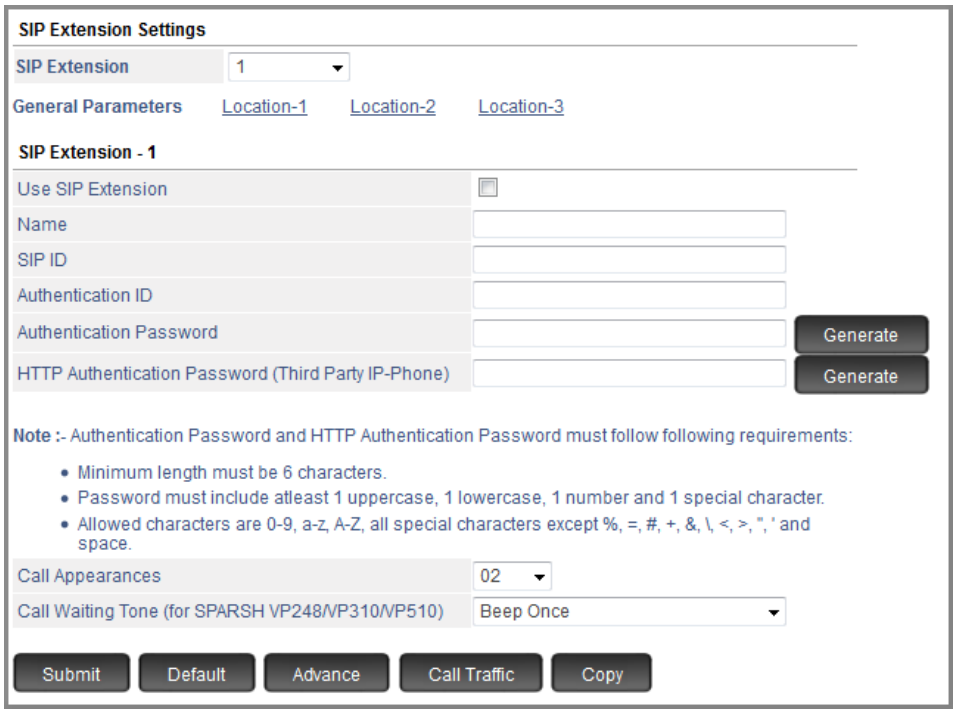
- SIP Extension General Parameters, see “Configuring SIP Extension General Parameters”.
- SIP Extension Settings, see “Configuring SIP Extension Settings using Jeeves”.
- Extended Phone Settings, see “Configuring Matrix Extended Phone Settings using Jeeves”.
- Voice Mail Settings, if you want to provide Voice Mail to the extension. See “Extension Voice Mail Settings”.

Configuring SIP Extension Settings using Jeeves

- Log into Jeeves as System Engineer.

- Under **Configuration**, click **VoIP Configuration**.

- Click **SIP Extension Settings**.



The page of SIP Extension 001 opens.

- You may select the **SIP Extension** number you want to configure.

  The parameters of the SIP Extension number you selected will appear on this page.

For SARVAM UCS upto 999 SIP Extensions can be registered with the system.

- Select the **Use SIP Extension** check box to enable the SIP extension. Default: disabled.
- In the **Name** field, enter a name of SIP Extension, which may be the name of the person who will use the UC Client/SIP Extension or the name of a Department. The name you enter here will be displayed as the Caller ID of the SIP Extension on the remote user's phone, when the SIP Extension makes calls.

  The name may consist of a maximum of 18 alphanumeric characters.

- Enter the **SIP ID** for the extension. The SIP ID is necessary for registering the SIP Extension with the Registrar of the system. It is the number with which you can call the SIP Extension. Any extension user of the SARVAM UCS can call a SIP Extension by dialing the SIP ID assigned to the SIP extension. SIP ID of each SIP Extension must be a unique number string of a maximum of 6 digits. Any combination of digits from 0 to 9 and the characters * and # are allowed.

  You cannot assign the same SIP ID to more than one extension.

  To assign SIP IDs according to your preference and requirement to a range of SIP Extensions, see “Assigning Access Codes to a Range of Extensions”.

  By default, the SIP IDs are Blank.

The SIP ID will be set to default value (blank), when you restore the default settings of the system.

- In the **Authentication ID** field, enter the number which you want the system to use for user authentication of the SIP messages received from the SIP Extension. You cannot keep this field blank and it must be unique for each SIP extension. The number may be a string of maximum 6 digits. All ASCII characters except < > and “ (double quote) are allowed. Default: Blank.

Make sure the User ID configured in “Digest Authentication” does not conflict with the Authentication ID configured above.

- In **Authentication Password**, enter the password manually or click **Generate** to automatically generate a unique password. This password will be used by the system to authenticate the SIP messages received from the SIP Extension. To avoid unauthorized access, we recommend you to change the password regularly. Make sure it is strong and is kept confidential. When you enter the password manually, the password must:

  - be of minimum 6 characters and can be a maximum of 12 characters.
  - include atleast one upper-case, one lower-case, one number and one special character.
  - all ASCII characters (except Percentage **%**, Hash **#,** Equal to **=,** Plus **+**, And **&**, Backslash **\,** Less than **<,** Greater than **>,** Apostrophe **’,** Double Quote **"** and **Space**) are allowed.
    Default: Blank.

  To provide additional security, when the Authentication fails 10 times consecutively due to wrong Authentication ID / Authentication Password, the system will blacklist the IP Address and Port for registration of this SIP Extension. However, you can remove the IP Address from the Blacklist IP Address list. See “Black List IP Address - SIP Extensions” for more details. This activity will be logged in the “System Activity Log” as well as “Simple Network Management Protocol (SNMP)”.

Make sure you note down or copy the Authentication Password in a confidential file.

- In **Call Appearances**, define the maximum number[161] of simultaneous incoming calls that the SIP Extension user should be allowed to receive. You can set up to 10 call appearances for a SIP Extension. Default: 2.

  When Call Appearance is set to 2, the SIP Extension can receive 2 calls at a time.

- Enter the **Email ID** of the extension user you wish to store. The Email ID can be a maximum of 64 characters. The Auto-Sign-In configuration mail will be sent to this ID.

- If you want the Mobile Clients to automatically configure and register with the Server, click the **Send** button adjacent to **Send Configuration Mail**.

  The Auto Sign-In Configuration Mail will be sent to the VARTA user on the **Email ID** configured above.

161. *For the calls that are routed through the CPU, the number of Vocoder channels that will be supported would be as per the license you purchase. Make sure the NX DBM VOCODER64 module is installed for SIP calls.*

The Auto Sign-In Mail button will appear only after you have enabled the SIP Extension and configured the SIP ID, Authentication ID and Password.

Make sure the Auto-Sig-In parameters have been configured. For details, refer to “Auto Sign-In Parameters”.

- The **Mail Status** will display either sent, failed or sending.

- Under **Authentication**, enable Authentication of any or all of the following SIP Message Options by selecting the respective check boxes:
  - INVITE Request
  - SUBSCRIBE Request

  By default, the SIP Message Options INVITE and SUBSCRIBE are enabled.

Make sure that the Authentication ID for the SIP Extension has been programmed.

- For secure conversations over SIP, enable **SRTP Mode**. The SARVAM UCS supports the following options:

  - **Disable:** SARVAM UCS uses normal RTP for transporting the speech packets.

  - **Optional:** SARVAM UCS uses SRTP for transporting the speech packets. If the remote user does not support SRTP, SARVAM UCS will use normal RTP for transporting the speech packets.

    - If you select this option, you must configure the **SRTP Media Type**. You may select **AVP** or **SAVP**. By default, AVP is selected as the SRTP Media Type.

  - **Forced:** SARVAM UCS uses only SRTP (SAVP) for transporting the speech packets. If the remote user does not support SRTP, SARVAM UCS will reject incoming calls from and drop outgoing calls made to such users.

    By default, SRTP Mode is Disabled.

- Assign a **SIP Hardware Template** to the SIP Extension. Default: 01. The “SIP Hardware Template” contains voice quality related features such as Voice Codec selection, Tx and Rx Gains, Echo Cancellation, Jitter Buffer and Fax-over-IP options and related parameters

  There are 32 different templates to choose from. Each template can also be altered to suit your requirement and preferences. By default, Template number 01 assigned to all SIP Extensions as well as to SIP Trunks.

  Check if the values in this template fulfill requirements of the SIP Extension. If Template 01 fulfills the feature requirements, retain Template 01.

  If a different set of SIP hardware features are to be allowed to this SIP Extensions, prepare another template and assign it to this extension. To do this,

  - Under **VoIP Configuration**, click the **SIP Hardware Template** link.

  - Select a Template number, for example 02.

- Customize Template number 02 and click **Submit**.
- Return to the **SIP Extension Settings** page.
- Select the number of the Template you customized, Template 02 in the **SIP Hardware Template** field.

Also see the topic "SIP Hardware Template" to know more about customizing the templates and applying on the SIP Extensions.

- Assign a **Station Basic Feature Template** to the SIP Extension. Default: The "Station Basic Feature Template" has a set of features like Time Table, Class of Service, Toll Control, Operator, Storage of Incoming and Outgoing Calls, Outgoing Trunk Bundle groups. There are 50 different templates to choose from. Each template can also be altered to suit your requirement and preferences.

  If the default Station Basic Feature Template 01 fulfills the feature requirements of the SIP Extension ("Class of Service (COS)", "Toll Control", "OG Trunk Bundle Group", etc.) retain this template, you may also customize this template. If you want to assign a different set of features to this SIP Extension, prepare a different Station Basic Feature Template and apply it to this extension. To do this,

  - Under **Configuration**, click the **Station Basic Feature Template** link.
  - Select a Template number, for example 05.
  - Customize Template number 05 and click **Submit**.
  - Return to the **SIP Extension Settings** page.
  - Select the number of the Template you customized, Template 05, in the **Station Basic Feature Template** field.
  - Click **Submit** to save changes.

  Also, see the topic "Station Basic Feature Template" to know more about customizing the templates and applying on extensions.

- Assign a **Station Advanced Feature Template** to the SIP Extension. Default: Template 01. The "Station Advanced Feature Template" has a set of advanced features for extensions such as Alarm Notification settings, Routing of Incoming Auto Attendant Calls, Call Duration Control, Floor Service, etc. There are 50 different templates to choose from. Each template can also be altered to suit your requirement and preferences.

  Check if the default template fulfills the feature requirements of the SIP Extension by clicking the **Station Advanced Feature Template** link.

  You may retain this template and customize it further, or customize another template if a different set of features are to be allowed to this SIP Extension. To customize/prepare another template,

  - Under **Configuration**, click the **Station Advanced Feature Template** link.
  - Select the Template number, for example 02, and customize this template.
  - Click **Submit** and return to the 'SIP Extension Settings' page.

- In the **Station Advanced Feature Template** field, select the number of the template you customized.

- Click **Submit** to save changes.

Also see the topic "Station Advanced Feature Template" for instructions on customizing these templates and applying them on the extensions.

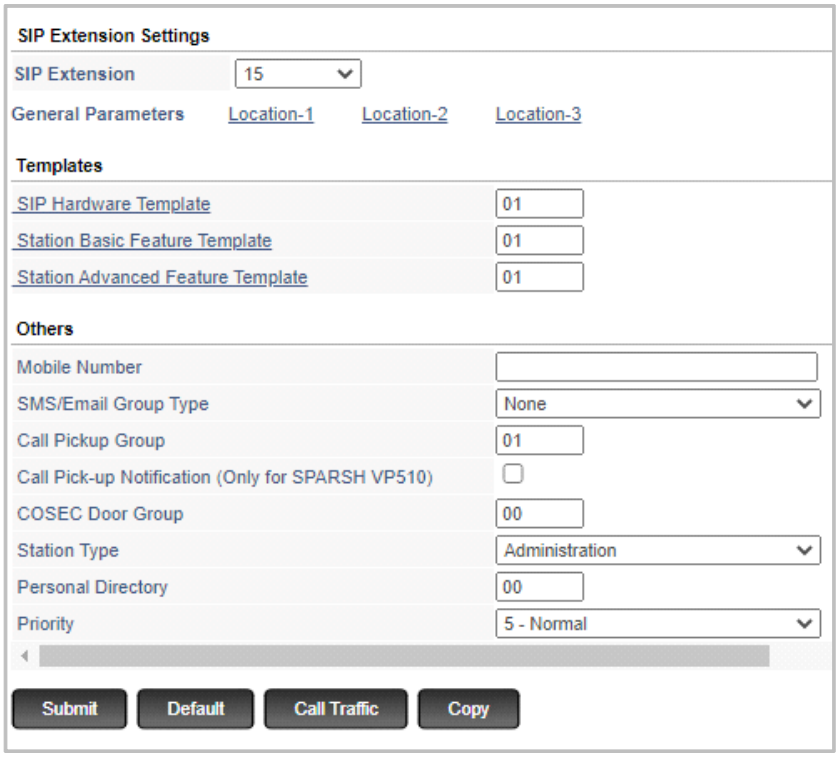
- In **Voice Mail Settings**, click the Voice Mail Settings link. The respective Extension Voice Mail Settings window will open. You may edit the parameters. For details, see "Extension Voice Mail Settings".

The Voice Mail Settings link will be visible only if you have configured the respective SIP ID.

Click **Close** to close the window.

Advanced Configuration Parameters

- If you want to provide other features like Personal Directory, Priority, or assign a Station Type to the SIP Extension, click the **Advanced** button at the bottom of the page.



- Enter the **Mobile Number** of the extension user you wish to store. The Number can be a maximum of 16 digits.

- You can assign the extension user to a Group. Select the desired **SMS/Email Group Type** from the list.The system clubs together extension users assigned the same Group. Default: None. For details, see "SMS/Email Group".

- Assign the SIP Extension to a **Call Pick-up Group**, if required.

  Call Pick Up allows the SIP Extension to 'pick up' (answer) calls ringing on any other extension, by using the respective icon from Client GUI, without physically going to the ringing extension. It also allows

incoming calls for the SIP Extension to be answered by the other extensions assigned the same Call Pick-Up group.
For this to work, both the ringing extension and the extension picking up the call must be in the same 'Call Pick Up Group'. Refer "Call Pick Up" for instructions on how to create groups. You can create as many as 99 groups numbered from 01 to 99.

Enter the number of the Call Pick-Up Group you created for this SIP Extension in this field.

- Select the **Call Pick-up Notification (Only for SPARSH VP510)** check box, if you want the Call Pick-up Notification to be displayed on the phone LCD. The Call Pick-up Notification will be displayed for internal as well as external calls. The notification will be displayed on SPARSH VP510 only if the ringing extension is in the same Call Pick-up Group. The notification will be displayed only when SPARSH VP510 is in idle or dial state. The notification will display the name/number of the Caller along with the name/number of the Called. For example, if the name and number of the Caller is ABC, 2001 and the name and number of Called is XYZ, 1001, the LCD notification will be displayed as ABC -> XYZ. Along with the LCD display the phone back-light will also be lit.

  For this to work, make sure you have enabled Call Pick-up in COS as well as assigned a Call Pick-up Group to the extension. Refer to "Call Pick Up" and "Class of Service (COS)". Call Pick-up Notifications will be displayed for DKP, SLT as well as SIP Extensions and for calls landing through CO, SIP as well as T1E1 Trunks. For details of the Notification, refer to the EON510_SPARSH VP510 V2 User Guide.

- If this is an Operator extension and you want the system to play beeps during a conference to the participants, to indicate the presence or absence of the Operator, select the **Station Type** as **Assistant**. If you are using the system in the *Hotel Mode*, select the **Station Type** for the SIP Extension as **Administration/Assistant** or **Guest**. The system will consider the options Administrator and Assistant as same.

- You may assign a **Personal Directory** number to the SIP Extension. Default: 00.
  A Personal Directory is a list of 25 frequently dialed numbers, each of which are stored by Index number (location code), Name and Trunk Access Codes ("Out Going Trunk Bundle Group Index"). The Personal Directory is necessary for using the features "Abbreviated Dialing" and "Dial By Name".

  When a Personal Directory is assigned to a SIP Extension, make sure you also configure this directory. The Personal Directory can be programmed by the SIP Extension users and by the System Engineer. Refer the topic "Abbreviated Dialing" for instructions on programming the Personal Directory.
  If Personal Directory is not to be assigned, enter 00 in this field.

- Select a **Priority** Level for the SIP Extension from 1 to 9. Default; 5-Normal.

  Each extension of the SARVAM UCS is assigned a Priority Level starting from 1, 2, 3... to 9, with '1' being lowest Priority and '9' being highest Priority. Whenever an extension (phone) with higher priority calls an extension with lower priority, a triple ring is placed on the called extension. To know more, read the feature description "Priority".

  If this SIP extension is assigned to Operator, you may want to set a higher priority for this extension.

- Click **Submit** to save your SIP Extension Settings.

Configuring Matrix Extended Phone Settings using Jeeves

You can connect the same or different types of Extended Phones/Soft Clients — SPARSH VP248, SPARSH VP330, SPARSH VP310, SPARSH VP510, SPARSH VP210, Matrix VARTA ADR100 Mobile UC Client, Matrix

VARTA AMP100 Mobile UC Client or MATRIX VARTA WIN200 Desktop UC Client — at each location. In this case we assume that Matrix VARTA Mobile UC Clients are registered at Location 1, 2 and 3.

If you have connected SPARSH VP248 refer to "Configuring Matrix Extended Phone Settings using Jeeves" in *Configuring Matrix SPARSH VP248 as SIP Extensions*.

If you have connected SPARSH VP310 at any of the locations, refer to "Configuring Matrix SPARSH VP310".

If you have connected SPARSH VP330 at any of the locations, refer to "Configuring Matrix SPARSH VP330".

If you have connected SPARSH VP510 at any of the locations, refer to "Configuring Matrix SPARSH VP510".

If you have connected Extended SPARSH VP710 at any of the locations, refer to "Configuring Matrix Extended SPARSH VP710".

If you have connected Extended SPARSH VP210 at any of the locations, refer to "Configuring Matrix SPARSH VP210".

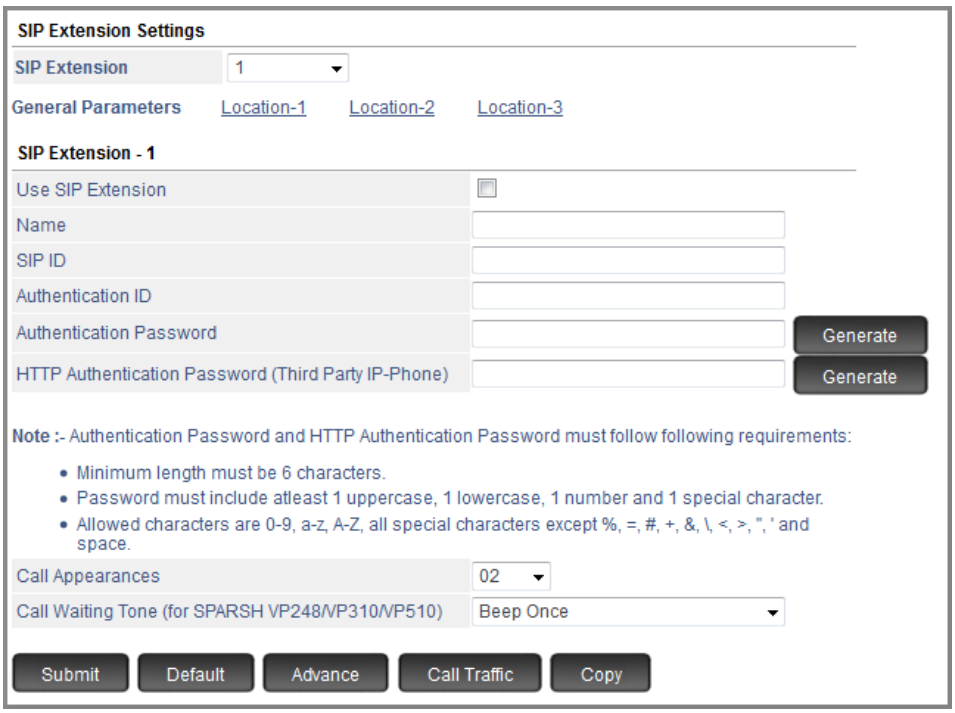
If you have registered Matrix VARTA Mobile UC Clients in any of the locations, refer to "Configuring Matrix VARTA ADR100/AMP100 UC Clients".

If you have registered MATRIX VARTA WIN200 Desktop UC Client in any of the locations, refer to "Configuring Matrix VARTA WIN200 UC Client".

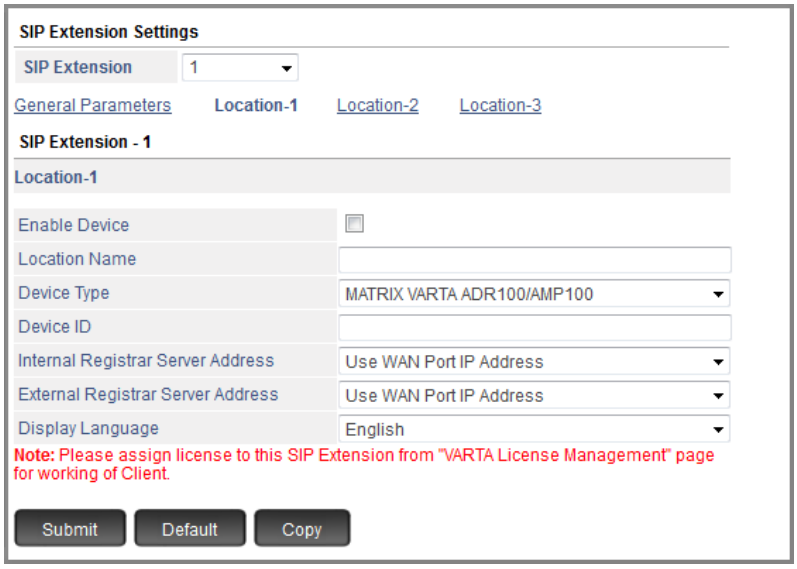
If you want to use more than one Matrix UC Clients as a SIP Extension, configure their settings at **Location 1**, **Location 2** and **Location 3**.

*If you want to use the IM functionality in the MATRIX VARTA ADR100/AMP100, you must configure it at **Location-1** only.*

- Click **Location 1**.



- The settings of the phone at **Location 1** appear.



- Select the **Enable Device** check box. Default: Disabled.

- Enter the **Location Name** for the phone to identify the phone. Location name may be the place where the phone is located (e.g.: Head office, branch, residence). The Location Name may consist of 18 characters (maximum). Default: Blank.

- Select **MATRIX VARTA ADR100/AMP100** (for the Android/iPhone Application) as the **Device Type** at this location. Make sure you assign the desired license to this SIP extension. For details, see “VARTA License Management”.

- Enter the **MAC Address/IMEI**[162]**/ESN Number** of the phone/tablet on which you have installed the application.

  If you are using an iPhone, enter the **Device ID** here. Default: blank.

  SARVAM UCS validates the phone/tablet on which you have installed the application on the basis of the IMEI/ESN Number or Device ID, and provides configuration on validation.

  As SARVAM UCS allows registration of the SIP Extension from three different locations, it identifies the SIP Extension in each location by the programmed IMEI/ESN Number/Device ID.

- Select the appropriate **Internal Registrar Server Address** to register the application with the SIP Registrar of SARVAM UCS within a private network. Select the appropriate option as per your installation scenario:

  - If you want the application to register using the WAN network, select **Use WAN Port IP Address** as the Internal Registrar Server Address.

  - If you want the application to register using the LAN network, select **Use LAN Port IP Address** as Registrar Server Address.

162. IMEI Number is the unique identification number of the GSM engine used in the Mobile handset.

- If Dynamic DNS is configured in the Network Parameters, select **Use Dynamic DNS Host Name** as Internal Registrar Server Address.

  By default, Use WAN Port IP Address is selected as the Internal Registrar Server Address.

- Select the appropriate **External Registrar Server Address** to register the application with the SIP Registrar of SARVAM UCS from a public network. Select the option according to your installation scenario:

  - If you want the application to register using the WAN network, select **Use WAN Port IP Address** as External Registrar Server Address.

  - If the application is connected in the Public Network and SARVAM UCS is located behind a Router, or behind a NAT Router and STUN is programmed, select **Use Router/STUN's IP Address** as External Registrar Server Address.

    Make sure you configure either the **Router's Public IP Address** or **Simple Traversal of UDPs through NATs (STUN)** in Network Parameters. For details, see "Configuring Network Parameters".

  - If Dynamic DNS is configured in the Network Parameters, select **Use Dynamic DNS Host Name** as External Registrar Server Address.

    By default, Use WAN Port IP Address is selected as the External Registrar Server Address.

Transport Mode and SRTP

- Select the protocol to be used to transport the SIP messages. You can select the **Transport Mode** as **TCP** or **TLS**.

If you select TCP, make sure the SIP Over TCP check box is selected in VoIP Parameters.

If you select TLS, make sure the SIP Over TLS check box is selected in VoIP Parameters.

For secure conversations over SIP, select the **Enable SRTP?** check box. The SIP messages will be transported over SRTP only.

The application supports RTP Relay. For detailed description, see "Configuring VoIP Parameters".

SMS Over IP

- If you want the VARTA users to send SMS to any extension user as well as receive IM from any extension user, select the **Enable SMS Over IP** check box. For detailed information, see "SMS over IP".

RTP Port

- Define **RTP Port**:

  - **RTP Listening Port:** This is the port on which the phone listens for RTP packets. This port is also used as the source port for sending RTP packets to the remote peer. The valid range for this port is 1025-65278. Default: 8000.

Quality of Service

- Set the **Quality of Service (QoS)** for SIP signaling:

  - **SIP DiffServe/ToS:** The system sends all the SIP signaling messages with this QoS setting. This field defines the priority bits for SIP messages. The Valid *DiffServe* range is from 00-63, default: 26.

  - **Voice DiffServe/ToS:** The system sends all the Voice packets with this QoS setting. This field defines the priority bits for Voice packet. It also improves the voice quality.The Valid *DiffServe* range is from 00-63, default: 46.

  - **Video DiffServe/ToS:** The system sends all the Video packets with this QoS setting. This field defines the priority bits for Video packet. It also improves the video quality.The Valid *DiffServe* range is from 00-63, default: 46.

    Configure any decimal value as per your requirement from the table mentioned below:

DSCP <=> IP Precedence Conversion Table			
DSCP Name	**DS Field Binary**	**Value Decimal**	**IP Precedence**
CS_0	000 000	0	0
CS_1	001 000	8	1
AF_{11}	001 010	10	1
AF_{12}	001 100	12	1
AF_{13}	001 110	14	1
CS_2	010 000	16	2
AF_{21}	010 010	18	2
AF_{22}	010 100	20	2
AF_{23}	010 110	22	2
CS_3	011 000	24	2
AF_{31}	011 010	26	3
AF_{32}	011 100	28	3
AF_{33}	011 110	30	3
CS_4	100 000	32	4
AF_{41}	100 010	34	4
AF_{42}	100 100	36	4
AF_{43}	100 110	38	4
C_{55}	101 000	40	5
EF	101 110	46	5
CS_6	110 000	48	6
CS_7	111 000	56	7
CS Class Selector (RFC_{2474})			
AFxy Assured Forwarding (x=class, y=drop precedence) (RFC_{2597})			
EF Expedited Forwarding (RFC_{3246})			

Timers

- Set the following **Timers** to the desired value, where required:

  - **SIP INVITE Timer (sec):** This is the time in seconds that the phone waits for a response from the called party after ending INVITE message. This timer starts after sending INVITE message to the called party and stops on receipt of the provisional response or the final response or when the user disconnects the call. On expiry of the timer, the phone terminates the call process and gives an error tone to the user. The range of the SIP INVITE TIMER is 10-180 seconds. Default: 30 seconds.

  - **SIP Provisional Timer (sec)**: This is the time in seconds that the phone waits for final response after receiving the provisional response from the called party. This timer starts on the receipt of the provisional response from the called party and stops on receipt of the final response from the called party or when the user disconnects the call. On expiry of the timer, the IP phone terminates the call process and gives error tone to the user. The range of SIP Provisional Timer is 10-180 seconds. Default: 60 seconds.

  - **General Request Timer (sec):** This is the time in seconds for which the phone waits for response of a transaction request. This timer starts on initiating a transaction. This timer stops on receipt of a response for the request. On expiry of the timer, the phone clears the transaction. This timer is used for Registration request, etc. The range of the General Request Timer is 10-60 seconds. Default: 20 seconds.

- Click **Submit** to save settings.

- If you have completed the configuration at Location 1, follow the same steps as described above to configure the application at Location 2 and Location 3.

  However, if you want to replicate the configuration of the MATRIX VARTA ADR100/AMP100 Settings same as Location 1 to Location 2 and Location 3, you can use the **Copy** button present at the bottom of the page. To know more, refer to “Copy Parameter Values”.

If you wish to Copy only the configurations from a location to all other locations where the same IP Phone is connected, make sure you clear the Device Type check box.

If you wish to copy the Device Type as well as all the configurations from a location to all the SIP Extensions, make sure all the check boxes are selected.

When you change any of the parameters listed below in the SIP Extension at Location 1, 2, 3, the phone will go in Auto Configuration mode automatically, if registered:

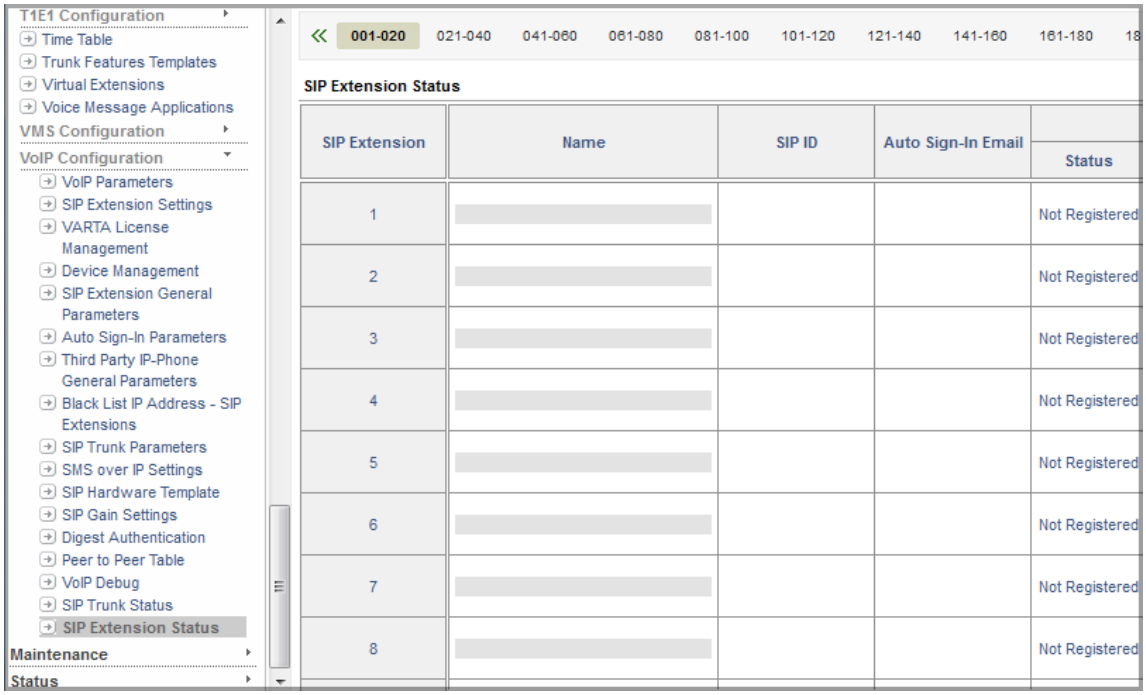
- Use SIP Extension
- SIP ID
- Name
- Authentication ID
- Authentication Password
- Internal Registrar Server Address
- External Registrar Server Address
- MAC Address/IMEI/ESN Number
- Enable Matrix Extended Phone Mode
- Extended Phone Type
- Transport Mode
- Enable SRTP
- QoS
- SIP/RTP Ports

- SIP Timers
- Class of Service
- Trunk Access Code
- The SE Password of SARVAM UCS is changed
- Specific parameters in Network Port parameters are changed
- Specific parameters in VoIP Parameters are changed
- You restart the System
- Set the System to Default

Viewing SIP Extension Status

You can view the Status of SIP Extension using Jeeves. To do this,

- Log into Jeeves as System Engineer.
- Under **Configuration**, click **VoIP Configuration**.
- Click **SIP Extension Status**.



- The SIP Extension Status page will open and display the following for each SIP Extension,
  - SIP Extension number
  - Name of the SIP extension
  - SIP ID assigned to the SIP Extension
  - Auto Sign-In Email; whether the mail is sent, failed or sending.
  - REGISTRATION status; whether the SIP Extension is registered or not.
  - Contact 1 (for *MATRIX VARTA AMP100/ADR100)*

When the device is Registered - It will display the SIP ID, IP Address and the Registration Expiry Timer.

When the device is in the Background - It will display Registered and the time remaining for the expiry of the VARTA Client Inactivity Timer.

When Unregistered - The existing details will be cleared and it will be blank.

To know more, refer “Apple Push Notification Service Support” or “Firebase Cloud Messaging (FCM) Support”.

- Contact 2 - same as above.
- Contact 3 - same as above.

- You may Log out of Jeeves.

*You can also view the SIP Extension Status from the **Status** link. To view, click the SIP Extension link under Status.*

Configuring SIP Extensions using Bulk Configuration

In today's competitive world, time is an important asset for all the modern organizations. So, most of the organizations demand for a system that accomplish most of its functionality at a click of a button. As, we know, SARVAM UCS supports 999[163] extensions and configuring these extensions one by one may be tedious and time consuming. So, to overcome this concern of the organizations, SARVAM UCS supports Bulk configuration.

Bulk configuration allows you to configure a large number of SIP Extensions by simply uploading a CSV File in the Server. Once the CSV File is uploaded, the extensions listed in the file are configured in the server.

You can also configure extensions one-by-one, to know more, refer to "Configuring SIP Extension Settings as per the Extended Phone Type".

How to works

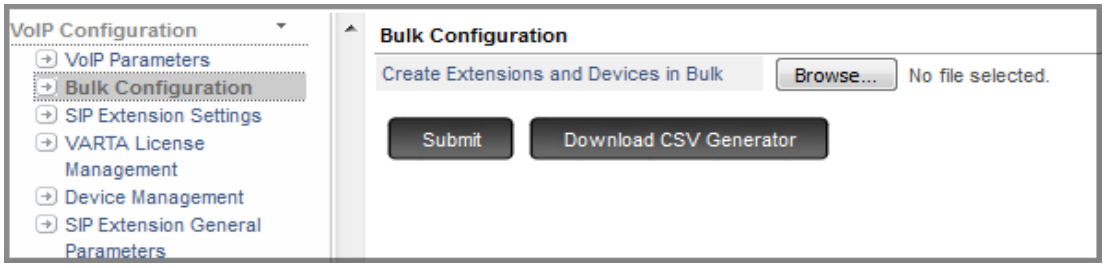
- Firstly, you need to download the CSV Generator Template from Jeeves. To know more, refer to "Download ing the CSV Generator Template".
- Secondly, customize the CSV Generator Template as per your requirement. To know how to customize the template, refer to "Preparing the CSV Generator Template".
- After you have prepared the Template, you need to generate the CSV File. To know more, refer to "Generating CSV File". After generating the file, you must export the file. To know how to export the file, refer to "Exporting the CSV File".
- Once the CSV file is exported, now you need to upload the file. To know how to upload the CSV File, refer to "Uploading the CSV File". The Extensions listed in the CSV File are configured in the Server.

Before configuring the extensions using Bulk Configuration, make sure Microsoft Excel is installed on your PC.

How to configure

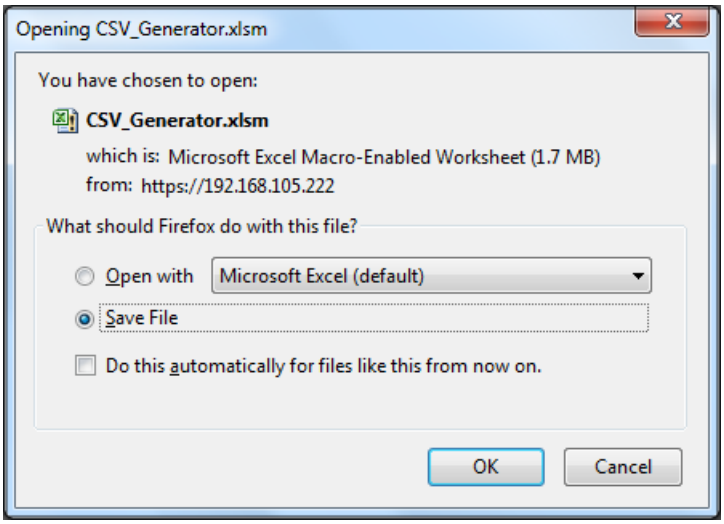
Downloading the CSV Generator Template

- Login as System Engineer.
- Under **VoIP Configuration**, click **Bulk configuration.**



163. *ETERNITY LENX/MENX supports 2000 SIP Extensions.*

- Click **Download CSV Generator** button to download the CSV Generator Template.
- The CSV_Generator.xlsm window opens.



- You can either open the CSV Generator Template or save the Template to a desired location on the local disk.

- *The above window display depends upon the browser you are using. Check the Download Settings of your browser and set the Download path accordingly.*
  OR
- *If your browser does not ask you for the location you want to save your file, it saves it in the default location according to the download path specified for that browser.*

Preparing the CSV Generator Template

After downloading the CSV Generator Template, you can now customize the template as per your requirement. The parameters listed in the CSV Generator Template are explained below in detail:

Parameters	Description
Start Extension Number	Enter the Extension Number for the first extension. For example: If you configure Start Extension Number as 5001, the system will start creating the extensions with 5001 as the first extension number followed by 5002, 5003.....for other extensions.
Total Extensions	Enter the total number of extensions you want to configure. For example: You want to configure 100 extensions, then enter 100 as the Total Extensions.

Parameters	Description
Start S/W Port No.	Enter the software port number from where you want to configure the first extension. For example: You want to configure the extensions from software port number 1, then enter 1 as the software port number.
Use SIP Extension	Select Yes to enable the SIP extensions. If you select No, the SIP extensions will not get registered with the server.
SIP Hardware Template	Enter the template number of the SIP Hardware Template you want to assign to the extensions. By default, the SIP Hardware Template number 01 is assigned. The SIP Hardware Template contains voice quality related features such as Voice Codec selection, Tx and Rx Gains, Echo Cancellation, Jitter Buffer and related parameters. Check if the values in this template fulfills the requirements of the extensions. If the Template number 01 fulfills the feature requirements, retain the Template number 01. If a different set of SIP hardware features are to be allowed to the extensions, then prepare another template and enter that template number in this field. To know how to customize the template, refer to “SIP Hardware Template”.
Station Basic Feature Template	Enter the template number of the Station Basic Feature Template you want to assign to the extensions. By default, Station Basic Feature Template number 01 is assigned. The Station Basic Feature Template has a set of features like Time Table, Class of Service, Toll Control, Operator, Storage of Incoming and Outgoing Calls, Outgoing Trunk Bundle groups. Check if the values in this template fulfills the requirements of the extensions. If the Template number 01 fulfills the feature requirements, retain the Template number 01. If a different set of features are to be allowed to the extensions, then prepare another template and enter that template number in this field. To know how to customize the template, refer to “Station Basic Feature Template”.

Parameters	Description
Station Advanced Feature Template	Enter the template number of the Station Advanced Feature Template you want to assign to the extensions. By default, Station Advanced Feature Template number 01 is assigned. The Station Advanced Feature Template has a set of advanced features for the extensions such as such as Alarm Notification settings, Routing of Incoming Auto Attendant Calls, Call Duration Control, Floor Service and related parameters. Check if the values in this template fulfills the requirements of the extensions. If the Template number 01 fulfills the feature requirements, retain the Template number 01. If a different set of features are to be allowed to the extensions, then prepare another template and enter that template number in this field. To know how to customize the template, refer to "Station Advanced Feature Template".
Call Pickup Group	Enter the Call Pickup Group number you want to assign to the extensions. By default, Call Pickup Group number 1 is assigned. Call Pick Up allows the extension user to 'pick up' (answer) calls ringing on any other extension, by dialing a feature code, without physically going to the ringing extension. It also allow incoming calls for the extension to be answered by the other extensions assigned the same Call Pick-Up group. For this to work, both the ringing extension and the extension picking up the call must be in the same 'Call Pick Up Group'. Refer "Call Pick Up" for instructions on how to create groups.
Station Type	Select the Station Type — *Administration/ Guest* you want to assign to the extensions.
Priority	Select the Priority level for the extensions from 1 to 9. By default, 5-Normal is selected as the Priority level. Each extension is assigned a Priority Level starting from 1,2, 3....to 9, with '1' being the lowest Priority and '9' being the highest Priority. To know more about the feature, refer to "Priority".

You can connect/ register three Matrix Extended IP Phones / UC clients/ Standard SIP Phones at three different location as a single SIP extension. You can register the same or different types of Extended IP Phones/ UC clients/ Standard SIP Phones — at each location.

For Example: You can configure SPARSH VP330 at Location 1, 2 and 3. If you want to use more than one SPARSH VP330 Extended IP Phones as a SIP Extension, configure their settings at Location 1, Location 2 and Location 3.

For Auto Provisioning, you must configure the Standard SIP phones at Location1 only. To know more, refer to "Configuring Standard SIP Phones".

Location 1	Description
Enable Device	Select Yes to enable the SIP extension.
Device Type	Select the desired Device Type for location 1.
Registrar Server Address	Select the appropriate Registrar Server Address to register the Extended IP Phones/ VARTA WIN200/ Standard SIP Phones with the SIP Registrar of SARVAM UCS, according to your installation scenario. If extensions are connected on the WAN network, select ***Use WAN Port IP Address*** as the Registrar Server IP Address. If extensions are connected on the LAN network, select ***Use LAN Port IP Address*** as the Registrar Server IP Address. If extensions are connected in the Global Network and SARVAM UCS is located behind a Router, or behind a NAT Router and STUN is configured, select ***Use Router/STUN's IP Address*** as the Registrar Server IP Address. Make sure you configure either the Router's Public IP Address or Simple Traversal of UDPs through NATs (STUN) in Network Parameters. For details, see "Configuring Network Parameters". If Dynamic DNS is configured in the Network Parameters, select ***Use Dynamic DNS Host Name*** as the Registrar Server IP Address. By default, ***Use WAN Port IP Address*** is selected as the Registrar Server IP Address.
Internal Registrar Server Address	Select the appropriate Internal Registrar Server Address to register the UC clients (VARTA ADR100/ VARTA AMP100) with the SIP Registrar of SARVAM UCS within a private network. Select the appropriate option as per your installation scenario: If you want the UC clients to register using the WAN network, select ***Use WAN Port IP Address*** as the Internal Registrar Server Address. If you want the UC clients to register using the LAN network, select ***Use LAN Port IP Address*** as the Internal Registrar Server Address. If Dynamic DNS is configured in the Network Parameters, select ***Use Dynamic DNS Host Name*** as the Internal Registrar Server Address. By default, ***Use WAN Port IP Address*** is selected as the Internal Registrar Server Address.

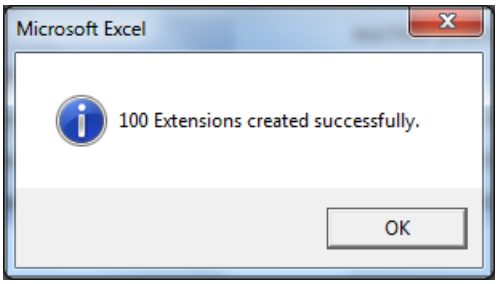
Location 1	Description
External Registrar Server Address	Select the appropriate External Registrar Server Address to register the UC clients (VARTA ADR100/ VARTA AMP100) with the SIP Registrar of SARVAM UCS from a public network. Select the option according to your installation scenario: If you want the UC clients to register using the WAN network, select ***Use WAN Port IP Address*** as the External Registrar Server Address. If the UC clients is connected in the Public Network and SARVAM UCS is located behind a Router, or behind a NAT Router and STUN is configured, select ***Use Router/STUN's IP Address*** as External Registrar Server Address. Make sure you configure either the Router's Public IP Address or Simple Traversal of UDPs through NATs (STUN) in Network Parameters. For details, see "Configuring Network Parameters". If Dynamic DNS is configured in the Network Parameters, select ***Use Dynamic DNS Host Name*** as the External Registrar Server Address. By default, ***Use WAN Port IP Address*** is selected as the External Registrar Server Address.

Similarly, you can configure the parameters listed in Location 2 and 3 as per your requirement.

Generating CSV File

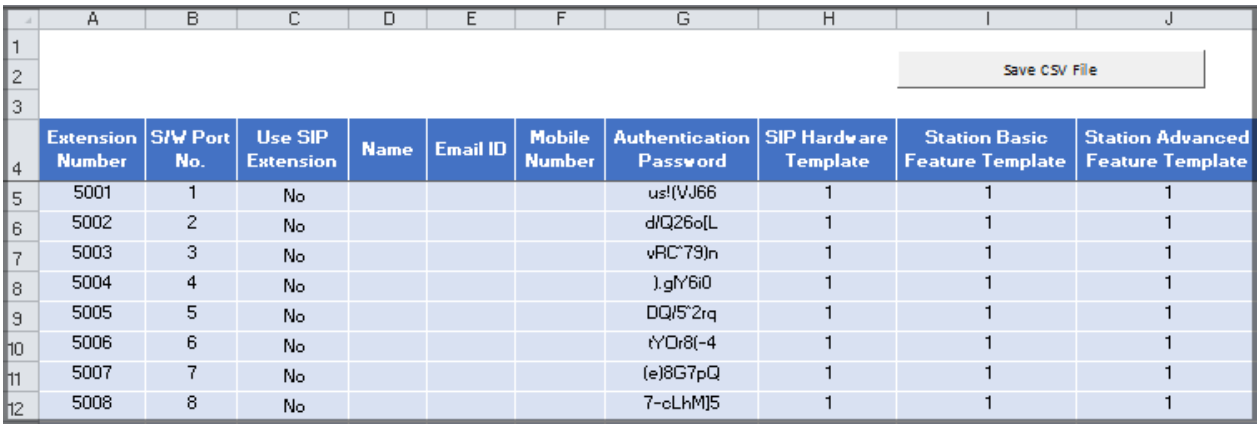
After you have configured the required parameters in the CSV Generator Template, you can now generate the CSV File. To do so,

- Click the **Generate csv** button present at the bottom of the CSV Generator Template.
- A pop window opens specifying the number of extensions that are created successfully.



- Click **OK**.

- The CSV File will be generated.



	A	B	C	D	E	F	G	H	I	J
1										
2									Save CSV File	
3										
4	Extension Number	S/W Port No.	Use SIP Extension	Name	Email ID	Mobile Number	Authentication Password	SIP Hardware Template	Station Basic Feature Template	Station Advanced Feature Template
5	5001	1	No				us!(VJ66	1	1	1
6	5002	2	No				d/Q26o[L	1	1	1
7	5003	3	No				vRC`79)n	1	1	1
8	5004	4	No				).gfY6i0	1	1	1
9	5005	5	No				DQ/5^2rq	1	1	1
10	5006	6	No				tYOr8(-4	1	1	1
11	5007	7	No				(e)8G7pQ	1	1	1
12	5008	8	No				7-cLhM]5	1	1	1

You can now modify the generated CSV File by adding the following details for each and every extension in the file as per your requirement.

- **Name:** Enter the name for the extensions listed in the file. The name you enter here will be displayed as the Caller ID of the extensions on the remote user's phone, when the extension user makes calls.

  It can be the name of the person who will use the extension or the name of a Department. You can configure a name of a maximum of 18 alphanumeric characters.

- **Email ID:** Enter the email address for the extensions listed in the file. The Email ID you enter here will be used to send the auto-sign email in Mobile UC clients — VARTA ADR100 and VARTA AMP100 applications. Refer "Auto Sign-In Parameters", to know more. The Email ID is also used for various server features.

  You can configure an Email ID of a maximum of 64 characters.

- **Mobile Number:** Enter the mobile number for the extensions listed in the file, if required. The mobile number can be a maximum of 16 digits.

- **Location Name:** Enter the location for the extension so as to identify the extension. Location Name may be the place where the extension is located (e.g: 2nd Floor/ Desk number -8, Head office, Branch office, residence).

  The Location Name may consist of 18 characters (maximum).

- **MAC Address / Device ID:** Enter the MAC Address / Device ID of the extensions.

Similarly you can configure Location Name and MAC Address / Device ID in location 2 and location 3, if required.

You can also modify the CSV File by editing or deleting the values of the parameters as per your requirement.

- *Authentication ID is used by the system for user authentication of the SIP messages received from the extensions. This parameter is not listed in the CSV File and is automatically configured by the system in the SIP Extension Settings Page, when you upload the CSV File. The system will configure the **Authentication ID** same as the **Extension Number** configured in the CSV File for the extensions.*

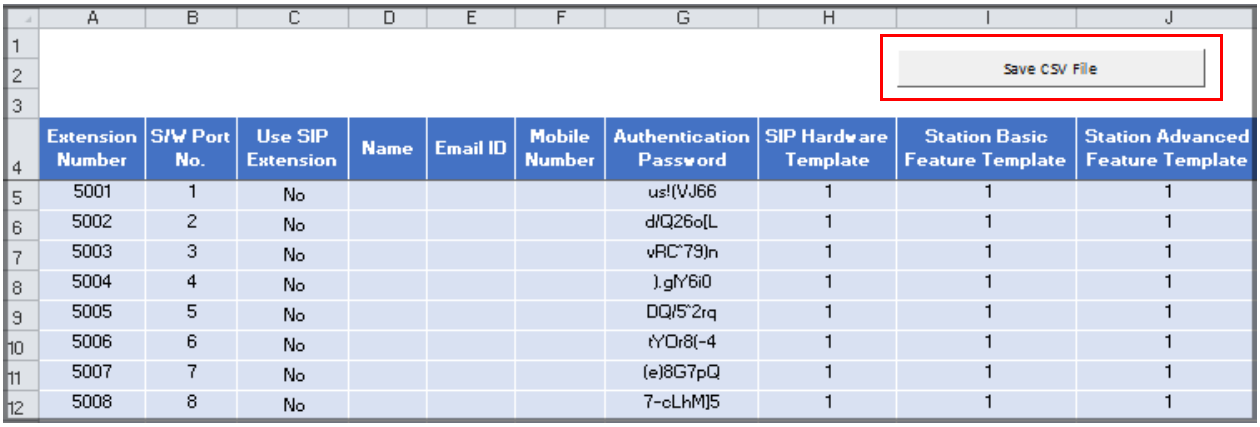
  You can change the Authentication ID, if required. The Authentication ID must be unique for each SIP extension. It may be a string of maximum 6 alphanumeric characters. All ASCII characters except < > and " (double quote) are allowed.

- *Corresponding to the Authentication ID, the system generates a random **Authentication Password** and assign to all the extensions listed in th CSV File. You can edit the authentication password for the extensions, if required. Make sure you note down or copy the Authentication Password in a confidential file.*

  All the other parameters that are not listed in the CSV File but are present in the SIP Extension Settings Page will contain its default value. You can modify the values of these parameters, after the extensions are configured in the system. You can also use the copy button present at the bottom of the SIP Extension Settings Page to replicate the changes made in the configuration of an extension to all the other extensions configured in the system. To know more, refer to "Copy Parameter Values".

Exporting the CSV File

After modifying the CSV File, you can now export the file. To do so,



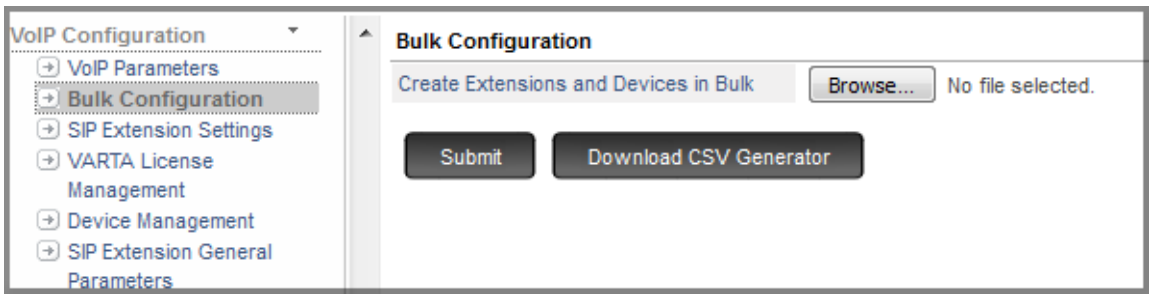
Extension Number	S/W Port No.	Use SIP Extension	Name	Email ID	Mobile Number	Authentication Password	SIP Hardware Template	Station Basic Feature Template	Station Advanced Feature Template
5001	1	No				us!(VJ66	1	1	1
5002	2	No				d/Q26o[L	1	1	1
5003	3	No				vRC^79)n	1	1	1
5004	4	No				).gY6i0	1	1	1
5005	5	No				DQ/5^2rq	1	1	1
5006	6	No				tY^Dr8(-4	1	1	1
5007	7	No				(e)8G7pQ	1	1	1
5008	8	No				7-cLhM]5	1	1	1

- Click the **Save CSV File** button present at the top of the CSV File.
- The CSV File with the name format **CSV-Exported-File-DD-Mmm-YYYY HH-MM-SS**, where **DD-Mmm-YYYY signifies** the current date and **HH-MM-SS** signifies the current time, will be exported on your desktop.

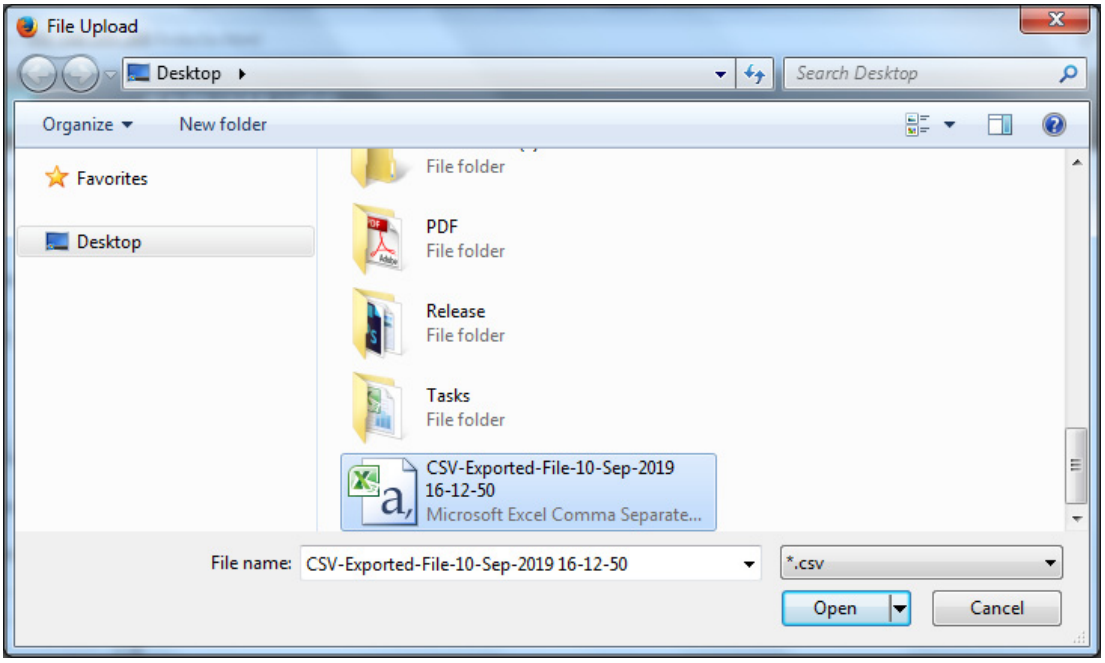
Uploading the CSV File

After you have exported the CSV File, you now have to upload the CSV File to create the extensions. To do so,

- Login as System Engineer.
- Under **VoIP Configuration**, click **Bulk configuration**.

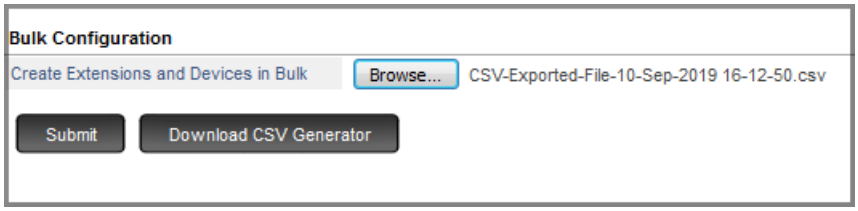


- Click the **Browse** button to select the CSV File from the local disk on the computer.



- After selecting the required CSV File, click **Open**.

  The system displays the name of the CSV File.



- Click **Submit** to upload the CSV File.

After you upload the CSV File, the following statuses will be displayed:

- **Successful:** The total number of extensions that are successful configured in the server.
- **Failed**: The total number of extensions that are not configured in the server. The list of extensions that are not configured are also displayed.

- *Consider extension 5001 is already configured at a particular location in the system, say location 1 of S/W Port No 1. Another extension 5002 with different set of configuration is listed at the same location, that is, location 1 of S/W Port No 1 in the CSV File, then in this case, when the CSV file is uploaded, the configuration of the extension 5001 will be overwritten with the configuration of the extension 5002.*
- *In case, you keep the Extension Number for any extension listed in the CSV File as blank, then, the system will not create this extension. Moreover, the system will consider it as an end of the list and any extension/s listed after this blank Extension Number will not be created.*
- *The total number of extensions the system creates will depend on the number of extensions configured in System Pre-requisites. For an instance, if you have configured 100 as the **SIP Extensions** in **System Pre-requisites** and upload a CSV File with a configuration of 500 extensions, then the system will create only the first 100 extensions.*
- *Make sure the SIP ID and MAC/ Device ID is unique for all the extensions configured in the system. In case, the SIP ID or MAC /Device ID listed in the CSV File conflicts with the SIP ID or MAC /Device ID of an extension already configured in the system, then in this case, the extension with the duplicate SIP ID or MAC/ Device ID will not be created.*
- *It may take about 3 minutes to create all the 2000 extensions in the system.*
- *The system will not create an extension listed in the CSV File, if the Extension Number configured for this extension in the CSV File matches with any of the access codes configured in the system. To know the access codes that are configured in the system, refer to "Access Codes".*
- *Make sure you do not leave the mandatory parameters listed in the CSV Generator Template as blank, otherwise the system will not create these extensions. All the parameters listed in the CSV Generator Template are mandatory for creating the extensions.*

To configure rest of the parameters which are not mentioned in the CSV File, refer to "Configuring SIP Extension Settings as per the Extended Phone Type".

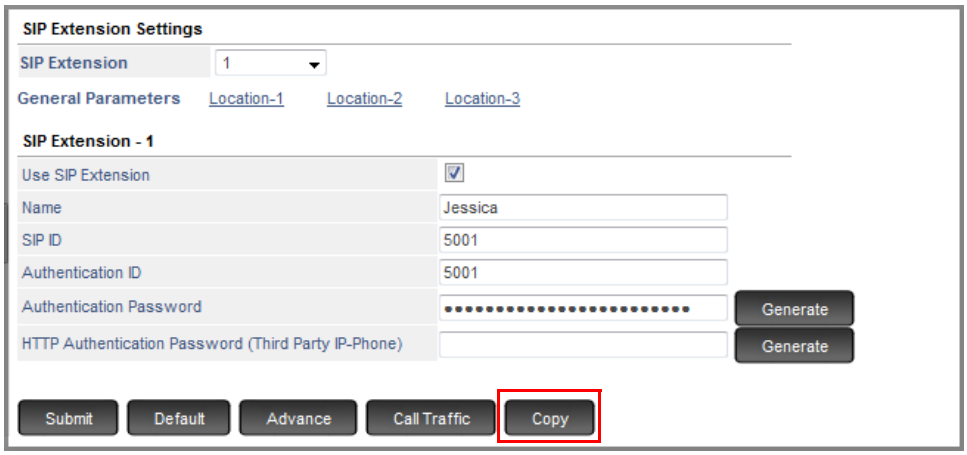
Copy Parameter Values

Once you have created the extensions, you can now configure the parameters which are not mentioned in the CSV File as per your requirement.

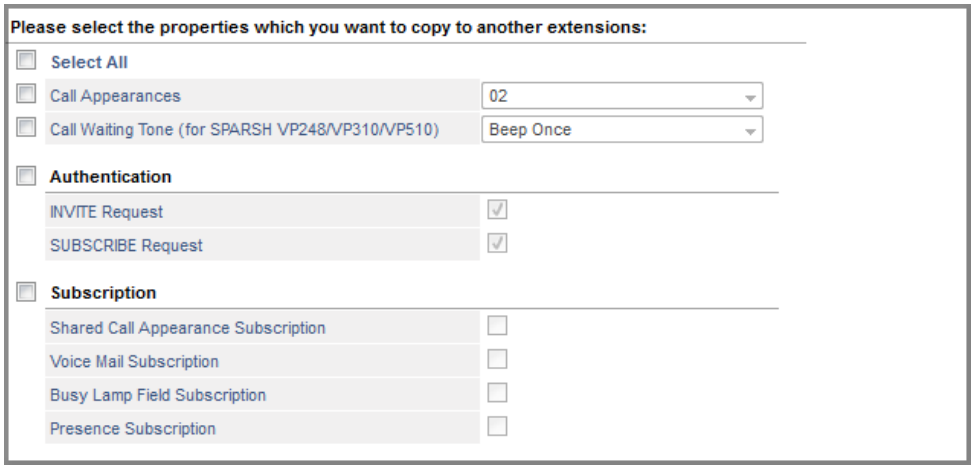
After you have configured the required parameters for an extension, you can now copy the configuration of this extension to all the other extensions by simply using the **Copy** button present at the bottom of the SIP Extension Settings page.

For an instance, you can configure the parameters like *DSS Key Settings, Quality of Service (QoS)* for an extension and copy the configuration of this extension to other extensions. To do so,

- Click **Copy**.



- The Copy page opens in another tab.

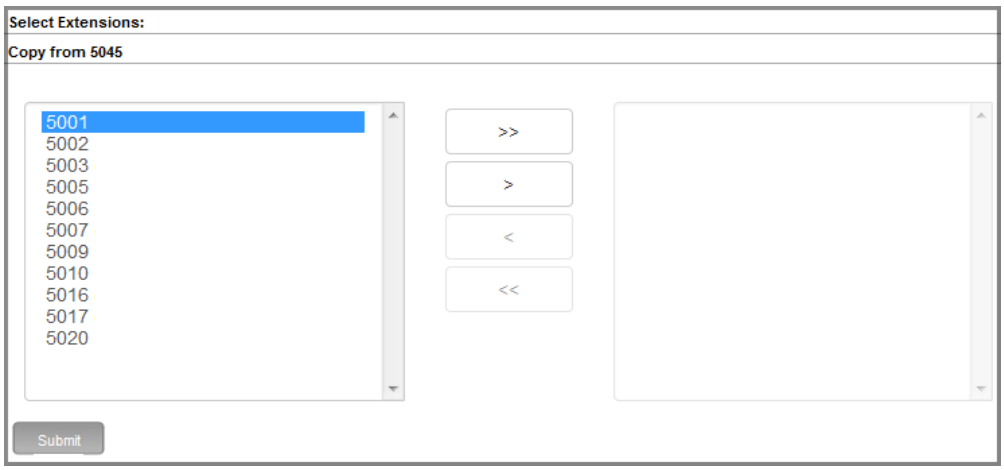


- Select the check boxes against the parameters you want to copy to other extensions. If you want to copy all the parameters to the other extensions, select the check box **Select All**.

If you wish to Copy only the configurations from a location to all other locations where the same IP Phone is connected, make sure you clear the Device Type check box.

If you wish to copy the Device Type as well as all the configurations from a location to all the SIP Extensions, make sure all the check boxes are selected.

- Click **Next**. The Select Extensions window opens.



- The number and/or name of the current extension is displayed in **Copy from**.
  The extensions to which you can copy the current extension settings are displayed on the left side box arranged sequentially in the increasing order.

  - **To select a extension from the List,**

    - Select the desired extension and click **>** button. The selected extension will appear on the right side box.

  - **To select multiple extensions from the List,**

    - Select the desired extensions and click **>** button. The selected extensions will appear on the right side box.

  - **To select all the extensions from the List,**

    - Click **>>** button. All the extensions will appear on the right side box.

  You can also de-select the selected extensions, if required.

  - **To de-select a extension from the List,**

    - Select the desired extension and click **<** button. The selected extension will appear on the left side box.

  - **To de-select multiple extensions from the List,**

    - Select the desired extensions and click **<** button. The selected extensions will appear on the left side box.

  - **To select all the extensions from the List,**

    - Click **<<** button. All the extensions will appear on the left side box.

- Click **Submit**. The settings of the current extension is copied to all the selected extensions.

Auto Sign-In Parameters

The system supports automatic configuration and registration of Mobile Clients — VARTA ADR100, VARTA AMP100 applications with the Server using Auto Sign-In.

Auto Sign-In enables Mobile Clients — VARTA ADR100, VARTA AMP100 applications — to configure and register with the server automatically at a click of a button.

For this you must:

- Configure the **Auto Sign-In Parameters**. For details, refer “Configuring Auto-Sign-In Parameters”
- Configure the **General Parameters** in **SIP Extensions Settings**. For details, refer “Configuring SIP Extension Settings using Jeeves”
- Make sure you send the **Auto Sign-In Configuration Mail**. For details, refer “Configuring SIP Extension Settings using Jeeves”

You can also view the status of Auto Sign-In Email in “Viewing SIP Extension Status”

How it Works

After you configure the required Auto Sign-In parameters and have sent the Auto Sign-In Mail to the Mobile Client users, they need to follow the instructions given below:

- The Auto Sign-In mail has an attachment that contains the necessary configuration details. The Mobile Client user must open the attachment in the Auto Sign-In mail using the VARTA ADR100 or VARTA AMP100 application. For more information refer to the respective User Guides.

- The Server will receive the request and process it. The client will get configured and registered automatically at any free Location1, 2 or 3 in SIP Extension Settings. If none of the Locations are free the request will not be served.

- You can check the Registration status in “Viewing SIP Extension Status”.

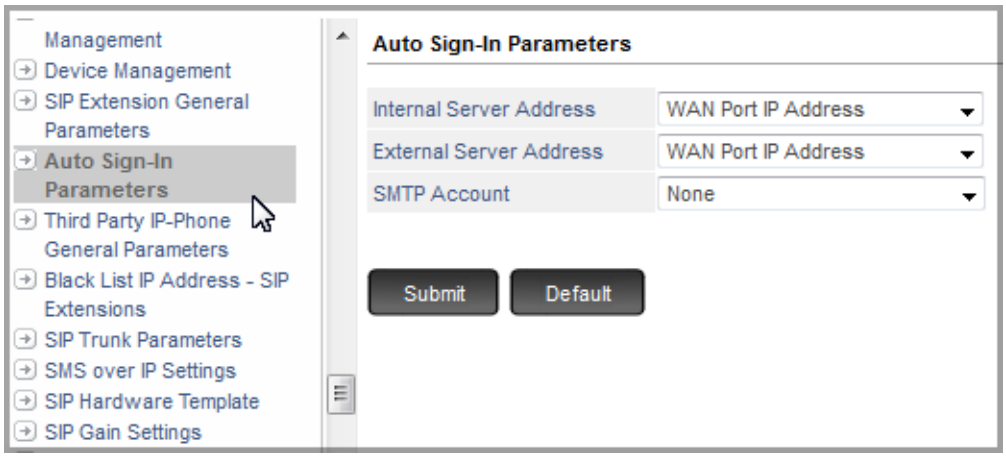
Configuring Auto-Sign-In Parameters

The information you configure in Auto Sign-In Parameters will be sent in the mail to the Mobile Client user, when you click the **Send Auto Sign-In Configuration Mail** button.

- Log into Jeeves as System Engineer.

- Under **Configuration**, click **VoIP Configuration**.

- Click **Auto Sign-In Parameters**.



- Select the appropriate **Internal Server Address** to register the application with the SIP Registrar of SARVAM UCS within a private network. Select the appropriate option as per your installation scenario:
  - If you want the application to register using the WAN network, select **Use WAN Port IP Address** as the Internal Server Address.
  - If you want the application to register using the LAN network, select **Use LAN Port IP Address** as Internal Server Address.
  - If Dynamic DNS is configured in the Network Parameters, select **Use Dynamic DNS Host Name** as Internal Server Address.
  - If you do not want to configure the Internal Server Address, select **Don't Send**.

    By default, Use WAN Port IP Address is selected as the Internal Server Address.
- Select the appropriate **External Server Address** to register the application with the SIP Registrar of SARVAM UCS from a public network. Select the option according to your installation scenario:
  - If you want the application to register using the WAN network, select **Use WAN Port IP Address** as External Server Address.
  - If the application is connected in the Public Network and SARVAM UCS is located behind a Router, or behind a NAT Router and STUN is programmed, select **Use Router/STUN's IP Address** as External Server Address.

    Make sure you configure either the **Router's Public IP Address** or **Simple Traversal of UDPs through NATs (STUN)** in Network Parameters. For details, see "Configuring Network Parameters".
  - If Dynamic DNS is configured in the Network Parameters, select **Use Dynamic DNS Host Name** as External Server Address.
  - If you do not want to configure the External Server Address, select **Don't Send**.

    By default, Use WAN Port IP Address is selected as the External Server Address.
- Select the **SMTP Account**[164] though which you want the email to be sent.

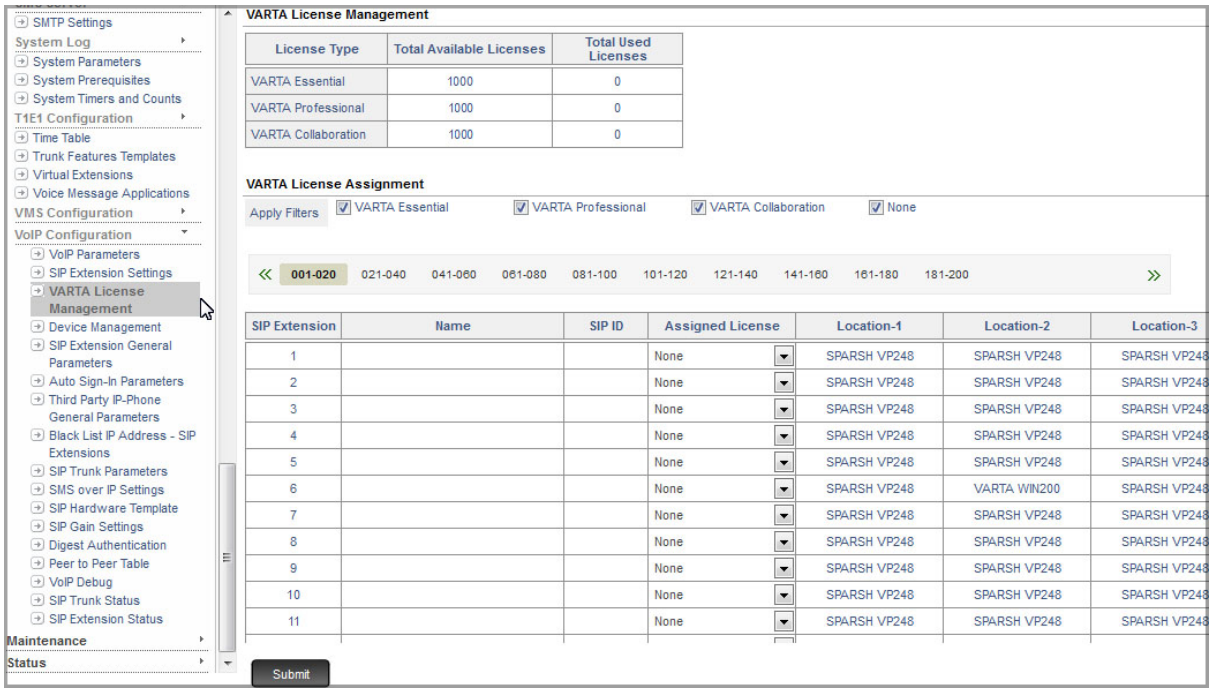
*If you select **Don't Send** in both Internal as well as External Server Address, the server will send the Auto Sign-In mail but the VARTA Mobile Clients will not get registered.*

164. Make sure that the SMTP settings are configured correctly.For more information, refer "SMTP Settings".

VARTA License Management

To view the VARTA License Status and to assign licenses to SIP Extensions,

- Log into Jeeves as System Engineer.
- Under **Configuration**, click **VoIP Configuration**.
- Click **VARTA License Management.**



When Demo Mode is activated, VARTA Collaboration Users license is by default assigned to all SIP Extensions internally. You do not need to configure it manually.

VARTA License Management

The following information will be displayed for the license you activated (VARTA Essential/Professional/ Collaboration).

- **License Type:** This displays the name of the licenses — Essential, Professional or Collaboration.
- **Total Available Licenses:** This displays the total number of licenses activated.
- **Total Used Licenses:** This displays the total number of VARTA users registered as SIP extensions.

VARTA License Assignment

After registering and configuring MATRIX VARTA WIN200/VARTA ADR100/VARTA AMP100 as a SIP Extension, you must select the desired license in the **Assigned License** field below. You can also filter the SIP Extensions as per the license assigned.

- **Apply Filters:** By default you can view all the SIP Extensions, as all the filters are enabled.

- Clear the **VARTA Essential** check box, if you do not want to view the SIP Extensions that are assigned this license.
- Clear the **VARTA Professional** check box, if you do not want to view the SIP Extensions that are assigned this license.

- Clear the **VARTA Collaboration** check box, if you do not want to view the SIP Extensions that are assigned this license.

- Clear the **None** check box, if you do not want to view the SIP Extensions that are not assigned any license.

- **SIP Extension:** This displays the SIP Extension Number with which you can register the VARTA UC Clients.

- **SIP ID:** This displays the SIP ID assigned to the SIP Extension.

- **Assigned License:** Select the license you wish to assign to the SIP Extension and click **Submit**.

- **Location 1, 2, 3:** This displays the Device Type selected on the SIP Extension Location 1, 2 and 3.

Configuring Standard SIP Phones

You can connect any of the following as SIP Extensions of the SARVAM UCS:

- Matrix SPARSH VP248
- Matrix SPARSH VP110
- Matrix SPARSH VP710
- Matrix SPARSH VP210
- Matrix SPARSH VP510
- Any Standard SIP Phone
- Any SIP-enabled device including PC based Soft-phone
- Analog Phone Adapter

For detailed product information and operational instructions, refer to the product documentation supplied with the Standard SIP Phone/device.

SARVAM UCS supports two separate methods of Provisioning the Standard SIP Phones. These two methods are:

- **Manual Provisioning:** In Manual Provisioning, the user must configure the required parameters of the Standard SIP Phone manually. So, it is not a simple plug-and-play solution for mass deployment, as it requires intervention of authorized technical personnel for phone configuration. To configure Standard SIP Phones using this method, see “Configuring Standard SIP Phones using Manual Provisioning”.

  Auto Provisioning: In Auto Provisioning, the Standard SIP Phone gets configured automatically by retrieving the required configuration file from the SARVAM UCS. The configuration file contains pre-programmed values of necessary parameters required by the Standard SIP Phone. Thus, it eliminates the necessity of manually configuring the Standard SIP Phone parameters. When the Standard SIP Phone starts, it then gets configured automatically. Here, SARVAM UCS acts as the Auto Provisioning Server by providing the configuration file to the Standard SIP Phones. Auto Provisioning enables mass deployment of Standard SIP Phones and provides a plug-and-play solution for them. To configure Standard SIP Phones using this method, see “Configuring Standard SIP Phones using Auto Provisioning”.

- *Auto Provisioning for Matrix SPARSH Phones and Third Party SIP Phones is supported only through the HTTP Port.*

- *For Auto Provisioning, you must configure the Standard SIP phones at **Location1** only. You will have to manually configure the Standard SIP phones at Location 2 and 3.*

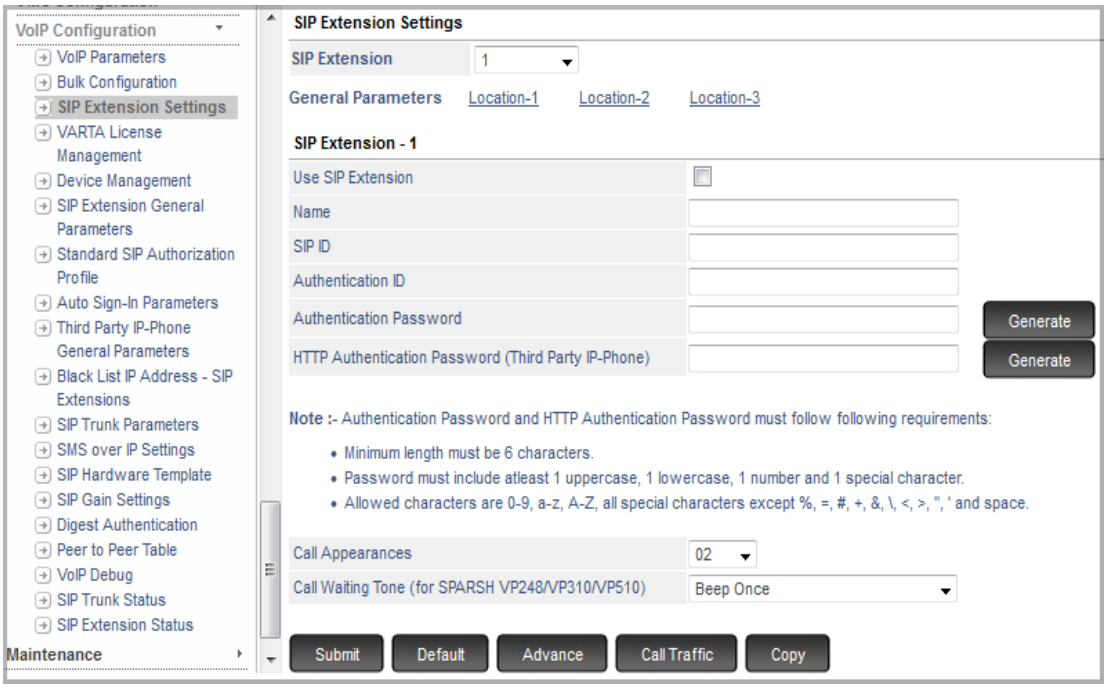
- *If you have already configured the phones at Location 2 and 3 and you upgrade the firmware, the phones will not function as Auto Provisioning is not supported at these locations. You will have to configure these phones again manually.*

To be able to use Standard SIP Phones/Devices as SIP Extensions, you must configure the following:

- SIP Extension General Parameters, see “Configuring SIP Extension General Parameters”.
- SIP Extension Settings, see “Configuring SIP Extension Settings using Jeeves”.
- Standard SIP Authorization Profile, see “Standard SIP Authorization Profile”.
- Voice Mail Settings, if you want to provide mailbox to the extensions. See “Extension Voice Mail Settings”.

Configuring SIP Extension Settings using Jeeves

- Log into Jeeves as System Engineer.
- Under **Configuration**, click **VoIP Configuration**.
- Click **SIP Extension Settings**.



The page of SIP Extension 001 opens.

- You may select the **SIP Extension** number you want to configure.

For SARVAM UCS upto 999 SIP Extensions can be registered with the system. SARVAM UCS supports IPv4 and IPv6 Addresses for registering Standard SIP Phones.

- Select the **Use SIP Extension** check box to enable the SIP extension. Default: disabled.
- In the **Name** field, enter a name for the SIP Extension, which may be the name of the person who will use the SIP Extension or the name of a Department. The name you enter here will be displayed as the Caller ID of the SIP Extension on the remote user's phone, when the SIP Extension user makes calls.

  The name may consist of a maximum of 18 alphanumeric characters.

If no name is assigned to the SIP Extension, the system will display the name received in the INVITE message from the SIP Extension user when making outgoing calls.

- Enter the **SIP ID** for the extension. The SIP ID is necessary for registering the SIP Extension with the Registrar of the system. It is the number with which you can call the SIP Extension. Any extension user of the system can call a SIP Extension by dialing the SIP ID assigned to the SIP Extension. SIP ID of each SIP Extension must be a unique number string of a maximum of 6 digits. Any combination of digits from 0 to 9 and the characters * and # are allowed.

  You cannot assign the same SIP ID to more than one extension.

By default, the SIP IDs are Blank.

The SIP ID will be set to default value (blank), when you restore the default settings of the system.

- In **Authentication ID**, enter the number which you want the system to use for user authentication of the SIP messages received from the SIP Extension. You cannot keep this field blank and it must be unique for each SIP extension. The number may be a string of maximum 6 digits. All ASCII characters except < > and “ (double quote) are allowed.Default: Blank.

Make sure the User ID configured in “Digest Authentication” does not conflict with the Authentication ID configured above.

- In **Authentication Password**, enter the password manually or click **Generate** to automatically generate a unique password. This password will be used by the system to authenticate the SIP messages received from the SIP Extension. To avoid unauthorized access, we recommend you to change the password regularly. Make sure it is strong and is kept confidential. When you enter the password manually, the password must:

  - be of minimum 6 characters and can be a maximum of 12 characters.
  - include atleast one upper-case, one lower-case, one number and one special character.
  - all ASCII characters (except Percentage **%**, Hash **#,** Equal to **=,** Plus **+**, And **&**, Backslash **\,** Less than **<,** Greater than **>,** Apostrophe **’,** Double Quote **"** and **Space**) are allowed.
    Default: Blank.

  To provide additional security, when the Authentication fails 10 times consecutively due to wrong Authentication ID / Authentication Password, the system will blacklist the IP Address and Port for registration of this SIP Extension. However, you can remove the IP Address from the Blacklist IP Address list. See “Black List IP Address - SIP Extensions” for more details. This activity will be logged in the “System Activity Log” as well as “Simple Network Management Protocol (SNMP)”.

Make sure you note down or copy the Authentication Password in a confidential file.

- In **HTTP Authentication Password** (Third Party IP-Phone), enter the password manually or click **Generate** to automatically generate a unique password.This password will be used by the system to authenticate the Standard SIP Phone connected to the system. SARVAM UCS validates the Standard SIP Phone on the basis of the SIP ID and HTTP Authentication Password (Third Party IP-Phone) and provides configuration on validation.

  To avoid unauthorized access, we recommend you to change the password regularly. Make sure it is strong and is kept confidential. The password must:

  - be of minimum 6 characters and can be a maximum of 12 characters.
  - include atleast one upper-case, one lower-case, one number and one special character.
  - all ASCII characters (except Percentage **%**, Hash **#,** Equal to **=,** Plus **+**, And **&**, Backslash **\,** Less than **<,** Greater than **>,** Apostrophe **’,** Double Quote **"** and **Space**) are allowed.
    Default: Blank.

  To provide additional security, when the HTTP Authentication fails 5 times consecutively within 10 minutes due to wrong Authentication ID / Authentication Password, the system will block the IP Address for configuration of this Standard SIP Phone. However, you can register again after 10 minutes. This activity will be logged in the “System Activity Log” as well as “Simple Network Management Protocol (SNMP)”.

You are recommended to note down or copy the Authentication Password and HTTP Authentication Password in a confidential file.

Make sure, you have configured SIP ID as HTTP User Name and HTTP Authentication Password as HTTP Password in the Standard SIP Phone.

- In **Call Appearances**, define the maximum number[165] of simultaneous calls that the SIP Extension should be allowed to make/receive. You can set up to 10 call appearances for a SIP Extension. Default: 2.

  When Call Appearance is set to 2, the SIP Extension can make/receive 2 calls at a time.

***Auto Sign-In parameters — Send Configuration Mail and Mail Status,** are applicable only for Mobile Clients — VARTA ADR100, VARTA AMP100 applications.*

*The **Send Configuration Mail** button will appear only after you have enabled the SIP Extension and configured the SIP ID, Authentication ID and Password.*

- Under **Authentication**, enable Authentication of any or all of the following SIP Message Options by selecting the respective check boxes:
  - **INVITE Request**
  - **SUBSCRIBE Request**

  By default, the SIP Message Options INVITE and SUBSCRIBE are enabled.

Make sure that the Authentication ID for the SIP Extension has been programmed.

- If you are going to register this SIP Extension with the same SIP ID at more than one location[166], you may enable the **Shared Call Appearance Subscription** check box on this SIP Extension. Default: Disabled.

  Shared Call Appearance provides notification on call states to all the Standard SIP Phones with the same SIP ID at different locations. To know more about this feature, see “Shared Call Appearance”.

- To provide voice mail facility to the SIP Extension, select the **Voice Mail Subscription** check box. Default: disabled.

- To allow the SIP Extension to monitor the status of another extension or Trunk, select the **Busy Lamp Field**[167] **Subscription** check box. Default: disabled. See “Busy Lamp Field for Trunks” to know more.

When extension's state is changed from Ringing (early state as defined in BLF) to Mature (confirm state, as defined in BLF) state, because of implementation of SARVAM UCS, it will send 'Terminate' state while moving from ringing to mature state. The interpretation of terminate message will vary from terminal to terminal.

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165. For the calls that are routed through the CPU, the number of Vocoder channels that will be supported would be as per the license you purchase.

166. SARVAM UCS allows you to register SIP Extension with the same SIP ID at three different locations.

167. Busy Lamp Field (BLF), a typical feature supported by the system and Key Telephone Systems, is also supported on SIP Extensions.

In the system and Key Telephone Systems, this feature is typically used by the Operator to monitor the status of another extension, that is, whether it is available, ringing or busy. The status of the other extensions is indicated on the special function keys programmed on the Operator's console. This helps the Operator decide whether to place the call, or transfer the call to that extension, or pick up the call ringing on that extension. With BLF Subscription enabled on the SIP Extension, the user can monitor the status of another extension or trunk.

- To allow the SIP Extension to view the status of other SIP-enabled Terminals, whether they are available or not, select the **Presence Subscription** check box. Default: disabled.

The SIP Extension, for which you have enabled Presence Subscription, will be able to view Presence of only those SIP Extensions which have PUBLISH enabled.

- To allow the SIP Extension to publish presence using the PUBLISH feature supported by the SIP Extension, select the **PUBLISH Enable** check box. Default: disabled.

  By default, **Authentication** for PUBLISH message is enabled. You may disable if you do not want to use Authentication.

You must configure the Authentication ID, if you have enabled both Publish and Authentication.

- For secure conversations over SIP, enable **SRTP Mode**. The SARVAM UCS supports the following options:

  - **Disable:** SARVAM UCS uses normal RTP for transporting the speech packets.

  - **Optional:** SARVAM UCS uses SRTP for transporting the speech packets. If the remote user does not support SRTP, SARVAM UCS will use normal RTP for transporting the speech packets.

    - If you select this option, you must configure the **SRTP Media Type**. You may select **AVP** or **SAVP**. By default, AVP is selected as the SRTP Media Type.

  - **Forced:** SARVAM UCS uses only SRTP (SAVP) for transporting the speech packets. If the remote user does not support SRTP, SARVAM UCS will reject incoming calls from and drop outgoing calls made to such users.

    By default, SRTP Mode is Disabled.

- **Key Templates** are not applicable to Standard SIP Phones registered with SARVAM UCS.

- Assign a **SIP Hardware Template** to the SIP Extension. Default: 01. The “SIP Hardware Template” contains voice quality related features such as Voice Codec selection, Tx and Rx Gains, Echo Cancellation, Jitter Buffer and Fax-over-IP options and related parameters

  There are 32 different templates to choose from. Each template can also be altered to suit your requirement and preferences. By default, Template number 01 assigned to all SIP Extensions as well as to SIP Trunks.

  Check if the values in this template fulfill requirements of the SIP Extension. If Template 01 fulfills the feature requirements, retain Template 01.

  If a different set of SIP hardware features are to be allowed to this SIP Extensions, prepare another template and assign it to this extension. To do this,

  - Under **VoIP Configuration**, click the **SIP Hardware Template** link.
  - Select a Template number, for example 02.
  - Customize Template number 02 and click **Submit**.
  - Return to the **SIP Extension Settings** page.
  - Select the number of the Template you customized, Template 02 in the **SIP Hardware Template** field.

Also see the topic “SIP Hardware Template” to know more about customizing the templates and applying on the SIP Extensions.

- Assign a **Station Basic Feature Template** to the SIP Extension. Default: The “Station Basic Feature Template” has a set of features like Time Table, Class of Service, Toll Control, Operator, Storage of Incoming and Outgoing Calls, Outgoing Trunk Bundle groups. There are 50 different templates to choose from. Each template can also be altered to suit your requirement and preferences.

  If the default Station Basic Feature Template 01 fulfills the feature requirements of the SIP Extension (“Class of Service (COS)”, “Toll Control”, “OG Trunk Bundle Group”, etc.) retain this template, you may also customize this template. If you want to assign a different set of features to this SIP Extension, prepare a different Station Basic Feature Template and apply it to this extension. To do this,

  - Under **Configuration**, click the **Station Basic Feature Template** link.
  - Select a Template number, for example 05.
  - Customize Template number 05 and click **Submit**.
  - Return to the **SIP Extension Settings** page.
  - Select the number of the Template you customized, Template 05, in the **Station Basic Feature Template** field.
  - Click **Submit** to save changes.

  Also, see the topic “Station Basic Feature Template” to know more about customizing the templates and applying on extensions.

- Assign a **Station Advanced Feature Template** to the SIP Extension. Default: Template 01. The “Station Advanced Feature Template” has a set of advanced features for extensions such as Alarm Notification settings, Routing of Incoming Auto Attendant Calls, Call Duration Control, Floor Service, etc. There are 50 different templates to choose from. Each template can also be altered to suit your requirement and preferences.

  Check if the default template fulfills the feature requirements of the SIP Extension by clicking the **Station Advanced Feature Template** link.

  You may retain this template and customize it further, or customize another template if a different set of features are to be allowed to this SIP Extension. To customize/prepare another template,

  - Under **Configuration**, click the **Station Advanced Feature Template** link.
  - Select the Template number, for example 02, and customize this template.
  - Click **Submit** and return to the 'SIP Extension Settings' page.
  - In the **Station Advanced Feature Template** field, select the number of the template you customized.
  - Click **Submit** to save changes.

  Also see the topic “Station Advanced Feature Template” for instructions on customizing these templates and applying them on the extensions.

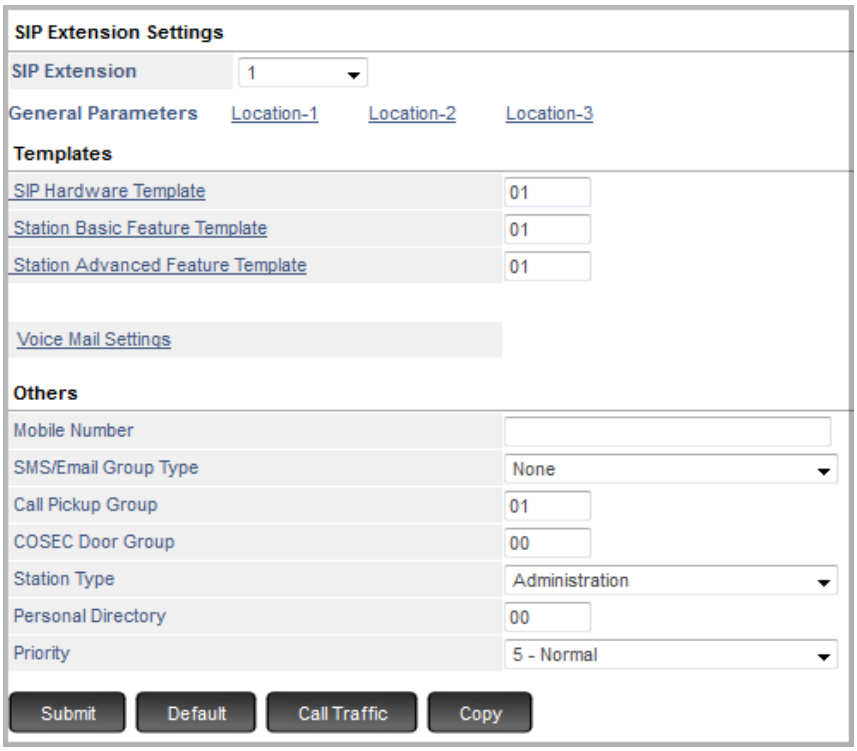
- In **Voice Mail Settings**, click the Voice Mail Settings link. The respective Extension Voice Mail Settings window will open. You may edit the parameters. For details, see “Extension Voice Mail Settings”.

The Voice Mail Settings link will be visible only if you have configured the respective SIP ID.

Click **Close** to close the window.

Advanced Configuration Parameters

- If you want to provide other features like Personal Directory, Priority, or assign a Station Type to the SIP Extension, click the **Advanced** button at the bottom of the page.



- Enter the **Mobile Number** of the extension user you wish to store. The Number can be a maximum of 16 digits.

- You can assign the extension user to a Group. Select the desired **SMS/Email Group Type** from the list.The system clubs together extension users assigned the same Group. Default: None. For details, see “SMS/Email Group”.

- Assign the SIP Extension to a **Call Pick-up Group**, if required.

  Call Pick Up allows the SIP Extension to 'pick up' (answer) calls ringing on any other extension, by dialing a feature code, without physically going to the ringing extension. It also allows incoming calls for the SIP Extension to be answered by the other extensions assigned the same Call Pick-Up group.

  For this to work, both the ringing extension and the extension picking up the call must be in the same 'Call Pick Up Group'. Refer “Call Pick Up” for instructions on how to create groups. You can create as many as 99 groups numbered from 01 to 99.

  Enter the number of the Call Pick-Up Group you created for this SIP Extension in this field.

- Select the **Call Pick-up Notification (Only for SPARSH VP510)** check box, if you want the Call Pick-up Notification to be displayed on the phone LCD. The Call Pick-up Notification will be displayed for internal as well as external calls. The notification will be displayed on SPARSH VP510 only if the ringing extension is in the same Call Pick-up Group. The notification will be displayed only when SPARSH VP510 is in idle or dial state. The notification will display the name/number of the Caller along with the name/number of the Called. For example, if the name and number of the Caller is ABC, 2001 and the name and number of

Called is XYZ, 1001, the LCD notification will be displayed as ABC -> XYZ. Along with the LCD display the phone back-light will also be lit.

For this to work, make sure you have enabled Call Pick-up in COS as well as assigned a Call Pick-up Group to the extension. Refer to "Call Pick Up" and "Class of Service (COS)". Call Pick-up Notifications will be displayed for DKP, SLT as well as SIP Extensions and for calls landing through CO, SIP as well as T1E1 Trunks. For details of the Notification, refer to the EON510_SPARSH VP510 V2 User Guide.

- You must assign the extension user to a **COSEC Door Group** for COSEC Integration. The users in the same group must be assigned the same group. You can create as many as 50 groups numbered from 00 to 50. Users who are assigned COSEC Door Group '00' are not a part of any group. See "COSEC Integration" for more information.

- If this is an Operator extension and you want the system to play beeps during a conference to the participants, to indicate the presence or absence of the Operator, select the **Station Type** as **Assistant**.

  If you are using the system in the *Hotel Mode*, select the **Station Type** for the SIP Extension as **Administration/Assistant** or **Guest**. The system will consider the options Administrator and Assistant as same.

- You may assign a **Personal Directory** number to the SIP Extension. Default: 00.

  A Personal Directory is a list of 25 frequently dialed numbers, each of which are stored by Index number (location code), Name and Trunk Access Codes ("Out Going Trunk Bundle Group Index"). The Personal Directory is necessary for using the features "Abbreviated Dialing" and "Dial By Name".

  When a Personal Directory is assigned to a SIP Extension, make sure you also configure this directory. The Personal Directory can be programmed by the SIP Extension users and by the System Engineer. Refer the topic "Abbreviated Dialing" for instructions on programming the Personal Directory.
  If Personal Directory is not to be assigned, enter 00 in this field.

- Select a **Priority** Level for the SIP Extension from 1 to 9. Default; 5-Normal.

  Each extension of the SARVAM UCS is assigned a Priority Level starting from 1, 2, 3... to 9, with '1' being lowest Priority and '9' being highest Priority. Whenever an extension (phone) with higher priority calls an extension with lower priority, a triple ring is placed on the called extension. To know more, read the feature description "Priority".

  If this SIP extension is assigned to Operator, you may want to set a higher priority for this extension.

- Click **Submit** to save your SIP Extension Settings.

Configuring Standard SIP Phones using Manual Provisioning

To be able to use Standard SIP Phones/devices with SARVAM UCS, you must configure the following:

- Configure the **SIP Extension Settings**. For details, see "Configuring SIP Extension Settings using Jeeves".[168]

168. Some of the parameters may not be applicable depending on the Standard SIP Phone you have connected to the system.

- Configure specific parameters (for example, SIP ID/User Name, Authentication ID, Password, Server Address/Domain Name etc.) in the Standard SIP Phone/device which are required to register it with system. For more information, refer the product documentation supplied with the Standard SIP Phone/ device you want to use.

Configuring Standard SIP Phones using Auto Provisioning

SARVAM UCS supports the following third party Standard SIP Phones for Auto Provisioning-

- Panasonic UTG200B
- Grandstream GXP110x
- Grandstream GXP2200
- Grandstream GXP21xx/116x/14xx
- Grandstream GXV3140/3175
- Yealink SIP-T19P/SIP-T21P/T22P/T26P/T28P/T4X
- Yealink SIP-T20P
- Yealink SIP-T3XG
- Cisco SPA50xG/51xG SIP Phone
- Cisco SPA525G SIP Phone
- Polycom IP Phone
- Snom IP Phone
- Htek 802
- Any Standard SIP Phone

Connecting the Standard SIP Phone

To be able to configure and register the Standard SIP Phone with SARVAM UCS using Auto Provisioning,

- Connect and reboot the Standard SIP Phone.
- Make sure that **DHCP** is selected as the *Connection Type* in the Standard SIP Phone.
- For Auto Provisioning, you must configure the Standard SIP phones at Location1 only.
- The phone will automatically fetch the configuration file(s) from SARVAM UCS and will get registered.

Using any third party DHCP Server in your LAN

You can use any third party DHCP Server in your LAN for assigning the ‘Auto Provisioning Server Address’ and ‘Server Port’ to the Standard SIP Phones.

- Make sure that the third party DHCP Server and your Standard SIP Phone, both are connected in the same subnet as that of the **LAN** Port of SARVAM UCS.
- Select **DHCP Option 66**, and configure **Data** type in that third party DHCP Server in the following format - **http://IP Address:SPARSH Port.**

If you are using Auto Provisioning and if the Standard SIP Phone has multiple SIP accounts, it is recommended to keep the first SIP account in default settings. After Auto Provisioning, the old configuration, if present for this account will be deleted automatically and it will get registered with the SARVAM UCS.

Configuring Panasonic Standard SIP Phones

You are recommended to complete the following steps before connecting **Panasonic UTG200B**:

- Decide the physical location of the Standard SIP Phone.
- Configure the settings in the third party DHCP Server. For instructions, see “Using any third party DHCP Server in your LAN”.
- Configure the **SIP Extension Settings** in SARVAM UCS. For details, see “Configuring SIP Extension Settings using Jeeves”[169]
- Configure the device specific settings applicable to your **Panasonic UTG200B** at any one of the Location1, 2 or 3 on the SIP Extensions page. To do so,
  - Under **Configuration**, click **VoIP Configuration**.
  - Click **SIP Extension Settings** link.
  - Click **Location 1** and configure the following parameters.

Parameter		Description/Action to be Taken	Applicable Standard SIP Phone Variant
Enable Device		Select the Enable Device check box.	Panasonic UTG200B
Location Name		Configure the Location Name to identify the phone. The Location Name can be a maximum of 18 characters. Default: Blank.	Panasonic UTG200B
Device Type		Select Device Type as Panasonic UTG200B.	Panasonic UTG200B
MAC Address		Enter the MAC Address of the Panasonic phone to be connected at this location. Default: Blank. SARVAM UCS validates the phone on the basis of the MAC Address and provides configuration on validation.	Panasonic UTG200B

169. *Some of the parameters may not be applicable depending on the Standard SIP Phone you have connected. Please refer the specific Standard SIP Phone manufacturer’s documentation for more details.*

Authenticate HTTP Provisioning request		Select the Authenticate HTTP Provisioning request check box to authenticate HTTP request for security reason. Default: Disabled. If you are using the Standard SIP Phone from the Public IP Address, we recommend you to enable this option. SARVAM UCS validates the Standard SIP Phone on the basis of the SIP ID and HTTP Authentication Password and provides configuration on validation. Make sure, you have configured SIP ID as HTTP User Name and HTTP Authentication Password as HTTP Password in the Standard SIP Phone.	Panasonic UTG200B
Registrar Server Address		Select the appropriate Registrar Server Address according to your installation scenario[a] to register the Standard SIP Phone with the SIP Registrar of SARVAM UCS. Default: Use WAN Port IP Address.	Panasonic UTG200B
Standard SIP Authorization Profile		Select the desired Standard SIP Authorization Profile from the list of profiles. Default: Panasonic	Panasonic UTG200B
Language		Displays the Standard SIP Phone language and it is non-editable.	Panasonic UTG200B
User Password		Configure the User Password[b] of the Standard SIP Phone. It can be a minimum of 6 characters to a maximum of 16 characters. Default: userpass	Panasonic UTG200B
Admin Password		Configure the Admin Password[c] of the Standard SIP Phone. It can be a minimum of 6 characters to a maximum of 16 characters. Default: adminpass	Panasonic UTG200B
Time Zone		Select the Time Zone of the region/country where the Standard SIP Phone is installed. Default: GMT + 05:30	Panasonic UTG200B
Daylight Saving Time	Enable DST	If DST is applicable, select the Enable DST check box. A list of DST parameters appear. Configure them as per your requirement. Default: Disabled.	Panasonic UTG200B
	DST Offset (min)	Configure the DST Offset in minutes. Valid Range: 0 to 720 minutes. Default: 60 minutes.	Panasonic UTG200B
	Start Day and Time of DST	To configure the time from when DST should be applied in the year, select the Month, Day, Week/Ordinal and Time (hh:mm) from the corresponding list boxes respectively.	Panasonic UTG200B
	End Day and Time of DST	To configure the time when DST should end, select the Month, Day, Week/Ordinal and Time (hh:mm) from the corresponding list boxes respectively.	Panasonic UTG200B
Primary NTP Server		Select the Primary NTP Server Address with which you want to synchronize the Date and Time of the Standard SIP Phone. Default: None. Make sure you configure the desired NTP Server Address/es in the “Third Party IP-Phone General Parameters”.	Panasonic UTG200B

a. If the Standard SIP phone is in the same network (LAN) as SARVAM UCS, select Use LAN Port IP Address as Registrar Server Address.

   If the Standard SIP phone is in the Global Network and SARVAM UCS is connected to Internet over WAN, select Use WAN Port IP Address as Registrar Server Address

   If the Standard SIP Phone is connected in the Global Network and SARVAM UCS is located behind a NAT Router, and STUN is programmed, select Use Router/STUN's IP Address as Registrar Server Address. Make sure the Router's Public IP Address is configured in the Network Parameters.

   If Dynamic DNS is configured in the Network Parameters, select Use Dynamic DNS Host Name as Registrar Server Address.

b. To avoid unauthorized access, we recommend you to change the User Password regularly. Make sure it is strong and is kept confidential.

c. To avoid unauthorized access, we recommend you to change the Admin Password regularly. Make sure it is strong and is kept confidential.

Configuring Grandstream Standard SIP Phones

You are recommended to complete the following steps before connecting any of the **Grandstream** Standard SIP Phones:

- Decide the physical location of the Standard SIP Phone.
- Configure the settings in the third party DHCP Server. For instructions, see "Using any third party DHCP Server in your LAN".
- Configure the **SIP Extension Settings** in SARVAM UCS. For details, see "Configuring SIP Extension Settings using Jeeves"[170]
- Configure the device specific settings applicable to your **Grandstream** at any one of the Location1, 2 or 3 on the SIP Extensions page. To do so,
  - Under **Configuration**, click **VoIP Configuration**.
  - Click **SIP Extension Settings** link.
  - Click **Location 1** and configure the following parameters.

Parameter	Description/Action to be Taken	Applicable Standard SIP Phone Variant
Enable Device	Select the Enable Device check box.	1. Grandstream GXP110x 2. Grandstream GXP2200 3. Grandstream GXP21xx/116x/14xx 4. Grandstream GXV3140/3175
Location Name	Configure the Location Name to identify the Standard SIP Phone. The Location Name can be a maximum of 18 characters. Default: Blank.	1. Grandstream GXP110x 2. Grandstream GXP2200 3. Grandstream GXP21xx/116x/14xx 4. Grandstream GXV3140/3175

170. Some of the parameters may not be applicable depending on the Standard SIP Phone you have connected. Please refer the specific Standard SIP Phone manufacturer's documentation for more details.

Device Type	Select Device Type as any of the desired Grandstream Standard SIP Phone you want to connect.	1. Grandstream GXP110x 2. Grandstream GXP2200 3. Grandstream GXP21xx/116x/ 14xx 4. Grandstream GXV3140/3175
MAC Address	Enter the MAC Address of the Grandstream phone to be connected at this location. Default: Blank. SARVAM UCS validates the phone on the basis of the MAC Address, and provides configuration on validation.	1. Grandstream GXP110x 2. Grandstream GXP2200 3. Grandstream GXP21xx/116x/ 14xx 4. Grandstream GXV3140/3175
Authenticate HTTP Provisioning request	Select the Authenticate HTTP Provisioning request check box to authenticate HTTP request for security reason. Default: Disabled. If you are using the Standard SIP Phone from the Public IP Address, we recommend you to enable this option. SARVAM UCS validates the Standard SIP Phone on the basis of the SIP ID and HTTP Authentication Password and provides configuration on validation. Make sure, you have configured SIP ID as HTTP User Name and HTTP Authentication Password as HTTP Password in the Standard SIP Phone.	1. Grandstream GXP110x 2. Grandstream GXP2200 3. Grandstream GXP21xx/116x/ 14xx 4. Grandstream GXV3140/3175
Registrar Server Address	Select the appropriate Registrar Server Address according to your installation scenario[a] to register the Standard SIP Phone with the SIP Registrar of SARVAM UCS. Default: Use WAN Port IP Address.	1. Grandstream GXP110x 2. Grandstream GXP2200 3. Grandstream GXP21xx/116x/ 14xx 4. Grandstream GXV3140/3175
Standard SIP Authorization Profile	Select the desired Standard SIP Authorization Profile from the list of profiles. Default: Grandstream	1. Grandstream GXP110x 2. Grandstream GXP2200 3. Grandstream GXP21xx/116x/ 14xx 4. Grandstream GXV3140/3175
Language	Select the desired Standard SIP Phone language. Default: English	1. Grandstream GXP110x 2. Grandstream GXP2200 3. Grandstream GXP21xx/116x/ 14xx 4. Grandstream GXV3140/3175
Send Phone Book	Select the Send Phone Book check box to enable downloading of the Phone Book (consisting of Extension and Global Directory Contacts) from the SARVAM UCS.	1. Grandstream GXP2200 2. Grandstream GXP21xx/116x/ 14xx 3. Grandstream GXV3140/3175
Phone Book Download interval (min)	Configure the Phone Book Download interval in minutes. Valid Range: 0 - 720 minutes. Default: 60 minutes	1. Grandstream GXP2200 2. Grandstream GXP21xx/116x/ 14xx 3. Grandstream GXV3140/3175

User Password	Configure the User Password[b] of the Standard SIP Phone. It can be a minimum of 1 character to a maximum of 16 characters. Default: user	1. Grandstream GXP110x 2. Grandstream GXP2200 3. Grandstream GXP21xx/116x/14xx 4. Grandstream GXV3140/3175
Admin Password	Configure the Admin Password[c] of the Standard SIP Phone. It can be a minimum of 1 character to a maximum of 16 characters. Default: admin	1. Grandstream GXP110x 2. Grandstream GXP2200 3. Grandstream GXP21xx/116x/14xx 4. Grandstream GXV3140/3175
Time Zone	Select the Time Zone of the region/country where the Standard SIP Phone is installed. Default: GMT + 05:30	1. Grandstream GXP2200 2. Grandstream GXP21xx/116x/14xx 3. Grandstream GXV3140/3175
Date Display Format	Select the Date Display Format for the Standard SIP Phone. Default: yyyy-mm-dd	1. Grandstream GXP2200 2. Grandstream GXP21xx/116x/14xx 3. Grandstream GXV3140/3175
Time Display Format	Select the Time Display Format for the Standard SIP Phone. Default: 24Hr	1. Grandstream GXP2200 2. Grandstream GXP21xx/116x/14xx 3. Grandstream GXV3140/3175
Primary NTP Server	Select the Primary NTP Server Address with which you want to synchronize the Date and Time of the Standard SIP Phone. Default: None. Make sure you configure the desired NTP Server Address/es in the “Third Party IP-Phone General Parameters”.	1. Grandstream GXP110x 2. Grandstream GXP2200 3. Grandstream GXP21xx/116x/14xx 4. Grandstream GXV3140/3175

a. If the Standard SIP phone is in the same network (LAN) as SARVAM UCS, select Use LAN Port IP Address as Registrar Server Address.

If the Standard SIP phone is in the Global Network and SARVAM UCS is connected to Internet over WAN, select Use WAN Port IP Address as Registrar Server Address

If the Standard SIP Phone is connected in the Global Network and SARVAM UCS is located behind a NAT Router, and STUN is programmed, select Use Router/STUN's IP Address as Registrar Server Address. Make sure the Router’s Public IP Address is configured in the Network Parameters.

If Dynamic DNS is configured in the Network Parameters, select Use Dynamic DNS Host Name as Registrar Server Address.

b. To avoid unauthorized access, we recommend you to change the User Password regularly. Make sure it is strong and is kept confidential.

c. To avoid unauthorized access, we recommend you to change the Admin Password regularly. Make sure it is strong and is kept confidential.

Configuring Yealink Standard SIP Phones

You are recommended to complete the following steps before connecting any of the **Yealink** Standard SIP Phones:

- Decide the physical location of the Standard SIP Phone. It must be connected in the same subnet as that of the LAN Port of SARVAM UCS.

- Configure the settings in the third party DHCP Server. For instructions, see “Using any third party DHCP Server in your LAN”.
- Configure the **SIP Extension Settings** in SARVAM UCS. For details, see “Configuring SIP Extension Settings using Jeeves”[171]
- Configure the device specific settings applicable to your **Yealink** at any one of the Location1, 2 or 3 on the SIP Extensions page. To do so,
  - Under **Configuration**, click **VoIP Configuration**.
  - Click **SIP Extension Settings** link.
  - Click **Location 1** and configure the following parameters.

Parameter		Description/Action to be Taken	Applicable Standard SIP Phone Variant
Enable Device		Select the Enable Device check box.	1. Yealink SIP-T19P/SIP-T21P/T22P/T26P/T28P/T4X 2. Yealink SIP-T20P 3. Yealink SIP-T3XG
Location Name		Configure the Location Name to identify the Standard SIP Phone. The Location Name can be a maximum of 18 characters. Default: Blank.	1. Yealink SIP-T19P/SIP-T21P/T22P/T26P/T28P/T4X 2. Yealink SIP-T20P 3. Yealink SIP-T3XG
Device Type		Select Device Type as any of the desired Yealink Standard SIP Phone you want to connect.	1. Yealink SIP-T19P/SIP-T21P/T22P/T26P/T28P/T4X 2. Yealink SIP-T20P 3. Yealink SIP-T3XG
MAC Address		Enter the MAC Address of the Yealink phone to be connected at this location. Default: Blank. SARVAM UCS validates the phone on the basis of the MAC Address, and provides configuration on validation.	1. Yealink SIP-T19P/SIP-T21P/T22P/T26P/T28P/T4X 2. Yealink SIP-T20P 3. Yealink SIP-T3XG

171. Some of the parameters may not be applicable depending on the Standard SIP Phone you have connected. Please refer the specific Standard SIP Phone manufacturer’s documentation for more details.

Authenticate HTTP Provisioning request		Select the Authenticate HTTP Provisioning request check box to authenticate HTTP request for security reason. Default: Disabled. If you are using the Standard SIP Phone from the Public IP Address, we recommend you to enable this option. SARVAM UCS validates the Standard SIP Phone on the basis of the SIP ID and HTTP Authentication Password and provides configuration on validation. Make sure, you have configured SIP ID as HTTP User Name and HTTP Authentication Password as HTTP Password in the Standard SIP Phone.	1. Yealink SIP-T19P/SIP-T21P/T22P/T26P/T28P/T4X 2. Yealink SIP-T20P 3. Yealink SIP-T3XG
Registrar Server Address		Select the appropriate Registrar Server Address according to your installation scenario[a] to register the Standard SIP Phone with the SIP Registrar of SARVAM UCS. Default: Use WAN Port IP Address.	1. Yealink SIP-T19P/SIP-T21P/T22P/T26P/T28P/T4X 2. Yealink SIP-T20P 3. Yealink SIP-T3XG
Standard SIP Authorization Profile		Select the desired Standard SIP Authorization Profile from the list of profiles. Default: Yealink	1. Yealink SIP-T19P/SIP-T21P/T22P/T26P/T28P/T4X 2. Yealink SIP-T20P 3. Yealink SIP-T3XG
Send Phone Book		Select the Send Phone Book check box to enable downloading of the Phone Book (consisting of Extension and Global Directory Contacts) from the SARVAM UCS. Default: Enabled	1. Yealink SIP-T19P/SIP-T21P/T22P/T26P/T28P/T4X 2. Yealink SIP-T20P 3. Yealink SIP-T3XG
Web User Interface Language		Select the desired language in which the Web User Interface of the selected Standard SIP Phone variant should be displayed. Default: English	1. Yealink SIP-T19P/SIP-T21P/T22P/T26P/T28P/T4X 2. Yealink SIP-T20P 3. Yealink SIP-T3XG
Phone User Interface		Select the desired language in which the Standard SIP Phone's User Interface should be displayed. Default: English	1. Yealink SIP-T19P/SIP-T21P/T22P/T26P/T28P/T4X 2. Yealink SIP-T20P 3. Yealink SIP-T3XG
User Password		Configure the User Password[b] of the Standard SIP Phone. It can be maximum of 16 characters long. Default: user	1. Yealink SIP-T19P/SIP-T21P/T22P/T26P/T28P/T4X 2. Yealink SIP-T20P 3. Yealink SIP-T3XG
Admin Password		Configure the Admin Password[c] of the Standard SIP Phone. It can be a maximum of up to 16 characters. Default: admin	1. Yealink SIP-T19P/SIP-T21P/T22P/T26P/T28P/T4X 2. Yealink SIP-T20P 3. Yealink SIP-T3XG

Time Zone		Select the Time Zone of the region/country where the Standard SIP Phone is installed. Default: + 05:30 India (Calcutta)	1. Yealink SIP-T19P/SIP-T21P/T22P/T26P/T28P/T4X 2. Yealink SIP-T20P 3. Yealink SIP-T3XG
Daylight Saving Time Mode		Select the Daylight Saving Time Mode that should be applied to the selected Standard SIP Phone. Default: Automatic.	1. Yealink SIP-T19P/SIP-T21P/T22P/T26P/T28P/T4X 2. Yealink SIP-T20P 3. Yealink SIP-T3XG
DST Type		Select the DST Type that should be applied to the Standard SIP Phone, either DST by Date or DST by Week.	1. Yealink SIP-T19P/SIP-T21P/T22P/T26P/T28P/T4X 2. Yealink SIP-T20P 3. Yealink SIP-T3XG
If DST Type = DST by Date	Start Date	Configure the time from when DST should be applied in the year by selecting the Month, Day and Hour from the corresponding list boxes respectively.	1. Yealink SIP-T19P/SIP-T21P/T22P/T26P/T28P/T4X 2. Yealink SIP-T20P 3. Yealink SIP-T3XG
	End Date	Configure the time when DST should end by selecting the Month, Day and Hour from the corresponding list boxes respectively.	1. Yealink SIP-T19P/SIP-T21P/T22P/T26P/T28P/T4X 2. Yealink SIP-T20P 3. Yealink SIP-T3XG
	Offset (minutes)	Configure the DST Offset timer value in minutes. Valid Range:-300 to +300.	1. Yealink SIP-T19P/SIP-T21P/T22P/T26P/T28P/T4X 2. Yealink SIP-T20P 3. Yealink SIP-T3XG

If DST Type = DST by Week	DST Start Month	Select the Month from when DST should be applied.	1. Yealink SIP-T19P/SIP-T21P/T22P/T26P/T28P/T4X 2. Yealink SIP-T20P 3. Yealink SIP-T3XG
	DST Start Day of Week	Select the Day of Week from when DST should be applied.	1. Yealink SIP-T19P/SIP-T21P/T22P/T26P/T28P/T4X 2. Yealink SIP-T20P 3. Yealink SIP-T3XG
	DST Start Day of Week Last in Month	Select the DST Start Day of Week Last in Month.	1. Yealink SIP-T19P/SIP-T21P/T22P/T26P/T28P/T4X 2. Yealink SIP-T20P 3. Yealink SIP-T3XG
	Start Hour of Day	Select the DST Start Hour of the Day.	1. Yealink SIP-T19P/SIP-T21P/T22P/T26P/T28P/T4X 2. Yealink SIP-T20P 3. Yealink SIP-T3XG
	DST Stop Month	Select the Month when DST should end.	1. Yealink SIP-T19P/SIP-T21P/T22P/T26P/T28P/T4X 2. Yealink SIP-T20P 3. Yealink SIP-T3XG
	DST Stop Day of Week	Select the Day of Week when DST should end.	1. Yealink SIP-T19P/SIP-T21P/T22P/T26P/T28P/T4X 2. Yealink SIP-T20P 3. Yealink SIP-T3XG
	DST Stop Day of Week Last in Month	Select the DST Stop Day of Week Last in Month.	1. Yealink SIP-T19P/SIP-T21P/T22P/T26P/T28P/T4X 2. Yealink SIP-T20P 3. Yealink SIP-T3XG
	End Hour of Day	Select the DST End Hour of the Day.	1. Yealink SIP-T19P/SIP-T21P/T22P/T26P/T28P/T4X 2. Yealink SIP-T20P 3. Yealink SIP-T3XG
Date Display Format		Select the Date Display Format for the Standard SIP Phone. Default:	1. Yealink SIP-T19P/SIP-T21P/T22P/T26P/T28P/T4X 2. Yealink SIP-T20P 3. Yealink SIP-T3XG
Time Display Format		Select the Time Display Format for the Standard SIP Phone. Default for T20P Phone: MM DD YY and for other phones WWW MMM DD	1. Yealink SIP-T19P/SIP-T21P/T22P/T26P/T28P/T4X 2. Yealink SIP-T20P 3. Yealink SIP-T3XG
Primary NTP Server		Select the Primary NTP Server Address with which you want to synchronize the Date and Time of the Standard SIP Phone. Default: None. Make sure you configure the desired NTP Server Address/es in the “Third Party IP-Phone General Parameters”.	1. Yealink SIP-T19P/SIP-T21P/T22P/T26P/T28P/T4X 2. Yealink SIP-T20P 3. Yealink SIP-T3XG

a. If the Standard SIP phone is in the same network (LAN) as SARVAM UCS, select Use LAN Port IP Address as Registrar Server Address.

   If the Standard SIP phone is in the Global Network and SARVAM UCS is connected to Internet over WAN, select Use WAN Port IP Address as Registrar Server Address

   If the Standard SIP Phone is connected in the Global Network and SARVAM UCS is located behind a NAT Router, and STUN is programmed, select Use Router/STUN's IP Address as Registrar Server Address. Make sure the Router's Public IP Address is configured in the Network Parameters.

   If Dynamic DNS is configured in the Network Parameters, select Use Dynamic DNS Host Name as Registrar Server Address.

b. To avoid unauthorized access, we recommend you to change the User Password regularly. Make sure it is strong and is kept confidential.

c. To avoid unauthorized access, we recommend you to change the Admin Password regularly. Make sure it is strong and is kept confidential.

Configuring Cisco Standard SIP Phones

You are recommended to complete the following steps before connecting any of the **Cisco** Standard SIP Phones:

- Decide the physical location of the Standard SIP Phone.
- Configure the settings in the third party DHCP Server. For instructions, see "Using any third party DHCP Server in your LAN".
- Configure the **SIP Extension Settings** in SARVAM UCS. For details, see "Configuring SIP Extension Settings using Jeeves"[172]
- Configure the device specific settings applicable to your **Cisco** at any one of the Location1, 2 or 3 on the SIP Extensions page. To do so,
  - Under **Configuration**, click **VoIP Configuration**.
  - Click **SIP Extension Settings** link.
  - Click **Location 1** and configure the following parameters.

Parameter		Description/Action to be Taken	Applicable Standard SIP Phone Variant
Enable Device		Select the Enable Device check box.	1. Cisco SPA50xG/51xG SIP Phone 2. Cisco SPA525G SIP Phone
Location Name		Configure the Location Name to identify the Standard SIP Phone. The Location Name can be a maximum of 18 characters. Default: Blank.	1. Cisco SPA50xG/51xG SIP Phone 2. Cisco SPA525G SIP Phone

172. *Some of the parameters may not be applicable depending on the Standard SIP Phone you have connected. Please refer the specific Standard SIP Phone manufacturer's documentation for more details.*

Device Type		Select Device Type as any of the desired Cisco Phone you want to connect.	1. Cisco SPA50xG/51xG SIP Phone 2. Cisco SPA525G SIP Phone
MAC Address		Enter the MAC Address of the Cisco phone to be connected at this location. Default: Blank. SARVAM UCS validates the phone on the basis of the MAC Address, and provides configuration on validation.	1. Cisco SPA50xG/51xG SIP Phone 2. Cisco SPA525G SIP Phone
Authenticate HTTP Provisioning request		Select the Authenticate HTTP Provisioning request check box to authenticate HTTP request for security reason. Default: Disabled. If you are using the Standard SIP Phone from the Public IP Address, we recommend you to enable this option. SARVAM UCS validates the Standard SIP Phone on the basis of the SIP ID and HTTP Authentication Password and provides configuration on validation. Make sure, you have configured SIP ID as HTTP User Name and HTTP Authentication Password as HTTP Password in the Standard SIP Phone.	1. Cisco SPA50xG/51xG SIP Phone 2. Cisco SPA525G SIP Phone
Registrar Server Address		Select the appropriate Registrar Server Address according to your installation scenario[a] to register the Standard SIP Phone with the SIP Registrar of SARVAM UCS. Default: Use WAN Port IP Address.	1. Cisco SPA50xG/51xG SIP Phone 2. Cisco SPA525G SIP Phone
Standard SIP Authorization Profile		Select the desired Standard SIP Authorization Profile from the list of the profiles. Default: Cisco	1. Cisco SPA50xG/51xG SIP Phone 2. Cisco SPA525G SIP Phone
Send Phone Book		Select the Send Phone Book check box to enable downloading of the Phone Book (consisting of Extension and Global Directory Contacts) from the SARVAM UCS. Default: Enabled	1. Cisco SPA50xG/51xG SIP Phone 2. Cisco SPA525G SIP Phone
Phonebook Name		Configure the Name of the Phonebook. This name will be displayed on the Phone LCD. It can be of maximum 32 characters. Default: Corporate Directory	1. Cisco SPA50xG/51xG SIP Phone 2. Cisco SPA525G SIP Phone
User Password		Configure the User Password[b] of the Standard SIP Phone. It can be a minimum of 1 character to a maximum of 16 characters. Default: Blank	1. Cisco SPA50xG/51xG SIP Phone 2. Cisco SPA525G SIP Phone

Admin Password		Configure the Admin Password[c] of the Cisco phone. It can be a minimum of 1 character to a maximum of 16 characters. Default: Blank	1. Cisco SPA50xG/51xG SIP Phone 2. Cisco SPA525G SIP Phone
Phone User Interface Language		Select the desired language in which the Standard SIP Phone's User Interface should be displayed. Default: English-US	1. Cisco SPA50xG/51xG SIP Phone 2. Cisco SPA525G SIP Phone
Language File Download Path		Select the path from which the Standard SIP Phone must fetch the Language files. Default: None. Make sure you configure the desired Path (Server Address/es) in the "Third Party IP-Phone General Parameters".	1. Cisco SPA50xG/51xG SIP Phone 2. Cisco SPA525G SIP Phone
Time Zone		Select the Time Zone of the region/country where the Standard SIP Phone is installed. Default: GMT + 05:30	1. Cisco SPA50xG/51xG SIP Phone 2. Cisco SPA525G SIP Phone
Enable DST		If DST is applicable, select the Enable DST check box. A list of DST parameters appear. Configure them as per your requirement. Default: Disabled	1. Cisco SPA50xG/51xG SIP Phone 2. Cisco SPA525G SIP Phone

Daylight Saving Time	DST Start Day of Week	Select the Day of Week from when DST should be applied.	1. Cisco SPA50xG/51xG SIP Phone 2. Cisco SPA525G SIP Phone
	DST Start Day of Week Last in Month	Select the DST Start Day of Week Last in Month.	1. Cisco SPA50xG/51xG SIP Phone 2. Cisco SPA525G SIP Phone
	Start Hour of Day	Select the DST Start Hour of the Day.	1. Cisco SPA50xG/51xG SIP Phone 2. Cisco SPA525G SIP Phone
	DST Stop Month	Select the Month when DST should end.	1. Cisco SPA50xG/51xG SIP Phone 2. Cisco SPA525G SIP Phone
	DST Stop Day of Week	Select the Day of Week when DST should end.	1. Cisco SPA50xG/51xG SIP Phone 2. Cisco SPA525G SIP Phone
	DST Stop Day of Week Last in Month	Select the DST Stop Day of Week Last in Month.	1. Cisco SPA50xG/51xG SIP Phone 2. Cisco SPA525G SIP Phone
	End Hour of Day	Select the DST End Hour of the Day.	1. Cisco SPA50xG/51xG SIP Phone 2. Cisco SPA525G SIP Phone
Date Display Format		Select the Date Display Format for the Standard SIP Phone. Default: month/day	1. Cisco SPA50xG/51xG SIP Phone 2. Cisco SPA525G SIP Phone
Time Display Format		Select the Time Display Format for the Standard SIP Phone. Default: 12Hr	1. Cisco SPA50xG/51xG SIP Phone 2. Cisco SPA525G SIP Phone
Offset Timer(min)		Configure the DST Offset timer value in minutes. Valid Range:-300 to +300. Default: 000	1. Cisco SPA50xG/51xG SIP Phone 2. Cisco SPA525G SIP Phone
Primary NTP Server		Select the Primary NTP Server Address with which you want to synchronize the Date and Time of the Standard SIP Phone. Default: None. Make sure you configure the desired NTP Server Address/es in the “Third Party IP-Phone General Parameters”.	1. Cisco SPA50xG/51xG SIP Phone 2. Cisco SPA525G SIP Phone

a. If the Standard SIP phone is in the same network (LAN) as SARVAM UCS, select Use LAN Port IP Address as Registrar Server Address.

   If the Standard SIP phone is in the Global Network and SARVAM UCS is connected to Internet over WAN, select Use WAN Port IP Address as Registrar Server Address

   If the Standard SIP Phone is connected in the Global Network and SARVAM UCS is located behind a NAT Router, and STUN is programmed, select Use Router/STUN's IP Address as Registrar Server Address. Make sure the Router's Public IP Address is configured in the Network Parameters.

   If Dynamic DNS is configured in the Network Parameters, select Use Dynamic DNS Host Name as Registrar Server Address.

b. To avoid unauthorized access, we recommend you to change the User Password regularly. Make sure it is strong and is kept confidential.

c. To avoid unauthorized access, we recommend you to change the Admin Password regularly. Make sure it is strong and is kept confidential.

Configuring Polycom Standard SIP Phones

You are recommended to complete the following steps before connecting any of the **Polycom** Standard SIP Phones:

- Decide the physical location of the Standard SIP Phone.
- Configure the settings in the third party DHCP Server. For instructions, see "Using any third party DHCP Server in your LAN".
- Configure the **SIP Extension Settings** in SARVAM UCS. For details, see "Configuring SIP Extension Settings using Jeeves"[173]
- Configure the device specific settings applicable to your **Polycom** at any one of the Location1, 2 or 3 on the SIP Extensions page. To do so,
  - Under **Configuration**, click **VoIP Configuration**.
  - Click **SIP Extension Settings** link.
  - Click **Location 1** and configure the following parameters.

Parameter		Description/Action to be Taken	Applicable Standard SIP Phone Variant
Enable Device		Select the Enable Device check box.	Polycom IP Phone
Location Name		Configure the Location Name to identify the Standard SIP Phone. The Location Name can be a maximum of 18 characters. Default: Blank.	Polycom IP Phone
Device Type		Select Device Type as Polycom IP Phone.	Polycom IP Phone

173. Some of the parameters may not be applicable depending on the Standard SIP Phone you have connected. Please refer the specific Standard SIP Phone manufacturer's documentation for more details.

MAC Address		Enter the MAC Address of the Polycom phone to be connected at this location. Default: Blank. SARVAM UCS validates the phone on the basis of the MAC Address, and provides configuration on validation.	Polycom IP Phone
Authenticate HTTP Provisioning request		Select the Authenticate HTTP Provisioning request check box to authenticate HTTP request for security reason. Default: Disabled. If you are using the Standard SIP Phone from the Public IP Address, we recommend you to enable this option. SARVAM UCS validates the Standard SIP Phone on the basis of the SIP ID and HTTP Authentication Password and provides configuration on validation. Make sure, you have configured SIP ID as HTTP User Name and HTTP Authentication Password as HTTP Password in the Standard SIP Phone.	Polycom IP Phone
Registrar Server Address		Select the appropriate Registrar Server Address according to your installation scenario[a] to register the Standard SIP Phone with the SIP Registrar of SARVAM UCS. Default: Use WAN Port IP Address	Polycom IP Phone
Standard SIP Authorization Profile		Select the desired Standard SIP Authorization Profile from the list of the profiles. Default: Polycom	Polycom IP Phone
Send Phone Book		Select the Send Phone Book check box to enable downloading of the Phone Book (consisting of Extension and Global Directory Contacts) from the SARVAM UCS. Default: Enabled	Polycom IP Phone
User Password		Configure the User Password[b] of the Standard SIP Phone. It can be maximum of 16 characters long. Default: 123	Polycom IP Phone
Admin Password		Configure the Admin Password[c] of the Standard SIP Phone. It can be a maximum of up to 16 characters. Default: 456	Polycom IP Phone
Phone User Interface Language		Select the desired language in which the Standard SIP Phone's User Interface should be displayed. Default: English Internal (en-in)	Polycom IP Phone

Enable DST		If DST is applicable, select the Enable DST check box. A list of DST parameters appear. Configure them as per your requirement. Default: Disabled	Polycom IP Phone
DST Type		Select the DST Type that should be applied to the Standard SIP Phone, either DST by Date or DST by Week.	Polycom IP Phone
If DST Type = DST by Date	Start Month	Select the Month from when DST should be applied.	Polycom IP Phone
	Start Day	Select the Day from when DST should be applied.	Polycom IP Phone
	Start Hour	Select the Hour from when DST should be applied.	Polycom IP Phone
	End Month	Select the Month when DST should end.	Polycom IP Phone
	End Day	Select the Day when DST should end.	Polycom IP Phone
	End Hour	Select the Hour when DST should end.	Polycom IP Phone
If DST Type = DST by Week	DST Start Month	Select the Month from when DST should be applied.	Polycom IP Phone
	DST Start Day of Week	Select the Day of Week from when DST should be applied.	Polycom IP Phone
	DST Start Day of Week Last in Month	Select the DST Start Day of Week Last in Month.	Polycom IP Phone
	Start Hour of Day	Select the DST Start Hour of the Day.	Polycom IP Phone
	DST Stop Month	Select the Month when DST should end.	Polycom IP Phone
	DST Stop Day of Week	Select the Day of Week when DST should end.	Polycom IP Phone
	DST Stop Day of Week Last in Month	Select the DST Stop Day of Week Last in Month.	Polycom IP Phone
	End Hour of Day	Select the DST End Hour of the Day.	Polycom IP Phone
Date Display Format		Select the Date Display Format for the Standard SIP Phone. Default: 1 Jan, Mon	Polycom IP Phone
Time Display Format		Select the Time Display Format for the Standard SIP Phone. Default: 24Hr	Polycom IP Phone

Primary NTP Server		Select the Primary NTP Server Address with which you want to synchronize the Date and Time of the Standard SIP Phone. Default: None. Make sure you configure the desired NTP Server Address/es in the "Third Party IP-Phone General Parameters".	Polycom IP Phone

a. If the Standard SIP phone is in the same network (LAN) as SARVAM UCS, select Use LAN Port IP Address as Registrar Server Address.

   If the Standard SIP phone is in the Global Network and SARVAM UCS is connected to Internet over WAN, select Use WAN Port IP Address as Registrar Server Address

   If the Standard SIP Phone is connected in the Global Network and SARVAM UCS is located behind a NAT Router, and STUN is programmed, select Use Router/STUN's IP Address as Registrar Server Address. Make sure the Router's Public IP Address is configured in the Network Parameters.

   If Dynamic DNS is configured in the Network Parameters, select Use Dynamic DNS Host Name as Registrar Server Address.

b. To avoid unauthorized access, we recommend you to change the User Password regularly. Make sure it is strong and is kept confidential.

c. To avoid unauthorized access, we recommend you to change the Admin Password regularly. Make sure it is strong and is kept confidential.

Configuring Snom Standard SIP Phones

You are recommended to complete the following steps before connecting any of the **Snom** Standard SIP Phones:

- Decide the physical location of the Standard SIP Phone. It must be connected in the same subnet as that of the LAN Port of SARVAM UCS.
- Configure the settings in the third party DHCP Server. For instructions, see "Using any third party DHCP Server in your LAN".
- Configure the **SIP Extension Settings** in SARVAM UCS. For details, see "Configuring SIP Extension Settings using Jeeves"[174]
- Configure the device specific settings applicable to your **Snom** at any one of the Location1, 2 or 3 on the SIP Extensions page. To do so,
  - Under **Configuration**, click **VoIP Configuration**.
  - Click **SIP Extension Settings** link.
  - Click **Location 1** and configure the following parameters.

Parameter	Description/Action to be Taken	Applicable Standard SIP Phone Variant
Enable Device	Select the Enable Device check box.	Snom IP Phone

174. Some of the parameters may not be applicable depending on the Standard SIP Phone you have connected. Please refer the specific Standard SIP Phone manufacturer's documentation for more details.

Location Name	Configure the Location Name to identify the Standard SIP Phone. The Location Name can be a maximum of 18 characters. Default: Blank.	Snom IP Phone
Device Type	Select Device Type as Snom IP Phone.	Snom IP Phone
MAC Address	Enter the MAC Address of the Snom phone to be connected at this location. Default: Blank. SARVAM UCS validates the phone on the basis of the MAC Address, and provides configuration on validation.	Snom IP Phone
Authenticate HTTP Provisioning request	Select the Authenticate HTTP Provisioning request check box to authenticate HTTP request for security reason. Default: Disabled. If you are using the Standard SIP Phone from the Public IP Address, we recommend you to enable this option. SARVAM UCS validates the Standard SIP Phone on the basis of the SIP ID and HTTP Authentication Password and provides configuration on validation. Make sure, you have configured SIP ID as HTTP User Name and HTTP Authentication Password as HTTP Password in the Standard SIP Phone.	Snom IP Phone
Registrar Server Address	Select the appropriate Registrar Server Address according to your installation scenario[a] to register the Standard SIP Phone with the SIP Registrar of SARVAM UCS. Default: Use WAN Port IP Address.	Snom IP Phone
Standard SIP Authorization Profile	Select the desired Standard SIP Authorization Profile from the list of the profiles. Default: Snom	Snom IP Phone
Send Phone Book	Select the Send Phone Book check box to enable downloading of the Phone Book (consisting of Extension and Global Directory Contacts) from the SARVAM UCS. Default: Enabled	Snom IP Phone
Web User Interface Language	Select the desired language in which the Web User Interface of the selected Standard SIP Phone should be displayed. Default: English	Snom IP Phone
Web User Language File Download Path	Select the path from which the Standard SIP Phone must fetch the Language files. Default: None. Make sure you configure the desired Path (Server Address/ es) in the “Third Party IP-Phone General Parameters”.	Snom IP Phone
Phone User Interface Language	Select the desired language in which the Standard SIP Phone’s User Interface should be displayed. Default: English	Snom IP Phone

Phone User Language File Download Path	Select the path from which the Standard SIP Phone must fetch the Language files. Default: None. Make sure you configure the desired Path (Server Address/es) in the "Third Party IP-Phone General Parameters".	Snom IP Phone
Time Zone	Select the Time Zone of the region/country where the Standard SIP Phone is installed. Default: India	Snom IP Phone
Date Display Format	Select the Date Display Format for the Standard SIP Phone. Default: mm/dd	Snom IP Phone
Time Display Format	Select the Time Display Format for the Standard SIP Phone. Default: 12Hr	Snom IP Phone
Call Progress Tone	Select the region to apply the Call Progress Tone prevailing there. Default: India.	Snom IP Phone
HTTP Server Login Username	Configure the HTTP Server Login User Name for web access. It can be a maximum of 16 characters. Default: Blank	Snom IP Phone
HTTP Server Login Password	Configure the HTTP Server Login Password[b] for web access. It can be a maximum of 16 characters. Default: Blank	Snom IP Phone
Admin Mode	Select Admin Mode check box to allow its access to the user. Default: Enabled	Snom IP Phone
Admin Mode Password	Configure the Admin Password[c] of the Standard SIP Phone. It can be a minimum of 1 character to a maximum of 16 characters. Default: 0000	Snom IP Phone
Primary NTP Server	Select the Primary NTP Server Address with which you want to synchronize the Date and Time of the Standard SIP Phone. Default: None. Make sure you configure the desired NTP Server Address/es in the "Third Party IP-Phone General Parameters".	Snom IP Phone

a. If the phone is in the same network (LAN) as SARVAM UCS, select Use LAN Port IP Address as Registrar Server Address.

If the phone is in the Global Network and SARVAM UCS is connected to Internet over WAN, select Use WAN Port IP Address as Registrar Server Address

If the phone is connected in the Global Network and SARVAM UCS is located behind a NAT Router, and STUN is programmed, select Use Router/STUN's IP Address as Registrar Server Address. Make sure the Router's Public IP Address is configured in the Network Parameters.

If Dynamic DNS is configured in the Network Parameters, select Use Dynamic DNS Host Name as Registrar Server Address.

b. To avoid unauthorized access, we recommend you to change the HTTP Server Login Password regularly. Make sure it is strong and is kept confidential.

c. To avoid unauthorized access, we recommend you to change the Admin Mode Password regularly. Make sure it is strong and is kept confidential.

Configuring Matrix SPARSH VP110 Standard SIP Phone

You are recommended to complete the following steps before connecting any of the **Matrix SPARSH VP110** Standard SIP Phones:

- Decide the physical location of the Standard SIP Phone.
- Configure the settings in the third party DHCP Server. For instructions, see “Using any third party DHCP Server in your LAN”.
- Configure the **SIP Extension Settings** in SARVAM UCS. For details, see “Configuring SIP Extension Settings using Jeeves”[175]
- Configure the device specific settings applicable to your **Matrix SPARSH VP110** at any one of the Location1, 2 or 3 on the SIP Extensions page. To do so,
  - Under **Configuration**, click **VoIP Configuration**.
  - Click **SIP Extension Settings** link.
  - Click **Location 1** and configure the following parameters.

Parameter		Description/Action to be Taken	Applicable Standard SIP Phone Variant
Enable Device		Select the Enable Device check box. Default: Disabled.	Matrix SPARSH VP110
Location Name		Configure the Location Name to identify the Standard SIP Phone. The Location Name can be a maximum of 18 characters. Default: Blank.	Matrix SPARSH VP110
Device Type		Select Device Type as MATRIX SPARSH VP110.	Matrix SPARSH VP110
MAC Address		Enter the MAC Address of the SPARSH VP110 phone to be connected at this location. Default: Blank. SARVAM UCS validates the phone on the basis of the MAC Address, and provides configuration on validation.	Matrix SPARSH VP110

175. Some of the parameters may not be applicable depending on the Standard SIP Phone you have connected. Please refer the specific Standard SIP Phone manufacturer’s documentation for more details.

Authenticate HTTP Provisioning request		Select the Authenticate HTTP Provisioning request check box to authenticate HTTP request for security reason. Default: Disabled. If you are using the Standard SIP Phone from the Public IP Address, we recommend you to enable this option. SARVAM UCS validates the Standard SIP Phone on the basis of the SIP ID and HTTP Authentication Password and provides configuration on validation. Make sure, you have configured SIP ID as HTTP User Name and HTTP Authentication Password as HTTP Password in the Standard SIP Phone.	Matrix SPARSH VP110
Registrar Server Address		Select the appropriate Registrar Server Address according to your installation scenario[a] to register the Standard SIP Phone with the SIP Registrar of SARVAM UCS. Default: Use WAN Port IP Address.	Matrix SPARSH VP110
Standard SIP Authorization Profile		Select the desired Standard SIP Authorization Profile from the list of profiles. Default: SPARSH VP110.	Matrix SPARSH VP110
Send Key Settings		Select the Send Phone Book check box to apply the Key Settings[b] to the Phone. Default: Enabled	Matrix SPARSH VP110
Dial Plan[c]		Select the desired Dial Plan. Default: 1 The Phone will detect end of dialing as per the rules configured in the Dial Plan selected here.	Matrix SPARSH VP110
Transport Mode		Select the protocol to be used to transport the SIP messages. Default: UDP	Matrix SPARSH VP110
Enable SRTP?		Select the Enable SRTP? check box for secure conversations over SIP. Default: Disabled	Matrix SPARSH VP110
SIP DiffServe/ ToS		Enter the desired SIP DiffServe/ToS to set the Quality of Service (QoS) for SIP packets Default: 26	Matrix SPARSH VP110
RTP DiffServe/ ToS		Enter the desired RTP DiffServe/ToS to set the Quality of Service (QoS) for RTP packets. Default: 46	Matrix SPARSH VP110
SIP Port		Enter the port on which the phone will listen for SIP messages. This port is used as source port in SIP messages.This port is also used to send SIP messages to the remote peer. Default: 5060	Matrix SPARSH VP110

Min RTP Port		To define a range of RTP ports, configure the minimum local RTP port. Default: 11780	Matrix SPARSH VP110
Max RTP Port		To define a range of RTP ports, configure the maximum local RTP port. Default: 11800	Matrix SPARSH VP110
Allow HTTP		Select the Allow HTTP check box to enable HTTP Web access. Default: Enabled	Matrix SPARSH VP110
HTTP Port		Enter the port number on which the HTTP access is to be given. Default: 80	Matrix SPARSH VP110
Allow HTTPS		Select the Allow HTTPS check box to enable HTTPS Web access. Default: Enabled	Matrix SPARSH VP110
HTTPS Port		Enter the port number on which the HTTPS access is to be given. Default: 443	Matrix SPARSH VP110
Local Phone Book[d]		Select the type of contacts (Extension or/and Global Directory Contacts) the phone must download from the SARVAM UCS. These will be stored in the phone's Local Phone Book. Default: Do not send	Matrix SPARSH VP110
Remote Phone Book		Select the type of contacts (Extension or/and Global Directory Contacts) the phone must download from the SARVAM UCS. These will be stored in the phone's Remote Phone Book. Default: Send first Extension Numbers, remaining Global Directory Numbers	Matrix SPARSH VP110
Web User Interface Language		Select the desired language in which the Web User Interface should be displayed. Default: English	Matrix SPARSH VP110
Phone User Interface Language		Select the desired language in which the Phone's User Interface should be displayed. Default: English	Matrix SPARSH VP110
User Password		Configure the User Password[e] of the Standard SIP Phone. It can be maximum of 16 characters long. Default: user	Matrix SPARSH VP110
Admin Password		Configure the Admin Password[f] of the Standard SIP Phone. It can be a maximum of up to 16 characters. Default: admin	Matrix SPARSH VP110
Ringer Device for Headset		When in Headset mode, select the ring destination for the SPARSH VP110. Default: Use Speaker	Matrix SPARSH VP110

Enable Distinctive Ring		Select the Enable Distinctive Ring check box, to set different ringing patterns to distinguish between different types of call events. The following types of call events are supported: • Internal Call • Trunk Call • Auto Call Back • Auto Redial • Alarm • Emergency • Operator Alarm • Message Wait • Priority • Emergency Conference Default: Disabled	Matrix SPARSH VP110
Call Progress Tone		Select the region to apply the Call Progress Tone prevailing there. Default: Custom.	Matrix SPARSH VP110
Time Zone		Select the Time Zone of the region/country where the Standard SIP Phone is installed. Default: + 05:30 India (Calcutta)	Matrix SPARSH VP110
Daylight Saving Time Mode		Select the Daylight Saving Time Mode that should be applied to the selected Standard SIP Phone. Default: Automatic.	Matrix SPARSH VP110
DST Type		Select the DST Type that should be applied to the Standard SIP Phone, either DST by Date or DST by Week. Default: DST by Date	Matrix SPARSH VP110
If DST Type = DST by Date	Start Month	Select the Month from when DST should be applied.	Matrix SPARSH VP110
	Start Day	Select the Day from when DST should be applied.	Matrix SPARSH VP110
	Start Hour	Select the Hour from when DST should be applied.	Matrix SPARSH VP110
	End Month	Select the Month when DST should end.	Matrix SPARSH VP110
	End Day	Select the Day when DST should end.	Matrix SPARSH VP110
	End Hour	Select the Hour when DST should end.	Matrix SPARSH VP110
	Offset Timer (min)	Configure the DST Offset timer value in minutes. Valid Range:-300 to +300. Default: Blank	Matrix SPARSH VP110

If DST Type = DST by Week	DST Start Month	Select the Month from when DST should be applied.	Matrix SPARSH VP110
	DST Start Day of Week	Select the Day of Week from when DST should be applied.	Matrix SPARSH VP110
	DST Start Day of Week Last in Month	Select the DST Start Day of Week Last in Month.	Matrix SPARSH VP110
	Start Hour of Day	Select the DST Start Hour of the Day.	Matrix SPARSH VP110
	DST Stop Month	Select the Month when DST should end.	Matrix SPARSH VP110
	DST Stop Day of Week	Select the Day of Week when DST should end.	Matrix SPARSH VP110
	DST Stop Day of Week Last in Month	Select the DST Stop Day of Week Last in Month.	Matrix SPARSH VP110
	End Hour of Day	Select the DST End Hour of the Day.	Matrix SPARSH VP110
Date Display Format		Select the Date Display Format for the Standard SIP Phone. Default: WWW MMM DD	Matrix SPARSH VP110
Time Display Format		Select the Time Display Format for the Standard SIP Phone. Default: 24 Hr	Matrix SPARSH VP110
Primary NTP Server		Select the Primary NTP Server Address with which you want to synchronize the Date and Time of the Standard SIP Phone. Default: None. Make sure you configure the desired NTP Server Address/es in the “Third Party IP-Phone General Parameters”.	Matrix SPARSH VP110

a. If the Standard SIP phone is in the same network (LAN) as SARVAM UCS, select Use LAN Port IP Address as Registrar Server Address.

If the Standard SIP phone is in the Global Network and SARVAM UCS is connected to Internet over WAN, select Use WAN Port IP Address as Registrar Server Address

If the Standard SIP Phone is connected in the Global Network and SARVAM UCS is located behind a NAT Router, and STUN is programmed, select Use Router/STUN's IP Address as Registrar Server Address. Make sure the Router’s Public IP Address is configured in the Network Parameters.If Dynamic DNS is configured in the Network Parameters, select Use Dynamic DNS Host Name as Regis trar Server Address.

b. To apply the desired Key Template:
   i) Select the **Send Key Settings** check box.
   ii) Click **Key Template** under **Configuration**.
   iii) Click the desired key template - Operator/ Executive/ Hotel Attendant/Guest of VP110. Assign features facilities to the keys in this template as per your requirement.
   For instructions, see "Customizing Extended IP Phone Templates using Jeeves"
   iv) Click **SIP Extension Settings** under **VoIP Configuration**. Select the SIP Extension Number on which SPARSH VP110 is registered.
   v) Under **General Parameters**, scroll to **Key Template** and select the template you configured as per your requirement.

The Personalized option of Key Template is not applicable for SPARSH VP110.

c. If you want to apply the rules of the Dial Plan configured in SARVAM UCS, see "Dial Plan for SIP Extension".
You can also configure rules for the Dial Plan from each phone. To do so, refer to the *SPARSH VP110 User Guide*.

d. If an option other than "Do not send" is selected in Local Phone Book, it will overwrite all Local Phone Book's contacts of the phone.

e. To avoid unauthorized access, we recommend you to change the User Password regularly. Make sure it is strong and is kept confidential.

f. To avoid unauthorized access, we recommend you to change the Admin Password regularly. Make sure it is strong and is kept confidential.

Configuring Htek 802 Standard SIP Phones

You are recommended to complete the following steps before connecting the **Htek 802** Standard SIP Phones:

- Decide the physical location of the Standard SIP Phone.
- Configure the settings in the third party DHCP Server. For instructions, see "Using any third party DHCP Server in your LAN".
- Configure the **SIP Extension Settings** in SARVAM UCS. For details, see "Configuring SIP Extension Settings using Jeeves"
- Configure the device specific settings applicable to **Htek** at any one of the Location1, 2 or 3 on the SIP Extensions page. To do so,
  - Under **Configuration**, click **VoIP Configuration**.
  - Click **SIP Extension Settings** link.
  - Click **Location 1** and configure the following parameters.

Parameter	Description/Action to be Taken	Applicable Standard SIP Phone Variant
Enable Device	Select the Enable Device check box.	Htek 802 IP Phone
Location Name	Configure the Location Name to identify the Standard SIP Phone. The Location Name can be a maximum of 18 characters. Default: Blank.	Htek 802 IP Phone
Device Type	Select Device Type as Htek 802.	Htek 802 IP Phone

MAC Address	Enter the MAC Address of the Htek 802 phone to be connected at this location. Default: Blank. SARVAM UCS validates the phone on the basis of the MAC Address, and provides configuration on validation.	Htek 802 IP Phone
Authenticate HTTP Provisioning request	Select the Authenticate HTTP Provisioning request check box to authenticate HTTP request for security reason. Default: Disabled. If you are using the Standard SIP Phone from the Public IP Address, we recommend you to enable this option. SARVAM UCS validates the Standard SIP Phone on the basis of the SIP ID and HTTP Authentication Password and provides configuration on validation. Make sure, you have configured SIP ID as HTTP User Name and HTTP Authentication Password as HTTP Password in the Standard SIP Phone.	Htek 802 IP Phone
Registrar Server Address	Select the appropriate Registrar Server Address according to your installation scenario[a] to register the Standard SIP Phone with the SIP Registrar of SARVAM UCS. Default: Use WAN Port IP Address.	Htek 802 IP Phone
Standard SIP Authorization Profile	Select the desired Standard SIP Authorization Profile from the list of profiles. Default: Htek	Htek 802 IP Phone
Admin Password	Configure the Admin Password[b] of the Standard SIP Phone. It can be a maximum of up to 16 characters. Default: admin	Htek 802 IP Phone
Assign Voice Mail Key	Enable this check box to assign the first programmable key of the phone as Voice Mail.	Htek 802 IP Phone
SIP Port	Enter the port on which the phone will listen for SIP messages. This port is used as source port in SIP messages.This port is also used to send SIP messages to the remote peer. Default: 5060	Htek 802 IP Phone
Min RTP Port	To define a range of RTP ports, configure the minimum local RTP port. Default: 5004	Htek 802 IP Phone
Max RTP Port	To define a range of RTP ports, configure the maximum local RTP port. Default: 5014	Htek 802 IP Phone
Apply System Time Zone	Select this check box if you want to apply the System's Time Zone to the Standard SIP Phone.	Htek 802 IP Phone
Time Zone	Select the Time Zone of the region/country where the Standard SIP Phone is installed. Default: GMT + 05:30	Htek 802 IP Phone

Primary NTP Server	Select the Primary NTP Server Address with which you want to synchronize the Date and Time of the Standard SIP Phone. Default: None. Make sure you configure the desired NTP Server Address/es in the "Third Party IP-Phone General Parameters".	Htek 802 IP Phone
Daylight Saving Time Mode	Select the Daylight Saving Time Mode that should be applied to the selected Standard SIP Phone. Default: Automatic.	Htek 802 IP Phone

a. If the phone is in the same network (LAN) as SARVAM UCS, select Use LAN Port IP Address as Registrar Server Address.

If the phone is in the Global Network and SARVAM UCS is connected to Internet over WAN, select Use WAN Port IP Address as Registrar Server Address

If the phone is connected in the Global Network and SARVAM UCS is located behind a NAT Router, and STUN is programmed, select Use Router/STUN's IP Address as Registrar Server Address. Make sure the Router's Public IP Address is configured in the Network Parameters.

If Dynamic DNS is configured in the Network Parameters, select Use Dynamic DNS Host Name as Registrar Server Address.

b. To avoid unauthorized access, we recommend you to change the Admin Password regularly. Make sure it is strong and is kept confidential.

Configuring Matrix SPARSH VP710 Standard SIP Phone

You are recommended to complete the following steps before connecting any of the **Matrix SPARSH VP710** Standard SIP Phones:

- Decide the physical location of the Standard SIP Phone. It must be connected in the same subnet as that of the LAN Port of SARVAM UCS.
- Configure the settings in the third party DHCP Server. For instructions, see "Using any third party DHCP Server in your LAN".
- Configure the **SIP Extension Settings** in SARVAM UCS. For details, see "Configuring SIP Extension Settings using Jeeves"
- Configure the device specific settings applicable to your **Matrix SPARSH VP710** at any one of the Location1, 2 or 3 on the SIP Extensions page. To do so,
  - Under **Configuration**, click **VoIP Configuration**.
  - Click **SIP Extension Settings** link.
  - Click **Location 1** and configure the following parameters.

Parameter		Description/Action to be Taken	Applicable Standard SIP Phone Variant
Enable Device		Select the Enable Device check box. Default: Disabled.	Matrix SPARSH VP710

Location Name		Configure the Location Name to identify the Standard SIP Phone. The Location Name can be a maximum of 18 characters. Default: Blank.	Matrix SPARSH VP710
Device Type		Select Device Type as MATRIX SPARSH VP710 - Standard SIP.	Matrix SPARSH VP710
MAC Address		Enter the MAC Address of the SPARSH VP710 phone to be connected at this location. Default: Blank. SARVAM UCS validates the phone on the basis of the MAC Address, and provides configuration on validation.	Matrix SPARSH VP710
Authenticate HTTP Provisioning request		Select the Authenticate HTTP Provisioning request check box to authenticate HTTP request for security reason. Default: Disabled. If you are using the Standard SIP Phone from the Public IP Address, we recommend you to enable this option. SARVAM UCS validates the Standard SIP Phone on the basis of the SIP ID and HTTP Authentication Password and provides configuration on validation. Make sure, you have configured SIP ID as HTTP User Name and HTTP Authentication Password as HTTP Password in the Standard SIP Phone.	Matrix SPARSH VP710
Registrar Server Address		Select the appropriate Registrar Server Address according to your installation scenario[a] to register the Standard SIP Phone with the SIP Registrar of SARVAM UCS. Default: Use WAN Port IP Address.	Matrix SPARSH VP710
Standard SIP Authorization Profile		Select the desired Standard SIP Authorization Profile from the list of profiles. Default: SPARSH VP710	Matrix SPARSH VP710
Dial Plan[b]		Select the desired Dial Plan. Default: 1 The Phone will detect end of dialing as per the rules configured in the Dial Plan selected here.	Matrix SPARSH VP710
Transport Mode		Select the protocol to be used to transport the SIP messages. Default: UDP	Matrix SPARSH VP710
Enable SRTP?		Select the Enable SRTP? check box for secure conversations over SIP. Default: Disabled	Matrix SPARSH VP710

DTMF Type		Select the appropriate DTMF Type to determine how the DTMF digits will be sent over the IP Network, when a DTMF digit is pressed.	Matrix SPARSH VP710
SIP DiffServe/ ToS		Enter the desired SIP DiffServe/ToS to set the Quality of Service (QoS) for SIP packets Default: 26	Matrix SPARSH VP710
RTP DiffServe/ ToS		Enter the desired RTP DiffServe/ToS to set the Quality of Service (QoS) for RTP packets. Default: 46	Matrix SPARSH VP710
SIP Port		Enter the port on which the phone will listen for SIP messages. This port is used as source port in SIP messages.This port is also used to send SIP messages to the remote peer. Default: 5060	Matrix SPARSH VP710
SIP TLS Port		Enter the port on which the phone will listen for SIP messages transported over TLS. This port is used as source port in SIP messages.This port is also used to send SIP messages to the remote peer. Default: 5061	Matrix SPARSH VP710
Min RTP Port		To define a range of RTP ports, configure the minimum local RTP port. Default: 11780	Matrix SPARSH VP710
Max RTP Port		To define a range of RTP ports, configure the maximum local RTP port. Default: 11800	Matrix SPARSH VP710
Allow HTTP		Select the Allow HTTP check box to enable HTTP Web access. Default: Enabled	Matrix SPARSH VP710
HTTP Port		Enter the port number on which the HTTP access is to be given. Default: 80	Matrix SPARSH VP710
Allow HTTPS		Select the Allow HTTPS check box to enable HTTPS Web access. Default: Enabled	Matrix SPARSH VP710
HTTPS Port		Enter the port number on which the HTTPS access is to be given. Default: 443	Matrix SPARSH VP710
Local Phone Book[C]		Select the type of contacts (Extension or/and Global Directory Contacts) the phone must download from the SARVAM UCS. These will be stored in the phone's Local Phone Book. Default: Do not send	Matrix SPARSH VP710
Send Personal Directory		Enable this check box if you want to allow usage of Personal Directory. These contacts will be stored in the phone's Local phone Book.	Matrix SPARSH VP710

Remote Phone Book		Select the type of contacts (Extension or/and Global Directory Contacts) the phone must download from the SARVAM UCS. These will be stored in the phone's Remote Phone Book. Default: Send first Extension Numbers, remaining Global Directory Numbers	Matrix SPARSH VP710
Web User Interface Language		Select the desired language in which the Web User Interface should be displayed. Default: English	Matrix SPARSH VP710
Phone User Interface Language		Select the desired language in which the Phone's User Interface should be displayed. Default: English	Matrix SPARSH VP710
User Password		Configure the User Password[d] of the Standard SIP Phone. It can be maximum of 16 characters long. Default: user	Matrix SPARSH VP710
Admin Password		Configure the Admin Password[e] of the Standard SIP Phone. It can be a maximum of up to 16 characters. Default: admin	Matrix SPARSH VP710
Ringer Device for Headset		When in Headset mode, select the ring destination for the SPARSH VP710. Default: Use Speaker	Matrix SPARSH VP710
Enable Distinctive Ring		Select the Enable Distinctive Ring check box, to set different ringing patterns to distinguish between different types of call events. The following types of call events are supported: • Internal Call • Trunk Call • Auto Call Back • Auto Redial • Alarm • Emergency • Operator Alarm • Message Wait • Priority • Emergency Conference Default: Disabled	Matrix SPARSH VP710
Call Progress Tone		Select the region to apply the Call Progress Tone prevailing there. Default: Custom.	Matrix SPARSH VP710
Time Zone		Select the Time Zone of the region/country where the Standard SIP Phone is installed. Default: + 05:30 India (Calcutta)	Matrix SPARSH VP710
Daylight Saving Time Mode		Select the Daylight Saving Time Mode that should be applied to the selected Standard SIP Phone. Default: Automatic.	Matrix SPARSH VP710

DST Type		Select the DST Type that should be applied to the Standard SIP Phone, either DST by Date or DST by Week. Default: DST by Date	Matrix SPARSH VP710
If DST Type = DST by Date	Start Month	Select the Month from when DST should be applied.	Matrix SPARSH VP710
	Start Day	Select the Day from when DST should be applied.	Matrix SPARSH VP710
	Start Hour	Select the Hour from when DST should be applied.	Matrix SPARSH VP710
	End Month	Select the Month when DST should end.	Matrix SPARSH VP710
	End Day	Select the Day when DST should end.	Matrix SPARSH VP710
	End Hour	Select the Hour when DST should end.	Matrix SPARSH VP710
	Offset (minutes)	Configure the DST Offset timer value in minutes. Valid Range:-300 to +300. Default: Blank	Matrix SPARSH VP710
If DST Type = DST by Week	Offset (minutes)	Configure the DST Offset timer value in minutes. Valid Range:-300 to +300. Default: Blank	Matrix SPARSH VP710
	DST Start Month	Select the Month from when DST should be applied.	Matrix SPARSH VP710
	DST Start Day of Week	Select the Day of Week from when DST should be applied.	Matrix SPARSH VP710
	DST Start Day of Week Last in Month	Select the DST Start Day of Week Last in Month.	Matrix SPARSH VP710
	Start Hour of Day	Select the DST Start Hour of the Day.	Matrix SPARSH VP710
	DST Stop Month	Select the Month when DST should end.	Matrix SPARSH VP710
	DST Stop Day of Week	Select the Day of Week when DST should end.	Matrix SPARSH VP710
	DST Stop Day of Week Last in Month	Select the DST Stop Day of Week Last in Month.	Matrix SPARSH VP710
	End Hour of Day	Select the DST End Hour of the Day.	Matrix SPARSH VP710
Date Display Format		Select the Date Display Format for the Standard SIP Phone. Default: WWW MMM DD	Matrix SPARSH VP710
Time Display Format		Select the Time Display Format for the Standard SIP Phone. Default: 24 Hr	Matrix SPARSH VP710

Primary NTP Server		Select the Primary NTP Server Address with which you want to synchronize the Date and Time of the Standard SIP Phone. Default: None. Make sure you configure the desired NTP Server Address/es in the "Third Party IP-Phone General Parameters".	Matrix SPARSH VP710

a. If the Standard SIP phone is in the same network (LAN) as SARVAM UCS, select Use LAN Port IP Address as Registrar Server Address.

If the Standard SIP phone is in the Global Network and SARVAM UCS is connected to Internet over WAN, select Use WAN Port IP Address as Registrar Server Address

If the Standard SIP Phone is connected in the Global Network and SARVAM UCS is located behind a NAT Router, and STUN is programmed, select Use Router/STUN's IP Address as Registrar Server Address. Make sure the Router's Public IP Address is configured in the Network Parameters.

If Dynamic DNS is configured in the Network Parameters, select Use Dynamic DNS Host Name as Registrar Server Address.

b. If you want to apply the rules of the Dial Plan configured in SARVAM UCS, see "Dial Plan for SIP Extension". You can also configure rules for the Dial Plan from each phone. To do so, refer to the *SPARSH VP710 User Guide*.

c. If an option other than "Do not send" is selected in Local Phone Book, it will overwrite all Local Phone Book's contacts of the phone.

d. To avoid unauthorized access, we recommend you to change the User Password regularly. Make sure it is strong and is kept confidential.

e. To avoid unauthorized access, we recommend you to change the Admin Password regularly. Make sure it is strong and is kept confidential.

Configuring Any Standard SIP Phone

You are recommended to complete the following steps before connecting Any Standard SIP Phone:

- Decide the physical location of the Standard SIP Phone.
- Configure the settings in the third party DHCP Server. For instructions, see "Using any third party DHCP Server in your LAN".
- Configure the SIP Extension Settings in SARVAM UCS. For details, see "Configuring SIP Extension Settings using Jeeves"[176].
- Configure the device specific settings applicable to your Standard SIP Phone at any one of the Location 1, 2 or 3 on the SIP Extensions page. To do so,
  - Under **Configuration**, click **VoIP Configuration**.
  - Click **SIP Extension Settings**.

176. *Some of the features may not be supported depending on the Standard SIP Phone you have connected. Please refer the specific Standard SIP Phone manufacturer's documentation for more details.*

- Click **Location 1** and configure the following parameters.

Parameter	Description/Action to be Taken	Applicable Standard SIP Phone Variant
Enable Device	Select the Enable Device check box. Default: Disabled.	Any Standard SIP Phone
Location Name	Configure the Location Name to identify the Standard SIP Phone. The Location Name can be a maximum of 18 characters. Default: Blank.	Any Standard SIP Phone
Device Type	Select Device Type as Any Standard SIP Phone.	Any Standard SIP Phone
MAC Address	Enter the MAC Address of the Standard SIP phone to be connected at this location. Default: Blank. SARVAM UCS validates the phone on the basis of the MAC Address, and provides configuration on validation.	Any Standard SIP Phone
Standard SIP Authorization Profile	Select the desired Standard SIP Authorization Profile from the list of profiles. Default: None.	Any Standard SIP Phone

*If you select the Device Type as **Any Standard SIP**, then you are recommended to configure the **Standard SIP Authorization Profile** to prevent any unauthorized access and misuse of the system.*

If you want to replicate the configuration of the SIP Phone Settings same as Location 1 to Location 2 and Location 3, you can use the **Copy** button present at the bottom of the page. To know more, refer to "Copy Parameter Values".

If you wish to Copy only the configurations from a location to all other locations where the same IP Phone is connected, make sure you clear the Device Type check box.

If you wish to copy the Device Type as well as all the configurations from a location to all the SIP Extensions, make sure all the check boxes are selected.

Standard SIP Authorization Profile

The Standard SIP Authorization Profile contains the list of default profiles of various Standard SIP Phones supported by the system.

Each Profile consists of details which you must configure for successful registration of the phone. Thus, the Standard SIP Authorization Profile ensures that only authorized phones are used as extensions of the system.

Using the Standard SIP Authorization Profile, you can register

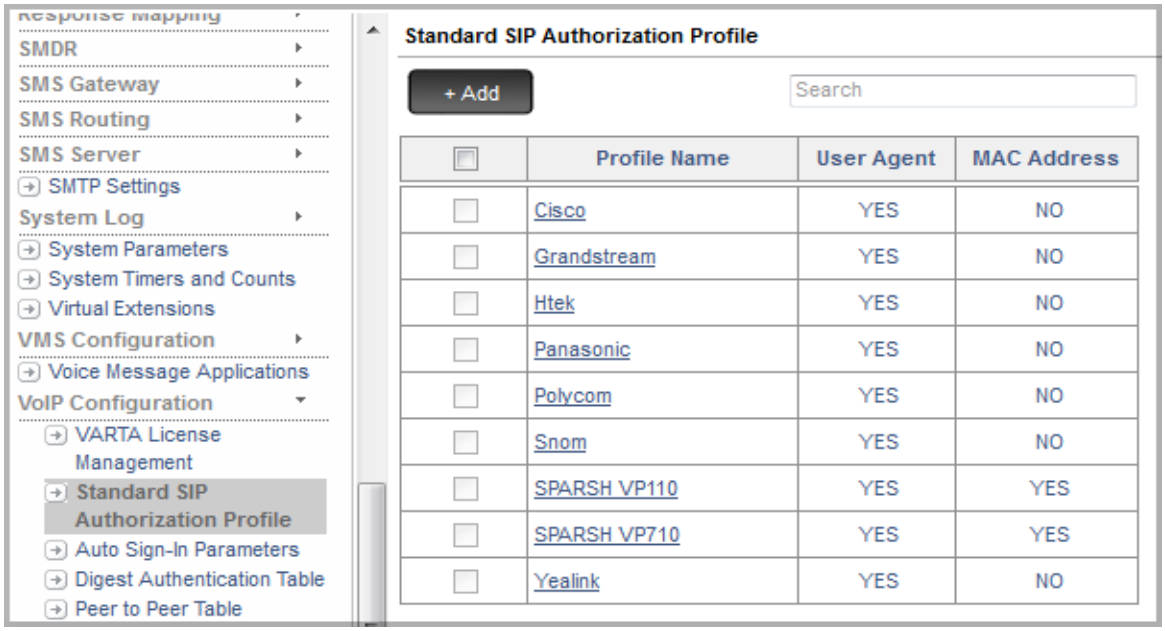
- Matrix SPARSH VP110
- Matrix SPARSH VP710
- Third-party Standard SIP Phone
- Any Standard SIP Phone

When you configure the Device Type in Location - 1/2/3 of **SIP Extension Settings**, the default Standard SIP Authorization Profile is assigned to the phone. However, you may change the profile by selecting the desired option in **Standard SIP Authorization Profile** drop down list. To know more, refer “Configuring Standard SIP Phones”.

Any changes in the assigned Standard SIP Authorization Profile may unregister the phones thereby, causing drop of ongoing calls.

Configuring Standard SIP Authorization Profile using Jeeves

- Login as System Engineer.
- Under **Configuration**, click **VoIP Configuration**.
- Click **Standard SIP Authorization Profile**.



The list of default profiles supported by the system is displayed.

You can also add, edit, search or delete the Standard SIP Authorization Profiles according to your preference.

You cannot delete the default Standard SIP Authorization Profiles supported by the system.

Let us consider, that you want to register a Cisco Standard SIP Phone,

- Click **Cisco**.

The Standard SIP Authorization Profile details are displayed.

- **Profile Name** displays the name of the profile of the Standard SIP Phone.
- Select the **Validate User Agent** check box if you want the system to validate the User Agent received during phone registration request. Default: Enabled.
- In **User Agent**, enter the details which you want the system to match with the User Agent field received from the phone. This parameter is applicable only if you have enabled the **Validate User Agent** check box.
- Select the **Validate MAC Address** check box if you want the system to validate the MAC Address received during phone registration request.

*You are recommended to enable **Validate MAC Address** to prevent any unauthorized access and misuse of the system.*

  - In **Fetch MAC Address From**, select the desired option - **User Agent** or **Custom Header**.

    If you select **User Agent**, the system will fetch the MAC Address from the User Agent field received during phone registration request.

    If you select **Custom Header**, the system will fetch the MAC Address from the configured Custom Header field during phone registration request.

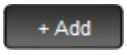
    - In **Custom Header**, enter the Header name from where the MAC Address is to be fetched. For example, MAC. This parameter is applicable only if you have selected the option Custom Header in Fetch MAC Address From.

- Click **Submit**.

Similarly, you can configure the **Standard SIP Authorization Profile** of any Standard SIP Phone.

Customizing Profile

To add a new Standard SIP Authorization Profile,

- Click 

and enter the details as per your requirement.
- Click **Submit**.

  The newly added Profile will get updated in the list.

To delete a Profile,

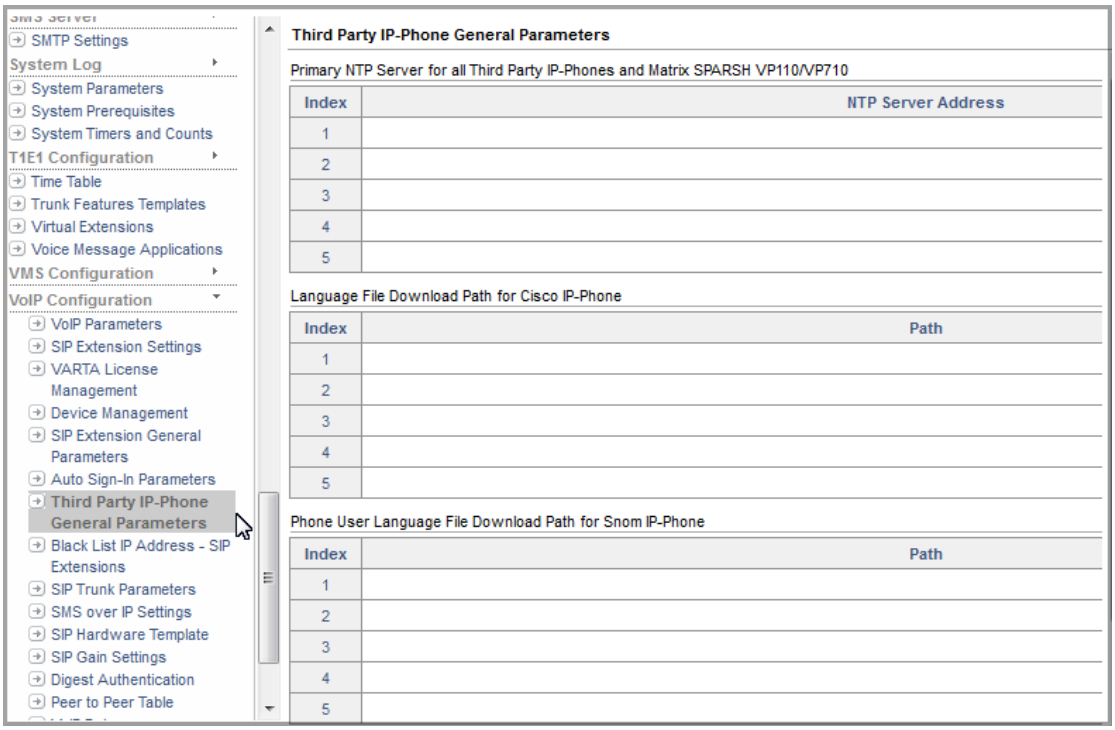
- Select the corresponding check box and click .

Third Party IP-Phone General Parameters

To configure the Third Party IP-Phone General Parameters,

- Under **Configuration**, click **VoIP Configuration**.
- Click **Third Party IP-Phone General Parameters** link.

  The Third Party IP-Phone General Parameters page opens.



Primary NTP Server for all Third Party IP-Phones and Matrix SPARSH VP110/VP710

- Under **NTP Server Address**, configure the Primary NTP Server Address with which you want to synchronize the Date and Time of the Standard SIP Phones. Default: Blank.

  You may configure maximum 5 different Server Addresses.

Language File Download Path for Cisco IP-Phone

- Under **Path**, configure the path from which you want the Standard SIP Phone to fetch the Language files. Default: Blank.

  You may configure maximum 5 different Paths.

Phone User Language File Download Path for Snom IP-Phone

- Under **Path**, configure the path from which you want the Standard SIP Phone to fetch the Language files for the Phone user. Default: Blank.
  You may configure maximum 5 different Paths.

Web User Language File Download Path for Snom IP-Phone

- Under **Path**, configure the path from which you want the Standard SIP Phone to fetch the Language files for the Web user. Default: Blank.
  You may configure maximum 5 different Paths.

- Click **Submit** to save changes.

The Server Addresses/Path you configure on this page will appear in the Combo box of the respective parameter on the SIP Extension page.

Device Management

SARVAM UCS supports Auto Detection and Auto Provisioning of third party IP Phones as well as Extended Clients. Device Management is a touch-free, plug and play feature and is an ideal solution for a large deployment of phones. To use this feature you must plug all the IP phones in same network (recommended) and also make sure that these phones are registered as SIP Extensions at **Location 1** only. The system will support Auto Detection and provisioning of all these phones (third party IP Phones and Extended Clients).

Once you connect the phones the details will be displayed on the Device Management page.To use this feature,

- Log into Jeeves as System Engineer.
- Under **Configuration**, click **VoIP Configuration**.
- Click **Device Management.**

- The following information will be displayed for each connected phone:
  - **Device Name:** It displays the name of the phone you connected/application you register.
  - **MAC Address / Device ID:** It displays the MAC or Device ID of the phone.
  - **IP Address:** It displays the IP Address of the phone.
  - **Last Seen:** It displays the date and time, when the system detects the connected phone.
  - **Assign:** If the phone is already configured, it displays the Name, Extension Number and Location Number. If you wish to edit the details, click on this detail link.

    If the device is not configured, it displays 'Not Assigned'. If you wish to assign an extension to the phone, click on the **Not Assigned** link.

    A new **Assign Extension** window opens. You can configure/edit the details:
    - **Select SIP Extension**: Select the SIP Extension Number you want to assign to the phone.
    - **Name**: Enter a name for the SIP Extension, which may be the name of the person who will use the SIP Extension or the name of a Department. The name you enter here will be displayed as the Caller ID of the SIP Extension on the remote user's phone, when the SIP Extension user makes calls. The name may consist of a maximum of 18 alphanumeric characters.
    - **SIP ID:** Enter the SIP ID for the extension. The SIP ID is necessary for registering the SIP Extension with the Registrar of the system. It is the number with which you can call the SIP Extension. Any extension user of the SARVAM UCS can call a SIP Extension by dialing the SIP ID

assigned to the SIP extension. SIP ID of each SIP Extension must be a unique number string of a maximum of 6 digits. Any combination of digits from 0 to 9 and the characters * and # are allowed.

You cannot assign the same SIP ID to more than one extension.

- **Authentication ID:** Enter the number which you want the system to use for user authentication of the SIP messages received from the SIP Extension. You cannot keep this field blank. The number may be a string of maximum 6 alphanumeric characters. All ASCII characters except < > and “ (double quote) are allowed.Default: Blank.

You must configure the Authentication ID, if any of the SIP Message Authentication Options, namely INVITE or SUBSCRIBE or PUBLISH, is enabled.

- In the **Authentication Password** field, enter the password manually or click **Generate** to automatically generate a unique password. This password will be used by the system to authenticate the SIP messages received from the SIP Extension.

  To avoid unauthorized access, we recommend you to change the password regularly. Make sure it is strong and is kept confidential. When you enter the password manually, it must:
  - be of minimum 6 characters and can be a maximum of 12 characters.
  - include atleast one upper-case, one lower-case, one number and one special character.
  - all ASCII characters (except Percentage **%**, Hash **#,** Equal to **=,** Plus **+**, And **&**, Backslash **\,** Less than **<,** Greater than **>,** Apostrophe **’,** Double Quote **"** and **Space**) are allowed. Default: Blank.

  To provide additional security, when the Authentication fails 10 times consecutively due to wrong Authentication ID / Authentication Password, the system will blacklist the IP Address and Port for registration of this SIP Extension. However, you can remove the IP Address from the Blacklist IP Address list. See “Black List IP Address - SIP Extensions” for more details. This activity will be logged in the “System Activity Log” as well as “Simple Network Management Protocol (SNMP)”.

- **Select Location:** Select the location 1, 2 or 3 on which you want the extension to register.

- **Location Name:** Enter the name that you want the system to display for the location you selected.

- **Registrar Server Address:** Select the appropriate Registrar Server Address to register the device with the SIP Registrar of SARVAM UCS, according to your installation scenario.

- Click **Submit**. The window closes and the details are displays on the Device Management page.

- **Reboot:** Click this button to reboot the phone remotely.

The phone will reboot only if it supports remote reboot.

For Grandstream, Snom and Polycom IP Phones reboot is not supported.

- Click **Clear All Assigned Entries**, to clear all the assignments.

Black List IP Address - SIP Extensions

Blacklist IP Address enables you to restrict unauthorized access to SARVAM UCS.

To use this feature, you must enable the *Allow SIP Extension Registration* check box and select the desired option for *Black List SIP Extension IP Address:Port on multiple Authentication Failure Attempts* in the "Security Settings". The SARVAM UCS blacklists the IP Address from which an unauthorized attempt is made for registration. When any user attempts to register as a SIP Extension using false credentials— Authentication ID or Authentication Password and the authentication attempt fails for 10 times consecutively within 10 minutes, SARVAM UCS blacklists the IP Address and port used for registration.

The blacklisted IP Address/es and ports are stored in the **Black List IP Address - SIP Extensions** table along with the date and time. This activity will also be logged in the "System Activity Log" as well as "Simple Network Management Protocol (SNMP)".

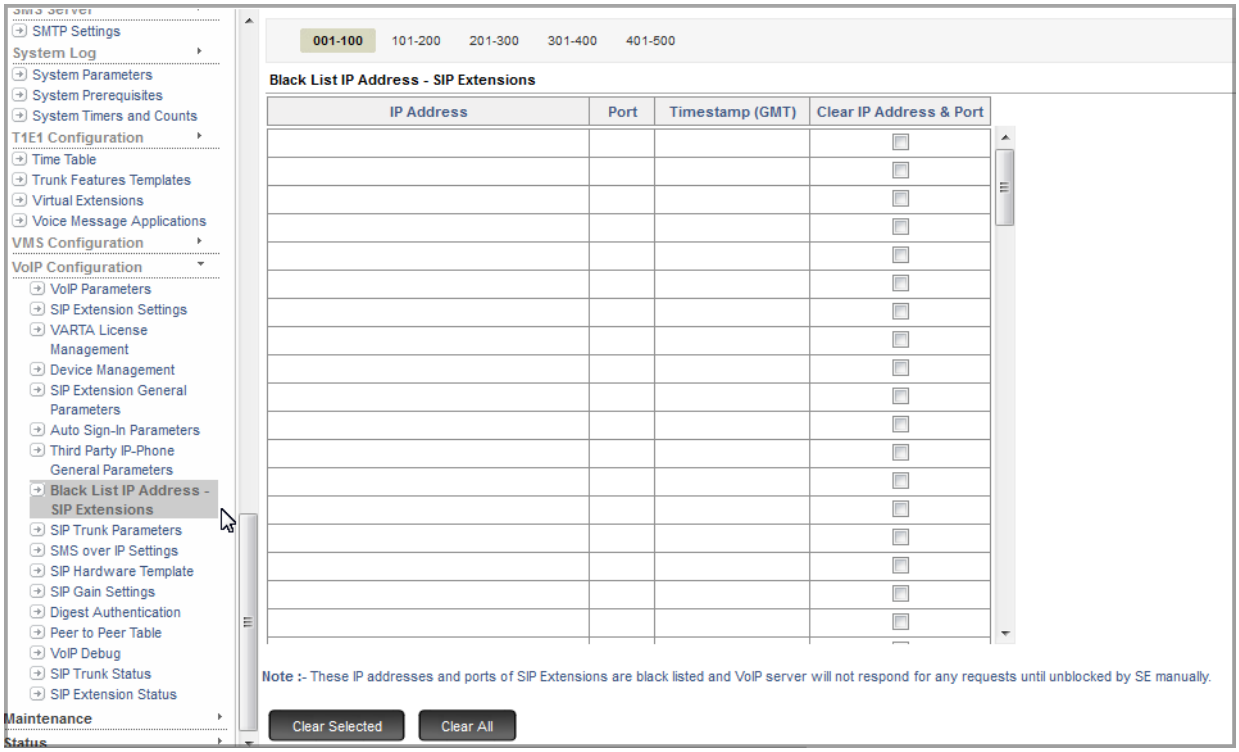
If you want to allow access of SARVAM UCS to such black listed IP Address, you must remove it from the **Black List IP Address - SIP Extensions** table. If this IP Address is a Trusted IP Address, you can configure it in the Trusted IP Address/es table to avoid further black listing. For details, see "Security Settings".

Blacklist IP Address table stores upto 500 entries. When the buffer is full, SARVAM UCS follows FIFO method to store further entries. You cannot edit any entries in this table. However, if required, you may remove a entry from this table.

To clear a entry from the **Black List IP Address - SIP Extensions** table,

- Log in as System Engineer.
- Under **Configuration**, click **VoIP Configuration**.
- Click the **Black List IP Address - SIP Extensions** link.

The Black List IP Address - SIP Extensions page opens.



- Select the **Clear IP Address & Port** check box of the entries that you want to remove from the Black List IP Address - SIP Extensions table.
- Click the **Clear Selected** button. The selected entries will be removed from the table.
- Click **Clear All** button to clear all the entries stored in the table.

Configuring DKP Extensions

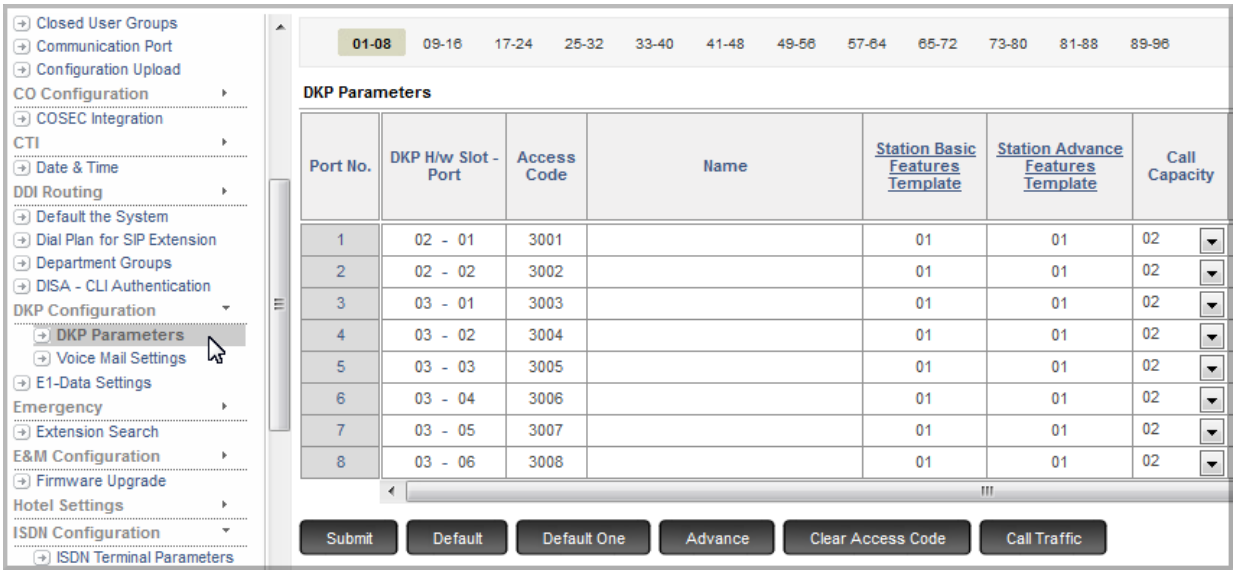
The number of DKP extensions available to you for configuration depends on the number of DKP ports supported by SARVAM UCS and the number of DKP ports you have specified on the "System Pre-requisites" page.

If you have enabled 'On-Site Configuration', the system will provide you only those ports that are actually present in the system for configuration.

Configure **DKP port parameters** using Jeeves or by dialing commands from a Telephone.

Configuring DKP Parameters using Jeeves

- Log into Jeeves as System Engineer.
- Under **Configuration**, click **DKP Configuration**.
- Click **DKP Parameters** to open the page.



The DKP Ports appear in tabs, with eight DKP Ports in each tab, 001-008, 009-016, 017-024, and so forth.

- For each DKP Port number, configure the following parameters:
  - **DKP H/w Slot - Port:** 'Slot' is the number of the Universal Slot in which the DKP Card is inserted. 'Port' is the number of the DKP hardware port on which the proprietary DKP EON is connected.

    The SARVAM UCS can automatically detect and assign the hardware slot and port numbers automatically to the DKP software ports.

    For example: if you have inserted the DKP8 Card in Slot number 03 and DKP16 Card in Slot number 04 of SARVAM UCS, the system will assign the hardware slot 03 and port numbers 01-08 to the DKP Software Ports from 001 to 008 respectively. The system will assign Slot 04 and port numbers 09-16 to the DKP Software Ports 009 to 024. Refer the topic "Software Port and Hardware ID" to know more.

    However, if required, you may change the Hardware Slot and Port assigned to the DKP software port. In this case, enter the desired Hardware Slot and Port number in this field.

If you want to de-assign the Hardware Slot and Port, Enter '00' in both fields.

- **Access Code:** Assign Station Access Codes to the DKP Port. Station Access Codes are commonly referred to as Extension Numbers. These may be a maximum of 6 digits, which are dialed to call the DKP port to which they are assigned.

  All DKP ports are assigned the following Station Access Codes as default.

DKP (Software) Port No.	Access Codes
001	3001
002	3002
003	3003
:	:
:	:
128	3128

  You may either apply the default Station Access Codes to the DKP ports or assign them according to your requirement and preferences.

  To assign Station Access Codes according to your preference and requirement to a range of DKP Ports, see “Assigning Access Codes to a Range of Extensions”.

If you decide to customize the Station Access Codes, make sure that the numbers do not clash with any other Access Code in the 'Dial' phase. Refer the topics “Access Codes” and “Conflict Dialing” to know more.

- **Name:** Assign a 'Name' to the DKP port. The name may be of the person who will use the DKP or the name of a department. This name will be displayed on the LCD of the DKP of the user and on other extension user's phones, provided these are also a model or EON or are equipped with Caller ID.

  You can program a name of maximum 18 alphanumeric characters.

- **Station Basic Feature Template:** Assign a “Station Basic Feature Template” to the DKP.

  A Station Basic Feature Template includes a set of features that completely define the behaviour of the Extension, such as Time Table, Operator access, Trunk Access, Class of Service, Toll Control, Call Budget, and Station Message Detail Records (storage of Incoming and Outgoing Call details).

  By default, Station Basic Feature Template 01 is assigned to all extensions of the system that includes DKP ports as well as DKP ports, ISDN Terminals, E&M Lines with Station as Orientation Type, and SIP Extensions.

  Check if the default template fulfills the feature requirements (like “Class of Service (COS)”, “Toll Control”, “OG Trunk Bundle Group”, etc.) of the DKP.

  If the default Template 01 fulfills the feature requirements and if the same features are to be allowed to all DKPs, retain Template 01.

  If different sets of features are to be allowed to different DKPs, then prepare separate Station Basic Feature Templates and apply them on the ports. To do this,

- Under **Configuration**, click the link “Station Basic Feature Template” to open the page.
- Select a Template number, for example 11.
- Customize Template number 11 and click 'Submit' at the bottom of the page.
- Now go back to the **DKP Parameters** page.
- Enter the number of the Template you customized, Template 11, in the **Station Basic Feature Template** field of the DKP Port, for instance DKP-005, on which you want to apply this template. If you want to apply this template to other ports too, like DKP-006, 007, and 008, assign the Template 11 to all these ports.
- Click **Submit** at the bottom of the page to save changes.
- Repeat the same steps to customize and assign a different Template to another DKP port.

  Also, refer the topic Station Basic Feature Templates to know more about customizing the templates and applying on the ports.

- **Station Advanced Feature Template:** Assign a Station Advanced Feature Template to the DKP. The Advanced Feature Template consists of a set of less commonly used features like Alarm Notification Type, Caller ID Presentation for Transferred Calls, DDI Incoming Call Routing, Storage of Internal Calls, Call Duration Control, Floor Service, Call Taping, etc.

  By default Station Advanced Feature Template 01 is assigned to all extensions of the SARVAM UCS, which includes DKP Ports, SLT Ports, ISDN Terminals, E&M Lines configured as Stations, and SIP Extensions.

  Check if this default template fulfills the feature requirements of the DKP Ports.

  If the default Template 01 fulfills the feature requirements, and if the same features are to be allowed to all DKP ports, retain Template 01.

  If different sets of features are to be allowed to different DKP Ports, then prepare separate Station Advanced Feature Templates and apply them on the ports.

  To do this,

  - Under **Configuration**, click **Station Advanced Feature Template** to open the page.
  - Select a Template number, for example 02.
  - Customize Template number 02.
  - Click **Submit** at the bottom of the page.
  - Now go back to the **DKP Parameters** page.
  - Enter the number of the Template you customized, Template 02 in the **Station Advanced Feature Template** field of the DKP Port, for instance DKP-001, on which you want to apply this template. If you want to apply this template to other ports too, like DKP-002, 003, and 004, assign the Template 04 to all these ports.

- Click **Submit** at the bottom of the page to save changes.
- Repeat the same steps to customize and assign a different Template to another DKP port.

  Also refer the topic “Station Advanced Feature Template” for instructions on customizing these templates and applying them on the ports.

- **Call Capacity:** Call Capacity is the number of Call Appearances (also referred to as 'call loops') assigned to a (DKP) extension. It is the ability of an Extension to handle multiple calls simultaneously. A Call Appearance allows an extension user to attend to more than one calling party at a time.

  A minimum of two Call Appearances must be assigned to a DKP Extension - Operator extension or Executive extension - so that the Extension user can put one party on hold while talking to another. A third Call Appearance allows the extension user to put two calls on hold, make/attend a third call and toggle between three calls.

  The higher the call capacity (the more the number of Call Appearances assigned to an extension), the more the number of calls the Extension user can handle.

  The SARVAM UCS supports a maximum of 10 Call Appearances as Call Capacity of the DKP extensions.

  DKP extensions for Executives are usually assigned 2 Call Appearances, while the Operator Extension is assigned 6 Call Appearances to handle 6 calls simultaneously. The default call capacity of the DKP ports is 02.

  Now, select the number of Call Appearances you wish to assign to the DKP port in the column, 'Call Capacity'.

- **Call Waiting Tone:** During an on-going conversation, if there is a second incoming call, the system plays beeps to indicate the second incoming call. You can set the frequency of the beeps as per your requirement. You can select from the following options:
  - Off
  - Beep Once
  - Beep until Answered

  Default: Beep Once

- **Key Map:** EON is designed to function as Operator, Executive, Hotel Attendant, and Hotel Guest extensions, providing default key settings (key maps) for all these functions. All you need to do is assign a Key Map Template according to the intended user of the DKP. For example, if the DKP is to be used by the Operator, select ‘Operator's Template’. The DKP will be assigned the key template with the special features required by Operators, such as more DSS keys for Trunk Access and Call Appearances, a Call Release Key, etc.

  Similarly, if the user of the DKP is a Hotel Attendant, select 'Hotel Attendant's Template'. The key map with the specific Front Desk User features such as Check-In, Check-Out, Guest In/Out, Change Room Clean Status, Room Shift, will be automatically assigned to the DKP.

  The option ’Personalized’ allows you to customize the key map of the DKP and assign functions to keys as per your requirement. Click **DSS Keys** to assign the features and facilities as per your requirement.

To know more about key maps, key templates and how to customize them, see “DSS Keys Programming”.

- **Call Pick-Up Group:** Program this parameter if you want to assign the DKP to a particular **“**Call Pick Up” group.

  Call Pick Up allows the DKP extension user to 'pick up' (answer) calls ringing on any other extension, by dialing a feature code, without physically going to the ringing extension. For this to work, both the ringing extension and the extension picking up the call must be in the same 'Call Pick Up Group'. Refer “Call Pick Up” for instructions on how to create groups. You can create as many as 99 groups, numbered from 01 to 99.

  Enter the number of the Call Pick-Up Group you created for this DKP in this field.

- You must assign the extension user to a **COSEC Door Group** for COSEC Integration. The users in the same group must be assigned the same group. You can create as many as 50 groups numbered from 00 to 50. Users who are assigned COSEC Door Group ‘00’ are not a part of any group. See “COSEC Integration” for more information.

- **Station Type:** Make sure you select the option Assistant, if it is the Operator Extension. The system will play beeps during the conference to the participants to indicate the presence or absence of the Operator. To know more, see “Conference-3 Party”, “Conference-Multiparty” and “Conference Dial-In”.

  For the Hotel Application of the SARVAM UCS, extensions are identified as — Administrator, Assistant or Guest extensions according to the intended user of the DKP. The system will consider the Administrator and Assistant options as same. When the Station Type is selected, the system will automatically assign the Guest and Administrator (Hotel Staff) features to the DKP. To know more, refer the *SARVAM UCS Hospitality System Manual*.

- **DSS Key Settings:** This parameter is to be configured if you wish to Personalize the phone Key Map or have attached a Direct Station Selection Console with the DKP.

  To Personalize the phone Key Map, refer to “Personalizing Key Maps” in “DSS Keys Programming”

  Matrix offers two models of DSS — DSS64, DSS532.

  If you are using EON48/310 you can attach two DSS64 to increase the number of DSS keys for Direct Station Calling or for one touch feature access.

  If you are using EON510 you can attach four DSS532 to increase the number of DSS keys for Direct Station Calling or for one touch feature access.

  Refer the topic “Installing DSS532 with EON510” for installation instructions.

  For instructions on configuring the DSS Keys of the Consoles, refer to “Programming DSS Console Keys” in “DSS Keys Programming”.

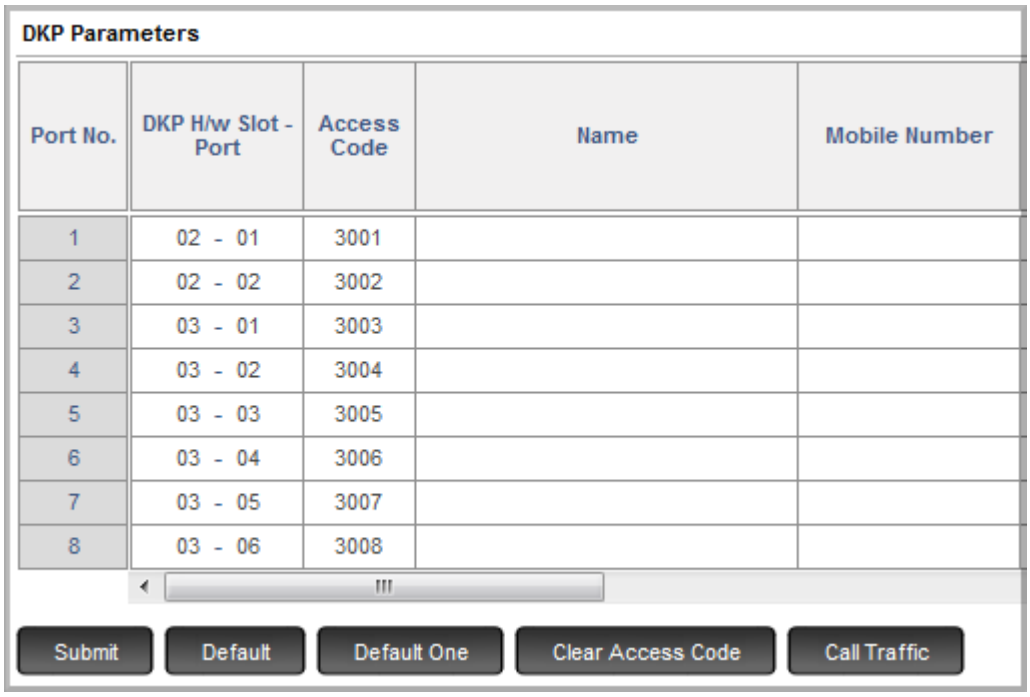
- In **Voice Mail Settings**, click the Voice Mail Settings link. The respective Extension Voice Mail Settings window will open. You may edit the parameters. For details, see “Extension Voice Mail Settings”.

The Voice Mail Settings link will be visible only if you have configured the respective access codes.

Click **Close** to close the window.

Advanced Configuration

The above listed parameters fulfill the basic DKP extension port configuration requirements of most users. However, it is anticipated that some users may need to configure the more advanced features like Personal Directory, Language Selection, DKP Settings - Backlight Brightness and Contrast, Headset/Handset/Speaker Volume levels, Select a Ringer Tune, attach Direct Station Selection Consoles, user details etc. For such users, you may click the 'Advanced' button.



- Configure the following advanced DKP Parameters:
- **Mobile Number:** Enter the Mobile Number of the extension user you wish to store. The Number can be a maximum of 16 digits.
- **Email ID:** Enter the Email ID of the extension user you wish to store. The Email ID can be a maximum of 64 characters.
- **SMS/Email Group Type:** You can assign the extension user to a Group. Select the desired **SMS/Email Group Type** from the list.The system clubs together extension users assigned the same Group. Default: None. For details, see “SMS/Email Group”.
- **Personal Directory:** Enter the number of the **“**Personal Directory**”** that you want to assign to the DKP. A Personal Directory is a list of 25 frequently dialed numbers, each of which are stored by Index number (location code), Name and Trunk Access Codes (“OG Trunk Bundle Group”). The Personal Directory is necessary for using the features “Abbreviated Dialing” and “Dial By Name”.

  To be able to assign a Personal Directory to a DKP you must first program it. Refer the topic “Abbreviated Dialing” for instructions on programming the Personal Directory.
- **Priority:** Select a Priority Level for the DKP from 1, 2, 3... to 9, with '1' being lowest Priority and '9' being highest Priority. Whenever an extension (DKP) with higher priority calls an extension with lower priority, a triple ring is placed on the called extension. To know more, read the feature description “Priority”.

By default, the Priority of all DKP ports is set to '5-Normal'.

- **CO CLIP Pattern:** This parameter allows you to select the type of Calling Line Presentation on the DKP for incoming calls from trunks. You can select any of the below options:
  - Name Only: only the name of the caller will be displayed.
  - Number Only: only the number of the caller will be displayed.
  - Number + Name: both the name and the number of the caller will be displayed.

  By default, Number + Name is selected as the CO CLIP Pattern for all DKPs.

- **Language:** SARVAM UCS provides language support for English, French, German, Spanish, Portuguese, and Italian. When you select any of these languages, all the command strings and prompts will appear in the selected language. By default English is selected.

DKP users can change the language by accessing and navigating through the phone menu.

The SA can change the Language by logging into the SA Jeeves.

- **Ringer Mode:** You can select a Ringer mode for each DKP from the four options:

  - Ring immediately (it rings immediately as a fresh calls lands on the DKP).
  - Ring if idle (rings only if the DKP is idle).
  - Ring after a delay (if the call is still not answered).
  - Silent.

  By default the Ringer Mode is set to 'Ring Immediately'. Change the Ringer Mode on the DKP as per the requirement and preferences of the DKP users.

- **Ring Delay Timer:** The Ring Delay Timer is the time in seconds the SARVAM UCS will wait to ring on receiving a call. This Timer needs to be set only if you have selected 'Ring after a delay' as the Ringer Mode for the DKP in the previous parameter.

  By default, the Ringer Delay Timer is set to 10 seconds. You may change the Ring Delay Timer according to the preferences and requirements of the DKP user.

- **Acknowledge Timer:** This Timer is to be programmed to enable the Ringer Auto Acknowledge mode. This mode determines when to stop the ring on the DKP. There are two options for ringer auto acknowledge:

  - Stop only when the call is answered.
  - Stop after a delay.

  To stop the ring on the DKP after a delay, the Acknowledge Timer must be programmed. The range of the timer is 01 to 99 seconds.

  To stop the ring only when the Call is answered or manually acknowledged, the Acknowledge Timer must be set to '00'. By default, Ring Auto Acknowledge is turned OFF.

- **Play Ring ON:** With this parameter, you can assign the Ring Destination for the DKP; you can choose whether the Ring should be played on the Speakerphone or Headset of the DKP. Default: Speakerphone is selected.

When you select the Headset as the destination, ensure that you have enabled Headset Connectivity flag and have connected a Headset to the DKP.

The speech path of both, the Headset and the Handset is common. If the Headset is not connected and you have selected the Headset as the ring destination, the ring will be played on the speaker of the Handset.

- **Ring Tune:** You can select from different ringer tunes for each DKP according to your preferences and requirement. By default, Ringer Tune is set to 1.
- **Ring Volume:** You can select from different ringer volumes for each DKP according to your preferences and requirement. By default, Ringer Volume is set to 5.
- **Handset Transmit Volume Level:** This parameter is used for increasing or decreasing the volume of outgoing speech (Transmit Gain) on the Handset of the DKP. Select the desired Handset Tx Volume Level from 0 to 9. By default, Handset Tx Volume Level is 4.
- **Handset Receive Volume Level:** This parameter is used for increasing or decreasing the volume of incoming speech (Receive Gain) on the Handset of the DKP. Select the desired Handset Rx Volume Level from 0 to 9. By default, Handset Rx Volume Level is 4.
- **Headset Transmit Volume Level:** This parameter is used for increasing or decreasing the volume of outgoing speech (Transmit Gain) on the Headset port of the DKP. Select the desired Headset Tx Volume Level from 0 to 9. By default, Headset Tx Volume Level is 4.
- **Headset Receive Volume Level:** This parameter is used for increasing or decreasing the volume of incoming speech (Receive Gain) on the Headset port of the DKP. Select the desired Headset Rx Volume Level from 0 to 9. By default, Headset Rx Volume Level is 4.
- **Hands-free Transmit Volume Level:** With this parameter you may change the Volume level of Transmit Gain of the Speaker phone MIC volume from 0 to 9, as per your preference. This parameter is to be used for increasing or decreasing the volume levels of outgoing speech on the Speaker of the DKP. By default, Hands-free Tx volume level is 4.
- **Hands-free Receive Volume Level:** With this parameter you may change the Volume level of Receive Gain of the Speaker phone MIC volume from 0 to 9, as per your preference. This parameter is to be used for increasing or decreasing the volume levels of incoming speech on the Speaker of the DKP. By default, Hands-free Rx volume level is 4.
- **Handsfree Parameters (For EON510):** These parameters are applicable only when EON510 is connected to the DKP Port.
  - **High Gain Mode:** Select this check box if you want to increase the volume of incoming speech in Handsfree mode (Speaker). By default, this check box is disabled
  - **AGC:** This parameter can be configured only if High Gain Mode check box is disabled.

    There may be disturbances during speech over CO Trunks, to resolve this you may select **ON for CO Ports** as the AGC option.

    If there are disturbances during speech over any trunk, select **ON for all ports** as the AGC option.

By default, AGC is OFF.

Handsfree Parameters (For EON510) is supported in EON510 firmware version V3R2 onwards.

If Handsfree High gain Mode is enabled, then the two-way speech quality may get degraded.

- **Handset High Gain Mode:** Select the check box, if you wish to increase the incoming speech volume on the handset. By default, it is disabled. This is useful for individuals with hearing aids.

This is applicable for EON510 and SPARSH VP510.

This is feature can also be enabled for SPARSH VP210, for details refer to the SPARSH VP210 (Extended) User Guide.

- **Key Click Volume Level:** You may change the Key Click Volume (Key DTMF Side tone) of the DKP. Key Click Volume is the tone you hear as you press the dial pad keys of EON. Select the desired volume level from 0 to 9. By default, the volume level is set to 5.

- **DTMF Generation Flag:** This flag is used to enable or disable DTMF dialing on the DKP. By default, the flag is enabled. To disable the flag, click the check box.

- **DTMF Transmit Level:** You can select the desired Transmit Level from 0 to 9 for DTMF generation from the DKP. By default, the DTMF Transmit Level is set to 2.

- **Headset Connected?:** Enable this parameter by selecting the check box if you want to use a Headset with the DKP.

  Make sure that you have also connected a Headset to the DKP.

- **Auto Answer:** Enable this parameter by selecting the check box if you want to set the "Auto Answer" feature on the DKP.
  When this feature is set, the DKP goes OFF-Hook automatically after a preset period of time, without the user having to pick up the handset or press the speaker or headset key.

- **Auto Answer Timer (sec):** This parameter is to be programmed if you have enabled the "Auto Answer" feature on the DKP.

  When the Auto Answer feature is enabled, the Auto Answer Timer must be programmed. This timer defines the time in seconds that the DKP should wait before going OFF-Hook. The range of this timer is 1 to 9 seconds. By default, the Auto Answer Timer is set to 1 second.

- **LCD Backlight Level:** You can change the LCD Backlight Brightness of EON. The intensity of the backlight brightness increases from 0 to 4, where '0' will cause the backlight to be turned OFF. '1' signifies minimum intensity, '4' signifies maximum intensity. Select any of the levels from 1 to 4 from the list.

- **LCD Backlight OFF Timer:** The backlight of the LCD display of EON can be kept switched ON continuously, or can be set to switch OFF automatically after a predefined period of time, known as the Backlight OFF Timer. The range of the Backlight OFF Timer is 000-999 seconds. By default it is set to switch OFF after 010 seconds.

- **LCD Contrast Level:** The EON offers 4-level contrast control for its LCD display. Level 1 signifies minimum and level 4 is the maximum. The contrast increases in steps of 1 to 4. By default the contrast is set to level 3. You may adjust it to the level comfortable to you. Select a level from 1-4.

- **Line Echo Cancellation (LEC):** Enable this parameter to cancel the echo generated on the other end.

  You must also configure the **Line Echo Cancellation Start Timer**. When DKP users go Off-Hook to make/ receive calls, the Line Echo Cancellation Timer starts. After the expiry of this timer the SARVAM UCS will start Line Echo Cancellation. To configure the Echo Cancellation Start Timer, see "System Timers and Counts".

- *You are recommended to enable Line Echo Cancellation only if you hear echo after you have set the AC Impedance value in the CO Hardware Template according to the Accurate Test conducted by you. For details see, "AC Impedance Test" and "CO Hardware Template".*
- *The Call Progress Tones heard by the DKP user may be affected, if you enable LEC.*
- *LEC is supported only in EON48D, Version V3R15 or later.*

- **CPLD Version:** This parameter displays the CPLD Version of the Digital Key Phone Card.

- The CPLD Version will be displayed only if supported by your DKP Card.

- **Current Firmware:** This parameter displays the current firmware of the Digital Key Phone connected to the DKP Port.

- **Upgrade Firmware:** Enable this parameter by selecting the check box, if you want to upgrade the firmware of the Digital Key Phone connected to the SARVAM UCS.

- *You can upgrade firmware of any two DKP phones at a time. Before upgrading, make sure both these phones are in idle state. You cannot use these phones while their upgradation is in process.*

- *You can view and Upgrade the current firmware of the DKP phone only from Jeeves.*

Configuring DKP Extensions using a Telephone

- Enter SE mode from a DKP/SLT.

  To assign Hardware Slot and Port, dial:
  - ***1102-DKP-Slot-Port offset on the card***
    Where,
    DKP is the Software Port number of the DKP port from 001 to 128.
    Slot is Slot number in which the DKP Card is inserted from 01 to 16
    Port Offset is the number of the DKP Port on the card from 01 to 99.

  To clear the hardware Slot and Port assigned to the DKP software port, dial:
  - ***1102-DKP-00-00***

  To assign an Access Code to a DKP Port, dial:
  - ***3102-1-DKP-Access Code-#****
    Where,
    DKP is the Software Port number of the DKP port from 001 to 128.

Access Code is a number string of any combination of 1, 2, 3, 4, 5 or 6 digits. Terminate the command with #* if the number string has fewer than 6 digits.

To clear the Access codes assigned to the DKP, dial:
- ***3102-1-DKP-#**** to clear the Access Code of a single DKP port.
- ***3102-2-DKP-DKP-#**** to clear the Access Codes of a range of DKP ports.
- ***3102-*-#**** to clear the Access Codes of all DKP Ports.

To assign default Station Access Codes to DKP, dial:
- ***3152-1-DKP*** to default Access Code of a single DKP Port.
- ***3152-2-DKP-DKP*** to default Access Codes of a range of DKP ports.
- ***3152-**** to default Access Codes of all DKP ports.

To assign a Name to a DKP Port, dial:
- ***5403-1-DKP-Name-#**** to assign a Name to a single DKP port.
- ***5403-2-DKP-DKP-Name-#**** to assign the same Name to a range of DKP ports.
- ***5403-*-Name-#**** to assign the same Name to all DKP ports.
  Where,
  DKP is the Software Port number of the DKP port from 001 to 128.
  Name is a string of a maximum of 18 alphanumeric characters. Terminate the commands with #* if the number string has fewer than 18 characters.

To clear the Name of the DKP Port, dial:
- ***5403-1-DKP-#**** to clear the Name of a single DKP port.
- ***5403-2-DKP-DKP-#**** to clear the Names of a range DKP ports.
- ***5403-*-#**** to clear the Names of all DKP ports.

To assign a Station Basic Feature Template to a DKP Port, dial:
- ***5504-1-DKP-Template Number*** to assign a template to a single DKP port.
- ***5504-2-DKP-DKP-Template Number*** to assign the same template to a range of DKP ports.
- ***5504-*-Template Number*** to assign the same template to all DKP ports.
  Where,
  DKP is the Software Port number of the DKP port from 001 to 128.
  Template Number is the number of the Station Basic Feature Template, from 01 to 50. Default: 01

To assign a Station Advanced Feature Template to a DKP Port, dial:
- ***5604-1-DKP-Template Number*** to assign a template to a single DKP port.
- ***5604-2-DKP-DKP-Template Number*** to assign the same template to a range of DKP ports.
- ***5604-*-Template Number*** to assign the same template to all DKP ports.
  Where,
  DKP is the Software Port number of the DKP port from 001 to 128.
  Template Number is the number of the Station Advanced Feature Template, from 01 to 50. Default: 01

To define the Call Capacity of a DKP Port, dial:
- ***1201-1-DKP-Call Capacity*** to define the call capacity for a single DKP port.
- ***1201-2-DKP-DKP- Call Capacity*** to define the same call capacity for a range of DKP ports.
- ***1201-*- Call Capacity*** to define the same call capacity for all DKP ports.
  Where,
  DKP is the Software Port number of the DKP port from 001 to 128.
  Call Capacity is from 02 to 10. Default: 02.

To assign a Key Map for a DKP Port, dial:
- ***1221-1-DKP-DKP Key Template Number*** to assign a Key Map to a single DKP port.

- ***1221-2-DKP-DKP-DKP Key Template Number*** to assign the same Key Map to a range of DKP ports.
- ***1221-*-DKP Key Template Number*** to assign the same Key Map to all DKP ports.
  Where,
  DKP is the Software Port number of the DKP port from 001 to 128.
  DKP Key Template Number is
  1 for Operator's Template
  2 Executive1 Template
  3 Hotel Attendant's Template
  4 Guest's template.
  5 Executive2 Template
  6 Executive3 Template
  Default: 1

To assign the DKP Port to a Call Pick-Up Group, dial:
- ***3902-1-DKP-Call Pick-Up Group*** to assign a single DKP port to a Call Pick-Up group.
- ***3902-2-DKP-DKP-Call Pickup Group*** to assign a range of DKP ports to the same Call Pick-Up group.
- ***3902-*-Call Pickup Group*** to assign all DKP ports to the same Call Pick-Up group.
  Where,
  DKP is the Software Port number of the DKP port from 001 to 128.
  Call Pickup Group is from 00 to 99. Default: 99.

To remove a DKP from a Call Pick Up group, dial:
- ***3902-1-DKP-00*** to remove a single DKP port from a Call Pick-Up group.
- ***3902-2-DKP-DKP-00*** to remove a range of DKP ports from a Call Pick-Up group.
- ***3902-*-00*** to remove all DKP ports from a Call Pick-Up group.

To select a Station Type for a DKP Port, refer the SARVAM UCS Hospitality System Manual.

- For Advanced Configuration of the DKP Ports, use the following commands:

  To assign a Personal Directory to a DKP Port, dial:
  - ***1906-1-DKP-Personal Directory*** to assign directory to a single DKP port.
  - ***1906-2-DKP-DKP- Personal Directory*** to assign the same directory to a range of DKP ports.
  - ***1906-*-Personal Directory*** to assign the same directory to all DKP ports.
    Where,
    DKP is the Software Port number of the DKP port from 001 to 128.
    Personal Directory number is from 01 to 50.

  To clear the Personal Directory assigned to a DKP Port, dial:
  - ***1906-1-DKP-00*** to clear the directory of a single DKP port.
  - ***1906-2-DKP-DKP-00*** to clear the directory of a range of DKP ports.
  - ***1906-*-00*** to clear the directory from all DKP ports.

  To define the Priority for a DKP Port, dial:
  - ***3912-1-DKP-Priority*** to define Priority for a single DKP port.
  - ***3912-2-DKP-DKP-Priority*** to define the same Priority for a range of DKP ports.
  - ***3912-*-Priority*** to define the same Priority for all DKP ports.
    Where,
    DKP is the Software Port number of the DKP port from 001 to 128.
    Priority is from 1 to 9. Default: 5-Normal

  To select CO CLIP Pattern for a DKP Port, dial:

- ***1243-1-DKP-CO CLIP Pattern*** to select the CLIP Pattern for a single DKP port.
- ***1243-2-DKP-DKP-CO CLIP Pattern*** to select the same CLIP Pattern for a range of DKP ports.
- ***1243-*-CO CLIP Pattern*** to select the same CLIP Pattern for all DKP ports.
  Where,
  DKP is the Software Port number of the DKP port from 001 to 128.
  CO CLIP Pattern is
  1 for Name only
  2 for Number only
  3 for Number+Name
  Default: Number+Name

To select Language for a DKP port, dial:
- ***1224-1-DKP-Language*** to select language for a single DKP port.
- ***1224-2-DKP-DKP-Language*** to select the same language for a range of DKP ports.
- ***1224-*-Language*** to select the same language for all DKP ports.
  Where,
  DKP is the Software Port number of the DKP port from 001 to 128.
  Language is
  1 for English
  2 for Français
  3 for Deutsch
  4 for Español
  5 for Português
  6 for Italian
  Default: English

To select Ringer Mode for a DKP Port, dial:
- ***1204-1-DKP-Ringer Mode*** to select Ringer mode for a single DKP port.
- ***1204-2-DKP-DKP-Ringer Mode*** to select Ringer mode for a range of DKP ports.
- ***1204-*-Ringer Mode*** to select Ringer mode for all DKP ports.
  Where,
  DKP is the Software Port number of the DKP port from 001 to 128.
  Ringer Mode is
  1 for Ring Immediate
  2 for Ring if Idle
  3 for Ring after Delay
  4 for Ring OFF
  Default: Ring Immediate

To select Ring Delay Timer for a DKP Port, dial:
- ***1205-1-DKP-Ring Delay Timer*** to select Delay Timer for a single DKP port.
- ***1205-2-DKP-DKP- Ringer Delay Timer*** to select Delay Timer for a range of DKP ports.
- ***1205-*- Ringer Delay Timer*** to select Delay Timer for all DKP ports.
  Where,
  DKP is the Software Port number of the DKP port from 001 to 128.
  Ring Delay Timer is 01 to 99 seconds. Default: 10 seconds.

To set the Acknowledgement Mode, dial:
- ***1206-1-DKP-Ringer Auto Acknowledge Mode*** to select mode for a single DKP port.
- ***1206-2-DKP-DKP-Ringer Auto Acknowledge Mode*** to select the same mode for a range of DKP ports.
- ***1206-*-Ringer Auto Acknowledge Mode*** to select the same mode for all DKP ports.
  Where,

DKP is the Software Port number of the DKP port from 001 to 128.
Ringer Auto Acknowledge Mode is
0 for OFF
1 for ON

To set the Ringer Auto Acknowledge Timer, dial:

- ***1207-1-DKP-Ringer Auto Acknowledge Timer*** to set Timer for a single DKP port.
- ***1207-2-DKP-DKP-Ringer Auto Acknowledge Timer*** to set the same Time for a range of DKP ports.
- ***1207-*-Ringer Auto Acknowledge Timer*** to set the same Time for all DKP ports.
  Where,
  DKP is the Software Port number of the DKP port from 001 to 128.
  Ringer Auto Acknowledge Timer is from 01 to 99 seconds. Default: 00.

To select Destination for 'Play Ring ON' for a DKP Port, dial:

- ***1220-1-DKP-Ring Destination*** to select Play Ring ON destination for a single DKP port.
- ***1220-2-DKP-DKP-Ring Destination*** to select the same Play Ring ON destination for a range of DKP ports.
- ***1220-*-Ring Destination*** to select the same Play Ring ON destination for all DKP ports.
  Where,
  DKP is the Software Port number of the DKP port from 001 to 128.
  Ring Destination is
  1 for Play Ring on Speaker Phone
  2 for Play Ring on Headset

To select Ring Tune for a DKP Port, dial:

- ***1202-1-DKP-Ring Tune*** to select a Tune for a single DKP port.
- ***1202-2-DKP-DKP-Ring Tune*** to select the same Tune for a range of DKP ports.
- ***1202-*-Ring Tune*** to select the same Tune for all DKP ports.
  Where,
  DKP is the Software Port number of the DKP port from 001 to 128.
  Ringer Tune is from 0 to 9. Default: 1.

To set Ringer Volume for a DKP Port, dial:

- ***1203-1-DKP-Ringer Volume*** to set Ringer Volume for a single DKP port.
- ***1203-2-DKP-DKP-Ringer Volume*** to set the same volume level for a range of DKP ports.
- ***1203-*-Ringer Volume*** to set the same volume level for all DKP ports.
  Where,
  DKP is the Software Port number of the DKP port from 001 to 128.
  Ringer Volume is from 0 to 9. Default: 5.

Setting Ringer Volume to '0' will cause the DKP volume to be turned OFF.

To set Handset Transmit (Tx) Volume Level for a DKP, dial:

- ***1208-1-DKP-Handset MIC Volume Level*** to set transmit volume for a single DKP port.
- ***1208-2-DKP-DKP-Handset MIC Volume Level*** to set the same transmit volume level for a range of DKP ports.
- ***1208-*-Handset MIC Volume Level*** to set the same transmit volume level for all DKP ports.
  Where,
  DKP is the Software Port number of the DKP port from 001 to 128.
  Handset MIC Volume Level is from 0 to 9. Default: 5.

Setting volume level to '0' will cause the Handset MIC volume to be turned OFF.

To set Handset Receive (Rx) Volume Level for a DKP, dial:

- ***1209-1-DKP-Handset Speaker Volume Level*** to set receive volume for a single DKP port.
- ***1209-2-DKP-DKP-Handset Speaker Volume Level*** to set the same receive volume level for a range of DKP ports.
- ***1209-*-Handset Speaker Volume Level*** to set the same receive volume level for all DKP ports.

  Where,

  DKP is the Software Port number of the DKP port from 001 to 128.

  Handset Speaker Volume Level from 0 to 9. Default: 5.

Setting volume level to '0' will cause the Handset Speaker volume to be turned OFF.

To set Right Handset Transmit (Tx) Volume Level for DKP Port, dial:

- ***1225-1-DKP-Right Handset MIC Volume Level*** to set transmit volume for a single DKP port.
- ***1225-2-DKP-DKP-Right Handset MIC Volume Level*** to set the same transmit volume level for a range of DKP ports.
- ***1225-*-Right Handset MIC Volume Level*** to set the same transmit volume level for all DKP ports.

  Where,

  DKP is the Software Port number of the DKP port from 001 to 128.

  Handset Volume Level is from 0 to 9. Default: 5.

Setting volume level to '0' will cause the Handset Speaker volume to be turned OFF.

To set Right Handset Receive (Rx) Volume Level for a DKP, dial:

- ***1226-1-DKP-Handset Speaker Volume Level*** to set receive volume for a single DKP port.
- ***1226-2-DKP-DKP-Handset Speaker Volume Level*** to set the same receive volume level for a range of DKP ports.
- ***1226-*-Handset Speaker Volume Level*** to set the same receive volume level for all DKP ports.

  Where,

  DKP is the Software Port number of the DKP port from 001 to 128.

  Handset Speaker Volume Level from 0 to 9. Default: 5.

Setting volume level to '0' will cause the Handset Speaker volume to be turned OFF.

To set Headset Transmit (Tx) Transmit Volume Level for a DKP, dial:

- ***1222-1-DKP-Headset MIC Volume Level*** to set receive volume for a single DKP port.
- ***1222-2-DKP-DKP- Headset MIC Volume Level*** to set the same receive volume level for a range of DKP ports.
- ***1222-*- Headset MIC Volume Level*** to set the same receive volume level for all DKP ports.

  Where,

  DKP is the Software Port number of the DKP port from 001 to 128.

  Headset MIC Volume Level is from 0 to 9. Default: 5.

To set Headset Receive (Rx) Volume Level for a DKP, dial:

- ***1223-1-DKP-Headset Speaker Volume Level*** to set receive volume for a single DKP port.
- ***1223-2-DKP-DKP- Headset Speaker Volume Level*** to set the same receive volume level for a range of DKP ports.
- ***1223-*- Headset Speaker Volume Level*** to set the same receive volume level for all DKP ports.

  Where,

  DKP is the Software Port number of the DKP port from 001 to 128.

  Headset Speaker Volume Level is from 0 to 9. Default: 5.

To set Hands-free Transmit (Tx) Volume Level for a DKP, dial:

- ***1210-1-DKP-Speaker Phone MIC Volume Level*** to set volume level for a single DKP port.

- ***1210-2-DKP-DKP-Speaker Phone MIC Volume Level*** to set the same volume level for a range of DKP ports.
- ***1210-*-Speaker Phone MIC Volume Level*** to set the same volume level for all DKP ports.
  Where,
  DKP is the Software Port number of the DKP port from 001 to 128.
  Speaker Phone MIC Volume Level from 0 to 9. Default: 5.

Setting volume level to '0' will cause the Speaker Phone MIC volume to be turned OFF.

To set Hands-free Receive (Rx) Volume Level for a DKP, dial:
- ***1211-1-DKP-Speaker Phone Speaker Volume Level*** to set volume level for a single DKP port.
- ***1211-2-DKP-DKP-Speaker Phone Speaker Volume Level*** to set the same volume level for a range of DKP ports.
- ***1211-*-Speaker Phone Speaker Volume Level*** to set the same volume level for all DKP ports.
  Where,
  DKP is the Software Port number of the DKP port from 001 to 128.
  Speaker Phone Speaker Volume is from 0 to 9. Default: 5.

To set Key Click Volume Level for a DKP, dial:
- ***1212-1-DKP-Key Click Volume*** to set volume level for a single DKP port.
- ***1212-2-DKP-DKP-Key Click Volume*** to set the same volume level for a range of DKP ports.
- ***1212-*-Key Click Volume*** to set the same volume level for all DKP ports.
  Where,
  DKP is the Software Port number of the DKP port from 001 to 128.
  Key Click Volume Level is from 0 to 9. Default: 5.

To enable/disable DTMF Generation Flag for a DKP, dial:
- ***1241-1-DKP-DTMF Generation*** to enable/disable the flag for a single DKP port.
- ***1241-2-DKP-DKP-DTMF Generation*** to enable/disable the flag for a range of DKP ports.
- ***1241-*-DTMF Generation*** to enable/disable the flag for all DKP ports.
  Where,
  DKP is the Software Port number of the DKP port from 001 to 128.
  DTMF Generation is
  0 for Disable
  1 for Enable
  Default: Enable.

To set DTMF Transmit Level for a DKP, dial:
- ***1218-1-DKP-DTMF Transmit Level*** to set Transmit level for a single DKP port.
- ***1218-2-DKP-DKP-DTMF Transmit Level*** to set Transmit level for a range of DKP ports.
- ***1218-*-DTMF Transmit Level*** to set Transmit level for all DKP ports.
  Where,
  DKP is the Software Port number of the DKP port from 001 to 128.
  DTMF Transmit Level is 0 to 9. Default: 2.

To enable/disable Headset Connectivity for a DKP, dial:
- ***1213-1-DKP-Headset Connectivity Flag*** to enable/disable the flag for a single DKP port.
- ***1213-2-DKP-DKP-Headset Connectivity Flag*** to enable/disable the flag for a range of DKP ports.
- ***1213-*-Headset Connectivity Flag*** to enable/disable the flag for all DKP ports.
  Where,
  DKP is the Software Port number of the DKP port from 001 to 128.
  Headset Connectivity Flag is
  0 for Disable

1 for Enable
Default: Disabled

To enable/disable Auto Answer for a DKP, dial:

- ***1214-1-DKP-Auto Call Answer Mode*** to enable/disable Auto Answer for a single DKP port.
- ***1214-2-DKP-DKP-Auto Call Answer Mode*** to enable/disable Auto Answer for a range of DKP ports.
- ***1214-*-Auto Call Answer Mode*** to enable/disable Auto Answer for all DKP ports.
  Where,
  DKP is the Software Port number of the DKP port from 001 to 128.
  Auto Call Answer Mode is
  1 for OFF (manual)
  0 for ON (auto)
  Default: OFF

To set Auto Answer Timer (sec) for a DKP, dial:

- ***1215-1-DKP-Auto Call Answer Timer*** to set Timer for a single DKP port.
- ***1215-2-DKP-DKP-Auto Call Answer Timer*** to set the same Timer duration for a range of DKP ports.
- ***1215-*-Auto Call Answer Timer*** to set the same Timer duration for all DKP ports.
  Where,
  DKP is the Software Port number of the DKP port from 001 to 128.
  Auto Call Answer Timer is from 1 to 9 seconds. Default: 3 seconds.

To set LCD Back Light Level of a DKP, dial:

- ***1216-1-DKP-LCD Backlight Level*** to set brightness of a single DKP.
- ***1216-2-DKP-DKP-LCD Backlight Level*** to set the same brightness level for a range of DKPs.
- ***1216-*-LCD Backlight Level*** to set the same brightness level for all DKPs.
  Where,
  DKP is the Software Port number of the DKP port from 001 to 128.
  Backlight Level is Brightness level from 1 to 4. Default: 3

To change LCD Backlight OFF Timer of a DKP, dial:

- ***1219-1-DKP-LCD Backlight OFF Timer*** to change the timer for a single DKP.
- ***1219-2-DKP-DKP-LCD Backlight OFF Timer*** to set the same timer duration for a range of DKPs.
- ***1219-*-LCD Backlight OFF Timer*** to set the same timer duration for all DKPs.
  Where,
  DKP is the Software Port number of the DKP port from 001 to 128.
  Backlight OFF Timer is from 000-999 seconds. Default: 010 seconds.

To change LCD Contrast Level of a DKP, dial:

- ***1217-1-DKP-LCD Contrast Level*** to set the Contrast level for a single DKP
- ***1217-2-DKP-DKP-LCD Contrast Level*** to set the same Contrast level for a range of DKPs.
- ***1217-*-LCD Contrast Level*** to set the same Contrast level for all DKPs.
  Where,
  DKP is the Software Port number of the DKP port from 001 to 128.
  LCD Contrast Level is from 1 to 4. Default: 3.

To assign Hardware Slot-Port to DSS connected to a DKP, dial:

- ***1103-DKP-DSS-Slot-Port offset on the card***
  Where,
  DKP is the Software Port number of the DKP port from 001 to 128.
  DSS is 1 for the first DSS Console and 2 for the second DSS Console attached to the DKP
  Slot is the number of the Universal Slot in which the DKP Card is located, to which the DSS Console is connected, from 01 to 16.

Port is the number of the hardware port on the card on which the DSS Console is connected.

To clear the hardware Slot-Port assigned to the DSS software port, dial:

- ***1103-DKP-DSS-00-00***

- Exit SE mode.

DKP extension users can change several phone settings according their preferences and requirement. These are referred to as DKP Personal Settings and include:

- *Ringer Volume*
- *Ringer Tune*
- *Ringer Mode*
- *Ringer Acknowledge Mode*
- *Speech Volume (Transmit/Receive)*
- *User Status (Present/Absent)*
- *Keypad Security (Lock/Open)*
- *Call Answer Type - Manual/Auto*
- *Headset/Handset Connectivity option*

To be able change the DKP personal settings, the DKP users must access and navigate the phone menu. Refer "Digital Key Phone-Operation".

DSS Keys Programming

The DSS (Direct Station Selection) Keys when personalized, provides you quick access to Stations, Trunks, Features/Functions of the SARVAM UCS. A few DSS Keys are provided on the phone itself. The number of DSS Keys provided on the phone depends on the type of phone. For details, see “Digital Key Phone-Operation” and “Extended IP Phone/VARTA UC Client - Operation”.

Matrix offers the DSS Consoles when attached to the phone, works as extensions to the phone. You can customize the DSS Console Keys as per your requirement by assigning the desired features/functions. For details, see “Direct Station Selection Console”.

To configure the DSS Keys on the phone, refer “Customizing Key Templates” and “Personalizing Key Maps”.

To configure the DSS Console Keys, refer “Programming DSS Console Keys”.

Key Templates

EON, the proprietary digital key phone (DKP) of Matrix and SPARSH IP Phones of Matrix, can be the extension of the Operators and Executives in an enterprise, and in hotels, it can be the extension of the Front Desk Attendants and Guests.

Each of these groups of users may require a different set of features on their phones. For example, when EON/SPARSH is an Operator's extension, for efficient call management, more DSS keys may be required for Trunk Access, Call Appearances, Call Release, Direct Station Calling, than for accessing features.

When EON/SPARSH is an Executive's extension, more DSS keys may be required for single-touch access to features, and fewer keys for Trunk Access and Direct Station Calling.

Similarly, when EON/SPARSH is a Hotel Attendant's extension, keys are required for specific hotel functions such as Check-In/Check-Out, Changing Room Clean Status, Room Shift, etc. But, a different set of keys with special functions are required when EON/SPARSH is provided as a guest extension, because guests are allowed only a limited number of features of the SARVAM UCS, such as calling the Front Desk/Floor Service, setting Do Not Disturb, Wake-up Calls, Call Forward, and checking Voice Mail.

Given the varying requirements of these groups of extension users, SARVAM UCS provides programmable Templates of Key Maps for the Operator, Executive1, Executive2, Executive3, Hotel Attendant, and Hotel Guest.

The default Key Maps in these Templates can be customized to match the requirement of the intended user group. The default Key Maps of Executive1, 2 and 3 are the same.

The customized Template is assigned to the DKPs/SPARSH IP Phones. For example, you may customize the Key Template for the Operator and assign it to the Operator DKPs. Likewise, you may customize the Key Template for Executive and assign it to the Executive extensions.

The default Key Maps vary according to the models of the DKP/SPARSH IP Phone in use, as illustrated below.

In case of Extended IP Phones, you can customize the Templates for the Operator, Executive1, Executive2, Executive3, Hotel Attendant, and Hotel Guest as well as add new templates. You can add upto 64 templates. The new template you add, can be edited or remove as per your requirement. However you cannot remove the default Templates — Operator, Executive1, Executive2, Executive3, Hotel Attendant, and Hotel Guest. These can only be customized.

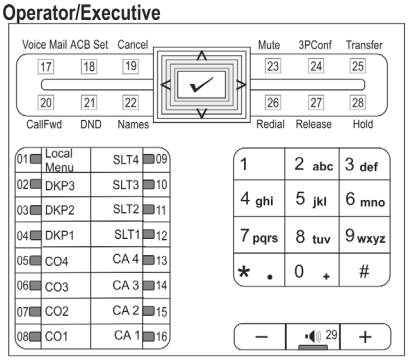
You can also Personalize the DKP/SPARSH IP Phone key maps, see “Personalizing Key Maps” for instructions.

The keys in the Key Templates are numbered only for the purpose of locating the keys when programming. Key numbers do not appear on the key labels on the phone body.

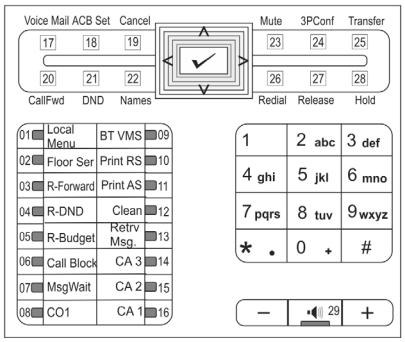
Default Key Maps

SPARSH VP248 Key Template (default)

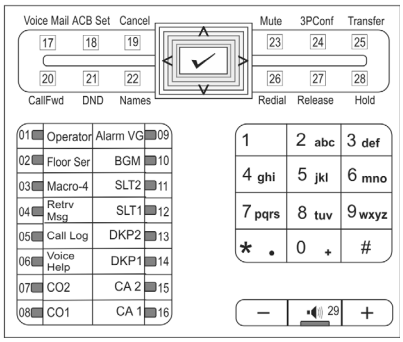
Operator/Executive



Hotel Attendant

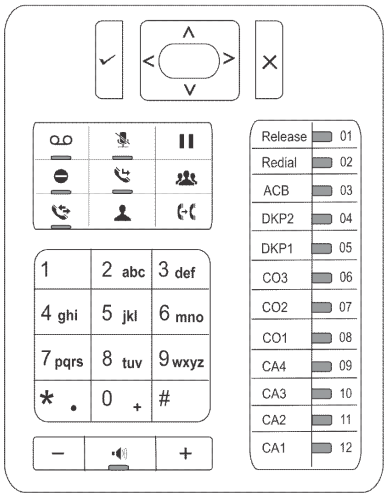


Guest

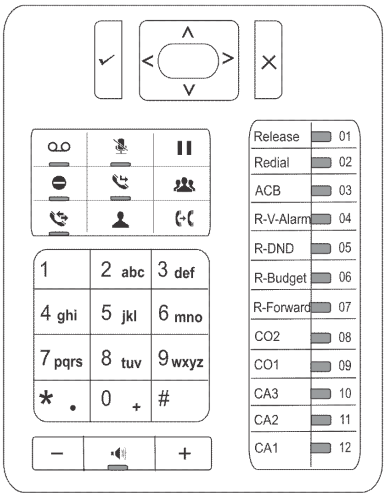


SPARSH VP310 Key Template (default)

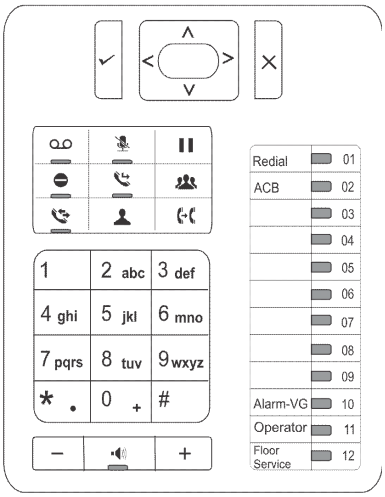
Operator/Executive Key Map



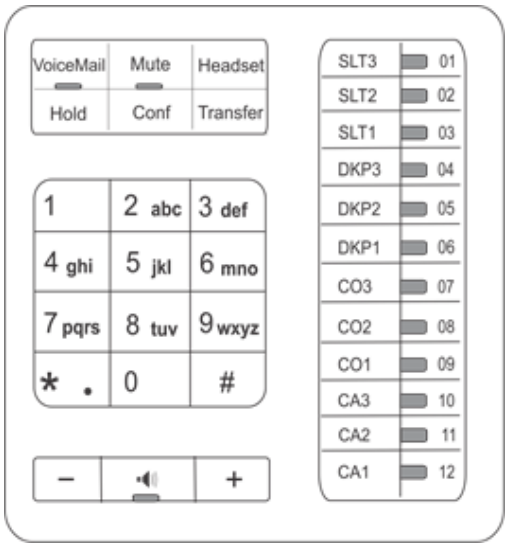
Hotel Attendant Key Map



Guest Key Map

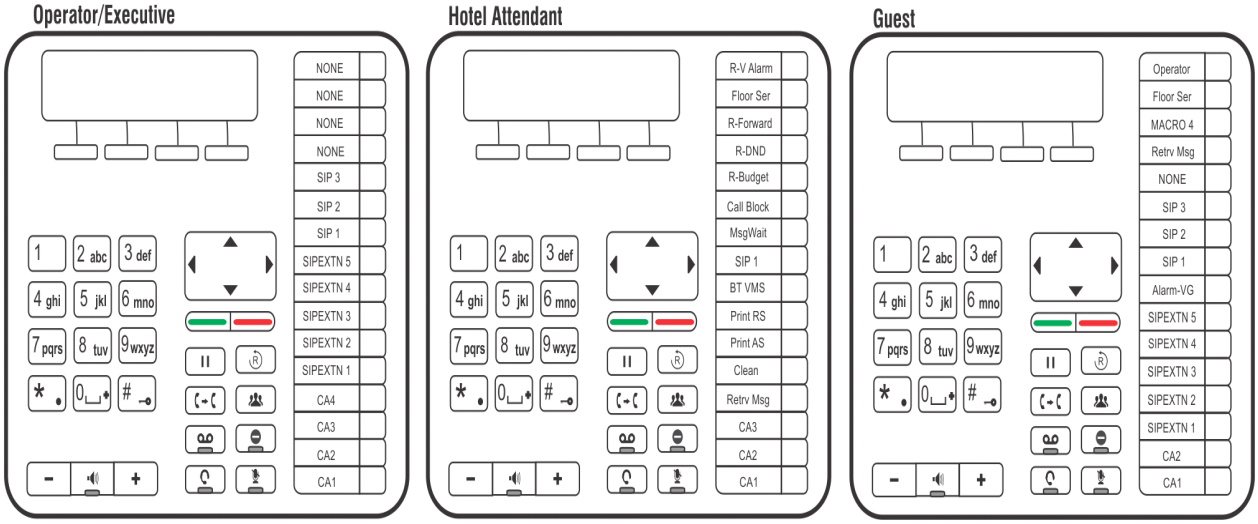


SPARSH VP330 Key Template (default)



The default Key Template is same for the Operator, Executive, Hotel Attendant and Guest.

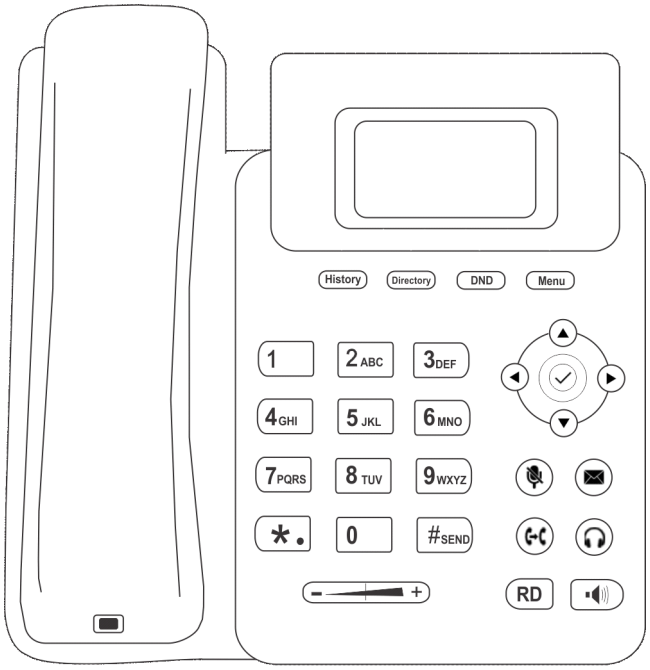
SPARSH VP510 Key Template (default)



SPARSH VP110

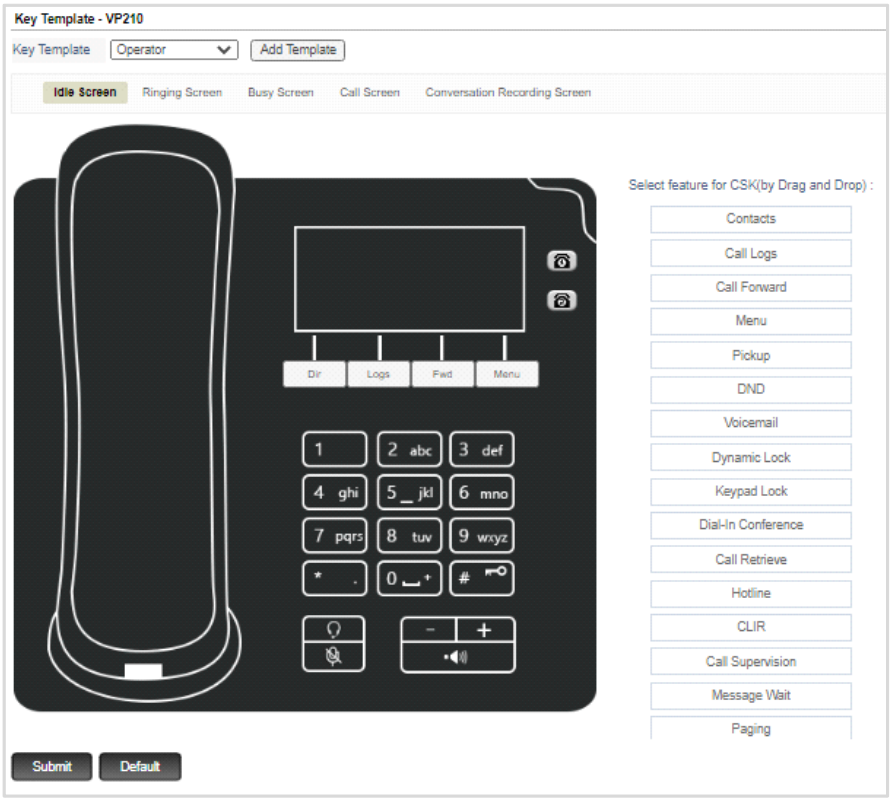
All the default templates are same for SPARSH VP110.

You can customize the following keys only — History, Directory, DND, Menu, Up, Down, Right, Left, OK, Mute and Transfer.



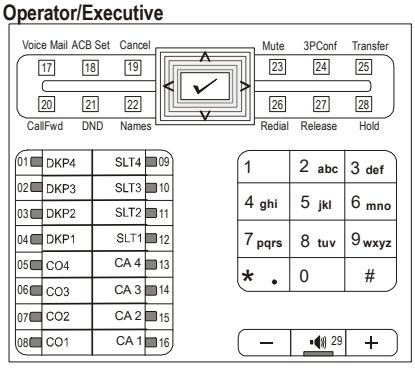
SPARSH VP210 (Default)

All the default templates are same for SPARSH VP210. You can customize the Context keys only.

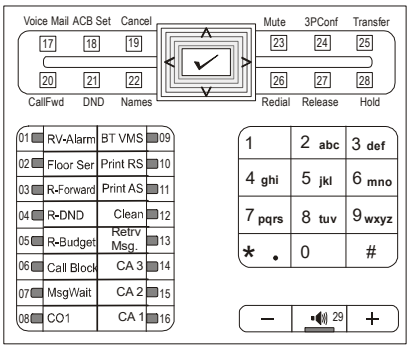


EON48 Key Templates (default)

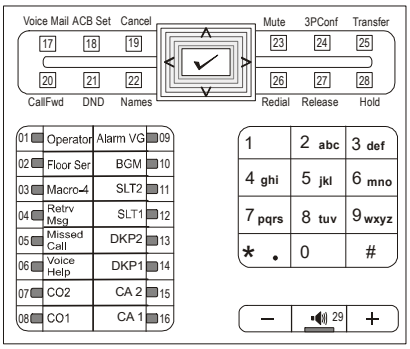
Operator/Executive



Hotel Attendant



Guest

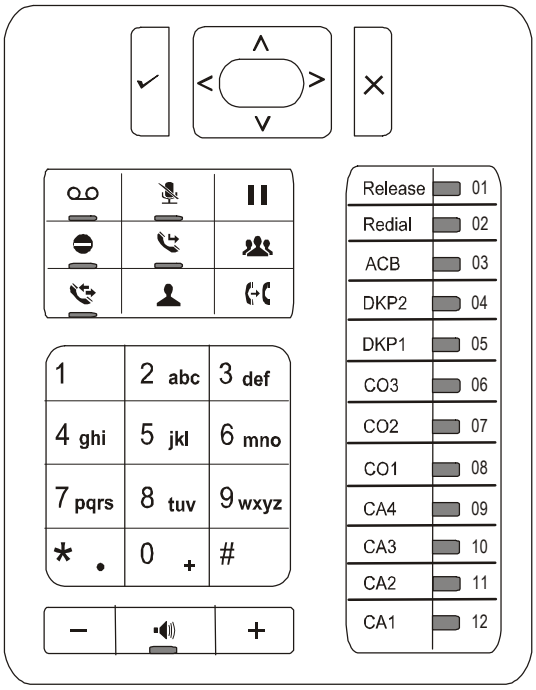


There is no LED on the following key numbers:

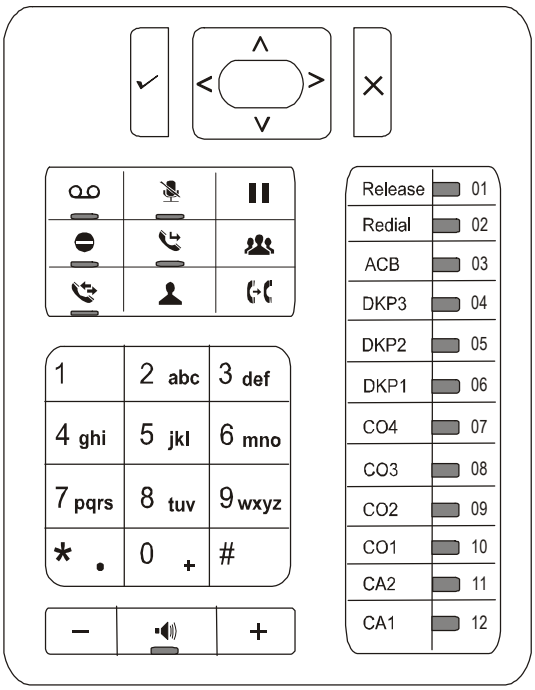
- *19 (Cancel)*
- *22 (Names)*
- *24 (Conference)*
- *25 (Transfer)*
- *26 (Redial)*
- *27 (Release)*
- *28 (Hold)*

EON310 Key Templates (default)

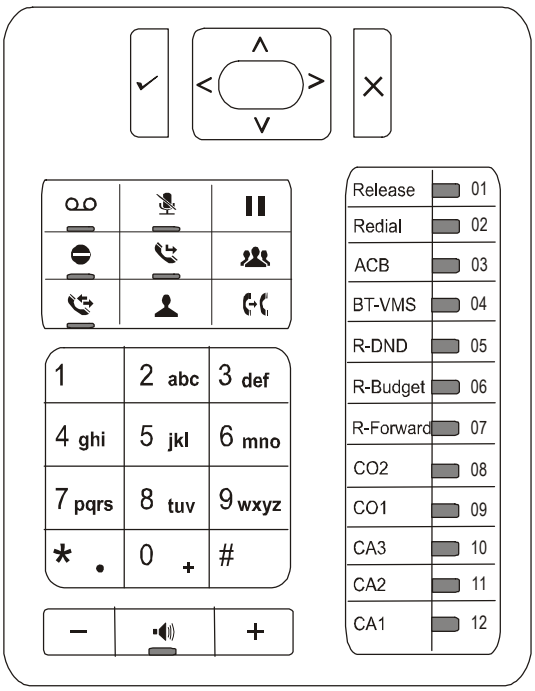
Operator Key Map



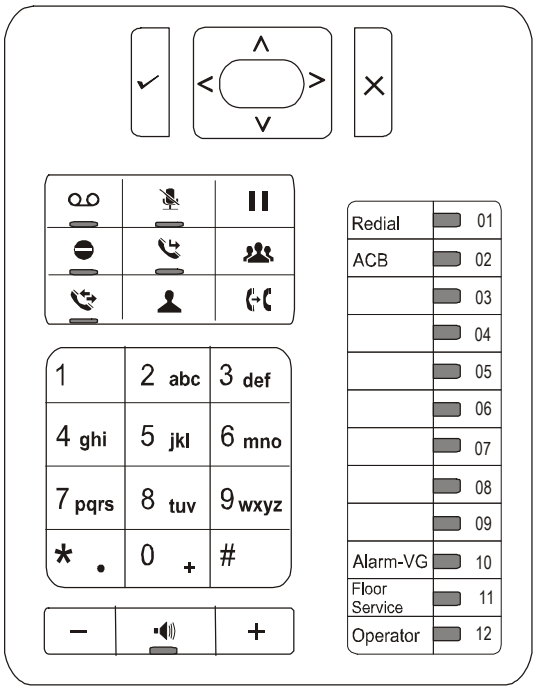
Executive Key Map



Hotel Attendant Key Map



Guest Key Map

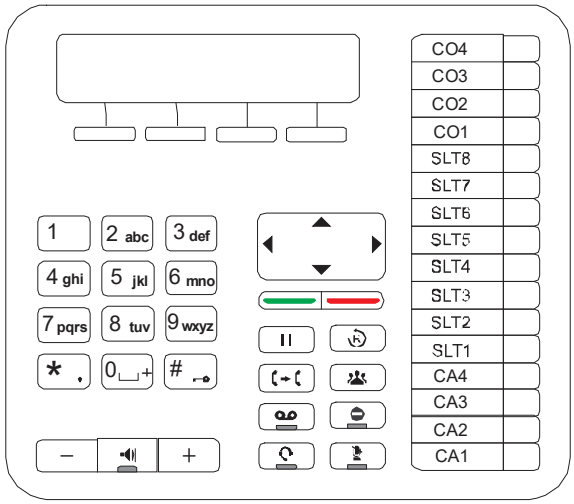


There is no LED on the following key numbers:

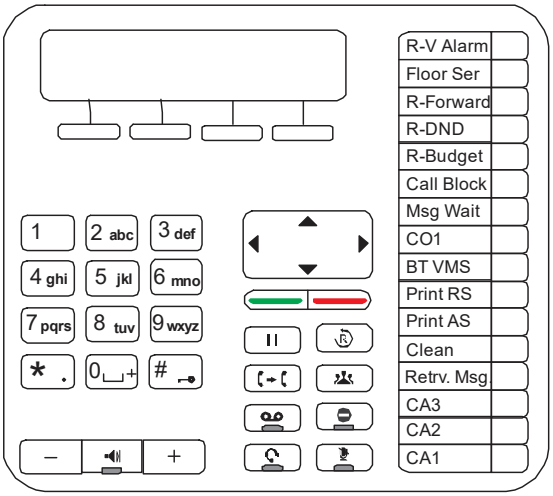
- 15 (Hold)
- 18 (Conference)
- 20 (Contacts)
- 21 (Transfer)

EON510 Key Templates (default)

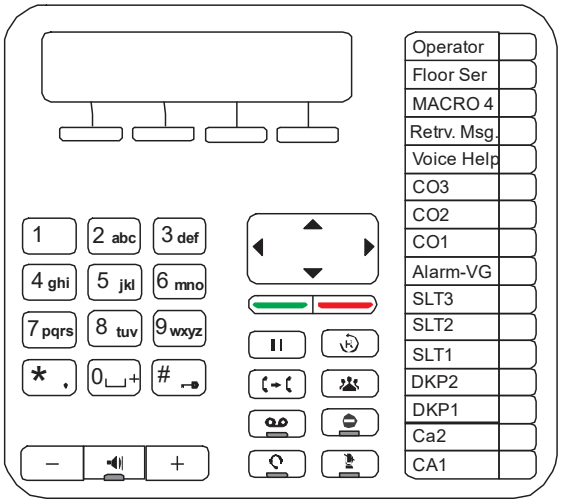
Operator / Executive



Hotel Attendant



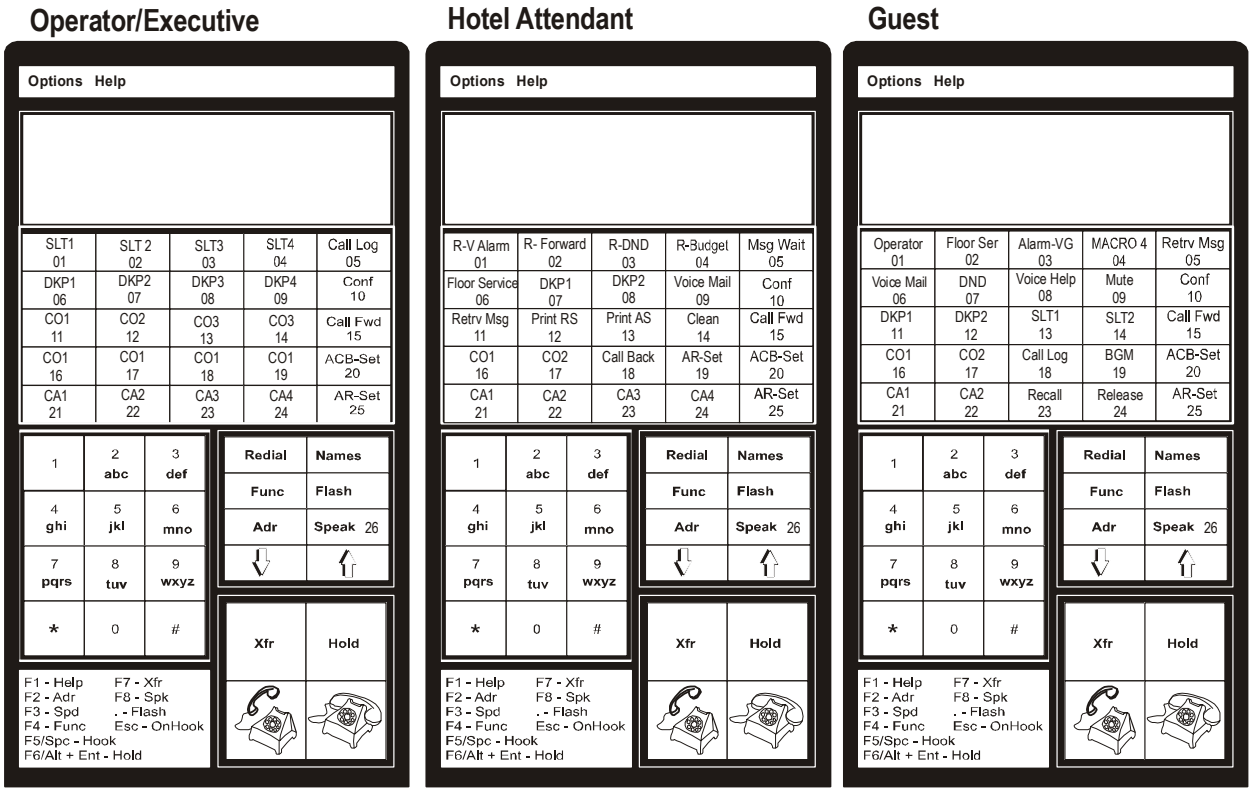
Guest



There is no LED on the following key numbers:

- *Hold*
- *Redial*
- *Transfer*
- *Conference*

EONSOFT Key Template (default)



By using Key Templates you can prepare and assign common key maps to all or as many DKPs and SPARSH IP Phones as you want, at one go.

SARVAM UCS also offers the flexibility to personalize the Key Maps of each DKP/SPARSH IP Phone, instead of using the Key Templates. For example, if you have assigned a common Executive Key Template to 12 DKPs, but you want to reassign some of the keys on two of these DKPs, SARVAM UCS allows you to selectively personalize the key maps of these two DKPs. To personalize the key maps for DKPs, see “Personalizing Key Maps” for instructions.

Customizing Key Templates

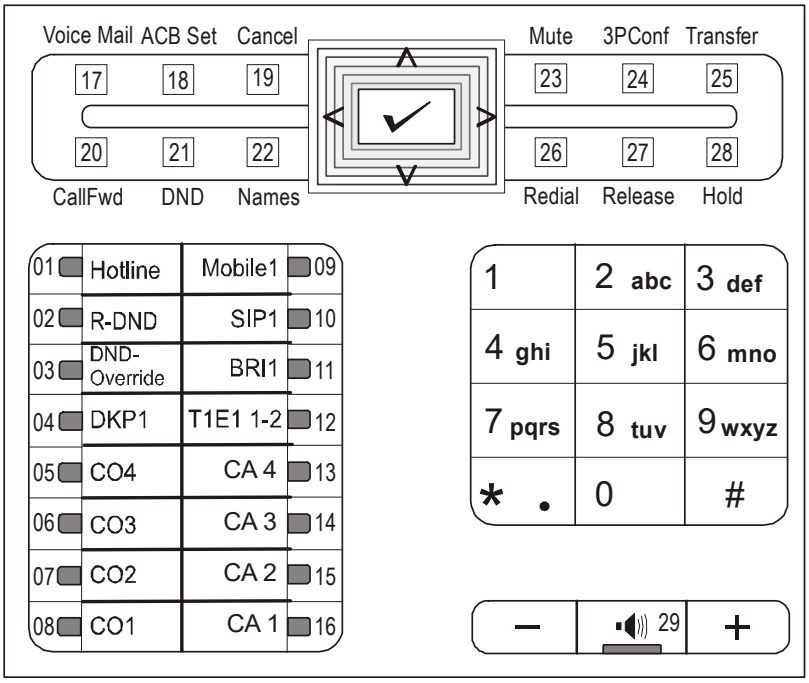
You can customize the Key Templates for:

- EON48 using Jeeves as well as a Telephone.
- EON310 using Jeeves only.
- EON510 using Jeeves only.
- SPARSH VP248 using Jeeves as well as a Telephone.
- SPARSH VP330 using Jeeves only.
- SPARSH VP310 using Jeeves only.
- SPASH VP110 using Jeeves only.
- SPARSH VP210 using Jeeves only.
- EONSOFT using Jeeves as well as a Telephone.

- *Before you begin programming the keys,*
  - *List the features/facilities that you want to change in each of the existing (default) Key Templates of Operator, Executive, Hotel Attendant, and Guest.*
  - *Create customized Key Templates on sheets of paper.*

- *For each template, decide the keys that will be reassigned the features you listed.*
- *You may use the key templates printed above to decide the position of keys.*
- *For each template that you customize, list down the DKPs which will be assigned the template, along with their Software port numbers and their corresponding Hardware Slot and Port Offset. Similarly list SPARSH IP Phones along with their Software port numbers (SIP Extension numbers).*
- *Similarly, list the DKPs and the SPARSH IP Phones which are to be assigned personalized Key Maps, along with their Software port numbers and their corresponding Hardware Slot and Port Offset.*

Illustrated below is an example of a customized Operator's Template for the EON48 model:



This customized Template is to be applied on 2 Operator DKPs:

- DKP-001, connected on H/w Slot-Port - 17-01
- DKP-002, connected on H/w Slot-Port - 17-02

You can customize the key template and assign it to DKPs using Jeeves or a Telephone.

EON48 and SPARSH VP248 have 12 Touch-sense feature keys. While you can reassign the features on these keys, you cannot re-label the keys. Avoid reassigning features on touch-sense keys.

EON310 has 9 feature keys, displaying feature icons as labels. While you can reassign the features on these keys, you cannot re-label the keys. Avoid reassigning features on these keys.

EON510 has 8 feature keys, displaying feature icons as labels. While you can reassign the features on these keys (except Headset and Mute key), you cannot re-label the keys. Avoid reassigning features on these keys.

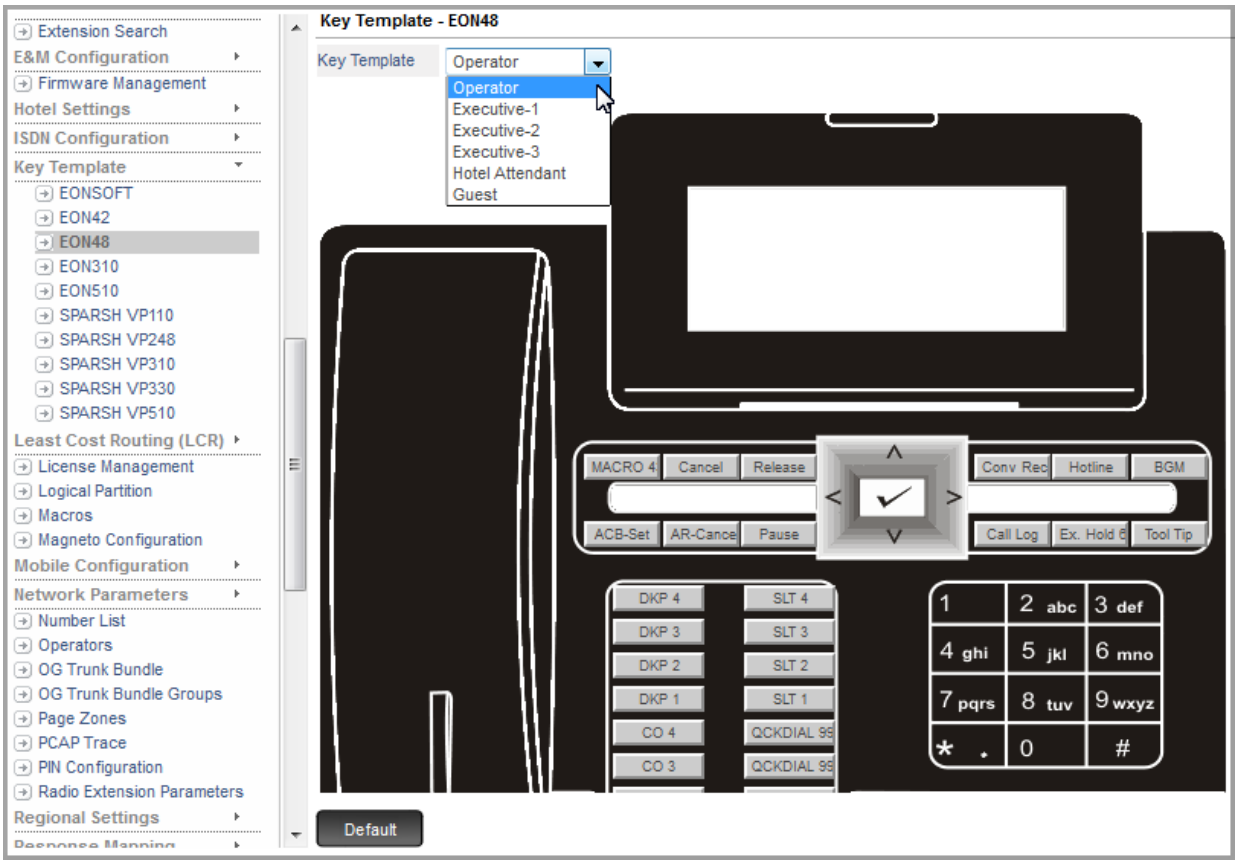
Customizing Digital Key Phone Templates using Jeeves

To customize key templates for EONSoft, EON42, EON310, EON510 follow the instructions given below.

- Log in as System Engineer.

- Under **Configuration**, click **Key Template** to open the page.
- There are separate links for each model of EON/IP Phone. Click the desired phone model.The respective phone model is displayed.
- Click **Key Template** to select the default key template — Operator, Executive1, Executive2, Executive3, Hotel Attendant and Guests — you want to customize.

  Let us attempt to program the sample Operator template we customized earlier for EON48.
- Click **EON48** to open the page. Select **Operator** as the **Key Template**.

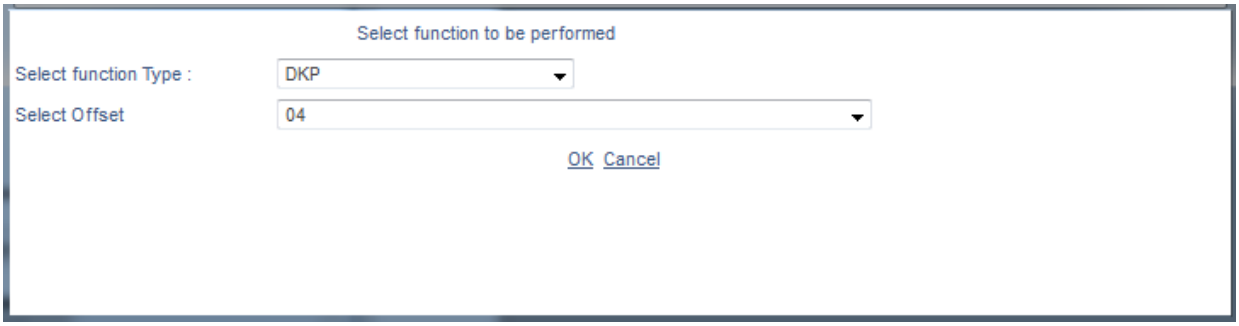


- The Operator Key template for EON48 appears.
- Refer the key template we customized on paper. As per the customized template the keys that need to be reassigned features are as follows:

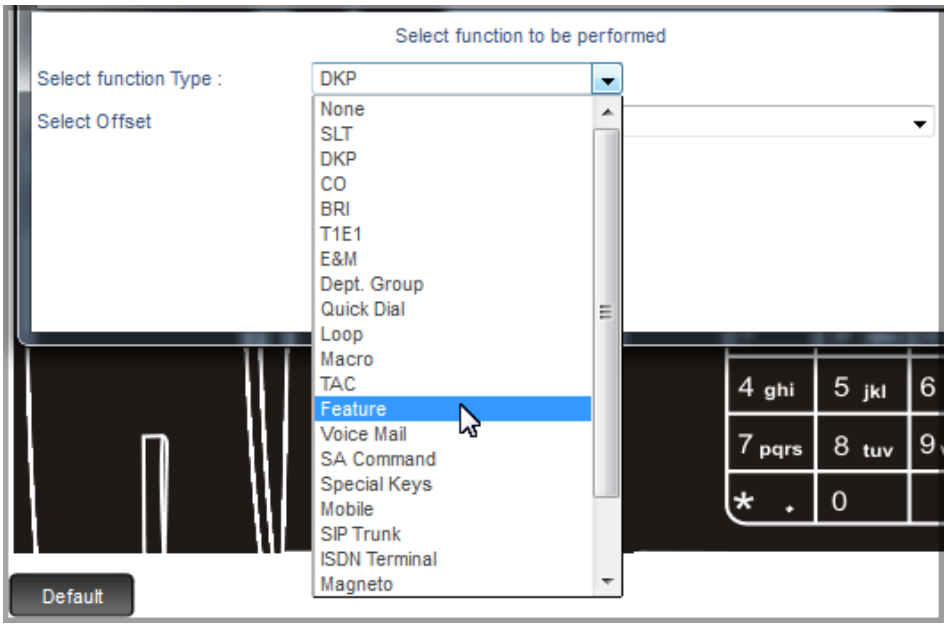
Existing function on the key	To be replaced by
DKP4	Hotline
DKP3	SA Command for “Remote DND”
DKP2	DND-Override
SLT4	Mobile Trunk 1 (MOBILE1)
SLT3	SIP Trunk 1 (SIP1)

Existing function on the key	To be replaced by
SLT2	BRI Trunk 1 (BRI 1)
SLT1	T1E1 Channel Number 2 of T1E1 Trunk 1

- To assign Hotline, click the DKP4 key. A new window opens.

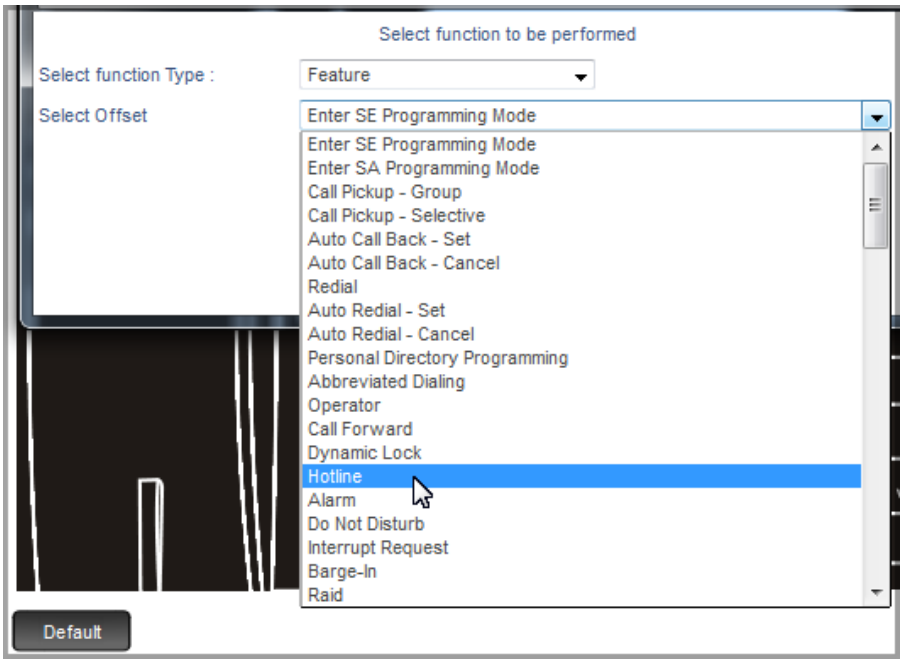


- As Hotline is a feature, in the **Select Function Type** list, click **Feature**.

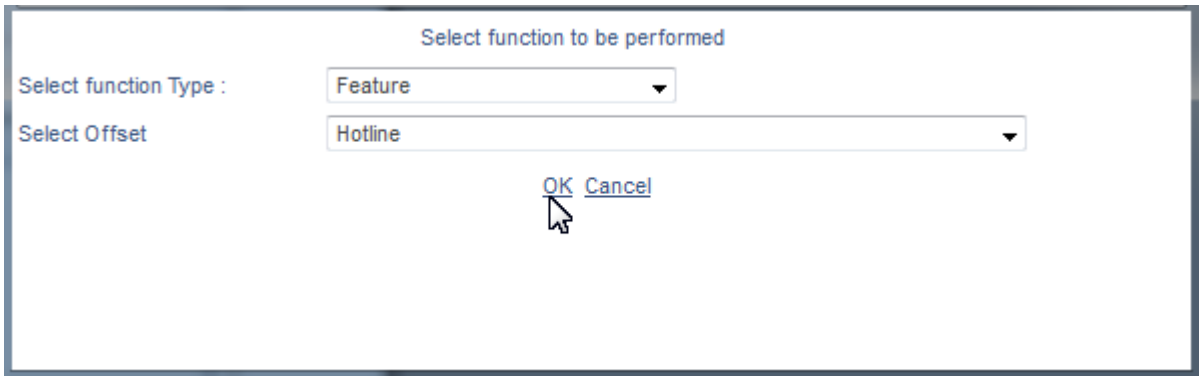


All features that can be assigned to keys will appear in the **Select Offset** list.

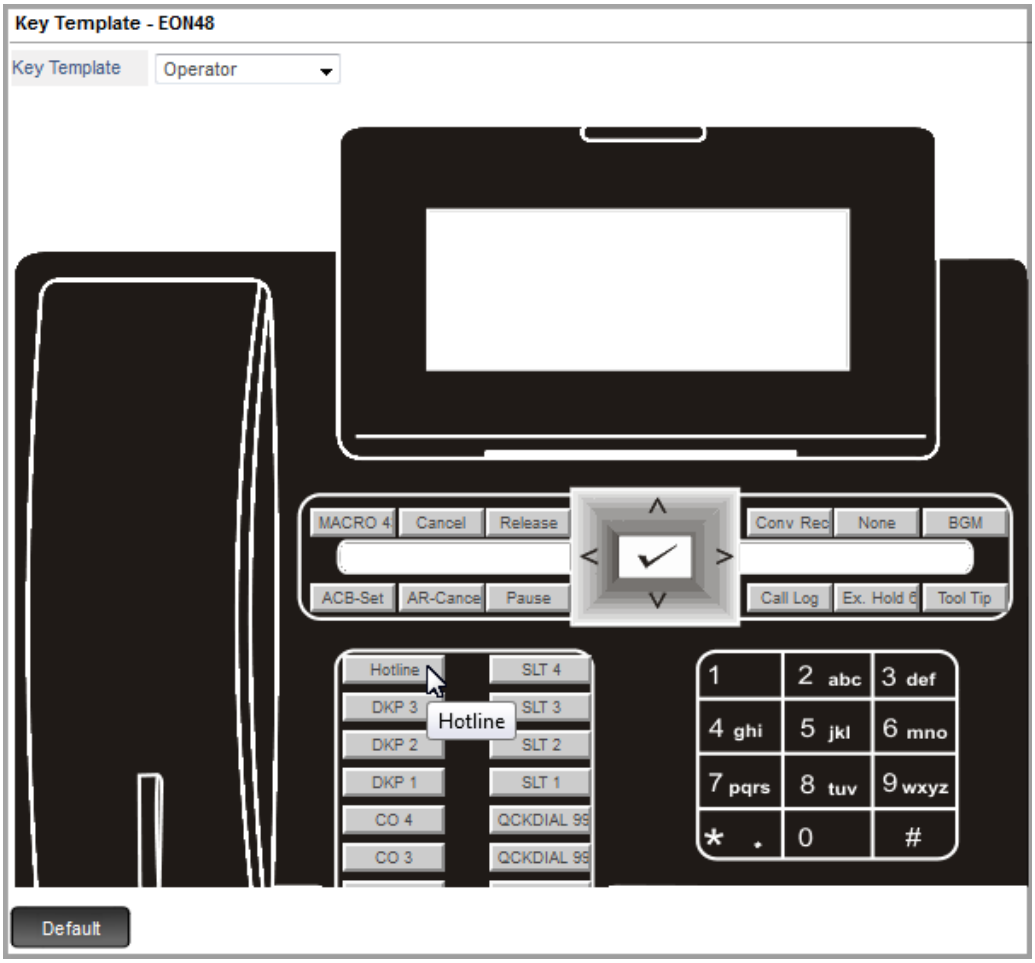
- In the **Select Offset** list, click **Hotline**.



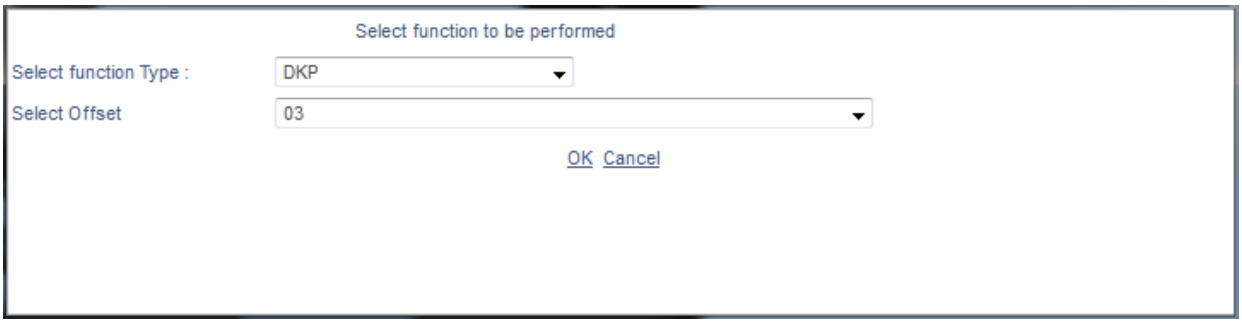
- Click on **OK** in the dialog box. The box will close.



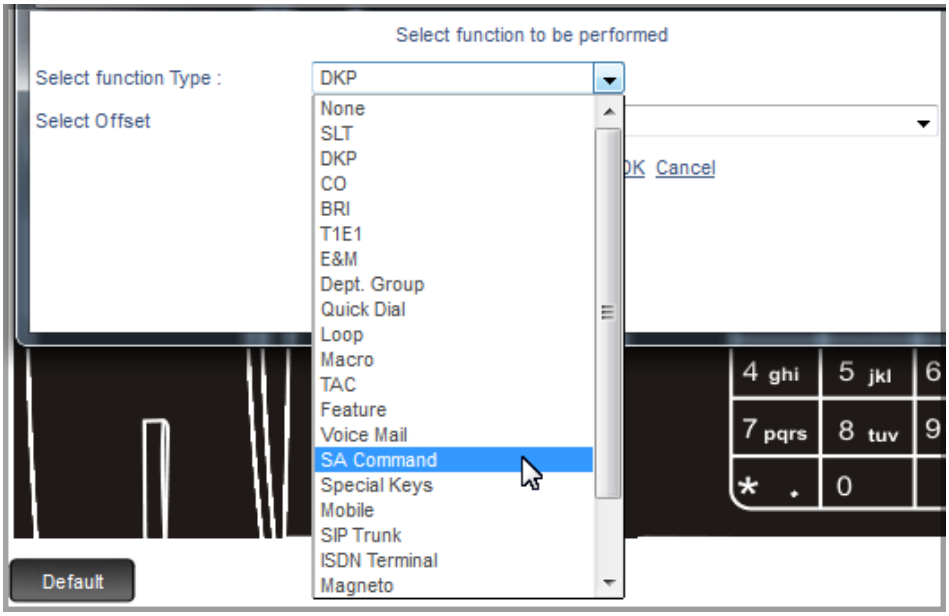
- The **Hotline** feature will appear on the key label.



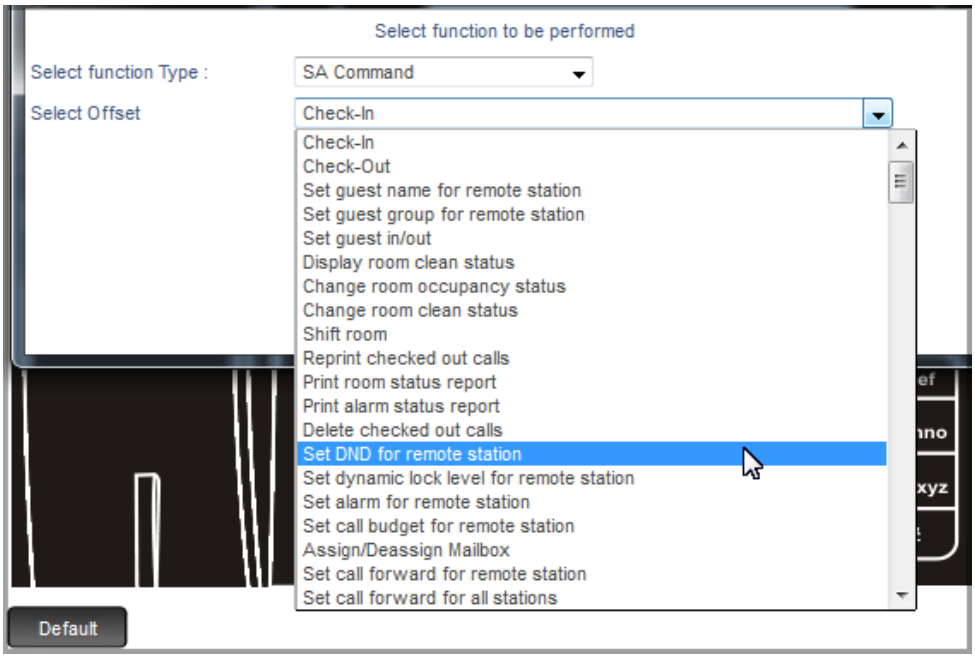
- To assign Remote DND, click DKP3 key.



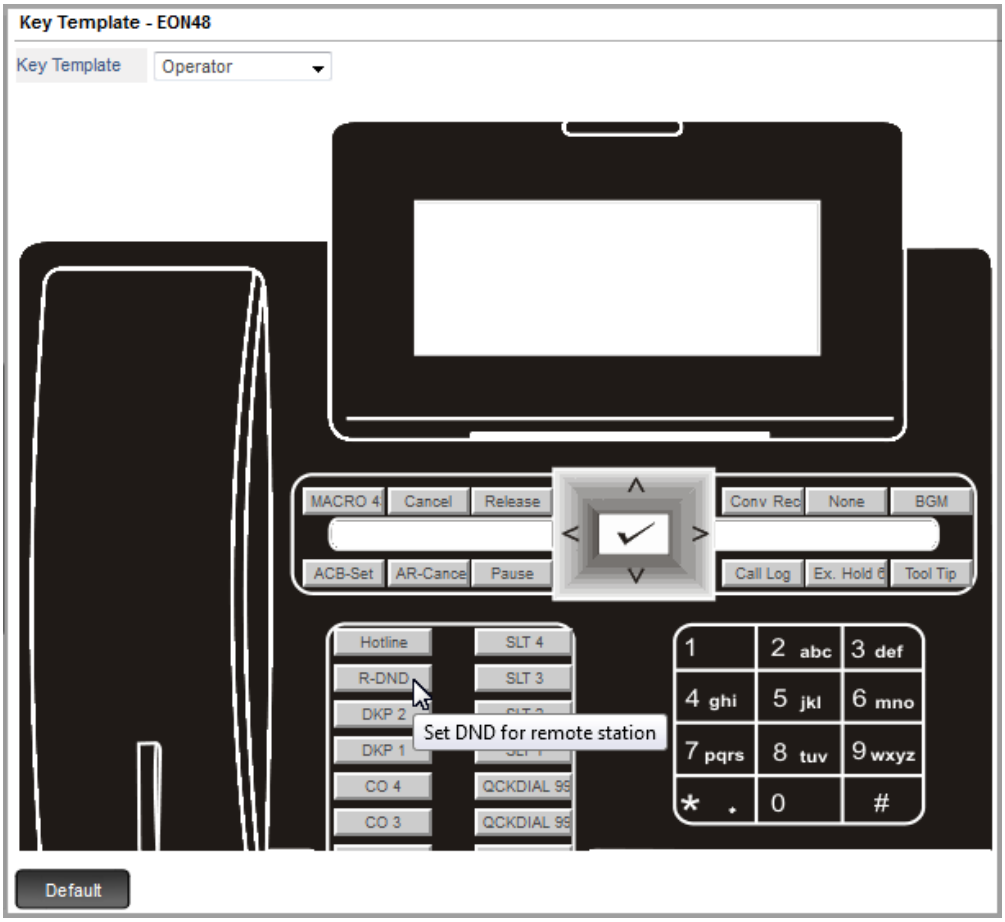
- As Remote DND is an SA Command, select the option **SA Command** in the **Select Function Type** list.



- In the **Select Offset** list, click **Set DND for remote station**.



- Click **OK**. The window closes. The Remote DND feature appears in abbreviated form as **R-DND** on the key label.



- Repeat these steps to reassign other keys, selecting the appropriate Function Type and the Offset for each feature/function.

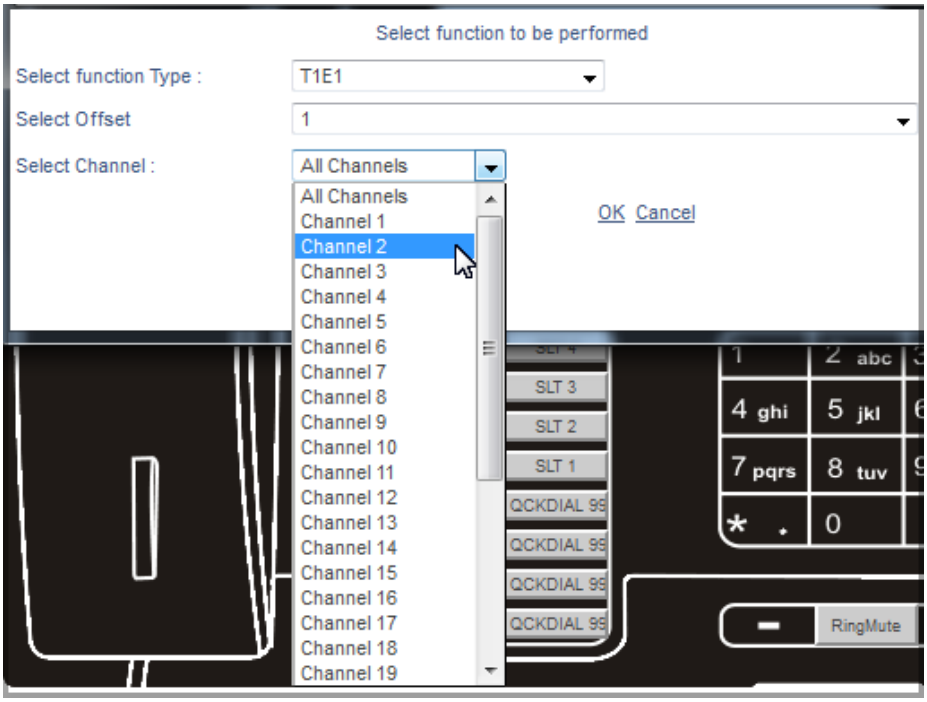
  For example, DND-Override is a feature, so select 'Feature' as the function type.

  To assign direct access to Mobile Trunk 1, select “Mobile” as function type and '01' as Offset.

  To assign direct access to SIP Trunk 1, select “SIP” as function type and '01' as Offset.

  To assign direct access to BRI Trunk 1, select “BRI” as function type and ‘01’ as Offset. When you select BRI as function type, you will be asked to ‘Select Channel’. Since you do not want to assign direct access to any particular BRI channel of this trunk, retain the option ‘All Channels” for ‘Select Channel’.

To assign direct access to Channel Number 2 of T1E1 Trunk 1, select 'T1E1' as function type, '01' as Offset.



When you select T1E1, as function type, you will be given the ‘Select Channel’ option. Since you want to assign direct access to Channel 2 of T1E1 Trunk 1, select the option ‘Channel 2’ in the ‘Select Channel’ list.

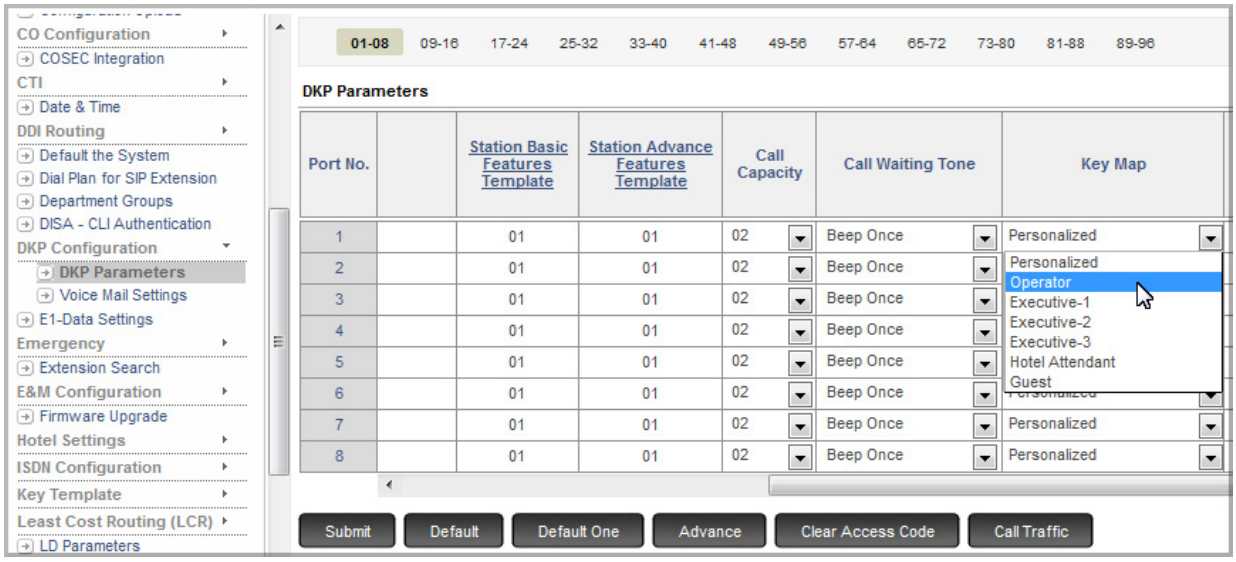
Always click **OK** when you select a Function Type and Offset.

- When you have completed assigning functions to keys, click **Submit** at the bottom of the page to save your settings.
- Follow the same steps to program the other Key Templates.
- Assign the key templates you created to the respective DKPs.

  To take the above example further, the Operator key template customized for EON48 is to be assigned to the DKPs 001 and 002. To do this,

  - Click the **DKP Parameters** link to open the page.
  - Scroll with the horizontal bar to reach the parameter **Key Map** of this DKP 001.

- Select **Operator** as the Key Map.



- Similarly, for DKP-002, select **Operator** as the Key Map.
- Click **Submit**.
- Also refer the topic “Configuring DKP Parameters using Jeeves” for instructions on assigning the Key Map in the DKP Parameters.

Customizing Extended IP Phone Templates using Jeeves

You can:

- customize the existing key templates — Operator, Executive1, Executive2, Executive3, Hotel Attendant and Guests.
- add new key templates.

Customizing Existing Key Templates

To customize existing key templates for SPARSH VP248, SPARSH VP330, SPARSH VP310, SPARSH VP510 and SPARSH VP110 follow the instructions given below.

To customize existing key templates for SPARSH VP210, refer to “Customizing Existing Key Templates - SPARSH VP210”.

- Log in as System Engineer.
- Under **Configuration**, click **Key Template** to open the page.
- There are separate links for each model of IP Phone. Click the desired phone model.The respective phone model is displayed.
- Click **Key Template** to select the desired key template — Operator, Executive1, Executive2, Executive3, Hotel Attendant and Guests — you want to customize.

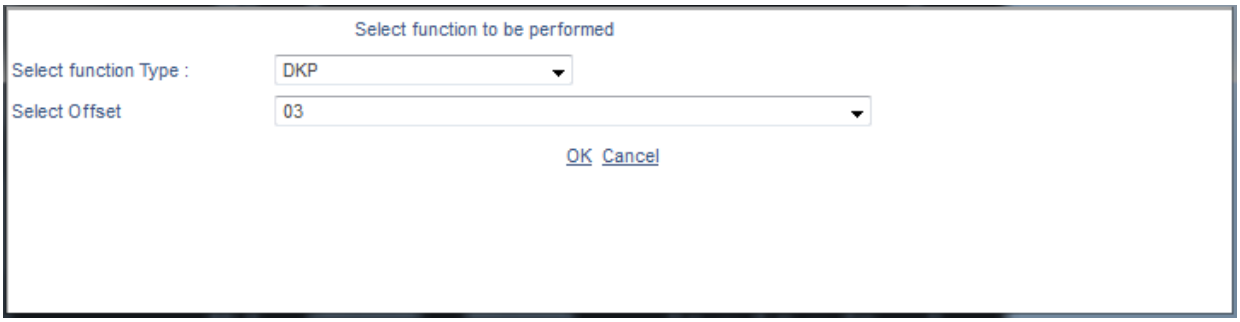
  Let us attempt to configure the sample Operator template for VP248.

Click **VP248** to open the page. Select **Operator** as the **Key Template**.

For example, to customize the key template the keys that need to be reassigned features are as follows:

Existing function on the key	To be replaced by
DKP3	Hotline
SLT3	SIP Trunk 1 (SIP1)

- To assign Hotline, click the DKP3 key. A new window opens.

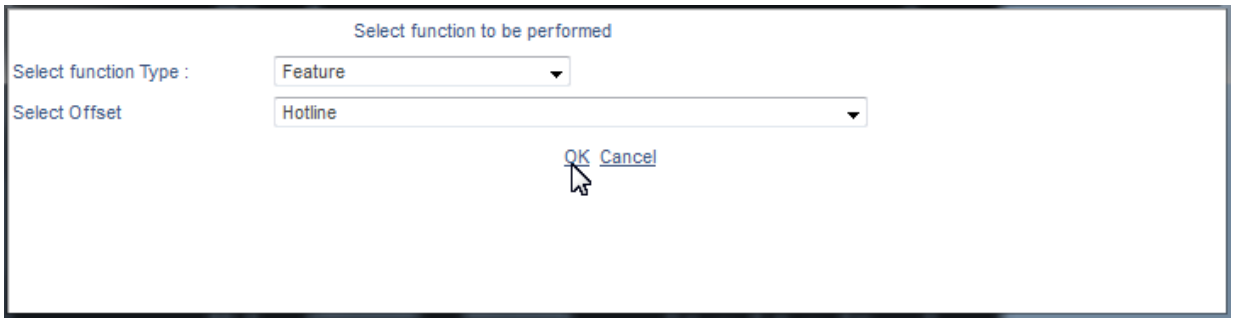


- As Hotline is a feature, in the **Select Function Type** list, click **Feature**.

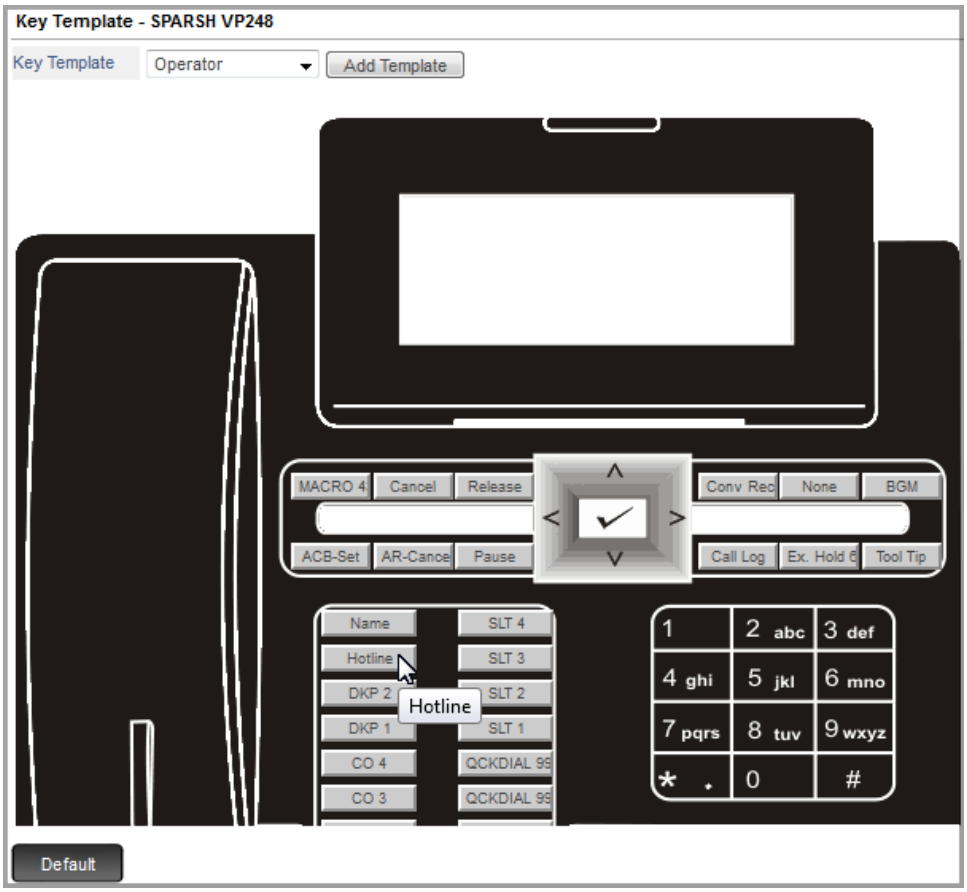
  All features that can be assigned to keys will appear in the **Select Offset** list.

- In the **Select Offset** list, click **Hotline**.

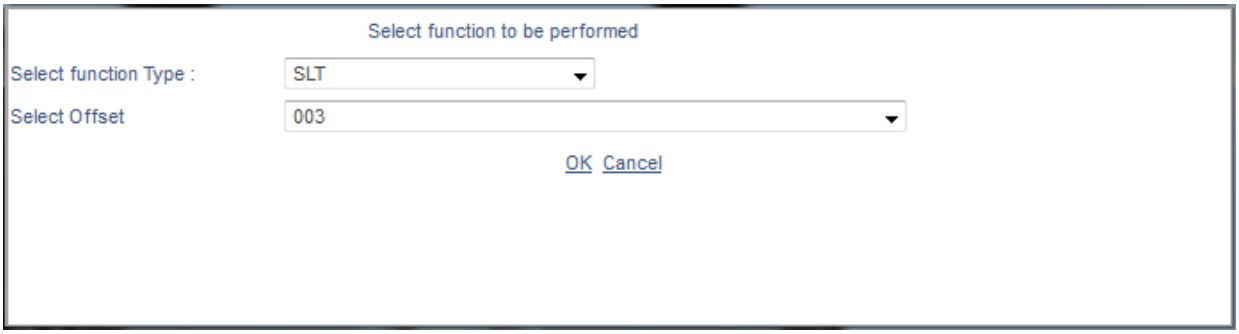
- Click on **OK** in the dialog box. The box will close.



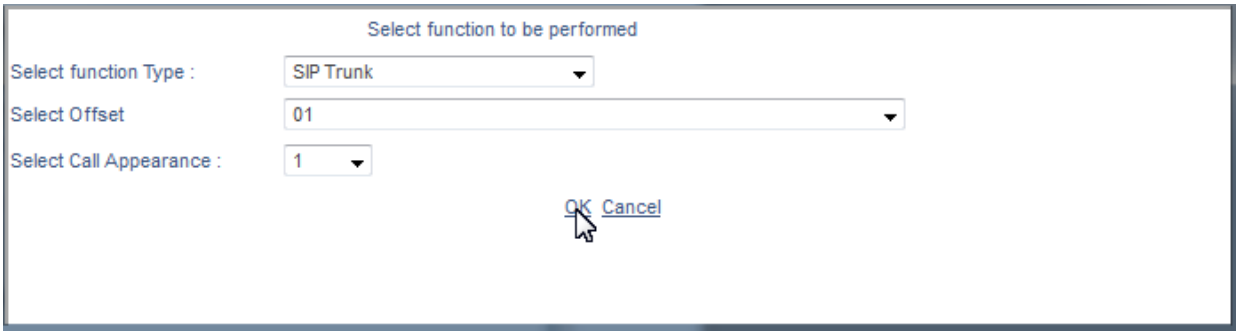
- The **Hotline** feature will appear on the key label.



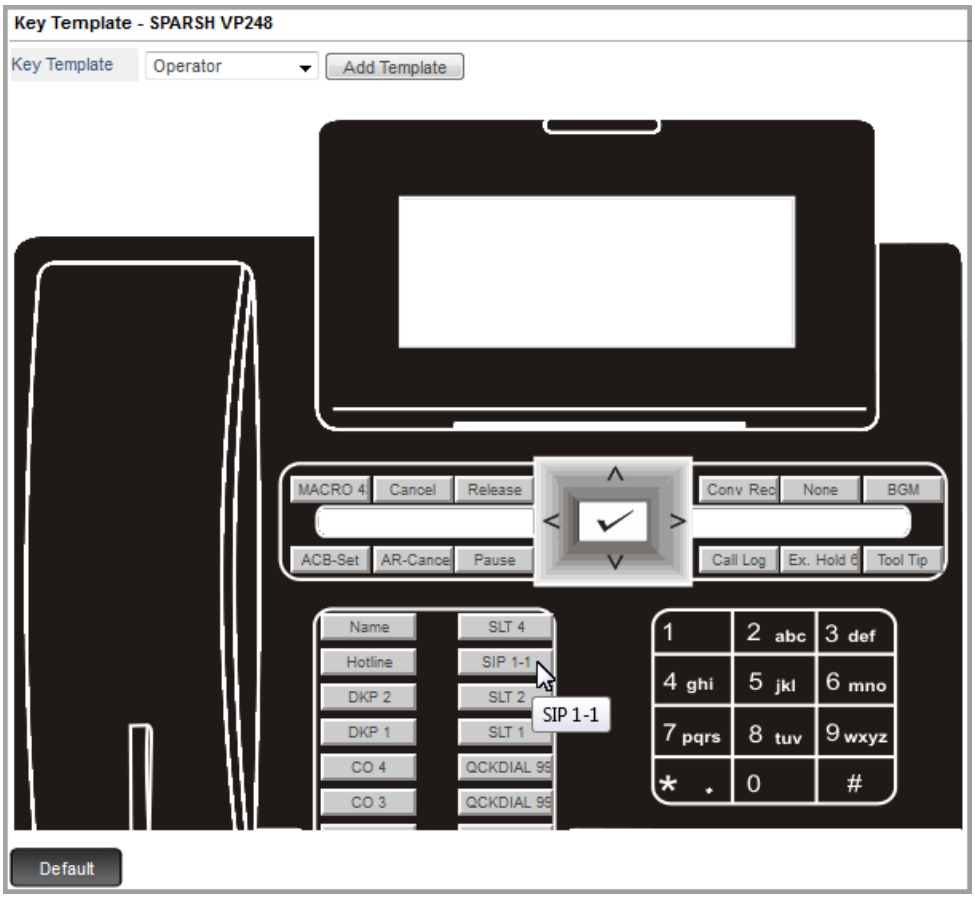
- To assign direct access to SIP Trunk 1, click SLT3 key.



Select **SIP Trunk** as function type and **01** as Offset.



SIP 1-1 appears as the key label.



When you have completed assigning functions to keys, click submit to save the settings.

Similarly, you can customize the other templates — Executive, Hotel Attendant, Guest.

To customize the templates for other phone models, follow the instructions as given above.

You can also add new templates. To do so, refer “Add, Edit or Remove Key Templates”.

Customizing Existing Key Templates - SPARSH VP210

SPARSH VP210 has the provision to program the four Context Keys. These keys enable you to access the most frequently used functions/features at the press of a single button.

The screens — Idle Screen, Ringing Screen, Busy Screen, Call Screen, Conversation Recording Screen, all have different set of features that can be accessed. SPRASH VP210, enables you to customize these by allowing you to set the priorities of the features in each type of screen as per your preference. You can assign the features to the Context Keys depending on the state of the call.

- In the Idle Screen you can assign the desired feature/function to the Context Keys as well as set their priorities as per your requirement.
- In the other Screens you can only set the priorities of the features.

Refer to the details mentioned below for Default Key Assignment and the Feature Key Assignment/Feature Priority Assignment asper the different Call States:

Idle Screen

Default Key Assignment

Parameter	Default Key Assigned
Context Key 1	Dir
Context Key 2	Logs
Context Key 3	Fwd
Context Key 4	Menu

Feature Key Assignment/Feature Priority Assignment

Type of Screen	Feature Priority Selection List
Idle Screen	Contacts
	Call Logs
	Call Forward
	Menu
	Pickup
	DND
	Voicemail
	Dynamic Lock
	Keypad Lock
	Dial-In Conference
	Call Retrieve
	Hotline
	CLIR
	Call Supervision
	Message Wait
	Paging
	Meet Me Paging
	Room Monitoring
	Intercom
	Follow Me
	Walk-In
	PIN Dialing
	Department Group Call Forward
	Open a Door
	User Status

Ringing Screen

Default Key Assignment

Parameter	Default Key Assigned
Context Key 1	Transfer Complete
Context Key 2	Auto Call Back
Context Key 3	Message Wait Set
Context Key 4	Next

Feature Priority Assignment

	Feature Priority Selection List
Ringing Screen	Transfer Complete
	Auto Call Back
	Message Wait Set
	Forced Answer
	Release
	End Call

Busy Screen

Default Key Assignment

Parameter	Default Key Assigned
Context Key 1	Auto Call Back
Context Key 2	Interrupt Request
Context Key 3	Barge-In
Context Key 4	Next

Feature Priority Assignment

	Feature Priority Selection List
Busy Screen	Auto Call Back
	Interrupt Request
	Barge-In
	Forced Call Disconnection
	Message Wait Set
	Transfer Complete
	Release
	Trunk Reservation
	End Call

Call Screen

Default Key Assignment

Parameter	Default Key Assigned
Context Key 1	Mute
Context Key 2	Hold
Context Key 3	Transfer
Context Key 4	Next

Feature Priority Assignment

	Feature Priority Selection List
Call Screen	Mute
	Hold
	Transfer
	Conference
	Personal Call Park
	VMS Blind Transfer
	Global Hold
	General Call Park
	Call Chaining
	Conversation Recording
	Release
	Flashing on Trunk
	Account Code
	Open a Door or ACK
	New Call
	End Call

Conversation Recoding Screen

Default Key Assignment

Parameter	Default Key Assigned
Context Key 1	Mute
Context Key 2	Hold
Context Key 3	Transfer
Context Key 4	Next

Feature Priority Assignment

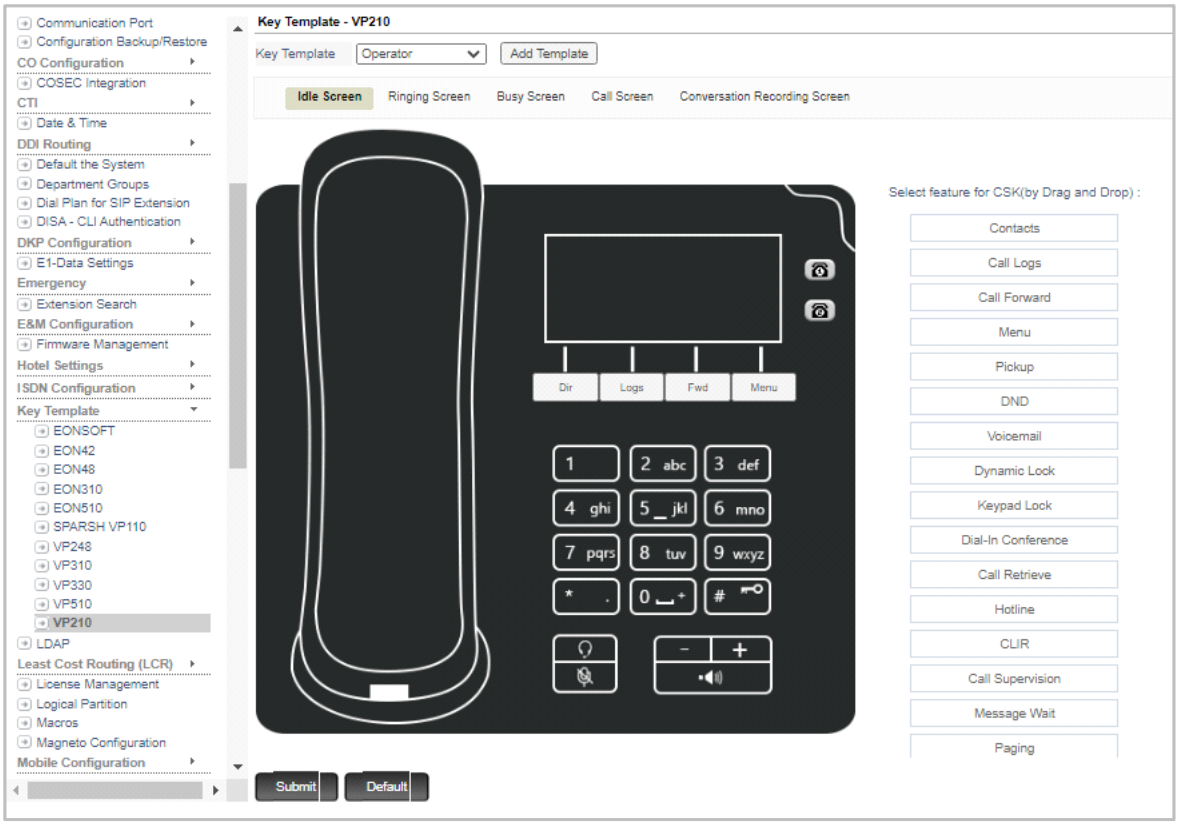
	Feature Priority Selection List
Conversation Recording Screen	Mute
	Hold
	Transfer
	Conference
	VMS Blind Transfer
	Global Hold
	Call Chaining
	Stop Recording
	Release
	End Call

To customize existing key templates for SPARSH VP210, follow the instructions given below.

- Log in as System Engineer.
- Under **Configuration**, click **Key Template** to open the page.
- There are separate links for each model of IP Phone. Click **VP210**.
- Click **Key Template** to select the desired key template — Operator, Executive1, Executive2, Executive3, Hotel Attendant and Guests — you want to customize.

  Let us attempt to configure the sample Operator template for VP210.

  Click **VP210** to open the page. Select **Operator** as the **Key Template**.



- Click **Idle Screen**.
- Each Context key, 1 to 4 can be assigned features.
- The feature assignment cum priority list appears on the right. You can change the feature assignments/ priorities as per your preference.
- To set the priority, drag and drop the features in the order of your preference. This will have two implications — the Context Key will be assigned the desired feature as well it will set the priority.
- Click **Submit**.
- The key map will refresh and the name of the Feature you selected (first four) will appear in abbreviated form as the key labels.

Menu must be assigned to one of the first four Context Keys.

- Similarly, you can click **Ringing Screen, Busy Screen**, **Call Screen** or **Conversation Recording Screen** and can set the feature priorities as per your preference.

  In these screens only priorities can be set and the 4th Context Key will always be assigned to **More >** feature. The first three Context keys will display the features assigned priorities 1, 2 and 3. To access other features press **More >**. The features will be displayed as per their set priorities.

Similarly, you can customize the other templates — Executive, Hotel Attendant, Guest.

You can also add new templates. To do so, refer “Add, Edit or Remove Key Templates”.

Add, Edit or Remove Key Templates

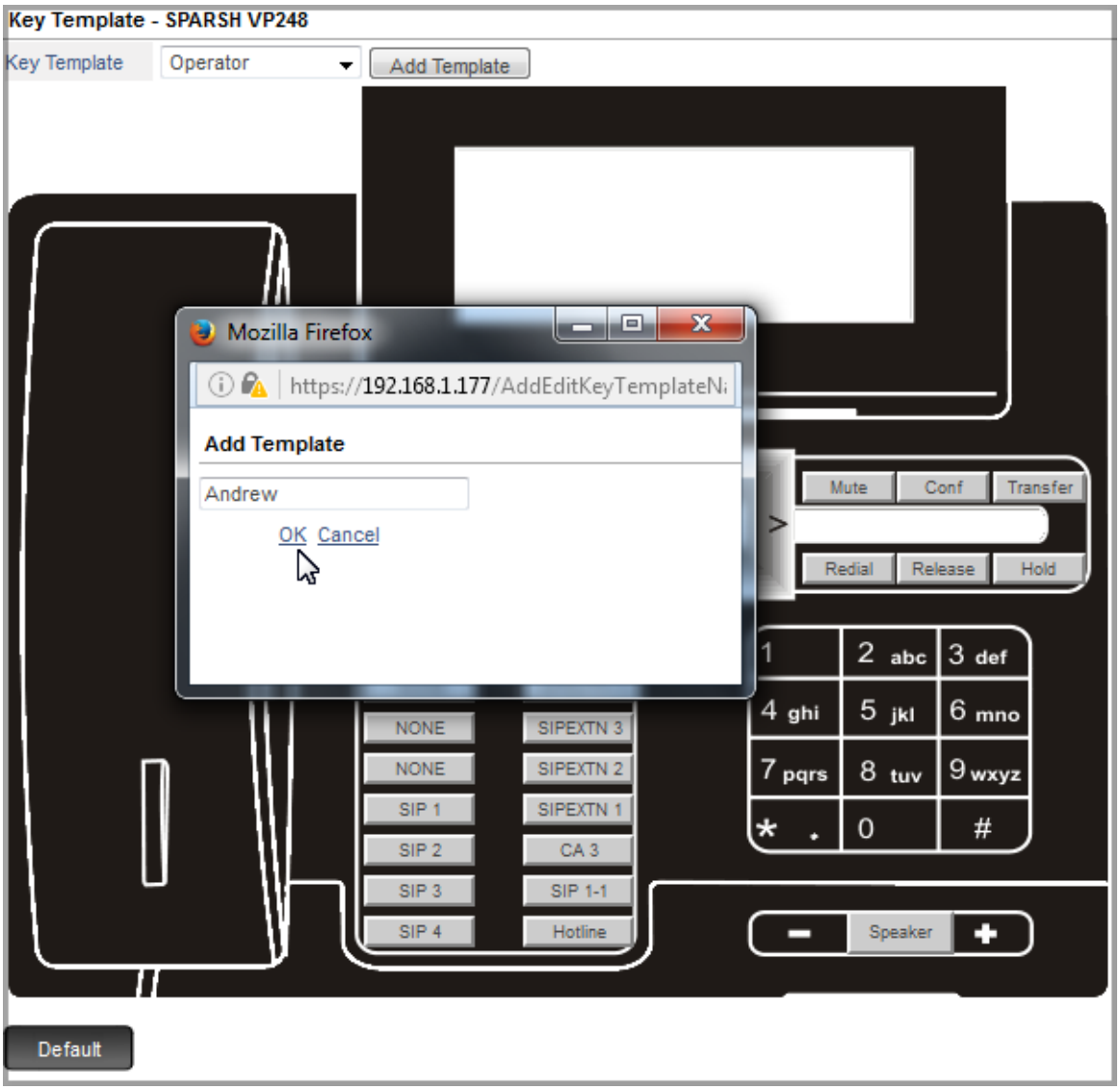
If you do not wish to customize any existing template, you can add new templates. By default, the Operator Template will be displayed as the phone key map. You can customize the key map as per your requirement or add a new template.

You can add upto 64 new templates for each phone model.

To add a template,

- Log in as System Engineer.
- Under **Configuration**, click **Key Template** to open the page.
- There are separate links for each model of IP Phone. Click the desired phone model.The respective phone model is displayed.

- Click **Add Template**. The Add Template window opens.



- Assign a name to the template. This template appears as one of the Key Template options.
- To customize the key template to match the requirement, click Andrew as the Key Template option.

  Follow the instructions given under “Customizing Key Templates”.
- When you have completed assigning functions to keys, follow the same steps if you wish to add other Key Templates.
- To assign the key templates you created to the respective SIP Extension location using Jeeves. Refer to “DSS Key Settings” in “Configuring Matrix SPARSH VP248”, “DSS Key Settings” in “Configuring Matrix SPARSH VP310”, “DSS Key Settings” in “Configuring Matrix SPARSH VP330”, “DSS Key Settings” in “Configuring Matrix SPARSH VP510” and “DSS Key Settings” in “Configuring Matrix SPARSH VP210” for instructions
- To Edit the template name, click **Edit Template**.
- To delete the template, click **Remove Template**.

Customizing Key Templates using a Telephone

EON310 / SPARSH VP110 / SPARSH VP210 Key Templates can be customized using Jeeves only.

When customizing key templates using a Telephone, two attributes of a key must be programmed, namely:

a. **Location** - the physical location of the key on the phone body.

b. **Function** - the function that the key should perform, that is, as direct access to a feature or an extension, a trunk type, an SA command, etc.

Follow the numbering of keys on the default Key Maps illustrated earlier in this topic.

- Enter SE mode from a DKP/SLT.

  To assign a feature to a key in a Template, dial:
  - ***1261-1-Key Template-Terminal Type-Key Number-Function Type-Function Number-Channel***

  To assign the same feature to the same key number in a range of templates, dial:
  - ***1261-2-Key Template-Key Template-Terminal Type-Key Number-Function Type-Function Number-Channel***

  To assign the same feature to the same key number in all the templates, dial:
  - ***1261-*-Terminal Type-Key Number-Function Type-Function Number-Channel***

  Where,
  Key Template is
  1 for Operator
  2 for Executive1
  3 for Hotel Attendant
  4 for Guest
  5 for Executive2
  6 for Executive3

  Terminal Type is
  1 for EON45/EONSOFT
  2 for EON42
  3 for EON48
  4 for SPARSH VP248
  6 for EON310
  7 for SPARSH VP330

  Key Number is the number of the key which is to be reassigned the feature.
  Key Numbers on EONSOFT are from 01 to 25
  Key Numbers on EON42 are from 01 to 25
  Key Numbers on EON48 are from 01 to 29.
  Key Numbers on SPARSH VP248 are from 01 to 29.
  Key Numbers on EON310 are from 01 to 21.
  Key Numbers on SPARSH VP330 are from 01 to 15.

  For numbering of the keys, refer the default Key Maps illustrated at the beginning of this topic.

  Function Type is the function to be performed by the key, from 00 to 34.

Function Number defines the exact function under each Function Type selected for the key.

Channel is the number of the channel in the BRI or PRI Line to be accessed.

Channel number for BRI Line is
01 for Channel 1
02 for Channel 2
00 for both Channels
Channel number for PRI Line is
01 to 30 for Channels 1 to 30 respectively.
00 for all Channels

Channel number for all function types other than BRI and PRI is ***00***
Refer the tables below for the complete list of function types and function numbers, and channel numbers.

Table: Function Types and Function Numbers

Function Type	Meaning	Function Number	Meaning
00	None	-	-
01	SLT (Direct Station Call)	001 to 512	Number of the SLT (software) port which is to be called when the key is pressed.
02	DKP (Direct Station Call)	001 to 128	Number of the DKP (software) which is to be called when the key is pressed.
03	CO (Direct Trunk Access)	001 to 128	Number of the CO (software) port which is to be called when the key is pressed.
04	BRI (Direct Trunk Access and Direct BRI Channel Access)	001 to 032	Number of the BRI (software) port which is to be called when the key is pressed. Number of the Channel in the BRI Line, which is to be accessed when the key is pressed. Channel numbers are: 01 for BRI Channel 1 02 for BRI Channel 2 00 for both Channels

Function Type	Meaning	Function Number	Meaning
05	T1E1 (Direct Trunk Access and Direct PRI Channel Access)	001 to 008	Number of the T1E1 (software) port which is to be called when the key is pressed. Number of the Channel in the PRI Line which is to be accessed when the key is pressed. Channel numbers are: 01 for T1E1 Channel 1 02 for T1E1 Channel 2 03 for T1E1 Channel 3 : : 30 for T1E1 Channel 30 00 for all Channels
06	E&M (Direct Trunk/ Station Access)	001 to 128	Number of the E&M (software) port which is to be called when the key is pressed.
11	Dept. Group (Direct Station Call)	01 to 24	Number of the Department Group which is to be called when the key is pressed.
12	Quick Dial (Abbreviated Dialing)	001 to 999	The Global Directory Index number on which the External Number is stored. Call will be made to that External Number when this key is pressed
13	LOOP (Loop Count or Call Capacity)	01 to 10	The number of Call Appearance/Call Loop which will be accessed when the key is pressed.
14	MACRO	01 to 25	The number string of the Key Board Macro which will be dialed when the key is pressed. Refer “Macros”
15	TAC (Direct Trunk Access)	1 to 6	The Trunk Access Code that will be dialed when the key is pressed.
18	FEATURE (Direct Feature Access)	01 to 87	The feature that will be invoked when the key is pressed. For a complete list of features, refer the “Table: “Function Type 18: Features””.
21	Voice Mail	1	The Voice Mail System Port to be dialed when the key is pressed.
22	SA Command (Direct Dialing of SA Command)	001 to 173	The SA Command to be invoked when the key is pressed. For a complete list of SA Commands, refer the “Table: “Function Type 22: SA Commands””.
24	Special Keys	01 to 15	The Special Digits to be dialed when the key is pressed. For a complete list of Special Keys, refer the “Table: “Function Type 24: Special Keys””.
25	MOBILE (Direct Trunk Access)	01 to 32	The number of the Mobile Port to be accessed when the key is pressed.
26	SIP (Direct Trunk Access)	01 to 32	The number of the SIP Trunk to be accessed when the key is pressed
28	ISDN Terminal (Direct Station Call)	01 to 64	The number of the ISDN Terminal (software) port which is to be dialed when the key is pressed.

Function Type	Meaning	Function Number	Meaning
29	Magneto (Direct Station Call)	001 to 128	The number of the Magneto (software) port to be dialed when the key is pressed.
31	ROOM (Direct Station Call)	001 to 512	The number of the Hotel Room to be dialed when the key is pressed.
34	SIP Extension (Direct Access)	001 to 999	The number of the SIP Extension to be dialed when the key is pressed.
36	Virtual Ext.	01 to 64	The number of the Virtual Extension to be dialed when the key is pressed.
37	General Orbit	2 to 9	The number of General Orbit where the call should be parked/retrieved from when the key is pressed.

Table: “Function Type 18: Features”

Feature	Feature Number	DSS LED Activity?
Enter SE Programming Mode	1	
Enter SA Programming Mode	2	
Call Pickup - Group	3	
Call Pickup - Selective	4	
Auto Call Back - Set	5	Yes
Auto Call Back - Cancel	6	
Redial	7	
Auto Redial - Set	8	Yes
Auto Redial - Cancel	9	
Personal Directory Programming	10	
Abbreviated Dialing	11	
Operator	12	
Call Forward	13	Yes
Dynamic Lock	14	
Hotline	15	Yes
Alarm	16	Yes
Do Not Disturb	17	Yes
Interrupt Request	18	
Barge-In	19	
Raid	20	
Trunk Reservation	21	
Call Toggle	22	
Conference	23	Yes

Feature	Feature Number	DSS LED Activity?
-	24	
Dial-In Conference	25	
Call Park	26	
Call Park - Retrieve	27	
Room Monitor	28	
Last Caller Recall	29	
Voice Help	30	
Walk-In Class of Service	31	
Change User Password	32	
Paging	33	
DISA Login	34	
Trunk to Trunk Call Release	35	
Cancel all Feature Settings	36	
Selective Port Access	37	
Flashing on Trunk	38	
User Absent/Present	39	Yes
Account Code by Number	40	
Account Code by Name	41	
Meet Me Paging	43	
Hot Desk	44	
Do Not Disturb Override	45	
Presence	46	Yes
Live Call Screening	47	Yes
Conversation Recording	48	
Forced Release	49	
Transfer	50	
Live Call Supervision	51	
Forced Answer	52	
Change Room Clean Status	53	
Guest Number Prefix	54	
Minibar Details	55	
Mute	56	Yes
Emergency Conference	57	
Self Ring Test	58	
Call Chaining	59	Yes

Feature	Feature Number	DSS LED Activity?
SA Command Prefix	60	
COSEC Door Open	61	
Floor Service	62	
Keypad Lock	63	Yes
CLI Restriction	64	Yes
Call Cost Display	65	
Reminder	66	Yes
Alarm - Voice Guided	67	
Reminder - Voice Guided	68	
Blind Transfer to Voice Mail	69	
Message Wait Set/Cancel	70	Yes
Retrieve New Message	71	Yes
PMS - User Defined Fields	72	
Global Call Park - Auto	73	
Open a Door	75	
Magneto Ring Enable	77	
Invoke RCOC	78	
Scheduled Call Forward	79	Yes
E&M Manual Priority Intrusion	80	
E&M Forced Release Order	81	
Department Call Forward	82	
General Mailbox	83	
Intercom	84	
Terminate Conference	86	
Leave Temp. / Rejoin Conf.	87	
Call Forward - When Not Registered	88	Yes
DSS Call Pick-up-Station	89	Yes
DSS Call Pick-up-Trunk	90	Yes
Trunk Access Code 1(TAC1)	-	
Trunk Access Code 2(TAC2)	-	
Trunk Access Code 3(TAC3)	-	
Trunk Access Code 4(TAC4)	-	
Trunk Access Code 5(TAC5)	-	
Trunk Access Code 6(TAC6)	-	
Voice Mail	-	Yes

Table: “Function Type 22: SA Commands”

SA Command Name	Function Number	DSS LED Activity?
Check-In	001	
Check-Out	002	
Guest Name	003	
Guest Group	004	
Guest-In/Out	005	
Guest Title	006	
Change Check-In Profile of Room	007	
Change Occupancy Status of Room / Extension	008	
Change Clean status of Room/Extension	009	
Room Shift	010	
Reprint Check Out Report	011	
Print Room Status Report	012	
Print Alarm Status Report	013	
Delete Checked Out calls	014	
Set DND-Remote	015	
Set Dynamic Lock settings - Remote	016	
Set Alarm -Remote	017	
Assign Call Budget to a station	018	
Assign / un-assign Mailbox to a Station - Remote	019	
Set Call Forward - Remote	020	
Set Call Forward for all Stations - Remote	021	
Assign Station User Greeting Message	022	
Display & Acknowledge System Activity	023	Yes
Display & Acknowledge System Fault	024	Yes
Station Budget Display - Remote	025	
Change User Password of a Station - Remote	026	
Lock/unlock DKP's Keypad - Remote	027	
User Absent / Present - Remote	028	
Change SA password	029	
Change SA mode timer	030	
Display Registered GSM Network ID	031	
Set Day/Night Mode	032	Yes
Terminate Security dialing	035	
Clear System Activity Log	036	

SA Command Name	Function Number	DSS LED Activity?
Start/Abort SAL in Offline mode	037	
Start/Abort SAL in Online mode	038	
-	039	
Cancel Dial in Conference	040	
Start/Abort SFL in Offline mode	041	
Start/Abort SFL in Online mode	042	
Display Port Parameters	043	
Start/Abort Online OG Report	044	
OG Print Filter: Print calls originated from station(s)	045	
OG Print Filter: To Print calls terminated from CO	046	
OG Print Filter: To Print calls terminated from BRI	047	
OG Print Filter: To Print calls terminated from T1E1	048	
OG Print Filter: To Print calls terminated from E&M	049	
OG Print Filter: To Print calls terminated from Mobile	050	
OG Print Filter: To Print calls terminated from SIP	051	
OG Print Filter: To Print calls Department Bill Group wise	052	
OG Print Filter: To print calls made on date(s)	053	
OG Print Filter: Print calls made between time	054	
OG Print Filter: To Print calls made to numbers matching with the numbers programmed in the Number List	055	
OG Print Filter: To Print calls of Duration more than this time	056	
OG Print Filter: To Print calls of Units more than the units programmed	057	
OG Print Filter: To Print calls made to account code	058	
Assign default OG Print filters	059	
Start/Abort offline report	060	
Enable/ Disable OG Schedule Reports	061	
Program Time for Daily OG Scheduled Reports	062	
Program Day and Time for OG Weekly Scheduled Reports	063	
Program Date and Time for OG Monthly Scheduled Reports	064	
Delete calls made by station(s)	065	
Delete calls made on/from date	066	
Clear SMDR OG buffer	067	
Start/Abort Internal calls Report	068	
Set filter to print Internal calls Report Station wise	069	

SA Command Name	Function Number	DSS LED Activity?
Set filter to print internal calls with duration greater than that given here	070	
Start/Abort Offline Internal Call Report	071	
Enable/ Disable Internal Scheduled Reports	072	
Program Time for Internal Daily Scheduled Reports	073	
Program Day and Time for Internal Weekly Scheduled Reports	074	
Program Date and Time for Internal Monthly Scheduled Reports	075	
Clear SMDR Internal Buffer	076	
Start/Abort Online - IC Report	077	
Set filter to print all Normal calls	078	
Set filter to print all Built-In Auto Attendant calls	079	
Set filter to print all Unanswered calls	080	
Set filter to print all Built-In Auto Attendant Unanswered calls	081	
Set filter to print all DISA calls	082	
Set filter to print all calls with speech duration More than timer	083	
Set filter to print all calls unanswered for duration More than timer	084	
Set filter to print all calls kept on hold for duration more than timer	085	
Set filter to print all IC calls received by the station	086	
Set filter to print all IC calls recd. On the CO	087	
Set filter to print all IC calls recd. On the BRI	088	
Set filter to print all IC calls recd. On the T1E1	089	
Set filter to print all IC calls recd. On the E&M	090	
Set filter to print all IC calls from Mobile	091	
Set filter to print calls received from SIP	092	
Set filter to print all IC calls recd. on/from date	093	
Set filter to print all IC calls recd. at/from-to Time	094	
Set filter to print all IC calls recd. From nos. matching the Number List	095	
Default IC Print filters	096	
Abort/Start IC Offline Report	097	
Enable/ Disable IC Scheduled Report	098	
Program Time for IC Daily Scheduled Reports	099	
Program Day and Time for IC Weekly Scheduled Reports	100	
Program Date and Time for IC Monthly Scheduled Reports	101	
Clear SMDR-IC buffer	102	

SA Command Name	Function Number	DSS LED Activity?
Start/Abort Printing of Online T1E1 Performance Report	103	
Start/Abort Offline T1E1 Performance Report	104	
Signal Strength of Mobile Port	105	
Enable/Disable Call Cost Display for a Station	106	
Start/Abort Hotel/Motel Activity log in Offline mode	107	
Start/Abort Hotel/Motel Activity log in Online mode	108	
Display and Acknowledge Hotel/Motel Activity	109	Yes
Change Guest VIP Status of Station	110	
Change Phone Ringing Pattern of Room	111	
Print Reminder Status Report	112	
Reminder - Remote	113	
Voice Guided Alarm - Remote	114	
Voice Guided Reminder - Remote	115	
Redirect Messages of a Station	116	
To record Holiday message	117	
Enable/Disable Scheduled Alarm Report	118	
Program Time for Scheduled Alarm Report	119	
Enable/Disable Scheduled Reminder Report	120	
Program Time for Scheduled Reminder Report	121	
Request Database Synchronization to PMS	122	
Enable/Disable Scheduled Room Status Report	123	
Program Time for Schedule Room Status Report	124	
Enable/Disable Scheduled Change of Room Clean Status	125	
Program Time for Schedule Change of Room Clean Status	126	
-	127	
Enable/Disable Internal Call Block For Guest Phones	128	Yes
Software Version Revision Display of CPU Card	129	
User Definable Fields	130	
OG Print Filter: To print calls originated from CO	131	
OG Print Filter: To print calls originated from BRI	132	
OG Print Filter: To print calls originated from T1E1	133	
OG Print Filter: To print calls originated from E&M	134	
OG Print Filter: To print calls originated from Mobile	135	
OG Print Filter: To print calls originated from SIP	136	
To change CPU from Active state to Standby (only for ULSB MK III)	137	

SA Command Name	Function Number	DSS LED Activity?
Set Budget Type for CO Port	138	
Program Budget Amount for CO	139	
Program Free Minutes on CO	140	
Set Budget Type for T1E1 Port	141	
Program Budget Amount for T1E1	142	
Program Free Minutes on T1E1	143	
Set Budget Type for BRI Port	144	
Program Budget Amount for BRI	145	
Program Free Minutes on BRI	146	
Set Budget Type for SIP	147	
Program Budget Amount for SIP	148	
Program Free Minutes on SIP	149	
Set Budget Type for MOBILE	150	
Program Budget Amount for MOBILE	151	
Program Free Minutes on MOBILE	152	
Call Budget Reset Mode for CO	153	
Scheduled Date to Rest Call Budget Statistics on CO	154	
Reset consumed Budget Amount /minutes on CO manually	155	
Call Budget Reset Mode for T1E1	156	
Scheduled Date to Reset Call Budget Statistics on T1E1	157	
Reset consumed Budget Amount /minutes on T1E1 manually	158	
Call Budget Reset Mode for BRI	159	
Scheduled Date to Rest Call Budget Statistics on BRI	160	
Reset consumed Budget Amount /minutes on BRI manually	161	
Call Budget Reset Mode for SIP	162	
Scheduled Date to Rest Call Budget Statistics on SIP	163	
Reset consumed Budget Amount /minutes on SIP manually	164	
Call Budget Reset Mode for MOBILE	165	
Scheduled Date to Rest Call Budget Statistics on MOBILE	166	
Reset consumed Budget Amount /minutes on MOBILE manually	167	
Swap Ports for CO8/SLT8 - MAG8 Card	168	
Reset mobile ‘avg call dur’ & ‘ans/seizure’	169	
Set/Cancel Scheduled Call Forward	170	
-	171	
-	172	

SA Command Name	Function Number	DSS LED Activity?
-	173	
-	174	
To Broadcast Message	175	

Table: “Function Type 24: Special Keys”

Special Key Name	Function Number	DSS LED Activity?
Hold	001	No
Exclusive Hold 1 to 8	002 to 009	Yes
Pause	010	No
One Touch Transfer	011	No
Ringer Mute	012	Yes
Release	015	No
Acknowledge	016	No
Enter	017	No
Speed	018	No
Emergency Alarm Log	019	Yes
Speaker	020	Yes
Wakeup Call Log	021	Yes
Headset	022	Yes
Cancel	023	No
Answer	024	No
Call Log	025	Yes (for Missed Call)
Local Menu	026	No

Examples:

Let us attempt to program the same sample Operator Template we customized earlier for EON48 using a telephone. For this, we need to know the location of the key, that is, the key number.

Existing function on the key	To be replaced by	Location of the key on the Key Map
DKP4	Hotline	Key No. 01
DKP3	SA Command for “Remote DND”	Key No. 02
DKP2	DND-Override	Key No. 03
SLT4	Mobile Trunk 1 (MOBILE1)	Key No. 09
SLT3	SIP Trunk 1 (SIP1)	Key No. 10
SLT2	BRI Trunk 1 (BRI 1)	Key No. 11

Existing function on the key	To be replaced by	Location of the key on the Key Map
SLT1	T1E1 Channel Number 2 of T1E1 Trunk 1	Key No. 12

To assign Hotline to the key currently assigned to DKP4 key (key 01) in the Operator Template, dial:

- ***1261-1-1-3-01-18-15-00***
  Where,
  1 is Operator Template
  3 is for EON48 Terminal
  01 is the Key Number
  18 is Function Type for Features
  15 is Function Number for Hotline feature.
  00 is Channel.

To assign Mobile Trunk 1 in place of SLT4 on DSS key 09 on the Operator Template, dial:

- ***1261-1-1-3-09-25-01-00***
  Where,
  1 is Operator Template
  3 is for EON48 Terminal
  09 is the Key Number
  25 is Function Type for Mobile Trunks
  01 is Function Number, that is, the port offset of the Mobile Port.
  00 is Channel

To assign BRI Trunk 1 in place of SLT2 on DSS Key 11 on the Operator Template, dial:

- ***1261-1-1-3-11-04-01-00***
  Where,
  1 is Operator Template
  3 is for EON48 Terminal
  11 is the Key Number
  04 is Function Type for BRI Trunks
  01 is Function Number, that is, number of the BRI port.
  00 is Channel number.

To assign Channel 2 of T1E1 Trunk 1 in place of SLT1 on DSS Key 12 on Operator Template, dial:

- ***1261-1-1-3-12-05-01-02***
  Where,
  1 is Operator Template
  3 is for EON48 Terminal
  12 is the Key Number
  05 is Function Type for T1E1 Trunks.
  01 is Function Number, that is, number of the T1E1 port.
  02 is Channel number.

To assign a DKP Key template to a DKP, dial:

- ***1221-1-DKP-DKP Key Template*** to assign the template to a single DKP.
- ***1221-2-DKP-DKP-DKP Key Template*** to assign the same template to a range of DKPs
- ***1221-*-DKP Key Template*** to assign the same template to all DKPs
  Where,
  DKP is number of the Software Port of the DKP to which the Template is to be assigned, from 001 to 128.
  DKP Key Template is
  0 for Personalized key map

1 for Operator Template
2 for Executive's Template
3 for Hotel Attendant's Template
4 for Guest Template.

For example, to assign the customized Operator Template to DKP 001 and DKP002, you may dial:

- ***1221-1-001-1*** to assign the template to DKP001
- ***1221-1-002-1*** to assign the template to DKP002
  OR
- ***1221-2-001-002-1*** to assign the same template to both DKPs at one go.

- Exit SE Mode.

Personalizing Key Maps

You can personalize the Key Maps of individual DKPs (EON) and SPARSH IP Phones, instead of using the Key Templates.

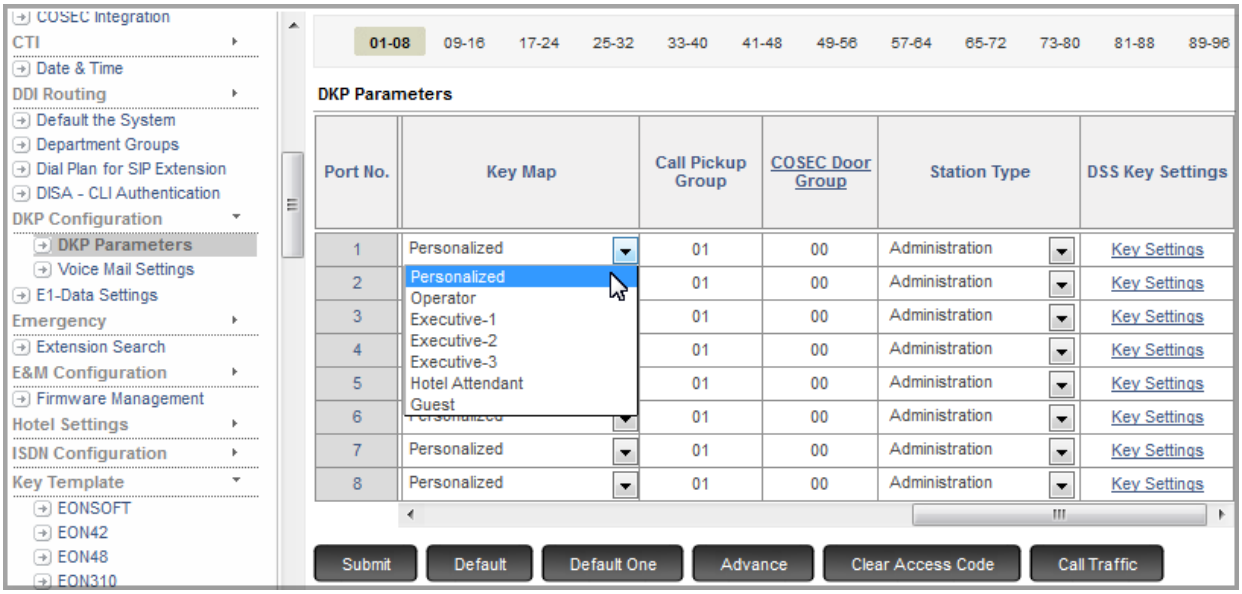
When you personalize the Key Map of a DKP, make sure 'Personalized' option is selected as the *Key Map* in the *DKP Parameters* of the DKP.

Similarly, when you personalize the Key Map of an SPARSH IP phone, make sure 'Personalized' option as the *Key Template* in the *SIP Extension Settings - Location* of the phone.

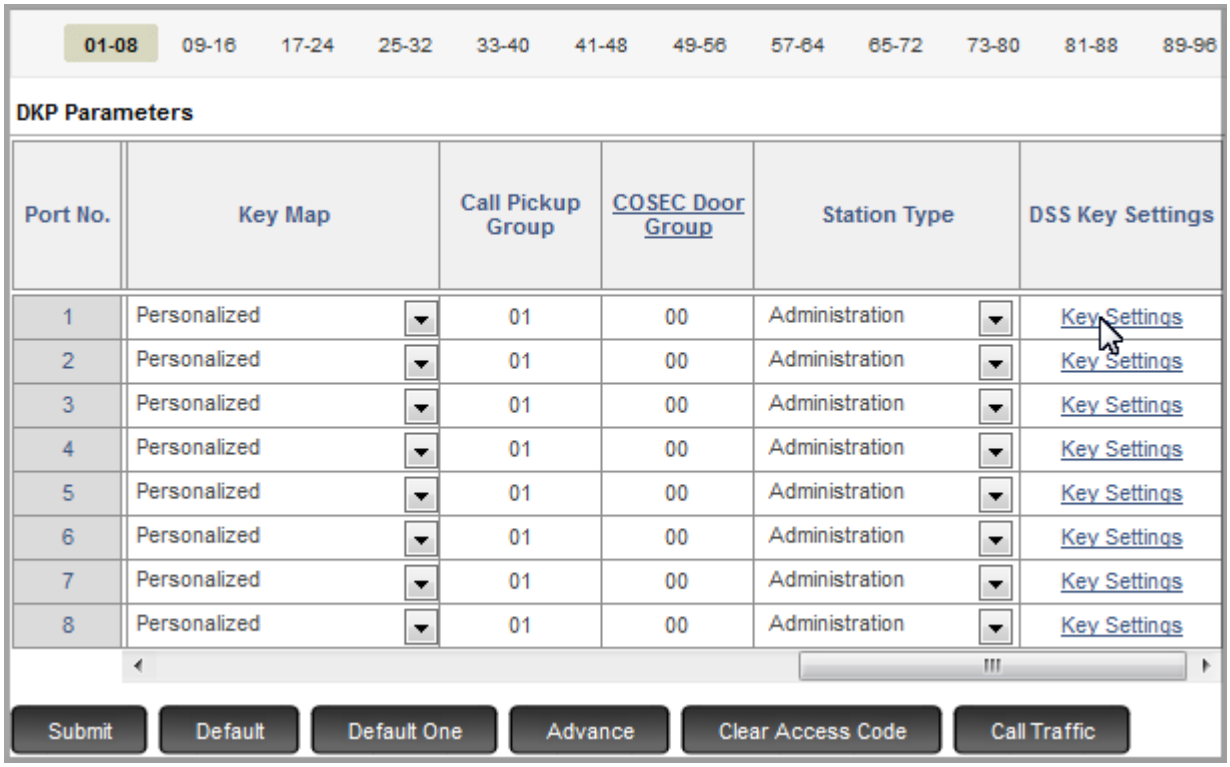
The Personalized option is not applicable for SPARSH VP110.

Personalizing Key Map using Jeeves

- Log in as System Engineer.
- Under **Configuration**, click **DKP Parameters**.
- Go to the DKP you want to assign a personalized key map, for example, DKP001.

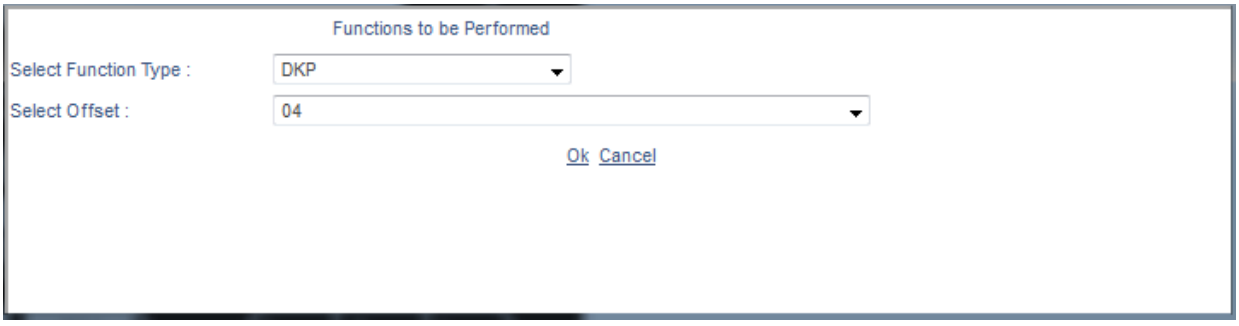


- Select **Personalized** as the **Key Map** option for the DKP.
- Click **Submit** to save Key Map selection.
- Click the **Key Settings** link under **DSS Key Settings**.

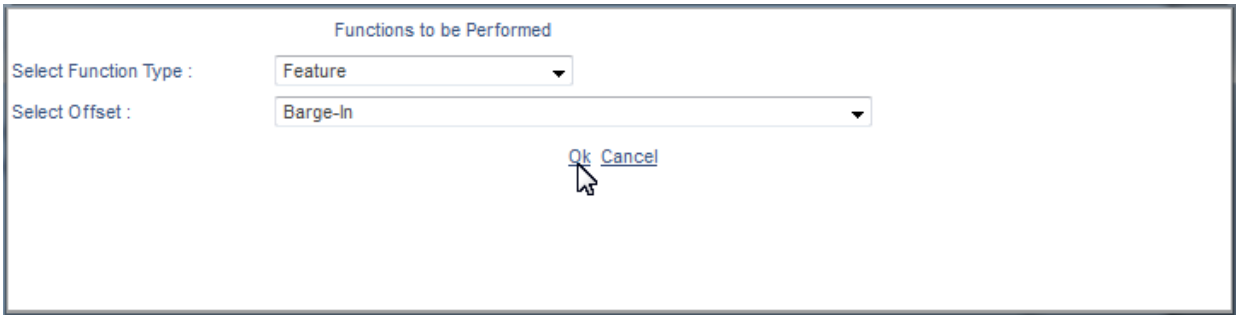


- The current Key Map of the DKP opens.

- To assign features to keys, follow the same steps as you did for customizing the key templates.
- Click the key you want to assign the function. For example, you want **Barge-In** on the key assigned to **DKP4**, click this key.
- The options for the Functions to be Performed by the key will open in a new window.



- Click the option **Feature** in the **Function Type** list.
- Click the option **Barge-In** in the **Select Offset** list.



- Click **OK**. The window will close. The new label will appear on the key.

- Repeat the same steps to program another key on this key map.
- When you have completed personalizing the key map, close the window.
- Click **Submit** to save.
- Follow the same steps to personalize the key map of another DKP.

For instructions on personalizing the Key Map of the Extended IP Phone, see “DSS Key Settings” in “Configuring Matrix SPARSH VP330”, “DSS Key Settings” in “Configuring Matrix SPARSH VP248”, “DSS Key Settings” in “Configuring Matrix SPARSH VP310”, “DSS Key Settings” in “Configuring Matrix SPARSH VP210” and “DSS Key Settings” in “Configuring Matrix SPARSH VP510”. The Personalized option is not applicable for SPARSH VP110.

Personalizing Key Maps using a Telephone

- Enter SE mode.

  To assign a Personalized Key map to a DKP, dial:
  - ***1221-1-DKP-0*** to assign personalized key map to a single DKP.
  - ***1221-2-DKP-DKP-0*** to assign the personalized key map to a range of DKPs.
  - ***1221-*-0*** to assign personalized key map to all DKPs.
    Where,
    DKP is number of the Software Port of the DKP to which the Template is to be assigned, from 001 to 128.
    0 is for Personalized key map.

    For example, to assign the personalized Key map to DKP 003, dial: 1221-1-003-0

  To assign a function to a DKP key in a personalized key map, dial:
  - ***1252-1-DKP-Key Number-Function Type-Function Number-Channel*** to assign a function to a DKP Key on a single DKP.
  - ***1252-2-DKP-DKP-Key Number-Function Type-Function Number-Channel*** to assign the same function on the same key number on a range of DKPs.
  - ***1252-*-Key Number-Function Type-Function Number-Channel*** to assign the same function on the same key number on all DKPs
    Where,
    DKP is the number of the Software Port of the DKP from 001 to 128.

    Key Number is the number of the key which is to be assigned the function/feature. The Key numbers vary according to the EON Terminal Type being used.
    Key Numbers on EONSOFT are from 01 to 25.
    Key Numbers on EON42SR are from 01 to 25.
    Key Numbers on EON48 are from 01 to 29.

    For numbering of the keys, refer the default Key Maps illustrated at the beginning of this topic.

    Function Type is the function to be performed by the key, from 00 to 34.

    Function Number defines the exact function under each Function Type selected for the key. Refer the tables above for the complete list of function types and function numbers.

    Channel is the number of the channel in the BRI or T1E1 PRI Line to be accessed.

    Channel number for BRI Line is
    01 for Channel 1
    02 for Channel 2
    00 for both Channels

    Channel number for T1E1 PRI Line is
    01 to 30 for Channels 1 to 30 respectively.
    00 for all Channels

    Channel number for all function types other than BRI and PRI is ***00***

  For example, you want Barge-In on key number 04 of DKP-003, dial:
  - ***1252-1-003-04-18-19-00***
    Where,
    003 is DKP-003

04 is the Key Number on DKP-003
18 is the Function Type for Features
19 is the Function Number for Barge-In
00 is Channel number

- Exit SE Mode.

After you have finished assigning key templates and key maps, test the functioning of the keys.

Programming DSS Console Keys

Direct Station Selection (DSS) Consoles are devices that function as extensional buttons for EON/SPARSH VP510 Phones, providing more buttons for single-touch Direct Station Calling and feature access.

You can attach two DSS64 Consoles to a single EON48/310 and four DSS532 Consoles to a single EON510/VP510 to further increase the number of keys. Refer *"Direct Station Selection Console"* to know more.

For installation instructions, refer the topics Installing EON, Installing DSS Consoles, under *"Installing ETERNITY LENX"*, *"Installing ETERNITY MENX"*, *"Installing ETERNITY GENX"* and *"Installing ETERNITY PENX".*

For instructions:

- to install the DSS532 with SPARSH VP510, see *"Installing DSS532 with SPARSH VP510"*.
- to install the DSS64/DSS532 with EON, see *"Installing DSS Consoles"*.

- *The system supports two soft embedded DSS Consoles for EONSOFT. These will not be displayed in DSS Status.*

Programming DSS Console Keys using Jeeves

You can configure the DSS Console either offline, that is before you have connected the DSS Console or after you have connected the Console.

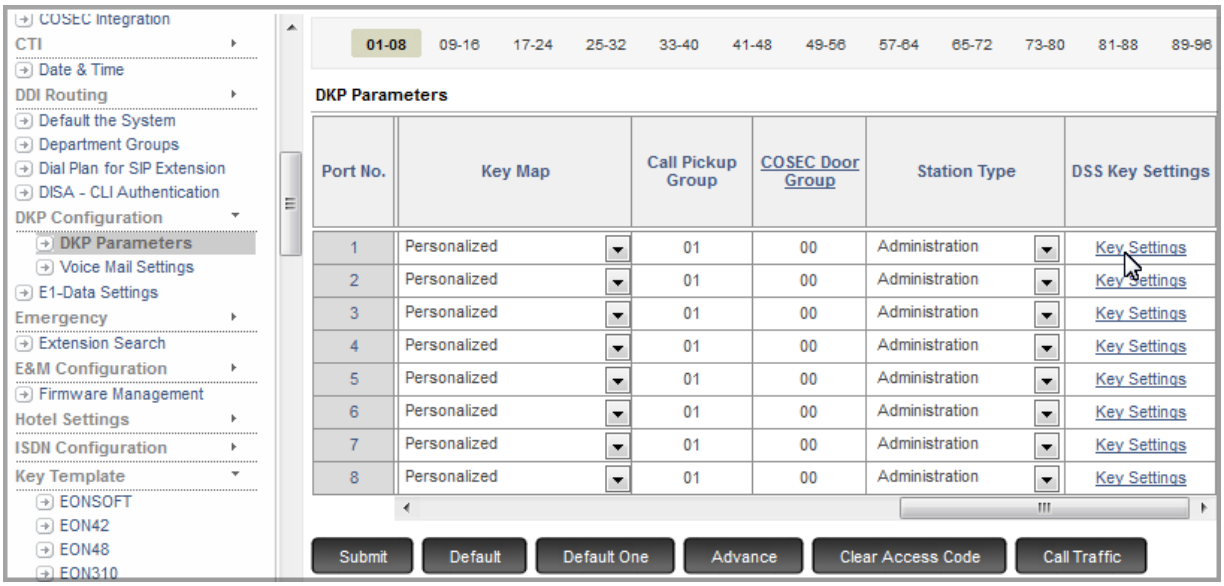
To configure DSS Console keys:

- refer to *"Configuring DSS Console Keys connected to DKP Phones (EON)"*, if the console is connected to a DKP.
- refer to *"Configuring DSS Console Keys connected to SPARSH VP510"*, if the console is connected to SPARSH VP510.

Configuring DSS Console Keys connected to DKP Phones (EON)

Configuring DSS Console Keys Offline,

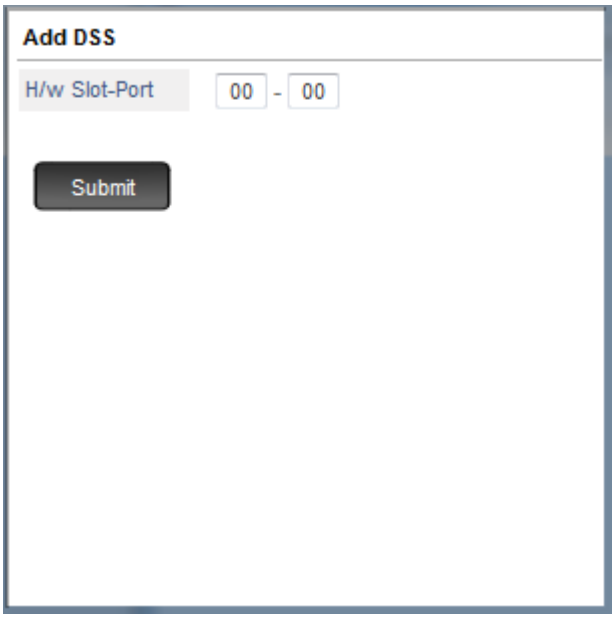
- Log into Jeeves as System Engineer
- Under **Configuration**, click **DKP Parameters**.
- Go to the DKP Port number to which you wish to attach the DSS Console.
- Using the horizontal scroll bar on the page, scroll to **DSS Key Settings**.
- Click **Key Settings**.



The **Key Settings** page opens.

- In **Device Type** select the type of DKP — EON48, EON310, EON510 — you wish to connect to this port.
- Click **ADD DSS**.

- If you have selected Device Type as EON48 or EON310, the **ADD DSS** window opens.

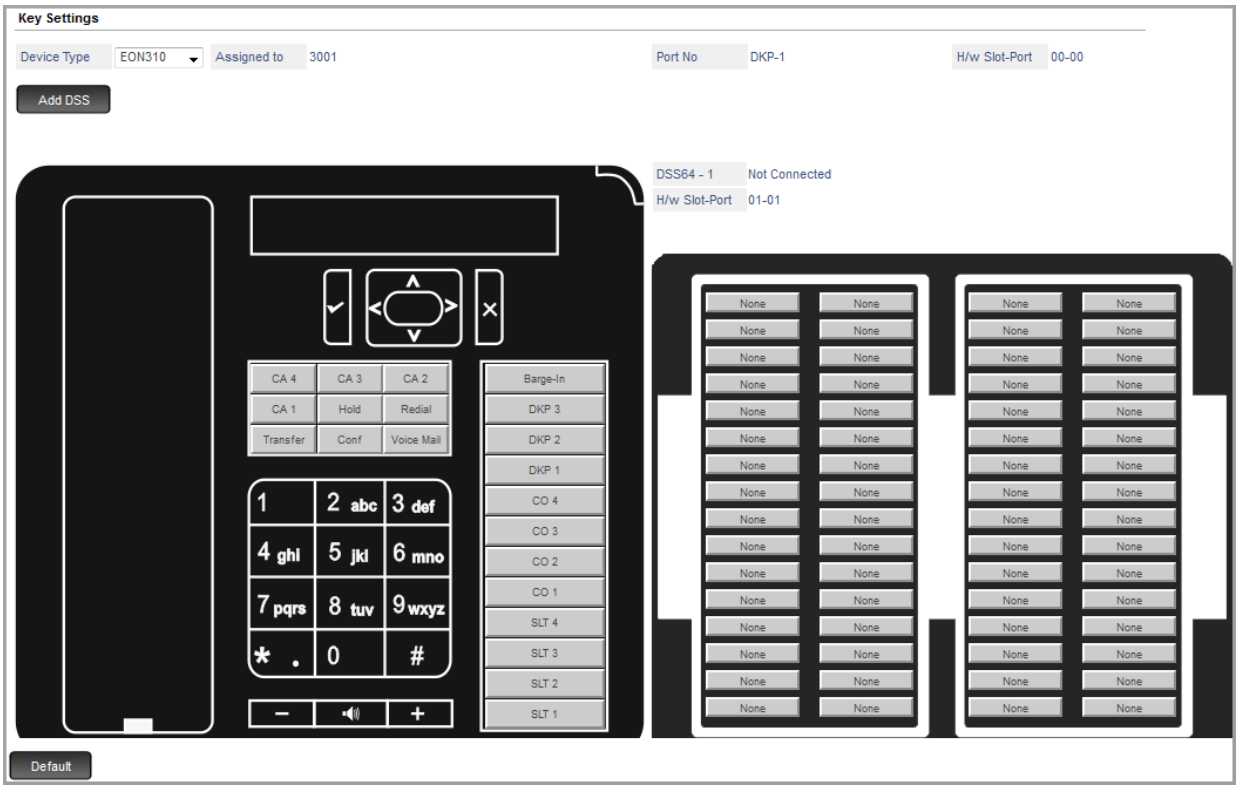


In **H/w Slot-Port,** enter the Hardware slot and port on which you wish to connect the DSS Console.

The Hardware slot port assigned now cannot be changed from Jeeves later.

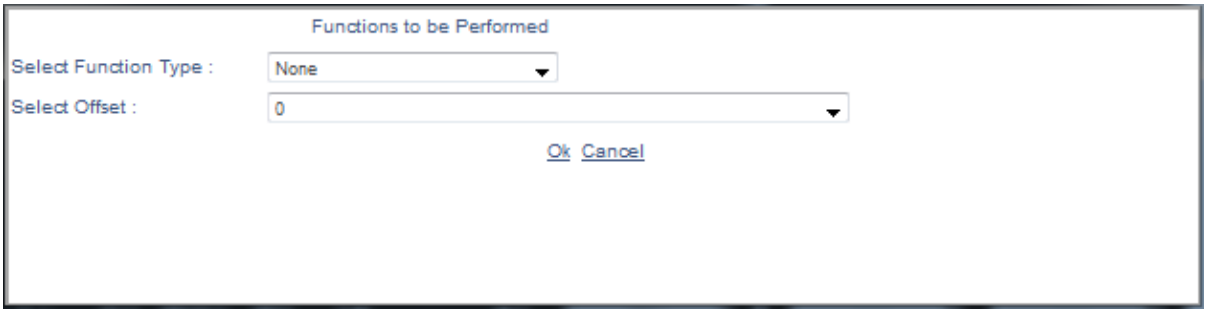
Click **Submit**. The ADD DSS window closes.

The default key map of DSS64 appears on your screen.

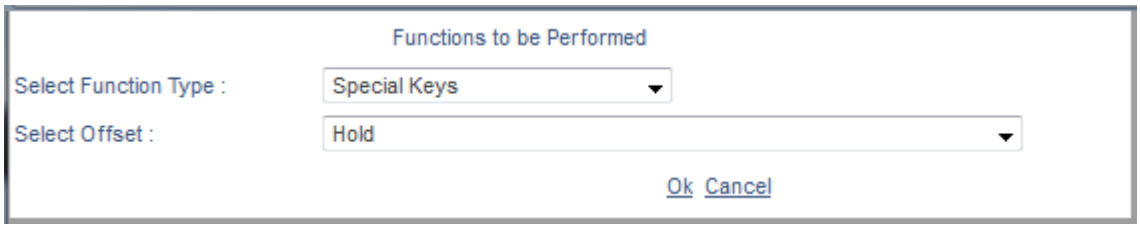


- For **EON510** as the Device Type, click on **ADD DSS** to add a DSS532 to the default key map of the phone.

- To add a DSS, click **ADD DSS**.
- By default **None** is assigned to all the DSS keys, you can now personalize the DSS Key map as per your requirement.
- Click the key on which you want to assign a feature/function.
- The options for the **Functions to be Performed** by the key will open in a new window.



- Select the desired **Function Type** to be assigned to the key and the desired **Offset** for the Function Type.



- Click **OK**.
- The function you selected will appear as the key label.

- Similarly, add another DSS and configure the keys.

- If you wish to clear all the key assignments, click the respective DSS Console check box and then click **Clear Key Assignment.**

- If you wish to interchange the key assignments of the DSS, click the check boxes of the respective DSS Consoles whose keys you wish to swap and then click **Interchange DSS Keys**.

The option Interchange DSS Keys is applicable only for DSS532.

- If you wish to remove a DSS Console, click **Remove DSS**.

- To view the status of all the DSS Consoles, click “DSS Status”**.**

Configuring DSS Console Keys after you have connected the phone as well as the Console,

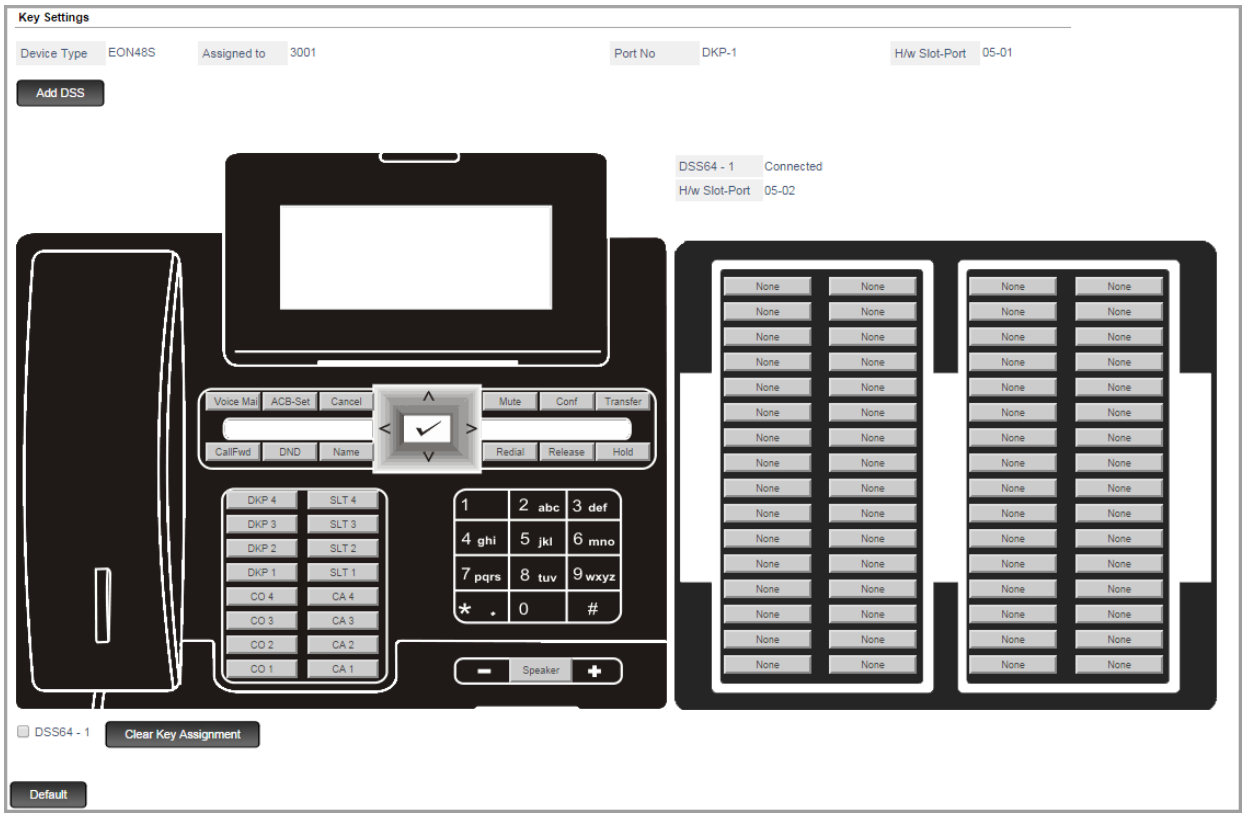
If you have connected the phone and the DSS Consoles, the system automatically detects them.

In case of DSS64 the connected DSS Console appears under **Unassigned DSS64** in **DSS Status**. You must first assign the DSS Console to a DKP Port and then you will be able to configure the DSS Keys. For instructions, see “DSS Status”.

If DSS532 is connected the default key map of the DSS Console will automatically appear on the screen and you will be able to configure the DSS Keys.

- Under **Configuration**, click **DKP Parameters**.

- Go to the DKP number to which DSS Console has been connected.
- Using the horizontal scroll bar on the page, scroll to **DSS Key Settings**.
- Click **Key Settings**. The **Key Settings** page opens.



The page displays the following:

Phone Details

- **Device Type** displays the model of EON connected — EON48, EON310 or EON510.
- **Assigned to** displays the Extension number assigned to the phone.
- **Port No** displays the software port assigned to the phone.
- **H/w Slot-Port** displays the Hardware slot and port on which the phone is connected.

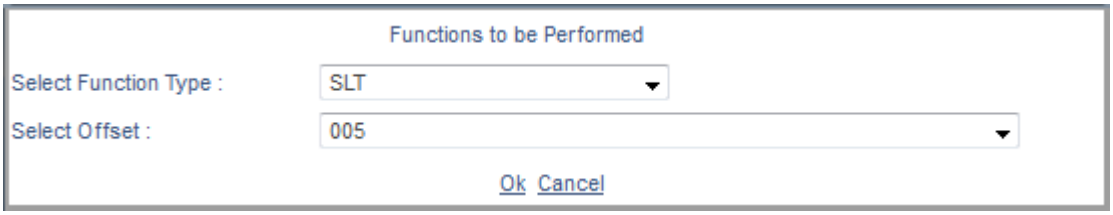
DSS Console Details

- Type of DSS Console connected along with the number and Status, that is, DSS64/DSS532 - 1 Connected or Not Connected.
- **H/w Slot-Port** displays the Hardware slot and port on which the Console is connected.

Hardware Slot-Port is applicable for DSS64 only.

- By default **None** is assigned to all the DSS keys, you can now personalize the DSS Key map as per your requirement.
- Click the key on which you want to assign a feature/function.

- The options for the **Functions to be Performed** by the key will open in a new window.



- Select the desired **Function Type** to be assigned to the key and the desired **Offset** for the Function Type.
- Click **OK**.
- The function you selected will appear as the key label.
- Similarly, configure other keys.
- If you wish to clear all the key assignments, click the respective DSS Console check box and then click **Clear Key Assignment.**
- If you wish to interchange the key assignments of the DSS, click the check boxes of the respective DSS Consoles whose keys you wish to swap and then click **Interchange DSS Keys**.

The option Interchange DSS Keys is applicable only for DSS532.

- To add another DSS Console, click **ADD DSS**.
- To view the status of all the DSS Consoles, click “DSS Status”**.**

Configuring DSS Console Keys connected to SPARSH VP510

You can connect DSS532 to SPARSH VP510 for quick access to Stations, Trunks, Features/Functions of the ETERNITY or at the touch of a single key, making call operations easy.

You can configure the DSS Console keys offline, that is before connecting the console or after you have connected the Console. If you have connected the DSS Console the system will automatically detect the same.

- Log into Jeeves as System Engineer
- Under **VoIP Configuration**, click **SIP Extension Settings**.
- Select the desired SIP Extension Number, click the desired Location 1 or 2 or 3.
- On the Location Page make sure you have selected SPARSH VP510 as the Device Type. Scroll to **DSS Key Settings**.

- Click **Key Settings**. The **Key Settings** page opens.

Phone Details

- **Device Type** displays SPARSH VP510.
- **Assigned to** displays the Extension number assigned to the phone.
- **Port No** displays the software port number as well as the Location, for example: SIP Extension -1 - Location - 1.

- Click **ADD DSS**.

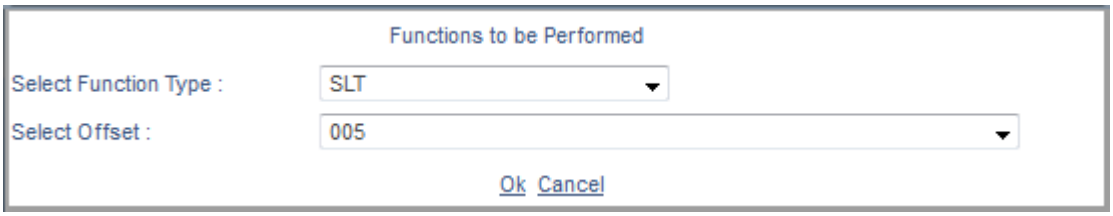
*If the DSS Consoles are connected the default Key maps will be displayed automatically. Click **ADD DSS** if you wish to add another DSS Console.*

- The default key map of DSS532 appears on your screen.

- By default **None** is assigned to all the DSS keys, you can now personalize the DSS Key map as per your requirement.

- Click the key on which you want to assign a feature/function.

- The options for the **Functions to be Performed** by the key will open in a new window.



- Select the desired **Function Type** to be assigned to the key and the desired **Offset** for the Function Type.
- Click **OK**.
- The function you selected will appear as the key label.
- Similarly, configure other keys.

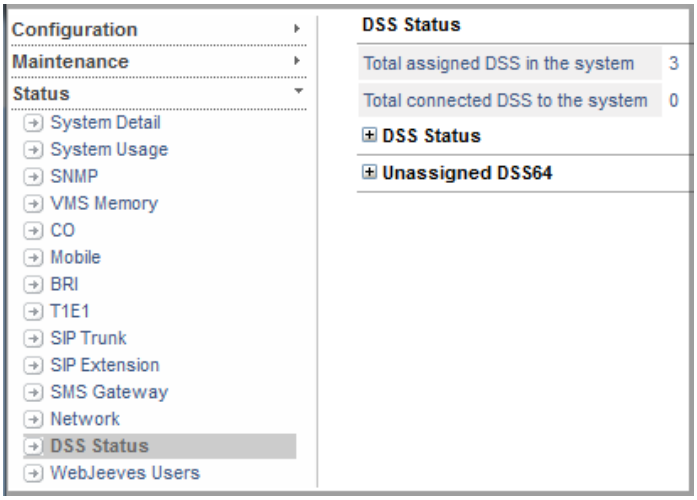
The following features cannot be assigned to DSS Keys:
- *Headset*
- *Speaker*
- *Ringer Acknowledge Key*
- *Local Menu*

- If you wish to clear all the key assignments, click the respective DSS Console check box and then click **Clear Key Assignment.**
- If you wish to interchange the key assignments of the DSS, click the check boxes of the respective DSS Consoles whose keys you wish to swap and then click **Interchange DSS Keys**.
- If you wish to remove a DSS Console, click **Remove DSS**.
- Click **ADD DSS** again to add another Console.
- To view the status of all the DSS Consoles, click “DSS Status”**.**

DSS Status

To know the details of the DSS Consoles at a glance,

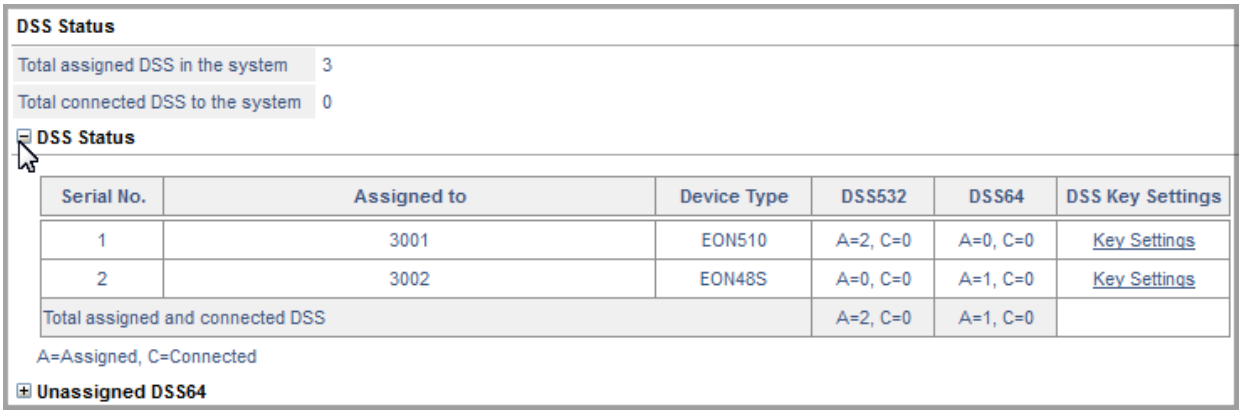
- Under **Status**, click **DSS Status**.



The following details are displayed:

- **Total assigned DSS in the system**
- **Total connected DSS to the system**

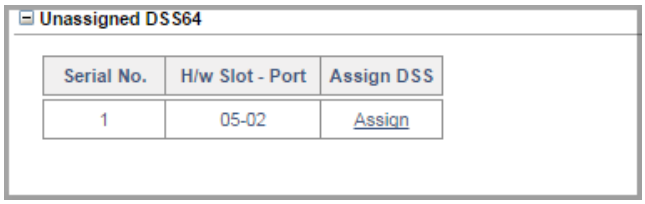
- Click **DSS Status** to expand.



The following information is displayed:

- **Serial Number**
- **Assigned to** displays the extension number to which the DSS Console is connected.
- **Device Type** displays the model of EON — EON48, EON310, EON510, SPARSH VP510 — to which the DSS Console is connected.
- **DSS532** displays the status of the console, that is Assigned or Connected.
- **DSS64** displays the status of the console, that is Assigned or Connected.
- **DSS Key Settings** provides the link to **Key Settings**. Click this link if you wish to configure the DSS Keys.

- Click **Unassigned DSS64** to expand.



If have connected any DSS64, you must assign it to a particular DKP Port and only then will you be able to configure the DSS Console Keys. You can view all unassigned DSS Consoles here.

For each unassigned console the following is displayed:

- **Serial Number**
- **H/w Slot - Port** is the Hardware Slot Port assigned to the DSS Console.
- **Assign DSS** displays a link **Assign**. To assign the DSS Console to a DKP Port click the link.
  - The **Assign DSS** window opens.

  - In **Select DKP Extension**, select the DKP Port number to which to wish to assign the console.
  - Click **Submit**.

- After you have assigned the DSS Console, under **Configuration** click **DKP Configuration.**

- Click **DKP Parameters**. Go to the DKP Port to which you assigned the DSS Console.

- Scroll to **DSS Key Settings** and click **Key Settings**.

- The DSS Console you assigned to this DKP Port appears on the screen. You can now personalize the DSS Key map as per your requirement.

Configuring ISDN Terminals

Depending on the number of BRI ports available to you and the type of Point-to-Multipoint Configuration (Short or Extended Passive Bus), a maximum of 64 ISDN Terminals can be connected to the SARVAM UCS.

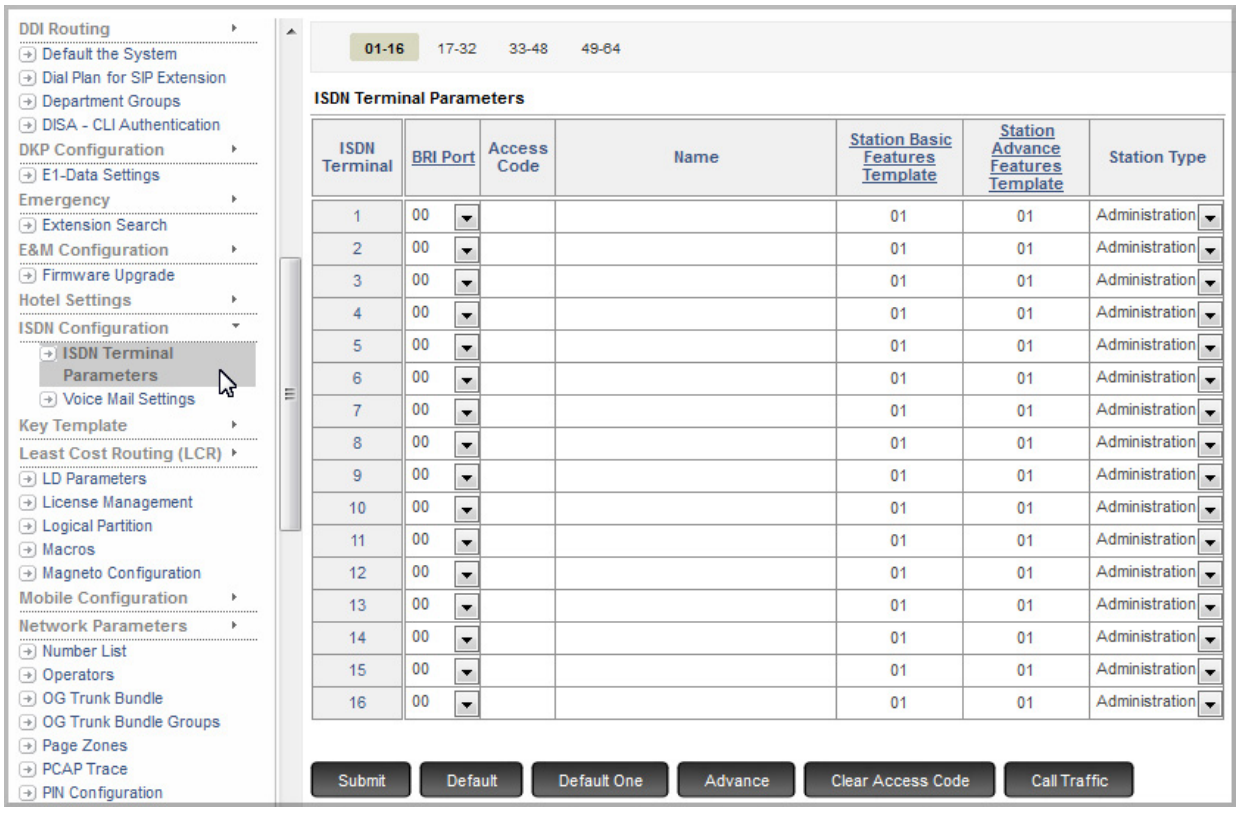
The number of ISDN Terminal extensions available to you for configuration also depends on the number of ISDN Terminals you have specified on the “System Pre-requisites” page.

If you have enabled **On-Site Configuration**, the system will provide you only those ISDN terminal ports that are actually present in the system for configuration.

Configure **ISDN Terminal Parameters** using Jeeves or by dialing commands from a Telephone.

Configuring ISDN Terminals using Jeeves

- Log into Jeeves as System Engineer.
- Under **Configuration**, click **ISDN Configuration**.
- Click **ISDN Terminal Parameters** to open the page.



- Configure the following parameters for each ISDN Terminal on this page:
  - **ISDN Terminal:** This non-editable field is the number of the software port of the ISDN Terminal.
  - **BRI Port:** This is the number of the BRI Software port to which the ISDN Terminal is connected on a Short/Extended Passive Bus line.

- **Access Code:** Assign Station Access codes to the ISDN Terminal. Station Access codes are commonly referred to as Extension Numbers. These may be a combination of a maximum of 6 digits, which are dialed to call the ISDN Terminal to which they are assigned.

  A maximum of 6 digits are allowed in an Access Code. By default, the Station Access Codes are blank for all ISDN Terminal ports.

  To assign Station Access Codes according to your preference and requirement to a range of ISDN Terminals, see “Assigning Access Codes to a Range of Extensions”.

If you decide to customize the Station Access Codes, make sure that the numbers do not clash with any other Access Code in the 'Dial' phase. Refer the topics “Access Codes” and “Conflict Dialing” to know more.

- **Name:** Assign a 'Name' to the ISDN Terminal. The name may be of the person who will use the ISDN Terminal or the name of a department. This name will be displayed to the called extension.

  You can program a name of maximum 18 alphanumeric characters.

- **Station Basic Feature Template:** Assign a “Station Basic Feature Template” to the ISDN Terminal.

  A Station Basic Feature Template includes a set of features that completely define the behaviour of the Extension, such as Time Table, Operator access, Trunk Access, Class of Service, Toll Control, Call Budget, and Station Message Detail Records (storage of Incoming and Outgoing Call details).

  By default, Station Basic Feature Template 01 is assigned to all extensions of the system that also includes SLT ports, ISDN ports and E&M Lines with Station as Orientation Type.

  Check if the default template fulfills the feature requirements (like “Class of Service (COS)”, “Toll Control”, “OG Trunk Bundle Group”, etc.) of the ISDN Terminal.

  If the default Template 01 fulfills the feature requirements and if the same features are to be allowed to all ISDN Terminals, retain Template 01.

  If different sets of features are to be allowed to different ISDN Terminals, then prepare separate Station Basic Feature Templates and apply them on the ports. To do this,

  - Under **Configuration**, click **Station Basic Feature Template** to open the page.
  - Select a Template number, for example Template number 12.
  - Customize Template number 12 and click 'Submit' at the bottom of the page.
  - Now go back to the **ISDN Terminal Parameters** page.
  - Enter the number of the Template you customized, Template 12, in the **Station Basic Feature Template** field of the ISDN Terminal, for instance, ISDN-01, on which you want to apply this template. If you want to apply this template to other terminals too, like ISDN-02, 03, and 04, assign the Template 12 to all them.
  - Click **Submit** at the bottom of the page to save changes.
  - Repeat the same steps to customize and assign a different Template to another ISDN Terminal.

Also, refer the topic “Station Basic Feature Template” to know more about customizing the templates and applying on extension ports.

- **Station Advanced Feature Template:** Assign a Station Advanced Feature Template to the ISDN Terminal. The Advanced Feature Template consists of a set of less commonly used features like Alarm Notification Type, Caller ID Presentation for Transferred Calls, DDI Incoming Call Routing, Storage of Internal Calls, Call Duration Control, Floor Service, Call Taping, etc.

  By default Station Advanced Feature Template 01 is assigned to all extensions of the SARVAM UCS, which includes ISDN ports, SLT ports, DKP Ports and E&M Lines configured as Stations.

  Check if this default template fulfills the feature requirements of the ISDN Terminal by clicking the 'Station Advanced Feature Template' link.

  If the default Template 01 fulfills the feature requirements, and if the same features are to be allowed to all ISDN Terminals, retain Template 01.

  If different sets of features are to be allowed to different ISDN Terminals, then prepare separate Station Advanced Feature Templates and apply them on the ports.

  To do this,

  - Under **Configuration**, click **Station Advanced Feature Template** to open the page.
  - Select a Template number, for example 04.
  - Customize Template number 04.
  - Click **Submit** at the bottom of the page.
  - Now go back to the **ISDN Terminal Parameters**' page.
  - Enter the number of the Template you customized, Template 04 in the **Station Advanced Feature Template** field of the ISDN Terminal, for instance, ISDN-01, on which you want to apply this template. If you want to apply this template to other terminals too, like ISDN-01, 02, and 03, assign the Template 04 to all these ports.
  - Click **Submit** at the bottom of the page to save changes.
  - Repeat the same steps to customize and assign a different Template to another ISDN Terminal.

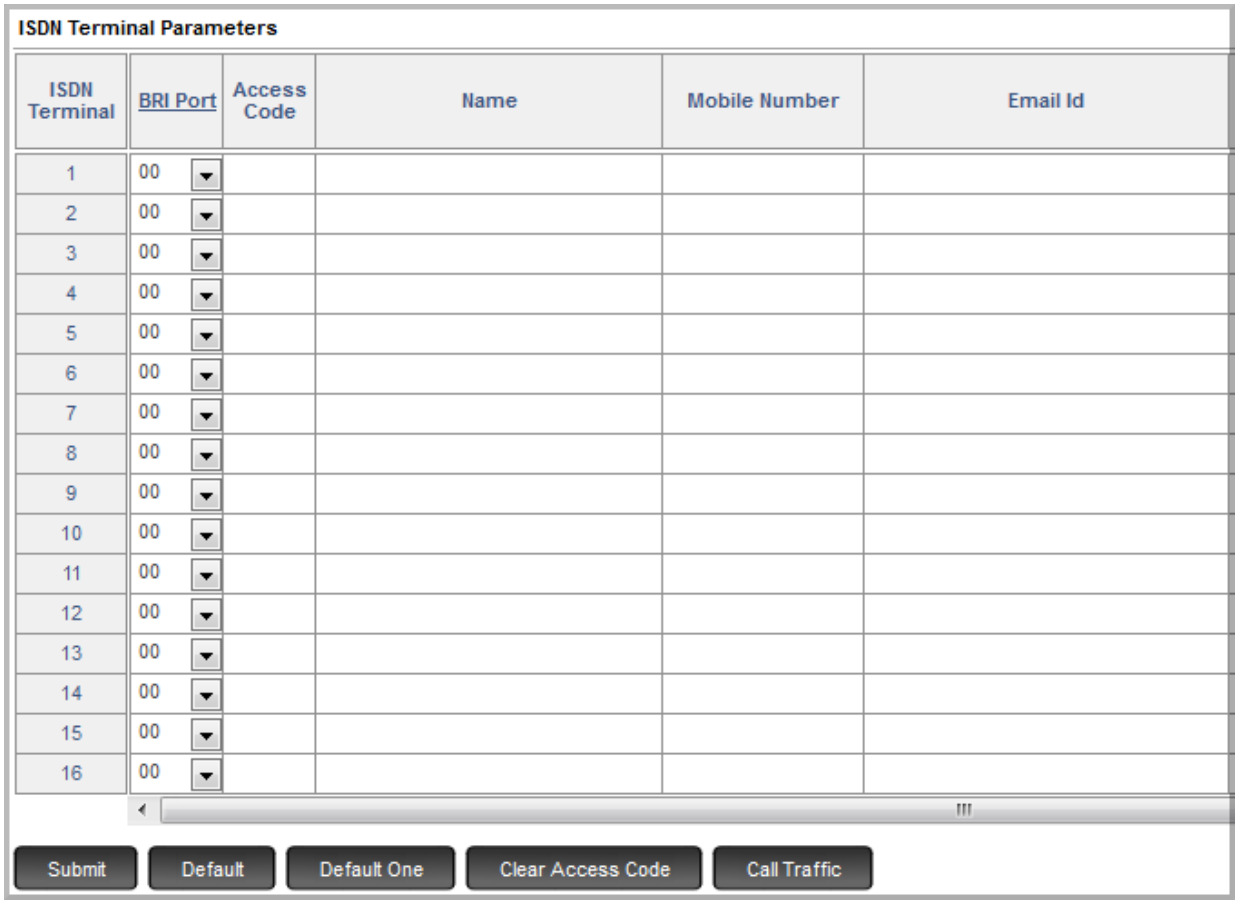
  Also refer the topic “Station Advanced Feature Template” for instructions on customizing these templates and applying them on the extension ports.

- **Station Type:** Make sure you select the option Assistant, if it is the Operator Extension. The system will play beeps during the conference to the participants to indicate the presence or absence of the Operator. To know more, see “Conference-3 Party”, “Conference-Multiparty” and “Conference Dial-In”.

  For the Hotel Application of the SARVAM UCS, extensions are identified as — Administrator, Assistant or Guest extensions according to the intended user of the ISDN Terminal. The system will consider the Administrator and Assistant options as same. When the Station Type is selected, the system will automatically assign the Guest and Administrator (Hotel Staff) features to the ISDN Terminal. To know more, refer the *SARVAM UCS Hospitality System Manual*.

Advanced features

The above listed parameters fulfill the ISDN Terminal configuration requirements of most users. If you need to use other features like Personal Directory, Call Pick-Up Groups, user details, you may click the **Advance** button.



Configure the following parameters:

- **Mobile Number:** Enter the Mobile Number of the extension user you wish to store. The Number can be a maximum of 16 digits.

- **Email ID:** Enter the Email ID of the extension user you wish to store. The Email ID can be a maximum of 64 characters.

- **SMS/Email Group Type:** You can assign the extension user to a Group. Select the desired **SMS/Email Group Type** from the list.The system clubs together extension users assigned the same Group. Default: None. For details, see “SMS/Email Group”.

- **Personal Directory:** Enter the number of the **“**Personal Directory**”** that you want to assign to the ISDN Terminal. A Personal Directory is a list of 25 frequently dialed numbers, each of which are stored by Index number (location code), Name and Trunk Access Codes (“OG Trunk Bundle Group”). The Personal Directory is necessary for using the features “Abbreviated Dialing” and “Dial By Name”.

  The Personal Directory number that you assign to an ISDN Terminal must also be programmed either by you, the System Engineer, or by the ISDN Terminal user. Refer the topic “Abbreviated Dialing” for instructions on programming the Personal Directory.

- **Priority:** Select a Priority Level for the ISDN Terminal from 1, 2, 3... to 9, with '1' being lowest Priority and '9' being highest Priority. Whenever an extension (ISDN Terminal) with higher priority calls an extension with lower priority, a triple ring is placed on the called extension. To know more, read the feature description "Priority".

  By default, the Priority of all ISDN Terminals is set to '5-Normal'.

- **Call Pick-Up Group:** Configure this parameter if you want to assign the ISDN Terminal to a particular "Call Pick Up" group.

  Call Pick Up allows the ISDN Terminal user to 'pick up' (answer) calls ringing on any other extension, by dialing a feature code, without physically going to the ringing extension. For this to work, both the ringing extension and the extension picking up the call must be in the same 'Call Pick Up Group'. Refer "Call Pick Up" for instructions on how to create groups. You can create as many as 99 groups numbered from 01 to 99.

  Enter the number of the Call Pick-Up Group you created for this ISDN Terminal in this field.

- Repeat the same steps to configure other ISDN Terminals.

- If you have completed configuration of all the above listed ISDN Terminal Parameters, click 'Submit' at the bottom of the page to save your changes.

Configuring ISDN Terminals using a Telephone

- Enter SE mode using a DKP/SLT.

  To assign BRI Software Port to an ISDN Terminal, dial:
  - ***7301-1-ISDN Terminal-BRI*** to assign BRI Software Port to a single ISDN Terminal.
  - ***7301-2-ISDN Terminal-ISDN Terminal-BRI*** to assign the same BRI Software Port to a range of ISDN Terminals.
  - ***7301-*-BRI*** to assign the same BRI Software Port to all ISDN Terminals.
    Where,
    ISDN Terminal is the Software Port number of the Terminal from 01 to 64.
    BRI is the number of the BRI Software Port to which the ISDN Terminal is connected, from 01 to 32.
    5. Default: No port is assigned, 00.

  To de-assign ISDN Terminal from a BRI Software Port, dial:
  - ***7301-1-ISDN Terminal-00*** to de-assign a single Terminal from a BRI port.
  - ***7301-2-ISDN Terminal-ISDN Terminal-00*** to de-assign a range of Terminals from a BRI port.
  - ***7301-*-00*** to de-assign all Terminals from a BRI port.

  To assign Access Code to an ISDN Terminal, dial:
  - ***3103-1- ISDN Terminal -Access Code-#****
    Where,
    ISDN Terminal is the Software Port number of the Terminal from 01 to 64.
    Access Code is a number string of any combination of a maximum of 6 digits. Terminate the command with #* if the number string has fewer than 6 digits. (0 to 9, #, * are allowed)

  To clear the Access codes assigned to the ISDN Terminal, dial:
  - ***3103-1-ISDN Terminal-#**** to clear the Access Code of a single terminal.
  - ***3103-2-ISDN Terminal-ISDN Terminal-#**** to clear the Access Code of a range of terminals.
  - ***3103-*-#**** to clear the Access Codes of all terminals.

To assign default Station Access Codes to ISDN Terminal, dial:

- ***3153-1-ISDN Terminal*** to default Access Code of a single terminal.
- ***3153-2-ISDN Terminal-ISDN Terminal*** to default Access Codes of a range of terminals.
- ***3153-**** to default Access Codes of all terminals.

To assign a Name to an ISDN Terminal, dial:

- ***5409-1-ISDN Terminal-Name-#**** to assign a Name to a single terminal.
- ***5409-2-ISDN Terminal-ISDN Terminal-Name-#**** to assign the same Name to a range of terminals.
- ***5409-*-Name-#**** to assign the same Name to all terminals.
  Where,
  ISDN Terminal is the Software Port number of the Terminal from 01 to 64.
  Name is a string of a maximum of 18 alphanumeric characters. Terminate the commands with #* if the number string has fewer than 18 characters.

To clear the Name of the ISDN Terminal, dial:

- ***5409-1-ISDN Terminal-#**** to clear the Name of a single terminal.
- ***5409-2-ISDN Terminal-ISDN Terminal-#**** to clear the Names of a range terminals.
- ***5409-*-#**** to clear the Names of all terminals.

To assign a Station Basic Feature Template to an ISDN Terminal, dial:

- ***5507-1-ISDN Terminal-Template Number*** to assign a template to a single terminal.
- ***5507-2-ISDN Terminal-ISDN Terminal-Template Number*** to assign the same template to a range of terminals.
- ***5507-*-Template Number*** to assign the same template to all terminals.
  Where,
  ISDN Terminal is the Software Port number of the Terminal from 01 to 64.
  Template Number is the number of the Station Basic Feature Template, from 01 to 50. Default: 01

To assign a Station Advanced Feature Template to an ISDN Terminal, dial:

- ***5607-1-ISDN Terminal-Template Number*** to assign a template to a single terminal.
- ***5607-2-ISDN Terminal-ISDN Terminal-Template Number*** to assign the same template to a range of terminals.
- ***5607-*-Template Number*** to assign the same template to all terminals.
  Where,
  ISDN Terminal is the Software Port number of the Terminal from 01 to 64.
  Template Number is the number of the Station Advanced Feature Template, from 01 to 50. Default: 01
  To select a Station Type for an ISDN Terminal, refer the SARVAM UCS Hospitality System Manual.

For Configuration of Advanced Features, use the following commands:

To assign a Personal Directory to an ISDN Terminal, dial:

- ***1907-1-ISDN Terminal-Personal Directory*** to assign directory to a single terminal.
- ***1907-2-ISDN Terminal-ISDN Terminal-Personal Directory*** to assign the same directory to a range of terminals.
- ***1907-*-Personal Directory*** to assign the same directory to all terminals.
  Where,
  ISDN Terminal is the Software Port number of the Terminal from 01 to 64.
  Personal Directory number is from 01 to 50.

To clear the Personal Directory assigned to an ISDN Terminal, dial:

- ***1907-1-ISDN Terminal-00*** to clear the directory of a single terminal.
- ***1907-2-ISDN Terminal-ISDN Terminal-00*** to clear the directory of a range of terminals.
- ***1907-**** to clear the directory from all terminals.

To define the Priority for an ISDN Terminal, dial:

- ***3913-1-ISDN Terminal-Priority*** to define Priority for a single terminal.
- ***3913-2-ISDN Terminal-ISDN Terminal-Priority*** to define the same Priority for a range of terminals.
- ***3913-*-Priority*** to define the same Priority for all terminals.
  Where,
  ISDN Terminal is the Software Port number of the Terminal from 01 to 64.
  Priority is from 1 to 9. Default: 5-Normal

To assign an ISDN Terminal to a Call Pick-Up Group, dial:

- ***3903-1-ISDN Terminal-Call Pick-Up Group*** to assign a single terminal to a Call Pick-Up group.
- ***3903-2-ISDN Terminal-ISDN Terminal-Call Pickup Group*** to assign a range of terminals to the same Call Pick-Up group.
- ***3903-*-Call Pickup Group*** to assign all terminals to the same Call Pick-Up group.
  Where,
  ISDN Terminal is the Software Port number of the Terminal from 01 to 64.
  Call Pickup Group is from 00 to 99. Default: 99.

To remove an ISDN Terminal from a Call Pick-Up Group, dial:

- ***3903-1-ISDN Terminal-00*** to remove a single terminal from a Call Pick-Up group.
- ***3903-2-ISDN Terminal-ISDN Terminal-00*** to remove a range of terminals from a Call Pick-Up group.
- ***3903-*-00*** to remove all terminals from a Call Pick-Up group.

- Exit SE mode.

Configuring SLT Extensions

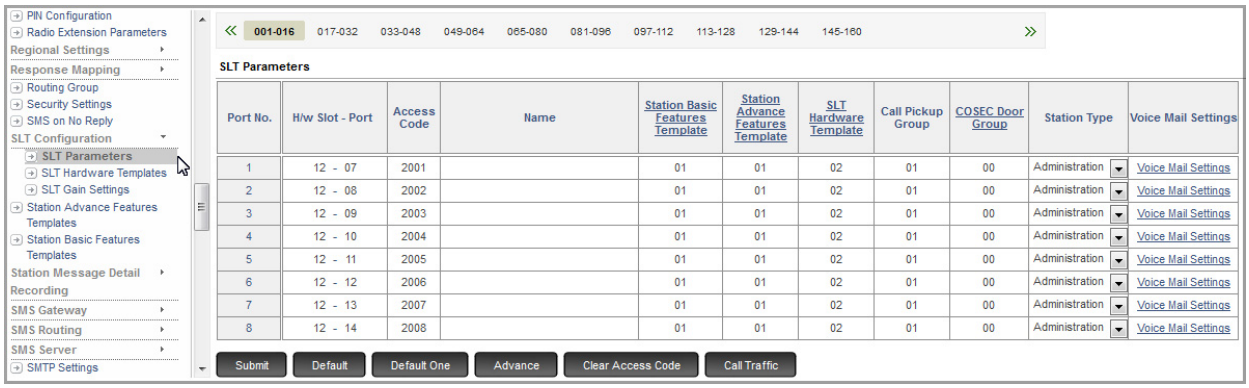
The number of SLT extensions available to you for configuration depends on the number of SLT ports supported by SARVAM UCS and the number of SLT ports you have specified on the "System Pre-requisites" page.

If you have enabled **On-Site Configuration**, the system will provide you only those ports that are actually present in the system for configuration.

Configure **SLT port parameters** using Jeeves or by dialing commands from a Telephone. If you have installed a Voice Mail module in SARVAM UCS, you may provide voice mail facility to all or selected SLT extensions. To provide voice mail to extensions, configure **VMS Settings**.

Configuring SLT Parameters using Jeeves

- Log into Jeeves as System Engineer.
- Under **Configuration**, click **SLT Configuration**.
- Click **SLT Parameters** to open the page.



- Configure the following parameters for each SLT port on this page:
  - **SLT Port No.:** This non-editable field is the number of the software port of the SLT port.
  - **Hardware Slot - Port:** 'Slot' is the number of the Universal Slot in which the SLT Card is inserted. 'Port' is the number of the SLT hardware port on which the telephone instrument is connected.

    The SARVAM UCS can automatically detect and assign the hardware slot and port numbers automatically to the SLT software ports.

    For example: if you have inserted the SLT8 Card in Slot number 02 and SLT16 Card in Slot number 03 of SARVAM UCS, the system will assign the hardware slot 02 and port numbers 01-08 to the SLT Software Ports from 001 to 008 respectively. The system will assign Slot 03 and port numbers 09-24 to the SLT Software Ports 009 to 024. Refer the topic "Software Port and Hardware ID" to know more.

    However, if required, you may change the Hardware Slot and Port assigned to the SLT software port. In this case, enter the desired Hardware Slot and Port number in this field.

    If you want to de-assign the Hardware Slot and Port, Enter '00' in both fields.

- **Access Code:** Assign Station Access Codes to the SLT Port[177]. Station Access Codes are commonly referred to as Extension Numbers. These may be a maximum of 6 digits, which are dialed to call the SLT port to which they are assigned.

  All SLT ports are assigned the following Station Access Codes as default.

SLT Software Port	Access Codes
001	2001
002	2002
003	2003
:	:
:	:
512	2512

  You may either apply the default Station Access Codes to the SLT ports or assign them according to your requirement and preferences.

  To assign Station Access Codes according to your preference and requirement to a range of SLT Ports, see “Assigning Access Codes to a Range of Extensions”.

If you decide to customize the Station Access Codes, make sure that the numbers do not clash with any other Access Code in the 'Dial' phase. Refer the topics “Access Codes” and “Conflict Dialing” to know more.

- **Name:** Assign a 'Name' to the SLT port. The name may be of the person who will use the SLT or the name of a department. This name will be displayed on the LCD of the remote user's phone, if it is equipped with Caller ID.

  You can program a name of a maximum of 18 alphanumeric characters.

- **Station Basic Feature Template:** Assign a “Station Basic Feature Template” to the SLT. A Station Basic Feature Template includes a set of features that completely define the behaviour of the extension port, such as Time Table, Operator access, Trunk Access, Class of Service, Toll Control, Call Budget, and Station Message Detail Records (storage of Incoming and Outgoing Call details).

  By default, Station Basic Feature Template 01 is assigned to all extensions of the system that includes DKP ports, ISDN Terminals, SIP extensions, and E&M Lines with Station as Orientation Type. Check if the default template fulfills the feature requirements (like “Class of Service (COS)”, “Toll Control”, “OG Trunk Bundle Group”, etc.) of the SLT extension user.

  If the default Template 01 fulfills the feature requirements and if the same features are to be allowed to all SLTs, retain Template 01.

  If different sets of features are to be allowed to different SLTs, then prepare separate Station Basic Feature Templates and apply them on the ports. To do this,

  - Under **Configuration**, click **Station Basic Feature Template** to open the page.

177. *The number of SLT ports vary according to the variant of SARVAM UCS.*

- Select a Template number, for example 10.
- Customize Template number 10 and click **Submit** at the bottom of the page.
- Now go back to the **SLT Parameters** page.
- Enter the number of the Template you customized, Template 10 in the **Station Basic Feature Template** field of the SLT Port, for instance SLT-003, on which you want to apply this template.
- Click 'Submit' at the bottom of the page to save changes.
- Repeat the same steps to customize and assign a different Template to another SLT port.

Also, refer the topic “Station Basic Feature Template” to know more about customizing the templates and applying on the ports.

- **Station Advanced Feature Template:** Assign a Station Advanced Feature Template to the SLT. The Advanced Feature Template consists of a set of less commonly used features like Alarm Notification Type, Caller ID Presentation for Transferred Calls, DDI Incoming Call Routing, Storage of Internal Calls, Call Duration Control, Floor Service, Call Taping, etc.

  By default Station Advanced Feature Template 01 is assigned to all extensions of the SARVAM UCS, which includes DKP Ports, SLT Ports, ISDN Terminals, SIP Extensions, and E&M Lines configured as Stations.

  Check if this default template fulfills the feature requirements of the SLT extension users by clicking the link **Station Advanced Feature Template**.

  If the default Template 01 fulfills the feature requirements, and if the same features are to be allowed to all SLT ports, retain Template 01.

  If different sets of features are to be allowed to different SLT Ports, then prepare separate Station Advanced Feature Templates and apply them on the ports.

  To do this,

  - Under **Configuration**, click **Station Advanced Feature Template** to open the page.
  - Select a Template number, for example 02.
  - Customize Template number 02.
  - Click **Submit** to save changes.
  - Return to **SLT Parameters** page.
  - Enter the number of the Template you customized, template 02, in the **Station Advanced Feature Template** field of the SLT Ports on which you want to apply this template.
  - Click **Submit** to save changes.
  - Repeat the same steps to customize and assign a different Template to another SLT port.

Also refer the topic “Station Advanced Feature Template” for instructions on customizing these templates and applying them on the ports.

- **SLT Hardware Template:** Assign an SLT Hardware Template to the SLT port. An SLT Hardware Template is a set of features that define the behavior of the SLT hardware port. The SLT Hardware Template allows you to configure according to user requirements a common set of features for all SLT Hardware Ports, like Caller ID Presentation (DTMF, FSK), Digit Pad Count, Ring Type, AC Impedance, Answer Signaling type, Speech Transmit and Receive Gains, Open Loop Disconnect, Loop Current, and Fax connectivity.

  There are 50 SLT Hardware Templates that can be customized and assigned to the SLT ports. By default SLT Hardware Template Number 01 is assigned to all the SLTs. This template has default values fulfilling the common requirements of a very broad user base.Check if the values in this template fulfill your requirements.

  If the default SLT Hardware Template 01 fulfills the feature requirements and if the same features are to be allowed to all SLTs, retain Template 01.

  If different sets of hardware features are to be allowed to different SLTs, then prepare separate SLT Hardware Templates and apply them on the ports. To do this,

  - Under **Configuration**, click **SLT Configuration**.
  - Click **SLT Hardware Template** to open the page.
  - Select a Template number, for example 02.
  - Customize Template number 02 and click **Submit** at the bottom of the page.
  - Now go back to the **SLT Parameters** page.
  - Enter the number of the Template you customized, template 02, in the **SLT Hardware Template** field of the SLT Ports on which you want to apply this template.
  - Click **Submit** to save changes.
  - Repeat the same steps to customize and assign a different SLT Hardware Template to another SLT port.

  Also, refer the topic “SLT Hardware Template” to know more about customizing the templates and applying on the SLT ports.

- **Call Pick-Up Group:** Configure this parameter if you want to assign the SLT to a particular “Call Pick Up” group.

  Call Pick Up allows the SLT extension user to 'pick up' (answer) calls ringing on any other extension, by dialing a feature code, without physically going to the ringing extension.

  For this to work, both extensions, the ringing extension and the extension picking up the call, must be in the same Call Pick Up Group. Refer “Call Pick Up” for instructions on how to create groups. You can create as many as 99 groups numbered from 01 to 99.

  Enter the number of the Call Pick-Up Group you created for this SLT in this field.

- **COSEC Door Group:** You must assign the extension user to a COSEC Door Group for COSEC Integration. The users in the same group must be assigned the same group. You can create as many as 50 groups numbered from 00 to 50. Users who are assigned COSEC Door Group ‘00’ are not a part of any group. See “COSEC Integration” for more information.

- **Station Type:** Make sure you select the option Assistant, if it is the Operator Extension. The system will play beeps during the conference to the participants to indicate the presence or absence of the Operator. To know more, see “Conference-3 Party”, “Conference-Multiparty” and “Conference Dial-In”.

  For the Hotel Application of the SARVAM UCS, extensions are identified as — Administrator, Assistant or Guest extensions according to the intended user of the SLT. The system will consider the Administrator and Assistant options as same. When the Station Type is selected, the system will automatically assign the Guest and Administrator (Hotel Staff) features to the SLT. To know more, refer the *SARVAM UCS Hospitality System Manual*.

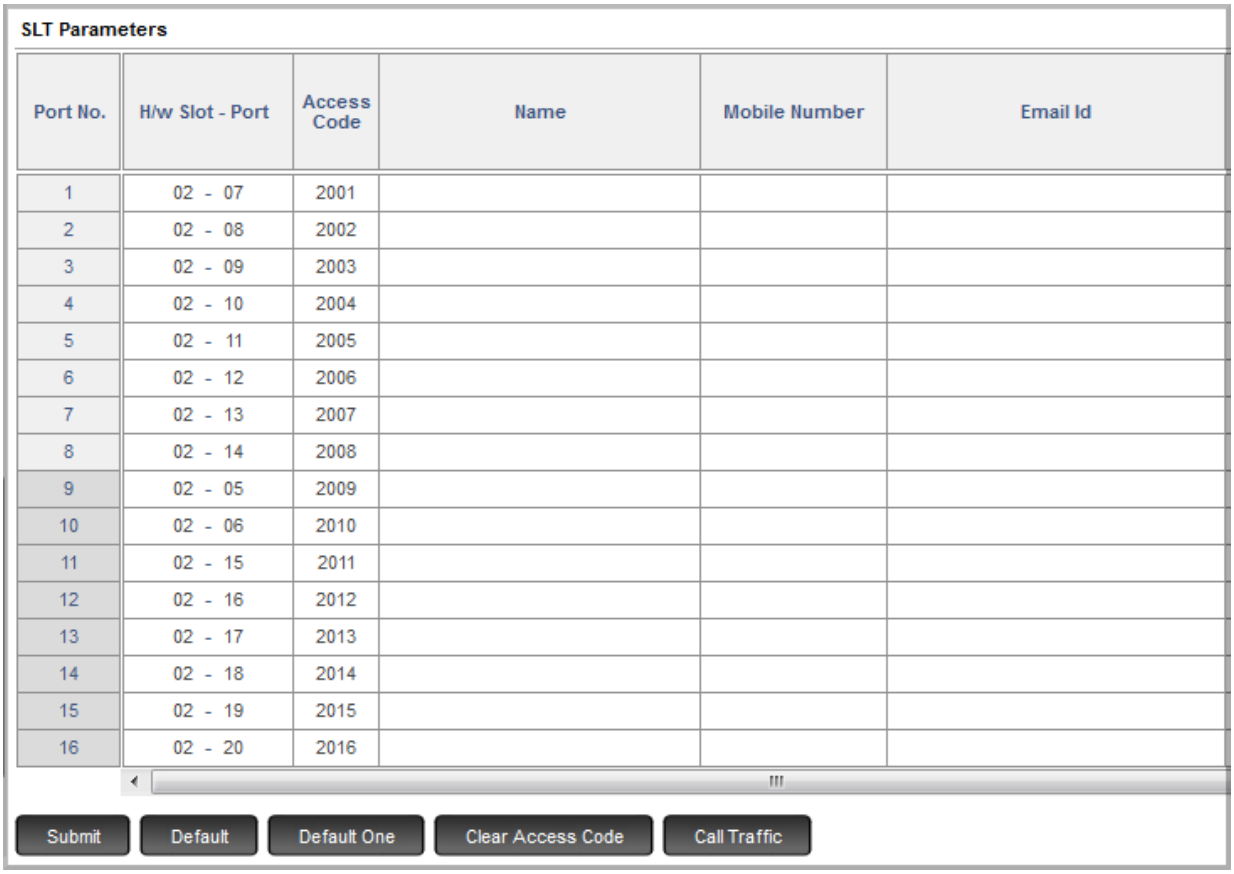
- In **Voice Mail Settings**, click the Voice Mail Settings link. The respective Extension Voice Mail Settings window will open. You may edit the parameters. For details, see “Extension Voice Mail Settings”.

The Voice Mail Settings link will be visible only if you have configured the respective access codes.

Click **Close** to close the window.

Advanced Configuration

For extension users who need to be provided features like Personal Directory or assigned a Priority, as well as to enter the user details, you may click the **Advance** button at the bottom of the page and program the following parameters:



- **Mobile Number:** Enter the Mobile Number of the extension user you wish to store. The Number can be a maximum of 16 digits.

- **Email ID:** Enter the Email ID of the extension user you wish to store. The Email ID can be a maximum of 64 characters.

- **SMS/Email Group Type:** You can assign the extension user to a Group. Select the desired **SMS/Email Group Type** from the list.The system clubs together extension users assigned the same Group. Default: None. For details, see “SMS/Email Group”.

- **Personal Directory:** Enter the number of the Personal Directory that you want to assign to this SLT. A Personal Directory is a list of 25 frequently dialed numbers, each of which are stored by Index number (location code), Name and Trunk Access Codes (“OG Trunk Bundle Group”). The Personal Directory is necessary for using the features “Abbreviated Dialing” and “Dial By Name”.

  When a Personal Directory is assigned to an SLT, it must also be configured. The Personal Directory can be configured by the SLT users and by the System Engineer. Refer the topic “Abbreviated Dialing” for instructions on configuring the Personal Directory.

- **Priority:** Select a Priority Level for the SLT.

  Each extension of the SARVAM UCS is assigned a Priority Level starting from 1, 2, 3... to 9, with '1' being lowest Priority and '9' being highest Priority. Whenever an extension phone with higher priority calls an extension with lower priority, a triple ring is placed on the called extension. To know more, read the feature description “Priority”.

By default, the Priority of all SLT ports is set to '5-Normal'. So, decide what Priority Level you will assign to each of the SLTs and set the desired level for each port.

- If you have completed configuring SLT parameters, click **Submit** at the bottom of the page to save your settings.

  It is possible to default all the parameters by clicking the **Default** button. You can also restore default values of the parameters of a single SLT by clicking the **Default One** button and specifying the SLT you want to set to default.

Configuring SLT Parameters using a Telephone[178]

- Enter SE mode from a DKP/SLT.

  To assign Hardware Slot-Port to an SLT Port, dial:
  - ***1101-SLT-Slot-Port offset on the card***
    Where,
    SLT is the Software Port number of the SLT port from 001 to 512.
    Slot is Slot number in which the SLT Card is inserted from 01 to 16
    Port Offset is the number of the SLT Port on the card from 01 to 99.

  To de-assign the hardware slot and the hardware port of an SLT port, dial:
  - ***1101-SLT-00-00***

  To assign Access Code to an SLT Port, dial:
  - ***3101-1-SLT-Access Code-#****
    Where,
    SLT is the Software Port number of the SLT port from 001 to 512.
    Access Code is a number string of any combination of 1, 2, 3 or 4 digits. Terminate the command with #* if the number string has fewer than 4 digits.

  To clear the Access codes assigned to the SLT, dial:
  - ***3101-1-SLT-#**** to clear the Access Codes of a single SLT port.
  - ***3101-2-SLT-SLT-#**** to clear the Access Codes of a range of SLT ports.
  - ***3101-*-#**** to clear the Access Codes of all SLT ports.

  To assign default Access Codes assigned to SLT, dial:
  - ***3151-1-SLT*** to default Access Codes of a single SLT port.
  - ***3151-2-SLT-SLT*** to default Access Codes of a range of SLT ports.
  - ***3151-**** to default Access Codes of all SLT ports.

  To assign a Name to an SLT, dial:
  - ***5402-1-SLT-Name-#**** to assign a Name to a single SLT port.
  - ***5402-2-SLT-SLT-Name-#**** to assign the same Name to a range of SLT ports.
  - ***5402-*-Name-#**** to assign the same Name to all SLT ports.
    Where,
    SLT is the Software Port number of the SLT port from 001 to 512.
    Name is a string of a maximum 18 alphanumeric characters. Terminate the commands with #* if the number string has fewer than 18 characters.

178. Using commands you can configure upto 999 ports in ETERNITY LENX. The remaining ports can be configured from Jeeves only.

To clear the name of the SLT, dial:

- ***5402-1-SLT-#**** to clear the Name of a single SLT port.
- ***5402-2-SLT-SLT-#**** to clear the Names of a range of SLT ports.
- ***5402-*-#**** to clear the Names of all SLT ports.

To assign an SLT Hardware Template to an SLT port, dial:

- ***5703-1-SLT-Template Number*** to assign a hardware template to a single SLT port.
- ***5703-2-SLT-SLT-Template Number*** to assign a hardware template to a range of SLT ports.
- ***5703-*-Template Number*** to assign a hardware template to all SLT ports.
  Where,
  SLT is the Software Port number of the SLT port from 001 to 512.
  Template Number is the number of the SLT Hardware Template, from 01 to 50. Default: 01.

To assign a Station Basic Feature Template to an SLT Port, dial:

- ***5503-1-SLT-Template Number*** to assign a Template to a single SLT port.
- ***5503-2-SLT-SLT-Template Number*** to assign the same Template to a range of SLT ports.
- ***5503-*-Template Number*** to assign the same Template to all SLT ports.
  Where,
  SLT is the Software Port number of the SLT port from 001 to 512.
  Template is the Station Basic Feature Template from 01 to 50. Default: 01.

To assign a Station Advanced Feature Template to an SLT Port, dial:

- ***5603-1-SLT-Template Number*** to assign a Template to a single SLT port
- ***5603-2-SLT-SLT-Template Number*** to assign the same Template to a range of SLT ports.
- ***5603-*-Template Number*** to assign the same Template to all SLT ports.
  Where,
  SLT is the Software Port number of the SLT port from 001 to 512.
  Template is the Station Advanced Feature Template from 01 to 50. Default: 01.

To assign a Call Pick-Up Group to an SLT Port, dial:

- ***3901-1-SLT-Call Pickup Group*** to assign a single SLT port to a Call Pick-Up Group.
- ***3901-2-SLT-SLT-Call Pickup Group*** to assign a range of SLT ports to the same Call Pick-Up Group.
- ***3901-*-Call Pickup Group*** to assign all SLT ports to the same Call Pick-Up Group.
  Where,
  SLT is the Software Port number of the SLT port from 001 to 512.
  Call Pickup Group is from 00 to 99. Default: 99.

To remove an SLT from a Call Pick Up group, dial:

- ***3901-1-SLT-00*** to remove a single SLT port from a group.
- ***3901-2-SLT-SLT-00*** to remove a range of SLT ports from a group.
- ***3901-**** to remove all SLT ports from a group.

To define the Station Type for an SLT Port, Refer SARVAM UCS Hospitality System Manual.

To assign a Personal Directory to an SLT Port, dial:

- ***1905-1-SLT-Personal Directory*** to assign a Personal Directory to a single SLT port.
- ***1905-2-SLT-SLT-Personal Directory*** to assign the same Personal Directory to a range of SLT ports.
- ***1905-*-Personal Directory*** to assign the same Personal Director to all SLT ports.
  Where,
  SLT is the Software Port number of the SLT port from 001 to 512.
  Personal Directory Number is from 01 to 50.

To clear the Personal Directory assigned to the SLT, dial:

- ***1905-1-SLT-00*** to clear the Personal Directory of a single SLT port.
- ***1905-2-SLT-SLT-00*** to clear the Personal Directory of a range of SLT ports.
- ***1905-**** to clear the Personal Directory of all SLT ports.

To define the Priority for an SLT Port, dial:

- ***3911-1-SLT-Priority*** to define Priority for a single SLT port.
- ***3911-2-SLT-SLT-Priority*** to define the same Priority for a range of SLT ports.
- ***3911-*-Priority*** to define the same Priority for all SLT ports.
  Where,
  SLT is the Software Port number of the SLT port from 001 to 512.
  Priority is from 1 to 9. Default: 5-Normal

- Exit SE mode.

Configuring Radio Interface

The Radio Card acts as an interface between the system and the Radio devices such as Radio Phones, Radio Repeaters, to add two-way radio functionality in SARVAM UCS. In Two-way radio, the speech can be transmitted as well as received by the radio devices. Radio devices are also called Radio Transceivers /Combat-Net Radio devices. The Two-way radio works on High Frequency (HF), Very High Frequency (VHF) or Ultra High Frequency (UHF).

A Radio Transceiver offers the following interfaces:

- Frequency tuner to set the frequency for receiving and sending audio messages
- Speaker to playback the audio received
- Microphone to send the audio message
- Push-to-Talk (PTT) button to activate radio transmission

Generally, the Radio Transceivers remain in the receiving (Rx) mode, so that the broadcasted audio messages can be heard. To transmit speech, you must press the PTT button on the Radio Transceiver.

Additional interfaces are also supported on Radio Transceivers, like Dial Pad, to dial out the DTMF digits as audio message. Such interfaces can be used where the Radio Transceivers are interfaced with the system, which allows you to dial the DTMF digits.

The Radio Interface is supported on SARVAM UCS. A maximum of 16 Radio Ports are supported.

Matrix does not supply Radio Transceivers.

How the Radio Interface works

The Radio Port of SARVAM UCS works as a Station.

Making a Call from the Radio Port

An extension user dials the Radio Port Access Code to call a Radio device.

- The Radio Port transmits the audio signal to the Radio device and the extension user gets Ring Back Tone for 3 seconds.
- Since the Radio Port is always in the receiving mode, the system connects the speech path between the Radio device user (connected to the Radio Port) and the extension user.
- The extension user (caller) can now converse with the Radio device user.
- When the Radio device user wants to talk, s/he must press the PTT button of the Radio device and talk. The extension user can now listen to the Radio device user.
- When the extension user wants talk to the Radio device user, s/he must press the '#' key. Similarly to listen to the Radio device user the extension user must press '#' key again.

*To connect/disconnect speech path using the '#' key, make sure the option **Using '#' key** is selected as PTT Activation in Radio Extension Parameter.*

- The extension user can disconnect the call by going On-hook.

If the Radio port is busy, the extension user will hear the Busy Tone.

The Radio extension user (3000) dials another Radio extension user (3001).

- Audio signal is detected on the Radio port (3001) and the Radio extension user (3000) gets Ring Back Tone for 3 seconds.
- Since the Radio Port (3001) is always in the receiving mode, the system connects the speech path between both the Radio extension users.
- Both users must press the PTT button of their Radio device to talk. At a time only one user can speak.
- If the system detects silence for the duration of the Radio Port Inactivity Timer, then the system releases the radio port and disconnects the call.

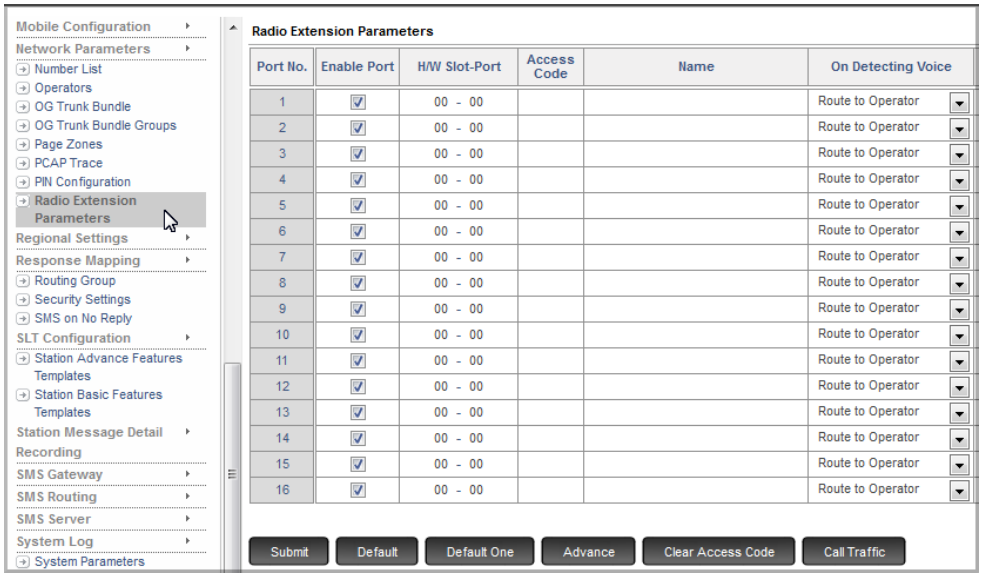
If the Radio port is busy, the extension user will hear the Busy Tone.

You must configure the Radio parameters so as to use the Radio functionality in the SARVAM UCS.

Configuring Radio Parameters using Jeeves

The following parameters are programmable using Jeeves only.

- Log into Jeeves as System Engineer.
- Under **Configuration**, click **Radio Extension Parameters** to open the page.



- Configure the following parameters.
  - **Enable Port:** This flag is for enabling or disabling a Radio port. When a Radio port is disabled, neither incoming nor outgoing calls can be made from that port.

    By default, the port is enabled. Clear the check box to disable the ports.

- **Hardware Slot and Port:** 'Slot' is the number of the Universal Slot in which the Radio Card has been inserted. 'Port' is the number of the hardware port on the card to which the Radio device is connected. By default the SARVAM UCS can detect and assign the hardware slot and port numbers automatically to the Radio (software) ports. However, you may change the Hardware Slot and Port assigned to the Radio software port. If required, enter the desired Hardware Slot and Port number in this field.

  If you want to de-assign the Hardware Slot and Port, enter '00' in both fields.

- **Access Code:** Assign Station Access Codes to the Radio Port. Station Access Codes are commonly referred to as Extension Numbers. These may be number strings of a maximum 6 digits, which are to be dialed to call the Radio port to which they are assigned.

  To assign Station Access Codes according to your preference and requirement to a range of Radio Ports, see “Assigning Access Codes to a Range of Extensions”.

  By default, the Station Access Codes are blank for all Radio ports.

- *If you decide to customize the Station Access Codes, make sure that the numbers do not clash with any other Access Code in the 'Dial' phase. Refer the topics “Access Codes” and “Conflict Dialing” to know more.*

- **Name:** Assign a 'Name' to the Radio port. The name may be of the person who will use the Radio device or the name of the department or location of the device. This name will be displayed on the LCD of the Operator/extension user's phone, if it is equipped with Caller ID.

  You can program a name of maximum 18 alphanumeric characters.

- **On Detecting Voice**: Select **Route to Operator**, to route incoming calls on the Radio Port to the Operator.

  Select **Greet to Dial**, if you want the callers to dial the desired extension number. The call will be routed to the dialed number. This option must be selected only when you have a Dial Pad connected to the Radio Device. You can customize the message played to the callers, if required. For detailed instructions, see “Voice Message Applications”.

  If you select Greet to Dial, the parameter **Time Between Consecutive PTT** will be un-editable and the parameter **PTT Count to Place a call** will be set to 1 and will be un-editable.

- **Station Basic Feature Template:** As the Radio Port functions as a station, assign a “Station Basic Feature Template” to the Radio port.

  Only the following features of the Station Basic Feature Template are applied on the Radio Port:
  - Time Table
  - Operator
  - Class of Service (Only Global Directory Part 1, 2 and 3, Basic Features and Privacy from Built-In Auto Attendant)
  - Call Budget
  - Call Privilege
  - Toll Control-Call Budget consumed
  - Outgoing Trunk Bundle Group (WH, BH and NH)
  - Store Outgoing Calls
  - Store Incoming Calls

By default, Station Basic Feature Template number 01 is also applied on Radio ports.

Check if the default settings of the features applied on the Radio ports (Time Table, Operator, Class of Service, Storage of Outgoing and Incoming Calls etc) match your requirements for the Radio ports. If yes, retain the default Station Basic Feature Template 01.

If you want to change any of the feature settings in for the Radio Ports, you may prepare a different Template[179], for example, Template 14 and apply it on the Radio Ports.

Also, if different feature settings are to be applied on different Radio Ports prepare separate Station Basic Feature Templates and apply them on the ports. To do this,

- Click the **Station Basic Feature Template** link to open the page.
- Select a Template number, for example 14.
- Customize the Radio Port related features (listed above) in Template number 14 and click **Submit** to save.
- Now go back to the **Radio Parameters** page.
- Enter the number of the Template you customized, Template 14 in the 'Station Basic Feature Template' field of the Radio Port, for example: Port 1, on which you want to apply this template. If you want to apply this template to other ports too, like 2, 3, and 4, assign the Template 14 to all these ports.
- Click **Submit** at the bottom of the page to save changes.
- If required, repeat the same steps to customize and assign a different Template to another Radio port.

Refer the topic “Station Basic Feature Template” to know more about customizing the templates and applying on the ports.

- **Station Advance Feature Template:** Assign a Station Advanced Feature Template to the Radio Port.

  Only the following features in Station Advance Feature Templates are applied on Radio Ports.
  - DDI IC Routing
  - Send DDI Number as CLI?
  - Internal Calls Storage
  - CDC Table
  - Call Tapping
  - Caller Category

  By default Station Advanced Feature Template 01 is assigned to all the Radio ports. Check if the default settings of the features applied on the Radio ports (Internal Call Storage flag and Call Forward Ring Timer) fulfill the feature requirements of the Radio ports, by opening the 'Station Advanced Feature Template' link.

  If the default template fulfills your requirement, retain the default Station Basic Feature Template 01 for the Radio ports.

179. *This is recommended because changing the values of the default Template will be applied on all other extension types to which the Template is assigned.*

However, if you want to change any of the feature settings in for the Radio Ports, you may prepare a different Template[180], for example, Template 05 and apply it on the Radio Ports. To do this,

- Click the **Station Advanced Feature Template** link to open the page.
- Select a Template number, for example 05.
- Customize Template number 05.
- Click **Submit** at the bottom of the page.
- Now go back to the **Radio Parameters** page.
- Enter the number of the Template you customized, Template 05 in the 'Station Advanced Feature Template' field of the Radio Port, for example, 1, on which you want to apply this template. If you want to apply this template to other terminals too, like 2, 3, and 4, assign the Template 05 to all these ports.
- Click **Submit** at the bottom of the page to save changes.
- Repeat the same steps to customize and assign a different Template to another Radio Port.

Also refer the topic “Station Advanced Feature Template” for instructions on customizing these templates and applying them on the station ports.

- **Priority:** Select a Priority Level for the Radio Port from 1, 2, 3... to 9, with '1' being lowest Priority and '9' being highest Priority. Whenever an extension (Radio Port) with higher priority calls an extension with lower priority, a triple ring is placed on the called extension. To know more, read the feature description “Priority”.

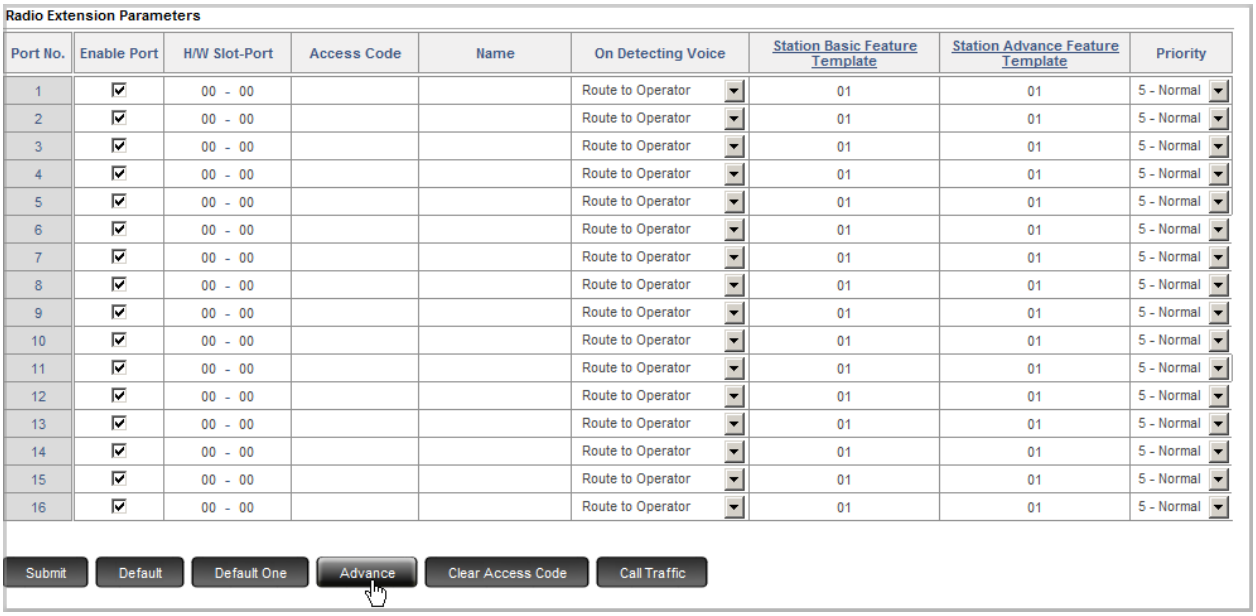
  By default, the Priority of all Radio ports is set to '5-Normal'.

Advanced Configuration

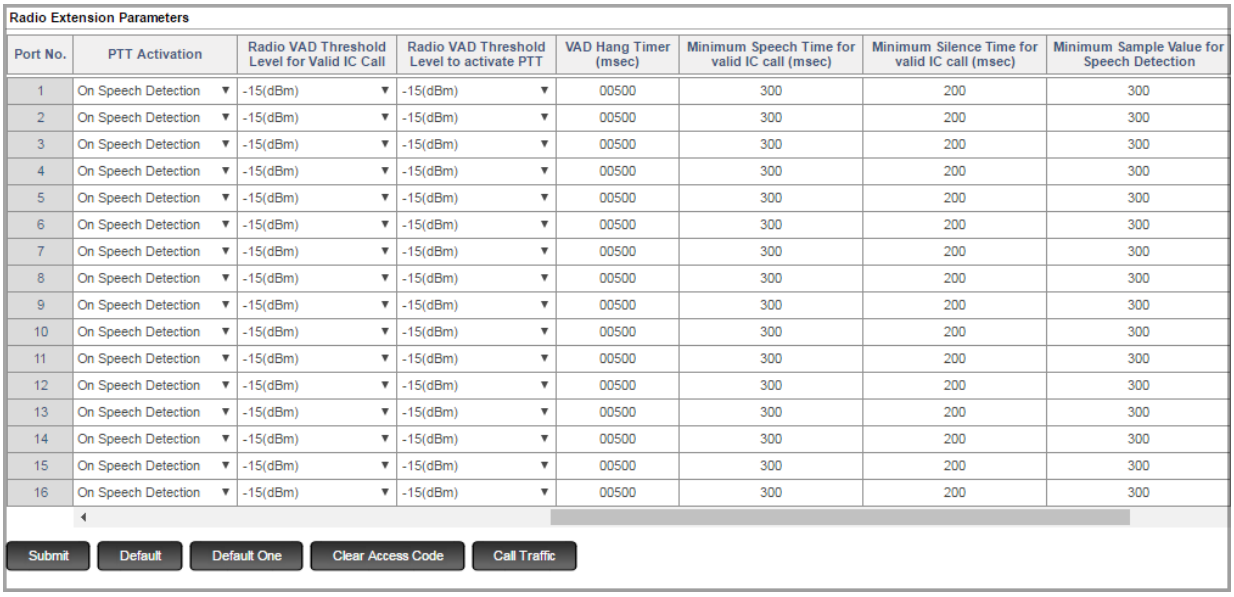
The above listed parameters fulfill the basic Radio extension port configuration requirements of most users. However, it is anticipated that some users may need to configure the more advanced features like Radio VAD

180. *This is recommended because changing the values of the default Template will be applied on all other extension types to which the Template is assigned.*

Threshold Level, Minimum Silence Time for valid IC call (msec), Minimum Speech Time to activate PTT (msec) etc. For such users, you may click the 'Advance' button.



- Scroll with the horizontal bar, and configure the following advanced Radio Parameters:



- **PTT Activation:** Select the desired option to activate PTT. You can select:
  - On Speech Detection
  - Using '#' Key

  If you select On Speech Detection, the PTT activation/deactivation will be done automatically. It will depend on the values of **Minimum Speech Time to activate PTT (msec)** and **Minimum Silence Time to deactivate PTT (msec).**

  If you select Using '#' Key, the user must press the '#' key to activate as well as deactivate PTT. It is not depend on the values of **Minimum Speech Time to activate PTT (msec)** and **Minimum Silence Time to deactivate PTT (msec).**

- **Radio VAD Threshold Level for Valid IC Call:** This parameter defines the level below which the Radio port would not validate the audio signal as valid speech packet for incoming call. Set the Radio VAD Threshold Level for Valid IC Call to the desired value. The range of this level is 0 to -96 dBm. By default, it is -15 dBm.

- **Radio VAD Threshold Level to activate PTT:** This parameter defines the level below which the Radio port would not validate the audio signal as valid speech packet to activate PTT. Set the Radio VAD Threshold Level to activate PTT to the desired value. The range of this level is 0 to -96 dBm. By default, it is -20 dBm.

- **VAD Hang Timer (msec):** This timer is applicable after the Minimum Silence Time for valid IC call (msec). The system will wait for the duration of this timer to expire after the expiry Minimum Silence Time for valid IC call (msec) timer to treat silence as valid silence before speech packets. The range is 50 - 10000. By default, it is 500 msec.

- **Minimum Speech Time for valid IC call (msec):** Set the duration for which the system must receive speech packets for a valid incoming call. The range is 100 to 9999 msec. By default, it is 500 msec.

- **Minimum Silence Time for valid IC call (msec):** Set the duration for which the system may detect and accept silence before actual speech packets, to treat this silence as a valid incoming call. The range is 100 to 9999 msec. By default, it is 1200 msec.

- **Minimum Sample Value for Speech Detection:** This parameter defines the minimum value for detecting speech on the Radio Port be considered by the system.The valid range is 100 to 900. By default, it is 300.

- **Time Between Consecutive PTT (msec):** This parameter defines the maximum Time Between Consecutive PTT. The valid range is 2000 to 6000 msec. By default, it is 4000 msec.

*This parameter is applicable only if you have selected **Route to Operator** as the **On Detecting Voice** option.*

- **PTT Count to Place call:** This parameter defines the count after which the system will place the call. The valid range is 1 to 5. By default, it is 3.

*This parameter is applicable only if you have selected **Route to Operator** as the **On Detecting Voice** option.*

- **Minimum Speech Time to activate PTT (msec):** Set this timer to ensure continuous communication, once the call matures. As per the set time, the PTT will be activated and speech will be transmitted. The range is 100 to 9999 msec. By default, it is 700 msec.

- **Minimum Silence Time to deactivate PTT (msec):** Set the duration of silence after which the system must deactivate PTT to disconnect the call. The range is 100 to 9999 msec. By default, it is 2000 msec.

- **Play Error Tone after Routing or Speech:** This flag is for enabling or disabling the tone to be played by the system. If you want the system to play error tone after routing the call or after disconnecting the speech select the check box to enable. By default, it is disabled.

- Click **Submit** to save changes.

Configuring Radio Port Inactivity Timer

When the Radio Port is in speech with another port and if no activity is detected to and from the Radio Port for the set duration, then SARVAM UCS releases the Radio Port and disconnects the call. The Radio Port Inactivity Timer can be configured using Jeeves only.

- Under **Configuration**, click **System Timers and Counts**.
- Scroll to reach **Other Features**.

- Set the **Radio Port Inactivity Timer (sec)** to the desired duration. Default: 60 seconds
- Click **Submit** to save your settings.

Configuring DTMF Detection on Radio

The DTMF Detection and Generation parameters are applicable only when there are Dial Pads connected to the Radio devices. These parameters are configurable using Jeeves only.

- Under **Configuration**, click **System Parameters**.
- Click **Radio** to expand.

- Configure the following parameters:
  - **DTMF Detection - Minimum Level (dB):** This parameter signifies the minimum level (dB) of the DTMF digit to be considered as valid. By default, Minimum levels set to -10.5dB.

- **DTMF Detection - Minimum ON Time (msec):** This parameter signifies the minimum time period for which the DTMF signal should be present in order to be detected. The valid range of this time is 17 to 204 milliseconds. By default, Minimum ON Time is set to 17 milliseconds.

- **DTMF Detection - Minimum OFF Time (msec):** This parameter signifies the minimum time period between successive DTMF digits. The valid range of this time is 17 to 204 milliseconds. By default, Minimum OFF Time is set to 17 milliseconds.

- **DTMF Generation - DTMF ON Time (msec):** It is the width of DTMF digit to be dialed out by the Radio port. By default the ON Time is set to 102 milliseconds.

- **DTMF Generation - Inter Digit Pause Time (msec):** When the Radio port dials out the DTMF digits, it waits for the Inter Digit Pause Timer, while dialing the DTMF digits. By default the timer is set to 102 milliseconds.

- Click **Submit**.

Configuring Radio Parameters using a Telephone

The following parameters are configurable using commands only.

Maximum Noise Suppression: This parameter identifies how much background noise can be suppressed.For example, if the background noise is -50dBm and you have set Maximum Noise Suppression as 12 dB, the background noise is reduced to -62 dBm.

Maximum Noise Suppression range is from 0 to 95dBm. Default value should be 25dBm.

- Enter SE mode from a DKP/SLT.

  - **4601-1-Radio-Maximum Noise Suppression** to assign the value to a single Radio port.
  - **4601-2-Radio-Radio-Maximum Noise Suppression** to assign the same value to a range of Radio ports.
  - **4601-*-Maximum Noise Suppression** to assign the same value to all the Radio ports.
    Where,
    Radio is the Software Port number of the Radio port from 01 to 16.
    Maximum Noise Suppression is from 0 to 95. Default: 25

VAD Noise Floor Threshold (in -dBm): This parameter specifies threshold above which VAD (Voice activity detection) decides that voice activity is present or below which VAD decides that only noise is present.

VAD Noise Floor Threshold range is from 0 to -96dBm. Default value should be -15dBm.

- Enter SE mode from a DKP/SLT.

  - 4602-1-Radio-**VAD Noise Floor Threshold** to assign the value to a single Radio port.
  - 4602-2-Radio-Radio-**VAD Noise Floor Threshold** to assign the same value to a range of Radio ports.
  - 4602-*-**VAD Noise Floor Threshold** to assign the same value to all the Radio ports.
    Where,
    Radio is the Software Port number of the Radio port from 01 to 16.
    VAD Noise Floor Threshold is from 00 to 96. Default:15

AGC Target Power In: This field identifies the target power level for speech output. Speech input power levels below or above this value will be raised or lowered as needed, towards this value.

AGC Target Power In range is from 0 to -20dBm. Default value should be -20dBm.

- Enter SE mode from a DKP/SLT.

  - **4603-1-Radio-AGC Target Power In** to assign the value to a single Radio port.
  - **4603-2-Radio-Radio-AGC Target Power In** to assign the same value to a range of Radio ports.
  - **4603-*-AGC Target Power In** to assign the same value to all the Radio ports.
    Where,
    Radio is the Software Port number of the Radio port from 01 to 16.
    AGC Target Power In is from 00 to 20. Default: 20

AGC Max Loss Limit In (in -dBm): This field identifies how much the speech input can be reduced by when the input is above the Target Power level.

AGC Max Loss Limit In range is from 0 to -23dBm. Default value should be -3dBm.

- Enter SE mode from a DKP/SLT.

  - **4604-1-Radio-AGC Max Loss Limit In** to assign the value to a single Radio port.
  - **4604-2-Radio-Radio-AGC Max Loss Limit In** to assign the same value to a range of Radio ports.
  - **4604-*-AGC Max Loss Limit In** to assign the same value to all the Radio ports.
    Where,
    Radio is the Software Port number of the Radio port from 01 to 16.
    AGC Max Loss Limit In is from 00 to 23. Default:03

AGC Max Gain Limit In: This field identifies how much the speech input can be increased by when the input is below the Target Power level.

AGC Max Gain Limit In range is from 0 to 23dBm. Default value should be 09dBm.

- Enter SE mode from a DKP/SLT.

  - **4605-1-Radio-AGC Max Gain Limit In** to assign the value to a single Radio port.
  - **4605-2-Radio-Radio-AGC Max Gain Limit In** to assign the same value to a range of Radio ports.
  - **4605-*-AGC Max Gain Limit In** to assign the same value to all the Radio ports.
    Where,
    Radio is the Software Port number of the Radio port from 01 to 16.
    AGC Max Loss Limit In is from 00 to 23. Default:03

To print the Radio Port parameters (only new parameters mentioned above) as debug:

- Enter SE mode from a DKP/SLT.

  - **4698-1-Radio** to print the parameters of a single Radio port.
  - **4698-2-Radio-Radio** to print the parameters of a range of Radio ports.
  - **4698-*** to print the parameters of all the Radio ports.
    Where,
    Radio is the Software Port number of the Radio port from 01 to 16.

To default the configuration of Radio Port parameters (only new parameters mentioned above):

- Enter SE mode from a DKP/SLT.

  - 4699-1-Radio to set the parameters of a single Radio port to default.
  - 4699-2-Radio-Radio to set the parameters of a range of Radio ports to default.
  - 4699-* to set the parameters of all the Radio ports to default.
    Where,
    Radio is the Software Port number of the Radio port from 01 to 16.

When you default the Radio Parameter Page (one or all) or when you Default the system, all the Radio parameters configured through Jeeves or Telephone will be set to default values.

Configuring Magneto Interface

A magneto telephone is a local battery telephone set, in which signaling current is provided by a hand generator, usually a magneto. The hand generator, commonly referred to as 'crank', is located on the right hand side of the telephone set and is turned to produce energy to ring other phones.

Tip and Ring wires of the magneto phone are dry, that is, they do not carry battery voltage when idle or in speech. This means the wires carry only AC voice signals and not the DC voltage of the battery.

The Battery is used to power the microphone only and ringing generator is used to provide ring voltage to alert the other phone.

Magneto Telephones are widely used by defense establishments as field phones in front lines, and by other establishments such as railroad companies (signaling emergencies, crossings, etc.), electric utilities, pipeline companies, who need to have their networks at places that are too remote to be serviced by public telephone networks.

SARVAM UCS supports the Magneto Interface. A maximum of 16 ports are supported in SARVAM UCS. The availability of ports depends on the number of Magneto Cards installed in the system.

Matrix does not supply Magneto Telephones.

How the Magneto Interface works

The Magneto port of SARVAM UCS works as a Station.

Making a Call to the Magneto Port

An extension user (SLT/DKP/ISDN Terminal) dials the Magneto Port Access Code to call a Magneto Field Telephone.

- Ring is played on the Magneto Field Telephone connected to the Magneto Port.
- The extension user gets Ring Back Tone for the duration of the Ring Back Tone Timer.
- The extension user must press # or the Magneto Ring Enable (MRE) Key (on the DKP) before the expiry of the Ring Back Tone Timer (programmable; default: 45 seconds) to check if the Magneto User has answered the call[181].
- The system considers pressing of the # or MRE Key as call maturity and connects the speech path between the Magneto Field Telephone (connected to the Magneto Port) and the extension phone.
- If the Magneto User has answered the call, the extension user may start speech on hearing the Magneto User's voice.
- If there is a period of silence, the extension user may press the # or MRE Key again to generate Ring on the Magneto Telephone, and press the # and MRE Key once again during the Ring Back Tone Timer to check speech with the Magneto User.

181. As there is no Answer Signaling or Call Disconnect feature on the Magneto line, there is no way for the Extension user to know whether speech has been established with the Magneto User, but to wait to hear the Magneto User's voice.

- Either party can disconnect the call:
    - The extension user either goes ON-Hook while in speech. Error Tone will be played to the Magneto user for the duration of the Error Tone Timer and the system releases the Magneto Port.
    - The Magneto User cranks the hand generator (Ring Down Signal) while in speech to disconnect the call. The system detects the Ring Current and releases the Magneto Port.

- *If the Magneto port is busy, the extension user will hear the Busy Tone.*
- *If the extension user goes ON-Hook while the Magneto Phone is ringing, the system will stop the ring on the Magneto phone.*

Making a Call from the Magneto Port

- The Magneto Field Telephone user cranks the hand generator to generate ringing current.
- The system detects the ringing current. If the ring current is present for more than 500msec, the system treats it as a Call request from the Magneto Port.
- The system rings the Operator extension assigned to the Magneto Port for the duration of the Ring Back Tone Timer and plays Ring Back Tone to the Magneto Telephone user.
- If the Operator goes OFF-Hook, before the expiry of the Ring Back Tone Timer, speech is established between the Operator and the Magneto User.
- The Operator can now transfer the call to another extension or external number.
- Either party can disconnect the call:
    - The Operator can go ON-Hook while in speech to disconnect. Error Tone will be played to the Magneto user.

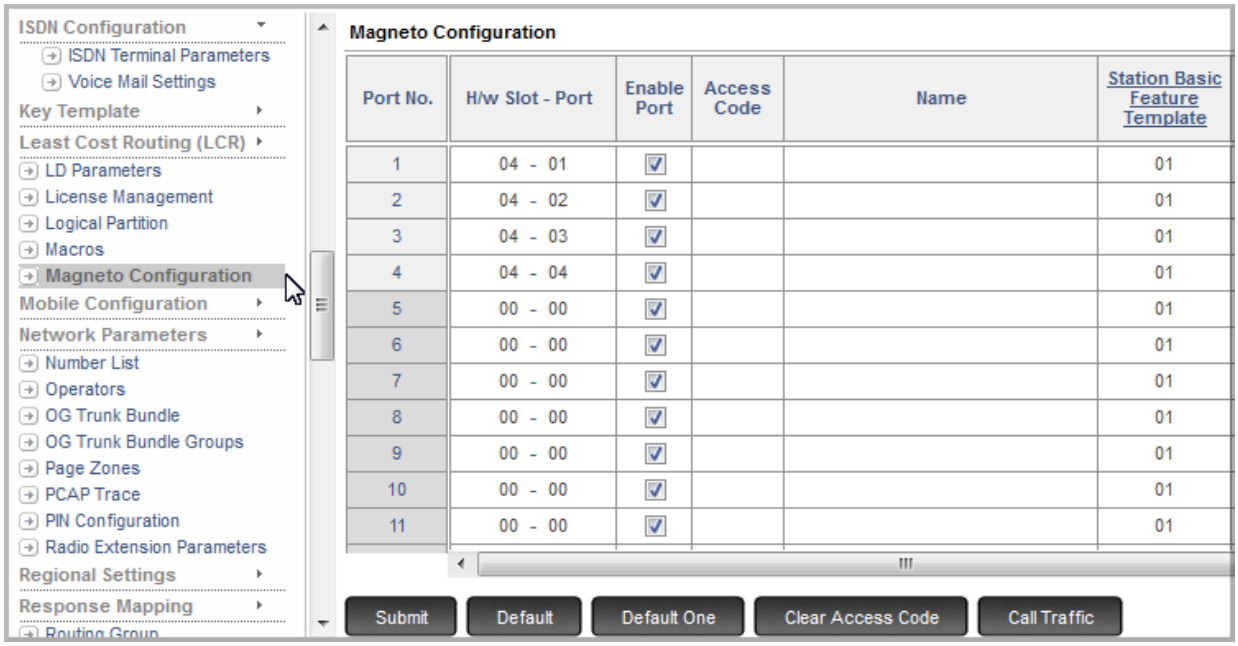
      OR

    - The Magneto User can cranks the hand generator again (Ring Down Signal) while in speech to disconnect the call.
- The system releases the Magneto Port and the port with which the Magneto User is in speech.
- If neither the Operator nor any of the two Magneto station users in speech disconnects the call, the system will automatically disconnect the call if silence (no speech) is detected for more than a specified duration of time.
- For this, the flag 'Enable Silence Detection on Magneto' must be enabled in the System Parameters and the 'Magneto Silence Disconnection Timer (default: 60 seconds) must be programmed. When Silence Disconnection is enabled on Magneto port, the system will start the Magneto Silence Disconnection Timer as soon as it detects silence. If the continuous silence is detected till the expiry of the timer the system considers the conversation as over and releases the Magneto stations. The call is disconnected. However, if speech is detected during this timer, the timer will be stopped.
- The call can be disconnected also using Forced Release by any extension user (Operator or any other extension user) having the feature 'Forced Release' in its Class of Service.

- *If the Operator is busy, Busy Tone is played to the Magneto user.*
- *The Magneto user may crank the hand generator again to send Ring Current on the expiry of the Ring Back Tone Timer. However, if the Magneto user sends the Ring Current again during the Ring Back Tone Timer, the system will treat it as a call disconnect event. It will stop ringing the Operator's extension and release the Magneto Port.*
- *The Magneto Ring Enable (MRE) Key should be programmed on the DKP extensions that are to be allowed access to call Magneto ports.*
- *On SLT extensions and ISDN Terminals, the # key will serve the function of the MRE key.*

Configuring Magneto Ports using Jeeves

- Log into Jeeves as System Engineer.
- Under **Configuration**, click **Magneto Configuration**.



Configure the following Magneto port parameters:

- **Magneto No.:** This non-editable field is the number of the software port of the Magneto port.
- **HW Slot-Port:** 'Slot' is the number of the Universal Slot in which the Magneto Card is inserted. 'Port' is the number of the Magneto hardware port on which the Magneto Field telephone instrument is connected.

  The SARVAM UCS can automatically detect and assign the hardware slot and port numbers automatically to the Magneto software ports.

  However, if required, you may change the Hardware Slot and Port assigned to the Magneto software port. In this case, enter the desired Hardware Slot and Port number in this field.

  If you want to de-assign the Hardware Slot and Port, Enter '00' in both fields.

- **Enable Port:** This flag is for enabling or disabling a Magneto port. When a Magneto port is disabled, neither incoming nor outgoing calls can be made from that port.

  By default, the port is enabled. You may disable ports that are not functioning by selecting the check box.

- **Access Code:** Assign Station Access Codes to the Magneto Port. Station Access Codes are commonly referred to as Extension Numbers. These may be number strings of a maximum 6 digits, which are to be dialed to call the Magneto port to which they are assigned.

  A maximum of 6 digits are allowed in an Access Code. By default, the Station Access Codes are blank for all Magneto ports.

  To assign Station Access Codes according to your preference and requirement to a range of Magneto Ports, see “Assigning Access Codes to a Range of Extensions”.

If you decide to customize the Station Access Codes, make sure that the numbers do not clash with any other Access Code in the 'Dial' phase. Refer the topics “Access Codes” and “Conflict Dialing” to know more.

- **Name:** Assign a 'Name' to the Magneto port. The name may be of the person who will use the Magneto telephone or the name of the department or location of the telephone. This name will be displayed on the LCD of the Operator/extension user's phone, if it is equipped with Caller ID.

  You can program a name of a maximum of 18 alphanumeric characters.

- **Station Basic Feature Template:** As the Magneto Port functions as a station, assign a “Station Basic Feature Template” to the Magneto port.

  Only the following features of the Station Basic Feature Template are applied on the Magneto Port:
  - Time Table
  - Operator
  - Class of Service
  - Store Outgoing Calls
  - Store Incoming Calls

  By default, Station Basic Feature Template number 01 is assigned to all extension types of the SARVAM UCS (SLT, DKP, ISDN Terminals, E&M Lines with Station as Orientation Type). Template 01 is also applied on Magneto ports by default.

  Check if the default settings of the features applied on the Magneto ports (Time Table, Operator, Class of Service, Storage of Outgoing and Incoming Calls) match your requirements for the Magneto ports. If yes, retain the default Station Basic Feature Template 01.

  If you want to change any of the feature settings in for the Magneto Ports, you may prepare a different Template[182], for example, Template 14 and apply it on the Magneto Ports.

  Also, if different feature settings are to be applied on different Magneto Ports prepare separate Station Basic Feature Templates and apply them on the ports. To do this,

  - Click the **Station Basic Feature Template** link to open the page.

182. *This is recommended because changing the values of the default Template will be applied on all other extension types to which the Template is assigned.*

- Select a Template number, for example 14.
- Customize the Magneto Port related features (listed above) in Template number 14 and click **Submit** to save.
- Now go back to the **Magneto Parameters** page.
- Enter the number of the Template you customized, Template 14 in the 'Station Basic Feature Template' field of the Magneto Port, for example: MAG-001, on which you want to apply this template. If you want to apply this template to other ports too, like MAG-002, 003, and 004, assign the Template 14 to all these ports.
- Click **Submit** at the bottom of the page to save changes.
- If required, repeat the same steps to customize and assign a different Template to another Magneto port.

Refer the topic “Station Basic Feature Template” to know more about customizing the templates and applying on the ports.

- **Station Advance Feature Template:** Assign a Station Advanced Feature Template to the Magneto Port.

  Only the following features in Station Advance Feature Templates are applied on Magneto Ports.
  - Internal Calls Storage
  - Call Forward Ring Timer

  By default Station Advanced Feature Template 01 is assigned to all extensions of the SARVAM UCS, which also includes DKP ports, SLT ports, ISDN Terminals and E&M Lines configured as Stations. Check if this default template fulfills the feature requirements of the Magneto Ports by opening the link 'Station Advanced Feature Template'.

  Check if the default settings of the features applied on the Magneto ports (Internal Call Storage flag and Call Forward Ring Timer) fulfill the feature requirements of the Magneto ports, by opening the 'Station Advanced Feature Template' link.

  If the default template fulfills your requirement, retain the default Station Basic Feature Template 01 for the Magneto ports.

  However, if you want to change any of the feature settings in for the Magneto Ports, you may prepare a different Template[183], for example, Template 05 and apply it on the Magneto Ports. To do this,

  - Click the **Station Advanced Feature Template** link to open the page.
  - Select a Template number, for example 05.
  - Customize Template number 05.
  - Click **Submit** at the bottom of the page.
  - Now go back to the **Magneto Parameters** page.

183. *This is recommended because changing the values of the default Template will be applied on all other extension types to which the Template is assigned.*

- Enter the number of the Template you customized, Template 05 in the 'Station Advanced Feature Template' field of the Magneto Port, for example, MAG-001, on which you want to apply this template. If you want to apply this template to other terminals too, like MAG-002, 003, and 004, assign the Template 05 to all these ports.

- Click **Submit** at the bottom of the page to save changes.

- Repeat the same steps to customize and assign a different Template to another Magneto Port.

Also refer the topic "Station Advanced Feature Template" for instructions on customizing these templates and applying them on the station ports.

- **Priority:** Assign a Priority Level from 1 to 9 to the Magneto Port, with '1' being lowest Priority and '9' being highest Priority. Whenever a Magneto Port with higher priority calls an extension with lower priority, a triple ring is placed on the called extension. To know more, read the feature description "Priority".

  By default, the Priority of all Magneto Ports is set to '5-Normal'.

- If you have completed configuration of the desired Magneto Ports, click **Submit** at the bottom of the page to save your settings.

- Now, configure the Magneto Ring Enable (MRE) Key for DKP extensions.

Configuring MRE Key for DKPs

The Magneto Ring Enable (MRE) Key is to be configured on DKP extensions that will allow calls to be made to the Magneto Ports.

Any DSS key may be configured as MRE Key. There are two ways to do this:

- Assign the MRE Key function in the DKP Key Template assigned to the DKPs, by customizing a template and assigning it to the DKPs.

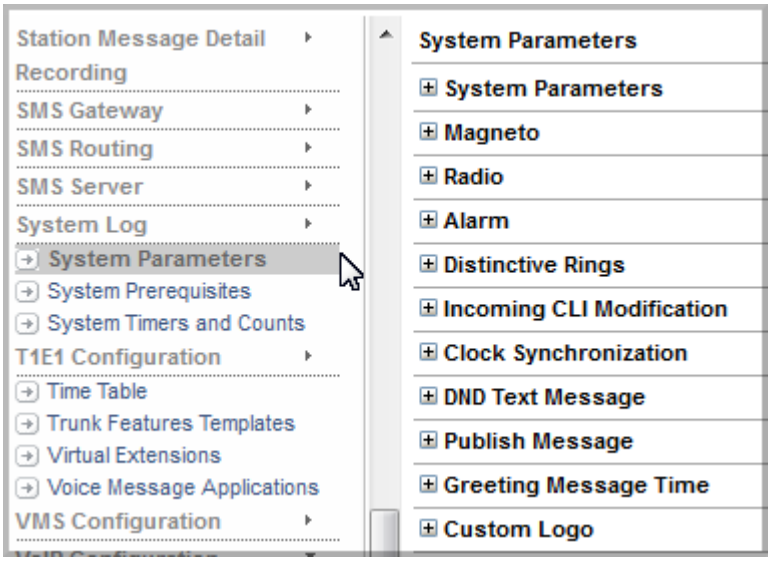
  OR
- Assign the MRE Key function individually in each DKP, by selecting 'Personalized' Key map for each DKP.

For detailed information and instructions, see "DSS Keys Programming".

Configuring Silence Detection on Magneto

If you want calls from Magneto ports to be disconnected automatically, if silence is detected, you must enable the **Silence Detection on Magneto** flag and program the **Magneto Silence Disconnection Timer**, if required.

- Under **Configuration**, click **System Parameters**.



- Click the **Magneto** sub-link to expand.
    - **Enable Silence Detection on Magneto**: By default this flag is enabled. If not, select the check box to enable this flag.
    - **Magneto Silence Disconnection Timer**: Set the timer to the desired value. The range of this timer is from 001 to 255 seconds. By default it is set to 30 seconds.
    - **Magneto VAD Threshold Level:** Set the level to the desired value. The range of this level of the 0 to -96 dBm. By default it is -25 dBm.
    - Click **Submit** at the bottom of the page to save your settings.
- Log out of Jeeves or continue with configuration tasks, as desired.

Configuring Magneto Parameters using a Telephone

- Enter SE mode from a DKP/SLT.

  To assign Slot-Port Assignment to a Magneto Port, dial:
  - ***1110-Magneto Trunk-Slot-Port Offset on the Card***
    Where,
    Magneto is the Software Port number of the Magneto port from 001 to 128.
    Slot is Slot number in which the Magneto Card is inserted from 01 to 16
    Port Offset is the number of the Magneto Port on the card from 00, 01 to 08.
    Default: Hardware Slot and Port Offset is automatically assigned.

  To enable/disable Magneto Port, dial:
  - ***6801-1-Magneto-Code*** to enable/disable a single Magneto port.
  - ***6801-2-Magneto-Magneto-Code*** to enable/disable a range of Magneto ports.
  - ***6801-*-Code*** to enable/disable all Magneto ports.
    Where,
    Magneto is the Software Port number of the Magneto port from 001 to 128.
    Code is

1 for Enable
2 for Disable
Default: Enable.

To program Access Code for Magneto Port, dial:

- ***3107-1-Magneto-Access Code-#****
  Where,
  Magneto is the Software Port number of the Magneto port from 001 to 128.
  Access Code is a number string of any combination of up to 6 characters, including the digits from to 9, and the characters #, *, A, B, C, D, F, P, and Blank space.
  Terminate the command with #* if the number string has fewer than 6 digits.
  Default: Blank.

To clear the Access Code assigned to the Magneto port, dial:

- ***3107-1-Magneto-#**** to clear Access Code assigned to a single Magneto port.
- ***3107-2-Magneto-Magneto-#**** to clear Access Code assigned to a range of Magneto ports.
- ***3107-*-#**** to clear the Access Codes assigned to all Magneto ports.
  Where,
  Magneto is from 001 to 128.

To assign default Access Codes to Magneto Port, dial:

- ***3156-1-Magneto*** to default Access Codes of a single Magneto port.
- ***3156-2-Magneto-Magneto*** to default Access Codes of a range of Magneto ports.
- ***3156-**** to default Access Codes of all Magneto ports.

To assign a Name to the Magneto Port, dial:

- ***5411-1-Magneto-Name-#**** to assign Name to a single Magneto port.
- ***5411-2-Magneto-Magneto-Name-#**** to assign the same to a range of Magneto ports.
- ***5411-*-Name-#**** to assign the same name to all Magneto ports.
  Where,
  Magneto is the Software Port number of the Magneto port from 001 to 128.
  Name is an alphanumeric string of Maximum 18 characters. Terminate the command with #* if fewer than 18 characters are to be used as name.

To clear the Name of the Magneto Port, dial:

- ***5411-1-Magneto-#**** to clear the name of a single Magneto port.
- ***5411-2-Magneto-Magneto-#**** to clear the name of a range of Magneto ports.
- ***5411-*-#**** to clear the names of all Magneto ports

To assign Priority level of a Magneto Port, dial:

- ***3919-1-Magneto-Priority*** to assign Priority Level to a single Magneto port.
- ***3919-2-Magneto-Magneto-Priority*** to assign the same Priority Level to a range of Magneto ports.
- ***3919-*-Priority*** to assign the same Priority Level to all Magneto ports.
  Where,
  Magneto is the Software Port number of the Magneto port from 001 to 128.
  Priority is from 1 to 9.
  Default: 5-Normal.

To assign a Station Basic Feature Template to Magneto Port, dial:

- ***5511-1-Magneto-Station Basic Feature Template Number*** to assign a template to a single Magneto port.
- ***5511-2-Magneto-Magneto- Station Basic Feature Template Number*** to assign the same template to a range of Magneto ports.

- ***5511-*- Station Basic Feature Template Number*** to assign the same template to all Magneto ports.
  Where,
  Magneto is the Software Port number of the Magneto port from 001 to 128.
  Station Basic Feature Template is from 01 to 50.
  Default: Station Basic Feature Template Number 01.

To assign a Station Advanced Feature Template for Magneto Port, dial:

- ***5611-1-Magneto-Station Advanced Feature Template Number*** to assign a template to a single Magneto port.
- ***5611-2-Magneto-Magneto- Station Advanced Feature Template Number*** to assign the same template to a range of Magneto ports.
- ***5611-*- Station Advanced Feature Template Number*** to assign the same template to all Magneto ports.
  Where,
  Magneto is the Software Port number of the Magneto port from 001 to 128.
  Station Advanced Feature Template is from 01 to 50.
  Default: Station Advanced Feature Template Number 01.

To enable or disable the "Enable Silence Detection on Magneto?” flag, dial:

- ***5357-Flag***
  Where,
  Flag is
  1 for Enable
  0 for Disable
  By default, the flag is enabled.

To set Magneto-Silence Detection Timer, dial:

- ***5356-Silence Detection Timer***
  Where,
  Silence detection timer value range is from 001 to 255 seconds.
  By default, the Timer is set to 30 seconds.

To set the Magneto Threshold Level, dial:

- **5358 - Magneto VAD Threshold Level**
  Where,
  The value of the Level is from 0 to -96.
  By default, the value of the Level is -25 dBm

- Exit SE mode.

Configuring 'Operator'

Users understand the term 'Operator' as a person who handles multiple simultaneous calls and functions as the link between callers and called parties.

For the system however, an 'Operator' is a Routing Group; a group of extensions to which calls made by extensions by dialing '9' are to be landed. This also includes Auto Attendant calls on trunks during which the caller dials '9'.

Depending on the size of the Enterprise and the amount of call traffic to be managed, more than one Operator may be employed. Also, it is not uncommon to have different Operator extensions according to the time of the day. For instance, during working hours calls may be handled by the Receptionists or Front Desk Personnel, whereas during non-working hours, calls may be handled by the Security Personnel.

To meet this requirement, SARVAM UCS offers configuration of up to 20 different Operators (Routing Group). However, at a time, only one Operator can be assigned to the extensions and trunks.

Each 'Operator' is assigned a Routing Group for the Time Zones - Working Hours, Break Hours and Non-Working Hours.

Each 'Operator' is assigned a Time Table, which defines the Working Hours, Break Hours and Non-working Hours for a week. The system follows this Time Table to assign a Routing Group as 'Operator' according to the current Time Zone.

Configuration of 'Operator' involves the following steps:

1. **Configuring Routing Groups as 'Operator':** A routing group may be made up of one or more than one extensions, depending on user requirement. If the user requires only one extension as 'Operator', include only one extension as member in the Routing Group for Operator. If the user requires five extensions as 'Operator', create a Routing Group of the five desired extensions to be used as 'Operator'.

   If the user requires Time-Zone based 'Operator', then prepare a different routing group for each Time Zone. If the user requires the same Operator for all Time Zones, use the same Routing Group number in all Time Zones.

2. **Configuring a Time Table for Operator:** This is applicable only when Operator extensions are different for different Time Zones.

   Define the Time Table to be followed for the Operator selection. The Time Table may be the same as the Time Tables assigned to trunks and extensions of the SARVAM UCS or may be configured to match with the timings of the persons who work as Operators. For example if Operators in the Enterprise are working in shifts, the Time Table can be configured to match their timings.

3. **Assignment of 'Operator' to Extensions:** SIP Users, UC Clients, SLT, DKP, ISDN Terminals and Virtual Extensions of the SARVAM UCS can be assigned to an 'Operator' in their Station Basic Feature Template.

   All extensions may be assigned to the same Operator, or different groups of extensions may be assigned to different Operators, so that call management is more efficient.

   Operator 1 is the default in the Station Basic Feature Templates. If you want to assign different extensions to different Operators, you must program separate Station Basic Feature Templates with a different Operator for each extension group.

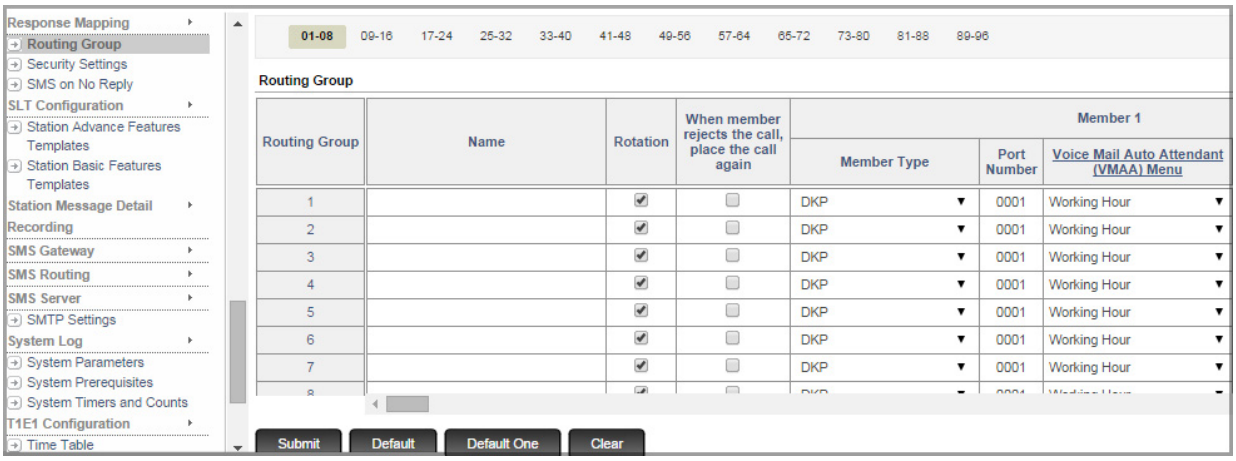
4. **Assignment of 'Operator' to Trunks:** Trunks are also assigned an 'Operator', so that when a caller dials '9' using Built-In Auto Attendant, the call is routed to the Routing Group defined as Operator for the trunk for a particular Time Zone. For example, the during working hours, a caller on trunk 001 dials '9', the call lands on 3001; when a caller on trunk 001 dials '9' during non-working hours the call lands on 3003 and when the caller dials '9' during break hours the call lands on 3002.

   Similarly, it is also possible to assign different Operators to different trunks.

Decide the number of Operators to be configured on the basis of the user's requirement.

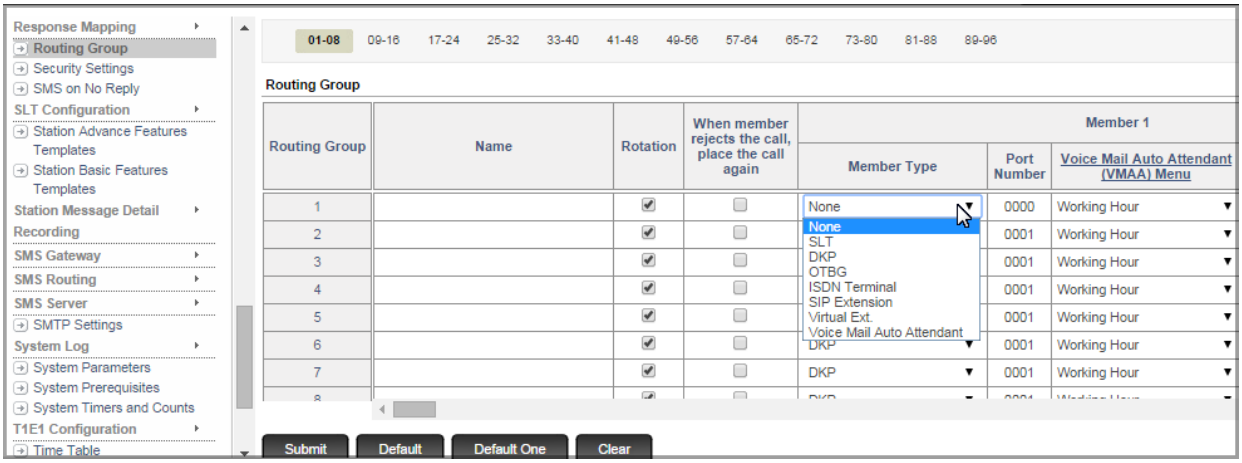
Configuring Operator using Jeeves

- Login as System Engineer.
- Under **Configuration**, click **Routing Group** to open the page.



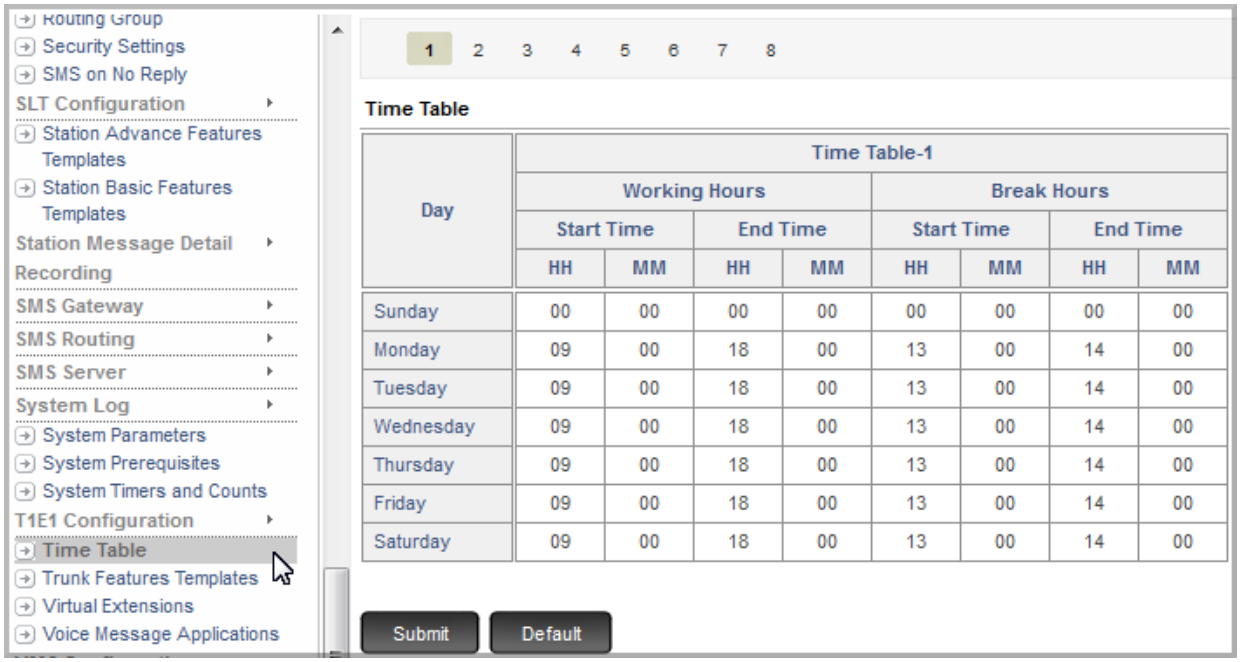
- Select a Routing Group you want to program for Operator. By default, Routing Group 32 is assigned to Operator. You may configure this group, or select another one.
- Select the type of extension to be included in the group as **Member Type**. The extension may be a DKP, an SLT, an ISDN terminal, an OTBG, a SIP Extension, a Virtual Extension or a Voice Mail Auto Attendant.
- Enter the **Port Number** (software port number) to which the SLT/DKP/ISDN Terminal/SIP extension is connected. If a Virtual Extension is selected as the Member Type, enter the Port number of the Landing Destination here. You can program as many as 32 members in the Routing Group. If the user requires only one extension as Operator, configure the first Member and disable all other 'members' of the routing group by setting **Member Type** to **None**.
- Assign the desired **Voice Mail Auto Attendant (VMAA) Menu,** if you have selected the Voice Mail Auto Attendant as the Port Type.

  You may click the *Voice Mail Auto Attendant (VMAA) Menu* link to edit the parameters of the desired VMAA Menu. For details, see “Voice Mail Auto-Attendant Menu”.



- Click **Submit** to save your changes.
- Repeat the same steps to configure another Routing Group.

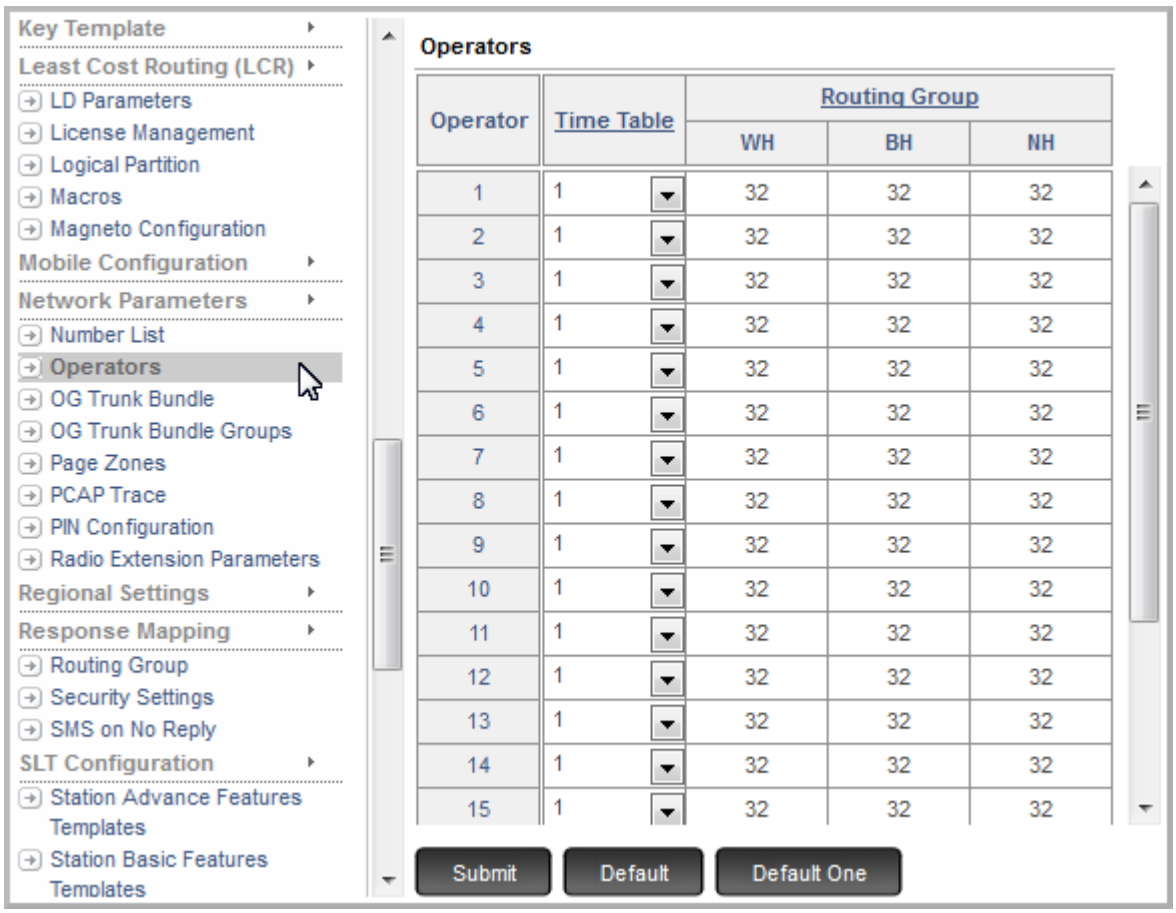
  If you have finished configuring Routing Groups for Operator, configure the Time Table for Operator.
- Under **Configuration**, click **Time Table** to open the page.



- By default Time Table 1 is assigned to all Operators. If this time table meets your requirement, retain it. If not, select another Time Table. Customize it by defining the Working Hours, Break Hours and Non-Working Hours for the week.
- Click **Submit** to save your changes.
- If you want to assign different Time Tables to different Operators, repeat the above steps to prepare the other Time Tables.

If you have completed configuring the Time Table,

- Under **Configuration**, click **Operators** to open the page.



- Select the Operator number you want to configure. By default Operator 1 is assigned to all extensions and trunks.
- Select the number of the Time Table you prepared for the selected Operator.
- Enter the number of the Routing Group you prepared for the selected Operator for Working Hours, for Break Hours and for Non-Working hours. If the same Routing Group is to be kept for all Time Zones, enter the same number in fields of all three time zones.
- Click **Submit** to save your changes.
- Repeat the above steps to configure another Operator.

Now, you may assign the 'Operator' groups you have configured to the SLT, DKP, ISDN Terminal, and SIP extensions by configuring the number of the Operator (1-20) in the “Station Basic Feature Template” applied on these extensions.

Similarly, you may assign the 'Operator' groups you have configured to the trunks in the “Trunk Feature Template” applied on these Trunks.

Configuring Operator using a Telephone

Prepare the required Routing Groups and Time Table first. Refer the topics “Routing Group” and “Time Tables” for SE commands.

- Enter SE mode from a DKP/SLT.

  To assign a Time Table to an Operator, dial:
  - ***1602-1-Operator-Time Table*** to assign a time table to a single Operator.
  - ***1602-2-Operator-Operator-Time Table*** to assign the same time table to a range of Operators.
  - ***1602-*-Time Table*** to assign the same time table to all Operators.
    Where,
    Operator is from 1 to 20.
    Time Table is from 1 to 8. Enter the number of the Time Table you customized for Operator.

  To define an Operator for Working Hours, dial:
  - ***1611-1-Operator-Routing Group***
    Where,
    Operator is from 1 to 20.
    Routing Group is from 01 to 96. Enter the number of the Routing Group you prepared for Operator for working hours. Default: Routing Group is 32.

  To define an Operator for Break Hours:
  - ***1612-1-Operator-Routing Group***
    Operator is from 1 to 20.
    Routing Group is from 01 to 96. Enter the number of the Routing Group you prepared for Operator for Break Hours. Default: Routing Group is 32.

  To define an Operator for Non-working Hours, dial:
  - ***1613-1-Operator-Routing Group***
    Operator is from 1 to 20.
    Routing Group is from 01 to 96. Enter the number of the Routing Group you prepared for Operator for Break Hours. Default: Routing Group is 32.

    For SE commands to assign an operator to extensions and trunks, refer the topics “Customizing Station Basic Feature Template using a Telephone” and “Customizing Trunk Feature Template using a Telephone”.

  To default an Operator, dial:
  - ***1601-1-Operator*** to restore default values of a single Operator.
  - ***1601-2-Operator-Operator*** to restore default values of a range of Operators.
  - ***1601-**** to restore default values of all Operators.
    Where,
    Operator is from 1 to 20.

    On issuing this command, Timetable 1 is assigned to Operator and Routing Group 32 is assigned to Operator for all Time Zones.

- Exit SE mode.

Extension Search

Using Extension Search, you can search for the extensions configured by you as well as find out the Software ports that are not assigned Extension Numbers.

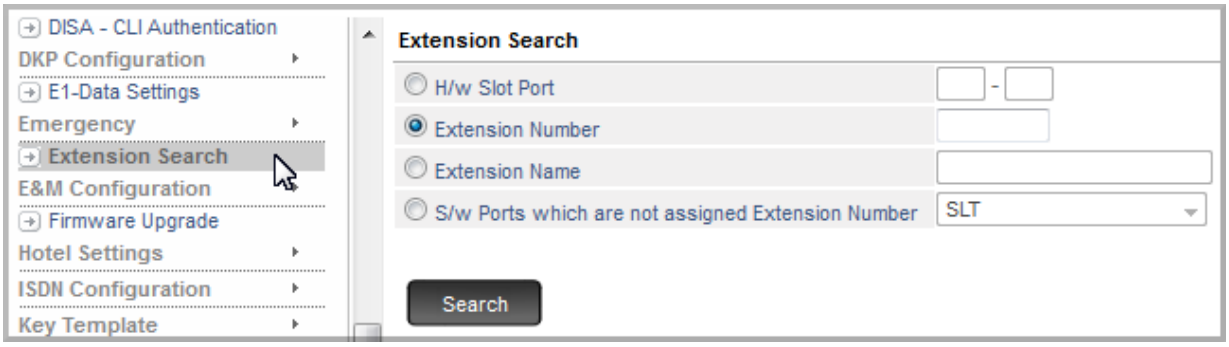
You can search for any extension by:

- H/w Slot Port
- Extension Number
- Extension Name
- Software Ports not assigned Extension Numbers

- *The number of extensions that the system will search for depends on the total extensions programmed in System Pre-requisites.*
- *If you are searching using H/W Slot-Port, the search is valid only for those extensions that are connected to the port.*
- *If you are searching using Extension Name, make sure you enter the name as configured in the system.*

Configuring Extension Search using Jeeves

- Log into Jeeves as System Engineer.
- Under **Configuration**, click **Extension Search**.



- Select the desired **Extension Search** option from the following:
  - **H/w Slot Port:** If you select this option, enter the *H/w Slot Port* of the extension you want to search. The system will perform the search for the active ports only.
  - **Extension Number:** If you select this option, enter the *Number* of the extension you want to search.
  - **Extension Name:** If you select this option, enter the *Name* of the extension you want to search.
  - **Software Ports not assigned Extension Numbers:** If you select this option, select the type of extension—SLT, DKP, Department Group, Magneto, SIP Extension, Virtual Extension, Extension Over Q-SIG, Radio Port—for which software ports are not assigned any extension numbers.
- Click **Search**. The search result appears on the screen.

Configuring Trunks

The SARVAM UCS supports the following types of trunk ports:

- SIP Trunks
- ISDN T1E1 PRI Lines
- ISDN BRI Lines
- Mobile Trunks
- CO/ Two-Wire Trunks
- E&M Lines

Templates for Configuring Trunk Lines

SARVAM UCS offers the following Hardware and Software Feature Templates to make the configuration of Trunks easy.

- SIP Hardware Template (for SIP Trunks and SIP Extensions)
- CO Hardware Template (for CO Trunk Lines).
- E&M Feature Template (for E&M Lines and T1E1PRI Lines that use E&M signaling)
- Trunk Feature Template (for All Trunk Types)

Using these templates, you can configure all Trunks that are to be assigned the same set of hardware and software features at one go, instead of configuring each trunk individually.

SIP Hardware Template

The SIP Hardware Template contains a set voice quality related features such as Voice Codec selection, Tx and Rx Gains, Echo Cancellation, Jitter Buffer and, Fax-over-IP options and related parameters

A SIP Hardware Template must be assigned to all SIP Trunks as well as SIP Extensions. Using the SIP Hardware Template, you can configure SIP Trunks and Extensions the same set of features at one go.

SARVAM UCS offers 32 SIP Hardware Templates, which can be customized as per the requirement and applied on SIP Trunks and Extensions.

SIP Hardware Template Parameters

Each of the SIP Hardware Template parameters is described below in brief.

- **Vocoder Preference:** Vocoders are the various voice codecs used to compress the data in RTP packets for optimum use of bandwidth and for ensuring voice quality. You can set 7 Vocoder options in the order of preference for a SIP trunk.

  The Vocoders supported by SARVAM UCS in the order of preference, from 1st to 7th, by default are: G.723, G.729AB, GSM FR, iLBC - 30 ms, iLBC - 20 ms, G. 711 μ-Law and G. 711 A-Law.

- *If you do not want to select any Vocoder, you can select the option 'None' in the Template. However, if all Vocoder Preferences from 1 to 7 are set to 'None', incoming and outgoing calls will be blocked.*

- *The eighth option in Vocoder, G.722 is applicable only when RTP Mode is set as RTP Relay or Direct RTP in "Configuring VoIP Parameters".*

- **G.723 Bit Rate (kbps):** You can select the Bit Rate for G.723 codec as 5.3 kbps or 6.3 kbps. When G.723 is negotiated, the selected Bit Rate will be applied only when sending the RTP packets. When receiving RTP packets from the remote end, both Bit Rates of G.723 will be accepted. The default G.723 Bit Rate is 6.3kpbs.

- **Silence Suppression:** This parameter suppresses the 'Silence' packets, allowing only the Voice packets through. SARVAM UCS supports Silence Suppression for all Vocoders except GSM FR. Default: Disabled.

Silence Suppression must be disabled if you have selected 'Pass Through' as the "Fax Type".

- **Send Silence Suppression Attribute:** By default this is flag is enabled, that is, the system will include the "silencesupp" media attribute in the SDP body.

  If you do not want the system to include the "silencesupp" media attribute in the SDP body, clear the flag.

Silence Suppression Attribute is not dependent on the Silence Suppression option set.

- **SIP Gain Settings Template:** You can increase or decrease the level of Incoming Speech (Receive Gain) and Outgoing Speech (Transmit Gain) on the SIP Trunks/Extensions by changing the Rx Gain and Tx Gain to the desired level. Different levels can be set for each port type in the SIP Gain Settings Template. By default, SIP Gain Template 1 is assigned to all the SIP Hardware Templates. If you want to assign a different Template, you must customize the SIP Gain Settings Template first and then assign the number of the SIP Gain Settings Template in this Template. To customize the SIP Gain Settings, see "Gain Settings".

- **DTMF Type:** This parameter will determine how the DTMF digits will be sent over the IP Network, when a DTMF digit is pressed. The SARVAM UCS supports three DTMF options: i) RTP (RFC 2833), ii) SIP Info, and iii) In-Band. You may select the appropriate option. By default RTP (RFC 2833) is selected.

- **RFC2833 Payload Type:** If you have selected RTP (RFC 2833) as the DTMF Type, you must configure the value of RFC2833 Payload Type. The RTP packets will be tagged as DTMF as per the set value. The value of RFC2833 Payload Type can be set from 96 to 127.

- **Call Hold Method:** You can select RFC 2543 or RFC 3264, as per your requirement.

  If you select RFC 3264, the following information will be sent in the SDP:
  - Connection Information: IP Address as used in Contact
  - Media attribute (a) : sendonly

  If you select RFC 2543, the following information will be sent in the SDP:
  - Connection Information: 0.0.0.0
  - Media attribute (a) : sendonly

  By default RFC 3264 is selected.

- **Echo Cancellation:** SARVAM UCS supports Echo Cancellation for SIP to CO trunk calls and SIP to Digital Trunks (BRI, T1E1, Mobile, SIP) and Extensions (DKP, ISDN Terminals). If you want to apply Echo Cancellation, you must enable configure the following parameters.

  - **Enable:** This flag is to be enabled to apply Echo Cancellation on SIP to CO and SIP to Digital Trunks/ Extensions. By default Echo Cancellation is enabled.

  - **Tail Length (msec) for CO:** Echo Cancellation Tail Length for SIP to CO trunks can be 32/64/128 milliseconds. By default, Echo Cancellation Tail Length for CO is set to 128 milliseconds.

- **Tail Length (msec) for Stations and Digital Trunks:** Echo Cancellation Tail Length for SIP to Digital Trunks/Extensions can be 32/64/128 milliseconds. By default, Echo Cancellation Tail Length for Digital Trunks/Extensions is set to 32 milliseconds.

- **Jitter Buffer:** The speed at which voice packets travel through a network depends on the condition of the network. All voice packets may not come at the same speed. This variation in the delay in receiving packets, known as Jitter, affects voice quality. Jitter Buffer helps overcome the delay in receiving voice packets and improves voice quality. Jitter Buffer receives voice packets, stores them and sends it to the DSP to process it at evenly spaced intervals, thus improving voice quality.

  SARVAM UCS supports two types of Jitter Buffer: Static and Dynamic. Static Jitter Buffer's internal delay is static, whereas, the Dynamic Jitter Buffer's internal delay adapts itself to the jitter in the network.

  - **Type:** Select the type of Jitter Buffer - Static or Dynamic - you want to use. If you selected Static Jitter Buffer, you may set the 'Minimum Delay'. The value configured in the Minimum Delay determines the size of the Static Jitter buffer.
    If you selected Dynamic Jitter Buffer, configure the 'Optimization Factor' and 'Minimum Delay'. The minimum size of the Dynamic Jitter buffer depends on the 'Minimum Delay' you configured. The Optimization Factor determines the rate of adaptation of the Dynamic Jitter Buffer to the jitter in the network.

  - **Minimum Delay (msec):** This parameter is to be configured for both Static and Dynamic Jitter Buffer. The Minimum Delay determines the size of the Static Jitter Buffer and When Jitter Buffer type is selected as Static, the Minimum Delay defines the size of the Static Jitter Buffer. The Static Jitter Buffer will store each received voice packets for the time you set and then it will send it to DSP for voice processing.

    When Jitter Buffer type is Dynamic, the Minimum Delay specifies the minimum time for which the Dynamic Jitter Buffer will store the received voice packet before sending it to the DSP for voice processing. 'Minimum Delay' can be from 10 to 280 milliseconds. By default Minimum Delay is set to 10 milliseconds.

- **Fax Type:** This parameter allows you to select the protocol of Fax over IP (FoIP). You can send/receive Fax from a Fax machine connected to the SLT port of the SARVAM UCS.

  The SARVAM UCS supports the fax options: None, T.38 (UDPTL), T.38 (RTP), and Pass Through.

  If you select None, the server will not detect the fax tone.

  If you select any other option both, calls and fax tone will be detected.

- *'Pass Through' and 'T.38' will work only if the peer devices also support the same option.*
- *If you select 'Pass through' as Fax type, you must disable 'Silence Suppression'.*
- *If the fax sent using T.38 is rejected, SARVAM UCS will use Pass Through for sending the Fax.*

- **T.38 Fax Parameters:** This parameter is relevant only if you have selected T.38 as the Fax Type for Fax over IP. Receiving a good quality fax over SIP trunk depends on high 'Eye Quality Monitor' (EQM). The higher the Eye Quality Monitor, the better the Fax quality. To improve the quality of T.38 fax reception, you may configure the below parameters.

  - **Version:** Configure this parameter as supported by the Remote Peer, which may be a Proxy Server or a SIP Device. While sending a fax, the Version will be sent in the re-INVITE Message to the Remote

Peer. While receiving a fax SARVAM UCS will accept a Version equal to or less than the configured Version.

A different Version can be configured for each SIP Trunk. This is useful when you have proxy SIP Trunks registered with different service providers supporting different versions.

The valid range for the Version is from 0 to 3. By default Version when Fax Type is T.38 (UDPTL) is 0 and when Fax Type is T.38 (RTP) it is 1.

- **Max Rate (Kbps):** This parameter controls the Fax image transfer speed. As EQM is inversely proportional to Fax Max Rate, if you receive poor quality fax, the Fax Max Rate should be reduced. The default Max rate is 14.4 kbps.

- **Image Redundancy Level:** The Fax Image Level is redundancy level for output Image, which can be None, 1 or 2.

  Fax Image transfer speed is inversely proportional to this parameter. If this parameter is low then fax is transferred faster. EQM is directly proportional to this parameter. If this parameter is high, good quality fax can be achieved.

  Increase the Fax Image Level, if the quality of fax does not improve with Fax Max Rate.

  Level None means no redundancy. By default Image Redundancy level is set to 2.

- **Data Redundancy Level:** This is a redundancy level for T.30 control data. Fax Data Level can be set as None, 1 or 2. Level None means no redundancy. The higher the Redundancy level, the slower would be the fax transmission.

  EQM is directly proportional to this parameter. The higher the Fax Data Redundancy Level, the better the EQM. By default, Data Redundancy Level is set to None.

- **Pass Through FAX Codec:** When the Fax option is selected as Pass Through, you must configure the Pass Through FAX Codec as supported by the Remote Peer. The Remote Peer may be a Proxy Server or a SIP Device.

  You may select the Codec as G.711 A-law or G.711 μ–Law. While sending a fax this Codec is sent in the re-INVITE message to the Remote Peer, but while receiving a fax SARVAM UCS will accept the fax with any Codec.

Customizing SIP Hardware Template

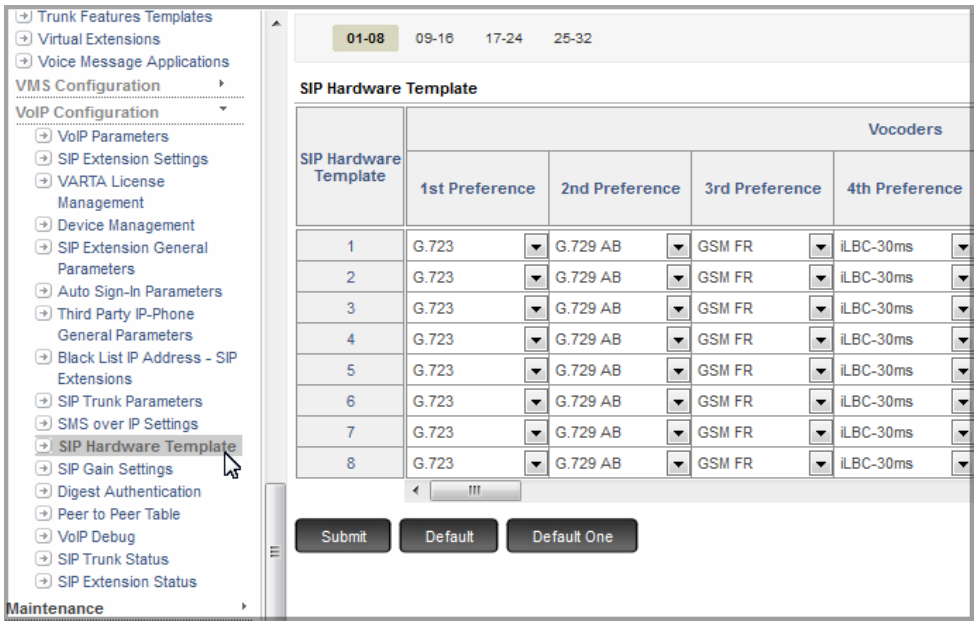
You can customize 32 SIP Hardware Templates using Jeeves or by dialing SE commands from a Telephone. By default, SIP Hardware Template 01 is assigned to all SIP Trunks as well as SIP Extensions.

If the default SIP Hardware Template 01 fulfills the user requirements, retain Template 01. If you want to change the values of certain SIP Hardware Parameters, but apply the same parameter values to all SIP Trunks and Extensions, simply customize the desired parameters in Template 01.

If different hardware parameters are to be applied to different SIP Trunks and SIP Extensions, customize different the SIP Hardware Templates using Jeeves or a Telephone and apply them to the SIP Trunks and SIP Extensions as appropriate.

Customizing SIP Hardware Template using Jeeves

- Log into Jeeves as System Engineer.
- Under **Configuration**, click **VoIP Configuration**.
- Click **SIP Hardware Template**.



- Select a Template number you wish to customize, for example Template 02.
- Change the values of the SIP Hardware Template parameters as desired.
- Click **Submit** to save your changes.
- Now, apply this Template 02 on the SIP Trunks and SIP Extensions.

To apply the customized template on SIP Trunks,

- Under **VoIP Configuration**, click **SIP Trunk Parameters**.
- Go to the SIP Trunks to which this Template is to be assigned, for example SIP Trunk 02, 03 and 04.
- Enter the number of the Template you customized, 02, in the field **SIP Hardware Template** of each of these SIP Trunks.
- Click **Submit** to save your template.

To apply the customized template on SIP Extensions,

- Under **VoIP Configuration**, click **SIP Extension Settings**.
- Go to the SIP Extension software ports to which the Template is to be assigned, for example SIP Extensions-005 to 008.

- Enter the number of the Template you customized, 02, in the field **SIP Hardware Template** of each of these SIP Extensions.

- Click **Submit** to save your settings.

- Repeat the same steps to customize another template and apply it on the SIP Trunks and Extensions.

CO Hardware Template

The CO Hardware Template contains a set of features, such as AC Termination Impedance, Pulse-Tone Dialing, Answer Supervision, Disconnect Supervision, DTMF detection, etc. that define the behavior of the hardware of the CO ports of the SARVAM UCS.

A CO Hardware template must be assigned to all the CO trunk ports. Using the CO templates, you can configure CO ports which are to be assigned the same set of features at one go, instead of configuring port-by-port.

The SARVAM UCS offers 50 CO Hardware Templates. These templates have commonly used values, but can be customized per the requirement and applied on the extensions.

SARVAM UCS supports only 'Loop Start' on CO Interface.

CO Hardware Template Parameters

The following is an overview of the parameters of these templates and a brief description of each.

- **Trunk Type:** Three types of Trunks can be interfaced to a CO port of the SARVAM UCS:

  - **Normal Dial type:** This is the conventional CO trunk available from the PSTN.

  - **Hotline type:** The CO trunk connecting two destinations immediately on grabbing the trunk.

  - **Delayed Hotline:** A special CO trunk available from the PSTN, which works as a normal dial type for some time and works as a hotline thereafter.

    By default all the CO ports are set as Normal Dial type. You may select the CO Dial Type you want to assign to the CO port.

- **Dial Type:** You can select the Dialing method as **Pulse** or **Tone** (with configurable Pulse Ratio and DTMF ON-OFF period) according to the Dialing method supported by the CO network to which the CO port is connected.

  By default, Tone is selected as the Dial Type.

- **Pulse Dial Ratio:** This parameter is to be configured if you have selected **Pulse** as the **Dial Type** in the previous parameter. The system supports the six different Pulse Dialing Ratios on CO ports. Select the appropriate Pulse Dial Ratio from the following according to the type of Pulse Dialing Ratio supported by your CO Network:
  - 10PPS, 1:2
  - 10PPS, 2:3
  - 10PPS, 1:1
  - 20PPS, 1:2
  - 20PPS, 2:3
  - 20PPS, 1:1

By default, 10PPS, 1:2 is selected as the Pulse Dial Ratio.

- **Rx CLI Type:** SARVAM UCS detects the CLI sent by the CO network and sends this information to the landing extension/operator with the ringing signal. You must select the CLI Type supported by your CO network from the following options:
  - Any ETSI DTMF format
  - Any FSK V.23 format
  - Any FSK Bellcore format
  - 1st Ring, ETSI DTMF, 2nd Ring
  - Polarity Reversal, ETSI DTMF, 1st Ring
  - 1st Ring, FSK, 2nd Ring
  - DT-AS, FSK, 1st Ring
  - RP-AS, FSK. 1st Ring
  - Polarity Reversal DT-AS, FSK, 1st Ring
  - Any DTMF Format (without Start/Stop Code)

  By default, Any ETSI DTMF format is selected as the Rx CLI Type.

- **AC Impedance:** The AC Termination Impedance of the CO port must match with the AC Termination Impedance supported by the PSTN network. The system supports the following AC Termination Impedance:
  - 600Ω
  - 900Ω
  - 270Ω + (750Ω || 150 nF)
  - 220Ω + (820Ω || 115 nF)
  - 370Ω + (620Ω || 310 nF)
  - 320Ω + (1050Ω || 230 nF)
  - 370Ω + (820Ω || 110 nF)
  - 275Ω + (780Ω || 115 nF)
  - 120Ω + (820Ω || 110 nF)
  - 350Ω + (1000Ω || 210 nF)
  - 200Ω + (680Ω || 100 nF)
  - 600Ω + 2.16 µF
  - 900Ω + 1 µF
  - 900Ω + 2.16 µF
  - 600Ω + 1 µF
  - Global complex impedance

  By default, the AC Termination Impedance is set as per the “Region” you have selected.

- **CO Termination:** This parameter allows you to increase near-end echo cancellation on the CO trunk. Near-end echo is primarily caused by the mismatch between AC Termination Impedance (presented by CO port of SARVAM UCS to the line) and CO Termination (Impedance presented by the Central Office to the line), and to some extent by the transmit and receive signal path.

  By correcting the line impedance mismatch, you can increase near-end echo cancellation. This is done by selecting the AC Termination Impedance and CO Termination, and selecting a Line Type that most closely models the line that connects the CO port of SARVAM UCS to the Central Office.

  In the **CO Termination** list, select the appropriate line impedance match. This would depend on the region where SARVAM UCS is deployed. For example, if AC Termination Impedance in your location is 600Ω and the CO Termination impedance is 900Ω in series with 2.16µF, select AC Impedance as 600Ω and CO Termination as 900Ω + 2.16µF. By default, None is selected.

Now, select the line model to be used from the CO Line Type list.

You are recommended to conduct the AC Impedance Test for the line connected to the CO Trunk port on which you will apply this template. The AC Impedance Test will help you determine the most appropriate values for the AC impedance, CO Termination and the CO Line Type. For more information see the topic "AC Impedance Test"

- **CO Line Type:** This parameter allows you to select the Line model for the CO Termination you have selected. You need to select a line type that most closely models the line connecting SARVAM UCS to the Central Office. In the **CO Line Type** list, you may select a specific EIA line model from the eight options (EIA-0 to EIA-7) or a specific wire gauge and length (2000 ft. 22/24/26awg). By default, None is selected.

- **CO Gain Settings Template:** You can increase or decrease the level of Incoming Speech (Receive Gain) and Outgoing Speech (Transmit Gain) on the Trunk by changing the Rx Gain and Tx Gain to the desired level. Different levels can be set for each port type in the CO Gain Settings Template. By default, CO Gain Template 1 is assigned to all the CO Hardware Templates. If you want to assign a different Template, you must customize the CO Gain Settings Template first and then assign the number of the CO Gain Settings Template in this Template. To customize the CO Gain Settings, see "Gain Settings".

- **Answer Supervision:** It is a signal from the CO network to indicate to call maturity. Whenever you make an outgoing call from CO trunk, the CO network will give answer signaling when the called party answers the call.

  This feature is particularly useful if you want to use "Call Cost Calculation (CCC)" to enable accurate billing. When the signal is received, the billing will start and in the absence of this signal, the call will not be billed, ensuring that unanswered and unsuccessful call attempts are not billed.

  Answer Supervision Signal has three options:

  - **Pseudo Answer:** It is used when no signaling is available from the PSTN. If this option is selected, the call will be considered as matured on the expiry of the 'Pseudo Answer Supervision Timer' (configurable; default 10 seconds), irrespective of whether or not the call actually gets matured. After this, the Call Duration Timer starts. Finally, the system starts detecting the "Disconnect Supervision" signal configured for the CO port.

Select this option only if there is no Answer Supervision Signal supported.

  - **Polarity Reversal:** It is used as maturity signal when the answer signaling is given in the form of Battery Reversal. If the battery polarity of the line is -ve for TIP and +ve for RING, when the called party has answered the call, the CO network will reverse the battery polarity, TIP becomes +ve and Ring -ve. After this, the Call Duration Timer is started. Finally system starts detecting the Disconnect Supervision signal configured for the CO port.

    By default, Answer Supervision is set as Pseudo Answer for each CO port.

Select the same Answer Supervision signal as provided by your CO Network. If the type of Answer Supervision signal selected in the system does not match with that of the CO network, the call will not be stored in the Station Message Detail Record (SMDR) buffer. For example, if the CO network does not support Answer Supervision, but you have set Polarity Reversal as Answer Supervision Type, the call will be considered as matured and will not be stored in the Station Message Detail Record (SMDR) buffer.

- **Pseudo Answer Supervision Timer:** Configure this timer if you have selected 'Pseudo Answer' as Answer Supervision Signal option.

  This is the time period for which the system will wait before treating a call as matured (regardless of whether or not it was answered). The range of this Timer is from 001 to 255 seconds. By default the Pseudo Answer Supervision Timer is set to 10 seconds.

When Pseudo Answer is selected as Answer Supervision signal, the call duration measured by the system will not accurately reflect the actual call duration because the Pseudo Answer Supervision Timer is not related to the actual call maturity. For example, if the Pseudo Answer Supervision Timer is set to 015 seconds, the call will be considered as matured after 015 seconds, even if it is answered after 20 seconds. Similarly, if this Timer is set to 080 seconds, but the call was answered after 020 seconds and disconnected after 040 seconds, this call will never be considered as matured as it ends before 080 seconds.

- **Disconnect Supervision:** It is a signal from the CO network to indicate call disconnection. Whenever a call (incoming or outgoing) made from the CO trunk is disconnected by the remote party, the CO network will send Disconnect signal to the CO port. SARVAM UCS will detect this signal and release the CO port.

  Disconnect Supervision signal is important when a PCO machine is connected to the (SLT Port) SARVAM UCS and or when SARVAM UCS is deployed in a Gateway application.

  In such application scenarios, it is desirable that calls that are disconnected by either end - calling party or called party - is terminated by the system and the port is released. If the called (remote) party has disconnected the call but the calling party (extension that made the outgoing call from SARVAM UCS) has not disconnected the call, the call remains live within the system.

  So, Disconnect Supervision signal is important, particularly when calls are routed from CO-to-CO ports, to indicate to the system that it needs to disconnect the call and release the port.
  Disconnect Supervision signal has three options:

  - **None:** When there this no signaling supported. Select this option only if there is no Disconnect Supervision signal supported.

  - **Polarity Reversal:** Call disconnection is signaled as Polarity Reversal when the call is disconnected by the remote user. For example, if the battery polarity of the CO port is '+ve' for TIP and '-ve' for RING in speech condition then on disconnection by the remote user, TIP will become '-ve' and Ring '+ve'. The user gets an Error tone and the CO port is released.

  - **Open Loop Disconnect:** Call Disconnection is signaled in the form of Open Loop, whereby the Battery voltage on the CO port is removed for a short duration. Voltage is restored after this short duration. However, the Polarity of Battery Voltage on the CO port is not changed.

    This option is to be selected when call disconnection is signaled in the form of Open Loop Disconnect pulse by the CO network. System will check Open Loop Disconnect signal for the time configured for Open Loop Disconnect Timer for each CO port. If the time of the Open Loop signal detected is less than the Open Loop Disconnect Timer configured, it will not be considered as valid Open Loop signal for releasing the CO port. But if the Open loop is detected continuously for at least for the time set as the Open Loop Disconnect timer, it is considered as a valid Disconnect Supervision signal. The call will be released and caller will get error tone.

    By default, Disconnect Supervision is set to None for each CO port.

Select the same Disconnect Supervision signal as provided by your CO Network.

- *Select the same Answer Supervision and Disconnect Supervision signal type supported by your CO network for the CO ports. Consider the following case:*
  - *The CO network supports Polarity Reversal signal as Answer and Disconnect Supervision.*
  - *But you have configured 'Pseudo Answer' as Answer Supervision signal and 'Polarity Reversal' as Disconnect Supervision signal for the CO ports in the system.*
  - *In this case, when a call is made through the CO port, the call will be considered as matured after the Pseudo Answer Supervision Timer.*
  - *Now, when the called party answers the call, the CO generates 'Polarity Reversal' as answer supervision signal on the CO port.*
  - *But as 'Polarity Reversal' is also configured as the Disconnect Supervision for the port, the system will interpret this (Answer Signaling) signal as Disconnect Supervision signal and disconnect the call.*

- **Open Loop Disconnect Timer (msec):** This parameter is applicable only if the option Open Loop Disconnect is selected as Disconnect Supervision on the CO port.

  The range of this timer is from 017 to 986 milliseconds. By default, the Timer is set to 204 msec.

- **Disconnect Tone Detection:** This parameter is to be configured if Call Disconnection is signaled by the CO network in the form of Disconnect Tone.

  When there is an incoming/outgoing call on/from the CO port is answered, the system will check whether the flag “Disconnect Tone detection is enabled. Only if the flag is enabled, the system will detect the Disconnect Tone.

  If Disconnect Tone is detected, the system will consider the call as ended and will release the CO port.

- **Disconnect Tone Cadence:** To enable the system to detect the Disconnect Tone accurately, you must set the Cadence (ON-OFF time) and Frequency of the Disconnect Tone, as supported by the CO network.

  Configure the following Disconnect Tone Cadence parameters:

  - **Frequency 1 (Hz):** Frequency 1 is from 300 to 1400 Hz. Default: 400Hz

  - **Operator:** This parameter has 3 options: No operator, Modulation (*), Addition (+). Default: No.

    If ‘No’ operator is selected frequency 2 will not be applicable.

    If Modulation is selected, frequency 1and frequency 2 will be used as modulation, F1* F2

    If Addition is selected, frequency 1 and frequency 2 will be used as addition, F1 + F2.

  - **Frequency 2 (Hz):** Frequency 2 is from 20 to 1400 Hz. Select Frequency 2 if the Disconnect Tone supported by the CO network is Dual Frequency. Default: 25Hz.
  - **ON Time 1 (ms), OFF Time 1 (ms):** Select Cadence for the first cycle ON Time1 and OFF Time 1. It may be 0 to 9999 milliseconds. Default: 750 ms ON Time 1, 750 ms OFF Time 1

  - **ON Time 2(ms), OFF Time 2 (ms):** Select Cadence for the second cycle ON Time 2 and OFF Time 2. It may be 0 to 9999 milliseconds. Default: 750 ms ON Time 2, 750 ms OFF Time 2.

- **ON Time 3(ms), OFF Time 3 (ms):** Select Cadence for the third cycle ON Time 3 and OFF Time 3. It may be 0 to 9999 milliseconds. Default: 0 ms ON Time 3, 0 ms OFF Time 3.

- **ON Time 4(ms), OFF Time 4 (ms):** Select Cadence for the fourth cycle ON Time 4 and OFF Time 4. It may be 0 to 9999 milliseconds. Default: 0 ms ON Time 4, 0 ms OFF Time 4.

  When disconnect tone detected on the port matches the Frequency and Cadences you have set, the call will be disconnected and the CO port will be released.

  When Disconnect cadence is zero, SARVAM UCS will skip that cadence and match the next cadence.

  SARVAM UCS will match the cadence for 3 cycles and then release the trunk.

- **Speech Delay Timer:** It is the time after which the system gives dial tone to the extension, when the extension user grabs the CO.

  To understand the significance of this timer, let us consider a situation. Extension 2001 does not have calling permission for long distance numbers. The user of extension 2001 grabs a CO trunk, and dials a number 1022-6305555. The system dials out this number, as it starts with '1', but since the actual dial tone from the CO comes after some time, the CO interprets this number as 022-6305555 and establishes speech. This way an extension user who does not have permission for long distance calling, can dial out a long distance number. This situation can be prevented by setting the Speech Delay Timer to an appropriate value.

  The range of this timer is from 000 to 255 seconds. By default it is set to '0' second.

- **Pause Timer:** This Timer is required for inserting delay while digits of a number string are out dialed from the CO trunk. The Pause Timer is applied when the features “Closed User Group (CUG)”, “Multi-Stage Dialing”,“Emergency Dialing”, “Last Number Redial”, “Auto Redial”, “Abbreviated Dialing”, “Call Back on Trunk Ports”, “Quick Dial”, “RCOC (Return Call to Original Caller)”, Least Cost Routing (“Configuring LCR”) are used to dial out the numbers from the CO port. The range of this timer is from 0500 to 2500 milliseconds. By default the timer is set to 1000 milliseconds.

- **Ring Cadence OFF Timer:** Configure this timer to set OFF time for Ring cadence. During the incoming call on CO port, if the CO gives ring in which the Ring OFF period is quite long, the system will consider that the ring has been stopped, and will stop ringing the SLT port, even though the incoming call is still present.

  To get accurate indication, the system supports Ring Cadence OFF timer on CO port so that ring can continue even for incoming calls with long Ring OFF period.

  The range of the Ring Cadence OFF timer is from 1 to 9 seconds. By default the timer is set to 6 seconds.

- **DTMF Out Dial:** While dialing out the DTMF digits from the CO port, the following attributes of DTMF signal are critical.

  - **DTMF Signal ON Time (ms):** It is the width of DTMF digit to be dialed out by the CO port and is configurable. By default the ON Time is set to 102 milliseconds.

  - **DTMF Inter-Digit Pause Timer (ms):** When the CO port dials out the DTMF digits on the CO, it waits for the Inter Digit Pause Timer, while dialing the DTMF digits on CO trunk. This timer is configurable. By default the timer is set to 102 milliseconds.

The 'level' of each DTMF digit is fixed, at -6.0 dB, but you may configure these parameters to match the CO network requirement.

These DTMF Out Dial attributes are applied when the features Redial, Auto Redial and Abbreviated Dialing are used to dial out the numbers from the CO port. These attributes are also applicable when you make a call from a DKP that has DTMF generation disabled.

- **DTMF Detection:** The default settings of DTMF Detection serve the requirements of most of the applications. However, you may fine tune the following parameters if you face any problems in DTMF detection.

  - **Minimum Level (dB):** This parameter signifies the minimum level (dB) of the DTMF digit to be considered as valid. By default, Minimum levels set to -4.5dB.

  - **Minimum ON Time (msec):** This parameter signifies the minimum time period for which the DTMF signal should be present in order to be detected. The valid range of this time is 17 to 204 milliseconds. By default, Minimum ON Time is set to 34 milliseconds.

  - **Minimum OFF Time (msec):** This parameter signifies the minimum time period between successive DTMF digits. The valid range of this time is 17 to 204 milliseconds. By default, Minimum OFF Time is set to 68.

- **Flash Timer (msec):** This parameter is relevant for dialing out Flash on the CO trunk to access some of the features of the PSTN. Configure the desired time of Flash to be generated on the CO trunk.
  The range of the timer is from 83 to 900 msecs. By default the Flash Timer is set to 600 msecs.

- **ON Hook Speed (msec):** This parameter allows you to set the amount of time for the line-side device to go on-hook.
  The ON-Hook speed specified is measured from the time the ON-Hook bit is cleared until loop current equals zero. Select the desired ON-Hook Speed from the following options:
  - <0.5ms
  - 3 ms
  - 26 ms

  By default, <0.5ms is selected as ON-Hook Speed.

- **OFF Hook Speed (ms):** This parameter defines the time to settle the line transients after which transmission or reception can occur. Select the desired OFF-Hook Speed from the following options:
  - 512 ms
  - 128 ms
  - 64 ms
  - 8 ms

  By default, OFF-Hook Speed is set to 8 milliseconds.

- **Current Limiting:** With this flag you can enable Loop Current Limiting mode. When this flag is enabled, the Loop Current will be limited to a maximum of 60mA.
  By default, the flag is disabled.

- **Minimum Loop Current (mA):** This parameter sets the minimum loop current at which DAA module of the CO port can operate. Select the minimum operational loop current from the following options as per your requirement:
  - 10
  - 12

- 14
- 16

The minimum Operational Loop Current set by default is set to '10 mA'.

- **Tip Ring Voltage (Volts):** This parameter allows you TIP/Ring Voltage Adjustment on the line side.

  Countries where Low voltage is required should use lower TIP/RING voltage. Adjust the values of the Tip Ring Voltage to match your country requirements from the following options:
  - 3.1
  - 3.2
  - 3.35
  - 3.5

  The default Tip/Ring voltage is 3.5.

- **Ringer Impedance:** Set the Ringer Impedance - High or Synthesized - for the CO port according your country-specific requirement.

  'High' signifies 20Mohm Ringer Impedance. This is the default Ringer Impedance provided on the line side by the DAA module of the CO port. The DAA Module can provide higher impedance when 'Synthesized' impedance is selected.

  Some countries like Poland, South Africa and Slovenia require higher ring impedance which is achieved by the DAA module, when Ringer Impedance is set to 'Synthesized' impedance.
  By default 'High' (20Mohm) is selected.

- **Ringer Threshold (Vrms):** This parameter defines the level below which the CO port would not validate the Ring signal and the level above which it validate the Ring signal. Set Ringer Threshold to the desired value from the following options:
  - 13.5 - 16.5
  - 19.35 - 23.65
  - 40.5 - 49.5

  By default 13.5 - 16.5 Vrms is set as Ringer Threshold.

- **PPDC:** 'Pre-PSTN Digit Count' or PPDC is parameter is to be configured, only if the CO Trunk ports on which the template is applied are in a “Behind the System Application”.

  PPDC is the number of digits (dialed by an extension) to be ignored by the system before toll control check is begun. It is the same as the number.

  In Behind the System Applications, another System may be connected to the SARVAM UCS, with some of its CO Trunks terminating into the Stations of the other System and other trunks directly connected to the PSTN.

  PPDC for CO Trunk ports directly connected to the PSTN must be set to '0'.

  For Trunk ports connected to stations of another System, PPDC must be configured as per the number of digits in the Trunk Access Code defined for that System.

  If the TAC is a single digit, select '1'. If TAC is double or triple digit number, accordingly select '2' or '3' as the PPDC.

To know more about this feature, refer "Behind the System Application".

- **Gateway Application - Answer Signaling:** This parameter is to be configured if the CO Port is being used in a gateway application as a source port (from where calls originate).

  The calls originating on the source port (CO port) are routed using another Trunk port, the terminating port, which may be any trunk port, for example: T1E1. When call made from the terminating port gets matured, this is signaled to the source port in the form of DTMF digits.

  .

  - **Use:** Enable this flag if you want the CO port to be used in a Gateway Application.

  - **DTMF String (max. 4 digits):** Configure the DTMF digits to be sent to signal call maturity to the source port.

- **Category (Logical Partitioning):** This parameter assigns the CO Port to a trunk category for the purpose of Logical Partitioning. By default all CO Ports are assigned to Category 1[184].

  If you have re-defined Category 1 or have assigned CO ports to a different category, say Category 3, enter the same number here.

  You may configure the call permission between the Category assigned to CO Ports and other Categories (assigned to other Trunk ports). Refer the feature description "Logical Partition" to know more.

- **Rx Gain at SIP Trunk (Pass Through Fax):** This parameter allows you to improve the quality of Fax over IP[185]. Configure this parameter if you have selected Pass Through Fax as the Type of Fax over IP on SIP Trunks, and if Pass Through Fax is to be received on a CO Trunk.

  - **Data Gain (dB)**: select the dB Level for Data Gain. By default, Data Gain is set to -11 dB.

  - **Bypass Gain (dB):** select the dB Level for Bypass Gain. By default, Bypass gain is set to -9 dB.

- **Idle Wait Timer:** This is the time taken by the Central Office to detect and release the line after the CO trunk has been released by SARVAM UCS. This time may vary from Central Office to Central Office. Set the Idle Wait Timer as per the time taken by your Central Office.

  Set this timer accurately. If the set time is more than the actual time taken by the CO to release the line, it will result in delay to the caller. If the set time is less than the actual time taken by the Central Office, no dial tone will be played to the caller.

  Valid range 001 to 255. By default, it is set as 002 seconds.

Customizing the CO Hardware Templates

The CO ports provide the interface to connect SARVAM UCS to the POTS network. The standards and features supported by POTS networks across the world vary. For example, some networks support Caller ID Presentation using DTMF signaling, while some support Caller ID Presentation using FSK signaling; some networks offer 600 Ohms Impedance, while others offer complex impedance.

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184. Trunk ports interfaced with PSTN /PLMN (Public Land Mobile Network) are assigned this category.

185. Normally, fax calls require less gain compared to voice calls. However, to improve fax reception, SARVAM UCS allows the configuring of gain settings for fax. Fax gain settings consist of Data Gain and Bypass Gain. SARVAM UCS supports Fax Receive Gain for SIP to CO Trunks, SIP to Digital Trunk calls as well as SIP to SLT Calls.

SARVAM UCS's versatile architecture allows it to be connected to such networks differing in their characteristics. You can configure the CO hardware features to match the standards supported by the POTS network of the country where the system is installed.

The 50 CO Hardware Templates offered by SARVAM UCS contain the default values of the above-listed parameters. The default parameter values of these are country specific and are loaded in each template according to the Country selected as the"Region".

For example, when India is selected as the Region, the default value of the AC Impedance is 600Ω, whereas it is 900 ohms when Philippines is selected as Region, and 320Ω + (1050Ω || 230 nF) for Region UK. Similarly, the default Rx CLI Type for Region India is 'Any ETSI DTMF format' while the same is 'Any FSK V.23 format' for Region UK.

By default, CO Hardware Template number 01 is assigned to all CO Trunk ports.

While the default CO Hardware Template number 01 has all with commonly used values to match your country-specific requirements, you can still customize each of the 50 Templates to match your preference or requirement.

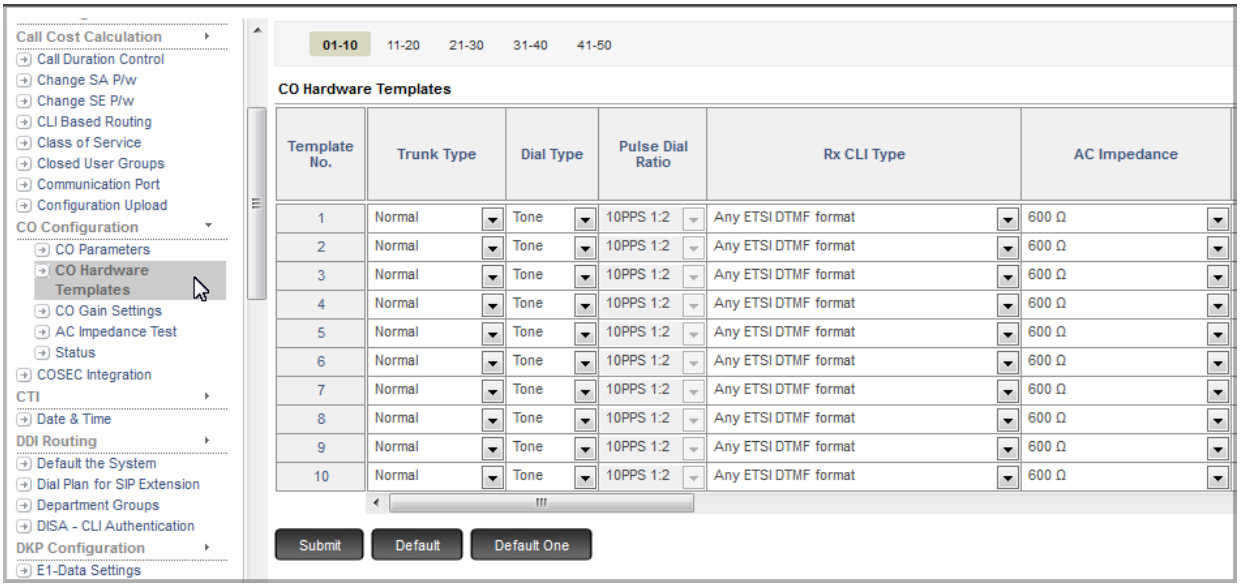
If the CO Hardware Template number 01 fulfills your requirements, and if the same features at to be applied on all CO trunk ports, retain Template 01. Similarly, if you want only a few changes to be made to Template 01 and apply it on all CO Ports, make the changes and retain the template.

However, if different sets of features are to be allowed to different CO hardware ports, then prepare separate CO Hardware Templates and apply them on the ports as required.

You can use Jeeves or a Telephone to customize the CO Hardware Template.

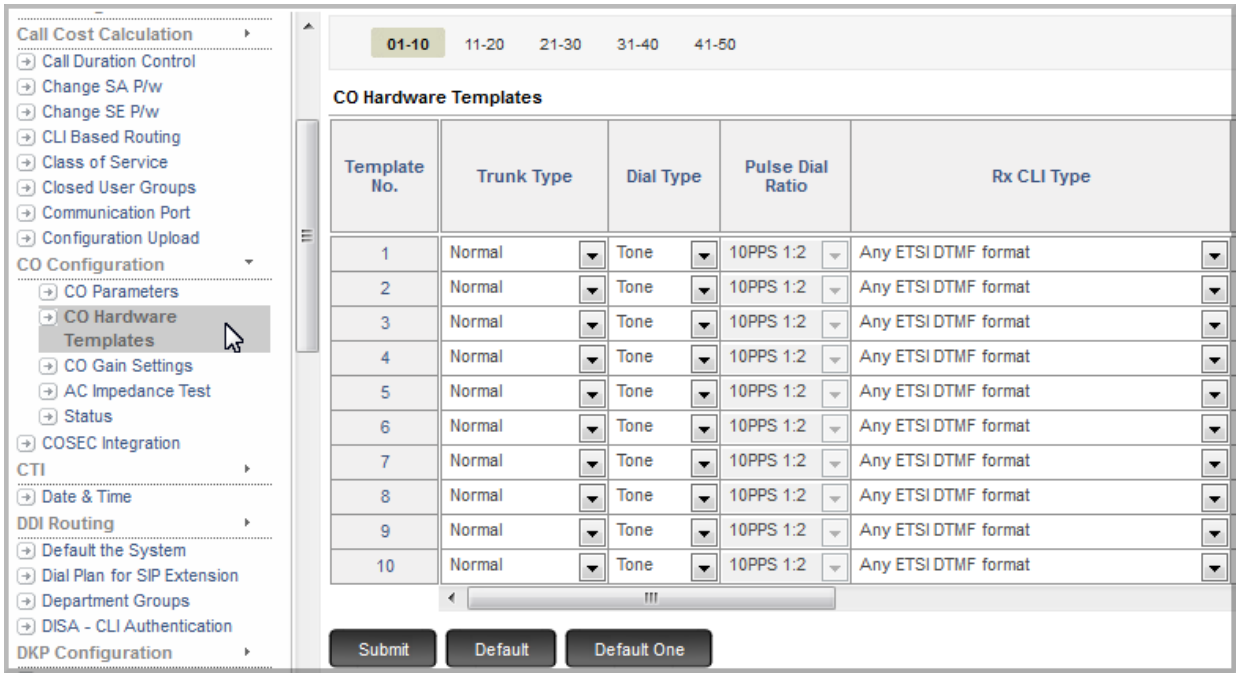
Customizing CO Hardware Templates using Jeeves

- Log in as System Engineer.
- Under **Configuration**, click **CO Configuration**.
- Click **CO Hardware Template** to open the page.



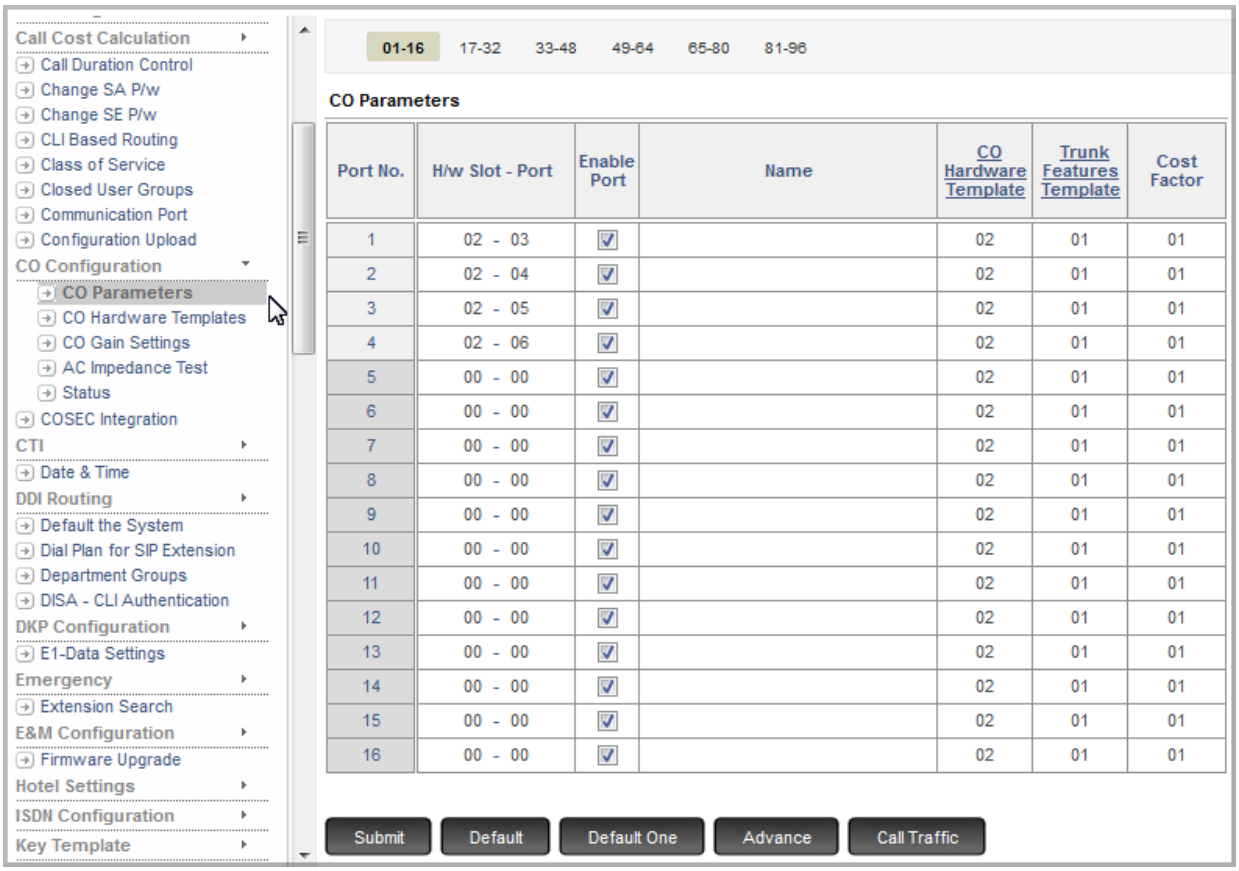
- Select a CO Hardware Template Number you wish to customize. For example, Template 07.

- Customize the CO Hardware Template number 07 by setting the parameters to the desired values.



- Click **Submit** to save changes.
- Now, apply this CO Hardware Template 07 on the CO ports.
- To do this, click the link **CO Parameters** to open the page.
- Go to the CO software ports to which this Template you customized (07) is to be assigned, for example CO-007 and 008.

- Enter the number of the Template you customized, 07, in the field **CO Hardware Template** of each of these CO ports.



- Click **Submit** to save your changes.

- Repeat the same steps to customize another template and apply it to the CO Port.

Customizing CO Hardware Templates using a Telephone

- Enter SE mode from a DKP/SLT.

  To change the default value of a CO Hardware Parameter in a Template, dial:
  - ***5902-1-CO Hardware Template Number-Feature Number-Code*** to change values of a parameter in a single template
  - ***5902-2- CO Hardware Template Number-CO Hardware Template Number-Feature Number-Code*** to set the same value for the parameter in a range of templates.
  - ***5902-*-Feature Number-Code*** to set the same value for the parameter in all templates.
    Template Number is the number of the CO Hardware Template from 01 to 50.
    Feature Number is the number of the CO Hardware Template Parameter from 01 to 50.
    Code is the value for each parameter from 0 to 58. Refer the following table.

Default Values of CO Hardware Templates (Region Code: India)

Parameter No.	01	03	04	05	02
Template No.	Trunk Type	Dial Type	Pulse Dial Ratio	Rx CLI Type	AC Termination Impedance
01	Normal	Tone	10PPS, 1:2	Any ETSI DTMF format	600 Ω
02 - 50	Same as Template 01				
Parameter Values					
0				None	
1	Normal	Pulse	10PPS, 1:2	Any ETSI DTMF format	600 Ω
2	Hotline	Tone	10PPS, 1:1	Any FSK V.23 format	900 Ω
3	-		20PPS, 1:2	Any FSK Bellcore format	270 Ω + (750 Ω || 150 nF) and 275 Ω + (780 Ω || 150 nF)
4			20PPS, 1:1	1st Ring, ETSI DTMF, 2nd Ring	220 Ω + (820 Ω || 120 nF) and 220 Ω + (820 Ω || 115 nF)
5				Polarity Reversal, ETSI DTMF, 1st Ring	370 Ω + (620 Ω || 310 nF)
6				1st Ring, FSK, 2nd Ring	320 Ω + (1050 Ω || 230 nF)
7				DT-AS, FSK, 1st Ring	370 Ω + (820 Ω || 110 nF)
8				RP-AS, FSK, 1st Ring	275 Ω + (780 Ω || 115 nF)
9				Polarity Reversal, DT-AS, FSK, 1st Ring	120 Ω + (820 Ω || 110 nF)
10				Any DTMF Format (without Start/Stop Code)	350 Ω + (1000 Ω || 210 nF)
11					200 Ω + (680 Ω || 100 nF)
12					600 Ω + 2.16 µF
13					900 Ω + 1 µF
14					900 Ω + 2.16 µF
15					600 Ω + 1 µF
16					Global complex impedance

Parameter No.	48	49		08	09
Template No.	CO Termination	CO Line Type	CO Gain Settings Template	Answer Supervision	Pseudo Answer Supervision Timer (sec)
01	None	None	1	Pseudo Answer	20
02 - 50	Same as Template 01				
Parameter Values					
0	None	None		Pseudo Answer	001 - 255
1	900 Ω + 2.16 µF	EIA-0	1	Polarity Reversal	
2	600 Ω	EIA-1	2		
3	1200 Ω + 376 Ω + 112 nF	EIA-2	3		
4	150 Ω + 510 Ω + 47 nF	EIA-3	4		
5	220 Ω + 820 Ω + 150 nF	EIA-4			
6	600 Ω + 1.5 µF	EIA-5			
7	220 Ω + 120 Ω + 115 nF	EIA-6			
8	220 Ω + 820 Ω + 115 nF	EIA-7			
9	370 Ω + 620 Ω + 310 nF	2000 ft. 22 awg			
10	220 Ω + 820 Ω + 120 nF	2000 ft. 24 awg			
11	300 Ω + 1000 Ω + 220 nF	2000 ft. 26 awg			
12	270 Ω + 750 Ω + 150 nF				
13	200 Ω + 560 Ω + 100 nF				
14					
15					
16					

Parameter No.	10	11	12	13	14	15
Template No.	Disconnect Supervision	Open Loop Disconnect Timer (msec)	Disconnect Tone Detection	Disconnect Tone Cadence		
				Frequency 1 (Hz)	Operator	Frequency 2 (Hz)
01	None	204	X	400	No	25
02 - 50	Same as Template 01					
Parameter Values						
0	None		X	300 - 1400	No	20-1400
1	Polarity Reversal	17	Ö		*	
2	Open Loop Disconnect	34			+	
3		51				
4		68				
5		85				
6		102				
7		119				
8		136				
9		153				
10		170				
11		187				
12		204				
13		221				
14		238				
15		255				
16		272				

Parameter No.	16	17	18	19	20	21	22	23
	Disconnect Tone Cadence							
Template No.	ON Time 1 (msec)	OFF Time 1 (msec)	ON Time 2 (msec)	OFF Time 2 (msec)	ON Time 3 (msec)	OFF Time 3 (msec)	ON Time 4 (msec)	OFF Time 4 (msec)
	750	750	750	750	0	0	0	0
02 - 50	Same as Template 01							
Parameter Values								
0	0, 40- 4000	0, 40- 4000	0, 40- 4000	0, 40- 4000	0, 40- 4000	0, 40- 4000	0, 40- 4000	0, 40- 4000
1								
2								
3								
4								
5								
6								
7								
8								
9								
10								
11								
12								
13								
14								
15								
16								

Parameter No.	24	25	26	27	28	29	30	31
Template No.	Speech Delay Timer (sec)	Pause Timer (msec)	Ring Cadence OFF Timer (sec)	DTMFOut Dial (Level = - 6.0dB, Fixed)		DTMF Detection		
				DTMF Signal ON Time (msec)	DTMF Inter Digit Pause Timer (msec)	MinimumLevel (dB)	Minimum ON Time (msec)	Minimum OFF Time (msec)
01	0	1000	6	102	102	-4.5	34	68
02 - 50	Same as Template 01							
Parameter Values								
0	000 to 255							
1		500	1	51	17	0	17	17
2		1000	2	68	34	-1.5	34	34
3		1500	3	85	51	-3.0	51	51
4		2000	4	102	68	-4.5	68	68
5		2500	5	119	85	-6.0	85	85
6			6	136	102	-7.5	102	102
7			7	153	119	-9.0	119	119
8			8	170	136	-10.5	136	136
9			9	187	153		153	153
10				204	170		170	170
11					187		187	187
12					204		204	204
13								
14								
15								
16								

Parameter No.	32	33	34	35	36	37	38
Template No.	Flash Timer (msec)	On-Hook Speed (msec)	Off-Hook Speed (msec)	Current Limiting	Minimum Loop Current (mA)	Tip-Ring Voltage (Volts)	Ringer Impednace
01	600	< 0.5	8	X	10	3.5	High
02 - 50	Same as Template 01						
Parameter Values							
0				X	10		
1	83	< 0.5	512	O	12	3.1	High
2	100	3	128		14	3.2	Synthesized
3	200	26	64		16	3.35	
4	300		8			3.5	
5	400						
6	500						
7	600						
8	700						
9	800						
10	900						
11							
12							
13							
14							
15							
16							

Parameter No.	39	40	41	42	43	46	47	50
Template No.	Ringer Threshold (Vrms)	PPDC	Enable	DTMF String	Category (Logical Partition)	Rx Gain at SIP Trunk (Pass Through FAX)		Idle Wait Timer (Sec.)
						Data Gain (dB)	Bypass Gain (dB)	
01	13.5 - 16.5	0	X	CCC	1	-11	-9	002
02 - 50	Same as Template 01							
Parameter Values						As per table 1		
0		0	X	4 digit max				
1	13.5 - 16.5	1	0		1			01 - 20
2	19.35 - 23.65	2			2			
3	40.5 - 49.5	3			3			
4		4			4			
5		5						
6		6						
7		7						
8		8						
9		9						
10								
11								
12								
13								
14								
15								
16								

Table 1:

Code	Open Loop Disconnect Timer (msec)	Code	Open Loop Disconnect Timer (msec)	Code	Open Loop Disconnect Timer (msec)
1	17	23	391	45	765
2	34	24	408	46	782
3	51	25	425	47	799
4	68	26	442	48	816
5	85	27	459	49	833
6	102	28	476	50	850
7	119	29	493	51	867
8	136	30	510	52	884
9	153	31	527	53	901
10	170	32	544	54	918
11	187	33	561	55	935
12	204	34	578	56	952
13	221	35	595	57	969
14	238	36	612	58	986
15	255	37	629		
16	272	38	646		
17	289	39	663		
18	306	40	680		
19	323	41	697		
20	340	42	714		
21	357	43	731		
22	374	44	748		

For example, to change the Rx CLI Type in Template 07 from 'Any ETSI DTMF format' to 'Any FSK V.23 format', dial ***5902-1-07-01-3***
Where,
07 is the template number
05 is the parameter number for Rx CLI Type
2 is the code for Any FSK V.23 format.

To default CO Hardware Template, dial:

- ***5901-1- CO Hardware Template Number*** to default a single template.
- ***5901-2- CO Hardware Template Number - CO Hardware Template Number*** to default a range of templates.
- **5901-*** to default all templates.

To assign a CO Hardware Template to a CO port, dial:

- ***5903-1-CO-CO Hardware Template Number*** to assign a hardware template to a single CO port.
- ***5903-2-CO-CO-CO Hardware Template Number*** to assign a hardware template to a range of CO ports.
- ***5903-*-CO Hardware Template Number*** to assign a hardware template to all CO ports.
  Where,
  CO is the Software Port number of the CO port from 001 to 128.
  Template Number is the number of the customized CO Hardware Template, from 01 to 50. Default: Template 01.

- Exit SE mode.

E&M Feature Template

The E&M Feature Template is a set of features specific to E&M signaling, which define the behaviour of the E&M ports, according to their 'Orientation Type', whether they are functioning as Stations, Trunks or Tie-Lines.

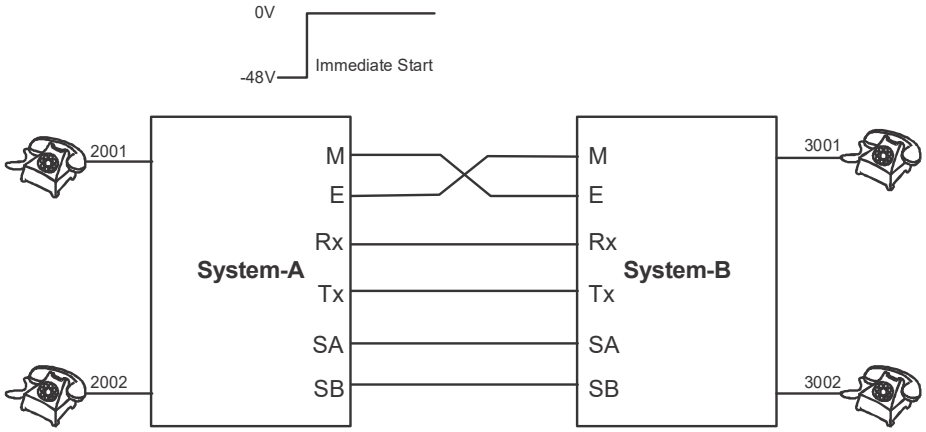
The E&M Feature Template is applied also on T1E1PRI trunks which use E&M signaling.The SARVAM UCS offers 50 such Templates. The E&M templates are loaded with default values that serve the requirements of a broad user base, but can be customized as per user requirements.

E&M Feature Template Parameters

The E&M Feature Template has the following parameters:

- **Seizure Type:** E&M Trunk Seizure Type is the line protocol that defines how the equipment seizes the E&M Trunk. It is also referred to as Start Dial Supervision Signaling Protocol. SARVAM UCS supports the following Seizure Types:

  - **Immediate:** The method of seizing the E&M Line using Immediate Start for Outgoing and Incoming calls is illustrated below.



Outgoing Call:
While making an outgoing call, when the extension user of System A seizes the E&M Port of System A, the status of the "M" wire of its E&M port will go high, indicating that it has seized the E&M line.
There will not be any signaling over the "E" wire of System A's E&M Port during seizure.

Incoming Call:
While receiving an incoming call over its E&M port, System-B will be ready to receive digits as soon as it detects high state on its "E" wire.

There will not be any signaling over the "M" wire of E&M Port of System B while receiving an incoming call.

- **Immediate with Ack:** The method of seizing the E&M Line using Immediate with Acknowledgment for Outgoing and Incoming calls is as follows:

Outgoing Call:
If this seizure type is selected, while an outgoing call is made by seizing the E&M Port, the 'M" wire will go high immediately.

The remote end will acknowledge this by making its 'M' wire high, which in turn will activates (high) 'E' wire of the E&M port of System-A.

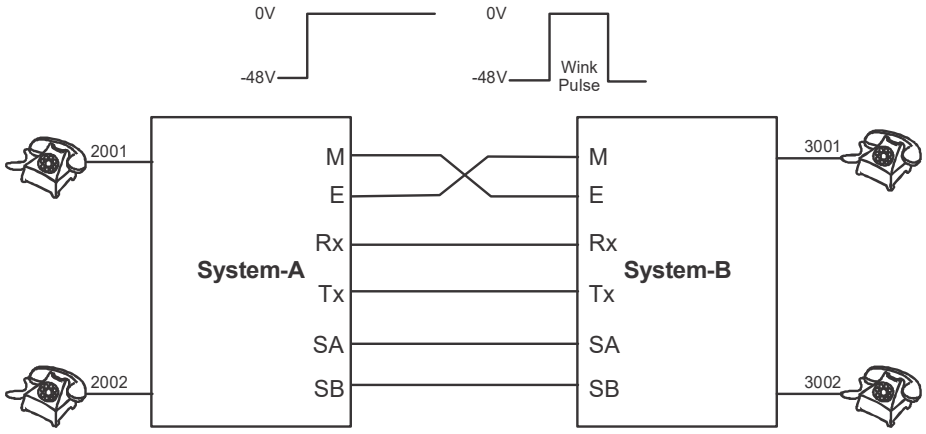
On sensing high signal on 'E' wire, System-A will start dialing out the DTMF/Pulse digits.

Incoming Call:
On detecting high signal on "E" wire of the E&M port, the system will consider it to be an incoming call seizure and hence it will immediately make its 'M" wire high, which will allow the remote end to dial out the DTMF/Pulse digits.

Call Disconnection:
If the parameter 'Release Type' is configured as 'Status Change' and for this type of seizure, the "M" wire at the remote end goes Low for some call condition, the call will be disconnected. For example, "M" wire at the remote end will go 'Low' in the following conditions:

- Remote end user dials invalid number and does not hang up on getting Error Tone.
- Remote end user dials valid extension number and after conversation remote end hangs up first.
- Remote end user dials valid number and extension does not reply, but the remote party does not disconnect the call.
- Remote end has made 'Orientation Type' of E&M port as 'Trunk' and for Incoming calls, all the extensions in the trunk landing group are busy and also second call is not allowed to the extension user, the system will disconnect the call after the expiry of the Ring Back Tone Timer, by making the 'M' wire Low.

- **Immediate + Wink:** The method of seizing the E&M Line using Immediate + Wink Start for Outgoing and Incoming calls is described below.



Outgoing Call:
While making an outgoing call when the System A attempts a seizure (grab), the state of the "M" wire of the E&M port of System A will go high.

To acknowledge this, the E&M port of System B will send Wink signal over its "M" wire, when System B is ready to receive digits.

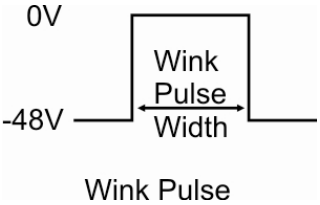
System A will wait for the duration of the 'Wait Wink Timer'. On receiving the acknowledgment in the form of Wink signal on the "E" wire of its E&M port, before the expiry of the Wait Wink Timer, System A will consider the trunk seizure as successful. Digits will be dialed out from E&M port.

If Wink is not received from System B within the ‘Wait Wink Timer’, System-A will drop the call.

Incoming Call:
While receiving an incoming call over the 'M' wire of its E&M port, System B will send the Wink signal to the System A, which has initiated the seizure (grab).

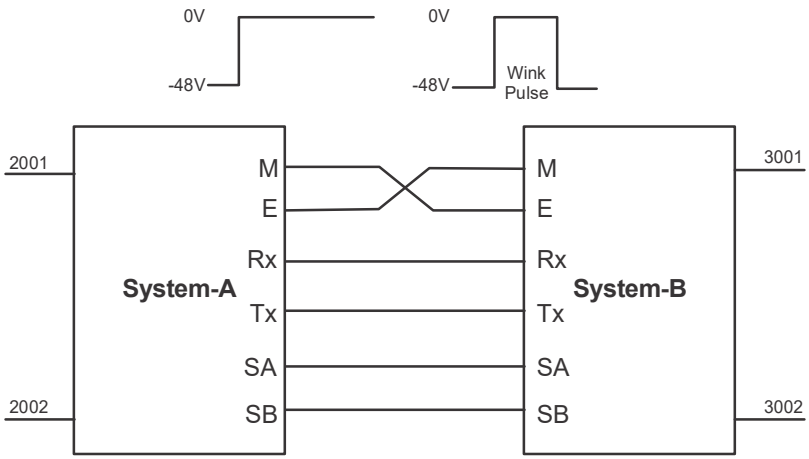
The ‘Wink’ signal will be sent by the System B when it is ready to receive the digits from the System A.

You can change the 'wink' pulse width by configuring the 'Wink Pulse Timer'.



The width of the Wink Pulse ranges from 0000 to 9999 milliseconds.

- **Immediate with Ack + Wink (MFC R2):** The method of seizing the E&M Line using Immediate with Ack + Wink (MFC R2) for outgoing and incoming calls is described below.



Outgoing Call:
When System-A attempts a seizure of (attempts to grab) the E&M Port, the 'M' Wire of the E&M Port of System-A will go high and wait for the E&M Port of System B to turn its 'M' wire high.

System-B detects this on its 'E' wire. To acknowledge this, the E&M port of System-B will turn its 'M' wire high and send a Wink signal over its 'M' Wire. Sending of the wink signal indicates the readiness of System-B to receive the digits of the called party number.

System-A will wait for the duration of the 'Wait Wink Timer' to receive the acknowledgment in form of the Wink Signal on the 'E' wire of its E&M port.

When System-A receives the acknowledgment from System-B, before the Wait Wink Timer expires, System-A considers the trunk seizure as successful and starts dialing out DTMF digits as per the MFCR2 Signaling protocol.

However, if System-A does not receive the Wink Signal within the ‘Wait Wink Timer', or if invalid Wink Pulse is received (not according to Wink Pulse Timer), System-A will drop the call by turning its 'M' wire low.

Incoming Call:
On detecting high signal on 'E' wire of the E&M port, System-B will consider it to be an incoming call seizure and hence it will immediately make its 'M' wire high and send the Wink signal on the 'M' wire of System-A to indicate its readiness to receive the called party number digits. The width of the Wink Pulse (referred to as 'Proceed To Send' in MFCR2 Signaling) can be changed by setting the 'Wink Pulse Timer'.

System-A dials out the digits as per the MFCR2 Signaling protocol.

When the called extension of System-B answers the call, System-B sends the Wink signal on the 'M' wire of its E&M Port to indicate the call maturity.

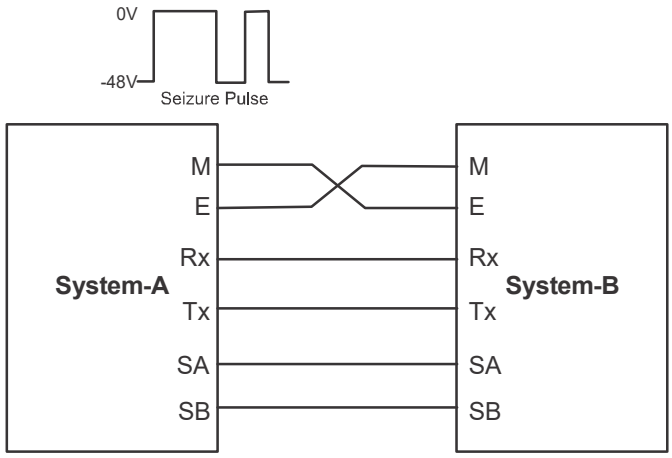
Call Disconnection:

The call can be disconnected by the calling party, System-A, or the called party, System-B by changing the status of the 'M' wire to Low.

Call Disconnection takes place when 'M' wire is low. So, it is recommended that the Call 'Release Type' of the E&M Port for this Seizure Type (Immediate with Ack+Wink) be set to 'Status Change'.

If you select Immediate with Ack+Wink as the Seizure Type, you must configure the MFCR2 Signaling parameters.

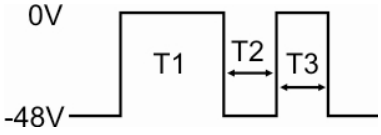
- **Seizure Pulse:** The method of seizing the E&M Line using Seizure Pulse for Outgoing and Incoming Calls is described below.



Outgoing Call:
While making an out going call from the E&M port of System-A, it will send Seizure Pulse over the "M" wire of its E&M port to seize the line.

Incoming Call:
While receiving an incoming call over the E&M Port System B will detect valid Seizure Pulse over the "E" wire of its E&M port.

Seizure Pulse can be set for various time periods T1, T2 and T3 as required.



The Seizure Pulse for T1, T2 and T3 ranges from 000 to 999 milliseconds.

- **Seizure Pulse + Wink:** The method of seizing the E&M Line using Seizure Pulse + Wink for Outgoing and Incoming Calls is as follows:

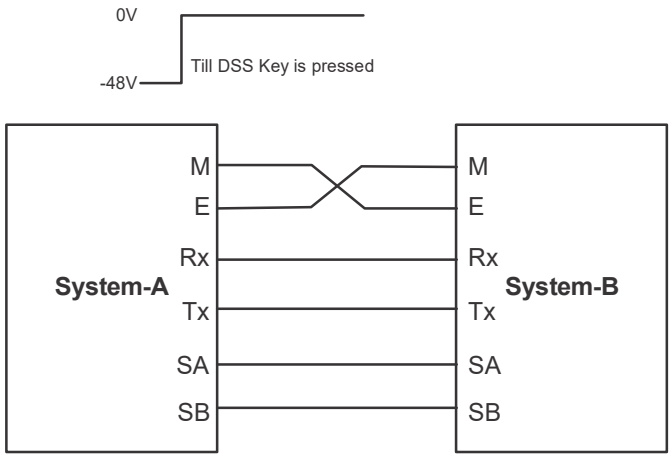
  Outgoing Call:
  While making an OG call, "Seizure Pulse" (as configured) will be sent on the "M" wire of E&M Port and will start ‘Wait Wink Timer’ and expect ‘Wink’ from the remote device.

  On receiving a valid Wink Pulse from the remote end within the Wait Wink Timer, digits will be dialed out on the E&M Port.

  If a valid Wink Pulse is not received from the remote end, digits will be dialed out on the expiry of ‘Wait Wink Timer’.

  Incoming Call:
  On detecting valid Seizure pulse (matching with configured value of seizure pulse) on "E" wire of the E&M Port, the E&M Port will send Wink pulse (of configured value) on "M" wire, and the call will be considered to be present.

- **Express:** Express Signaling works as illustrated below.



  To make a call from System A to System B, the caller from System A presses the DSS Key of the desired E&M port.

  For as long as the DSS key is pressed by caller from System A, the signal on the "M" wire of E&M port of System A will be high.

  When the destination extension on System B answers, the caller from System A releases the DSS key, as the line seizure is successful.

  When the caller from System A releases the DSS key, the signal on the 'M' wire of the E&M port of System A, and the "E" wire of the E&M port of System B will go low.

- **Radio A:** This seizure type to be used for supporting Combat Net Radio (CNR) signaling on the E&M port.

  The E&M port of the SARVAM UCS will detect this pulse on the 'E' wire of its E&M port and recognize it as a seizure signal (incoming call indication).

The length of this input signal (pulse) can be defined by setting the 'Minimum Pulse Width for Radio Seizure' Timer. The SARVAM UCS will recognize the input signal on the 'E' wire of its E&M port as a seizure signal only if the signal is present for the duration of this timer.

Once the incoming call is detected by the E&M port, the call is routed to the Routing Group number as per the call routing logic configured for the E&M port.

When Routing Group member (extension) answers the call, speech is established with the CNR user.

Outgoing Call:
When any extension user of the System wants to contact the CNR user (Soldier), extension user must grab the E&M port by dialing TAC/Selective Trunk Access / DSS key of E&M Port.

When E&M Port is grabbed by the extension user, the 'M' wire of the E&M port is made high, to indicate the seizure signal to the radio equipment. The radio equipment then passes on the call to the CNR user's wireless phone.

Incoming Call:
When status of "E" wire of the E&M port goes high for greater than or equal to the 'Minimum Pulse Width for Radio Call' it is considered to be an incoming call, and the call is routed as per the current call routing logic.

The Minimum Pulse Width for Radio Call' timer will be applicable only if the seizure type is configured as 'Radio A' or 'Radio B'. The range of this timer is from 0000 to 9999 milliseconds. The default value of this pulse is 150 milliseconds.

When an extension user of the System answers an incoming call, the 'M' wire of the E&M Port will be made high.

The call between the extension user and the CNR user can be disconnected only if the extension user disconnects the call. So, it is recommended that the call 'Release Type' of the E&M port be set to 'None'.

- **Radio B:** Characteristics of M and E wire for seizure type 'Radio B' are as follows:

  Outgoing Call:
  When any extension user of the System wants to contact the CNR user (Soldier), extension user must grab the E&M port by dialing TAC/Selective Trunk Access / DSS key of E&M Port.

  When E&M Port is grabbed by the extension user, the 'M' wire of the E&M port is made high, to indicate the seizure signal to the radio equipment. The radio equipment then passes on the call to the CNR user's wireless phone.

  Incoming Call:
  When the status of "E" wire of the E&M port goes high for greater than or equal to 'Minimum Pulse Width for Radio Call' SARVAM UCS considers it to be an incoming call, and routes the call as per the current call routing logic.

  When an extension user answers the call, there is no signaling on the 'M' wire of the E&M port by SARVAM UCS.

The call between the extension user and the CNR user can be disconnected only if the extension user disconnects the call. So, it is recommended that the call 'Release Type' of the E&M port be set to 'None'.

The following table shows the status of the "M" wire of an E&M Port while making Outgoing calls and while receiving Incoming calls.

E&M Seizure Type	Status of 'M' Wire after seizure and in conversation in an OG Call	Status of 'M' Wire when an IC call is initiated	Status of 'M' Wire on answering and when in conversation in an IC Call
Immediate	High	Low	High
Immediate with Ack	High	Low	High
Immediate + Wink	High	Wink	High
Immediate with Ack +Wink (MFC R2)	High	Wink	High
Seizure Pulse	Low	Low	Low
Seizure Pulse + Wink	Low	Wink	Low
Express	Low	Low	Low
Radio A	High	Low	High
Radio B	High	Low	Low

- **Orientation Type:** Configure the Orientation Type of the E&M Port according to your installation scenario.

  Select 'Station' if the E&M port is to function as Extension. All Extension-related parameters, the "Station Basic Feature Template" and the "Station Advanced Feature Template", will be applicable to this port.

  Select 'Trunk' as orientation type if the port is to be assigned the feature of a trunk line. All trunk-related parameters, "E&M Feature Template", will be applicable to this port.

  Select 'Tie Line'[186] if the E&M port is to function as both a Station and a Trunk. The system will regard the port as a Station for incoming calls and as a Trunk for outgoing calls. The Station Basic and Advanced Feature Templates as well as the Trunk Feature Template will be applied on this port.

  By default, the Orientation Type of all E&M ports is 'Station'.

- **Dial Type:** Digits can be dialed over E&M Tie Line by two methods:

  - **Tone:** In this Dial Type, the DTMF signals will be sent on the "Tx" of the E&M port of the originating side and it will be received over the "Rx" of the E&M port of the terminating side.

  - **Pulse:** In this Dial Type, the dialed digits will be sent on the "M" wire of the E&M port of the originating side and will be received over the "E" wire of the E&M port of the terminating side.

186. Such as in Systems used as Transit Exchanges as in a PLCC Network.

The way digits are sent varies according to the Trunk Seizure Type, as described for each Seizure Type below.

- ***Express:*** *the caller can make call by pressing the DSS key.*
- ***Immediate:*** *the System seizes the Tie Line and the system will start the Pause Timer. On the expiry of the Pause Timer the System sends the digits to the remote System.*
- ***Seizure Pulse + Wink:*** *on receiving Wink signal from the terminating end, the originating side (which initiates seizure) will start the Pause Timer and on expiry of Pause Timer it will start sending digits.*
- ***Seizure Pulse:*** *the originating System system will start the Pause Timer after sending the Seizure Pulse. On expiry of the Seizure Pulse Timer it will send digits to the terminating end.*

- **Pulse Dial Ratio:** This parameter is to be configured if 'Pulse' is selected as Dial Type in the previous parameter. Select the Pulse Dial Ratio from any of the following values:
  - 10PPS, 1:2
  - 10PPS, 2:3
  - 10PPS, 1:1
  - 20PPS, 1:2
  - 20PPS, 2:3
  - 20PPS, 1:1

  The default Pulse Dial Ratio is 10PPS, 1:2

- **Wink Wait Timer (sec):** This parameter is to be configured, if 'Immediate + Wink' or 'Seizure Pulse + Wink' have been selected as Seizure Type for the E&M port.

  The Wink Wait Timer is the Time period for which the system waits for the acknowledgment in the form of Wink Signal on "E" wire of the E&M port of the SARVAM UCS to consider it as a successful seizure.

  The range of this timer is from 000 to 255 seconds. By default the timer is set to '000' seconds.

- **Wink Pulse Timer (msec):** This parameter is to be configured, if 'Immediate + Wink' or 'Seizure Pulse + Wink' have been selected as Seizure Type for the E&M port.

  The Wink Pulse Timer defines the width of the Wink Pulse. The range of this timer is from 0000-9999 milliseconds. By default the timer is set to '0000' milliseconds.

- **Seizure Pulse Timers (msec):** This parameter is to be configured, if 'Seizure Pulse' is selected as Seizure Type for the E&M port.

  - **T1:** This is the time period of the first ON period of the 'Seizure Pulse'.

  - **T2:** this is the time period of the second ON period of the 'Seizure Pulse'

  - **T3:** this is the time period of the third ON period of the 'Seizure Pulse'.

  The range of T1, T2 and T3 is from 000-999 milliseconds. By default the timer is set to '000' milliseconds.

- **Minimum Pulse Width for Radio Seizure (msec):** This Timer is to be configured if 'Radio A' or 'Radio B' have been selected as the Seizure Type for the E&M ports.

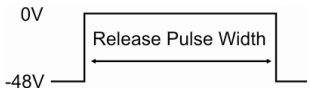
  The Minimum Pulse Width for Radio Seizure defines the time for which the SARVAM UCS will wait to detect the width of the pulse sent by the Radio Interface Device[187] on the 'E' wire of the E&M port and recognize it as a seizure signal (incoming call indication).

The range of this timer is from 000 to 999 milliseconds. By default the timer is set to '150' milliseconds.

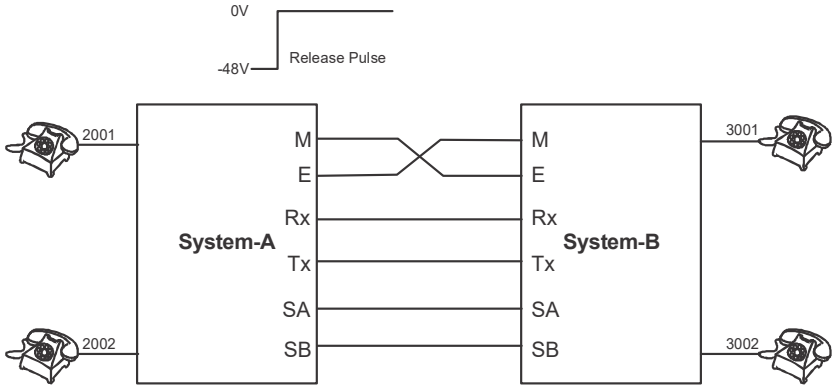
- **Release Type:** SARVAM UCS supports four methods to 'release' E&M calls based on, which end will release the call and 'release pulse width'. These are:

  - **None:** Select this option if you have selected "Express" as Trunk Seizure Type for the E&M port. It is advisable to keep the Release Type as "None" in case the protocol does not support any signaling for disconnecting the E&M port, for the reason that 'Trunk-to-Trunk Inactivity Timer' will be started if the E&M port with Release type "None" is involved in a Trunk-to-Trunk call.

  - **Release Pulse:** Select this option if the specific Pulse width of the Release Pulse is to be used to disconnect the call. This Pulse width is configurable, as shown below:



The call can be disconnected by either party by sending Release Pulse. Consider the following example:



SLT 2001 is connected to System-A and SLT 3001 is connected to System-B.

SLT 2001 user goes ON-Hook.
- System-A will send Release Pulse over the "M" wire of its E&M port.
- System-A will release its E&M port.
- The Release Pulse sent by System-A will be received and detected on the "E" wire of the E&M port of the System-B.
- System-B will release its E&M port.

SLT 3001 user goes ON-Hook.
- System-B will send Release Pulse over the "M" wire of its E&M port.
- System-B will release its E&M port.
- The Release pulse sent by System-B will be received and detected on the "E" wire of the E&M port of the System-A.
- System-A will release its E&M port.

  - **Status Change:** Select this option, if Status change of 'M' wire (‘M’ wire is low) is to be considered for release of the E&M call.

    By default, 'Status Change' is selected as the Release Type for all E&M ports.

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187. Connected on the 'M' wire of the E&M port. The Radio Interface Device sends pulse of approximately 100msec or higher.

If you selected ‘Immediate with Ack + WInk’ as Seizure Type for the E&M Port, select ‘Status Change’ as Release Type for the port.

- **Release Pulse Timer (msec):** This timer is to be configured if you selected the option 'Release Pulse' as the Release Type for E&M calls in the previous parameter.

  This timer defines the specific Pulse width of the Release Pulse is to be used to disconnect the call. The range of this timer is from 0000-9999 milliseconds. By default the timer is set to '0000' milliseconds.

- **CCS - When End Point:** CCS (Compander Control Signal) is a type of signal used by PLCC Networks to improve the quality of speech transmission. The PLCC network awaits this signal from the System when speech is established. SARVAM UCS supports CCS. The system sends CCS signal to the PLCC panel.

  This parameter is relevant if the E&M line is being used in a Power Line Communication Network (PLCC).

  This flag should be enabled if the E&M port is used as an Endpoint in a PLCC network. When the E&M Port is used as an Endpoint, the system sends CCS to the PLCSS panel while making an outgoing call through the E&M port and when receiving an incoming call on the E&M port.

  By default, the flag is enabled.

- **CCS - When Transit Exchange:** This flag is to be enabled if the E&M port is used as a Transit exchange in a PLCC network.

  When the E&M Port is used as a Transit Exchange: The system sends CCS to the PLCC panel when there is an Incoming/Outgoing Transit call through the E&M port.

  By default, the flag is enabled.

- **DTMF Detection:** This flag is relevant when the "Dial Type' for the E&M Port is selected as 'Tone'. This flag is of significance while receiving incoming calls.

- **Max. OG Pulse Digit Count:** This count defines the maximum number of digits that can be dialed out to make a call. When dialing out the number, if the number of digits exceeds this count, the port which is used for dialing these numbers is released automatically.

- **Idle Wait Timer (sec):** This timer signifies the time after which the codes could be simply (Station Numbers or Station Numbers with Exchange ID) dialed over the E&M trunk.

  The Idle Wait Timer is useful in two conditions:

  - ***When Forced Disconnection is used.*** For example, two exchanges A and B are connected through E&M trunk. Extension 2002 of System A is talking to extension 3001 of System B over the E&M line.

    Extension 2001 of System A calls extension 3001 of System B and finds it to be busy.

    Extension 2001 is allowed to use forced disconnection feature. Extension 2001 issues the forced disconnection command. The System A disconnects the Extension 2002. It then waits for the Idle Wait Timer to expire and then dials 3001 over the E&M trunk.

- ***To stop any station from grabbing the E&M trunk until call is released.*** For example, when Extension 2001 of System A goes On-Hook, System A sends a release signal over the E&M trunk to System B. In turn, System B sends a release signal to System A as an acknowledgment.

  The E&M Idle Wait Timer set in System A does not allow any other extension of System A to grab the E&M trunk. Similarly, E&M Idle Wait Timer set in System B does not allow any extension of System to grab the E&M trunk.

- **Flash Timer (msec):** This Timer is significant when the System acts as a Transit exchange for a call. The flash received on one E&M Port is generated on another E&M Port involved in a Transit call.

- **Pause Timer (msec):** This Timer defines the time for which the system waits before dialing the outside number after grabbing the E&M trunk.

  Sometimes an extension user may not get a dial tone immediately on grabbing a trunk, in this case, the extension user may wait for the dial tone before dialing out the number. However, when the system dials out the number, if there is no pause time, it is possible that the system may dial out the number before getting the dial tone. This may result in a wrong number being dialed out. The Pause Timer helps avoid this.

  The range of this timer is from 0000 to 9999 msec. By default the Timer is set to 800 msec.

- **Pseudo Answer Supervision Timer (sec):** This is the time period after which, the system will consider the call as matured, irrespective of whether the call was answered or not. At the end of the Timer, the system will start detecting Disconnect Supervision.

  The range of this timer is from 000 to 255 seconds. By default the Timer is set to 030 seconds.

- **Ring Timer (sec):** This is the time for which the extensions connected to SARVAM UCS ring for incoming calls.

  The Ring Timer is useful in situations where the users may not be able to immediately answer on the first few rings. The range of this timer is from 000 to 255 seconds. By default the Timer is set to 255 seconds.

- **Inter-Digit Pause Timer (msec):** This Timer defines the time gap to be inserted between digits of a number string being dialed by the system.

  The range of this timer is from 000 to 999 milliseconds. By default the Timer is set to 750 milliseconds.

- **DTMF Out Dial:** This parameter is of significance when the system dials out DTMF digits to enable the device at the remote end (in this case a System) to detect and decode the Tones. You must configure both the DTMF ON Time and the Level (dB) according to the DTMF digit detection capacity of the remote System.

  For example, System A and System B are connected over E&M Line. System B detects DTMF digits only if the tone remains present (ON) for 100 milliseconds frequency and at a transmit level of 4 dB. The DTMF Out Dial parameter for System A should be configured accordingly. The DTMF Out Dial ON Time should be set to 100ms and Level to 4dB.

  - **DTMF ON Time (msec):** This is the Time for which the DTMF digit tone will remain ON, while being dialed out by the SARVAM UCS. The range of this timer is 50 to 500 milliseconds. By default the ON Time is set to 100 milliseconds.

This Timer must be configured according to the DTMF digit detection capacity of the remote device.

- **DTMF Tx Level (dB):** This is the Transmit Level of the DTMF digit dialed out by the system. The range of DTMF Out Dial 'Level' is from 0 to 7. By default DTMF Out Dial Level is set to 3.

- **MFC R2 Signaling:** This parameter is relevant only if you selected 'Immediate with Ack + Wink' as the Seizure Type. Configure the following timers related to MFCR2 Signaling.

  - **Forward Tone Maximum ON Time (T1) (sec):** the range of this timer is from 1 to 99 seconds. By default it is set to 15 seconds.

  - **Forward Tone Maximum OFF Timer (T2) (sec):** The range of this timer is from 1 to 99. By default it is set to 24 seconds.

  - **Maximum Compelled Cycle Timer (T3) (sec):** the range of this timer is from 1 to 99. By default it is set to 15 seconds.

  - **Pulse Duration for Pulse Signal (msec):** The range of this timer is from 001 to 999. By default, it is set to 150 seconds.

  - **Pulse Signal Maximum Wait Timer (sec):** The range of this timer is from 1 to 99. By default, it is set to 15 seconds.

  - **First Forward Tone Wait Timer (sec):** The range of this timer is from 8 to 24. By default, it is set to 15 seconds.

  - **Minimum MF Signal Persist Timer (msec):** The range of this timer is from 1 to 255 seconds. By default, it is set to 20 seconds.

  - **Inter Digit Wait Timer (sec):** By default it is set to 10 seconds.

  - **Ask CLI:** By default, it is enabled.

- **Prefix String:** This parameter is useful when the System's connected using E&M lines do not send 0 when the user dials a number of another exchange. You must configure this parameter as zero along with the exchanges's ID and appropriately configure the strip digit count parameter in the CUG Table.

- **Category (Logical Partitioning):** This parameter assigns the E&M Port to a trunk category for the purpose of Logical Partitioning. By default all E&M Ports are assigned to Category 3[188].

  If you have re-defined Category 3 or have assigned E&M ports to a different category, say Category 2, enter the same number here.

  You may configure the call permission between the Category assigned to E&M Ports and other Categories. Refer the feature description "Logical Partition" to know more.

188. Trunk ports used to interconnect two Systems are assigned this category.

Customizing E&M Feature Template

By default E&M Feature Template 01 is applied on all E&M Ports. This template has 'Station' as the default Orientation Type.

If all the E&M Ports are to be configured as 'Stations', then retain this template.

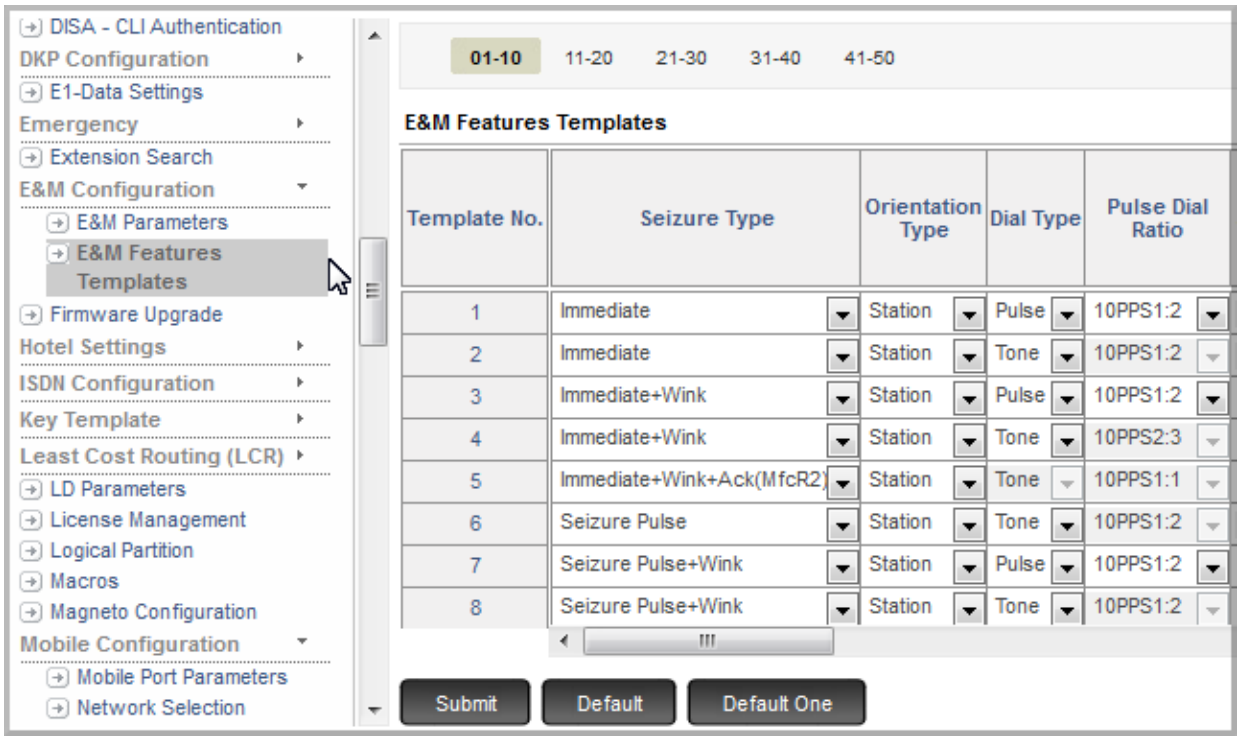
If all the E&M Ports are to be configured as 'Trunks', use the default E&M Feature Templates 09 and 10 which have 'Trunks' as Orientation Type.

If some of the E&M Ports are to be configured as Stations, some as Trunks and yet others as Tie Lines, prepare different E&M Feature Template for each Orientation Type and apply them to the related ports.

The E&M Feature Template can be customized using Jeeves and a Telephone.

Customizing E&M Feature Template using Jeeves

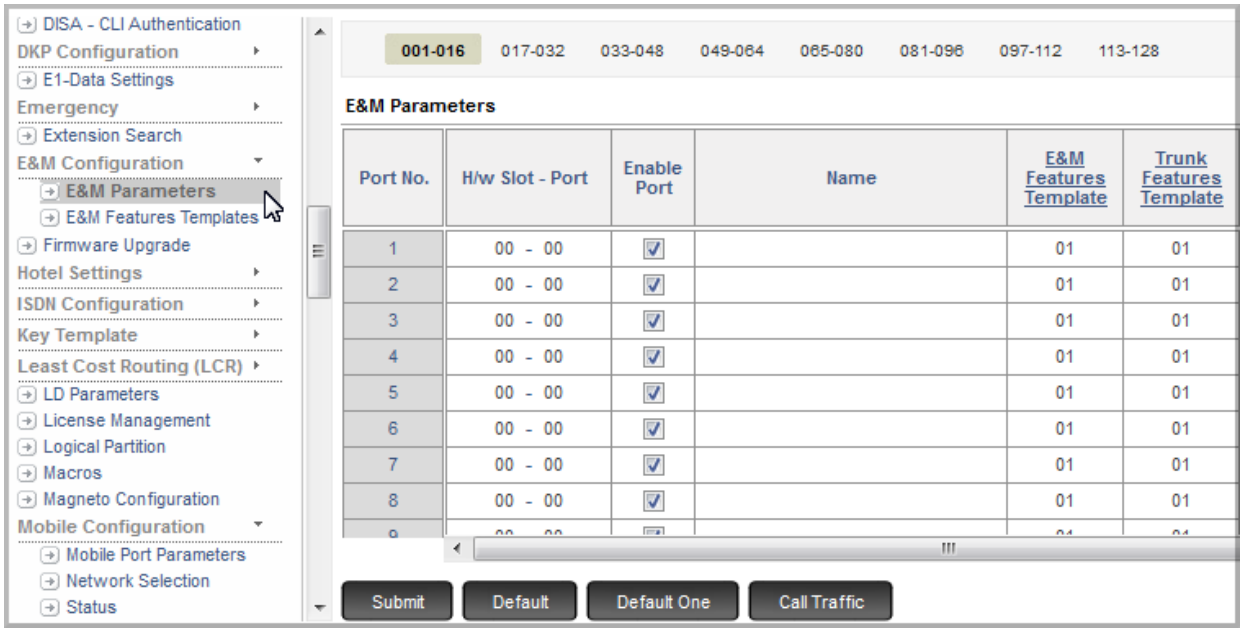
- Log in to Jeeves as System Engineer.
- Under **Configuration**, click **E&M Configuration**.
- Click **E&M Feature Template**.



- Select an E&M Feature Template Number.
- Customize the E&M Feature Template.
- Click **Submit** to save changes.
- Now, apply the E&M Feature Template you customized to the E&M Ports and T1E1PRI ports.

To apply E&M Feature Template on E&M ports:

- Click the **E&M Parameters** to open the page.
- Apply the E&M Feature Template you customized to the E&M Port by entering the template number in the **E&M Feature Template** field of this port.

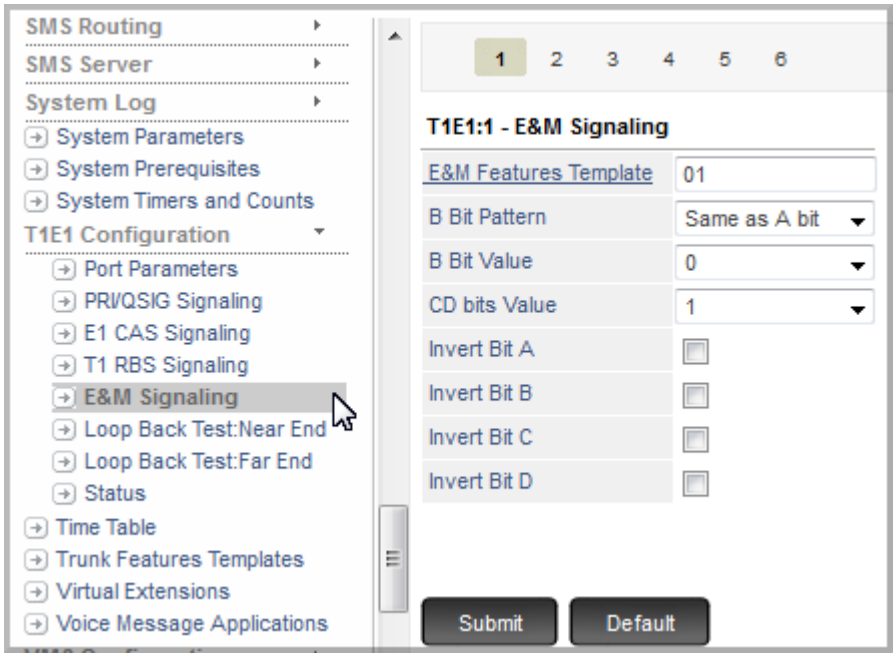


- Click **Submit** to save changes.

To apply E&M Feature Template on T1E1PRI ports:

- Click **T1E1 Configuration**.
- Click the **E&M Signaling**.

- Click the tab of the desired T1E1 trunk port number (1 to 8), you wish to apply the E&M feature Template.



- Enter the number of the Template you customized in the field **E&M Feature Template** of the selected T1E1 trunk port.

- Click **Submit** to save changes.

Customizing E&M Feature Template using a Telephone

- Enter SE mode from a DKP/SLT.

  To change the default values of a Parameter in an E&M Feature Template:
  - ***6002-1-Template Number-Parameter Number-Code*** to change the value of a parameter in a single template.
  - ***6002-2-Template Number-Template Number-Parameter Number-Code*** to set the same value for the parameter in a range of templates.
  - ***6002-*-Parameter Number-Code*** to set the same value for the parameter in all templates.
    Where,
    Template Number is the number of the E&M Feature Template from 01 to 50.
    Parameter Number is the number of the E&M Feature Template Parameter from 01 to 35.
    Code is the parameter value.

  Refer the following table for the Parameter Number and default parameter values of all E&M Feature Templates.

Default E&M Feature Templates

Para.No.	01	02	03	04	05	06	07	08	09	10	11	12	13	14	15	16
Template No.	Seizure Type	Orientation Type	Dial Type	Pulse Dial Ratio	Wait Wink Timer (sec)	Wink Pulse Timer (msec)	Seizure Pulse Timers (msec)			Minimum Pulse Width for Radio Seizure (msec)	Release Type	Release Pulse Timer (msec)	CCS - When End Point	CCS - When Transit Exchange	DTMF Detection	Max. OG Pulse Digit Count
							T1	T2	T3							
01	Immediate	Station	Pulse	10PPS, 1:2	000	0000	000	000	000	0000	Status Change	0000	√	√	X	24
02	Immediate	Station	Tone	10PPS, 1:2	000	0000	000	000	0000	0000	Status Change	0000	X	X	X	24
03	Immdiate+Wink	Station	Pulse	10PPS, 1:2	002	0200	000	0000	0000	0000	Status Change	0000	√	√	X	24
04	Immdiate+Wink	Station	Tone	10PPS, 2:3	002	0200	000	0000	0000	0000	Status Change	0000	X	X	X	24
05	Immediate with Ack+ Wink (MFCR2)	Station	Tone	10PPS, 1:1	005	0040	000	0000	0000	0000	Status Change	0000	X	X	X	24
06	Seizure Pulse	Station	Tone	10PPS, 1:2	000	0000	230	0080	0080	0000	Release Pulse	0900	X	X	X	24
07	Seizure Pulse +Wink	Station	Pulse	10PPS, 1:2	002	0200	230	0080	0080	0000	Release Pulse	0900	X	X	X	24
08	Seizure Pulse +Wink	Station	Tone	10PPS, 1:2	002	0200	230	0080	0080	0000	Release Pulse	0900	X	X	X	24
09	Express	Trunk	Pulse	10PPS, 1:2	000	0000	000	0000	0000	0000	None	0000	X	X	X	24
10	Express	Trunk	Tone	20PPS, 1:2	000	0000	000	0000	0000	0000	None	0000	X	X	X	24
11 to 50	Same as 01															

Parameter Values

0											None		X	X	X	01-24
1	Immediate	Trunk	Pulse	10PPS, 1:2	000-255	0000-9999	000-999	000-999	000-999	000-999	Release Pulse	0000-9999	√	√	√	
2	Immediate with Ack	Station	Tone	10PPS, 2:3							Status Change					
3	Immediate+Wink	Tie Line		10PPS, 1:1												
4	Seizure Pulse			20PPS, 1:2												
5	Seizure Pulse + Wink			20PPS,2:3												
6	Express			20PPS, 1:1												
7	Radio A															
8	Radio B															
9	Immediate with Ack+ Wink (MFCR2)															

Para.No.	01	17	18	19	20	21	22	23	24	25	26	27	28	29	30	31	32	33	34	35
Template No.	Seizure Type	Idle Wait Timer (sec)	Flash Timer (msec)	Pause Timer (msec)	Pseudo Answer Supervision Timer (sec)	Ring Timer (sec)	Inter Digit Pause Timer (msec)	DTMFOut Dial		MFCR2 Signaling									Prefix String	Category (Logical Partitioning)
								ON Time (msec)	Level (dB)	Forward Tone Maximum On Timer (T1) (sec)	Forward Tone Maximum Off Timer (T2) (sec)	Maximum Compelled Cycle Timer (T3) (sec)	Pulse Duration for Pulse Signal (msec)	Pulse Signal Maximum Wait Timer (sec)	First Forward Tone Wait Timer (sec)	Minimum MF Signal Persist Timer (msec)	Interdigit Wait Timer (sec)	Ask CLI		
01	Immediate	002	600	800	030	255	750	100	-6.0	15	24	15	150	15	15	20	10	√		3
02	Immediate	002	600	800	030	255	750	100	-6.0	15	24	15	150	15	15	20	10	√		3
03	Immdiate+Wink	002	600	800	030	255	750	100	-6.0	15	24	15	150	15	15	20	10	√		3
04	Immdiate+Wink	002	600	800	030	255	750	100	-6.0	15	24	15	150	15	15	20	10	√		3
05	Immediate with Ack+ Wink (MFCR2)	002	600	800	030	255	750	100	-6.0	15	24	15	150	15	15	20	10	√		3
06	Seizure Pulse	002	600	800	030	255	750	100	-6.0	15	24	15	150	15	15	20	10	√		3
07	Seizure Pulse +Wink	002	600	800	030	255	750	100	-6.0	15	24	15	150	15	15	20	10	√		3
08	Seizure Pulse +Wink	002	600	800	030	255	750	100	-6.0	15	24	15	150	15	15	20	10	√		3
09	Express	000	83	800	030	255	750	100	-6.0	15	24	15	150	15	15	20	10	√		3
10	Express	000	83	800	030	255	750	100	-6.0	15	24	15	150	15	15	20	10	√		3
11 to 50	Same as 01																			

Parameter Values

0		000-255	83-999	000-9999	000-255	000-255	000-999	50-500		1 to 99	1 to 99	1 to 99	001 to 999	1 to 99	8 to 24	1 to 255		X	0000-9999	
1	Immediate								-10.5									√		1
2	Immediate with Ack								-9.0											2
3	Immediate+Wink								-7.5											3
4	Seizure Pulse								-6.0											4
5	Seizure Pulse + Wink								-4.5											
6	Express								-3.0											
7	Radio A								-1.5											
8	Radio B								0											
9	Immediate with Ack+ Wink (MFCR2)																			

For example, you want to Immediate with Ack +Wink in the E&M Template number 2.

- Dial ***6002-1-2-01-9***
  Where
  2 is for E&M Template number 2
  01 is parameter number for Seizure Type
  9 is the value of Immediate with Ack+Wink

To restore default values to the Parameters of an E&M Feature Template:

- Dial ***6001-1-Template Number*** to restore default values of the parameters of a single template.
- Dial ***6001-2-Template Number-Template Number*** to restore the default values of a range of templates.
- Dial ***6001-**** to restore the default values of all templates.

To assign an E&M Feature Template to an E&M port

- Dial ***6003-1-E&M-Template Number*** to assign a template to a single E&M port.
- Dial ***6003-2-E&M-E&M-Template Number*** to assign the same template to a range of ports.
- Dial ***6003-*-Template Number*** to assign the same template to all ports.
  Where,
  E&M is the number of the E&M Software port from 001 to 128.
  Template Number is the number of the E&M Feature Template from 01 to 50.
  Default: Template 01 is assigned to all E&M ports.

To assign E&M Feature Template to a T1E1 Port:

- Dial ***6004-1-T1E1-Template Number*** to assign a template to a single T1E1 port.
- Dial ***6004-2-T1E1-T1E1-Template Number*** to assign the same template to a range of T1E1 ports.
- Dial ***6004-*-Template Number*** to assign the same template to all T1E1 ports.
  Where,
  T1E1 is the number of the T1E1PRI Software port, 01 to 08.
  Template Number is the number of the E&M Feature Template from 01 to 50.
  Default: Template 01 is assigned to all T1E1 ports.

- Exit SE mode.

Trunk Feature Template

The Trunk Feature Template is a set of general features that define the behavior of a Trunk port. SARVAM UCS offers 50 such Templates.

A Trunk Feature Template is assigned to all the Trunk types: SIP, T1E1 PRI, BRI, Mobile, CO and E&M.

Trunk Feature Template Parameters

The Trunk Feature Template contains the following features:

- **Time Table:** Select a Time Table for the Trunk ports.

  A Time Table is a schedule of the three Time Zones, namely: Working Hours, Break Hours, Non-Working hours for a week.

  Certain features of the SARVAM UCS like Operator, Auto Attendant, DISA, Trunk Landing Group, require the trunk to behave differently in each Time Zone. For example, it can be made to land on the Operator extension during working hours, and on the extension of the dining area during Break (lunch) hours, and on the extension of the Security Personnel during non-working hours.

  So, a Time Table is assigned to extensions defining the Time Zones for the entire week, so that the system can execute the Time Zone-dependent features and facilities according to the Time Table.

  There are 8 different Time Table templates to select from. By default, the Time Table 1 is assigned to all Trunk Feature Templates. All seven days of the week are 'working hours 9:00-18:00' with break hours '13:00 -14:00 hrs'.

  You may also customize the default Time Table 1 OR customize and assign a different Time Table to the Trunk Feature Template. Please refer the topic “Time Tables” for more details.

- **Operator:** Define the Operator for the Trunks on which the template is applied. Operator is used to route the call when the caller dials '9' during an Auto Attendant call. This parameter is of significance only if Built-In Auto Attendant/DISA is enabled on the trunk.

The system supports multiple Operators. In each Time Zone any one of the 20 Operators can be selected.

Trunks may be assigned to a single Operator, or different groups of Trunks may be assigned to different Operators, so that call management is more efficient. For instance certain Trunks may be assigned to Operator 1, while some may be assigned to Operator 2 and the rest to Operator 3.

Operator 1 is the default in the Trunk Feature Template. If you want to assign different trunks to different Operators, you must create a separate Trunk Feature Template with a different Operator for each trunk group.

Refer the topic "Operator" to know more.

- **CLI Based Routing:** Select the check box to enable CLI Based Routing on the Trunk for each Time Zone: Working Hours (WH), Break Hours (BH) and Non-working Hours (NH). Default: disabled.

  If you enable CLI Based Routing on the trunk for a Time Zone, make sure you also configure the CLI Based Routing Table. To know more, refer the feature description "CLI Based Routing".

- **Trunk Landing Group (Routing Group):** This parameter allows you to configure the group of extensions on which incoming calls on the trunks (to which this template is assigned) are to be landed. This group of extensions is referred to as 'Trunk Landing Group' (TLG).

  To configure the TLG, you must first configure Routing Groups. Refer "Trunk Landing Group (TLG)" for instructions on configuring trunk landing groups. Also refer the topic "Routing Group".

  There are as many as 96 Routing Groups which can be assigned as TLG. By default, Routing Group 01 is assigned as TLG for all Time Zones. If you have prepared a different TLG for each Time Zone, for example, Routing Group 02 for Working Hours and Break Hours, Routing Group 3 for Non-Working Hours, then enter the number of these Routing Groups in the TLG field.

- **Auto Attendant:** This parameter is to be configured if you want to enable "Auto Attendant" on the trunk ports on which you will apply the template.

  Auto Attendant can be enabled or disabled for each Time Zone, namely Working Hours (WH), Break Hours (BH) and Non-Working Hours (NH).

  - For each Time Zone, you may select the desired **Auto Attendant Type** from the following:

    - **OFF:** Select this option if you want to disable Auto Attendant for the Time Zone.

    - **Built-In Auto Attendant:** Select this option if you want the calls to be answered by the built-in Auto Attendant of the SARVAM UCS. In Built-In Auto Attendant, SARVAM UCS answers the call using Voice Modules, if assigned, or it answers the call and plays the appropriate call progress tone - Dial tone, Ring Back tone, Busy tone - for each call state.

      If you select this option, make sure you also configure the Built-In Auto Attendant related Timers and Flags, record and assign the Built-In Auto Attendant related Voice Message and set the Start Time for the Greeting Messages. Refer the topics "Auto Attendant" , "Voice Message Applications" and "Greeting Message Time" in *System Parameters* for instructions.

    - **Voice Mail Auto Attendant:** Select this option if you want to the calls to be answered by the Auto Attendant of the Voice Mail System. The Voice Mail System of SARVAM UCS answers calls and processes them according to the Voice Mail Auto Attendant Menu assigned to the trunk.

By default, Auto Attendant is disabled (OFF) for all the Time Zones.

- **Auto Attendant Delayed Timer:** Set this Timer, if you want to enable "Delayed Auto Attendant" on the trunk.

  When you enable Delayed Auto Attendant, SARVAM UCS routes the incoming call on the trunk to the Trunk Landing Group assigned to this trunk. It waits for the duration of the Auto Attendant Delayed Timer for any of the extensions in the Trunk Landing Group to answer the call.

  If none of the extensions in the Trunk Landing Group answers the call before the expiry of the Auto Attendant Delayed Timer, SARVAM UCS processes the call according to the type of Auto Attendant - Built-In Auto Attendant or Voice Mail Auto Attendant, set for the trunk.

  To enable Delayed Auto Attendant, set the timer to the desired value from the list. By default, it is set as Never that is it is disabled.

- **Voice Mail Auto Attendant (VMAA) Menu:** if you have selected the *Voice Mail Auto Attendant* as the Auto Attendant Type, select the VMAA Menu to assign to the respective Trunk Feature Template.

  You may click the *Voice Mail Auto Attendant (VMAA) Menu* link to edit the parameters of desired VMAA Menu. For details, see "Voice Mail Auto-Attendant Menu".

- **DISA:** This parameter is to be configured if you want to enable "Direct Inward System Access (DISA)" on the trunk ports on which you will apply the template.

  DISA can be enabled or disabled for each Time Zone, namely Working Hours (WH), Break Hours (BH) and Non-Working Hours (NH).

  For each Time Zone, you may select the desired DISA option from the following:

  - **Disabled:** Select this option if you want to disable DISA.

  - **PIN Auth.-Multiple calls:** Select this option if you want to enable DISA with PIN Authentication and allow multiple calls during the DISA login session.

  - **CLI Auth.-Multiple calls:** Select this option if you want to enable DISA login with CLI Authentication and allow multiple calls to be made during the DISA login session.

    Caller numbers that do not match with the CLI Table will be routed as per the logic of the Trunk Feature Template.

  - **CLI Auth.-One call:** Select this option if you want to enable DISA session with CLI Authentication, and allow only a single call to be made during the DISA login session. This form of DISA is used when SARVAM UCS is installed in a Gateway application. This form of DISA is applicable on CO Trunks only.

    By default, DISA is disabled for all Time Zones.

- **Trunk Auto Answer:** This parameter is relevant only if you want to enable the "Trunk Auto Answer" feature on the Trunk ports on which this template is applied.

  Trunk Auto Answer enables calls landing on a trunk to be answered automatically by greeting the caller with a voice message before the call is actually handled.
  You can set the following types of Trunk Auto Answer:

- **OFF:** Select this option if you do not want Trunk Auto Answer on the trunk.
- **For all Calls:** Select this option if you want all incoming calls landing on the trunk line to be answered.
- **When Busy:** Select this option if you want the system to answer incoming calls on the trunk to be answered if the landing destination is busy.
- **Delayed:** Select this option if you want SARVAM UCS to answer the incoming calls on the trunk if not answered by the landing destination within a certain time period, set as the *Delayed Trunk Auto Answer Timer*.

  The system first routes the incoming calls to the Trunk Landing Group. It waits for the duration of the Delayed Trunk Auto Answer timer for any of the extensions in the Trunk Landing Group to answer the call.
  If the call is not answered by any of the extensions, before the expiry of this timer, the system answers the call and routes it as per the Trunk Auto Answer logic.

  - If you select **Delayed** as the Trunk Auto Answer option, you must also configure the **Delayed Trunk Auto Answer** timer. Valid Range of the timer is 01 to 99 seconds. By default, it is set as 10 seconds.

Trunk Auto Answer can be enabled or disabled for each Time Zone, namely Working Hours (WH), Break Hours (BH) and Non-Working Hours (NH).

By default, Trunk Auto Answer is Off.

If you have enabled Trunk Auto Answer for All Calls, When Busy or Delayed, you must also set the Trunk Auto Answer Greeting Message.

- **Trunk Auto Answer Greeting Message:** This parameter is to be configured only if you have enabled Trunk Auto Answer **For All Calls** or **When Busy** for a Time Zone in the previous parameter.

  Assign the number of the Trunk Auto Answer Greeting with which callers will be greeted. For this you must first record a Voice Module with the desired Greeting Message and assign it to this parameter.

  You can assign up to 4 Greetings Messages for Trunk Auto Answer and assign a different Greeting message for each Time Zone. Refer the topic “Voice Message Applications” for instructions on configuring the greetings.

  Enter the number of the Greeting Message you want to be played for each Time Zone. By default, Trunk Auto Answer Greeting number 1 is assigned to all Time Zones.

- **Trunk Auto Answer RBT Message Type:** This parameter is to be configured only if you have enabled Trunk Auto Answer 'For All Calls' or 'When Busy' for a Time Zone in the previous parameter.

  When Trunk Auto Answer is enabled on a trunk, the system will answer the caller with a Greeting message once, and play the Ring Back Tone (RBT) Message Type you have selected. You can select an RBT Message Type from the following options:
  - **None:** The system will play Ring Back Tone to the caller after the Trunk Auto Answer Greeting Message.

- **Internal MOH:** The system will play internal music-on-hold to the caller after playing the Trunk Auto Answer Greeting Message.

- **RBT Message:** The system will play a voice message continuously to the caller.

  If you select this option, you must first record a Voice Module with the desired RBT Greeting Message.

  You can set up to 4 RBT Messages. You can also assign a different RBT message for each Time Zone. Refer the topic "Voice Message Applications" for instructions on configuring the RBT Message. Assign the number of the RBT Message you want to be played to callers in each Time Zone.

By default, 'None' is selected as RBT Message Type for all Time Zones.

- **Trunk Auto Answer Busy Bye Message:** This parameter is to be configured only if you have enabled Trunk Auto Answer ('For All Calls' or 'When Busy') for a Time Zone.

  When Trunk Auto Answer is enabled on a trunk, the system will answer the caller with a Greeting message once, and play the Ring Back Tone (RBT) Message Type you have selected (see previous parameter) continuously for the duration of the Built-In Auto Attendant Inactivity Timer. If the landing destination (called extension) is busy on the expiry of this Timer, the system will inform the caller about the busy state in two ways, which you can select from the following options:

  - **None:** The system will play a Busy Tone to the caller.

  - **Bye Message:** The system will play a voice message to the caller.

    If you select this option, you must first record a Voice Module with the desired Bye Message.

    You can set up to 4 Busy Bye Message. You can also assign a different Busy Bye message for each Time Zone. Refer the topic "Voice Message Applications" for instructions on configuring the Busy Bye Message.

    Assign the number of the Bye Message you want to be played to callers in each Time Zone.

  By default 'None' is selected as Busy Bye Message for all Time Zones.

- **Priority:** Select a Priority Level for the trunks on which the template will be applied.

  Each trunk of the SARVAM UCS can be assigned a Priority Level starting from 1, 2, 3... to 9, with '1' being lowest Priority and '9' being highest Priority.

  Whenever there are incoming calls on multiple trunks, the call on the trunk with higher priority will be answered by the Operator extension first. To know more, read the feature description "Priority".

  By default, the Priority of all trunks is set to '9-Highest'. Decide what Priority Level you will assign to the trunks and set the desired level for the trunk.

- **SMDR-OG Storage:** This flag is used to enable or disable the storage of details of outgoing calls from the trunk. Please refer the topic "Station Message Detail Recording-Storage" for more details. By default, storage of outgoing calls is enabled.

- **SMDR-IC Storage:** This flag is used to enable or disable storage of details of incoming calls on the trunk. Please refer the topic "Station Message Detail Recording-Storage" to know more. By default, storage of incoming calls is enabled.

- **Hold on DSS Key Press:** This flag defines the 'Hold' state of the external called party, when an extension user presses a DSS key to dial another port.

  For example, the DKP extension user (on DKP-001 port) is in the middle of speech with an external party on a Trunk port CO-002.

  If extension user of DKP-001 presses a DSS key to call another extension port DKP-003, two situations are possible, depending on whether the Hold on DSS Key Press flag is enabled or disabled:

  - **When the Hold Flag is enabled:** CO-002 will be played music-on-hold. DKP-001 will hear Ring Back Tone and the call will be placed on DKP-003.

  - **When Hold Flag is disabled:** CO-002 will be disconnected. DKP-001 will hear Ring Back Tone, and call will be placed on DKP-003.

  By default the Hold On DSS Key Press flag is enabled.

- **Forced Account Code:** This parameter is related to the "Account Codes" feature of the SARVAM UCS. This flag must be enabled, if the feature Forced Account Code is to be applied on the trunks.

  When this flag is enabled, the system will prompt extension users to dial the Account Code whenever they grab a trunk to dial out a number. The system will allow extension users to dial out numbers only when after they have dialed the Account Code or Name.

  By default, the flag is disabled. Refer the feature description for "Account Codes" to know more.

Account Codes feature must also be enabled in the Class of Service of extension users who are to be allowed this feature.

- **Call Cost Calculation Pulse Rate Option:** This parameter is to be configured only if you want to apply the "Call Cost Calculation (CCC)" feature on the trunks on which the template is applied.

  You have four options for Pulse Rate Types. Select from Pulse Rate Type for Pulse Rate Option 1 to 4 which you want to apply on the trunks.

- **Call Cost Calculation Time Schedule:** This parameter is to be configured only if you want to apply the "Call Cost Calculation (CCC)" feature on the trunks on which the template is applied.

  The Pulse Rates offered by service providers may vary according to the time of the day. In such cases, you must first define the Time Zone (time of the day) for which a particular Pulse Rate should be applied and the Time Schedule for each Time Zone.

  You can configure up to four different Time Zones - T1, T2, T3 and T4 with different Pulse rates in the CCC-"Configuring Pulse Rate Types".

  Now, configure the Call Cost Calculation Time Schedule, by specifying the Start Time and the End time (in 24hours: minutes format) for each Time Zone.

The default Time Schedule (starts and end time) for each Time Zone Index are as follows:

Time Zone Index	Start Time	End Time
T1	00:00	23:59
T2	00:00	23:59
T3	00:00	23:59
T4	00:00	23:59

If your service provider offers the same Pulse Rate for the entire day,

- configure only one Time Zone Index with the Pulse Rate, for instance, T1, in the CCC-Normal Pulse Rate Table.
- Now, set the Time Schedule for Time Zone, T1, with the start and end time in Hours: Minutes format;
- set the start and end time of the other Time Zone Index, T2 to T4, to 00:00 (hours: minutes).

Similarly, if your service provider supports two different Pulse Rates in a day, set the Start and the End time for two Time Zones and set the other two to 00:00.

- **Call Duration Control:** This parameter is to be configured only if you want to apply the "Call Duration Control (CDC)" feature on the trunks on which the template is applied.

  By default the Call Duration Control - **For IC Calls** check box is selected (enabled). Clear the check box to disable.

  By default the Call Duration Control - **For OG Calls** check box is selected (enabled). Clear the check box to disable.

- **Call Taping:** This parameter is to be configured only if you want to apply the "Call Taping" feature on the trunks on which the template is applied.

  Select the Call Taping check box to enable.

- Click **Submit**.

Customizing Trunk Feature Templates

By default, Trunk Feature Template 01 is assigned to all Trunk Types: SIP, T1E1 PRI, BRI, Mobile, CO and E&M. The default values of the parameters of this template are sufficient to meet the common requirements of most users.

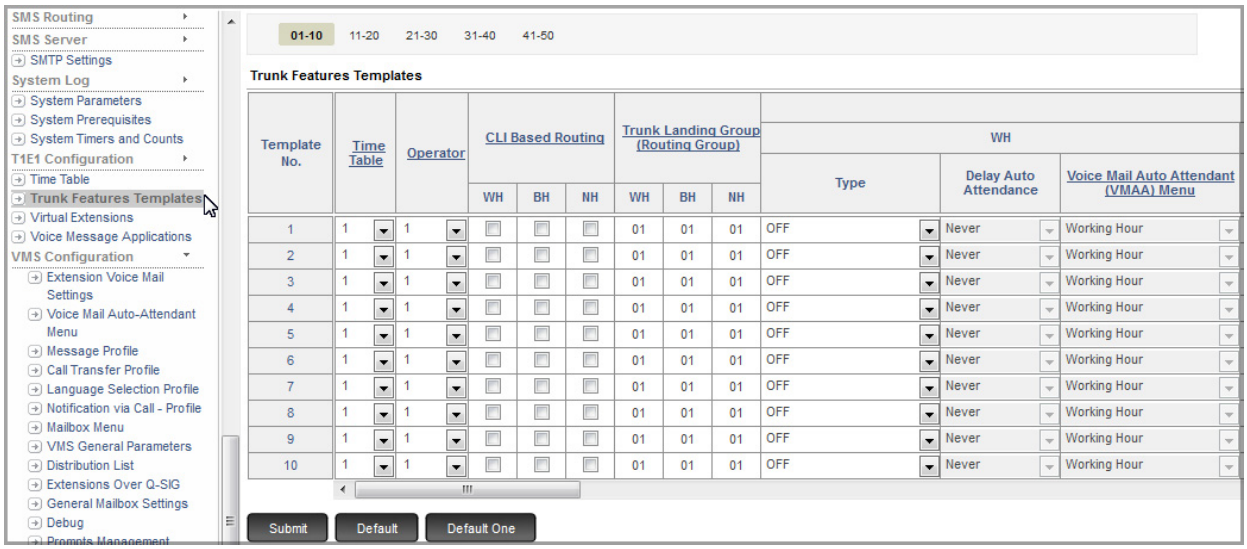
If the default values of the Template fulfill the requirements of all Trunk types, retain this template. If you want to change some of the feature settings and apply the template to all trunk types, you may simply customize this template.

However, if you want to assign different feature settings for different trunk types, you are recommended to prepare and apply separate Trunk Feature Templates for each Trunk type.

This can be done using Jeeves and from a telephone.

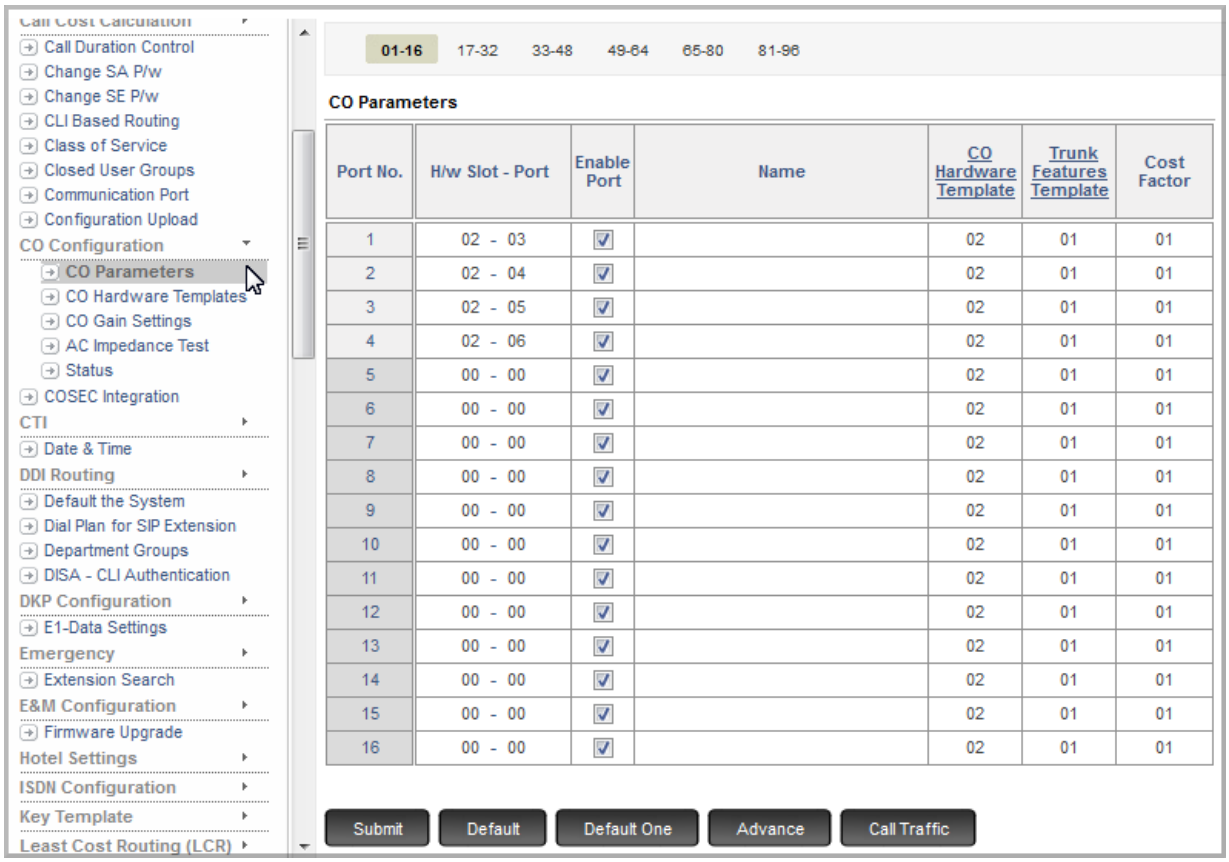
Customizing Trunk Feature Template using Jeeves

- Log in as System Engineer.
- Under **Configuration**, click **Trunk Feature Template**.



- Select a Trunk Feature Template Number, for instance 02.
- Customize the Template by changing the values of the desired parameters.
- Click **Submit** to save changes.
- Apply the Trunk Feature Template you customized to the CO Port type.
- To apply the template on CO Trunks,
  - Open **CO Parameters** page.
  - Go to the CO software ports to which this Template you customized is to be assigned, for instance CO-001 to 003.

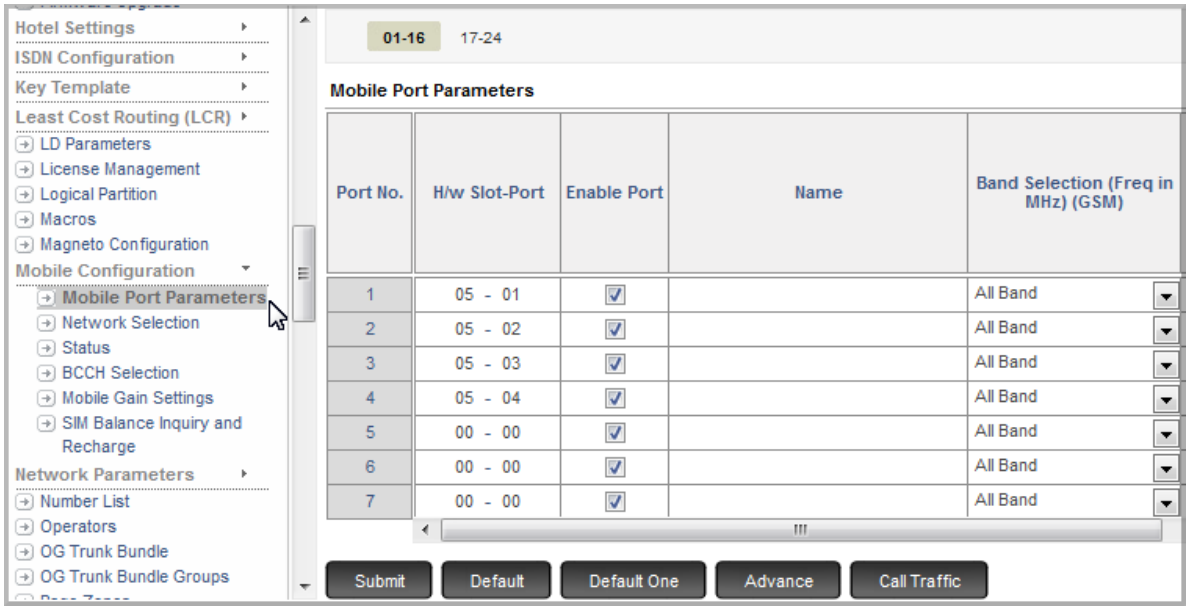
- Enter the number of the Trunk Feature Template you customized, 02, in the **Trunk Feature Template** field of this port.



- Click **Submit** to save changes.

- To apply the template on Mobile Ports,
  - Click **Mobile Configuration**.
  - Click **Mobile Port Parameters** to open the page.

- Enter the number of the template you customized in the field **Trunk Feature Template** of the Mobile Software ports to which you want to assign this template.



- Click **Submit** to save changes.

- Repeat the same steps to create another template and apply it on the desired Trunk Port type.

  To apply the template on E&M ports, open **E&M Parameters** page under E&M Configuration.

  To apply the template on SIP Trunks, open **SIP Parameters** page under VoIP Configuration.

  To apply the template to BRI Trunks, open **BRI Parameters** page under BRI Configuration.

  To apply the template on T1 lines, open **T1 Port Parameters** page under T1E1 Configuration.

  To apply the template on E1 lines, open **E1 Port Parameters** page under T1E1 Configuration.

- Remember to click **Submit** to save your settings on each page.

Customizing Trunk Feature Template using a Telephone

- Enter SE mode from a DKP/SLT.

  To change the default value of a Trunk Feature Parameter in a Template, dial:

  - ***5802-1-Trunk Feature Template Number-Feature Number-Code*** to change the value of a parameter in a single template
  - ***5802-2-Trunk Feature Template Number-Trunk Feature Template Number-Feature Number-Code*** to set the same value for the parameter in a range of templates.
  - ***5802-*-Feature Number-Code*** to set the same value for the parameter in all templates.
    Template Number is the number of the Trunk Feature Template from 01 to 50.
    Parameter Number is the number of the Trunk Feature Template Parameter from 01 to 40.
    Code is the value for each parameter from 0 to 9.
    Refer the following table for the parameter numbers and the values for the codes.

Trunk Feature Templates - Parameter Numbers and Default Values

Para. No	01	02	03	04	05	06	07	08	09	10	11	12
Template No.	Time Table	Operator	CLI Based Routing			Trunk Landing Group (Routing Group)			Auto Attendant			Auto Attendant Delayed Timer (sec)
			WH	BH	NH	WH	BH	NH	WH	BH	NH	
01	1	1	x	x	x	01	01	01	OFF	OFF	OFF	Never
02 to 50	Same As Template No. 01											
Parameters Values Code												
0	0	1				01 - 96	01 - 96	01 - 96	OFF	OFF	OFF	Never
1	1	2	√	√	√				Built-in Auto Attendant	Built-in Auto Attendant	Built-in Auto Attendant	After 10 sec
2	2	3							Voice Mail Auto Attendant	Voice Mail Auto Attendant	Voice Mail Auto Attendant	After 15 sec
3	3	4										After 20 sec
4	4											After 30 sec
5	5											After 40sec
6	6											After 50 sec
7	7											After 60 sec
8	8											
9												

Para. No	13	14	15	16	17	18	40	19	20	21
Template No.	DISA			Trunk Auto Answer			Delayed Trunk Auto Answer	Trunk Auto Answer Greeting Message		
	WH	BH	NH	WH	BH	NH		WH	BH	NH
01	Disable	Disable	Disable	Disabled	Disabled	Disabled	10	Greeting Msg. 1	Greeting Msg. 1	Greeting Msg. 1
02 to 50	emplate No. 01									
Parameters Values Code										
0	Disable	Disable	Disable	Disabled	Disabled	Disabled	01-99	Disabled	Disabled	Disabled
1	PIN Auth.- Multiple calls	PIN Auth.- Multiple calls	PIN Auth.- Multiple calls	For all Calls	For all Calls	For all Calls		Greeting Msg. 1	Greeting Msg. 1	Greeting Msg. 1
2	CLI Auth.- Multiple calls	CLI Auth.- Multiple calls	CLI Auth.- Multiple calls	When Busy	When Busy	When Busy		Greeting Msg. 2	Greeting Msg. 2	Greeting Msg. 2
3	CLI Auth.- One call- Ans. Sig.	CLI Auth.- One call- Ans. Sig.	CLI Auth.- One call- Ans. Sig.	Delayed	Delayed	Delayed		Greeting Msg. 3	Greeting Msg. 3	Greeting Msg. 3
4								Greeting Msg. 4	Greeting Msg. 4	Greeting Msg. 4
5										
6										
7										
8										
9										

Para. No	22	23	24	25	26	27	28	29	30	31	32
Template No.	Trunk Auto Answer RBT Message Type			Trunk Auto Answer Busy Bye Message			Priority	SMDR OG Storage	SMDR IC Storage	Hold on DSS Key Press	Force Account Code
	WH	BH	NH	WH	BH	NH					
01	None	None	None	None	None	None	9	√	√	√	X
02 to 50	Same As Template No. 01										
Parameters Values Code											
0	None	None	None	None	None	None	0	X	X	X	X
1	Internal MoH	Internal MoH	Internal MoH	Bye Msg. 1	Bye Msg. 1	Bye Msg. 1	1	√	√	√	√
2	External Music (AIP)	External Music (AIP)	External Music (AIP)	Bye Msg. 2	Bye Msg. 2	Bye Msg. 2	2				
3	RBT Msg. 1	RBT Msg. 1	RBT Msg. 1	Bye Msg. 3	Bye Msg. 3	Bye Msg. 3	3				
4	RBT Msg. 2	RBT Msg. 2	RBT Msg. 2	Bye Msg. 4	Bye Msg. 4	Bye Msg. 4	4				
5	RBT Msg. 3	RBT Msg. 3	RBT Msg. 3				5				
6	RBT Msg. 4	RBT Msg. 4	RBT Msg. 4				6				
7							7				
8							8				
9							9				

Para. No	34	35				36				37				38			
Template No.	Call Cost Calculation Pulse Rate Option	Call Cost Calculation Time Schedule															
		T1				T2				T3				T4			
		Start Time		End Time		Start Time		End Time		Start Time		End Time		Start Time		End Time	
		HH	MM	HH	MM	HH	MM	HH	MM	HH	MM	HH	MM	HH	MM	HH	MM
01	1	00	00	23	59	00	00	23	59	00	00	23	59	00	00	23	59
02 to 50	Same As Template No. 01																
Parameters Values Code																	
0		00-23	00-59	00-23	00-59	00-23	00-59	00-23	00-59	00-23	00-59	00-23	00-59	00-23	00-59	00-23	00-59
1	1																
2	2																
3	3																
4	4																
5																	
6																	
7																	
8																	
9																	

For example: Assign Trunk Landing Group 03 to Working Hours and Break Hours and Trunk Landing Group 04 to Non-Working Hours in Template 02.

To assign TLG 03 to Working Hours, dial:

- ***5802-1-02-03-03***
  Where,
  02 is the template number
  03 is the parameter number for TLG-Working Hours
  03 is the code for the TLG to be assigned to Working Hours.

To assign TLG 03 to Break Hours, dial:

- ***5802-1-02-04-03***
  Where,
  02 is the template number
  04 is the parameter number for TLG-Break Hours
  03 is the code for the TLG to be assigned to Break Hours.

To assign TLG 04 to Non-Working Hours, dial:

- ***5802-1-02-05-03***
  Where,
  02 is the template number
  05 is the parameter number for TLG-Non-Working Hours
  03 is the code for the TLG to be assigned to Non-Working Hours.

To default Trunk Feature Templates, dial:

- ***5801-1-Trunk Feature Template Number*** to default a single template.
- ***5801-2-Trunk Feature Template Number-Trunk Feature Template Number*** to default a range of templates.
- ***5801-**** to default all templates.
  Where,
  Trunk Feature Template is the template number from 01 to 50.

- To assign a Trunk Feature Template to a Trunk Port type:

  For CO trunks, dial:
  - ***5803-1-CO- Trunk Feature Template Number*** to assign a feature template to a single CO port.
  - ***5803-2-CO-CO- Trunk Feature Template Number*** to assign a feature template to a range of CO ports.
  - ***5803-*- Trunk Feature Template Number*** to assign a feature template to all CO ports.
    Where,
    CO is the Software Port number of the CO port from 001 to 128.
    Template Number is the number of the customized Trunk Feature Template, from 01 to 50. Default: Template 01.

  For BRI trunks, dial:
  - ***5804-1-BRI- Trunk Feature Template Number*** to assign a feature template to a single BRI port.
  - ***5804-2-BRI-BRI- Trunk Feature Template Number*** to assign a feature template to a range of BRI ports.
  - ***5804-*- Trunk Feature Template Number*** to assign a feature template to all BRI ports.
    Where,
    BRI is the Software Port number of the CO port from 01 to 32.
    Template Number is the number of the customized Trunk Feature Template, from 01 to 50. Default: Template 01.

  For E&M Trunks, dial:
  - ***5805-1-E&M- Trunk Feature Template Number*** to assign a template to a single E&M port.
  - ***5805-2-E&M-E&M- Trunk Feature Template Number*** to assign the same template to a range of E&M ports.
  - ***5805-*- Trunk Feature Template Number*** to assign the same template to all E&M ports.
    Where,
    E&M is the Software Port number of the port from 001 to 128.
    Template Number is the number of the customized Trunk Feature Template, from 01 to 50. Default: Trunk Feature Template 01.

  For T1E1 Trunks, dial:
  - ***5806-1-T1E1- Trunk Feature Template Number*** to assign a template to a single T1E1 port.
  - ***5806-2-T1E1- Trunk Feature Template Number*** to assign the same template to a range of T1E1 ports.
  - ***5806-*- Trunk Feature Template Number*** to assign the same template to all T1E1 ports.
    Where,
    T1E1is the Software Port number of the port from 01 to 08.
    Template Number is the number of the customized Trunk Feature Template, from 01 to 50. Default: Trunk Feature Template 01.

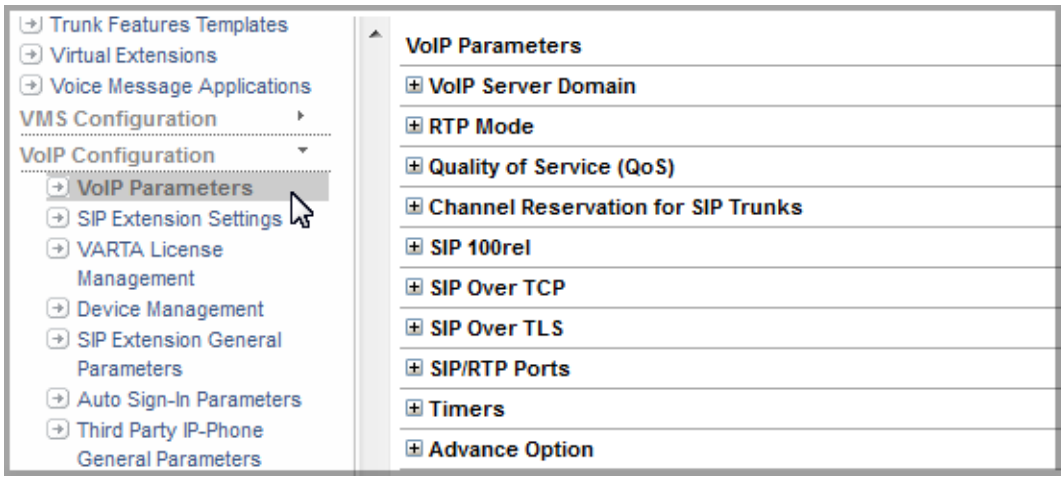
  For Mobile ports, dial:
  - ***5807-1-Mobile- Trunk Feature Template Number*** to assign a template to a single Mobile port.
  - ***5807-2-Mobile-Mobile- Trunk Feature Template Number*** to assign the same template to a range of Mobile Ports.
  - ***5807-*- Trunk Feature Template Number*** to assign the same template to all Mobile ports.
    Where,
    Mobile Port Number is the number of the software port from 001 to 064.
    Template Number is the number of the Trunk Feature Template, from 01 to 50.
    Default: Trunk Feature Template 01.

- Exit SE mode.

Configuring VoIP Parameters

Configuring VoIP Parameters using Jeeves

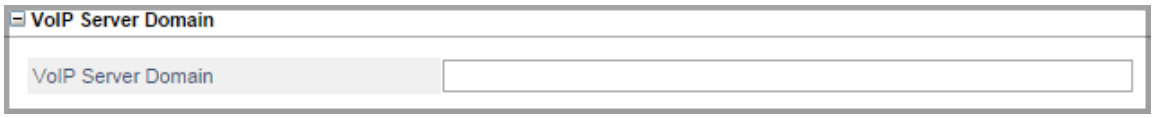
- Login as System Engineer.
- Under **Configuration**, click **VoIP Configuration**.
- Click **VoIP Parameters** to open the page.



Changing any parameter on this page will result in call drop of all ongoing VoIP calls. All Open SIP Extensions registered will start functioning only after the next registration.

VoIP Server Domain

Click **VoIP Server Domain** to expand.



- **VoIP Server Domain:** This parameter is of relevance only if you are configuring SIP Extensions.

  The system is capable of maintaining a domain for registering SIP clients (any SIP-enabled device) as SIP Extensions.

  Configure the Server Domain if you want SIP clients to register with the Registrar Server of the system using the domain handled by the system[189]. The Domain Name can be a maximum of 40 characters. Default: Blank.

189. *SIP clients can be registered with the system either using the domain handled by the CPU or using the WAN or LAN Port IP Address.*

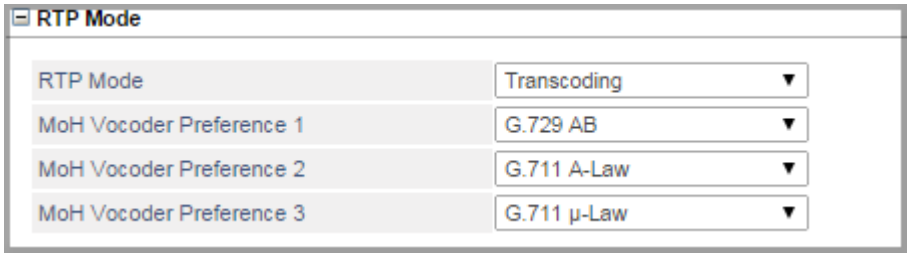
     If domain is programmed, the system will listen for the SIP message which is redirected to the programmed domain only. It will also listen for SIP messages on the WAN IP address and LAN IP address.

     But if domain is not programmed, the system will listen for SIP messages only on the WAN IP Address and LAN IP address.

If you configure Server Domain for registration of SIP clients, you must also map the Domain Name and the IP Address of the WAN Port to the DNS Server in the network.

RTP Mode

Click **RTP Mode** to expand.



- **RTP Mode:** Select the desired RTP mode using which you want SARVAM UCS to route speech in SIP to SIP calls.

  You can select from the following:

  - **Transcoding:** When this option is selected, RTP packets will be routed using Vocoder channels[190] for SIP to SIP calls. SIP users (Extended SIP Clients and Standard SIP Clients) will be able to access all the features of SARVAM UCS. This option uses two Vocoder channels for SIP to SIP calls. Thus the maximum number of SIP to SIP calls per system is equal to the number of Vocoder channels divide by 2.

  - **RTP Relay:** When this option is selected, RTP packets will be routed without using Vocoder channels for SIP to SIP calls. The system will use Vocoder channels to route SIP to TDM calls and vice versa. System will use Vocoder channels for some features and Standard SIP Clients will be able to use limited features of SARVAM UCS. Refer to “SARVAM UCS Features supported with RTP/Direct RTP” for more details.

  - **Direct RTP:** When this option is selected, no Vocoder channel will be used for SIP to SIP calls and RTP packets will be sent to and fro directly between SIP end points. If transfer of RTP packets is not possible between SIP end points, system will use RTP Relay as the fallback option. The system will use Vocoder channels to route SIP to TDM calls and vice versa. System will use Vocoder channels for some features and Standard SIP Clients will be able to use limited features of SARVAM UCS. Refer to “SARVAM UCS Features supported with RTP/Direct RTP” for more details.

- *Direct RTP is not supported on SIP Trunks and MATRIX VARTA ADR100/AMP100/WIN200, therefore the system will use RTP Relay.*

- *If RTP Relay or Direct RTP is selected,*
  - *Maximum 550 Audio calls[191] or 55 Video calls are supported in SARVAM UCS.*
  - *Certain voice messages will not be played. See “Voice Message Applications” for more details.*

- *If you change the RTP Mode, the system will release all ongoing calls.*

190. *The number of Vocoder channels that will be supported would be as per the license you purchase.*
191. *ETERNITY PENX supports 64 Direct RTP Audio Calls and 16 Video Calls.*

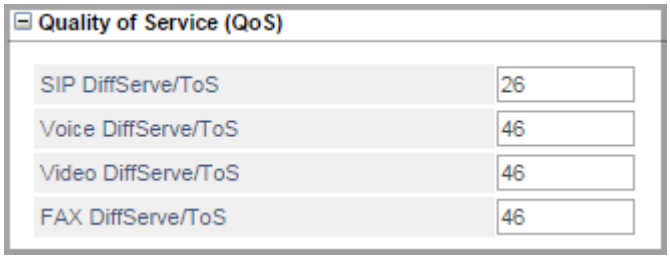
- **MoH Vocoder Preference:** If RTP Mode is set as RTP Relay or Direct RTP, the system will play the MoH stored in the Vocoder module when the system holds any SIP end point. The MoH is played as per the configured MoH Vocoder Preference.

  Select the Vocoders in the order of their preference, for **MoH Vocoder Preference 1**, **MoH Vocoder Preference 2** and **MoH Vocoder Preference 3**.

  If required, you can customize the MoH to be played as per your requirement. For detailed instructions, see “Uploading Custom MoH”.

Quality of Service (QoS)

Click **Quality of Service (QoS)** to expand.



- **Quality of Service (QoS):** QoS refers to priority of IP packets on network layer. It can be programmed for both signaling (SIP) and media (Voice, Video and Fax). Configure the following types QoS:

  - **SIP DiffServe/ToS:** The system sends all the SIP signaling messages with this QoS setting. This field defines the priority bits for SIP messages. The Valid *DiffServe* range is from 00-63, default: 26

  - **Voice DiffServe/ToS:** The system sends all the Voice packets with this QoS setting. This field defines the priority bits for Voice packet. The Valid *DiffServe* range is from 00-63, default: 46

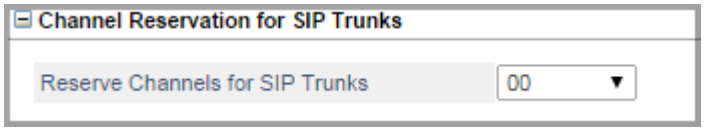
  - **Video DiffServe/ToS:** The system sends all the Video packets with this QoS setting. This field defines the priority bits for Video packet. The Valid *DiffServe* range is from 00-63, default: 46

  - **Fax DiffServe/ToS:** The system sends all the Fax packets with this QoS setting. This field defines the priority bits for Fax packet. The Valid *DiffServe* range is from 00-63, default: 46

QoS parameters are applicable for all packets (SIP/ Media) leaving both LAN and WAN port as well as TCP connection.

Channel Reservation for SIP Trunks

Click **Channel Reservation for SIP Trunks** to expand.



- **Channels Reserved for SIP Trunks:** The Vocoder module supports up to 128 voice channels, which can be used by SIP Extensions and SIP Trunks.

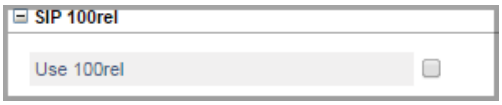
  It may happen that SIP Extension users use most of the channels of the Vocoder module, leaving too few or none for making/receiving SIP Trunk calls.

This can be avoided by reserving some voice channels exclusively for SIP Trunk calls.

Specify the minimum number of voice channels you want to reserve for SIP Trunk calls. By default, no channel is reserved.

SIP 100rel

Click **SIP 100rel** to expand.



- **SIP 100rel:** This parameter is to be configured if you want to support reliable transmission of (SIP) provisional responses. Enable 100rel by selecting the check box, if you want the Vocoder module to use 100rel for reliable transmission of SIP provisional responses and to use PRACK (Provisional Acknowledgment). Default: Disabled.

SIP Over TCP

Click **SIP Over TCP** to expand.

- **SIP Over TCP:** SARVAM UCS supports transporting of SIP messages over User Datagram Protocol (UDP), Transfer Control Protocol (TCP) as well as Transport Layer Security connection (TLS). Despite the advantages that SIP over TCP and SIP over TLS offer, it is more common to use UDP to transport SIP messages.

  By default, SIP Over TCP is enabled. To be able to send SIP messages over TCP, you must configure ‘TCP’ as the ‘Default Transport for Outgoing Messages’ on the ‘SIP Trunk Parameters’ page.

  If you do not want to transport SIP messages using TCP, clear the SIP Over TCP check box.

SIP Over TLS

Click **SIP Over TLS** to expand.

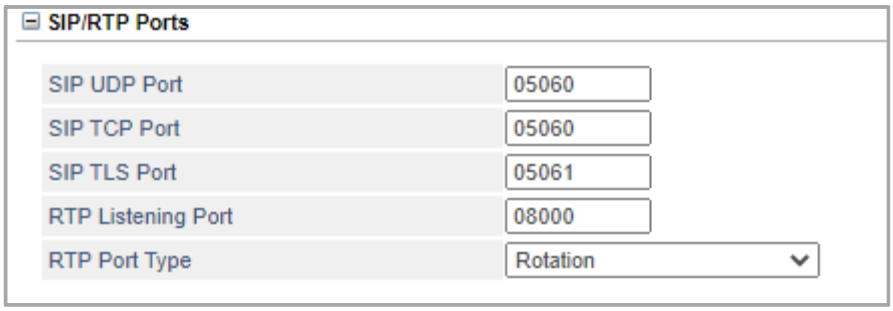
- **SIP Over TLS:** SARVAM UCS supports transporting of SIP messages over Transport Layer Security. TLS offers secure SIP signaling.

  By default, SIP Over TLS is enabled. To be able to send SIP messages over TLS, you must configure ‘TLS’ as the ‘Default Transport for Outgoing Messages’ on the ‘SIP Trunk Parameters’ page.

  If you do not want SIP messages to be transported using TLS, clear the SIP Over TLS check box.

SIP/RTP Ports

Click **SIP/RTP Ports** to expand.



- **SIP UDP Port:** This port defines the port on which SARVAM UCS listens for SIP messages transported over UDP. This port is also used as the source port for sending SIP messages to the remote peer. The valid range for this port is 1025-65535. Default: 05060.

- **SIP TCP Port:** This port defines the port on which SARVAM UCS listens for SIP messages transported over TCP. This port is also used as the source port for sending SIP messages to the remote peer. The valid range for this port is 1025-65535. The default SIP TCP Port is 05060.

- **SIP TLS Port:** This port defines the port on which SARVAM UCS listens for SIP messages transported over TLS. This port is also used as the source port for sending SIP messages to the remote peer. The valid range for this port is 1025-65535. The default SIP TLS Port is 05061.

- **RTP Listening Port:** This port defines the port on which SARVAM UCS listens for RTP Packets. This port is also used as the source port for sending RTP packets to the remote peer. The valid range for this port is 1025-56734. The default RTP Listening Port is 08000.

Valid Port Range for MENX/LENX: 1025 - 56734.
Valid Port Range for PENX: 1025 - 64510.

- **RTP Port Type:** Select the desired Port Type — Rotation, First Free — from the drop-down.The default RTP Port Type is Rotation.

  If you select Rotation, for each new session the system checks and allocates the next available RPT port in a sequential manner. Make sure all the RTP Ports are forwarded.

  If you select First Free, for each new session the system allocates the first available RTP Port regardless of its sequence.That is, the system will start checking from the starting RTP port for every new session and allocate the first free port available.In this case the audio and video ports need to forwarded separately. However, for a video call both audio and video ports will be consumed. For a video call maximum 400 ports can be forwarded.

  For LENX/MENX and GENX
  RTP Listening port = configured RTP Listening port
  Ending number of audio port = RTP Listening port + (number of audio calls * 4)
  Range of audio port = RTP Listening port to Ending number of audio port
  Starting number of video port = RTP Listening Port + 4400
  Ending number of video port = The total number of video port starting range + (number of video calls * 4)
  Range of Video port = Starting number of video port to Ending number of video port

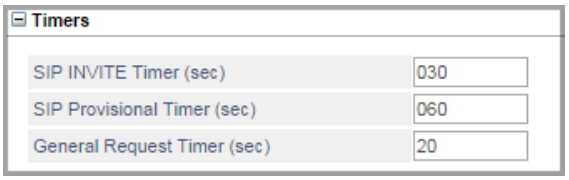
Let us understand this with the help of an example:
Number of audio calls to be made 100 and number of video calls to be made 50
RTP Listening port = 12000
Ending number of audio port = 12000 + (100*4) = 12400
Range of audio port = Range 12000 to 12400
Starting number of video port = 12000 + 4400 = 16400
Ending number of video port = 16400 + (50 * 4) = 16600
Range of Video port = 16400 to 16600

For PENX
RTP Listening port = configured RTP Listening port
Ending number of audio port = RTP Listening port + (number of audio calls * 4)
Range of audio port = RTP Listening port to Ending number of audio port
Starting number of video port = RTP Listening Port + 512
Ending number of video port = The total number of video port starting range + (number of video calls * 4)
Range of Video port = Starting number of video port to Ending number of video port

Let us understand this with the help of an example:
Number of audio calls to be made 50 and number of video calls to be made 8
RTP Listening port = 12000
Ending number of audio port = 12000 + (50*4) = 12200
Range of audio port = Range 12000 to 12200
Starting number of video port = 12000 + 512 = 12512
Ending number of video port = 12512 + (8 * 4) = 12544
Range of Video port = 12512 to 12544

Timers

Click **Timers** to expand.



- **SIP Invite Timer (sec):** This is the time in seconds for which the system waits for a response from the called party after ending INVITE message. This timer starts after sending INVITE message to the called party and stops on receipt of the provisional response or the final response or when the user disconnects the call. On expiry of the timer, the call process is terminated by the SARVAM UCS and an error tone is played to the user. The range of the SIP Invite Timer is 010-180 seconds. Default: 30 seconds.

- **SIP Provisional Timer (sec):** This is the time in seconds for which the system waits for the final response after receiving the provisional response from the called party. This timer starts on the receipt of the provisional response from the called party and stops on receipt of the final response from the called party or when the user disconnects the call. On expiry of the timer, the SARVAM UCS terminates the call process and plays error tone to the user. The range of SIP provisional Timer is 010-180. Default: 60 seconds.

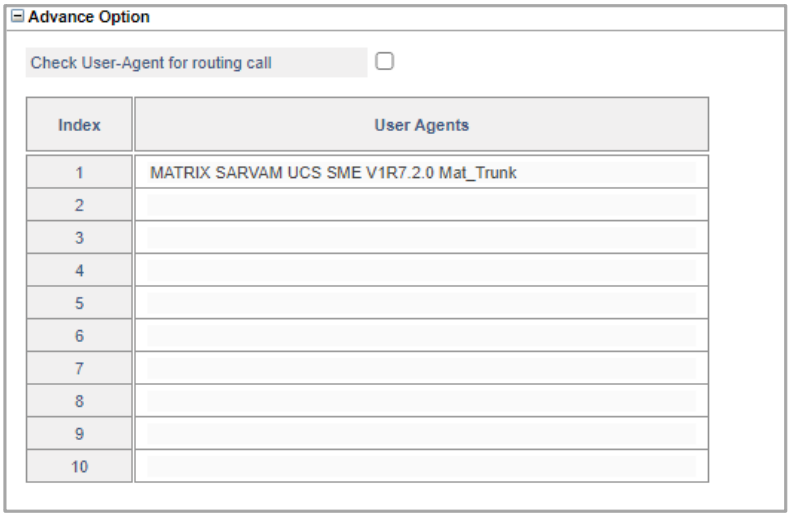
- **General Request Timer (sec):** The time in seconds for which the system waits for response for a transaction request. This timer starts on the initiation of a transaction. This timer stops on receipt of a

response for the request. On expiry of the timer, the system clears the transaction. This timer is used for Registration request, etc. The range of the General Request Timer is 10-60 seconds. Default: 20 seconds.

The Timers will be applicable only after System Restart.

Advance Option

Click **Advance Option** to expand.



- **Check User-Agent for routing call:** Default: Disabled. Select the check box to enable. If enabled SARVAM UCS will check if the call is originated from a trunk or an extension.

  In the User Agent table, under **User Agent**s, configure the User Agent Strings of the other PBXs from which you want incoming call to land on the extensions. In this table the first index is reserved for Matrix Systems and is not editable. You can configure the User Agent Strings of the other PBXs from index 2 to 10.

  Now, let us consider a scenario:
  - We have two PBX's, either Matrix at both the ends or at one end Matrix and at the other any Third party PBX. Let us consider PBX A (Matrix PBX) and PBX B (Third Party PBX).
  - Both the PBX's have the same extension numbers configured, for example 2001.
  - You have configured a P2P trunk between the two PBX's.

  Now, when 2001 of PBX A receives an incoming call from 2001 of PBX B through the P2P trunk, then the system checks if **Check User-Agent for routing call** is enabled.

  If enabled, then the system will check the User Agent in the INVITE request and compares the same with the entries in User Agent Table. Only is an exact match is found the system will place the call, else it will reject it.

Configuring SIP Trunks

SIP trunks may be Proxy or Non-Proxy. All SIP trunks are considered as Proxy trunks by default.

Regardless of whether a SIP Trunk is Proxy or Non-Proxy, it must be assigned to the LAN/WAN Port, which is to be used for that SIP trunk and the SIP trunk must be enabled. VoIP Calls can be initiated after suitable configuration of the SIP trunk number in the Outgoing Trunk Bundle Group.

For Proxy SIP Trunks, you must program the following parameters required for registration with the Proxy Server.

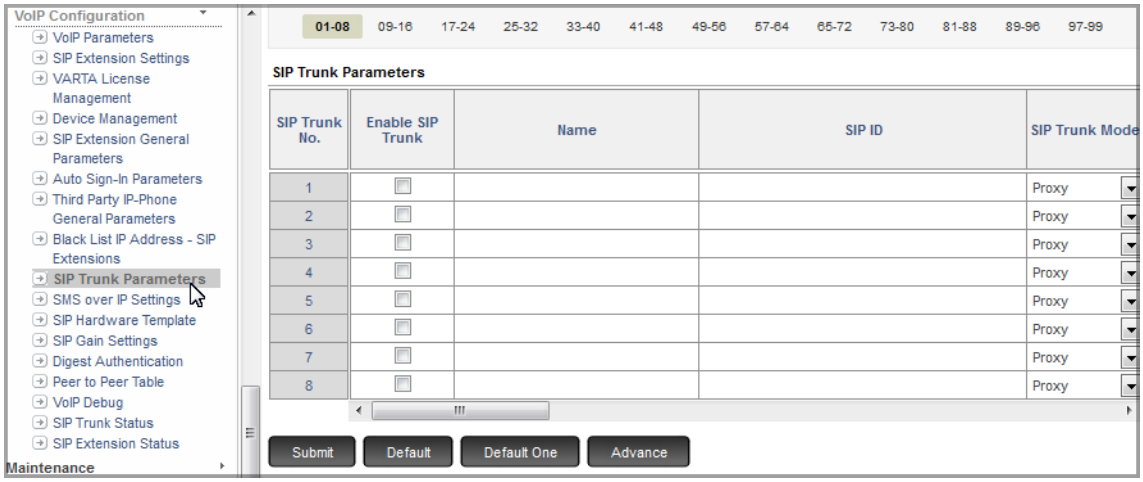
- Enable the SIP trunk.
- Program the SIP ID, registrar Server Address, Registrar Server Port, Authentication ID, Authentication Password as provided by your ITSP.
- If your ITSP uses Outbound Proxy, Enable the Outbound Proxy for the SIP trunk and also program the Outbound Proxy Server Address and Port as provided by your ITSP.

The maximum number of SIP trunks supported by SARVAM UCS is 99.

If you have installed a single VoIP module, you can configure all SIP trunks on the same module.

Configuring SIP Trunks using Jeeves

- Log in as System Engineer.
- Under **Configuration**, click **VoIP Configuration**.
- Click **SIP Trunk Parameters** to open the page.



Configure the following SIP Trunk parameters for the SIP trunks:

- **SIP Trunk No.:** This non-editable field is the number of the SIP trunk. The SIP trunks are numbered from 01 to 99.
- **Enable SIP Trunk:** This flag is for enabling or disabling the SIP trunk. To be able to make incoming and outgoing calls from the SIP trunk, click to enable the SIP trunk. By default, the SIP trunk is disabled, disallowing incoming and outgoing calls. You may disable SIP trunks that are not functioning.

- **Name:** You may assign a 'Name' to each SIP trunk to facilitate identification. Whenever there is an incoming call without CLI on this port, the Name you have programmed will be displayed on the landing extension, provided it is a DKP.

  The Name assigned to the SIP trunk may consist of a maximum of 18 characters. The Name of the port may be the name of the ITSP the SIP trunk is subscribed with (recommended).

- **SIP ID:** Enter the SIP User ID provided by the ITSP. SIP User ID is the ID that callers will use to call this SIP trunk.

  The SIP User ID may be a number or text for remote parties to call on the SIP trunk. For example, if SIP URI provided by the ITSP is 12345@abc.com, enter 12345 in this field. SIP User ID may consist of a maximum of 40 characters. All ASCII characters are allowed.

- **SIP Trunk Mode**: You may select SIP Trunk Mode as **Proxy** or **Peer-to-Peer**, according to your requirement.

  If you are using the services of an Internet Telephony Service Provider (ITSP), select **Proxy** to register this SIP trunk with the ITSP.

  If you are not using this service, select **Peer-to-Peer.**

- During Incoming Call Routing to select a SIP trunk, SARVAM UCS compares the SIP ID received in the Request URI of the INVITE message with the SIP ID configured on the SIP Trunk.

  If you do not want SARVAM UCS to check the SIP ID received in the Request URI of the INVITE message, clear the **Check SIP ID During Incoming Call** check box. Default: Enabled.

- **Treat Incoming calls as**: If you select Peer-to-Peer as the SIP Trunk mode, you may configure the trunk to **Treat Incoming call as: Trunk or Station.**

  If you select **Trunk**, the incoming calls will be routed as per the **Trunk Feature Template** assigned to the SIP Trunk.

  If you select **Station**, the system will route the incoming call as follows:

  - When only a number is received in the “To:” field of the INVITE message, SARVAM UCS will check the number in the Closed User Group Table. If a match is found in the CUG table the call will be routed as per the corresponding Outgoing Trunk Bundle Group.

  - If the CUG Table is not configured or if no match is found for the number received in the “To:” field of the INVITE message, the system will check if there is an extension number that matches with the number received in the “To:” field of the INVITE message. If a match is found the call is routed to the desired extension number.

  - When a Trunk Access Code and a number is received in the “To:” field of the INVITE message, the system will route the call as per the Outgoing Trunk Bundle Group assigned in the Station Basic Feature Template of the SIP Trunk.

  By default, Trunk is selected.

- *If Station is selected as the option for Treat Incoming call as, the user will only be able to:*
  - *Dial Flexible Numbers*

- *Dial Operator Code*
- *Dial Trunk Access Code for making outgoing calls*
- *Access the Global Directory*
- *Make calls within the Closed User Group*

- **Registrar Server Address:** Enter the Proxy/Registrar Server Address. Both IPv4 and IPv6 addresses are supported. The Server Address may be an IP Address or a Domain name, of maximum 40 characters

- **Registrar Server Port:** Enter the Registrar Server Listening Port. The valid rang is from 1025 to 65535. By default, 5060 is set as the Listening Port of the Registrar Server.

- **Re-registration Time (sec):** The Registrar Server deletes an entry of its client from its database on expiry of a fixed timer, which is set by the Registrar Server. SARVAM UCS sends a registration request before this Timer expires to remain registered on the server.

  Enter the value of the Timer after which the system should send registration request to maintain registration binding with the server. The valid range of this timer is from 00001- 65535. By default the Timer is set to 3600 seconds.

- **Registration Retry Time (sec):** This Timer stands for the period between retries for registration. If the registration attempt fails, SARVAM UCS sends the registration request on the expiry of this Timer again. The system continues to send the registration request till it gets registered. The valid range of this timer is from 00001- 65535. By default the Timer is set to 00010 seconds.

- **Authentication User ID:** Enter the Authentication ID provided by the ITSP for this SIP trunk. The Authentication User ID may be a string of 40 characters (maximum), including ASCII characters.

- **Authentication Password:** Enter the Authentication Password provided by the ITSP for this SIP trunk. The Authentication Password may be a string of 128 characters (maximum), including ASCII characters.

- **Outbound Proxy:** This parameter is relevant only if the ITSP has a SIP outbound server to handle voice calls. If yes, program the following parameters:

  - **Enable:** Select this check box to enable Outbound Proxy. By default the flag is disabled.

  - **Server Address:** Enter the Outbound Proxy Server's address. Both IPv4 and IPv6 addresses are supported. It may be an IP Address or Domain name. A maximum of 48 characters, including ASCII characters are allowed.

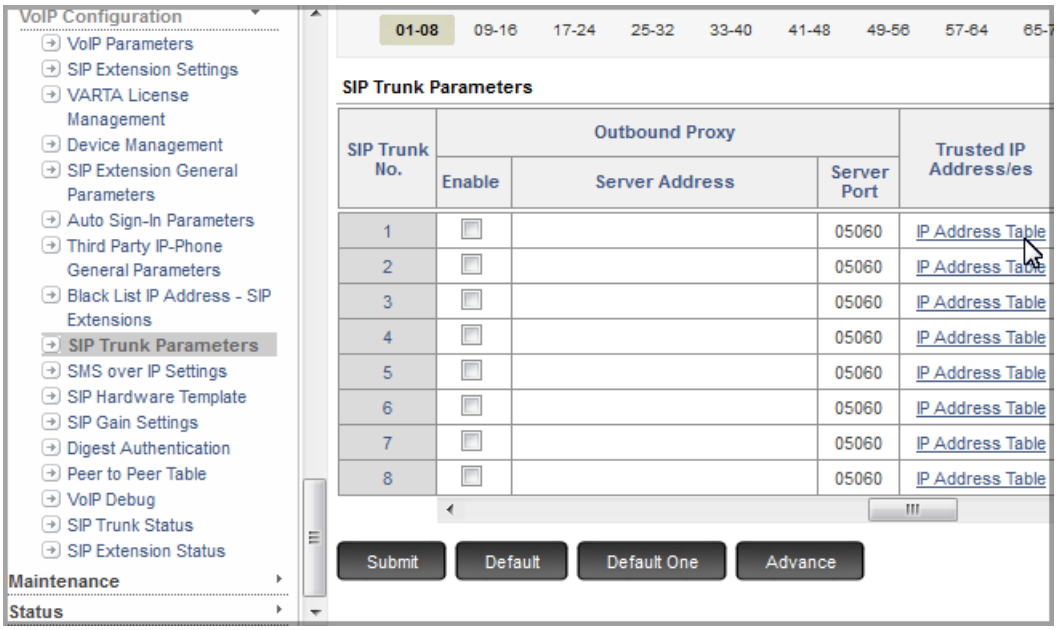
  - **Server Port:** Enter the Outbound Proxy Server's Listening Port. The valid range for this is 1025-65535. By default the Server Port is 5060.

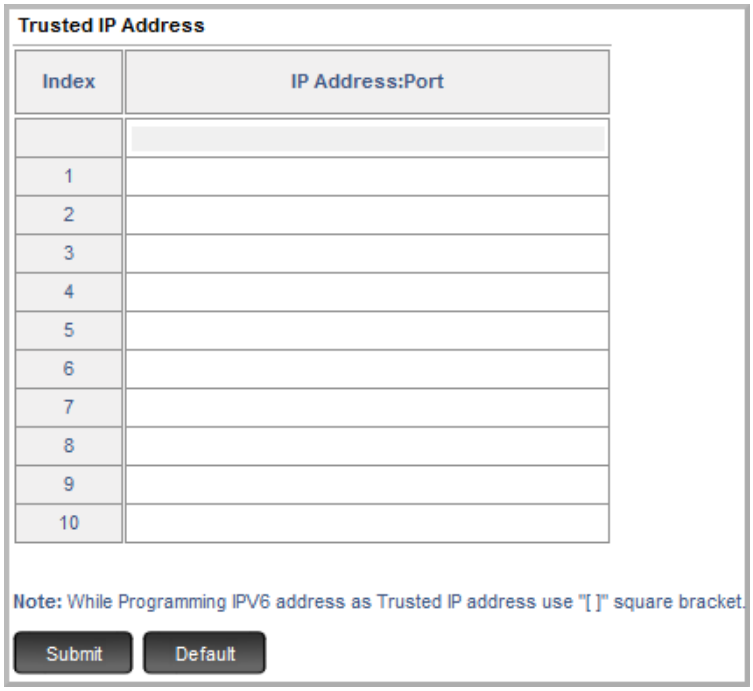
- **Trusted IP Address/es:** You must configure IP Address table to allow incoming calls from specific IP addresses on this SIP Trunk. Both IPv4 and IPv6 addresses are supported.
  If you select *Peer to Peer* as the SIP Trunk Mode and you do not configure this table, incoming calls on this SIP Trunk will be rejected.

  If you select *Proxy* as the SIP Trunk Mode and you do not configure this table, incoming calls will be allowed only from the Registrar Server Address or Outbound Proxy Address. All other calls on this SIP Trunk will be rejected.

To configure the IP Address table,

- Click **IP Address Table**.





If you have selected SIP Trunk mode as Peer to Peer, configure the following.

- **Allow from all IP Addresses:** Enable this check box, if you want to allow incoming calls on this SIP Trunk from all IP Addresses. Default: Disabled.

- **Apply Digest Authentication:** Enable this check box, if you want to allow incoming calls from callers only after the callers have authenticated themselves (with their User ID and Password). If the caller does not enter valid credentials in two attempts, the system will reject the call.

If you have enabled *Allow from all IP Addresses* check box, Apply Digest Authentication check box will be enabled automatically. You must configure the Digest Authentication table.

To configure the table, click the **Apply Digest Authentication** link. For detailed instructions, see "Digest Authentication".

- **Consider Peer to Peer Table for Trusted IP Address:** Enable this check box, if you also want to allow incoming calls from the domain names or IP Addresses configured in the Peer to Peer table. Default: Enabled. To know more about Peer to Peer table, see "Peer-to-Peer Calling".

If you have selected SIP Trunk mode as Proxy or Peer to Peer, configure the Trusted IP Address table.

The first entry in the table will display the *Proxy/Registrar Server Address:Port* or *Outbound Proxy Address: Port* as configured for this SIP Trunk (applicable only for Proxy Mode). For the Index numbers 1 to 10,

- Enter the **IP Address** and the corresponding **Port** from which you want to allow incoming calls.

  Do not configure the port, if you want to allow incoming calls from all the ports for a particular IP Address.

- Click **Submit** to save your entries and close the window.

- **SIP Hardware Template:** Assign a "SIP Hardware Template" to the SIP Trunk. The SIP Hardware Template contains voice quality related features such as Voice Codec selection, Tx and Rx Gains, Echo Cancellation, Jitter Buffer and, Fax-over-IP options and related parameters.

  There are 32 different templates to choose from. Each template can also be altered to suit your requirement and preferences. By default, Template number 01 assigned to all SIP Trunks. Template number 01 is also assigned to all SIP Extensions.

  First, check if the values in Template 01 fulfill the feature requirements of the SIP Trunks. Retain this template, if it fulfills the feature requirements of all SIP Trunks and if the same features are to be allowed to all SIP Trunks.

  If different sets of SIP hardware features are to be allowed to different SIP Trunks, then prepare separate SIP Hardware Templates and apply them on the SIP Trunks. To do this,

  - Under 'VoIP Configuration', click the 'SIP Hardware Template' link.
  - Select a Template number, for example 03.
  - Customize Template number 03 and click 'Submit'.
  - Now go back to the 'SIP Trunks' page.
  - Enter the number of the Template you customized, Template 03 in the 'SIP Hardware Template' field of the SIP Trunk on which you want to apply this template.
  - Click 'Submit' at the bottom of the page to save changes.

  - Repeat the same steps to customize and assign a different SIP Hardware Template to another SIP Trunk.

  - Also, refer the topic "SIP Hardware Template" to know more about customizing the templates and applying on the SIP Trunks.

- **Trunk Feature Template:** A Trunk Feature Template is a set of features like Time Table, Operator, Auto Attendant, DISA, Trunk Auto Answer, Trunk Landing Group, SMDR Storage, etc., that defines the behavior of a Trunk. Apply a Trunk Feature Template to the SIP trunk. By default, Trunk Feature Template 01 is applied on all SIP trunks as well as all other trunk types. Refer the topic "Trunk Feature Template" to know more.

  Click the 'Trunk Feature Template' link to open the page. Check if the default Template 01 fulfills your requirement for the SIP trunk.

  If the default Template 01 does not fulfill your requirement, prepare another Trunk Feature Template[192], and enter the newly prepared Template number for the SIP trunk.

- **Cost Factor:** This parameter is of relevance only if 'Least Cost Routing' feature is applied on the SIP trunk.

  Cost Factor is a number assigned to each trunk for identification. This number also serves as a preference number for the trunk. The Cost Factor can be from 1 to 99. Trunks having the same preference must be assigned the same Cost Factor. Different trunk types can also be assigned the same Cost Factor. These trunks are used for routing calls.

  Assign a Cost Factor to the SIP trunk, for instance, 02 and program Least Cost Routing Table accordingly.

  For example, if you want to route all outgoing calls starting with number '9' through the SIP trunk 01 only,

  - You must first assign a Cost Factor (01-99) to SIP trunk 01, for example, 02.
  - Click the 'Least Cost Routing - Number Based' link to open the page.
  - Enter '9' in the 'Number' column, Cost Factor '02' as Preference 1, 2, 3 and 4.
  - Click 'Submit' to save your settings.

    All outgoing calls assigned Cost Factor trunk 02 will be made from SIP trunk 01.

- **Simultaneous Calls**[193]**:** This parameter is for defining the number of simultaneous calls to be allowed on the SIP trunk. Default:32.

  The number of simultaneous SIP calls depends on the number of simultaneous calls allowed by the ITSP with whom you have subscribed this SIP Trunk.

  If the ITSP supports less than 32 simultaneous calls on SIP Trunks, you must program this parameter accordingly.

- Click **Submit** to save your configuration settings.

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192. The default template is applied on the ports of all trunk types supported by SARVAM UCS. Changes to the default template will be applied on all trunk types also. So, you are advised to prepare a new template and apply it to the desired trunk types.

193. This parameter is only applicable if you have set the RTP Mode as Transcoding in "Configuring VoIP Parameters". The number of simultaneous calls supported will depend on the number of Vocoder channels available. The channels supported depends on the license purchased. For details, refer to Vocoder Channels in "License Management".

Advanced Configuration Parameters

The above listed parameters fulfill the basic SIP trunk configuration requirements of most users. However, it is anticipated that some users may desire other less commonly used parameters such as RCOC, Accept Anonymous Calls on SIP trunks, Call Budget on SIP trunks, etc.

For such users, you may click the **Advance** button and configure the following parameters:

- **Return Call to Original Caller (RCOC):** Enable this flag if you want to apply the 'Return Call to Original Caller' on this SIP trunk.

  If this feature is enabled on the SIP trunk, the system will route calls returned by remote parties back to the extensions that originally made the call from this Trunk (the original callers' extensions). To know more, refer the feature description for "RCOC (Return Call to Original Caller)".

- **Station Basic Feature Template:** Assign a "Station Basic Feature Template" to the SIP trunk. There are 50 different templates to choose from. Each template can also be altered to suit your requirement and preferences. Default: Template number 01 assigned to all SIP trunks.

- **Station Advanced Feature Template:** Assign a "Station Advanced Feature Template" to the SIP trunk. There are 50 different templates to choose from. Each template can also be altered to suit your requirement and preferences. Default: Template number 01 assigned to all SIP trunks.

- **Add 'rinstance' in Register:** 'rinstance' is any random value which can be used by the WAN Port to fetch its own contact binding, that is, to know the Registration Expiry Timer assigned by the server. By default, this flag is enabled.

- **Send REGISTER Message:** With this parameter you can select whether or not the system should send REGISTER message from the SIP trunk. By default, this flag is enabled allowing REGISTER message to be sent from the SIP trunk.

- **Allow OG Calls without Registration:** This parameter is to be enabled to allow outgoing calls to be made from the SIP trunk, even when the SIP trunk is not registered. If this flag is disabled the system will not

allow outgoing calls to be made if the status of the SIP trunk is 'not registered'. By default, this flag is disabled.

- **Send OPTIONS as Heartbeat:** With this parameter you can select whether or not the system should send the OPTIONS message periodically to the Proxy Server to monitor its availability. Calls can be made and received only if the Proxy Server is alive.

  If the Proxy Server is unavailable, like no response is received, the status of the SIP Trunk will display “Heartbeat Failed” along with the Reason for Failure.

*To view status of the Proxy Server, go to **SIP Trunk Status**.*

If you enable **Send OPTIONS as Heartbeat**, you must configure the Heartbeat Interval.

- **Heartbeat Interval**: Define the **Heartbeat Interval (Seconds)**, the time period, from 10 to 999 seconds, after which SARVAM UCS should send the OPTIONS message to the Proxy Server. Default: 30 seconds.

- **DNS Record Type:** Set this parameter as provided to you by your Service Provider. You may select any of the following options:

  - **A/AAAA Record:** Select this option, if you want SARVAM UCS to send the query to the DNS Server to fetch the IP Address of the Target Server on which further SIP messages are to be sent.

    A Record maps to an IPv4 Address of the Target Server, while AAAA (quad-A) Record maps to an IPv6 Address.

  - **SRV Record:** Select this option, if you want SARVAM UCS to send the SRV query to the DNS Server to fetch the Destination Port. The system will also make a DNS A query to fetch the IP Address of the Target Server on which further SIP messages are to be sent.

  - **NAPTR/SRV Record:** Select this option, if you want SARVAM UCS to send queries to the DNS Server to fetch the Transport, Port and IP Address of the Target Server on which further SIP messages are to be sent.

    By default, A Record is set as the DNS Record Type.

This parameter will be checked when:
- *SIP Trunk Mode is configured as Proxy.*
- *Registrar Server Address is configured as Domain and Outbound Proxy Address is blank **OR** Registrar Server Address is configured as Domain/IP Address and Outbound Proxy Address is configured as Domain.*

- **Send CLI in FROM field:** This parameter allows you to configure the CLI of the SIP Trunk to be sent to the remote party on outgoing calls made using the SIP trunk. You may select any of the following options as desired:

  - **CLIR:** Select this option if you do not want the CLI to be sent.

  - **SIP ID:** You may select this option if you want the SIP ID programmed on the SIP Trunk to be sent as CLI.

- **Calling Party Wise:** Select this option if you want to send the Calling Extension Number (the number of the extension making the outgoing call through the SIP trunk) as CLI.

  When reverse DDI is programmed on the SIP Trunk, the DDI number of the calling extension will be sent, instead of its extension number.

  If the calling extension has disabled the parameter 'Send DDI as CLI' in its Station Advanced Feature Template, then its Pilot number configured in the Outgoing Reference Table will be sent as CLI.
  If calling extension has enabled CLIR, no CLI will be sent on the SIP Trunk.

- **Fixed Number:** Select this option if you want a specific number to be sent as CLI. When you select this option, you must also define the number to be sent as CLI.

  You may select this option if you wish to send any of your trunk line numbers as CLI on the SIP Trunk so as to enable the called party to call back the calling party using this CLI.

  Since it is not possible to call back a SIP ID, Fixed Number offers you a solution, using which you can send a trunk line number as CLI on the SIP Trunk. Using this CLI, the called party can call back the calling party.

  If you select this option, you must configure the **Fixed Number**. The Fixed Number may consist of a maximum of 40 characters. Valid characters are from 0-9 and plus (+).

  By default, SIP ID is set as the Send CLI in FROM field for all SIP Trunks.

When extension number of the calling extension is blank, and the 'Send CLI in FROM field' programmed for the SIP Trunk is other than "SIP ID", then also SIP ID will be sent as CLI.

When Emergency numbers are dialed using the SIP Trunk and even if CLIR is set as the 'Send CLI in FROM field' option, the system will send the number of the caller as CLI.

- **Send Called Party Number In:** SARVAM UCS provides you the option to Send Called Party Number in "To" or "Request URI" field. You may select — "To, Request URI", "To", "Request URI". Default: To, Request URI.

  - If you select *To*, then the called party number will be sent in "To" field, whereas SIP ID configured on the trunk will be sent in the "Request URI" field.

  - If you select *Request URI*, then the called party number will be sent in the "Request URI" field, while SIP ID configured on the trunk will be sent in the "To" field.

  - If you select *To, Request URI*, then the called party number will be sent in both the fields.

    If the SIP ID is not configured and you select the option — *To* or *Request URI*, then the called party number will be sent in both the fields.

    If the called party number is not available in any of the above cases then the remote server address will be sent in the selected field.

- **Allow Incoming CLI Modification:** To apply Incoming CLI Modification on the SIP trunk, select the **Allow Incoming CLI Modification** check box. By default, it is disabled.

Incoming CLI Modification is useful in countries where the Calling Line Identification (CLI) received by the System extension users must be suitably modified before it can be used to dial out the number. To know more, see “Incoming CLI Modification”.

If you receive CLI in dialable format, there is no need to use this feature. In such case, keep the flag disabled.

*For an incoming call on the SIP trunk, the Incoming CLI Modification will be applied only when both — the **Allow Incoming CLI Modification** check box and the **Enable Incoming CLI Modification** check box in System Parameters — are enabled.*

- **Use “tel” URI type in:** If your ITSP requires “tel” in the URI so that it can handle/ route calls to/from global numbers, select the desired option from any of the following:
  - TO, Request URI
  - FROM
  - TO, Request URI and FROM

  The system will send 'tel' URI in the selection headers and Request URI according to the selection you make. This will be used while making an Outgoing call.

  By default **None** is selected.

- **Send "user=phone":** Select this check box, if you want SARVAM UCS to add user=phone in the Request URI/FROM/TO header of the INVITE message. Default: Disabled.

  SARVAM UCS will send user=phone, only if the SIP ID is numeric.

- **P-Asserted-Identity/P-Preferred-Identity for Outgoing calls:** Configure the following **P-Asserted-Identity/P-Preferred-Identity for Outgoing calls** parameters as provided to you by your Service Provider.

  - Select **Add P-Asserted-Identity/P-Preferred-Identity** check box, if supported by your Service Provider.
  - Select **P-Preferred-Identity** or **P-Asserted-Identity** as **Identity**.

  - Select the desired **Caller ID** option:
    - **SIP ID:** If you want the SIP ID configured on the SIP Trunk to be displayed as the Caller ID.

    - **Calling Party Wise:** If you want the Calling Extension Number (the number of the extension making the outgoing call through the SIP trunk) to be displayed as the Caller ID.

    - **Fixed Number:** If you want a specific number to be sent as the Caller ID. When you select this option, you must also define the number to be sent as the Caller ID.

      The Fixed Number may consist of a maximum of 40 characters. Valid characters are from 0-9 and plus (+) .

  By default P-Asserted-Identity/P-Preferred-Identity for Outgoing calls is disabled.

- **P-Asserted-Identity/P-Preferred-Identity for Incoming calls:** Configure the following **P-Asserted-Identity/P-Preferred-Identity for Incoming calls** parameters as provided to you by your Service Provider.

- Select **Get CLI from P-Asserted-Identity/P-Preferred-Identity** check box, if supported by your Service Provider.
- Select **P-Preferred-Identity** or **P-Asserted-Identity** as the **Preference**.
- By default **Handle Privacy Header** is enabled. If the system receives Privacy Header as:
  - **Privacy: ID**, anonymous will be displayed as the CLI to the called extension.
  - **Privacy: None**, the CLI received from the Service Provider will be displayed to the called extension.

    If No Privacy header is received the CLI received from the Service Provider will be displayed to the called extension.
  - If you do not want the system to handle the Privacy Header, clear the check box.

By default P-Asserted-Identity/P-Preferred-Identity for Incoming calls is disabled.

- **Redirection:** When an incoming call is diverted/forwarded to an external number using the same SIP Trunk, configure the following parameters as required by your ITSP:
  - Select the desired **Call Redirection Type**. You may select **Generate new Call** or **Send 302**. Default: Generate new Call.
  - If you want the system to send 181 response to the source upon redirection, enable the **Sending of 181 responses upon Redirection** check box. Default: Disabled.
  - If you have selected **Generate new Call** as the **Call Redirection Type**, you may select the **Include "Diversion" while diverting the call** check box. The system includes incoming call request information in Diversion header, along with the reason of redirection. Default: Disabled.
  - If you have selected **Generate new Call** as the **Call Redirection Type**, you may select the **Include "History-Info" while diverting the call** check box. The system includes the incoming call request information as well as the new call information in the INVITE/REINVITE requests generated on the target. Default: Disabled.
  - If you have selected **Generate new Call** as the **Call Redirection Type**, select the desired option in **CLI while Diversion/Redirection**. You may select **Self** or **Received**. Default: Self.

    If you select **Self**, the system will send SIP Trunks' Identity and if you select **Received** the system will send the received CLI to the target.
- **Call Transfer Type:** Select the **Call Transfer Type as** supported by your ITSP. System will send the Call Transfer request to the transferee only if calls are routed through the same SIP Trunk. You may select:
  - **System:** In this option, the SARVAM UCS will handle the Call Transfer locally.
  - **Network:** In this option, the Call Transfer is handled by the Network.

  Default: System.

- **Accept Anonymous Calls?:** The option is for accepting calls without CLI that land on the SIP trunk. By default this option is disabled, that is, calls without CLI will not be allowed on this SIP trunk port. Select the check box to enable.

- **Source Port IP Address:** Select the Source Port IP Address for the SIP trunk. You may select from any of the following options, as applicable to your installation scenario.

  - **Use WAN Port IP Address:** Select this option, if the WAN Port is connected directly to the public Internet.

  - **Use LAN Port IP Address:** Select this option, if the LAN Port is connected within the LAN network.

  - **Use IP Address fetched using STUN:** Select this option, if the WAN Port is located behind a NAT router other than Symmetric.

  - **Use Router's Public IP Address:** Select this option if the WAN Port is located behind a NAT router (any type).

- **Handle rport:** Select the desired option from:

  - **Force NAT:** By default, this option is selected. The system will not check Contact/Via header etc. while sending SIP messages and will follow Symmetric Signaling.

  - **RFC 3581:** Select this option, if you want the system to follow Standard RFC's while sending SIP messages.

  Default: Force NAT

- **Use Symmetric RTP?:** The use of Symmetric RTP makes it possible for a SIP device to send the RTP on the same connection on which it is listening for RTP. This is done only on peer to peer SIP trunks.

  Enable this flag, if the WAN Port is located on a public IP and you want outgoing calls to the SIP Client located behind the NAT Router. OR if you need to receive incoming calls from the SIP Client located behind the NAT router. By default, Use of Symmetric RTP is disabled.

- **Accept RTP Packets from Random Port:** By default, this is disabled, that is, the System will accept RTP packets from the negotiated RTP port only.

  Enable this flag, if you want the system to accept RTP packets from any random port.

  - *Accept RTP Packets from Random Port parameter is applicable only if:*
    - *Use Symmetric RTP? is disabled.*
    - *RTP Packets flag in PCAP is disabled. For details, see "PCAP Trace".*
    - *RTP Mode is set as Transcoding or RTP Relay. For details, see "Configuring VoIP Parameters".*

  - *The System security might be at risk, if you enable this option as packets from all RTP ports are accepted by the System.*

- **Default Transport for Outgoing Message:** Configure this parameter for Proxy SIP Trunks only. The SARVAM UCS supports three options for transporting outgoing SIP messages:

  - **UDP:** Outgoing messages are transported using UDP. By default this option is selected.

- **TCP:** Outgoing messages are transported using TCP. If you select this option, you must enable ‘SIP Over TCP’ on the ‘VoIP Parameters’ page.

- **TLS:** Outgoing messages are transported using TLS. If you select this option, you must enable ‘SIP Over TLS’ on the ‘VoIP Parameters’ page.

  By default, UDP is selected as the Default Transport for Outgoing Message.

The Default Transport for Outgoing Message options are checked only if you have enabled SIP over TCP or SIP over TLS.

If the SIP over TCP and SIP over TLS are disabled, all outgoing SIP messages will be transported over UDP only.

- **SRTP Mode:** SARVAM UCS supports SRTP (Secure Real Time Protocol) for secure conversations over SIP. The SARVAM UCS supports the following options:

  - **Disable:** SARVAM UCS uses normal RTP for transporting the speech packets.

  - **Optional:** SARVAM UCS uses SRTP for transporting the speech packets. If the remote user does not support SRTP, SARVAM UCS will use normal RTP for transporting the speech packets.

    - If you select this option, you must configure the **SRTP Media Type**. You may select **AVP** or **SAVP**. By default, AVP is selected as the SRTP Media Type.

  - **Forced:** SARVAM UCS uses only SRTP (SAVP) for transporting the speech packets. If the remote user does not support SRTP, SARVAM UCS will reject incoming calls from and drop outgoing calls made to such users.

    By default, SRTP Mode is Disabled.

- **Call Budget:** By default, Call Budget is enabled on the trunk. If you wish to change the default configuration or disable it for this SIP trunk, configure the parameters as per your requirement:

  - **Type:** Select the type of Call Budget on Trunk—Amount or Minutes or Calls—to be applied on this SIP trunk. By default, Minutes is selected as the Call Budget type.To disable select Type as None.

  - **Amount:** If you selected 'Amount' as the Call Budget Type, enter the Budget Amount in this field. By default the Amount is set to 99999.

  - **Minutes:** If you selected 'Minutes' as the Call Budget Type, enter the number of Minutes in this field. By default the number of minutes is set as 000300.

  - **Calls:** If you selected 'Calls' as the Call Budget Type, enter the number of Calls in this field. By default the number of calls is set to 9999.

  - **Scheduled Reset:** Enable this flag if you want the Call Budget Amount/Minutes/Number of Calls to be reset on a particular date of every month.

  - **Scheduled (Date):** Select the date of the month (Daily or 1-31) on which you want the Call Budget Amount/Minutes/Number of Calls to be reset every month. You may select 'Daily' if your plan suggests so.

The consumed Call Budget Amount/Minutes/Number of Calls can be reset also using SA and SE commands, referred to as Manual Reset. Refer the feature description "Call Budget on Trunk".

- **Call Back:** This parameter is related to the "Call Back on Trunk Port' feature. If you want to enable the 'Call Back on Trunk Port' feature on this SIP trunk, configure the following parameters:

  - **Enable Call Back:** Enable this flag to activate the Call Back on Trunk Port feature. By default, this flag is disabled on all trunk port types. By default, the flag is disabled.

  - **Call Back Timer:** This is the duration for which the system waits for the caller to disconnect before applying the Call Back. The range of this timer is from 01 to 99 seconds. By default, it is set to 10 seconds for all SIP Trunks.

  - **Call Back Mode:** Select from the following options how a 'Call Back' call answered by the remote party should be routed:
    - Built-In Auto Attendant
    - PIN Authentication - Multiple Calls
    - CLI Authentication - Multiple Calls
    - CLI Authentication - Single Call - Answer Signaling
    - Operator

    By default, Operator is selected as the Call Back Mode.

  - **Call Back on:** This parameter allows you to select if the call back should be made to the same number that was received or to a different number. If you want the call back to be made to the same number select the 'CLI number". If you want the call back to be made to a different number, select 'Alternate Number'.
    By default, CLI number is selected for Call Back.

  - **Incoming Number List:** Program the number strings that are eligible for Call Back in this List. By default, Number List 15 is assigned to Call Back Incoming Number List.

    Number List 15 is also assigned to all SIP trunks as well as all other Trunk port types. If you want the same numbers strings to be programmed commonly for all SIP trunks and Trunk Port types, retain this list.

    If you want a different set of number strings to be programmed for this SIP Trunk, select a different Number List, and assign it to the SIP trunk port.

    You may program the Incoming Number List either from the 'Number List' page or by clicking the 'Incoming Number List' link to reach the Number List page.

    Refer the topic "Number Lists" to know more, and for configuration instructions.

  - **Outgoing Number List:** Program the number strings that are to be called back in this List. For each number string you programmed in the 'Incoming Number List', you must program in the corresponding index in the Outgoing Number List a number to which the call back is to be made. For example, for the number string programmed at Index 1 in the Incoming Number List, a corresponding number string at the same Index, Index 1, should be programmed in the 'Outgoing Number List'.

    By default, Number List 16 is assigned to Outgoing Number List.The same Number List 16 is also assigned to all SIP trunks as well as all other Trunk port types.

You may program the default number list, or a different number list and assign it to this SIP Trunk port.

You may program the Outgoing Number List either from the ‘Number List’ page or by clicking the ‘Outgoing Number List’ link to reach the Number List page.

Refer the topic “Number Lists” to know more, and for configuration instructions.

- **Call Back from:** This parameter determines the trunk port to be used to make the call back. The call back can be made using the same port or an “OG Trunk Bundle Group”.

  Select ‘Same port’ if you want the call back to be made using the same port on which the missed call is received. If you select OGTBG, the call back will be made using the OGTBG, which you have defined.

  By default, Same port is selected.

- **OGTB Group:** If you selected OGTBG for making the call back in the previous parameter, you must define the OGTBG that must be used in this parameter.

  By default, OGTBG 01 is assigned to all SIP trunks.

  If you want the system to select the lowest cost trunk for making the call back, enable Least Cost Routing on the OGTBG that you define here for Call Back.

- **Incoming (IC) Reference ID:** Assign an Incoming Reference ID for the SIP trunk for Working Hours, Break Hours, Non-Working Hours. By default, 00 is assigned as Incoming Reference ID for all three time zones.

- **Outgoing (OG) Reference ID:** Assign an Outgoing Reference ID for the SIP Trunk, from 00 to 99. By default, 00 is assigned as Outgoing Reference ID.

- **Pause Timer (sec):** This Timer is required for inserting delay while digits of a number string are out dialed from the SIP trunk. The Pause Timer will be applicable when the letter 'P' is configured in the DTMF number string which is to be out dialed as DTMF digits on the SIP trunk. The range of this timer is from 1 to 9 seconds. By default the Timer is set to 3 seconds.

  For example, if 'PPP3' is to be out dialed and Pause timer is programmed as 3 seconds, the SARVAM UCS will out dial the digit 3 after 9 seconds, after a delay of individual P (3+3+3 =9). The range of this Timer is from 1 to 9.

  This parameter is used for the “Multi-Stage Dialing” feature.

- **DTMF ON Timer:** This is the time for which the DTMF digit will remain ON, while being out dialed by the SARVAM UCS. This parameter finds its application in the feature “Multi-Stage Dialing”. The range of this timer is from 051 to 255 milliseconds. By default, ON Time is defined as 102 msec.

- **DTMF Inter Digit Pause (msec):** This is the time for which the SARVAM UCS will wait before dialing the successive digits.

- **Category (Logical Partition):** This parameter assigns the SIP Trunks to a trunk category for the purpose of Logical Partitioning. By default all SIP Trunks are assigned to Category 4. Do not change the default setting.

If you want to change the call permission between the SIP Trunks and other trunks, click the 'Category' link to open the Logical Partitioning page. You may program the call permission between Category 4 (assigned to SIP Trunks) and other Categories. Refer the feature description "Logical Partition" to know more.

- **Gateway Application - Answer Signaling:** This parameter is to be programmed if the SIP trunk is being used in a gateway application as a source port (from where calls originate). The calls originated on the source port (SIP trunk) are routed using another Trunk port, the terminating port, which may be any trunk port, like T1E1. When call made from the terminating port gets matured, this is signaled to the source port in the form of DTMF digits.

  - **Use?:** Enable this flag if you want the SIP trunk to be used in a Gateway Application.

  - **DTMF String:** Program the DTMF digits to be sent to signal call maturity to the source port.

- **Send Re-INVITE over SIP Trunk on Hold:** With this parameter you can select whether or not the system should send Re-INVITE message from the SIP Trunk to the Remote Peer, when an external call over the SIP Trunk is put on hold by the extension user. Set this parameter as per the requirement of the Remote Peer. The Remote Peer can be a Proxy Server or a SIP Device.

  By default, Send Re-Invite over SIP Trunk on Hold is disabled. For more information see "Call Hold".

- **Delayed Offer:** Enable this flag, if you want the SIP Trunk to generate INVITE without SDP. Default: Disabled.

- **Fetch Called Party Number From**: SARVAM UCS extensions may be assigned DDI numbers provided by the ITSP. When INVITE is received from ITSP, the ITSP may send the DDI number either in the "Request URI" of the INVITE message or in the "To:" field of the INVITE message.

  By default, Request URI is selected. Ask your ITSP if you need to change this parameter.

- **Called Party Number as CLI:** If you want SARVAM UCS to display the called party number received in the INVITE message as the CLI, select the Display Called Party Number as CLI check box. By default, Display Called Party Number as CLI option is disabled.

  This parameter is useful when a single SIP Trunk having DDI Numbers and Operator are shared by more than one organization. If you enable this option, make sure:
  - you configure the names and corresponding numbers of the organizations sharing the SIP Trunk in the Global Directory of SARVAM UCS.
  - the Operator has a DKP or an Extended IP Phone or a Mobile UC Client.

  With this option enabled the Operator will be able to handle calls more efficiently. When there is an incoming call, SARVAM UCS matches the number with the numbers in the Global Directory. If a match is found SARVAM UCS displays the company name configured for that entry to the Operator, that is, the CLI will display the called party number and name.

  After the Operator answers the call, the CLI will change and display the calling party number and name (if configured in the Global Directory).

  If you keep this option disabled, the calling party number and name will be displayed as the CLI, both during an incoming call and after the call is answered by the Operator.

You can configure the Display Called Party Number as CLI option only from Jeeves.

- **On Connecting Media Send**: If Built-In Auto Attendant or DISA is enabled on the SIP Trunk, select the response you want the system to send on Connecting Media, that is **200 OK** or **183 Session Progress**.

- **Answer Source Trunk on Receiving**: When a call received on any trunk of SARVAM UCS is routed through the SIP Trunk, select the response after which the call on the source trunk must be answered. You can select **Early Media** or **200 OK**.

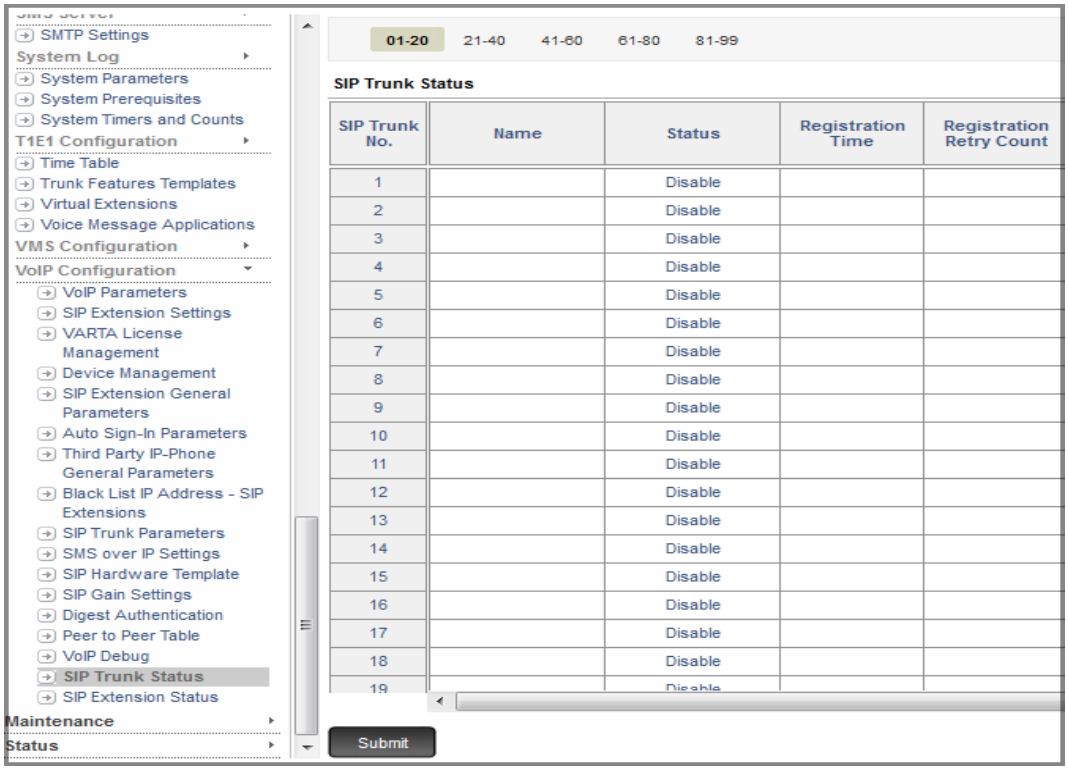
- If you have configured the above parameters, click **Submit** at the bottom of the page to save your configuration settings.

Viewing SIP Trunk Status

You can also view the settings of the SIP Trunk Parameters on Jeeves.

To do this,

- Under **VoIP Configuration**, click the **SIP Trunk Status** link.



- For each SIP trunk (number), the following settings will be displayed:
  - SIP Trunk Number
  - Name
  - Status
  - Registration Time
  - Registration Retry Count
  - Reason for Failure
  - Call Budget Type
  - Allotted Amount/Minutes/Calls
  - Consumed Amount/Minutes/Calls
  - Scheduled Reset

- Budget Reset Scheduled (Date)
- Reset Consumed Budget (this is not a status indicator. It is for resetting the Consumed Call Budget manually)

*You can also view the SIP Trunk Status from the **Status** link. To view, click the SIP Trunk link under Status.*

Configuring PRI Trunks

What's this?

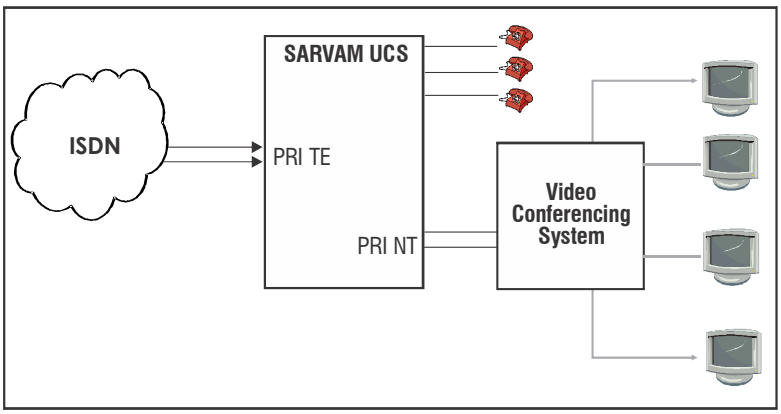
- This topic explains the connection of PRI line, and application of PRI.
- For MSN Numbering System, please refer "Configuring BRI Trunks".
- PRI port of the SARVAM UCS configured for NT mode can be connected to the PRI Port of another System configured for TE mode. In such case, the SARVAM UCS will behave as Transit Exchange.
- Dialing method on the PRI port is not programmable. The PRI port (configured for TE mode) will send the called party number in Enbloc mode.
- All the switch variants are applicable to the PRI port whether programmed in TE or NT mode.

Applications:

Applications for PRI-NT Port are as described below:

- Video conferencing system connected to the PRI-NT port with IC/OG calls.
- Data Calls support.
- Networking of Systems.

Video conferencing system connected to the PRI-NT port:



- Connect the Video Conference system (H.320) to the PRI-NT port of the SARVAM UCS.
- This feature works only if it is supported by the Service Provider. Video Conference is established mainly by the Video Conferencing (VC) equipment.
- Video Conferencing requires 6 B-channels. But Video Conferencing can also be done at lower bit rates also using the "aggregation" of 6 B channels which must be supported by the VC equipment.
- The VC equipment uses two methods:H.221 and H.242 BONDING, for aggregation.
- More than one Video conferencing system can be connected to the SARVAM UCS to each PRI-NT port.
- It also supports internal calling like, one Video Conferencing system to another.

OG calls from VC

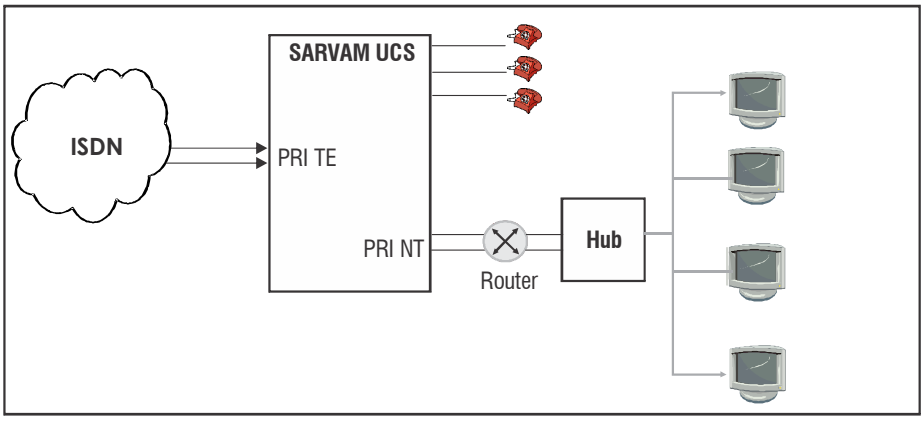
- After suitable connections to the VC equipment, user follows following sequence:
- Program the OGTBG (OG Trunk Bundle Group) such that the user gets at least 6 B-channels of the same PRI port.
- Go OFF-Hook.
- Dial the trunk access code to grab the PRI trunk.
- Dial the destination number.
- If VC at the called party responds, the call is established which occupies 6-B Channels.
- If 6 B-channels on the same PRI port are not available, user at VC will get busy tone.
- The user at calling party's VC gets dial tone of the ISDN exchange.
- The user at VC starts dialing the destination number. The destination number is sent in keypad IE by the VC user to the PRI-NT port whereas it is dialed on the PRI-TE port in the method programmed for the port that is, Enbloc.
- Rest of the signaling is done between the VC equipment and the called party's VC equipment.
- If VC equipment supports Phone Book feature, user can make call using the feature.
- At the end of VC, all the 6 B-channels are freed and available for other users.
- It is preferable that the SE will assign an OGTBG to the Video Conferencing equipment in such a manner that the same group is not assigned to any other station. This is to allow Video Conferencing call at any time.

IC Video Call:

- Following steps are followed for IC call:
    - Program the system to place IC video calls to the PRI-NT port using DDI Routing Table.
    - For this, program the DDI Routing Table. Refer related chapter.
    - You can also prepare the Trunk Landing Group (Routing Group) to land the call to PRI-NT port. For this purpose, Routing Group is to be modified.

Data Calls from PRI-NT:

- SARVAM UCS PRI-NT port supports data communication also.
- For data-communication, connect the Router supporting ISDN-PRI or a Computer with SARVAM UCS Card T1E1PRI to the PRI-NT port as shown below:



Using this feature following applications are supported:

- The computers connected in the LAN can browse the net through the PRI.
- Remote LAN Access the Computers in the LAN can access the computer/computers in LAN at the remote end (Branch office/Home office).
- Files can be transferred from one LAN to another.

OG Data calls from the Router

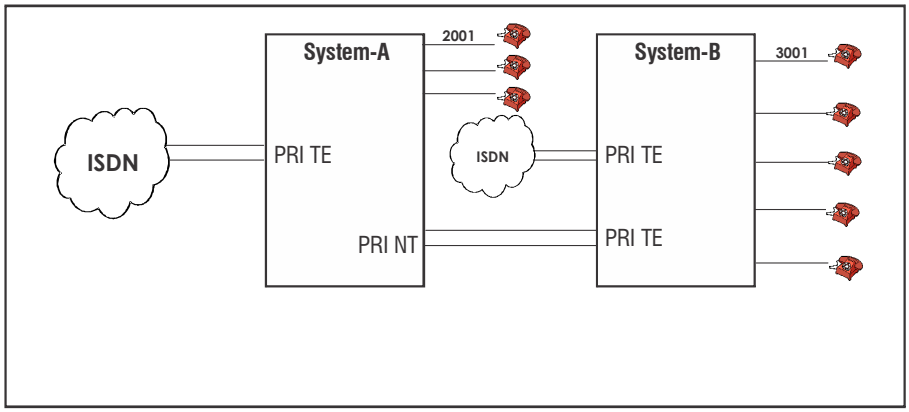
- The data call can be made by the router requesting desired number of channels. This establishes a live connection between the Router and the ISP through the System. The users on the LAN can browse the net as normal using Internet Explorer or Netscape Navigator.
- For this, the SARVAM UCS will allocate data channels only on the PRI-TE port so as to leave other channels for speech calls when the system detects the call to be a data call.
- Similarly, a Remote Computer can be accessed (Remote LAN Access) by dialing the Remote users' number (The remote end System should be so programmed that the call made to a number lands directly on the Router.) This establishes a permanent connection between the two Routers (and hence two LAN networks).
- Now the user at System-A can access the computer in LAN at the remote end in the same way as accessing another computer on the same network.

IC Data calls to the Router:

- Program the Trunk Landing Group such that all data calls will land on the PRI-NT port to which the Router is connected using CLI based Routing logic or DDI based routing logic.
- However, while routing call on the PRI-NT port, the System will check that the data channels reserved for data communication on the PRI-NT port are enough to establish the call. Otherwise the call will be rejected.
- The call will be rejected if the number of channels, reserved for data calls are already busy with one data-call.

Networking of Systems:

One of the applications is also for connecting multiple Systems using PRI-NT and PRI-TE ports. A simple connection of only two Systems is explained below. Refer following figure for connecting another System to the PRI-NT port.



Various cases of Calls from System-A to System-B can be:

Case-1: (2001 calls 3001)

- A station 2001 of System-A can call a station 3001 of System-B by dialing 3001.
- Such a call can be routed using CUG table of System-A. Refer chapter “Closed User Group (CUG)”.

Case-2: (2001 calls a subscriber of ISDN network connected to System-B)

Method 1

- Program (for System-A) an OGTBG containing the PRI-NT port and assign Trunk access code to it.
- Grab the PRI-NT by dialing access code. User of 2001 will get dummy dial tone of the System-A.
- Dial a station number 3001 of System-B.

Method 2

- User of 2001 can dial trunk access code of System-A, and dial a trunk access code of System-B. He will be connected to ISDN network through System-B.
- Then he can make OG call on PRI-NT port as per method programmed for the PRI-NT port. (For example, PRI-NT port can be programmed as a Trunk port and when station user of System-B grabs the PRI-NT port, his call will be placed on the destination programmed for PRI-NT port in the Trunk Landing Group).

Case-3: (3001 to 2001)

Station 3001 of System-B will go Off-Hook.

- Using CUG feature of System-B, 3001 can call 2001 through PRI-NT port. Refer chapter “Closed User Group (CUG)” for more details.
- 3001 can also dial Trunk Access Code to grab its PRI-TE port. He will get dial tone of PRI-NT Port of System-A. Now 3001 can dial 2001 or trunk access code to make a call to the ISDN network connected to System-A.

How to configure

For programming of the PRI, please refer topic “Configuring E1 Trunks” and “Configuring T1 Trunks” for more details.

- *Trunk software ports are automatically assigned to the PRI port by the system depending on the slot in which they are inserted.*
- *Please take care to **terminate the PRI line on the HDSL interface only** of the ISDN modem.*
- *The DDI Routing and Routing Tables shall be programmed as explained in chapters on DDI.*

Configuring T1 Trunks

What's this?

Digital Signal Level 1 (T1E1) trunks use Bit-Oriented Signaling (BOS) and multiplexes 24 channels (T1 service) or 32 channels (E1 service) into a single data stream. T1E1 can be used for voice or voice-grade data and for data-transmission protocols. T1 trunk service multiplexes 24 channels into a single 1.544-Mbps data stream. E1 trunk service multiplexes 32 channels into a single 2.048-Mbps stream. Both T1 and E1 provide a digital interface for trunk groups.

Signaling Modes

Common Channel Signaling (CCS) is an industry-standard technique where any one of a group of channels carries the signals for the other channels. Matrix uses the 24th channel of a group for signaling. This signaling technique differs from 24-channel signaling. When the system is configured for Facility-Associated Signaling, 24-channel signaling uses the 24th channel in a T1E1 facility to carry signals. This technique also is called clear channel, out-of-band or alternate voice data (AVD) signaling.

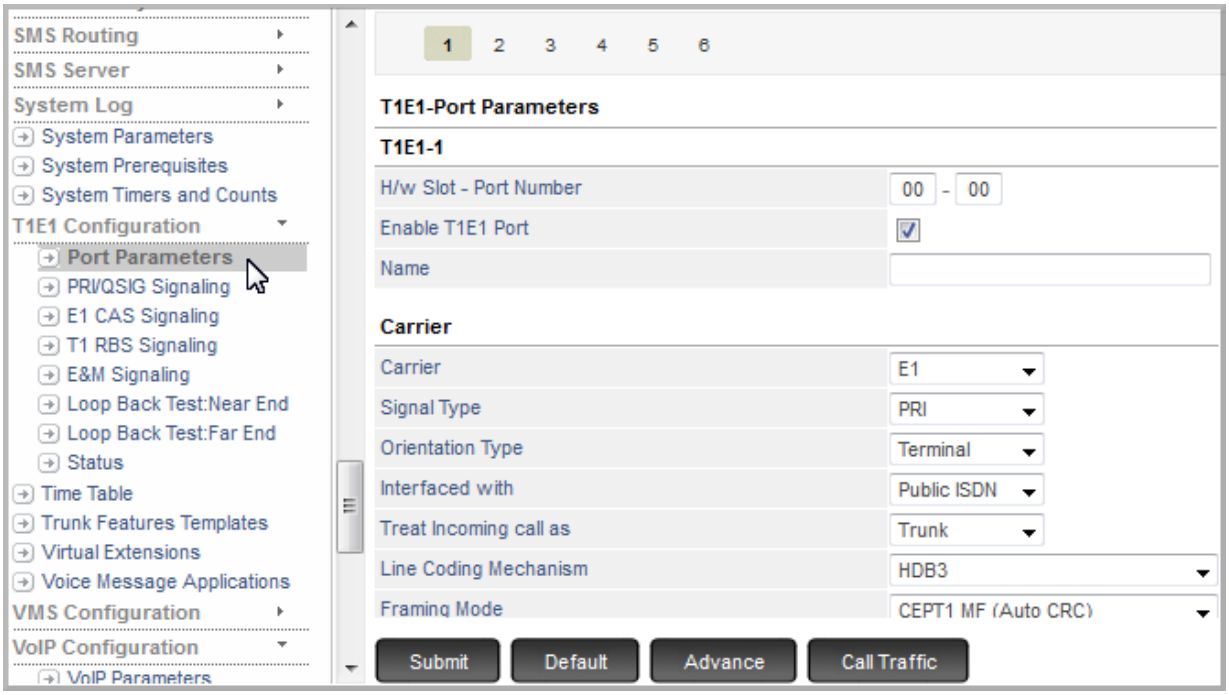
ROBBED-BIT signaling is a per-channel in-band signaling technique for transmitting signaling bits on each channel in a T1E1 facility. The least-significant bit in every 6th transmitted information frame is removed and replaced by a signaling bit. This technique is also called in-band signaling. The maximum transmission rate for each bearer channel with ROBBED-BIT signaling is 56 Kbps. T1 RBS Protocol supports 24 channels (from 01 to 24) and the protocol doesn't consume any channel for signaling so that there are total 24 channels available for the users.

ISDN-PRI signaling is carried on the 24th channel for a 1.544 Mbps (T1) connection and on the 16th channel for a 2.048 Mbps (E1) connection. In case of T1 PRI Protocol supports 24 channels (from 01 to 24), in which channel no. 24 is used for the signaling, so effectively there are 23 Voice channels are available.

QSIG is an ISDN based protocol for signaling between nodes of a Private Integrated Services network.
Any of the common trunks, except for PCOL (Personal Central Office Line) trunks, can be analog or digital. (PCOL trunks can only be analog.) Administering a digital trunk group is very similar to administering its analog counterpart, but digital trunks must connect to a T1E1 port and this port must be administered separately.

Configuring T1E1 Port Parameters using Jeeves

- Log into Jeeves as System Engineer.
- Under **Configuration**, click **T1E1 Configuration**.
- Click **Port Parameters** to open the page.



- Select the T1E1 Port number you want to configure by clicking the respective tab, and program the following port parameters:

  - SARVAM UCS will assign the **Hardware Slot-Port** automatically, when any card is inserted in the system.

    Hardware slot is the number of the Universal slot of SARVAM UCS in which the T1E1 Card is inserted. Range of slot number is 01-16. Port is the number of T1E1 hardware port on the card to which the T1E1 line is connected. Range of Port is from 00-99.

    If you want to de-assign the Hardware Slot and Port, Enter '00' in both fields. By default, Hardware slot-Port is 00–00.

  - Keep the **Enable T1E1 Port** flag enabled.

    Clear the flag, only if you do not want to use this port. By default, it is enabled.

  - You may assign a **Name** to the T1E1 Port for identification of the port. The Name may consist of a maximum of 12 characters. By default, it is blank.

  - Select the **Carrier** type as **T1**. The Carrier type will automatically be assigned to the port when you select the region. You may it if required.

  - Select **Signal Type**. Signal Type signifies the type of signaling to be used on E1 line. The E1 Signaling supported are:
    - PRI
    - RBS
    - QSIG
    - E&M

    Default: PRI

If you select **PRI or QSIG** as the Signal Type, configure the **PRI/QSIG Parameters**.

If you select **RBS** as the Signal Type, configure the **RBS Parameters**.

If you select **E&M** as the Signal Type, configure the **E&M Parameters**.

- Set the **Orientation Type**. Select the orientation type from the following options:
  - Terminal
  - Network
  - Tie-Line

  Default: Terminal

- If you have set **Terminal** as the **Orientation Type**, you must select the type of network with which it is to be **Interfaced With**. The network may be:
  - Public ISDN
  - Private ISDN

  Default: Public ISDN

- You may configure the port to **Treat Incoming call as Trunk** or **Station**.

  If you select **Trunk**, the system will treat all incoming calls as external calls landing on the trunk. The calls will be routed as per the **Trunk Feature Template** assigned to the T1E1 Port.

  If you select **Station**, you must also assign a Station Basic Feature Template and Station Advanced Feature Template to the T1E1 Port.

  When you select Station, the system will treat the calling party as an extension user. The user will have access to all the features and facilities of the system, as per the Station Basic Feature Template and Station Advanced Feature Template assigned to the T1E1 Port.

  By default, Trunk is selected.

- *If Point-to-Point is selected as the Interface Type, you can select the option **Trunk** or **Station** for the parameter **Treat Incoming call as**.*

- *If Point-to-Multipoint is selected as the Interface Type, only **Station** can be set as the option for the parameter **Treat Incoming call as**.*

- *If **Station** is selected as the option for **Treat Incoming call as**, the user will only be able to:*
  - *Dial Flexible Numbers*
  - *Dial Operator Code*
  - *Dial Trunk Access Code for making outgoing calls*
  - *Access the Global Directory*
  - *Make calls within the Closed User Group*

- Line coding is a pattern that data assumes as it is propagated over a communication channel. Select the **Line Coding Mechanism** from the following:
  - AMI-Basic
  - B8ZS
  - NRZ (Fiber Optic)

  Default: AMI-Basic.

- Framing means to form a set of 24 or 32, 8 bits time slot that is to be treated as single transmission unit. The **Framing Modes** supported by SARVAM UCS are:
  - SF (D4)
  - ESF

  Default: SF (D4)

- **Line Buildout Parameter** field reduces the outgoing signal strength by a fixed amount. The appropriate level of loss depends on the distance between your switch (measured by cable length from the smart jack) and the nearest repeater. Where another switch is at the end of the circuit, as in campus environments, use the cable length between the 2 switches to select the appropriate setting from the list. This field is relevant if the Near-end CSU type field is integrated. By default, 0-133ft is selected as the Line Buildout Parameter.

- Configure **Outgoing (OG) Reference ID**. By default, OG Reference ID is 00.

- Configure **Incoming (IC) Reference ID** for working hours, non-working hours and break hours. By default, IC Reference ID is 00.

- Assign **Trunk Feature Template** to the T1E1 Port. Trunk Feature Template is a set of general features that define the behavior of a Trunk Port. By default, Template 01 is assigned to all T1E1 Ports.

  For more details, see “Trunk Feature Template”.

- **Station Basic Feature Template** assigned to the T1E1 Port is displayed in this field. Station Basic Feature Template is a set of general features that define the basic behavior of a station. By default, Template 01 is assigned to all T1E1 Ports.

  For more details, see “Station Basic Feature Template”.

- **Station Advanced Feature Template** assigned to the T1E1 Port is displayed in this field. Station Advanced Feature Template is a set of advanced features, to be applied on extensions such as CLIP, Floor Service, Walk-in Class of Service. By default, Template 01 is assigned to all T1E1 Ports.

  For more details, see “Station Advanced Feature Template”.

- Select **Priority** for the T1E1 Port. Priority is the precedence given to certain trunks and extensions over others in being answered by the destination extension. You can select from 1 to 9. By default, Priority 5-Normal is set for all T1E1Ports.

  For know more about Priority feature, see “Priority”.

- Assign a **Cost Factor** to the T1E1 Port. By default, all the T1E1 Ports are assigned Cost Factor 01.

  For more details, see “Cost Factor”.

- ISDN glare occurs if the system initiates an outgoing call on a B-Channel at the same time the network initiates an incoming call on that same B-channel. You may configure the **Glare Option** as **Proceed** or **Held Back**. While processing a glare condition, the configured Glare Option on T1E1 port will be considered.

- Select **Category (Logical Partition)** for the T1E1 Port. You may select from the following options:
  - 1
  - 2

- 3
- 4

By default, 1 is selected for all T1E1 Ports. Refer the feature description "Logical Partition" to know more.

- **Idle Code** is the 8-bit sequence that occupies the time slot on a E1/T1 trunk channel when it is not being used. By default, 127 is configured as the Idle Code.

- If you want SARVAM UCS to display the called party number as the CLI for incoming calls, select the **Display Called Party Number as CLI** check box. By default, Display Called Party Number as CLI option is disabled.

  This option is useful when a single T1E1 line connection and Operator are shared by more than one organization. If you enable this option, make sure:
  - you configure the names and corresponding numbers of the organizations sharing the line in the Global Directory of SARVAM UCS.
  - the Operator has a DKP or an Extended IP Phone or a Mobile UC Client.

  With this option enabled the Operator will be able to handle calls more efficiently. When there is an incoming call, SARVAM UCS matches the number with the numbers in the Global Directory. If a match is found SARVAM UCS displays the company name configured for that entry to the Operator, that is, the CLI will display the called party number and name.

  After the Operator answers the call, the CLI will change and display the calling party number and name (if configured in the Global Directory).

  If you keep this option disabled, the calling party number and name will be displayed as the CLI, both during an incoming call and after the call is answered by the Operator.

You can configure the Display Called Party Number as CLI option only from Jeeves.

- Select the **Allow Incoming CLI Modification** check box if you want to apply 'Allow Incoming CLI Modification' on the T1E1 Port. By default, it is disabled.

  Incoming CLI Modification is useful in countries where the Calling Line Identification (CLI) received by the System extension users must be suitably modified before it can be used to dial out the number. To know more, see "Incoming CLI Modification".

If you receive CLI in dialable format, there is no need to use this feature. In such case, keep the flag disabled.

*For an incoming call on the T1 trunk, the Incoming CLI Modification will be applied only when both — the **Allow Incoming CLI Modification** check box and the **Enable Incoming CLI Modification** check box in System Parameters — are enabled.*

- Select **Return Call to Original Caller (RCOC)** flag to enable this feature on the T1E1 Port. By default, RCOC flag is disabled.

  For know more about RCOC feature, see "RCOC (Return Call to Original Caller)".

- In **Channel Reserved for Data Call**, configure the channels you want to reserve for Data Calls. By default, 00 channels are reserved.

- In **Channel Reserved for Outgoing Call**, configure the channels you want to reserve for making outgoing calls. By default, 30 channels are reserved.

- In **Channel Reserved for Incoming Call**, configure the channels you want to reserve for receiving incoming calls. By default, 30 channels are reserved.

- When a caller dials the trunk access code or selective trunk access code for dialing the number directly on the trunk port, the caller waits for the dial tone before dialing the number. But some exchanges do not give Dial Tone for the T1E1 Port. For Example, when T1E1 port as E1CAS type is used in Delhi, it is observed that the exchange does not give dial tone when direct dialing on the trunk is used.

  Enable the **Feed Dial Tone** flag. SARVAM UCS will provides the dial tone to the caller when the T1E1 Port is accessed.

  It is applicable only when Online dialing is used as, Store and Forward dialing, the dial tone is given to the caller. The dial tone is played as per the Dial Tone Timer.
  By default, Feed Dial Tone flag is disabled.

- *When dial tone flag is disabled, user will hear the dial tone of the exchange if provided, otherwise, user will hear the silence.*

- *If the user is making the call from the SLT port and dial tone is not provided by the exchange, user will not know when to start dialing the number. In this case, it is possible that some digits are not out dialed on the port and wrong number is dialed out because system will out dial the number only if Outgoing call Acknowledge is received but the user is not aware of this condition. Hence, it is required to enable this flag, if exchange is not providing the dial tone.*

- When Online dialing or Store and Forward dialing are used, some exchanges do not provide any tone while routing/processing the call. Thus, the caller does not know whether the call is being processed or not as there is silence. In such case, You must enable the **Feed Routing Tone** flag, SARVAM UCS will play the routing tone to the caller. SARVAM UCS stops playing the routing tone will be stopped when an alert message or connect message or disconnect message is received from the T1E1 Card.

  By default, the Feed Routing Tone flag is disabled.

- To customize the pulse width option and set the pulse shapes configure the **Custom Pulse** parameters. SARVAM UCS generates pulse shapes which match the country standard, where it is installed. However, if the standard pulse shape does not match, SARVAM UCS enables you to customize the pulse width to match your exact requirements.

  To use customize pulse width option and set the pulse shape in 1 to 4 phases, keep the **T1/E1 Custom Pulse Width (CPW) flag** enabled.

  - **Word 1:** Set the pulse width for setting pulse shape in the 1st phase. The range of Custom Pulse Width-Word 1 is 001 to 127. Default: 109.

  - **Word 2:** Set the pulse width for setting pulse shape in the 2nd phase. The range of Custom Pulse Width-Word 2 is 001 to 127. Default: 107.

  - **Word 3:** Set the pulse width for setting pulse shape in the 3rd phase. The range of Custom Pulse Width-Word 3 is 001 to 127. Default: 64.

- **Word 4:** Set the pulse width for setting pulse shape in the 4th phase. The range of Custom Pulse Width-Word 4 is 001 to 127. Default: 64.

- Select the **Bearer Service** supported by your service provider. You can select from:
  - Speech
  - 3.1 KHz Audio

  By default, Speech is selected.

- The **Overlap Receiving Timer** is relevant while receiving the called party number information in overlap receiving mode. It is not relevant for the port in overlap sending mode.

  Range of Overlap Receiving Timer is from 01 to 99 seconds. Default: 15 seconds.

- Configure **Pause Timer** for theT1E1 Port. Range of Pause Timer is from 1 to 9 seconds. By default, it is set to 3 seconds for all T1E1 Ports.

  This Timer is required to insert delay between the digits while dialing out DTMF digits on the T1E1 port. One of the applications for using this parameter is Multi-stage dialing. Refer chapter "Multi-Stage Dialing".

  For example, if PPP2 is to be outdialed and Pause timer is programmed as 3 seconds, the SARVAM UCS will out dial the digit 2 after 9 seconds i.e delay of individual P i.e 3+3+3 =9.

- When the SETUP Message is sent by SARVAM UCS to the network (ISDN exchange), the exchange responds by sending SETUP ACK (Acknowledgment), and dial tone is played to the caller. The time taken by the exchange to respond to the SETUP message may vary from exchange to exchange. Set the **SETUP Response Timer (sec)** as per the time taken by the network to respond to the SETUP message and play dial tone to the caller.

  Valid Range of the timer is 01 to 20 seconds. By default it is set to 4 seconds.

Change the default settings only if required. If the time you set is less than the time taken by the exchange to respond, no dial tone will be played to the caller.

You can configure the SETUP Message Timer only from Jeeves.

- Configure **DTMF On Time** for theT1E1 Port. Range of DTMF On Time is from 051 to 255 ms. By default, it is set to 102 ms for all T1E1 Ports.

  The DTMF On Time is the time for which the DTMF digit which is to be outdialed by the SARVAM UCS remain On. One of the applications for using this parameter is Multi-stage dialing. Refer chapter "Multi-Stage Dialing".

- Configure **DTMF Inter Digit Pause Timer** for the T1E1 Port. Range of Inter Digit Pause Timer is from 051 to 255 ms. By default, it is set to 102 ms for all T1E1 Ports.

  Inter Digit Pause Timer is the time for which the system will wait while receiving the dialing digits to consider it as end-of-dialing.

- Configure the **Minimum ON Time (msec)** for which the DTMF signal should be present in order to be detected. The valid range of this time is 10 to 200 milliseconds. By default, Minimum ON Time is set to 20 milliseconds.

- Configure the **Minimum OFF Time (msec)**. This parameter signifies the minimum time period between successive DTMF digits. The valid range of this time is 10 to 200 milliseconds. By default, Minimum OFF Time is set to 20 milliseconds.

  Configure the **Minimum Level (dB)** for the DTMF digit to be considered as valid. The valid range of this time is 0 to -36.5 dB. By default, Minimum levels is set to -36.5dB.

- If SARVAM UCS is to be used as a Gateway, enable **Gateway Application-Answer Signaling** on the T1E1 Port and configure **DTMF String**. By default, Gateway Application-Answer Signaling is disabled and CCC is configured as DTMF String.

  For more details, see “Gateway Application-Answer Signaling”.

- Configure **Call Budget** parameters for the T1E1 Ports. Call Budget is an expense control feature of SARVAM UCS that allows you to keep track of the cost of phone calls made from the T1E1 Port.By default, Call Budget is enabled on the trunk. If you wish to change the default configuration or disable it for this T1E1 Port, configure the parameters as per your requirement:

  - **Type:** Select the type of Call Budget, that is, Amount or Minutes or Calls to be applied on the T1E1 Port. By default, Minutes is selected as the Call Budget type.To disable select Type as None.

  - **Amount:** If you selected 'Amount' as the Call Budget Type, enter the Budget Amount in this field. By default the Amount is set to 999999.

  - **Minutes:** If you selected 'Minutes' as the Call Budget Type, enter the number of Minutes in this field. By default the number of minutes is set as 000300.

  - **Calls:** If you selected 'Calls' as the Call Budget Type, enter the number of Calls in this field. By default the number of calls is set to 9999.

  - **Scheduled Reset:** Enable this flag if you want the Call Budget Amount/Minutes/Number of Calls to be reset on a particular date of every month.

  - **Scheduled (Date):** Select the date of the month (Daily or 1-31) on which you want the Call Budget Amount/Minutes/Number of Calls to be reset every month. You may select 'Daily' if your plan suggests so.

- **Call Back:** This parameter is related to the ‘Call Back on Trunk Port’ feature. If you want to enable the 'Call Back on Trunk Port' feature on this T1E1 Port, configure the following parameters:

  - **Enable Call Back:** Enable this flag to activate the Call Back on Trunk Port feature. By default, this flag is disabled on all trunk port types. By default, the flag is disabled.

  - **Call Back Timer (sec):** This is the duration for which the system waits for the caller to disconnect before applying the Call Back. The range of this timer is from 01 to 99 seconds. By default, it is set to 10 seconds.

  - **Call Back Mode:** Select from the following options how a ‘Call Back’ call answered by the remote party should be routed:
    - Built-In Auto Attendant
    - PIN Authentication - Multiple Calls
    - CLI Authentication - Multiple Calls

- CLI Authentication - Single Call - Answer Signaling
- Operator

By default, Operator is selected as the Call Back Mode.

- **Call Back on:** This parameter allows you to select if the call back should be made to the same number that was received or to a different number. If you want the call back to be made to the same number select the ‘CLI number”. If you want the call back to be made to a different number, select ‘Alternate Number’.
  By default, CLI number is selected for Call Back.

- **Incoming Number List:** Program the number strings that are eligible for Call Back in this List. By default, Number List 15 is assigned to Call Back Incoming Number List.

  Number List 15 is also assigned to all T1E1 Ports as well as all other Trunk port types. If you want the same numbers strings to be programmed commonly for all T1E1 Ports and Trunk Port types, retain this list.

  If you want a different set of number strings to be programmed for this T1E1 Port, select a different Number List, and assign it to the T1E1 Port.
  You may program the Incoming Number List either from the ‘Number List’ page or by clicking the ‘Incoming Number List’ link to reach the Number List page.

  Refer the topic “Number Lists” to know more, and for configuration instructions.

- **Outgoing Number List:** Program the number strings that are to be called back in this List.

  For each number string you programmed in the ‘Incoming Number List’, you must program in the corresponding index in the Outgoing Number List a number to which the call back is to be made. For example, for the number string programmed at Index 1 in the Incoming Number List, a corresponding number string at the same Index, Index 1, should be programmed in the ‘Outgoing Number List’.

  By default, Number List 16 is assigned to Outgoing Number List.The same Number List 16 is also assigned to all T1E1 Ports as well as all other Trunk port types.

  You may program the default number list, or a different number list and assign it to this T1E1 Port.

  You may program the Outgoing Number List either from the ‘Number List’ page or by clicking the ‘Outgoing Number List’ link to reach the Number List page.

  Refer the topic “Number Lists” to know more, and for configuration instructions.

- **Call Back from:** This parameter determines the trunk port to be used to make the call back. The call back can be made using the Same Port or an “OG Trunk Bundle Group”.

  Select ‘Same Port’ if you want the call back to be made using the same port on which the missed call is received. If you select OGTB Group, the call back will be made using the OGTB Group, which you have defined.

  By default, Same Port is selected.

- **OGTB Group:** If you selected OGTB Group for making the call back in the previous parameter, you must define the OGTB Group that must be used in this parameter.

By default, OGTB Group 01 is assigned.

If you want the system to select the lowest cost trunk for making the call back, enable Least Cost Routing on the OGTB Group that you define here for Call Back.

- FDL is used for communicating general maintenance information or for transmitting user defined information within the T1 link. General maintenance information is in the form of Performance Message Report which is generated by the SARVAM UCS Card T1E1PRI and depending upon the FDL Protocol, the Performance Message Report is sent every second or on request.

  Select the **FDL** flag to enable. This parameter is applicable only if Framing = ESF. If the Network (Public or Private) to which the SARVAM UCS is connected does not support FDL then FDL will be disabled. By default, the T1 FDL is disabled.

  If you have enabled FDL, configure the **FDL Protocol**. SARVAM UCS supports ANSI T1.403 and AT&T 54016 protocols of reporting the performance monitoring. By default, the T1 FDL Protocol is ANSI T1.403.

  For more information, see “E1/T1 Maintenance”

- Enter appropriate **Debug Code (Level 1 to 4)**, to obtain debug information of various parts of T1E1 Card on the COM Port. By default, debug is off for all T1E1 ports for all levels.

  Code is the value for the specified level to turn ON the debug for the parameters. Code range is from 000 to 255. Code value '000' for each level will turn off that level's debug.

  Level and Code for T1E1 Port are as specified below:

  Level 1:

Unused	Unused	Unused	Unused	Layer 4	CAS DSP	MFC R2	CAS

001	CAS
002	MFC R2
004	CAS DSP
008	Layer 4
000	Debug Off

  Level 2:

Unused	Unused	Unused	HDLC (D-Channel)	FDL	ABCD Bits	Counters	Alarms

001	Alarms
002	Counters
004	ABCD Bits
008	FDL

016	HDLC (D Channel)
000	Debug Off

Level 3:

Unused	Flow Debug	NLS Debug	LAP Debug	SVC Primitives	Variables	State	Primitives

001	Primitives
002	State
004	Variables
008	SVC Primitives
016	LAP Debug
032	NLS Debug
064	Flow Debug
000	Debug Off

Level 4:

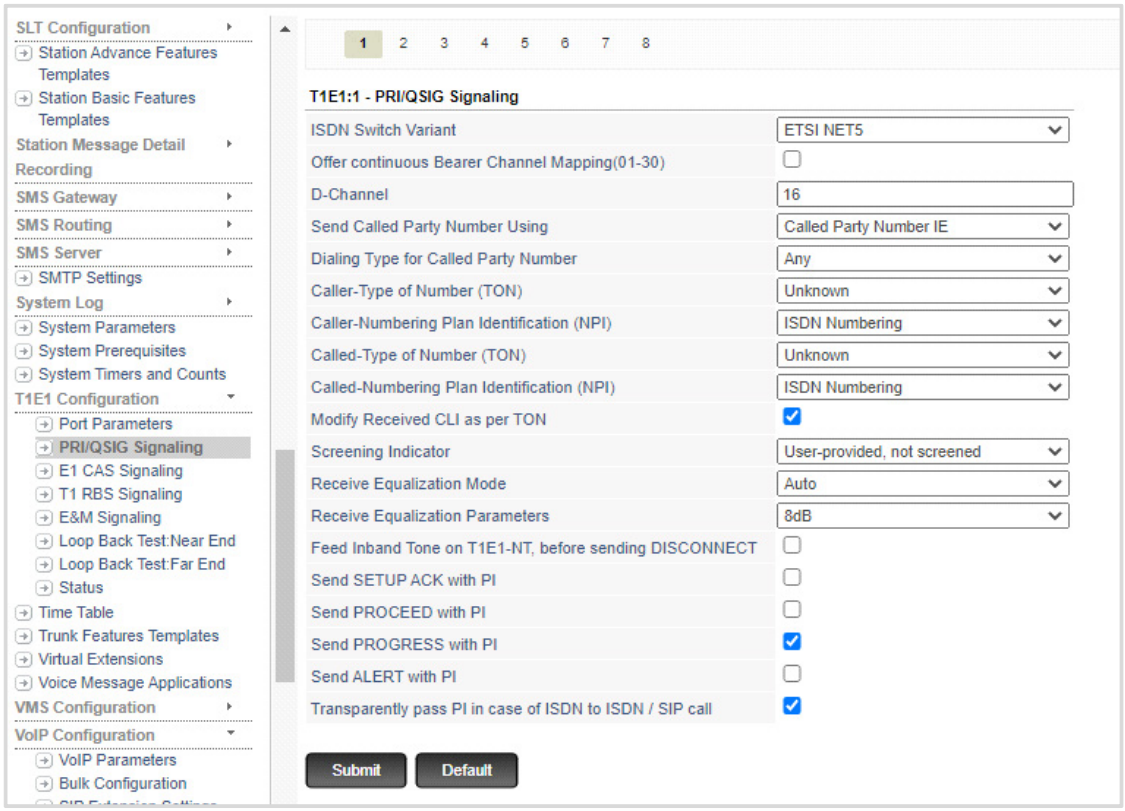
Unused	Unused	Unused	Unused	Unused	Unused	NI Debug	OS Task

001	OS Task
002	NI Debug
000	Debug Off

PRI/QSIG Parameters

If you have selected **PRI** as **Signal Type** on the Port Parameters page,

- Under **T1E1 Configuration**, click **PRI/QSIG Signaling** and configure the PRI/QSIG parameters.



- **ISDN Switch Variant:** ISDN supports a variety of service provider switches. Different countries use specific type of ISDN switch. This switch is designed using ISDN standard protocol. The type of switch determines various factors such as how many ISDN devices would be handled, which B-channel will support voice, video, data etc. Select the ISDN Switch Variant from the list. Default: ETSI NET5.

- **Offer continuous Bearer Channel Mapping(01-30):** Select this check box for continuous Bearer Channel Mapping for E1-QSIG/E1-PRI.

- **D-Channel:** Enter the Channel number that is used for Data Signaling. By default, D-Channel is 16. Valid Range is 1 to 31.

- **Send Called Party Number Using:** Select the appropriate option from the following for Send Called Party Number Using:
  - Called Party Number IE (Information Element)
  - Keypad Facility IE (Information Element)

  Default: Called Party Number IE is selected.

- **Dialing Type for Called Party Number:** Select the option from the following as supported by your exchange:
  - Enbloc
  - Digit-by-Digit
  - Any

  Default: Any.

- **Caller - Type of Numbering Plan (TON):** Select the appropriate option from the following for sending the type of numbering plan of the calling party:
  - Unknown
  - International
  - National
  - Network Specific
  - Subscriber
  - Abbreviated
  - Reserved

  Default: Unknown.

- **Caller- Numbering Plan Identification (NPI):** Select the appropriate option from the following for sending the numbering plan identification of the calling party:
  - Unknown
  - ISDN Numbering
  - Data Numbering
  - Telex Numbering
  - National Numbering
  - Private
  - Reserved

  Default: ISDN Numbering.

- **Called - Type of Numbering Plan (TON):** Select the appropriate option from the following for sending the type of numbering plan of the called party:
  - Unknown
  - International
  - National
  - Network Specific
  - Subscriber
  - Abbreviated
  - Reserved

  Default: Unknown.

- **Called - Numbering Plan Identification (NPI):** Select the appropriate option from the following for sending the numbering plan identification of the called party:
  - Unknown
  - ISDN Numbering
  - Data Numbering
  - Telex Numbering
  - National Numbering
  - Private
  - Reserved

  Default: ISDN Numbering.

- **Modify Received CLI as per TON:** This is applicable for PRI-TE mode only. If this check box is selected then the Calling Party Number received in SETUP message will be changed as per the Type of Number received in the SETUP message from Calling Party Number field by the system. Default: Disabled.

- **Screening Indicator:** Select the option as provided to you by your exchange. You can select from:
  - User provided, not screened
  - User Provided, verified and passed
  - User Provided, verified and failed
  - Network Provided

By default, User provided, not screened.

- **Receive Equalization Mode:** You can set the Receive Equalization Mode as Auto or Manual. By default, Auto is selected as the Receive Equalization Mode.

- **Receive Equalization Parameters:** This field increases the strength of incoming signals by a fixed amount to compensate for line losses. Select the required option from the list. By default, the receive equalization parameters of T1E1 is 8dB.

- **Feed Inband Tones on T1E1-NT, before sending DISCONNECT:** Select this flag, if you want to feed inband tones on T1E1-NT before sending DISCONNECT message. This flag is applicable only when T1E1 port is configured as 'Network'. When this flag is enabled, inband tones shall be feed for 15 seconds (fixed, non programmable) before sending DISCONNECT message.

  When this flag is disabled, inband tones (Busy/Error as applicable for the state of the call) shall not be feed before sending the DISCONNECT message. However when DISCONNECT message is sent from T1E1-NT port, inband tones will always be sent with 'progress indicator 8'. By default, this flag is disabled.

- **Send SETUP ACK with PI:** Select this flag if you want the system to send PI (Progress indicator) element in Setup Ack message to other end. This option is relevant only when call landing on the T1E1 port are to be routed to SLT/CO/Mobile/DKP port. If T1E1 port is configured as 'Network', by default it is enabled and if T1E1 port is configured as 'Terminal', by default it is disabled.

- **Send PROCEED with PI:** Select this flag if you want the system to send PI (Progress indicator) in Proceed message to other end. This option is relevant only when call landing on the T1E1 port are to be routed to SLT/CO/Mobile/DKP port. If T1E1 port is configured as 'Network', by default it is enabled and if T1E1 port is configured as 'Terminal', by default it is disabled.

- **Send PROGRESS with PI:** Select this flag if you want the system to send PI (Progress indicator) in Progress message to other end. This option is relevant only when call landing on the T1E1 port are to be routed to SLT/CO/Mobile/DKP port. By default it is enabled.

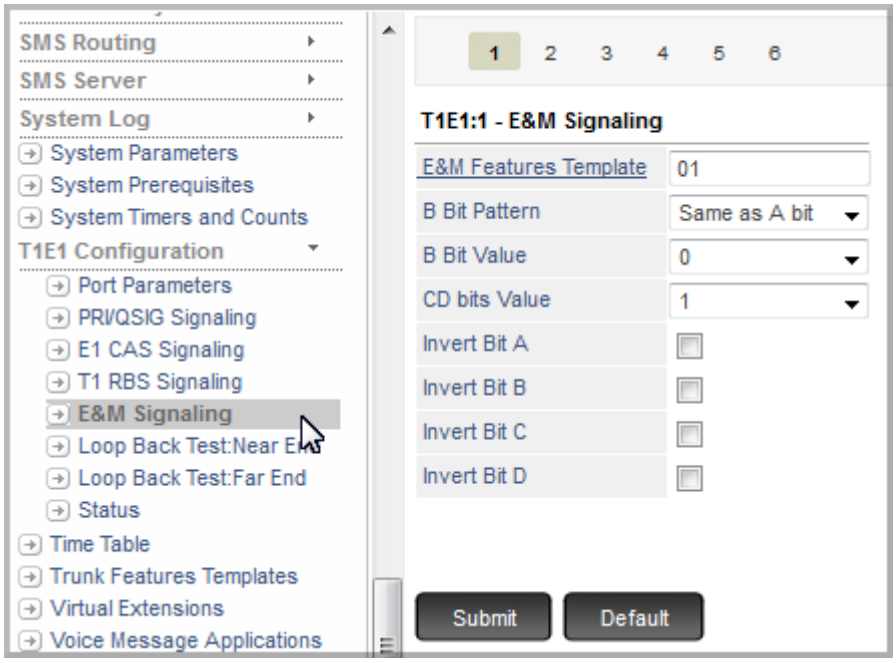
- **Send ALERT with PI:** Select this flag if you want the system to send PI (Progress indicator) in Alert message to other end. This option is relevant only when call landing on the T1E1 port are to be routed to SLT/CO/Mobile/DKP port. If T1E1 port is configured as 'Network', by default it is enabled and if T1E1 port is configured as 'Terminal', by default it is disabled.

- **Transparently pass PI in case of ISDN to ISDN / SIP call:** Select this flag, if you want the system to just pass the PI as received to the other end.This option is relevant only when call landing on the T1E1 port are to be routed to ISDN or SIP ports. By default it is enabled.

- Click **Submit** to save changes.

E&M Signaling

Under **T1E1 Configuration**, click **E&M Signaling** and configure the E&M parameters.



- **E&M Feature Template**: Assign an E&M Feature Template to the T1E1 Port. The E&M Feature Template is a set of features specific to E&M signaling, which define the behavior of the E&M ports, according to their 'Orientation Type', whether they are functioning as Stations, Trunks or Tie-Lines. By default, Template 01 is assigned to all T1E1 Ports.

  For more details, see “E&M Feature Template”.

- **B Bit Pattern:** Select the Bit Pattern from Same as Bit A or Fixed Value. By default, the Code is 1 (Same as A bit).

- **B Bit Value:** Configure the B bit value, the value can be 0 or 1. By default, B bit value is 0.

- **CD Bit Value:** Configure the CD bit value, the valid range of the value is 1 to 3. By default, B bit value is 0.

- **Invert Bit A:** This parameter signifies whether A-bit is to be inverted before transmitting and on receiving. Select the check box to Invert Bit A.

  Default: Disabled (Do Not Invert Bit A).

- **Invert Bit B:** This parameter signifies whether B-bit is to be inverted before transmitting and on receiving. Select the check box to Invert Bit B.

  Default: Disabled (Do Not Invert Bit B1)

- **Invert Bit C:** This parameter signifies whether C-bit is to be inverted before transmitting and on receiving. Select the check box to Invert Bit C.

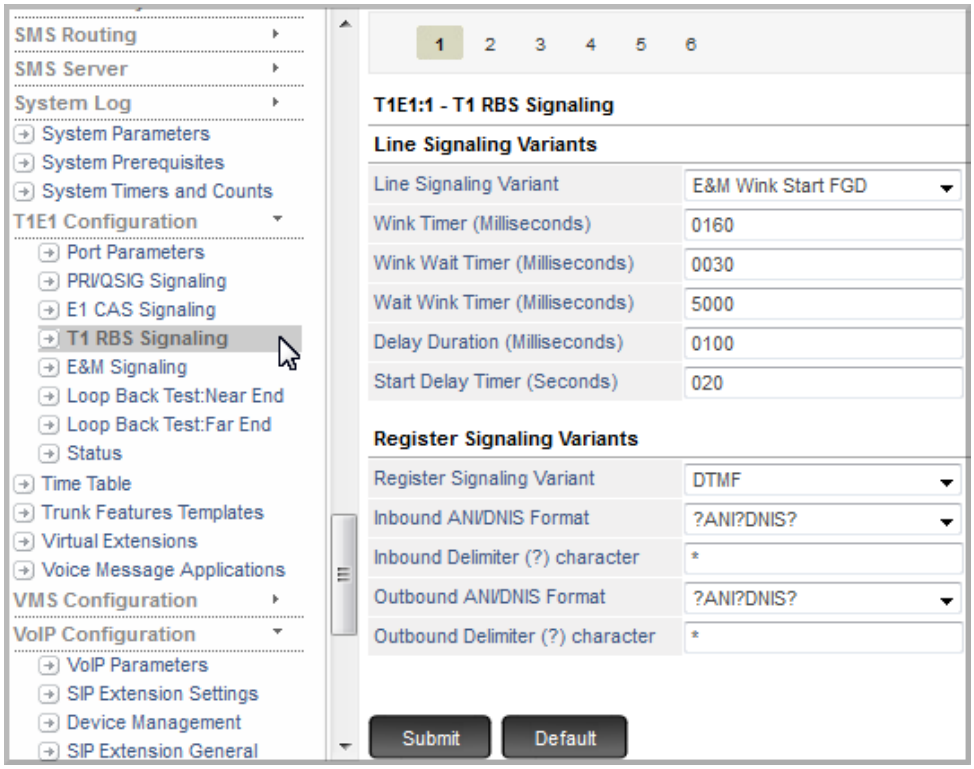
  Default: Disabled (Do Not Invert Bit C).

- **Invert Bit D:** This parameter signifies whether D-bit is to be inverted before transmitting and on receiving. Select the check box to enable, that is to Invert Bit D.

Default: Disabled (Do Not Invert Bit D).

RBS Parameters

Under **T1E1 Configuration**, click **T1 RBS Signaling** and configure the RBS parameters.



- **Line Signaling Variant:** Select the T1 Line Signaling Variant from following options:
  - SLT Loop Start
  - CO Loop Start
  - SLT Ground Start
  - CO Ground Start
  - E&M Immediate Dial/Start
  - E&M Wink Start
  - E&M Wink Start FGD

  Default: E&M Wink Start FGD.

- **Wink Timer (milliseconds)**: Wink timer refers to the momentary Off-Hook condition to acknowledge end of making an outgoing call. The Wink Timer ranges from 0001 ms to 9999 ms. Default: 160 msec.

- **Wink Wait Timer (milliseconds):** Wink Wait Timer signifies the maximum time the system should wait before sending a wink start signal after an incoming seizure is detected. Wink Wait Timer ranges from 0001 to 9999 msec. Default: 30msec.

Ensure that this timer is greater than the Wink Wait Timer of the other end.

- **Wait Wink Timer (milliseconds):** Wait Wink Timer signifies the time for which SARVAM UCS will wait for receiving the DNIS after sending the outgoing seizure signal. Wait Wink Timer ranges from 0001 to 9999 msec. Default: 5000 msec.

Make sure that this timer is greater than the Wait Wink Timer of the other end.

- **Delay Duration (milliseconds):** This duration signifies the time after which the DNIS information is to be sent while making an outgoing call. Range of the Delay Duration is from 0001 to 9999 msec. Default: 100 msec.

- **Start Delay Timer:** Start Delay Timer signifies the time for which SARVAM UCS waits for receiving DNIS from the network. This timer is loaded on receiving the Off-hook (I/C Seizure) on the receive channel (while receiving an incoming call). The Start Delay Timer ranges from 0001 to 9999 ms. Default: 20 msec.

- **Register Signaling Variant:** The Register Signaling Variant for T1/E1 Ports is set as DTMF.

- **Inbound ANI/DNIS Format:** Select the Inbound ANI/DNIS Format for T1/E1 Ports from the following options:
  - ANI
  - DNIS
  - ?ANI?
  - ?DNIS?
  - ?ANI?DNIS?
  - ?DNIS?ANI?

  Default: ?ANI?DNIS?.

- **Inbound Delimiter (?) Character:** Define the Inbound Delimiter Character in this field. Characters supported in this field are 0-9, #, *, A, B, C and D. Default: *

- **Outbound ANI/DNIS Format:** Select the Outbound ANI/DNIS Format for T1/E1 Port from the following options:
  - ANI
  - DNIS
  - ?ANI?
  - ?DNIS?
  - ?ANI?DNIS?
  - ?DNIS?ANI?

  Default: ?ANI?DNIS?.

- **Outbound Delimiter (?) Character:** Define the Outbound Delimiter Character in this field. Characters supported in this field are 0-9, #, *, A, B, C and D. Default: **

Configuring T1E1 Port Parameters using Jeeves

The commands explained below should be referred as:
• To program a single port: *XXXX-1*
• To program a range of ports: *XXXX-2*
• To program all the ports: *XXXX-**

Port Parameters

T1E1-1

Hardware Slot-Port

Use following command to assign hardware ID to a T1E1 software port.
1107-T1E1-Slot-Port offset

Where,
T1E1 is from 01 to 08.
Slot is the number of the universal slot, where the T1E1 Card is installed, from 01 to 16.
Port is the number of theT1E1 port on the card, from 01 to 32.

Use following command to de-assign the hardware slot and the hardware port assigned to the T1E1 software port.
1106-T1E1-00-00

Port Status

Use the following command to enable/disable the port:
6101-1-T1E1-Port Status
6101-2-T1E1-T1E1-Port Status
6101-*-Port Status
Where,
T1E1 is from 01 to 08.

Port Status	Meaning
0	Disable
1	Enable

By default, the T1E1 Port is enabled.

Name

Use the following command to assign a name to the port:
5407-1-T1E1-Name
5407-2-T1E1-T1E1-Name
5407-*-Name
Where,
T1E1 is from 01 to 08.
Name can be of upto 18 characters.

Carrier

Use the following command to select the carrier:
6108-1-T1E1-Carrier Type
6108-2-T1E1-T1E1-Carrier Type
6108-*-Carrier Type
Where,
T1E1 is from 01 to 08.

Carrier Type	Meaning
1	E1
2	T1

Line type

Use following command to program signaling type/ Line type of a T1E1:
6105-1-T1E1-Line Type
6105-2-T1E1-T1E1-Line Type

6105-*-Line Type
Where,
T1E1 is from 01 to 08.

Line Type	Meaning
1	ISDN_E1_PRI
2	ISDN_T1_PRI
3	ISDN_E1_CAS
4	ISDN_T1_RBS
5	ISDN_E1_QSIG
6	ISDN_T1_QSIG
7	ISDN_E1_E&M
8	ISDN_T1_E&M

By default, signaling type/ of a T1E1 is 1.

*DDI Routing is not supported on T1/E1 trunk line if you have selected **E&M** as the **Signal Type**.*

Orientation Type

Use following command to program 'Orientation Type' for the T1E1 port:
6106-1-T1E1-Orientation Type
6106-2.T1E1-T1E1-Orientation Type
6106-*-Orientation Type
Where,
T1E1=01 to 08.

Orientation	Meaning
1	Terminal
2	Network
3	Tie Line

By default Type = 1.

When Orientation = Terminal, the port will be regarded as trunk. All the trunk related parameters will be applicable.
When Orientation = Network, the port will be regarded as station. All the station related parameters will be applicable.
When Orientation = Tie-line, the port will be regarded as station for all IC calls to it and as trunk for all OG calls to be made through it.

Line Coding Mechanism

Use following command to program the Line Coding Mechanism for the T1E1 port:
6103-1-T1E1-Line Coding
6103-2-T1E1-T1E1-Line Coding
6103-*-Line Coding

Where,
T1E1 is from 01 to 08.

Line Coding	Meaning
1	AMI-Basic
2	B8ZS
3	CMI

By default, Line Coding is AMI-Basic.

Set Line Coding = AMI or B8ZS for T1 line. However, B8ZS is recommended.
Set Line Coding = AMI or HDB3 for E1 line. However, HDB3 is recommended.
CMI is used in Japan and since SARVAM UCS does not support Japan, this option will never be used.

Framing Mode

Use following command to program the Framing Mode for the T1E1 port:
6104-1-T1E1-Framing
6104-2-T1E1-T1E1-Framing
6104-*-Framing
Where,

Framing	Meaning
1	SF (D4) for T1
2	ESF for T1

By default, Framing is SF (D4).

Set Framing = SF or ESF for T1 line. However, ESF is recommended since it supports advanced features like CRC and FDL, which provide the performance reports.

Line Buildout Parameters

Use the following command to program the line build out parameters of a T1E1:
6162-1-T1E1-Code
6162-2-T1E1-T1E1-Code
6162-*-Code
Where,
T1E1 is from 01 to 08.

Code	Meaning
1	0-133ft
2	133-266ft
3	266-399ft
4	399-533ft
5	533-665ft

Code	Meaning
6	-7.5dB or equidistance
7	-16dB or equidistance
8	-22.5dB or equidistance

By default, the Line Build Out Parameters is 0-133ft.

DDI Routing

OG Reference ID

Use the following command to assign OG Reference ID to T1E1 port:
6131-1-T1E1-OG Reference ID
6131-2-T1E1-T1E1-OG Reference ID
6131-*-OG Reference ID
Where,
T1E1 is from 01 to 08.
OG Reference ID is from 00 to 99.
By default, OG Reference ID is 00.

IC Reference ID for Working Hour

Use the following command to assign IC Reference ID for Working Hour:
6132-1-T1E1-IC Reference ID
6132-2-T1E1-T1E1-IC Reference ID
6132-*-IC Reference ID
Where,
T1E1 is from 01 to 08.
IC Reference ID is from 00 to 99.
By default, IC Reference ID is 00.

IC Reference ID for Break Hour

Use the following command to assign IC Reference ID for Break Hour:
6133-1-T1E1-IC Reference ID
6133-2-T1E1-T1E1-IC Reference ID
6133-*-IC Reference ID
Where,
T1E1 is from 01 to 08.
IC Reference ID is from 00 to 99.
By default, IC Route Reference ID is 00.

IC Route Reference ID for Non-working Hour

Use the following command to assign IC Route Reference ID for Non-working Hour:
6134-1-T1E1-IC Reference ID
6134-2-T1E1-T1E1-IC Reference ID
6134-*-IC Reference ID
Where,
T1E1 is from 01 to 08.
IC Reference ID is from 00 to 99.
By default, IC Route ID is 00.

Templates

Trunk Feature Template

Use the following command to assign a Trunk Feature Template to the T1E1 Trunks, dial:
5806-1-T1E1- Trunk Feature Template Number to assign a template to a single T1E1 port.
5806-2-T1E1- Trunk Feature Template Number to assign the same template to a range of T1E1 ports.
5806-*- Trunk Feature Template Number to assign the same template to all T1E1 ports.
Where,
T1E1is the Software Port number of the port from 01 to 08.
Template Number is the number of the customized Trunk Feature Template, from 01 to 50. Default: Trunk Feature Template 01.

Station Basic Feature Template

To assign a Station Basic Feature Template to a T1E1PRI port, dial:
5506-1-T1E1PRI-Template Number to assign a template to a single T1E1 port.
5506-2-T1E1PRI-T1E1PRI-Template Number to assign the same template to a range of T1E1 ports.
5506-*-Template Number to assign the same template to all T1E1 ports.
Where,
T1E1PRI is the number of the T1E1PRI Software port, from 01 to 08.
Template Number is the number of the Station Basic Feature Template, from 01 to 50.
Default: Template 01 is assigned to all T1E1 ports.

Station Advance Features Template

To assign a Station Advanced Feature Template to a T1E1PRI port
5606-1-T1E1PRI-Template Number to assign a template to a single T1E1 port.
5606-2-T1E1PRI-T1E1PRI-Template Number to assign the same template to a range of T1E1 ports.
5606-*-Template Number to assign the same template to all T1E1 ports.
Where,
T1E1PRI is the number of the T1E1PRI Software port, from 01 to 08.
Template Number is the number of the Station Advanced Feature Template, from 01 to 50.
Default: Template 01 is assigned to all T1E1 ports.

Others

Priority

To assign priority to T1E1
3914-1-T1E1-Priority to assign a template to a single T1E1 port.
3914-2-T1E1PRI-T1E1PRI-Priority to assign the same priority to a range of T1E1 ports.
3914***-*-Template Number*** to assign the same Priority to all T1E1 ports.
Where,
T1E1PRI is the number of the T1E1PRI Software port, from 01 to 08.
Priority is from 1 to 9. Default: 5-Normal.

Cost Factor

Use following command to assign a name to the T1E1 port:
6102-1-T1E1-SP
6102-2-T1E1-T1E1-SP
6102-*-SP
Where,
T1E1 is from 01 to 08.

SP is from 01 to 99.
By default, Service Provider is 01.

Glare Option

Use following command to program Glare Option for the T1E1 port:
6112-1-T1E1-Glare Option
6112-2-T1E1-T1E1-Glare Option
6112-*-Glare Option
Where,
T1E1 is from 01 to 08.

Glare Option	Meaning
1	Proceed
2	Held Back

By default, Glare Option is 2.

Set Glare=Proceed, if SARVAM UCS is to be given priority in event of Glare.
Set Glare=Held Back, if the other end of the link is to be given priority in event of Glare.

The Glare settings should be complimentary on either side of the link.

Category (Logical Partitioning)

Use following command to set the Category (Logical Partitioning) for the T1E1 port:
6121-1-T1E1-Category
6121-2-T1E1-T1E1-Category
6121-*-Category
Where,
T1E1 is from 01 to 08.
Category is from 1 to 4.
By default, Category is set as 1.

Idle Code

Use the following command to program the Idle Code of a T1E1:
6113-1-T1E1-Idle Code
6113-2-T1E1-T1E1-Idle Code
6113-*-Idle Code
Where,
T1E1 is from 01 to 08.
Idle Code is from 000 to 255 (corresponding to 8 bits).
By default, the idle code is 127 (7F).
The binary equivalent of the programmed value (000 to 255) is sent on the channel to signify that the channel is idle. (or Unused) This setting depends on the network. Most commonly applicable values are 7F and FF (Binary equivalent is 0111 1111 and 1111 1111, decimal equivalent is 127 and 255).

Use message mode of the Digital Switch IC to send the idle channel code.

RCOC

To enable RCOC on T1E1 Trunk
Dial ***6145-1-T1E1-Code*** to enable the feature on a single trunk.
Dial ***6145-2-T1E1-T1E1-Code*** to enable the feature on a range of trunks.
Dial ***6145-*-Code*** to enable the feature on all trunks.
Where,
T1E1 is the software port number of the trunk from 01 to 08.
Code is
0 for Disable
1 for Enable
Default: Disable

Channels

Return Call to Original Caller (RCOC)

Use the following command to enable RCOC on T1E1 Trunk:
6145-1-T1E1-Code to enable the feature on a single trunk.
6145-2-T1E1-T1E1-Code to enable the feature on a range of trunks.
6145-*-Code to enable the feature on all trunks.
Where,
T1E1 is the software port number of the trunk from 01 to 08.
Code is
0 for Disable
1 for Enable
Default: Disable

Channel Reserved for Data Call

Use the following command to reserve channels for data transmission on T1E1:
6135-1-T1E1-Channel Count (Data)
6135-2-T1E1-T1E1-Channel Count (Data)
6135-*-Channel Count (Data)
Where,
T1E1 is from 01 to 08.
Channel Count (Data) is from 00 to 30.
By default, Channel Count for data transmission is 00.

Channel Reserved for Outgoing Call

Use the following command to program number of channels reserved for OG channel count:
6136-1-T1E1-Channel Count (OG)
6136-2-T1E1-T1E1-Channel Count (OG)
6136-*-Channel Count (OG)
Where,
T1E1 is from 01 to 08.
Channel Count (OG) is from 00 to 30. "It specifies the number of channels to be reserved for making an OG calls. For example, If OG channel count is programmed as 15, simultaneous 15 (maximum) OG calls can be made from the T1E1 port".
By default, OG Channel Count is 30.

Channel Reserved for Incoming Call

Use the following command to program number of channels reserved for IC channel count:
6137-1-T1E1-Channel Count (IC)
6137-2-T1E1-T1E1-Channel Count (IC)

6137-*-Channel Count (IC)
Where,
T1E1 is from 01 to 08.
Channel Count (IC) is from 00 to 30. "It specifies the number of channels to be reserved for making an IC calls. For example, If IC channel count is programmed as 10, simultaneous 10 (max.) IC calls can be received on the T1E1 port".
By default, Channel Count (IC) is 30.

Tone

Feed Dial Tone

Use following command to program the dial tone flag for T1E1 port:
6115-1-T1E1-Flag
6115-2-T1E1-T1E1-Flag
6115-*-Flag
Where,
T1E1 is from 01 to 08.

Flag	Meaning
0	Disable
1	Enable

By default, Dial Tone Flag is '0' for all the T1E1 ports.

Feed Routing Tone

Use following command to program the routing tone flag for T1E1 port:
6116-1-T1E1-Flag
6116-2-T1E1-T1E1-Flag
6116-*-Flag
Where,
T1E1 is from 01 to 08.

Flag	Meaning
0	Disable
1	Enable

By default, Routing Tone Flag is '0' for all the T1E1 ports.

Custom Pulse

T1/E1 Custom Pulse Width (CPW)

Use following command to enable/disable Custom Pulse Width (CPW) Flag for the T1E1 port for T1 signaling:
6171-1-T1E1-Flag
6171-2-T1E1-T1E1-Flag
6171-*-Flag
Where,

T1E1 is from 01 to 08.

Flag	Meaning
0	Disable
1	Enable

By default, Custom Pulse Width Flag is '0'.

T1/E1Custom Pulse Width Word 1

This command is used for Pulse shaping in first phase.

Use following command to program Custom Pulse Width Word 1 for the T1E1 port for T1 signaling:
6172-1-T1E1-Custom Pulse Width Word 1
6172-2-T1E1-T1E1-Custom Pulse Width Word 1
6172-*-Custom Pulse Width Word 1
Where,
T1E1 is from 01 to 08.
Custom Pulse Width Word 1 is from 000 to 127 volt.
By default, Custom Pulse Width Word 1 is 63 volt.

T1/E1Custom Pulse Width Word 2

This command is used for Pulse shaping in second phase.

Use following command to program Custom Pulse Width Word 2 for the T1E1 port for T1 signaling:
6173-1-T1E1-Custom Pulse Width Word 2
6173-2-T1E1-T1E1-Custom Pulse Width Word 2
6173-*-Custom Pulse Width Word 2
Where,
T1E1 is from 01 to 08.
Custom Pulse Width Word 2 is from 000 to 127 volt.
By default, Custom Pulse Width Word 2 is 58 volt.

T1/E1Custom Pulse Width Word 3

This command is used for Pulse shaping in third phase.

Use following command to program Custom Pulse Width Word 3 for the T1E1 port for T1 signaling:
6174-1-T1E1-Custom Pulse Width Word 3
6174-2-T1E1-T1E1-Custom Pulse Width Word 3
6174-*-Custom Pulse Width Word 3
Where,
T1E1 is from 01 to 08.
Custom Pulse Width Word 3 is from 000 to 127 volt.
By default, Custom Pulse Width Word 3 is 76 volt.

T1/E1Custom Pulse Width Word 4

This command is used for Pulse shaping in fourth phase.

Use following command to program Custom Pulse Width Word 4 for the T1E1 port for T1 signaling:
6175-1-T1E1-Custom Pulse Width Word 4
6175-2-T1E1-T1E1-Custom Pulse Width Word 4

6175-*-Custom Pulse Width Word 4
Where,
T1E1 is from 01 to 08.
Custom Pulse Width Word 4 is from 000 to 127 volt.
By default, Custom Pulse Width Word 4 is 00 volt.

Timer

Pause Timer

Use following command to program Pause Timer:
6109-1-T1E1- Pause Timer
6109-2-T1E1-T1E1-Pause Timer
6109-*-Pause Timer
Where,
T1E1 is from 01 to 08
Pause Timer is from 1 to 9 seconds
By default, Pause Timer is 3 seconds.

DTMF ON Time

Use following command to program DTMF ON Time:
6117-1-T1E1-DTMF ON Time
6117-2-T1E1-T1E1-DTMF ON Time
6117-*-DTMF ON Time
Where,
T1E1 is from 01 to 08.
DTMF ON Time is from 051 to 255 msec.
By default, DTMF ON Time is 102 msec.

DTMF Inter Digit Pause Timer

Use following command to program DTMF Inter digit Pause Timer:
6118-1-T1E1-DTMF Inter digit Pause Time
6118-2-T1E1-T1E1-DTMF Inter digit Pause Time
6118-*-DTMF Inter digit Pause Time
Where,
T1E1 is from 01 to 08
DTMF Inter digit Pause Time is from 051 to 255 msec.
By default, DTMF Inter digit Pause Time is 102 msec.

Gateway

Use Gateway Application - Answer Signaling?

Use following command to set flag for 'Gateway Application-Answer Signaling' on T1E1 trunk:
6119-1-T1E1-Gateway Application-Answer Signaling flag
6119-2-T1E1-T1E1-Gateway Application-Answer Signaling flag
6119-*- Gateway Application-Answer Signaling flag
Where,
T1E1 is from 01 to 08.

Flag	Meaning
0	Disable

Flag	Meaning
1	Enable

By default, Gateway Application-Answer Signaling flag is 'disable'.

Gateway Application - Answer Signaling DTMF Digits

Use following command to program DTMF digits string to be dialed as Gateway Application-Answer Signaling:
6120-1-T1E1-Gateway Application-Answer Signaling DTMF String
6120-2-T1E1-T1E1-Gateway Application-Answer Signaling DTMF String
6120-1-T1E1-Gateway Application-Answer Signaling DTMF String
Where,
T1E1 is from 01 to 08.
DTMF Digits allowed for DTMF string are from (0 - 9), *, #, A, B, C, D.
Maximum 4 DTMF digits can be programmed. If you need less than 4 digits for DTMF string, terminate the command using #*.

To program #, *, A, B, C, D use following codes:

Digit	Code for programming through command
A	#4
B	#5
C	#6
D	#7
*	**
#	##

By default Gateway Application-Answer Signaling DTMF String is 'CCC'.

Call Budget

Call Budget Type

To program Call Budget Type on T1E1 Port, dial:
6122-1-T1E1-Budget Type to program call budget type for a single trunk port.
6122-2-T1E1-T1E1-Budget Type to program the same call budget type for a range of trunk ports.
6122-*-Budget Type to program the same call budget type for all trunk ports.
Where,
T1E1is the number of the T1E1 software port from 01 to 08.
Budget Type is
0 for None
1 for Amount
2 for Minutes
3 for Number of Calls
By default, Budget Type is None.

Call Budget Amount

To program Call Budget Amount on T1E1 Port, dial:
6123-1-T1E1-Budget Amount to program call budget amount for a single trunk port.

6123-2-T1E1-T1E1-Budget Amount to program the same amount for a range of trunk ports.
6123-*-Budget Amount to program the same amount for all trunk ports.
Where,
T1E1is the number of the T1E1 software port from 01 to 08.
Budget Amount is of 6 digits max. Use leading zeros if amount to be programmed has fewer than 6 digits.
By default Budget Amount is 999999.

Call Budget Minutes

To program Call Budget Minutes on T1E1 Port:
6124-1-T1E1-Minutes to program minutes for a single trunk port.
6124-2-T1E1-T1E1-Minutes to program the same minutes for a range of trunk ports.
6124-*-Minutes to program the same minutes for all trunk ports.
Where,
T1E1is the number of the T1E1 software port from 01 to 08.
Minutes is of 6 digits max. Use leading zeros if Minutes to be programmed has less than 6 digits.
By default, Minutes is 999999.

Call Budget Number of Calls

To program Call Budget Number of Calls on T1E1 Port, dial:
6125-1-T1E1-Number of calls to program number of calls for a single trunk port.
6125-2-T1E1-T1E1-Number of calls to program the same number of calls for a range of trunk ports.
6125-*-Number of calls to program the same number of calls for all trunk ports.
Where,
T1E1is the number of the T1E1 software port from 01 to 08.
Number of Calls is of 4 digits from 0001 to 9999. Use leading zeros if number of calls to be programmed has fewer than 4 digits.
By default, Number of calls is 9999.

Call Budget Reset Mode

To program Call Budget Reset Mode for T1E1, dial:
6138-1-T1E1-Call Budget Reset Mode to program reset mode for a single trunk port.
6138-2-T1E1-T1E1-Call Budget Reset Mode to program the same reset mode for a range of trunk ports.
6138-*-Call Budget Reset Mode to program the same reset mode for all trunk ports.
Where,
T1E1is the number of the T1E1 software port from 01 to 08.
1 for Scheduled reset
2 for Manual reset
By default, Call Budget Reset Mode is Scheduled.

Call Budget Date for Scheduled Reset mode

To program the Date for Scheduled Reset mode for T1E1, dial:
6139-1-T1E1-Date to program date for a single trunk port.
6139-2-T1E1-T1E1-Date to program the date for a range of trunk ports.
6139-*-Date to program the date for all trunk ports.
Where,
T1E1is the number of the T1E1 software port from 01 to 08.
Date is
01 to 31 for Scheduled date to reset every month.
00 for Scheduled reset Daily.
By default, Reset date is 1st. of every month.

Call Back

Use the following commands to program Call Back on T1E1 Trunk ports. To know more about this feature, refer the topic Call Back on Trunk Ports.

To enable/disable Call Back on T1E1 port:
6176-1-T1E1- Call Back Flag
6176-2-T1E1-T1E1-Call Back Flag
6176- *-Call Back Flag
Where,
T1EI is from 01 to 08

Call Back Flag	Meaning
0	Disable
1	Enable

By default, Call Back flag is disabled.

Call Back Timer

To program Call Back Timer for T1E1 port:
6177-1-T1E1-Call Back Timer
6177-2-T1E1-T1E1-Call Back Timer
6177-*-Call Back Timer
Where,
T1EI is from 01 to 08
Pause Timer Range is from 01 to 99 Sec.
By default, Pause Timer is 10 Seconds

Call Back Mode

To program Call Back Mode on T1E1 port:
6178-1-T1E1-Call Back Mode
6178-2-T1E1-T1E1-Call Back Mode
6178-*-Call Back Mode
Where,
T1EI is from 01 to 08
Call Back Mode is 1 to 5.

Call Back Mode	Meaning
1	Built-In Auto Attendant
2	PIN Auth. - Multiple Calls
3	CLI Auth. - Multiple Calls
4	CLI Auth. - Single Call - Ans. Sig.
5	Operator

By default, Call Back mode is Operator

Call Back On

To program Call Back On method for T1E1 port:

6179-1-T1E1-Call Back on selection
6179-2-T1E1-T1E1-Call Back on selection
6179-*-Call Back on selection
Where,
T1E1 is from 01 to 08
Call back on selection is

Call Back on	Meaning
1	CLI Number
2	Alternate Number

By default, Call Back on selection is CLI Number.

Incoming Number List

To assign Call Back - Incoming Number List to a T1E1 port:
6180-1-T1E1-Incoming Number List
6180-2-T1E1-T1E1-Incoming Number List
6180-*-Incoming Number List
Where,
T1E1 is from 01 to 08
Incoming Number List is from 01 to 16.
By default, Incoming Number List is 15.

Outgoing Number List

To assign a Call Back - Outgoing Number List to a T1E1 port:
6147-1-T1E1-Outgoing Number List
6147-2-T1E1-T1E1-Outgoing Number List
6147-*-Outgoing Number List
Where,
T1E1 is from 01 to 08
Outgoing Number List is from 01 to 16.
By default, Outgoing Number List is 16.

Call Back From

To define Call Back From for a T1E1 port:
6148-1-T1E1-Call Back From
6148-2-T1E1-T1E1-Call Back From
6148-*-Call Back From
Where,
T1E1 is from 01 to 08
Call Back From is
1 for Same Port
2 for OGTB Group
By default, Same Port is selected as Call Back From.

OGTB Group

To assign a Call Back - OGTB Group for a T1E1 port:
6149-1-T1E1-OGTB Group
6149-2-T1E1-T1E1-OGTB Group
6149-*-OGTB Group

Where,
T1E1 is from 01 to 08
OGTB Group is from 01 to 32
By default, OGTBG is 01.

FDL

FDL Flag - copied from T1 Maintenance

Use following command to enable/disable T1 FDL on a T1E1PRI port:
6164-1-T1E1PRI-T1 FDL
6164-2-T1E1PRI-T1E1PRI-T1 FDL
6164-*-T1 FDL
Where,
T1E1PRI is from 01 to 08.

T1 FDL	Meaning
0	Disable
1	Enable

By default, the T1 FDL is disabled.

FDL Protocol

Use following command to program the T1 FDL Protocol for a T1E1PRI port:
6165-1-T1E1PRI-T1 FDL Protocol
6165-2-T1E1PRI-T1E1PRI-T1 FDL Protocol
6165-*-T1 FDL Protocol

Where,
T1E1PRI is from 01 to 08.

T1 FDL Protocol	Meaning
0	Disable
1	AT&T 54016
2	ANSI T1.403

By default, the T1 FDL Protocol is ANSI T1.403.

Refer “E1/T1 Maintenance”

Debug

SARVAM UCS supports debug of parameters (debug codes) depending on the Level of debug. On issuing this command the SARVAM UCS Card T1E1 will send the debug details to the COM port of the T1E1 port.

Debug Level	Command
Debug Level-1	***6191-1-T1E1-1-XXX***

Debug Level	Command
Debug Level-2	***6191-1-T1E1-2-XXX***
Debug Level-3	***6192-1-T1E1-1-XXX***
Debug Level-4	***6192-1-T1E1-2-XXX***

Option 1
Use following command to start/stop debug the parameters for the T1E1 port:
6191-1-T1E1-Level-Debug Code
6191-2-T1E1-T1E1-Level-Debug Code
6191-*-Level-Debug Code
Where,
T1E1 Port is from 01 to 08.
Level is from 1 to 4 (As shown below).
Code is the value for the specified level to turn ON the debug for the parameters. Code range is from 000 to 255.
Code value '000' for each level will turn off that level's debug.

Level 1:

Unused	Unused	Unused	Unused	Layer 4	CAS DSP	MFC R2	CAS

001	CAS
002	MFC R2
004	CAS DSP
008	Layer 4
000	Debug Off

Level 2:

Unused	Unused	Unused	HDLC (D-Channel)	FDL	ABCD Bits	Counters	Alarms

001	Alarms
002	Counters
004	ABCD Bits
008	FDL
016	HDLC (D Channel)
000	Debug Off

Level 3:

Unused	Flow Debug	NLS Debug	LAP Debug	SVC Primitives	Variables	State	Primitives

001	Primitives
002	State

004	Variables
008	SVC Primitives
016	LAP Debug
032	NLS Debug
064	Flow Debug
000	Debug Off

Level 4:

Unused	Unused	Unused	Unused	Unused	Unused	NI Debug	OS Task

001	OS Task
002	NI Debug
000	Debug Off

Default: Debug Code = ‘Debug OFF’ for all T1E1 ports for all levels.

T1E1:1 - PRI/QSIG Signaling

ISDN Switch Variant

Use following command to program the ISDN PRI Switch Variant:
6107-1-T1E1-ISDN Switch Variant
6107-2-T1E1-T1E1-ISDN Switch Variant
6107-*-ISDN Switch Variant
Where,
T1E1 is from 01 to 08.

ISDN PRI Variant	Meaning
1	ATT 4ESS
2	ATT 5ESS
3	Australia
4	DMS
5	ETSI NET5
6	NTT INS64
7	SWV Hongkong
8	US NI12
9	QSIG E1
10	QSIG T1

By default, ISDN PRI Switch Variant of a T1E1 is ETSI NET5.

Send Called Party Number Using

Use the following command to program overlap receiving timer:
6146-1-T1E1-Type
6146-2-T1E1-T1E1-Type
6146-*-Type
Where,
T1E1 is from 01 to 08.
Type is
1 for Keypad Facility IE
0 for Called Party IE
Default: Called Party Number IE

Overlap Receiving Timer

Use the following command to program overlap receiving timer:
6114-1-T1E1-Timer
6114-2-T1E1-T1E1-Timer
6114-*-Timer
Where,
T1E1 is from 01 to 08.
Timer is from 00 to 99.
By default, Overlap Receiving Timer is 15 seconds.

Caller-Type of Number (TON)

Use following command to program the Caller-Type of Number (TON) for a T1E1:
6126-1-T1E1-Source TON
6126-2-T1E1-T1E1-Source TON
6126-*-Source TON
Where,
T1E1 is from 01 to 08.

Source TON	Meaning
1	Unknown: This is used when the user or network has no a prior information about the numbering plan. In this case, the Address Value field is organized according to the network dialing plan. For example, prefix or escape digits might be present.
2	International Number.
3	National Number: Prefix or escape digits shall not be included.
4	Network Specific Number: This is used to indicate administration/Service number specific to the serving network. For example, used to access an operator.
5	Subscriber Number: This is used when a specific short number representation is stored in one or more SCs as part of a higher layer application.
6	Abbreviated Number.
7	Reserved Number.

By default, Caller-Type of Number (TON) of T1E1 is 1.

Caller-Numbering Plan Identification (NPI)

Use following command to program Caller-Numbering Plan Identification (NPI) for T1E1:
6127-1-T1E1-Source NPI
6127-2-T1E1-T1E1-Source NPI

6127-*-Source NPI
Where,
T1E1 is from 01 to 08.

Source NPI	Meaning
1	Unknown
2	ISDN Numbering Plan
3	Data Numbering Plan
4	Telex Numbering
5	National Numbering Plan
6	Private Numbering Plan
7	Reserved for Extension

By default, Caller-Numbering Plan Identification (NPI) for T1E1 is 2.

Called-Type of Number (TON)

Use following command to program Called-Type of Number (TON) for a T1E1:
6128-1-T1E1-Destination TON
6128-2-T1E1-T1E1-Destination TON
6128-*-Destination TON
Where,
T1E1 is from 01 to 08.

Destination TON	Meaning
1	Unknown: This is used when the user or network has no a prior information about the numbering plan. In this case, the Address Value field is organized according to the network dialing plan. For example, prefix or escape digits might be present.
2	International Number.
3	National Number: Prefix or escape digits shall not be included.
4	Network Specific Number: This is used to indicate administration/Service number specific to the serving network. For example, used to access an operator.
5	Subscriber Number: This is used when a specific short number representation is stored in one or more SCs as part of a higher layer application.
6	Abbreviated Number.
7	Reserved Number.

By default, Called-Type of Number (TON) of T1E1 is 1.

Called-Numbering Plan Identification (NPI)

Use following command to program Called-Numbering Plan Identification (NPI) for T1E1:
6129-1-T1E1-Destination NPI
6129-2-T1E1-T1E1-Destination NPI
6129-*-Destination NPI
Where,

T1E1 is from 01 to 08.

Destination NPI	Meaning
1	Unknown
2	ISDN Numbering Plan
3	Data Numbering Plan
4	Telex Numbering
5	National Numbering Plan
6	Private Numbering Plan
7	Reserved for Extension

By default, Called-Numbering Plan Identification (NPI) for T1E1 is 2.

Receive Equalization Mode

Use the following command to program auto receive equalization mode:
6110-1-T1E1-Mode
6110-2-T1E1-T1E1-Mode
6110-*-Mode
Where,
T1E1 is from 01 to 08.

Mode	Meaning
0	Manual
1	Auto

By default, Receive Equalization Mode is 1.

Receive Equalization Parameters

Use the following command to program the receive equalization parameters of a T1E1:
6111-1-T1E1-Receive Equalization Parameters
6111-2-T1E1-T1E1-Receive Equalization Parameters
6111-*-Receive Equalization Parameters
Where,
T1E1 is from 01 to 08.

Receive Equalization Parameters	Meaning
1	None
2	8 dB
3	16 dB
4	24 dB
5	32 dB
6	40 dB
7	48 dB

By default, the receive equalization parameters of T1E1 is 1.

Feed Inband Tones on T1E1-NT, before sending DISCONNECT

Use following command to program to select whether the inband tones should be feed on T1E1-NT before sending DISCONNECT message?
6130-1-T1E1-Flag
6130-2-T1E1-T1E1-Flag
6130-*-Flag
Where,

Flag	Meaning
0	No
1	Yes

Default = NO.

E&M Signaling

E&M Feature Template

Assign E&M Feature Template to T1E1 using following command:
Use Following command to assign E&M Feature Template to T1E1 Port
6004-1-T1E1-Template Number
6004-2-T1E1-T1E1- Template Number
6004-*- Template Number
Where,
T1E1 is from 01 to 08.
Template Number is 01 to 50.
By default, Template 01 is assigned to T1E1.

Now proceed to program other parameters for E&M on T1E1 using following commands:

B Bit Pattern

Use following command to select B Bit Pattern
7191-1-T1E1-Code
7191-2-T1E1-T1E1-Code
7191-*-Code
Where,
T1E1 is from 01 to 08.

Code	Meaning
1	Same as A bit
2	Fixed Value

By default, the Code is 1 (Same as A bit)

B Bit Value

Use following command to program B Bit Value
7192-1-T1E1-B Bit Value

7192-2-T1E1-T1E1- B Bit Value
7192-*- B Bit Value
Where,
T1E1 is from 01 to 08.
B bit value can be 0 or 1.
By default, B bit value is 0.

CD Bit Value

Use following command to program CD Bit Value
7193-1-T1E1-CD Bit Value
7193-2-T1E1-T1E1-CD Bit Value
7193-*- CD Bit Value
Where,
T1E1 is from 01 to 08.
CD bit value can be 1 or 3.
By default, CD bit value is 1.

Invert Bit A

Use following command to program to invert/don't invert Bit A for the T1E1 port:
7162-1-T1E1-Invert Bit A
7162-2-T1E1-T1E1-Invert Bit A
7162-*-Invert Bit A
Where,
T1E1 is from 01 to 08.

Invert Bit A	Meaning
0	Disable
1	Enable

By default, Invert Bit A is 0.

Invert Bit B

Use following command to program to invert/don't invert Bit B for the T1E1 port:
7163-1-T1E1-Invert Bit B
7163-2-T1E1-T1E1-Invert Bit B
7163-*-Invert Bit B
Where,
T1E1 is from 01 to 08.

Invert Bit B	Meaning
0	Disable
1	Enable

By default, Invert Bit B is 0.

Invert Bit C

Use following command to program to invert/don't invert Bit C for the T1E1 port:
7164-1-T1E1-Invert Bit C
7164-2-T1E1-T1E1-Invert Bit C

7164-*-Invert Bit C
Where,
T1E1 is from 01 to 08.

Invert Bit C	Meaning
0	Disable
1	Enable

By default, Invert Bit C is 0.

Invert Bit D
Use following command to program to invert/don't invert Bit D for the T1E1 port:
7165-1-T1E1-Invert Bit D
7165-2-T1E1-T1E1-Invert Bit D
7165-*-Invert Bit D

Where,
T1E1 is from 01 to 08.

Invert Bit D	Meaning
0	Disable
1	Enable

By default, Invert Bit D is 0.

RBS Signaling

To assign a Line Signaling Variants to the T1E1 port, dial:
6181-1-T1E1-T1 Line Signaling Variants
6181-2-T1E1-T1E1-T1 Line Signaling Variants
6181-*-T1 Line Signaling Variants
Where,
T1E1 is from 1 to 7.

T1 Line Signaling Variants	Meaning
1	SLT Loop Start
2	CO Loop Start
3	SLT Ground Start
4	CO Ground Start
5	E&M Immediate Start/Dial
6	E&M Wink Start
7	E&M Wink Start FGD

Default: E&M Wink Start.

To set the Wink Timer, dial:

6182-1-T1E1-Wink Timer
6182-2-T1E1-T1E1-Wink Timer
6182-*-Wink Timer
Where,
T1E1 is from 01 to 08.
Wink Timer is from 0001 to 9999 ms.
By default, T1 Wink Timer is 0200 ms.

Use the following command to program the T1 wink wait timer for T1E1:
6183-1-T1E1-Wink Wait Timer
6183-2-T1E1-T1E1-Wink Wait Timer
6183-*-Wink Wait Timer
Where,
T1E1 is from 01 to 08.
Wink Wait Timer is from 0001 to 9999 ms.
By default, T1 Wink Wait Timer is 0200 ms.

Use the following command to program the T1 wait wink timer for T1E1:
6184-1-T1E1-Timer
6184-2-T1E1-T1E1-Timer
6184-*-Timer
Where,
T1E1 is from 01 to 08.
Timer is from 0001 to 9999 ms.
By default, T1 Wait Wink Timer is 5000 ms.

Use the following command to program the T1 delay duration for T1E1:
6185-1-T1E1-Delay Duration
6185-2-T1E1-T1E1-Delay Duration
6185-*-Delay Duration
Where,
T1E1 is from 01 to 08.
Delay Duration is from 0001 to 9999 ms.
By default, T1 Delay Duration is 0140 ms.

Use the following command to program the T1 Start Delay Timer for T1E1:
6186-1-T1E1-Start Delay Timer
6186-2-T1E1-T1E1-Start Delay Timer
6186-*-Start Delay Timer
Where,
T1E1 is from 01 to 08.
Start Delay Duration is from 001 to 255 seconds.
By default, T1 Start Delay Timer is 020 seconds.

To assign a Register Signaling Variant to the T1E1 port, dial:
6161-1-T1E1-T1 Register Signaling Variant
6161-2-T1E1-T1E1-T1 Register Signaling Variant
6161-*-T1 Register Signaling Variant
Where,

T1E1 is from 01 to 08.

T1 Register Signaling Variant	Meaning
1	T1 RBS DTMF

T1 RBS DTMF: DNIS is transmitted in the corresponding speech channel using the DTMF signals as per the ITU-T Q.23.

To select the Inbound ANI/DNIS Format, dial:
6166-1-T1E1-Code
6166-2-T1E1-T1E1-Code
6166-*-Code
Where,
T1E1 is from 01 to 08.
Code is

T1 Register Signaling Variant	Code
ANI	1
DNIS	2
?ANI?	3
?DNIS?	4
?ANI?DNIS?	5
?DNIS?ANI?	6

By default,?ANI?DNIS? is selected.

To assign the Inbound Delimiter (?) character, dial:
6167-1-T1E1-Character
6167-2-T1E1-T1E1-Character
6167-*-Character
Where,
T1E1 is from 01 to 08.
Character can be #, *, A,B,C,D and 0-9.
By default, * is set.

To assign the Outbound ANI/DNIS Format, dial:
6168-1-T1E1-Code
6168-2-T1E1-T1E1-Code
6168-*-Code
Where,
T1E1 is from 01 to 08.
Code is

T1 Register Signaling Variant	Code
ANI	1
DNIS	2
?ANI?	3

T1 Register Signaling Variant	Code
?DNIS?	4
?ANI?DNIS?	5
?DNIS?ANI?	6

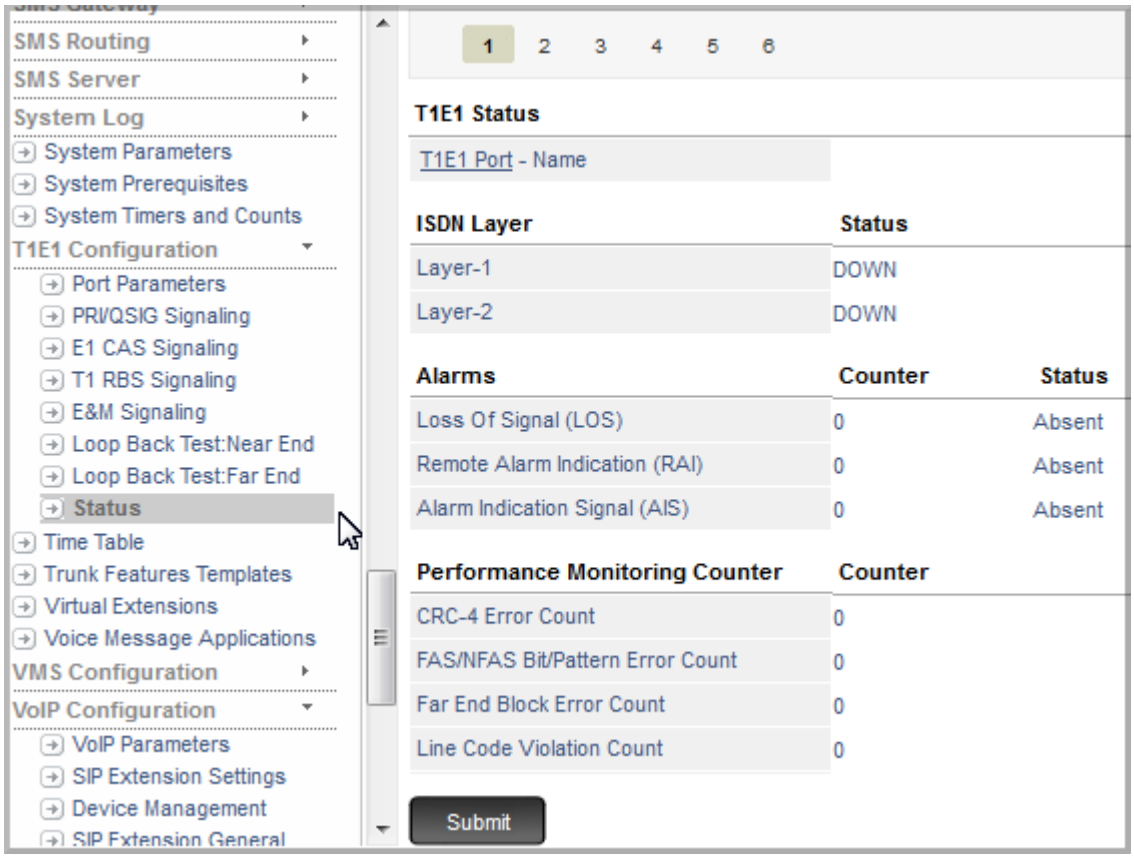
By default,?ANI?DNIS? is selected.

To assign the Outbound Delimiter (?) character
6169-1-T1E1-Character
6169-2-T1E1-T1E1-Character
6169-*-Character
Where,
T1E1 is from 01 to 08.
Character can be #, *, A,B,C,D and 0-9.
By default, * is set.

Viewing T1E1 Trunk Status

You can view the status of T1E1 Trunks on Jeeves only. To do this,

- Under **T1E1 Configuration**, click **Status**.



- For each T1E1 Trunk, the following settings will be displayed:
  - T1E1 Port No.
  - Name
  - Layer 1

- Layer 2
- Alarms
- Performance Monitoring Counter
- Call Budget Type
- Allotted Amount/Minutes/Calls
- Consumed Amount/Minutes/Calls
- Call Budget Reset Mode
- Call Budget Reset Scheduled (Date)
- Reset Consumed (this is not a status indicator. It is for resetting the Consumed Call Budget manually)

*You can also view the T1E1 Trunk Status from the **Status** link. To view, click the T1E1 link under Status.*

T1 RBS Parameters

What's this?

Some countries like North America support the standard of 1.544Mbps of PCM trunk. This is known as T1 Trunks. The T1 type of PCM Trunks use Robbed Bit Signaling. ROBBED-BIT signaling is a per-channel signaling technique for transmitting signaling bits on each channel in a T1E1 facility. The least-significant bit in every 6th transmitted information frame is removed and replaced by a signaling bit. This technique is also called in-band signaling. The maximum transmission rate for each bearer channel with ROBBED-BIT signaling is 56 Kbps.

ISDN-PRI signaling is carried on the 24th channel for a 1.544 Mbps connection and on the 16th channel for a 2.048 Mbps connection. There are two types of parameters:

- Line Signaling (ABCD Bits)
- Register Signaling

Line signaling is described by following types:

- E&M Wink Start FGD
- E&M Wink Start
- E&M Immediate Dial/Start
- SLT Ground start
- SLT Loop Start
- CO Loop Start
- CO Ground Start

T1 Line signaling type is applicable when the Line Type is programmed as T1 RBS for the T1E1 Port.

The Register signaling supported by SARVAM UCS is DTMF.

T1 Line signaling is applicable when the Line Type is programmed as ISDN_T1_RBS for the T1E1 Port. Refer chapter "Configuring T1 Trunks" to program Line Type as T1 RBS.

The various Line Signaling Variant are explained below:

E&M Wink Start

Bits A and B are set to 1 to indicate Off-hook. Bits A and B are set to 0 to indicate On-Hook. Bits C and D follow bits A and B. The application for making calls and receiving calls is explained below:

Making an OG Call

- To transmit Off-Hook, bits A, B, C and D on the transmit channel are set to 1.
- The far end (network) sends a wink (a momentary OFF-Hook for 200ms), that is, bits A, B, C and D on the receive channel receive a pulse (Active High) of 200ms.
- On receipt of the wink signal, DNIS (Dialed Number Identification Signal) is sent on the speech channels using the Register Signaling type (DTMF or Decadic (A-Bit) or R1 MFC or R2 MFC).

DNIS is Dialed Number In Service. It is the ISDN number that is being dialed. This is provided by the telco in the call setup messages. DNIS can be used to provide differentiated service to dialing users.

The call goes through when the called party answers the call.

Receiving an IC Call

- Bit A, B, C and D =1 are received on the receive channel.
- The System sends a wink.
- The far end sends the DNIS in the speech channels using Register Signaling.

Disconnect

- By the Network-Bits A and B on the receive channel are 0. Bits C and D are also 0 in ESF. Following this, the bits on the transmit channel are set to 0 by the System.
- By the System-Bit A and B on the transmit channel are set to 0. Bit C and D are also set to 0 in case of ESF. Following this, the bits A and B (C and D in ESF) are received as 0 on the receive channel.

E&M Wink Start FGD

- Bits A and B are set to 1 to indicate OFF-Hook. Bits A and B are set to 0 to indicate ON-Hook. Bits C and D follow bits A and B (Incase of ESF).

Making an OG Call

- To transmit OFF-Hook, bits A, B, C and D on the transmit channel are set to 1.
- The far end (network) sends a wink (a momentary OFF-Hook for 200ms.), that is, bits A, B, C and D on the receive channel receive a pulse (Active High) of 200ms.
- On receipt of the wink signal, DNIS (Dialed Number Identification Signal). It is the ISDN number that is being dialed. This is provided by the telco in the call setup messages. DNIS can be used to provide differentiated service to dialing users.) is sent on the speech channels using the Register Signaling type.
- The call goes through when the called party answers the call.

Receiving an IC Call

- Bit A, B, C and D =1 are received on the receive channel.
- The System sends a wink.
- The far end sends the DNIS in the speech channels using DTMF.
- On expiry of DTMF Inter-digit timer, the System sends a wink again.
- The System goes off hook when the call is answered by the called party.

Disconnect

- By the Network-Bits A and B on the receive channel are 0. Bits C and D are also 0 in ESF. Following this, the bits on the transmit channel are set to 0 by the System.
- By the System-Bit A and B on the transmit channel are set to 0. Bit C and D are also set to 0 in case of ESF. Following this, the bits A and (C and D in ESF) are received as 0 on the receive channel.

Please note that while using T1 RBS, only DID is being sent/received and not the CLI/ANI.

E&M Immediate Start/Dial

Same as E&M Wink Start. But here the O/G side sends the DNIS immediately after sending Off-hook on its transmit channel.

SLT Loop Start

The T1E1 port acts as a SLT port whereas the network acts as CO port. When configured for this type of signaling, the SLT side (the System) uses the A-bit whereas the CO side (the Network) uses the B-bit. Bits C and D follow A and B bits.

Direction	State	A	B	C	D
Transmit	ON-Hook	0	1	0	1
Transmit	OFF-Hook/Loop Closed	1	1	1	1
Receive	ON-Hook	0	1	0	1
Receive	OFF-Hook	0	1	0	1
Receive	Ringing	1	1	1	1

Making an OG Call

- To transmit OFF-Hook, bits A (and Bit C if ESF) on the transmit channel is set to 1.
- The far end provides dial tone. There is no signaling change.
- The SARVAM UCS sends the DNIS (Dialed Number Identification Signal) on the speech channels using the DTMF signals.
- The call goes through when the called party answers the call.

Receiving an IC Call

- Bit B (and bit D if ESF) toggles as per the ringing pattern.
- The Network sends the DNIS on the speech channel using DTMF signals.
- The System detects toggling of bit B. When the called station of the System answers, the System transmits Off-hook state by changing bit-A from 0 to 1.

Disconnect

By the Network:

- No indication from the Network. The System will detect error tone to detect a disconnect from the network. On detecting on-hook from the network, the System transmits on-hook by setting Bit A from 1 to 0.

By the System:

- The System transmits on-hook by setting Bit A from 1 to 0.

SLT Ground Start

The T1E1 port acts as a SLT port whereas the network acts as CO port. When configured for this type of signaling, the SLT side (the System) uses the A-bit (however, the System can use B-bit but generally it is ignored by the Network) whereas the CO side (the Network) uses the A and B-bits both. Bits C and D follow A and B bits.

Direction	State	A	B	C	D
Transmit	ON-Hook/Loop Open	0	1	0	1
Transmit	Ground on Ring	0	0	0	0
Transmit	OFF-Hook/Loop Closed	1	1	1	1
Receive	ON-Hook/No TIP Ground	1	1	1	1
Receive	OFF-Hook/TIP Ground	0	1	0	1
Receive	Ringing	0	0	0	0

Making an OG Call

- To transmit Off-hook, bits A (and Bit C if ESF) and B (and Bit C if ESF) on the transmit channel is set to 0.
- The network detects this change and goes off-hook. The A-bit on the receive channel goes from 1 to 0. The B-bit is set to 1.
- Bit A and Bit B are set to 1.
- Network sends dial tone.
- System detects dial tone and sends DNIS (DTMF digits) on the corresponding speech channel.
- The call goes through when the called party answers the call.

Receiving an IC Call

- Bit A (and bit C if ESF) goes from 1 to 0.
- Bit B (and bit D if ESF) toggles as per the ringing.
- The Network sends the DNIS on the speech channel using DTMF signals.
- The System detects toggling of bit B and the seizure on Bit A.
- When the called station of the System answers, the PBS transmits Off-hook state by changing bit-A from 0 to 1.
- The Network stops toggling of bit B. Bit B is set to 1.

Disconnect

- By the Network-A bit on the receive channel of the System goes from 0 to 1. On detecting on-hook from the network, the System transmits on-hook by setting Bit A from 1 to 0.
- By the System-The System transmits on-hook by setting Bit A from 1 to 0. On detecting on-hook from the system, the network transmits on-hook by setting Bit A from 0 to 1.

- CO Loop Start or CO Ground Start is used when the SARVAM UCS is connected to the Network, that is, when the T1E1 port is configured for Terminal mode (Connection mode).

- Whereas SLT Loop Start or SLT Ground Start is used when the SARVAM UCS is connected to another SARVAM UCS, that is, when the T1E1 port is configured for Network mode.

CO Loop Start-This is complementary to SLT Loop Start explained above.
CO Ground Start-This is complementary to SLT Ground Start explained above.

Configuring T1E1 Port Parameters using Jeeves

- Log into Jeeves as System Engineer.

- Under **Configuration**, click **T1E1 Configuration**.

- Click **T1 RBS Parameters** to open the page and configure the following:

  - **Line Signaling Variant:** Select the T1 Line Signaling Variant from following options:
    - SLT Loop Start
    - CO Loop Start
    - SLT Ground Start
    - CO Ground Start
    - E&M Immediate Dial/Start
    - E&M Wink Start
    - E&M Wink Start FGD

    Default: E&M Wink Start FGD.

  - **Wink Timer (milliseconds)**: Wink timer refers to the momentary Off-Hook condition to acknowledge end of making an outgoing call. The Wink Timer ranges from 0001 ms to 9999 ms. Default: 160 msec.

  - **Wink Wait Timer (milliseconds):** Wink Wait Timer signifies the maximum time the system should wait before sending a wink start signal after an incoming seizure is detected. Wink Wait Timer ranges from 0001 to 9999 msec. Default: 30msec.

Ensure that this timer is greater than the Wink Wait Timer of the other end.

  - **Wait Wink Timer (milliseconds):** Wait Wink Timer signifies the time for which SARVAM UCS will wait for receiving the DNIS after sending the outgoing seizure signal. Wait Wink Timer ranges from 0001 to 9999 msec. Default: 5000 msec.

Make sure that this timer is greater than the Wait Wink Timer of the other end.

  - **Delay Duration (milliseconds):** This duration signifies the time after which the DNIS information is to be sent while making an outgoing call. Range of the Delay Duration is from 0001 to 9999 msec. Default: 100 msec.

  - **Start Delay Timer:** Start Delay Timer signifies the time for which SARVAM UCS waits for receiving DNIS from the network. This timer is loaded on receiving the Off-hook (I/C Seizure) on the receive channel (while receiving an incoming call). The Start Delay Timer ranges from 0001 to 9999 ms. Default: 20 msec.

  - **Register Signaling Variant:** The Register Signaling Variant for T1/E1 Ports is set as DTMF.

- **Inbound ANI/DNIS Format:** Select the Inbound ANI/DNIS Format for T1/E1 Ports from the following options:
  - ANI
  - DNIS
  - ?ANI?
  - ?DNIS?
  - ?ANI?DNIS?
  - ?DNIS?ANI?

  Default: ?ANI?DNIS?.

- **Inbound Delimiter (?) Character:** Define the Inbound Delimiter Character in this field. Characters supported in this field are 0-9, #, *, A, B, C and D. Default: *

- **Outbound ANI/DNIS Format:** Select the Outbound ANI/DNIS Format for T1/E1 Port from the following options:
  - ANI
  - DNIS
  - ?ANI?
  - ?DNIS?
  - ?ANI?DNIS?
  - ?DNIS?ANI?

  Default: ?ANI?DNIS?.

- **Outbound Delimiter (?) Character:** Define the Outbound Delimiter Character in this field. Characters supported in this field are 0-9, #, *, A, B, C and D. Default: **

Configuring T1E1 Port Parameters using Telephone

- Enter SE mode from a DKP/SLT.

  - To assign a Line Signaling Variants to the T1E1 port, dial:
    6181-1-T1E1-T1 Line Signaling Variants
    6181-2-T1E1-T1E1-T1 Line Signaling Variants
    6181-*-T1 Line Signaling Variants
    Where,
    T1E1 is from 01 to 08.

T1 Line Signaling Variants	Meaning
1	SLT Loop Start
2	CO Loop Start
3	SLT Ground Start
4	CO Ground Start
5	E&M Immediate Start/Dial
6	E&M Wink Start
7	E&M Wink Start FGD

    Default: E&M Wink Start.

- To set the Wink Timer, dial:
  6182-1-T1E1-Wink Timer
  6182-2-T1E1-T1E1-Wink Timer
  6182-*-Wink Timer
  Where,
  T1E1 is from 01 to 08.
  Wink Timer is from 0001 to 9999 ms.
  By default, T1 Wink Timer is 0200 ms.

- To configure the T1 wink wait timer for T1E1, dial:
  6183-1-T1E1-Wink Wait Timer
  6183-2-T1E1-T1E1-Wink Wait Timer
  6183-*-Wink Wait Timer
  Where,
  T1E1 is from 01 to 08.
  Wink Wait Timer is from 0001 to 9999 ms.
  By default, T1 Wink Wait Timer is 0200 ms.

- To program the T1 wait wink timer for T1E1, dial:
  6184-1-T1E1-Timer
  6184-2-T1E1-T1E1-Timer
  6184-*-Timer
  Where,
  T1E1 is from 01 to 08.
  Timer is from 0001 to 9999 ms.
  By default, T1 Wait Wink Timer is 0200 ms.

  To program the T1 delay duration for T1E1, dial:
  6185-1-T1E1-Delay Duration
  6185-2-T1E1-T1E1-Delay Duration
  6185-*-Delay Duration
  Where,
  T1E1 is from 01 to 08.
  Delay Duration is from 0001 to 9999 ms.
  By default, T1 Delay Duration is 0140 ms.

- To program the T1 Start Delay Timer for T1E1, dial:
  6186-1-T1E1-Start Delay Timer
  6186-2-T1E1-T1E1-Start Delay Timer
  6186-*-Start Delay Timer
  Where,
  T1E1 is from 01 to 08.
  Start Delay Duration is from 001 to 255 seconds.
  By default, T1 Start Delay Timer is 020 seconds.

- To assign a Register Signaling Variant to the T1E1 port, dial:
  6161-1-T1E1-T1 Register Signaling Variant
  6161-2-T1E1-T1E1-T1 Register Signaling Variant
  6161-*-T1 Register Signaling Variant
  Where,

T1E1 is from 01 to 08.

T1 Register Signaling Variant	Meaning
1	T1 RBS DTMF

T1 RBS DTMF: DNIS is transmitted in the corresponding speech channel using the DTMF signals as per the ITU-T Q.23.

- To select the Inbound ANI/DNIS Format, dial:
  6166-1-T1E1-Code
  6166-2-T1E1-T1E1-Code
  6166-*-Code
  Where,
  T1E1 is from 01 to 08.
  Code is

T1 Register Signaling Variant	Code
ANI	1
DNIS	2
?ANI?	3
?DNIS?	4
?ANI?DNIS?	5
?DNIS?ANI?	6

By default, ?ANI?DNIS? is selected.

- To assign the Inbound Delimiter (?) character, dial:
  6167-1-T1E1-Character
  6167-2-T1E1-T1E1-Character
  6167-*-Character
  Where,
  T1E1 is from 01 to 08.
  Character can be #, *, A,B,C,D and 0-9.
  By default, * is set.

- To assign the Outbound ANI/DNIS Format, dial:
  6168-1-T1E1-Code
  6168-2-T1E1-T1E1-Code
  6168-*-Code
  Where,
  T1E1 is from 01 to 08.
  Code is

T1 Register Signaling Variant	Code
ANI	1
DNIS	2
?ANI?	3

T1 Register Signaling Variant	Code
?DNIS?	4
?ANI?DNIS?	5
?DNIS?ANI?	6

By default, ?ANI?DNIS? is selected.

- To assign the Outbound Delimiter (?) character, dial:
  6169-1-T1E1-Character
  6169-2-T1E1-T1E1-Character
  6169-*-Character
  Where,
  T1E1 is from 01 to 08.
  Character can be #, *, A,B,C,D and 0-9.
  By default, * is set.

- Exit SE mode.

Configuring E1 Trunks

What's this?

Digital Signal Level 1 (T1E1) trunks use Bit-Oriented Signaling (BOS) and multiplexes 24 channels (T1 service) or 32 channels (E1 service) into a single data stream. T1E1 can be used for voice or voice-grade data and for data-transmission protocols. T1 trunk service multiplexes 24 channels into a single 1.544-Mbps data stream. E1 trunk service multiplexes 32 channels into a single 2.048-Mbps stream. Both T1 and E1 provide a digital interface for trunk groups.

Signaling Modes

Common Channel Signaling (CCS) is an industry-standard technique where any one of a group of channels carries the signals for the other channels. Matrix uses the 24th channel of a group for signaling. This signaling technique differs from 24-channel signaling. When the system is configured for Facility-Associated Signaling, 24-channel signaling uses the 24th channel in a T1E1 facility to carry signals. This technique also is called clear channel, out-of-band or alternate voice data (AVD) signaling.

Channel Associated Signaling (CAS) is similar to common-channel signaling and is used only when the Bit Rate is 2.048 Mbps (the trunk is used with an E1 interface). Signaling is carried on the 16th channel.

Common-channel signaling and channel associated signaling provide a maximum transmission rate of 64 Kbps for bearer channels.

ROBBED-BIT signaling is a per-channel in-band signaling technique for transmitting signaling bits on each channel in a T1E1 facility. The least-significant bit in every 6th transmitted information frame is removed and replaced by a signaling bit. This technique is also called in-band signaling. The maximum transmission rate for each bearer channel with ROBBED-BIT signaling is 56 Kbps.

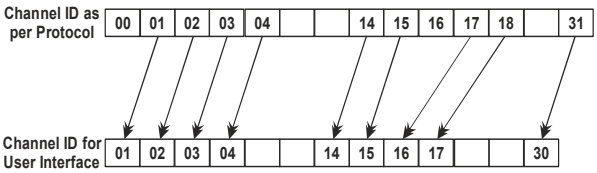
ISDN-PRI signaling is carried on the 24th channel for a 1.544 Mbps (T1) connection and on the 16th channel for a 2.048 Mbps (E1) connection.

QSIG is an ISDN based protocol for signaling between nodes of a Private Integrated Services network. Any of the common trunks, except for PCOL (Personal Central Office Line) trunks, can be analog or digital. (PCOL trunks can only be analog.) Administering a digital trunk group is very similar to administering its analog counterpart, but digital trunks must connect to a T1E1 port and this port must be administered separately.

User interface for E1_PRI and E1_CAS channels:

- In case of ISDN_E1_PRI and ISDN_E1_CAS the protocol supports 32 Channels ranging from 00 to 31, out of which 2 channels (channel no. 00 and 16) are used for framing/signaling. So effectively user has 30 channels for OG/IC calls.

- For better understanding of the user the channel IDs are mapped as shown below. Thus for the E1_PRI and E1_CAS the T1E1 port supports total 30 channels ranging from 01 to 30, which you can use for making and receiving calls.



Mapping of Actual Channel ID with User Interface (E1_PRI and E1_CAS)

- The system Debug is for trouble shooting and so the channel ID in debug will be as the actual channel ID as supported by the E1_PRI and E1_CAS protocols.

- Similarly, in case of T1 PRI, Protocol supports 24 channels (from 01 to 24), in which channel no. 24 is used for the signaling, so effectively there are 23 Voice channels are available.

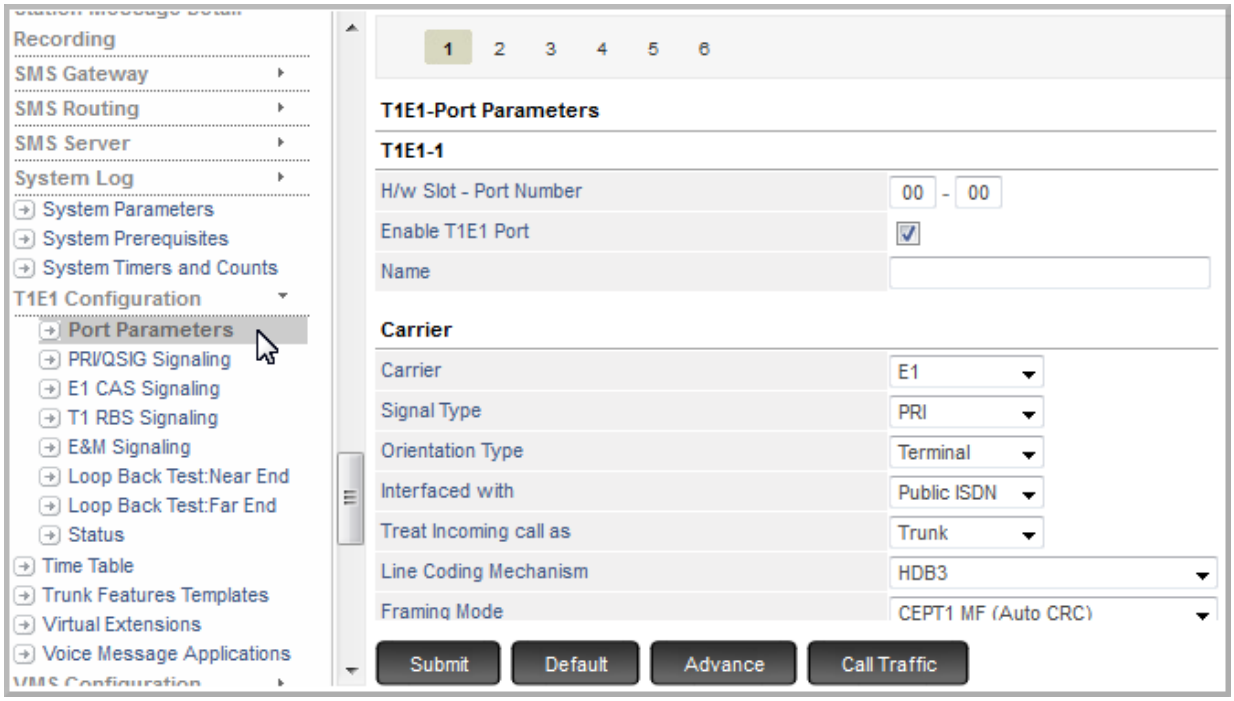
- But in case of T1 RBS, Protocol supports 24 channels (from 01 to 24) and the protocol doesn't consume any channel for signaling so that there are total 24 channels available for the users.

Configuring T1E1 Port Parameters using Jeeves

- Log into Jeeves as System Engineer.

- Under **Configuration**, click **T1E1 Configuration**.

- Click **Port Parameters** to open the page.



- Select the T1E1 Port number you want to configure by clicking the respective tab, and program the following port parameters:

    - SARVAM UCS will assign the **Hardware Slot-Port** automatically, when any card is inserted in the system.

Hardware slot is the number of the Universal slot of SARVAM UCS in which the T1E1 Card is inserted. Range of slot number is 01-16. Port is the number of T1E1 hardware port on the card to which the T1E1 line is connected. Range of Port is from 00-99.

If you want to de-assign the Hardware Slot and Port, Enter '00' in both fields. By default, Hardware slot-Port is 00–00.

- Keep the **Enable T1E1 Port** flag enabled.

  Clear the flag, only if you do not want to use this port. By default, it is enabled.

- You may assign a **Name** to the T1E1 Port for identification of the port. The Name may consist of a maximum of 12 characters. By default, it is blank.

- Select the **Carrier** type as **E1**. The Carrier type will automatically be assigned to the port when you select the region. You may it if required.

- Select **Signal Type**. Signal Type signifies the type of signaling to be used on E1 line. The E1 Signaling supported are:
  - PRI
  - CAS
  - QSIG
  - E&M

  Default: PRI

  If you select **PRI or QSIG** as the Signal Type, configure the **PRI/QSIG Parameters**.

  If you select **CAS** as the Signal Type, configure the **CAS Parameters**.

  If you select **E&M** as the Signal Type, configure the **E&M Parameters**.

- Set the **Orientation Type**. Select the orientation type from the following options:
  - Terminal
  - Network
  - Tie-Line

  Default: Terminal

- If you have set **Terminal** as the **Orientation Type**, you must select the type of network with which the port is to be **Interfaced With**. Select the type of network from the following:
  - Public ISDN
  - Private ISDN

  Default: Public ISDN

- You may configure the port to **Treat Incoming call as Trunk** or **Station**.

  If you select **Trunk**, the system will treat all incoming calls as external calls landing on the trunk. The calls will be routed as per the **Trunk Feature Template** assigned to the T1E1 Port.

  If you select **Station**, you must also assign a Station Basic Feature Template and Station Advanced Feature Template to the T1E1 Port.

When you select Station, the system will treat the calling party as an extension user. The user will have access to all the features and facilities of the system, as per the Station Basic Feature Template and Station Advanced Feature Template assigned to the T1E1 Port.

By default, Trunk is selected.

- *If Point-to-Point is selected as the Interface Type, you can select the option **Trunk** or **Station** for the parameter **Treat Incoming call as**.*

- *If Point-to-Multipoint is selected as the Interface Type, only **Station** can be set as the option for the parameter **Treat Incoming call as**.*

- *If **Station** is selected as the option for **Treat Incoming call as**, the user will only be able to:*
  - *Dial Flexible Numbers*
  - *Dial Operator Code*
  - *Dial Trunk Access Code for making outgoing calls*
  - *Access the Global Directory*
  - *Make calls within the Closed User Group*

- Line coding is a pattern that data assumes as it is propagated over a communication channel. Select the **Line Coding Mechanism** from the following:
  - AMI-Basic
  - HDB3
  - NRZ (Fiber Optic)

  Default: HDB3.

- Framing means to form a set of 24 or 32, 8 bits time slot that is to be treated as single transmission unit. The **Framing Modes** supported by SARVAM UCS are:
  - CEPT1 MF (No CRC)
  - CEPT1 MF (Forced CRC)
  - CEPT1 MF (Auto CRC)

  Default: CEPT1 MF (Auto CRC).

- Configure **Outgoing (OG) Reference ID**. By default, OG Reference ID is 00.

- Configure **Incoming (IC) Reference ID** for working hours, non-working hours and break hours. By default, IC Reference ID is 00.

- Assign **Trunk Feature Template** to the T1E1 Port. Trunk Feature Template is a set of general features that define the behavior of a Trunk Port. By default, Template 01 is assigned to all T1E1 Ports.

  For more details, see “Trunk Feature Template”.

- **Station Basic Feature Template** assigned to the T1E1 Port is displayed in this field. Station Basic Feature Template is a set of general features that define the basic behavior of a station. By default, Template 01 is assigned to all T1E1 Ports.

  For more details, see “Station Basic Feature Template”.

- **Station Advanced Feature Template** assigned to the T1E1 Port is displayed in this field. Station Advanced Feature Template is a set of advanced features, to be applied on extensions such as CLIP, Floor Service, Walk-in Class of Service. By default, Template 01 is assigned to all T1E1 Ports.

For more details, see “Station Advanced Feature Template”.

- Select **Priority** for the T1E1 Port. Priority is the precedence given to certain trunks and extensions over others in being answered by the destination extension. You can select from 1 to 9. By default, Priority 5-Normal is set for all T1E1Ports.

  For know more about Priority feature, see “Priority”.

- Assign a **Cost Factor** to the T1E1 Port. By default, all the T1E1 Ports are assigned Cost Factor 01.

  For more details, see “Cost Factor”.

- ISDN glare occurs if the system initiates an outgoing call on a B-Channel at the same time the network initiates an incoming call on that same B-channel. You may configure the **Glare Option** as **Proceed** or **Held Back**. While processing a glare condition, the configured Glare Option on T1E1 port will be considered.

- Select **Category (Logical Partition)** for the T1E1 Port. You may select from the following options:
  - 1
  - 2
  - 3
  - 4

  By default, 1 is selected for all T1E1 Ports. Refer the feature description “Logical Partition” to know more.

- **Idle Code** is the 8-bit sequence that occupies the time slot on a E1/T1 trunk channel when it is not being used. By default, 127 is configured as the Idle Code.

- If you want SARVAM UCS to display the called party number as the CLI for incoming calls, select the **Display Called Party Number as CLI** check box. By default, Display Called Party Number as CLI option is disabled.

  This option is useful when a single T1E1 line connection and Operator are shared by more than one organization. If you enable this option, make sure:
  - you configure the names and corresponding numbers of the organizations sharing the line in the Global Directory of SARVAM UCS.
  - the Operator has a DKP or an Extended IP Phone or a Mobile UC Client.

  With this option enabled the Operator will be able to handle calls more efficiently. When there is an incoming call, SARVAM UCS matches the number with the numbers in the Global Directory. If a match is found SARVAM UCS displays the company name configured for that entry to the Operator, that is, the CLI will display the called party number and name.

  After the Operator answers the call, the CLI will change and display the calling party number and name (if configured in the Global Directory).

  If you keep this option disabled, the calling party number and name will be displayed as the CLI, both during an incoming call and after the call is answered by the Operator.

You can configure the Display Called Party Number as CLI option only from Jeeves.

- Select the **Allow Incoming CLI Modification** check box if you want to apply 'Allow Incoming CLI Modification' on the T1E1 Port. By default, it is disabled.

Incoming CLI Modification is useful in countries where the Calling Line Identification (CLI) received by the System extension users must be suitably modified before it can be used to dial out the number. To know more, see “Incoming CLI Modification”.

If you receive CLI in dialable format, there is no need to use this feature. In such case, keep the flag disabled.

*For an incoming call on the E1 trunk, the Incoming CLI Modification will be applied only when both — the **Allow Incoming CLI Modification** check box and the **Enable Incoming CLI Modification** check box in System Parameters — are enabled.*

- Select **Return Call to Original Caller (RCOC)** flag to enable this feature on the T1E1 Port. By default, RCOC flag is disabled.

  For know more about RCOC feature, see “RCOC (Return Call to Original Caller)”.

- In **Channel Reserved for Data Call**, configure the channels you want to reserve for Data Calls. By default, 00 channels are reserved.

- In **Channel Reserved for Outgoing Call**, configure the channels you want to reserve for making outgoing calls. By default, 30 channels are reserved.

- In **Channel Reserved for Incoming Call**, configure the channels you want to reserve for receiving incoming calls. By default, 30 channels are reserved.

  When a caller dials the trunk access code or selective trunk access code for dialing the number directly on the trunk port, the caller waits for the dial tone before dialing the number. But some exchanges do not give Dial Tone for the T1E1 Port. For Example, when T1E1 port as E1CAS type is used in Delhi, it is observed that the exchange does not give dial tone when direct dialing on the trunk is used.

- Enable the **Feed Dial Tone** flag. SARVAM UCS will provides the dial tone to the caller when the T1E1 Port is accessed.

  It is applicable only when Online dialing is used as, Store and Forward dialing, the dial tone is given to the caller. The dial tone is played as per the Dial Tone Timer.

  By default, Feed Dial Tone flag is disabled.

- *When dial tone flag is disabled, user will hear the dial tone of the exchange if provided, otherwise, user will hear the silence.*

- *If the user is making the call from the SLT port and dial tone is not provided by the exchange, user will not know when to start dialing the number. In this case, it is possible that some digits are not out dialed on the port and wrong number is dialed out because system will out dial the number only if Outgoing call Acknowledge is received but the user is not aware of this condition. Hence, it is required to enable this flag, if exchange is not providing the dial tone.*

- When Online dialing or Store and Forward dialing are used, some exchanges do not provide any tone while routing/processing the call. Thus, the caller does not know whether the call is being processed or not as there is silence. In such case, You must enable the **Feed Routing Tone** flag, SARVAM UCS will play the routing tone to the caller. SARVAM UCS stops playing the routing tone when an alert message or connect message or disconnect message is received from the T1E1 Card.

By default, the **Feed Routing Tone** flag is disabled.

- To customize the pulse width option and set the pulse shapes configure the **Custom Pulse** parameters. SARVAM UCS generates pulse shapes which match the country standard, where it is installed. However, if the standard pulse shape does not match, SARVAM UCS enables you to customize the pulse width to match your exact requirements.

  To use customize pulse width option and set the pulse shape in 1 to 4 phases, keep the **T1/E1 Custom Pulse Width (CPW) flag** enabled.

  - **Word 1:** Set the pulse width for setting pulse shape in the 1st phase. The range of Custom Pulse Width-Word 1 is 001 to 127. Default: 109.
  - **Word 2:** Set the pulse width for setting pulse shape in the 2nd phase. The range of Custom Pulse Width-Word 2 is 001 to 127. Default: 107.
  - **Word 3:** Set the pulse width for setting pulse shape in the 3rd phase. The range of Custom Pulse Width-Word 3 is 001 to 127. Default: 64.
  - **Word 4:** Set the pulse width for setting pulse shape in the 4th phase. The range of Custom Pulse Width-Word 4 is 001 to 127. Default: 64.

- Select the **Bearer Service** supported by your service provider. You can select from:
  - Speech
  - 3.1 KHz Audio

  By default, Speech is selected.

- The **Overlap Receiving Timer** is relevant while receiving the called party number information in overlap receiving mode. It is not relevant for the port in overlap sending mode.

  Range of Overlap Receiving Timer is from 01 to 99 seconds. Default: 15 seconds.

- Configure **Pause Timer** for theT1E1 Port. Range of Pause Timer is from 1 to 9 seconds. By default, it is set to 3 seconds for all T1E1 Ports.

  This Timer is required to insert delay between the digits while dialing out DTMF digits on the T1E1 port. One of the applications for using this parameter is Multi-stage dialing. Refer chapter “Multi-Stage Dialing”.

  For example, if PPP2 is to be outdialed and Pause timer is programmed as 3 seconds, the SARVAM UCS will out dial the digit 2 after 9 seconds i.e delay of individual P i.e 3+3+3 =9.

- When the SETUP Message is sent by SARVAM UCS to the network (ISDN exchange), the exchange responds by sending SETUP ACK (Acknowledgment), and dial tone is played to the caller. The time taken by the exchange to respond to the SETUP message may vary from exchange to exchange. Set the **SETUP Response Timer (sec)** as per the time taken by the network to respond to the SETUP message and play dial tone to the caller.

  Valid Range of the timer is 01 to 20 seconds. By default it is set to 4 seconds.

Change the default settings only if required. If the time you set is less than the time taken by the exchange to respond, no dial tone will be played to the caller.

You can configure the SETUP Message Timer only from Jeeves.

- Configure **DTMF On Time** for theT1E1 Port. Range of DTMF On Time is from 051 to 255 ms. By default, it is set to 102 ms for all T1E1 Ports.

  The DTMF On Time is the time for which the DTMF digit which is to be outdialed by the SARVAM UCS remain On. One of the applications for using this parameter is Multi-stage dialing. Refer chapter "Multi-Stage Dialing".

- Configure **DTMF Inter Digit Pause Timer** for the T1E1 Port. Range of Inter Digit Pause Timer is from 051 to 255 ms. By default, it is set to 102 ms for all T1E1 Ports.

  Inter Digit Pause Timer is the time for which the system will wait while receiving the dialing digits to consider it as end-of-dialing.

- Configure the **Minimum ON Time (msec)** for which the DTMF signal should be present in order to be detected. The valid range of this time is 10 to 200 milliseconds. By default, Minimum ON Time is set to 20 milliseconds.

- Configure the **Minimum OFF Time (msec)**. This parameter signifies the minimum time period between successive DTMF digits. The valid range of this time is 10 to 200 milliseconds. By default, Minimum OFF Time is set to 20 milliseconds.

  Configure the **Minimum Level (dB)** for the DTMF digit to be considered as valid. The valid range of this time is 0 to -36.5 dB. By default, Minimum levels is set to -36.5dB.

- If SARVAM UCS is to be used as a Gateway, enable **Gateway Application-Answer Signaling** on the T1E1 Port and configure **DTMF String**. By default, Gateway Application-Answer Signaling is disabled and CCC is configured as DTMF String.

  For more details, see "Gateway Application-Answer Signaling".

- Configure **Call Budget** parameters for the T1E1 Ports. Call Budget is an expense control feature of SARVAM UCS that allows you to keep track of the cost of phone calls made from the T1E1 Port. By default, Call Budget is enabled on the trunk. If you wish to change the default configuration or disable it for this T1E1 Port, configure the parameters as per your requirement:

  - **Type:** Select the type of Call Budget, that is, Amount or Minutes or Calls to be applied on the T1E1 Port. By default, Minutes is selected as the Call Budget type.To disable select Type as None.

  - **Amount:** If you selected 'Amount' as the Call Budget Type, enter the Budget Amount in this field. By default the Amount is set to 999999.

  - **Minutes:** If you selected 'Minutes' as the Call Budget Type, enter the number of Minutes in this field. By default the number of minutes is set as 000300.

  - **Calls:** If you selected 'Calls' as the Call Budget Type, enter the number of Calls in this field. By default the number of calls is set to 9999.

  - **Scheduled Reset:** Enable this flag if you want the Call Budget Amount/Minutes/Number of Calls to be reset on a particular date of every month.

- **Scheduled (Date):** Select the date of the month (Daily or 1-31) on which you want the Call Budget Amount/Minutes/Number of Calls to be reset every month. You may select 'Daily' if your plan suggests so.

- **Call Back:** This parameter is related to the ‘Call Back on Trunk Port’ feature. If you want to enable the 'Call Back on Trunk Port' feature on this T1E1 Port, configure the following parameters:

  - **Enable Call Back:** Enable this flag to activate the Call Back on Trunk Port feature. By default, this flag is disabled on all trunk port types. By default, the flag is disabled.

  - **Call Back Timer (sec):** This is the duration for which the system waits for the caller to disconnect before applying the Call Back. The range of this timer is from 01 to 99 seconds. By default, it is set to 10 seconds.

- **Call Back Mode:** Select from the following options how a ‘Call Back’ call answered by the remote party should be routed:
  - Built-in Auto Attendant
  - PIN Authentication - Multiple Calls
  - CLI Authentication - Multiple Calls
  - CLI Authentication - Single Call - Answer Signaling
  - Operator

  By default, Operator is selected as the Call Back Mode.

- **Call Back on:** This parameter allows you to select if the call back should be made to the same number that was received or to a different number. If you want the call back to be made to the same number select the ‘CLI number”. If you want the call back to be made to a different number, select ‘Alternate Number’.

  By default, CLI number is selected for Call Back.

- **Incoming Number List:** Program the number strings that are eligible for Call Back in this List. By default, Number List 15 is assigned to Call Back Incoming Number List.

  Number List 15 is also assigned to all T1E1 Ports as well as all other Trunk port types. If you want the same numbers strings to be programmed commonly for all T1E1 Ports and Trunk Port types, retain this list.

  If you want a different set of number strings to be programmed for this T1E1 Port, select a different Number List, and assign it to the T1E1 Port.

  You may program the Incoming Number List either from the ‘Number List’ page or by clicking the ‘Incoming Number List’ link to reach the Number List page.

  Refer the topic “Number Lists” to know more, and for configuration instructions.

- **Outgoing Number List:** Program the number strings that are to be called back in this List.

  For each number string you programmed in the ‘Incoming Number List’, you must program in the corresponding index in the Outgoing Number List a number to which the call back is to be made. For example, for the number string programmed at Index 1 in the Incoming Number List, a corresponding number string at the same Index, Index 1, should be programmed in the ‘Outgoing Number List’.

By default, Number List 16 is assigned to Outgoing Number List.The same Number List 16 is also assigned to all T1E1 Ports as well as all other Trunk port types.

You may program the default number list, or a different number list and assign it to this T1E1 Port.

You may program the Outgoing Number List either from the 'Number List' page or by clicking the 'Outgoing Number List' link to reach the Number List page.

Refer the topic "Number Lists" to know more, and for configuration instructions.

- **Call Back from:** This parameter determines the trunk port to be used to make the call back. The call back can be made using the Same Port or an "OG Trunk Bundle Group".

  Select 'Same Port' if you want the call back to be made using the same port on which the missed call is received. If you select OGTB Group, the call back will be made using the OGTB Group, which you have defined.

  By default, Same Port is selected.
- **OGTB Group:** If you selected OGTB Group for making the call back in the previous parameter, you must define the OGTB Group that must be used in this parameter.

  By default, OGTB Group 01 is assigned.

  If you want the system to select the lowest cost trunk for making the call back, enable Least Cost Routing on the OGTB Group that you define here for Call Back.

- FDL is used for communicating general maintenance information or for transmitting user defined information within the T1 link. General maintenance information is in the form of Performance Message Report which is generated by the SARVAM UCS Card T1E1PRI and depending upon the FDL Protocol, the Performance Message Report is sent every second or on request.

  Select the **FDL** flag to enable. This parameter is applicable only if Framing = ESF. If the Network (Public or Private) to which the SARVAM UCS is connected does not support FDL then FDL will be disabled. By default, the T1 FDL is disabled.

  If you have enabled FDL, configure the **FDL Protocol**. SARVAM UCS supports ANSI T1.403 and AT&T 54016 protocols of reporting the performance monitoring. By default, the T1 FDL Protocol is ANSI T1.403.

  For more information, see "E1/T1 Maintenance"

- Enter appropriate **Debug Code (Level 1 to 4)**, to obtain debug information of various parts of T1E1 Card on the COM Port. By default, debug is off for all T1E1 ports for all levels.

  Code is the value for the specified level to turn ON the debug for the parameters. Code range is from 000 to 255. Code value '000' for each level will turn off that level's debug.

  Level and Code for T1E1 Port are as specified below:

  Level 1:

Unused	Unused	Unused	Unused	Layer 4	CAS DSP	MFC R2	CAS

001	CAS
002	MFC R2
004	CAS DSP
008	Layer 4
000	Debug Off

Level 2:

Unused	Unused	Unused	HDLC (D-Channel)	FDL	ABCD Bits	Counters	Alarms

001	Alarms
002	Counters
004	ABCD Bits
008	FDL
016	HDLC (D Channel)
000	Debug Off

Level 3:

Unused	Flow Debug	NLS Debug	LAP Debug	SVC Primitives	Variables	State	Primitives

001	Primitives
002	State
004	Variables
008	SVC Primitives
016	LAP Debug
032	NLS Debug
064	Flow Debug
000	Debug Off

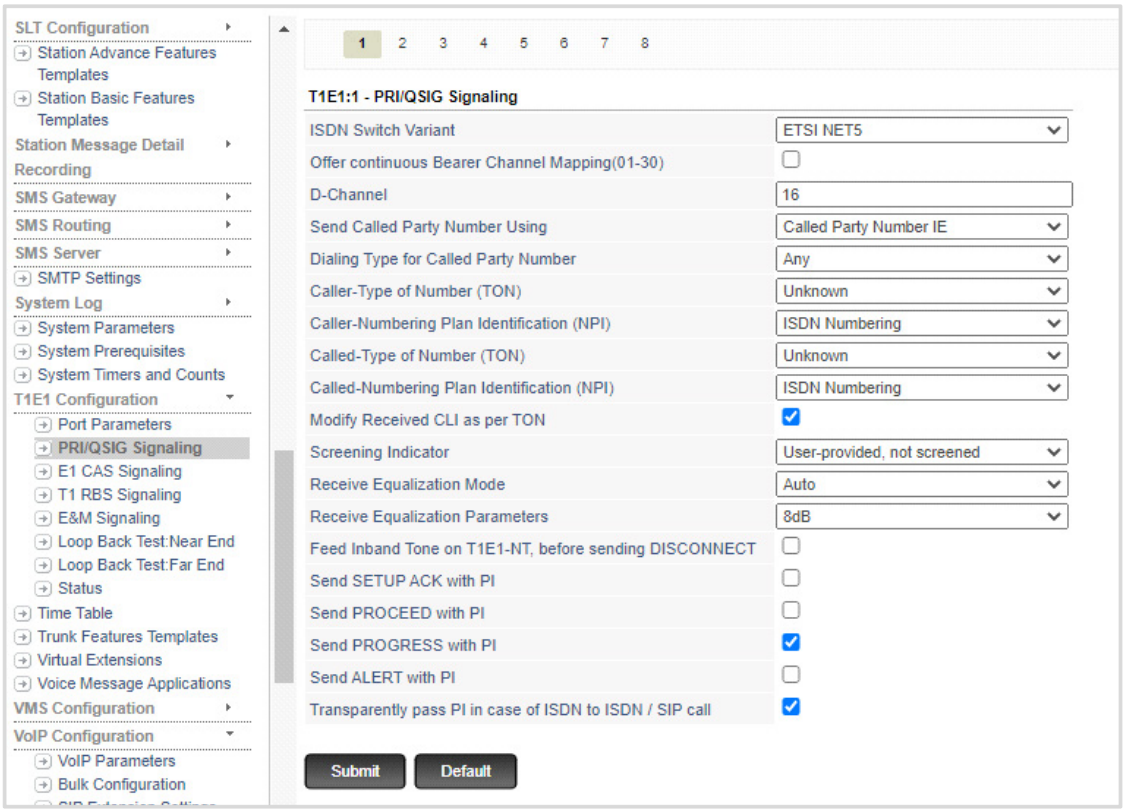
Level 4:

Unused	Unused	Unused	Unused	Unused	Unused	NI Debug	OS Task

001	OS Task
002	NI Debug
000	Debug Off

PRI/QSIG Parameters

- Under **T1E1 Configuration**, click **PRI/QSIG Signaling** and configure the PRI/QSIG parameters.



- **ISDN Switch Variant:** ISDN supports a variety of service provider switches. Different countries use specific type of ISDN switch. This switch is designed using ISDN standard protocol. The type of switch determines various factors such as how many ISDN devices would be handled, which B-channel will support voice, video, data etc. Select the ISDN Switch Variant from the list. By default, **ETSI NET5** is selected as the ISDN Switch Variant.

- **Offer continuous Bearer Channel Mapping(01-30):** Select this check box for continuous Bearer Channel Mapping for E1-QSIG/E1-PRI.

- **D-Channel:** Enter the Channel number that is used for Data signaling. By default, D-Channel is 16. Valid Range is 1 to 31.

- **Send Called Party Number Using:** Select the appropriate option from the following for Send Called Party Number Using:
  - Called Party Number IE (Information Element)
  - Keypad Facility IE (Information Element)

  By default, Called Party Number IE is selected.

- **Dialing Type for Called Party Number:** Select the type of dialing supported by your exchange. You can select:
  - Enbloc
  - Digit -by - Digit
  - Any

  By default, Any is selected as the Dialing Type for Called Party Number.

- **Caller - Type of Numbering Plan (TON):** Select the appropriate option from the following for sending the type of numbering plan of the calling party:
  - Unknown
  - International
  - National
  - Network Specific
  - Subscriber
  - Abbreviated
  - Reserved

  Default: Unknown.

- **Caller- Numbering Plan Identification (NPI):** Select the appropriate option from the following for sending the numbering plan identification of the calling party:
  - Unknown
  - ISDN Numbering
  - Data Numbering
  - Telex Numbering
  - National Numbering
  - Private
  - Reserved

  Default: ISDN Numbering.

- **Called - Type of Numbering Plan (TON):** Select the appropriate option from the following for sending the type of numbering plan of the called party:
  - Unknown
  - International
  - National
  - Network Specific
  - Subscriber
  - Abbreviated
  - Reserved

  Default: Unknown.

- **Called - Numbering Plan Identification (NPI):** Select the appropriate option from the following for sending the numbering plan identification of the called party:
  - Unknown
  - ISDN Numbering
  - Data Numbering
  - Telex Numbering
  - National Numbering
  - Private
  - Reserved

  Default: ISDN Numbering.

- **Modify Received CLI as per TON:** This is applicable for PRI-TE mode only. If this check box is selected then the Calling Party Number received in SETUP message will be changed as per the Type of Number received in the SETUP message from Calling Party Number field by the system. Default: Disabled.

- **Screening Indicator:** Select the option as provided to you by your exchange. You can select from:
  - User provided, not screened
  - User Provided, verified and passed
  - User Provided, verified and failed
  - Network Provided

By default, User provided, not screened.

- **Receive Equalization Mode:** You can set the Receive Equalization Mode as Auto or Manual. By default, Auto is selected as the Receive Equalization Mode.

- **Receive Equalization Parameters:** This field increases the strength of incoming signals by a fixed amount to compensate for line losses. Select the required option from the list. By default, the receive equalization parameters of T1E1 is 8dB.

- **Feed Inband Tones on T1E1-NT, before sending DISCONNECT:** Select this flag, if you want to feed inband tones on T1E1-NT before sending DISCONNECT message. This flag is applicable only when T1E1 port is configured as 'Network'. When this flag is enabled, inband tones shall be feed for 15 seconds (fixed, non programmable) before sending DISCONNECT message.

  When this flag is disabled, inband tones (Busy/Error as applicable for the state of the call) shall not be feed before sending the DISCONNECT message. However when DISCONNECT message is sent from T1E1-NT port, inband tones will always be sent with 'progress indicator 8'. By default, this flag is disabled.

- **Send SETUP ACK with PI:** Select this flag if you want the system to send PI (Progress indicator) element in Setup Ack message to other end. This option is relevant only when call landing on the T1E1 port are to be routed to SLT/CO/Mobile/DKP port. If T1E1 port is configured as 'Network', by default it is enabled and if T1E1 port is configured as 'Terminal', by default it is disabled.

- **Send PROCEED with PI:** Select this flag if you want the system to send PI (Progress indicator) in Proceed message to other end. This option is relevant only when call landing on the T1E1 port are to be routed to SLT/CO/Mobile/DKP port. If T1E1 port is configured as 'Network', by default it is enabled and if T1E1 port is configured as 'Terminal', by default it is disabled.

- **Send PROGRESS with PI:** Select this flag if you want the system to send PI (Progress indicator) in Progress message to other end. This option is relevant only when call landing on the T1E1 port are to be routed to SLT/CO/Mobile/DKP port. By default it is enabled.

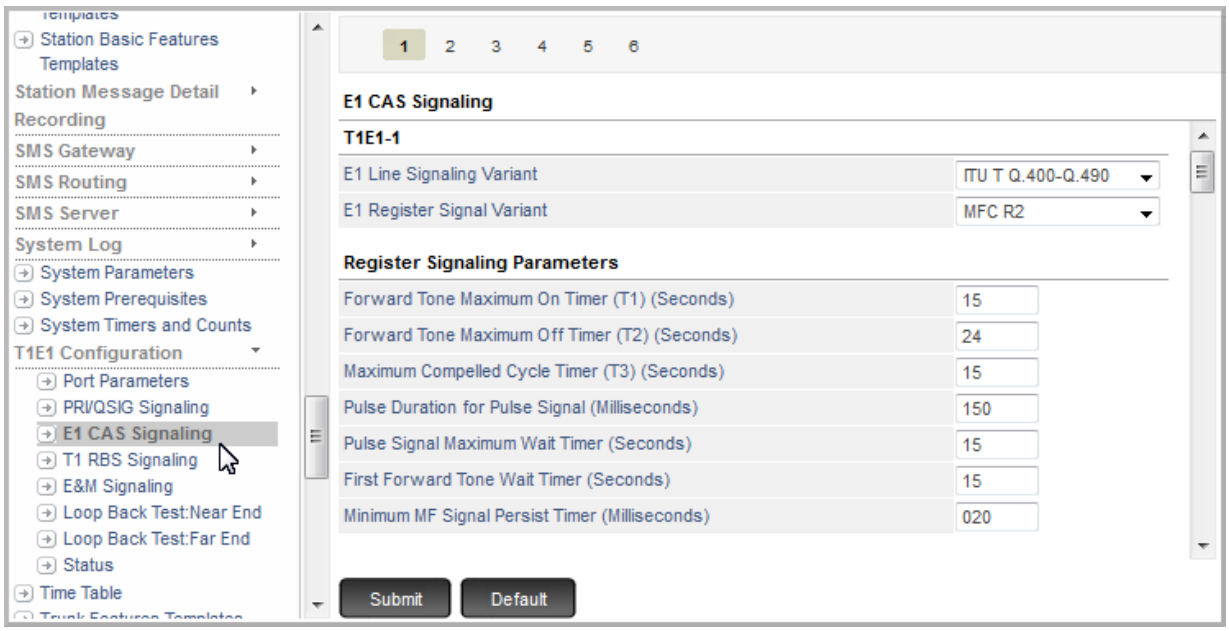
- **Send ALERT with PI:** Select this flag if you want the system to send PI (Progress indicator) in Alert message to other end. This option is relevant only when call landing on the T1E1 port are to be routed to SLT/CO/Mobile/DKP port. If T1E1 port is configured as 'Network', by default it is enabled and if T1E1 port is configured as 'Terminal', by default it is disabled.

- **Transparently pass PI in case of ISDN to ISDN / SIP call:** Select this flag, if you want the system to just pass the PI as received to the other end.This option is relevant only when call landing on the T1E1 port are to be routed to ISDN or SIP ports. By default it is enabled.

- Click **Submit** to save changes.

E1 CAS Parameters

Under **T1E1 Configuration**, click **E1 CAS Signaling** and configure the CAS parameters.



- **E1 Line Signaling Variant:** The E1 Line Signaling Variant supported by SARVAM UCS is ITU T Q.400-Q.490.

- **E1 Register Signaling Variant:** Select the appropriate option from the following for the E1 Register Signaling Variant:

  - DTMF - DNIS/ANI is transmitted in the corresponding speech channel using the DTMF signals as per ITU-T Q.23.

  - MFC R2 - DNIS/ANI is transmitted in the corresponding speech channel using the MFC R2 signals as per ITU-T Q.400-Q490.

    By default, E1 Register Signaling Variants is MFC R2.

Register Signaling Parameters

- **Forward Tone Maximum On Timer (T1) (Seconds):** This timer signifies the maximum time for which the forward signal remains ON, from the outbound end.

  The range of Forward Time Maximum ON Timer is from 01 to 99 seconds. Default: 15 seconds.

  The DSP (Digital Signaling Processor) device will send the forward tone for this timer and will expect backward signal within this timer. If no backward signal is received during this time, a timeout condition will occur in this case, an alert signal will be sent to the CPU Card, error tone be issued to the calling party and a clear forward signal will be sent on the line.

- **Forward Tone Maximum Off Timer (T2) (Seconds):** This timer signifies the maximum time between two out going forward signals. During this time the forward tone will remain OFF. If the outbound end does not send a forward signal for this time, the inbound end will interpret it as per its condition and shall take action accordingly.

The range of Forward Time Maximum OFF Timer is from 01 to 99 seconds. Default: 24 seconds.

- **Maximum Compelled Cycle Timer (T3) (Seconds):** This timer signifies the maximum time within which one compelled signaling cycle shall end.

  The range of Maximum Compelled Cycle Timer is from 01 to 99 seconds. Default: 15 seconds.

- **Pulse Duration for Pulse Signal (Milliseconds):** Backward signals A-3, A-4, A-6 and A-15 are pulsed to the outbound end. Pulse duration of these signals vary from country to country.

  The range of Pulse Duration for Pulsed Signals is from 001 to 999 ms. Default: 150 ms.

It is recommended that tolerance be fixed at +/- 25 ms.

- **Pulsed Signal Maximum Wait Timer (Seconds):** This timer signifies the time for which the outbound end waits for the pulsed signal. If the pulsed signal is not received during this time, the compelling signaling is said to be complete.

  The range of Pulsed Signal Maximum Wait Timer is from 01 to 99 seconds. Default: 15 seconds.

- **First Forward Tone Wait Timer (Seconds):** This timer signifies the time between receipt of line seizure signal and the first forward signal.

  The range of the First Forward Tone Wait Timer is from 08 to 24 seconds. Default:15 seconds.

- **Minimum MF Signal Persist Timer (Milliseconds):** This timer signifies the minimum time for which the forward/backward signal shall be sustained on the line by the receiving end.

  The range of Minimum MF Signal Persist Timer is from 001 to 255 ms. Default: 20 ms.

Outbound Parameters

- **Dialed Number Identification Signal (DNIS) End Type:** This parameter is applicable only when DNIS length is set to 99 (that is, variable). The outbound end indicates end of DNIS using a group I tone or using time out.

  Range of DNIS End Type is from 00, 11 to 15; where 00 indicates End of DNIS as time out. 11 to 15 indicates group I tone to declare End of DNIS. Default: 15.

- **Address Number Information (ANI) Send Position:** This parameter signifies the number of DNIS digits after which address number information is to be sent. Address number information is usually sent on receiving the backward tone Send next digit or Send next ANI digit.

  If send next address number information tone is received then this parameter is not applicable. But if same tone is used by the inbound end to request the next ANI digit and the next DNIS digit, ANI is sent after the number of digits as set in this field.

  The range of ANI Send Position is from 00 to 99. Default: 00.

- **Is Address Number Information (ANI) Available:** This parameter indicates Group A tone (received from the inbound tone) that is to be interpreted as a question by the inbound end asking the outbound end whether the outbound end has ANI digits to be sent.

The range of Is ANI Available, Group A tone is from 01 to 15. Default: 05.

- **Positive Response to Is ANI Available:** This parameter signifies the Group 1 tone that the outbound end will send to the inbound end as a response to Is ANI Available tone from the inbound end. The tone defined in this parameter indicates the Group 1 tone with which the Outbound end will respond to the inbound end to indicate that it has ANI digits to be sent.

  The range of Positive Response to Is ANI Available, Group 1 tone is from 01 to 15. Default: 01.

- **Negative Response to Is ANI Available:** This parameter signifies the Group 1 tone that the outbound end will send to the inbound end as a response to Is ANI Available tone from the inbound end. The tone defined in this parameter indicates the Group 1 tone with which the Outbound end will respond to the inbound end to indicate that it does not have ANI digits to be sent.

  The range of Negative Response to Is ANI Available, Group 1 tone is from 01 to 15. Default: 10.

- **ANI End Tone Presentation Allowed**: This parameter signifies the Group 1 tone used to signify end of ANI digits with Presentation Allowed.

  The range of End of ANI with Presentation Allowed, Group 1 tone is from 00, 11 to 15. Default:15.

- **ANI End Tone Presentation Restricted**: This parameter signifies the Group 1 tone used to signify end of ANI digits with Presentation Restricted.

  The range of End of ANI with Presentation Restricted, Group 1 tone is from 00, 11 to 15. Default: 00.

Inbound Parameters

- **Dialed Number Identification Signal (DNIS) End Type:** This parameter is applicable only when the DNIS length is set to 99 (that is, variable). The outbound end indicates end of DNIS using a group I tone or using time out.

  The range of DNIS End Type is from 00, 11 to 15; where 00 indicates End of DNIS as time out, 11 to 15 indicates group I tone to declare End of DNIS. Default:15.

- **Dialed Number Identification Signal (DNIS) Digit Length:** This parameter signifies the number of DNIS digits required by inbound end to indicate the Called party number during MFC R2 signaling.
  The range of DNIS Length is from 01 to 99. Default: 99.

  DNIS Digit length (01 to 98) will be expected by the inbound end. (Practical value would be 01 to 10)

  DNIS Digit length 99 indicates DNIS length is variable. Further action is taken after timeout or on receipt of I-15. Refer parameter 'DNIS End Type (Inbound)'.

- **Address Number Information (ANI) Request Position:** The inbound end may or may not request ANI digits. It may request ANI digits after receiving the first DNIS or after receiving second DNIS or even after receiving all the DNIS digits.

  The range of ANI Request Position is as follows:

ANI Request	Meaning
00	Never request ANI digits

ANI Request	Meaning
01-98	Request ANI digits on receipt of these many DNIS digits
99	Request after receiving all the DNIS digits (complete DNIS)

Default: 99.

- **Address Number Information (ANI) Length:** This parameter signifies the number of ANI digits that would be expected by the inbound side as Calling Party Number during MFC R2 signaling. This parameter at the inbound side guides the inbound register to switch from requesting ANI digits back to requesting DID digits.

  The range of ANI Length is from 00 to 99. Default: 99.

  - ANI Length = 00, ANI is not sent by the Outbound end.

  - ANI Length = 99, ANI Length is variable. If ANI length is variable, the logic waits for End of ANI from the outbound side. The inbound end will sense for I-12 and I-15. I-12 is used to signify that no ANI digits are available whereas I-15 is used to signify end of ANI digits. Some countries like China use I-15 to signify both the events viz. End of ANI and no ANI digits available.

- **Ask Address Number Information (ANI):** This parameter specifies the backward group A tone used to ask the outbound end whether it has ANI digits to be sent. This parameter is also known as **Request ANI Category**.

  The range of Ask ANI is from 00, 01 to 15. If no tone is sent by the inbound end, set this parameter to 00. For India, this parameter is set to 04. Default: 05.

- **Positive Response to Ask ANI:** This parameter specifies that the Group 1 forward tone is to be received by the inbound end from the outbound which in turn indicates that outbound end has ANI digits to be sent. This parameter is also known as ANI category.

  The range of Positive Response to Ask ANI is from 01 to 15. Default: 01.

  For example, In India I-1 or I-10 is sent by the outbound end. In Kuwait, I-6 is sent. This parameter cannot be zero because; Is ANI Available request will be made by the inbound end only if the country supports this protocol.

- **Negative Response to Ask ANI:** This parameter specifies the Group 1 forward tone to be received by the inbound end from the outbound which would indicate that outbound end has ANI digits to be sent. This parameter is also known as ANI category.

  The range of Negative Response to Ask ANI is from 01 to 15. Default: 10.

  For example, in India I-1 or I-10 is sent by the outbound end. In Kuwait, I-6 is sent. This parameter cannot be zero because; Is ANI Available request will be made by the inbound end only if the country supports this protocol.

- **ANI End Tone Presentation Allowed:** This parameter specifies the Group I tone that the inbound end should expect from the outbound end to consider End of ANI digits with information that the Presentation of ANI by the outbound end is allowed.

The range of ANI End Tone Presentation Allowed is 00 or from 11 to 15. Default: 15.

If no tone is sent, set this parameter to 00. For India use A-4, for China use A-1.

- **ANI End Tone Presentation Restricted:** This parameter specifies the group I tone that the inbound end shall expect from the outbound end to consider End of ANI digits with an information that the Presentation of ANI by the outbound end is Restricted.

  The range of ANI End Tone Presentation Restricted is 00 or from 11 to 15. Default: 00.

  If no tone is sent, set this parameter to 00. For India use A-4, for China use A-1.

- **Ask Calling Party Sub Category:** This parameter specifies the group 1 tone that the inbound end shall expect from the outbound end to consider End of ANI digits with an information that the Presentation of ANI by the outbound end is Restricted. Select this flag to enable. Default: Disabled.

Forward Group II

- **Ordinary Subscriber:** This parameter specifies the forward group II tone used to inform the inbound end that the calling party is an Ordinary Subscriber. This signal is sent in response to Calling Party Category signal Request from the inbound end.

  Ordinary Subscriber is 00, 01 to 15. Default: 01. If this parameter is not applicable, assign 00.

- **Priority Subscriber:** This parameter specifies the forward group II tone used to inform the inbound end that the calling party is a Priority Subscriber. This signal is sent in response to Calling Party Category signal Request from the inbound end.

  Priority Subscriber is 00, 01 to 15. Default: 02. If this parameter is not applicable, assign 00.

- **Maintenance Equipment:** This parameter specifies the forward group II tone used to inform the inbound end that the calling party is Maintenance equipment.

  Maintenance Equipment is 00, 01 to 15. Default: 03. If this parameter is not applicable, assign 00.

- **Operator:** This parameter specifies the forward group II tone used to inform the inbound end that the calling party is Operator.

  Operator is from 00, 01 to 15. Default: 05. If this parameter is not applicable, assign 00.

- **Pay Phone:** This parameter specifies the forward group II tone used to inform the inbound end that the calling party is Pay Phone (Coin box).

  Pay Phone is from 00, 01 to 15. Default: 00. If this parameter is not applicable, assign 00.

- **Data Transmission:** This parameter specifies the forward group II tone used to inform the inbound end that the call is a Data Call.

  Data Transmission is from 00, 01 to 15. Default: 06. If this parameter is not applicable, assign 00.

- **Interception Operator:** This parameter specifies the forward group II tone used to inform the inbound end that the call is from Interception Operator.
  Interception Operator is from 00, 01 to 15. Default: 00. If this parameter is not applicable, assign 00.

Backward Group A

- **Send Next Digit (N+1) (DNIS):** This parameter specifies the backward group A tone used to request next digit. Be it ANI digit or DNIS digit.

  Send next Digit range is 00, 01 to 15. Default: 01. If you do not want to use any tone, assign 00.

  For India, use A-1 to signify Send DNIS Digit event.

- **Send Last But One Digit (N-1) (DNIS):** This parameter specifies the backward group A tone used to request last but one digit that is, N-1 digit. Be it ANI digit or DNIS digit.

  Send last but one digit range is 00, 01 to 15. Default: 02. If you do not want to use any tone, assign 00.

  For India, use A-9 to signify Send last but one digit event.

- **Send Last But Two Digits (N-2) (DNIS)**: This parameter specifies the backward group A tone used to request last but two digits that is, N-2 digit. Be it ANI digit or DNIS digit.

  Send last but two digits range is 00, 01 to 15. Default: 07. If you do not want to use any tone, assign 00.

  For India, use A-7 to signify Send last but two digits event.

- **Send Last But Three Digits (N-3) (DNIS):** This parameter specifies the backward group A tone used to request last but three digits that is, N-3 digit. Be it ANI digit or DNIS digit.

  Send last but three digits range is 00, 01 to 15. Default: 08. If you do not want to use any tone, assign 00.

  For India use A-8 to signify Send last but three digits event.

- **Send Caller Party Category and ANI Digit:** It is to send calling party's category requests transmission of a group II signal. Range is 00 to 15. By Default, it is 05.

- **Address Completed, Change over of Group B:** This parameter specifies the backward group A tone used to inform the inbound end that the incoming register at the inbound end needs no additional address digit and is about to go over to transmission of a group B signal conveying the status of equipment at the subscriber at the inbound end.

  Address-Complete, Changeover to reception of Group B signal range is 00, 01 to 15. Default: 03. If you do not want to use any tone, assign 00.

- **Send Calling Party Category and Change to Group C:** This parameter specifies the backward group A tone used by the inbound end to request Calling Party Category from the outbound end. This tone also informs the outbound end to change to reception of Group C signal.

  Send Calling Party Category and Change to Group C range is from 00, 01 to 15. Default: 00. If you do not want to use any tone, assign 00.

- **Congestion in the National Network:** This parameter specifies the backward group A tone used to inform the congestion at the inbound end.

  Congestion in National Network range is 00, 01 to 15. Default: 04. If you do not want to use any tone, assign 00.

- **Send Calling Party Category:** This parameter specifies the backward group A tone used to request calling party category.

  Send calling party's category range is 00, 01 to 15. Default: 05. If you do not want to use any tone, assign 00.

  For India, use A-7 to signify 'Send calling party's category' event.

- **Address Completed, Charge, Set Speech Condition:** This parameter specifies the backward group A tone used to inform the inbound end that the incoming register at the inbound end needs no additional address digit, but will not send Group B signals. Also charge the call on answer.

  The range of Address-Complete, Charge, Set-up Speech conditions is 00, 01 to 15. Default: 06. If you do not want to use any tone, assign 00.

- **Repeat DNIS Digits from Beginning:** This parameter specifies the backward group A tone used to inform the outbound end to send all the DNIS digits from the beginning.

  The range of Repeat DNIS digits from beginning is 00, 01 to 15. Default: 00. If you do not want to use any tone, assign 00.

- **Send Next ANI Digits:** This parameter specifies the backward group A tone used to request next (first) ANI digit.

  The range of Send Next ANI Digit is 00 or 01 to 15.Default: 00. If no such tone is sent, set this parameter to 00.

  A few countries use different tone to request next ANI digit and next DNIS digits. For example, India uses A-4, China uses A-1.

Backward Group B

- **Send Special Information Tone:** This parameter specifies the backward group B tone used to inform the outbound end that the call cannot be made through because of reasons beyond those which are considered by the Protocol. Hence Special Information tone will be sent to the calling party. SARVAM UCS will send only the Group B signal and then disconnect the call.

  The range of Send Special Information Tone is from 00, 01 to 15. Default: 02. If you do not want to use any tone, assign 00.

- **Send Special Information Tone and Setup Speech Conditions:** This parameter specifies the backward group B tone used to inform the outbound end that the call cannot be made through because of reasons beyond those which are considered by the Protocol. Hence, Special information tone will be sent to the calling party and request the outbound end to setup speech conditions.

  In this case, SARVAM UCS shall connect the calling party to the voice message of the system informing the caller that the call cannot be connected.

  The range of Send Special Information Tone and setup speech conditions is from 00, 01 to 15. Default: 02. If you do not want to use any tone, assign 00.

- **Subscriber Line Busy:** This parameter specifies the backward group B tone used to inform the outbound end that the called subscriber is busy.

Subscriber Line busy range is from 00, 01 to 15. Default: 03. If you do not want to use any tone, assign 00.

- **Subscriber Line Free, Charge:** This parameter specifies the backward group B tone used to inform the outbound end that the called subscriber is free and the call is to be charged on answer.

  The range of Subscriber Line free, Charge is from 00, 01 to 15. Default: 06. If you do not want to use any tone, assign 00.

- **Subscriber Line Free, No charge:** This parameter specifies the backward group B tone used to inform the outbound end that the called subscriber is free, but the call is not to be charged on answer. This signal permits non-chargeable calls without the need for transferring **no charge** information by line signals.

  The range of Subscriber Line free, NO Charge is 00, 01 to 15. Default: 07. If you do not want to use any tone, assign 00.

- **Congestion:** This parameter specifies the backward group A tone used to inform that congestion is encountered after changeover from Group-A to Group-B signals.

  Congestion range is from 00, 01 to 15. Default: 04. If you do not want to use any tone, assign 00.

- **Unallocated Number:** This parameter specifies the backward group B tone used to inform the outbound end that the number received is not in use.

  The range of Unallocated Number is from 00, 01 to 15. Default: 05. If you do not want to use any tone, assign 00.

- **Subscriber Line Out of Order:** This parameter specifies the backward group B tone used to inform the outbound end that the called subscriber's line is out of order.

  Range of Subscriber's Line out of order is from 00, 01 to 15. Default: 08. If you do not want to use any tone, assign 00.

- **Call Rejected, No Indication:** This parameter specifies the Group B backward tone used to inform the outbound end that the call is rejected but there is no indication of cause.

  Call rejected, No indication range is from 00, 01 to 15. Default: 00. If this parameter is not applicable, assign 00.

- **Alternative Answer Tone:** This parameter specifies the Group B backward tone used to inform the outbound end that the call is accepted and the speech path is made through.

  Alternative Answer Tone range is from 00, 01 to 15. Default: 00. If this parameter is not applicable, assign 00.

- **Changed Number:** This parameter specifies the Group B backward tone used to inform the outbound end that the number dialed by the calling party is changed. However, this parameter is rarely used.

  The range of Changed Number (announcement on line) is from 00, 01 to 15. Default: 00. If this parameter is not applicable, assign 00.

Backward Group C

- **Send Next ANI Digit:** This parameter specifies the backward group C tone to request next (even first) ANI digit from the outbound end.

  The range of Send next ANI digit (Group C) is from 00, 01 to 15. Default: 00. If this parameter is not applicable, assign 00.

- **Request Transition Back to Group A and Restart from First DNIS (Group C):** This parameter specifies the backward group C tone to restart from the first DNIS and request transition to Group A.

  The range of Request transition to Group A and restart from first DNIS is from 00, 01 to 15. Default: 00. If this parameter is not applicable, assign 00.

- **Address Completed, Change to Reception of Group B:** This parameter specifies the backward group C tone used to signify Address completed, change to reception of Group B signal.

  The range of Address completed, change to reception of Group B signal is from 00, 01 to 15. Default: 00. If this parameter is not applicable, assign 00.

- **Congestion:** This parameter specifies the backward group C tone used to signify Congestion.

  The range of Congestion is from 00, 01 to 15. Default: 00. If this parameter is not applicable, assign 00.

- **Request Transition Back to Group A and Sent Next DNIS:** This parameter specifies the backward group C tone used to signify request transition back to group A, and send next DNIS.

  The range of Request transition back to group A, and send next DNIS is from 00, 01 to 15. Default: 00. If this parameter is not applicable, assign 00.

- **Request Transition Back to Group A and Repeat the Last DNIS:** This parameter specifies the backward group C tone used to signify request transition back to group A, and repeat the last DNIS. The range of Request transition back to group A, and repeat the last DNIS is from 00, 01 to 15. Default: 00. If this parameter is not applicable, assign 00.

Line Signal Parameters

- **Line Signaling:** By default this flag is enabled. If you are unable to make outgoing calls, check with your Service Provider and disable this option. This option will be relevant only when you Tie-up SARVAM UCS with another PBX.

- **C & D Bits:** This parameter indicates the default values of C and D bits when the T1/E1 Port transmits line signals.

CD Bits	Meaning (Binary Value)
0	00 (C=0, D=0)
1	01
2	10
3	11

  Default: 01 that is, C and D Bit is 1

The C and D bits received during an IC call should be ignored by the system.

- **Invert Bit A:** This parameter signifies whether A-bit is to be inverted before transmitting and on receiving. Select the check box to Invert Bit A.

  Default: Disabled (Do Not Invert Bit A).

- **Invert Bit B:** This parameter signifies whether B-bit is to be inverted before transmitting and on receiving. Select the check box to Invert Bit B.

  Default: Disabled (Do Not Invert Bit B1)

- **Invert Bit C:** This parameter signifies whether C-bit is to be inverted before transmitting and on receiving. Select the check box to Invert Bit C.

  Default: Disabled (Do Not Invert Bit C).

- **Invert Bit D:** This parameter signifies whether D-bit is to be inverted before transmitting and on receiving. Select the check box to enable, that is to Invert Bit D.

  Default: Disabled (Do Not Invert Bit D).

- **E1 Metering Bit:** This parameter signifies the bit used by the network to signal metering pulses.
  You can select from the following options:
  - None
  - Bit-A
  - Bit-B
  - Bit-C
  - Bit-D

  Default: Bit-A.

- **E1 Metering Pulse Minimum Timer (Milliseconds):** This timer signifies the minimum time for which the metering bit is changed, to be recognized as a genuine metering pulse subject to E1 Metering Pulse Minimum timer.

  All Changes occurred for time less than this timer is ignored. The range of E1 Metering Pulse Minimum timer is from 20ms to 1000ms. Default: 150ms.

- **Clear Back Signal:** This parameter signifies the signal used to signify that the called party has disconnected the line first. This is indicated in two ways: Release Guard (Ab =1) or Forced Release (Bb = 0). This parameter is country specific.

  Default: Release Guard.

- **Release Timer (Milliseconds):** This timer signifies the time for which the clear back signal should persist on the line to be recognized as a genuine clear back signal. This is also known as Clear Back timer.

  The range of Release Timer is from 20ms to 1000ms. Default: 400 ms.

- **Line Seizure Acknowledge Wait Timer (Milliseconds):** This timer signifies the time for which the outbound end waits for seizure acknowledgement from the inbound end after sending the line seizure

signal. On expiry of this timer, clear forward signal is sent by the outbound end. Alarm is to be generated. This timer is applicable only when acting as outbound end.

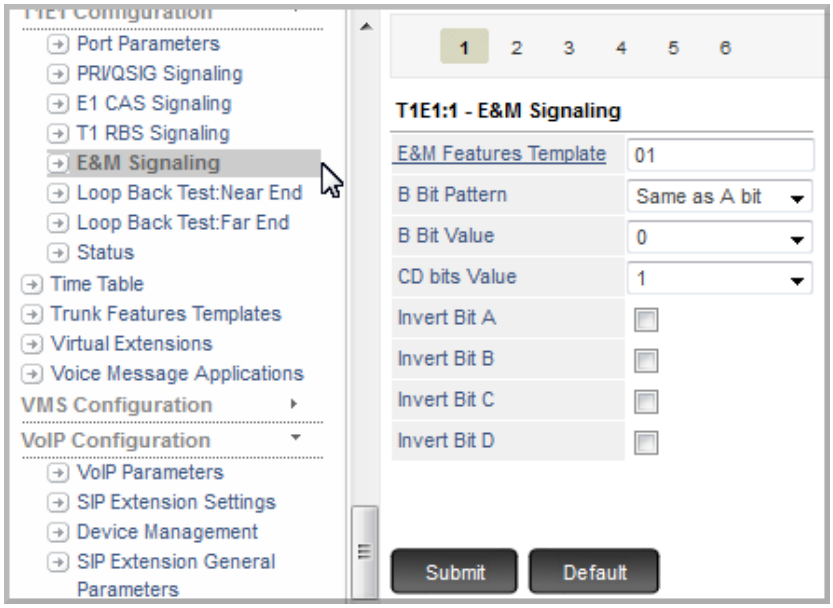
The range of Line Seizure acknowledge Wait Timer is from 0001ms to 9999 ms. Default: 200ms.

- **Release Guard Timer (Milliseconds):** This timer signifies the time for which inbound register waits before declaring the channel idle (sending idle signal) when clear forward line signal is received from the outbound end. This timer is applicable for Forced Release signal. This timer is applicable only when acting as inbound end. This timer depends on the speed of switching and processing.

  The range of Release Guard Timer is from 0000 ms to 9999 ms. Default: 200ms.

E&M Signaling

Under **T1E1 Configuration**, click **E&M Signaling** and configure the E&M parameters.



- **E&M Feature Template**: Assign an E&M Feature Template to the T1E1 Port. The E&M Feature Template is a set of features specific to E&M signaling, which define the behavior of the E&M ports, according to their 'Orientation Type', whether they are functioning as Stations, Trunks or Tie-Lines. By default, Template 01 is assigned to all T1E1 Ports.

  For more details, see “E&M Feature Template”.

- **B Bit Pattern:** Select the Bit Pattern from Same as Bit A or Fixed Value. By default, the Code is 1 (Same as A bit).

- **B Bit Value:** Configure the B bit value, the value can be 0 or 1. By default, B bit value is 0.

- **CD Bit Value:** Configure the CD bit value, the valid range of the value is 1 to 3. By default, B bit value is 0.

- **Invert Bit:** This parameter signifies whether A-bit is to be inverted before transmitting and on receiving. Select the check box to Invert Bit A.

  Default: Disabled (Do Not Invert Bit A).

- **Invert Bit B:** This parameter signifies whether B-bit is to be inverted before transmitting and on receiving. Select the check box to Invert Bit B.

  Default: Disabled (Do Not Invert Bit B1)

- **Invert Bit C:** This parameter signifies whether C-bit is to be inverted before transmitting and on receiving. Select the check box to Invert Bit C.

  Default: Disabled (Do Not Invert Bit C).

- **Invert Bit D:** This parameter signifies whether D-bit is to be inverted before transmitting and on receiving. Select the check box to enable, that is to Invert Bit D.

  Default: Disabled (Do Not Invert Bit D).

Configuring T1E1 Port Parameters using Jeeves

The commands explained below should be referred as:
• To program a single port: *XXXX-1*
• To program a range of ports: *XXXX-2*
• To program all the ports: *XXXX-**

Port Parameters

T1E1-1

Hardware Slot-Port

Use following command to assign hardware ID to a T1E1 software port.
1107-T1E1-Slot-Port offset
Where,
T1E1 is from 01 to 08.
Slot is the number of the universal slot, where the T1E1 Card is installed, from 01 to 16.
Port is the number of theT1E1 port on the card, from 01 to 32.

Use following command to de-assign the hardware slot and the hardware port assigned to the T1E1 software port.
1106-T1E1-00-00

Port Status

Used to enable/disable the port. When the Port is disabled, it will not be allotted to the user on grabbing the port. Instead the user will get error tone.

Use the following command to enable/disable the port:
6101-1-T1E1-Port Status
6101-2-T1E1-T1E1-Port Status
6101-*-Port Status
Where,

T1E1 is from 01 to 08.

Port Status	Meaning
0	Disable
1	Enable

By default, the T1E1 Port is enabled.

Name

Use the following command to assign a name to the port:
5407-1-T1E1-Name
5407-2-T1E1-T1E1-Name
5407-*-Name
Where,
T1E1 is from 01 to 08.
Name can be of upto 18 characters.

Carrier

Use the following command to select the carrier:
6108-1-T1E1-Carrier Type
6108-2-T1E1-T1E1-Carrier Type
6108-*-Carrier Type
Where,
T1E1 is from 01 to 08.

Carrier Type	Meaning
1	E1
2	T1

Line type

Use following command to program signaling type/ Line type of a T1E1:
6105-1-T1E1-Line Type
6105-2-T1E1-T1E1-Line Type
6105-*-Line Type
Where,
T1E1 is from 01 to 08.

Line Type	Meaning
1	PRI
2	QSIG
3	CAS
4	E&M

By default, signaling type of a T1E1 is 1.

*DDI Routing is not supported on T1/E1 trunk line if you have selected **E&M** as the **Signal Type**.*

Orientation Type

Use following command to program 'Orientation Type' for the T1E1 port:
6106-1-T1E1-Orientation Type
6106-2-T1E1-T1E1-Orientation Type
6106-*-Orientation Type

Where,
T1E1=01 to 08.

Orientation	Meaning
1	Terminal
2	Network
3	Tie Line

By default Type = 1.

When Orientation = Terminal, the port will be regarded as trunk. All the trunk related parameters will be applicable.
When Orientation = Network, the port will be regarded as station. All the station related parameters will be applicable.
When Orientation = Tie-line, the port will be regarded as station for all IC calls to it and as trunk for all OG calls to be made through it.

Line Coding Mechanism

Use following command to program the Line Coding Mechanism for the T1E1 port:
6103-1-T1E1-Line Coding
6103-2-T1E1-T1E1-Line Coding
6103-*-Line Coding
Where,
T1E1 is from 01 to 08.

Line Coding	Meaning
1	AMI-Basic
2	HDB3
3	NZR (Fiber Optic)

By default, Line Coding is HDB3.

Framing Mode

Use following command to program the Framing Mode for the T1E1 port:
6104-1-T1E1-Framing
6104-2-T1E1-T1E1-Framing
6104-*-Framing

Where,

Framing	Meaning
1	CEPT1 MF (No CRC)
2	CEPT1 MF (Forced CRC)
3	CEPT1 MF (Auto CRC)

By default, Framing is CEPT1 MF (Auto CRC).

DDI Routing

OG Reference ID

Use the following command to assign OG Reference ID to T1E1 port:
6131-1-T1E1-OG Reference ID
6131-2-T1E1-T1E1-OG Reference ID
6131-*-OG Reference ID
Where,
T1E1 is from 01 to 08.
OG Reference ID is from 00 to 99.
By default, OG Reference ID is 00.

IC Reference ID for Working Hour

Use the following command to assign IC Reference ID for Working Hour:
6132-1-T1E1-IC Reference ID
6132-2-T1E1-T1E1-IC Reference ID
6132-*-IC Reference ID
Where,
T1E1 is from 01 to 08.
IC Reference ID is from 00 to 99.
By default, IC Reference ID is 00.

IC Reference ID for Break Hour

Use the following command to assign IC Reference ID for Break Hour:
6133-1-T1E1-IC Reference ID
6133-2-T1E1-T1E1-IC Reference ID
6133-*-IC Reference ID
Where,
T1E1 is from 01 to 08.
IC Reference ID is from 00 to 99.
By default, IC Route Reference ID is 00.

IC Route Reference ID for Non-working Hour

Use the following command to assign IC Route Reference ID for Non-working Hour:
6134-1-T1E1-IC Reference ID
6134-2-T1E1-T1E1-IC Reference ID
6134-*-IC Reference ID
Where,
T1E1 is from 01 to 08.
IC Reference ID is from 00 to 99.

By default, IC Route ID is 00.

Templates

Trunk Feature Template

Use the following command to assign a Trunk Feature Template to the T1E1 Trunks, dial:
5806-1-T1E1- Trunk Feature Template Number to assign a template to a single T1E1 port.
5806-2-T1E1- Trunk Feature Template Number to assign the same template to a range of T1E1 ports.
5806-*- Trunk Feature Template Number to assign the same template to all T1E1 ports.
Where,
T1E1is the Software Port number of the port from 01 to 08.
Template Number is the number of the customized Trunk Feature Template, from 01 to 50. Default: Trunk Feature Template 01.

Station Basic Feature Template

To assign a Station Basic Feature Template to a T1E1PRI port, dial:
5506-1-T1E1PRI-Template Number to assign a template to a single T1E1 port.
5506-2-T1E1PRI-T1E1PRI-Template Number to assign the same template to a range of T1E1 ports.
5506-*-Template Number to assign the same template to all T1E1 ports.
Where,
T1E1PRI is the number of the T1E1PRI Software port, from 01 to 08.
Template Number is the number of the Station Basic Feature Template, from 01 to 50.
Default: Template 01 is assigned to all T1E1 ports.

Station Advance Features Template

To assign a Station Advanced Feature Template to a T1E1PRI port
5606-1-T1E1PRI-Template Number to assign a template to a single T1E1 port.
5606-2-T1E1PRI-T1E1PRI-Template Number to assign the same template to a range of T1E1 ports.
5606-*-Template Number to assign the same template to all T1E1 ports.
Where,
T1E1PRI is the number of the T1E1PRI Software port, from 01 to 08.
Template Number is the number of the Station Advanced Feature Template, from 01 to 50.
Default: Template 01 is assigned to all T1E1 ports.

Others

Priority

To assign priority to T1E1
3914-1-T1E1-Priority to assign a template to a single T1E1 port.
3914-2-T1E1PRI-T1E1PRI-Priority to assign the same priority to a range of T1E1 ports.
3914***-*-Template Number*** to assign the same Priority to all T1E1 ports.
Where,
T1E1PRI is the number of the T1E1PRI Software port, from 01 to 08.
Priority is from 1 to 9. Default: 5-Normal.

Cost Factor

Assign a Cost Factor to the T1E1 port. This is useful in LCR.

Use following command to assign a name to the T1E1 port:
6102-1-T1E1-SP

6102-2-T1E1-T1E1-SP
6102-*-SP
Where,
T1E1 is from 01 to 08.
SP is from 01 to 99.
By default, Service Provider is 01.

Glare Option

Use following command to program Glare Option for the T1E1 port:
6112-1-T1E1-Glare Option
6112-2-T1E1-T1E1-Glare Option
6112-*-Glare Option
Where,
T1E1 is from 01 to 08.

Glare Option	Meaning
1	Proceed
2	Held Back

By default, Glare Option is 2.

Set Glare=Proceed, if SARVAM UCS is to be given priority in event of Glare.
Set Glare=Held Back, if the other end of the link is to be given priority in event of Glare.

The Glare settings should be complimentary on either side of the link.

Category (Logical Partitioning)

Use following command to set the Category (Logical Partitioning) for the T1E1 port:
6121-1-T1E1-Category
6121-2-T1E1-T1E1-Category
6121-*-Category
Where,
T1E1 is from 01 to 08.
Category is from 1 to 4.
By default, Category is set as 1.

Idle Code

Use the following command to program the Idle Code of a T1E1:
6113-1-T1E1-Idle Code
6113-2-T1E1-T1E1-Idle Code
6113-*-Idle Code
Where,
T1E1 is from 01 to 08.
Idle Code is from 000 to 255 (corresponding to 8 bits).
By default, the idle code is 127 (7F).

Use message mode of the Digital Switch IC to send the idle channel code.

RCOC

To enable RCOC on T1E1 Trunk, dial:
6145-1-T1E1-Code to enable the feature on a single trunk.
6145-2-T1E1-T1E1-Code to enable the feature on a range of trunks.
6145-*-Code to enable the feature on all trunks.
Where,
T1E1 is the software port number of the trunk from 01 to 08.
Code is
0 for Disable
1 for Enable
Default: Disable

Channels

Return Call to Original Caller (RCOC)

Use the following command to enable RCOC on T1E1 Trunk:
6145-1-T1E1-Code to enable the feature on a single trunk.
6145-2-T1E1-T1E1-Code to enable the feature on a range of trunks.
6145-*-Code to enable the feature on all trunks.
Where,
T1E1 is the software port number of the trunk from 01 to 08.
Code is
0 for Disable
1 for Enable
Default: Disable

Channel Reserved for Data Call

Use the following command to reserve channels for data transmission on T1E1:
6135-1-T1E1-Channel Count (Data)
6135-2-T1E1-T1E1-Channel Count (Data)
6135-*-Channel Count (Data)
Where,
T1E1 is from 01 to 08.
Channel Count (Data) is from 00 to 30.
By default, Channel Count for data transmission is 00.

Channel Reserved for Outgoing Call

Use the following command to program number of channels reserved for OG channel count:
6136-1-T1E1-Channel Count (OG)
6136-2-T1E1-T1E1-Channel Count (OG)
6136-*-Channel Count (OG)
Where,
T1E1 is from 01 to 08.
Channel Count (OG) is from 00 to 30. "It specifies the number of channels to be reserved for making an OG calls. For example, If OG channel count is programmed as 15, simultaneous 15 (maximum) OG calls can be made from the T1E1 port".
By default, OG Channel Count is 30.

Channel Reserved for Incoming Call

Use the following command to program number of channels reserved for IC channel count:
6137-1-T1E1-Channel Count (IC)

6137-2-T1E1-T1E1-Channel Count (IC)
6137-*-Channel Count (IC)
Where,
T1E1 is from 01 to 08.
Channel Count (IC) is from 00 to 30. "It specifies the number of channels to be reserved for making an IC calls. For example, If IC channel count is programmed as 10, simultaneous 10 (max.) IC calls can be received on the T1E1 port".
By default, Channel Count (IC) is 30.

Tone

Feed Dial Tone

Use following command to program the dial tone flag for T1E1 port:
6115-1-T1E1-Flag
6115-2-T1E1-T1E1-Flag
6115-*-Flag
Where,
T1E1 is from 01 to 08.

Flag	Meaning
0	Disable
1	Enable

By default, Dial Tone Flag is '0' for all the T1E1 ports.

Feed Routing Tone

Use following command to program the routing tone flag for T1E1 port:
6116-1-T1E1-Flag
6116-2-T1E1-T1E1-Flag
6116-*-Flag
Where,
T1E1 is from 01 to 08.

Flag	Meaning
0	Disable
1	Enable

By default, Routing Tone Flag is '0' for all the T1E1 ports.

Custom Pulse

T1/E1 Custom Pulse Width (CPW)

Use following command to enable/disable Custom Pulse Width (CPW) Flag for the T1E1 port for T1 signaling:
6171-1-T1E1-Flag
6171-2-T1E1-T1E1-Flag
6171-*-Flag
Where,

T1E1 is from 01 to 08.

Flag	Meaning
0	Disable
1	Enable

By default, Custom Pulse Width Flag is '0'.

T1/E1Custom Pulse Width Word 1

This command is used for Pulse shaping in first phase.

Use following command to program Custom Pulse Width Word 1 for the T1E1 port for T1 signaling:
6172-1-T1E1-Custom Pulse Width Word 1
6172-2-T1E1-T1E1-Custom Pulse Width Word 1
6172-*-Custom Pulse Width Word 1
Where,
T1E1 is from 01 to 08.
Custom Pulse Width Word 1 is from 000 to 127 volt.
By default, Custom Pulse Width Word 1 is 63 volt.

T1/E1Custom Pulse Width Word 2

This command is used for Pulse shaping in second phase.

Use following command to program Custom Pulse Width Word 2 for the T1E1 port for T1 signaling:
6173-1-T1E1-Custom Pulse Width Word 2
6173-2-T1E1-T1E1-Custom Pulse Width Word 2
6173-*-Custom Pulse Width Word 2
Where,
T1E1 is from 01 to 08.
Custom Pulse Width Word 2 is from 000 to 127 volt.
By default, Custom Pulse Width Word 2 is 58 volt.

T1/E1Custom Pulse Width Word 3

This command is used for Pulse shaping in third phase.

Use following command to program Custom Pulse Width Word 3 for the T1E1 port for T1 signaling:
6174-1-T1E1-Custom Pulse Width Word 3
6174-2-T1E1-T1E1-Custom Pulse Width Word 3
6174-*-Custom Pulse Width Word 3
Where,
T1E1 is from 01 to 08.
Custom Pulse Width Word 3 is from 000 to 127 volt.
By default, Custom Pulse Width Word 3 is 76 volt.

T1/E1Custom Pulse Width Word 4

This command is used for Pulse shaping in fourth phase.

Use following command to program Custom Pulse Width Word 4 for the T1E1 port for T1 signaling:
6175-1-T1E1-Custom Pulse Width Word 4
6175-2-T1E1-T1E1-Custom Pulse Width Word 4

6175-*-Custom Pulse Width Word 4
Where,
T1E1 is from 01 to 08.
Custom Pulse Width Word 4 is from 000 to 127 volt.
By default, Custom Pulse Width Word 4 is 00 volt.

Timer

Pause Timer

Use following command to program Pause Timer:
6109-1-T1E1- Pause Timer
6109-2-T1E1-T1E1-Pause Timer
6109-*-Pause Timer
Where,
T1E1 is from 01 to 08
Pause Timer is from 1 to 9 seconds
By default, Pause Timer is 3 seconds.

DTMF ON Time

Use following command to program DTMF ON Time:
6117-1-T1E1-DTMF ON Time
6117-2-T1E1-T1E1-DTMF ON Time
6117-*-DTMF ON Time
Where,
T1E1 is from 01 to 08.
DTMF ON Time is from 051 to 255 msec.
By default, DTMF ON Time is 102 msec.

DTMF Inter Digit Pause Timer

Use following command to program DTMF Inter digit Pause Timer:
6118-1-T1E1-DTMF Inter digit Pause Time
6118-2-T1E1-T1E1-DTMF Inter digit Pause Time
6118-*-DTMF Inter digit Pause Time
Where,
T1E1 is from 01 to 08
DTMF Inter digit Pause Time is from 051 to 255 msec.
By default, DTMF Inter digit Pause Time is 102 msec.

Gateway

Use Gateway Application - Answer Signaling?

Use following command to set flag for 'Gateway Application-Answer Signaling' on T1E1 trunk:
6119-1-T1E1-Gateway Application-Answer Signaling flag
6119-2-T1E1-T1E1-Gateway Application-Answer Signaling flag
6119-*- Gateway Application-Answer Signaling flag
Where,
T1E1 is from 01 to 08.

Flag	Meaning
0	Disable

Flag	Meaning
1	Enable

By default, Gateway Application-Answer Signaling flag is 'disable'.

Gateway Application - Answer Signaling DTMF Digits

Use following command to program DTMF digits string to be dialed as Gateway Application-Answer Signaling:
6120-1-T1E1-Gateway Application-Answer Signaling DTMF String
6120-2-T1E1-T1E1-Gateway Application-Answer Signaling DTMF String
6120-1-T1E1-Gateway Application-Answer Signaling DTMF String
Where,
T1E1 is from 01 to 08.
DTMF Digits allowed for DTMF string are from (0 - 9), *, #, A, B, C, D.
Maximum 4 DTMF digits can be programmed. If you need less than 4 digits for DTMF string, terminate the command using #*.

To program #, *, A, B, C, D use following codes:

Digit	Code for programming through command
A	#4
B	#5
C	#6
D	#7
*	**
#	##

By default Gateway Application-Answer Signaling DTMF String is 'CCC'.

Call Budget

Call Budget Type

To program Call Budget Type on T1E1 Port, dial:
6122-1-T1E1-Budget Type to program call budget type for a single trunk port.
6122-2-T1E1-T1E1-Budget Type to program the same call budget type for a range of trunk ports.
6122-*-Budget Type to program the same call budget type for all trunk ports.
Where,
T1E1is the number of the T1E1 software port from 01 to 08.
Budget Type is
0 for None
1 for Amount
2 for Minutes
3 for Number of Calls
By default, Budget Type is None.

Call Budget Amount

To program Call Budget Amount on T1E1 Port, dial:
6123-1-T1E1-Budget Amount to program call budget amount for a single trunk port.

6123-2-T1E1-T1E1-Budget Amount to program the same amount for a range of trunk ports.
6123-*-Budget Amount to program the same amount for all trunk ports.
Where,
T1E1is the number of the T1E1 software port from 01 to 08.
Budget Amount is of 6 digits max. Use leading zeros if amount to be programmed has fewer than 6 digits.
By default Budget Amount is 999999.

Call Budget Minutes

To program Call Budget Minutes on T1E1 Port:
6124-1-T1E1-Minutes to program minutes for a single trunk port.
6124-2-T1E1-T1E1-Minutes to program the same minutes for a range of trunk ports.
6124-*-Minutes to program the same minutes for all trunk ports.
Where,
T1E1is the number of the T1E1 software port from 01 to 08.
Minutes is of 6 digits max. Use leading zeros if Minutes to be programmed has less than 6 digits.
By default, Minutes is 999999.

Call Budget Number of Calls

To program Call Budget Number of Calls on T1E1 Port, dial:
6125-1-T1E1-Number of calls to program number of calls for a single trunk port.
6125-2-T1E1-T1E1-Number of calls to program the same number of calls for a range of trunk ports.
6125-*-Number of calls to program the same number of calls for all trunk ports.
Where,
T1E1 is the number of the T1E1 software port from 01 to 08.
Number of Calls is of 4 digits from 0001 to 9999. Use leading zeros if number of calls to be programmed has fewer than 4 digits.
By default, Number of calls is 9999.

Call Budget Reset Mode

To program Call Budget Reset Mode for T1E1, dial:
6138-1-T1E1-Call Budget Reset Mode to program reset mode for a single trunk port.
6138-2-T1E1-T1E1-Call Budget Reset Mode to program the same reset mode for a range of trunk ports.
6138-*-Call Budget Reset Mode to program the same reset mode for all trunk ports.
Where,
T1E1 is the number of the T1E1 software port from 01 to 08.
1 for Scheduled reset
2 for Manual reset
By default, Call Budget Reset Mode is Scheduled.

Call Budget Date for Scheduled Reset mode

To program the Date for Scheduled Reset mode for T1E1, dial:
6139-1-T1E1-Date to program date for a single trunk port.
6139-2-T1E1-T1E1-Date to program the date for a range of trunk ports.
6139-*-Date to program the date for all trunk ports.
Where,
T1E1 is the number of the T1E1 software port from 01 to 08.
Date is
01 to 31 for Scheduled date to reset every month.
00 for Scheduled reset Daily.
By default, Reset date is 1st. of every month.

Call Back

Use the following commands to program Call Back on T1E1 Trunk ports. To know more about this feature, refer the topic Call Back on Trunk Ports.

To enable/disable Call Back on T1E1 port:
6176-1-T1E1- Call Back Flag
6176-2-T1E1-T1E1-Call Back Flag
6176- *-Call Back Flag
Where,
T1E1 is from 01 to 08

Call Back Flag	Meaning
0	Disable
1	Enable

By default, Call Back flag is disabled.

Call Back Timer

To program Call Back Timer for T1E1 port:
6177-1-T1E1-Call Back Timer
6177-2-T1E1-T1E1-Call Back Timer
6177-*-Call Back Timer
Where,
T1E1 is from 01 to 08
Pause Timer Range is from 01 to 99 Sec.
By default, Pause Timer is 10 Seconds

Call Back Mode

To program Call Back Mode on T1E1 port:
6178-1-T1E1-Call Back Mode
6178-2-T1E1-T1E1-Call Back Mode
6178-*-Call Back Mode
Where,
T1E1 is from 01 to 08
Call Back Mode is 1 to 5.

Call Back Mode	Meaning
1	Built-in Auto Attendant
2	PIN Auth. - Multiple Calls
3	CLI Auth. - Multiple Calls
4	CLI Auth. - Single Call - Ans. Sig.
5	Operator

By default, Call Back mode is Operator

Call Back On

To program Call Back On method for T1E1 port:

6179-1-T1E1-Call Back on selection
6179-2-T1E1-T1E1-Call Back on selection
6179-*-Call Back on selection

Where,
T1E1 is from 01 to 08
Call back on selection is

Call Back on	Meaning
1	CLI Number
2	Alternate Number

By default, Call Back on selection is CLI Number.

Incoming Number List

To assign Call Back - Incoming Number List to a T1E1 port:
6180-1-T1E1-Incoming Number List
6180-2-T1E1-T1E1-Incoming Number List
6180-*-Incoming Number List
Where,
T1E1 is from 01 to 08
Incoming Number List is from 01 to 16.
By default, Incoming Number List is 15

Outgoing Number List

To assign a Call Back - Outgoing Number List to a T1E1 port:
6147-1-T1E1-Outgoing Number List
6147-2-T1E1-T1E1-Outgoing Number List
6147-*-Outgoing Number List
Where,
T1E1 is from 01 to 08
Outgoing Number List is from 01 to 16.
By default, Outgoing Number List is 16.

Call Back From

To define Call Back From for a T1E1 port:
6148-1-T1E1-Call Back From
6148-2-T1E1-T1E1-Call Back From
6148-*-Call Back From
Where,
T1E1 is from 01 to 08
Call Back From is
1 for Same Port
2 for OGTB Group
By default, Same Port is selected as Call Back From.

OGTB Group

To assign a Call Back - OGTB Group for a T1E1 port:
6149-1-T1E1-OGTB Group
6149-2-T1E1-T1E1-OGTB Group

6149-*-OGTB Group
Where,
T1E1 is from 01 to 08
OGTB Group is from 01 to 32
By default, OGTBG is 01.

FDL

FDL Flag - copied from T1 Maintenance

T1 FDL can be enabled/disabled. This parameter is applicable only if Framing = ESF. If the Network (Public or Private) to which the SARVAM UCS is connected does not support FDL then T1 FDL will be disabled.

Use following command to enable/disable T1 FDL on a T1E1PRI port:
6164-1-T1E1PRI-T1 FDL
6164-2-T1E1PRI-T1E1PRI-T1 FDL
6164-*-T1 FDL
Where,
T1E1PRI is from 01 to 08.

T1 FDL	Meaning
0	Disable
1	Enable

By default, the T1 FDL is disabled.

FDL Protocol

SARVAM UCS will support both the protocols of reporting the performance monitoring. This parameter is applicable only if T1 FDL is enabled and Framing = ESF. This parameter will match the protocol expected by the other end of the link.

Use following command to program the T1 FDL Protocol for a T1E1PRI port:
6165-1-T1E1PRI-T1 FDL Protocol
6165-2-T1E1PRI-T1E1PRI-T1 FDL Protocol
6165-*-T1 FDL Protocol
Where,
T1E1PRI is from 01 to 08.

T1 FDL Protocol	Meaning
0	Disable
1	AT&T 54016
2	ANSI T1.403

By default, the T1 FDL Protocol is ANSI T1.403.

Refer “E1/T1 Maintenance”.

Debug

SARVAM UCS supports debug of parameters (debug codes) depending on the Level of debug. On issuing this command the T1E1 Card will send the debug details to the COM port of the T1E1 port.

Debug Level	Command
Debug Level-1	***6191-1-T1E1-1-XXX***
Debug Level-2	***6191-1-T1E1-2-XXX***
Debug Level-3	***6192-1-T1E1-1-XXX***
Debug Level-4	***6192-1-T1E1-2-XXX***

Use following command to start/stop debug the parameters for the T1E1 port:
6191-1-T1E1-Level-Debug Code
6191-2-T1E1-T1E1-Level-Debug Code
6191-*-Level-Debug Code
Where,
T1E1 Port is from 01 to 08.
Level is from 1 to 4 (As shown below).
Code is the value for the specified level to turn ON the debug for the parameters. Code range is from 000 to 255.
Code value '000' for each level will turn off that level's debug.

Level 1:

Unused	Unused	Unused	Unused	Layer 4	CAS DSP	MFC R2	CAS

001	CAS
002	MFC R2
004	CAS DSP
008	Layer 4
000	Debug Off

Level 2:

Unused	Unused	Unused	HDLC (D-Channel)	FDL	ABCD Bits	Counters	Alarms

001	Alarms
002	Counters
004	ABCD Bits
008	FDL
016	HDLC (D Channel)
000	Debug Off

Level 3:

Unused	Flow Debug	NLS Debug	LAP Debug	SVC Primitives	Variables	State	Primitives

001	Primitives
002	State
004	Variables
008	SVC Primitives
016	LAP Debug
032	NLS Debug
064	Flow Debug
000	Debug Off

Level 4:

Unused	Unused	Unused	Unused	Unused	Unused	NI Debug	OS Task

001	OS Task
002	NI Debug
000	Debug Off

Default: Debug Code = 'Debug OFF' for all T1E1 ports for all levels.

Refer "T1 RBS Parameters"

T1E1:1 - PRI/QSIG Signaling

ISDN Switch Variant

This field selects the variant for ISDN PRI when ISDN PRI is selected as signaling mode. This must match with the other end of the link.

Use following command to program the ISDN PRI Switch Variant:
6107-1-T1E1-ISDN Switch Variant
6107-2-T1E1-T1E1-ISDN Switch Variant
6107-*-ISDN Switch Variant
Where,
T1E1 is from 01 to 08.

ISDN PRI Variant	Meaning
1	ATT 4ESS
2	ATT 5ESS
3	Australia
4	DMS

ISDN PRI Variant	Meaning
5	ETSI NET5
6	NTT INS64
7	SWV Hongkong
8	US NI12
9	QSIG E1
10	QSIG T1

By default, ISDN PRI Switch Variant of a T1E1 is ETSI NET5.

Send Called Party Number Using

Use the following command to program overlap receiving timer:
6146-1-T1E1-Type
6146-2-T1E1-T1E1-Type
6146-*-Type
Where,
T1E1 is from 01 to 08.
Type is
1 for Keypad Facility IE
0 for Called Party IE
Default: Called Party Number IE

Overlap Receiving Timer

Use the following command to program overlap receiving timer:
6114-1-T1E1-Timer
6114-2-T1E1-T1E1-Timer
6114-*-Timer
Where,
T1E1 is from 01 to 08.
Timer is from 00 to 99.
By default, Overlap Receiving Timer is 15 seconds.

Caller-Type of Number (TON)

Use following command to program the Caller-Type of Number (TON) for a T1E1:
6126-1-T1E1-Source TON
6126-2-T1E1-T1E1-Source TON
6126-*-Source TON
Where,
T1E1 is from 01 to 08.

Source TON	Meaning
1	Unknown: This is used when the user or network has no a prior information about the numbering plan. In this case, the Address Value field is organized according to the network dialing plan. For example, prefix or escape digits might be present.
2	International Number.
3	National Number: Prefix or escape digits shall not be included.

Source TON	Meaning
4	Network Specific Number: This is used to indicate administration/Service number specific to the serving network. For example, used to access an operator.
5	Subscriber Number: This is used when a specific short number representation is stored in one or more SCs as part of a higher layer application.
6	Abbreviated Number.
7	Reserved Number.

By default, Caller-Type of Number (TON) of T1E1 is 1.

Caller-Numbering Plan Identification (NPI)

Use following command to program Caller-Numbering Plan Identification (NPI) for T1E1:
6127-1-T1E1-Source NPI
6127-2-T1E1-T1E1-Source NPI
6127-*-Source NPI
Where,
T1E1 is from 01 to 08.

Source NPI	Meaning
1	Unknown
2	ISDN Numbering Plan
3	Data Numbering Plan
4	Telex Numbering
5	National Numbering Plan
6	Private Numbering Plan
7	Reserved for Extension

By default, Caller-Numbering Plan Identification (NPI) for T1E1 is 2.

Called-Type of Number (TON)

Use following command to program Called-Type of Number (TON) for a T1E1:
6128-1-T1E1-Destination TON
6128-2-T1E1-T1E1-Destination TON
6128-*-Destination TON
Where,
T1E1 is from 01 to 08.

Destination TON	Meaning
1	Unknown: This is used when the user or network has no a prior information about the numbering plan. In this case, the Address Value field is organized according to the network dialing plan. For example, prefix or escape digits might be present.
2	International Number.
3	National Number: Prefix or escape digits shall not be included.

Destination TON	Meaning
4	Network Specific Number: This is used to indicate administration/Service number specific to the serving network. For example, used to access an operator.
5	Subscriber Number: This is used when a specific short number representation is stored in one or more SCs as part of a higher layer application.
6	Abbreviated Number.
7	Reserved Number.

By default, Called-Type of Number (TON) of T1E1 is 1.

Called-Numbering Plan Identification (NPI)

Use following command to program Called-Numbering Plan Identification (NPI) for T1E1:
6129-1-T1E1-Destination NPI
6129-2-T1E1-T1E1-Destination NPI
6129-*-Destination NPI
Where,
T1E1 is from 01 to 08.

Destination NPI	Meaning
1	Unknown
2	ISDN Numbering Plan
3	Data Numbering Plan
4	Telex Numbering
5	National Numbering Plan
6	Private Numbering Plan
7	Reserved for Extension

By default, Called-Numbering Plan Identification (NPI) for T1E1 is 2.

Receive Equalization Mode

Use the following command to program auto receive equalization mode:
6110-1-T1E1-Mode
6110-2-T1E1-T1E1-Mode
6110-*-Mode
Where,
T1E1 is from 01 to 08.

Mode	Meaning
0	Manual
1	Auto

By default, Receive Equalization Mode is 1.

Receive Equalization Parameters

This field increases the strength of incoming signals by a fixed amount to compensate for line losses.

Use the following command to program the receive equalization parameters of a T1E1:
6111-1-T1E1-Receive Equalization Parameters
6111-2-T1E1-T1E1-Receive Equalization Parameters
6111-*-Receive Equalization Parameters
Where,
T1E1 is from 01 to 08.

Receive Equalization Parameters	Meaning
1	None
2	8 dB
3	16 dB
4	24 dB
5	32 dB
6	40 dB
7	48 dB

By default, the receive equalization parameters of T1E1 is 1.

Feed Inband Tones on T1E1-NT, before sending DISCONNECT

Use following command to program to select whether the inband tones should be feed on T1E1-NT before sending DISCONNECT message?
6130-1-T1E1-Flag
6130-2-T1E1-T1E1-Flag
6130-*-Flag
Where,

Flag	Meaning
0	No
1	Yes

Default = NO.

E1 CAS Signaling

T1E1-1

E1 Line Signaling Variant

The following command is applicable when the Bit Rate=E1.

Use following command to program E1 Line Signaling Variant for the T1E1 port:
6152-1-T1E1-E1 Line Signaling Variant
6152-2-T1E1-T1E1-E1 Line Signaling Variant
6152-*-E1 Line Signaling Variant

Where,
T1E1 is from 01 to 08.

E1 Line Signaling Variant	Meaning
01	ITU T Q.400-Q.490

By default, E1 Line Signaling Variant is 01.

E1 Register Signaling Variant

Use following command to program E1 Register Signaling Variant for the T1E1 port:
6153-1-T1E1-E1 Register Signaling Variant
6153-2-T1E1-T1E1-E1 Register Signaling Variant
6153-*-E1 Register Signaling Variant
Where,
T1E1 is from 01 to 08.

E1 Register Signaling Variant	Meaning
1	DTMF
2	MFC R2

By default, E1 Register Signaling Variants is 2.

- **DTMF:** DNIS/ANI is transmitted in the corresponding speech channel using the DTMF signals as per ITU-T Q.23.
- **MFC R2:** DNIS/ANI is transmitted in the corresponding speech channel using the MFC R2 signals as per ITU-T Q.400-Q490.

Register Signaling Parameters

Forward tone Maximum ON timer (T1)

This timer signifies the maximum time for which the forward signal can be ON, from the outbound end.
Use the following command to program Forward Tone Maximum ON Timer:
7101-1-T1E1-Forward Tone Maximum ON Timer
7101-2-T1E1-T1E1-Forward Tone Maximum ON Timer
7101-*-Forward Tone Maximum ON Timer
Where,
T1E1 is from 01 to 08.
Forward Tone Maximum ON Timer is from 01 to 99 seconds.
By default, Forward Tone Maximum ON Timer is 15 secs.

Forward Tone Maximum OFF Timer (T2)

Use the following command to program Forward Tone Maximum Off Timer:
7102-1-T1E1-Forward Tone Maximum OFF Timer
7102-2-T1E1-T1E1-Forward Tone Maximum OFF Timer
7102-*-Forward Tone Maximum OFF Timer
Where,
T1E1 is from 01 to 08.
Forward Tone Maximum OFF Timer is from 01 to 99 seconds.

By default, Forward Tone Maximum Off Timer is 24 seconds. This timer will be less than the Incoming R2 register timeout timer programmed by the inbound end.

Maximum Compelled Cycle Time (T3)

Use the following command to program the Maximum compelled cycle time:
7103-1-T1E1-Maximum Compelled Cycle Time
7103-2-T1E1-T1E1-Maximum Compelled Cycle Time
7103-*-Maximum Compelled Cycle Time
Where,
T1E1 is from 01 to 08.
Total Call Set Up Timer is from 01 to 99 seconds.
By default, Maximum Compelled Cycle Time is 15 secs.

Pulse Duration for Pulse Signal

Use the following command to program Pulse duration for pulsed signals:
7104-1-T1E1-Pulse Duration for Pulse Signal
7104-2-T1E1-T1E1-Pulse Duration for Pulse Signal
7104-*-Pulse Duration for Pulse Signal
Where,
T1E1 is from 01 to 08.
Tolerance is fixed at +/-25ms.
By default, the Pulse Duration for Pulsed Signals is 150ms.

Pulse Signal Maximum Wait Timer

Use the following command to program the Pulsed Signal Maximum Wait Timer:
7105-1-T1E1-Pulse Signal Maximum Wait Timer
7105-2-T1E1-T1E1-Pulse Signal Maximum Wait Timer
7105-*-Pulse Signal Maximum Wait Timer
Where,
T1E1 is from 01 to 08.
Pulsed Signal Maximum Wait Timer is from 01 to 99 seconds.
By default, Pulsed Signal Maximum Wait Timer is 15 secs.

First Forward Tone Wait Timer

Use the following command to program first forward tone wait timer:
7106-1-T1E1-First Forward Tone Wait Timer
7106-2-T1E1-T1E1-First Forward Tone Wait Timer
7106-*-First Forward Tone Wait Timer
Where,
T1E1 is from 01 to 08.
First forward tone wait timer is from 08 to 24 seconds.
By default, the First Forward Tone Wait Timer is 15 seconds.

Minimum MF Signal Persist Timer

Use following command to program Minimum MF Signal Persist timer:
7107-1-T1E1-Minimum MF Signal Persist Timer
7107-2-T1E1-T1E1-Minimum MF Signal Persist Timer
7107-*-Minimum MF Signal Persist Timer
Where,
T1E1 is from 01 to 08.
Minimum MF Signal Persist timer is from 001 to 255 ms.

By default, Minimum MF Signal Persist timer is 20 ms.

Outbound Parameters

Dialed Number Identification Signal (DNIS) END Type

Use following command to program to set DNIS END Type (outbound) for T1E1 port:
7108-1-T1E1-End of DNIS
7108-2-T1E1-T1E1-End of DNIS
7108-*-End of DNIS
Where,
T1E1 is from 01 to 08.
End of DNIS is from 00, 11 to 15.
00 indicates End of DNIS as time out.
01 to 15 indicates group 1 tone to declare End of DNIS.
By default, DNIS End Type (Outbound) is 15.

Address Number Information (ANI) Send Position

Use following command to program the ANI Send Position:
7110-1-T1E1-ANI Send Position
7110-2-T1E1-T1E1-ANI Send Position
7110-*-ANI Send Position
Where,
T1E1 is from 01 to 08.
ANI Send Position is from 00 to 99.
By default, ANI Send Position is 00.

Is Address Number Information (ANI) Available

Use following command to program Is ANI Available (Outbound):
7111-1-T1E1-Is ANI Available (Outbound)
7111-2-T1E1-T1E1-Is ANI Available (Outbound)
7111-*-Is ANI Available (Outbound)
Where,
T1E1 is from 01 to 08.
Is ANI Available (Outbound) is Group A tone from 00 to 15. If no question by the inbound end, set this parameter to 00.
By default, Is ANI Available is '05'.

Positive Response to Is ANI Available

Use the following command to program the Positive Response to Is ANI Available (Outbound):
7112-1-T1E1-Positive Response to Is ANI Available
7112-2-T1E1-T1E1-Positive Response to Is ANI Available
7112-*-Positive Response to Is ANI Available
Where,
T1E1 is from 01 to 08.
Positive Response to Is ANI Available is a Group 1 tone from 01 to 15.
By default, Positive Response to Is ANI Available is 01.

Negative Response to Is ANI Available

Use following command to program the Negative Response to Is ANI Available (Outbound):
7113-1-T1E1-Negative Response to Is ANI Available
7113-2-T1E1-T1E1-Negative Response to Is ANI Available

7113-*-Negative Response to Is ANI Available
Where,
T1E1 is from 01 to 08.
Negative Response to Is ANI Available is a Group 1 tone from 01 to 15.
By default, Negative Response to Is ANI Available is 10.

ANI End Tone Presentation Allowed

Use the following command to program the End of ANI with Presentation Allowed (Outbound):
7114-1-T1E1-ANI End Tone with Presentation Allowed (Outbound)
7114-2-T1E1-T1E1-ANI End Tone with Presentation Allowed (Outbound)
7114-*-ANI End Tone with Presentation Allowed (Outbound)
Where,
T1E1 is from 01 to 08.
ANI End Tone with Presentation Allowed (Outbound) is a Group 1 tone from 00, 11 to 15. If no such tone is sent, set this parameter to 00.
By default, ANI End Tone with Presentation Allowed (Outbound) is 15.

ANI End Tone Presentation Restrict

Use the following command to program the End of ANI with Presentation Restricted (Outbound):
7115-1-T1E1-ANI End Tone with Presentation Restrict (Outbound)
7115-2-T1E1-T1E1-ANI End Tone with Presentation Restrict (Outbound)
7115-*-ANI End Tone with Presentation Restrict (Outbound)
Where,
T1E1 is from 01 to 08.
ANI End Tone with Presentation Restrict (Outbound) is a Group 1 tone from 00, 11 to 15. If no such tone is sent, set this parameter to 00.
By default, ANI End Tone with Presentation Restrict (Outbound) is 00.

Inbound Parameters

Dialed Number Identification Signal (DNIS) End Type

Use following command to program DNIS End Type:
7109-1-T1E1-DNIS End Type
7109-2-T1E1-T1E1-DNIS End Type
7109-*-DNIS End Type
Where,
T1E1 is from 01 to 08.
DNIS End Type is from 00, 11 to 15.
00 indicates End of DNIS as time out.
01 to 15 indicates group 1 tone to declare End of DNIS.
By Default, DNIS End Type (Inbound) is 15.

Dialed Number Identification Signal (DNIS) Digit Length

Use following command to program the DNIS Digit Length:
7116-1-T1E1-DNIS Digit Length
7116-2-T1E1-T1E1-DNIS Digit Length
7116-*-DNIS Digit Length
Where,
T1E1 is from 01 to 08.
DNIS Digit Length is from 01 to 99.
By default, DNIS Digit Length is 99.

- DNIS Digit length (01 to 98) will be expected by the inbound end. (Practical value would be 01 to 10)
- DNIS Digit length 99 indicates DNIS length is variable. Further action is taken after timeout or on receipt of I-15. Refer parameter 'DNIS End Type (Inbound)'.

Address Number Information (ANI) Request Position

Use following command to program the number of DNIS digits after which ANI digits should be requested by the inbound end:
7117-1-T1E1-ANI Request Position
7117-2-T1E1-T1E1-ANI Request Position
7117-*-ANI Request Position
Where,
T1E1 is from 01 to 08.
ANI Request Position is from 00, 01 to 99.
ANI Request Position=00 indicates Never request ANI digits.
ANI Request Position=01 to 98 indicates Request ANI digits on receipt of these many DNIS digits.
ANI Request Position = 99 indicates Request after receiving all the DNIS digits (complete DNIS).
By default, ANI Request Position is 99.

Address Number Information (ANI) Length

Use following command to program ANI Length (Inbound):
7118-1-T1E1-ANI Length
7118-2-T1E1-T1E1-ANI Length
7118-*-ANI Length
Where,
T1E1 is from 01 to 08.
ANI Length = 00 to 99.
By default, ANI Length is 99.

- ANI Length=00 indicates ANI is not sent by the Outbound end.
- ANI Length = 99 indicates ANI Length is variable. If ANI length is variable, the logic waits for End of ANI from the outbound side. The inbound end will sense for I-12 and I-15. I-12 is used to signify that no ANI digits are available whereas I-15 is used to signify end of ANI digits. Some countries like China use I-15 to signify both the events viz. End of ANI and no ANI digits available.

Ask Address Number Information (ANI)

Use following command to program the ASK ANI available:
7119-1-T1E1-ASK ANI
7119-2-T1E1-T1E1-ASK ANI
7119-*-ASK ANI
Where,
T1E1 is from 01 to 08.
ASK ANI Available is from 00 or 01 to 15.
If no such tone is sent by the inbound end, set this parameter to 00. For India, this should be set to 04
By default, Ask ANI (Inbound) is '05'.

Positive Response to Ask ANI

Use following command to program Positive Response to Ask ANI:
7120-1-T1E1-Positive Response to Ask ANI
7120-2-T1E1-T1E1-Positive Response to Ask ANI
7120-*-Positive Response to Ask ANI
Where,
T1E1 is from 01 to 08.

Positive Response to Ask ANI is from 01 to 15.
By default, Positive Response to Ask ANI is 01.

- This cannot be zero. This is because; Is ANI Available request would be made by the inbound end only if the country supports this protocol. In such event, Is ANI Available request will be responded to.

  For example, In India I-1 or I-10 is sent by the Outbound end. In Kuwait, I-6 is sent.

Negative Response to Ask ANI

Use following command to program Negative Response to Ask ANI:
7121-1-T1E1-Negative Response to Ask ANI
7121-2-T1E1-T1E1-Negative Response to Ask ANI
7121-*-Negative Response to Ask ANI
Where,
T1E1 is from 01 to 08.
Negative Response to Ask ANI is from 01 to 15.
By default, Negative Response to Ask ANI is 10.

- This cannot be zero. This is because; Is ANI Available request would be made by the inbound end only if the country supports this protocol. In such event, Is ANI Available request will be responded to.

  For example, In India I-1 or I-10 is sent by the Outbound end. In Kuwait, I-6 is sent.

ANI End Tone Presentation Allowed

Use following command to program the ANI End Tone Presentation Allowed:
7122-1-T1E1-ANI End Tone Presentation Allowed
7122-2-T1E1-T1E1-ANI End Tone Presentation Allowed
7122-*-ANI End Tone Presentation Allowed
Where,
T1E1 is from 01 to 08.
By default, ANI End Tone Presentation Allowed is 15.

- ANI End Tone Presentation Allowed is 00 or 11 to 15. If no such tone is sent, set this parameter to 00. For For example, India uses A-4, China uses A-1, etc.

ANI End Tone Presentation Restricted

Use following command to program the ANI End Tone Presentation Restricted:
7123-1-T1E1-ANI End Tone Presentation Restricted
7123-2-T1E1-T1E1-ANI End Tone Presentation Restricted
7123-*-ANI End Tone Presentation Restricted
Where,
T1E1 is from 01 to 08.
By default, ANI End Tone Presentation Restricted (Inbound) is 00.

ANI End Tone Presentation Restricted is 00 or 11 to 15. If no such tone is sent, set this parameter to 00. For example: India uses A-4, China uses A-1, etc.

Ask Calling Party Sub Category

Use following command to program the Ask Calling Party Sub Category:
7160-1-T1E1-Code
7160-2-T1E1-T1E1-Code
7160-*-Code

Where,
T1E1 is from 01 to 08.
Code is
1 to enable
0 to disable

Forward Group B

Ordinary Subscriber

Use following command to program the Ordinary Subscriber:
7124-1-T1E1-Ordinary Subscriber
7124-2-T1E1-T1E1-Ordinary Subscriber
7124-*-Ordinary Subscriber
Where,
T1E1 is from 01 to 08.
Ordinary Subscriber is from 00, 01 to 15. Use '00' when this parameter is not applicable.
By default, Ordinary Subscriber is 01.

Priority Subscriber

Use following command to program the Priority Subscriber:
7125-1-T1E1-Priority Subscriber
7125-2-T1E1-T1E1-Priority Subscriber
7125-*-Priority Subscriber
Where,
T1E1 is from 01 to 08.
Priority Subscriber is from 00, 01 to 15. Use '00' when this parameter is not applicable.
By default, Priority Subscriber is 02.

Maintenance Equipment

Use following command to program the Maintenance Equipment:
7126-1-T1E1-Maintenance Equipment
7126-2-T1E1-T1E1-Maintenance Equipment
7126-*-Maintenance Equipment
Where,
T1E1 is from 01 to 08.
Maintenance Equipment is from 00, 01 to 15. Use '00' when this parameter is not applicable.
By default, Maintenance Equipment is 03.

Operator

Use following command to program the Operator:
7127-1-T1E1-Operator
7127-2-T1E1-T1E1-Operator
7127-*-Operator
Where,
T1E1 is from 01 to 08.
Operator is from 00, 01 to 15. Use '00' when this parameter is not applicable.
By default, Operator is 05.

Pay Phone

Use following command to program the Pay Phone:
7128-1-T1E1-Pay Phone

7128-2-T1E1-T1E1-Pay Phone
7128-*-Pay Phone
Where,
T1E1 is from 01 to 08.
Operator is from 00, 01 to 15. Use '00' when this parameter is not applicable.
By default, Pay Phone is 00.

Data Transmission

Use following command to program the Data Transmission:
7129-1-T1E1-Data Transmission
7129-2-T1E1-T1E1-Data Transmission
7129-*-Data Transmission
Where,
T1E1 is from 01 to 08.
Data Transmission is from 00, 01 to 15. Use '00' when this parameter is not applicable.
By default, Data Transmission is 06.

Interception Operator

Use following command to program the Interception Operator:
7130-1-T1E1-Interception Operator
7130-2-T1E1-T1E1-Interception Operator
7130-*-Interception Operator
Where,
T1E1 is from 01 to 08.
Interception Operator is from 00, 01 to 15. Use '00' when this parameter is not applicable.
By default, Interception Operator is 00.

Backward Group A

Send next Digit (N+1) (DNIS)

Use following command to program the Send next Digit:
7131-1-T1E1-Send Next Digit
7131-2-T1E1-T1E1-Send Next Digit
7131-*-Send Next Digit
Where,
T1E1 is from 01 to 08.
Send next Digit is 00, 01 to 15. 00 is used for No tone.
By default, Send Next Digit is 01.

Send last but one Digit (N-1) (DNIS)

Use following command to program the Send last but one Digit:
7132-1-T1E1-Send Last But One Digit
7132-2-T1E1-T1E1-Send Last But One Digit
7132-*-Send Last But One Digit
Where,
T1E1 is from 01 to 08.
Send Last But One Digit is 00, 01 to 15. 00 is used for No tone.
By default, Send Last But One Digit (N-1) is 02.

Send last but two digit (N-2) (DNIS)

Use following command to program the Send last but two digit:
7134-1-T1E1-Send Last But Two Digit
7134-2-T1E1-T1E1-Send Last But Two Digit
7134-*-Send Last But Two Digit
Where,
T1E1 is from 01 to 08.
Send Last But Two Digit is 00, 01 to 15. 00 is used for No tone.
By default, Send Last But Two Digit (N-2) is 07.

Send last but three Digit (N-3) (DNIS)

This parameter specifies the backward group A tone used to request last but three digit, that is, N-3 digit. Be it ANI digit or DNIS digit.

Use following command to program the Send Last But Three Digit:
7135-1-T1E1-Send Last But Three Digit
7135-2-T1E1-T1E1-Send Last But Three Digit
7135-*-Send Last But Three Digit
Where,
T1E1 is from 01 to 08.
Send last but three digit is 00, 01 to 15. 00 is used for No tone.
By default, Send Last But Three Digit is 08.

Send Caller Party Category and ANI Digit

Use following command to program the Send Caller Party Category and ANI Digit:
7135-1-T1E1-Send Caller Party Category and ANI Digit
7135-2-T1E1-T1E1-Send Caller Party Category and ANI Digit
7135-*-Send Caller Party Category and ANI Digit
Where,
T1E1 is from 01 to 08.
Send Caller Party Category and ANI Digit is 00 to 15.

By default, Send Last But Three Digit is 08.

Address-Complete, Change over to Group B

Use following command to program the Address-Complete, Change over to reception of Group B signals:
7136-1-T1E1-Address-Complete, Change Over to Reception of Group B Signals
7136-2-T1E1-T1E1-Address-Complete, Change Over to Reception of Group B Signals
7136-*-Address-Complete, Change Over to Reception of Group B Signals
Where,
T1E1 is from 01 to 08.
Address-Complete, Change Over to Reception of Group B Signals is 00, 01 to 15. 00 is used for No tone.
By default, Address-Complete, Change Over to Reception of Group B Signals is 03.

Send Calling Party Category and Change to Group C

Use following command to program the Send Calling Party Category and Change to Group C:
7137-1-T1E1-Send Calling Party Category and Change to Group C
7137-2-T1E1-T1E1-Send Calling Party Category and Change to Group C
7137-*-Send Calling Party Category and Change to Group C
Where,
T1E1 is from 01 to 08.

Send Calling Party Category and change to Group C is from 00, 01 to 15. 00 is used for No tone.
By default, Send Calling Party Category and Change to Group C is 00.

Congestion in National Network

Use following command to program congestion in the National Network
7138-1-T1E1-Congestion in National Network
7138-2-T1E1-T1E1-Congestion in National Network
7138-*-Congestion in National Network
Where,
T1E1 is from 01 to 08.
Congestion in National Network is 00, 01 to 15. 00 is used for No tone.
By default, Congestion in National Network is 04.

Send Calling Party Category

Use following command to program the Send caller party's category:
7139-1-T1E1-Send Calling Party Category
7139-2-T1E1-T1E1-Send Calling Party Category
7139-*-Send Calling Party Category
Where,
T1E1 is from 01 to 08.
Send Caller Party Category is 00, 01 to 15. 00 is used for No tone.
By default, Send Calling Party's Category is 05.

Address-Completed, Charge, Set Speech Conditions

Use following command to program the Address-Complete, Charge, Set-up Speech Conditions of Group B signals:
7140-1-T1E1-Address-Complete, Charge, Set-up Speech Conditions
7140-2-T1E1-T1E1-Address-Complete, Charge, Set-up Speech Conditions
7140-*-Address-Complete, Charge, Set-up Speech Conditions
Where,
T1E1 is from 01 to 08.
Address-Complete, Charge, Set-up Speech Conditions is 00, 01 to 15. 00 is used for No tone.
By default, Address-Complete, Charge Set-up Speech Conditions is 06.

Repeat DNIS digits from Beginning

Use following command to program the repeat DNIS digits from beginning of Group B signals:
7141-1-T1E1-Repeat DNIS Digits from Beginning
7141-2-T1E1-T1E1-Repeat DNIS Digits from Beginning
7141-*-Repeat DNIS Digits from Beginning
Where,
T1E1 is from 01 to 08.
Repeat DNIS Digits from Beginning is from 00, 01 to 15. 00 is used for No tone.
By default, Repeat DNIS Digits from Beginning is 00.

Send Next ANI Digit

Use following command to program the Send Next ANI Digit
7142-1-T1E1-Send Next ANI Digit
7142-2-T1E1-T1E1-Send Next ANI Digit
7142-*-Send Next ANI Digit
Where,
T1E1 is from 01 to 08.
Send Next ANI Digit is 00 or 01 to 15. If no such tone is sent, set this parameter to 00.

By default, Send Next ANI Digit is 05.

- Few countries use different tone to request next ANI digit and next DNIS digits.
- For example: India uses A-4, China uses A-1, etc.

Backward Group B

Send Special Information Tone

Use following command to program the Send Special Information Tone:
7143-1-T1E1-Send Special Information Tone
7143-2-T1E1-T1E1-Send Special Information Tone
7143-*-Send Special Information Tone
Where,
T1E1 is from 01 to 08.
Send Special Information Tone is from 00, 01 to 15. 00 is used for No tone.
By default, Send Special Information Tone is 02.

Send Special Information Tone and Setup Speech Conditions

Use following command to program the Send Special Information Tone and Setup Speech Condition:
7144-1-T1E1-Send Special Information Tone and Setup Speech Conditions
7144-2-T1E1-T1E1-Send Special Information Tone and Setup Speech Conditions
7144-*-Send Special Information Tone and Setup Speech Conditions
Where,
T1E1 is from 01 to 08.
Send Special Information Tone, and Setup Speech Conditions is from 00, 01 to 15. 00 is used for No tone.
By default, Send Special Information Tone and Setup Speech Conditions is 02.

Subscriber Line Busy

Use following command to program the subscriber line busy:
7145-1-T1E1-Subscriber Line Busy
7145-2-T1E1-T1E1-Subscriber Line Busy
7145-*-Subscriber Line Busy
Where,
T1E1 is from 01 to 08.
Subscriber Line Busy is from 00, 01 to 15. 00 is used for No tone.
By default, Subscriber Line Busy is 03.

Subscriber Line Free, Charge

Use following command to program the subscriber line free, charge:
7146-1-T1E1-Subscriber Line Free, Charge
7146-2-T1E1-T1E1-Subscriber Line Free, Charge
7146-*-Subscriber Line Free, Charge
Where,
T1E1 is from 01 to 08.
Subscriber Line Free, Charge is from 00, 01 to 15. 00 is used for No tone.
By default, Subscriber Line Free, Charge is 06.

Subscriber Line Free, No charge

Use following command to program the subscriber line free, no charge:
7147-1-T1E1-Subscriber Line Free, NO Charge
7147-2-T1E1-T1E1-Subscriber Line Free, NO Charge

7147-*-Subscriber Line Free, NO Charge
Where,
T1E1 is from 01 to 08.
Subscriber Line Free, NO Charge is from 00, 01 to 15. 00 is used for No tone.
By default, Subscriber Line Free, NO Charge is 07.

Congestion

Use following command to program Congestion:
7148-1-T1E1-Congestion
7148-2-T1E1-T1E1-Congestion
7148-*-Congestion
Where,
T1E1 is from 01 to 08.
Congestion is from 00, 01 to 15. 00 is used for No tone.
By default, Congestion is 04.

Unallocated Number

Use following command to program the Unallocated Number:
7149-1-T1E1-Unallocated Number
7149-2-T1E1-T1E1-Unallocated Number
7149-*-Unallocated Number
Where,
T1E1 is from 01 to 08.
Unallocated Number is from 00, 01 to 15. 00 is used for No tone.
By default, Unallocated Number is 05.

Subscriber's Line Out of Order

Use following command to program the subscriber's line out of order:
7150-1-T1E1-Subscriber's Line out of Order
7150-2-T1E1-T1E1-Subscriber's Line out of Order
7150-*-Subscriber's Line out of Order
Where,
T1E1 is from 01 to 08.
Subscriber's Line out of order is from 00, 01 to 15. 00 is used for No tone.
By default, Subscribers Line Out of Order is 08.

Call Rejected, No Indication

Use following command to program the Call rejected, No indication of cause:
7151-1-T1E1-Call rejected, No indication
7151-2-T1E1-T1E1-Call rejected, No indication
7151-*-Call rejected, No indication
Where,
T1E1 is from 01 to 08.
Call rejected, No indication is from 00 to 15.
By default, Call rejected, No indication is 00.

Alternative Answer Tone

Use following command to program the Alternative Answer Tone:
7152-1-T1E1-Alternative Answer Tone
7152-2-T1E1-T1E1-Alternative Answer Tone
7152-*-Alternative Answer Tone

Where,
T1E1 is from 01 to 08.
Alternative Answer Tone is from 00, 01 to 15. Use '00' when this parameter is not applicable.
By default, Alternative Answer Tone is 00.

Changed Number

Use following command to program the Changed Number (announcement on line):
7153-1-T1E1-Changed Number
7153-2-T1E1-T1E1-Changed Number
7153-*-Changed Number
Where,
T1E1 is from 01 to 08.
Changed Number (announcement on line) is from 00, 01-15. Use '00' when this parameter is not applicable.
By default, Changed Number (announcement on line) is 00.

Backward Group C

Send Next ANI digit

Use following command to program the Send next ANI digit:
7154-1-T1E1-Send Next ANI Digit
7154-2-T1E1-T1E1-Send Next ANI Digit
7154-*-Send Next ANI Digit
Where,
T1E1 is from 01 to 08.
Send next ANI digit (Group C) is from 00, 01 to 15. Use '00' when this parameter is not applicable.
This parameter is applicable in Mexico only.
By default, Send Next ANI Digit (Group C) is 00.

Request Transition to Group A and Restart From First DNIS

Use following command to program the Request transition to Group A and restart from first DNIS.
7155-1-T1E1-Request Transition to Group A and Restart from First DNIS
7155-2-T1E1-T1E1-Request Transition to Group A and Restart from First DNIS
7155-*-Request Transition to Group A and Restart from First DNIS
Where,
T1E1 is from 01 to 08.
Request Transition to Group A and Restart from First DNIS is from 00, 01 to 15. Use '00' when this parameter is not applicable.

By default, Request Transition to Group A and Restart from First DNIS is 00.

Address Completed, Change to Reception of Group B

Use following command to program the Address completed, change to reception of Group B signal:
7156-1-T1E1-Address Completed, Change to Reception of Group B Signal
7156-2-T1E1-T1E1-Address Completed, Change to Reception of Group B Signal
7156-*-Address Completed, Change to Reception of Group B Signal
Where,
T1E1 is from 01 to 08.
Address Completed, Change to Reception of Group B Signal is from 00, 01 to 15. Use '00' when this parameter is not applicable.
By default, Address Completed, Change to Reception of Group B Signal is 00.

Congestion

This parameter specifies the backward group C tone used to signify Congestion

Use following command to program the tone for Congestion:
7157-1-T1E1-Congestion
7157-2-T1E1-T1E1-Congestion
7157-*-Congestion
Where,
T1E1 is from 01 to 08.
Congestion is from 00, 01 to 15. Use '00' when this parameter is not applicable.
By default, Congestion is 00.

Request Transition Back to Group A, and Sent Next DNIS

Use following command to program the tone for request transition back to group A, and send next DNIS signal:
7158-1-T1E1-Request Transition Back to Group A, and Send Next DNIS
7158-2-T1E1-T1E1-Request Transition Back to Group A, and Send Next DNIS
7158-*-Request Transition Back to Group A, and Send Next DNIS
Where,
T1E1 is from 01 to 08.
Request transition back to group A, and send next DNIS is from 00, 01 to 15. Use '00' when this parameter is not applicable.
By default, Request Transition Back to Group A, and Send Next DNIS is 00.

Request Transition Back to Group A, and Restart the Last DNIS

Use following command to program the tone for request transition back to group A, and repeat the last DNIS:
7159-1-T1E1-Request Transition Back to Group A, and Restart the Last DNIS
7159-2-T1E1-T1E1-Request Transition Back to Group A, and Restart the Last DNIS
7159-*-Request Transition Back to Group A, and Restart the Last DNIS
Where,
T1E1 is from 01 to 08.
Request Transition Back to Group A, and Restart the Last DNIS is from 00, 01 to 15. Use '00' when this parameter is not applicable.
By default, Request Transition Back to Group A and Restart the Last DNIS is 00.

Line Signal Parameters

C and D Bits

This parameter indicates the default values of C and D bits when the SARVAM UCS transmits line signals.

Use following command to program the C and D Bits for T1E1 port:
7161-1-T1E1-C and D Bits
7161-2-T1E1-T1E1-C and D Bits
7161-*-C and D Bits
Where,
T1E1 is from 01 to 08.

CD Bits	Meaning (Binary Value)
0	00 (C=0, D=0)
1	01

CD Bits	Meaning (Binary Value)
2	10
3	11

By default, CD Bits is 1.

The C and D bits received during an IC call are ignored by the SARVAM UCS.

Invert Bit A

Use following command to program to invert/don't invert Bit A for the T1E1 port:
7162-1-T1E1-Invert Bit A
7162-2-T1E1-T1E1-Invert Bit A
7162-*-Invert Bit A
Where,
T1E1 is from 01 to 08.

Invert Bit A	Meaning
0	Disable
1	Enable

By default, Invert Bit A is 0.

Invert Bit B

Use following command to program to invert/don't invert Bit B for the T1E1 port:
7163-1-T1E1-Invert Bit B
7163-2-T1E1-T1E1-Invert Bit B
7163-*-Invert Bit B
Where,
T1E1 is from 01 to 08.

Invert Bit B	Meaning
0	Disable
1	Enable

By default, Invert Bit B is 0.

Invert Bit C

Use following command to program to invert/don't invert Bit C for the T1E1 port:
7164-1-T1E1-Invert Bit C
7164-2-T1E1-T1E1-Invert Bit C
7164-*-Invert Bit C
Where,

T1E1 is from 01 to 08.

Invert Bit B	Meaning
0	Disable
1	Enable

By default, Invert Bit C is 0.

Invert Bit D

Use following command to program to invert/don't invert Bit D for the T1E1 port:
7165-1-T1E1-Invert Bit D
7165-2-T1E1-T1E1-Invert Bit D
7165-*-Invert Bit D
Where,
T1E1 is from 01 to 08.

Invert Bit D	Meaning
0	Disable
1	Enable

By default, Invert Bit D is 0.

E1 Metering Bit

Use following command to program the E1 Metering Bit for the T1E1 port:
7166-1-T1E1-E1 Metering Bit
7166-2-T1E1-T1E1-E1 Metering Bit
7166-*-E1 Metering Bit
Where,
T1E1 is from 01 to 08.

E1 Metering Bit	Meaning
0	None
1	Bit-A
2	Bit-B
3	Bit-C
4	Bit-D

By default, E1 Metering Bit is 1.

E1 Metering Pulse Minimum Timer

Use following command to program the Metering Pulse Minimum timer for the T1E1 port:
7167-1-T1E1-E1 Metering Pulse Minimum Timer
7167-2-T1E1-T1E1-E1 Metering Pulse Minimum Timer
7167-*-E1 Metering Pulse Minimum Timer

Where,
T1E1 is from 01 to 08.
E1 Metering Pulse Minimum timer is from 20ms to 1000ms.
By default, E1 Metering Pulse Minimum Timer is 150ms.

Clear Back Signal

Use following command to program the Clear Back Signal for the T1E1 port:
7168-1-T1E1-Clear Back Signal
7168-2-T1E1-T1E1-Clear Back Signal
7168-*-Clear Back Signal
Where,
T1E1 is from 01 to 08.

Clear Back Signal	Meaning
1	Release Guard (Ab=1)
2	Forced Release (Bb=0)

By default, Clear Back Signal is 1.

Release Timer

Use following command to program the Release Timer for the T1E1 port:
7169-1-T1E1-Release Timer
7169-2-T1E1-T1E1-Release Timer
7169-*-Release Timer
Where,
T1E1 is from 01 to 08.
Release Timer is from 20ms to 1000ms.
By default, Release Timer is 400 ms.

Line Seizure Acknowledge Wait Timer

Use following command to program Line seizure acknowledge wait timer:
7170-1-T1E1-Line Seizure Acknowledge Wait Timer
7170-2-T1E1-T1E1-Line Seizure Acknowledge Wait Timer
7170-*-Line Seizure Acknowledge Wait Timer
Where,
T1E1 is from 01 to 08.
Line Seizure Acknowledge Wait Timer is from 0001 to 9999 milli seconds.
By default, Line Seizure Acknowledge Wait Timer is 200ms.

Release Guard Timer

Use following command to program Release Guard Timer:
7171-1-T1E1-Release Guard Timer
7171-2-T1E1-T1E1-Release Guard Timer
7171-*-Release Guard Timer
Where,
T1E1 is from 01 to 08.
T1E1 Release Guard Timer is from 0000 to 9999 milliseconds.
By default, Release Guard Timer is 200ms. This timer depends on the speed of switching and processing.

E&M Signaling

E&M Feature Template

Assign E&M Feature Template to T1E1 using following command:
Use Following command to assign E&M Feature Template to T1E1 Port
6004-1-T1E1-Template Number
6004-2-T1E1-T1E1- Template Number
6004-*- Template Number
Where,
T1E1 is from 01 to 08.
Template Number is 01 to 50.
By default, Template 01 is assigned to T1E1.

B Bit Pattern

Use following command to select B Bit Pattern
7191-1-T1E1-Code
7191-2-T1E1-T1E1-Code
7191-*-Code
Where,
T1E1 is from 01 to 08.

Code	Meaning
1	Same as A bit
2	Fixed Value

By default, the Code is 1 (Same as A bit)

B Bit Value

Use following command to program B Bit Value
7192-1-T1E1-B Bit Value
7192-2-T1E1-T1E1- B Bit Value
7192-*- B Bit Value
Where,
T1E1 is from 01 to 08.
B bit value can be 0 or 1.
By default, B bit value is 0.

CD Bit Value

Use following command to program CD Bit Value
7193-1-T1E1-CD Bit Value
7193-2-T1E1-T1E1-CD Bit Value
7193-*- CD Bit Value
Where,
T1E1 is from 01 to 08.
CD bit value can be 1 or 3.
By default, CD bit value is 1.

Invert Bit A

Use following command to program to invert/don't invert Bit A for the T1E1 port:
7162-1-T1E1-Invert Bit A

7162-2-T1E1-T1E1-Invert Bit A
7162-*-Invert Bit A
Where,
T1E1 is from 01 to 08.

Invert Bit A	Meaning
0	Do not invert
1	Invert

By default, Invert Bit A is 0.

Invert Bit B

Use following command to program to invert/don't invert Bit B for the T1E1 port:
7163-1-T1E1-Invert Bit B
7163-2-T1E1-T1E1-Invert Bit B
7163-*-Invert Bit B
Where,
T1E1 is from 01 to 08.

Invert Bit B	Meaning
0	Do not invert
1	Invert

By default, Invert Bit B is 0.

Invert Bit C

Use following command to program to invert/don't invert Bit C for the T1E1 port:
7164-1-T1E1-Invert Bit C
7164-2-T1E1-T1E1-Invert Bit C
7164-*-Invert Bit C
Where,
T1E1 is from 01 to 08.

Invert Bit C	Meaning
0	Do not invert
1	Invert

By default, Invert Bit C is 0.

Invert Bit D

Use following command to program to invert/don't invert Bit D for the T1E1 port:
7165-1-T1E1-Invert Bit D
7165-2-T1E1-T1E1-Invert Bit D
7165-*-Invert Bit D
Where,

T1E1 is from 01 to 08.

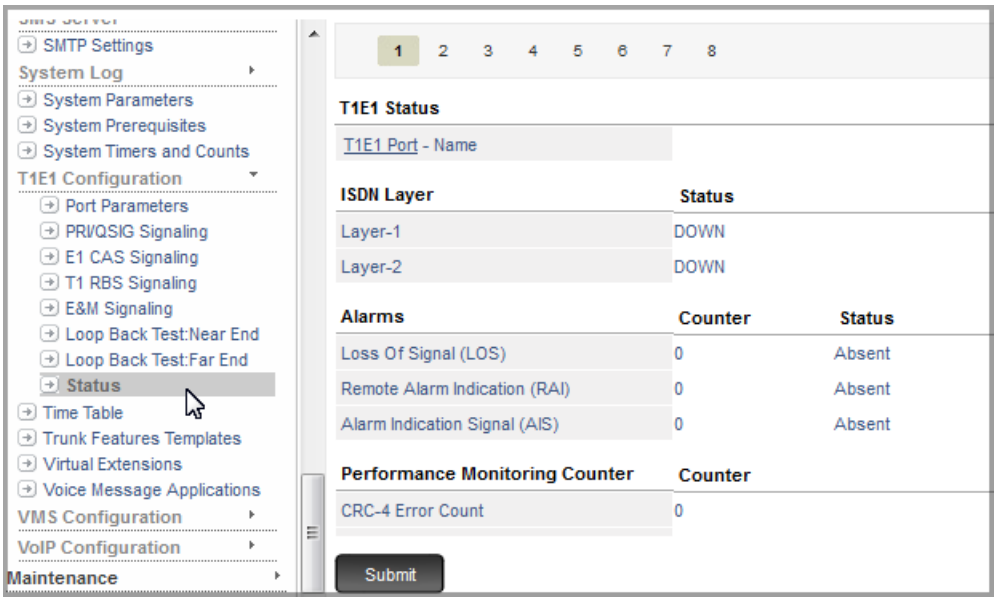
Invert Bit D	Meaning
0	Do not invert
1	Invert

By default, Invert Bit D is 0.

Viewing T1E1 Trunk Status

You can view the status of T1E1 Trunks on Jeeves only. To do this,

- Under **T1E1 Configuration**, click **Status**.



- For each T1E1 Trunk, the following settings will be displayed:
  - T1E1 Port No.
  - Name
  - Layer 1
  - Layer 2
  - Alarms
  - Performance Monitoring Counter
  - Call Budget Type
  - Allotted Amount/Minutes/Calls
  - Consumed Amount/Minutes/Calls
  - Call Budget Reset Mode
  - Call Budget Reset Scheduled (Date)
  - Reset Consumed (this is not a status indicator. It is for resetting the Consumed Call Budget manually)

*You can also view the T1E1 Trunk Status from the **Status** link. To view, click the T1E1 link under Status.*

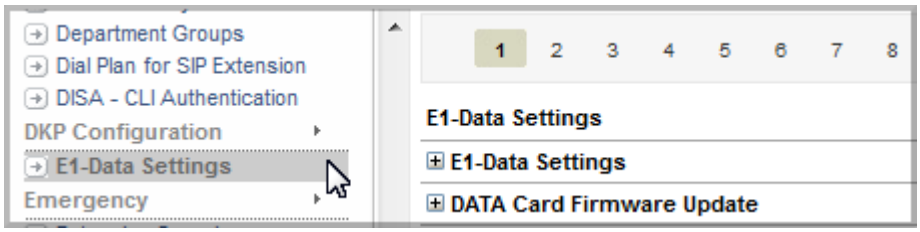
Configuring Data Parameters

The SARVAM UCS supports 4 Data ports. For more information about this card, see the topic “The Data Card”.

You can configure the Data Port Parameters only when you have set **E1** as the **Carrier** type in the **Port Parameters** under **T1E1 Configuration**. See “Configuring E1 Trunks” for more details. To configure the Data Port Parameters, see “E1 Data Settings”.

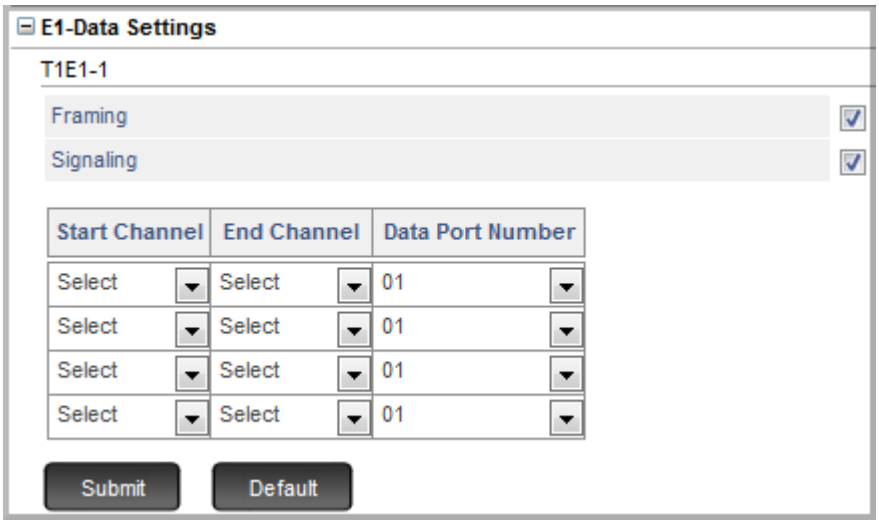
Configuring Data Parameters using Jeeves

- Log into Jeeves as System Engineer.
- Under **Configuration**, click **E1-Data Settings** to open the page.



E1 Data Settings

- Click **E1-Data Settings** to expand.



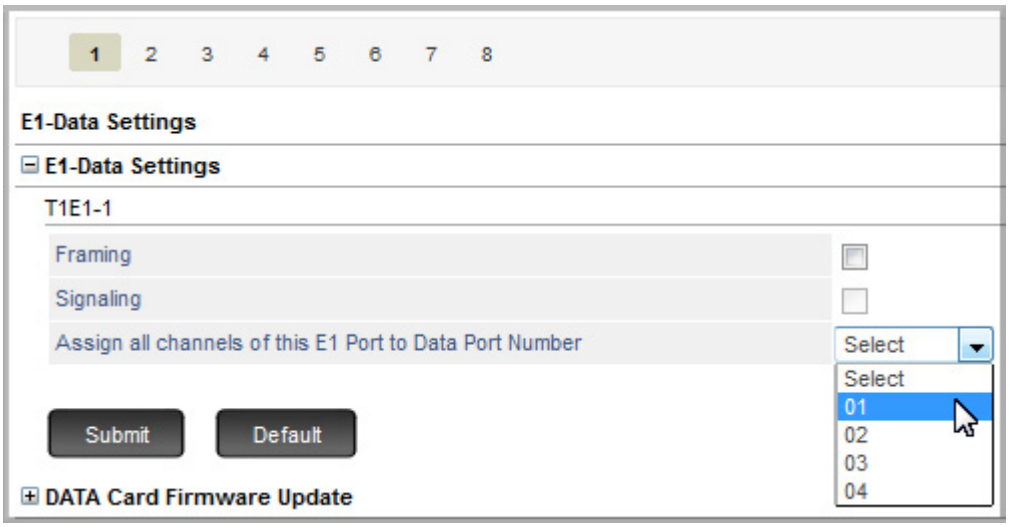
- Select the **T1E1 Port** tab, for which you want to configure the data parameters.
- By default, both **Framing** and **Signaling** are enabled. You can divide the E1 channels to route Voice calls as well as for Data communication.
- Select the number of E1 Channels to be used for routing data from E1 to the desired Data port in **Start Channel** and **End Channel**. Default: Start and End Channel are blank.
- In **Data Port Number**, select the desired Data Port to which the E1 Port is to be mapped. Default: 1.

- The E1 Channels not configured above to route Data, shall be used to route Voice Calls.

- Click **Submit** to save.

  - By default, **Framing** is enabled. You can use this E1 Port for both Voice Calling and Data Communication.

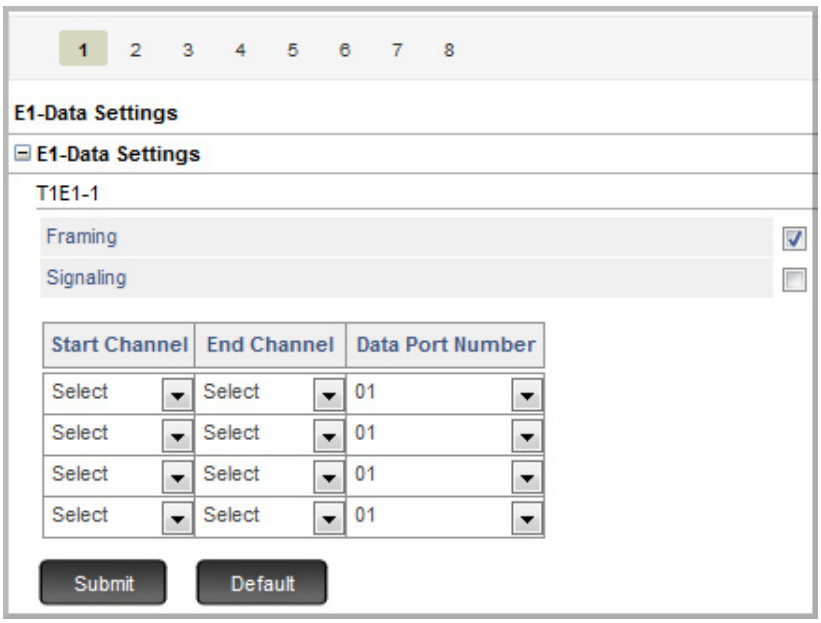
    Clear this check box, if you want to use this E1 Port for Data Communication only. When you clear this check box, Framing is disabled and you can allot all the E1 channels to a single data port only.



    In **Assign all channels of this E1 Port to Data Port Number**, select the desired Data Port to which you want to assign all the E1 channels.

  - By default, **Signaling** is enabled. You can use this E1 Port for both Voice Calling and Data Communication.

Clear this check box, if you want to use this E1 Port for Data Communication only. When you clear this check box, Signaling is disabled and you can allot different E1 channels to different data ports.

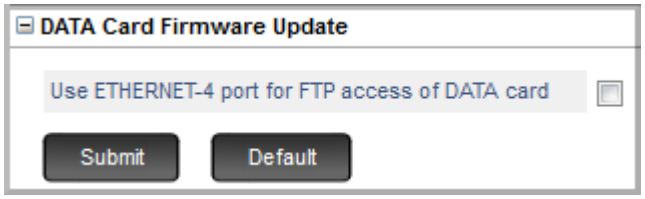


- Select the number of E1 Channels to be used for routing data from E1 to the desired Data port in **Start Channel** and **End Channel**. Default: Start and End Channel are blank.
- In **Data Port Number**, select the desired Data Port to which the E1 Port is to be mapped. Default: 1.

- Click **Submit** to save.
- Repeat the same steps to map another E1 port to a Data port and select the channels to be used for routing data between the two ports.

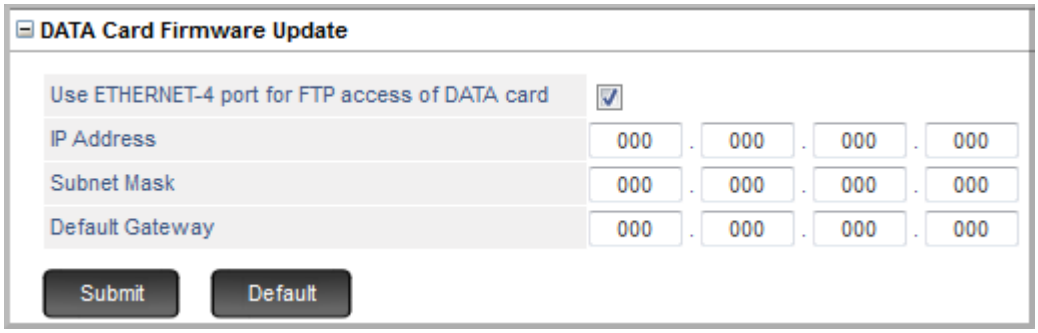
DATA Card Firmware Upgrade

- Click **DATA Card Firmware Update** to expand.



- Select the **Use ETHERNET-4 port for FTP access of DATA Card** check box to access the FTP.

- Click **OK**.



- Enter the **IP Address** through which the FTP of the Data Card can be accessed. Default: Blank
- Enter the **Subnet Mask**. Default: Blank
- Enter the **Default Gateway**. Default: Blank

- Click **Submit**.

Configuring BRI Trunks

BRI port of the SARVAM UCS configured for NT mode can be connected to the BRI Port of another System configured for TE mode. In such case, the SARVAM UCS will behave as Transit Exchange.

Dialing method (en bloc/overlap) on the BRI port is programmable. This is because many PSTN exchanges support only Overlap receiving. Hence, in such cases the BRI port (configured for TE mode) will send the called party number in overlap mode.

SARVAM UCS supports a software port entity called "ISDN Terminal" which can be connected to the BRI-NT port and will be treated as stations of the System.

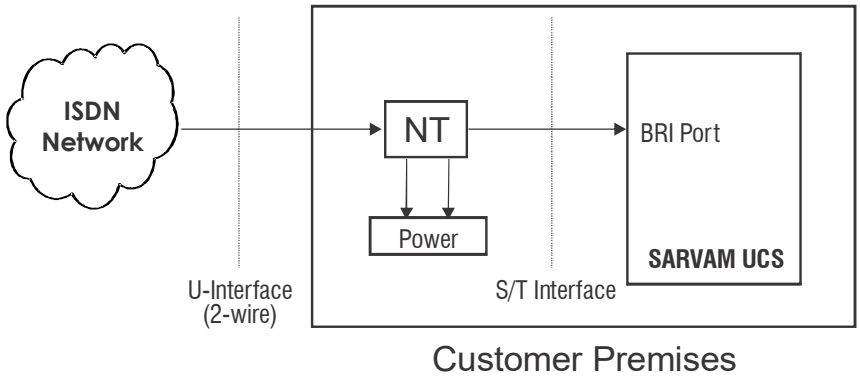
How it works

- The ISDN numbering plan with SARVAM UCS revolves around Multiple Subscriber Number (MSN) and Direct Dialing In (DDI).
- With MSN feature, practically all the station users or other specific terminals can be given a unique telephone number. One station can be assigned a number 2765400, other station can be assigned 2765401, etc. Hence, for one subscriber the Service Provider can assign multiple numbers. Hence this feature is called Multiple Subscriber Number.
- The SARVAM UCS can be programmed to land all the calls coming through various channels of the BRI line on the Operator just like in case of normal trunk.
- Alternatively, using DDI feature of ISDN the calls can be made to land directly on the desired stations. To accomplish this requirement, each station should be given a unique directory number. On assigning directory number, a table is formed internally called DDI table as shown below:

Station (S/w Port No.)	Directory Number
000	03
005	04
006	05
008	06
009	07

- When a caller calls a MSN number, the call lands on the System. The System compares the incoming number with the DDI table. If the incoming number matches with any number in the DDI table, it routes the call to the specific station. If the incoming number does not match with any number in the DDI table then it is matched with CLI number. If it matches with any number in the CLI table, the call is routed according to the CLI table. If the number does not match with either of these tables then the call is routed to the landing destination.

How to connect the BRI line?



- Most of the Service Providers provide the NT1 along with the BRI line.
- At the Customer's Premises, the BRI line is terminated on the NT1. The S/T interface of the NT1 is connected to BRI port of the SARVAM UCS.

  The configuration details of U interface (RJ-45) at NT1 are given below:

Pin Number	Pin Detail
4	Tx
5	Rx

  The configuration details of S/T interface (RJ-45) on NT1 are given below:

Pin Number	Pin Detail
3	Rx1
4	Tx1
5	Tx2
6	Rx2

Association of BRI port and ISDN Terminal:

- The 'ISDN terminal' is defined as a real port type in the system.
- Since 32 BRI ports are supported at present and as per protocol a maximum of 8 terminals can be connected to the BRI port; at present 64 software ports are supported for ISDN terminals, namely 01 to 64 (Maximum 64 Terminals).
- ISDN terminals (software ports) do not have any hardware slot and port Id of their own.
- ISDN terminals (software ports) are associated to the BRI software port and the BRI software port has a hardware slot and port ID of its own.
- Each ISDN terminal is assigned an access code (flexible number), station basic template, station advanced template and a CPU group.

The ISDN terminals do not have a direct software port number as other port types have. Rather the ISDN terminals have a derived software port number. Such derived software port numbers are used while programming extension name or assigning Station Basic Feature or Station Advanced Feature templates or programming Routing Group.

For Call routing from the BRI NT:

- *When call is made from the ISDN terminal the BRI port will check the calling party number sent by the ISDN Terminal. If the calling party number = Programmed access code of the ISDN Terminal which are assigned to the BRI port, the BRI Terminal will process the call further as per the Station Basic Feature Template and Station Advanced Feature Template assigned to the ISDN Terminal.*

- *If the ISDN terminal doesn't send the calling party number while making the OG call, the Station Basic/ Advanced Feature template assigned to the BRI port will be used, for call processing.*

- *When the ISDN terminal doesn't send the calling party number = Access code programmed for the BRI port on which it is connected, while making the OG call, the Station Basic/Advanced Feature Template assigned to the BRI port will be applied.*

- *When ISDN terminal sends calling party number which is not programmed for any ISDN terminals assigned to that particular BRI Port, neither its access code of that particular BRI port, the call will be dropped, by the BRI port.*

BRI port Orientation Type

- BRI port can be connected to a public ISDN exchange, private ISDN exchange or to an ISDN terminal (BRI Access).

- When the BRI port is connected to a Public ISDN exchange, it behaves as a terminal. SARVAM UCS supports this function of BRI Access by assigning it a parameter viz. Orientation = Terminal. The signaling used for this Orientation is Q.931-protocol.

- When the BRI port is connected to an ISDN Phone or ISDN Video phone, it behaves as a 'network'. SARVAM UCS supports this function of BRI Access by assigning it a parameter viz. Orientation = Network. In this case, the BRI port behaves as network. The signaling used for this Orientation is Q.931- protocol.

- When the BRI port is connected to a private ISDN (the main application of this configuration is CUG, feature transparency, etc.), the BRI port is used as a pipe of 128Kbps. It is of no significance which end acts as network and which acts as terminal (User) since the role of the BRI port in such case will be to route the call depending on the signaling protocol applied on the BRI port (128Kbps link). SARVAM UCS supports this function of BRI Access by assigning it a parameter viz. Orientation = Tie-line. In this case, the BRI port behaves as 128Kbps link. The signaling used for this Orientation can be any of the Inter-exchange signaling protocol for BRI Access. The most commonly used is QSIG.

TEI negotiation on BRI port

- BRI port supports both automatic and fixed TEI negotiation, so that it can be integrated with a system of another vendor. If 'Fixed' mode is selected, its value is required to be configured using SE commands or Jeeves. TEI is 'Terminal Endpoint Identifier' protocol for negotiation used while connecting to the BRI port with remote BRI port.

Applications of BRI-NT Port:

Two applications are described below:

- Video Phone
- Making Data Call

Video Phone

- An important application of BRI-NT is establishing a Video Call using Video phone. It is used just like other ISDN Phones.

- Video Phone can be used with SARVAM UCS supporting a BRI connection. SARVAM UCS does not support call transfer from one Video Terminal to another Terminal. But the call routing is implemented by preparing suitable OG Trunk Bundle Group (OGTBG).

- The OGTBG for the Video Phone should be so formed by the SE that a BRI channel will be allotted by the Call Processing logic (preferably).

- If the OGTBG formed by the SE contains a non-ISDN trunk and if by the OGTBG logic this non-ISDN trunk is to be allotted to the ISDN phone which has requested 2 B-channels then the call will be dropped. The non-ISDN trunk is allowed only if the ISDN user makes an audio call. The system will allot first channel even if other channels are available.

OG Call using Video Phone:

- The SARVAM UCS will support supplementary feature viz. Hold only. If the terminal equipment uses Hold feature to hold a video call, it will be applicable.

- The SARVAM UCS does not support logic for converting audio call to video call.

- Instead, the Video Phone user will have to use one Trunk Access Code (and hence OGTBG) to make and receive Video call and another TAC to make an audio call.

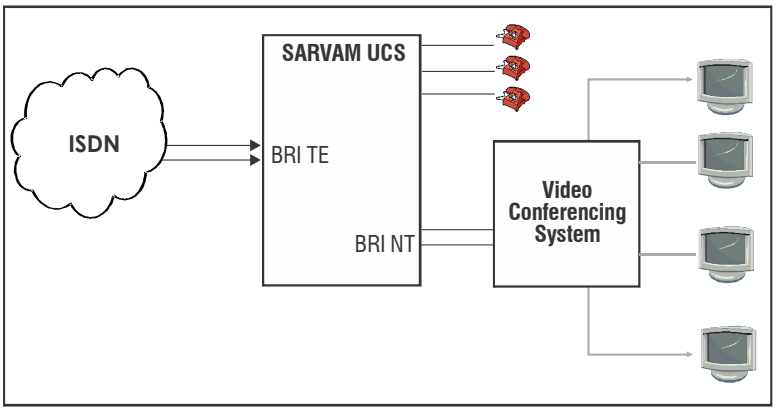
IC call to a Video Phone:

- User can program normal DDI routing logic to route the IC call to the Video Phone using the first or additional channels.

- Depending upon terminal equipment call, the call will be answered by the Operator and transferred to the Video Phone.

Video Conference

As shown in the figure SARVAM UCS supports Video conference from BRI-NT port.



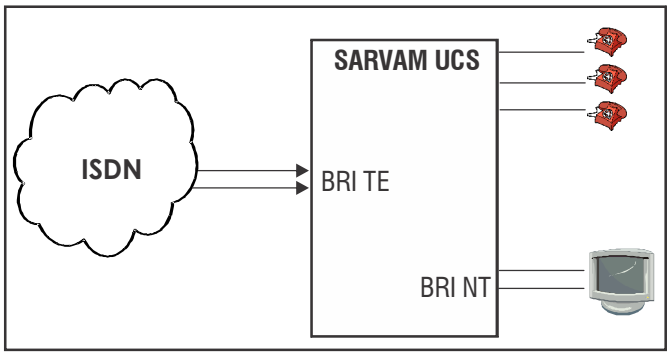
- This feature works only if it is supported by the Service Provider. Video Conference is established mainly by the Video Conferencing (VC) equipment.
- A Video Conference system (H.320) can be connected to any of the BRI-NT port.
- For Video conferencing, three BRI-NT ports of SARVAM UCS are connected to the three ports of the VC equipment (The remaining one port of the Video Conference system remains free).
- Thus user will assign at least 6 B-channels to the OGTBG that is to be assigned to the VC equipment. (User can assign BRI-NT software port nos. 01 to 03 to the Video Conferencing equipment).
- The user will program the routing of IC calls to the Video Conferencing equipment using DDI routing table for placing IC video conferencing calls.
- More than one VC-equipment can be connected to a BRI-NT Port.

- *Video Conferencing generally requires 6 B-channels. But Video Conferencing can also be done at lower bit rates also using the "aggregation" of 6 B channels which must be supported by the VC equipment.*
- *Please note that a Video Conference call cannot be transferred or kept on Hold.*

Making Data Calls

- SARVAM UCS BRI-NT port supports data communication also.

- For data-communication, connect the Router supporting ISDN-BRI or a Computer with the BRI Card (BRI-NT port) as shown below:



Using this feature following applications are supported:

- The computers connected in the LAN can browse the net through the BRI.
- Remote LAN Access, the Computers in the LAN can access the computer/computers in LAN at the remote end (Branch office/Home office).
- Files can be transferred from one LAN to another.

OG Data calls from the Router:

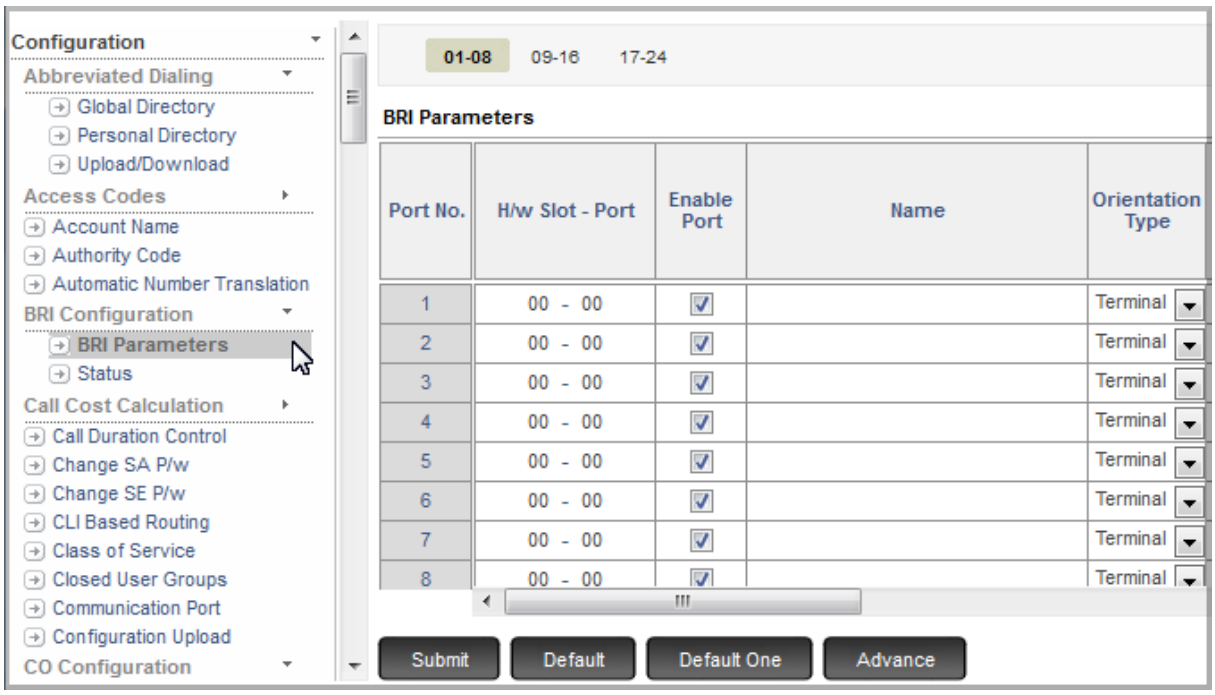
- The data call can be made by the router requesting desired number of channels. This establishes a live connection between the Router and the ISP through the system. The users on the LAN can browse the net as normal using Internet Explorer or Netscape Navigator.
- For this, the SARVAM UCS will allocate data channels only on the BRI-TE port so as to leave other channels for speech calls when the system detects the call to be a data call.
- Similarly, a Remote Computer can be accessed (Remote LAN Access) by dialing the Remote users' number (The remote end system should be so programmed that the call made to a number lands directly on the Router.) This establishes a permanent connection between the two Routers (and hence two LAN networks).
- Now the user of the system can access the computer in LAN at the remote end in the same way as accessing another computer on the same network.

IC Data calls to the Router:

- Program the Trunk Landing Group such that all data calls will land on the BRI-NT port to which the Router is connected using CLI based Routing logic or DDI based routing logic.
- However, while routing call on the BRI-NT port, the system will check that the data channels reserved for data communication on the BRI-NT port are enough to establish the call. Otherwise the call will be rejected.
- The call will be rejected if the number of channels reserved for data calls, are already busy with one data-call.

Configuring BRI Parameters using Jeeves

- Log in as System Engineer.
- Under **Configuration**, click **BRI Configuration**.
- Click **BRI Parameters**.



- SARVAM UCS will assign the **Hardware Slot-Port** automatically, when any card is inserted in the system.

  Hardware slot is the number of the Universal slot of SARVAM UCS in which the BRI Card is inserted. Range of slot number is 01-16. Port is the number of BRI hardware port on the card to which the BRI line is connected. Range of Port is from 00-99.

  If you want to de-assign the Hardware Slot and Port, Enter '00' in both fields. By default, Hardware slot-Port is 00–00.

- Keep the **BRI Port** enabled.

  Clear the **Enable Port** check box, only if you do not want to use this port. By default, it is enabled.

- You may assign a **Name** to the BRI Port for identification of the port. The Name may consist of a maximum of 12 characters. By default, it is blank.

- Set **Orientation Type** for BRI Port. You may select from the following options:
  - Terminal
  - Network
  - Tie Line

  By default, Orientation Type is Terminal.

Orientation Type - Terminal

If you select **Terminal** as **Orientation Type**, configure the following:

- Enable **Power Feed** flag, if you want SARVAM UCS to feed power to the terminal equipment. By default, it is disabled.
- Select **Interface Type** as Point-to-Point or Point-to-Multipoint according to your installation. By default, it is Point-to-Multipoint.
- The **TEI Negotiation** mode will be set automatically as per the Interface Type selected for the BRI Port.
  - If the Point-to-Point is selected as the Interface Type, the **TEI Negotiation** will be set as **Fixed**.
    - You must also configure **TEI Negotiation Value (for Fixed Mode)**. TEI Value range is from 00 to 63. By default, it is 00. TEI Value of BRI (TE) port of SARVAM UCS should match with the TEI Value expected by the NT equipment at other end.
  - If Point-to-Multipoint is selected as the Interface Type, the **TEI Negotiation** will be set as **Auto**.

When you change the TEI mode on any port, the BRI Card will restart.

- You may configure the port to **Treat Incoming call as Trunk** or **Station**.

  If you select **Trunk**, the system will treat all incoming calls as external calls landing on the trunk. The calls will be routed as per the **Trunk Feature Template** assigned to the BRI Port.

  If you select **Station**, you must also assign a Station Basic Feature Template and Station Advanced Feature Template to the BRI Port.

  When you select Station, the system will treat the calling party as an extension user. The user will have access to all the features and facilities of the system, as per the Station Basic Feature Template and Station Advanced Feature Template assigned to the BRI Port.

  By default, Trunk is selected.

- *If Point-to-Point is selected as the Interface Type, you can select the option **Trunk** or **Station** for the parameter **Treat Incoming call as**.*
- *If Point-to-Multipoint is selected as the Interface Type, only **Station** can be set as the option for the parameter **Treat Incoming call as**.*
- *If **Station** is selected as the option for **Treat Incoming call as**, the user will only be able to:*
  - *Dial Flexible Numbers*
  - *Dial Operator Code*
  - *Dial Trunk Access Code for making outgoing calls*
  - *Access the Global Directory*
  - *Make calls within the Closed User Group*

- Different countries use specific type of ISDN switch. The type of switch determines various factors such as how many ISDN devices would be handled, which B-channel will support voice, video, data etc. Select **ISDN Switch Variant** supported by your country. You may select from the following options:
  - ATT_4ESS
  - ATT_5ESS

- AUSTRALIA
- AUTO CONFIG
- DMS_100
- ETSI_NET3
- NTT_INS64
- SWV_HONG_KONG
- US_NI1
- US_NI2
- VN_X

By default, it is ETSI_NET3.

- Configure **Outgoing (OG) Reference ID** for working hours, non-working hours and break hours. By default, OG Reference ID is 00.

- Configure **Incoming (IC) Reference ID** for working hours, non-working hours and break hours. By default, IC Reference ID is 00.

- Assign **Trunk Feature Template** to the BRI Port. Trunk Feature Template is a set of general features that define the behavior of a Trunk Port. By default, Template 01 is assigned to all BRI Ports.

  For more details, see “Trunk Feature Template”.

- Assign a **Cost Factor** to the BRI Port. By default, all the BRI Ports are assigned Cost Factor 01.

  For more details, see “Cost Factor”.

- **Station Basic Feature Template** assigned to the BRI Port is displayed in this field. Station Basic Feature Template is a set of general features that define the basic behavior of a station. By default, Template 01 is assigned to all BRI Ports.

  For more details, see “Station Basic Feature Template”.

- **Station Advanced Feature Template** assigned to the BRI Port is displayed in this field. Station Advanced Feature Template is a set of advanced features, to be applied on extensions such as CLIP, Floor Service, Walk-in Class of Service. By default, Template 01 is assigned to all BRI Ports.

  For more details, see “Station Advanced Feature Template”.

- If you have set **Terminal** as the **Orientation Type**, you must select the type of network with which the port is to be **Interfaced With**. Select the type of network from the following:
  - Public ISDN
  - Private ISDN

  Default: Public ISDN

- If you want SARVAM UCS to display the called party number as the CLI for incoming calls, select the **Display Called Party Number as CLI** check box. By default, Display Called Party Number as CLI option is disabled.

  This option is useful when a single BRI line connection and Operator are shared by more than one organization. If you enable this option, make sure:
  - you configure the names and corresponding numbers of the organizations sharing the line in the Global Directory of SARVAM UCS.

- the Operator has a DKP or an Extended IP Phone or a Mobile UC Client.

With this option enabled the Operator will be able to handle calls more efficiently. When there is an incoming call, SARVAM UCS matches the number with the numbers in the Global Directory. If a match is found SARVAM UCS displays the company name configured for that entry to the Operator, that is, the CLI will display the called party number and name.

After the Operator answers the call, the CLI will change and display the calling party number and name (if configured in the Global Directory).

If you keep this option disabled, the calling party number and name will be displayed as the CLI, both during an incoming call and after the call is answered by the Operator.

You can configure the Display Called Party Number as CLI option only from Jeeves.

- Select the **Allow Incoming CLI Modification** check box if you want to apply 'Allow Incoming CLI Modification' on the BRI Port. By default, it is disabled.

  Incoming CLI Modification is useful in countries where the Calling Line Identification (CLI) received by the System extension users must be suitably modified before it can be used to dial out the number. To know more, see “Incoming CLI Modification”.

If you receive CLI in dialable format, there is no need to use this feature. In such case, keep the flag disabled.

  *For an incoming call on the BRI trunk, the Incoming CLI Modification will be applied only when both — the **Allow Incoming CLI Modification** check box and the **Enable Incoming CLI Modification** check box in System Parameters — are enabled.*

- Select **Layer 1 Mode** depending upon the type of line terminated on BRI port of SARVAM UCS.

  The Public ISDN provides different types of lines in different countries as mentioned below:
  - On Demand
  - Always ON

  In 'On Demand' type of line, layer 1 (physical layer) remains 'down' when the line is idle. When the network places incoming call, it activates the layer 1 and places the call. When the terminal makes out going call, the layer 1 gets activated automatically.

  In 'Always ON' type of line, layer 1 is always 'up', in normal condition. By default, Layer 1 Mode is 'Always ON'.

  When the system detects 'layer 1 down', it removes the unhealthy line from the OGTBG. If you have assigned a DSS Key to the BRI Port/channel, the faulty condition is indicated by LED indication.

  DSS key LED behavior

  When Layer 1 of BRI line is down or not connected, DSS key LED will glow 1 sec On and 3 sec Off in violet color.

  If Layer 1 of BRI line is up, the DSS key LED will be turned off.

- *SARVAM UCS uses the BRI Port to place call only if the layer 1 is 'up'. When layer 1 goes 'down', SARVAM UCS considers the line as un-healthy and will not use this BRI as destination port. SARVAM UCS will place the call using the alternate port programmed in the same routing group.*

- *The default Layer 1 Mode is 'Always ON'. Hence, if the interfaced line is of the type 'On Demand', the calls will not get routed through BRI port, unless the 'Layer 1 Mode' is changed to 'On Demand'.*

- *When the port is un-healthy, the SARVAM UCS routes the call using other healthy port. However, this depends upon the member selection method and other ports programmed in the OG Trunk Bundle Group (OGTBG). Refer* chapter *"OG Trunk Bundle Group"* for more details.

- Select **Priority** for the BRI Port. Priority is the precedence given to certain trunks and extensions over others in being answered by the destination extension. You can select from 1 to 9. By default, Priority 5-Normal is set for all BRI Ports.
  For know more about Priority feature, see "Priority".

- Select **Return Call to Original Caller (RCOC)** check box to enable this feature on the BRI Port. By default, RCOC flag is disabled.

  For know more about RCOC feature, see "RCOC (Return Call to Original Caller)".

- Set **Overlap Receiving Timer** for BRI Port. This timer is relevant while receiving the called party number information in overlap receiving mode. By default, it is set to 15 seconds.

- When the SETUP Message is sent by SARVAM UCS to the network (ISDN exchange), the exchange responds by sending SETUP ACK (Acknowledgment), and dial tone is played to the caller. The time taken by the exchange to respond to the SETUP message may vary from exchange to exchange. Set the **SETUP Response Timer (sec)** as per the time taken by the network to respond to the SETUP message and play dial tone to the caller.

  Valid Range of the timer is 01 to 20 seconds. By default it is set to 4 seconds.

Change the default settings only if required. If the time you set is less than the time taken by the exchange to respond, no dial tone will be played to the caller.

You can configure the SETUP Message Timer only from Jeeves.

- Configure **Idle Code** for the BRI Port. Range of Idle Code is from 000 to 255. By default, it is 127 (7F).

  The binary equivalent of the configured value (000 to 255) is sent on the channel to signify that the channel is idle (or Used). This setting depends on the network. Most commonly applicable values are 7F and FF (Binary equivalent is 0111 1111 and 1111 1111, decimal equivalent is 127 and 255).

- Select the number of **Channels** to be **reserved** for **Data Communication**. Channel count is from 0 to 2. By default, number of channels reserved for Data Communication is 02.

- Select the number of **Channels** to be **reserved** for **Outgoing Calls**. Channel count is from 0 to 2. By default, number of channels reserved for Outgoing Calls is 02.

- Select the number of **Channels** to be **reserved** for **Incoming Calls**. Channel count is from 0 to 2. By default, number of channels reserved for Incoming Calls is 02.

- Select the required option for sending the **Caller- Type of Numbering Plan (TON)** from the following:
  - Unknown
  - International
  - National
  - Network Specific
  - Subscriber
  - Abbreviated
  - Reserved

  By default, Unknown is selected.

- Select the required option for sending the **Caller- Numbering Plan Identification (NPI)** from the following:
  - Unknown
  - ISDN Numbering
  - Data Numbering
  - Telex Numbering
  - National Numbering
  - Private
  - Reserved

  By default, ISDN Numbering is selected.

- Select the required option for sending the **Called-Type of Numbering Plan (TON)** from the following:
  - Unknown
  - International
  - National
  - Network Specific
  - Subscriber
  - Abbreviated
  - Reserved

  By default, Unknown is selected.

- Select the required option for sending the **Called-Numbering Plan Identification (NPI)** from the following:
  - Unknown
  - ISDN Numbering
  - Data Numbering
  - Telex Numbering
  - National Numbering
  - Private
  - Reserved

  By default, ISDN Numbering is selected.

- Select the **Screening Indicator** as provided to you by your exchange. You can select from:
  - User provided, not screened
  - User Provided, verified and passed
  - User Provided, verified and failed
  - Network Provided

  By default, User provided, not screened.

- Select the **Bearer Service** supported by your service provider. You can select from:
  - Speech

- 3.1 KHz Audio

  By default, Speech is selected.

- Configure **Call Budget** parameters for the BRI Ports. Call Budget is an expense control feature of SARVAM UCS that allows you to keep track of the cost of phone calls made from the BRI Port.By default, Call Budget is enabled. If you wish to change the default configuration or disable it for this BRI port, configure the parameters as per your requirement:

  - **Type:** Select the type of Call Budget, that is, Amount or Minutes or Calls to be applied on the BRI Port. By default, Minutes is selected as the Call Budget type.To disable select Type as None.

  - **Amount:** If you selected 'Amount' as the Call Budget Type, enter the Budget Amount in this field. By default the Amount is set to 999999.

  - **Minutes:** If you selected 'Minutes' as the Call Budget Type, enter the number of Minutes in this field. By default the number of minutes is set as 000300.

  - **Calls:** If you selected 'Calls' as the Call Budget Type, enter the number of Calls in this field. By default the number of calls is set to 9999.

  - **Scheduled Reset:** Enable this flag if you want the Call Budget Amount/Minutes/Number of Calls to be reset on a particular date of every month.

  - **Scheduled (Date):** Select the date of the month (Daily or 1-31) on which you want the Call Budget Amount/Minutes/Number of Calls to be reset every month. You may select 'Daily' if your plan suggests so.

- **Call Back:** This parameter is related to the ‘Call Back on Trunk Port’ feature. If you want to enable the 'Call Back on Trunk Port' feature on this BRI Port, configure the following parameters:

  - **Enable Call Back:** Enable this flag to activate the Call Back on Trunk Port feature. By default, this flag is disabled on all trunk port types. By default, the flag is disabled.

  - **Call Back Timer (sec):** This is the duration for which the system waits for the caller to disconnect before applying the Call Back. The range of this timer is from 01 to 99 seconds. By default, it is set to 10 seconds.

  - **Call Back Mode:** Select from the following options how a ‘Call Back’ call answered by the remote party should be routed:
    - Built-in Auto Attendant
    - PIN Authentication - Multiple Calls
    - CLI Authentication - Multiple Calls
    - CLI Authentication - Single Call - Answer Signaling
    - Operator

    By default, Operator is selected as the Call Back Mode.

  - **Call Back on:** This parameter allows you to select if the call back should be made to the same number that was received or to a different number. If you want the call back to be made to the same number select the ‘CLI number”. If you want the call back to be made to a different number, select ‘Alternate Number’.

By default, CLI number is selected for Call Back.

- **Incoming Number List:** Program the number strings that are eligible for Call Back in this List. By default, Number List 15 is assigned to Call Back Incoming Number List.

  Number List 15 is also assigned to all BRI Ports as well as all other Trunk port types. If you want the same numbers strings to be programmed commonly for all BRI Ports and Trunk Port types, retain this list.

  If you want a different set of number strings to be programmed for this BRI Port, select a different Number List, and assign it to the BRI Port.

  You may program the Incoming Number List either from the ‘Number List’ page or by clicking the ‘Incoming Number List’ link to reach the Number List page.

  Refer the topic “Number Lists” to know more, and for configuration instructions.

- **Outgoing Number List:** Program the number strings that are to be called back in this List. For each number string you programmed in the ‘Incoming Number List’, you must program in the corresponding index in the Outgoing Number List a number to which the call back is to be made. For example, for the number string programmed at Index 1 in the Incoming Number List, a corresponding number string at the same Index, Index 1, should be programmed in the ‘Outgoing Number List’.

  By default, Number List 16 is assigned to Outgoing Number List.The same Number List 16 is also assigned to all BRI Ports as well as all other Trunk port types.

  You may program the default number list, or a different number list and assign it to this BRI Port.

  You may program the Outgoing Number List either from the ‘Number List’ page or by clicking the ‘Outgoing Number List’ link to reach the Number List page.

  Refer the topic “Number Lists” to know more, and for configuration instructions.

- **Call Back from:** This parameter determines the trunk port to be used to make the call back. The call back can be made using the same port or an “OG Trunk Bundle Group”.

  Select ‘Same Port’ if you want the call back to be made using the same port on which the missed call is received. If you select OGTB Group, the call back will be made using the OGTB Group, which you have defined.

  By default, Same Port is selected.

- **OGTB Group:** If you selected OGTB Group for making the call back in the previous parameter, you must define the OGTB Group that must be used in this parameter.

  By default, OGTB Group 01 is assigned.

  If you want the system to select the lowest cost trunk for making the call back, enable Least Cost Routing on the OGTB Group that you define here for Call Back.

- Configure **Pause Timer** for the BRI Port. Range of Pause Timer is from 1 to 9 seconds. By default, it is set to 3 seconds for all BRI Ports.

This Timer is required to insert delay between the digits while dialing out DTMF digits on the BRI port. One of the applications for using this parameter is Multi-stage dialing. Refer chapter "Multi-Stage Dialing".

For example, if PPP2 is to be outdialed and Pause timer is programmed as 3 seconds, the SARVAM UCS will out dial the digit 2 after 9 seconds i.e delay of individual P i.e 3+3+3 =9.

- Configure **DTMF On Time** for the BRI Port. Range of DTMF On Time is from 051 to 255 ms. By default, it is set to 102 ms for all BRI Ports.

  The DTMF On Time is the time for which the DTMF digit which is to be outdialed by the SARVAM UCS remain On. One of the applications for using this parameter is Multi-stage dialing. Refer chapter "Multi-Stage Dialing".

- Configure **DTMF Inter Digit Pause Timer** for the BRI Port. Range of Inter Digit Pause Timer is from 051 to 255 ms. By default, it is set to 102 ms for all BRI Ports.
  Inter Digit Pause Timer is the time for which the system will wait while receiving the dialing digits to consider it as end-of-dialing.

- Select **Category (Logical Partition)** for the BRI Port. You may select from the following options:
  - 1
  - 2
  - 3
  - 4

  By default, 1 is selected for all BRI Ports. Refer the feature description "Logical Partition" to know more.

- If SARVAM UCS is to be used as a Gateway, enable **Gateway Application-Answer Signaling** on the BRI Port and configure **DTMF String**. By default, Gateway Application-Answer Signaling is disabled and CCC is configured as DTMF String.

  For more details, see "Gateway Application-Answer Signaling".

- Enter appropriate **Debug Code (Level 1 to 4)** to obtain debug information of various parts of BRI Card on the COM Port. By default, debug is off for all BRI Ports for all levels.

  Code is the value for the specified level to turn ON the debug for the parameters. Code range is from 000 to 255. Code value '000' for each level will turn off that level's debug.

  Level and Code for BRI Port are as specified below:

  Level 1

Unused	Flow	NLS	LAP	SVC Primitives	Variables	State	Primitives

Code	Meaning
001	Primitives
002	State
004	Variables
008	SVC Primitives
016	LAP
032	NLS

Code	Meaning
064	Flow Debug
000	Debug Off

Level 2

Unused	Unused	Unused	DTMF Digit	Unused	Layer 4	Unused	Unused

Code	Meaning
004	Layer 4
016	DTMF Digit
000	Debug Off

Level 3

Unused	Unused	Unused	HDLC D-Channel	Unused	Unused	Unused	Unused

Code	Meaning
016	HDLC D-Channel
000	Debug Off

Level 4

Unused	Unused	Unused	Unused	Unused	Unused	OS Task	NI

Code	Meaning
001	Network Interface (NI)
002	OS Task
000	Debug Off

Example:

- If '004' decimal value is entered as Debug Code for Level 1, then its binary equivalent '100' (0000100) indicates that debug for "Variables" will get enabled.

  If "007" decimal value is entered as a Debug Code for the Level 1, then its binary equivalent "111" (00000111) indicates that debug for "Primitives", "States" and "Variables" will get enabled.

Orientation Type - Network

If you select Network as Orientation Type, configure the following:

- Select **BRI-NT Interface Type** as Point-to-Point or Point-to-Multipoint according to your installation. By default, it is Point-to-Multipoint.
- The **TEI Negotiation** mode will be set automatically as per the Interface Type selected for the BRI Port.

- If the Point-to-Point is selected as the Interface Type, the **TEI Negotiation** will be set as **Fixed**.

    - You must also configure **TEI Negotiation Value (for Fixed Mode)**. TEI Value range is from 00 to 63. By default, it is 00. TEI Value configured in the BRI (NT) should match with the TEI Value configured in the Terminal Equipment connected with it.

- If the Point-to-Multipoint is selected as the Interface Type, the **TEI Negotiation** will be set as **Auto**.

When you change the TEI mode on any port, the BRI Card will reset.

- You may configure the port to **Treat Incoming call as Trunk** or **Station**.

  If you select **Trunk**, the system will treat all incoming calls as external calls landing on the trunk. The calls will be routed as per the **Trunk Feature Template** assigned to the BRI Port.

  If you select **Station**, you must also assign a Station Basic Feature Template and Station Advanced Feature Template to the BRI Port.

  When you select Station, the system will treat the calling party as an extension user. The user will have access to all the features and facilities of the system, as per the Station Basic Feature Template and Station Advanced Feature Template assigned to the BRI Port.

  By default, Trunk is selected.

- *If Point-to-Point is selected as the Interface Type, you can select the option **Trunk** or **Station** for the parameter **Treat Incoming call as**.*

- *If Point-to-Multipoint is selected as the Interface Type, only **Station** can be set as the option for the parameter **Treat Incoming call as**.*

- *If **Station** is selected as the option for **Treat Incoming call as**, the user will only be able to:*
    - *Dial Flexible Numbers*
    - *Dial Operator Code*
    - *Dial Trunk Access Code for making outgoing calls*
    - *Access the Global Directory*
    - *Make calls within the Closed User Group*

- Different countries use specific type of ISDN switch. The type of switch determines various factors such as how many ISDN devices would be handled, which B-channel will support voice, video, data etc. Select **ISDN Switch Variant** supported by your country. You may select from the following options:
    - ATT_4ESS
    - ATT_5ESS
    - AUSTRALIA
    - AUTO CONFIG
    - DMS_100
    - ETSI_NET3
    - NTT_INS64
    - SWV_HONG_KONG
    - US_NI1
    - US_NI2
    - VN_X

By default, it is ETSI_NET3.

- Configure **Outgoing (OG) Reference ID** for working hours, non-working hours and break hours. By default, OG Reference ID is 00.

- Configure **Incoming (IC) Reference ID** for working hours, non-working hours and break hours. By default, IC Reference ID is 00.

- Assign **Trunk Feature Template** to the BRI Port. Trunk Feature Template is a set of general features that define the behavior of a Trunk Port. By default, Template 01 is assigned to all BRI Ports.
  For more details, see “Trunk Feature Template”.

- Assign a **Cost Factor** to the BRI Port. By default, all the BRI Ports are assigned Cost Factor 01.

  For more details, see “Cost Factor”.
- **Station Basic Feature Template** assigned to the BRI Port is displayed in this field. Station Basic Feature Template is a set of general features that define the basic behavior of a station. By default, Template 01 is assigned to all BRI Ports.

  For more details, see “Station Basic Feature Template”.

- **Station Advanced Feature Template** assigned to the BRI Port is displayed in this field. Station Advanced Feature Template is a set of advanced features, to be applied on extensions such as CLIP, Floor Service, Walk-in Class of Service. By default, Template 01 is assigned to all BRI Ports.

  For more details, see “Station Advanced Feature Template”.

- If you want SARVAM UCS to display the called party number as the CLI for incoming calls, select the **Display Called Party Number as CLI** check box. By default, Display Called Party Number as CLI option is disabled.

  This option is useful when a single BRI line connection and Operator are shared by more than one organization. If you enable this option, make sure:
  - you configure the names and corresponding numbers of the organizations sharing the line in the Global Directory of SARVAM UCS.
  - the Operator has a DKP or an Extended IP Phone or a Mobile UC Client.

  With this option enabled the Operator will be able to handle calls more efficiently. When there is an incoming call, SARVAM UCS matches the number with the numbers in the Global Directory. If a match is found SARVAM UCS displays the company name configured for that entry to the Operator, that is, the CLI will display the called party number and name.

  After the Operator answers the call, the CLI will change and display the calling party number and name (if configured in the Global Directory).

  If you keep this option disabled, the calling party number and name will be displayed as the CLI, both during an incoming call and after the call is answered by the Operator.

You can configure the Display Called Party Number as CLI option only from Jeeves.

- Select the **Allow Incoming CLI Modification** check box if you want to apply 'Allow Incoming CLI Modification' on the BRI Port. By default, it is disabled.

Incoming CLI Modification is useful in countries where the Calling Line Identification (CLI) received by the System extension users must be suitably modified before it can be used to dial out the number. To know more, see “Incoming CLI Modification”.

If you receive CLI in dialable format, there is no need to use this feature. In such case, keep the flag disabled.

*The Incoming CLI Modification will be applied only when both — **Allow Incoming CLI Modification** check box and **Enable Incoming CLI Modification** check box in System Parameters — are enabled.*

- Select **Layer 1 Mode** depending upon the type of line terminated on BRI port of SARVAM UCS.

  The Public ISDN provides different types of lines in different countries as mentioned below:
  - On Demand
  - Always ON

  In 'On Demand' type of line, layer 1 (physical layer) remains 'down' when the line is idle. When the network places incoming call, it activates the layer 1 and places the call. When the terminal makes out going call, the layer 1 gets activated automatically.

  In 'Always ON' type of line, layer 1 is always 'up', in normal condition. In this type of line, 'layer 1 down' indicates fault condition and calls get failed.

  By default, Layer 1 Mode is 'Always ON'.

  - *When Layer 1 mode is set as 'Always ON', the SARVAM UCS uses this BRI to place call only if the layer 1 is 'up'. When layer 1 goes 'down', SARVAM UCS considers the line as un-healthy and will not use this BRI as destination port. SARVAM UCS will place the call using the alternate port programmed in the same routing group.*
  - *When Layer 1 mode is set as 'On Demand', SARVAM UCS will not check the Layer 1 condition while placing call through this BRI.*
  - *The default Layer 1 Mode is 'Always ON'. Hence, if the interfaced line is of the type 'On Demand', the calls will not get routed through BRI port, unless the 'Layer 1 Mode' is changed to 'On Demand'.*
  - *When the port is un-healthy, the SARVAM UCS routes the call using other healthy port. However, this depends upon the member selection method and other ports programmed in the OG Trunk Bundle Group (OGTBG). Refer* chapter *“OG Trunk Bundle Group”* for more details.

- Select **Priority** for the BRI Port. Priority is the precedence given to certain trunks and extensions over others in being answered by the destination extension. You can select from 1 to 9. By default, Priority 5-Normal is set for all BRI Ports.

  For know more about Priority feature, see “Priority”.

- Select **Return Call to Original Caller (RCOC)** check box to enable this feature on the BRI Port. By default, RCOC flag is disabled.

  For know more about RCOC feature, see “RCOC (Return Call to Original Caller)”.

- Set **Overlap Receiving Timer** for BRI Port. This timer is relevant while receiving the called party number information in overlap receiving mode. By default, it is set to 15 seconds.

- When the SETUP Message is sent by SARVAM UCS to the network (ISDN exchange), the exchange responds by sending SETUP ACK (Acknowledgment), and dial tone is played to the caller. The time taken by the exchange to respond to the SETUP message may vary from exchange to exchange. Set the **SETUP Response Timer (sec)** as per the time taken by the network to respond to the SETUP message and play dial tone to the caller.

  Valid Range of the timer is 01 to 20 seconds. By default it is set to 4 seconds.

Change the default settings only if required. If the time you set is less than the time taken by the exchange to respond, no dial tone will be played to the caller.

You can configure the SETUP Message Timer only from Jeeves

- Configure **Idle Code** for the BRI Port. Range of Idle Code is from 000 to 255. By default, it is 127 (7F).

  The binary equivalent of the configured value (000 to 255) is sent on the channel to signify that the channel is idle (or Used). This setting depends on the network. Most commonly applicable values are 7F and FF (Binary equivalent is 0111 1111 and 1111 1111, decimal equivalent is 127 and 255).

- Select the number of **Channels** to be **reserved** for **Data Communication**. Channel count is from 0 to 2. By default, number of channels reserved for Data Communication is 02.

- Select the number of **Channels** to be **reserved** for **Outgoing Calls**. Channel count is from 0 to 2. By default, number of channels reserved for Outgoing Calls is 02.

- Select the number of **Channels** to be **reserved** for **Incoming Calls**. Channel count is from 0 to 2. By default, number of channels reserved for Incoming Calls is 02.

- Select the required option for sending the **Caller- Type of Numbering Plan (TON)** from the following:
  - Unknown
  - International
  - National
  - Network Specific
  - Subscriber
  - Abbreviated
  - Reserved

  By default, Unknown is selected.

- Select the required option for sending the **Caller- Numbering Plan Identification (NPI)** from the following:
  - Unknown
  - ISDN Numbering
  - Data Numbering
  - Telex Numbering
  - National Numbering
  - Private
  - Reserved

  By default, ISDN Numbering is selected.

- Select the required option for sending the **Called-Type of Numbering Plan (TON)** from the following:
  - Unknown
  - International
  - National
  - Network Specific
  - Subscriber
  - Abbreviated
  - Reserved

  By default, Unknown is selected.

- Select the required option for sending the **Called-Numbering Plan Identification (NPI)** from the following:
  - Unknown
  - ISDN Numbering
  - Data Numbering
  - Telex Numbering
  - National Numbering
  - Private
  - Reserved

  By default, ISDN Numbering is selected.

- Select the **Bearer Service** supported by your service provider. You can select from:
  - Speech
  - 3.1 KHz Audio

  By default, Speech is selected.

- Configure **Call Budget** parameters for the BRI Ports. Call Budget is an expense control feature of SARVAM UCS that allows you to keep track of the cost of phone calls made from the BRI Port.

  - **Type:** Select the type of Call Budget, that is, Amount or Minutes or Calls to be applied on the BRI Port. By default, no Call Budget type is selected.

  - **Amount:** If you selected 'Amount' as the Call Budget Type, enter the Budget Amount in this field. By default the Amount is set to 999999.

  - **Minutes:** If you selected 'Minutes' as the Call Budget Type, enter the number of Minutes in this field. By default the number of minutes is set to 999999.

  - **Calls:** If you selected 'Calls' as the Call Budget Type, enter the number of Calls in this field. By default the number of calls is set to 9999.

  - **Scheduled Reset:** Enable this flag if you want the Call Budget Amount/Minutes/Number of Calls to be reset on a particular date of every month.

  - **Scheduled (Date):** Select the date of the month (Daily or 1-31) on which you want the Call Budget Amount/Minutes/Number of Calls to be reset every month. You may select 'Daily' if your plan suggests so.

- Configure **Call Back** parameters for the BRI Ports. Call Back is used to respond to missed calls from particular numbers on the BRI Port.

- **Enable Call Back:** Enable this flag to activate the Call Back on Trunk Port feature. By default, the flag is disabled.

- **Call Back Timer (sec):** This is the duration for which the system waits for the caller to disconnect before applying the Call Back. The range of this timer is from 01 to 99 seconds. By default, it is set to 10 seconds.

- **Call Back Mode:** From the following options select how a 'Call Back' call answered by the remote party should be routed:
  - Built-in Auto Attendant
  - PIN Authentication - Multiple Calls
  - CLI Authentication - Multiple Calls
  - CLI Authentication - Single Call - Answer Signaling
  - Operator

  By default, Operator is selected as the Call Back Mode.

- **Call Back on:** This parameter allows you to select if the call back should be made to the same number that was received or to a different number. If you want the call back to be made to the same number select the 'CLI number". If you want the call back to be made to a different number, select 'Alternate Number'.

  By default, CLI number is selected for Call Back.

- **Incoming Number List:** Program the number strings that are eligible for Call Back in this List. By default, Number List 15 is assigned to Call Back Incoming Number List.
  Number List 15 is also assigned to all the BRI Ports as well as all other Trunk port types. If you want the same numbers strings to be programmed commonly for all the BRI Ports and Trunk Port types, retain this list.

  If you want a different set of number strings to be programmed for this CO Trunk, select a different Number List, and assign it to the CO trunk port.

  You may program the Incoming Number List either from the 'Number List' page or by clicking the 'Incoming Number List' link to reach the Number List page.

  Refer the topic "Number Lists" to know more, and for configuration instructions.

- **Outgoing Number List:** Program the number strings that are to be called back in this List.
  For each number string you programmed in the 'Incoming Number List', you must program in the corresponding index in the Outgoing Number List a number to which the call back is to be made. For example, for the number string programmed at Index 1 in the Incoming Number List, a corresponding number string at the same Index, Index 1, should be programmed in the 'Outgoing Number List'.

  By default, Number List 16 is assigned to Outgoing Number List.The same Number List 16 is also assigned to all BRI Ports as well as all other Trunk port types.

  You may program the default number list, or a different number list and assign it to this BRI port.

  You may program the Outgoing Number List either from the 'Number List' page or by clicking the 'Outgoing Number List' link to reach the Number List page.

  Refer the topic "Number Lists" to know more, and for configuration instructions.

- **Call Back from:** This parameter determines the trunk port to be used to make the call back. The call back can be made using the same port or an "OG Trunk Bundle Group".

  Select 'Same port' if you want the call back to be made using the same port on which the missed call is received. If you select OGTBG, the call back will be made using the OGTBG, which you have defined.

  By default, Same port is selected.

- **OGTB Group:** If you selected OGTBG for making the call back in the previous parameter, you must define the OGTBG that must be used in this parameter.
  By default, OGTBG 01 is assigned.

  If you want the system to select the lowest cost trunk for making the call back, enable Least Cost Routing on the OGTBG that you define here for Call Back.

- Configure **Pause Timer** for the BRI Port. Range of Pause Timer is from 1 to 9 seconds. By default, it is set to 3 seconds for all BRI Ports.

  This Timer is required to insert delay between the digits while dialing out DTMF digits on the BRI port. One of the applications for using this parameter is Multi-stage dialing. Refer chapter "Multi-Stage Dialing".

  For example, if PPP2 is to be outdialed and Pause timer is programmed as 3 seconds, the SARVAM UCS will out dial the digit 2 after 9 seconds i.e delay of individual P i.e 3+3+3 =9.

- Configure **DTMF On Time** for the BRI Port. Range of DTMF On Time is from 051 to 255 ms. By default, it is set to 102 ms for all BRI Ports.

  The DTMF On Time is the time for which the DTMF digit which is to be outdialed by the SARVAM UCS remain On. One of the applications for using this parameter is Multi-stage dialing. Refer chapter "Multi-Stage Dialing".

- Configure **DTMF Inter Digit Pause Timer** for the BRI Port. Range of Inter Digit Pause Timer is from 051 to 255 ms. By default, it is set to 102 ms for all BRI Ports.

  Inter Digit Pause Timer is the time for which the system will wait while receiving the dialing digits to consider it as end-of-dialing.

- Select **Category (Logical Partition)** for the BRI Port. You may select from the following options:
  - 1
  - 2
  - 3

  By default, 1 is selected for all BRI Ports.

- If SARVAM UCS is to be used as a Gateway, enable **Gateway Application-Answer Signaling** on the BRI Port and configure **DTMF String**. By default, Gateway Application-Answer Signaling is disabled and CCC is configured as DTMF String.

  For more details, see "Gateway Application-Answer Signaling".

- Enter appropriate **Debug Code (Level 1 to 4)**, to obtain debug information of various parts of BRI Card on the COM Port. By default, debug is off for all BRI ports for all levels.

Code is the value for the specified level to turn ON the debug for the parameters. Code range is from 000 to 255. Code value '000' for each level will turn off that level's debug.

Level and Code for BRI Port are as specified below:

Level 1

Unused	Flow	NLS	LAP	SVC Primitives	Variables	State	Primitives

Code	Meaning
001	Primitives
002	State
004	Variables
008	SVC Primitives
016	LAP
032	NLS
064	Flow Debug
000	Debug Off

Level 2

Unused	Unused	Unused	DTMF Digit	Unused	Layer 4	Unused	Unused

Code	Meaning
004	Layer 4
016	DTMF Digit
000	Debug Off

Level 3

Unused	Unused	Unused	HDLC D-Channel	Unused	Unused	Unused	Unused

Code	Meaning
016	HDLC D-Channel
000	Debug Off

Level 4

Unused	Unused	Unused	Unused	Unused	Unused	OS Task	NI

Code	Meaning
001	Network Interface (NI)

Code	Meaning
002	OS Task
000	Debug Off

Configuring BRI Parameters using Commands

The commands explained below should be referred as:
To program a single port: ***XXXX-1***
To program a range of ports: ***XXXX-2***
To program all the ports: ***XXXX-****

Hardware Slot-Port

Use following command to assign hardware ID to a BRI software port:

1106-BRI-Slot-Port offset
Where,
BRI is from 01 to 32.
Slot is the number of the universal slot, where the BRI Card is installed, from 01 to 16.
Port is the number of the BRI port on the card, from 00 to 99.

Use following command to de-assign the hardware slot and the hardware port assigned to the BRI software port.
1106-BRI-00-00

Name

Use following command to assign a name to the BRI port:
5405-1-BRI-Name
5405-2-BRI-BRI-Name
5405-*-Name
Where,
BRI is from 01 to 32.
Name is an alpha-numeric string of 12 characters. Terminate the command with #*.
By default, Name field is Blank.

Enable BRI Port

This parameter is used to enable/disable the port. When the Port configured in TE mode is disabled, it will not be allotted to the user on grabbing the port. Instead the user will get error tone. Also no IC call will be allowed.

Likewise, when the port is configured in NT mode is disabled, the IC call will not be allowed to land on this port. The port will be treated as absent and accordingly other activities will be performed like routing the call to other stations in the group, etc.

Use following command to enable/disable BRI port:
6201-1-BRI-Flag
6201-2-BRI-BRI-Flag
6201-*-Flag
Where,
BRI is from 01 to 32.
Flag is 0 for disable, 1 for enable.
By default, the BRI Port is enabled.

Orientation Type

Use following command to program Orientation Type for the BRI port:
6204-1-BRI-Orientation Type
6204-2-BRI-BRI-Orientation Type
6204-*-Orientation Type
Where,

Orientation Type	Meaning
1	Terminal
2	Network
3	Tie Line

By default Type = 1.

When Orientation=Terminal, the port will be regarded as trunk. All the trunk related parameters will be applied.

When Orientation=Network, the port will be regarded as station. All the station related parameters will be applied.

When Orientation = Tie-line, the port will be regarded as station for all incoming calls on it and as a trunk for all outgoing calls made through it.

Power Feed

Use following command to Feed Power to the BRI port:
6227-1- BRI Port- Feed Power to Port
6227-2- BRI Port-BRI Port- Feed Power to Port
6227-*- Feed Power to Port
Where,
BRI Port is from 01 to 32
Feed Power to port is 0 for disable, 1 for enable.
Default it is disabled.

BRI NT Interface Type

This parameter is relevant only when BRI port is used as Network Interface, i.e BRI NT mode.

Use the following command to select the Interface type as Point-to-Point or Point-to-Multipoint:
6226-1-BRI - Interface Type
6226-2-BRI-BRI-Interface Type
6226-*-Interface Type
Where,
Interface Type is
1 for Point-to-Point
2 for Point-to-Multipoint
Default =Point to Point

When ISDN Phones are to be connected with BRI NT of the SARVAM UCS, this parameter should be set as 'Point to Multipoint'.

When an ISDN System is to be interfaced with BRI-NT of the SARVAM UCS, this parameter should be set as 'Point to Point'.

BRI ISDN Switch Variant

Use following command to program the ISDN BRI Switch Variant of the BRI port:
6203-1-BRI-ISDN BRI Switch Variant
6203-2-BRI-BRI-ISDN BRI Switch Variant
6203-*-ISDN BRI Switch Variant
Where,
BRI is from 01 to 32.

ISDN BRI Variant	Meaning
01	ATT_4ESS
02	ATT_5ESS
03	AUSTRALIA
04	AUTO CONFIG
05	DMS_100
06	ETSI_NET3
07	NTT_INS64
08	SWV_HONG_KONG
09	US_NI1
10	US_NI2
11	VN_X

By default, ISDN BRI Switch Variant is 06.

OG Reference ID-WH

Use following command to program an OG Reference ID to a BRI port:
6231-1-BRI-OG Reference ID
6231-2-BRI-BRI-OG Reference ID
6231-*-OG Reference ID
Where,
BRI is from 01 to 32.
OG Reference ID is from 00 or 01 to 99.
By default, OG Reference ID for BRI is 00.

This command is significant only if the BRI port is configured in TE mode.

OG Reference ID-BH

Use following command to program an OG Reference ID to a BRI port:
6241-1-BRI-OG Reference ID
6241-2-BRI-BRI-OG Reference ID
6241-*-OG Reference ID
Where,
BRI is from 01 to 32.
OG Reference ID is from 00 or 01 to 99.

By default, OG Reference ID for BRI is 00.

OG Reference ID-NH

Use following command to program an OG Reference ID to a BRI port:
6242-1-BRI-OG Reference ID
6242-2-BRI-BRI-OG Reference ID
6242-*-OG Reference ID
Where,
BRI is from 01 to 32.
OG Reference ID is from 00 or 01 to 99.
By default, OG Reference ID for BRI is 00.

This command is significant only if the BRI port is configured in TE mode.

IC Reference ID-WH

Use following command to program an IC Reference ID-WH on the BRI port:
6232-1-BRI-IC Reference ID
6232-2-BRI-BRI-IC Reference ID
6232-*-IC Reference ID
Where,
BRI is from 01 to 32.
IC Reference ID is from 01 to 99.
By default, IC Reference ID-WH for BRI is 00.

This command is significant only if the BRI port is configured in TE mode.

IC Reference ID-BH

Use following command to program an IC Reference ID-BH on the BRI port:
6233-1-BRI-IC Reference ID
6233-2-BRI-BRI-IC Reference ID
6233-*-IC Reference ID
Where,
BRI is from 01 to 32.
IC Reference ID is from 00 or 01 to 99.
By default, IC Reference ID-BH for BRI is 00.
This command is significant only if the BRI port is configured in TE mode.

IC Reference ID-NH

Use following command to program an IC Reference ID-NH on the BRI port:
6234-1-BRI-IC Reference ID
6234-2-BRI-BRI-IC Reference ID
6234-*-IC Reference ID
Where,
BRI is from 01 to 32.
IC Reference ID is from 00 or 01 to 99.
By default, IC Reference ID-NH for BRI is 00.

This command is significant only if the BRI port is configured in TE mode.

Trunk Feature Template

Use the following command to assign a Trunk feature Template to the BRI Port:

5804-1-BRI- Trunk Feature Template Number
5804-2-BRI-BRI- Trunk Feature Template Number.
5804-*- Trunk Feature Template Number.
Where,
BRI is from 01 to 32.
Template Number is the number of the customized Trunk Feature Template, from 01 to 50. Default: Template 01.

Cost Factor

Use following command to assign a Cost Factor to the BRI port:
6202-1-BRI-SP
6202-2-BRI-BRI-SP
6202-*-SP
Where,
BRI is from 01 to 32.
Cost Factory is from 01 to 99.
By default, all the BRI ports are assigned Cost Factor = 01.

This parameter is insignificant when BRI port is configured in NT mode.

Station Basic Feature Template

Use the following command to assign a Station Basic Feature Template to a BRI port:
5509-1-BRI-Template Number
5509-2-BRI-BRI-Template Number.
5509-*-Template Number
Where,
BRI is from 01 to 32.
Template is the number of the Station Basic Feature Template, from 01 to 50.
Default: Template 01 is assigned to all BRI ports.

Station Advanced Feature Template

Use the following command to assign a Station Advanced Feature Template to a BRI port:
5609-1-BRI-Template Number
5609-2-BRI-BRI-Template Number
5609-*-Template Number

Where,
BRI is from 01 to 32.
Template is the number of the Station Advanced Feature Template, from 01 to 50.
Default: Template 01 is assigned to all BRI ports.

Layer 1 Mode

- This parameter is applicable for BRI port, when orientation type is 'Terminal'.
- The Public ISDN provides different types of lines in different countries as mentioned below:
  - On Demand
  - Always ON
- In 'On Demand' type of line, layer 1 (physical layer) remains 'down' when the line is idle.
- When the network places incoming call, it activates the layer 1 and places the call.

- When the terminal makes out going call, the layer 1 gets activated automatically.
- In 'Always ON' type of line, layer 1 is always 'up', in normal condition.
- In this type of line, 'layer 1 down' indicates fault condition and calls get failed.

Use following command to program 'Layer 1 Mode' for BRI port:
6225-1-BRI-Layer 1 Mode
Where,
BRI is from 01 to 32.

Layer 1 Mode	Meaning
1	Always ON
2	On Demand

Select the 'Layer 1 Mode' parameter, depending upon the type of line, terminated on BRI port of SARVAM UCS.

By default, Layer 1 Mode is 'Always ON'.

TEI Negotiation

SE can select the Automatic or Fixed TEI negotiation for each BRI port as required. If Fixed TEI negotiation is selected, the value of fixed TEI negotiation is required to be programmed.
Use following command to select TEI Negotiation on a BRI port:
6238-1-BRI-TEI Negotiation Mode
6238-2-BRI-BRI-TEI Negotiation Mode
6238-*-TEI Negotiation Mode
Where,
BRI is from 00 to 32.

TEI Negotiation mode	Meaning
0	Automatic (Non-fixed)
1	Fixed

By default, TEI Negotiation Mode is Automatic.

TEI Value (for Fixed Mode)

Use following command to program TEI Negotiation value when programmed as Fixed:
6239-1-BRI-TEI Value
6239-2-BRI -BRI-TEI Value
6239-*-TEI Value
Where,
BRI is from 00 to 32.
TEI Value is from 00 to 63
By default, TEI Value is 00.

The System Engineer should take care of the following points when using the above commands for TEI negotiation:

- When you change the TEI mode on any port, the BRI Card will get reset.

- If you have selected 'Fixed' mode, program the value as per the value of port connected at remote end as explained below:

    - TEI Value programmed in the BRI (NT) should match with the TEI value programmed in the Terminal equipment connected with it.

    - TEI value of BRI (TE) port of SARVAM UCS should match with the TEI value expected by the NT equipment at other end.

Priority

This command is applicable when BRI port is configured to act as a station.
Use following command to assign priority to BRI:
3916-1-BRI-Priority
3916-2-BRI-BRI-Priority
3916-*-Priority
Where,
BRI is from 01 to 32
Priority is from 1 to 9
By default, Priority for BRI is 5-Normal.

Return Call to Original Caller (RCOC)

Use the following command to enable RCOC on BRI trunk:
6220-1-BRI-Code
6220-2-BRI-BRI-Code
6220-*-Code
Where,
BRI is from 01 to 32.
Code is
0 for Disable
1 for Enable
Default: Disabled

Overlap Receiving Timer

Use following command to set overlap receiving timer for BRI:
6208-1-BRI-Timer
6208-2-BRI-BRI-Timer
6208-*-Timer
Where,
BRI is from 01 to 32
Timer is from 00 to 99 seconds.
Default: 15 seconds.

This timer is relevant while receiving the called party number information in overlap receiving mode. It is not relevant for overlap sending mode.

Idle Code

Use following command to program the Idle Code for the BRI port:
6207-1-BRI-Idle Code
6207-2-BRI-BRI-Idle Code
6207-*-Idle Code
Where,
BRI is from 01 to 32.

Idle Code is from 000 to 255.

The binary equivalent of the programmed value (000 to 255) is sent on the channel to signify that the channel is idle. (or Unused) This setting depends on the network. Most commonly applicable values are 7F and FF (Binary equivalent is 0111 1111 and 1111 1111, decimal equivalent is 127 and 255).

Default: Idle Code for the BRI port is 127 (7F).

Channel Reserved for Data Communication

Use the following command to program number of channels reserved for data transmission:
6235-1-BRI-Channel Count
6235-2-BRI-BRI-Channel Count
6235-*-Channel Count
Where,
BRI is from 01 to 32.
Channel Count is from 00 to 02.
By default, Number of Channels is 02.

Channels Reserved for OG Calls

Use following command to reserve the number of channels for OG calls on a BRI port.
6236-1-BRI-Channel Count
6236-2-BRI-BRI-Channel Count
6236-*-Channel Count
Where,
BRI is from 01 to 32.
Channel Count is from 0 to 2.
By default, Channel is reserved for OG Calls for BRI is 2. Both the channels are used for receiving OG calls.

Channels reserved for IC Calls

Use following command to reserve the number of channels for IC calls on a BRI port.
6237-1-BRI-Channel Count
6237-2-BRI-BRI-Channel Count
6237-*-Channel Count
Where,
BRI is from 00 to 32.
Channel Count is from 0 to 2.
By default, Channel is reserved for IC Calls for BRI is 2. Both the channels are used for receiving IC calls.

Caller - Type of Number (TON)

Use following command to program a Caller TON for the BRI port:
6221-1-BRI-Caller TON
6221-2-BRI-BRI-Caller TON
6221-*-Caller TON
Where,
BRI is from 01 to 32.

Caller TON	Meaning
1	Unknown
2	International Number

Caller TON	Meaning
3	National Number
4	Network Specific Number
5	Subscriber Number
6	Abbreviated Number
7	Reserved Number

By default, Caller TON for BRI is Unknown.

Caller Numbering Plan Identification (NPI)

Use following command to program a Caller NPI for the BRI port:
6222-1-BRI-Caller NPI
6222-2-BRI-BRI-Caller NPI
6222-*-Caller NPI
Where,
BRI is from 01 to 32.

Caller NPI	Meaning
1	Unknown
2	ISDN Numbering
3	Date Numbering
4	Telex Numbering
5	National Numbering
6	Private
7	Reserved

By default, Caller NPI for BRI is ISDN Numbering.

Called - Type of Number (TON)

Use following command to program a Called Party destination TON for the BRI port:
6223-1-BRI-Called Party TON
6223-2-BRI-BRI-Called Party TON
6223-*-Called Party TON
Where,
BRI is from 01 to 32.

Called Party TON	Meaning
1	Unknown
2	International Number
3	National Number
4	Network Specific Number
5	Subscriber Number
6	Abbreviated Number

Called Party TON	Meaning
7	Reserved Number

By default, Called Party TON for BRI is Unknown.

Called Numbering Plan Identification (NPI)

Use following command to program a Called Party NPI for the BRI port:
6224-1-BRI-Called Party NPI
6224-2-BRI-BRI-Called Party NPI
6224-*-Called Party NPI
Where,
BRI is from 01 to 32.

Called Party NPI	Meaning
1	Unknown
2	ISDN Numbering
3	Date Numbering
4	Telex Numbering
5	National Numbering
6	Private
7	Reserved

By default, Called Party NPI for BRI is ISDN Numbering.

Call Budget

Refer the topic "Call Budget on Trunk" for command strings.

Call Back

Use the following commands to program Call Back on BRI Trunk ports. To know more about this feature, refer the topic Call Back on Trunk Ports.

To enable/disable Call Back on BRI port:
6242-1-BRI- Call Back Flag
6242-2-BRI-BRI-Call Back Flag
6242- * - Call Back Flag
Where,
BRI is from 01 to 32

Call Back Flag	Meaning
0	Disable
1	Enable

By default, Call Back flag is disabled.

To program Call Back Timer for BRI port:

6243-1-BRI-Call Back Timer
6243-2-BRI-BRI-Call Back Timer
6243-*-Call Back Timer
Where,
BRI is from 01 to 32
Pause Timer Range is from 01 to 99 Sec.
By default, Pause Timer is 10 Seconds

To program Call Back Mode on BRI port:
6244-1-BRI-Call Back Mode
6244-2-BRI-BRI-Call Back Mode
6244-*-Call Back Mode
Where,
BRI is from 01 to 32
Call Back Mode is 1 to 5

Call Back Mode	Meaning
1	Built-in Auto Attendant
2	PIN Auth. - Multiple Calls
3	CLI Auth. - Multiple Calls
4	CLI Auth. - Single Call - Ans. Sig.
5	Operator

By default, Call Back mode is Operator.
To program Call Back On method for BRI port:
6245-1-BRI-Call Back on selection
6245-2-BRI-BRI-Call Back on selection
6245-*-Call Back on selection
Where,
BRI is from 01 to 32
Call back on selection is

Call Back on	Meaning
1	CLI Number
2	Alternate Number

By default, Call Back on selection is CLI Number

To assign Call Back - Incoming Number List to a BRI port:
6246-1-BRI-Incoming Number List
6246-2-BRI-BRI-Incoming Number List
6246-*-Incoming Number List
Where,
BRI is from 01 to 32
Incoming Number List is from 01 - 16.
By default, Incoming Number List is 15.

To assign a Call Back - Outgoing Number List to a BRI port:
6247-1-BRI-Outgoing Number List

6247-2-BRI-BRI-Outgoing Number List
6247-*-Outgoing Number List
Where,
BRI is from 01 to 32
Outgoing Number List is from 01 - 16.
By default, Outgoing Number List is 16.

To define Call Back From for a BRI port:
6248-1-BRI-Call Back From
6248-2-BRI-BRI-Call Back From
6248-*-Call Back From
Where,
BRI is from 01 to 32
Call Back From is
1 for Same Port
2 for OGTB Group
By default, Same Port is selected as Call Back From.

To assign a Call Back - OGTB Group for a BRI port:
6249-1-BRI-OGTB Group
6249-2-BRI-BRIOGTB Group
6249-*-OGTB Group
Where,
BRI is from 01 to 32
OGTB Group is from 01 to 32
By default, OGTBG is 01.

Pause Timer

Use following command to program Pause Timer:
6209-1-BRI- Pause Timer
6209-2-BRI-BRI-Pause Timer
6209- * - Pause Timer
Where,
BRI is from 01 to 32
Pause Timer is from 1 to 9 seconds
By default, Pause Timer is 3 seconds.

This Timer is used to provide delay for the number dialing by the BRI port. Pause timer will be applicable when any 'P' digit is configured in the DTMF number string which is to be outdialed as DTMF digits on BRI port. One of the applications for using this parameter is Multi-stage dialing. Refer chapter “Multi-Stage Dialing”.

For example, if PPP2 is to be outdialed and Pause timer is programmed as 3 seconds, the SARVAM UCS will out dial the digit 2 after 9 seconds i.e delay of individual P i.e 3+3+3 =9.

DTMF ON Time (msec)

Use following command to program DTMF ON Time:
6210-1-BRI-DTMF ON Time
6210-2-BRI-BRI-DTMF ON Time
6210-*-DTMF ON Time
Where,
BRI is from 01 to 32
DTMF ON Time is from 051 to 255 msec.
By default, DTMF ON Time is 102 msec.

This parameter decides for how much time the DTMF digit will be ON, while out dialed by the SARVAM UCS. One of the applications for using this parameter is Multi-stage dialing. Refer chapter "Multi-Stage Dialing".

DTMF Inter digit Pause Timer

Use following command to program DTMF Inter digit Pause Timer:
6211-1-BRI- DTMF Inter digit Pause Time
6211-2-BRI-BRI- DTMF Inter digit Pause Time
6211-*-DTMF Inter digit Pause Time
Where,
BRI is from 01 to 32
DTMF Inter digit Pause Time is from 051 to 255 msec.
By default, DTMF Inter digit Pause Time is 102 msec.

This parameter decides how much time the pause (gap) should be present between two digits while dialed by the SARVAM UCS. One of the applications for using this parameter is Multi-stage dialing. Refer chapter "Multi-Stage Dialing".

Category (Logical Partition)

Use the following command to program 'Category (Logical Partition)' for BRI port:

6213-1-BRI- Category
6213-2-BRI-BRI- Category
6213-*-Category
Where,
BRI is from 01 to 32
Category is from 1 to 3.
1 for CO
2 for Leased
3 for Private.
By default, Category is CO.

Gateway Application-Answer Signaling

This command is used to enable Gateway Application-Answer Signaling on BRI trunk. Refer chapter "Gateway Application-Answer Signaling" for more details.

Use following command to set flag for Gateway Application-Answer Signaling on BRI trunk:
6206-1-BRI-Gateway Application-Answer Signaling Flag
6206-2-BRI-BRI-Gateway Application-Answer Signaling Flag
6206-*-Gateway Application-Answer Signaling Flag
Where,
BRI is from 01 to 32

Flag	Meaning
0	Disable
1	Enable

By default, Gateway Application-Answer Signaling Flag is 'disable'.

Use following command to program DTMF digits string to be dialed as Gateway Application-Answer Signaling:
6212-1-BRI-Gateway Application-Answer Signaling DTMF String

6212-2-BRI-BRI-Gateway Application-Answer Signaling DTMF String
6212-*-Gateway Application-Answer Signaling DTMF String
Where,
BRI is from 01 to 32.
DTMF Digits allowed for DTMF string are from (0 - 9), *, #, A, B, C, D.
Maximum 4 DTMF digits can be programmed. If you need less than 4 digits for DTMF string, terminate the command using #*.

To program #, *, A, B, C, D use following codes:

Digit	Code for programming through command
A	#4
B	#5
C	#6
D	#7
*	**
#	##

By default, Gateway Application-Answer Signaling DTMF String is 'CCC'.

Debug Code

Debug information of various parts of the card can be obtained on the COM port of the card.
Use following command to get appropriate debug information for the BRI port:
6291-1-BRI-Level-Code
6291-2-BRI-BRI-Level-Code
6291-*-Level-Code
Where,
BRI is from 01 to 32.
Level is from 1 to 4 (As shown below).
Code is the value for the specified level to turn ON the debug for the parameters Code range = 000 to 255. Code value '000' for each level will turn off that level's debug.

Level and Code for BRI port are as specified below:

Level 1

Unused	Flow	NLS	LAP	SVC Primitives	Variables	State	Primitives

Code	Meaning
001	Primitives
002	State
004	Variables
008	SVC Primitives
016	LAP
032	NLS
064	Flow Debug

Code	Meaning
000	Debug Off

Level 2

Unused	Unused	Unused	DTMF Digit	Unused	Layer 4	Unused	Unused

Code	Meaning
004	Layer 4
016	DTMF Digit
000	Debug Off

Level 3

Unused	Unused	Unused	HDLC D-Channel	Unused	Unused	Unused	Unused

Code	Meaning
016	HDLC D-Channel
000	Debug Off

Level 4

Unused	Unused	Unused	Unused	Unused	Unused	OS Task	NI

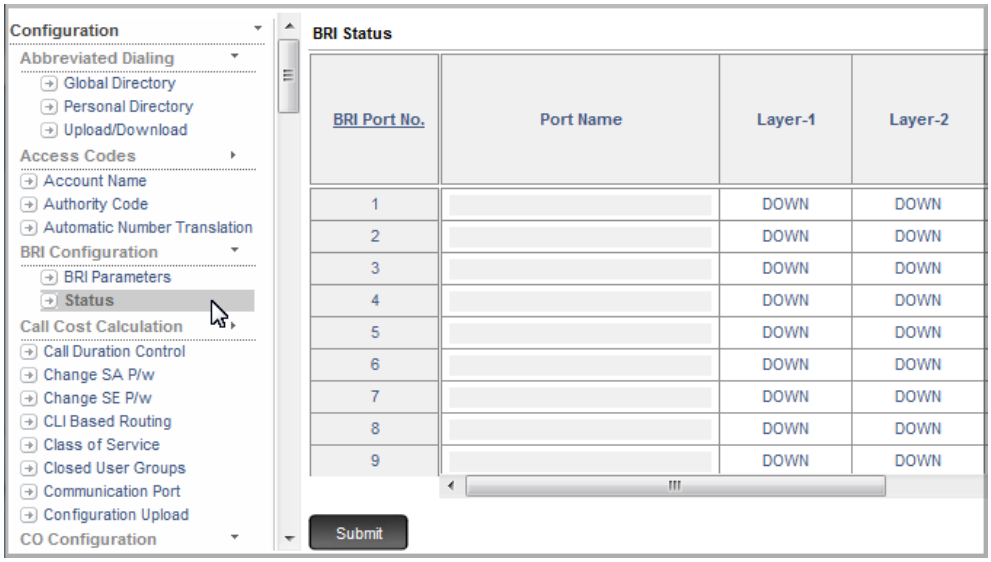
Code	Meaning
001	Network Interface (NI)
002	OS Task
000	Debug Off

By Default Debug = off for all BRI ports for all levels.

Viewing BRI Port Status

You can view the status of BRI ports on Jeeves only. To do this,

- Under **BRI Configuration**, click **Status**.



- For each BRI Port, the following settings will be displayed:
  - BRI Port Number
  - Port Name
  - Layer 1
  - Layer 2
  - Call Budget Type
  - Call Budget Reset Mode
  - Call Budget Reset Scheduled (Date)
  - Allotted Amount/Minutes/Calls
  - Consumed Amount/Minutes/Calls
  - Reset Consumed (this is not a status indicator. It is for resetting the Consumed Call Budget manually)

If you have assigned a DSS Key to the BRI Port/channel and when Layer 1 of BRI line is down or not connected, DSS key LED will glow 1 sec On and 3 sec Off in violet color to indicate the faulty condition.

*You can also view the BRI Port Status from the **Status** link. To view, click the BRI link under Status.*

Configuring Mobile Trunks

The system supports a maximum of 40 Mobile ports. Before you begin configuring the Mobile ports, ensure that the Mobile Card has been installed correctly. The system supports 2G, 3G as well as 4G LTE network thereby providing higher bandwidth, greater connection speed and better underlying technology for VoIP calls. Access to these networks is made possible by mounting any of these —2G, 3G or 4G— GSM modules on the Mobile card.

Configuration of Mobile Trunks involves:

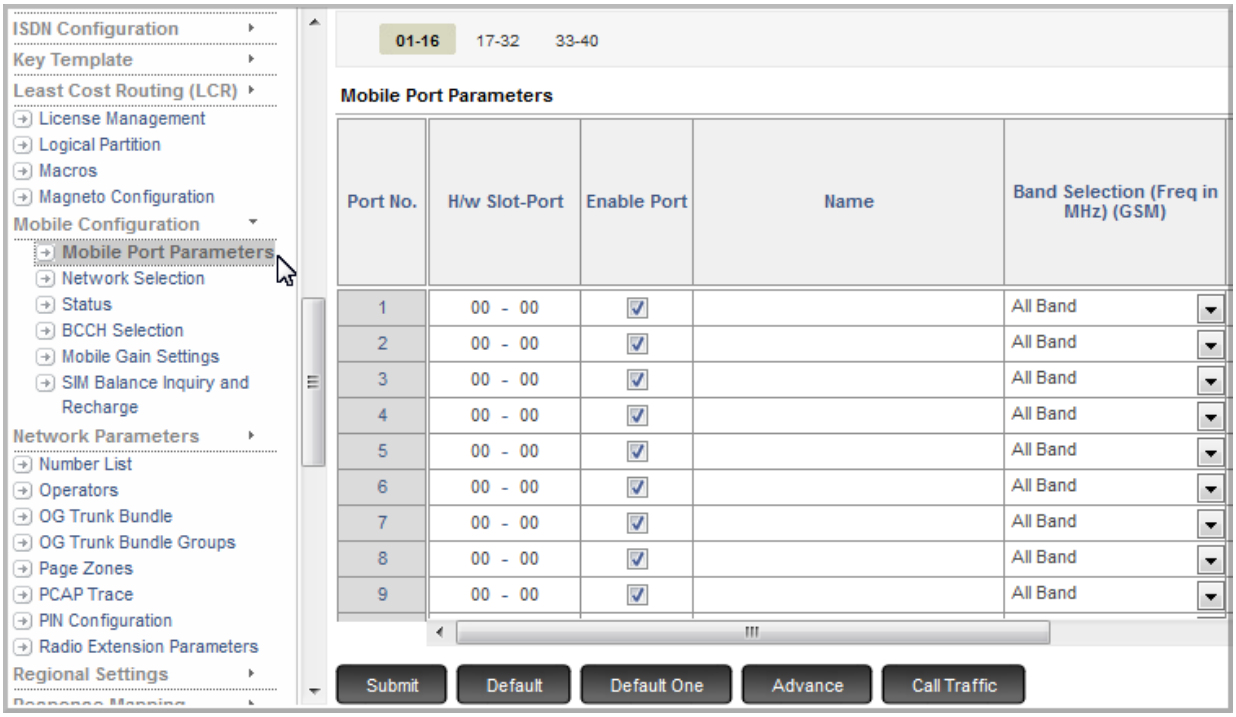
1. Customizing the Mobile Port Parameters
2. Network Selection
3. Configuring VoLTE parameters. For details refer to “VoLTE Configuration”
4. Uploading MBN files, For details refer to “MBN File Upload”
5. Viewing Mobile Port Status

You can customize mobile port parameters and select the network using Jeeves and a Telephone. However, mobile port status can be viewed using Jeeves only.

Mobile Port Parameters

Configuring Mobile Port Parameters using Jeeves

- Log into Jeeves as System Engineer.
- Under **Configuration**, click **Mobile Configuration**.
- Click **Mobile Port Parameters** to open the page.



Configure the following port parameters:

- **Hardware Slot and Port:** 'Slot' is the number of the Universal Slot in which the Mobile Card has been inserted. 'Port' is the number of the Mobile port in which you have installed the SIM card.

  The number of the Mobile Port depends on the configuration of the Mobile Card. For instance, if you have installed GSM8 Card, the number of the ports will be from 1 to 8.

  By default the SARVAM UCS can detect and assign the hardware slot and port numbers automatically to the mobile (software) ports. However, if required, you may change the Hardware Slot and Port assigned to the mobile software port. In this case, enter the desired Hardware Slot and Port number in this field.

  If you want to de-assign the Hardware Slot and Port, enter '00' in both fields.

- **Enable Port:** This flag is for enabling or disabling a Mobile Trunk port. When a Mobile Trunk port is disabled, neither incoming nor outgoing calls can be made from that port.

  By default, the port is enabled. You may disable ports that are not functioning by clicking the check box.

- **Name:** You may assign a 'Name' to each Mobile Trunk to facilitate identification. Whenever there is an incoming call without CLI on this port, the Name you have programmed will be displayed on the landing extension.

  The Name of the port may be the name of the Service Provider of the SIM card you have installed on this port (recommended) or the phone number assigned to the SIM card on this port.

  The Name may comprise a maximum of 18 characters.

- **Band Selection[194] (Freq in MHz) (GSM):** The Frequency Band supported by the mobile networks varies from country to country. SARVAM UCS's mobile card supports frequency bands of most countries. Select the Frequency Band used by your GSM/ UMTS Provider for the mobile port.

  By default, Mobile Frequency Band 'All Bands' is selected.

This is applicable only for GSM Mode.

When you change the Frequency Band, the change will be effected after the next system restart or the next Mobile Port restart.

- **Preferred Network Mode[195]:** Select the Preferred Network Mode for the Mobile Port — Any (GSM/ UMTS/LTE), GSM only, UMTS only or LTE only. By default, Any (GSM/UMTS/LTE) is selected.

If you select the Preferred Network Mode as LTE only, calling functionality will be possible only if the Service provider supports calling through VoLTE.

194. *Not applicable if CDMA Mobile Card is installed in your system.*
195. *Not applicable if CDMA Mobile Card is installed in your system.*

- **SIM PIN**[196]**:** If you have enabled PIN protection and changed the SIM PIN of the card to the default value '1234' using a mobile handset[197], you can assign a new PIN to the SIM card from the SARVAM UCS.

  Make sure that the PIN stored on the SIM card and that of the system are the same.

- *You must click 'Submit' after you enter the new PIN. Wait for 5 seconds, and then refresh this page to view the new SIM PIN.*

- *If you have enabled PIN protection, and the SIM PIN on the Card and the SIM PIN programmed in the SARVAM UCS are not the same, the SIM card may get blocked and would require the Personal Unblocking Number (PUK) from the Service Provider to reactivate it again.*

The SIM PIN will not be set to default value, when you set the system to default.

- **Incoming Calls:** Select the Incoming Call Mode on the Mobile Port from the options:

  - '**Accept':** incoming calls will be allowed and incoming call logic is applied.

  - **'Ignore':** incoming calls will not be processed further and call logic will not be applied.

  - **'Reject':** incoming calls will be rejected immediately and mobile port will be freed (released).

  By default, incoming calls are accepted. This feature is particularly useful in outbound call centers for blocking incoming calls on a SIM number (mobile port) of the SARVAM UCS.

- **Trunk Feature Template:** A Trunk Feature Template is a set of features like Time Table, Operator, Auto Attendant, DISA, Trunk Auto Answer, Trunk Landing Group, SMDR Storage, etc., that defines the behavior of a Trunk. Apply a Trunk Feature Template to the Mobile Trunk. By default, Trunk Feature Template 01 is applied on all Mobile Trunks as well as all other trunk types. Refer the topic "Trunk Feature Template" to know more.

  Click the 'Trunk Feature Template' link to open the page. Check if the default Template 01 fulfills your requirement for the mobile port.

  If the default Template 01 does not fulfill your requirement, prepare another Trunk Feature Template[198], and enter the newly prepared Template number for the Mobile port.

- **CLIR**[199]**:** Enable this flag if you want to activate CLIR for all outgoing calls made through the Mobile Port.

  When CLIR is enabled, the called party will not be able to see the subscriber number of the Mobile Port.

  By default CLIR is disabled for all Mobile ports.

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196. Not applicable if CDMA Mobile Card is installed in your system.

*197. Refer the topic **Installing the Mobile Card** for instructions. If you have not enabled PIN protection before installing the GSM Card, you will not be able to change the SIM PIN.*

198. The default template is applied on the ports of all trunk types supported by SARVAM UCS. Changes to the default template will be applied on all trunk types also. So, you are advised to prepare a new template and apply it to the desired trunk types.

199. Not applicable if CDMA Mobile Card is installed in your system.

This feature will work only if subscribed/supported by your mobile service provider.

- **Return Call to Original Caller (RCOC)**[200]**:** Enable this flag if you want to apply the RCOC feature.

  If this feature is enabled on the Mobile trunk port, the system routes calls returned by remote parties back to the extensions that originally made the call from this port (the original callers' extensions). To know more, refer the feature description for Return Call to Original Caller (RCOC).
- **Cost Factor:** This parameter is of relevance only if 'Least Cost Routing' feature is applied on the mobile port.

  Cost Factor is a number assigned to each trunk for identification. This number also serves as a preference number for the trunk. The Cost Factor can be from 1 to 99. Trunks having the same preference must be assigned the same Cost Factor. Different trunk types can also be assigned the same Cost Factor. These trunks are used for routing calls.

  Assign a Cost Factor to the Mobile Trunk port, for example, 03 and program Least Cost Routing Table accordingly.

  For example, if you want to route all outgoing calls starting with number '9' through the SIM installed in Mobile Port Number 01 only,

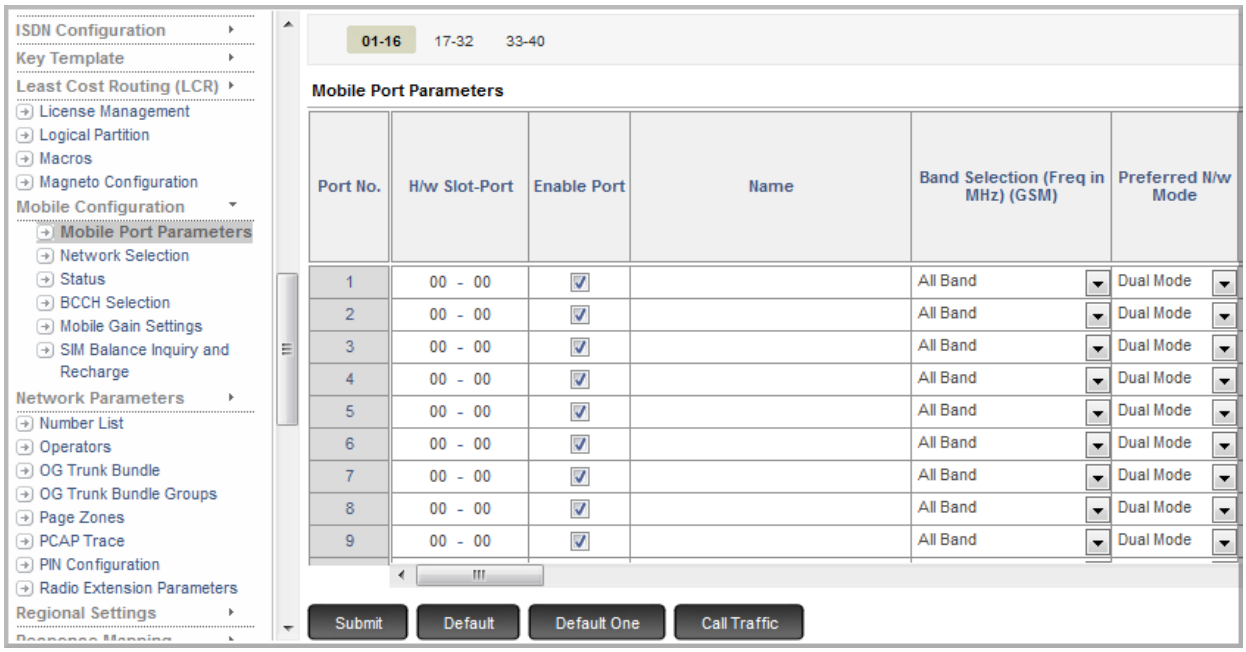
  - You must first assign a Cost Factor (01-99) to Mobile Port 01, for example, 03.
  - Click 'Least Cost Routing - Number Based' to open the page.
  - Enter '9' in the 'Number' column, Cost Factor '03' as Preference 1, 2, 3 and 4.
  - Click **Submit** at the bottom of the page to save your setting.

  All outgoing calls assigned Cost Factor trunk 03 will be made from Mobile Port 01.

200. Not applicable if CDMA Mobile Card is installed in your system.

Advanced Configuration

The above listed parameters fulfill the basic mobile trunk port configuration requirements of most users. However, it is anticipated that some users may need to configure other less commonly used features on the mobile ports, such as Call Budget, Call Back on Mobile Port, or they may want to use the Mobile port as a Gateway application.



For such users, you may click the 'Advance' button and program the following parameters:

- **N/w Registration Retry Count:** The mobile port is programmed to automatically locate and register with the Network that supports the SIM card installed on it. Also, at each power ON, the mobile port (SIM) will automatically register with the Network that supports the SIM on it.

  However, if the Mobile port fails to register, it will restart the process of network registration on the expiry of the Network Registration Retry Timer[201]. On the expiry of this timer, the system will retry registration for the programmed Count (number of times) and with each re-try attempt, the count will be decremented by one.

  By default the Retry Count is set to 2. If required you may change the Count to the desired value.

- **Mobile Gain Settings Template:** You can increase or decrease the level of Incoming Speech (Receive Gain) and Outgoing Speech (Transmit Gain) on the Mobile port by changing the Rx Gain and Tx Gain to the desired level. Different levels can be set for each port type in the Mobile Gain Settings Template. By default, Mobile Gain Template 1 is assigned. If you want to assign a different Template, you must customize the Mobile Gain Settings Template first and then assign the number of the Mobile Gain Settings Template here. To customize the Mobile Gain Settings, see “Gain Settings”.

If you change the Tx or Rx Gain during an active call, the change you made will not apply on the current call. It will be applied on the next call.

- **Call Back:** This parameter is related to the ‘Call Back on Trunk Port’ feature. If you want to enable the 'Call Back on Trunk Port' feature on this Mobile trunk, configure the following parameters:

201. *The Network Registration Retry Timer defines the time for which the Mobile port, which has failed to register with the network, should wait before attempting to re-register with the network. Network registration retry timer is 2 minutes and is non-programmable.*

- **Enable Call Back:** Enable this flag to activate the Call Back on Trunk Port feature. By default, this flag is disabled on all trunk port types. By default, the flag is disabled.

- **Call Back Timer:** This is the duration for which the system waits for the caller to disconnect before applying the Call Back. The range of this timer is from 01 to 99 seconds. By default, it is set to 10 seconds.

- **Call Back Mode:** Select from the following options how a 'Call Back' call answered by the remote party should be routed:
  - Built-In Auto Attendant
  - PIN Authentication - Multiple Calls
  - CLI Authentication - Multiple Calls
  - CLI Authentication - Single Call - Answer Signaling
  - Operator

  By default, Operator is selected as the Call Back Mode.

- **Call Back on:** This parameter allows you to select if the call back should be made to the same number that was received or to a different number. If you want the call back to be made to the same number select the 'CLI number". If you want the call back to be made to a different number, select 'Alternate Number'.

  By default, CLI number is selected for Call Back.

- **Incoming Number List:** Program the number strings that are eligible for Call Back in this List. By default, Number List 15 is assigned to Call Back Incoming Number List.
  Number List 15 is also assigned to all Mobile trunks as well as all other Trunk port types. If you want the same numbers strings to be programmed commonly for all Mobile trunks and Trunk Port types, retain this list.

  If you want a different set of number strings to be programmed for this Mobile Trunk, select a different Number List, and assign it to the Mobile trunk port.

  You may program the Incoming Number List either from the 'Number List' page or by clicking the 'Incoming Number List' link to reach the Number List page.

  Refer the topic "Number Lists" to know more, and for configuration instructions.

- **Outgoing Number List:** Program the number strings that are to be called back in this List.
  For each number string you programmed in the 'Incoming Number List', you must program in the corresponding index in the Outgoing Number List a number to which the call back is to be made. For example, for the number string programmed at Index 1 in the Incoming Number List, a corresponding number string at the same Index, Index 1, should be programmed in the 'Outgoing Number List'.

  By default, Number List 16 is assigned to Outgoing Number List.The same Number List 16 is also assigned to all Mobile trunks as well as all other Trunk port types.

  You may program the default number list, or a different number list and assign it to this Mobile Trunk port.

  You may program the Outgoing Number List either from the 'Number List' page or by clicking the 'Outgoing Number List' link to reach the Number List page.

Refer the topic “Number Lists” to know more, and for configuration instructions.

- **Call Back from:** This parameter determines the trunk port to be used to make the call back. The call back can be made using the same port or an “OG Trunk Bundle Group”.

  Select ‘Same port’ if you want the call back to be made using the same port on which the missed call is received. If you select OGTBG, the call back will be made using the OGTBG, which you have defined.

  By default, Same port is selected.

- **OGTB Group:** If you selected OGTBG for making the call back in the previous parameter, you must define the OGTBG that must be used in this parameter.

  By default, OGTBG 01 is assigned.

If you want the system to select the lowest cost trunk for making the call back, enable Least Cost Routing on the OGTBG that you define here for Call Back.

- **Call Budget:** By default, Call Budget is enabled on the trunk. If you wish to change the default configuration or disable it for this Mobile trunk port, configure the parameters as per your requirement:

  - **Type:** Select the type of Call Budget on Trunk—Amount, Minutes or Number of Calls—to be applied on this mobile trunk port. By default, Minutes is selected as the Call Budget type.To disable select Type as None.

  - **Amount:** If you selected 'Amount' as the Call Budget Type, enter the Budget Amount in this field. By default the Amount is set to 999999.

  - **Minutes:** If you selected 'Minutes' as the Call Budget Type, enter the number of Minutes in this field. By default the number of minutes is set as 000300.

  - **Calls:** If you selected ‘Calls’ as the Call Budget Type, enter the number of calls in this field. By default, the number of calls is set to 9999.

  - **Scheduled Reset:** Enable this flag if you want the Call Budget Amount/Minutes/Number of Calls to be on a particular date of every month.

  - **Scheduled (Date):** Enter the date of the month (Daily or 1-31) on which you want the Call Budget Amount/Minutes/Number of Calls to be every month. You may select ‘Daily’ if your plan suggests so.

The consumed Call Budget Amount/Minutes/Number of Calls can be reset from SE and SA Mode, referred to as Manual Reset. Refer the feature description “Call Budget on Trunk”.

- **Accept Anonymous Calls:** The check box is for accepting calls without CLI that land on the mobile port. By default the option is disabled, that is, calls without CLI will not be allowed on this Mobile port. Select the check box to enable.

- **Pause Timer:** This Timer is used for providing delay in number dialing from the Mobile port. The Pause Timer will be applicable when the digit 'P' is configured in the DTMF number string which is to be out dialed as DTMF digits on the Mobile port.

For example, if PPP2 is to be out dialed and Pause timer is programmed as 3 seconds, the SARVAM UCS will out dial the digit 2 after 9 seconds, that is, after a delay of individual P (3+3+3 =9). The range of this time is from 1 to 9. By default the Timer is set to 1 seconds.

This parameter is used for the "Multi-Stage Dialing" feature.

- **DTMF Outdial Option:** You can select whether to send the DTMF digits from the Mobile Ports **Inband** or through signaling, that is, **AT Command**. By default, DTMF Outdial Option is Inband.

  When you select DTMF Outdial using AT Command, the length of the DTMF digits will be determined by the DTMF ON Time you set.

*If CDMA Mobile Card is installed in your system, select **DTMF Outdial option** as AT Command for efficient functionality.*

- **DTMF ON Timer:** This parameter determines the time for which the DTMF digit will remain ON, while being out dialed by the system. This parameter finds its application in the feature "Multi-Stage Dialing"' and in *DTMF Outdialing using AT Command*. By default, DTMF ON Timer is 100 ms.

- **DTMF Detection Mode:** You can select whether to detect the DTMF digits using the GSM Modules or through DSP. Default: Using DSP.

  When you select DTMF Detection Using Module, the length of the DTMF digits will be determined by the DTMF Detection Timer you set.

*If CDMA Mobile Card is installed in your system, select **DTMF Detection Mode** as Using Module (GSM) for efficient functionality.*

- **DTMF Detection Duration:** This is the minimum ON to consider the DTMF digit as a valid digit. The valid range of this timer is 20 to 100ms. Default: 30ms.

*If CDMA Mobile Card is installed in your system, set the **DTMF Detection Duration** as 20ms.*

- **Min. Level (dB):** This parameter signifies the minimum level (dB) of the DTMF digit to be considered as valid. By default, Minimum level is set to -30dB.

  If the Min. Level set is very low, the DTMF digits might be detected in Voice and if it is very high, the DTMF digits may be lost.

  If received DTMF digit level is higher than or equal to the set value of Min. Level (dB), the system will accept the DTMF.

  If received DTMF digit level is lower than the set value of Min. Level (dB), the system will ignore the DTMF.

- **Category (Logical Partition):** This parameter assigns the Mobile Port to a trunk category for the purpose of Logical Partitioning. By default all Mobile Ports are assigned to Category 1. Do not change the default setting.

  If you want to change the call permission between the mobile port and other trunks, click the 'Category' link to open the Logical Partitioning page. You may program the call permission between Category 1 (assigned to Mobile Trunk Ports) and other Categories. Refer the feature description "Logical Partition" to know more.

- **Gateway Application-Answer Signaling:** This parameter is to be programmed if the Mobile Trunk Port is being used in a gateway application as a source port (from where calls originate). The calls originated on

the source port (mobile port) are routed using another Trunk port, the terminating port, which may be any trunk port, for example: T1E1. When call made from the terminating port gets matured, this is signaled to the source port in the form of DTMF digits.

- **Enable:** Enable this flag if you want the Mobile port to be used in a Gateway Application.
- **DTMF String (max. 4 digits):** Program the DTMF digits to be sent to signal call maturity to the source port.

- **SMS Parameters**[202]**:** If you want to use this Mobile Trunk Port for the SMS Server application to send/ receive SMS, configure the following parameters:

  - **SMS Center Number:** The system displays the default SMS Center Number of the network. If required, you can change the SMS Center Number. The SMS Center Number can be a maximum of 16 digits. Valid Range: 0 to 9 and +

  - **Send SMS:** Enable this check box if you want to use this Mobile Trunk Port to send SMS.

  - **Number of SMS to be sent (Total):** Enter the maximum number of SMS that can be sent using this Mobile Trunk Port. Default: 0999999999. Valid Range: 1 to 4294967296.

  - **Number of SMS to be sent (Daily):** Enter the maximum number of SMS that can be sent using this Mobile Trunk Port daily. Default: 20000. Valid Range: 1 to 20000.

  - **Receive SMS:** Enable this check box if you want to use this Mobile Trunk Port to receive SMS.

- **Debug:** Enable this flag by selecting the check box if you want to initiate debugging for the Mobile Port. By default, debugging is disabled.

- **Extended CLI:** Select this check box if the CLI of the current incoming call displays the last caller's CLI. When the flag is enabled, the current caller's CLI will be displayed. By default, it is disabled.

- **Allow Incoming CLI Modification:** Select the **Allow Incoming CLI Modification** check box if you want to apply Incoming CLI Modification on the Mobile trunk. By default, it is disabled.

  Incoming CLI Modification is useful in countries where the Calling Line Identification (CLI) received by the System extension users must be suitably modified before it can be used to dial out the number. To know more, see "Incoming CLI Modification".

If you receive CLI in dialable format, there is no need to use this feature. In such case, keep the flag disabled.

*For an incoming call on the Mobile trunk, the Incoming CLI Modification will be applied only when both — the **Allow Incoming CLI Modification** check box and the **Enable Incoming CLI Modification** check box in System Parameters — are enabled.*

- Repeat the same steps to configure other mobile ports.

- If you have completed configuration of all the above listed Mobile Port Parameters, click **Submit** at the bottom of the page to save your changes.

Configuring Mobile Port Parameters using a Telephone

202. Not applicable if CDMA Mobile Card is installed in your system.

- Enter SE mode from a DKP/SLT.

  To assign Hardware Slot and Port, dial:
  - ***1108-Mobile-Slot-Port Offset on the card***
    Where,
    Mobile Port Number is the number of the software port from 001 to 064
    Slot is the number of the universal slot from 01 to 16
    Port Offset on the card is 00, 01 to 99.

  To de-assign the Hardware Slot and Port, dial:
  - ***1108-Mobile-00-00***

  To enable/disable the Mobile Port, dial:
  - ***8000-1-Mobile-Flag*** to enable/disable a single mobile port.
  - ***8000-2-Mobile-Mobile-Flag*** to enable/disable a range of mobile ports.
  - ***8000-*-Flag*** to enable/disable all mobile port.
    Where,
    Mobile Port Number is the number of the software port from 001 to 064
    Flag is
    1 for Enable
    0 for Disable

  To assign a Name to the Mobile Port, dial:
  - ***5408-1-Mobile-Name-#**** to assign a name to a single mobile port.
  - ***5408-2-Mobile-Mobile-Name-#**** to assign the same name to a range of mobile ports.
  - ***5408-*-Name-#**** to assign the same name to all mobile ports.
    Where,
    Mobile Port Number is the number of the software port from 001 to 064
    Name is a string of a maximum of 18 alphanumeric characters. Terminate the command with #* if the number string is fewer than 18 characters.

  To clear the Name of the Mobile Port, dial:
  - ***5408-1-Mobile-#**** to clear the name of a single mobile port.
  - ***5408-2-Mobile-Mobile-#**** to clear the names of a range of mobile ports.
  - ***5408-*-#**** to clear the names of all mobile ports.

  To select the frequency Band[203] for the Mobile Port, dial:
  - ***8009-1-Mobile Port Number-Mobile Frequency Band Code*** to select the band for a single mobile port.
  - ***8009-2-Mobile Port Number -Mobile Port Number -Mobile Frequency Band Code*** to select the same band for a range of mobile ports.
  - ***8009-*-Mobile Frequency Band Code*** to select the same band for all mobile ports.
    Where,
    Mobile Port is the software port number of the mobile port, from 001 to 064.
    Mobile Frequency Band Code is
    1 for 900
    2 for 1800
    3 for1900
    4 for 850+1900
    5 for 900+1800
    Default: 900 + 1800

203. Not applicable if CDMA Mobile Card is installed in your system.

To select the Preferred Network Mode[204] for the Mobile Port, dial:

- ***8038-1-Mobile Port Number-Preferred Network Mode*** to select the band for a single mobile port.
- ***8038-2-Mobile Port Number -Mobile Port Number -Preferred Network Mode*** to select the same band for a range of mobile ports.
- ***8038-*-Preferred Network Mode*** to select the same band for all mobile ports.
  Where,
  Mobile Port is the software port number of the mobile port, from 001 to 064.
  Preferred Network Mode is:
  1 for Any (LTE/UMTS/GSM)
  2 for GSM
  3 for UMTS
  4 for LTE

To assign SIM PIN[205] to the Mobile Port, dial:

- ***8006-1-Mobile Port Number-SIM PIN-#**** to assign SIM PIN to a single mobile port.
  Where,
  Mobile Port Number is the software port number of the mobile port, from 001 to 064.
  SIM PIN is the new SIM PIN, any combination of 4 to 8 digits. Terminate the command with '#*' if SIM PIN is fewer than 8 digits.

To select Incoming Call mode on the Mobile port, dial:

- ***8005-1-Mobile-Mode*** to select the mode for a single mobile port.
- ***8005-2-Mobile-Mobile-Mode*** to select the same mode for a range of mobile ports.
- ***8005-*-Mode*** to select the same mode for all mobile ports.
  Where,
  Mode is
  1 for Accept Incoming Calls.
  2 for Ignore Incoming Calls
  3 for Reject Incoming Calls
  Default: Accept

To assign a Trunk Feature Template to the Mobile Port, dial:

- ***5807-1-Mobile Port Number-Template Number*** to assign a template to a single mobile port.
- ***5807-2-Mobile Port Number-Mobile Port Number-Template Number*** to assign the same template to a range of mobile ports.
- ***5807-*-Template Number*** to assign the same template to all mobile ports.
  Where,
  Mobile Port Number is the number of the software port from 001 to 064.
  Template Number is the number of the Trunk Feature Template from 01 to 50.

To enable/disable CLIR[206] on the Mobile Port, dial:

- ***8031-1-Mobile Port Number-CLIR*** to enable/disable CLIR on a single mobile port.
- ***8031-2-Mobile Port Number-Mobile Port Number-CLIR*** to enable/disable CLIR on a range of mobile ports.
- ***8031-*- CLIR*** to enable/disable CLIR on all mobile ports.
  Where,
  Mobile Port Number is the number of the software port from 001 to 064.
  CLIR is

204. Not applicable if CDMA Mobile Card is installed in your system.
205. Not applicable if CDMA Mobile Card is installed in your system.
206. Not applicable if CDMA Mobile Card is installed in your system.

0 for Disable
1 for Enable
Default: Disable

To enable RCOC[207] on Mobile Port, dial:

- ***8030-1-Mobile -Code*** to enable the feature on a single mobile port.
- ***8030-2-Mobile-Mobile-Code*** to enable the feature on a range of mobile ports.
- ***8030-*-Code*** to enable the feature on all mobile ports.
  Where,
  Mobile Port Number is the number of the software port from 001 to 064.
  Code is
  0 for Disable
  1 for Enable
  Default: Disabled

To assign Cost Factor to a Mobile Port, dial:

- ***8001-1-Mobile-Cost Factor*** to assign cost factor to a single mobile port.
- ***8001-2-Mobile-Mobile-Cost Factor*** to assign the same cost factor to a range of mobile ports.
- ***8001-*-Cost Factor*** to assign the same cost factor to all mobile ports.
  Where,
  Mobile Port Number is the number of the software port from 001 to 064.
  Cost Factor is from 01 to 99.

- For Advanced Configuration of the Mobile Ports, use the following commands:

  To set Network Registration Retry Count, dial:

  - ***8004-1-Mobile-N/w Registration Retry Count*** to set the count for a single mobile port.
  - ***8004-2-Mobile-Mobile-N/w Registration Retry Count*** to set the same count for a range of mobile ports.
  - ***8004-*-N/w Registration Retry Count*** to set the same count for all mobile ports.
    Where,
    Mobile Port Number is the number of the software port from 001 to 064.
    N/w Registration Retry Count is from 001 - 255: Default: 002

  - For commands to program Call Budget on Mobile Ports, refer the topic “Call Budget on Trunk”.

  To enable/disable Call Back on Mobile Port, dial:

  - ***8010-1-Mobile-Code*** to enable/disable Call Back on a single mobile port.
  - ***8010-2-Mobile-Mobile-Code*** to enable/disable Call Back on a range of mobile ports.
  - ***8010-*-Code*** to enable/disable Call Back on all mobile ports.
    Where,
    Mobile Port Number is the number of the software port from 001 to 064.
    Code is
    0 for Disable
    1 for Enable
    Default: Disabled

  To change Call Back Timer for the Mobile Port, dial:

  - ***8011-1-Mobile-Call Back Timer*** to change Call Back Timer for a single mobile port.
  - ***8011-2-Mobile-Mobile-Call Back Timer*** to set the same Call Back Timer duration for a range of mobile ports.
  - ***8011-*-Call Back Timer*** to set the same Call Back Timer duration for all mobile ports.

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207. Not applicable if CDMA Mobile Card is installed in your system.

Where,
Mobile Port Number is the number of the software port from 001 to 064.
Call Back Timer is from 01 to 99 seconds.
Default: 010 seconds.

To select the Call Back Mode, dial:

- ***8012-1-Mobile-Call Back Mode*** to select Call Back Mode for a single mobile port.
- ***8012-2-Mobile-Mobile Call Back Mode*** to select the same Call Back Mode for a range of mobile ports.
- ***8012-*-Call Back Mode*** to select the same Call Back Mode for all mobile ports.
  Where,
  Mobile Port Number is the number of the software port from 001 to 064.
  Call Back Mode is
  1 for Built-In Auto Attendant
  2 for PIN Authentication - Multiple Calls
  3 for CLI Authentication - Multiple Calls
  4 for CLI Authentication - Single Call - Answer Signaling
  5 for Operator
  Default: Operator

To program Call Back on for the Mobile Port, dial:

- ***8013-1-Mobile-Call Back on*** to program call back on for a single mobile port.
- ***8013-2-Mobile-Mobile-Call Back on*** to program the same call back on for a range of mobile ports.
- ***8013-*-Number List*** to program the same call back on for all mobile ports.
  Where,
  Mobile Port Number is the number of the software port from 001 to 064.
  Call Back on is
  1 for CLI Number
  2 for Alternate Number.

To assign a Call Back - Incoming Number List to a Mobile Port, dial:

- ***8034-1-Mobile-Incoming Number List*** to assign a list to a single mobile port.
- ***8034-2-Mobile-Mobile-Incoming Number List*** to assign the same list to a range of mobile port.
- ***8034-*-Incoming Number List*** to assign the same list to all mobile ports.
  Where,
  Mobile Port Number is the number of the software port from 001 to 064.
  Incoming Number List is from 01 to 16
  Default: 15

To assign a Call Back - Outgoing Number List to a Mobile Port, dial:

- ***8035-1-Mobile-Outgoing Number List*** to assign a list to a single mobile port.
- ***8035-2-Mobile-Mobile-Outgoing Number List*** to assign the same list to a range of mobile port.
- ***8035-*-Outgoing Number List*** to assign the same list to all mobile ports.
  Where,
  Mobile Port Number is the number of the software port from 001 to 064.
  Outgoing Number List is from 01 to 16
  Default: 16

To select Call Back From for a Mobile Port, dial:

- ***8036-1-Mobile-Call Back From*** to select call back from for a single mobile port.
- ***8036-2-Mobile-Mobile-Call Back From*** to select the same call back from option for a range of mobile ports.
- ***8036-* -Call Back From*** to select the same call back from option for all mobile ports.
  Where,
  Mobile Port Number is the number of the software port from 001 to 064.

Call Back From is
1 for Same Port
2 for OGTB Group
Default: Same Port

To assign a Call Back - OGTB Group for a Mobile Port, dial:

- ***8037-1-Mobile-OGTB Group*** to assign an OGTBG to a single mobile port.
- ***8037-2-Mobile-Mobile-OGTB Group*** to assign the same OGTBG to a range of mobile ports.
- ***8037-*-OGTB Group*** to assign the same OGTBG to all mobile ports.
  Where,
  Mobile Port Number is the number of the software port from 001 to 064.
  OGTB Group is from 01 to 32
  Default: 01

To enable/disable Accept Anonymous Calls on Mobile Port, dial:

- ***8029-1-Mobile-Code*** to enable/disable Anonymous Calls on Mobile Port on a single mobile port.
- ***8029-2-Mobile-Mobile-Code*** to enable/disable Anonymous Calls on Mobile Port on a range of mobile ports.
- ***8029-*-Code*** to enable/disable Anonymous Calls on Mobile Port on all mobile ports.
  Where,
  Mobile Port Number is the number of the software port from 001 to 064.
  Scheduled Reset Date is from 01 to 31.

To program the Pause Timer on Mobile Port, dial:

- ***8014-1-Mobile-Pause Timer*** to set the timer on a single mobile port.
- ***8014-2-Mobile-Mobile-Pause Timer*** to set the same timer on a range of mobile ports.
- ***8014-*-Pause Timer*** to set the same timer on all mobile ports.
  Where,
  Mobile Port Number is the number of the software port from 001 to 064.
  Pause Timer is from 1 to 9 seconds. Default: 1 second.

To program DTMF ON Time on Mobile Port, dial:

- ***8015-1-Mobile-DTMF ON Time*** to set the time on a single mobile port.
- ***8015-2-Mobile-Mobile-DTMF ON Time*** to set the same time on a range of mobile ports.
- ***8015-*-DTMF ON Time*** to set the same time on all mobile ports.
  Where,
  Mobile Port Number is the number of the software port from 001 to 064.
  DTMF ON Time is
  1 for 100 msec
  2 for 200 msec
  Default: 100 msec

To set DTMF Detection Mode on Mobile Port, dial:

- ***8051-1-Mobile-Mode*** to set the DTMF Detection Mode on a single mobile port.
- ***8051-2-Mobile-Mode*** to set DTMF Detection Mode on a range of mobile ports.
- ***8051-*-Mobile-Mode*** to set the same DTMF Detection Mode on all mobile ports.
  Where,
  Mobile Port Number is the number of the software port from 001 to 064.
  DTMF Detection Mode is
  1 for Using DSP
  2 for Using Module
  Default: Using DSP

To program DTMF Detection Duration on Mobile Port, dial:

- ***8052-1-Mobile-Duration*** to set the DTMF Detection Duration on a single mobile port.
- ***8052-2-Mobile-Duration*** to set the DTMF Detection Duration on a range of mobile ports.
- ***8052-*-Mobile-Duration*** to set the DTMF Detection Duration on all mobile ports.
  Where,
  Mobile Port Number is the number of the software port from 001 to 064.
  DTMF Detection Duration range is from 20 to 100 msec.
  Default: 30 msec

To assign the Mobile Port to a 'Category' for Logical Partition, dial:

- ***8018-1-Mobile-Category*** to assign a single mobile port to a category.
- ***8018-2-Mobile-Mobile-Category*** to assign a range of mobile ports to the same category.
- ***8018-*-Category*** to assign all mobile ports to the same category.
  Where,
  Mobile Port Number is the number of the software port from 001 to 064.
  Category is from 1 to 3.
  1 for Trunk ports interfaced with PSTN /PLMN (Public Land Mobile Network)
  2 for Leased lines terminated in the trunk ports.
  3 Trunk ports used to interconnect two Systems.
  4 SIP Trunks interfaced with ISP/ITSP
  Default: 1

To enable/disable Gateway Application on the Mobile Port, dial:

- ***8016-1-Mobile-Gateway Application flag*** to enable/disable on a single mobile port.
- ***8016-2-Mobile-Mobile-Gateway Application flag*** to enable/disable on a range of mobile ports.
- ***8016-*-Gateway Application flag*** to enable/disable on all mobile ports.
  Where,
  Mobile Port Number is the number of the software port from 001 to 064.
  Gateway Application flag is.
  0 for Disable
  1 for Enable
  Default: Disable

To program the DTMF String for the Gateway Application on the Mobile Port, dial:

- ***8017-1-Mobile-DTMF String*** to program the string on a single mobile port.
- ***8017-2-Mobile-Mobile-DTMF String*** to program the same string on a range of mobile ports.
- ***8017-*-DTMF String*** to program the same string on all mobile ports.
  Where,
  Mobile Port Number is the number of the software port from 001 to 064.
  DTMF String is a maximum of 4 digits. Default: CCC

To enable/disable Debug on a Mobile Port, dial:

- ***8028-1-Mobile port number-Debug Code*** to enable/disable debug on a single mobile port.
- ***8028-2-Mobile port number-Mobile port number-Debug Code*** to enable/disable debug on a range of mobile ports.
- ***8028-*-Debug Code*** to enable/disable debug all mobile ports.
  Where,
  Mobile Port Number is the number of the software port from 001 to 064.
  Debug Code is
  1 for Enable
  0 for Disable
  Default: Disable

To program SIP Rx Gain on the Mobile Port, dial:

- ***8041-1-Mobile-SIP Rx Gain*** to program the SIP Rx Gain on a single mobile port.
- ***8041-2-Mobile-Mobile-SIP Rx Gain*** to program the same SIP Rx Gain on a range of mobile ports.
- ***8041-*-SIP Rx Gain*** to program the same SIP Rx Gain on all mobile ports.
  Where,
  Mobile Port Number is the number of the software port from 001 to 064.
  Rx and Tx Gain is 01 to 63 dB: Default: 4.
  For Rx and Tx Gain at SIP codes, see the tables below.

To program SIP Tx Gain on the Mobile Port, dial:

- ***8042-1-Mobile-SIP Tx Gain*** to program the SIP Tx Gain on a single mobile port.
- ***8042-2-Mobile-Mobile-SIP Tx Gain*** to program the same SIP Tx Gain on a range of mobile ports.
- ***8042-*-SIP Tx Gain*** to program the same SIP Tx Gain on all mobile ports.
  Where,
  Mobile Port Number is the number of the software port from 001 to 064.
  Rx and Tx Gain is 01 to 63 dB: Default: 4.
  For Rx and Tx Gain at SIP codes, see the tables below.

Code	Rx Gain at SIP	Tx Gain at SIP
0	--	--
1	-31	-31
2	-30	-30
3	-29	-29
4	-28	-28
5	-27	-27
6	-26	-26
7	-25	-25
8	-24	-24
9	-23	-23
10	-22	-22
11	-21	-21
12	-20	-20
13	-19	-19
14	-18	-18
15	-17	-17

Code	Rx Gain at SIP	Tx Gain at
16	-16	-16
17	-15	-15
18	-14	-14
19	-13	-13
20	-12	-12
21	-11	-11
22	-10	-10
23	-9	-9
24	-8	-8
25	-7	-7
26	-6	-6
27	-5	-5
28	-4	-4
29	-3	-3
30	-2	-2
31	-1	-1

Code	Rx Gain at SIP	Tx Gain at SIP
32	0	0
33	1	1
34	2	2
35	3	3
36	4	4
37	5	5
38	6	6
39	7	7
40	8	8
41	9	9
42	10	10
43	11	11
44	12	12
45	13	13
46	14	14
47	15	15

Code	Rx Gain at SIP	Tx Gain at SIP
48	16	16
49	17	17
50	18	18
51	19	19
52	20	20
53	21	21
54	22	22
55	23	23
56	24	24
57	25	25
58	26	26
59	27	27
60	28	28
61	29	29
62	30	30
63	31	31

- Exit SE mode.

Network Selection

After the Mobile Card is successfully installed and powered on, the mobile port is programmed to automatically locate and register with the Network that supports the SIM card installed in. Also, at each power ON, the mobile port (SIM) will automatically register with the Network that supports the SIM on it. However, if the Mobile port fails to register, it will restart the process of network selection on the expiry of the Network Registration Retry Timer[208].

If the SARVAM UCS is located in a border area where more than one Network Operator is available, it is possible that the SIM card may register with another available network and result in 'Roaming' charges. To avoid this, you must disable automatic network selection and program manual network selection.

When you enable manual network selection, you must program the Network Operator Priority Table. This table requires you to program the Network Operator Codes (MCC-MNC)[209] in order of priority for a Mobile Port. So, whenever you register with the network manually, select the Network Operator that matches in order of priority. If

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*208. The **Network Registration Retry Timer** defines the time for which the Mobile port, which has failed to register with the network, should wait before attempting to re-register with the network. Network registration retry timer is 2 minutes and is non-programmable.*

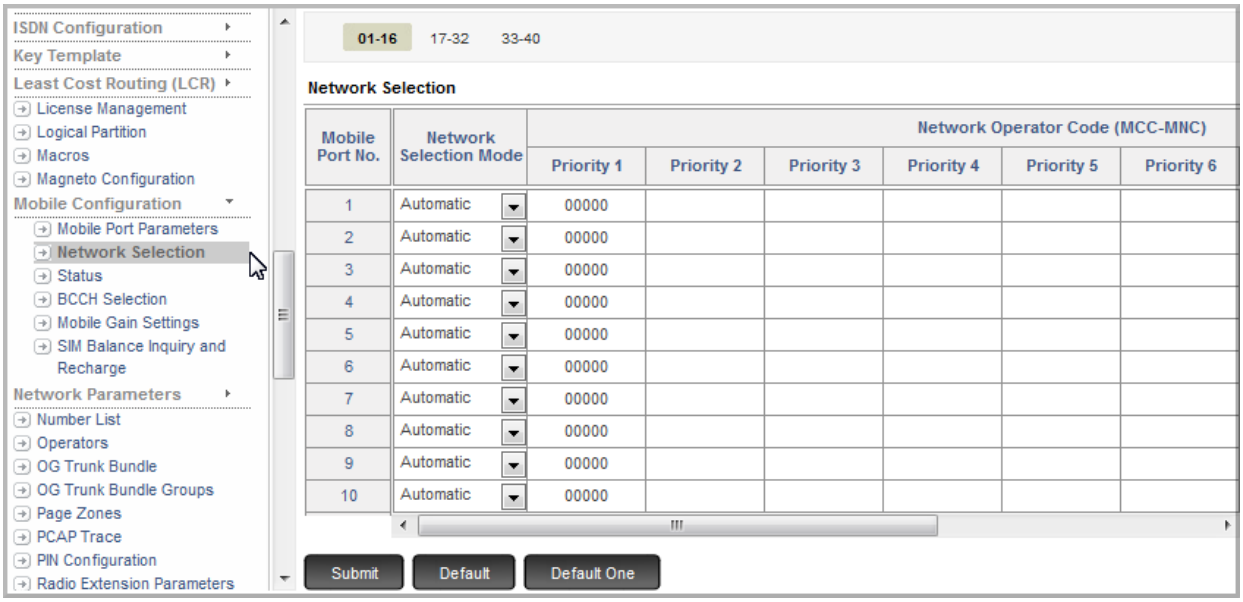
the Mobile port fails to register, it will restart the process of network selection on the expiry of the Network Registration Retry Timer.

If no match is found, the Mobile port (SIM) will not get registered with any of the available network operators and no calls can be made or received on this port.

Configuring Network Selection using Jeeves

While still logged in as System Engineer,

- Under **Mobile Configuration**, click **Network Selection**.



- Set the **Network Selection Mode** to **Manual**.

If CDMA Mobile Card is installed in your system, keep the Network Selection Mode as Automatic. Manual mode is not applicable in this case.

- Enter the **Network Operator Codes (MCC-MNC)** in order of priority. The codes must not exceed 8 digits. You can store up to 9 Network Operator Codes in the order of priority.

- Repeat the same steps to set network selection mode for other mobile ports.

- Click **Submit** at the bottom of the page to save your settings.

- *When you change the Network Selection Mode to 'Manual' and the Network Operator Code manually, the change you made will not come into effect until you have restarted the Mobile Port.*

- *To restart the Mobile Port,*
  - *go back to **Mobile Port Parameters**.*
  - *Click the check box to disable the **Enable Port** flag.*

209. ***The Network Operator Code*** *comprises of the Mobile Country Code (MCC) appended by the Mobile Network Code (MNC). The MCC is usually a 3-digit code that identifies a country. A single country may be assigned more than one MCC. For example the MCC assigned to India is 404, but same code applies to all network operators in the country.*

The MNC is usually a 2/3-digit code. The MCC-MNC combination uniquely identifies the home network of the mobile terminal or the mobile user. For example, AirTel, a GSM network operator in India, has different MNC assigned to its networks in various states. The MNC for AirTel in the state of Maharashtra is 90, while the same for the state of Gujarat is 98.

- *Click **Submit** at the bottom of the page.*
- *Wait for the page to refresh.*
- *Click the check box to enable the **Enable Port** flag.*
- *Click **Submit** at the bottom of the page.*
- *The Mobile Port will be restarted.*

Configuring Network Selection using a Telephone

- Enter SE mode from a DKP/SLT.

  To select Network Selection mode, dial:
  - ***8007-1-Mobile Port Number-Code*** to select network selection mode for a single mobile port.
  - ***8007-2-Mobile Port Number-Mobile Port Number-Code*** to select the same network selection mode for a range of mobile ports.
  - ***8007-*-Code*** to select the same network selection mode for all mobile ports.
    Where,
    Mobile Port Number is the number of the software port from 001 to 064.
    Code is
    1 for Automatic Selection
    2 for Manual Selection
    Default: Automatic Selection

    For example, to program Manual Network Selection on Mobile Port number 1, dial ***8007-1-01-2***

  To program the network operator codes in order of priority for Mobile port, dial:
  - ***8008-1-Mobile-Priority-Network Operator Code-#**** to program network operator codes for a single mobile port.
  - ***8008-2-Mobile-Mobile-Priority-Network Operator Code-#**** to program the same network operator codes for a range of mobile ports.
  - ***8008-*-Priority-Network Operator Code-#**** to program the same network operator codes of all mobile ports.
    Where,
    Mobile Port Number is the number of the software port from 001 to 064.
    Priority is from 1 to 9.
    Network Operator Code is MCC-NCC maximum 8 digits. Terminate the command with #* if the number of digits is fewer than 8.

    For example, following Network Operator Codes need to be programmed in the order of priority on Mobile Port number 01:
    Priority 1=40498
    Priority 2 = 40425
    Priority 3 = 40421

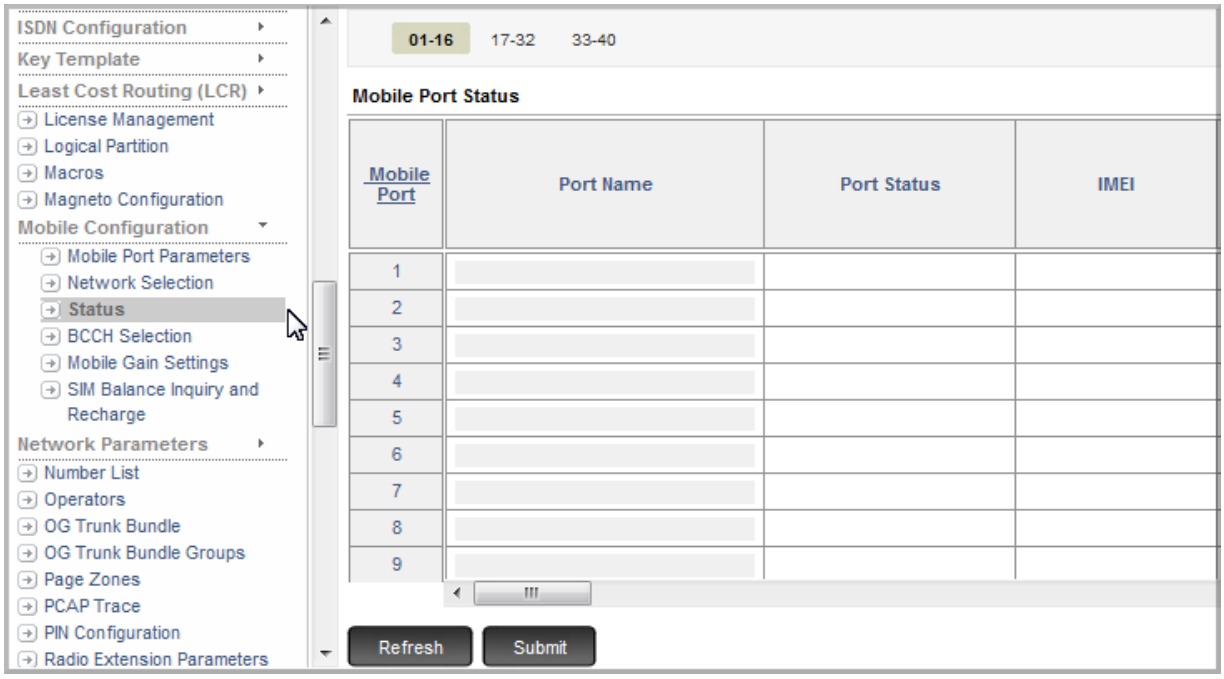
    Dial:
    8008-1-01-1-40498-#*
    8008-1-01-2-40425-#*
    8008-1-01-3-40421-#*

- Exit SE mode.

Viewing Mobile Port Status

You can view the status of Mobile ports on Jeeves only. To do this,

- Under **Mobile Configuration**, click **Status**.
- Wait for 4-5 seconds after the Port Status page is opened.
- Click **Refresh** at the bottom of the page for current view.



- The Port status page will display the following parameters for all Mobile ports that have been enabled:
  - **Port Name:** This is the name by which the Mobile Port is programmed.
  - **Port Status:** This is the status of the connection - showing Initialization with the Network, Registering with the Network, Idle or Busy state of the network. It also shows errors and alerts when SIM is absent, the wrong SIM PIN has been entered, SIM PUK is required.
  - **IMEI:** This is the unique identification number of (the GSM engine) each Mobile port.

If CDMA Mobile Card is installed in your system, ESN No.(Electronic Serial Number) will be displayed as the parameter name instead of IMEI.

In absence of RUIM Card, "00000000" will be displayed as the ESN No.

- **Network Operator Code:** This is the MCC-MNC code of the network with which the mobile port is registered.
- **Network Operator Name:** This is the name of the service provider/network operator with which the Mobile Port is registered.
- **SMSC Number:** This is the number of the SMS Center of the network operator. If CDMA Mobile Card is installed in your system, SMSC Number status will be displayed as blank.
- **Signal Strength (dBm):** This is the signal strength in '-dBm' as received from the network with which the Mobile port is registered.

- **Bit Error Rate (BER):** BER is Bit Error Rate which defines the quality of the channel.

- **Ec/Io (dB)[210]:** This is the ratio of the received energy per chip (= code bit) and the interference level, given in dB. In case no true interference is present, the interference level is equal to the noise level. This parameter is significant only when the GSM module is registered with the 3G network.

  The range of this parameter can be from 0 to 63dB. A ratio of 10dB to 14dB is normal, numbers going higher than that is progressively worse.

- **Call Duration:** This is the total call duration of matured outgoing calls[211] on the Mobile port. This data is used for calculating Answer Seizure Ratio (ASR) for the port. It is displayed in MMMMMM:SS format.

- **Dialed Calls:** This is the total number of outgoing calls[212] made from the Mobile port. This data is used for calculating Answer Seizure Ratio (ASR) for the port.

- **Successful Calls:** This is the total number of matured outgoing calls made from the Mobile port. This data is used for calculating Answer Seizure Ratio (ASR) and Average Call Duration for the port.

- **ASR % :** This is the Answer Seizure Ratio (ASR) calculated by the system for the Mobile port, in terms of percentage. ASR is the sum of all outgoing matured calls from the Mobile port, divided by the total number of outgoing calls made from the Mobile port, multiplied by 100. The system calculates ASR after the completion of the outgoing call.

- **ACD:** This is the Average Call Duration (ACD) of outgoing calls made from the Mobile port. It is an indicator for monitoring the network condition. Decreasing ACD is indicative of trouble in the network condition.

  The system calculates ACD after the completion of the outgoing calls, by dividing the total call duration by the number of outgoing matured calls.

- **Reset ASR and ACD:** This field allows the System Engineer to manually the ASR and the ACD of the Mobile port.

  The parameters Total Call duration, Number of matured calls, Total Number of OG Calls, ASR and ACD are saved in the configuration, and are not on Power OFF condition. The system maintains the statistics for the last 999 calls. When the total number of outgoing calls exceeds 999, the system will stop calculating ACD and ASR and will display ASR and ACD calculated on the basis of the last 999 calls only.

  Therefore, the System Engineer must manually ASR and ACD when the total number of calls reaches 999. When you ASR and ACD the number of call matured and the number of calls dialled is to 0.

  ASR and ACD can be anytime, even when the total number of calls is less than 999.

When ACD is, only the 'Total Call Duration' maintained for the ACD calculation will be. The 'Total Call Duration' of the Call Budget, the consumed minutes maintained for the Call Budget on the mobile port will remain unaffected.

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210. Not applicable if CDMA Mobile Card is installed in your system.
211. Matured calls are outgoing calls for which 'CONNECT' message was received from the network.
212. The total number of outgoing calls made includes the number of times the ATD has been sent from the Mobile port to the network.

- **SIM ID**[213]**:** This is the Integrated Circuit Card ID (ICC-ID) of the SIM card inserted in the Mobile port. Each SIM is internationally identified by its ICC-ID. ICC-IDs are stored in the SIM card and are also printed on the SIM card body.

- **IMSI**[214]**:** International Mobile Subscriber Identity (IMSI) is a unique number stored in the SIM card.

- **Cell ID:** This is the 16-bit identifier that identifies the cell. The cell is the radio coverage area given by one BTS (Base Transceiver Station).

If CDMA Mobile Card is installed in your system, NID (Network Identification) will be displayed as the parameter name instead of Cell ID.

- **Location Area Code (LAC):** The LA (Location Area) is a group of cells defined by the Operator. The LAC (Location Area Code) uniquely identifies a LA within a PLMN (Public Land Mobile Network).

If CDMA Mobile Card is installed in your system, SID (System Identification) will be displayed as the parameter name instead of LAC.

- **Call Budget Type:** This shows the Call Budget Type, whether Amount, Minutes or Number of Calls, are set on the Mobile port.

- **Allotted Amount/ Minutes/Calls:** This shows the sum/number of minutes/number of calls allotted as Call Budget on the Mobile port.

- **Consumed Amount/ Minutes/Calls:** This shows the sum/number of minutes/number of calls of the allotted Call Budget that has been used up on the Mobile port.

- **Scheduled Reset:** This shows if you have enabled Scheduled Reset.

- **Call Budget Reset Schedule (Date):** This shows whether the consumed Call Budget on the Mobile port is to be Daily or on a particular date of a month.

- **Reset Consumed Amount/Minutes/Calls:** This editable field allows the you to reset the consumed Call Budget Amount/Minutes/Calls at any time, manually.

- SMS Budget **Scheduled Reset**: This shows if you have enabled Scheduled Reset for the SMS budget.

If CDMA Mobile Card is installed in your system, any parameters related to SMS will not be supported.

- SMS **Budget Reset Schedule (Date):** This shows the date on which the SMS Budget will be reset.

- SMS Budget **Consumed SMS (Total):** This shows the number of SMS of the allotted SMS Budget that has been used up on the Mobile port.

- **Consumed SMS (Daily):** This shows the number of SMS of the allotted Daily SMS Budget that has been used up on the Mobile port.

- **Reset Consumed SMS (Total) Budget:** This editable field allows the you to reset the consumed SMS Budget, manually. It will reset both **Consumed SMS (Total)** and **Consumed SMS (Daily)**.

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213. Not applicable if CDMA Mobile Card is installed in your system.
214. Not applicable if CDMA Mobile Card is installed in your system.

The Consumed SMS Budget can be reset from the System Engineer mode as well as the System Administrator mode manually at any time or on a scheduled date from the System Administrator mode only. Refer Resetting SMS Budget.

- **Registered with Network:** This shows the type of network with which the Mobile port is registered, whether GSM, GSM Compact, 3G or UMTS, CDMA, 4G or LTE or Not Registered.
- **Firmware Version of Engine:** This shows the firmware version of the mobile Engine.

*You can also view the Mobile Port Status from the **Status** link. To view, click the Mobile link under Status.*

Resetting SMS Budget[215]

You manually reset the Consumed SMS Budget from the SE mode from the **Status** page (as explained above) or from the SA mode.

To reset the SMS Budget from the SA mode, follow the steps given below:

- Log in as System Administrator.
- Under **Trunks Properties**, click the **Mobile** link.
  - **Consumed SMS (Total):** This shows the number of SMS of the allotted SMS Budget that has been used up on the Mobile port.
  - **Consumed SMS (Daily):** This shows the number of SMS of the allotted Daily SMS Budget that has been used up on the Mobile port.
  - **Scheduled reset consumed SMS (Total) budget:** Enable this check box if you want only the **Consumed SMS (Total)** Budget to be reset on a particular date of every month.
  - **Budget Reset Schedule (Date):** Select the date of the month (Daily or 1-31) on which you want the SMS Budget to be reset every month.
  - **Reset Consumed SMS (Total) Budget:** Enable this check box to reset the consumed SMS Budget, manually. It will reset both **Consumed SMS (Total)** and **Consumed SMS (Daily)**.

Resetting ASR and ACD using a Telephone

- Enter SE mode from a DKP/SLT.

  To ASR and ACD calculation for Mobile Port, dial:
  - ***8032-1-Mobile-1*** to ASR-ACD calculation for a single mobile port.
  - ***8032-2-Mobile-Mobile-1*** to ASR-ACD calculation for a range of mobile ports.
  - ***8032-*-1*** to ASR-ACD calculation for all mobile ports.
- Exit SE mode.

The System Administrator can also

215. Not applicable if CDMA Mobile Card is installed in your system.

- *view the ID of the Mobile Network Operator with which the Mobile Port is currently registered.*
- *check Signal Strength of a mobile port, whenever there is trouble placing calls over a Mobile port to rule out weak signal as the cause.*
- *Reset ASR and ACD calculation.*

To do this,

- *Enter SA mode from a DKP.*
- *Dial **1072-017-Mobile Port Number** to view ID of Network Operator.*
  You will get a confirmation tone.
  If the port is successfully registered, the name of the network Operator will be displayed on the LCD of the DKP.
  If the port is not registered, the message 'Error' will be displayed on the LCD.
- *Dial **1072-032-Mobile Port Number** to check signal strength.*
- *Wait for the confirmation tone.*
  When the network responds, the Mobile Network Signal Strength will be displayed on the LCD of your DKP and programming beeps will be played.

  The Signal Strength values are in -dBm. '-113' indicates weak signal, whereas '-51' indicates maximum signal strength.
- *Dial **1072-022-Mobile Port Number***

  You will get the confirmation tone and the confirmation message “ASR-ACD Reset” will appear on your phone’s display.

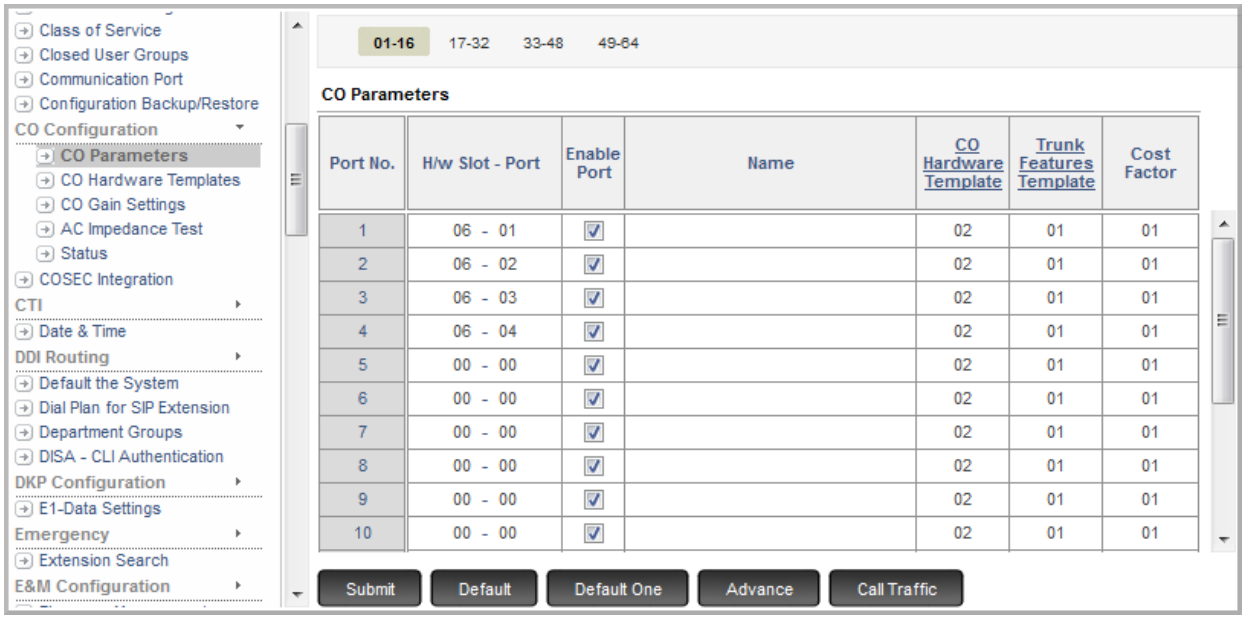
Configuring CO Trunks

The SARVAM UCS supports a maximum of 64 Analog Two-Wire Trunk Lines. Before you begin configuring the CO trunk ports, ensure that the CO trunk Card has been installed correctly.

You may configure the CO ports from Jeeves and using a Telephone.

Configuring CO Trunks using Jeeves

- Log into Jeeves as System Engineer.
- Under **Configuration**, click **CO Configuration**.
- Click **CO Parameters.**



Configure the following port parameters:

- **CO No.:** This non-editable field is the number of the software port of the CO Trunk.
- **Hardware Slot and Port:** 'Slot' is the number of the Universal Slot in which the CO Card has been inserted. 'Port' is the number of the CO trunk port on that card.

  By default the SARVAM UCS can detect and assign the hardware slot and port numbers automatically to the CO (software) ports. However, if required, you may change the Hardware Slot and Port assigned to the CO software port. In this case, enter the desired Hardware Slot and Port number in this field.

  If you want to de-assign the Hardware Slot and Port, Enter '00' in both fields.

- **Enable Port:** This flag is for enabling or disabling a CO Trunk port. When a CO Trunk port is disabled, neither incoming nor outgoing calls can be made from that port.

  By default, the port is enabled. You may disable ports that are not functioning by clearing this check box.

  You may disable CO port in case of trouble with the CO line.

- **Name:** You may assign a 'Name' to each CO Trunk to facilitate identification. Whenever there is an incoming call without CLI on this port, the Name you have programmed will be displayed on the landing extension.

  The Name of the port may be the name of the Service Provider of this Trunk Line (recommended).

  The Name may comprise a maximum of 18 characters.

- **CO Hardware Template:** A CO Hardware Template is a set of features that completely define the behavior of the hardware port of the CO, such as Type of CO Trunk, AC Termination Impedance, Pulse-Tone Dialing, Answer Supervision, Disconnect Supervision, DTMF detection, etc.

  Apply a CO Hardware template to the CO trunk port. The SARVAM UCS offers 50 CO Hardware Templates. By default, CO Hardware Template number 01 is assigned to all CO Trunks. Refer the topic "CO Hardware Template" to know more.

  Check if this default template fulfills the feature requirements of the CO Hardware Ports by clicking the 'CO Hardware Template' link.

  If CO Hardware Template 02 fulfills your requirements, and if the same features at to be applied on all CO trunk ports, retain Template 02. Similarly, if you want only a few changes to be made to Template 02 and apply it on all CO Ports, make the changes and retain the template.

  However, if different sets of features are to be allowed to different CO hardware ports, then prepare separate CO Hardware Templates and apply them on the ports as required. To do this,

  - Click the **CO Hardware Template** link to open the page.
  - Select a CO Hardware Template Number.
  - Customize the CO Hardware Template.
  - Click **Submit** at the bottom of the page.
  - Go back to the **CO Parameters** page.
  - Apply the CO Hardware Template you customized to the CO Port by entering the template number in the **CO Hardware Template** field of this port.
  - Click **Submit** at the bottom of the page.
  - Repeat the same steps to customize another template and apply it to the CO Port.

  To know more about the hardware port features and customizing templates, refer the topic "CO Hardware Template".

- **Trunk Feature Template:** A Trunk Feature Template is a set of features like Time Table, Operator, Auto Attendant, DISA, Trunk Auto Answer, Trunk Landing Group, SMDR Storage, etc., that defines the behavior of a Trunk. Apply a Trunk Feature Template to the CO Trunk port. By default, Trunk Feature Template 01 is applied on all CO Trunks as well as all other trunk types like ISDN BRI, ISDN T1/E1/PRI, GSM, and VoIP. Refer the "Trunk Feature Template" topic to know more.

Click the **Trunk Feature Template** link to open the page. Check if the default Template 01 fulfills your requirement for the CO Trunk port.

If the default Template 01 does not fulfill your requirement, you may prepare a different Trunk Feature Template and apply on all CO Ports. For this,

- Click the **Trunk Feature Template** link.
- The Trunk Feature Template page will open.
- Prepare a new template, for example 02 to match your requirements.
- Click **Submit** at the bottom of the page to save changes.
- Go back to the **CO Parameters** page.
- Go to the CO Software Port Number you want to assign the Template you prepared.
- Enter the number of the Template you prepared (02) in the **Trunk Feature Template** field.
- Click **Submit** at the bottom of the page to save changes.

You may also prepare different Templates for different CO Ports, for example Template 02 for certain ports, Template 03 for others. In this case, follow the steps described above. For each CO Port, enter the number of the template you have prepared for that port.

To know more about customizing templates, refer the topic “Trunk Feature Template”.

- **Cost Factor:** This parameter is of relevance only if 'Least Cost Routing' feature is applied on the CO Trunk port.

  Cost Factor is a number assigned to each trunk for identification. This number also serves as a preference number for the trunk. The Cost Factor can be from 1 to 99. Trunks having the same preference must be assigned the same Cost Factor. Different trunk types can also be assigned the same Cost Factor. These trunks are used for routing calls.
  Assign a Cost Factor to the CO Trunk port, for instance 02, and program Least Cost Routing Table accordingly.

  For example, if you want to route all outgoing calls starting with number '6' through the CO Trunk Port 001 only,

  - You must first assign a Cost Factor (01-99) to CO Port 001, for example, 02.
  - Click the link **Least Cost Routing**.
  - Click the **Least Cost Routing - Number Based** link to open the page.

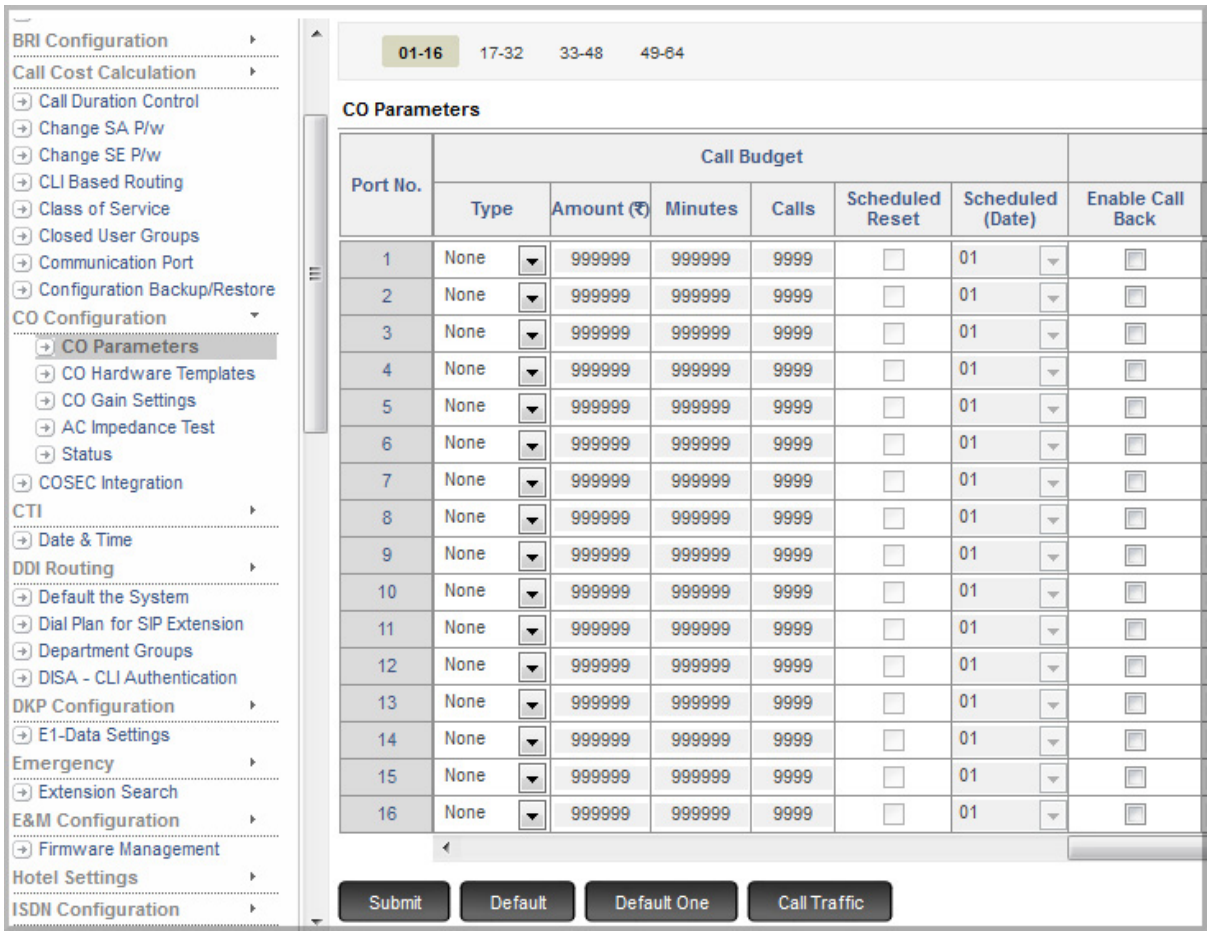
- Enter '6' in the **Number** column, Cost Factor 02 as **Preference 1, 2, 3 and 4**.

- Click **Submit** at the bottom of the page to save your settings.

  All outgoing calls assigned Cost Factor trunk 02 will be made from CO Trunk Port 001.

Advanced Configuration

The above listed parameters fulfill the basic CO trunk port configuration requirements of most users. However, for users who need advanced features like Call Budget on the CO trunk ports, you may click the **Advance** button and program the following parameters:



- **Call Budget:** By default, Call Budget is enabled on the trunk. If you wish to change the default configuration or disable it for this CO trunk port, configure the parameters as per your requirement:
    - **Type:** Select the type of Call Budget on Trunk—Amount, Minutes or Number of Calls—to be applied on this CO trunk port. By default, Minutes is selected as the Call Budget type.To disable select Type as None.
    - **Amount:** If you selected 'Amount' as the Call Budget Type, enter the Budget Amount in this field. By default the Amount is set to 999999.
    - **Minutes:** If you selected 'Minutes' as the Call Budget Type, enter the number of Minutes in this field. By default the number of minutes is set as 000300.
    - **Calls:** If you selected ‘Calls’ as the Call Budget Type, enter the number of calls in this field. By default, the number of calls is set to 9999.
    - **Scheduled Reset:** Enable this flag if you want the Call Budget Amount/Minutes to be reset on a particular date of every month.

- **Scheduled (Date):** Enter the date of the month (Daily or 1-31) on which you want the Call Budget Amount/Minutes/Number of Calls to be reset every month. You may select ‘Daily’ if your plan suggests so.

The consumed Call Budget Amount/Minutes/Number of Calls can be reset from SE and SA Mode, referred to as Manual Reset. Refer the feature description “Call Budget on Trunk”.

- **Call Back:** This parameter is related to the ‘Call Back on Trunk Port’ feature. If you want to enable the 'Call Back on Trunk Port' feature on this CO trunk, configure the following parameters:

  - **Enable Call Back:** Enable this flag to activate the Call Back on Trunk Port feature. By default, this flag is disabled on all trunk port types. By default, the flag is disabled.

  - **Call Back Timer:** This is the duration for which the system waits for the caller to disconnect before applying the Call Back. The range of this timer is from 01 to 99 seconds. By default, it is set to 10 seconds.

  - **Call Back Mode:** Select from the following options how a ‘Call Back’ call answered by the remote party should be routed:
    - Built-in Auto Attendant
    - PIN Authentication - Multiple Calls
    - CLI Authentication - Multiple Calls
    - CLI Authentication - Single Call - Answer Signaling
    - Operator

    By default, Operator is selected as the Call Back Mode.

  - **Call Back on:** This parameter allows you to select if the call back should be made to the same number that was received or to a different number. If you want the call back to be made to the same number select the ‘CLI number”. If you want the call back to be made to a different number, select ‘Alternate Number’.

    By default, CLI number is selected for Call Back.

  - **Incoming Number List:** Program the number strings that are eligible for Call Back in this List. By default, Number List 15 is assigned to Call Back Incoming Number List.

    Number List 15 is also assigned to all CO trunks as well as all other Trunk port types. If you want the same numbers strings to be programmed commonly for all CO trunks and Trunk Port types, retain this list.

    If you want a different set of number strings to be programmed for this CO Trunk, select a different Number List, and assign it to the CO trunk port.

    You may program the Incoming Number List either from the ‘Number List’ page or by clicking the ‘Incoming Number List’ link to reach the Number List page.

    Refer the topic “Number Lists” to know more, and for configuration instructions.

  - **Outgoing Number List:** Program the number strings that are to be called back in this List.
    For each number string you programmed in the ‘Incoming Number List’, you must program in the corresponding index in the Outgoing Number List a number to which the call back is to be made. For

example, for the number string programmed at Index 1 in the Incoming Number List, a corresponding number string at the same Index, Index 1, should be programmed in the ‘Outgoing Number List’.

By default, Number List 16 is assigned to Outgoing Number List.The same Number List 16 is also assigned to all CO trunks as well as all other Trunk port types.

You may program the default number list, or a different number list and assign it to this CO Trunk port.

You may program the Outgoing Number List either from the ‘Number List’ page or by clicking the ‘Outgoing Number List’ link to reach the Number List page.

Refer the topic “Number Lists” to know more, and for configuration instructions.

- **Call Back from:** This parameter determines the trunk port to be used to make the call back. The call back can be made using the Same Port or an Outgoing Trunk Bundle Group (OGTBG).

  Select ‘Same Port’ if you want the call back to be made using the same port on which the missed call is received. If you select OGTB Group, the call back will be made using the OGTB Group, which you have defined.

  By default, Same Port is selected.

- **OGTB Group:** If you selected OGTB Group for making the call back in the previous parameter, you must define the OGTB Group that must be used in this parameter.

  By default, OGTB Group 01 is assigned.

  If you want the system to select the lowest cost trunk for making the call back, enable Least Cost Routing on the OGTB Group that you define here for Call Back.

- **Allow Incoming CLI Modification:** Select the **Allow Incoming CLI Modification** check box if you want to apply Incoming CLI Modification on the CO trunk. By default, it is disabled.

  Incoming CLI Modification is useful in countries where the Calling Line Identification (CLI) received by the System extension users must be suitably modified before it can be used to dial out the number. To know more, see “Incoming CLI Modification”.

If you receive CLI in dialable format, there is no need to use this feature. In such case, keep the flag disabled.

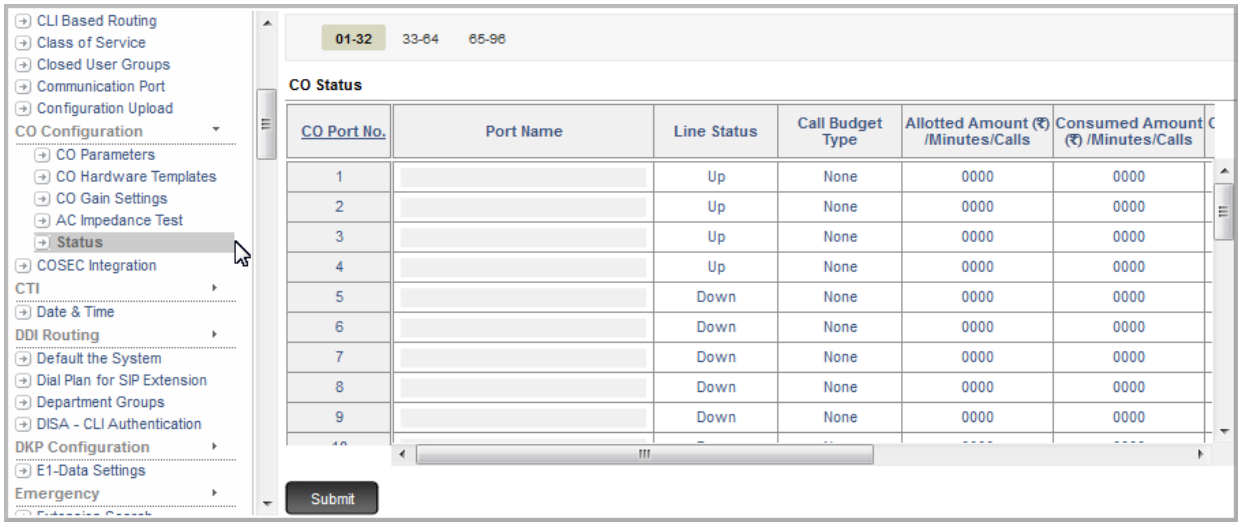
*For an incoming call on the CO trunk, the Incoming CLI Modification will be applied only when both — the **Allow Incoming CLI Modification** check box and the **Enable Incoming CLI Modification** check box in System Parameters — are enabled.*

- If you have completed configuration of all the above listed CO Parameters, click 'Submit' at the bottom of the page to save your changes.

Viewing CO Trunk Status

You can view the status of CO Trunks on Jeeves only. To do this,

- Under **CO Configuration**, click **Status**.



- For each CO Trunk, the following settings will be displayed:
  - CO Port Number
  - Port Name
  - Line Status
  - Call Budget Type
  - Allotted Amount/Minutes/Calls
  - Consumed Amount/Minutes/Calls
  - Call Budget Reset Mode
  - Call Budget Reset Scheduled (Date)
  - Reset Consumed (this is not a status indicator. It is for resetting the Consumed Call Budget manually)

*You can also view the CO Trunk Status from the **Status** link. To view, click the CO link under Status.*

Configuring CO Trunks using a Telephone

- Enter SE mode from a DKP/SLT.

  To assign Hardware Slot and Port to the CO Port, dial:
  - ***1104-CO-Slot-Port offset on the card***
    Where,
    CO is the Software port number of the CO port from 001 to 128.
    Slot is Slot number in which the CO is inserted from 01 to 16
    Port Offset is the number of the Port on the card from 01 to 99.

  To clear the Hardware ID assigned to a CO Port, dial:
  - ***1104-CO-00-00***

  To program a Name for a CO Port, dial:
  - ***5404-1-CO-Name-#**** to program a name for a single CO port.
  - ***5404-2-CO-CO-Name-#**** to program the same name for a range of CO ports.
  - ***5404-*-Name-#**** to program the same name for all CO ports.
    Where,
    CO is the Software port number of the CO port from 001 to 128.
    Name is a string of alphanumeric characters (maximum) 18 characters.
    Terminate the command with #* if the name string has fewer than 18 characters.

To clear the Name of a CO trunk, dial:

- ***5404-1-CO-#**** to clear the name of a single CO port.
- ***5404-2-CO-CO-#**** to clear the names for a range of CO ports.
- ***5404-*-#**** to clear the names of all CO ports.

To enable/disable the CO Trunk, dial:

- ***3307-1-CO-Flag*** to enable/disable a single CO port.
- ***3307-2-CO-CO-Flag*** to enable/disable a range of CO ports.
- ***3307-* -Flag*** to enable/disable all CO ports.
  Where,
  CO is the Software port number of the CO port from 001 to 128.
  Flag is
  0 for Disable
  1 for Enable
  Default: Enable

To assign a Trunk Feature Template to a CO Port, dial:

- ***5803-1-CO-Trunk Feature Template Number*** to assign a feature template to a single port.
- ***5803-2-CO-CO- Trunk Feature Template Number*** to assign the same template to a range of ports.
- ***5803-*- Trunk Feature Template Number*** to assign the same template to all ports.
  Where,
  CO is the Software port number of the CO port from 001 to 128.
  Trunk Feature Template number is from 01 to 50.
  Default: Template 01.

To assign a CO Hardware Template to a CO Port, dial:

- ***5903-1-CO-Hardware Template Number*** to clear the name of a single port.
- ***5903-2-CO-CO-Hardware Template Number*** to clear the names for a range of ports.
- ***5903-*-Hardware Template Number*** to clear the names of all ports.
  Where,
  CO is the Software port number of the CO port from 001 to 128.
  Hardware Template Number is CO Hardware Template from 01 to 50.
  Default: Template 01.

To assign a Cost Factor to a CO Port, dial:

- ***3308-1-CO-Cost Factor*** to assign cost factor for a single CO port.
- ***3308-2-CO- CO-Cost Factor*** to assign cost factor for a range of CO ports.
- ***3308-*-Cost Factor*** to assign cost factor for all CO ports.
  Where,
  CO is the Software port number of the CO port from 001 to 128.
  Cost Factor is from 01 to 99. Default: 01

- For commands to program Call Budget on CO trunks, refer the topic “Call Budget on Trunk”.

To enable / disable Call Back on CO port, dial:

- ***3310-1-CO-Code*** to enable/disable Call Back on a single CO trunk.
- ***3310-2-CO-CO-Code*** to enable/disable Call Back on a range of CO trunks.
- ***3310-*-Code*** to enable/disable Call Back on all CO trunks.
  Where,
  CO is the number of the software port of the CO trunk from 001 to 128.
  Code is
  0 for Disable
  1 for Enable

Default: Disabled

To program Call Back Timer on CO port, dial:

- ***3311-1-CO-Call Back Timer*** to set timer for a single CO trunk.
- ***3311-2 -CO-CO-Call Back Timer*** to set same timer duration for a range of CO trunks.
- ***3311-*-Call Back Timer*** to set the same timer duration for all CO trunks.
  Where,
  CO is the number of the software port of the CO trunk from 001 to 128.
  Timer is from 01 to 99 Sec.
  Default: 10 Seconds

To select Call Back Mode for CO port, dial:

- ***3312 -1-CO-Call Back Mode*** to set call back mode for a single CO trunk.
- ***3312 -2-CO-CO-Call Back Mode*** to set same call back mode for a range of CO trunks.
- ***3312-*-Call Back Mode*** to set the same call back mode for all CO trunks.
  Where,
  CO is the number of the software port of the CO trunk from 001 to 128.
  Call Back Mode is from 1 to 5
  1 for Built-in Auto Attendant
  2 for PIN Auth. - Multiple Calls
  3 for CLI Auth. - Multiple Calls
  4 for CLI Auth. - Single Call - Ans. Sig.
  5 for Operator
  Default: Operator

To program Call Back On for CO port, dial:

- ***3313-1-CO-Call Back on*** to program call back on for a single CO trunk.
- ***3313-2-CO-CO-Call Back on*** to program the same call back on option for a range of CO trunks.
- ***3313-*Call Back on*** to program the same call back on option for all CO trunks.
  Where,
  CO is the number of the software port of the CO trunk from 001 to 128.
  Call back on is
  1 for CLI Number
  2 for Alternate Number
  Default: CLI Number

To assign Call Back - Incoming Number List to a CO port, dial:

- ***3314-1 -CO-Incoming Number List*** to assign a list to a single CO trunk.
- ***3314-2-CO-CO-Incoming Number List*** to assign the same list to a range of CO trunks.
- ***3314-*-Incoming Number List*** to assign the same list to all CO trunks.
  Where,
  CO is the number of the software port of the CO trunk from 001 to 128.
  Incoming Number List is from 01 to 16.
  Default: 15

To assign Call Back - Outgoing Number List to a CO port, dial:

- ***3315-1 -CO-Outgoing Number List*** to assign a list to a single CO trunk.
- ***3315-2-CO-CO-Outgoing Number List*** to assign the same list to a range of CO trunks.
- ***3315-*-Outgoing Number List*** to assign the same list to all CO trunks.
  Where,
  CO is the number of the software port of the CO trunk from 001 to 128.
  Incoming Number List is from 01 to 16.
  Default: 16

To select Call Back From port for a CO Trunk port, dial:

- ***3316-1-CO-Call Back From*** to select Call Back From for a single CO trunk.
- ***3316-2-CO-CO-Call Back From*** to select the same Call Back From for a range of CO trunks.
- ***3316-*-Call Back From*** to select the same Call Back From for all CO trunks.
  Where,
  CO is the number of the software port of the CO trunk from 001 to 128.
  Call Back From is
  1 for Same Port
  2 for OGTB Group
  Default: Same Port

To assign Call Back - OGTB Group to a CO port, dial:

- ***3317-1-CO-OGTB Group*** to assign an OGTBG to a single CO trunk.
- ***3317-2-CO-CO-OGTB Group*** to assign the same OGTBG to a range of CO trunks.
- ***3317-*-OGTB Group*** to assign the same OGTBG to all CO trunks.
  Where,
  CO is the number of the software port of the CO trunk from 001 to 128.
  OGTB Group is from 01 to 32.
  Default: 01

- Exit SE mode.

Configuring E&M Lines

The SARVAM UCS supports E&M interface. A maximum of 32 ports are supported in SARVAM UCS.

If you have correctly installed the E&M Cards and observed the Reset Cycle, you may now program the E&M Ports using Jeeves or a Telephone, depending on your installation scenario.

An E&M port of SARVAM UCS can be programmed to take on the function of:

- a Subscriber (Station) - works like an extension interface, receiving incoming calls.

  OR

- a Trunk - works like a trunk interface when any of the extensions of the System makes an outgoing call through it.

  OR

- a Tie Line - takes on a dual personality: functioning as both a station and a trunk. The E&M port works like an extension interface for incoming calls. It works like a trunk interface when any extension makes an outgoing call through it.

  This dual function is used in Systems that are used as Transit Exchanges as in a PLCC Network. Read "PLCC-An Introduction" to know more.

The E&M Card of SARVAM UCS supports the following features:

- E&M Interface - Types IV and V
- Speech Interface - Two-wire and four-wire.
- E&M Trunk Seizure Type[216]: Immediate, Immediate with Ack, Immediate + Wink, Immediate with Ack+Wink (MFCR2)[217], Seizure Pulse, Seizure Pulse + Wink, Express, and Radio.
- Address Signaling: Pulse dial (Pulse 10PPS, Pulse 20PPS) and Tone Dial (DTMF).

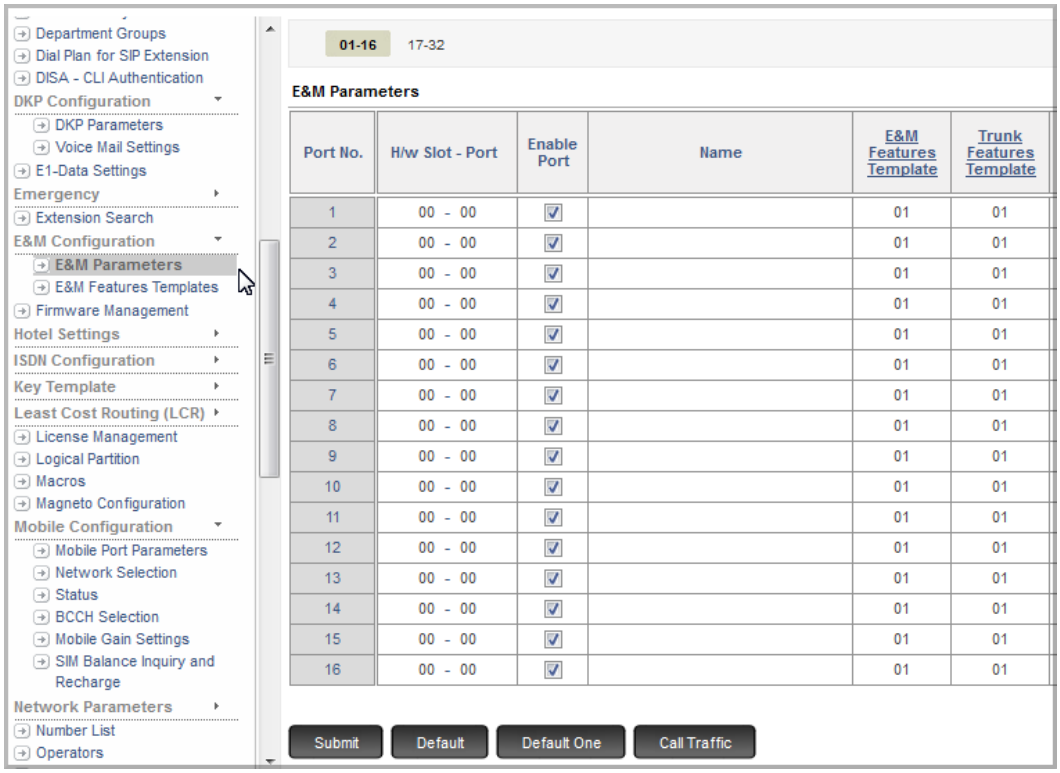
The E&M Interface (Type IV and Type V connection) and the Speech Interface (2-wire speech or 4-wire) are selected at the time of installation by changing the Jumper settings.

Configuring the E&M using Jeeves

- Log into Jeeves as System Engineer.
- Under **Configuration**, click **E&M Configuration**.

216. *This is the line protocol that defines how the equipment seizes the E&M trunk. Also referred to as Start Dial Supervision Signaling Protocol.*
217. *Currently supported only on ETERNITY GENX E&M4 Card.*

- Click the **E&M Parameters**.



Configure the following parameters for each E&M port on this page:

- **E&M Port No.:** This non-editable field is the software port number of the E&M Port. Refer the topic "Software Port and Hardware ID" to know more.

- **H/w (Hardware) Slot - Port:** 'Slot' is the number of the number of the Universal Slot in which the E&M Card is inserted. 'Port' is the number of the E&M hardware port on which the Tie Line equipment (System, Router, Leased Line, etc.) is connected.

  The SARVAM UCS can automatically detect and assign the hardware slot and port numbers automatically to the E&M software ports.

  For example: if you have inserted the E&M8 Card in Slot 05 and E&M4 Card in Slot 06 of SARVAM UCS, the system will assign the Hardware Slot 05 and port numbers 01-08 to the E&M Software Ports from 001 to 008 respectively. The system will assign hardware Slot 6 and port numbers 01-04 to the E&M Software Ports 009 to 012. Refer the topic "Software Port and Hardware ID" to know more.

  However, if required, you may change the Hardware Slot and Port assigned to the E&M software port. In this case, enter the desired Hardware Slot and Port number in this field.

  If you want to de-assign the Hardware Slot and Port, Enter '00' in both fields.

- **Enable Port:** This flag is for enabling or disabling an E&M port. When an E&M port is disabled, neither incoming nor outgoing calls can be made from that port.

  By default, the port is enabled. You may disable ports that are not functioning.

- **Name:** You may assign a Name to the E&M Port. Whenever there is an incoming call on this Port, this name will be displayed on the destination extension, while receiving the call.

  The Name may comprise a maximum of 18 characters.

- **E&M Feature Template:** This parameter is applicable to all E&M Ports. The E&M Feature Template is a complete set of E&M features to be applied on E&M Ports according to their 'Orientation Type', whether they are Stations, Trunks or Tie-Lines.

  By default E&M Feature Template 01 is applied on all E&M Ports. This template has 'Station' as the default Orientation Type.

  If all the E&M Ports are to be programmed as 'Stations' retain this template.

  If all the E&M Ports are to be programmed as 'Trunks' use the default E&M Feature Templates 09 and 10 which have 'Trunks' as Orientation Type.

  If some of the E&M Ports are to be programmed as Stations, some as Trunks and yet others as Tie Lines, prepare different E&M Feature Template for each Orientation Type and apply them to the related ports.

  - Under **Configuration**, click **E&M Feature Template** to open the page.
  - Select an E&M Feature Template Number.
  - Customize the E&M Feature Template.
  - Click **Submit** to save changes.
  - Go to the **E&M Parameters** page.
  - Apply the E&M Feature Template you customized to the E&M Port by entering the template number in the 'E&M Feature Template' field of this port.
  - Click **Submit** at the bottom of the page.
  - Repeat the same steps to customize another template and apply it to another E&M Port.

  Refer the topic “E&M Feature Template” for more details on customizing the templates and applying them on E&M Ports.

- **Trunk Feature Template:** This parameter is relevant only if the E&M Port is to be programmed to function as a Trunk or a Tie-Line[218]. To know more, refer “Trunk Feature Template”.

  Assign an E&M Feature Template to the E&M Port. A Trunk Feature Template is a set of features like Time Table, Operator, Auto Attendant, DISA, Trunk Auto Answer, Trunk Landing Group, SMDR Storage, etc., that defines the behavior of a Trunk. Apply a Trunk Feature Template to the E&M Port.

  By default, Trunk Feature Template 01 is applied on all E&M (trunk) Ports. This Template is also the default template commonly applied on all other trunk types (CO, ISDN BRI, ISDN T1E1PRI, GSM, and VoIP).

218. *To program the E&M Port as a Trunk or a Tie-Line, you must set the 'Orientation Type' of the E&M Port to 'Trunk' or 'Tie-Line' in the “E&M Feature Template” applied on the port.*

Click the link **Trunk Feature Template** to open the page. Check if the default Template 01 fulfills your requirement for the E&M Trunk port.

If not, you may prepare a different Trunk Feature Template and apply on all E&M Ports. For this,
If the default Template 01 does not fulfill your requirement,

You may prepare a different Trunk Feature Template and apply on all E&M Ports. For this,

- Click the **Trunk Feature Template** link.
- The Trunk Feature Template page will open.
- Prepare a new template, for example 04 to match your requirements.
- Click **Submit** at the bottom of the page to save changes.
- Go to the **E&M Parameters**.
- Go to the E&M Software Port Number you want to which you want to assign the Template you prepared.
- Enter the number of the Template you prepared (04) in the **Trunk Feature Template** field.
- Click **Submit** at the bottom of the page to save changes.

You may also prepare different Templates for different E&M Ports, for example Template 04 for certain ports, Template 05 for others. In this case, for each E&M Port, enter the number of the template you have prepared for that port.

To know more about customizing templates, refer the topic “Trunk Feature Template”.

- **Station Basic Feature Template:** This parameter is applicable on when the E&M Port is to be programmed to function as a Station (Orientation type = Station or Tie-Line).

  Assign a “Station Basic Feature Template” to the E&M Port functioning as a Station. By default, Station Basic Feature Template 01 is assigned to all extensions, that includes SLT and DKP ports.

  Check if the default template fulfills the feature requirements (like “Class of Service (COS)”, “Toll Control”, “OG Trunk Bundle Group”, etc.) of the E&M Ports functioning as Stations.

  If the default Template 01 fulfills the feature requirements and if the same features are to be allowed to all E&M (station) ports, retain Template 01.

  If different sets of features are to allowed to different E&M (station) Ports, then prepare separate Station Basic Feature Templates and apply them on the ports. To do this,

  - Click the link **Station Basic Feature Template** to open the page.
  - Select a Template number, for example 05.
  - Customize Template number 05 and click **Submit** at the bottom of the page.
  - Now go back to the **E&M Parameters** page.

- Enter the number of the Template you customized, Template 05 in the 'Station Basic Feature Template' field of the E&M Port (for example: E&M No. 003) on which you want to apply this template. If you want to apply this template to other ports too, like E&M No. 004, 005, and 006, assign the Template 05 to all these ports.

- Click **Submit** at the bottom of the page to save changes.

- Repeat the same steps to customize and assign a different Template to another E&M port.

Also, refer the topic "Station Basic Feature Template" to know more about customizing the templates and applying on the ports.

- **Station Advanced Feature Template:** This parameter is applicable only when the E&M Port is to be programmed to function as a Station.

  By default Station Advanced Feature Template 01 is assigned to all extensions, that includes SLT and DKP ports as well as E&M ports with the orientation type 'Station'.

  Check if this default template fulfills the feature requirements of the E&M Ports (with 'station' as orientation type) by selecting the 'Station Advanced Feature Template' link.

  If the default Template 01 fulfills the feature requirements, and if the same features are to be allowed to all E&M (station) ports, retain Template 01.

  If different sets of features are to be allowed to different E&M (station) Ports, then prepare separate Station Advanced Feature Templates and apply them on the ports.

  To do this,

  - Click the **Station Advanced Feature Template** link to open the page.

  - Select a Template number, for example 03.

  - Customize Template number 03.

  - Click **Submit** at the bottom of the page.

  - Go to the **E&M Parameters** page.

  - Enter the number of the Template you customized, Template 03 in the 'Station Basic Feature Template' field of the E&M Port (for example, E&M No. 003) on which you want to apply this template. If you want to apply this template to other ports too, like E&M No. 004, 005, and 006, assign the Template 03 to all these ports.

  - Click **Submit** at the bottom of the page to save changes.

  - Repeat the same steps to customize and assign a different Template to another E&M (Station) port.

  Also refer the topic "Station Advanced Feature Template" for instructions on customizing these templates and applying them on the ports.

- **Priority:** This parameter is applicable only when the E&M Port is to be programmed to function as a Station[219]. To know more, refer "E&M Feature Template".

Each station of the SARVAM UCS is assigned a Priority Level starting from 1, 2, 3... to 9, with '1' being lowest Priority and '9' being highest Priority. Whenever an extension (phone) with higher priority calls an extension with lower priority, a triple ring is placed on the called extension. To know more, read the feature description “Priority”.

By default, the Priority of all E&M Ports functioning as Stations is set to '9-Highest'. So, decide what Priority Level you will assign to each of the E&M Ports functioning as Stations and set the desired level for each port.

- **Cost Factor:** This parameter is of relevance only if 'Least Cost Routing' feature is applied on the E&M Trunk port.

  Cost Factor is a number assigned to each trunk for identification. This number also serves as a preference number for the trunk. The Cost Factor can be from 1 to 99. Trunks having the same preference must be assigned the same Cost Factor. Different trunk types can also be assigned the same Cost Factor. These trunks are used for routing calls.

  Assign a Cost Factor to the E&M Trunk port, for example, 03 and program Least Cost Routing Table accordingly.

  For example, if you want to route all outgoing calls starting with number '6' through the E&M Trunk Port 001 only,

  - You must first assign a Cost Factor (01-99) to E&M Trunk Port 001, for example 03.
  - Click the **Least Cost Routing - Number Based** link to open the page.
  - Enter '6' in the 'Number' column, Cost Factor '03' as Preference 1, 2, 3 and 4.
  - Click **Submit** at the bottom of the page to save your setting.

    All outgoing calls assigned Cost Factor trunk 03 will be made from E&M Trunk Port 001.

- **Allow Incoming CLI Modification:** Select the **Allow Incoming CLI Modification** check box if you want to apply Incoming CLI Modification on the E&M trunk. By default, it is disabled.

  Incoming CLI Modification is useful in countries where the Calling Line Identification (CLI) received by the System extension users must be suitably modified before it can be used to dial out the number. To know more, see “Incoming CLI Modification”.

If you receive CLI in dialable format, there is no need to use this feature. In such case, keep the flag disabled.

*For an incoming call on the E&M trunk, the Incoming CLI Modification will be applied only when both — the **Allow Incoming CLI Modification** check box and the **Enable Incoming CLI Modification** check box in System Parameters — are enabled.*

- Repeat the same steps to configure other E&M Trunk/Station ports.

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219. Recall that E&M ports can function as trunks, as stations and have both functions. When an E&M Port is programmed as a Station Interface, it can only receive incoming calls. To program the E&M Port as a Station, you must set the 'Orientation Type' of the E&M Port to 'Station' in the E&M Feature Template applied on the port.

- If you have completed configuration of all the above listed E&M Parameters, click 'Submit' at the bottom of the page to save your changes.

Configuring the E&M using a Telephone

- Enter SE mode from a DKP/SLT.

  To assign Hardware ID to an E&M Software Port, dial:
  - ***1105-E&M-Slot-Port offset on the card***
    Where,
    E&M is the Software Port number of the E&M port from 001 to 128.
    Slot is Slot number in which the E&M Card is inserted from 01 to 16
    Port Offset is the number of the E&M Port on the card from 01 to 99.

  To clear the Hardware ID assigned to an E&M Software Port, dial:
  - ***1105-E&M-00-00***

  To enable/disable an E&M Port, dial:
  - ***3321-1-E&M-Code*** to enable/disable a single E&M port.
  - ***3321-2-E&M-E&M-Code-#**** to enable/disable a range of E&M ports.
  - ***3321-*-Code*** to enable/disable all E&M ports.
    Where,
    E&M is the Software Port number of the port from 001 to 128.
    Code is
    0 for Disable
    1 for Enable
    Default: Enable

  To program a Name for an E&M Port, dial:
  - ***5406-1-E&M-Name-#**** to program a name for a single E&M port.
  - ***5406-2-E&M-E&M-Name-#**** to program the same name for a range of E&M ports.
  - ***5406-*-Name-#**** to program the same name for all E&M ports.
    Where,
    E&M is the Software Port number of the port from 001 to 128.
    Name is a string of alphanumeric characters 18 characters (maximum).
    Terminate the command with #* if the name string has fewer than 18 characters.

  To clear a name assigned to an E&M port, dial:
  - ***5406-1-E&M-#**** to clear the name of a single E&M port.
  - ***5406-2-E&M-E&M-#**** to clear the names for a range of E&M ports.
  - ***5406-*-#**** to clear the names of all E&M ports.

  To assign an E&M Feature Template to an E&M Port, dial:
  - ***6003-1-E&M-Template Number*** to assign a template to a single port.
  - ***6003-2-E&M-E&M-Template Number*** to assign the same template to a range of ports.
  - ***6003-*-Template Number*** to assign the same template to all ports.
    Where,
    E&M is the Software Port number of the port from 001 to 128.
    Template Number is from 01 to 50.
    Default: E&M Feature Template 01.

  To assign a Trunk Feature Template to an E&M Port, dial:
  - ***5805-1-E&M-Template Number*** to assign a template to a single port.

- ***5805-2-E&M-E&M-Template Number*** to assign the same template to a range of ports.
- ***5805-*-Template Number*** to assign the same template to all ports.
  Where,
  E&M is the Software Port number of the port from 001 to 128.
  Template Number is from 01 to 50.
  Default: Trunk Feature Template 01.

To assign a Station Basic Feature Template to an E&M Port, dial:
- ***5505-1-E&M-Template Number*** to assign a template to a single port.
- ***5505-2-E&M-E&M-Template Number*** to assign the same template to a range of ports.
- ***5505-*-Template Number*** to assign the same template to all ports.
  Where,
  E&M is the Software Port number of the port from 001 to 128.
  Template Number is from 01 to 50.
  Default: Station Basic Feature Template 01.

To assign a Station Advanced Feature Template to an E&M Port, dial:
- ***5605-1-E&M-Template Number*** to assign a template to a single port.
- ***5605-2-E&M-E&M-Template Number*** to assign the same template to a range of ports.
- ***5605-*-Template Number*** to assign the same template to all ports.
  Where,
  E&M is the Software Port number of the port from 001 to 128.
  Template Number is from 01 to 50.
  Default: Station Advanced Feature Template 01

To set a Priority Level for an E&M Port, dial:

This command is applicable only when the E&M Port is configured to function as a station!

- ***3915-1-E&M-Priority*** to set Priority for a single E&M Port.
- ***3915-2-E&M-E&M-Priority*** to set the same Priority for a range of E&M Ports.
- ***3915-*-Priority*** to set the same Priority for all E&M Ports.
  Where,
  E&M is the Software Port number of the port from 001 to 128.
  Priority is from 1 to 9.
  Default: 9-Highest

To assign a Cost Factor to an E&M port, dial:
- ***3322-1-E&M-Cost Factor*** to assign Cost Factor to a single E&M port.
- ***3322-2-E&M-E&M-Cost Factor*** to assign Cost Factor to a range of E&M ports.
- ***3322-*-Cost Factor*** to assign Cost Factor to all E&M ports.
  Where,
  E&M is the Software Port number of the port from 001 to 128.
  Cost Factor is from 01 to 99. Default: 01

- Exit SE mode.

Configuring LCR

Least Cost Routing (also referred to as Automatic Route Selection) is an expense control feature of SARVAM UCS.

Least Cost Routing (LCR) is useful when there are different trunk lines for making outgoing calls, and the service providers of these trunks offer different tariffs for calls made to certain locations or numbers or during a particular time of the day.

When a call is made from an extension of the SARVAM UCS, LCR recognizes where the call is going. Depending upon how the LCR is programmed, the system routes the call through the assigned trunks.

The system can be programmed to select the most cost effective trunk for the time of the day when the call is made from the extension or to select the most cost effective trunk for the destination number dialed from the extension or to select the most cost effective trunk considering both time of the day and destination number.

Accordingly, SARVAM UCS supports four types of LCR which can be programmed, namely:

1. **Time-based LCR:** This type of LCR may be used when you have trunk lines of more than one service provider, and each offers a different tariff according to the time of the day.

   For example, Service Provider 1 offers a lower tariff for calls made between 9am to 8pm, while Service Provider 2 offers a lower tariff for calls made between 8pm to 9am.

   When Time-based LCR is programmed, the system uses the Online-dialing logic, whereby digits dialed by the user are directly passed on to the trunk.

2. **Number based LCR:** This type of LCR may be used when you have trunk lines of more than one service provider, and each offers different tariffs according to the area or distance, or phone numbers dialed. For instance, Service Provider 1 provides cheaper calling rates for calls made from City A to City B, than Service Provider 2 and Service Provider 3.

3. **Time and Number based LCR:** This type of LCR is a combination of number and time based LCR, that is, the service providers offer different tariffs according to the time of the day as well as area/distance.

   For example, Service Provider 1 offers lower rates for calls made from City A to City B during peak hours 9am to 8pm, as compared to Service Provider 2, whereas Service Provider 2 offers cheaper rates for calls made from City A to City B during off peak hours (8pm to 9 am).

   When Time+Number-based LCR is programmed, the system uses Store and Forward dialing logic, whereby digits dialed by the user are first stored at a memory location in the system, and then dialed out on the assigned trunk.

4. **Service Provider-based LCR:** This type of LCR may be used when the same Service Providers offer different rates for calls made to numbers within their own network and for calls made to numbers of another Service Provider's network. For example, Service Provider 1 offers lower rates to call a Service Provider1 number in City A and in City B, than for calling numbers of Service Provider 2 in the same cities.

   This type of LCR may also be used when the same Service Providers apply different charges for different subscriber services provided by them. For example, Service Provider 1 offers both Fixed Line as well as GSM services and applies different charges for fixed line and GSM services.

When Service Provider-based LCR is programmed, whenever a number is dialed out, the system ignores the area code, checks the number in the 'Service Provider-based LCR table', and routes the call according to the trunk programmed for that number.

SARVAM UCS also supports LCR based on Carrier Pre-Selection. This type of LCR is useful where there exist different service providers for local and long distance calls. Refer the topic "Least Cost Routing-Carrier Pre-Selection" to know more.

Cost Factor

For LCR to work, all trunks that are allotted to extensions for making outgoing calls, must first be assigned a Cost Factor.

Cost Factor is a number assigned to each trunk for identification. This number also serves as a preference number for the trunk. The Cost Factor can be from 1 to 99. Trunks having the same preference must be assigned the same Cost Factor. Different trunk types can also be assigned the same Cost Factor. These trunks are used for routing calls.

By default all trunks are assigned Cost Factor number 01.

After assigning Cost Factor to Trunks, you must configure the Type of LCR to be used on Trunks in the Outgoing Trunk Bundle Group (OGTBG) allotted to the extensions for making calls.

Assigning Cost Factor to Trunks

- On a sheet of paper, make a list of all the trunks assigned to extensions for making outgoing calls.
- Make a table of the trunk types and assign a cost factor to each trunk type, as shown below.

Trunk Type and Number	Service Provider	Cost Factor
CO-001	BSNL	01
CO-002	BSNL	02
MOB-001	Reliance	03
MOB-002	BSNL	04
MOB-003	Airtel	05
MOB-004	Vodafone	06
BRI-01	BSNL	07
BRI-02	Reliance	08

- Program the Cost Factor number you assigned to the Trunk types in their respective trunk parameters. For instance, assign Cost Factor 01 to CO-001 and Cost Factor 002 to CO-002 in the "Configuring CO Trunks". Similarly, assign Cost Factor 03 to Mobile Trunk 001, Cost factor 04 to Mobile Trunk 002, Cost Factor 05 to Mobile Trunk port 003, and Cost Factor 06 to Mobile Trunk port 004 in the "Mobile Port Parameters". Assign Cost Factor 07 to BRI Trunk port 01 and Cost Factor 08 to BRI Trunk port 02 in "Configuring BRI Trunks".

Configuring Time-based LCR

- You can configure Time-based LCR for as many as 8 different Time Zones.
- On a sheet of paper, make a table for Time-based LCR.
- Define the Time Zone, that is, the start and end time, when the LCR should be applied for the outgoing calls. The Time Zone you define is stored at an Index number from 1 to 8.
- For each Time Zone that you define, select the Trunk as your first preference, that is, Preference 1. Select the trunk of your second, third and fourth preference. When the trunk you selected as first preference is busy, the system will route the call through the next trunk you have set that is free.
- Refer to the table you prepared for assigning Cost Factor to trunks.
- For example, you want calls made during 9am to 8pm to be routed through BSNL CO trunks (CO-001 and CO-002). If these trunks are busy, you want the system to route calls through the BRI line of BSNL trunk. When this line is busy, you want the system to attempt to route calls through the BRI line of Reliance.
- You want calls made between 8pm to 9am to be routed through BSNL CO trunk 001 only.
- At Time Zone Index 1, define the Time Zone start and end time in 24 Hours:Minutes format, enter the Cost Factor you assigned to CO-001 (01) and CO-002 (02) as Preference 1 and Preference 2 respectively. Enter the cost factor you assigned to BRI-01 (07) and BRI02 (08) as Preference 3 and Preference 4 respectively.

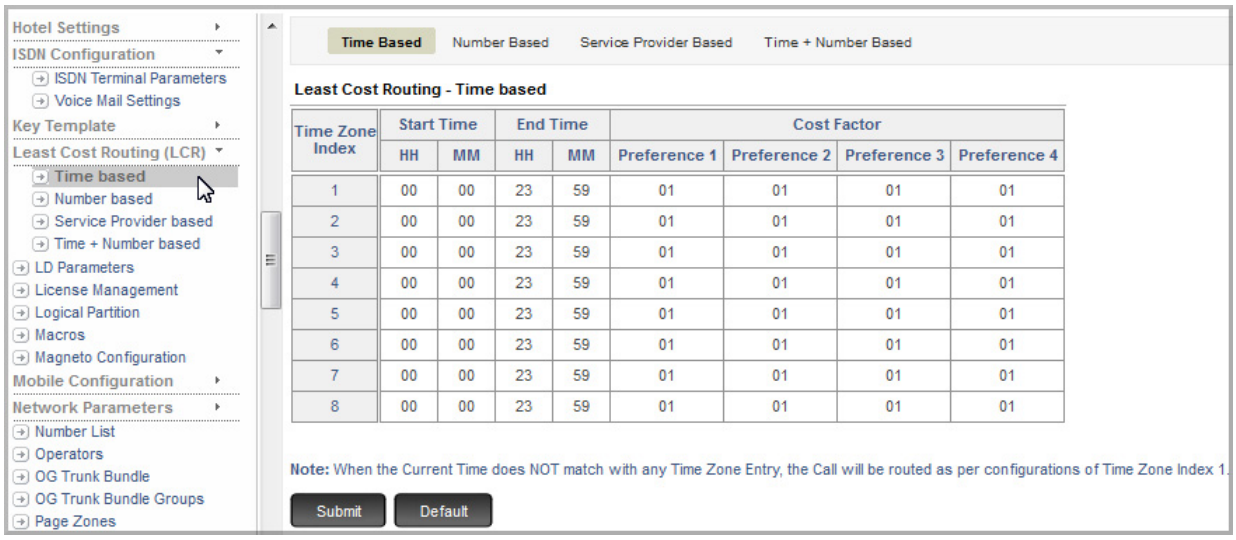
Time Zone Index	Time Zone		Cost Factor			
	Start Time (HH:MM)	End Time (HH:MM)	Preference 1	Preference 2	Preference 3	Preference 4
1	09:00	20:00	01	02	07	08
2	20:01	08:59	01	01	01	01
3						
4						
5						
6						
7						
8						

- Similarly, at Time Zone Index 2, define the Time Zone in 24 Hours: Minutes format. Enter the Cost factor you assigned to CO-001, that is, 01 as Preference 1, 2, 3, and 4. When calls are made during this time period, they will be routed through CO-001 only.
- If you have finished defining Time Zones and the preferred trunks for the time zones, configure the Time-based LCR using Jeeves or a Telephone.

Configuring Time-based LCR using Jeeves

- Log in as System Engineer.
- Under **Configuration**, click **Least Cost Routing (LCR)**.

  The links to the LCR options appear.
- Click **Time-based** to open the page.



- Enter the values of the Time-based LCR you prepared on the sheet of paper in the appropriate fields.
- Click **Submit** at the bottom of the page to save your settings.
- Log out of Jeeves or continue programming, as required.

Configuring Time-based LCR using a Telephone

- Enter SE mode.

  To program Time Zone at a Time Zone index, dial:
  - ***3402-Time Zone Index-Start Time-End Time***
    Where,
    Time Zone Index is from 01 to 08.
    Start Time is the time in HH:MM format when the Time zone starts.
    End Time is the time in HH:MM format when the Time zone ends.
    For example to program 09:00 to 20:00 hours at Time Zone Index 1, dial ***3402-01-0900-2000***

    By default, Time Zone is 00.00 to 23.59.

  To program the Cost Factor (Service Provider preference) for the Time Zone, dial:
  - ***3403-Time Zone Index-CF1-CF2-CF3-CF4***
    Where,
    Time Zone Index is from 01 to 08.
    CF1is the first preferred (the cheapest) service provider.
    CF2 is the second preferred (second cheapest) service provider.
    CF3 is the third preferred service provider.

CF4 is the fourth preferred service provider.

For example, to program the preferred trunks for the Time Zone 09:00 to 20:00 hours, dial ***3403-01-01-02-07-08***

It is mandatory to complete this command with CF1 to CF4. If you have only one service provider, program the same as CF1, CF2, CF3, CF4.

To default the Time Zone based LCR table, dial:
- ***3401***

- Exit SE mode.

Configuring Number-based LCR

- You can configure Number-based LCR for as many as 99 different Numbers, which are stored against Index numbers from 01 to 99.

- On a sheet of paper, make a table for Number-based LCR.

- Enter each of the number strings at an Index number from 01 to 99. A Number string may be a complete telephone number, a truncated phone number or an area code. The number can be a maximum of 64 characters (Digits + Wildcards). Valid characters: 0 to 9, *, #, X, T, Comma [,], Hyphen [-], Caret [^]. See "Wildcard Characters" to know the various number patterns you can use.

- For each number string you enter, select a Trunk as your first preference, that is, Preference 1. Select the trunk of your second, third and fourth preference. When the trunk you selected as first preference is busy, the system will route the call through the next trunk you have set that is free.

- Refer to the table you prepared for assigning Cost Factor to trunks.

  For example, you want all mobile numbers to be routed through the Mobile Trunk ports, all local numbers to be routed through the CO ports.

  All mobile numbers start with the number '9', which is prefixed with a '0' when making long distance mobile calls, so enter '9' and '09' as the number strings. For '9' as well as '09', select the Mobile trunks through which the calls should be made in order of preference.

  Similarly, all local numbers start with 2, so enter this number in the number string column, and select the CO trunk in the order of preference. As in this example, you have only two CO trunks, so you may keep the same two trunks as your preference.

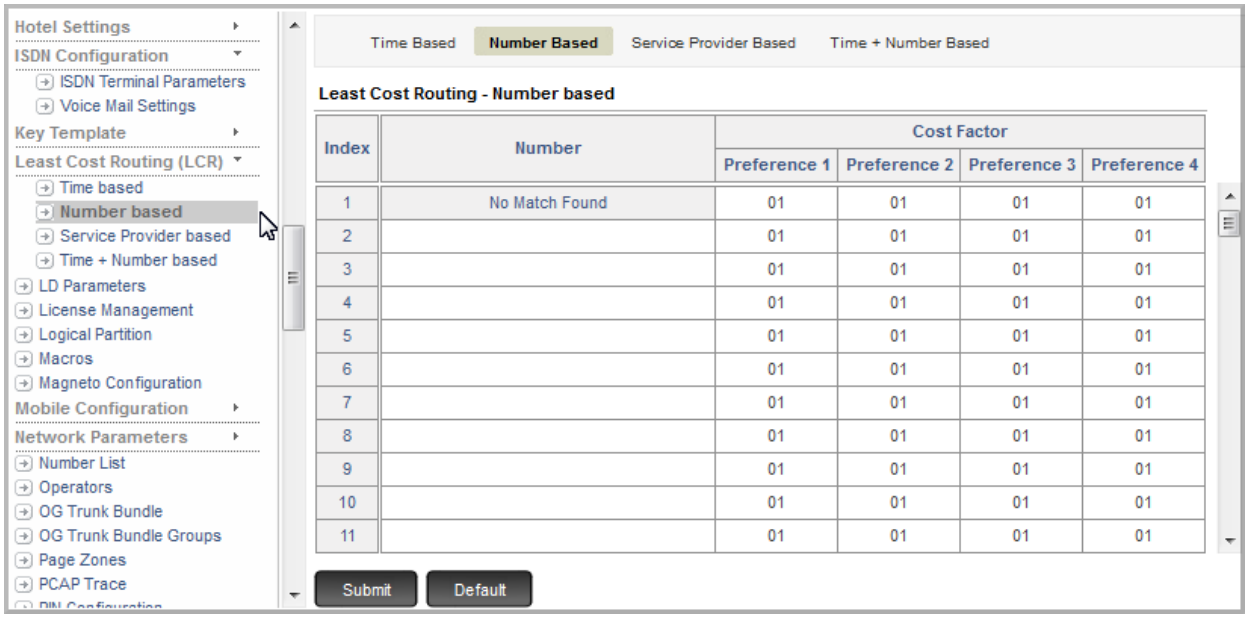
Index	Number	Cost Factor			
		Preference 1	Preference 2	Preference 3	Preference 4
1	9	04	03	05	06
2	09	04	06	05	03
3	2	01	02	01	02
4					
5					

Index	Number	Cost Factor			
		Preference 1	Preference 2	Preference 3	Preference 4
:					
:					
99					

- If you have finished entering the number strings, and selecting the preferred trunks for the numbers, configure the Number-based LCR using Jeeves or a Telephone.

Configuring Number-based LCR using Jeeves

- Log in as System Engineer.
- Under **Configuration**, click **Least Cost Routing (LCR)**.
- Click **Number-based** to open the page.



- Enter the values of the Number-based LCR you prepared on the sheet of paper in the appropriate fields.
- In Number you can enter may be a complete telephone number, a truncated phone number or an area code. You may enter upto 64 characters (Digits + "Wildcard Characters") in this field. Valid characters: 0 to 9, *, #, X, T, Comma [,], Hyphen [-], Caret [^]. Default: Blank.
- Click **Submit** at the bottom of the page to save your settings.

Wildcard Characters

SARVAM UCS supports following characters.

Character	Description
X (letter X)	X represents any single digit from 0 to 9.

#	When # is configured in a number string, it will not be considered as End of Dialing.
*****	When * is configured in a number string, it will not be considered as End of Dialing.
+	+ (plus) can be configured as a first character of the Destination Number string in the *SIP Trunk* only.
[-]	Hyphen within the bracket, defines a range. Only digits 0-9 are allowed within a bracket.
[,]	Comma within a bracket is used as a separator between the groups of numbers.
[^]	Caret within a bracket is used to deny or restrict the number or range defined after the symbol. Only digits 0-9 are allowed after the caret.
T (letter T)	Character T can be configured only as a last character in a number string. When configured in a number string, the system waits for End of Dialing.

Refer the following table to understand how you can configure the Numbers.

Numbers	Description
1XX	Allows you to dial any number in a range from 100 to 199.
[2-5]XX	Allows you to dial any 3 digit number in a range from 200-599.
[2,3,8]XX	Allows you to dial any 3 digit number in the range from 200-299, 300-399, 800-899.
[2-9]XXXXXX	Allows you to dial any 7 digit number in the range from 2000000-9999999.
23[^2]1	Allows you to dial a 4 digit number: 2301, 2311, 2331, 2341, 2351, 2361, 2371, 2381, 2391.
2630[500-550]	Allows you to dial a 7 digit number in the range from 2630500-2630550.
[^6-7]X	Allows you to dial a 2 digit number in the range from 00 to 99 except the numbers from 60 to 79.
1234	Allows you to dial 1234 number only.
011T	Allows you to dial any number starting with 011. The number must be of minimum 3 digits and maximum digits must be as configured for the port.

Configuring Number-based LCR using a Telephone

- Enter SE mode.

  To program Cost Factor (Service Provider preference) for the each Number, dial:
  - ***3412-Number Index-CF1-CF2-CF3-CF4***
    Where,
    Number Index is from 01 to 99.
    CF1is the first preferred (the cheapest) service provider.
    CF2 is the second preferred (second cheapest) service provider.
    CF3 is the third preferred service provider.
    CF4 is the fourth preferred service provider.

    For example, to program Cost Factor for number '9' at Index 01, dial 3412-01-04-03-05-06

It is mandatory to complete this command with CF1 to CF4. If you have only one service provider, program the same as CF1, CF2, CF3, CF4.

To default the Number-based LCR table, dial:

- ***3410***

Configuring Time and Number-based LCR

- This is a combination of the Time Zone-based and Number-based LCR. You may use this feature if your service providers offer lower call rates for calls made to certain numbers during a certain time of the day.
- On a sheet of paper, make a table for Time and Number-based LCR.
- Define the Time Zones when the service providers offer lower tariff. You can define up to 8 time zones.
- For each Time Zone you define, specify the Number strings on which lower tariff is applied during that Time Zone. The number can be a maximum of 64 characters (Digits + Wildcards). Valid characters: 0 to 9, *, #, X, T, Comma [,], Hyphen [-], Caret [^]. See “Wildcard Characters” to know the various number patterns you can use.
- For each Number string you enter for a particular time zone, assign Cost Factor. Select a trunk as your first preference. Select trunks of your second, third and fourth preference. Refer to the table you prepared for assigning Cost Factor.
- You can enter up to 99 different number strings, which are stored at Index numbers from 01 to 99. The Number strings may be complete telephone numbers, truncated phone numbers or area codes.

  When the trunk you selected as first preference is busy, the system will route the call through the next trunk you set as preference if it is free.

  For example, service provider of CO-001 and CO-002 (assigned Cost Factor 01 and 02) offers the lowest rate for calls made to Area Code 022 between 8am to 12pm, followed by service providers of Mobile Trunk-02 (assigned cost factor 04) and Mobile Trunk-01 (assigned cost factor 03).
- Define Time Zone 1 Start and End time as 08 to 12:00 hours
- Enter area code 022 as Number string at Number Index 1

- Assign Cost Factor preference for the number string in this sequence: 01, 02, 04, 03

		Time Zone1				Time Zone2				Time Zone3				:	Time Zone 8
		HH		MM		HH		MM		HH		MM			
Start Time		08		00		12		00		09		00			
End Time		12		00		18		00		20		00			
		Cost Factor				Cost Factor				Cost Factor					
Index	Number	Preference 1	Preference 2	Preference 3	Preference 4	Preference 1	Preference 2	Preference 3	Preference 4	Preference 1	Preference 2	Preference 3	Preference 4		
1	022	01	02	04	03	04	03	02	01	03	05	06	04		
2	011	01	02	04	05										
3	080	01	02	05	06					03	05	06	04		
:															
99															

- If you have finished defining the time zones, entering the number strings, and selecting the preferred trunks for the number strings, configure the Number and Time-based LCR using Jeeves or a Telephone.

Configuring Time+Number-based LCR using Jeeves

- Log in as System Engineer.
- Under **Configuration**, click **Least Cost Routing (LCR)**.
- Click **Time + Number based** to open the page.

- Enter the values of the Time+Number-based LCR you prepared on the sheet of paper in the appropriate fields.

- Click **Submit** at the bottom of the page to save your settings.

Configuring Time+Number-based LCR using a Telephone

- Enter SE mode.

  To define Time Zone for Time+Number-based LCR, dial:
  - ***3421-Time Zone Index-Start Time-End Time***
    Where,
    Time Zone Index is from 01 to 08.
    Start Time is the time in HH:MM format when the Time zone starts.
    End Time is the time in HH:MM format when the Time zone ends.
    By default, Time Zone is 00.00 to 23.59.

    For example, to define 08:00 to 12:00 as start and end time of Time Zone 1, dial ***3421-1-0800-1200***

  To program Cost Factor (Service Provider preference) for the each Number and Time Zone, dial:
  - ***3423-Number Index-Time Zone Index-CF1-CF2-CF3-CF4***
    Where,
    Number Index is from 01 to 99.
    Time Zone Index is from 01 to 08.
    CF1is the first preferred (the cheapest) service provider.
    CF2 is the second preferred (second cheapest) service provider.
    CF3 is the third preferred service provider.
    CF4 is the fourth preferred service provider.

    For example, to program Cost factor for Area code 022 during time zone 08 to 12:00 hours as entered in the sample table, dial ***3423-01-1-01-02-04-03***

  To default the Time and Number-based LCR table, dial:
  - ***3420***

Configuring Service Provider-based LCR

- In Service Provider-based LCR, whenever a number is dialed out, the system ignores the area code, and starts checking the numbers in the 'Service Provider-based LCR table' and routes the call according to the trunk programmed for that number. For this, you must program the two parameters Area Code and Ignore Digit Count in the Area Code Table.

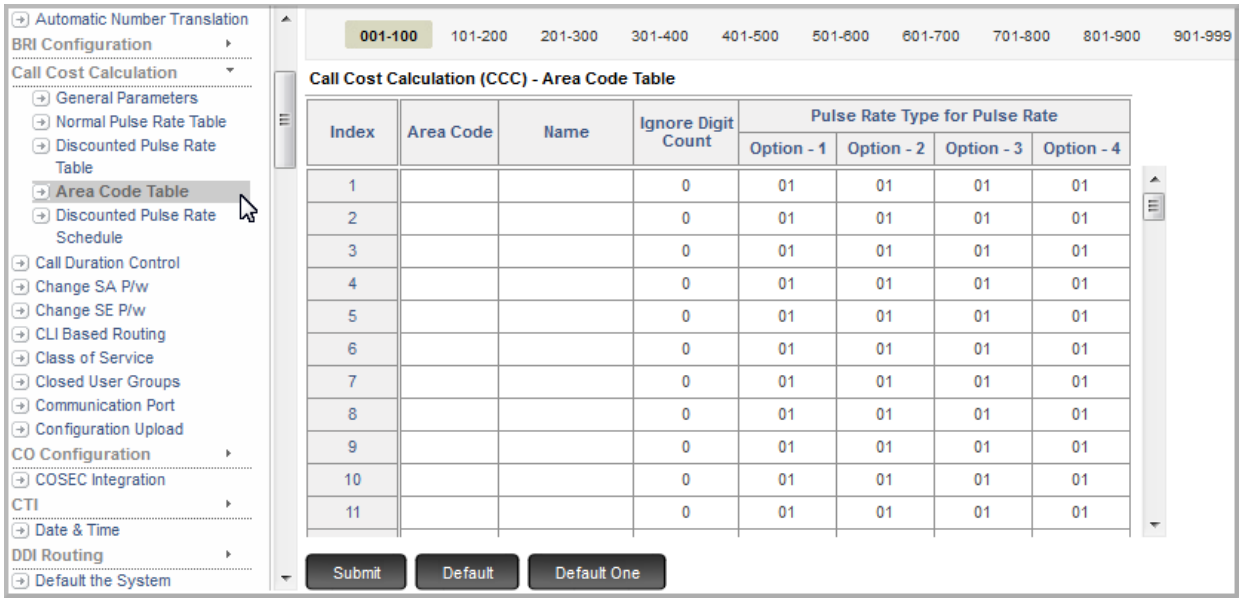
- On a sheet of paper make a table for Service Provider-based LCR.

Index No.	Number	Area Code	Ignore Digit Count	Cost Factor			
				Preference 1	Preference 2	Preference 3	Preference 4
01	3	080	3	08	07	01	02
02	6	022	3	07	01	02	08
:	:		:				
99	2	03852	5	01	02	01	02

- As you can see, the Service Provider-based LCR Table is similar to the Number-based LCR table.
- You can program as many as 99 different numbers which are stored against Index numbers from 01 to 99.
- The number strings may be the complete telephone number, a truncated phone number or the first digit of the phone number. The number can be a maximum of 64 characters (Digits + Wildcards). Valid characters: 0 to 9, *, #, X, T, Comma [,], Hyphen [-], Caret [^]. See "Wildcard Characters" to know the various number patterns you can use.
- For each number string that you enter against an Index number, you must also specify the Area Code and the Ignore Digit Count.
- The Ignore Digit Count is the number of digits in the area code that the system should ignore before checking the Service Provider-based LCR table. For each area code that you enter, the corresponding Ignore Digit Count will be the number of digits in the area code. For example, the area code for the number starting with '3' is 080, which consists of 3 digits. So, the Ignore Digit Count for the number/area code 080 will be 3.
- For each number string and area code that you enter, assign the Trunk of the service provider that you prefer as your first, second, third and forth preference for dialing that number/area code. Refer the table you prepared for assigning Cost Factor to trunks.
- If you have finished entering the number strings, their corresponding area codes and the Ignore Digit Count, and the preferred trunks, configure Service Provider-based LCR using Jeeves or a Telephone.

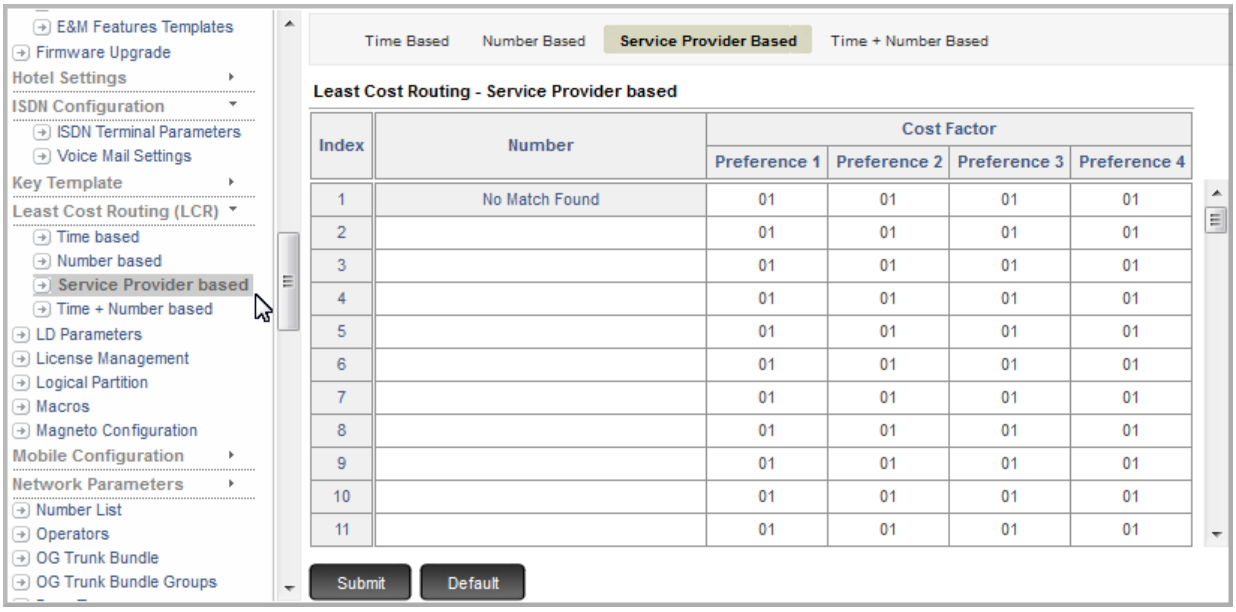
Configuring Service Provider-based LCR using Jeeves

- Log in as System Engineer.
- To program Area Code and Ignore Digit Count, Under **Configuration**, click **Call Cost Calculation**.
- Click **Area Code Table** to open the page.



Each area code is stored at an index number.

- Enter the Area Codes and the corresponding Ignore Digit Counts from the sheet you prepared for Service Provider based-LCR. You may also enter the respective name for each area code, if desired.
- Click **Submit** at the bottom of the page to save your changes.
- Now, click the **Least Cost Routing (LCR)**. The links for the LCR options appear.
- Click **Service Provider based** to open the page.



- Enter the values of the Service Provider-based LCR you prepared on the sheet of paper in the appropriate fields.
- Click **Submit** at the bottom of the page to save your settings.

Configuring Service Provider-based LCR using a Telephone

- Enter SE mode.

  To program Area Code in the Area Code Table, dial:
  - ***2620-Area Code Index-Area Code-#****
    Where,
    Area Code Index is from 001 to 999.
    Area Code is a string of maximum 7 digits. Terminate the command with #* if the number string has fewer than 7 digits.
    By default, Area Code is 'Blank'.

    For example, to program Number string '080' at Number Index 001, dial ***2620-001-080-#****

  To clear an Area Code in the Area Code Table, dial:
  - ***2620-Area Code Index-#****

  To program Ignore Digit Count for an Area Code, dial:
  - ***2623-Area Code Index-Ignore Digit Count***
    Where,
    Area Code Index is from 001 to 999.

Ignore Digit Count is from 0 to 9.
By default, Ignore Digit Count is '0'.

Refer the topic "Area Code Table" under Call Cost Calculation to know more.

To program Cost Factor (Service Provider preference) for the each Number, dial:

- ***3442-Number Index-CF1-CF2-CF3-CF4***
  Where,
  Number Index is from 01 to 99.
  CF1is the first preferred (the cheapest) service provider.
  CF2 is the second preferred (second cheapest) service provider.
  CF3 is the third preferred service provider.
  CF4 is the fourth preferred service provider.

  For example, to program Cost Factor for number '3' at Index 01, dial ***3412-01-08-07-01-02***

It is mandatory to complete this command with CF1 to CF4. If you have only one service provider, program the same as CF1, CF2, CF3, CF4.

To default the Number-based LCR table, dial:

- ***3440***

- Exit SE mode.

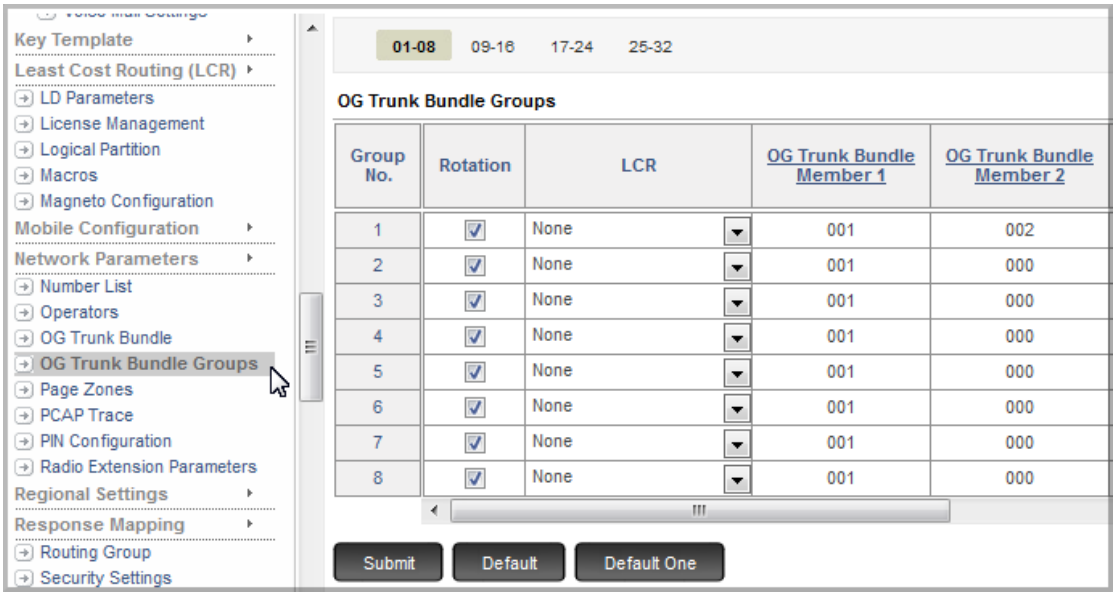
Configuring LCR Type on Trunks

After assigning Cost Factor to Trunks and programming the LCR Type - Time-based, Number-based, Time and Number-based, Service Provider-based - you must now apply the desired LCR Type on the Outgoing Trunk Bundle Group (OGTBG) allotted to the extensions. This can be done using Jeeves or a Telephone.

Configuring LCR Type in OGTB using Jeeves

- Log in as System Engineer.

- Under **Configuration**, click **Outgoing Trunk Bundle Group**.



- For each OGTBG number assigned to extensions, select the desired LCR Type: Time-based, Number-based, Time+Number based, Service Provider-based (Cost Factor).

- Click **Submit** to save your settings.

You can find the OGTBG number assigned to each extension from the Station Basic Feature Template assigned to the extension.

Configuring LCR Type in OGTB using a Telephone

- Enter SE mode.

  To select LCR type in OG Trunk Bundle Group, dial:
  - ***1404-1-OGTBG-LCR Type*** to select an LCR Type for a single group.
  - ***1404-2-OGTBG-OGTBG-LCR Type*** to select the same LCR Type for a range of OGTB groups.
  - ***1404-*-LCR Type*** to select the same LCR Type for all OGTB groups.

  Where,
  OGTBG is from 01 to 32.
  LCR Type is
  0 for No LCR
  1 for Time zone-based LCR
  2 for Number-based LCR
  3 for Time+Number-based LCR
  4 for Service Provider-based (Cost Factor) LCR
  By default, 'No LCR' is selected as LCR Type.

- Exit SE mode.

Configuring Emergency Number Dialing

If you are using the system in the Hospitality mode, to dial the Emergency Number 911, you must purchase the E911 license. For details, refer "License Management".

SARVAM UCS supports dialing of Emergency Numbers immediately without any blocking.

The system will disregard features such as Toll Control, Call Budget, Automatic Number Translation, Call Duration Control on the extensions when dialing out Emergency Numbers. To know more, refer the topic "Emergency Dialing".

For Emergency Number Dialing, you need to configure the Emergency Number Table. In this table, each Emergency Number is to be assigned an Outgoing Trunk Bundle Groups (OGTB) through which the Emergency Number is to be routed.

The system loads the default Emergency Numbers and Outgoing Trunk Bundle Group (OGTBG) in the Emergency Number Table as per the "Region" you selected for the system. The Emergency Numbers loaded by default in the Emergency Number Table are non-editable, but you can re-assign the default OGTB, as per your requirement.

The OGTBG, by default do not have any OG Trunk Bundles assigned. Make sure these are assigned, as only then emergency calls will be routed through the OGTBG.

To use the feature "Emergency Calls (911) - Reporting to PSAP", make sure you assign only T1 PRI lines in the outgoing trunk bundle group for dialing the number 911.

The default Emergency table as per the Region you select is given below:

Feature	Emergency Number 1	Emergency Number 2	Emergency Number 3	Emergency Number 4	Emergency Number 5	Emergency Number 6	Emergency Number 7	Emergency Number 8	Emergency Number 9	Emergency Number 10
Australia	000	106	112							
OGTB Group Number	32	30	31	1	1	1	1	1	1	1
Banglade sh	999									
OGTB Group Number	1	1	1	1	1	1	1	1	1	1
Belgium	101	100	112							
OGTB Group Number	1	1	1	1	1	1	1	1	1	1
Bhutan	110	112	113							
OGTB Group Number	1	1	1	1	1	1	1	1	1	1
Canada	911									

Feature	Emergency Number 1	Emergency Number 2	Emergency Number 3	Emergency Number 4	Emergency Number 5	Emergency Number 6	Emergency Number 7	Emergency Number 8	Emergency Number 9	Emergency Number 10
OGTB Group Number	32	1	1	1	1	1	1	1	1	1
China	110	120	119							
OGTB Group Number	1	1	1	1	1	1	1	1	1	1
Germany	110	112								
OGTB Group Number	1	1	1	1	1	1	1	1	1	1
India										
OGTB Group Number	1	1	1	1	1	1	1	1	1	1
Indonesia	110	118	119	113	112					
OGTB Group Number	1	1	1	1	1	1	1	1	1	1
Italy	112									
OGTB Group Number	1	1	1	1	1	1	1	1	1	1
Jordan	191	199								
OGTB Group Number	1	1	1	1	1	1	1	1	1	1
Kazakhstan	03									
OGTB Group Number	1	1	1	1	1	1	1	1	1	1
Kenya	999									
OGTB Group Number	1	1	1	1	1	1	1	1	1	1
Kuwait	777									
OGTB Group Number	1	1	1	1	1	1	1	1	1	1
Malaysia	999	112								

Feature	Emergency Number 1	Emergency Number 2	Emergency Number 3	Emergency Number 4	Emergency Number 5	Emergency Number 6	Emergency Number 7	Emergency Number 8	Emergency Number 9	Emergency Number 10
OGTB Group Number	1	1	1	1	1	1	1	1	1	1
Maldives	102	108								
OGTB Group Number	1	1	1	1	1	1	1	1	1	1
Mauritius	999	115	144							
OGTB Group Number	1	1	1	1	1	1	1	1	1	1
Mexico	911									
OGTB Group Number	1	1	1	1	1	1	1	1	1	1
Namibia	911									
OGTB Group Number	1	1	1	1	1	1	1	1	1	1
Nepal	100									
OGTB Group Number	1	1	1	1	1	1	1	1	1	1
New Zealand	111	112	911	08						
OGTB Group Number	1	1	1	1	1	1	1	1	1	1
Oman	9999									
OGTB Group Number	1	1	1	1	1	1	1	1	1	1
Pakistan	15	115	16	911	112					
OGTB Group Number	1	1	1	1	1	1	1	1	1	1
Philippines	117	911	112							
OGTB Group Number	1	1	1	1	1	1	1	1	1	1
Poland	997	999	998	112						

Feature	Emergency Number 1	Emergency Number 2	Emergency Number 3	Emergency Number 4	Emergency Number 5	Emergency Number 6	Emergency Number 7	Emergency Number 8	Emergency Number 9	Emergency Number 10
OGTB Group Number	1	1	1	1	1	1	1	1	1	1
Russia	02	03	01	112						
OGTB Group Number	1	1	1	1	1	1	1	1	1	1
Singapore	999	995	112	911						
OGTB Group Number	1	1	1	1	1	1	1	1	1	1
South Africa	10111	10177	1022							
OGTB Group Number	1	1	1	1	1	1	1	1	1	1
Spain	091	061	080	085	112					
OGTB Group Number	1	1	1	1	1	1	1	1	1	1
Sri Lanka	119	110	111							
OGTB Group Number	1	1	1	1	1	1	1	1	1	1
Sudan	110	112	113							
OGTB Group Number	1	1	1	1	1	1	1	1	1	1
Sweden	112									
OGTB Group Number	1	1	1	1	1	1	1	1	1	1
Taiwan	110	119								
OGTB Group Number	1	1	1	1	1	1	1	1	1	1
Thailand	191	1669	199							
OGTB Group Number	1	1	1	1	1	1	1	1	1	1
Turkey	155	112	110							

Feature	Emergency Number 1	Emergency Number 2	Emergency Number 3	Emergency Number 4	Emergency Number 5	Emergency Number 6	Emergency Number 7	Emergency Number 8	Emergency Number 9	Emergency Number 10
OGTB Group Number	1	1	1	1	1	1	1	1	1	1
UAE	999	998	997	112						
OGTB Group Number	1	1	1	1	1	1	1	1	1	1
UK	999	112								
OGTB Group Number	1	1	1	1	1	1	1	1	1	1
USA	911									
OGTB Group Number	32	1	1	1	1	1	1	1	1	1

For example, if you selected USA as Region, the default Emergency Number Table would look like this:

Default Emergency Number Table for USA

Index	Emergency Number	OG Trunk Bundle Group
01	911	32
02		01
03		01
04		01
05		01
06		01
07		01
08		01
09		01
10		01

If you selected Australia as Region, the default Emergency Number Table would look like this:

Default Emergency Number Table for Australia

Index	Emergency Numbers	OG Trunk Bundle Group
01	000	32
02	106	30
03	112	31
04		01

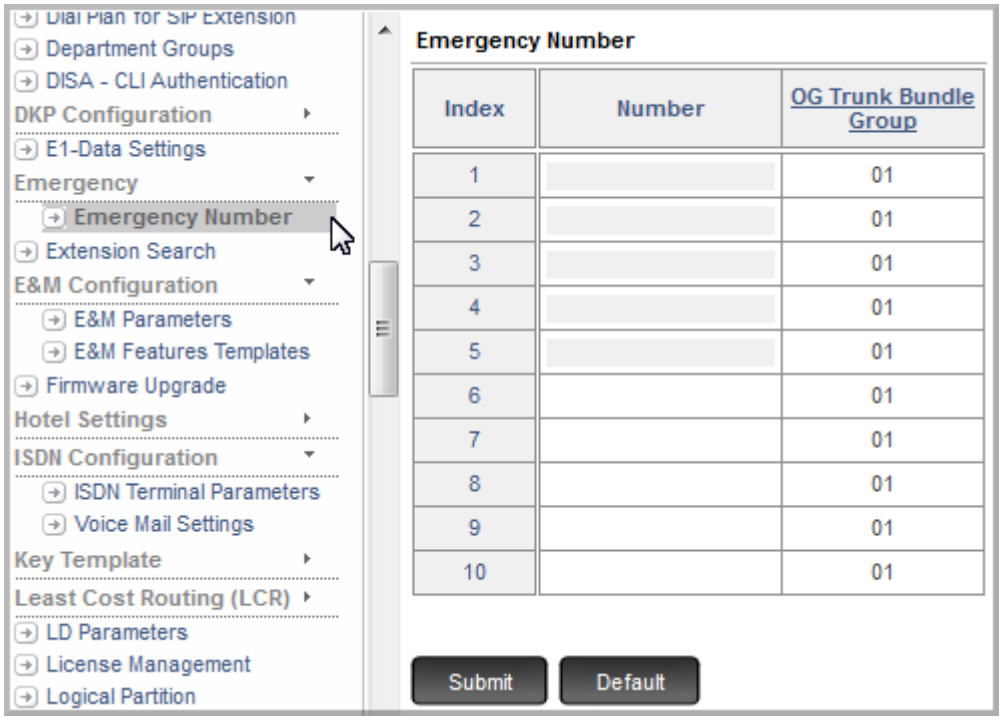
Index	Emergency Numbers	OG Trunk Bundle Group
05		01
06		01
07		01
08		01
09		01
10		01

Each Number is stored at an index from 01 to 10. The Emergency Number fields from index 01 to 5 are non-editable, but you can select a different OGTBG for each of these default Emergency Numbers.

You can add Emergency Numbers at index 06 to 10 in the table, and select the OGTBG as required.

Configuring Emergency Number Table using Jeeves

- Log in as System Engineer.
- After you have selected the **Region.**
- Under **Configuration**, click **Emergency.**
- Click **Emergency Number** to open the page.



Click the **Default** button, the system will upload the default Emergency Numbers as per the Region you have selected.

*If you upgrade the firmware make sure you click **Default**. The system will upload the default Emergency Numbers as per the Region.*

The first five entries, at Index 01 to 05 on this table, are uneditable. These fields will be populated with the default Emergency Numbers of your country (which you selected as *Region*).

If the Emergency Numbers loaded by default are applicable for your region/country, all you need to do is re-assign, if required, the **Outgoing Trunk Bundle Group** assigned by default to each number.

If the Emergency Numbers loaded by default are not applicable for your region/country, you may add the Emergency Numbers and their OGTBG at index 06 to 10 in this table.

Make sure that the trunks configured in the OGTBG for each Emergency Number belong to the correct network and the ports through which the calls are to be routed are not disabled. For example, '112' is the default Emergency Number for the mobile network. So, make sure that the Mobile Trunk to be used for dialing this number is included in the OGTBG you assign to this number in the Table.

- Click **Submit** to save changes.

Configuring Emergency Number Table using a Telephone

- Enter SE mode from DKP/SLT.

  To add Emergency Numbers in the table, dial:
  - ***3116-Index-Emergency Number-#****
    Where,
    Index is the location at which the Emergency Number is stored in the table, from 06 to 10 (01 to 05 are reserved for default numbers).
    Emergency Number is a string of 6 digits max. Use digits 0-9 only.

  To assign an OG Trunk Bundle Group (OTBG) to an emergency number, dial:
  - ***3117-Index-OG Trunk Bundle Group***
    Where,
    Index is the location at which the Emergency Number is stored in the table, from 01 to 10.
    OGTBG is from 01 to 32.

- Exit SE mode.

CHAPTER 11

Configuring Voice Mail System

Before you begin configuration of the VMS related parameters, consider the following points:

- Make sure you have installed the VMS module correctly and observed the Reset Cycle. Refer the topic "Installing the VMS Module" for instructions on installing the VMS module on the CPU Card. Also refer the topic "VMS Channels" for license related information.

- The number of VMS channels will be considered as the maximum channels available for reservation for Voice Mail Auto-Attendant parameter.

  - The channel reserved for Voice Mail Auto-Attendant configuration will not be changed if the number of channels configured in *Channel Reserved for Voice Mail Auto Attendant* parameter is increased.

  - The channels reserved for Voice Mail Auto-Attendant configuration will be changed to the number of VMS channels if the number of VMS channels available are less than the number of channels configured in *Channel Reserved for Voice Mail Auto Attendant* parameter. To know more, refer "Configuring VMS General Parameters".

- Decide which extension users are to be provided voice mail. Make a list of these extensions by their port type and (software) port number, access codes. Configure the VMS parameters for these extensions. See "Extension Voice Mail Settings".

- If you intend to use the VMS Auto Attendant on trunks,
  - make a list of the trunks by their port type (SIP, T1E1 PRI, BRI, Mobile, CO and E&M) and (software) port number.
  - configure welcome and greeting messages. You may record custom welcome messages that meet your requirements. Refer to "Voice Mail Auto-Attendant Menu".
  - configure Voice Mail Auto Attendant (VMAA) Menu for the time zones in the Trunk Feature Template(s) of the trunks on which you want to use the VMS Auto Attendant. Refer to "Trunk Feature Template".
  - assign the desired Voice Mail Auto Attendant (VMAA) Menu to the trunks in their Trunk Feature Template.

- The prompts used to route the call using the Voice Mail Auto Attendant can be customized as per your requirement. Refer to "Prompts Management".

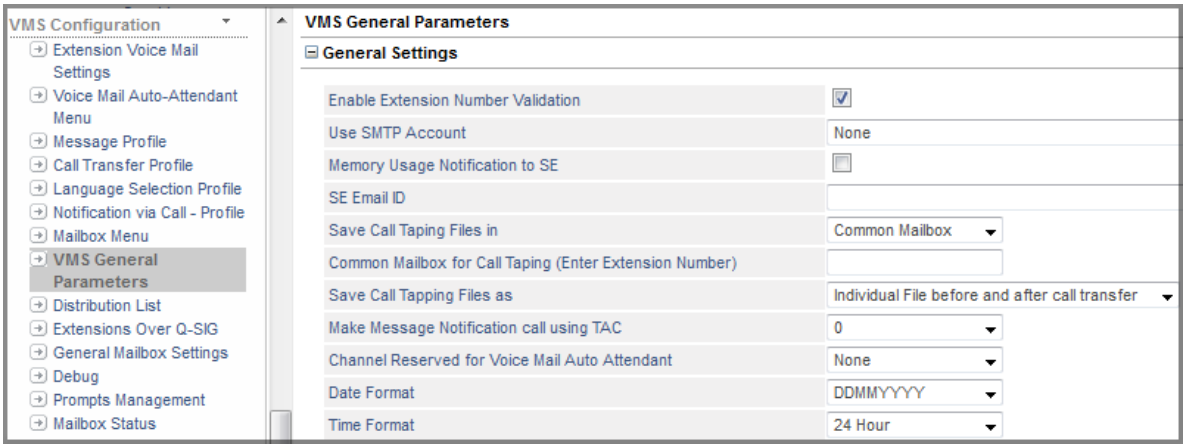
Configuring VMS General Parameters

VMS General Parameters allows you to customize VMS settings as per your requirement. You can:

- configure the VMS General Settings
- set the default language
- configure the Play Settings
- define the Filename format
- customize Voice Mail Notification Messages

Configuring VMS General Parameters using Jeeves

- Log in as System Engineer.
- Under **Configuration**, click **VMS Configuration**.
- Click **VMS General Parameters**.



- Check the default values of the following parameters, and change them, if required, to the desired values.

General Settings

- **Enable Extension Number Validation:** The VMS Auto Attendant allows callers to directly reach the desired party in an organization, by giving them the option of dialing the extension number.

  When the Extension Number Validation flag is enabled, the VMS compares the extension number dialed by the external caller with the extension numbers configured in the system. If no match is found, the VMS responds with a message "*Invalid Number*".

  By default, Extension Number Validation is enabled when SARVAM UCS is operating in the Enterprise mode. This flag is disabled when SARVAM UCS is operating in the Hospitality mode.

- **Use SMTP Account:** Select the SMTP Account you wish to use for VMS Notifications.

  The SMTP Account you configure will be used for all VMS Email Notifications — Message Notifications and Memory Usage Notifications.

  You may add a new SMTP Account. To do so,
  - Select *Add New* option for Use SMTP Account.

- Click **Settings** to configure the parameters of the New SMTP Account you created. For more information, see “SMTP Settings”.

If you select None as the option, the system will not send any VMS Notifications — Memory Usage Notification to SE,VMS E-Mail Notification, Message Wait Notification via E-Mail — even if you have enabled and configured the respective notification.

- **Memory Usage Notification to SE[220]:** The VMS allows all the memory related notifications — VMS memory usage and mailbox memory usage — to be sent to the System Engineer via email. Enable this flag, if you want memory related notifications to be sent via email to the System Engineer.

  You may customize the Notification Messages as per your requirement. For details, see “VMS E-Mail Notification”.

  You must also specify the email address to which the notifications are to be sent in *SE Email ID*.

- **SE Email ID:** Enter the email address on which the notifications should be sent to the System Engineer. The System Engineer must have access to this email ID. The email ID may consist of a maximum of 64 characters.

- **Save Call Taping Files in:** When Call Taping feature is enabled on your extension, you can save the Call Taping files either in the Common Mailbox or in your Personal Mailbox. By default, Call Taping files are saved in the Common Mailbox.

- **Common Mailbox for Call Taping (Enter Extension Number):** If you choose to save Call Taping files in Common Mailbox, you must specify the Access Code of any SLT, DKP, SIP Extension, Department Group or General Mailbox, whose mailbox you want to assign for Call Taping.

  If Call Taping files are saved in Common Mailbox, only the extension users who have access to the Common Mailbox will be able to retrieve and listen to the recorded conversations.

- **Save Call Tapping Files as:** If you select Common Mailbox, then select the type of file you want the system to generate for saving the taped conversation. You may select — *Individual File before and after call transfe*r or *Single File before and after call transfer* as per your requirement.

  If you select *Individual File before and after call transfe*r, the system will generate two separate files for saving the taped conversation, that is, one file containing the conversation taped before the call is transferred and another file containing the conversation taped after the call is transferred. However, if you select *Single File before and after call transfer,* the system will generate only one single file for saving the conversation taped before and after the call is transferred.

- **Make Message Notification calls using TAC:** Select the Trunk Access Code to be used by the system to make outgoing notification calls to external numbers.

- **Channel Reserved for Voice Mail Auto Attendant:** Select the number of channels of the VMS that you wish to reserve for the Voice Mail Auto Attendant. These channels will be used to answer incoming calls landing on Voice Mail Auto Attendant enabled Trunks only. These channels even if free will not be available to extension users to access their Mailbox.

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*220. This parameter will not function if you have selected None as the **Use SMTP Account** option. You may configure this parameter for future use.*

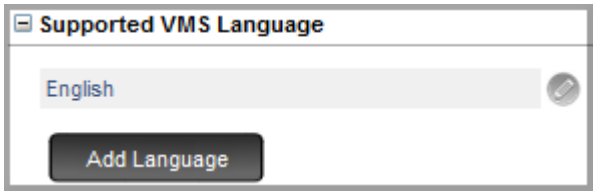
- **Date Format:** Select the Date Format — DD-MM-YYYY or MM-DD-YYYY. The Date format selected here will be applicable for all the VMS features, VMS E-Mail Notifications, etc.

*This Date format is not applicable for **Message Profile**. The "Date Playback Format" will be applicable there.*

- **Time Format:** Select the Time Format — 24 Hour or 12 Hour. The Time format selected here will be applicable for all the VMS features, VMS E-Mail Notifications, etc.

Supported VMS Language

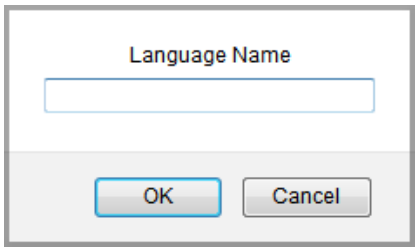
You may add a new language, edit or delete the existing languages.



To add a new language,

- Click the **Add Language** button.

  A window will pop up, asking you the language name.



- Enter the language name in the field. The allowed characters are A to Z, a to z, 0 to 9, - and _.
- Click OK.

To Edit a Language Name,

- Click .
- A window will pop up. You can edit the language name here.
- Click OK.

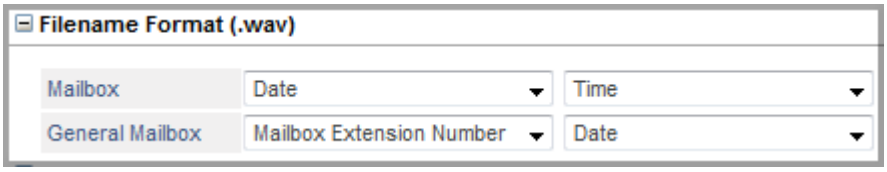
To delete a language, click .

If the language you wish to delete is already configured for any other parameter, the system will set English as the default language for that parameter.

Play Settings

- **Play '#' as**: Select the desired option — Hash or Pound. The system will play the selected option when '#' is pressed.
- **Play '*' as**: Select the desired option — Star or Asterisk. The system will play the selected option when '*' is pressed.
- **For all Options Menu, play digit**: Select the desired option — after each Option, before each Option. The system will play the digit as per the selected option for all Menu options.
- **For all Options Menu, for digit dialing play**: Select the desired option — Press or Dial. The system will play the selected option for digit dialing.

Filename Format (.wav)



You can define the Filename format for Mailbox (Personal) and General Mailbox by setting the sequence of the various parameters — None, Date, Time, Message Type, Calling Number, Called Number and Mailbox Extension Number[221]— as per requirement.

While selecting the Filename format, make sure:
- None is not selected as the first two options.
- Date and Time are included in the filename.
- The same option (other than None) is not selected twice.

The table given below defines the parameters. You may sequence it as per your requirement.

Option	Meaning
None	The system will skip this option and move to the next option configured.
Date	Select this option if you want the system to add the Date[a] in the filename.
Time	Select this option if you want the system to add the Time[b] in the filename.
Message Type	Select this option if you want the system to add the Message Type[c] for the message stored.

*221. The **Mailbox Extension Number** parameter is only applicable for General Mailbox Filename Format.*

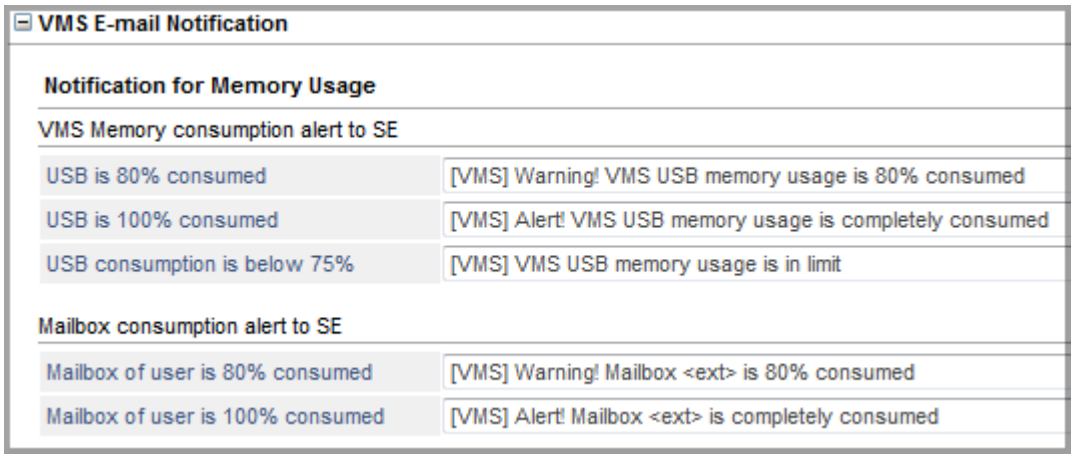
Option	Meaning
Calling Number	Select this option if you want the system to add the calling number. If calling number is Blank, the system will add 'No Number'.
Called Number	Select this option if you want the system to add the called number. If Called number is not available, system will ignore this option and select the next option. This option is generally required when you wish to use the feature Call Tapping.
Mailbox Extension Number	This option is applicable only for General Mailbox. Select this option if you want the system to add the mailbox extension number from which the message is transferred to the General Mailbox.

a. The Date Format here will be as per the Date Format you have selected in General Settings.
b. The Time Format here will be as per the Time Format you have selected in General Settings.
c. The Message Types supported by the system are: Call forward(CF), Call Taping(CT), Conversation Recording(CR), Broadcast Message(BM), Transfer to Mailbox(TM), Leave Message(LM), Send Message(SM), Call forward with LCS – No reply, all(LS), Redirect Message(RM), Message forward(MF).

The default sequential Filename Format is as given below:

- For Mailbox(Personal): *Date-Time-Message Type-Calling Number-Called Number.wav*
- For General Mailbox: *Mailbox Extension Number-Date-Time-Message Type-Calling Number-Called Number.wav*

VMS E-Mail Notification



Make sure you have configured the user Email ID for sending the notifications to user. For details, see "Message Wait Notification via E-Mail" in "Extension Voice Mail Settings".

To know about the Message Wait Notification feature, see "Email Based Notification".

The table below displays the default events for which notifications will be sent to the SE. You may customize it as per your requirement.

Event	Message	Description
Notification for Memory Usage		
VMS Memory consumption alert to SE		

Event	Message	Description
USB 80% is consumed	[VMS] Warning! VMS USB memory usage is 80% consumed	This email will be sent to the SE, when 80% of the USB memory has been consumed.
USB 100% is consumed	[VMS] Alert! VMS USB memory usage is completely consumed	This email will be sent to the SE, when 100% of the USB memory has been consumed.
USB consumption is below 75%	[VMS] VMS USB memory usage is in limit	This email will be sent to SE, when USB memory consumption is below 70%.
Mailbox consumption alert to SE		
Mailbox of user is 80% consumed	[VMS] Warning! Mailbox <ext> is 80% consumed	This email will be sent to the SE, when 80% of the mailbox memory has been consumed, along with the Extension Number (if configured).
Mailbox of user is 100% consumed	[VMS] Alert! Mailbox <ext> is completely consumed	This email will be sent to the SE, when 100% of the mailbox memory has been consumed, along with the Extension Number (if configured).
Mailbox consumption alert to User		
Mailbox is 80% consumed	[VMS] Warning! Mailbox <ext> is 80% consumed	This email will be sent to the User, when 80% of the personal mailbox memory has been consumed, along with the Extension Number (if configured).
Mailbox is 100% consumed	[VMS] Alert! Mailbox <ext> is completely consumed, please delete old messages to allow storing of new messages	This email will be sent to the User, when 100% of the personal mailbox memory has been consumed, along with the Extension Number (if configured).
General Mailbox is 80% consumed	[VMS] Warning! General Mailbox <ext> is 80% consumed	This email will be sent to the User, when 80% of the general mailbox memory has been consumed, along with the Extension Number (if configured).
General Mailbox is 100% consumed	[VMS] Alert! General Mailbox <ext> is completely consumed, please delete old messages to allow storing of new messages	This email will be sent to the User, when 100% of the general mailbox memory has been consumed, along with the Extension Number (if configured).
Notification for New Message in Mailbox		
Normal Message	[VMS] <msg_type> Message received from <cli>	This email will be sent to user's personal mailbox, when a message is received along with the Message Type and the Caller Number[a].
Conversation Recording Message	[VMS] Message received for conversation with <num1>	This email will be sent to user's personal mailbox, when the conversation between the extension user and the caller was recorded.
Call Tapping Message	[VMS] Message received for call recording between <num1> and <num2>	This email will be sent to user's personal mailbox, when the conversation between two Numbers was recorded.
Notification for New Message in General Mailbox		

Event	Message	Description
New Message	[VMS] <msg_type> Message received for <ext> from <cli>	This mail will be sent to the SE Email ID configured in the General Settings, when a message with the message type[b] is received for an Extension Number (if configured) from the Caller Number.
Conversation Recording Message	[VMS] Message received for <ext> and conversation was with <num1>	This mail will be sent to the SE Email ID configured in the General Settings, when a message is received for an Extension Number (if configured) and conversation was with the Number.

a. If Caller Number is unavailable, Caller Name will be displayed. If both are unavailable, 'Unknown' will be displayed. The CLI Number can be an External Caller Number or an Internal Caller Number.
b. Message Type may be Normal or Urgent.

General Mailbox Settings

A General Mailbox is a common mailbox in the VMS, with which more than one extension users are associated. When the personal mailbox of any extension user is full, all new messages are diverted to the General Mailbox.

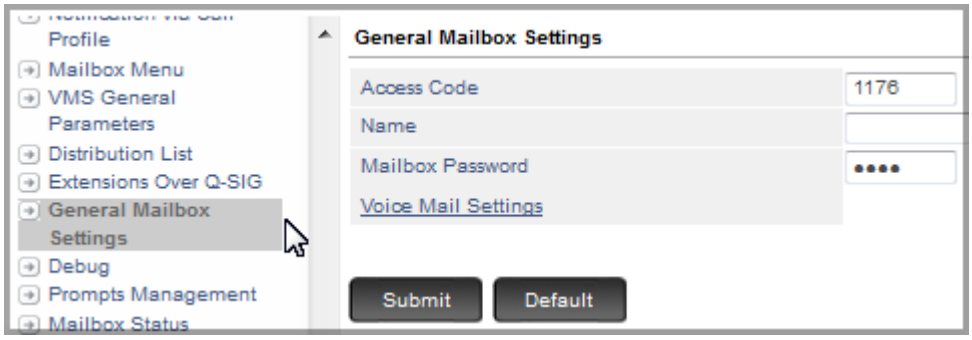
To access the General Mailbox, the extension users must have the feature General Mailbox enabled in their CoS.

The extension users can listen to the messages in the General Mailbox, by dialing the General Mailbox access code (programmable; default: 1176).

You can change the default settings of General Mailbox parameters using Jeeves.

Configuring General Mailbox Parameters using Jeeves

- Log in as System Engineer.
- Under **Configuration**, click **VMS Configuration**.
- Click **General Mailbox Settings**.



- Check the default values of the following parameters, and change them, if required, to the desired values:
  - **Access Code:** By default, 1176 is the access code for the General Mailbox. Extension users must dial this number, if they want to access the General Mailbox.

    If required, you may assign a different access code to the General Mailbox. The Access Code you assign may consist of a maximum of 16 digits. Digits 0-9, # and * are allowed.
  - **Name:** You can assign a Name to the General Mailbox. The name you assign can be a maximum of 18 characters. The Name must not have space as it's first character.

    <, >, :, ", /, \, |, ? and * characters are not allowed.
  - **Mailbox Password:** If you have selected the **Ask Password to access Mailbox** check box in *Extension Voice Mail Settings*, extensions users can access the General Mailbox by dialing the default Password-1111.

To avoid unauthorized access, we recommend you to change the default password. The password you assign may consist of a maximum of 4 digits. Valid Range: 0000 to 9999. Make sure the new password is strong and is provided to the extension users who need to access the General Mailbox only.

- **Voicemail Settings**: Click on this link to configure the Voicemail Settings for the General Mailbox. The Extension Voicemail Settings window opens.

  By default, the General Mailbox with access code-1176 will open. For information regarding the configuration, see “Extension Voice Mail Settings”.

- Click **Submit** to save the settings.

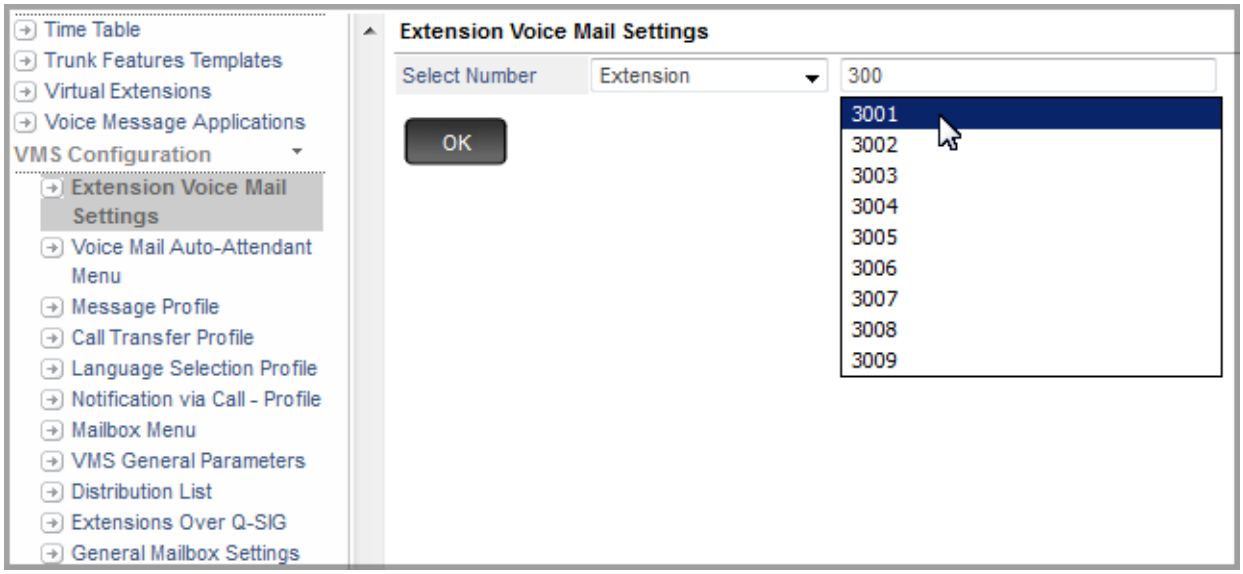
For more information and instructions on how to access the General Mailbox, see “Accessing the General Mailbox”.

Extension Voice Mail Settings

Extension Voice Mail Settings allows you to configure the various VMS parameters assigned to — an Extension, a Department Group, an Operator or a General Mailbox.

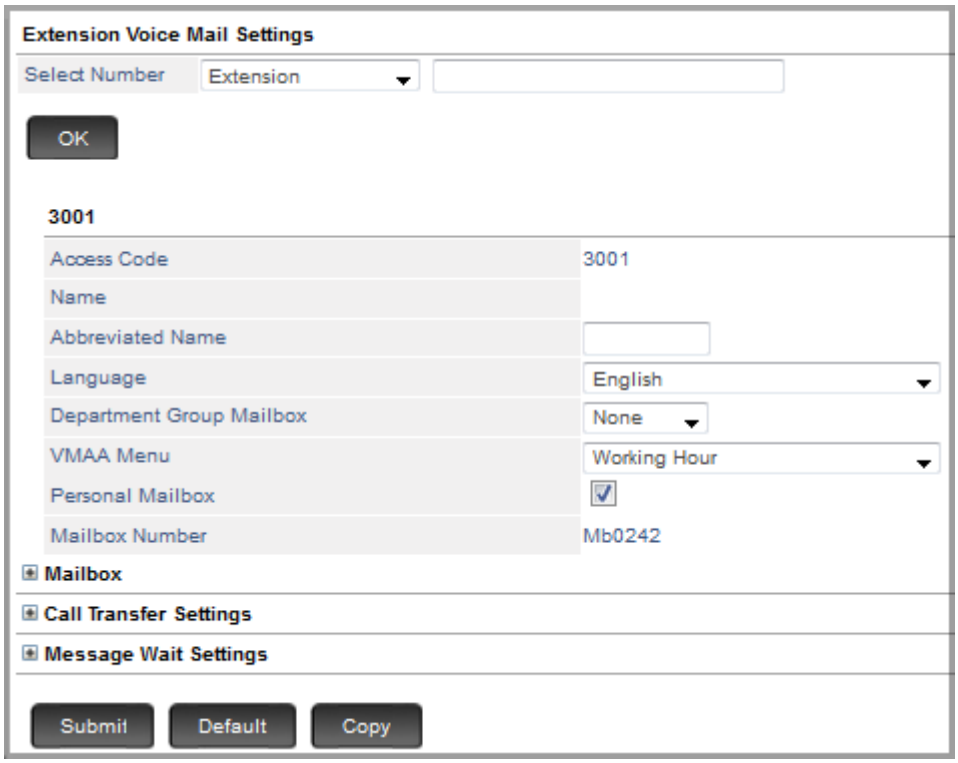
Configuring Extension Voice Mail Settings

- Log in as System Engineer.
- Under **Configuration**, click **VMS Configuration**.
- Click **Extension Voice Mail Settings**.



- **Select Number:** You may select — Extension, Department Group, Operator or General Mailbox.
  - For **Extension**, enter the Extension Number/Name and then select the same from the drop down list.
  - For **Department Group**, select the desired Department Group Number from the drop down list.
  - For **Operator**, select the desired Operator from the drop down list.
- Click **OK**.

You may now configure the respective VMS parameters.



- **Access Code:** The Access Code of the respective Extension, Department Group or General Mailbox is displayed as a status.

- **Abbreviated Name:** You may configure the abbreviated name for the respective Extension or Department Group. It must be a minimum of 3 alphabetic characters and a maximum of 8 alphabetic characters.

  Abbreviated Name is applicable when you use *Dial by Name* feature to transfer the call to any extension using VMAA Menu. For details, refer “Dial By Name”.

- **Language:** Select the Language which you want the system to use when accessing the Personal Mailbox. The languages displayed as options in the drop down list is as per the *Supported VMS Language* you configure in VMS General Settings.

All the prompts related to Personal Mailbox will be played in the language you select here.

- **Department Group Mailbox:** Select the Department Group Mailbox Number which you want to assign to the respective Extension. You may select None or the Department Group Mailbox Number. The system will allow the access to the department group mailbox you select here.

  If you select None, the department group mailbox will not be accessible by the Extension even if the Extension is a member of the Department Group.

- **VMAA Menu:** Select the VMAA Menu Number which you want to assign to the respective Extension or Department Group. For more information, see “Voice Mail Auto-Attendant Menu”.

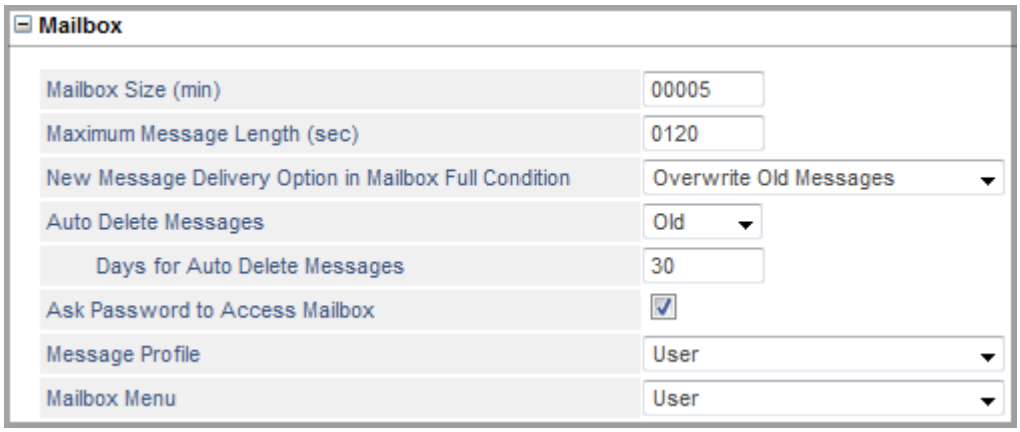
- **Personal Mailbox:** Keep the check box enabled if you want to assign the Personal Mailbox to the respective Extension, Department Group or General Mailbox. It allows you to access your Personal Mailbox.

For Extension and Department Group:
If you disable the check box, the Personal Mailbox will not be assigned and the system will not allow you to access your Personal Mailbox.

For General Mailbox:
If you disable the check box, the General Mailbox will not be assigned and the system will not allow you to access the General Mailbox.

- **Mailbox Number:** The Mailbox Number is displayed as a status when the Personal Mailbox is assigned to the respective Extension, Department Group or General Mailbox, if you have enabled the *Personal Mailbox* check box. If Personal Mailbox is not assigned, the Mailbox Number will be displayed as Blank.

Mailbox[222]



- In **Mailbox Size (min)**, configure the maximum allowed size for message storage. You may configure from 1 to 60000 minutes.

- In **Maximum Message Length (sec)**, configure the maximum time for which a message can be recorded by the caller. You may configure from 1 to 3600 seconds.

In case the Maximum Message Length is more than the Mailbox Size, the Mailbox size will be considered for recording.

- In **New Message Delivery Option in Mailbox Full Condition**, select an option for delivering the New Message when your Personal Mailbox is full.

  You can select from any of the options described below:

  - **Do not offer to leave message**, if you do not want the system to allow any new message to be delivered when the Personal Mailbox is full.

  - **Deliver to General Mailbox**, if you want the system to deliver the new message to the General Mailbox when Personal Mailbox is full.

  - **Overwrite Old Messages**, if you want the system to delete the old messages and allow the new message to be stored when the Personal Mailbox is full.

222. *Not applicable when you select **Operator** as the "Select Number" option.*

In case, the recorded message size is greater than the old message that is to be overwritten then the recorded message wll not be delivered. A prompt will be played for the same. Note: The old message will be deleted.

The prompt will not be played in case of multiple recipients.

If the mailbox has no old messages, the recorded message will not be stored.

- **Overwrite New Messages**, if you want the system to delete the new messages to allow the new message to be stored when the Personal Mailbox is full.

  The number of messages that will be deleted would be as per the *Maximum Message Length* allowed.

If the mailbox has no new messages, the recorded message will not be stored.

- **Overwrite All (Old + New)**, if you want the system to delete the old and new messages and allow the new message to be stored when the Personal Mailbox is full.

- In **Auto Delete Message**, select an option to delete the messages automatically by the system.You may select None, Old or All.

  - Select **None** if you do not want the system to delete the message automatically from the Personal Mailbox.

  - Select **Old** if you want the system to delete the old read messages automatically after the number of days you configure in *Days for Auto Delete Messages.*

  - Select **All** if you want the system to delete all messages — read or unread — automatically after the number of days you configure in *Days for Auto Delete Messages.*

    - In **Days for Auto Delete Messages**, configure the number of days after which you want the system to automatically delete the messages.

- In **Ask Password to Access Mailbox**,by default the access to the mailbox is password protected. The "User Password" is required to access the mailbox. Whenever the mailbox owner accesses the mailbox, the VMS will ask for the (user) password.

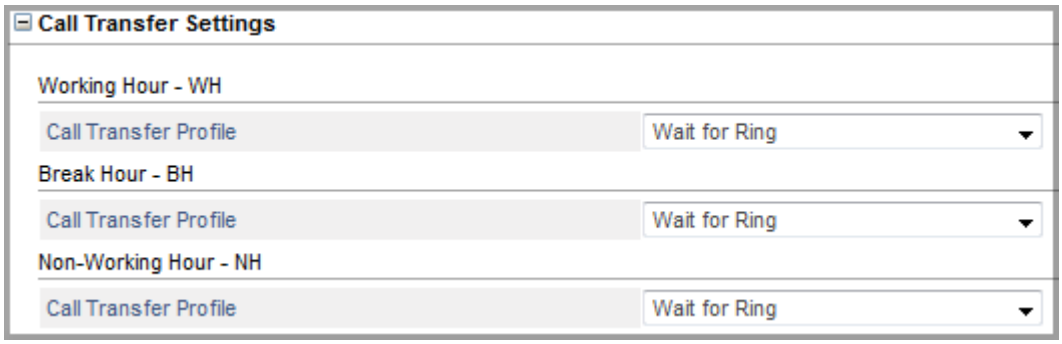
  If you want to remove password protection, clear this check box.

Since a Mailbox can be accessed using the default User Password, 1111, extension users who are assigned a mailbox are recommended to change their User Password. To avoid unauthorized access, we recommend extension users to change the password regularly. Make sure it is strong and is kept confidential.

- In **Message Profile**, select the Message Profile you want to assign. For more information, see "Message Profile".

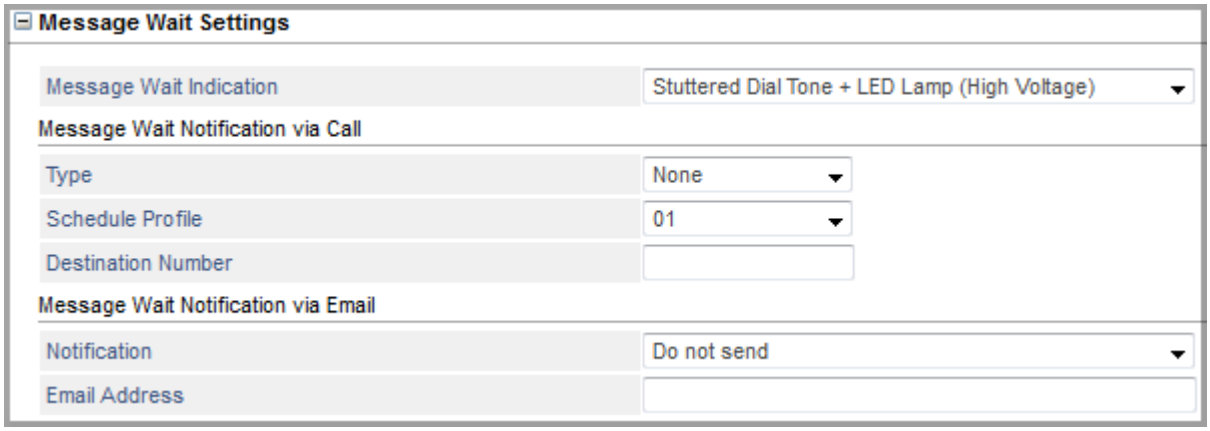
- In **Mailbox Menu**, select the Mailbox Menu you want to assign. For more information, see "Mailbox Menu".

Call Transfer Settings[223]



- Under **Working Hour - WH**, select the Call Transfer Profile you want system to use for transferring the calls during Working Hours. The drop down list will include the profiles you configure in the “Call Transfer Profile”.
- Under **Break Hour - BH**, select the Call Transfer Profile you want system to use for transferring the calls during Break Hours. The drop down list will include the profiles you configure in the “Call Transfer Profile”.
- Under **Non-Working Hour - NH**, select the Call Transfer Profile you want system to use for transferring the calls during Non-Working Hours. The drop down list will include the profiles you configure in the “Call Transfer Profile”.

Message Wait Settings[224]



- In **Message Wait Indication**, select the type of indication to be given to the extension user for new messages in the mailbox and message wait set by another extension user. This is only applicable when you select Extension as the “Select Number” option.

  You can select from any of the four types of indicators described below for new messages:

  - **Stuttered Dial Tone/Voice Message:** When the extension user goes OFF-Hook, s/he will hear a voice message, if a pre-recorded Voice Module has been assigned for Message Wait Notification. If no voice module is recorded and assigned, the extension user will hear a stuttered dial tone instead.

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*223. Not applicable when you select **General Mailbox** as the “Select Number” option.*
*224. Not applicable when you select **Operator** as the “Select Number” option.*

If you want voice message to be played as message wait notification, record and assign a Voice Module. Refer "Voice Message Applications" for instructions.

SARVAM UCS can play only 9 Voice Modules simultaneously. The Voice Module for Message Wait Notification will not be played if there are already 9 being played simultaneously. In this case, Stuttered Dial Tone will be played as Message Wait Indication, when the extension user goes OFF-Hook.

- **Ring:** The extension will ring for the duration of the Message Wait Ring Timer (configurable; default: 30 seconds), for as many times as the Message Wait Ring Count (configurable; default: 10 times), at the interval set as the Message Wait Ring Timer Interval (configurable; default: 30 minutes).

  When the extension user answers the call, the VMS informs the user of the new message and allows the extension user to access it.

The Ring option is not applicable for SIP Extensions.

- **Stuttered Dial Tone + LED Lamp (High Voltage):** When the extension user goes OFF-Hook, the extension user will hear a stuttered dial tone and if the SLT has a 'Message Wait' lamp, the lamp will blink continuously using High Voltage. When the extension user retrieves all the waiting messages, the LED will be turned off and the stuttered dial tone will stop.

- **Stuttered Dial Tone + LED Lamp (Polarity Reversal):** When the extension user goes OFF-Hook, the extension user will hear a stuttered dial tone and if the SLT has a 'Message Wait' lamp, the lamp will blink continuously using Polarity Reversal. When the extension user retrieves all the waiting messages, the LED will be turned off and the stuttered dial tone will stop.

- **LED Lamp (High Voltage):** If the SLT has a 'Message Wait' lamp, it will blink continuously using High Voltage. When the extension user retrieves all the waiting messages, the LED will be turned off.

- **LED Lamp (Polarity Reversal):** If the SLT has a 'Message Wait' lamp, it will blink continuously using Polarity Reversal. When the extension user retrieves all the waiting messages, the LED will be turned off.

Default: Stuttered Dial Tone + LED Lamp (High Voltage).

Refer the feature description "Message Wait" to know more.

Message Wait Notification via Call[225]

The message wait notification will be sent to a number(destination number). This number can be an internal or an external number.

- In **Type**, you may select — Immediate, Scheduled or None.

  - Select **Immediate**, if you want the notifications to be sent as soon as a new message arrives in the mailbox of the extension user.

  - Select **Scheduled**, if you want the notification to be sent at fixed time schedules.

  - Select **None**, if you do not want to set message wait notification via call.
    Default: None.

*225. Applicable only when you select **Extension** as the "Select Number" option.*

- In **Schedule Profile**, select the **Notification via Call - Profile** number according to which you want the system to send the notifications. The Notification via Call Profile determines how notification calls are to be made to the destination numbers. To know more, see “Message Wait Notification via Call”.

- In **Destination Number**, configure the number that you want the system to use for sending the notification via calls.

  The destination number can be an internal or an external number. The destination number can be a maximum of 16 digits. Valid digits are 0 to 9, # and *.

  When the notification call is answered, the VMS informs the callee about the new message and allows the callee to access it.

  Refer the feature description “Message Wait Notification via Call” to know more.

Message Wait Notification via E-Mail[226]

The message wait notification will be sent to the e-mail address of the extension user.

- In **Notification**, you may select — Do not send, Without Attachment, With Attachment or With Attachment and mark voicemail as read. Default: Do not send.

  - Select **Do not send**, if you do not want the system to send email notification to the user for new message even if the email ID is configured. In this case, the system will send the Mailbox Memory Usage notification.

  - Select **Without Attachment**, if you want the system to send email notification to the user for new message if the email ID is configured.

  - Select **With Attachment**, if you want the system to send email notification to the user for new message along with the message as attachment. Make sure the email ID is configured. The attachment will be sent only if the message size is less than or equal to 20 MB, else the email notification will be sent without the attachment.

  - Select **With Attachment and mark voicemail as read**, if you want the system to send email notification to the user for new message along with the message as attachment and also mark the voicemail as read. Make sure the email ID is configured. The attachment will be sent only if the message size is less than or equal to 20 MB, else the email notification will be sent without the attachment.

- **E-mail Address:** Enter the email ID of the extension user to which the notification is to be sent. Maximum allowable length of the email ID is 64 characters. Default: blank.

*Extension users will receive notifications only for the mailbox memory utilization, if you configure the **E-mail Address** and select **Do not sent** as the **Notification** option.*

Refer the feature description “Email Based Notification” to know more.

- Click **Submit** to save Extension Voice Mail settings.

- Click **Copy** if you wish to copy the respective Extension’s Voice Mail settings to other extensions.

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*226. Not applicable when you select **Operator** as the “Select Number” option.*

Copy Voice Mail Settings to window will open.

You can copy the Voice Mail Settings to a single Extension, multiple extensions or all extensions.

For single Extension, select **Extension Number** and enter the Extension Number.

For multiple extensions, select **Extension Numbers from** and enter the Extension Number range.

For all extensions, select **All Extensions**.

The Voice Mail Settings — Abbreviated Name, Message Wait Notification via Call parameters and Message Wait Notification via Email parameters will not be copied.

Voice Mail Auto-Attendant Menu

Voice Mail Auto-Attendant Menu is applicable when trunk call is directly routed to VMS.

Each trunk can be assigned a VMAA Menu. For details, see “Trunk Feature Template”.

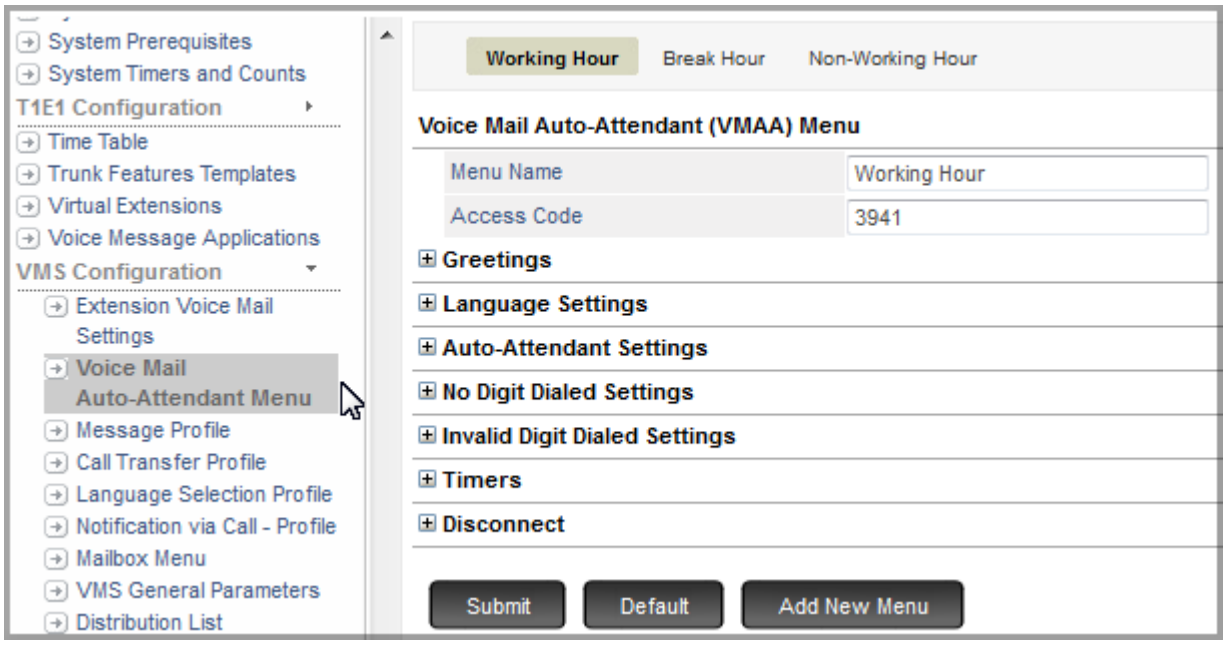
If VMAA Menu assigned to any action/trunk is deleted, the system will consider the first VMAA Menu for that action/trunk by default.

VMS supports a maximum of 128 VMAA Menu.

By default, three VMAA Menus — Working Hour, Break Hour and Non-Working Hour— are provided to you. These three VMAA menus cannot be deleted but you may edit their settings as per your requirement.

How to Configure

- Log in as System Engineer.
- Under **Configuration**, click **VMS Configuration**.
- Click **Voice Mail Auto-Attendant Menu**.



You may add a new menu, edit the default menus or delete a VMAA menu.

To add a new VMAA Menu,

- Click **Add New Menu**. A new *VMAA Menu xxx* will be created. You may now configure this as per your requirement.

To delete a menu, click **Delete**.

The default VMAA Menus cannot be deleted.

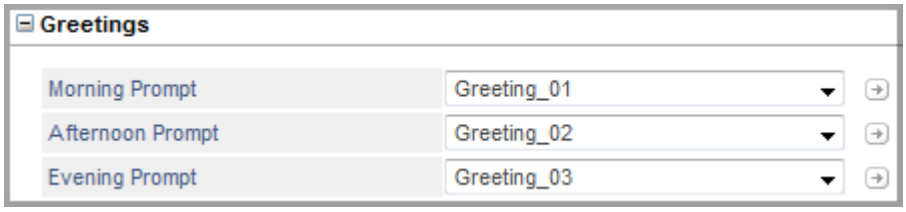
To edit a VMAA Menu,

- Click on the **VMAA Menu** tab you wish to edit.
- In **Menu Name**, configure the name of the VMAA Menu. By default, it is *VMAA Menu xxx* where xxx is the VMAA Menu Number from 001 to 128.
- In **Access Code**, configure the Access Code you wish to assign to the respective VMAA Menu. The caller can transfer the call to a VMAA Menu by dialing the respective access code. Access Code can be a maximum of 6 digits.

Make sure, the Access Codes assigned to the VMAA Menus do not conflict with the existing access codes. System will not save the configured VMAA access code if the same code is already assigned.

Greetings

Greetings allows you to select the prompts to greet the caller.



- In **Morning Prompt**, select the prompt which you wish to play to the caller as the Morning Greeting.

  Select **Do Not Play**, if you do not wish to play any prompt.

  - Click **Settings** . The **Prompts Management** page opens.

    You may add a new prompt from here. For further details, see “Prompts Management”.

  Once the files are uploaded these appear as options for Morning Prompt.

- In **Afternoon Prompt**, select the prompt which you wish to play to the caller as the Afternoon Greeting.

  Select **Do Not Play**, if you do not wish to play any prompt.

  - Click **Settings** . The **Prompts Management** page opens.

    You may add a new prompt from here. For further details, see “Prompts Management”.

  Once the files are uploaded these appear as options for Afternoon Prompt.

- In **Evening Prompt**, select the prompt which you wish to play to the caller as the Evening Greeting.

Select **Do Not Play**, if you do not wish to play any prompt.

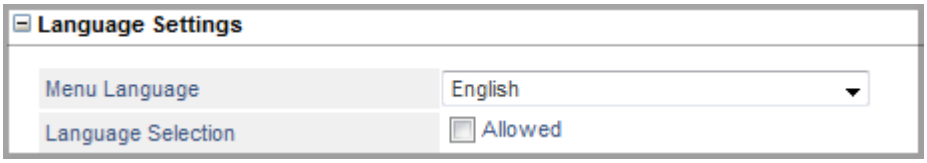
- Click **Settings** . The **Prompts Management** page opens.

  You may add a new prompt from here. For further details, see “Prompts Management”.

Once the files are uploaded these appear as options for Evening Prompt.

Language Settings

Language Settings allows you to set a menu language for the respective VMAA Menu and also gives the caller a choice to select the language.

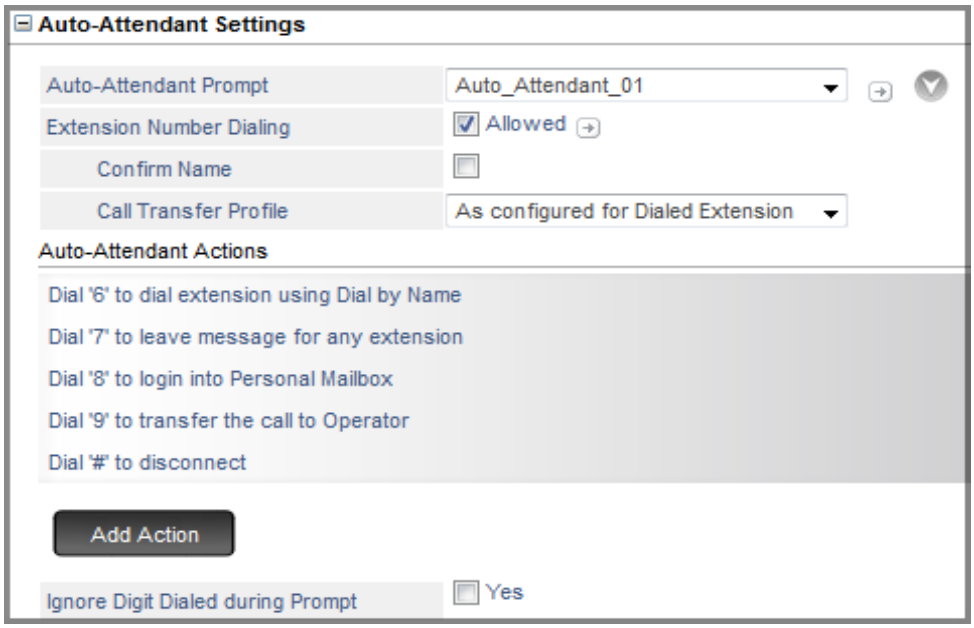


- In **Menu Language**, select the language which you wish to set as the default language for the respective VMAA Menu. For all the calls routed to the VMAA Menu, the prompts and greetings will be played as per the Menu Language set.
- Select the **Language Selection** check box if you wish to allow the caller to choose a language.
  - In **Language Selection Profile**, select the language profile you want to assign to the respective VMAA Menu.
    - Click **Settings** to configure the parameters of the selected profile. For more information, see “Language Selection Profile”.

    From here, the caller can select a different language other than the Menu Language configured. All the VMS prompts henceforth will be played in the language selected here.

Auto-Attendant Settings

Auto-Attendant Settings allows you to customize the Auto-Attendant Actions.



- In **Auto Attendant Prompt**, select the prompt which you wish to play to the caller according to the Auto-Attendant Actions.

  Select **Do Not Play**, if you do not wish to play any prompt.

  - Click **Settings** . The **Prompts Management** page opens.

    You may add a new prompt from here. For further details, see “Prompts Management”.

  Once the files are uploaded these appear as options for Auto Attendant Prompt.

  To add more Auto-Attendant Prompts, click . You can add a maximum of 4 Auto Attendant Prompts in each VMAA Menu.

  After playing all Auto-Attendant prompts, the system will wait for the input from the caller as per the timers configured. For details, see “Timers”.

- Select the **Extension Number Dialing** check box to allow the caller to dial the Extension Number while the Auto-Attendant prompts are being played.

  - Click **Settings** to configure the No Match Found Settings.

    - In **No Match Found Prompt**, select the prompt which you wish to play when caller has dialed an invalid extension number.

      Select **None**, if you do not wish to play any prompt.

      You may add a new Prompt. To do so,

      - Click **Settings** . The **Prompts Management** page opens.

You may add a new prompt from here. For further details, see "Prompts Management".

Once the files are uploaded these will appear as the options for No Match Found Prompt.

- Select the **No Match Found Retry** check box to prompt the caller to dial the Extension Number again.
- In **No Match Found Retry Count**, select the number of times you wish to prompt the caller for dialing the extension number. The prompt will be played repeatedly till the Retry Count expires.
- In **Retry Count Expiry Prompt**, select the prompt you wish to play when the Retry Count expires.

  Select **None**, if you do not wish to play any prompt.

  - Click **Settings** . The **Prompts Management** page opens.

    You may add a new prompt from here. For further details, see "Prompts Management".

  Once the files are uploaded these appear as options for Retry Count Expiry Prompt.

- In **No Match Found Action**, you can select — Disconnect, Transfer to Operator, Transfer to Extension, Transfer to Department Group, Go to VMAA Menu.

  - If you select **Transfer to Operator** or **Transfer to Department Group** or **Transfer to Extension**, select the respective Operator number/Department Group number or enter the desired extension number.

    - In **Call Transfer Profile**, select the desired Call Transfer Profile as per your requirement. By default, *As configured for Transfer Number* is selected.

      Click **Settings** to configure the parameters of the selected profile. For more information, see "Call Transfer Profile".

  - If you select **Go to VMAA Menu**, select the desired VMAA Menu.
  - If you select **Disconnect**, the system will disconnect the call.

- Select the **Confirm Name** check box, if you want the system to play the prompt asking the caller to enter the name of the called party, after dialing the extension number. To know more, refer to "Dial by Extension Number".
- In **Confirm Name Prompt,** select the prompt which you wish to play to the caller.

  - Click **Settings** . The **Prompts Management** page opens.

    You may add a new prompt from here. For further details, see "Prompts Management".

    Once the files are uploaded these appear as options for Confirm Name Prompt.

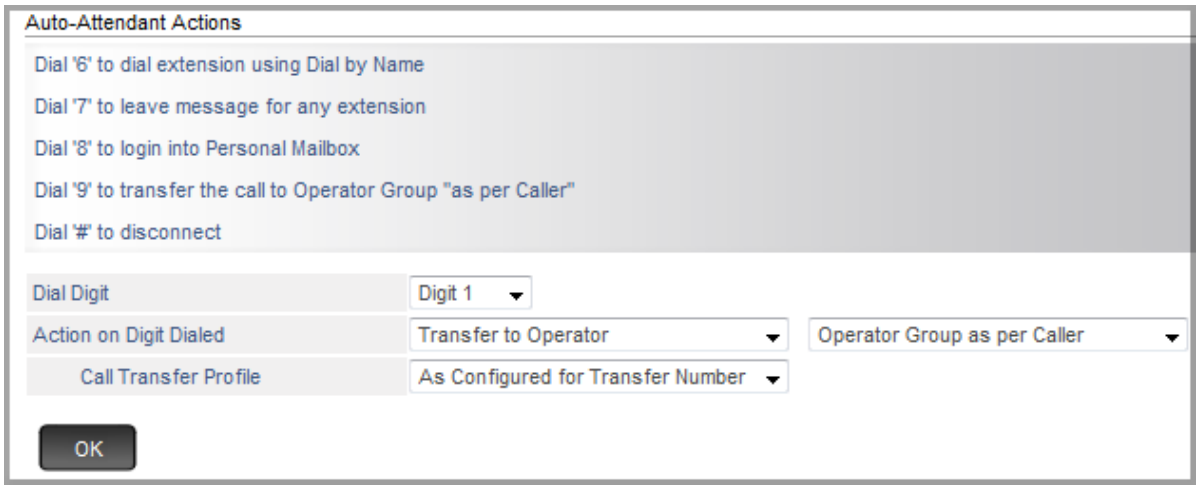
  The system will place the call only when the entered name matches with the **Abbreviated Name** in *Extension Voice Mail Settings*. To know more, refer to "Extension Voice Mail Settings".

- In **Call Transfer Profile**, select the desired Call Transfer Profile you want the system to use. Make sure you have configured the parameters for this profile. By default, *As configured for Dialed Extension* is selected.
    - Select **As configured for Dialed Extension** if you want the VMS Auto Attendant to use the Call Transfer Profile which is assigned to the dialed extension.
    - Select **Wait for Ring** if you want the VMS Auto Attendant to wait for the extension to start ringing and then transfer the call.
    - Select **Blind** if you want the VMS Auto Attendant to transfer the call on the extension without checking whether it is busy or free.
    - Select **Attended** if you want the VMS Auto Attendant to transfer the call when the extension answers (goes OFF-Hook).
    - Click **Settings** to configure the parameters of the selected profile. For more information, see "Call Transfer Profile".

When the Call Transfer Profiles are added, or any existing profile is edited, these changes will be reflected in the options displayed.

Auto-Attendant Actions

The default Auto-Attendant actions taken by the system, when there is an incoming trunk call on the VMS are displayed here.



You may add a new action, edit or delete the default actions.

To delete an Auto-Attendant action, mouse over on the respective action and click .

To add/edit an Auto-Attendant action,

- Click on the **Add Action** button to add a new action or click to edit an action.
- In **Dial Digit**, select the digit to be dialed for the respective action.
- In **Action to Digit Dialed**, you can select any one of the following:

- Transfer to Operator
- Transfer to Extension
- Transfer to Department Group
- Go to VMAA Menu
- Play Information
- Dial Extension Number by Name
- Leave Voice Mail
- Personal Mailbox Access
- Change Language
- Repeat Prompt
- Go to Previous Menu
- Disconnect
- If you select **Transfer to Operator** or **Transfer to Extension** or **Transfer to Department Group**, select the respective Operator number or desired extension number or Department Group number.

    - In **Call Transfer Profile**, select the desired Call Transfer Profile as per your requirement. By default, *As configured for Transfer Number* is selected.

- If you select **Go to VMAA Menu**, select the desired VMAA Menu.

- If you select **Play Information**, click **Settings** to configure the parameters. For details, see “Play Information”.

- If you select **Dial Extension Number by Name**, the system will allow caller to dial any extension number using name. Click **Settings** to configure the parameters. For details, see “Dial by Name”.

- If you select **Leave Voice Mail**, the system will allow caller to leave the message directly to the extension number or department group. Click **Settings** to configure the parameters. For details, see “Leave Voice Mail”.

- If you select **Personal Mailbox Access**, the system will provide the mailbox access to any user from remote location. Click **Settings** to configure the parameters. For details, see “Mailbox Access”.

- If you select **Change Language**, select the Language which you want the system to use for the VMAA Menu instead of the previously selected Language.

  Click **Settings** to configure the parameters of the selected language profile. For more information, see “Language Selection Profile”.

  All the VMS prompts henceforth will be played in the language selected here.

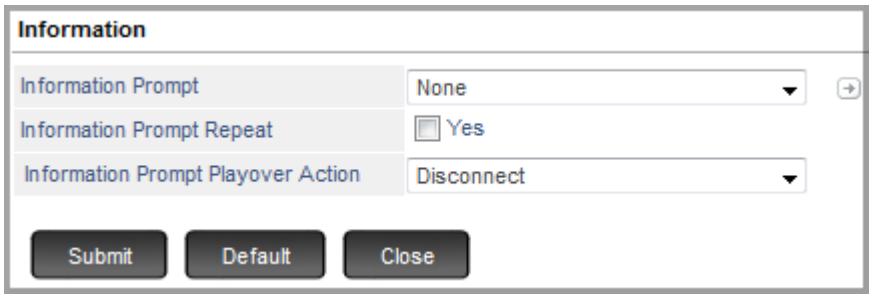
- If you select **Repeat prompt**, the system will repeat all the Auto-Attendant prompts configured in sequence.

- If you select **Go to Previous Menu**, the system will provide an option to the caller to go back to the previous VMAA Menu.

- If you select **Disconnect**, the system will disconnect the call.

- Select the **Ignore Digit Dialed during Prompt** check box, if you want the system to ignore the digits dialed by the caller while the prompts are being played.

Play Information



- In **Information Prompt**, select the prompt which has the information to be played to the caller. The information may be about the company or about the different products etc.

  Select **None**, if you do not wish to play any information prompt to the caller.

  - Click **Settings** . The **Prompts Management** page opens.

    You may add a new prompt from here. For further details, see “Prompts Management”.

  Once the files are uploaded these appear as options for Information Prompt.

- Select the **Information Prompt Repeat** check box to allow the Information prompt to be repeated to the caller.

- In **Information Prompt Repeat Count**, select the number of times you wish to play information prompt to the caller. The prompt will be played repeatedly till the Repeat Count expires.

- In **Repeat Count Expiry Prompt**, select the prompt you wish to play when the Information Prompt Repeat Count expires.

  Select **None**, if you do not wish to play any prompt.

  - Click **Settings** . The **Prompts Management** page opens.

    You may add a new prompt from here. For further details, see “Prompts Management”.

  Once the files are uploaded these appear as options for Repeat Count Expiry Prompt.

- In **Information Prompt Playover Action**, select — Transfer to Operator, Transfer to Department Group, Transfer to Extension, go to VMAA Menu or Disconnect.

  - If you select **Transfer to Operator** or **Transfer to Department Group** or **Transfer to Extension**, select the respective Operator number/Department Group number or enter the desired extension number.

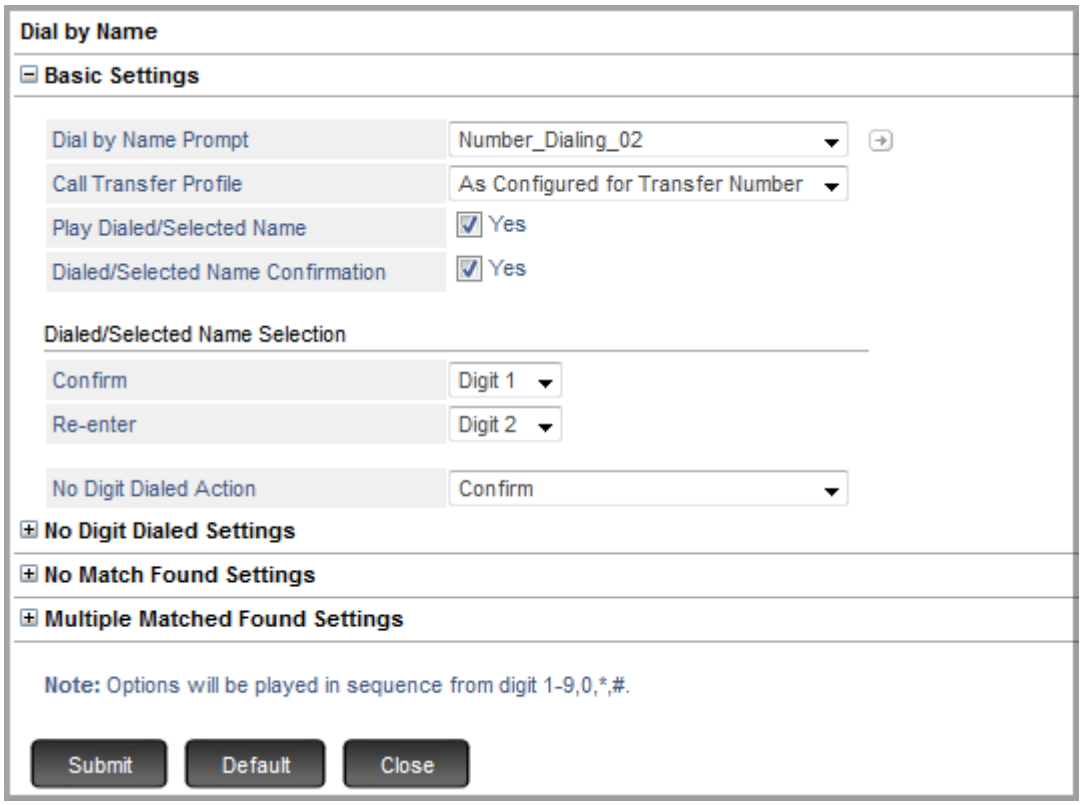
    - In **Call Transfer Profile**, select the desired Call Transfer Profile as per your requirement. By default, *As configured for Transfer Number* is selected.

    Click **Settings** to configure the parameters of the selected profile. For more information, see “Call Transfer Profile”.

- If you select **Go to VMAA Menu**, select the desired VMAA Menu.
- If you select **Disconnect**, the system will disconnect the call.

- Click **Close** to close the window.

Dial by Name



Basic Settings

- In **Dial by Name Prompt**, select the prompt which you wish to play to the caller allowing him/her to dial the extension number using name.

  Select **None**, if you do not wish to play any prompt.

  - Click **Settings** . The **Prompts Management** page opens.

    You may add a new prompt from here. For further details, see "Prompts Management".

  Once the files are uploaded these appear as options for Dial by Name Prompt.

- In **Call Transfer Profile**, select the desired Call Transfer Profile as per your requirement. By default, *As configured for Transfer Number* is selected.

  Click **Settings** to configure the parameters of the selected profile. For more information, see "Call Transfer Profile".

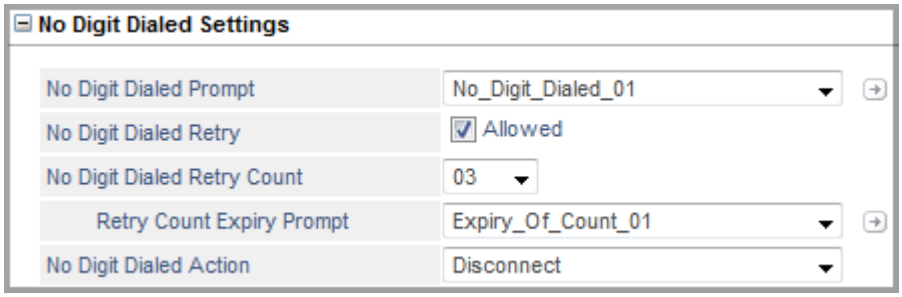
- Select the **Play Dialed/Selected Name** check box, if you want the system to play the name dialed by the caller and check the option **Dialed/Selected Name Confirmation**.

If you disable the check box, the system will transfer the call directly to the extension number dialed by the caller as per the Call Transfer Profile configured.

- Select the **Dialed/Selected Name Confirmation** check box, if you want the system to confirm with the caller before transferring the call to the extension number dialed.
  - In **Confirm**, assign the Digit (0-9, *, # or None) that you want the system to play to the caller for confirming the Name dialed before transferring the call. If you assign *None*, the option will not be played.
  - In **Re-enter**, assign the Digit (0-9, *, # or None) that you want the system to play to prompt the caller to dial the extension number again.
  - In **No Digit Dialed Action**, select the option — Confirm or Re-enter. This is applicable when the caller has not dialed any digit for Selected Name Confirmation. The system will function as per the option you select.

No Digit Dialed Settings

No Digit Dialed Settings are applicable when caller has not dialed any digit and the first digit wait timer has expired.



- In **No Digit Dialed Prompt**, select the prompt which you wish to play when caller has not dialed any digit.

  Select **None**, if you do not wish to play any prompt when caller has not dialed any digit.
  - Click **Settings** . The **Prompts Management** page opens.

    You may add a new prompt from here. For further details, see “Prompts Management”.

  Once the files are uploaded these appear as options for No Digit Dialed Prompt.
- Select the **No Digit Dialed Retry** check box to prompt the caller to dial the digit again.
  - In **No Digit Dialed Retry Count**, select the number of times you wish to prompt the caller for dialing the digit. The prompt will be played repeatedly till the Retry Count expires.
  - In **Retry Count Expiry Prompt**, select the prompt you wish to play when the Retry Count expires.

    Select **None**, if you do not wish to play any prompt.
  - Click **Settings** . The **Prompts Management** page opens.

    You may add a new prompt from here. For further details, see “Prompts Management”.

Once the files are uploaded these appear as options for Retry Count Expiry Prompt.

- In **No Digit Dialed Action**, you can select — Disconnect, Transfer to Operator, Transfer to Extension, Transfer to Department Group, Go to VMAA Menu.

  - If you select **Transfer to Operator** or **Transfer to Department Group** or **Transfer to Extension**, select the respective Operator number/Department Group number or enter the desired extension number.

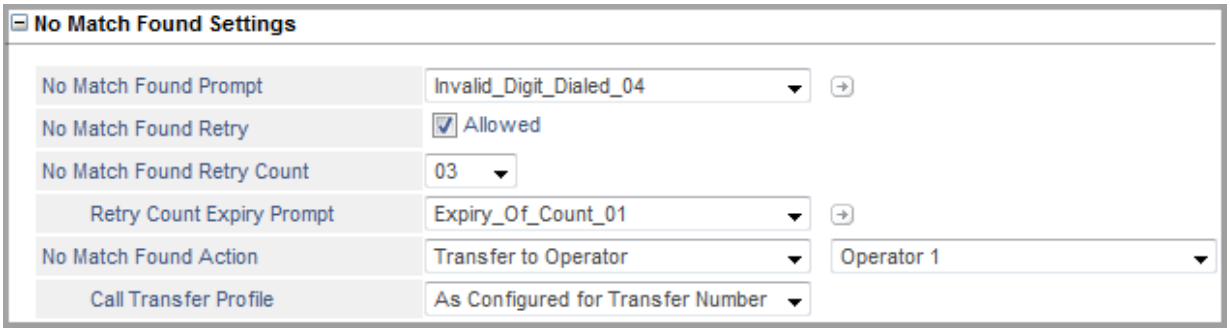
    In **Call Transfer Profile**, select the desired Call Transfer Profile as per your requirement. By default, *As Configured for Transfer Number* is selected.

    Click **Settings** to configure the parameters of the selected profile. For more information, see "Call Transfer Profile".

  - If you select **Go to VMAA Menu**, select the desired VMAA Menu.

  - If you select **Disconnect**, the system will disconnect the call.

No Match Found Settings



- In **No Match Found Prompt**, select the prompt which you wish to play when caller has dialed an invalid extension number.

  Select **None**, if you do not wish to play any prompt.

  - Click **Settings**. The **Prompts Management** page opens.

    You may add a new prompt from here. For further details, see "Prompts Management".

  Once the files are uploaded these appear as options for No Match Found Prompt.

- Select the **No Match Found Retry** check box to prompt the caller to dial the Extension Number again.

- In **No Match Found Retry Count**, select the number of times you wish to prompt the caller for dialing the extension number. The prompt will be played repeatedly till the Retry Count expires.

- In **Retry Count Expiry Prompt**, select the prompt you wish to play when the Retry Count expires.

  Select **None**, if you do not wish to play any prompt.

- Click **Settings** . The **Prompts Management** page opens.

  You may add a new prompt from here. For further details, see “Prompts Management”.

Once the files are uploaded these appear as options for Retry Count Expiry Prompt.

- In **No Match Found Action**, you can select — Disconnect, Transfer to Operator, Transfer to Extension, Transfer to Department Group, Go to VMAA Menu.

  - If you select **Transfer to Operator** or **Transfer to Department Group** or **Transfer to Extension**, select the respective Operator number/Department Group number or enter the desired extension number.

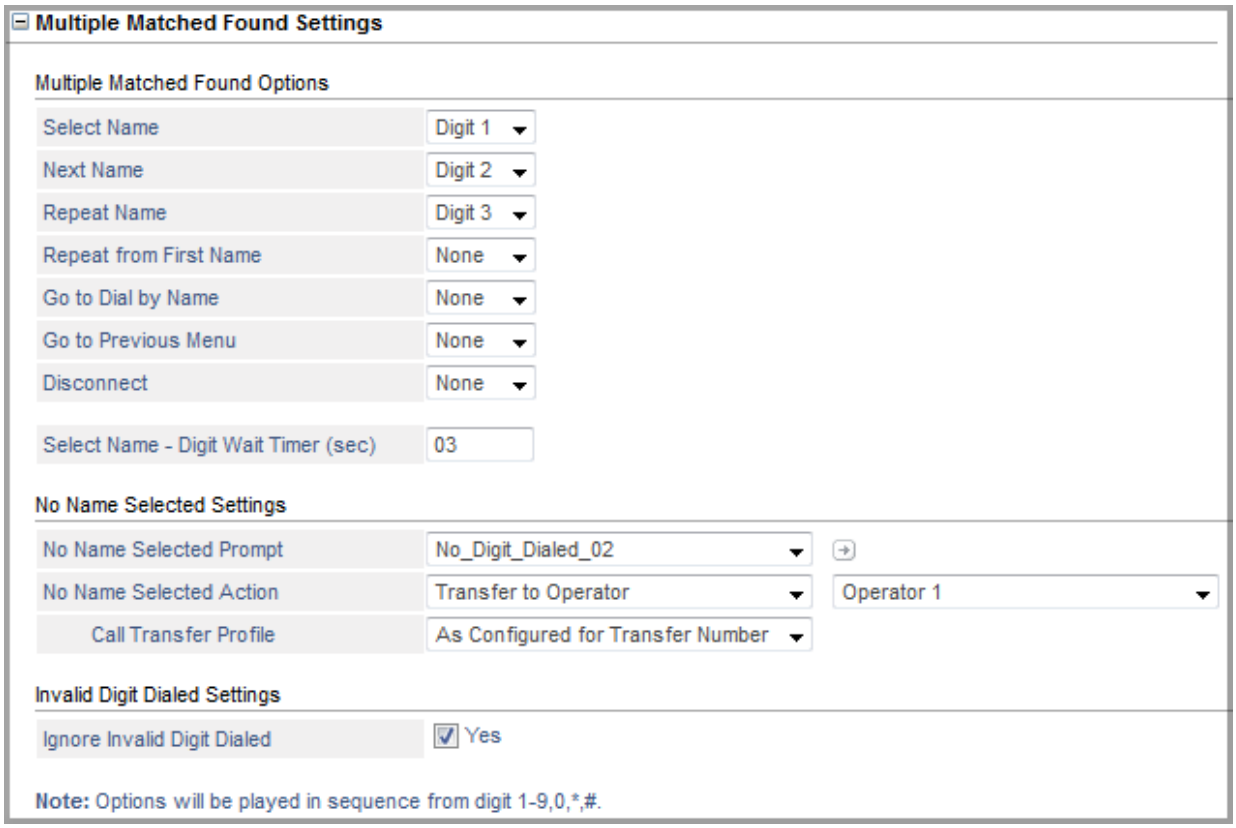
    In **Call Transfer Profile**, select the desired Call Transfer Profile as per your requirement. By default, *As Configured for Transfer Number* is selected

    Click **Settings** to configure the parameters of the selected profile. For more information, see “Call Transfer Profile”.

  - If you select **Go to VMAA Menu**, select the desired VMAA Menu.

  - If you select **Disconnect**, the system will disconnect the call.

Multiple Match Found Settings



Multiple Match Found Options

You can assign the Digits (0-9, *, # or None) to each of the options — Select Name, Next Name, Repeat Name, Repeat from First Name, Go to Dial by Name, Go to Previous Menu or Disconnect.

Make sure you do not assign the same digit to multiple options.

The digit you configure will be played as the digit to be dialed by the caller to select the respective option. The options will be played in the sequence — 1-9, 0, *, #. The option for which the Digit selected is None will not be played.

After the digits have been assigned, the system will proceed further as per the options selected by you. These options will be played for each of the Names included in the Multiple Match Found list.

Brief description of each option is explained below:

- **Select Name**: The system will play a Name from the multiple match found list. The caller may select the this name for if he wishes to transfer the call to this Extension.
- **Next Name**: The system will play the next name from the multiple match found list.
- **Repeat Name**: The system will re-play the last name played from the multiple match found list.
- **Repeat from First Name**: The system will play the multiple match found list of names from the first matched name.
- **Go to Dial by Name**: The system will clear the multiple match found list and prompt caller to dial the Extension by Name again.
- **Go to Previous Menu**: The system will provide an option to the caller to go back to the Previous Menu.
- **Disconnect**: The system will disconnect the call after playing the Disconnect prompt.

In **Select Name - Digit Wait Timer (sec)**, enter the time for which you want the system to wait to play the next name while playing the multiple match found list. Default: 03 seconds

No Name Selected Settings

- In **No Name Selected Prompt**, select the prompt you wish to play when no name is selected by the caller.

  Select **None**, if you do not wish to play any prompt.

  - Click **Settings** . The **Prompts Management** page opens.

    You may add a new prompt from here. For further details, see “Prompts Management”.

  Once the files are uploaded these appear as options for No Name Selected Prompt.

- In **No Name Selected Action**, you can select — Transfer to Operator, Transfer to Extension, Transfer to Department Group, Go to VMAA Menu, Go to Previous Menu or Disconnect.

  - If you select **Transfer to Operator** or **Transfer to Department Group** or **Transfer to Extension**, select the respective Operator number/Department Group number or enter the desired extension number.

In **Call Transfer Profile**, select the desired Call Transfer Profile as per your requirement. By default, *As Configured for Transfer Number* is selected.

Click **Settings** to configure the parameters of the selected profile. For more information, see "Call Transfer Profile".

- If you select **Go to VMAA Menu**, select the desired VMAA Menu.
- If you select **Go to Previous Menu**, the system will go back to the previous menu.
- If you select **Disconnect**, the system will disconnect the call.

Invalid Digit Dialed Settings

- Clear the **Ignore Invalid Digit Dialed** check box, if you do not want the system to ignore the invalid digit dialed and prompt the caller for the same. By default, it is enabled.
  - In **Invalid Digit Dialed Prompt**, select the prompt you wish to play when the caller has dialed an invalid digit for option selection.

    Select **None**, if you do not wish to play any prompt when an invalid digit is dialed.
  - Click **Settings**. The **Prompts Management** page opens.

    You may add a new prompt from here. For further details, see "Prompts Management".

  Once the files are uploaded these appear as options for Invalid Digit Dialed Prompt.
- In **Invalid Digit Dialed Action**, you can select — Transfer to Operator, Transfer to Extension, Transfer to Department Group, Go to VMAA Menu, Go to Previous menu or Disconnect.
  - If you select **Transfer to Operator** or **Transfer to Department Group** or **Transfer to Extension**, select the respective Operator number/Department Group number or enter the desired extension number.

    In **Call Transfer Profile**, select the desired Call Transfer Profile as per your requirement. By default, *As Configured for Transfer Number* is selected.

    Click **Settings** to configure the parameters of the selected profile. For more information, see "Call Transfer Profile".
  - If you select **Go to VMAA Menu**, select the desired VMAA Menu.
  - If you select **Go to Previous Menu**, the system will go back to the previous menu.
  - If you select **Disconnect**, the system will disconnect the call.
- Click **Close** to close the window.

Leave Voice Mail

- In **Leave Voice Mail to**, select an option where you want to leave the voice mail message.
  - If you select **Extension Number** or **Department Group**, the system will prompt the caller to leave the message for the Extension Number or Department Group configured.
  - If you select **Extension number dialed by caller**, the system will prompt the caller to enter the number to leave the message for the Extension Number or Department Group.

    Click **Settings** to configure the *Leave Voice Mail - Dialed Extension Number* parameters.
    - In **Leave Voice Mail Prompt**, select the prompt you wish to play to prompt the caller to leave the message.

      Select **None**, if you do not wish to play any prompt.
      - Click **Settings**. The **Prompts Management** page opens.

        You may add a new prompt from here. For further details, see "Prompts Management".

      Once the files are uploaded these appear as options for Leave Voice Mail Prompt.
    - For **No Match Found Settings**, refer "No Match Found Settings".
    - For **No Digit Dialed Settings**, refer "No Digit Dialed Settings".
- Select the **Override Message Leave Settings of Extension** check box, if you want the system to apply the Leave Voice Mail Settings configured on this page. Configure the "Message Leave Settings" and "No Mailbox Action".

  Clear the check box, if you want the system to apply the Leave Voice Mail settings configured for the extension.
- In **Action after Leaving Voice Mail**, you can select — Disconnect, Transfer to Operator, Transfer to Extension, Transfer to Department Group, Go to VMAA Menu.
  - If you select **Transfer to Operator** or **Transfer to Department Group** or **Transfer to Extension**, select the respective Operator number/Department Group number or enter the desired extension number.

- In **Call Transfer Profile**, select the desired Call Transfer Profile as per your requirement. By default, *As configured for Transfer Number* is selected.

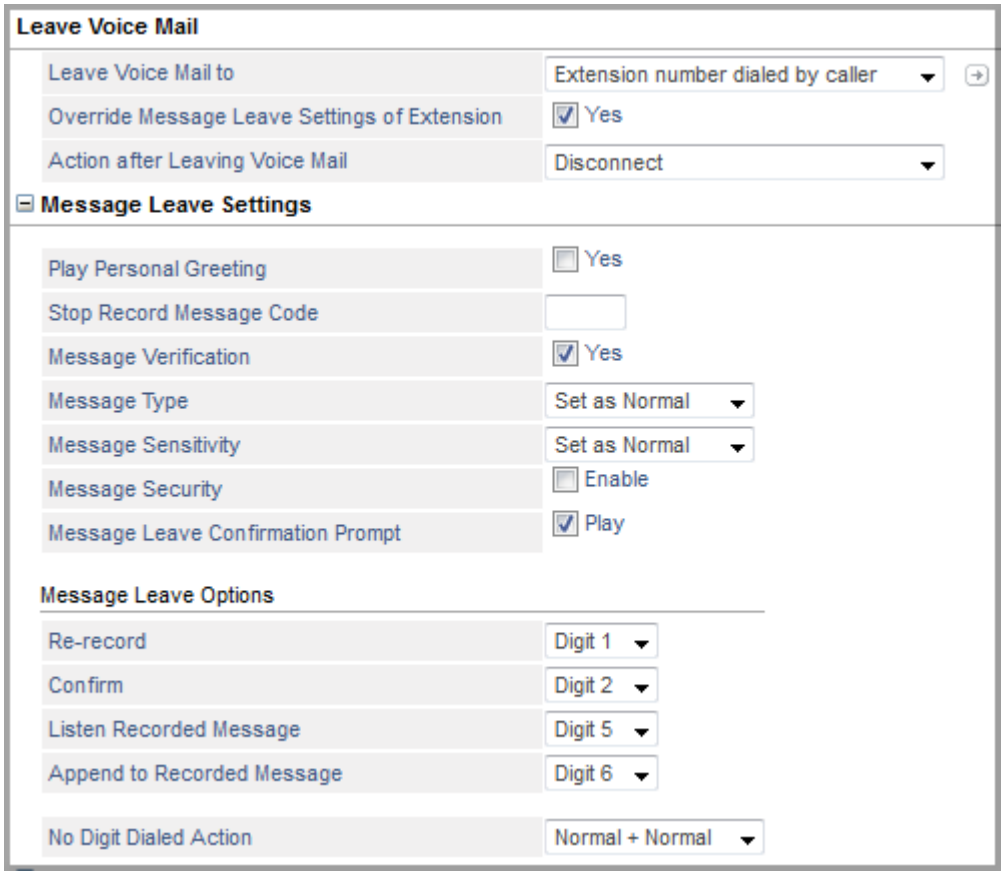
  Click **Settings** to configure the parameters of the selected profile. For more information, see "Call Transfer Profile".

- If you select **Go to VMAA Menu**, select the desired VMAA Menu.

- If you select **Disconnect**, the system will disconnect the call.

Message Leave Settings

Message Leave Settings is applicable if you have enabled *Overide Message Leave Settings of Extension* parameter. It allows you to configure the settings for the messages you receive from the caller i.e the message left for you by the caller.



- Select the **Play Personal Greeting** check box to allow the personal greetings to be played to the caller.

If the personal greeting is not recorded or is unavailable, no prompt will be played.

Personal Greetings will not be applicable if your mailbox is defined as the destination for Call Tapping or Conversation Recording messages.

- Select the **Play Conditional Greeting** check box to allow the conditional greetings to be played to the caller.

If the conditional greeting is not recorded or is unavailable, no prompt will be played.

Conditional Greeting will be applicable only if your mailbox is defined as the destination for Call Transfer Unsuccessful.

- In **Stop Record Message Code**, enter the digits (0-9,* or #) that you want the caller to dial to stop the recording of the message. The system will play this to the caller along with the prompt before he starts recording the message. You may configure a maximum of upto 3 digits.

If you do not assign any digit and keep it blank, the system will play the prompt without the stop code.

- Select the **Message Verification** check box to allow the recorded message to be verified by the caller before storing it in the Personal Mailbox or re-record it. You must configure the "Message Leave Options".

  If you disable the check box, the system will directly store the recorded message in the Personal Mailbox.

- In **Message Type**, select the option according to which you want the system to store the recorded messages. You may select — Set as Normal, Set as Urgent or Ask caller.

  - Select **Set as Normal**, if you want the system to store the recorded message in the mailbox as Normal.
  - Select **Set as Urgent**, if you want the system to store the recorded message in the mailbox as Urgent.
  - Select **Ask Caller**, if you want the system to ask the caller to select the message type before storing the recorded message in the mailbox.

- In **Message Sensitivity**, select the option according to which you want the system to prioritize the messages as per their importance. You may select — Set as Normal, Set as Private or Ask caller.

  - Select **Set as Normal**, if you want the system to store the recorded message in the mailbox as Normal.
  - Select **Set as Private**, if you want the system to store the recorded message in the mailbox as Private. The message received is considered as confidential and forwarding the message is restricted.
  - Select **Ask Caller**, if you want the system to ask the caller to select the message sensitivity before storing the recorded message in the mailbox.

- Select the **Message Security** check box, if you want the system to restrict the forwarding and downloading of the messages.

Message Security has higher priority than Message Type and Message Sensitivity set for the message.

If you disable the check box, the system will not set any security for the message.

- Select the **Message Leave Confirmation Prompt** check box if you want the system to play the Message Leave Confirmation prompt to the caller after leaving the message. The Message Leave Confirmation prompt plays the Message Type, Message Sensitivity and Message Security that you configured.

  If you disable the check box, the system will not play the Message Leave Confirmation prompt to the caller after leaving the message.

Message Leave Options

You can assign the Digits (0-9, *, # or None) to each of the options — Re-record, Confirm, Urgent, Private(Confidential), Listen Recorded Message and Append to Recorded Message.

Make sure you do not assign the same digit to multiple options.

The digit you configure will be played as the digit to be dialed by the caller to select the respective option. The options will be played in the sequence — 1-9, 0, *, #. The option for which the Digit selected is None will not be played.

After the digits have been assigned, the system will proceed further as per the options selected by you. Given below is the brief description of each option:

- **Re-record**: The system will clear the recorded message and let the caller record a new message again. This is applicable only if you have enabled the Message Verification check box.

- **Confirm**: The system will save the recorded message as per the option set by you in No Digit Dialed Action. This is applicable only if you have enabled the Message Verification check box or selected the Ask Caller as the Message Type and/or Message Sensitivity option.

- **Urgent**: The system will save the recorded message along with the Message Type as *Urgent*. This is applicable only if you have selected the Ask Caller as the Message Type option.

- **Private(Confidential)**: The system will save the recorded message along with the Message Sensitivity as *Private*. This is applicable only if you have selected the Ask Caller as the Message Sensitivity option.

- **Listen Recorded Message**: The system will play the recorded message to the caller and then play the message leave options again. This is applicable only if you have enabled the Message Verification check box.

- **Append to Recorded Message**: The system will let the caller record a new message again and add it to the existing recorded message. This is applicable only if you have enabled the Message Verification check box.

System will check — Maximum Message Length, New Message Delivery Option in Mailbox Full Condition etc — before allowing the caller to append the recorded message

The Message, that is, the recorded message along with the appended message cannot exceed the Maximum Message Length. Hence, the system will allow recording of the appending message only till it reaches the Maximum Message Length.

- In **No Digit Dialed Action**, if no digit is dialed by the caller, the system will take action as per the Message Type and Message Sensitivity configured.

  If you have configured *Ask Caller* option in Message Type and/or Message Sensitivity, you can select the desired action — Normal+Normal, Urgent+Normal, Normal+Private or Urgent+Private.

No Mailbox Action

No Mailbox Action is applicable if you have enabled *Overide Message Leave Settings of Extension* parameter.

- In **No Mailbox Action**, you can select — Disconnect, Transfer to Operator, Transfer to Extension, Transfer to Department Group, Go to VMAA Menu.

- If you select **Transfer to Operator** or **Transfer to Department Group** or **Transfer to Extension**, select the respective Operator number/Department Group number or enter the desired extension number.

  - In **Call Transfer Profile**, select the desired Call Transfer Profile as per your requirement. By default, *As configured for Transfer Number* is selected.

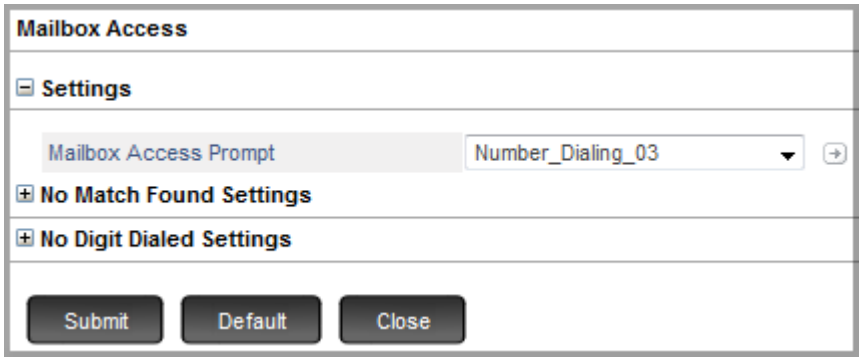
    Click **Settings** to configure the parameters of the selected profile. For more information, see "Call Transfer Profile".

- If you select **Go to VMAA Menu**, select the desired VMAA Menu.

- If you select **Disconnect**, the system will disconnect the call.

- Click **Close** to close the window.

Mailbox Access



Settings

- In **Mailbox Access Prompt**, select the prompt you wish to play when the caller has selected *Personal Mailbox Access* as the Action on Digit Dialed option.

  Select **None**, if you do not wish to play any prompt.

  - Click **Settings**. The **Prompts Management** page opens.

    You may add a new prompt from here. For further details, see "Prompts Management".

  Once the files are uploaded these appear as options for Mailbox Access Prompt.

No Match Found Settings

- For details, refer "No Match Found Settings".

No Digit Dialed Settings

- For details, refer "No Digit Dialed Settings".

- Click **Close** to close the window.

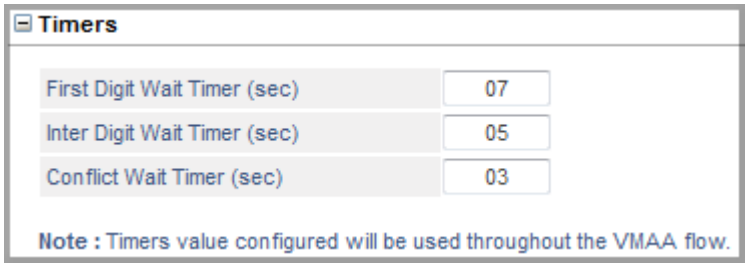
No Digit Dialed Settings

- For details, refer “No Digit Dialed Settings”.

Invalid Digit Dialed Settings

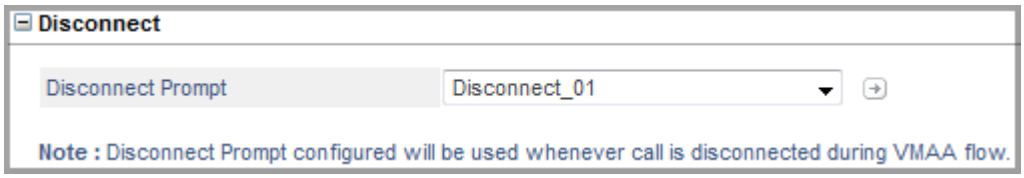
- For details, refer “Invalid Digit Dialed Settings”.

Timers



- In **First Digit Wait Timer**, enter the time for which you want the system to wait for the caller to dial the first digit for menu option selection or for entering the Extension or Department Group number. On the expiry of the First Digit Wait Timer, the system will apply “No Digit Dialed Settings”. Default: 07 seconds
- In **Inter Digit Wait Timer**, enter the time for which you want the system to wait for the caller to dial the subsequent digit after the first digit is dialed. On the expiry of the Inter Digit Wait Timer, the system will apply “No Match Found Settings”. Default: 05 seconds
- In **Conflict Wait Timer**, enter the time for which you want the system to wait for the extension user to dial the next digit to resolve conflicting access codes dialed by the extension user. Default: 03 seconds

Disconnect



- In **Disconnect Prompt**, select the prompt you wish to play when the caller has selected *Disconnect* as the Action on Digit Dialed option.

  Select **None**, if you do not wish to play any prompt.

  - Click **Settings** . The **Prompts Management** page opens.

    You may add a new prompt from here. For further details, see “Prompts Management”.

  Once the files are uploaded these appear as options for Disconnect Prompt.

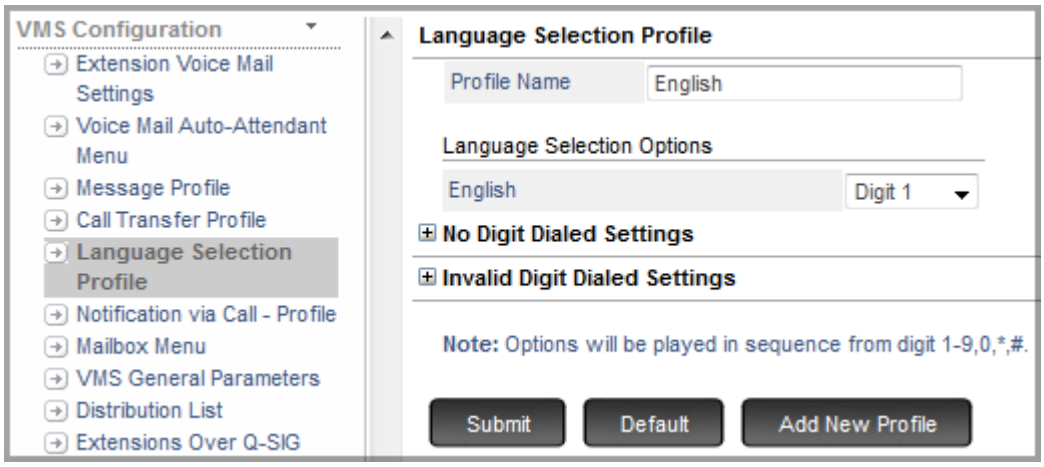
Language Selection Profile

Language Selection Profile is a menu which allows you to select the language. The languages you configure in the "Supported VMS Language" will be displayed under Language selection options. You may add a maximum of 8 profiles.

If Language Selection Profile assigned to any parameter is deleted, the system will consider the first Language Selection Profile.

How to Configure

- Log in as System Engineer.
- Under **Configuration**, click **VMS Configuration**.
- Click the **Language Selection Profile**.



You may add a new profile, edit the default profile or delete a Language Profile.

To add a new Language Profile,

- Click **Add New Profile**. A new *Language Profile x* will be created. You may now configure this as per your requirement.

To delete a profile, click **Delete**.

The default Language Profile cannot be deleted.

To edit a Language Profile,

- In **Profile Name**, you may assign a name to the Language profile. By default, it is *Language Selection x,* where x is the Language Profile Number from 1 to 8.

Language Selection Options

The languages that are added in the system are displayed here. For details, see “Configuring VMS General Parameters”.

You can assign the Digits(0-9, *, # or None) to each of the Languages.

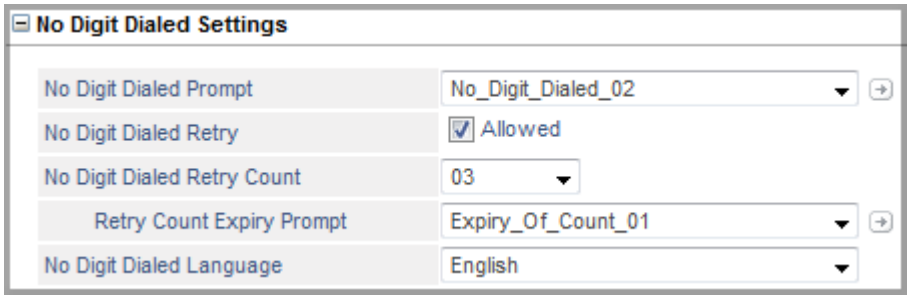
The digit you configure will be played as the digit to be dialed by the caller to select the respective language. The options will be played in the sequence — 1-9, 0, *, # — as per the Digits configured. The language for which the Digit selected is None will not be played.

The option you configure for a language will be played in the respective language.

For Example, you selected the Digit 5 to be played for Italian language. The option will be played along with the digit to dial (as configured) in Italian Language.

No Digit Dialed Settings

No Digit Dialed Settings are applicable when caller has not dialed any digit to select the desired Language and the first digit wait timer has expired.



- In **No Digit Dialed Prompt**, select the prompt you wish to play when caller has not dialed any digit for option selection.

  Select **None**, if you do not wish to play any prompt.

  - Click **Settings** . The **Prompts Management** page opens.

    You may add a new prompt from here. For further details, see “Prompts Management”.

  Once the files are uploaded these appear as options for No Digit Dialed Prompt.

- Select the **No Digit Dialed Retry** check box to allow the Language selection options to be played again if no digit has been dialed.

  - In **No Digit Dialed Retry Count**, select the number of times you wish to prompt the caller for language selection. The language selection options will be played repeatedly till the Retry Count expires.

  - In **Retry Count Expiry Prompt**, select the prompt you wish to play when the Retry Count expires.

    Select **None**, if you do not wish to play any prompt.

    - Click **Settings** . The **Prompts Management** page opens.

You may add a new prompt from here. For further details, see “Prompts Management”.

Once the files are uploaded these appear as options for Retry Count Expiry Prompt.

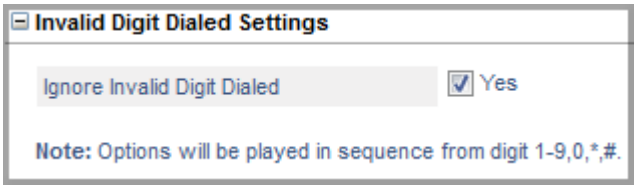
- In **No Digit Dialed Language**, select — Use Language configured in Menu, Disconnect, or the language configured as Supported VMS Language.

  - Select **Use Language configured in Menu**, if you want the system to use the Language you configured as Menu Language in VMAA Menu.
  - Select **Disconnect**, if you want the system to disconnect the call.

  If you select a language from the Supported VMS Languages you configured, the system will use the Language you select. See “Supported VMS Language”.

Invalid Digit Dialed Settings

Invalid Digit Dialed Settings are applicable when caller has dialed an invalid digit — a digit which is not configured as the Language Selection option.



- Clear the **Ignore Invalid Digit Dialed** check box, if you do not want the system to ignore the invalid digit dialed and prompt the caller for the same. By default, it is enabled.

  - In **Invalid Digit Dialed Prompt**, select the prompt you wish to play when the caller has dialed an invalid digit for option selection.

    Select **None**, if you do not wish to play any prompt when an invalid digit is dialed.

    - Click **Settings** . The **Prompts Management** page opens.

      You may add a new prompt from here. For further details, see “Prompts Management”.

    Once the files are uploaded these appear as options for Invalid Digit Dialed Prompt.

  - Select the **Invalid Digit Dialed Retry** check box to allow the Language selection options to be played again if an invalid digit has been dialed.

    - In **Invalid Digit Dialed Retry Count**, select the number of times you wish to prompt the caller for language selection. The language selection options will be played repeatedly till the Retry Count expires.

    - In **Retry Count Expiry Prompt**, select the prompt you wish to play when the Retry Count expires.

      Select **None**, if you do not wish to play any prompt.

      - Click **Settings** . The **Prompts Management** page opens.

You may add a new prompt from here. For further details, see “Prompts Management”.

Once the files are uploaded these appear as options for Retry Count Expiry Prompt.

- In **Invalid Digit Dialed Language**, select — Use Language configured in Menu, Disconnect, or the language configured as Supported VMS Language.
  - Select **Use Language configured in Menu**, if you want the system to use the Language you configured as Menu Language in VMAA Menu.
  - Select **Disconnect**, if you want the system to disconnect the call.
  - Select a language from the Supported VMS Languages you configured, if you want the system to use the Language you select. See “Supported VMS Language”.

Message Profile

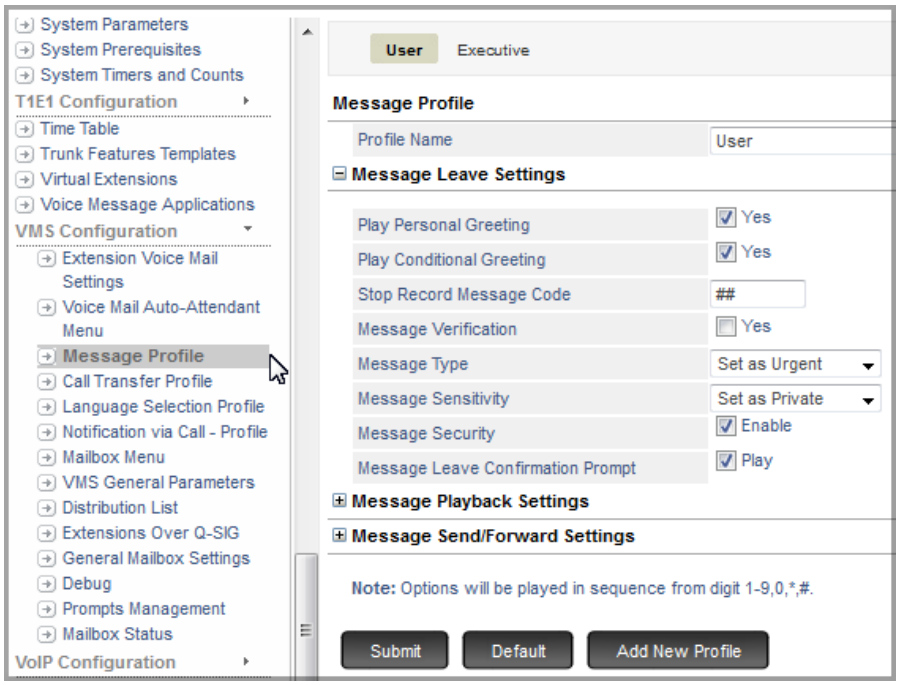
Message Profile allows you to customize the Message Leave Settings, Message Playback Settings and Message Sent/Forward Settings.

VMS supports a maximum of 24 Message Profiles.

By default, two Message Profiles — User and Executive — are provided to you. These two profiles cannot be deleted but you may edit their settings as per your requirement.

How to Configure

- Log in as System Engineer.
- Under **Configuration**, click **VMS Configuration**.
- Click **Message Profile**.



You may add a new profile, edit the default profile or delete a message profile.

To add a new Message Profile,

- Click **Add New Profile**. A new *Message Profile xx* will be created. You may now configure this as per your requirement.

To delete a profile, click **Delete**.

The default Message Profiles cannot be deleted.

To edit a Message Profile,

- In **Profile Name**, you may configure the name of the Message profile you want. By default, it is *Message Profile xx* where xx is the Message Profile Number from 01 to 24.

Message Leave Settings

Message Leave Settings are applicable only when the external caller reaches to your mailbox or internal caller replies to your message.

- Select the **Play Personal Greeting** check box to allow the personal greetings to be played to the caller. The Personal Greetings will be played as per the timezone - Working Hours, Break hours or Non-Working hours.

If the personal greeting is not recorded or is unavailable, no prompt will be played.

- Select the **Play Conditional Greeting** check box to allow the conditional greetings to be played to the caller. The conditional greetings will be played when the call forward is set to VMS for Busy, No Reply or Unconditional.

If the conditional greeting is not recorded or is unavailable, no prompt will be played.

- In **Stop Record Message Code**, enter the digits (0-9,* or #) that you want the caller to dial to stop the recording of the message. The system will play this to the caller along with the prompt before he starts recording the message. You may configure a maximum of upto 3 digits.

If you do not assign any digit and keep it blank, the system will play the prompt without the stop code. In this case, you may dial any digit to stop recording.

- Select the **Message Verification** check box to allow the recorded message to be verified by the caller before storing it in the Personal Mailbox or to re-record it. You must configure the “Message Leave Options”.

  If you disable the check box, the system will directly store the recorded message in the Personal Mailbox.

- In **Message Type**, select the option according to which you want the system to store the received messages as per the priority. You may select — Set as Normal, Set as Urgent or Ask caller.

  - Select **Set as Normal**, if you want the system to store the messages received in the mailbox as Normal. The messages left by the external callers or replied by the internal callers will be considered as normal messages.

  - Select **Set as Urgent**, if you want the system to store the messages received in the mailbox as Urgent. This is useful when you want the messages left by the external callers or replied by the internal callers to be considered as urgent messages with higher priority.

  - Select **Ask Caller**, if you want the system to ask the caller to select the message type before leaving the recorded message on your extension. If the call gets disconnected before the caller selects the message type, the message will be set as normal by default. You must configure the “Message Leave Options”.

  While retrieving, the Urgent and Normal messages will be played separately.

- In **Message Sensitivity**, select the option according to which the system will decide whether the messages left by the external callers or replied by the internal callers are allowed to be forwarded or not.

  - Select **Set as Normal**, if you want the system to store the messages received in the mailbox as Normal. The messages left by the external callers or replied by the internal callers will be considered as normal messages.

  - Select **Set as Private**, if you want the system to store the messages received in the mailbox as Private. The message left by the external callers or replied by the internal callers will be considered as confidential and forwarding the message will be restricted.

  - Select **Ask Caller**, if you want the system to ask the caller to select the message sensitivity before storing the recorded message in the mailbox. If the call gets disconnected before the caller selects the message sensitivity, the message will be set as normal by default. You must configure the "Message Leave Options".

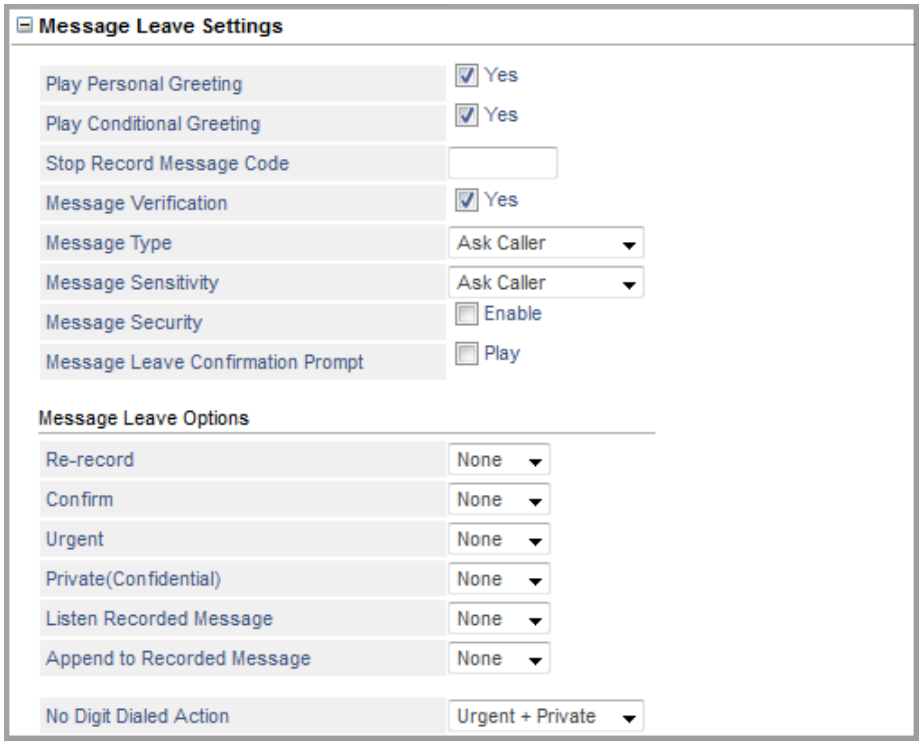
- The **Message Security** parameter is reserved for future use. Keep this check box disabled.

- Select the **Message Leave Confirmation Prompt** check box if you want the system to play the prompt to the caller after leaving the message. The Message Type and Message Sensitivity that you configured will be played.

  If you disable the check box, the system will not play the Message Leave Confirmation prompt to the caller after leaving the message.

Message Leave Options

Message Leave Options will be played based on the configuration of the parameters — Message Verification, Message Type and Message Sensitivity.



You can assign the Digits (0-9, *, # or None) to each of the options — Re-record, Confirm, Urgent, Private(Confidential), Listen Recorded Message and Append to Recorded Message.

Make sure you do not assign the same digit to multiple options.

The digit you configure will be played as the digit to be dialed by the caller to select the respective option. The options will be played in the sequence — 1-9, 0, *, #. The option for which the digit selected is None will not be played.

After the digits have been assigned, the system will proceed further as per the options selected by you. Given below is the brief description of each option:

- **Re-record**: The system will clear the recorded message and let the caller record a new message again. This is applicable only if you have enabled the Message Verification check box.

- **Confirm**: The system will save the recorded message as per the option set by you in No Digit Dialed Action. This is applicable only if you have enabled the Message Verification check box or selected the Ask Caller as the Message Type and/or Message Sensitivity option.

- **Urgent**: The system will save the recorded message as Urgent message. This is applicable only if you have selected the Ask Caller as the Message Type option.

- **Private(Confidential)**: The system will save the recorded message as Private message. This is applicable only if you have selected the Ask Caller as the Message Sensitivity option.

- **Listen Recorded Message**: The system will play the recorded message to the caller and then play the message leave options again. This is applicable only if you have enabled the Message Verification check box.

- **Append to Recorded Message**: The system will let the caller record a new message again and add it to the existing recorded message. This is applicable only if you have enabled the Message Verification check box.

System will check — Maximum Message Length, New Message Delivery Option in Mailbox Full Condition etc — before allowing the caller to append the recorded message.

The Message, that is, the recorded message along with the appended message cannot exceed the Maximum Message Length. Hence, the system will allow recording of the appending message only till it reaches the Maximum Message Length.

- In **No Digit Dialed Action**, if no digit is dialed by the caller, the system will take action as per the Message Type and Message Sensitivity configured.

  This is applicable only if you have enabled the Message Verification check box or selected the Ask Caller as the Message Type and/or Message Sensitivity option.

  Based on the configuration, you may select the desired action — Normal+Normal, Urgent+Normal, Normal+Private or Urgent+Private.

Message Playback Settings

Message Playback Settings allows you to configure the settings applicable when extension user accesses his/her Personal Mailbox and plays the messages.

- In **Message Playback Direction**, select the option according to which you want the system to play the messages to the extension user. You may select — Play oldest recorded message first or Play new recorded message first.
    - Select **Play oldest recorded message first**, if you want the system to play the old messages first and then play the new messages.
    - Select **Play new recorded message first**, if you want the system to play the new messages first and then play the old messages.
- In **New Message Playback**, select the option defining the priority according to which you want the system to play the new messages to the extension user. You may select — Play Normal Messages and thereafter Urgent Messages or Play Urgent Messages first and thereafter Normal Messages.
    - Select **Play Normal Messages and thereafter Urgent Messages**, if you want the system to first play the new messages with message type as Normal and then play the new messages with the message type as Urgent.
    - Select **Play Urgent Messages first and thereafter Normal Messages**, if you want the system to first play the new messages with message type as Urgent and then play the new messages with the message type as Normal.
- In **Old Message Playback**, select the option defining the priority according to which you want the system to play the old messages to the extension user. You may select — Play Normal Messages and thereafter Urgent Messages or Play Urgent Messages first and thereafter Normal Messages.
    - Select **Play Normal Messages and thereafter Urgent Messages**, if you want the system to first play the old messages with message type as Normal and then play the old messages with the message type as Urgent.
    - Select **Play Urgent Messages first and thereafter Normal Messages**, if you want the system to first play the old messages with message type as Urgent and then play the old messages with the message type as Normal.
- **Date Playback Format:** Select the Date Playback Format — DD-MM-YYYY or MM-DD-YYYY— you want the system to play while playing the message details to the extension user.

*This Date Playback format is applicable only for **Message Profile**.*

- In **Message Type and Sensitivity**, select the option according to which you want the system to play the Message Type and Sensitivity of the stored messages to the extension user. You may select — Do Not Play, Play before Message or Play after Message.
    - Select **Do Not Play**, if you do not want the system to play the Message Type and Sensitivity.
    - Select **Play before Message**, if you want the system to play the Message Type and Sensitivity before playing the message.
    - Select **Play after Message**, if you want the system to play the Message Type and Sensitivity after playing the message.
- In **Message Details**, select the option according to which you want the system to play the Message Details of the stored messages to the extension user. You may select — Play before Message, Play after Message or Play on Demand.
    - Select **Play before Message**, if you want the system to play the Message Details before playing the message.
    - Select **Play after Message**, if you want the system to play the Message Details after playing the message.
    - Select **Play on Demand**, if you want the system to play the Message Details only when the caller wants. You must select the desired option to listen to the message details.

The System will first play the Message Type and Sensitivity and then play the Message Details.

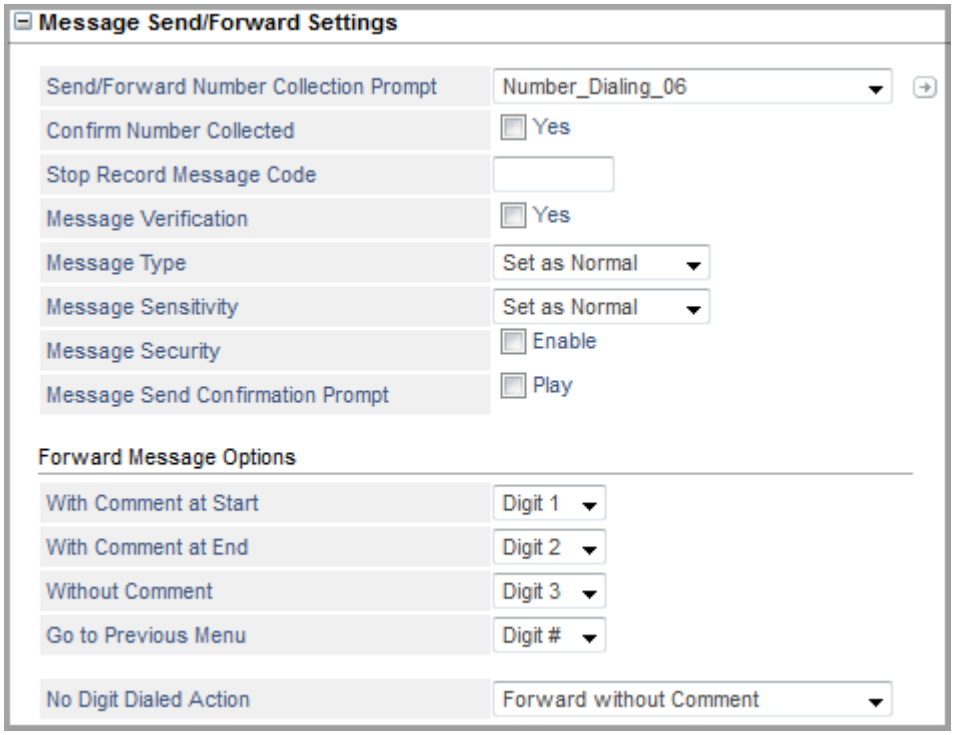
- Select the **Message Count** check box if you want the system to play the Message sequence number to the extension user before playing the message. The message count is just to differentiate between various messages played to the caller.

The message count has no interaction with the number of old/new/total messages present in the mailbox.

If you disable the check box, the system will not play the Message sequence number before playing the message to the extension user.

Message Send/Forward Settings

Message Send/Forward Settings allows you to configure the settings for the messages you send or forward.



- In **Send/Forward Number Collection Prompt**, select the prompt which you wish to play when you have selected the Send Message or Forward Message option. This is to prompt the extension user for entering the destination number/s for sending/forwarding the message.

  Select **None**, if you do not wish to play any prompt.

  - Click **Settings** . The **Prompts Management** page opens.

    You may add a new prompt from here. For further details, see “Prompts Management”.

  Once the files are uploaded these appear as options for Send/Forward Number Collection Prompt.

- Select the **Confirm Number Collected** check box to allow the system to check the Destination Number/s collected by the extension user. Configure the “Destination Number Confirmation” settings.

- In **Stop Record Message Code**, enter the digits (0-9,* or #) that you want the extension user to dial to stop the recording of the message. The system will play this to the extension user along with the prompt before s/he starts recording the message. You may configure a maximum of upto 3 digits.

If you do not assign any digit and keep it blank, the system will play the prompt without the stop code. In this case, you may dial any digit to stop recording.

- Select the **Message Verification** check box to allow the recorded message to be verified by the extension user before storing it in the Personal Mailbox or re-record it.

  If you disable the check box, the system will directly store the recorded message in the Personal Mailbox.

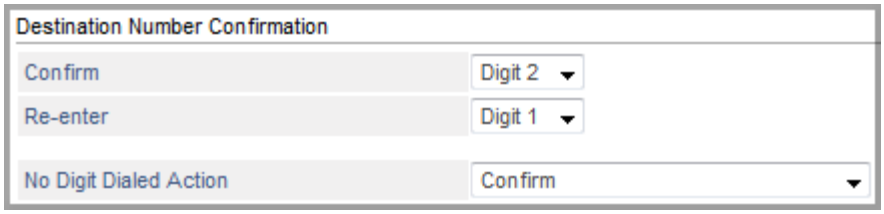
- In **Message Type**, select the option according to which you want the system to send/forward the recorded messages. You may select — Set as Normal, Set as Urgent or Ask caller.
    - Select **Set as Normal**, if you want the system to send/forward the recorded message in the mailbox as Normal.
    - Select **Set as Urgent**, if you want the system to send/forward the recorded message in the mailbox as Urgent with higher priority.
    - Select **Ask Caller**, if you want the system to ask the extension user to select the message type before storing the recorded message in the mailbox. If the call gets disconnected before the message type is selected, the message will be set as normal by default. You must configure the "Message Send Options".

    While retrieving, the Urgent and Normal messages will be played separately.
- In **Message Sensitivity**, select the option according to which you want the system to prioritize the messages as per their importance. You may select — Set as Normal, Set as Private or Ask caller.
    - Select **Set as Normal**, if you want the system to send/forward the recorded message in the mailbox as Normal.
    - Select **Set as Private**, if you want the system to send/forward the recorded message in the mailbox as Private. The message received is considered as confidential and forwarding the message is restricted.
    - Select **Ask Caller**, if you want the system to ask the extension user to select the message sensitivity before storing the recorded message in the mailbox. If the call gets disconnected before the message sensitivity is selected, the message will be set as normal by default. You must configure the "Message Send Options".
- The **Message Security** parameter is reserved for future use. Keep this check box disabled.
- Select the **Message Send Confirmation Prompt** check box if you want the system to play the prompt to the extension user after sending the message on the destination number/s i.e. to the recipients. The Message Type and Message Sensitivity that you configured will be played.

  If you disable the check box, the system will not play the Message Send Confirmation prompt to the extension user after sending the message.

Destination Number Confirmation

This option is applicable only if you have selected the **Confirm Number Collected** check box.



You can assign the Digits (0-9, *, # or None) to each of the options — Confirm, Re-enter.

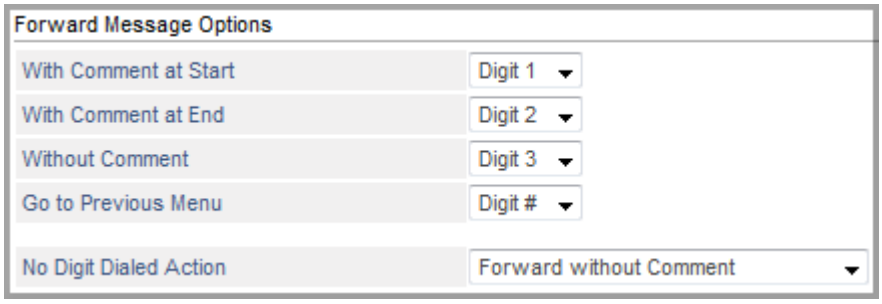
Make sure you do not assign the same digit to both the options.

The digit you configure will be played as the digit to be dialed by the extension user to select the respective option. The options will be played in the sequence — 1-9, 0, *, #. The option for which the Digit selected is None will not be played.

After the digits have been assigned, the system will proceed further as per the options selected by you. Given below is a brief description of each option:

- **Confirm**: The system will save the destination number/s collected considering they are valid.
- **Re-enter**: The system will clear all the destination numbers collected and ask the extension user to record the destination number/s again.
- In **No Digit Dialed Action**, select the option — Confirm or Re-enter. This is applicable when the extension user has not dialed any digit for Destination Number Confirmation. The system will function as per the option you select.

Forward Message Option



You can assign the Digits (0-9, *, # or None) to each of the options — With Comment at Start, With Comment at End, Without Comment or Go to Previous Menu.

Make sure you do not assign the same digit to multiple options.

The digit you configure will be played as the digit to be dialed by the extension user to select the respective option. The options will be played in the sequence — 1-9, 0, *, #. The option for which the Digit selected is None will not be played.

After the digits have been assigned, the system will proceed further as per the options selected by you. Given below is a brief description of each option:

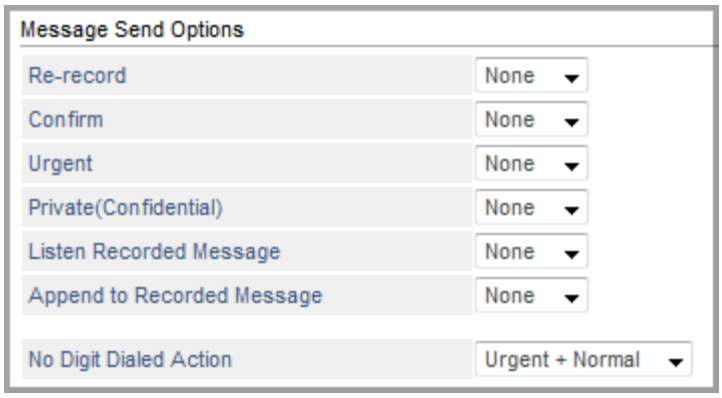
- **With Comment at Start**: The system will provide an option to the extension user for recording a message in addition to the existing recorded message before forwarding the message to the destination number/s. The additional message will be added as the initial part of the message i.e. will be added at the beginning of the message.
- **With Comment at End**: The system will provide an option to the extension user for recording a message in addition to the existing recorded message before forwarding the message to the destination number/s. The additional message will be added as the final part of the message i.e. will be added at the end of the message.
- **Without Comment**: The system will provide an option to the extension user for sending the recorded message to the destination number/s without adding any additional comment to it.

- **Go to Previous Menu**: The system will provide an option to the extension user to go back to the Mailbox Access Main Menu.

- In **No Digit Dialed Action**, select the option — With Comment at Start, With Comment at End or Without Comment. This is applicable when the extension user has not dialed any digit for Forward Message Option. The system will carry out the function when no digit is dialed as per the option you select.

Message Send Options

Message Send Options will be played based on the configuration of the parameters — Message Verification, Message Type and Message Sensitivity.



You can assign the Digits (0-9, *, # or None) to each of the options — Re-record, Confirm, Urgent, Private(Confidential), Listen Recorded Message and Append to Recorded Message.

Make sure you do not assign the same digit to multiple options.

The digit you configure will be played as the digit to be dialed by the extension user to select the respective option. The options will be played in the sequence — 1-9, 0, *, #. The option for which the digit selected is None will not be played.

After the digits have been assigned, the system will proceed further as per the options selected by you. Given below is the brief description of each option:

- **Re-record**: The system will clear the recorded message and let the extension user record a new message again.
  This is applicable only if you have enabled the Message Verification check box.

- **Confirm**: The system will save the recorded message as per the option set by you in No Digit Dialed Action. This is applicable only if you have enabled the Message Verification check box or selected the Ask Caller as the Message Type and/or Message Sensitivity option.

- **Urgent**: The system will save the recorded message as Urgent message. This is applicable only if you have selected the Ask Caller as the Message Type option.

- **Private(Confidential)**: The system will save the recorded message as Private message. This is applicable only if you have selected the Ask Caller as the Message Sensitivity option.

- **Listen Recorded Message**: The system will play the recorded message to the extension user and then play the message leave options again. This is applicable only if you have enabled the Message Verification check box.

- **Append to Recorded Message**: The system will let the extension user record a new message again and add it to the existing recorded message. This is applicable only if you have enabled the Message Verification check box.

System will check — Maximum Message Length, New Message Delivery Option in Mailbox Full Condition etc — before allowing the extension user to append the recorded message.

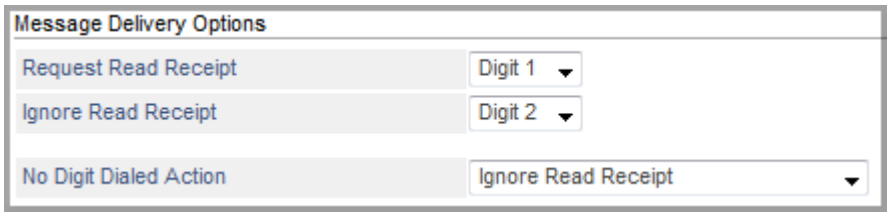
The Message, that is, the recorded message along with the appended message cannot exceed the Maximum Message Length. Hence, the system will allow recording of the appending message only till it reaches the Maximum Message Length.

- In **No Digit Dialed Action**, if no digit is dialed by the extension user, the system will take action as per the Message Type and Message Sensitivity configured.

  This is applicable only if you have enabled the Message Verification check box or selected the Ask Caller as the Message Type and/or Message Sensitivity option.

  Based on the configuration, you may select the desired action — Normal+Normal, Urgent+Normal, Normal+Private or Urgent+Private.

Message Delivery Options



You can assign the Digits (0-9, *, # or None) to each of the options — Request Read Receipt or Ignore Read Receipt.

Make sure you do not assign the same digit to multiple options.

The digit you configure will be played as the digit to be dialed by the extension user to select the respective option. The options will be played in the sequence — 1-9, 0, *, #. The option for which the Digit selected is None will not be played.

After the digits have been assigned, the system will proceed further as per the options selected by you. Brief description of each option is explained below:

- **Request Read Receipt**: The system will provide an option to the extension user for requesting the read receipt to be played whenever the message is read by the destination number. In case of multiple destination numbers, it will be played individually for each number after the message has been read.

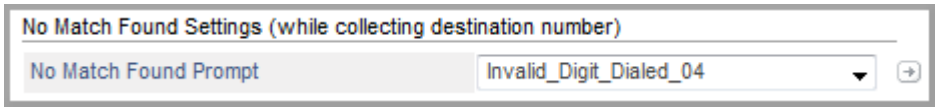
  The read receipt includes the name, destination number and the first five seconds of the message sent or forwarded.

- **Ignore Read Receipt**: The system will provide an option to the extension user for ignoring the read receipt i.e. no read receipt will be played back to the extension user.

- In **No Digit Dialed Action**, select the option — Request Read Receipt, Ignore Read Receipt. This is applicable when the extension user has not dialed any digit for Message Delivery Option. The system will carry out the function when no digit is dialed as per the option you select.

No Match Found Settings (while collecting destination number)

No Match Found Settings are applicable when no valid match is found as the Destination Number.



- In **No Match Found Prompt**, select the prompt which you wish to play when the extension user has dialed an invalid destination number/s for sending/forwarding the message.

  Select **None**, if you do not wish to play any prompt when extension user has dialed an invalid destination number.

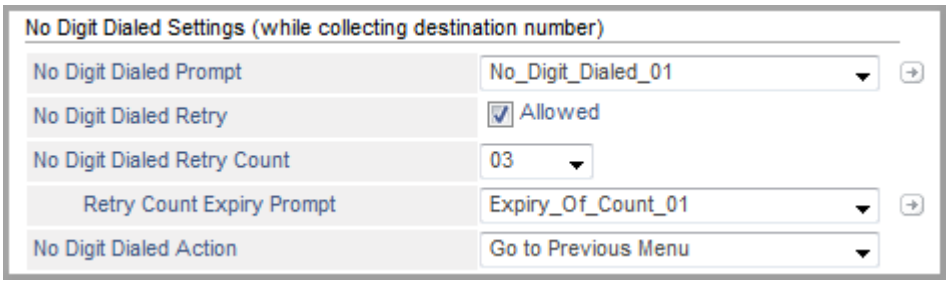
  - Click **Settings** . The **Prompts Management** page opens.

    You may add a new prompt from here. For further details, see “Prompts Management”.

  Once the files are uploaded these appear as options for No Match Found Prompt.

No Digit Dialed Settings (while collecting destination number)

No Digit Dialed Settings are applicable when extension user has not dialed the destination number and the first digit wait timer has expired.



- In **No Digit Dialed Prompt**, select the prompt which you wish to play when extension user has not dialed any digit for destination number.

  Select **None**, if you do not wish to play any prompt when extension user has not dialed any digit.

  - Click **Settings** . The **Prompts Management** page opens.

    You may add a new prompt from here. For further details, see “Prompts Management”.

  Once the files are uploaded these appear as options for No Digit Dialed Prompt.

- Select the **No Digit Dialed Retry** check box to prompt the extension user to dial the Destination Number again.
    - In **No Digit Dialed Retry Count**, select the number of times you wish to prompt the extension user for dialing the destination number. The prompt will be played repeatedly till the Retry Count expires.
    - In **Retry Count Expiry Prompt**, select the prompt you wish to play when the Retry Count expires.

      Select **None**, if you do not wish to play any prompt.

        - Click **Settings** . The **Prompts Management** page opens.

          You may add a new prompt from here. For further details, see “Prompts Management”.

      Once the files are uploaded these appear as options for Retry Count Expiry Prompt.
- In **No Digit Dialed Action**, select — Go to Previous Menu or Disconnect.
    - Select **Go to Previous Menu**, if you want the system to provide an option to the extension user to go back to the Mailbox Access Main Menu.
    - Select **Disconnect**, if you want the system to disconnect the call.

Call Transfer Profile

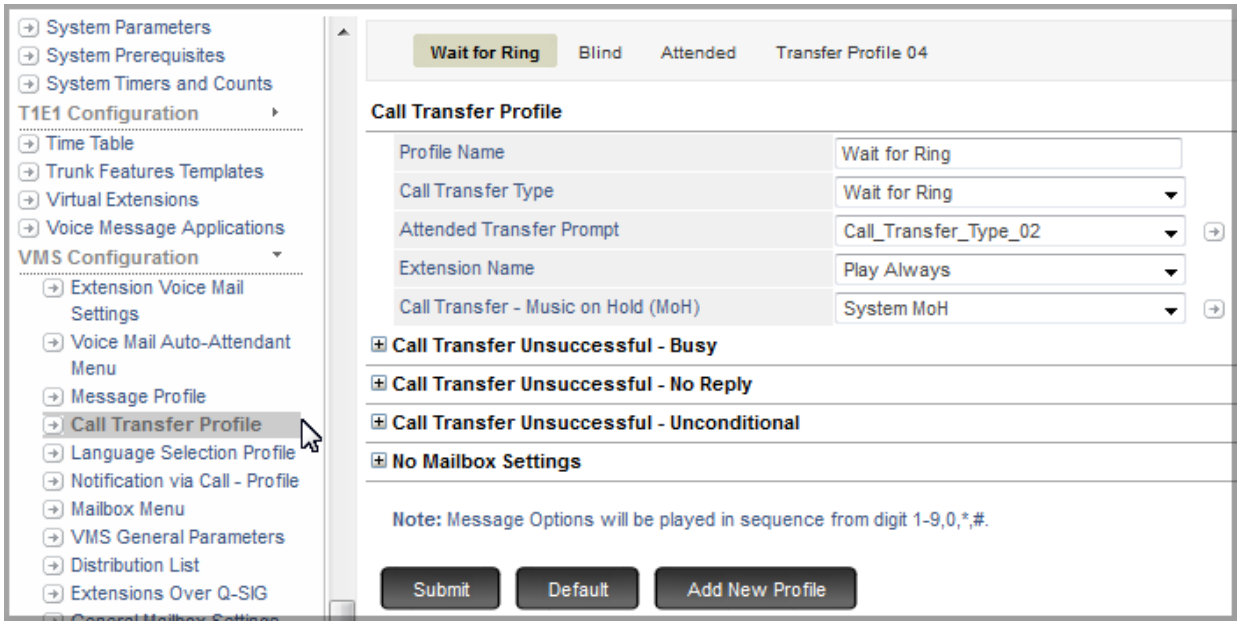
Call Transfer Profile allows you can select the Call Transfer Type — Blind, Wait for Ring, Wait for Answer, Screened, None — and customize parameters related to it as per your requirement.

VMS supports a maximum of 64 Call Transfer Profiles.

By default, three Call Transfer Profiles — Wait for Ring, Blind and Attended — are provided to you. These three profiles cannot be deleted but you may edit their settings as per your requirement.

How to Configure

- Log into Jeeves.
- Under **Configuration**, click **VMS Configuration**.
- Click **Call Transfer Profile**.



You may add a new profile, edit the default profile or delete a Call Transfer Profile.

To add a new Call Transfer Profile,

- Click **Add New Profile**. A new *Transfer Profile xx* will be created. You may now configure this as per your requirement.

To delete a profile, click **Delete**.

The default Call Transfer Profiles cannot be deleted.

To edit a Call Transfer Profile,

- In **Profile Name**, you may configure the name of the Call Transfer profile you want. By default, it is *Transfer Profile xx* where xx is the Call Transfer Profile Number from 01 to 64.
- In **Call Transfer Type**, you can select — None, Blind, Wait for Ring, Wait for Answer or Screened.
  - If you select **None**, the caller will be transfered to the transferor's mailbox directly. You must configure the "Leave Voice Mail" parameters.
  - If you select **Blind**, the caller will be transfered to the transfer number directly.
  - If you select **Wait for Ring,** the caller will be transfered to the transfer number after the number starts ringing.
  - If you select **Wait for Answer,** the caller will be transfered to the transfer number only after the transferor answers the call.
  - If you select **Screened**, the caller will be transfered only after the transferor confirms to speak to the caller. You must configure the "Call Transfer Type -Screened" parameters.
- In **Attended Transfer Prompt**, select the prompt you want the system to play while the call is being transfered. This is applicable only if you have selected Wait for Ring or Wait for Answer as the Call Transfer Type option.

  Select **None**, if you do not wish to play any prompt.

  - Click **Settings** . The **Prompts Management** page opens.

    You may add a new prompt from here. For further details, see "Prompts Management".

  Once the prompts are uploaded these appear as options for Attended Transfer Prompts.
- In **Blind Transfer Prompt**, select the prompt you want the system to play when the call is being transfered. This is applicable only if you have selected Blind as the Call Transfer Type option.

  Select **None**, if you do not wish to play any prompt. You may also add a new Prompt. To do so, follow the same steps as given in Attended Transfer Prompt.
- In **Screened Transfer Prompt**, select the prompt you want the system to play when the call is being transfered. This is applicable only if you have selected Screened as the Call Transfer Type option.

  Select **None**, if you do not wish to play any prompt. You may also add a new Prompt. To do so, follow the same steps as given in Attended Transfer Prompt.
- In **Wait for Answer Timer (sec)**, configure the time for which you want the system to wait before transferring the call. This is applicable only if you have selected Wait for Answer or Screened as the Call Transfer Type option.

*If the Wait for Answer Timer value is greater than Ring Back Tone timer, then Ring Back Tone Timer will expire first and the next action will be taken as per the **Call Transfer Unsuccessful - Unconditional** configuration.*

- In **Extension Name**, you can select — Do not play or Play Always.

  Select **Play Always**, if you want the system to play the recorded Extension Name as well as the respective Transfer prompt during the transfer.

  Select **Do not Play**, if you want the system to only play the respective Transfer prompt and not the Extension Name during the transfer.

- In **Call Transfer - Music On Hold (MOH)**, you may select System MoH or add a new MoH. By default, System MoH will be played. This parameter is applicable only if you have selected Wait for Ring or Wait for Answer as the Call Transfer Type.

  - Click **Settings** . The **Prompts Management** page opens.

    You may add a new prompt from here. For further details, see “Prompts Management”.

  Once the files are uploaded, these appear as options for Call Transfer - MoH.

- In **Action after Leaving Voice Mail**, you can select — Disconnect, Transfer to Operator, Transfer to Extension, Transfer to Department Group or Go to VMAA Menu. This parameter is applicable only if you have selected None as the Call Transfer Type.

  - If you select **Transfer to Operator** or **Transfer to Department Group** or **Transfer to Extension**,

    - Select the respective Operator number/Department Group number or enter the desired extension number.

    - In Call Transfer Profile, select the desired Call Transfer Profile as per your requirement. By default, Call Transfer Profile for Transfer Number is selected.

  - If you select **Go to VMAA Menu**, select the desired VMAA Menu.

  - If you select **Disconnect**, the system will disconnect the call.

Leave Voice Mail

This option is applicable only if you select None as the Call Transfer Type.

- Select the **Override Message Leave Settings of Extension** check box, if you want the system to apply the Leave Voice Mail Settings configured on this page.

  Clear the check box, if you want the system to apply the Leave Voice Mail settings configured for the extension.

Message Leave Settings

Message Leave Settings allows you to configure the settings for the messages you receive from the caller i.e the message left for you by the caller.

- Select the **Play Personal Greeting** check box to allow the personal greetings to be played to the caller.

If the personal greeting is not recorded or is unavailable, no prompt will be played.

Personal Greetings will not be applicable if your mailbox is defined as the destination for Call Tapping or Conversation Recording messages.

- Select the **Play Conditional Greeting** check box to allow the conditional greetings to be played to the caller.

If the conditional greeting is not recorded or is unavailable, no prompt will be played.

Conditional Greeting will be applicable only if your mailbox is defined as the destination for Call Transfer Unsuccessful.

- In **Stop Record Message Code**, enter the digits (0-9,* or #) that you want the caller to dial to stop the recording of the message. The system will play this to the caller along with the prompt before he starts recording the message. You may configure a maximum of upto 3 digits.

If you do not assign any digit and keep it blank, the system will play the prompt without the stop code.

- Select the **Message Verification** check box to allow the recorded message to be verified by the caller before storing it in the Personal Mailbox or re-record it. You must configure the "Message Leave Options".

  If you disable the check box, the system will directly store the recorded message in the Personal Mailbox.

- In **Message Type**, select the option according to which you want the system to store the recorded messages. You may select — Set as Normal, Set as Urgent or Ask caller.

  - Select **Set as Normal**, if you want the system to store the recorded message in the mailbox as Normal.
  - Select **Set as Urgent**, if you want the system to store the recorded message in the mailbox as Urgent.
  - Select **Ask Caller**, if you want the system to ask the caller to select the message type before storing the recorded message in the mailbox.

- In **Message Sensitivity**, select the option according to which you want the system to prioritize the messages as per their importance. You may select — Set as Normal, Set as Private or Ask caller.

  - Select **Set as Normal**, if you want the system to store the recorded message in the mailbox as Normal.
  - Select **Set as Private**, if you want the system to store the recorded message in the mailbox as Private. The message received is considered as confidential and forwarding the message is restricted.
  - Select **Ask Caller**, if you want the system to ask the caller to select the message sensitivity before storing the recorded message in the mailbox.

- Select the **Message Security** check box, if you want the system to restrict the forwarding and downloading of the messages.

Message Security has higher priority than Message Type and Message Sensitivity set for the message.

  If you disable the check box, the system will not set any security for the message.

- Select the **Message Leave Confirmation Prompt** check box if you want the system to play the Message Leave Confirmation prompt to the caller after leaving the message. The Message Leave Confirmation prompt plays the Message Type, Message Sensitivity and Message Security that you configured.

  If you disable the check box, the system will not play the Message Leave Confirmation prompt to the caller after leaving the message.

Message Leave Options

You can assign the Digits (0-9, *, # or None) to each of the options — Re-record, Confirm, Urgent, Private(Confidential), Listen Recorded Message and Append to Recorded Message.

Make sure you do not assign the same digit to multiple options.

The digit you configure will be played as the digit to be dialed by the caller to select the respective option. The options will be played in the sequence — 1-9, 0, *, #. The option for which the Digit selected is None will not be played.

After the digits have been assigned, the system will proceed further as per the options selected by you. Given below is the brief description of each option:

- **Re-record**: The system will clear the recorded message and let the caller record a new message again. This is applicable only if you have enabled the Message Verification check box.

- **Confirm**: The system will save the recorded message as per the option set by you in No Digit Dialed Action. This is applicable only if you have enabled the Message Verification check box or selected the Ask Caller as the Message Type and/or Message Sensitivity option.

- **Urgent**: The system will save the recorded message along with the Message Type as *Urgent*. This is applicable only if you have selected the Ask Caller as the Message Type option.

- **Private(Confidential)**: The system will save the recorded message along with the Message Sensitivity as *Private*. This is applicable only if you have selected the Ask Caller as the Message Sensitivity option.

- **Listen Recorded Message**: The system will play the recorded message to the caller and then play the message leave options again. This is applicable only if you have enabled the Message Verification check box.

- **Append to Recorded Message**: The system will let the caller record a new message again and add it to the existing recorded message. This is applicable only if you have enabled the Message Verification check box.

System will check — Maximum Message Length, New Message Delivery Option in Mailbox Full Condition etc — before allowing the caller to append the recorded message.

The Message, that is, the recorded message along with the appended message cannot exceed the Maximum Message Length. Hence, the system will allow recording of the appending message only till it reaches the Maximum Message Length.

- In **No Digit Dialed Action**, if no digit is dialed by the caller, the system will take action as per the Message Type and Message Sensitivity configured.

  If you have configured *Ask Caller* option in Message Type and/or Message Sensitivity, you can select the desired action — Normal+Normal, Urgent+Normal, Normal+Private or Urgent+Private.

Call Transfer Unsuccessful - Busy

If the called party is busy and has set Call Forward-Busy to VMS, the Call Transfer Unsuccessful - Busy parameters will be applicable. You can customize these parameters as per your requirement.

The VMS allows you to:

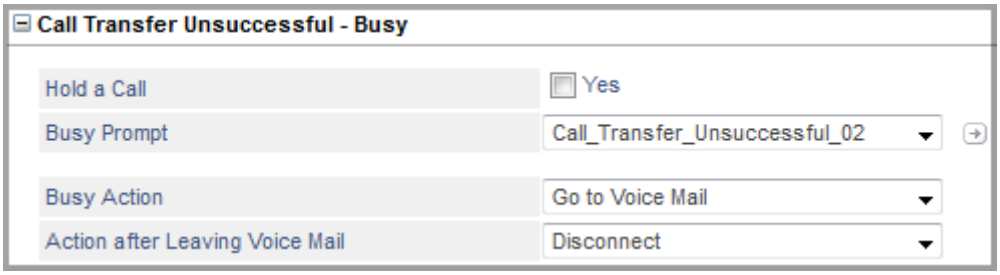
- process the call immediately for the busy condition as per the options you select.
  or
- process the call after being held for a certain duration as per the options you select.

Configure the following parameters, if you want the VMS to process the call immediately when the desired extension is busy,

- Busy Prompt
- Busy Action
- Action after Leaving Voice Mail

Configure the following parameters, if you want the VMS to put the call on hold when the desired extension is busy,

- Busy Prompt
- Busy Hold Prompt
- Provide Advanced Options during Hold
- Busy Extension Status check time interval (sec) and Time Interval Expiry Prompt
- Busy Extension Status check retry count and Retry Count Expiry Prompt
- Busy Hold - Music on Hold
- Busy Action
- Action after Leaving Voice Mail



- Select **Hold a Call** check box, if you want the call to be held. This option is applicable only for internal calls.
- In **Busy Prompt**, select the prompt you wish to play to the caller when the dialed extension is busy.

  Select **None**, if you do not wish to play any prompt.

  - Click **Settings** . The **Prompts Management** page opens.

    You may add a new prompt from here. For further details, see “Prompts Management”.

  Once the files are uploaded these appear as options for Busy Prompt.
- In **Busy Hold Prompt**, select the prompt you wish to play to the caller when put on hold as the dialed extension is busy.

  Select **None**, if you do not wish to play any prompt.

  - Click **Settings** . The **Prompts Management** page opens.

    You may add a new prompt from here. For further details, see “Prompts Management”.

  Once the files are uploaded these appear as options for Busy Hold Prompt.

  - Select **Provide Advanced Options during Hold** check box, if you want the system to play Advanced options to the caller.

You can assign the Digits (0-9, *, # or None) to each of the options — Leave Voice Mail, Transfer to Operator, Transfer to Assistant, Transfer to Alternate/Mobile Number, Dial Extension Number, Go to Main Menu, Go to Previous Menu and Disconnect.

Make sure you do not assign the same digit to multiple options.

The digit you configure will be played as the digit to be dialed by the caller to select the respective option. The options will be played in the sequence — 1-9, 0, *, #. The option for which the Digit selected is None will not be played.

After the digits have been assigned, the system will proceed further as per the options selected by you. Brief description of each option is explained below:

- **Leave Voice Mail:** After you assign the digit, click **Settings** to configure the parameters.
  - Select the **Override Message Leave Settings of Extension** check box, if you want the system to apply Message Leave Settings customized on this page and not that of the extension or department group number.

    Clear the check box if you want the system to apply Message Leave Settings configured for the extension or department group number.
- **Transfer to Operator:** After you assign the digit, click **Settings** to configure the parameters.
  - In **Operator Group**, select the desired Operator Group Number. Make sure you have configured the parameters for this group. For detailed instructions, see “Extension Voice Mail Settings”. By default, As Configured in Extension Settings is selected, that is, the system will use the Operator Group configured for the transfer number.
  - In **Call Transfer Profile**, select the desired Call Transfer Profile you want the system to use. By default, *As configured for Transfer Number* is selected.
    - Select **As configured for Dialed Extension** if you want the VMS Auto Attendant to use the Call Transfer Profile which is assigned to the dialed extension.
    - Select **Wait for Ring** if you want the VMS Auto Attendant to wait for the extension to start ringing and then transfer the call.
    - Select **Blind** if you want the VMS Auto Attendant to transfer the call on the extension without checking whether it is busy or free.
    - Select **Attended** if you want the VMS Auto Attendant to transfer the call when the extension answers (goes OFF-Hook).

      Click **Settings** to configure the parameters of the selected profile.

When the Call Transfer Profiles are added, or any existing profile is edited, these changes will be reflected in the options displayed.

- **Transfer to Assistant**: If you assign a digit for this option, make sure you have programmed the Assistant Number. For detailed instructions, refer “Number Programming (Assistant/Personal)” in “Mailbox Menu”.

Click **Settings** to configure the parameter.

- In **Call Transfer Profile**, select the desired Call Transfer Profile as per your requirement. By default, *Wait for Ring* is selected.

  Click **Settings** to configure the parameters of the selected profile.

- **Transfer to Alternate/Mobile Number**: If you assign a digit for this option, make sure you have programmed the Alternate/Mobile Number. For detailed instructions, refer “Number Programming (Assistant/Personal)” in “Mailbox Menu”.

- **Dial Extension Number:** After you assign the digit, click **Settings** to configure the parameters.

  - In **Dial Extension Number Prompt**, select the prompt you wish to play to the caller to dial the desired extension number.

    Select **None**, if you do not wish to play any prompt.

    - Click **Settings**. The **Prompts Management** page opens.

      You may add a new prompt from here. For further details, see “Prompts Management”.

    Once the files are uploaded these appear as options for Dial Extension Number Prompt.

    - In **Call Transfer Profile**, select the desired Call Transfer Profile as per your requirement. By default, *As configured for Dialed Extension* is selected.

      Click **Settings** to configure the parameters of the selected profile.

  No Match Found Settings: No Match Found Settings are applicable when a Extension Number is found invalid, the system will play the *No Match Found* prompt.

  - In N**o Match Found Prompt**, select the prompt which you wish to play when caller has dialed an invalid extension number.

    Select **None**, if you do not wish to play any prompt.

    - Click **Settings**. The **Prompts Management** page opens.

      You may add a new prompt from here. For further details, see “Prompts Management”.

    Once the files are uploaded these appear as options for No Match Found Prompt.

  - Select the **No Match Found Retry** check box to prompt the caller to dial the Extension Number again.

  - In **No Match Found Retry Count**, select the number of times you wish to prompt the caller for dialing the extension number. The prompt will be played repeatedly till the Retry Count expires.

- In **Retry Count Expiry Prompt**, select the prompt you wish to play when the Retry Count expires.

  Select **None**, if you do not wish to play any prompt.

  - Click **Settings** . The **Prompts Management** page opens.

    You may add a new prompt from here. For further details, see “Prompts Management”.

  Once the files are uploaded these appear as options for Retry Count Expiry Prompt.

- In **No Match Found Action**, you can select — Disconnect, Transfer to Operator, Transfer to Extension, Transfer to Department Group, Go to VMAA Menu.

  - If you select **Transfer to Operator** or **Transfer to Department Group** or **Transfer to Extension**, select the respective Operator number/Department Group number or enter the desired extension number.

    In **Call Transfer Profile**, select the desired Call Transfer Profile as per your requirement. By default, *As Configured for Transfer Number* is selected

    Click **Settings** to configure the parameters of the selected profile.

  - If you select **Go to VMAA Menu**, select the desired VMAA Menu.

  - If you select **Disconnect**, the system will disconnect the call.

No Digit Dialed Settings: No Digit Dialed Settings are applicable when caller has not dialed the extension number and the first digit wait timer has expired.

- In **No Digit Dialed Prompt**, select the prompt which you wish to play when caller has not dialed any digit.

  Select **None**, if you do not wish to play any prompt.

  - Click **Settings** . The **Prompts Management** page opens.

    You may add a new prompt from here. For further details, see “Prompts Management”.

  Once the files are uploaded these appear as options for No Digit Dialed Prompt.

- Select the **No Digit Dialed Retry** check box to prompts the caller to dial the Extension Number again.

- In **No Digit Dialed Retry Count**, select the number of times you wish to prompt the caller for dialing the extension number. The prompt will be played repeatedly till the Retry Count expires.

- In **Retry Count Expiry Prompt**, select the prompt you wish to play when the Retry Count expires.

  Select **None**, if you do not wish to play any prompt.

- Click **Settings** . The **Prompts Management** page opens.

  You may add a new prompt from here. For further details, see “Prompts Management”.

Once the files are uploaded these appear as options for Retry Count Expiry Prompt.

- In **No Digit Dialed Action**, you can select — Disconnect, Transfer to Operator, Transfer to Extension, Transfer to Department Group, Go to VMAA Menu.
  - If you select **Transfer to Operator** or **Transfer to Department Group** or **Transfer to Extension**, select the respective Operator number/Department Group number or enter the desired extension number.

    In **Call Transfer Profile**, select the desired Call Transfer Profile as per your requirement. By default, *As Configured for Transfer Number* is selected.

    Click **Settings** to configure the parameters of the selected profile.
  - If you select **Go to VMAA Menu**, select the desired VMAA Menu.
  - If you select **Disconnect**, the system will disconnect the call.
- **Go to Main Menu:** Assign a digit for this option if you want the system to re-direct the caller to the Main Menu.
- **Go to Previous Menu:** Assign a digit for this option if you want the system to re-direct the caller to the Previous Menu.
- **Disconnect:** Assign a digit for this option if you want the system to provide the option to disconnect the call.

- In **Busy Extension Status check time interval (sec)**, enter the interval of time after which you want the system to inform the caller that the extension number is busy. The system will play the Time Interval Expiry Prompt.
  - In **Time Interval Expiry Prompt**, select the prompt you wish to play when the time interval expires.

    Select **None**, if you do not wish to play any prompt.
    - Click **Settings** . The **Prompts Management** page opens.

      You may add a new prompt from here. For further details, see “Prompts Management”.

  Once the files are uploaded these appear as options for Time Interval Expiry Prompt.
- In B**usy Extension Status check retry count**, select the number of times you wish the system to check if the transfer number is free. The prompt will be played repeatedly till the Retry Count expires.
  - In **Retry Count Expiry Prompt**, select the prompt you wish to play when the Retry Count expires.

    Select **None**, if you do not wish to play any prompt.

- Click **Settings** . The **Prompts Management** page opens.

  You may add a new prompt from here. For further details, see “Prompts Management”.

Once the files are uploaded these appear as options for Retry Count Expiry Prompt.

- In **Busy Hold - Music on Hold**, select the prompt you wish to play. This option is applicable only is you have enabled the Hold a Call check box.

  Select **None**, if you do not wish to play any prompt.

  - Click **Settings** . The **Prompts Management** page opens.

    You may add a new prompt from here. For further details, see “Prompts Management”.

  Once the files are uploaded these appear as options for Busy Hold - Music on Hold.

- In **Busy Action,** you can select — Disconnect, Transfer to Operator, Transfer to Extension, Transfer to Department Group, Go to VMAA Menu, Go to Voice Mail, Give Advanced Options.

  - If you select **Transfer to Operator** or **Transfer to Department Group** or **Transfer to Extension**, select the respective Operator number/Department Group number or enter the desired extension number.

    In **Call Transfer Profile**, select the desired Call Transfer Profile as per your requirement. By default, *As Configured for Transfer Number* is selected.

    Click **Settings** to configure the parameters of the selected profile.

  - If you select **Go to VMAA Menu**, select the desired VMAA Menu.

  - If you select **Disconnect**, the system will disconnect the call.

  - If you select **Give Advanced Option**, see “Advanced Options”

- In **Action after Leaving Voice Mail**, you can select — Disconnect, Transfer to Operator, Transfer to Extension, Transfer to Department Group, Go to VMAA Menu. This option is applicable only is you select Go to Voice Mail or Give Advanced options as the Busy Action option.

  - If you select **Transfer to Operator** or **Transfer to Department Group** or **Transfer to Extension**, select the respective Operator number/Department Group number or enter the desired extension number.

    In **Call Transfer Profile**, select the desired Call Transfer Profile as per your requirement. By default, *As Configured for Transfer Number* is selected.

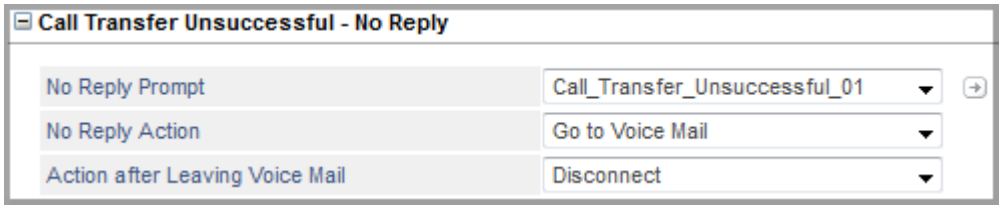
    Click **Settings** to configure the parameters of the selected profile.

  - If you select **Go to VMAA Menu**, select the desired VMAA Menu.

  - If you select **Disconnect**, the system will disconnect the call.

Call Transfer Unsuccessful - No Reply

If the called party has set Call Forward-No Reply to VMS, the Call Transfer Unsuccessful - No Reply parameters will be applicable. You can customize these parameters as per your requirement.



- In **No Reply Prompt,** select the prompt you wish to play to the caller if the dialed extension does not reply.
- In **No Reply Action,** you can select — Go to Voice Mail, Give Advanced Options, Transfer to Operator, Transfer to Extension, Transfer to Department Group, Go to VMAA Menu, Disconnect.
    - If you select **Transfer to Operator** or **Transfer to Department Group** or **Transfer to Extension**, select the respective Operator number/Department Group number or enter the desired extension number.

      In **Call Transfer Profile**, select the desired Call Transfer Profile as per your requirement. By default, *As Configured for Transfer Number* is selected.

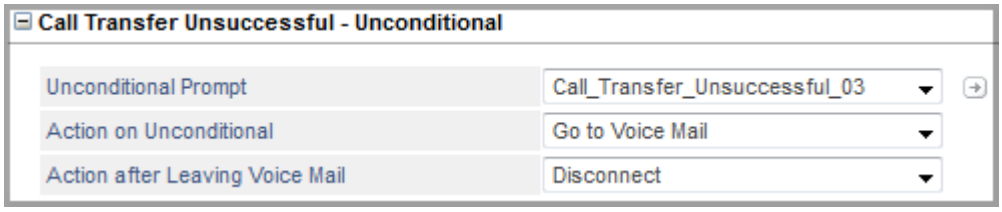
      Click **Settings** to configure the parameters of the selected profile.
    - If you select **Go to VMAA Menu**, select the desired VMAA Menu.
    - If you select **Disconnect**, the system will disconnect the call.
    - If you select **Give Advanced Option**, see “Advanced Options”
- In **Action after Leaving Voice Mail**, you can select — Transfer to Operator, Transfer to Extension, Transfer to Department Group, Go to VMAA Menu, Disconnect. This option is applicable only is you select Go to Voice Mail or Give Advanced options as the No Reply Action option.
    - If you select **Transfer to Operator** or **Transfer to Department Group** or **Transfer to Extension**, select the respective Operator number/Department Group number or enter the desired extension number.

      In **Call Transfer Profile**, select the desired Call Transfer Profile as per your requirement. By default, *As Configured for Transfer Number* is selected.

      Click **Settings** to configure the parameters of the selected profile.
    - If you select **Go to VMAA Menu**, select the desired VMAA Menu.
    - If you select **Disconnect**, the system will disconnect the call.

Call Transfer Unsuccessful - Unconditional

If the called party has set Call Forward-Unconditional to VMS, the Call Transfer Unsuccessful - Unconditional parameters will be applicable. You can customize these parameters as per your requirement.



- In **Unconditional Prompt,** select the prompt you wish to play to the caller.
- In **Action on Unconditional,** you can select — Go to Voice Mail, Give Advanced Options, Transfer to Operator, Transfer to Extension, Transfer to Department Group, Go to VMAA Menu, Disconnect.
    - If you select **Transfer to Operator** or **Transfer to Department Group** or **Transfer to Extension**, select the respective Operator number/Department Group number or enter the desired extension number.

      In **Call Transfer Profile**, select the desired Call Transfer Profile as per your requirement. By default, *As Configured for Transfer Number* is selected.

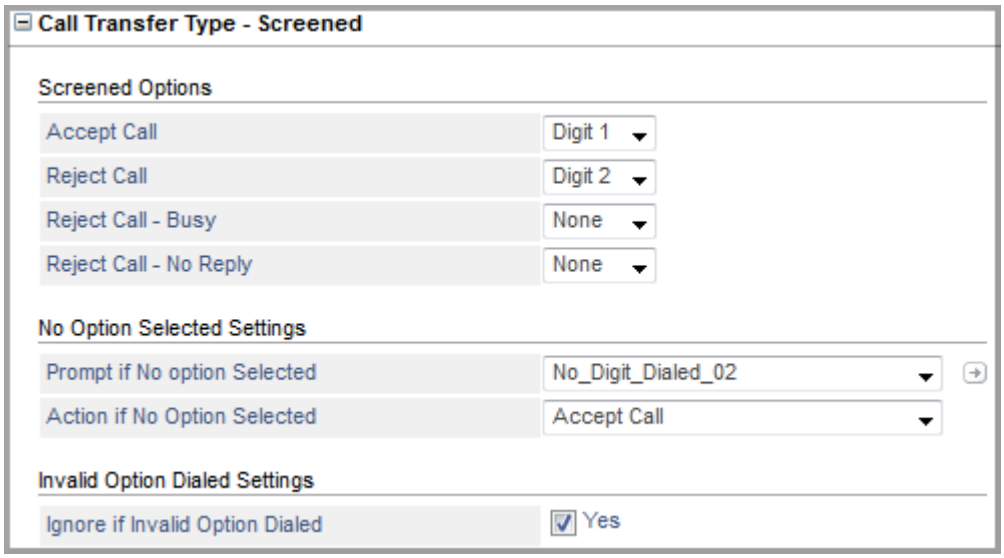
      Click **Settings** to configure the parameters of the selected profile.
    - If you select **Go to VMAA Menu**, select the desired VMAA Menu.
    - If you select **Disconnect**, the system will disconnect the call.
    - If you select **Give Advanced Option**, see “Advanced Options”
- In **Action after Leaving Voice Mail**, you can select — Transfer to Operator, Transfer to Extension, Transfer to Department Group, Go to VMAA Menu, Disconnect. This option is applicable only is you select Go to Voice Mail or Give Advanced options as the No Reply Action option.
    - If you select **Transfer to Operator** or **Transfer to Department Group** or **Transfer to Extension**, select the respective Operator number/Department Group number or enter the desired extension number.

      In **Call Transfer Profile**, select the desired Call Transfer Profile as per your requirement. By default, *As Configured for Transfer Number* is selected.

      Click **Settings** to configure the parameters of the selected profile.
    - If you select **Go to VMAA Menu**, select the desired VMAA Menu.
    - If you select **Disconnect**, the system will disconnect the call.

Call Transfer Type -Screened

Screened Transfer is when the VMS connects the caller to the transfer number after confirmation from the transfer number. If you have selected Call Transfer Type as Screened, you may customize the Call Transfer Type - Screened parameters as per your requirement.



Screened Options

The system will play Screened Options — Accept Call, Reject Call, Reject Call - Busy, Reject Call - NoReply — to the caller. You can assign the Digits (0-9, *, # or None) to each of the options.

Make sure you do not assign the same digit to multiple options.

The digit you configure will be played as the digit to be dialed by the caller to select the respective option. The options will be played in the sequence — 1-9, 0, *, #. The option for which the Digit selected is None will not be played.

After the digits have been assigned, the system will proceed further as per the options selected by you. Brief description of each option is explained below:

- Accept Call - When this option is selected, the system will transfer the call to the desired number.
- Reject Call - When this option is selected, system will consider it as Call Transfer Unsuccessful - Unconditional and process the call further as per the settings of “Call Transfer Unsuccessful - Unconditional”.
- Reject Call - Busy - When this option is selected, system will consider it as Call Transfer Unsuccessful - Busy and process the call further as per the settings of “Call Transfer Unsuccessful - Busy”.
- Reject Call - No Reply - When this option is selected, system will consider it as Call Transfer Unsuccessful - No Reply and process the call further as per the settings of “Call Transfer Unsuccessful - No Reply”.

No Option Selected Settings

While the Screened Options are being played, if the caller does not select any options, the following parameters will be applicable.

- In **Prompt if No Option Selected**, select the prompt you wish to play to the caller, if no Screened Option is selected by the caller.

Select **None**, if you do not wish to play any prompt.

- Click **Settings** . The **Prompts Management** page opens.

  You may add a new prompt from here. For further details, see “Prompts Management”.

Once the files are uploaded these appear as options for Prompt if No Option Selected.

- In **Action if No Option Selected,** you can select — Accept Call, Reject Call, Reject Call with Busy, Reject Call with No Reply, Disconnect.

Invalid Option Selected Settings

While the Screened Options are being played, if the caller dials a digit which is not programmed, the following parameters will be applicable.

- Clear the **Ignore if Invalid Option Dialed** check box, if you do not want the system to ignore the invalid digit dialed and prompt the caller for the same. By default, it is enabled.
  - In **Prompt if Invalid Option Selected,** select the prompt which you wish to play when the caller has dialed an invalid digit.

    Select **None**, if you do not wish to play any prompt.

    - Click **Settings** . The **Prompts Management** page opens.

      You may add a new prompt from here. For further details, see “Prompts Management”.

    Once the files are uploaded these appear as options for Prompt if Invalid Option Selected.

  - In **Action if No Option Selected,** you can select — Accept Call, Reject Call, Reject Call with Busy, Reject Call with No Reply, Disconnect.

No Mailbox Settings

If for any option *Leave Voice Mail* is selected and a mailbox is not assigned to the number, the system will play a prompt to the caller informing the caller that a mailbox is not assigned. The system will then proceed further as per the action you select here.

- In **No Mailbox Action**, you can select — Transfer to Operator, Transfer to Extension, Transfer to Department Group, Go to VMAA Menu, Disconnect.
  - If you select **Transfer to Operator** or **Transfer to Department Group** or **Transfer to Extension**, select the respective Operator number/Department Group number or enter the desired extension number.

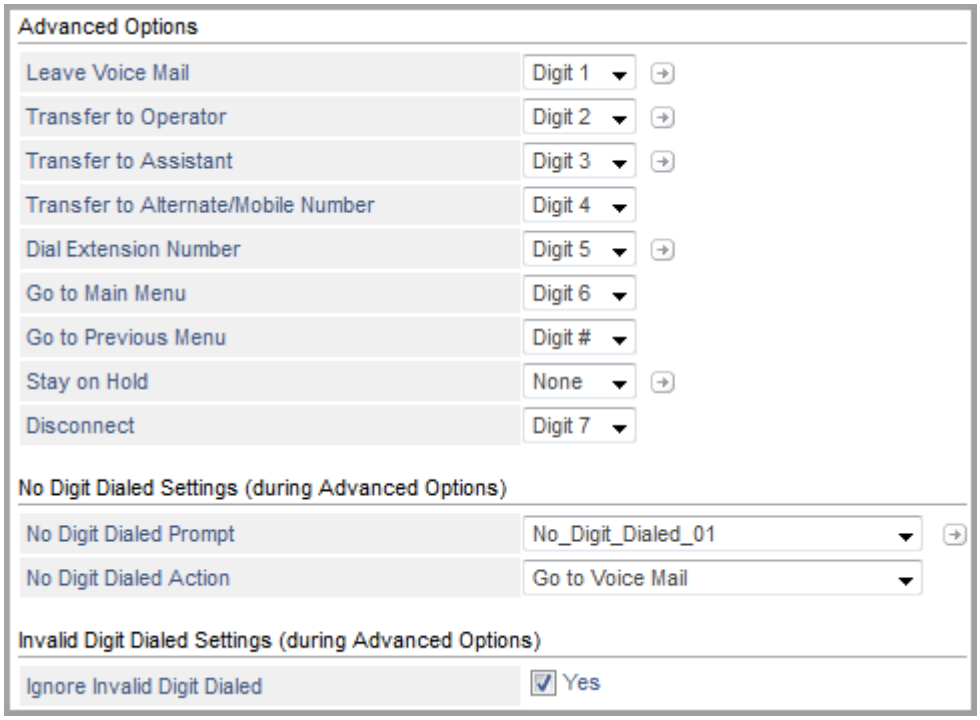
    In **Call Transfer Profile**, select the desired Call Transfer Profile as per your requirement. By default, *Wait for Ring* is selected.

Click **Settings** to configure the parameters of the selected profile.

- If you select **Go to VMAA Menu**, select the desired VMAA Menu.
- If you select **Disconnect**, the system will disconnect the call.

Advanced Options

The system will play Advanced options— Leave Voice Mail, Transfer to Operator, Transfer to Assistant, Transfer to Alternate/Mobile Number, Dial Extension Number, Go to Main Menu, Go to Previous Menu, Stay on Hold, Disconnect — to the caller. You can assign the Digits (0-9, *, # or None) to each of the options.



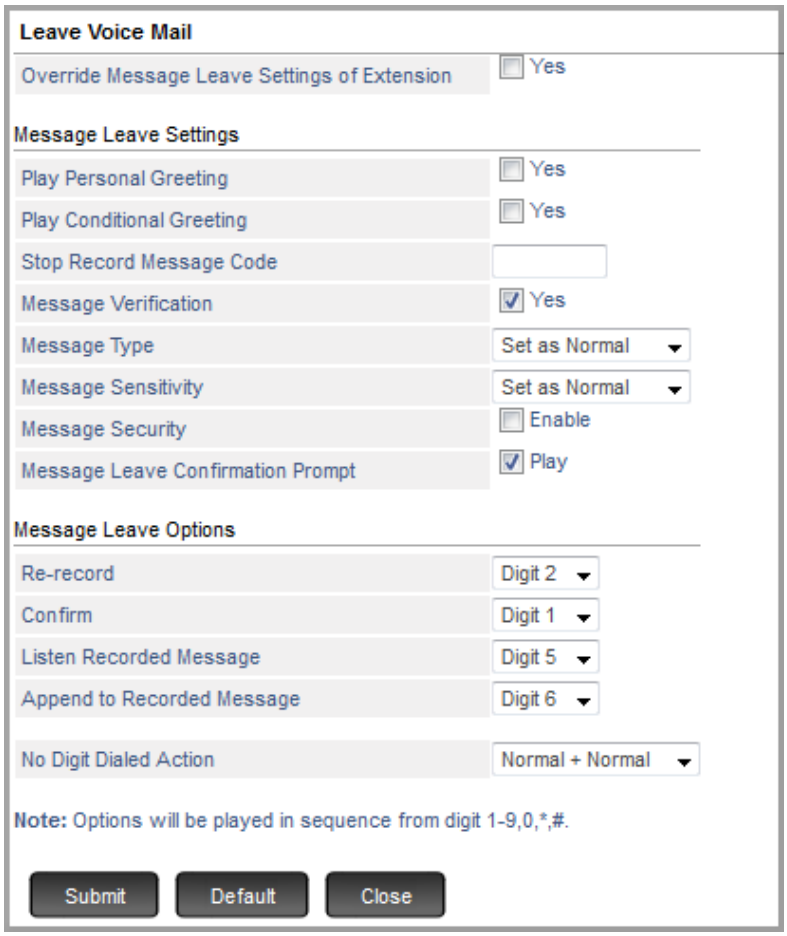
Make sure you do not assign the same digit to multiple options.

The digit you configure will be played as the digit to be dialed by the caller to select the respective option. The options will be played in the sequence — 1-9, 0, *, #. The option for which the Digit selected is None will not be played.

After the digits have been assigned, the system will proceed further as per the options selected by you. Brief description of each option is explained below:

- Leave Voice Mail
- Transfer to Operator
- Transfer to Assistant
- Transfer to Alternate Number/Mobile Number
- Dial Extension Number
- Go to Main Menu
- Go to Previous Menu
- Stay on Hold
- Disconnect

Leave Voice Mail

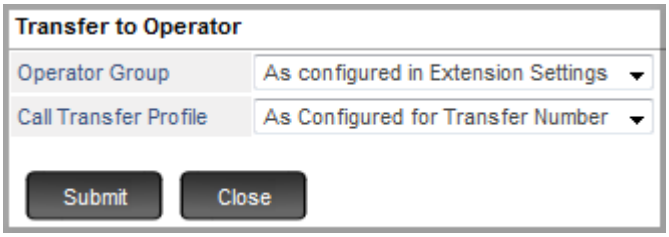


After you assign the digit, click **Settings** to configure the parameters.

- Select the **Override Message Leave Settings of Extension** check box, if you want the system to apply Message Leave Settings customized on this page and not that of the extension or department group number.

  Clear the check box if you want the system to apply Message Leave Settings configured for the extension or department group number.

Transfer to Operator



After you assign the digit, click **Settings** to configure the parameters.

- In **Operator Group**, select the desired Operator Group Number. Make sure you have configured the parameters for this group. For detailed instructions, see "Extension Voice Mail Settings". By default, As Configured in Extension Settings is selected, that is, the system will use the Operator Group configured for the Transfer Number.

- In **Call Transfer Profile**, select the desired Call Transfer Profile as per your requirement. By default, *As Configured for Transfer Number* is selected.

  Click **Settings** to configure the parameters of the selected profile.

Transfer to Assistant

If you assign a digit for this option, make sure you have programmed the Assistant Number. For detailed instructions, refer "Number Programming (Assistant/Personal)" in "Mailbox Menu".

Click **Settings** to configure the parameter.

- In **Call Transfer Profile**, select the desired Call Transfer Profile you want the system to use. Make sure you have configured the parameters for this profile. By default, Wait for Ring is selected, that is, the system will use the Call Transfer profile configured for transfer number.

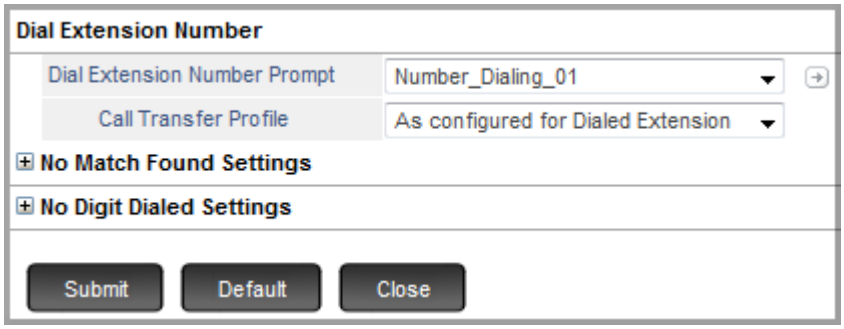
  - In **Call Transfer Profile**, select the desired Call Transfer Profile as per your requirement. By default, *Wait for Ring* is selected.

    Click **Settings** to configure the parameters of the selected profile.

Transfer to Alternate/Mobile Number

If you assign a digit for this option, make sure you have programmed the Alternate/Mobile Number. For detailed instructions, refer "Number Programming (Assistant/Personal)" in "Mailbox Menu".

Dial Extension Number



After you assign the digit, click **Settings** to configure the parameters.

- In **Dial Extension Number Prompt**,select the prompt you wish to play to the caller to dial the desired extension number.

  Select **None**, if you do not wish to play any prompt.

  - Click **Settings**. The **Prompts Management** page opens.

    You may add a new prompt from here. For further details, see “Prompts Management”.

  Once the files are uploaded these appear as options for Dial Extension Number Prompt.

- In **Call Transfer Profile**, select the desired Call Transfer Profile as per your requirement. By default, *As configured for Dialed Extension* is selected.

  Click Settings to configure the parameters of the selected profile.

No Match Found Settings

No Match Found Settings are applicable when the caller has dialed a digit or a number which does not match the Extension Number list present in the system, the system will play the *No Match Found* prompt.

- In **No Match Found Prompt**, select the prompt which you wish to play when caller has dialed a digit that does not match the Extension Number List.

  Select **None**, if you do not wish to play any prompt.

  - Click **Settings**. The **Prompts Management** page opens.

    You may add a new prompt from here. For further details, see “Prompts Management”.

  Once the files are uploaded these appear as options for No Match Found Prompt.

- Select the **No Match Found Retry** check box to prompt the caller to dial the Extension Number again.
- In **No Match Found Retry Count**, select the number of times you wish to prompt the caller for dialing the extension number. The prompt will be played repeatedly till the Retry Count expires.

  - In **Retry Count Expiry Prompt**, select the prompt you wish to play when the Retry Count expires.

    Select **None**, if you do not wish to play any prompt.

    - Click **Settings**. The **Prompts Management** page opens.

      You may add a new prompt from here. For further details, see “Prompts Management”.

    Once the files are uploaded these appear as options for Retry Count Expiry Prompt.

- In **No Match Found Action**, you can select — Disconnect, Transfer to Operator, Transfer to Extension, Transfer to Department Group, Go to VMAA Menu.

- If you select **Transfer to Operator** or **Transfer to Department Group** or **Transfer to Extension**, select the respective Operator number/Department Group number or enter the desired extension number.

    - In **Call Transfer Profile**, select the desired Call Transfer Profile as per your requirement. By default, *As configured for Transfer Number* is selected.

      Click **Settings** to configure the parameters of the selected profile.

- If you select **Go to VMAA Menu**, select the desired VMAA Menu.

- If you select **Disconnect**, the system will disconnect the call.

No Digit Dialed Settings

No Digit Dialed Settings are applicable when caller has not dialed the extension number and the first digit wait timer has expired.

- In **No Digit Dialed Prompt**, select the prompt which you wish to play when caller has not dialed any digit.

  Select **None**, if you do not wish to play any prompt.

  - Click **Settings**. The **Prompts Management** page opens.

    You may add a new prompt from here. For further details, see “Prompts Management”.

  Once the files are uploaded these appear as options for No Digit Dialed Prompt.

- Select the **No Digit Dialed Retry** check box to prompts the caller to dial the Extension Number again.

- In **No Digit Dialed Retry Count**, select the number of times you wish to prompt the caller for dialing the extension number. The prompt will be played repeatedly till the Retry Count expires.

  - In **Retry Count Expiry Prompt**, select the prompt you wish to play when the Retry Count expires.

    Select **None**, if you do not wish to play any prompt.

    - Click **Settings**. The **Prompts Management** page opens.

      You may add a new prompt from here. For further details, see “Prompts Management”.

    Once the files are uploaded these appear as options for Retry Count Expiry Prompt.

- In **No Digit Dialed Action**, you can select — Disconnect, Transfer to Operator, Transfer to Extension, Transfer to Department Group, Go to VMAA Menu.

  - If you select **Transfer to Operator** or **Transfer to Department Group** or **Transfer to Extension**, select the respective Operator number/Department Group number or enter the desired extension number.

    - In **Call Transfer Profile**, select the desired Call Transfer Profile as per your requirement. By default, *As configured for Transfer Number* is selected.

Click **Settings** to configure the parameters of the selected profile.

- If you select **Go to VMAA Menu**, select the desired VMAA Menu.
- If you select **Disconnect**, the system will disconnect the call.

Go to Main Menu

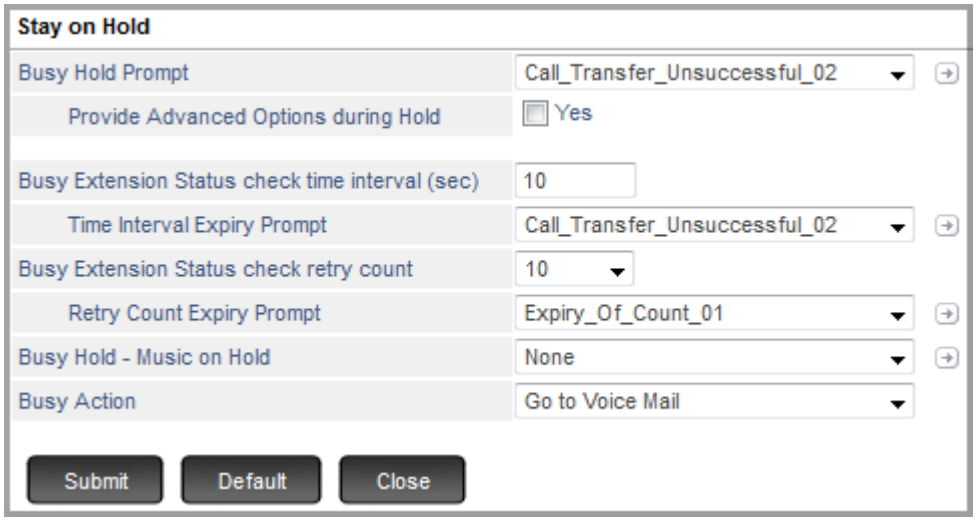
Assign a digit for this option if you want the system to re-direct the caller to the Main Menu.

Go to Previous Menu

Assign a digit for this option if you want the system to re-direct the caller to the Previous Menu.

Stay on Hold

This option is applicable for Call Transfer Unsuccessful - Busy only.



After you assign the digit, click **Settings** to configure the parameters.

- In **Busy Hold Prompt**, select the prompt to be played to the caller when put on hold.

  Select **None**, if you do not wish to play any prompt.

  - Click **Settings**. The **Prompts Management** page opens.

    You may add a new prompt from here. For further details, see “Prompts Management”.

  Once the files are uploaded these appear as options for Busy Hold Prompt.

  - Select **Provide Advanced Options during Hold** check box, if you want the system to play Advanced options to the caller.

You can assign the Digits (0-9, *, # or None) to each of the options — Leave Voice Mail, Transfer to Operator, Transfer to Assistant, Transfer to Alternate/Mobile Number, Dial Extension Number, Go to Main Menu, Go to Previous Menu and Disconnect.

Make sure you do not assign the same digit to multiple options.

The digit you configure will be played as the digit to be dialed by the caller to select the respective option. The options will be played in the sequence — 1-9, 0, *, #. The option for which the Digit selected is None will not be played.

After the digits have been assigned, the system will proceed further as per the options selected by you. Brief description of each option is explained below:

- **Leave Voice Mail:** After you assign the digit, click **Settings** to configure the parameters.
    - Select the **Override Message Leave Settings of Extension** check box, if you want the system to apply Message Leave Settings customized on this page and not that of the extension or department group number.

      Clear the check box if you want the system to apply Message Leave Settings configured for the extension or department group number.

- **Transfer to Operator:** After you assign the digit, click **Settings** to configure the parameters.
    - In **Operator Group**, select the desired Operator Group Number. Make sure you have configured the parameters for this group. For detailed instructions, see "Extension Voice Mail Settings". By default, As Configured in Extension Settings is selected, that is, the system will use the Operator Group configured for the Transfer Number.
        - In **Call Transfer Profile**, select the desired Call Transfer Profile as per your requirement. By default, *As configured for Transfer Number* is selected.

          Click **Settings** to configure the parameters of the selected profile.

- **Transfer to Assistant**: If you assign a digit for this option, make sure you have programmed the Assistant Number. For detailed instructions, refer "Number Programming (Assistant/Personal)" in "Mailbox Menu".

  Click **Settings** to configure the parameter.
    - In **Call Transfer Profile**, select the desired Call Transfer Profile as per your requirement. By default, *As configured for Transfer Number* is selected.

      Click **Settings** to configure the parameters of the selected profile.

- **Transfer to Alternate/Mobile Number**: If you assign a digit for this option, make sure you have programmed the Alternate/Mobile Number. For detailed instructions, refer "Number Programming (Assistant/Personal)" in "Mailbox Menu".

- **Dial Extension Number:** After you assign the digit, click **Settings** to configure the parameters.

Select **None**, if you do not wish to play any prompt.

- Click **Settings** . The **Prompts Management** page opens.
  You may add a new prompt from here. For further details, see “Prompts Management”.

Once the files are uploaded these appear as options for Dial Extension Number Prompt.

- In **Call Transfer Profile**, select the desired Call Transfer Profile as per your requirement. By default, *As configured for Dialed Extension* is selected.

  Click **Settings** to configure the parameters of the selected profile.

No Match Found Settings: No Match Found Settings are applicable when a Extension Number is found invalid, the system will play the *No Match Found* prompt.

- In **No Match Found Prompt**, select the prompt which you wish to play when caller has dialed an invalid extension number.

  Select **None**, if you do not wish to play any prompt.

  - Click **Settings** . The **Prompts Management** page opens.

    You may add a new prompt from here. For further details, see “Prompts Management”.

  Once the files are uploaded these appear as options for No Match Found Prompt.

- Select the **No Match Found Retry** check box to prompt the caller to dial the Extension Number again.

- In **No Match Found Retry Count**, select the number of times you wish to prompt the caller for dialing the extension number. The prompt will be played repeatedly till the Retry Count expires.

- In **Retry Count Expiry Prompt**, select the prompt you wish to play when the Retry Count expires.

  Select **None**, if you do not wish to play any prompt.

  - Click **Settings** . The **Prompts Management** page opens.

    You may add a new prompt from here. For further details, see “Prompts Management”.

  Once the files are uploaded these appear as options for Retry Count Expiry Prompt.

- In **No Match Found Action**, you can select — Disconnect, Transfer to Operator, Transfer to Extension, Transfer to Department Group, Go to VMAA Menu.

  - If you select **Transfer to Operator** or **Transfer to Department Group** or **Transfer to Extension**, select the respective Operator number/Department Group number or enter the desired extension number.

    In **Call Transfer Profile**, select the desired Call Transfer Profile as per your requirement. By default, *As configured for Transfer Number* is selected.

Click **Settings** to configure the parameters of the selected profile.

- If you select **Go to VMAA Menu**, select the desired VMAA Menu.
- If you select **Disconnect**, the system will disconnect the call.

No Digit Dialed Settings: No Digit Dialed Settings are applicable when caller has not dialed the extension number and the first digit wait timer has expired.

- In **No Digit Dialed Prompt**, select the prompt which you wish to play when caller has not dialed any digit.

  Select **None**, if you do not wish to play any prompt.

  - Click **Settings**. The **Prompts Management** page opens.

    You may add a new prompt from here. For further details, see “Prompts Management”.

  Once the files are uploaded these appear as options for No Digit Dialed Prompt.

- Select the **No Digit Dialed Retry** check box to prompts the caller to dial the Extension Number again.
- In **No Digit Dialed Retry Count**, select the number of times you wish to prompt the caller for dialing the extension number. The prompt will be played repeatedly till the Retry Count expires.
- In **Retry Count Expiry Prompt**, select the prompt you wish to play when the Retry Count expires.

  Select **None**, if you do not wish to play any prompt.

  - Click **Settings**. The **Prompts Management** page opens.

    You may add a new prompt from here. For further details, see “Prompts Management”.

  Once the files are uploaded these appear as options for Retry Count Expiry Prompt.

- In **No Digit Dialed Action**, you can select — Disconnect, Transfer to Operator, Transfer to Extension, Transfer to Department Group, Go to VMAA Menu.

  - If you select **Transfer to Operator** or **Transfer to Department Group** or **Transfer to Extension**, select the respective Operator number/Department Group number or enter the desired extension number.

    In **Call Transfer Profile**, select the desired Call Transfer Profile as per your requirement. By default, *As configured for Transfer Number* is selected.

    Click **Settings** to configure the parameters of the selected profile.

  - If you select **Go to VMAA Menu**, select the desired VMAA Menu.
  - If you select **Disconnect**, the system will disconnect the call.

- **Go to Main Menu:** Assign a digit for this option if you want the system to re-direct the caller to the Main Menu.
- **Go to Previous Menu:** Assign a digit for this option if you want the system to re-direct the caller to the Previous Menu.
- **Disconnect:** Assign a digit for this option if you want the system to provide the option to disconnect the call.

- In **Busy Extension Status check time interval (sec)**, enter the interval of time after which you want the system to inform the caller that the extension number is busy. The system will play the Time Interval Expiry Prompt.
  - In **Time Interval Expiry Prompt**, select the prompt you wish to play when the time interval expires.

    Select **None**, if you do not wish to play any prompt.
  - Click **Settings** . The **Prompts Management** page opens.

    You may add a new prompt from here. For further details, see “Prompts Management”.

  Once the files are uploaded these appear as options for Time Interval Expiry Prompt.
- In **Busy Extension Status check retry count**, select the number of times you wish the system to check if the transfer number is free. The prompt will be played repeatedly till the Retry Count expires.
  - In **Retry Count Expiry Prompt**, select the prompt you wish to play when the Retry Count expires.

    Select **None**, if you do not wish to play any prompt.
    - Click **Settings** . The **Prompts Management** page opens.

      You may add a new prompt from here. For further details, see “Prompts Management”.

  Once the files are uploaded these appear as options for Retry Count Expiry Prompt.
- In **Busy Hold - Music on Hold**, select the prompt you wish to play. This option is applicable only is you have enabled the Hold a Call check box.

  Select **None**, if you do not wish to play any prompt.

  You may add a new Prompt. To do so,
  - Click **Settings** . The **Prompts Management** page opens.

    You may add a new prompt from here. For further details, see “Prompts Management”.

  Once the files are uploaded these appear as options for Busy Hold - Music on Hold.
- In **Busy Action,** you can select — Disconnect, Transfer to Operator, Transfer to Extension, Transfer to Department Group, Go to VMAA Menu, Go to Voice Mail, Give Advanced Options.

- If you select **Transfer to Operator** or **Transfer to Department Group** or **Transfer to Extension**, select the respective Operator number/Department Group number or enter the desired extension number.

  In **Call Transfer Profile**, select the desired Call Transfer Profile as per your requirement. By default, *As configured for Transfer Number* is selected.

  Click **Settings** to configure the parameters of the selected profile.

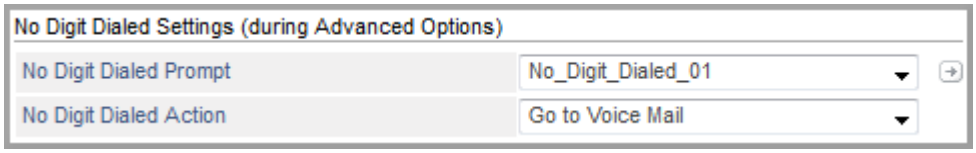
- If you select **Go to VMAA Menu**, select the desired VMAA Menu.
- If you select **Disconnect**, the system will disconnect the call.
- If you select **Give Advanced Option**, see "Advanced Options".

Disconnect

Assign a digit for this option if you want the system to provide the option to disconnect the call.

No Digit Dialed Settings (during Advanced Options)

No Digit Dialed Settings are applicable when caller has not dialed the extension number and the first digit wait timer has expired.



- In **No Digit Dialed Prompt**, select the prompt which you wish to play when caller has not dialed any digit.

  Select **None**, if you do not wish to play any prompt.

  - Click **Settings**. The **Prompts Management** page opens.

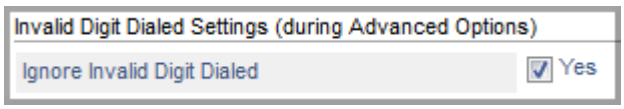
    You may add a new prompt from here. For further details, see "Prompts Management".

  Once the files are uploaded these appear as options for No Digit Dialed Prompt.

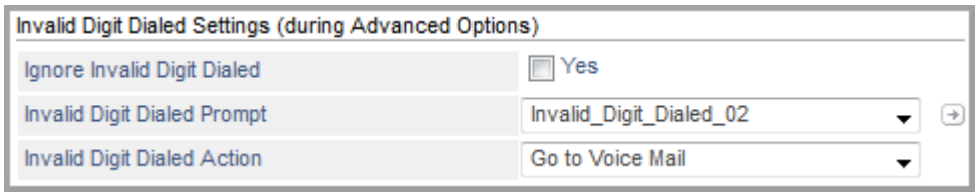
- In **No Digit Dialed Action**, you can select — Disconnect, Go to Voice Mail, Transfer to Operator, Transfer to Assistant, Transfer to Alternate Number, Go to Main Menu, Go to Previous Menu, Stay on Hold. Refer to the details given above under "Advanced Options".

Invalid Digit Dialed Settings (during Advanced Options)

Invalid Digit Dialed Settings are applicable when caller has dialed an invalid digit — a digit which is not configured as the Advanced Options.



- Clear the **Ignore Invalid Digit Dialed** check box, if you do not want the system to ignore the invalid digit dialed and prompt the caller for the same. By default, it is enabled.



- In **Invalid Digit Dialed Prompt**, select the prompt which you wish to play when the caller has dialed an invalid digit.

  Select **None**, if you do not wish to play any prompt.

  - Click **Settings** . The **Prompts Management** page opens.

    You may add a new prompt from here. For further details, see “Prompts Management”.

  Once the files are uploaded these appear as options for Invalid Digit Dialed Prompt.

- In **Invalid Digit Dialed Action**, you can select — Disconnect, Go to Voice Mail, Transfer to Operator, Transfer to Assistant, Transfer to Alternate Number, Go to Main Menu, Go to Previous Menu, Stay on Hold.

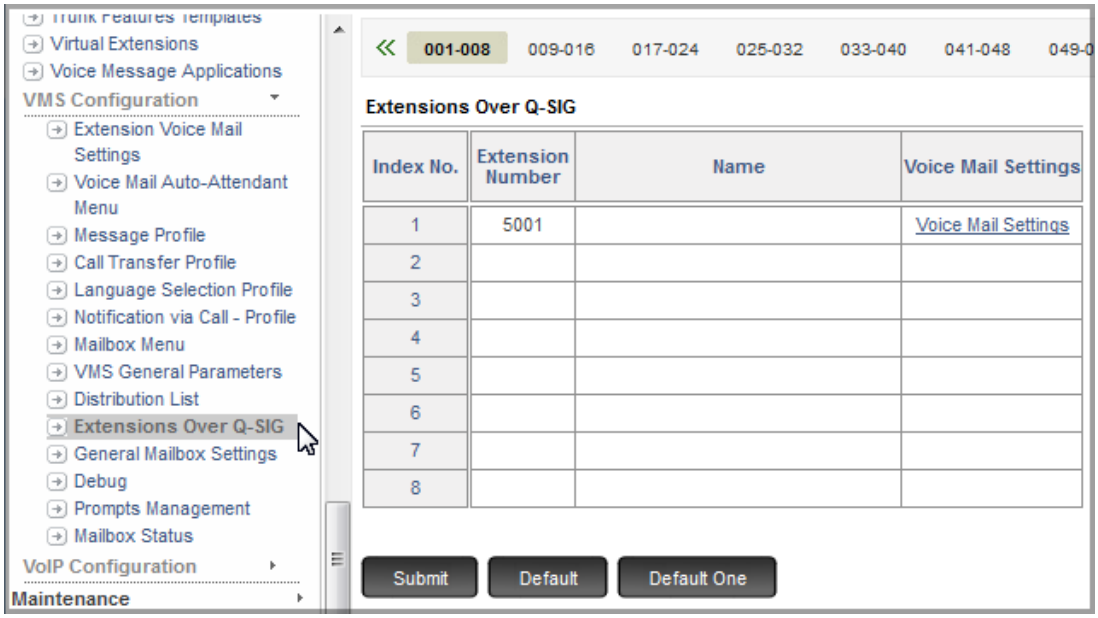
  Refer to the details given above under “Advanced Options”.

Extensions Over Q-SIG

SARVAM UCS supports Voice Mail for extension users connected over QSIG. These are extensions of the remote System (may be SARVAM UCS or any other system) connected with SARVAM UCS over QSIG.

To provide Voice Mail to the remote system extension users you must configure the Voice Mail Settings. To do so,

- Log in as System Engineer.
- Under **Configuration**, click **VMS Configuration**.
- Click **Extensions over Q-SIG**.



- For the extensions of the system you wish to assign voice mail, configure the following:
  - In **Extension Number**, enter the number assigned to the extensions of the remote System. The number must be entered along with the Exchange ID (if applicable).

    The Extension Number can be a maximum of 6 digits max. Valid Digits are 0-9, * ,# (*, # can be used as fist digit only).
  - In **Name**, enter a name which you wish to assign to the extension. The name may be of the person using the extension number of the remote System or it can be the name of a department.

    It can be a maximum of 18 alphanumeric characters. Default: Blank.
  - In **Voice Mail Settings**, click the Voice Mail Settings link. The respective Extension Voice Mail Settings window will open. You may edit the parameters. For details, see “Extension Voice Mail Settings”.

    Click **Close** to close the window.
- Click **Submit** to save.

- Click **Default** to default the page.
- Click **Default One** to default only one entry.

  A new window will open. Select the **Index Number** which you wish to default.

  Click **OK**.

Notification via Call-Profile

What's this?

The VMS supports Notification via Call to inform the extension users about the arrival of new messages in their mailbox.

Extension users can receive new message notification calls on a phone number of their choice. This number may be another extension number or an external number. You can set the Type of notification calls as:

- **Immediate:** Users will receive notifications as soon as a new message arrives in their mailbox.
  Or
- **Scheduled:** Extension users will receive notification at specified time intervals.

You can set the preferred time slots in a day during which notification calls should be made to extension users. In addition to the time slot preference, you can also choose to receive notification calls on a Holiday.

Message Notification via Call for Department Group will not work if the destination number is an external number.

How it works

For this feature to work, you must do the following configuration for the extension:

- Select the type of Notification call.
- Define the preferred time slots by configuring Time Zones. You can configure four different Time Zones, defining the Start Time and End Time for each Time Zone.
- Configure the phone number to which the notification call is to be made. If the number is an external number, configure the Trunk Access Code to be used for making the calls.

When **Immediate** is selected as the Type of notification,

- A new message arrives in the mailbox of the extension user.
- The system checks the preferred start and end time of the time zones configured for the extension. If the message has arrived within the preferred time slot (Start and End Time) it immediately makes the notification call on the number configured for the user.

  If the number is an external number, the system dials out the number using the Trunk Access Code (TAC) assigned for making notification calls.
- When the call is answered, the extension user gets connected to the VMS and can listen to the message.
- If the notification call is not answered, by default, the system makes three attempts (Message Notification Retry Count; programmable) at an interval of 5 minutes (Message Notification Interval; programmable) between each attempt.
- If the notification call remains unanswered after the third attempt, the system will not make any more attempts to place this notification call. The next notification call will be made only when another new message arrives in the mailbox of the user between the start and end time of the configured time zone.

When **Schedule** is selected as the Type of notification,

- A new message arrives in the mailbox of the extension user.
- The system checks the start time of the time zone(s) configured and the notification call will be made on the number at the subsequent start time.

  If the number is an external number the system dials out the number through the Trunk Access Code (TAC) assigned for making notification calls.
- When the call is answered the extension user gets connected to the VMS and can listen to the message.
- If the notification call is not answered, the system makes three attempts (Message Notification Retry Count; programmable) at an interval of 5 minutes (Message Notification Interval; programmable) for each time zone. The system will continue to make attempts to place the notification call till the call is answered.

Thus, when Notification type is Immediate, notification call is made for each message that is received within the start and end time configured in the time zone.

When Notification type is Scheduled, notification call is made for all messages received before the start time configured in the time zone. Where multiple time zones are configured, notification call will be made at the start time of the next time zone.

How to configure

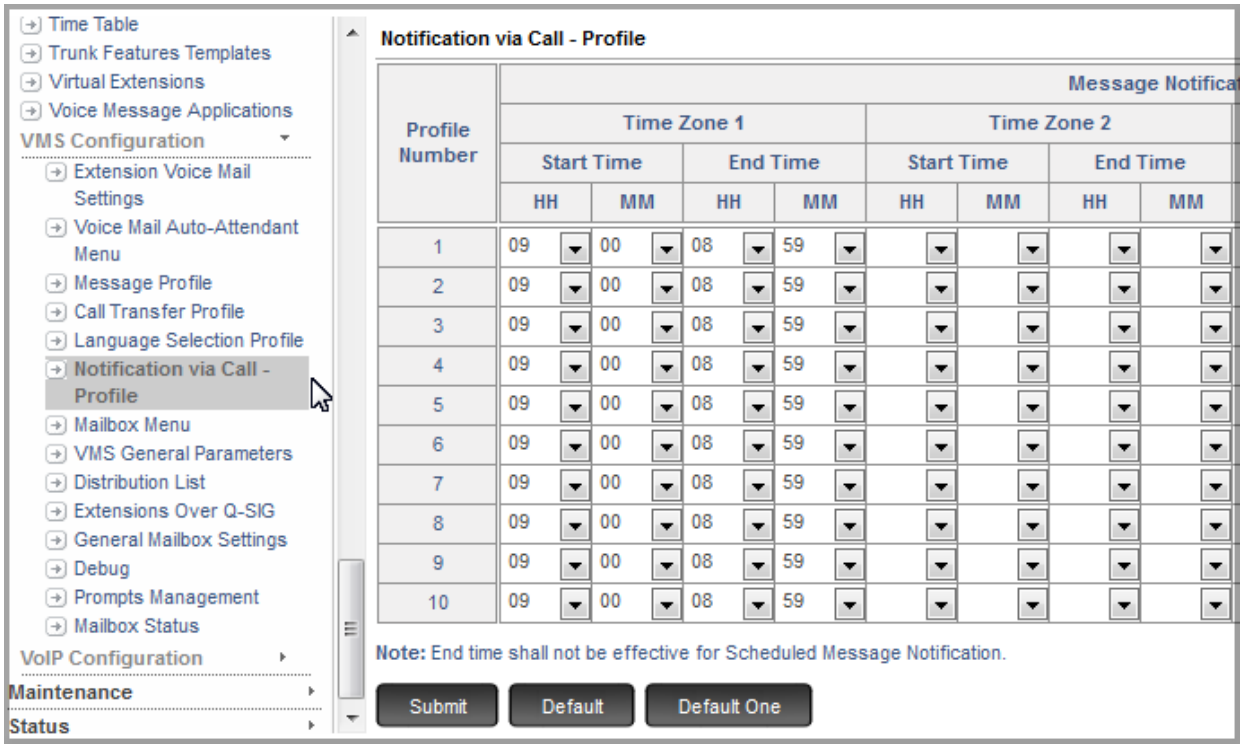
For Message Wait Notification via Call, you need to configure:

- the parameters for **Message Wait Notification via Call** under Message Wait Settings in Extension Voice Mail Settings.
- select the desired profile in Schedule Profile. For instructions to configure the profile parameters, see “Configuring Notification via Call - Profile”.
- Make Message Notification call using TAC for calls to be made to external numbers. See “Configuring VMS General Parameters”.
- if required, the Message Notification Retry Count, Message Notification Interval and Message Notification Ring. See “System Timers and Counts”.

Configuring Notification via Call - Profile

- Log into Jeeves as System Engineer.
- Under **Configuration**, click **VMS Configuration**.

- Click **Notification via Call - Profile** to open the page.



- Select the **Profile Number** which you want to assign to Message Wait Notification via Call.

  The Message Wait Notification Profile determines how notification calls are to be made to the desired numbers. You can configure upto 16 different profiles. In each profile, you can set different time zones according to the user preferences.

  Configure the following parameters against the Profile Number you select:

  - For Each Time Zone, **Time Zone 1** to **4**, configure the **Start Time** and **End Time**. The valid range is 00:00 to 23:59.
  - If you want to receive notifications on a holiday, select the **Notify on Holiday** check box.
  - Click **Submit** to save changes.
  - Now, assign the profile numbers to the desired extensions. Make sure the Message Wait Notification via Call parameters have been configures in the Extension Voice Mail Settings of these extensions.

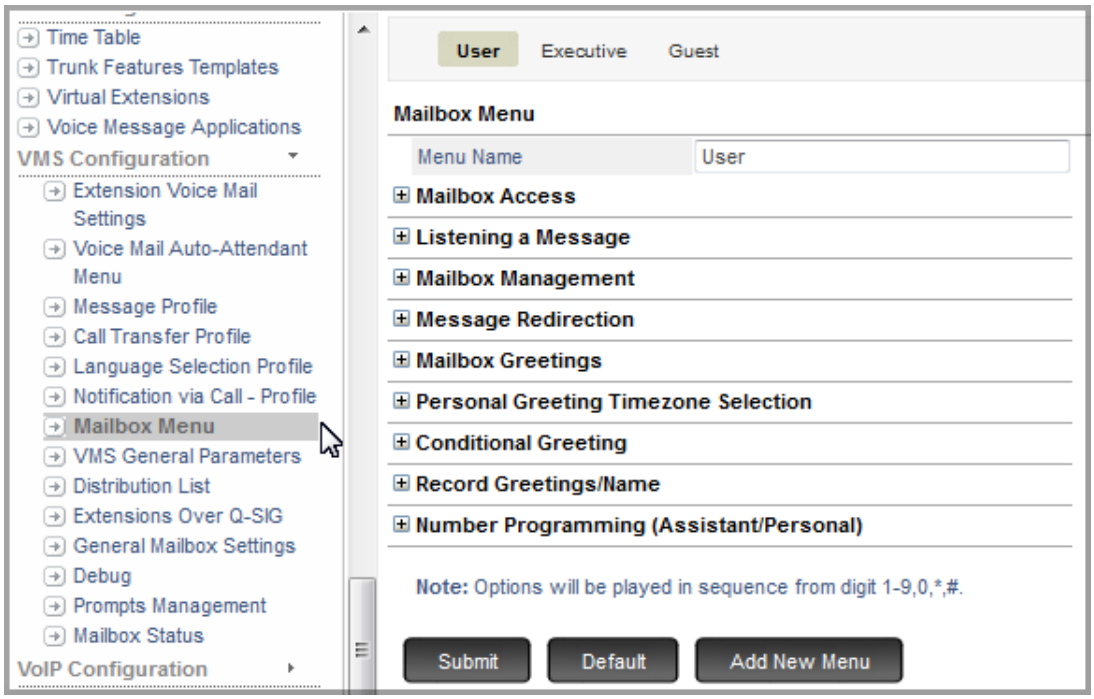
Mailbox Menu

Mailbox Menu offers you a group of parameters that will be used when the caller accesses his/her personal mailbox. VMS supports a maximum of 12 Mailbox Menus.

By default, three mailbox Menus — User, Executive and Guest— are provided to you. These three mailbox menus cannot be deleted but you can edit their settings as per your requirement.

How to Configure

- Log into Jeeves.
- Under **Configuration**, click **VMS Configuration**.
- Click **Mailbox Menu**.



You may add a new menu, edit the default menus or delete a Mailbox menu.

To add a new Mailbox Menu,

- Click **Add New Menu**. A new *Mailbox Menu xxx* will be created. You may now configure this as per your requirement.

To delete a menu, click **Delete**.

The default Mailbox Menus cannot be deleted.

To edit a Mailbox Menu,

- In **Menu Name**, configure the name of the Mailbox Menu you want. By default, it is *Mailbox Menu xx* where xx is the Mailbox Menu Number from 01 to 12.

Mailbox Menu Features

You can assign the Digits (0-9, *, # or None) to each of the options given under the features — Mailbox Access, Listening a Message, Mailbox Management, Message Redirection, Mailbox Greetings, Personal Greetings Timezone Selection, Conditional Greeting, Record Greetings/Name and Number Programming (Assistant/ Personal).

Make sure you do not assign the same digit to multiple options under the respective feature. For example, the digits assigned to various options under Listening a Message feature must be unique.

Select the **Play** check box to play the respective option.

If you assign the digit to an option but do not select the Play check box for the same, then the option will not be played to the caller.
The digits you configure will be played as the digit to be dialed by the caller to select the respective option. The options will be played in the sequence — 1-9, 0, *, #. The option for which the digit selected is None will not be played.

Brief description of each option in the respective features are given below:

Mailbox Access

Mailbox Access is the main menu which will be played to the caller whenever s/he accesses his/her Personal Mailbox.

- **Listen New Messages**: The system will play the new(unread) messages present in the caller's personal mailbox. The new messages will be played as per the *Message Playback Settings* configured in the Message Profile. For details, refer "Message Playback Settings".

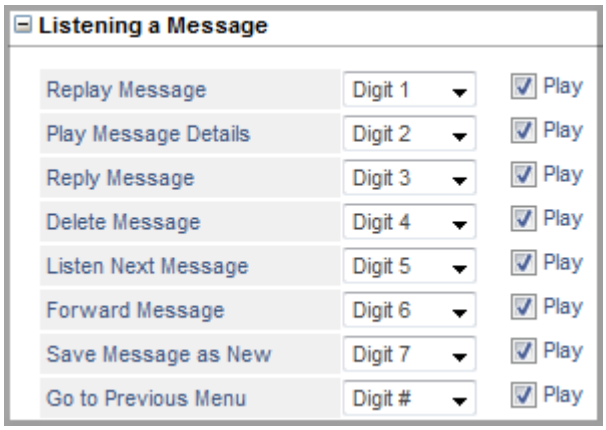
- **Listen Old Messages**: The system will play the old(read) messages present in the caller's personal mailbox. The old messages will be played as per the *Message Playback Settings* configured in the Message Profile. For details, refer "Message Playback Settings".

- **Send Message**: The system will play a prompt asking the caller to enter the Destination Number of the recipient for sending the message. The Destination Number may include a Distribution List. The Message will be send as per the *Message Send/Forward Settings* configured in the Message Profile. For details, refer "Message Send/Forward Settings".

- **Mailbox Management**: The system allows user to manage his/her personal mailbox. For further information, see "Mailbox Management".

Listening a Message

Listening a Message is a menu which will be played to the caller after every message (old/new).



- **Replay Message**: The system will play the message again.
- **Play Message Details**: The system will play the message details — Message Date/Time and the Name of sender. The message details will be played as per the *Message Playback Settings* configured in the Message Profile. For details, refer “Message Playback Settings”.
- **Reply Message**: The system will allow the caller to reply the sender with a message. The Message will be sent as per the *Message Leave Settings* configured in the Message Profile. For details, refer “Message Leave Settings”.
  After sending the reply message, the Mailbox Access menu will be played again.

  Reply Message will not be played if:
  - the message does not include the calling party number.
  - the message includes the calling party number but no Mailbox is assigned.
  - the message includes the calling party number as an external number.
  - it is a Broadcasted or a Call Tapping message.
- **Delete Message**: The system will delete the message and play the next message(old/new).
- **Listen Next Message**: The system will play the next message(old/new) as per the last message played.

  After playing the next message, the *Mailbox Access menu* will be played again.
- **Forward Message**: The system will play a prompt asking caller to enter the Destination Number of the recipient for forwarding the message.

  The Destination Number may include a Distribution List. A maximum of 10 Destination Numbers can be added. The Message will be sent as per the *Message Send/Forward Settings* configured in the Message Profile. For details, refer “Message Send/Forward Settings”.

  After forwarding the message, the *Mailbox Access menu* will be played again.

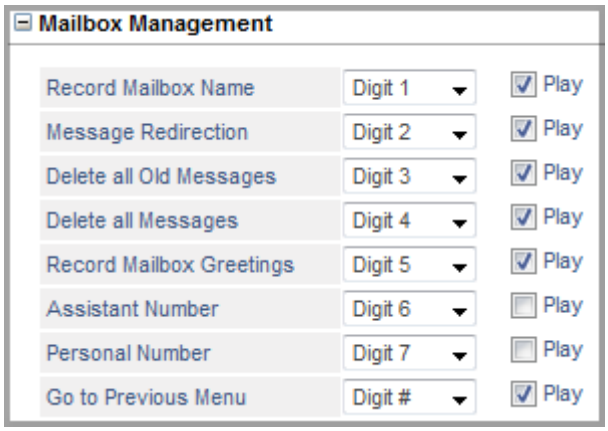
  The message will not be forwarded if the *Message Sensitivity* is set as *Private*. For details, refer “Message Profile”.

- **Save Message as New**: The system will save the message as a New Message keeping all the message properties — Date and Time, Caller, Sensitivity and Security — unchanged.

- **Go to Previous Menu**: The system will provide an option to the caller to go back to the "Mailbox Access" menu.

Mailbox Management

Mailbox Management is a menu used to manage the Personal Mailbox.



- **Record Mailbox Name**: The system will allow the caller to record, play or erase the Mailbox Name.

  The *Record Greetings/Name* options will be played where the caller can choose whether s/he want to Record, Play or Erase the Name. For details, refer "Record Greetings/Name".

- **Message Redirection**: The system will allow the caller to set or cancel Message Redirection. Message Redirection is used when the caller wants to forward all the messages to the other user's personal mailbox.

  The *Message Redirection* options will be played where caller can choose if s/he wants to set or cancel this feature. For details, refer "Message Redirection".

- **Delete all Old Messages**: The system will delete all the old messages from the caller's Personal Mailbox.

- **Delete all Messages**: The system will delete all the messages (old/new) from the caller's Personal Mailbox.

- **Record Mailbox Greetings**: The system will allow the caller to record, play or erase the Mailbox Personal and Conditional Greetings.

  The options will be played in the following sequence:

  - The *Mailbox Greetings* options will be played asking the caller to select the type of greetings — Personal or Conditional — which he wishes to Record, Play or Erase. For details, refer "Mailbox Greetings".

    - If the caller selects *Personal*, the *Personal Greeting Time zone Selection* options will be played. The caller must select the timezone — Working Hour, Break Hour or Non-Working Hour for which s/he wishes to Record, Play or Erase the personal greeting. For details, refer "Personal Greeting Timezone Selection".

- If the caller selects *Conditional*, the *Conditional Greetings* options will be played. The caller must select the type of Conditional Greeting — Busy, No-Reply or Unconditional for which s/he wishes to Record, Play or Erase the greetings. For details, refer “Conditional Greeting”.

- The *Record Greetings/Name* options will be played where the caller can choose whether s/he wants to Record, Play or Erase the Mailbox Greetings. For details, refer “Record Greetings/Name”.

- **Assistant Number**: The system will allow the caller to enter, play or clear the Assistant Number.

  The caller can call this number when the user is not available to answer the call.

  The *Number Programming (Assistant/Personal)* options will be played from where the caller can choose whether s/he wants to Enter, Play or clear the Assistant Number. For details, refer “Number Programming (Assistant/Personal)”

- **Personal Number**: The system will allow the caller to enter, play or clear the Personal Number.

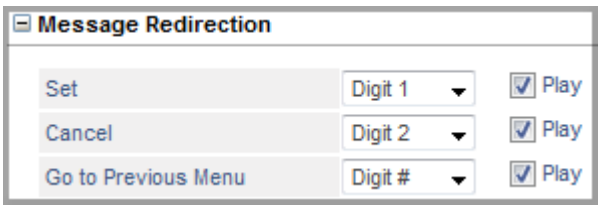
  The Personal Number is used as an alternate number. The caller can call this number when the user is not available to answer the call.

  The *Number Programming (Assistant/Personal)* options will be played from where the caller can choose whether s/he want to Enter, Play or clear the Personal Number. For details, refer “Number Programming (Assistant/Personal)”.

- **Go to Previous Menu**: The system will provide an option to the caller to go back to the “Mailbox Access” menu.

Message Redirection

Message Redirection is a menu which is applicable if the caller selects the *Message Redirection* option in “Mailbox Management”.



- **Set**: The system will set the Message Redirection feature and play a prompt to the caller for entering the Message Redirect Number.

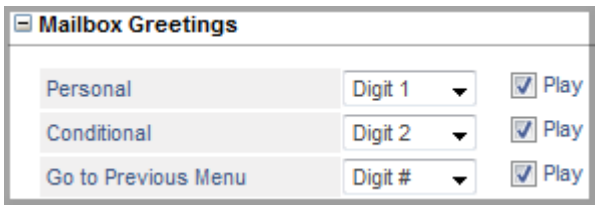
  The Message Redirect Number must be a valid Extension Number, Department Group Number or Extension over QSIG Number. A Mailbox must be assigned to this Number.

- **Cancel**: The system will clear the Message Redirect Number programmed by the caller.

- **Go to Previous Menu**: The system will provide an option to the caller to go back to the “Mailbox Management” menu.

Mailbox Greetings

Mailbox Greetings is a menu which is applicable if the caller selects the *Record Mailbox Greetings* option in the Mailbox Management.



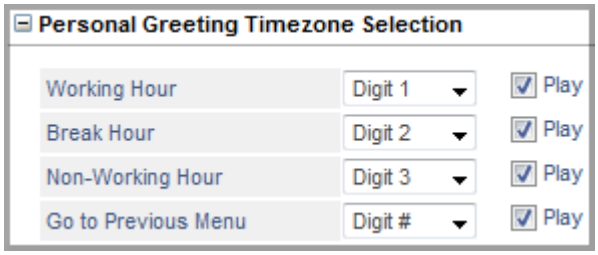
- **Personal**: The system will play the *Personal GreetingTimezone Selection* options. The caller must select the timezone — Working Hour, Break Hour or Non-Working Hour for which s/he wishes to Record, Play or Erase the Personal greetings. For details, refer “Personal Greeting Timezone Selection”.

- **Conditional**: The system will play the *Conditional Greetings* option. The caller must select the type of Conditional Greeting — Busy, No-Reply or Unconditional for which s/he wishes to Record, Play or Erase. For details, refer “Conditional Greeting”.

- **Go to Previous Menu**: The system will provide an option to the caller to go back to the “Mailbox Management” menu.

Personal Greeting Timezone Selection

Personal Greeting Timezone Selection is a menu which is applicable if the caller selects the *Personal* Greetings option in the Record Mailbox Greetings. You can record the personal greetings like “Good Morning”, “Good Afternoon” or “ Good Evening” to greet the caller.



The various Timezones are:

- **Working Hour**: The system will play the *Record Greetings/Name* options and allow the caller to Record, Play or Erase a personal greeting for the Working Hour Timezone.

- **Break Hour**: The system will play the *Record Greetings/Name* options and allow the caller to Record, Play or Erase a personal greeting for the Break Hour Timezone.

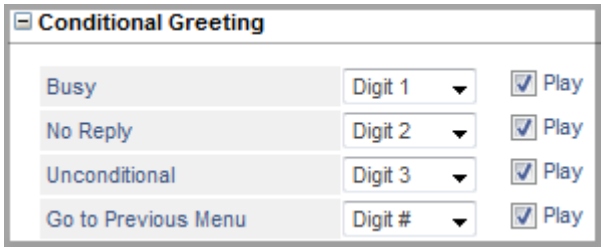
- **Non-Working Hour**: The system will play the *Record Greetings/Name* options and allow the caller to Record, Play or Erase a personal greeting for the Non-Working Hour Timezone.

- **Go to Previous Menu**: The system will provide an option to the caller to go back to the Record Mailbox Greetings Menu.

  For details, refer “Record Greetings/Name”.

Conditional Greeting

Conditional Greeting is a menu which is applicable if the caller selects the *Conditional* option in the Record Mailbox Greetings. Conditional Greetings are applicable if any call forward — Busy, No Reply or Unconditional — is set on the calling extension.



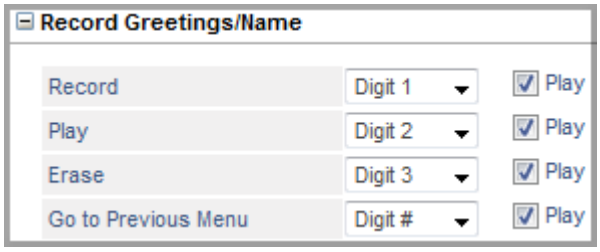
The various Conditional Greeting types are:

- **Busy**: The system will play the *Record Greetings/Name* options and allow the caller to Record, Play or Erase the Conditional Greeting of type — Busy.
- **No-Reply**: The system will play the *Record Greetings/Name* options and allow the caller to Record, Play or Erase the Conditional Greeting of type — No-Reply.
- **Unconditional**: The system will play the *Record Greetings/Name* options and allow the caller to Record, Play or Erase the Conditional Greeting of type — Unconditional.
- **Go to Previous Menu**: The system will provide an option to the caller to go back to the Record Mailbox Greetings Menu.

  For details, refer “Record Greetings/Name”.

Record Greetings/Name

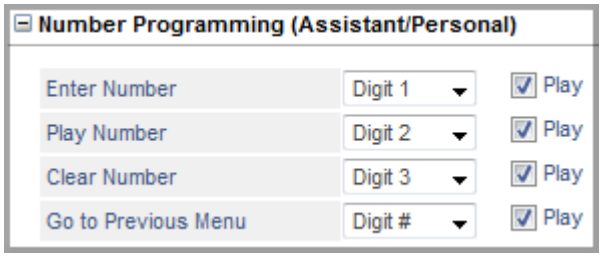
Record Greetings/Name is a menu which allows the caller to Record, Play or Erase the Mailbox Name or the Personal/Conditional Greetings.



- **Record**: The system will allow the caller to Record the Mailbox Name or the Personal/Conditional Greeting. Maximum allowed length for recording a name/greeting is 120 seconds.
- **Play**: The system will Play the Mailbox Name or the Personal/Conditional Greeting to the caller.
- **Erase**: The system will clear the recorded Mailbox Name or the Personal/Conditional Greeting.
- **Go to Previous Menu**: The system will provide an option to the caller to go back to the Previous Menu — *Record Mailbox Name, Personal Greeting Timezone Selection* or *Conditional Greeting*.

Number Programming (Assistant/Personal)

Number Programming (Assistant/Personal) is a menu which allows the caller to program the Assistant Number or Personal Number.



- **Enter Number**: The system will allow the caller to program the Assistant Number or Personal Number. For Assistant Number, only the Assistant Number List present in the system will be allowed.

Make sure the Assistance Number is an extension number and the Personal Number is an external (Mobile) number.

Make sure you do not enter the Department Group Mailbox Number as the Assistant Number or the Personal Number.

- **Play Number**: The system will play the Assistant Number or Personal Number to the caller.
- **Clear Number**: The system will clear the Assistant Number or Personal Number programmed by the caller.
- **Go to Previous Menu**: The system will provide an option to the caller to go back to the “Mailbox Management”.

Distribution List

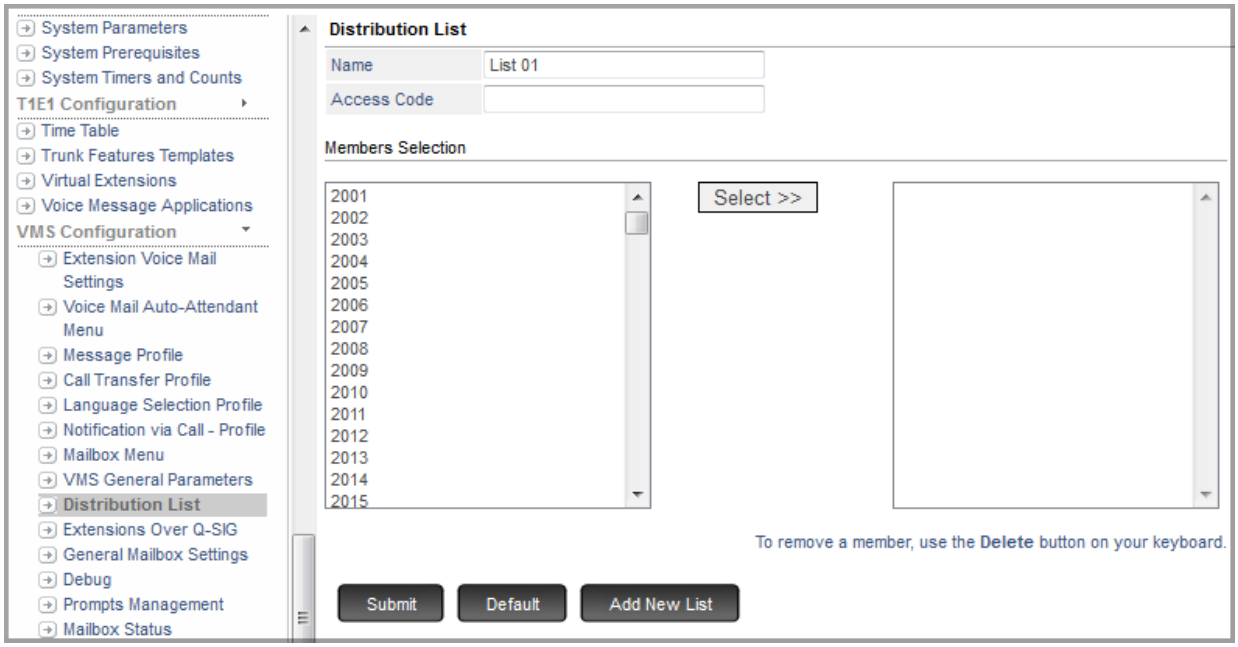
A Distribution List enables extension users to send the same message to a group of extensions at the same time.

Any extension with a mailbox can be included in a Distribution List. You can create upto 30 Distribution Lists of 50 members each.

How to configure

To configure Distribution List using Jeeves,

- Log in as System Engineer.
- Under **Configuration,** click **VMS Configuration**.
- Click **Distribution List**.



You may add a new list, edit the default list or delete a Distribution List.

To add a new List,

- Click **Add New List**. A new *List xx* will be created. You may now configure this as per your requirement.

To delete a list, click **Delete**.

The default Distribution List cannot be deleted.

To edit a Distribution List,

- In **Name**, configure the name of the Distribution List. By default, it is *List xx* where xx is the Distribution List Number from 01 to 30.

- In **Access Code**, configure the Access Code you wish to assign to the respective Distribution List. The caller can send or forward the message to a Distribution List by dialing the respective access code. It can be a maximum of 6 digits.

Make sure, the Access Codes assigned to the Distribution Lists do not conflict with the existing access codes. System will not save the configured Distribution List access code if the same code is already assigned.

Members Selection

- All the Extension Numbers appear in the left side box arranged sequentially in the increasing order.

  - To select the members for the Distribution List,

    - Place your cursor on the desired Extension and click **Select>>** button. The selected extension will appear on the right side box. It is also possible to select a range of extensions at a time by pressing 'SHIFT' and 'Down' arrow keys.

      A maximum of 50 extensions can be selected.

  - To delete the members from the Distribution List,

    - Place your cursor on the desired Extension and click delete key from your keyboard. The selected extension will be deleted. It is also possible to select a range of extensions at a time by pressing 'SHIFT' and 'Down' arrow keys.

- Click **Submit** to save changes.

Recording Voice Messages

The VMS of SARVAM UCS supports voice messages for different functions, which are broadly classified as:

- **System Greetings**: These are messages are played to the caller when a new call lands on the VMS. Callers are greeted according to the time of the day - morning, afternoon, evening (Time Zone). You can customize the Time Zones as per your requirement. For detailed instructions, see “Greeting Message Time”. A different System Greeting can also be played to callers on holidays.

  System Greetings are played to callers when the VMS Auto Attendant feature is enabled on trunks.

- **Personal Greetings:** These messages are played to callers when they are diverted to the extension user’s mailbox to leave a message. Extension users can record personal mailbox greeting messages of their choice.

- **Conditional Greetings:** These messages are played to callers when they are diverted to the extension user’s mailbox for certain conditions—busy, no reply or unconditional/unregistered Call Forward. Extension users can record a different message for each condition.

- **Welcome Messages:** These are messages are played to callers who call the VMS. Welcome messages help the callers navigate through the VMS. Welcome messages are played according to the time of the day, that is, the Time Zone programmed in the system.

  Welcome Messages are played to callers when the VMS Auto Attendant feature is enabled on trunks.

- **Holiday Messages:** These messages are played to callers who call the VMS on a Holiday. These messages are played in place of the Welcome messages and help the callers navigate through the VMS. A different message can be played for each holiday.

  Holiday Messages are played to callers when the VMS Auto Attendant feature is enabled on trunks and the Holiday Table is configured.

- **Prompts/Responses:** These are voice guidance messages that are played to the caller in response to the action taken (that is, when the caller dials a digit).

For all of these message types, audio files containing the appropriate recorded voice guidance messages are loaded in the configuration of the VMS Module. The VMS plays the messages related to the function it is performing.

For example, if Voice Mail Auto Attendant (the VMS Auto Attendant feature) is enabled on a trunk, the VMS plays messages relevant to the Voice Mail Auto Attendant Menu programmed for the current Time Zone. This helps the caller navigate through various options as the VMS plays the related message, as explained below:

- The VMS plays the default System Greeting and Welcome message to the caller according to the time of the day, e.g.: *“Good Morning”. “Welcome! Please dial the extension number or To dial by name press ‘6’,To leave a message press ‘7’, To access your Personal Mailbox press ‘8’, For further assistance press ‘9’, To disconnect the call press ‘#’ (hash)”.*

- As the caller navigates, the VMS plays the pre-recorded voice messages related to the particular option selected by the caller. If the caller dials 6 to dial by name, the VMS plays relevant voice message, e.g.: "Please enter first three letters of the name" and then plays “More than one match found. Matching Names

will be played one by one. To Select the name press ‘1’, to Skip the name press ‘2’, To Repeat the last name press ‘3’.”

- If the caller dials 1 to select the name, the VMS plays the prompt: “To confirm press ‘1’, to Re-enter press ‘2’.”

- If the caller dials 1 the VMS transfers the call as per the transfer type assigned to the selected station.

In the same way, when you set a voice guided alarm, the VMS plays the alarm-related voice prompts, like: *“Enter the time, HH MM in twenty four hour format*”. Thus, for every voice mail related function or feature, the VMS plays the appropriate voice message.

The VMS gives you the option of either using the default voice guidance messages loaded in the VMS Module configuration, *or* recording custom messages that better suit your purpose.

All VMS default voice messages are in English only. If you want, you may record voice messages in your local language.

No special programming is required for using the default voice messages. However, if you want to use custom messages, you must first:

- record the message (Greetings, Welcome Messages, Holiday Messages, Voice Guidance prompt).
- upload the new recorded message file in the VMS configuration.

Recording Voice Messages

When you record messages of your choice, consider these important points:

- The custom messages must be in WAV format.

- Make sure the message files have the following attributes:
  - Audio Format: CCIT u-law
  - Channel: 1 (mono)
  - Sampling Frequency: 8 KHz
  - Audio Sample Size: 8 bit

- You must record the custom messages from an external source.

- You must verify the message and then these messages must be uploaded on to the VMS configuration files.

- Extension users can record their personal mailbox greetings on their own. Refer the topic “Recording Personal Greetings” to know more.

- Extension users can record the conditional greetings on their own. Refer the topic “Recording Conditional Greetings” to know more.

Uploading Custom Voice Messages

As mentioned earlier, when you record voice messages from an external source, make sure that:

- The audio file is recorded in the prescribed format (.wav) and attributes.

- The SARVAM UCS is connected to a computer (standalone or LAN).
- You can upload a single prompt or the entire folder. For detailed instructions, see “Prompts Management”

Prompts Management

Prompts Management allows you to view all the prompt folders present in the system for each of the configured languages. You may upload or download a prompt file to/from the VMS Prompt folders for any language. The default prompts recorded in English language are provided to you. See “VMS Prompts” for details.

You may overwrite the default prompts or upload new prompts. While uploading the prompts for other languages, you must record them with reference to the default prompts listed under the topic ***VMS Prompts***.

There are 21 Prompt folders present in the system. The first 11 folders are flexible while the others are fixed.

The Flexible Folders are:

- Greeting
- Auto-Attendant
- Number Dialing
- No Digit Dialed
- Invalid Digit Dialed
- Expiry of Count
- Call Transfer Type
- Call Transfer Unsuccessful
- Information
- MoH
- Disconnect

The Fixed Folders are:

- Language
- Dial by Name
- Call Transfer
- Message Record
- Message Send Forward
- Mailbox Access
- Mailbox Access Menu
- Number
- Alarm
- Miscellaneous

The Flexible Folders include the prompt files which the system uses according to the configuration done by you. You may upload new prompt files, download and delete the existing files from the system.

The Fixed Folders include the prompt files which the system uses internally and are non-configurable. You may overwrite or download the existing files but cannot delete these from the system.

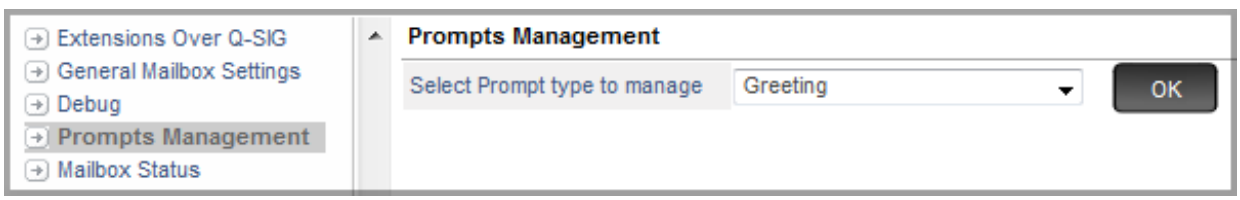
The following table defines the maximum number of prompt files supported in each folder.

Folder Name	Maximum Prompt files supported	Maximum length allowed for each prompt (seconds)
Greeting	24	180
Auto-Attendant	99	180

Folder Name	Maximum Prompt files supported	Maximum length allowed for each prompt (seconds)
Number Dialing	12	120
No Digit Dialed	12	120
Invalid Digit Dialed	12	120
Expiry of Count	12	120
Call Transfer Type	12	120
Call Transfer Unsuccessful	12	120
Information	24	180
MoH	12	180
Disconnect	12	120
Language	01	120
Dial by Name	13	120
Call Transfer	16	120
Message Record	19	120
Message Send Forward	30	120
Mailbox Access	29	120
Mailbox Access Menu	50	120
Number	46	120
Alarm	53	120
Miscellaneous	24	120

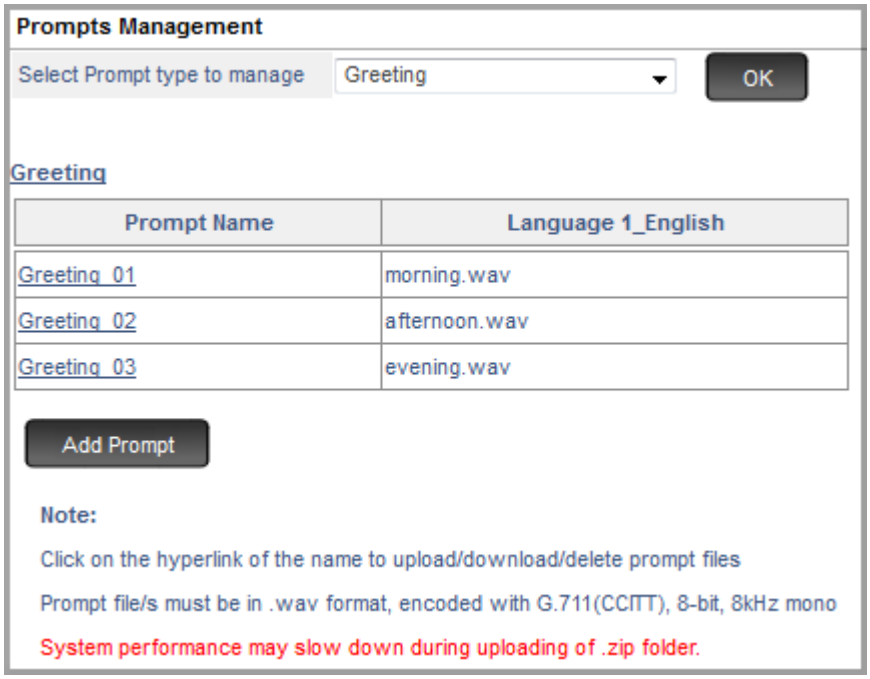
How to Configure

- Log into Jeeves.
- Under **Configuration**, click **VMS Configuration**.
- Click **Prompts Management**.



- In **Select Prompt type to manage**, select the desired folder.
- Click **OK**.

All the prompt files present in the respective language folders of the selected Prompt Type will be displayed.



You can upload or download the entire folder for the respective Prompt Type for each of the languages.

To add a new prompt in a Flexible Folder,

- Click the **Add Prompt** button at the bottom of the page.

  The **Add/Edit Prompt** window opens.

- Click the **Browse** button to reach the location on the local disk where the respective prompts are stored in your PC.

- Click to upload.

Make sure the folder you upload is a zip file.

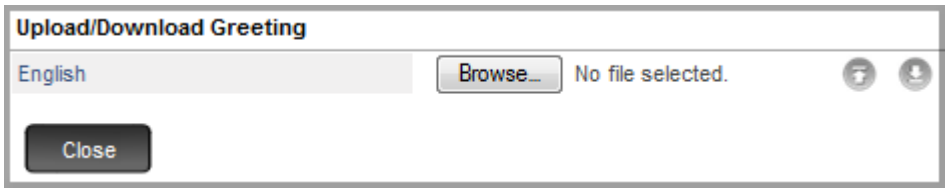
Make sure the prompt files within the folder are in .wav format and have the following attributes:

- *Audio Format: u-law*
- *Channel : 1 (mono)*
- *Sampling Frequency: 8 KHz*
- *Audio Sample Size: 8 bit*

To Upload or Download the entire Prompt Type folder,

- Click on the **Prompt Type** link.

The **Upload/Download Prompt Type** Window opens.

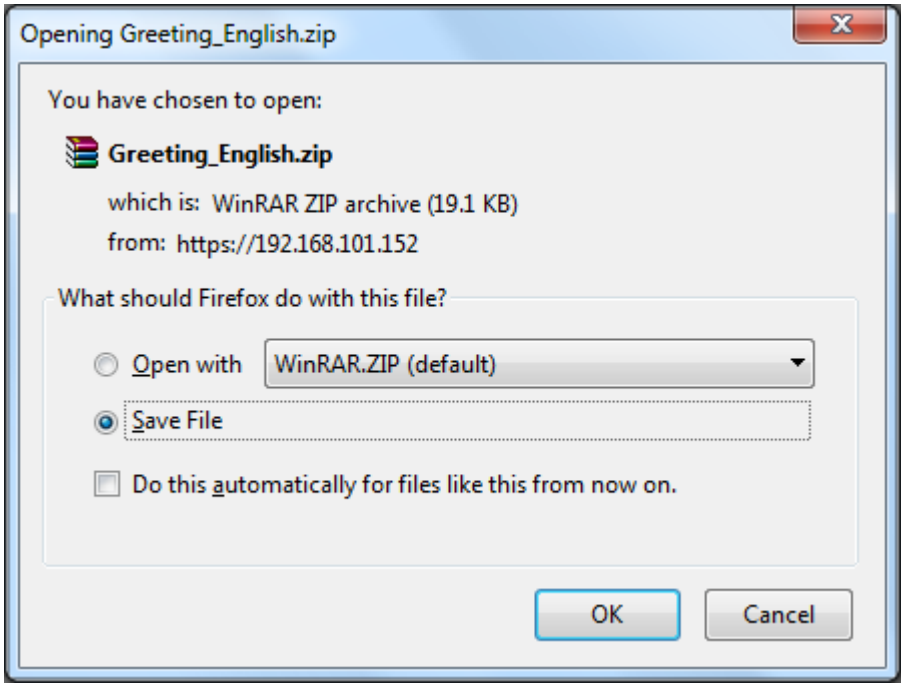


- The language-wise folders will be displayed.
- Click the **Browse** button of the desired language to reach the location on the local disk where the respective prompt folder is stored in your PC.
- Click to upload.

Make sure, the file to be uploaded is a zip file.
The filename can be a maximum of 64 characters.
The allowed characters for the filename are A to Z, a to z, 0 to 9, - and _.

To download all the prompts of the respective prompt type and language selected, click .

The **Opening Prompt Type_Language.zip** window will open.

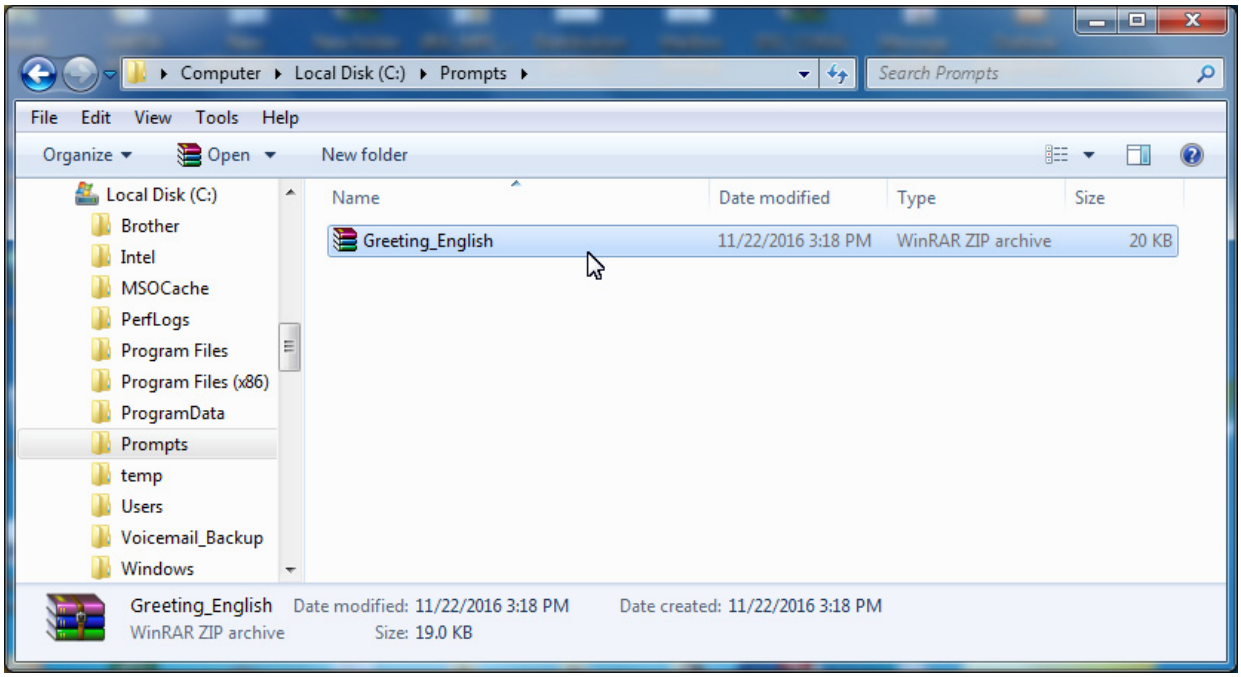


- You can either open the zip file or save the file to a location.

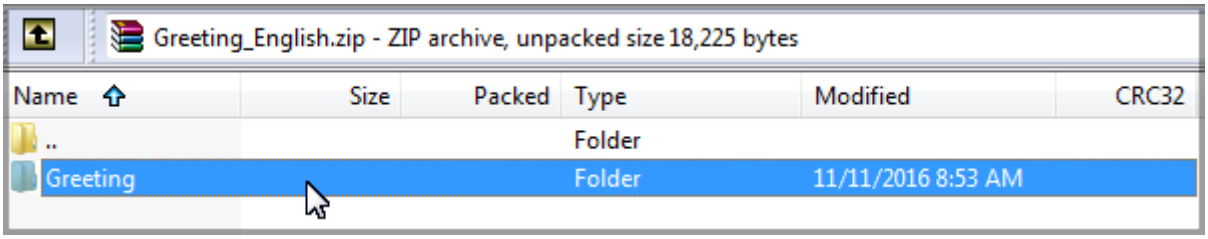
*The above window display depends upon the browser you are using. Check the **Download Settings** of your browser and set the Download path accordingly.*
OR
If your browser does not ask you for the location you want to save your file, it saves it in the default location according to the download path specified for that browser.

*If you are using Mozilla Firefox (version 3.5 recommended), before you save the configuration files, set the **Downloads** option of your browser as **Always ask me where to save files**.*

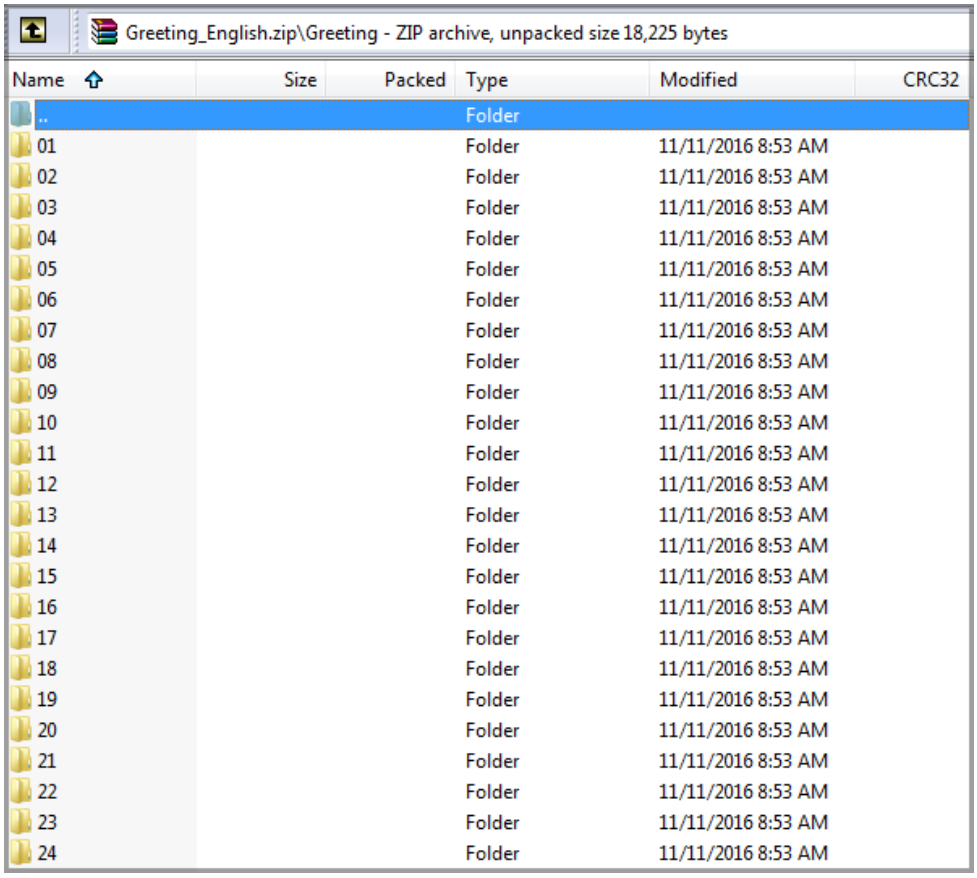
- Save the file on the local disk.



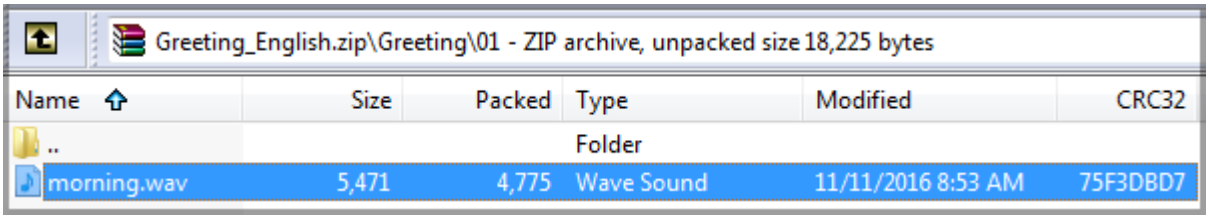
- To view the files,
  - Open the **Prompt Type_Language** folder.



- Now, open the **Prompt Type folder** to view all the subfolders containing the prompts.



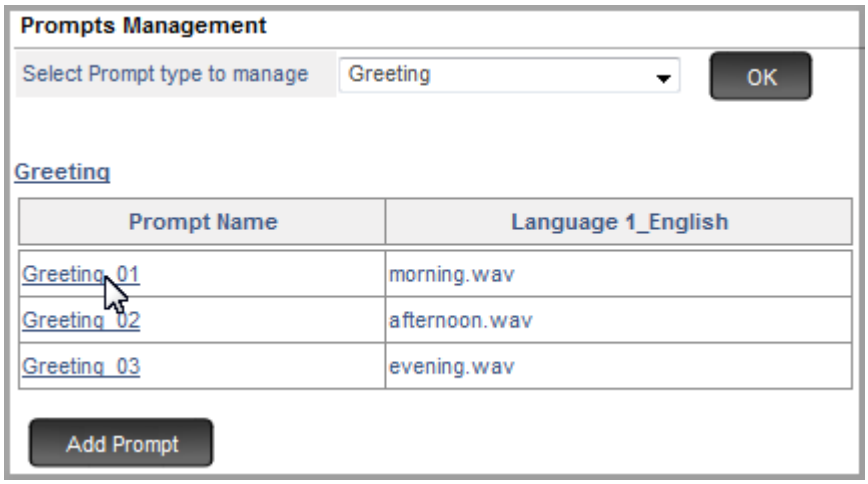
- Open the subfolder containing the prompt.



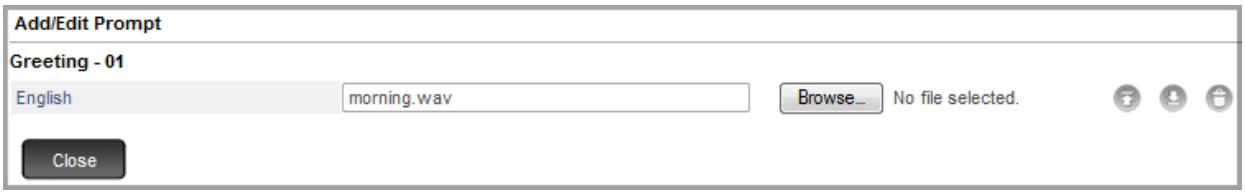
You may now upload, download or delete a prompt file to/from a folder. The files present in the Fixed Folders cannot be deleted or added, but the existing files can be overwritten by a new one.

To Upload or Download a Prompt,

- Click on the respective **Prompt Name** link.



The **Add/Edit Prompt** window will open.

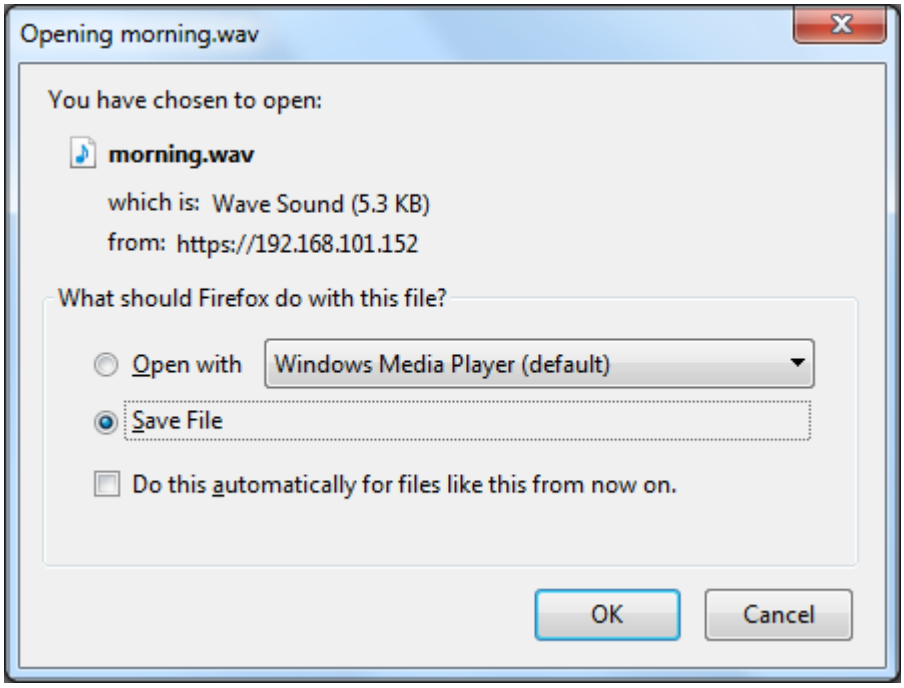


- To upload or overwrite a prompt,
  - Click the **Browse** button to reach the location on the local disk where the respective prompts are stored in your PC.
  - Click to upload.

Make sure, the file to be uploaded is a wav file.
The filename can be a maximum of 64 characters.
The allowed characters for the filename are A to Z, a to z, 0 to 9, - and _.

- To download a prompt, click .

The **Opening Prompt Name_xx.wav** window will open; where xx signifies the prompt number.

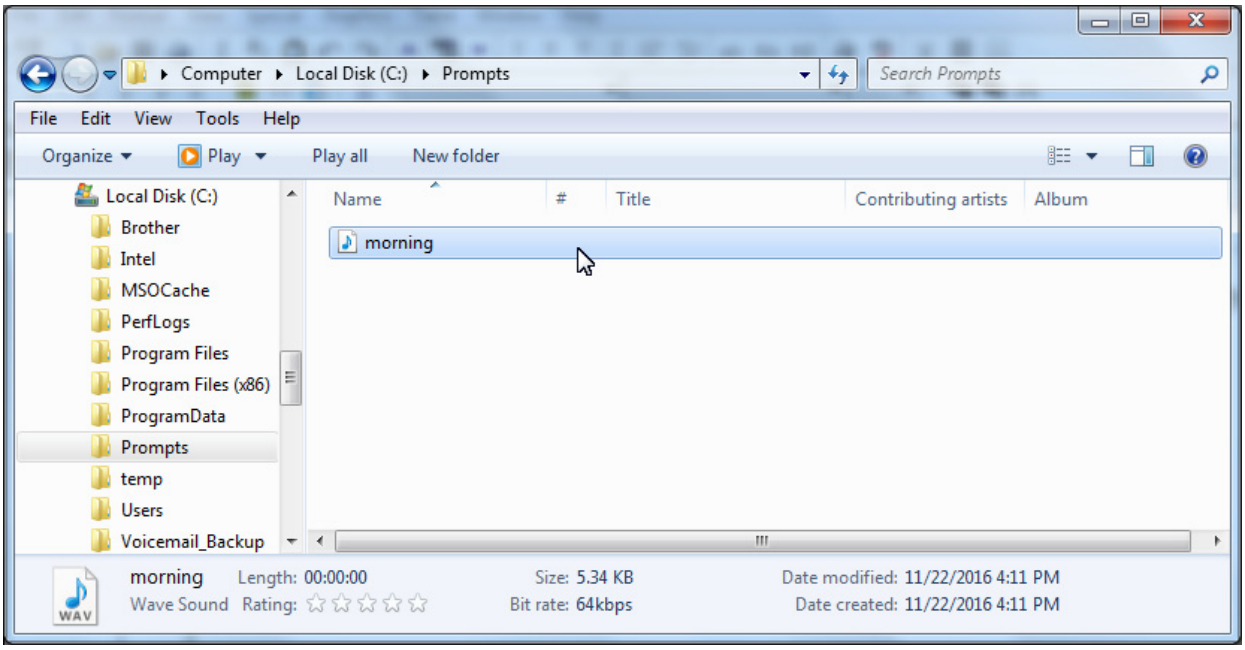


- You can either open the wav file or save the file to a location.

*The above window display depends upon the browser you are using. Check the **Download Settings** of your browser and set the Download path accordingly.*
OR
If your browser does not ask you for the location you want to save your file, it saves it in the default location according to the download path specified for that browser.

*If you are using Mozilla Firefox (version 3.5 recommended), before you save the configuration files, set the **Downloads** option of your browser as **Always ask me where to save files**.*

- Save the file on the local disk.



- To delete a prompt, click .

For Firmware Versions earlier than V1R7.1,

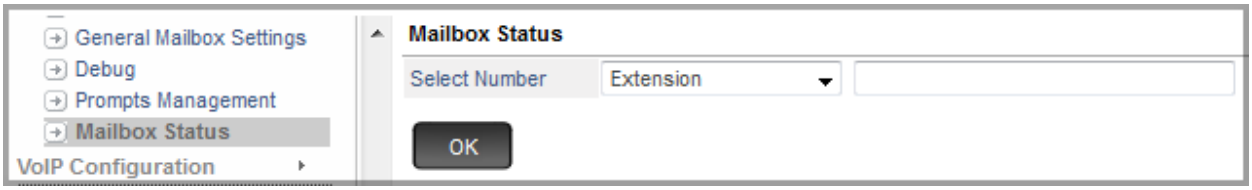
- *If you wish to upload customized prompts in the Miscellaneous folder, follow the same instructions as mentioned above to download the folder and copy the customized prompts in the desired folders.*
- *You also need to make sure that the Folder number 01 has only one file you need, delete the other file. Only then you will be able to upload the folder again with customized prompts.*

Mailbox Status

The Mailbox Status displays the mailbox details of the respective Extension, Department Group or General Mailbox.

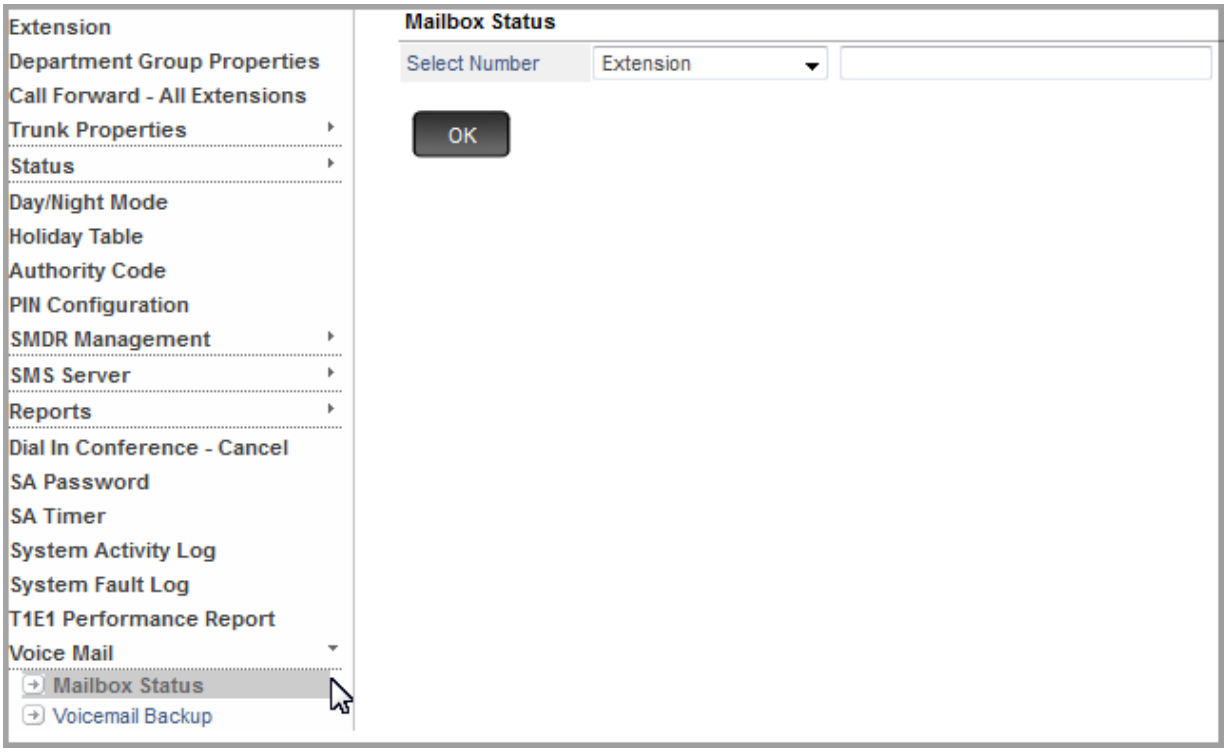
Viewing Mailbox Status

- Login as System Engineer.
- Under **Configuration**, click **VMS Configuration**.
- Click **Mailbox Status**.



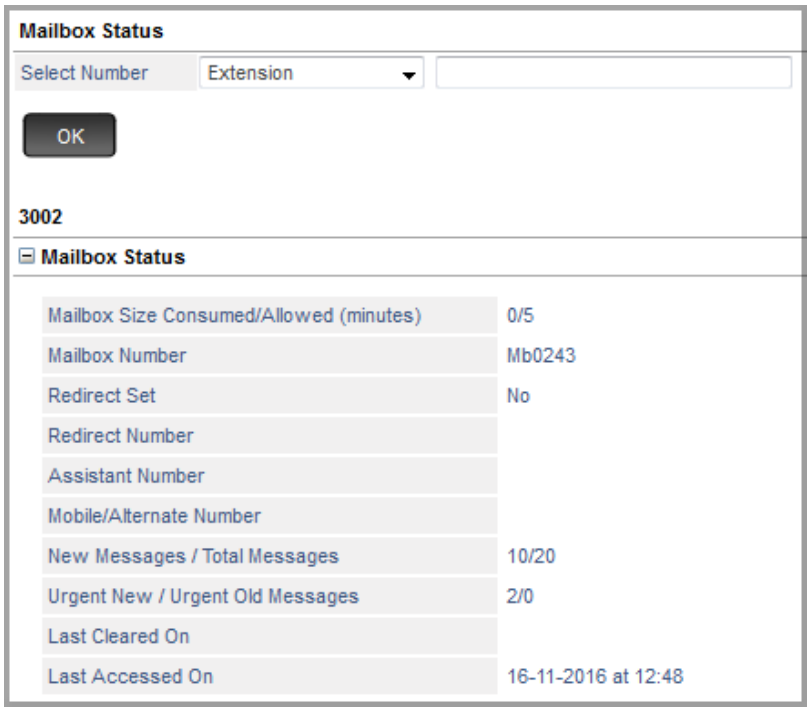
OR

- Log in as System Administrator.
- Under **Voice mail**, click **Mailbox Status**.



- **Select Number:** You may select — Extension, Department Group or General Mailbox — whose mailbox status you wish to view.
    - For **Extension**, enter the Extension Number/Name and then select the same from the drop down list.
    - For **Department Group**, select the desired Department Group Number from the drop down list.
- Click **OK**.

- Click **Mailbox Status** to expand.



The respective mailbox parameters along with their status will be displayed.

- **Mailbox Size Consumed / Allowed (minutes)**: It displays the minutes consumed by the Mailbox messages out of the maximum minutes allowed for the Mailbox.
- **Mailbox Number**: It displays the mailbox number assigned to the respective Extension or Department Group or the General Mailbox.
- **Redirect Set**: It displays 'Yes' if the Redirect is set and 'No' if the Redirect is not set for the respective mailbox user.
- **Redirect Number**: It displays the Redirect number, if configured, for the mailbox user.
- **Assistant Number**: It displays the Assistant number, if configured, for the mailbox user.
- **Mobile/Alternate Number**: It displays the Mobile/Alternate number, if configured, for the mailbox user.
- **New Messages / Total Messages**: It displays the number of New Messages out of the Total Messages available in the respective mailbox.

- **Urgent New / Urgent Old Messages**: It displays the number of New Urgent Messages as well as the Old Urgent Messages available in the respective mailbox.
- **Last Cleared On**: It displays the date and time when all the Mailbox Messages (old and new) were last cleared by the user or SE.
- **Last Accessed On**: It displays the date and time when the Mailbox was last accessed.

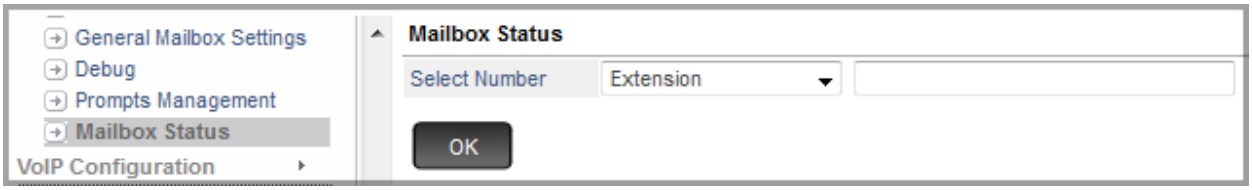
Delete Voice Messages

Using Jeeves

***Delete Voice Messages** will be displayed only when the Mailbox is assigned and it contains old or new voice messages.*

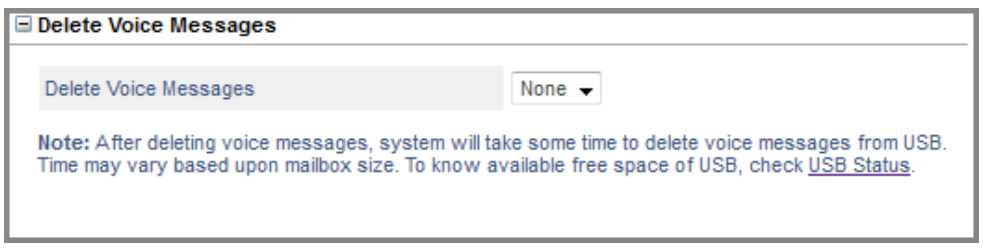
Delete Voice Messages from SE mode,

- Log into Jeeves as System Engineer.
- Under **Configuration**, click **VMS Configuration**.
- Click **Mailbox Status.**



- **Select Number:** You may select — Extension, Department Group or General Mailbox — whose mailbox messages you wish to delete.
  - For **Extension**, enter the Extension Number/Name and then select the same from the drop down list.
  - For **Department Group**, select the desired Department Group Number from the drop down list.
  - For **General Mailbox**, select the General Mailbox Number from the drop down list.
- Click **OK**.

- Click **Delete Voice Messages** to expand.



- From the **Delete Voice Messages** list, select the type of messages you want to delete — None, All, New or Old. By default, None is selected.
- Click on the **Delete** button.
  - A confirmation message is displayed. Click **OK** to delete the messages.

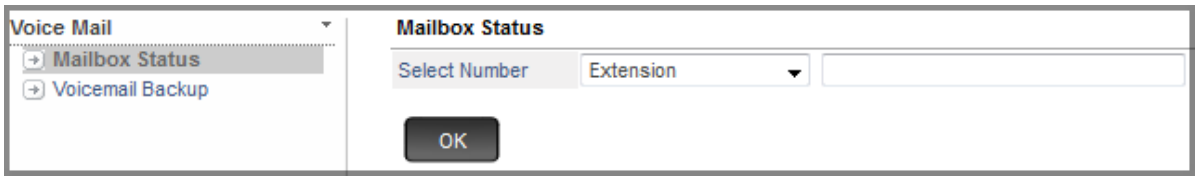
- Log out of Jeeves, or continue as desired.

Delete voice messages *is not applicable for Personal Greetings, Conditional Greetings and Extension Name.*

After deleting voice messages, system will take some time to clear them. The time taken to clear the messages will vary depending upon the mailbox size.The USB Status will be updated only after these messages have been cleared.

Delete Voice Messages from SA mode

- Log into Jeeves as System Administrator.
- Click **Voice Mail** to expand.
- Under **Voice Mail,** click **Mailbox Status.**



- Follow the same steps as given above to delete the voice messages.
- Log out of Jeeves, or continue as desired.

Using SA Command[227]

You can delete voice messages using SA command also.

- Enter SA mode from a DKP/SLT.

  To delete the messages, dial:

  - ***1072-315-Extension Number-Message type***
    Where,
    Extension Number can be the number of SLT, DKP or SIP Extensions, ISDN Terminals, Extension over QSIG, Department Group or the General Mailbox.

    Message Type is
    1 - All Messages
    2 - Old Messages

227. SA Commands for deleting voice messages cannot be executed from VARTA UC Clients, SPARSH VP330 or SPARSH VP210.

3 - New Messages.

- Exit SA mode.

Voicemail Backup

The Voicemail Backup allows you to store the Backup of the desired — Extensions, Department Groups, General Mailbox— voicemail messages on the Network Drive. For details regarding the Network Drive configuration, see “Network Drive Settings”.

SARVAM UCS allows you to take the voicemail backup of a single extension, range of extensions, all extensions or selected extensions.

You can take Voicemail Backup, either

- Manually: The backup is taken whenever you want.
  Or
- as per Schedule: The backup is taken on a preset Day, Date and Time.

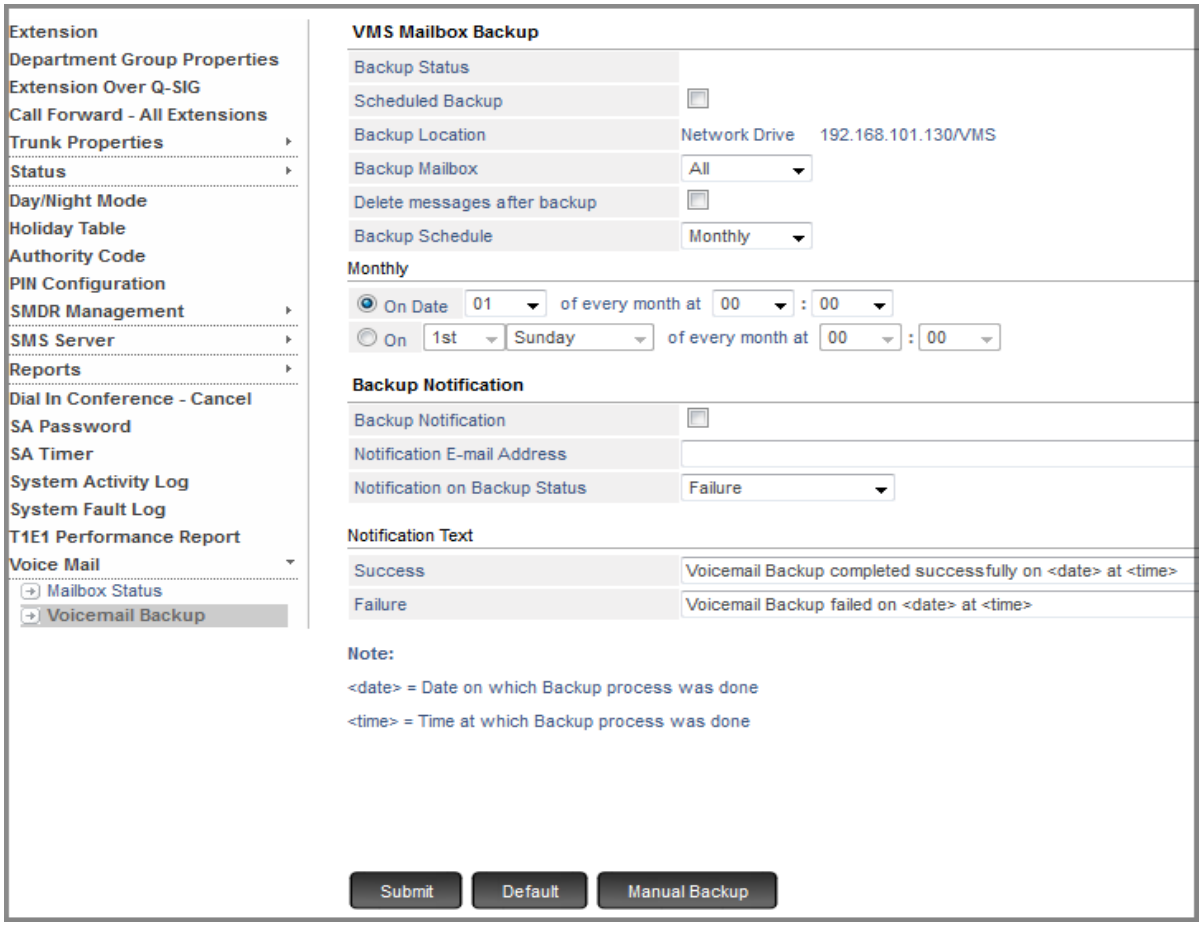
- *If you want to store the backup files in a PC having Windows as the Operating System, make sure it has IPv4 address.*
- *Backup is not possible in Apple PCs.*

How to Configure

Scheduled Backup

- Log in as System Administrator.
- Click the **Voice Mail** link.

- Under **Voice Mail**, click **Voicemail Backup**.



To schedule automatic Voicemail Backup configure the following:

- **Backup Status:** Displays the last scheduled/manual backup status (Successful or Failed) with date, time, type of backup done and location at which backup is stored. By default, this field is blank. When the Voicemail Backup is on-going, this displays the progress status of the backup. However, if you wish to abort the backup midway, you may click the **Abort** button.

***Abort** button will be visible only when the Voicemail Backup is in progress.*

- **Scheduled Backup:** Allows you to schedule automatic Voicemail Backup on a specific day, date and time. To set a scheduled backup, enable this check box. By default, it is disabled.

- **Backup Location:** Displays the path of Network drive with the Folder Name configured in the Network Drive Settings through SE mode. If the parameters have not been configured for the Network Drive, it will display the error message here.

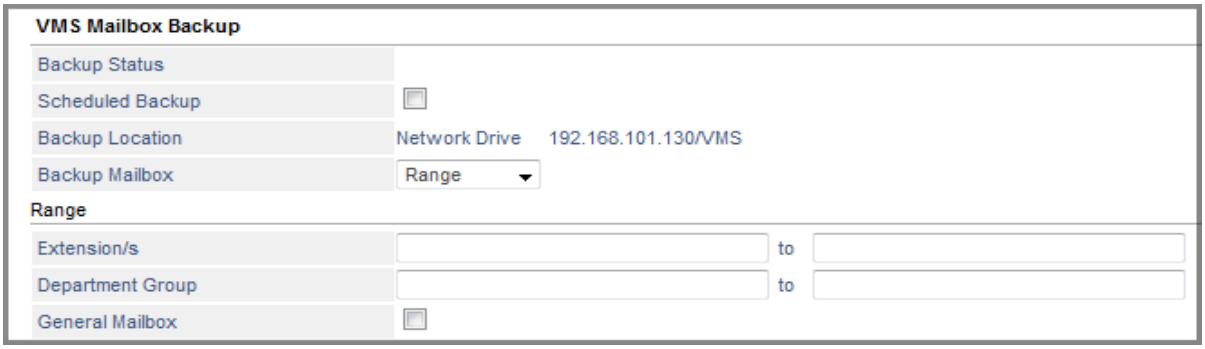
If storage capacity of the Network drive is full, then backup will not be taken and an error is displayed.

- **Backup Mailbox:** This allows you to select the mailbox/es for which the backup needs to be taken.

  - To backup voicemail of all extensions, select **All** Extensions. This would take the backup of all mailboxes — Extensions, Department Groups and General mailbox. By default, All option is selected.

- To backup voicemail of a single extension, select **Extension**. Enter the Extension Number/Name and then select the same from the drop down list.

- To backup voicemail of multiple extensions in sequence, select **Range**.

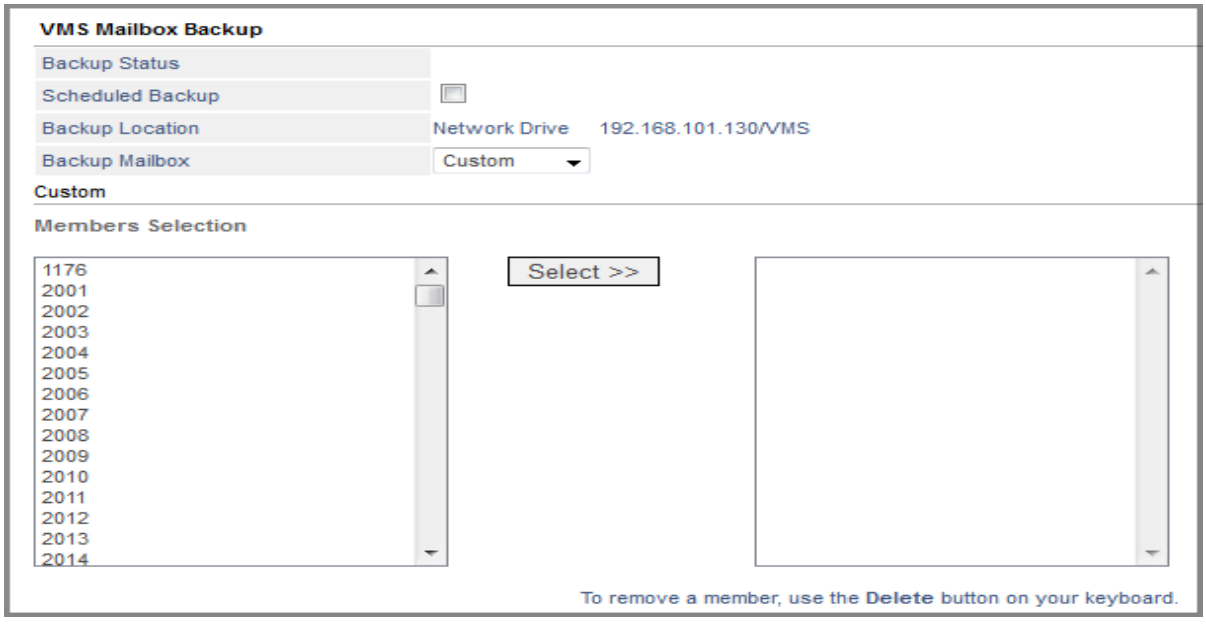


  - To backup voicemail of a range of Personal mailboxes, enter the starting and ending **Extension** Numbers and then select the same from the drop down list.

  - To backup voicemail of a range of **Department Group** mailboxes, enter the starting and ending Department Group Numbers and then select the same from the drop down list.

  - To backup voicemail of the **General Mailbox**, enable this check box.

- To backup voicemail of randomly selected extensions from the list, select **Custom**.



  - Select the required Extension numbers, Department Group or General mailbox from the **Member Selection** list and click **Select**. The selected mailboxes will be displayed in the right-side panel.

  - To remove a selected member, click on the desired extension in the right-side panel and press Delete key on your keyboard.

- Select the **Delete messages after backup** check box if you want the system to delete the respective extension/s voicemail messages after their backup has been stored on the network drive.

- To set an automatic **Backup Schedule**, select the desired option — Hourly, Daily, Weekly or Monthly.

Voicemail Backup will be taken only when the System's time matches with the configured time in the **Backup Schedule.**

To receive Backup Notification, configure the following:

- You will receive **Backup Notification** only if this check box is enabled. By default, it is disabled.
- If you have opted for Backup Notification, then enter the desired email id in **Notification E-mail Address** to which the notification should be sent.

Voicemail Backup E-mail notification will be sent only when SMTP account is configured in System Log Notification. For instructions, see "System Log Notification" and "SMTP Settings".

- In **Notification of Backup Status**, select the desired status for which you require notification — Failure, Success or Success + Failure. By default, Failure is selected.
- In **Notification Text: Success**, enter the text you would like to receive as subject line in email when the Voicemail Backup has been successful. By default, the text is **Voicemail Backup completed successfully on <date> at <time>.**
- In **Notification Text: Failure**, enter the text you would like to receive as subject line in email when the Voicemail Backup has failed. By default, the text is **Voicemail Backup failed on <date> at <time>.**

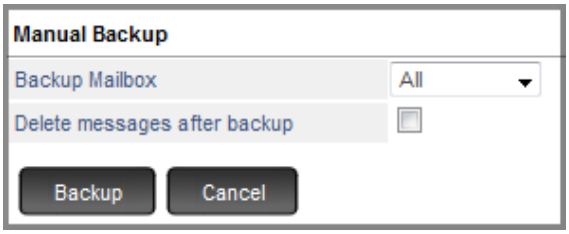
  Date-Time will be replaced with the date and time of the system when manual/scheduled Voicemail Backup was taken.

Manual Backup

If the Network drive is not configured, then Manual Backup option will not be displayed.

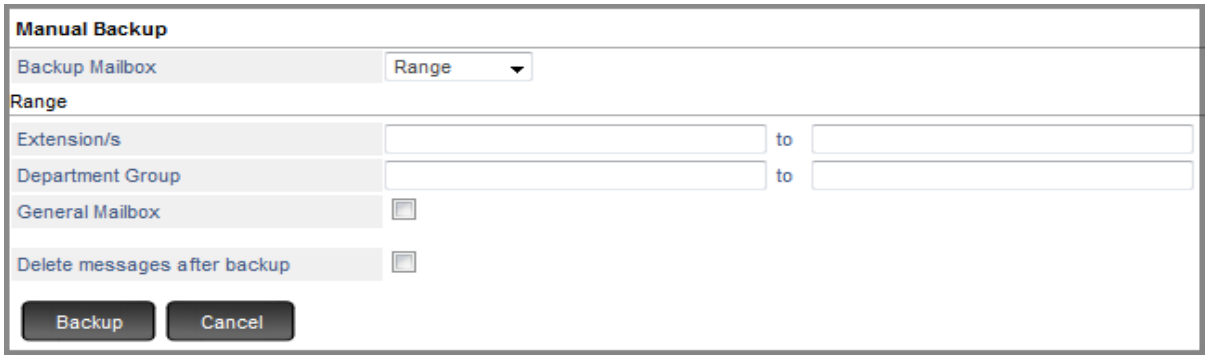
To take the Voicemail backup manually,

- Click on **Manual Backup** button. A new window for Manual Backup is opened.



- **Backup Mailbox** allows you to select the mailbox/es of which the backup needs to be taken.
  - To backup voicemail of all extensions, select **All** Extensions. This would take the backup of all mailboxes — Extensions, Department Groups and General mailbox. By default, All option is selected.
  - To backup voicemail of a single extension, select **Extension**. Enter the Extension Number/Name and then select the same from the drop down list.

- To backup voicemail of multiple extensions in sequence, select **Range**.

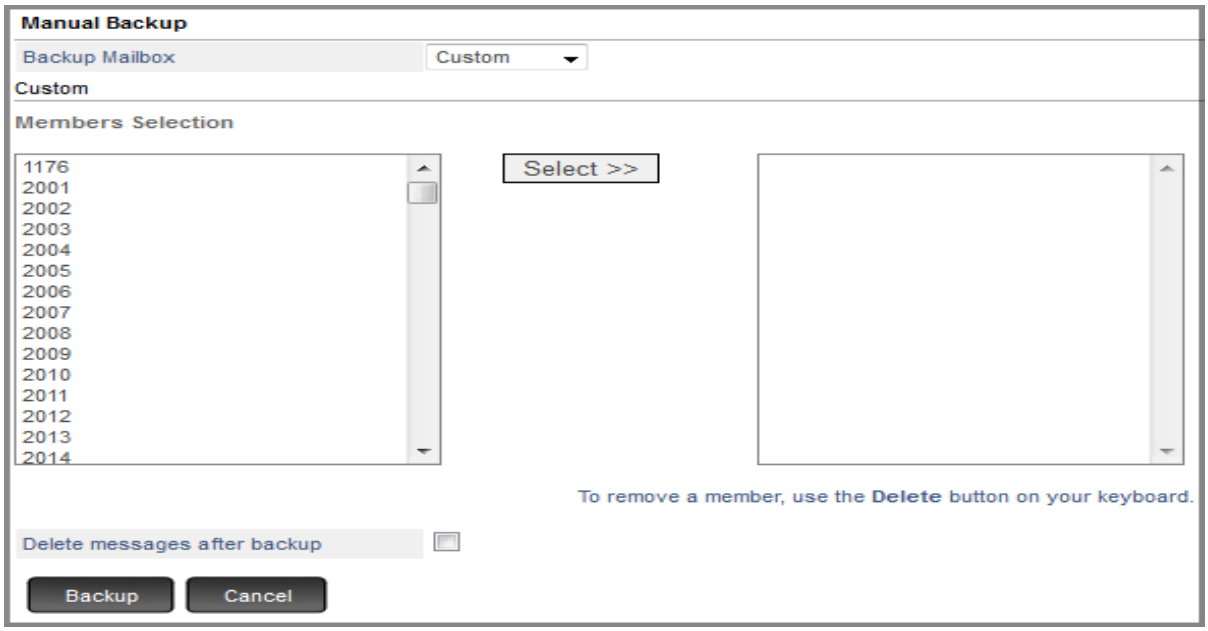


  - To backup voicemail of a range of Personal mailboxes, enter the starting and ending **Extension** Numbers and then select the same from the drop down list.

  - To backup voicemail of a range of **Department Group** mailboxes, enter the starting and ending Department Group Numbers and then select the same from the drop down list.

  - To backup voicemail of the **General Mailbox**, enable this check box.

- To backup voicemail of randomly selected extensions from the list, select **Custom**.



  - Select the required Extension numbers, Department Group or General mailbox from the **Member Selection** list and click **Select**. The selected mailboxes will be displayed in the right-side panel.

  - To remove a selected member, click on the desired extension in the right-side panel and press Delete key on your keyboard.

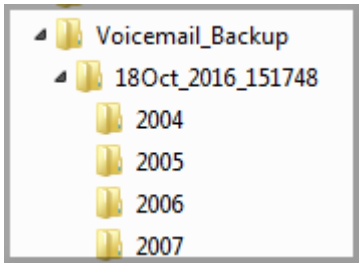
- Select the **Delete messages after backup** check box if you want the system to delete the respective extension/s voicemail messages after their backup has been stored on the network drive.

*If you click **Backup** and a Scheduled Backup is in progress, then Manual Backup will not be taken and an error is displayed.*

Folder Structure

The Voicemail Backup of the respective Extension/s will be stored in the *Shared Folder* you configure in Network Drive Settings.

The folder structure will be as shown below:



The **Voicemail_Backup** folder consists of the folders with the folder names defined as per the Date and Time of the backup.

The Folder Names will be as per the Date and Time Format you configure in *VMS General Parameters*.

Supported formats for folder names are DDMonthname_YYYY_HHMMSS and DDMonthname_YYYY_HHMMSS_PM/AM.

Where,
DDMonthname_YYYY represents the Date, *HHMMSS* represents the Time in 24 Hour format and *HHMMSS_PM/AM* represents the Time in 12 Hour format.

The sub-folders with the Extension Names contains the respective extension's voicemail files.

VMS Debug

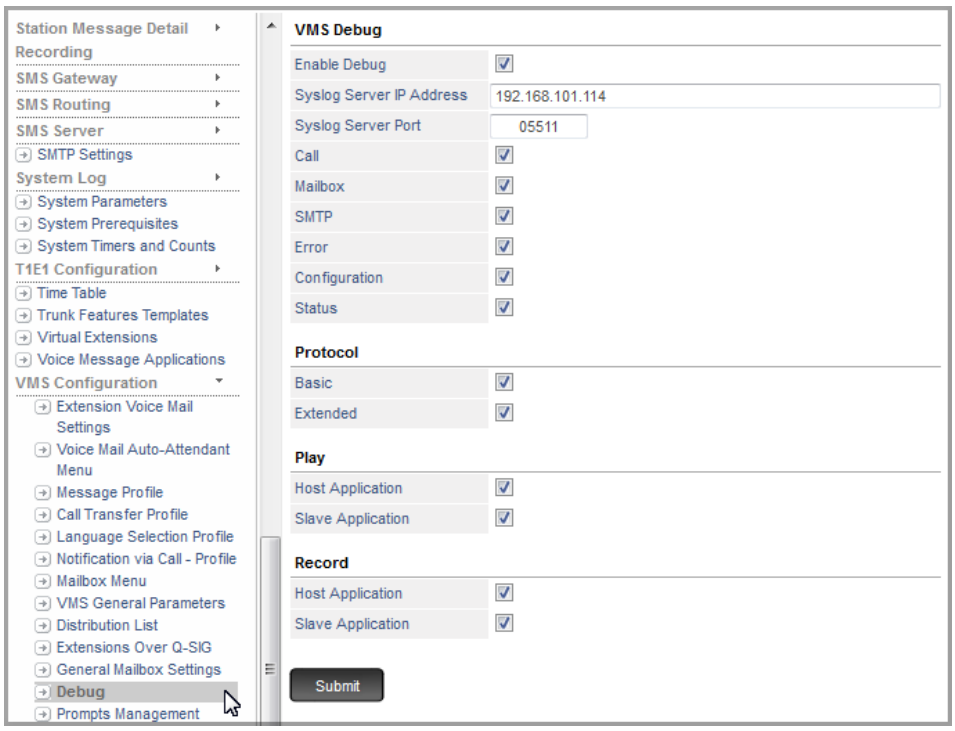
The VMS supports debug for the VMS Application and SMTP. You can view debug messages on the Syslog Server[228].

To be able to use Syslog for debug, you will need to:

- connect a PC to the LAN/WAN Port of the ETERNITY GENX.
- select the Ethernet Port as the Destination Port.
- configure the Syslog Server Address and the Server Port on which Syslog will listen for debug messages.

Configuring VMS Debug using Jeeves

- Log in as System Engineer.
- Under **Configuration**, click **VMS Configuration**.
- Click the **Debug**.



- Select the **Enable Debug** check box if you wish to view the VMS Debug
  - In **Syslog Server IP Address**, configure the IP Address of the Syslog Server.
  - In **Syslog Server Port**, configure the address of the Listening Port of the Syslog Server. Valid port range is: 1025 to 65535; 514. By default, the remote server port address is '514'.

If you disable the check box, the system will not send the VMS Debug to the Syslog Server.

228. *The SARVAM UCS supports Syslog Client, which enables the VMS to send debug messages in syslog format to the remote 'Syslog Server' on the IP network. You can view the system debug messages on the remote Syslog server or any other application which can capture the Syslog debug messages.*

The system supports following VMS Debug Levels. You may select the respective checkbox of the debug levels you wish to include in the VMS Debug.

- Select the **Call** check box to view the debug for all types of call flow.
- Select the **Mailbox** check box to view the debug for all operations related to Mailbox.
- Select the **SMTP** check box to view the debug related to the SMTP Client handled by VMS.
- Select the **Error** check box to view the debug for all types of errors.
- Select the **Configuration** check box to view the debug for configuration update.
- Select the **Status** check box to view the different status of all the VMS functions — total running calls, USB status, pending mail notification count, total messages, etc.

Protocol

- Select the **Basic** check box to view the debug for data received with its IE.
- Select the **Extended** check box to view the debug for data received in raw hex format.

Play

- Select the **Host Application** check box to view the debug related to play from VMS application side.
- Select the **Slave Application** check box to view the debug related to play from VMS Layer side.

Record

- Select the **Host Application** check box to view the debug related to the records from VMS application side.
- Select the **Slave Application** check box to view the debug related to the records from VMS Layer side.

Click **Submit**.

CHAPTER 12

SMS Server

This feature is not applicable if CDMA Mobile Card is installed in your system.

The SMS Server Application enables you to:

- Send/ receive SMS to/from individuals or groups using the Mobile Port of SARVAM UCS.
- Forward SMS received on Mobile Port as Emails to users through the Email Client.
- Forward Email of the users as SMS to the Mobile users through the Mobile Port.
- Configure Personal Directory via Email. For details, see"Configuring Personal Directory via Email" under "Abbreviated Dialing".

The SMS Server application works as an intermediary between the GSM Short Message Service and the SARVAM UCS. The Server supports multipart, 7 bit text messages as well as UNICODE messages.

The Server functions as an SMTP Client to send emails and as a POP3 Client to receive emails. SMS Server supports three types of Emails—Plain Text, HTML and MIME— from its mail clients.

To use this feature, you must purchase the SMS Server License. Refer to the topic, "License Management" to know more.

Make sure your Email Server uses SMTP and POP3 to send/receive emails.

If you have activated both, the SMS Gateway license as well as the SMS Server license, the SMS Gateway will be given priority. If you disable the SMS Gateway functionality, you need to restart the system to resume the SMS Server functionality.

To forward SMS as IM on SIP Extensions and IM from SIP Extensions as SMS, see "SMS over IP".

How it works

For this feature to work,

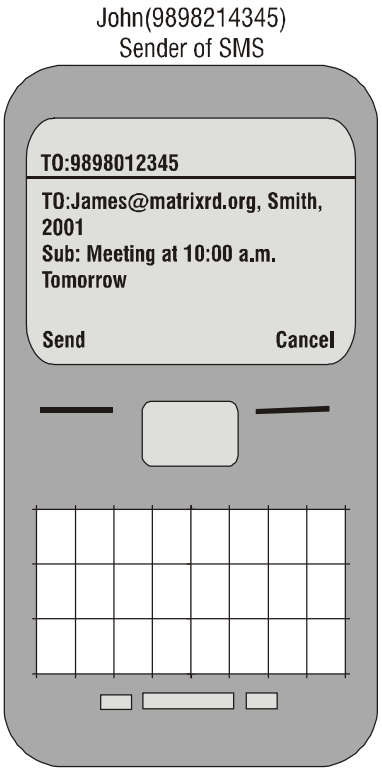
- you must have the SMS Server license. See "License Management".
- make sure your Email Server uses SMTP to send messages and POP3 to receive messages.
- you must configure the SMTP Client and POP3 Client parameters in SMS Server of SARVAM UCS. See "SMS Server - Mail Settings"
- the users must have valid Email ID's.
- you must configure the desired Groups and assign them to the extension users and/or Global/Personal Directory contacts. See "SMS/Email Group".
- configure the SMS parameters for the desired extension users and/or Global/Personal Directory contacts, that is, Mobile Number, Email ID and SMS/Email Group. See "Configuring SLT Extensions", "Configuring

DKP Extensions", "Configuring ISDN Terminals", "Configuring Matrix SPARSH VP330", "Configuring Matrix VARTA ADR100/AMP100 UC Clients", "Configuring Matrix SPARSH VP248","Configuring Matrix SPARSH VP310", "Configuring Matrix SPARSH VP510","Configuring Matrix SPARSH VP210" "Configuring Personal Directory using Jeeves" and "Configuring Global Directory using Jeeves" under "Abbreviated Dialing".

- you must define the Mobile port through which the messages are to be sent/received (Fixed/LCR). See "SMS Routing".
- configure the SMS parameters and the SMS Budget parameters (if required) on the respective Mobile ports, see "Configuring Mobile Trunks".
- you must configure the SMS Server parameters as well as the multipart SMS parameters for sending/ receiving SMS (if required). See "How to Configure".

Forwarding an incoming SMS as Email

- To forward an SMS to an Email ID, the SMS received on the Mobile Port must be in a specific format. Illustrated below is as example of the format.



Here,

- The sender of the SMS is John (9898214345).
- The sender must send the message to the SIM Number of the Mobile Port of SARVAM UCS, that is 9898012345.
- The body of the message must contain the following:
  - **To:** This is the destination where the Email is to be sent. In this case, James@matrixrd.org, Smith, 2001.
  - **Sub:** This is the message that will be displayed to the recipients. In this case, 'Meeting at 10.00 a.m. tomorrow'.
- When the SMS is received on Mobile Port (9898012345) in the above format, the system checks the senders number (9898214345 - John) in the Denied numbers list of the SMS Server.

If the SMS is received in multiple parts the SMS Server combines it into a single SMS.

- If a match is found in the Denied list, the SMS will be rejected. If no match is found, then it checks the destination where the SMS is to be forwarded as an email.

  The destination can be the recipients Extension number, Name/Group Name or Email ID. The **To** field in the message body specifies the destination. To send the same message to multiple destinations, enter the destinations separating them with a comma or semicolon.

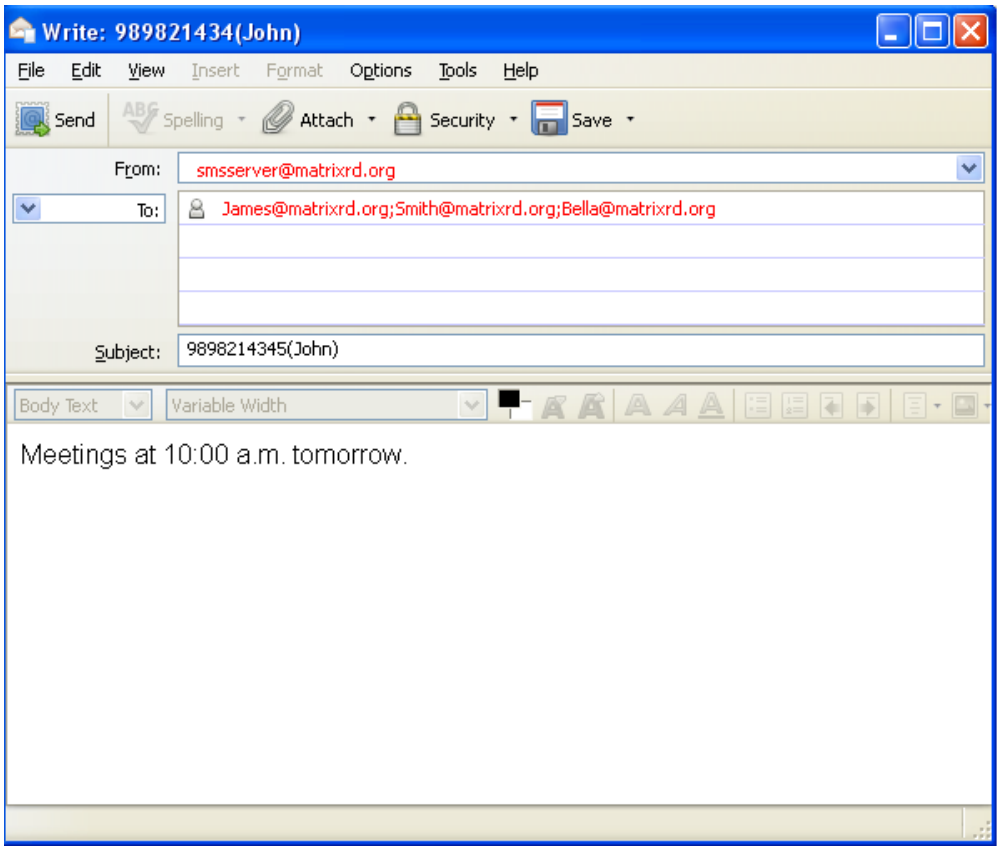
  When the destination is an Email ID, the system will forward the SMS to the recipients Email ID. In this case, James@matrixrd.org

  When the destination is a Name or Extension Number, the system will search for the Name/Number in its database. When a match is found, the message will be sent to their corresponding Email IDs. In this case the Email ID of Smith is Smith@matrixrd.org and the Email ID of the Extension user 2001 is Bella@matrixrd.org.

  If a match is not found for the Email ID, Name/Group Name or Extension Number, by default the system rejects the message. The system also provides you the option to send the message to a specific recipient (Send to Default recipient), if you do not want to reject the message.

- The SMS Server will convert this SMS to an Email. The email will be sent by the SMS Server to the Email Server and the Email Server will finally deliver it to the recipients.

- When the recipients download their emails, it will be as per the format displayed below.

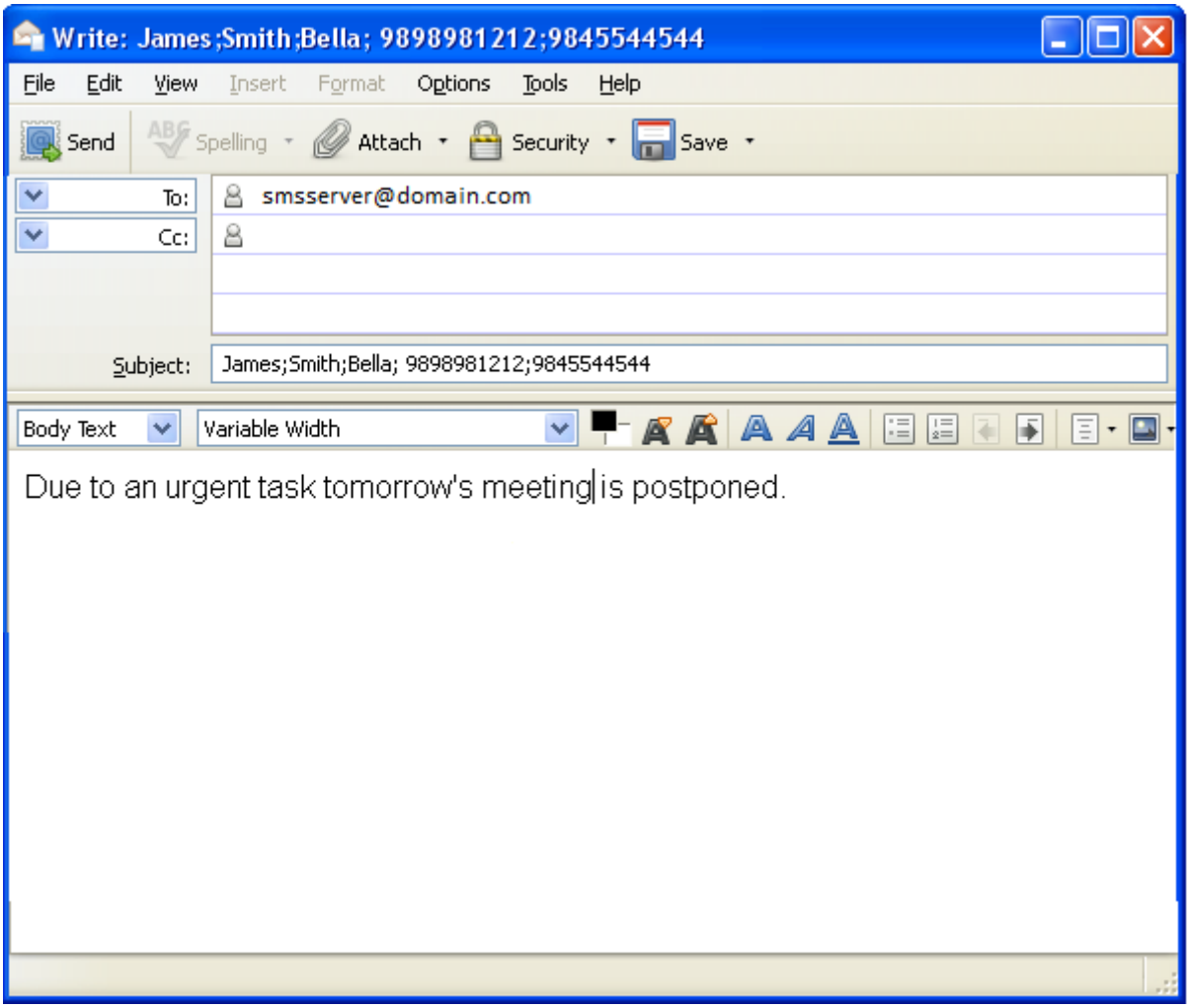


In this Email, the **From** field contains the Email address of the SMS Server. The **To** field contains the Email IDs of the recipients, the **Subject** contains the Name and/or Number of the sender and the **Body** of the message contains the message for the recipients.

The Name will be displayed only if it has been configured and it is found in the system database.

Forwarding an incoming Email as SMS

- To forward an Email as an SMS, the sender must send the Email to the SMS Server in a specific format. Illustrated below is as example of the format.



Here,

1. The sender of the Email is John, John@matrixrd.org.
2. The **To** field must be the Email ID of the SMS Server, smsserver@domain.com.
3. The **Subject** must contain the destination where the SMS to be delivered.

   The destination can be a Name/Numbers of the users to whom the SMS is to be sent. To send the SMS to multiple destinations, enter the destinations separating them with a comma or semicolon. Here the destination, that is the recipients are James, Smith, Bella, 9898981212, 9845544544.

4. The **body** must contain the message to be sent to the recipients, that is ‘Due to an urgent task tomorrow’s meeting is postponed.’

5. The senders Email ID or Name will be displayed to the recipients, if you have configured the parameter **Send Footer/Signature in SMS**.

*It is recommended that you configure the parameter **Send Footer/Signature in SMS**, so that the recipient knows the sender of the message. If this parameter is not configured, only the mobile port SIM number will be displayed to the recipient.*

- This mail will be sent to the Email Server and then the Email Server forwards it to the SMS Server.

- The SMS Server can receive emails from extensions users (system users) or from external users. You can allow or deny emails from users as per your requirement.

  If you want to receive Emails from extension users only, select the **Enable Email to SMS forwarding** check box.

  If you want to receive emails from extension users as well as external users, select the **Enable Email to SMS forwarding for External Users** check box.

- Then, the Server checks the Email ID of the sender in the Denied Email list.

- If a match is found in the Denied list, the Email will be rejected. If no match is found, then it checks the destination where the SMS is to be forwarded.

  The destination can be the recipients Number or Name. The **Subject** field of the message specifies the destination. To send the same SMS to multiple destinations, enter the destinations separated by comma or semicolon.

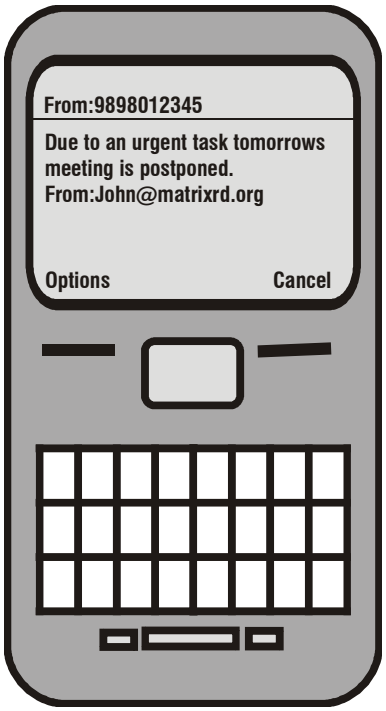
  If the destination is a Number, the server will check the number in the Denied list. If a match is found the email will be rejected. If a match is not found, the Server will send the SMS directly to the number using the Mobile Port of SARVAM UCS.

  If the destination is a Name, the system will search for the Name in its database. When a match is found, the SMS will be sent to the corresponding Number. In this case, the Numbers of James, Smith, Bella.

  The Server sends a return email as well as a delivery status report to the SA/Sender or to Both, informing that the SMS has been delivered/not delivered.

  If a match is not found for the Name, a reply mail is sent to the Sender, informing that the Name is not found.

The SMS Server converts the Email to a SMS. In this case the SMS delivered to each recipient, will appear as given below.



In the SMS sent to the recipients, the **From** field contains the SIM Number through which the SMS Server sent the SMS. The **Body** of the message contains the message for the recipients and the Footer/Signature that is, the Email ID/Name of the sender (if the Signature has been added).

*It is recommended that you configure the parameter **Send Footer/Signature in SMS**, so that the recipient knows the sender of the message. If this parameter is not configured, only the mobile port SIM number will be displayed to the recipient.*

Bulk SMS

The SMS Server supports Bulk SMS, that is a single message can be sent to multiple numbers. The message can be sent using the email client by specific users.

How it works

For this feature to work,

- make sure you have enabled Email to SMS Forwarding. See “Email to SMS Forwarding”.
- configure the Minimum time delay required between sending two consecutive SMS. See “SMS Configuration”.
- configure the Bulk SMS parameters and the Email IDs of the users allowed to send Bulk SMS. See “Bulk SMS”
- you must define the Mobile port through which the messages are to be sent (Fixed or LCR). See “SMS Routing”
- make sure your Email Server uses SMTP to send messages and POP3 to receive messages.
- you must configure the SMTP Client and POP3 Client parameters in the SMS Server of SARVAM UCS. See “SMS Server - Mail Settings”
- users must have valid Email IDs.

To send the Bulk SMS the email must be received in the following format:

- the email must be sent with the Subject “Bulk SMS”.
- the content received in the email will be considered as the SMS text.The text in the email must not exceed 160 characters.
- the mail must contain an attachment of the numbers only or numbers and names in csv format. The csv file can have a maximum of 1000 numbers. The csv file can have two columns Name and Number or have only one column with numbers. Make sure the columns do not have any header.

When the system receives the email from the user who has requested for Bulk SMS, the system will check this Email ID in the Allowed Email IDs List for sending Bulk SMS. If a match is found, the system will serve the request. If a match is not found the Bulk SMS request will be rejected.

*The system will check the **Allowed Email IDs for sending Bulk SMS** list, only if you have enabled the **Allowed Email IDs to send Bulk SMS** check box. If the **Allowed Email IDs for sending Bulk SMS option** is disabled, all the system users (extension users) will be able to send Bulk SMS requests.*

The system will serve only one Bulk SMS request at a time. If one Bulk SMS is in progress and the system receives another request, the system will reject it. Bulk SMS supports a single SMS of 160 characters only. It does not support multi-part SMS.

While the Bulk SMS request is in progress, the system also provides you the option to stop the process at any point of time if required. The sender must contact the System Administrator to do so.

After the Bulk SMS request has been served, the system updates the csv file and adds another column, Status. For each entry the Status of the SMS is updated. The Status column may contain any one of the following:

- Sent - When any SMS is sent successfully by the system i.e OK is received from GSM engine.
- Failed - When GSM engine gives any error response
- Invalid - For any entry where data is found other than number in the .csv file
- Denied - When SMS number is found it the Denied list
- Limit Exceed - When Daily or Monthly limit is exceeded for all the Mobile ports used for sending the SMS for any entry.
- No Port Available - When all Mobile Ports are configured as "Not allowed to send SMS" which are to be used for sending the SMS for any entry.

This report is then emailed to the user who had requested for Bulk SMS. The Subject of the report email is the same as received for Bulk SMS request.

In certain cases the system may not be able to serve the Bulk SMS request. In such cases the system sends a reply email with the Error message to the user who requested for Bulk SMS. The possible Error conditions and messages are mentioned below:

Condition	Error Message
If Mail Subject is written as “bulk SMS” (case insensitive) but No file is attached in Mail	No attachment found to process bulk SMS. Please attach valid “.csv” file for Bulk SMS.
If multiple attached file is received	Only one attachment is allowed. Attached file format should be ".csv". Attachment file size should be less than 1 MB.
If multiple attached file is received and from that multiple attached file one file format is .csv	Only one attachment is allowed. Attached file format should be ".csv". Attachment file size should be less than 1 MB.

If attached file format is found to be other than .csv	No attachment found to process bulk SMS. Please attach valid “.csv” file for Bulk SMS.
If attached file is .csv but file size is more than we support. (i.e 2MB)	There is No such Error cause for this condition. In this case, the incoming email will get ignored.
If attached file format is found to be .csv but in subject it is not received as "Bulk SMS" (string required for this Bulk SMS feature) and not applicable for Master also.	Attachment is not allowed for normal Mail to SMS.
If attached file format is found to be .csv, subject is matched for Bulk SMS but feature is disable	Bulk SMS feature is disabled
If attached file format is found to be .csv, subject is matched for Bulk SMS but user is not allowed to send Bulk SMS (user means from which email is received)	User not allowed to send Bulk SMS
If attached file format is found to be .csv, subject is matched for Bulk SMS, user is allowed Bulk SMS feature (user means from which email is received) but Text data contains more than 160 characters (i.e consider that text data is such that sms is to be send in multi-part)	More than 160 character text is not supported for Bulk SMS
If attached file format is found to be .csv, subject is matched for Bulk SMS, user is allowed Bulk SMS feature (user means from which email is received), Text data is also for single part sms but attached file data is not as required.	Invalid ".csv" file.It should contain data only in first 2 columns (i.e.Name, Phone Number or Phone Number, Name) without any header.
If already one .csv file for Bulk sms is in process and other attached valid file is received.	Already one file is in process. Please send the file again after x minute(s) approximately. Here x is the time duration for which current csv file will be processed.
If Bulk SMS file is processed, send report	Please find the attached report of Bulk SMS file "xxx.csv" received on "Date-Time" . where xxx.csv should be replaced with filename Date-Time should be replaced with the date and time when email was received by system.

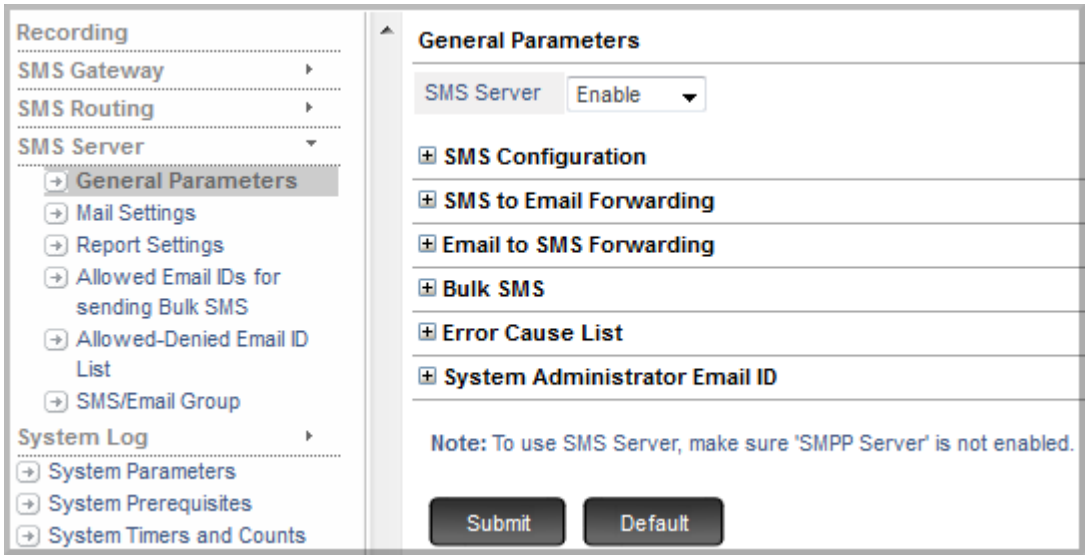
How to Configure

For the SMS Server to function, you need to configure the following parameters,

General Parameters

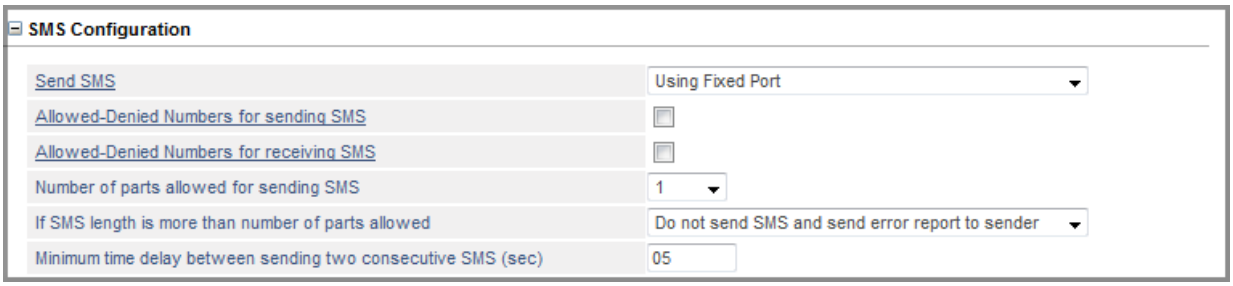
- Log in as System Engineer.
- Under **Configuration**, click **SMS Server**.
- Click **General Parameters**. The SMS Server-General Parameters page opens.

- By default, the **SMS Server** is disabled. To use the SMS Server feature, select **Enable**.



SMS Configuration

Click **SMS Configuration** to expand.



Configure the SMS related parameters:

- You can **Send SMS** through certain fixed Mobile Ports or through different Mobile Ports according to specific numbers and time.

  To send messages through fixed Mobile Ports, select **Using Fixed Mobile Port** and configure the **SMS Routing-Fixed Port** table. For detailed information, see “Fixed Port Routing (SMS Server)”.

  To send messages to specific numbers through certain preferred Mobile Port/s during a defined time interval, select **Based on specific Time-Number** and configure the **SMS Routing-LCR** table. For detailed information, see “Least Cost Routing”.

  Default: Using Fixed Mobile Port.

- Configure the **Allowed-Denied Numbers for sending SMS** list, if you want to allow or restrict sending of messages on specific numbers. To do this,
  - Select the **Allowed-Denied Numbers for sending SMS** check box.
  - Click on **Allowed-Denied Numbers for sending SMS** and the **Allowed-Denied Numbers for sending SMS** table opens. You can configure upto 250 numbers.

- In the Allowed column, enter the numbers on which messages can be sent and in the Denied column, enter the numbers on which messages cannot be sent.

- You can also configure this list by clicking the **Allowed-Denied Numbers for sending SMS** link under **SMS Routing**.

  Default: Disabled

- Configure the **Allowed-Denied Numbers for receiving SMS** list, if you want to allow or restrict receiving messages from specific numbers. To do this,
  - Select the **Allowed-Denied Numbers for receiving SMS** check box.
  - Click on **Allowed-Denied Numbers for receiving SMS** and the **Allowed-Denied Numbers for receiving SMS** table opens. You can configure upto 250 numbers.
  - In the Allowed column, enter the numbers from which messages can be received and in the Denied column, enter the numbers from which messages cannot be received.
  - You can also configure this list by clicking the **Allowed-Denied Numbers for receiving SMS** link under **SMS Routing**.

  Default: Disabled

If you want to allow multipart messages configure the following parameters:

- Select the **Number of parts allowed for sending SMS** from the list. Default:1

- If the **SMS length is more than number of parts allowed**, you can select either **Ignore remaining part of the SMS** or **Do not send SMS and send error report to sender**. Default: Do not send SMS and send error report to sender.

- If you select **Ignore remaining part of the SMS**, you can select the **Send error report to sender when SMS length is more than allowed parts** check box, if you want to send an error report to the sender.

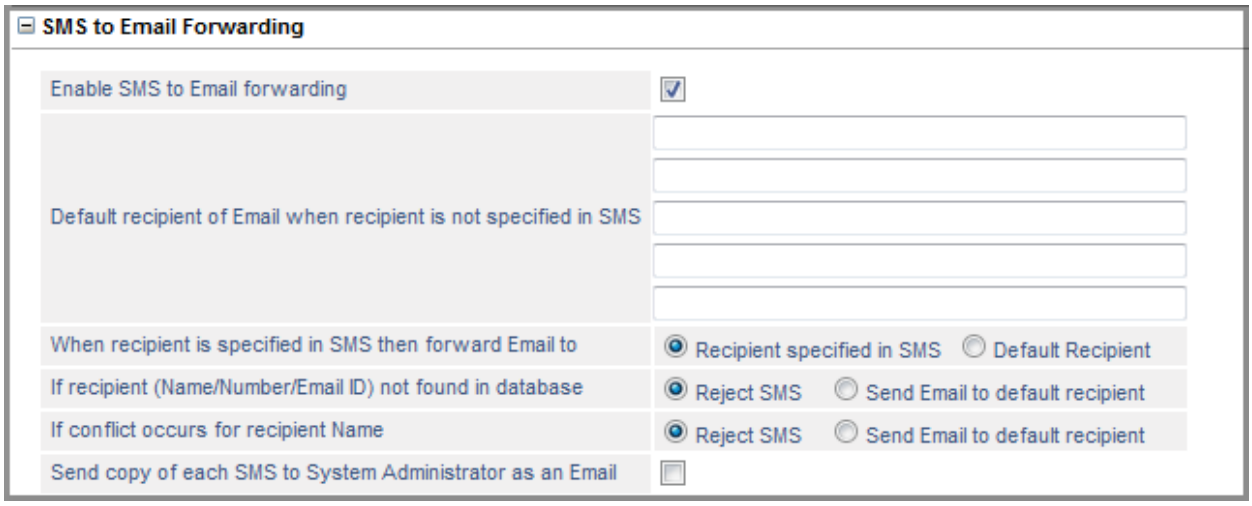
- If you enable **Send error report to sender when SMS length is more than allowed parts**, enter the message you want to send to the sender in the email in **Reply error report as an Email to sender containing text**. Default text: Some texts of message are ignored as length of mail is more than allowed characters.

- Configure the **Minimum time delay between sending two consecutive SMS (sec)** as supported by the network. Default: 05.

Click **Submit** to save the changes you made.

SMS to Email Forwarding

Click **SMS to Email Forwarding** to expand.



Configure the following parameters:

- By default the **Enable SMS to Email forwarding** is selected (enabled). If you do not want the SMS Server to forward the SMS as Emails to the users, clear the check box.
- Configure the **Default recipient of Email when recipient is not specified in SMS**. When an SMS is received without any recipient, the system delivers the SMS as an Email to the default recipient/s configured here. You can configure upto 5 Email IDs. Default: Blank.
- Select the desired option for, **When recipient is specified in SMS then forward Email to**. You can select **Recipient specified in SMS** or **Default Recipient**. Default: Recipient specified in SMS.

  If you want all incoming SMS to be delivered as Email to a specific recipient only, select **Default Recipient**.

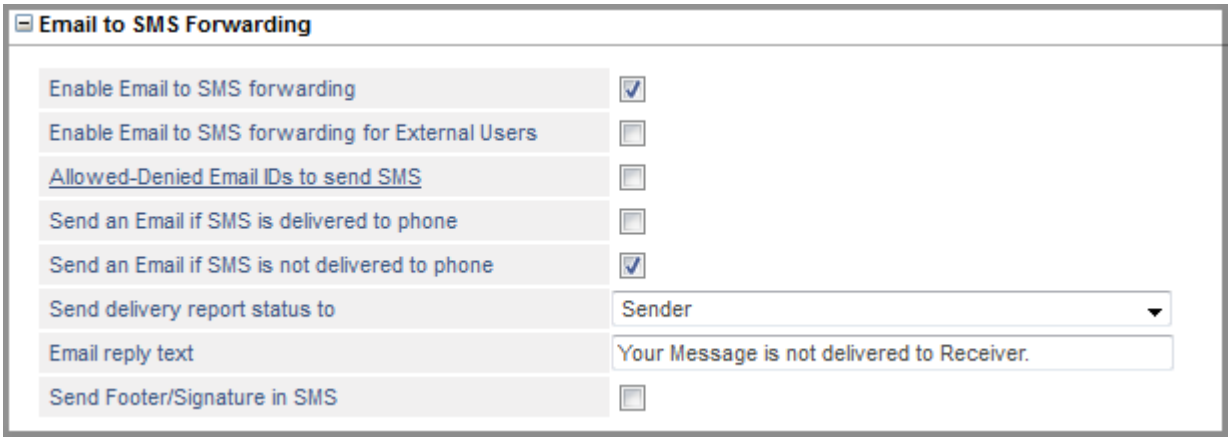
  If you want the incoming SMS to be delivered as Email to the recipients specified in the SMS, select **Recipient specified in SMS.**
- Select the desired option for, **If recipient (Name/Number/Email ID) not found in database**. You can select **Reject SMS** or **Send Email to default recipient.** Default: Reject SMS.
- Select the desired option for, **If conflict occurs for recipient Name**. You can select **Reject SMS** or **Send Email to default recipient**. Default: Reject SMS.
- Enable the **Send copy of each SMS to System Administrator as an Email** check box, if you want to send a copy of SMS as an Email to the System Administrator. Default: Disabled.

  If you enable this check box, you must configure atleast one Email address of the System Administrator under “System Administrator Email ID”.

Click **Submit** to save the changes you made.

Email to SMS Forwarding

Click **Email to SMS Forwarding** to expand.



Configure the following parameters:

- By default the **Enable Email to SMS forwarding** check box is selected (enabled). Emails received from extension users only will be forwarded as SMS. If you do not want the SMS Server to forward Emails as SMS to the users, clear the check box.

- By default the **Enable Email to SMS forwarding for External Users** check box is clear (disabled). Select this check box, if you want the SMS Server to receive Emails from extension users as well as external users and then forward them as SMS.

- Configure the **Allowed-Denied Email IDs to send SMS** list, if you want to allow or restrict certain Email IDs to send SMS. To do this,

  - Select the **Allowed-Denied Email IDs to send SMS** check box.
  - Click on **Allowed-Denied Email IDs to send SMS** and the **Allowed-Denied Email ID list** table opens. You can configure upto 500 Email IDs.
  - Select the desired option **Allow all except programmed in Denied List** or **Deny all except programmed in Allowed List**.
  - If you select **Allow all except programmed in Denied List,** in **Denied Email ID** column, enter the Email IDs from which SMS cannot be sent.
  - If you select **Deny all except programmed in Allowed List,** in **Allowed Email ID** column, enter the Email IDs from which SMS can be sent.
  - You can also configure this list by clicking the **Allowed-Denied Email ID List** link under **SMS Server**.

  Default: Disabled.

- Select the **Send an Email if SMS is delivered to phone** check box, if you want a confirmation email to be sent when the delivery report is received by the Server from the network. Default: Disabled.

  - If you have enabled **Send an Email if SMS is delivered to phone**, in **Send delivery report status to**, select the recipient to whom the delivery status report must be sent. You can select **Sender**, **System Administrator** or **Both**. Default: Sender.

  - If you select **System Administrator**, the email will be sent to the email IDs configured in “System Administrator Email ID”.

- In **Email Reply Text**, enter the message you want to send in the Email. The message can a maximum of 100 characters. Default text: Your Message is delivered to Receiver.

- Select the **Send an Email if SMS is not delivered to phone** check box, if you want a confirmation email to be sent when the delivery report is not received by the Server from the network. Default: Disabled.

  - If you have enabled **Send an Email if SMS is not delivered to phone**, in **Send delivery report status to**, select the recipient to whom the delivery status report must be sent. You can select **Sender**, **System Administrator** or **Both**. Default: Sender.

  - If you select **System Administrator**, the email will be sent to the email IDs configured in "System Administrator Email ID".

  - In **Email Reply Text**, enter the message you want to send in the Email. The message can a maximum of 100 characters. Default text: Your Message is not delivered to Receiver.

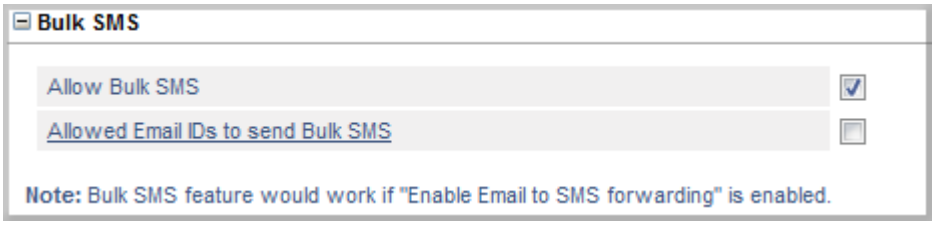
- Enable the **Send Footer/Signature in SMS** check box and select the desired option to be sent:
  - Specific Text: Enter the text/message to be sent to the recipient as signature in the SMS.
  - Send Email ID of Sender: Enter the Email ID of the Sender to be sent to the recipient as signature in the SMS.
  - Send Name of Sender: Enter the Name of the Sender to be sent to the recipient as signature in the SMS.

It is recommended that you configure the parameter Send Footer/Signature in SMS, so that the recipient knows the sender of the message. If this parameter is not configured, only the mobile port SIM number will be displayed to the recipient.

Click **Submit** to save the changes.

Bulk SMS

Click **Bulk SMS** to expand.



Configure the following parameters:

- By default **Bulk SMS** is disabled. If you want the SMS Server to send Bulk SMS, select the check box.

- Select the **Allowed Email IDs to send Bulk SMS** check box and configure the **Allowed Email IDs to send Bulk SMS** list. Default: Disabled. Only these users will be able to send Bulk SMS. To configure the Email IDs,
  - Click on **Allowed Email IDs to send Bulk SMS** and the **Allowed Email IDs for sending Bulk SMS** table opens. You can configure upto 64 Email IDs.
  - Enter the **Email IDs** of the users who can send Bulk SMS.
  - You can also configure this list by clicking the **Allowed Email IDs for sending Bulk SMS** link under **SMS Server**.

- Click **Submit** to save the entries.

Error Cause List

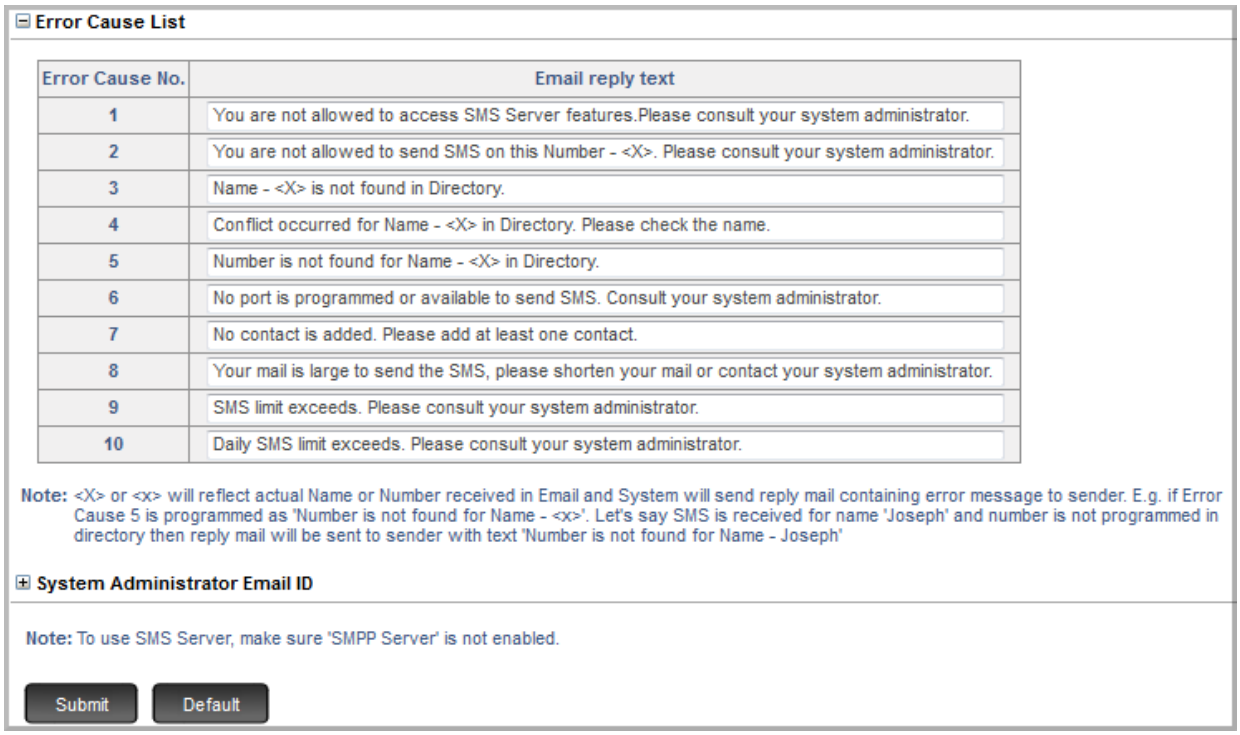
There are different activities/events/error conditions handled by the SMS Server. The Server sends an SMS/Email as reply for each of these to the sender.

Refer to the table below, to known the possible Condition/Activity/Event:

Error Cause	Condition/ Activity/Event	Email reply text
Error Cause 1	When Email user sends Email to SMS Server for sending SMS and Email ID is programmed in Denied List, then reject Email.	You are not allowed to access SMS Server features. Please consult your administrator.
Error Cause 2	When Number is programmed in Denied List for Sending SMS/Receiving SMS, then reject Email (for email to SMS query) or Reject SMS (for SMS to email query).	You are not allowed to send SMS on this Number <X>. Please consult your administrator. Where <X> is the number.
Error Cause 3	When Email user sends Email to SMS Server for sending SMS and Name is written in mail but name is not found in any Directory; then reject Email.	"Name <X>" is not found in Directory Where <X> is name written as recipient in subject line.
Error Cause 4	If conflict occurs for the Recipient Name in Directory (SMS to Email or Email to SMS), then reject it.	Conflict occurred for "Name <X>" in Directory. Please check the Name. Where <X> is the Name written as recipient by sender.
Error Cause 5	If Number is not programmed for the Recipient Name in Directory request (SMS to Email or Email to SMS), then reject it.	Number is not found for Name <X> in the Directory.
Error Cause 6	When there is an Email for SMS forwarding but there is no port is programmed for sending SMS, request for sending SMS is rejected.	No port is programmed to send SMS. Consult your System Administrator.
Error Cause 7	When Email User sends Mail to SMS Server having subject line blank, reject this mail.	No contact is added. Please add at least one contact.
Error Cause 8	When Email user sends Email to SMS Server for sending SMS and length of message body is more than allowed characters. The email will be rejected or the remaining part will be ignored, depending on the configuration.	Your mail is large to send the SMS, please shorten your mail or contact your administrator.
Error Cause 9	When SMS credit exceeds, send email to sender informing about the status.	SMS Limit Exceeds. Please consult your System Administrator.
Error Cause 10	When Daily SMS credit exceeds, send email to sender informing about the status.	Daily SMS Limit Exceeds. Please consult your System Administrator.

If required you can modify the Email reply text as per your requirement. To do this,

- Click the **Error Cause List** link to expand.

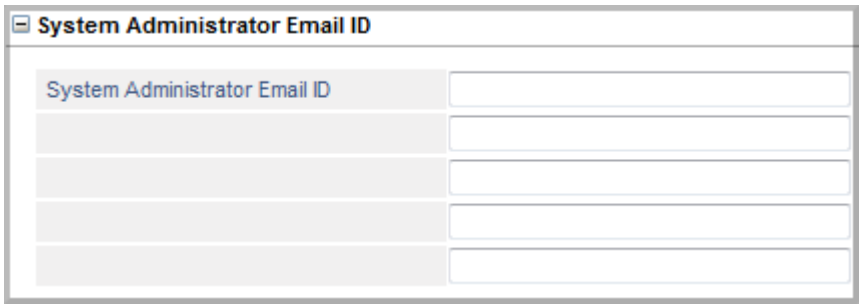


- The Error Cause table displays the Error Cause number with the corresponding Email reply text.
- Select the Error Cause Number for which you need to edit/change the reply text.
- In **Email reply text**, enter the desired text.
- Click **Submit**.

You can generate a report of all the errors, see “SMS Server Reports” for more information. These Error Causes are also logged into the Fault Log. See “System Fault Log” for more information.

System Administrator Email ID

Click the **System Administrator Email ID** link to expand.



- You must Configure the **System Administrator Email ID** to which a copy of SMS must be sent by the SMS Server, if you have enabled **Send copy of each SMS to System Administrator as an Email** check box under “SMS to Email Forwarding”.

  You can configure upto 5 Email ID’s. The Email IDs can be of maximum 64 characters.

- Click **Submit** to save changes.

SMS Routing

The SMS Server can send SMS using any of the following methods:

- Fixed Port Routing - through a single/fixed group of Mobile Ports.
- Least Cost Routing - through selective preferred Mobile Ports grouped together in order to utilise the benefits offered by the service providers, such as 1000 free SMS in a month, reduced rates to send messages etc.

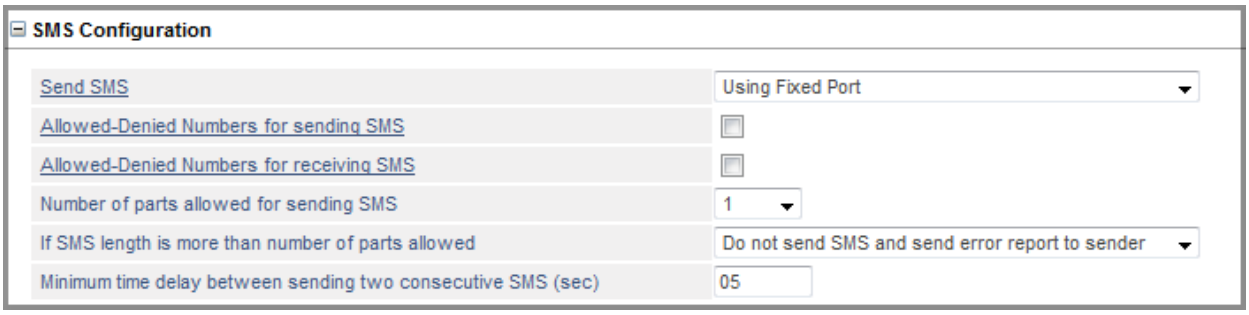
The system allows you to configure different Fixed Port Routing Tables for the SMS Server application and SMS over IP application. But, the Least Cost Routing table is common for both these applications.

Fixed Port Routing (SMS Server)

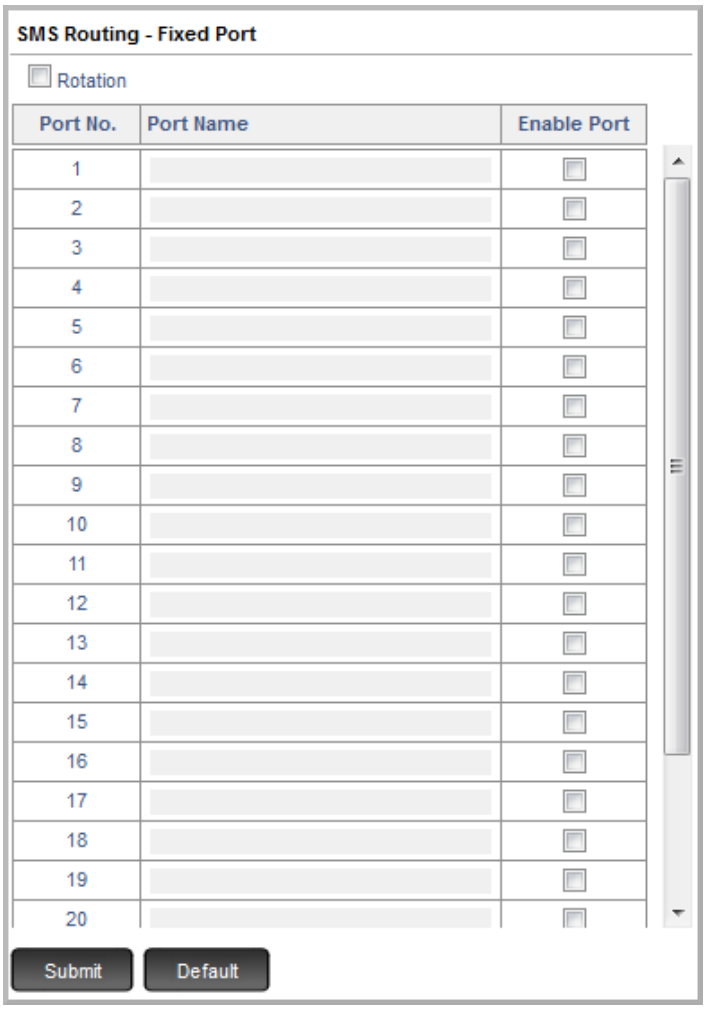
You can route the SMS through a single or fixed group of ports. To do this, you must determine the ports through which you want to send the SMS and then configure the SMS Routing-Fixed Port table.

Configuring Fixed Port Routing

- Log in as System Engineer.
- Under **Configuration**, click **SMS Server.**
- Click **General Parameters**.
- Click **SMS Configuration** to expand.
- As the **Send SMS** option select **Using Fixed Port**.



- Click on the **Send SMS** link, the **SMS Routing-Fixed Port** table opens.



The number of Mobile Ports displayed, depend on the variant of SARVAM UCS.

- Configure the following parameters:
  - **Rotation:** By default, all the messages will be sent using the first Mobile Port enabled by you.

    Select the **Rotation** check box, when you want the first SMS to be sent through the first Mobile Port, the subsequent SMS through the next Mobile Port and so on. For example, if three Ports MOB Port 1, MOB Port 2, MOB Port 3 are selected and there is a request to send 5 SMS. Then, the first SMS will be sent through MOB Port 1, second SMS through MOB Port 2, third SMS through MOB Port 3, fourth SMS through MOB Port 1 and so on.

    If the Rotation check box is cleared, all the SMS will be sent using the first Mobile Port enabled by you. For example, if three Ports MOB Port 1, MOB Port 2, MOB Port 3 are selected and there is a request to send 5 SMS. Then, each SMS will be sent through MOB Port 1 only.

  - **Port No.:** This is the Mobile Port number.

  - **Name:** This is the name assigned to the Mobile Port. This will be displayed only if you have assigned a name to the Mobile Port on the Mobile Port Parameters page. See “Configuring Mobile Trunks”.

- **Enable:** Select the Enable check box corresponding to the Mobile Port you want to use for sending the SMS.
- Click **Submit**.

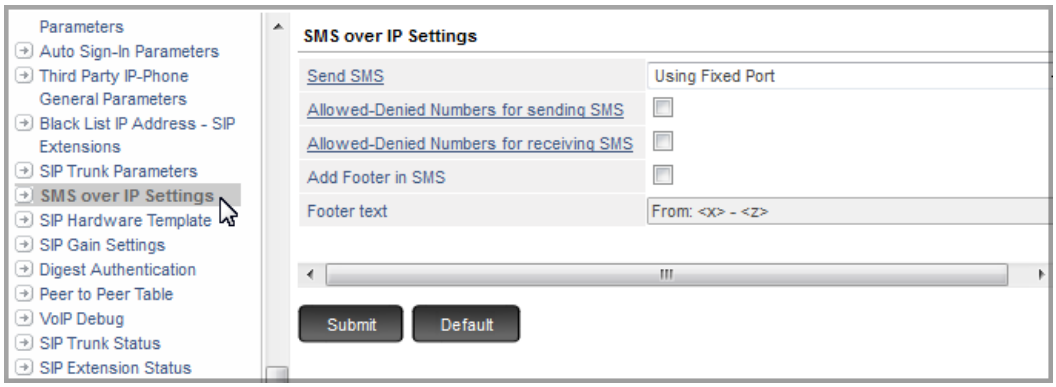
*Make sure that you have selected the **Send SMS** check box for this Mobile Port. See "Configuring Mobile Trunks".*

Fixed Port Routing (SMS over IP)

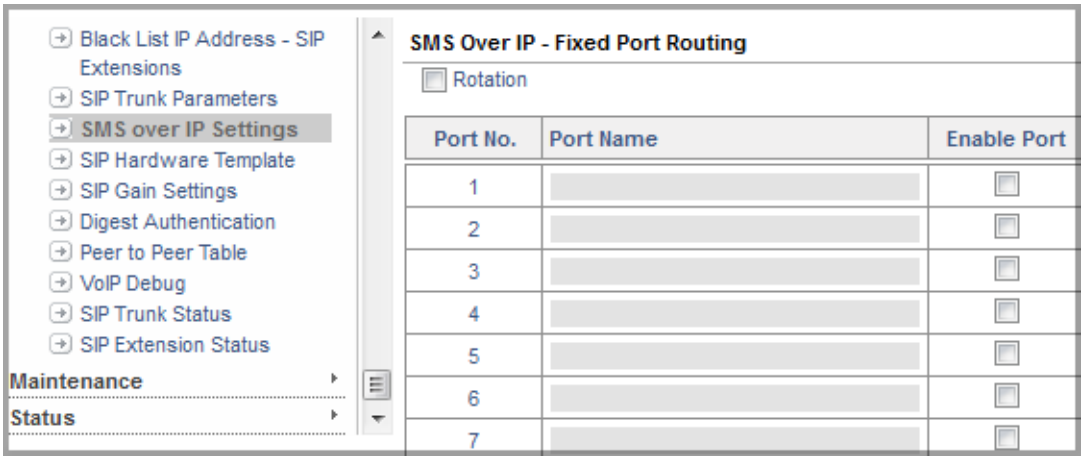
You can route the SMS through a single or fixed group of ports. To do this, you must determine the ports through which you want to send the SMS and then configure the SMS over IP-Fixed Port table.

Configuring Fixed Port Routing

- Log in as System Engineer.
- Under **Configuration**, click **VoIP Configuration.**
- Click **SMS over IP Settings**.
- As the **Send SMS** option select **Using Fixed Mobile Port**.



- Click on the **Send SMS** link.
- The **SMS over IP-Fixed Port Routing** table opens



The number of Mobile Ports displayed, depend on the variant of SARVAM UCS.

- Configure the following parameters:
    - **Rotation:** By default, all the messages will be sent using the first Mobile Port enabled by you.

      Select the **Rotation** check box, when you want the first SMS to be sent through the first Mobile Port, the subsequent SMS through the next Mobile Port and so on. For example, if three Ports MOB Port 1, MOB Port 2, MOB Port 3 are selected and there is a request to send 5 SMS. Then, the first SMS will be sent through MOB Port 1, second SMS through MOB Port 2, third SMS through MOB Port 3, fourth SMS through MOB Port 1 and so on.

      If the Rotation check box is cleared, all the SMS will be sent using the first Mobile Port enabled by you. For example, if three Ports MOB Port 1, MOB Port 2, MOB Port 3 are selected and there is a request to send 5 SMS. Then, each SMS will be sent through MOB Port 1 only.
    - **Port No.:** This is the Mobile Port number.
    - **Name:** This is the name assigned to the Mobile Port. This will be displayed only if you have assigned a name to the Mobile Port on the Mobile Port Parameters page. See “Configuring Mobile Trunks”.
    - **Enable:** Select the Enable check box corresponding to the Mobile Port you want to use for sending the SMS.
    - Click **Submit**.

*Make sure that you have selected the **Send SMS** check box for this Mobile Port. See “Configuring Mobile Trunks”.*

Least Cost Routing

Least Cost Routing (also referred to as Automatic Route Selection) is an expense control feature of SARVAM UCS.

Least Cost Routing (LCR) is useful when there are different Mobile Ports for sending messages, and the service providers offer different schemes for SMS. These schemes may be for certain numbers or during a particular time of the day.

When an SMS is sent from the system, LCR recognizes where the SMS is going to be delivered. It selects the lowest cost port from among all the ports allotted for sending SMS, depending upon how the LCR is configured.

The system can be configured to select the most cost effective trunk for the time of the day when the SMS is sent, or to select the most cost effective trunk for the destination number to which the SMS is sent, or to select the most cost effective trunk considering both time of the day and destination number.

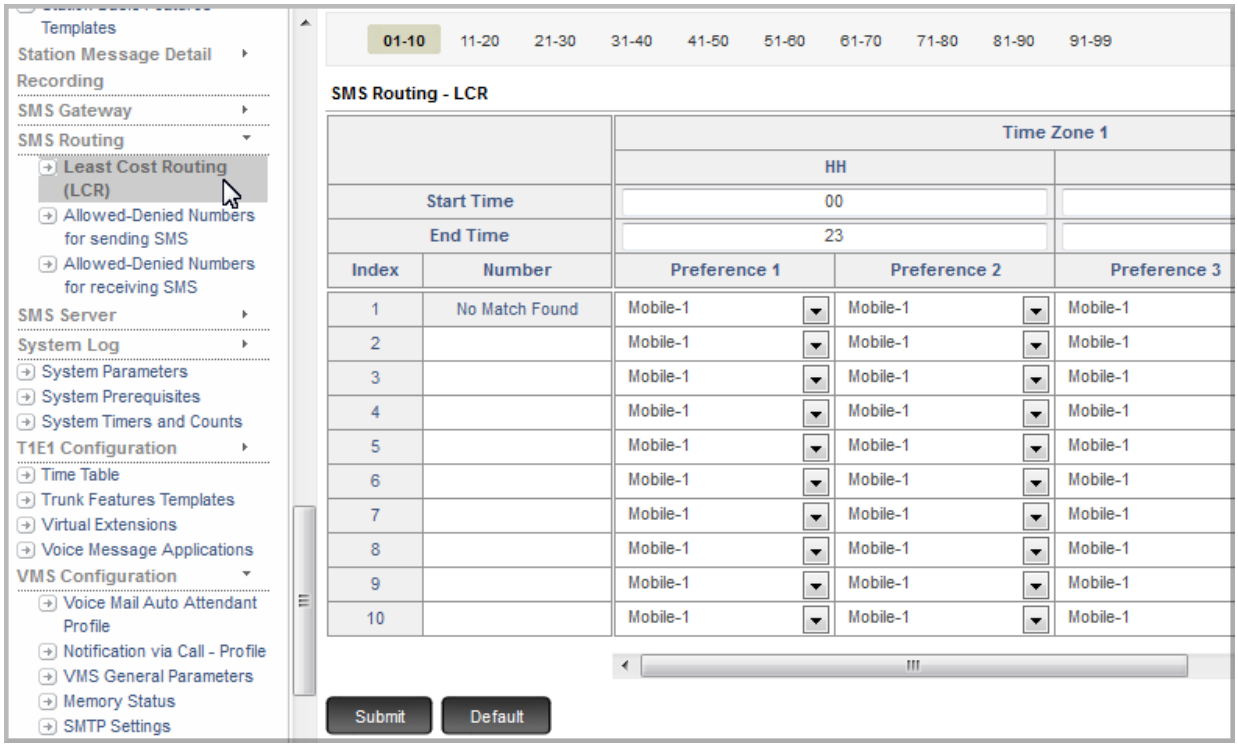
You can configure Time or Number or both as per your requirement.

- If you have configured only the Time, the system will check the time while sending the SMS and then route it according to the selected preference. For example you want to send SMS from one group of trunks during 9:00 a.m. to 2:00 p.m. and from 3:00 p.m. to 8:00 a.m. through another group.
- If you have configured only the Numbers, the system will check the destination number while sending the SMS and then route it according to the selected preference. For example SMS to numbers that begin with 99 and 97 can be routed through different trunk groups.

- If you have configured both, the time and number, the system will check the time as well as the destination number while sending the SMS and then route it according to the selected preference.

Configuring LCR

- Log in as System Engineer.
- Under **Configuration**, click **SMS Routing**.
- Click **Least Cost Routing**. The SMS Routing - LCR table opens.



The number of Mobile Ports displayed, depend on the variant of SARVAM UCS.

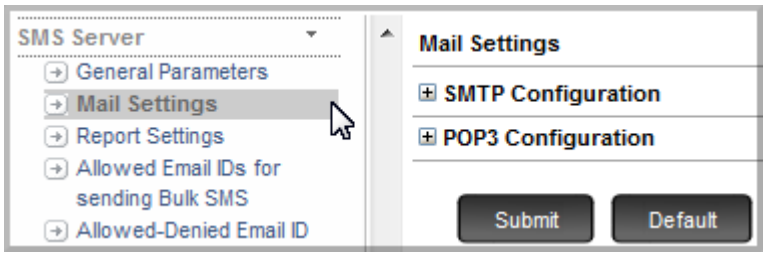
- Configure the following parameters:
- **Time Zone 1** to **4:** Configure the **Start Time** and **End Time** for each time zone. You can configure four different time zones—Time Zone1, 2, 3 and 4 as per your requirement.
- **Number:** Configure the destination numbers to which the messages are to be sent.
- **Preference1** to **4**: Select the trunks in the order of preference through which you want to send the SMS. You can select upto 4 preferences.
- Click **Submit** to save your settings.

SMS Server - Mail Settings

The Server functions as an SMTP Client to send emails and as a POP3 Client to receive emails. The SMS Server will be able to send/receive emails only after you have configured the relevant SMTP and POP3 parameters in the SMS Server.

How to configure

- Log in as System Engineer.
- Under **Configuration**, click **SMS Server**.
- Click **Mail Settings**.



SMTP Configuration

Contact your Network Administrator for the following information and configure the parameters as per the configurations done in the Email Server to register the SMS Server as an SMTP Client.

Click the **SMTP Configuration** link to expand.

- In **Use SMTP Account** select the account you want the SMS Server application to use.

  You may add a new SMTP Account. To do so,

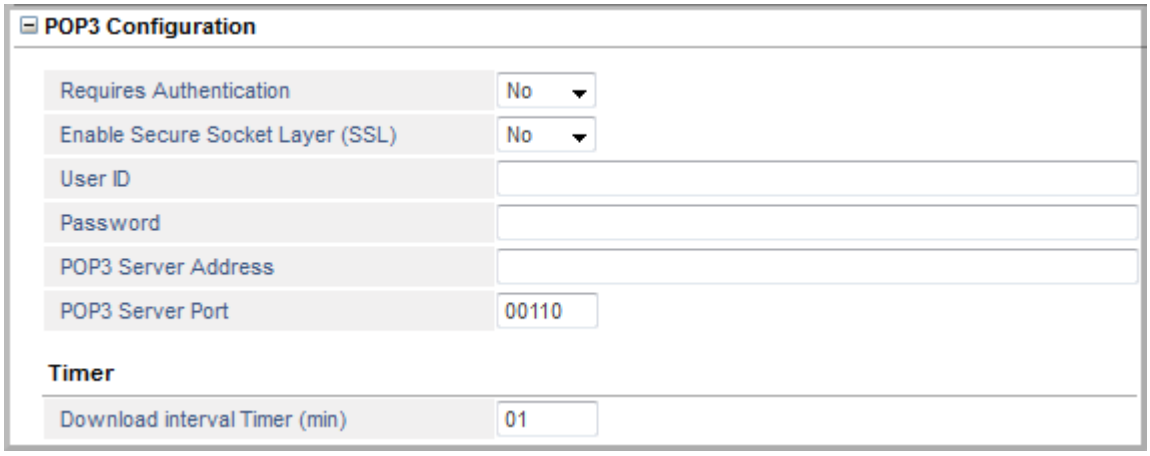
  - Select *Add New* option for Use SMTP Account.
  - Click **Settings** to configure the parameters of the New SMTP Account you created. For more information, see “SMTP Settings”.

Click **Submit** to save settings.

POP3 Configuration

Contact you Network Administrator for the following information and configure the parameters as per the configurations done in the Email Server to register the SMS Server as a POP3 Client.

Click the **POP3 Configuration** link to expand.



- If the Email Server uses authentication, select **Requires Authentication** as **Yes**. Default: No. If your Email Server uses authentication, you must also configure the *User ID* and the *Password*.

- To transport all data in a secure manner, select **Enable Secure Socket Layer (SSL)** as **Yes**. All the data to the Email Server will be transported over secure layer. Default: No.

- If you have enabled authentication, configure the **User ID** and the Authentication **Password** as provided to you by your network administrator. The User ID may consist of a maximum of 40 characters and the Password can be a maximum of 24 characters. Default: Blank.

- Configure the **POP3 Server Address** and **POP3 Server Port**. This is the Server's IP Address and Port number that is used to download incoming mails. Both IPv4 and IPv6 addresses are supported. For example, Email Server address is 192.168.1.1 and port is 1400, then configure the Server Address as 192.168.1.1 and port as 1400. If port is not programmed, use the default port value equal to 110. Valid Port range: 110;995;1025 to 65535. The Server Address can be a maximum of 46 characters.

Click **Submit** to save settings.

Timer

- **Download interval Timer (min.)** is the time interval after which the POP3 Client of the SMS Server retries to fetch new mail from the Email Server. Valid Range is 01 to 99 minutes. Default: 01 minute.

Test POP3 Settings

- After you have configured the POP3 parameters and submitted them, the **Click to Test POP3 Settings** button appears.
- Click the **'Click to Test POP3 Settings'** button to check if the POP3 parameters have been configured correctly.

  When you click this button, the alert message appears: *"Testing POP3 can take up to 99 seconds. Would you like to continue?"* Click the **OK** button.
  The message *"Please refresh the web browser after few seconds to check the test mail status"* appears. Click **OK** button.

  Refresh the web browser after a few seconds. The Test Result will be displayed in the 'Test Status' field.

- **Test Status:** Any one of the results listed below may appear in this field:

Test Status Message	Description
"Login to POP3 Mail Server is Successful"	When connection to POP3 server is established successfully.
"Login to POP3 Mail Server is Failed"	When connection to POP3 server is not established successfully

SMS Server Reports

The SARVAM UCS maintains a database with details of all the Email and SMS transactions as well as the Errors that occurred during these transactions.

The system has a buffer for 999 successful SMS/Email transactions and 100 Error transactions. When the buffer is full, the system overwrites these transactions on First In First Out (FIFO) basis. The buffer can be cleared at any time from the System Administrator mode.

You can generate different type of reports by setting the desired filters. SARVAM UCS can generate reports whenever you want or you can obtain it Online, immediately after the Email or SMS has been sent/received.

You can generate Reports, either

- Manually: The report is generated whenever you want.
  Or
- as per Schedule: The report is generated on a preset Day, Date and Time.

SARVAM UCS allows you to set a variety of filters for printing the SMS Server Reports. SARVAM UCS supports Syslog Client for SMS Server Reports. The Syslog Client enables the system to send records in syslog format to the remote ‘Syslog Server’. You can view the records on the remote server and print.

The Report contains the following information for each transaction:

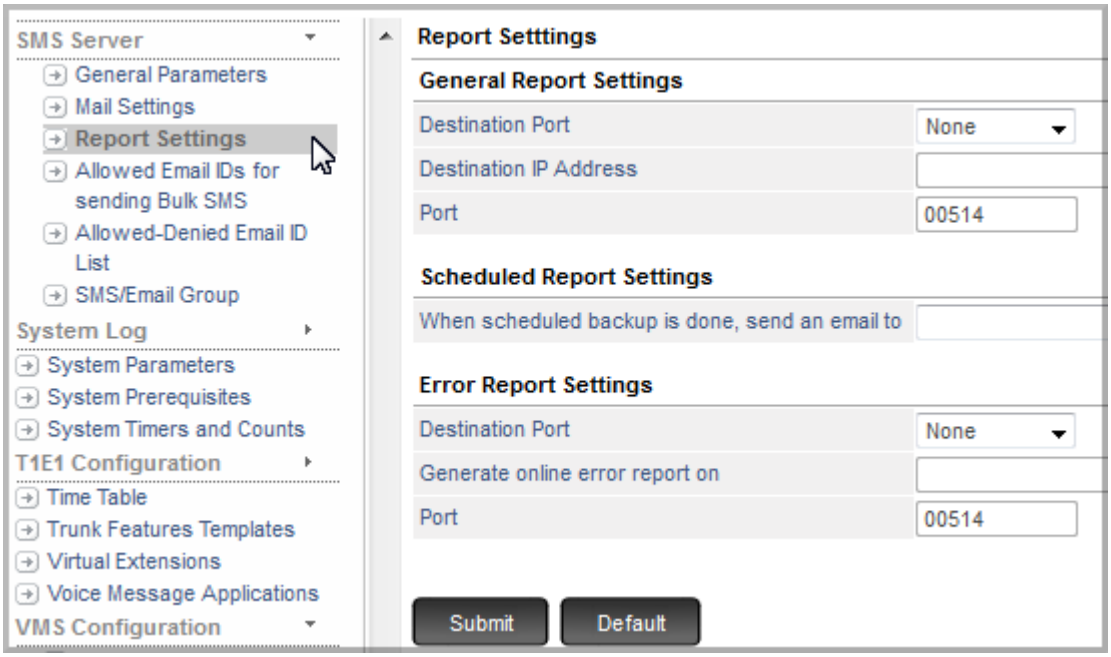
- Index
- Email
  - Direction
  - Email Address
  - Status
  - Date
  - Time
- SMS
  - Direction
  - Number
  - Status
  - Date
  - Time
  - Part of SMS
- Mobile Port
- Text

To generate the reports you must configure the following parameters:

- General Report Settings
- Scheduled Report Settings
- Error Report Settings
- Report Backup Parameters
- Report Filters

Configuring Report Parameters

- Log in as System Engineer.
- Under **Configuration**, click **SMS Server**.
- Click **Report Settings**.



- Under **General Report Settings**,
  - Select **Ethernet** as the **Destination Port**.
  - In **Destination IP Address**, enter the IP Address of the remote Syslog Server. Both IPv4 and IPv6 addresses are supported.
  - In **Port**, enter the port of the remote Syslog Server. Valid port range is: 514; 1025 to 65535. Default: 514.
- Under **Scheduled Report Settings,**
  - If you have opted for Scheduled Reports, in **When scheduled backup is done, send an email to**, enter the desired email ID. The report will be generated and sent to this email ID.

To select the Scheduled Report options log into the SA mode.

- Under **Error Report Settings**,
  - Select **Ethernet** as the **Destination Port**.
  - In **Destination IP Address**, enter the IP Address of the remote Syslog Server. Both IPv4 and IPv6 addresses are supported.

- In **Port**, enter the port of the remote Syslog Server. Valid port range is: 514; 1025 to 65535. Default: 514.

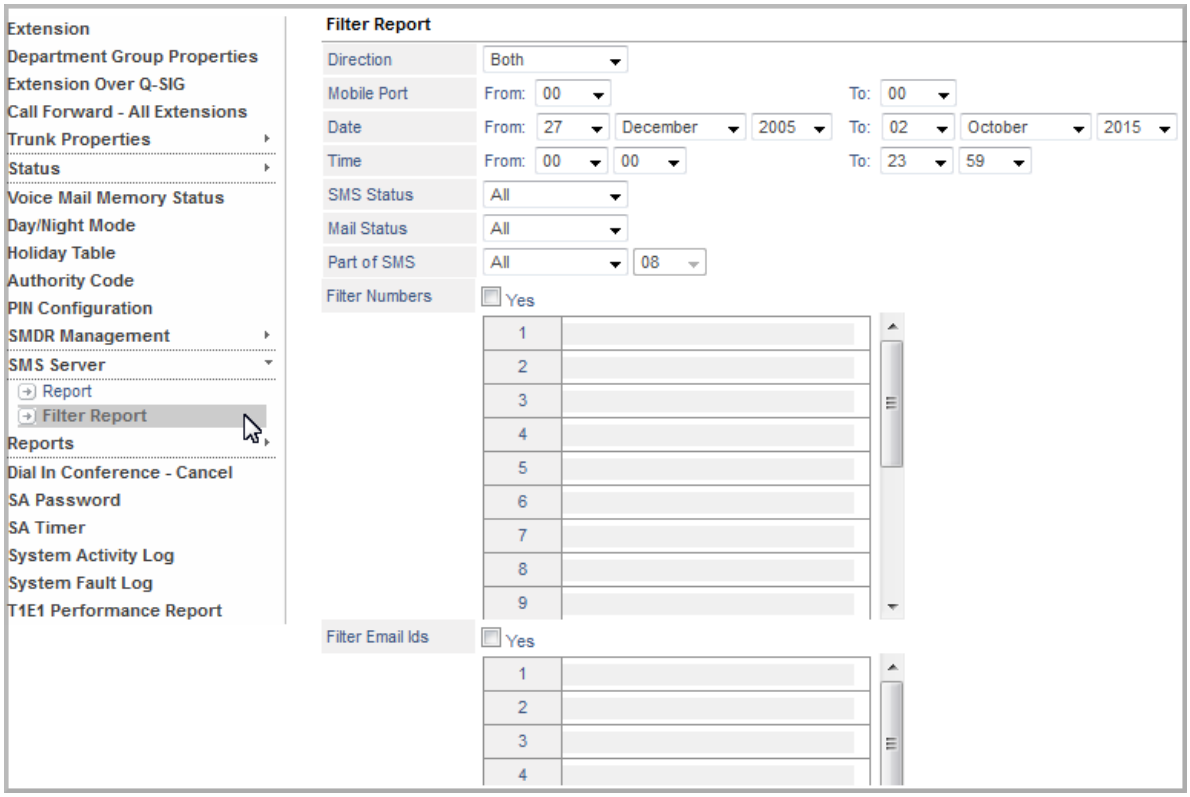
How to use

You can print Reports whenever you want or schedule printing of the report from the System Administrator mode.

You must set the filters as per your requirement before you print the Report. To do this,

- Open Jeeves.
- Log in as System Administrator.
- Click **SMS Server** to expand.

Settings Report Filters



- Click **Filter Report** to set the various filters as per your requirement:
  - Select the **Direction**. You can select **SMS to Email** or **Email to SMS** or **Both**.
  - Select the **Mobile Port/s** using which the SMS are sent/received. You can select the desired range in the **From** and **To** fields.
  - Select the **Dates** during which the SMS/Emails are sent/received.You can select the desired range in the **From** and **To** fields.

- Select the **Time** duration during which the SMS/Emails are sent/received.You can set the desired range in the **From** and **To** fields.

- Select the type of **SMS Status**. You can select from the following:
  - All
  - Pending
  - Sent
  - Delivered
  - Not Delivered
  - Received
  - Failed

- Select the type of **Mail Status**. You can select from the following:
  - All
  - Sent
  - Received
  - Pending
  - Failed

- The minimum and maximum number of parts in which an SMS can be sent is from 1 to 8. Select the **Part of SMS** for which you want to generate the report. After you have selected the Part of SMS value, select the desired filter—**All**, **Equal to**, **Less Than** or **More than**—to be applied to that value. For example is you select 5 as the Part of the SMS and More than as filter, the report will be generated for all the SMS sent in more than 5 parts.

- If you want reports to be generated for certain numbers, select the **Filter Numbers** check box and configure the desired numbers in the table.

- If you want reports to be generated for certain Email IDs, select the **Filter Email Ids** check box and configure the desired Email IDs in the table.

- Click **Submit**.

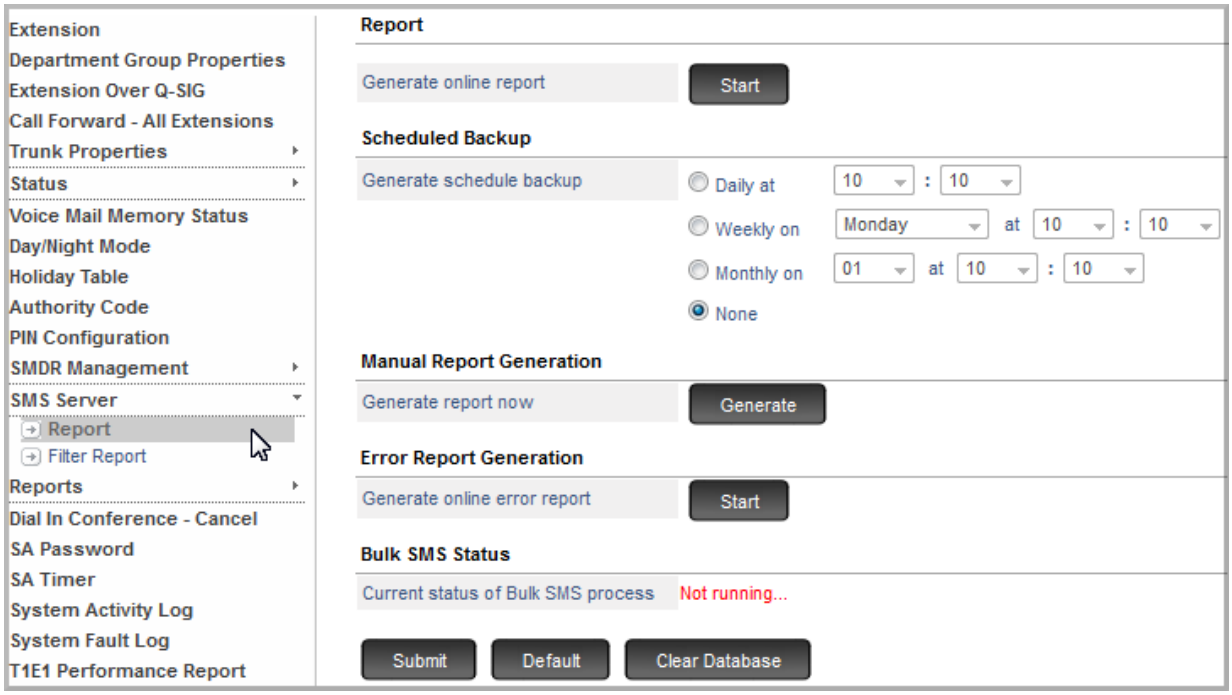
- To view the report, click the **Generate Report** button.

- The report is displayed in a new window.

Online Report Generation

- Click the **Report** under **SMS Server**.

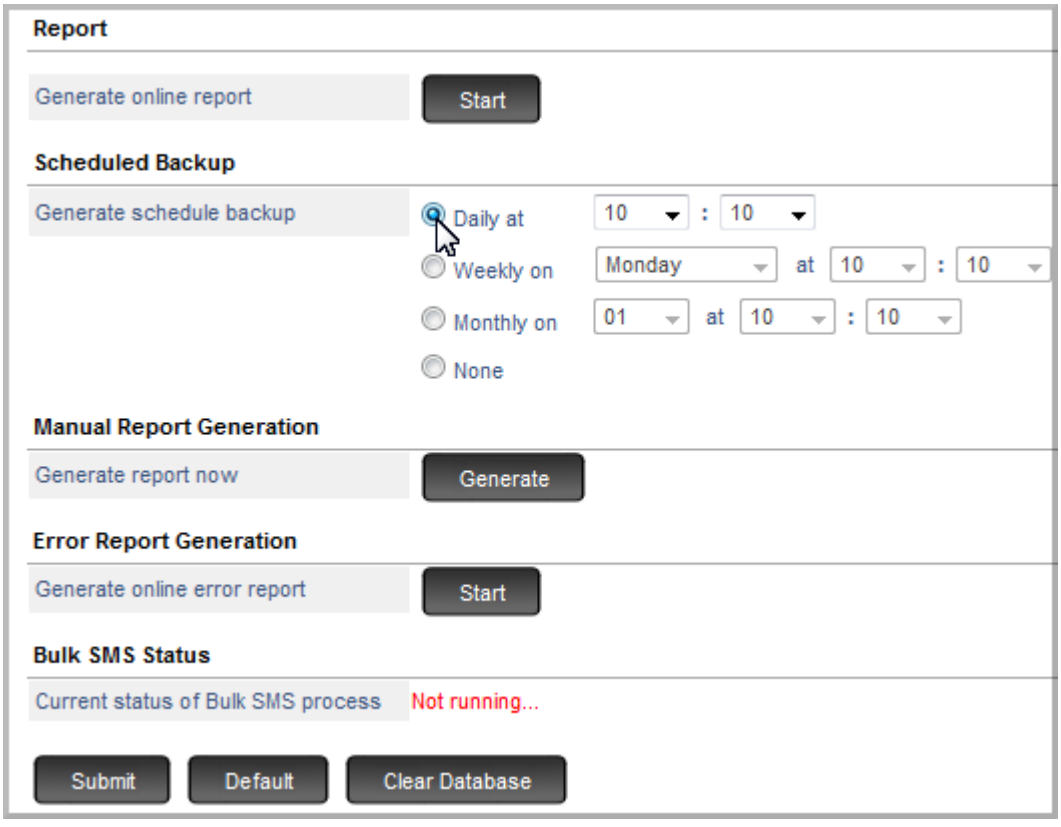
- To **Generate online report,** click the **Start** button.



- To stop printing, click the **Stop** button.

Scheduled Report Generation

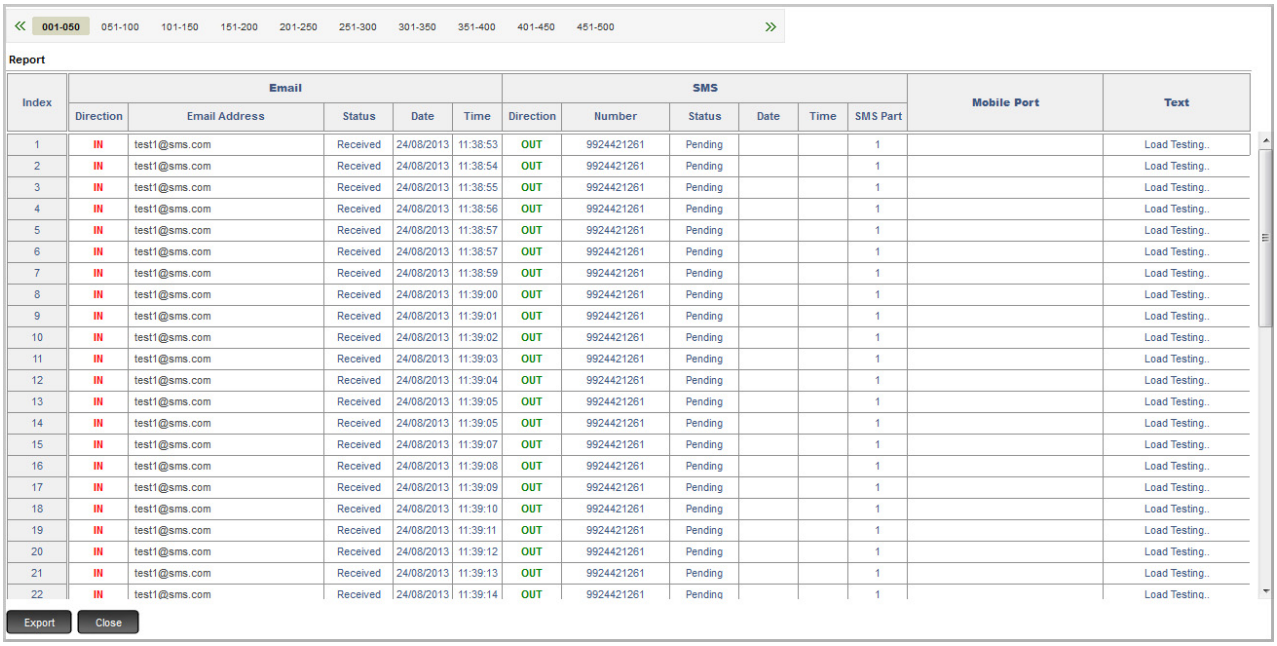
- Click the **Report** under **SMS Server**.
- To **Generate schedule backup**, select the desired option a particular day, day of the week, or day of the month.



- Click **Submit** to save.

Manual Report Generation

- To generate the report manually, click the **Report** under **SMS Server**.
- To **Generate report now**, click the **Generate** button.
- The report is generated and appears in a new window.

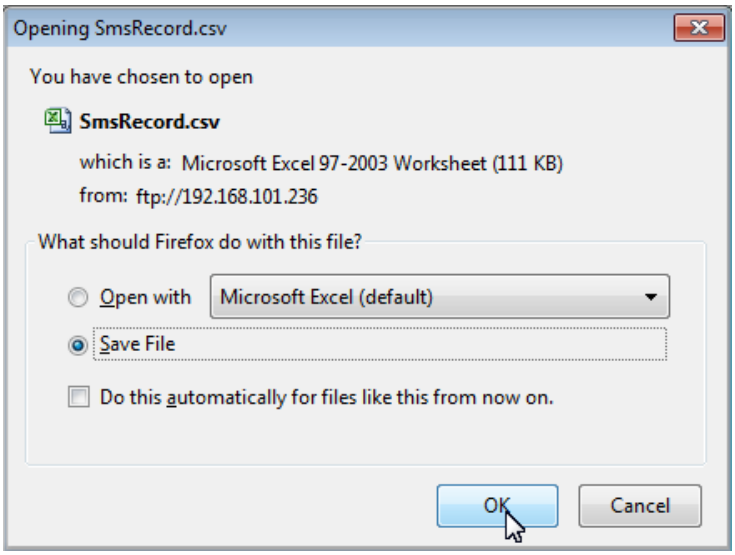


Report

Index	Email					SMS						Mobile Port	Text
	Direction	Email Address	Status	Date	Time	Direction	Number	Status	Date	Time	SMS Part		
1	IN	test1@sms.com	Received	24/08/2013	11:38:53	OUT	9924421261	Pending			1		Load Testing..
2	IN	test1@sms.com	Received	24/08/2013	11:38:54	OUT	9924421261	Pending			1		Load Testing..
3	IN	test1@sms.com	Received	24/08/2013	11:38:55	OUT	9924421261	Pending			1		Load Testing..
4	IN	test1@sms.com	Received	24/08/2013	11:38:56	OUT	9924421261	Pending			1		Load Testing..
5	IN	test1@sms.com	Received	24/08/2013	11:38:57	OUT	9924421261	Pending			1		Load Testing..
6	IN	test1@sms.com	Received	24/08/2013	11:38:57	OUT	9924421261	Pending			1		Load Testing..
7	IN	test1@sms.com	Received	24/08/2013	11:38:59	OUT	9924421261	Pending			1		Load Testing..
8	IN	test1@sms.com	Received	24/08/2013	11:39:00	OUT	9924421261	Pending			1		Load Testing..
9	IN	test1@sms.com	Received	24/08/2013	11:39:01	OUT	9924421261	Pending			1		Load Testing..
10	IN	test1@sms.com	Received	24/08/2013	11:39:02	OUT	9924421261	Pending			1		Load Testing..
11	IN	test1@sms.com	Received	24/08/2013	11:39:03	OUT	9924421261	Pending			1		Load Testing..
12	IN	test1@sms.com	Received	24/08/2013	11:39:04	OUT	9924421261	Pending			1		Load Testing..
13	IN	test1@sms.com	Received	24/08/2013	11:39:05	OUT	9924421261	Pending			1		Load Testing..
14	IN	test1@sms.com	Received	24/08/2013	11:39:05	OUT	9924421261	Pending			1		Load Testing..
15	IN	test1@sms.com	Received	24/08/2013	11:39:07	OUT	9924421261	Pending			1		Load Testing..
16	IN	test1@sms.com	Received	24/08/2013	11:39:08	OUT	9924421261	Pending			1		Load Testing..
17	IN	test1@sms.com	Received	24/08/2013	11:39:09	OUT	9924421261	Pending			1		Load Testing..
18	IN	test1@sms.com	Received	24/08/2013	11:39:10	OUT	9924421261	Pending			1		Load Testing..
19	IN	test1@sms.com	Received	24/08/2013	11:39:11	OUT	9924421261	Pending			1		Load Testing..
20	IN	test1@sms.com	Received	24/08/2013	11:39:12	OUT	9924421261	Pending			1		Load Testing..
21	IN	test1@sms.com	Received	24/08/2013	11:39:13	OUT	9924421261	Pending			1		Load Testing..
22	IN	test1@sms.com	Received	24/08/2013	11:39:14	OUT	9924421261	Pending			1		Load Testing..

Export Close

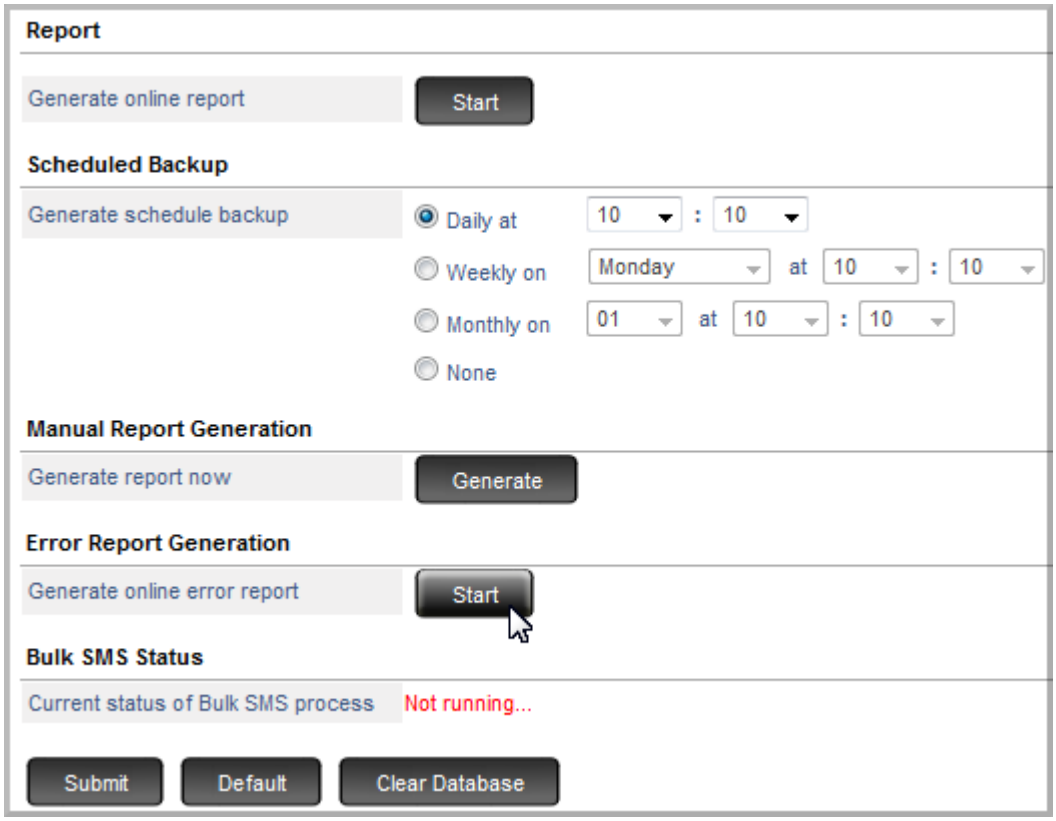
- Click **Export**, if you wish to save the report at the desired location.



The report will be saved in CSV format.

Online Error Report Generation

- Click the **Report** under **SMS Server**.
- To **Generate Online error report,** click the **Start** button.



- To stop printing, click the **Stop** button.

Bulk SMS Status

- Click the **Report** under **SMS Server**.
- In **Current status of Bulk SMS process,** the status of the Bulk SMS process is displayed.
- If the **Current status of Bulk SMS process** displays *Running*, click the **Abort** button to stop the ongoing process midway.

To clear all the database, click the **Clear Database** button.

SMS over IP

The Unified Messaging Functionality of SARVAM UCS includes SMS over IP.

SMS over IP allows you to:

- Forward SMS received on Mobile Port from any extension user as IM to Matrix VARTA UC Users/ Extended SPARSH VP710 Users.
- Forward IM from Matrix VARTA UC Users/Extended SPARSH VP710 Users as SMS to any extension user through the desired Mobile Port.

How it works

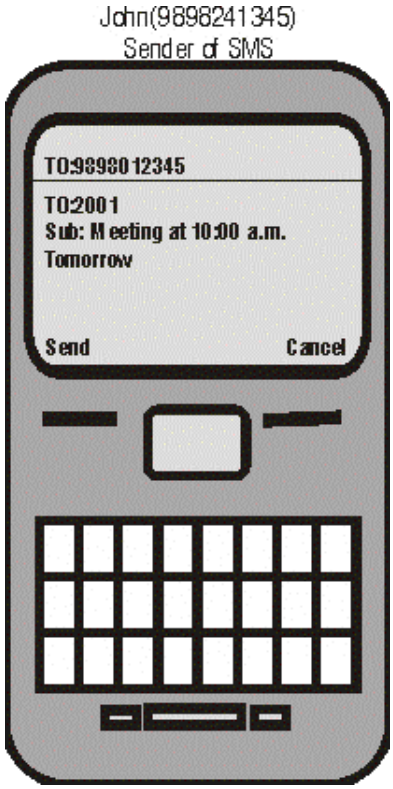
For this feature to work,

- you must define the Mobile port through which the messages are to be sent/received (Fixed/LCR). See “SMS Routing”.
- configure the SMS parameters and the SMS Budget parameters (if required) on the respective Mobile ports, see “Configuring Mobile Trunks”.
- enable **SMS over IP** for Matrix VARTA UC Users. See “Configuring Matrix VARTA ADR100/AMP100 UC Clients” and .“Configuring Matrix Extended SPARSH VP710”.
- configure the **SMS over IP Settings**. See “How to Configure”.

SMS over IP Functionality requires SMS Server License. Make sure you purchase the license for using this feature. For more information, see “License Management”.

Forwarding an incoming SMS as an IM

- To forward SMS as an IM, the SMS received on the Mobile Port must be in a specific format. Illustrated below is as example of the format.



Here,

- The sender of the SMS is John (9898214345).
- The sender must send the message to the SIM Number of the Mobile Port of SARVAM UCS, that is 9898012345.

- The body of the message must contain the following:
  - **To:** This is the destination where the IM is to be sent. In this case 2001.
  - **Sub:** This is the message that will be displayed to the recipients. In this case, ‘Meeting at 10.00 a.m. tomorrow’.

- When the SMS is received on Mobile Port (9898012345) in the above format, the system checks the senders number (9898214345 - John) in the Denied numbers list of the SMS Server.

If the SMS is received in multiple parts, the first 160 characters only will be sent as an IM.

- If a match is found in the Denied list, the SMS will be rejected. If no match is found, then it checks the destination where the SMS is to be forwarded as an IM.

  The destination can be the recipients Extension number or Name. The **To** field in the message body specifies the destination.

You cannot send the same message to multiple destinations at the same time. You must send the same message to individual recipients separately.

- When the destination is an Email ID, the further handling of the SMS will be done by the SMS Server Application. See “SMS Server” for more details.

- When the destination is an Extension Number, the system will search for the Extension Number in its database. When a match is found and it is a SIP extension, the system will check if the option **SMS over IP** is enabled for this SIP extension. If it is enabled the SMS will be forwarded as an IM. If this option is disabled, the further handling of the SMS will be done by the SMS Server Application. See “SMS Server” for more details.

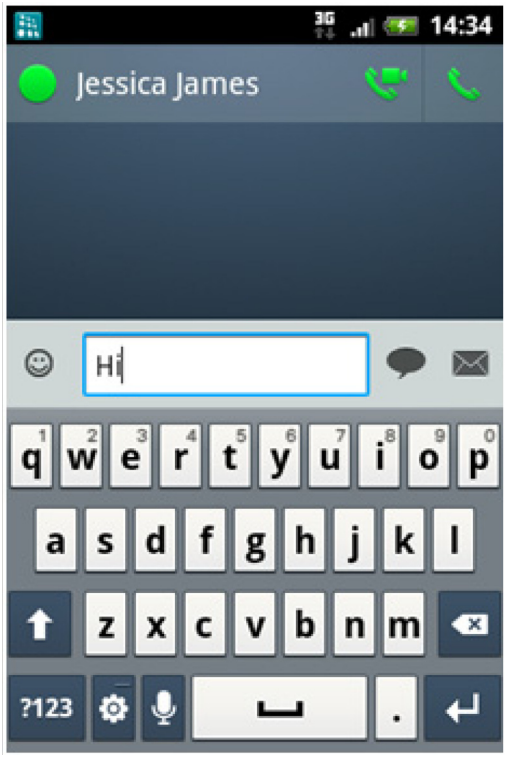
  When the destination is an Extension Number other than SIP extension, the further handling of the SMS will be done by the SMS Server Application. See “SMS Server” for more details.

- When the destination is a Name, the system will search for the Name in its database. When a match is found, the system will check the corresponding Extension number. If it is a SIP extension the SMS will be forwarded as an IM. If the Extension Number is any other extension or if a match is not found the further handling of the SMS will be done by the SMS Server Application. See “SMS Server” for more details.

Forwarding an IM as an SMS

You can send a message as an IM or SMS using Matrix VARTA Mobile UC Clients only.

To forward an IM as an SMS, refer to the illustration below:



Here,

- The SIP extension user is typing a message to Jessica James. This message can either be sent as an IM or an SMS.

- To send an SMS, user taps **Message** .
- The system will check if the SMS is sent by a SIP extension. If it is a SIP extension, then it checks if the **SMS over IP** option is enabled for the SIP extension. If it is enabled then the SMS will be sent using the desired Mobile Port. The Mobile Port using which the SMS will be sent to the destination will depend on the configurations— Fixed/LCR— in SMS over IP Settings.
- When the SMS is received on Mobile Port, the system checks the **Send SMS** and the **Call Budget** parameters configured for the Mobile Port.

  If Send SMS is disabled or the Call Budget exceeds the SMS will be rejected.

  If **Send SMS** is enabled and the **Call Budget** has not exceeded, the system checks the senders number in the Denied numbers list as configured in the SMS over IP Settings.

- If a match is found in the Denied list, the SMS will be rejected. If no match is found, then it checks the destination where the SMS is to be delivered. The destination can be an Extension Number/ External Number or Name.

  When the destination is an Extension Number/Name the system will search for the Extension Number in its database and then send an SMS to its corresponding Mobile Number.

  When the destination is a Mobile number the system will send the SMS to this Mobile number.
- The senders Name and/or Number will be displayed to the recipients, if you have enabled **Add Footer in SMS** and configured the parameter **Footer text**.

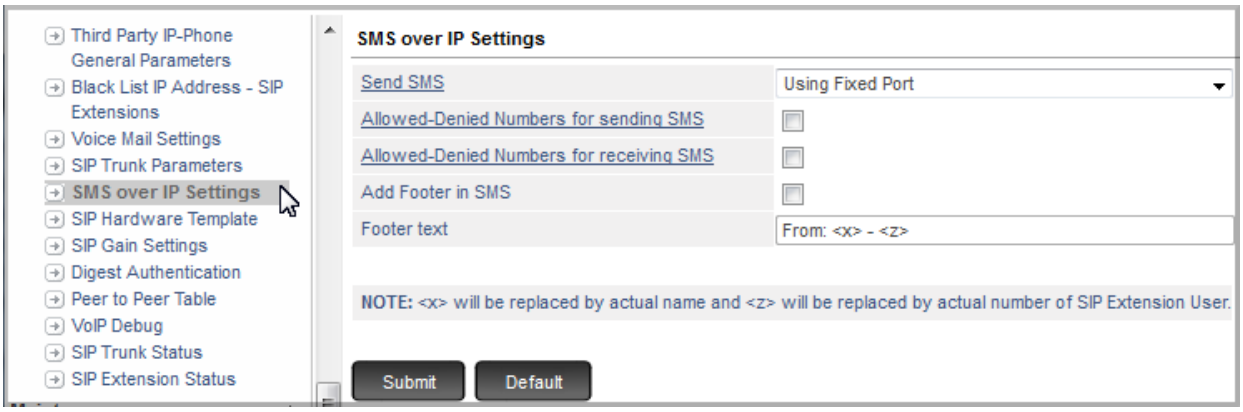
*It is recommended that you configure the parameters **Add Footer in SMS and Footer text**, so that the recipient knows the sender of the message. If this parameter is not configured, only the mobile port SIM number will be displayed to the recipient.*

How to Configure

You need to configure the following SMS over IP parameters,

- Log in as System Engineer.
- Under **Configuration**, click **VoIP Configuration**.

- Click **SIP over IP Settings**.



- Configure the following parameters:

  - You can **Send SMS** through certain fixed Mobile Ports or through different Mobile Ports according to specific numbers and time.

    To send messages through fixed Mobile Ports, select **Using Fixed Mobile Port** and configure the **SMS over IP - Fixed Port Routing** table. For detailed information, see “Fixed Port Routing (SMS over IP)”.

    To send messages to specific numbers through certain preferred Mobile Port/s during a defined time interval, select **Based on LCR Table** and configure the **SMS Routing-LCR** table. For detailed information, see “Least Cost Routing”.

    Default: Using Fixed Mobile Port.

  - Configure the **Allowed-Denied Numbers for sending SMS** list, if you want to allow or restrict sending of messages on specific numbers. To do this,
    - Select the **Allowed-Denied Numbers for sending SMS** check box.
    - Click on **Allowed-Denied Numbers for sending SMS** and the **Allowed-Denied Numbers for sending SMS** table opens. You can configure upto 999 numbers.
    - In the Allowed column, enter the numbers on which messages can be sent and in the Denied column, enter the numbers on which messages cannot be sent.
    - You can also configure this list by clicking the **Allowed-Denied Numbers for sending SMS** link under **SMS Routing**.

      Default: Disabled

  - Configure the **Allowed-Denied Numbers for receiving SMS** list, if you want to allow or restrict receiving messages from specific numbers. To do this,
    - Select the **Allowed-Denied Numbers for receiving SMS** check box.
    - Click on **Allowed-Denied Numbers for receiving SMS** and the **Allowed-Denied Numbers for receiving SMS** table opens. You can configure upto 999 numbers.
    - In the Allowed column, enter the numbers from which messages can be received and in the Denied column, enter the numbers from which messages cannot be received.
    - You can also configure this list by clicking the **Allowed-Denied Numbers for receiving SMS** link under **SMS Settings**.

      Default: Disabled.

- Enable the **Add Footer in SMS** check box and in **Footer text** enter the desired text/message to be sent to the recipient as signature in the SMS. By default, the senders Name and Number will be sent.

- Click Submit to save the changes.

SMS/Email Group

What's this?

In an organization you can group together extension users as per your requirements. Groups can be made according to the departments, project-wise, product-wise, according to a particular task. The system clubs together the extension users assigned the same Group.

You can reach all the members simultaneously in a Group, by sending a SMS or an Email.

How to configure

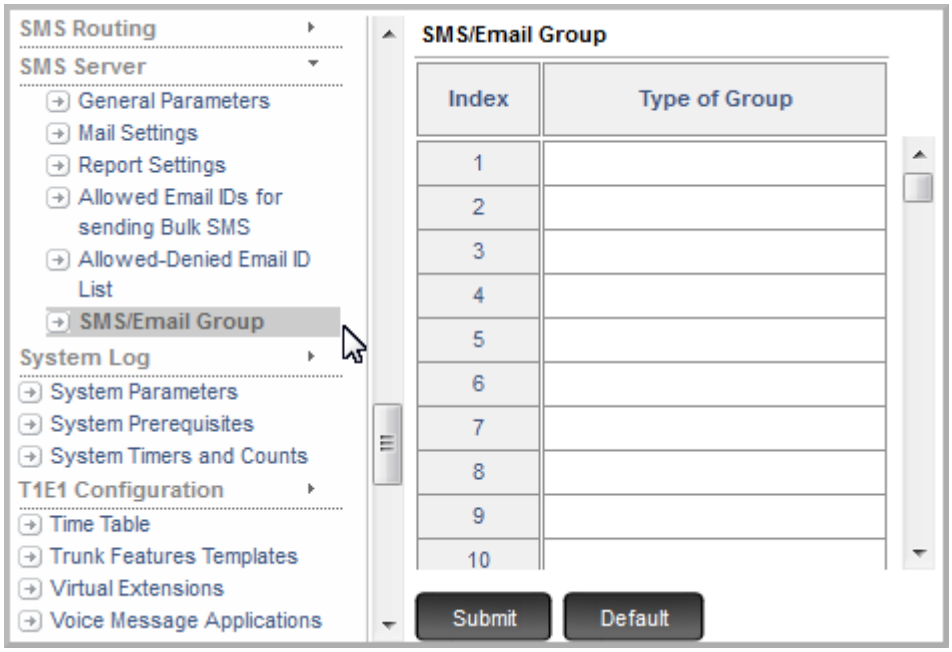
To use SMS/Email Group feature,

- Decide the type of groups you want to create.
- Make a list of extensions to be assigned to a particular group.
- Configure the type of Groups at each index in the SMS/Email Group table. For instructions, see "Configuring SMS/Email Group table".
- Assign the desired extensions to the SMS/Email Group you created as per your requirement. For instructions, see "Configuring SLT Extensions", "Configuring DKP Extensions", "Configuring ISDN Terminals", "Configuring Matrix SPARSH VP330", "Configuring Matrix SPARSH VP248", "Configuring Matrix VARTA ADR100/AMP100 UC Clients", "Configuring Matrix SPARSH VP310", "Configuring Matrix SPARSH VP510", "Configuring Matrix SPARSH VP210"
- Assign a Global/Personal Directory contact to the SMS/Email Group you created as per your requirement. For instructions, see "Configuring Personal Directory using Jeeves" and "Configuring Global Directory using Jeeves" under "Abbreviated Dialing".

Configuring SMS/Email Group table

- Log in as System Engineer.
- Under **Configuration**, click **SMS Server.**

- Click **SMS/Email Group**.



You can configure maximum 99 Groups,

- At the desired index, in **Type of Group** configure the Group Name. The Group Name can be a maximum of 16 characters. Default: Blank.

- Click **Submit** to save your entries.

CHAPTER 13

Computer Telephony Integration (CTI)

CTI stands for Computer Telephony Integration. CTI is a technology that integrates a telephone and a computer. Using CTI, you can control your telephone with your computer and vice versa. This technology contributes to the success of a modern business.

The Matrix TAPI Service Provider (TSP) acts as a link to integrate the interactions between your telephones and your computers. The computer in which you install this application functions as a Client and SARVAM UCS functions as a Server. In an organization, it may happen that the data is stored in different computers. In this scenario, the Matrix TAPI Service Provider can be installed in three (maximum) different computers, if required.

Connecting your telephone with your computer has advantages such as — initiate a call at a click of a mouse directly from a CRM system (Customer Relationship Management) or from a database, view missed calls directly on your screen and can call back with just one click, transfer and call recording can be initiated via a mouse click.

- *The Matrix TAPI Service Provider (TSP) is developed using TAPI Standard 2.2.*
- *To use this feature, you must purchase the CTI License. Refer "License Management" for more details.*
- *To develop your own TAPI Application, refer to the **Matrix TAPI Developer's Guide**. The documentation can be found at https://www.matrixcomsec.com/support/telecom-product-manuals/*

How it works

Using CTI, only following features and facilities of SARVAM UCS can be accessed:

Features	SLT	DKP/Extended IP Phone	MATRIX VARTA WIN200 Desktop UC Client	Standard IP Phone
Make a call	✓	✓	✓	x
Answer/receive a call	x	✓	✓	x
Put a call on hold	x	✓	✓	x
Retrieve a held call	x	✓	✓	x
Toggle between two calls	✓	✓	✓	x
Transfer a call	✓	✓	✓	x
Disconnect/Release a call	✓	✓	✓	✓

Features	SLT	DKP/Extended IP Phone	MATRIX VARTA WIN200 Desktop UC Client	Standard IP Phone
Conference - Add/Remove a party	✓	✓	✓	x
Dial digits for Multi-Stage dialing	✓	✓	✓	x

For CTI to work:

- You must install the Matrix TAPI Service Provider (TSP) in your computer (Client). For instructions refer to the TAPI User Guide. The documentation can be found at https://www.matrixcomsec.com/support/telecom-product-manuals/
- Configure the Matrix TSP parameters. For instructions refer to the TAPI User Guide.
- Configure the CTI parameters in SARVAM UCS. See “Configuring CTI Parameters”.
- Make sure the necessary third party TAPI applications are installed.

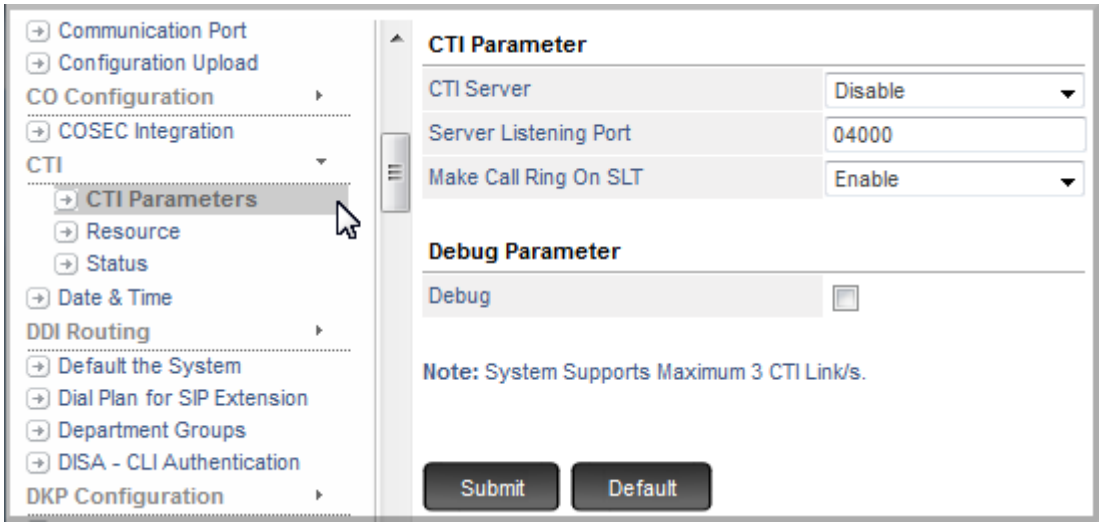
This is how CTI works,

- After successful installation of the Matrix TSP in your Computer (Client) and configuring the necessary parameters, a binding (CTI link) is established between the Client and SARVAM UCS. The SARVAM UCS communicates with the Client via Matrix TSP.
- All the information is passed by SARVAM UCS to the Matrix TSP and vise versa.
- The third party softwares in the Client can also communicate with the Matrix TSP to get the required information.
- You can handle calls from your computer (desktop application) as well as your extension phone.

Configuring CTI Parameters

You must configure the following CTI parameters in SARVAM UCS. To do so,

- Log in as System Engineer.
- Under **Configuration**, click **CTI**.
- Click **CTI Parameters**.



- Enable **CTI Server**.
- Enter **Server Listening Port**. When the CTI Link is to be established between Matrix TSP and SARVAM UCS, then the binding will be done on this port.

Make sure the same port is configured as the **Listening Port of System— SARVAM UCS** *in the Client. For more details, see* ***Matrix TSP User Guide****.*

- Enable **Make Call Ring on SLT**, if you want the SLT to ring when an outgoing call is initiated from the CTI Client.

If Extended IP Phone or DKP is connected as an extension, the speaker of the phone is automatically turned on when an outgoing call is initiated from the CTI Client.

- Select the **Debug** check box to log the CTI Server events.

- Click **Resources**, to view the details of the CTI events and activities. These details are used by the Technical Team only.
- Click **Status**. It displays the following details of each Client:
  - **IP Address:** It displays the IP Address of the Client in which you have installed the Matrix TSP.
  - **Port:** It displays the Port of the Client on which the CTI binding with the Server is established.
  - **TSP Version:** It displays the Software Version of the Matrix TSP installed in the Client.
  - **Status:** It displays the status of the Client and Server binding, that is whether the link is up or down.

CHAPTER 14

Incoming and Outgoing Call Routing

Incoming Call Routing

What's this?

SARVAM UCS offers various options for routing incoming calls on trunks. You can configure incoming call routing in the following ways:

- Land the call on a group of extensions, referred to as "Trunk Landing Group (TLG)".
- Route the call to the Built-In Auto Attendant.
- Route call to the Voice Mail Auto Attendant.
- Greet the caller with Trunk Auto Answer, and then route the call to the Trunk Landing Group.
- Route the call to a specific extension on the basis of the CLI received.
- Route the call according to the DDI numbers[229].
- Route the call to a group of trunks to use the Gateway functionality of SARVAM UCS.

It is also possible to configure a different routing option for each time zone—Working Hours, Break Hours, Non-working hours—according to your call routing requirement.

Besides the type of routing you configure, there are certain features supported by SARVAM UCS on trunks that also determine how an incoming call will be routed, when these features are enabled on the trunk.These features are:

- "Call Back on Trunk Ports": Missed calls received on a trunk are returned to the same or to an alternative number.

- "Direct Inward System Access (DISA)": From a remote location, users can make a call to an extension of SARVAM UCS. The call is routed to an extension after authentication and the users can use the system resources (make calls, access features, configure the system).

- "RCOC (Return Call to Original Caller)"[230]: When an unanswered outgoing call is made by an extension is returned by the called party, this call is routed to the very extension that originally made the outgoing call.

229. Supported on ISDN and SIP trunks only.
230. Supported on digital trunks—BRI, T1E1PRI, Mobile and SIP.

How it works

When a call lands on a trunk, the system checks the trunk features and the routing option configured for the current time zone on the trunk in the following sequence and accordingly routes the call.

1. Call Back on Trunks
2. DISA - CLI Authentication
3. RCOC (Return Call on Original Caller)
4. DDI Routing and DISA - CLI Authentication
5. DDI Routing and DISA - PIN Authentication
6. DDI Routing
7. DISA - PIN Authentication
8. Auto Attendant - Built-In or Voice Mail
9. Trunk Auto Answer
10. Trunk Landing Group

In the default Incoming Call Routing configuration, incoming calls on a trunk of SARVAM UCS are routed to a "Trunk Landing Group (TLG)".

Routing to a Trunk Landing Group

A Trunk Landing Group (TLG) is a "Routing Group", consisting of a group of extension types. These extensions may be Digital Key Phones (DKP), Single Line Telephone (SLT), SIP, ISDN Terminals or Virtual Extensions. A Routing Group can have a single extension type or as many as 32 extension types as members.

A Routing Group can also have a Voice Mail Auto Attendant and an Outgoing Trunk Bundle Group as members, the significance of which is described later in this topic under *Routing to the Voice Mail Auto Attendant* and *Routing to an Outgoing Trunk Bundle Group*.

- The call is landed on the The Trunk Landing Group (TLG) assigned to the trunk for the current time zone.

  By default, Trunk Landing Group number 01 assigned to all trunks for all time zones and has DKP software port 001, SLT software ports 001 and 002 as members.

- The member extensions in the TLG start ringing in the sequence in which they are arranged in the TLG.

- Each member extension rings for the duration of the *Ring Timer* (programmable; default: 15 seconds).

- If *Continuous Ring* is enabled, each extension will ring continuously till the call is answered. The extension continues to ring even as other extensions in the group are hunted.

  If the call remains unanswered even after the last extension in the group has been hunted, the system will loop back and start hunting from the first extension, all over again.

- If *Rotation* is enabled on a TLG, for each new call on a trunk, the system will land the call on the extension next to the one that received the last call. Thus, ensuring an equal distribution of incoming calls on all member extensions of the TLG.

  When Rotation is disabled, for each new call on a trunk, the system will land the call on the first free extension of the TLG.

  To know more about how TLG works, see "Trunk Landing Group (TLG)".

You can also route calls that remain unanswered by extensions in the Trunk Landing Group to the Voice Mail Auto Attendant.

Routing Calls to the Auto Attendant

- When a call lands on the trunk, the system checks the trunk features and the routing option configured for the current time zone on the trunk in the sequence mentioned earlier.

- When Auto Attendant is configured for the current time zone, the system checks for the type of Auto Attendant configured for the current time zone.

Built-In Auto Attendant

- If Built-In Auto Attendant is configured,
  - The system greets the caller with a pre-recorded voice message and prompts the caller to dial the desired number.
  - The call is placed on the extension number dialed by the caller.
  - If the caller fails to dial any extension number the call is routed to the TLG configured for the current time zone.
  - If the caller dials ‘9’ for Operator, the call is routed to the Operator group assigned to the trunk. To know more, see “Configuring 'Operator'”.

  For a detailed description of how the Built-In Auto Attendant works, see “Auto Attendant”.

Voice Mail Auto Attendant

- If Voice Mail Auto Attendant is configured,
  - The call is routed to the Voice Mail System (VMS).
  - The VMS greets the caller with the Welcome message and the Greeting message selected for the current time zone.
  - The VMS processes the call as per the Voice Mail Auto Attendant Menu assigned to the trunk.

  For a detailed description of how the Voice Mail Auto Attendant works, see “Auto Attendant”.

You can also route calls that remain unanswered by extensions in the Trunk Landing Group to the Voice Mail Auto Attendant.

Delayed Auto Attendant

- If Delayed Auto Attendant is configured,
  - The system first places the call on the TLG.
  - It waits for the configured *Auto Attendant Delayed Timer* to expire.
  - If the call remains unanswered by the TLG, on the expiry of the Auto Attendant Delayed Timer, the system checks for the type of Auto Attendant selected for the current time zone.

  - If Built-In Auto Attendant is selected, the call is routed accordingly. See description for“Built-In Auto Attendant” in this topic.

  - If Voice Mail Auto Attendant is selected the call is processed as per the Voice Mail Auto Attendant Menu assigned to the trunk.

  To know more about Delayed Auto Attendant, see “Auto Attendant”.

Greeting with Trunk Auto Answer

- When a call lands on the trunk, the system checks the trunk features and the routing option configured for the current time zone on the trunk in a sequence mentioned earlier.

- When Trunk Auto Answer is enabled on a trunk, the system checks for the type of Trunk Auto Answer set on the trunk.

- If ***Trunk Auto Answer - For All Calls*** is selected,
  - the call immediately answered with a Greeting message, known as the *Trunk Auto Answer Greeting.*
  - The call is then routed to the landing destination selected for the current time zone.
  - The system waits for the duration of the *Built-In Auto Attendant Inactivity Timer* (default: 60 seconds) as it waits for the call to be answered.
  - If the call is answered by the landing destination, within this timer, the caller is connected to the answering extension.
  - If the call remains unanswered until the expiry of this timer, the system plays the *Trunk Auto Answer Busy Bye* message or a Busy Tone and releases the trunk port.

- If ***Trunk Auto Answer - When Busy*** is selected,
  - the system checks will first check the state of the landing destination for the current time zone.
  - If the landing destination is busy, the system answers the call with the *Trunk Auto Answer Greeting* message.
  - The system waits for the duration of the *Built-In Auto Attendant Inactivity Timer* (default: 60 seconds) for the landing destination to become free.
  - If the landing destination is free before the timer expires, the call is placed.
  - If the landing destinations is busy till the end of this timer, the system plays the Trunk Auto Answer Busy Bye message or a Busy tone to the caller and releases the trunk port.

- If ***Trunk Auto Answer - Delayed*** is selected,
  - the system first routes the incoming calls to the TLG selected for the current time zone.
  - It waits for the duration of the *Delayed Trunk Auto Answer Timer* (programmable; default:10 seconds) for any of the extensions in the TLG to answer the call.
  - If the call is not answered by any of the extensions within this timer, the system answers the call with the *Trunk Auto Answer Greeting* message, and rings the landing destination.
  - The system starts the *Built-In Auto Attendant Inactivity Timer* (default: 60 seconds) as it waits for the call to be answered and repeatedly plays the *Ring Back Tone* message or tone to the caller.
  - If the call is answered before the Inactivity Timer by the landing destination, the system stops the timer and the Ring Back tone message/tone and connects the caller to the destination.
  - If the call remains unanswered till the expiry of the Inactivity Timer, the system plays the Trunk Auto Answer Busy Bye message or a Busy tone to the caller and releases the trunk port.

Feature Interactions

As mentioned earlier, certain trunk features also determine incoming call routing. Consider the following feature interactions between Trunk Auto Answer and other trunk features when configuring Trunk Auto Answer as a routing option on a trunk.

- **Call Back on Trunks:** This feature should be disabled when using Trunk Auto Answer.

- **DISA - CLI Authentication:** This feature should be disabled when using Trunk Auto Answer.

- **Return Call to Original Caller (RCOC):** If both Trunk Auto Answer and RCOC are enabled on the same trunk, the call will first be answered as per Trunk Auto Answer, and then placed on the destination as per RCOC.

- **CLI Based Routing:** If both Trunk Auto Answer and CLI Based Routing are enabled on the same trunk, the call will first be answered as per the Trunk Auto Answer, and then placed on the destination as per the CLI Based Routing Table.

- **DISA - PIN Authentication:** If both Trunk Auto Answer and DISA PIN Authentication are enabled on the same trunk, only DISA - PIN Authentication will work. The call will be treated as a DISA call and routed as per the DISA logic.

- **Auto Attendant:** If both Trunk Auto Answer and Auto Attendant are enabled on the same trunk, only Auto Attendant will work. The call will be processed by the Auto Attendant.

Routing Calls based on CLI Received

It is possible to land calls from a particular telephone number on a particular extension or group of extensions using "CLI Based Routing" on the trunk. For this, you need to configure the CLI of the calling parties and their corresponding landing destinations in the *CLI Based Routing Table*.

- When CLI Based Routing is configured for the current time zone, the system checks the entries of the CLI Based Routing Table.

- If the calling party's number is found in the CLI Based Routing Table, the call is placed on the corresponding landing destination.

- If the calling party's number does not exist in the CLI Based Routing Table, the call will be routed according to the incoming call routing option you have configured.

  For a detailed feature description, see "CLI Based Routing".

Routing Calls on the basis of DDI Numbers

Direct Dialing-In (DDI) is a feature of ISDN (T1E1PRI and BRI) and SIP trunks which enables calls to be directly placed to the desired extensions without operator intervention. The DDI Numbers given by the Service Provider are mapped to extensions. Callers can reach the extensions directly by dialing the DDI number of the respective extension.

- DDI Routing works on the basis of the *IC Reference Table* and the *DDI Routing Table*.

- The T1E1PRI, BRI and SIP trunks ports are assigned IC Reference ID, which may be different for each time zone. The IC Reference ID is the reference number that acts as an identifier to the translation logic programmed in the IC Reference Table. When a call lands on the ISDN/SIP trunk, the system checks the IC Reference ID assigned to it. It checks the IC Reference Table for the corresponding DDI Routing Reference ID for call resolving.

- Using the DDI Routing Reference ID, the system checks the DDI Routing Table for mapping the received DDI number to a flexible number (target extension).

- The system first compares the received DDI number (called party number) with the DDI numbers programmed in the DDI Routing Table.

- Once a perfect match is found, the system checks for the Route on First Destination flag in the IC reference table. If the flag is enabled, the call lands on the first extension of the MSN number in the DDI Routing Table. If the flag is disabled, the call is routed to the identified target extension.

- Once the target extension is identified, the system checks the *DDI IC routing* flag of the extension in its "Station Advanced Feature Template". If the flag is enabled, the call lands on the extension else the call is routed to the TLG assigned to the trunk.

- When the call lands on the DDI extension, the caller gets the Ring Back Tone. The extension rings for duration of the *DDI Ring Timer*.

- If the call is not answered within the DDI Ring Timer, the system checks for the *When No reply* option in the IC reference Table for action to be taken.

- Similarly, if the DDI extension is busy, the caller gets Busy Tone. The system checks the *When Busy* option in the IC Reference Table for action to be taken.

- The options for action to be taken the DDI extension does not No Reply or is Busy are:
  - Disconnect the call
  - Route the call to Trunk Landing Group
  - Answer the call automatically, greet the caller with a voice message, and on completion of message disconnect the call.
  - Answer the call, greet the caller with a voice message, and on completion of the message route the call to Trunk Landing Group.
  - Route the call to Voice Mail; to the DDI extension's Mail Box.

To know more, refer the topics "Direct Dialing-In (DDI)" .

DDI Routing will not work if Auto Attendant or DISA are enabled on a trunk.

Routing Calls to Outgoing Trunk Bundle Groups

To use SARVAM UCS's gateway functionality, you can route incoming calls on a trunk to a group of outbound trunks, referred to as "OG Trunk Bundle Group". The Called Number (number dialed by the caller) received in the incoming call is out dialed from Outgoing Trunk Bundle Groups.

Routing to Outgoing Trunk Bundle Groups is supported only on SIP, ISDN T1E1PRI, ISDN BRI and E&M Trunks (except Express line), as they support Called Number. CO and Mobile trunks support only CLI.

To do this, you must configure the desired outbound trunks in an "Outgoing Trunk Bundle" and create an "Outgoing Trunk Bundle Group (OGTB)". You must then include this OGTB as a member of the Trunk Landing Group you assign to the SIP/ISDN T1E1PRI/ISDN BRI/E&M trunk.

When there is an incoming call on the SIP/ISDN T1E1PRI/ISDN BRI/E&M trunk,

- the system checks the trunk features and the routing option configured for the current time zone on the trunk in a sequence as mentioned earlier.
- On finding OGTB as member in the TLG of the current time zone, the call is routed to the OGTB number assigned.
- The system out dials the Called Number received in the incoming call from the OGTB.

How to configure

Incoming call routing options for a trunk are to be configured on the Trunk Feature Template assigned to the trunk.

Configuring Trunk Landing Group

To land calls on a group of extensions, you must do the following:

- First configure a Trunk Landing Group. You may configure the same Trunk Landing Group for working, break and non-working hours or you different Trunk Landing Group for all three time zones.
- To land calls on the Operator extension first, make sure the Operator extension is included in the TLG you configure. Also make sure that the Operator extension is the first member in the group.

  You can also keep the same Routing Group you configured as Operator group as the Trunk Landing Group.
- Assign the Trunk Landing Group to the trunk in the Trunk Feature Template for each time zone.
- To have a number of extensions in the group ring simultaneously, enable *Continuous Ring* on these extensions and set the *Ring Timer* for these extensions to *'00'* seconds.
- To set equal distribution of incoming calls on all extensions in the group, enable *Rotation* for the entire group (default: disabled).

For detailed instructions, see "How to configure" in *Trunk Landing Group*.

If you want incoming calls to be routed to Trunk Landing Group only, do not enable any other incoming call features on the trunk.

Configuring Auto Attendant

To route calls to the **Built-In-Auto Attendant**, you must do the follow:

- Make a list of the trunks by their port type (CO, Mobile, SIP, BRI, T1E1PRI) and port number on which you want to use the Built-In Auto Attendant.
- Configure a Trunk Feature Template with Built-In Auto Attendant enabled for the desired time zones.
- Assign this Trunk Feature Template to the desired trunks. The calls landing on this trunk will be answered by the Built-In Auto Attendant. See "Trunk Feature Template", for more information.
- Set the Start Time for the Morning, Afternoon and Evening Greeting Messages. Refer "Greeting Message Time" in *System Parameters* for instructions.
- Assign *Voice Modules* for Built-In Auto Attendant Messages. To play to callers pre-recorded voice messages as Built-In Auto Attendant greetings and to play voice prompts at each stage of the call, you need to assign Voice Modules for the following Built-In Auto Attendant Messages:

  For detailed instructions, see "How to configure" in "Auto Attendant".
- If you want the incoming calls to first land on extensions, you must configure the Auto Attendant Delayed Timer (sec). The call will be placed fist on the extensions in the Trunk Landing Group. If the call remains

unanswered till the expiry of the timer, it shall be answered by the Built-In Auto Attendant. This is known as **Delayed Auto Attendant** on trunks.

For detailed instructions, see “How to configure” in “Auto Attendant”.

To route the call to the Voice Mail Auto Attendant, you need to configure:

- Make a list of the trunks by their port type (CO, Mobile, SIP, BRI, T1E1PRI) and port number on which you want to use the Voice Mail Auto Attendant.
- Configure a Trunk Feature Template with Voice Mail Auto Attendant. enabled for the desired time zones and assign the Voice Mail Auto Attendant (VMAA) Menu.
- Assign this Trunk Feature Template to the desired trunks. The calls landing on this trunk will be answered by the Voice Mail Auto Attendant. See “Trunk Feature Template”, for more information.

  For detailed instructions, see “How to configure” in “Auto Attendant”.
- Complete the VMS related configuration.
  - Configure Welcome and Greeting messages. You may either use the default, pre-recorded welcome messages of the VMS, or record the custom welcome messages that meet your requirements, in .WAV file format.
  - Configure the parameters of the Voice Mail Auto Attendant (VMAA) Menu.

  For more information and instructions, see the “Configuring Voice Mail System”.
- If you want the incoming calls to first land on extensions, you must configure the Auto Attendant Delayed Timer (sec). The call will be placed fist on the extensions in the Trunk Landing Group. If the call remains unanswered till the expiry of the timer, it shall be answered by the Voice Mail Auto Attendant. This is known as **Delayed Auto Attendant** on trunks.

  For detailed instructions, see “How to configure” in “Auto Attendant”.

Configuring CLI Based Routing

To route the call to a specific extension on the basis of the CLI received:

- Select the trunk on which to want to enable CLI Based Routing.
- Configure the CLI Based Routing Table. Enter the numbers of the calling parties and the numbers of the corresponding destination extensions.
- Enable CLI Based Routing on the desired trunks according to time zones in their “Trunk Feature Template”.

  For more details, see “How to configure” in *CLI Based Routing*.

Configuring Trunk Auto Answer

- Select the trunk on which you want to enable Trunk Auto Answer.

- Enable Trunk Auto Answer in the Trunk Feature Template of the desired trunk.
- Select the **Trunk Auto Answer Greeting message**, the **Trunk Auto Answer Ring Back Tone Message**, and the **Trunk Auto Answer Busy Bye Message** for the Working Hours, Break Hours and Non-Working Hours.Create a Trunk Landing Group with the desired extensions.

- Configure the Trunk Auto Answer related **Timers**, if required. The following Timers are of relevance to the Trunk Auto Answer Feature:
  - The DID Inactivity Timer (default: 60 seconds)
  - The Ring Back Tone Timer (default: 45 seconds)
  - The Busy Tone Timer (default: 7 seconds)

  - You may change the duration of these timers from the"System Timers and Counts" page.

The Ring Back Timer and the Busy Tone Timer are also applicable for the Ring Back Tone and the Busy Tone played for internal calls.

- Record and assign Voice Modules for the following Voice Messages related to this feature:
  - Trunk Auto Answer Greeting Message
  - Trunk Auto Answer Ring Back Tone Message
  - Trunk Auto Answer Busy Bye Message

  For each of these messages, you can record four different messages.

  See the topic "Voice Message Applications" for instructions on recording and assigning voice modules to greeting messages.

  For detailed instructions, see "How to configure" in *Trunk Auto Answer.*

Configuring Direct Dial-In (DDI)

- DDI can be enabled on ISDN T1E1PRI, BRI as well on SIP Trunks. Select the trunks on which you want to enable DDI.

- Make a list of the Extensions to whom you want to assign DDI Numbers.

- Assign an Incoming (IC) Reference ID in the respective port parameters of the trunks. For instructions, see "Configuring E1 Trunks", "Configuring T1 Trunks", "Configuring BRI Trunks" and "Configuring SIP Trunks".

- Configure the DDI Routing Table and Incoming Reference Table. For instructions, see "Direct Dialing-In (DDI)".

Configuring Routing to OGTB

To route the call to an outgoing trunk, when using the gateway functionality of SARVAM UCS,

- You must make the Outgoing Trunk Bundles. See "Outgoing Trunk Bundle"

- Make an Outgoing Trunk Bundle Group by assigning these Outgoing Trunk Bundles. See "OG Trunk Bundle Group" for detailed instructions.

- Select OTBG as member in the “Routing Group”.
- Assign this Routing Group Number in the Trunk Landing Group.
- You may configure the same Trunk Landing Group for working, break and non-working hours or you different Trunk Landing Group for all three time zones.
- Assign the Trunk Landing Group to the trunk in the Trunk Feature Template for each time zone.
- For detailed instructions, see “How to configure” in Trunk Landing Group.

Outgoing Call Routing

What's this?

SARVAM UCS provides you a platform wherein you can integrate and use the services from different networks - VoIP, ISDN, GSM, CO.

The extensions users (UC/SIP Users, DKP, ISDN Terminals, SLT) of SARVAM UCS can make outgoing calls to external numbers or to internal numbers. The process of identifying the call and then routing the call through the most economical network, constitutes Outgoing Call Routing.

An outgoing call can be a call:

- to another extension number, that is a number dialed without grabbing a trunk line.
- to an external number, that is a number dialed after grabbing a trunk line.
- to a number in a closed user group, that is a number which is a part of a private network. A private network is when a few Systems are connected to each other using SIP, T1/E1 QSIG, T1/E1 PRI, E&M etc.

When an outgoing call is made by the extension user, the system simultaneously checks for:

- the features and facilities enabled for the extension user.
- the features enabled on the trunk assigned in the OG Trunk Bundle Group, for the Trunk Access Code dialed by the user.

With the default setting in the system, outgoing calls (except calls between extensions) will not be routed. Follow the steps below to ensure call routing:

- **enable the desired features in class of service assigned to the extensions. For details, see** “Class of Service (COS)”.
- **assign the desired toll control level to the extensions for making outgoing calls. For details, see** “Toll Control”.
- **assign the OGTBG to the extensions through which the calls can be routed. For details, see** “Outgoing Trunk Bundle” **and** “OG Trunk Bundle Group”.
- **by default Call Duration Control is enabled on the trunks and extension, hence the calls will be disconnected after the expiry of the timer. You can change these settings as per your requirement. For details, see** “Call Duration Control (CDC)”**.**
- **by default Call Budget is enabled on the Trunks. After the expiry of the budget outgoing calls will not be routed through the trunks. You can change these settings as per your requirement. For details, see** “Call Budget on Trunk”.

If you have purchased a new system with Firmware later than V1R6.7, the new default settings will be applied automatically. Refer to “Modified default parameter values for Firmwares later than V1R6.7”.

If you are upgrading the system, refer to “After updating Firmware later than V1R6.7” and “Modified default parameter values for Firmwares later than V1R6.7”.

Features and Facilities that can be enabled in the Station Basic Features Template assigned to the extension user:

- The **Class of Service** assigned to the extension user. Class of Service (CoS) defines the permission an extension will have on a System. It defines the set features of the System that the extension is to be allowed access to.

  Feature requirements vary among users and with time. Similarly, certain features that are required during working hours may not be required during break or non-working hours.

For details, see “Class of Service (COS)”.

- The **Call Budget** assigned to the extension user: The system keeps a tab on the total cost of phone calls made by extension users. If the budget assigned to the user has been consumed, the system will not allow the user to make further outgoing calls, but will be able to make internal calls.

  The extension user can be assigned a fresh budget, after which s/he can resume making calls.

  Call Budget can be enabled on all the extensions as well as on selected extensions. Each extension can be assigned a different amount depending on the user requirement.

  For details, see “Call Budget on Extension”.

- The **Call Privilege** assigned to the extension. In Call Privilege you can define the Toll Control Levels. Toll Control (or Toll Restriction) is an expense control feature of SARVAM UCS. It enables you to program the system so that each extension has a designated calling permission referred to as 'Call Privilege'.

  Each type Call Privilege allows the extension to call certain areas and restricts it from calling others. The extension can also be restricted from the dialing of specific telephone numbers.

  Bu default, the No Calls is assigned to all the Toll Control levels. You must configure the parameters as per your requirement to ensure calling.

  For details, see “Toll Control”.

Features enabled on the trunk:

- A Trunk Access Code is a short digit sequence dialed from an extension phone instructing the System to assign a trunk line or any trunk line from a group of trunks (“OG Trunk Bundle Group”) to the user to dial an external number. For making outgoing calls each extension user must dial a Trunk Access Code.

- The outgoing calls are routed through the OG Trunk Bundle Groups assigned to the extensions in the Station Basic Features Template. All the trunks connected to the system can be bunched in different groups called OG Trunk Bundle Groups and these OG Trunk Bundle Groups can be allotted to each extension.

  Each OG Trunk Bundle Group consists of a single OG Trunk Bundle or multiple OG Trunk Bundles. The Outgoing (OG) Trunk Bundle is set of parameters that completely define the grouping of similar channels/ trunks. Bundles of similar trunks/channels only can be formed. The system can hunt for a free trunk within the bundle as per the set option - Ascending, Descending or Cyclic. For more information, see “Outgoing Trunk Bundle”.

  With the default OGTBG assigned outgoing calls will not be possible. Refer to the instructions in “Outgoing Trunk Bundle” and “OG Trunk Bundle Group”.

  An Extension can be allotted different OG Trunk Bundle Group during different timings of the day.

- Since there are different trunk lines for making calls and the service providers of these trunks offer different tariffs for calls made to certain locations or numbers or during a particular time of the day, you can enable Least Cost Routing on the trunks.

When a call is made from an extension of the SARVAM UCS, using LCR the system selects the lowest cost trunk from among all the trunks allotted to that extension to make the outgoing call, depending upon the type of LCR configured.

For more information, see “Configuring LCR”.

- For LCR to work, all trunks that are allotted to extensions for making outgoing calls, must first be assigned a **Cost Factor**.

  Cost factor is used for grading trunks in the order of increasing cost of routing calls, from 01 to 99, where 01 signifies least cost and 99 signifies the highest cost. Thus you can grade up to 99 trunks according to the increasing cost of routing calls.

  After assigning Cost Factor to Trunks, you must configure the Type of LCR to be used on Trunks in the Outgoing Trunk Bundle Group (OGTBG) allotted to the extensions for making calls.

  If LCR is not enabled, the outgoing call will be routed through the free trunk from the OG Trunk Bundle Group.

  For more information, see “Cost Factor” in *Configuring LCR*.

- The **Call Budget** assigned to the Trunk. Each trunk can be allotted a 'budget' limit for outgoing calls. This budget limit can be programmed to be reloaded manually each time it is exceeded or at a scheduled date, either daily or at a particular date of the month.

  By default Call Budget Type - Minutes is enabled for 300 Minutes.

  For more information, see “Call Budget on Trunk”.

How it works

When the user dials a number without grabbing a trunk, that is, without dialing a Trunk Access Code, the system checks:

- The Closed User Groups table, if a match is found the call is routed through the OGTBG assigned in the table. For more information, see “Closed User Group (CUG)”.

- If no match is found, the checks for an internal extension number. For the Extension user it checks for the features enabled in the Class of Service (CoS). If Basic Features are disabled the extension user will not be able to make internal calls. If in the CoS the Basic Features are enabled and a match is found in the internal group, the call is placed on the dialed extension.

- If still the system is unable to find a match, the systems plays an error tone to the caller.

When the user dials a number after grabbing a trunk line, that is after dialing a Trunk Access Code, the system checks:

- The Station Basic Feature Template assigned to the user. In this template it checks for the type of Toll Control and Call Budget.

- The Outgoing Trunk Bundle Group assigned to the user for making outgoing calls, for the Trunk Access Code dialed.

- The features enabled on the Trunks in the OGTBG, that is Least Cost Routing and Call Budget.
    - If LCR is not enabled the system allots a free trunk from the OG Trunk Bundle Group to route the call.
    - If LCR is enabled, the system checks for the type of LCR enabled and routes the call using the cheapest free trunk from the group. If the cheapest trunk is not free, the system hunts for the second cheap trunk in group and routes the call. If none of the trunks are free, the system plays a busy tone to the user.
    - After checking LCR, for the free trunk the system checks the type of Call Budget enabled on the trunk. The calls will be routed using this trunk as long as the budget limit set for the trunk (i.e. the Amount or Minutes or the maximum number of Calls) is not crossed. As soon as the limit is crossed the Trunk is automatically disabled.

How to configure

For the extension user configure the following in the Station Basic Features Template:

- The Basic Features are enabled in the Class of Service (CoS). If you want to restrict dialing of internal calls by the user, disable the Basic Features. Select a CoS group number, for example 19 and disable the Basic Features. In the Station Basic Features Template assigned to the user, enter 19 as the CoS for each time zone.
- To restrict calling, enable Call Budget and configure the following parameters for the feature to work:
    - select the Call Budget check box to enable
    - select the Toll Control-Call Budget Consumed option
    - set the Preset Call Budget Amount
- Set the Call Privilege for the different time zones.
- For each the Trunk Access Code in each time zone, assign an Outgoing Trunk Bundle Group.

For detailed instructions to assign a CoS, Call Budget, Call Privilege and Outgoing Trunk Bundle Group to an extension user, see “Station Basic Feature Template” and “Configuring DKP Extensions”, “Configuring SLT Extensions”, “Configuring ISDN Terminals”, “Configuring SIP Extensions”.

You need to provide access of the trunks to the users for making calls, to do so, follow the steps given below:

- To create an Outgoing Trunk Bundle put similar trunk types are put together.
- Assign the Outgoing Trunk Bundles as members in an Outgoing Trunk Bundle Group. For detailed instructions, see “OG Trunk Bundle Group”.
- For the Outgoing Trunk Bundle Group, enable Least Cost Routing, if required. Select the type of LCR required and also configure the respective LCR tables. For detailed instructions, see “Configuring LCR”.
- Enter this Outgoing Trunk Bundle Group number, as per the time zone in the Station Basic Features Template assigned to the extension user.
- Determine the cost factor you want to assign each trunk. Assign the Cost Factor to each trunk type in their respective trunk parameters.

- Enable Call Budget on Trunks. Select the type of Call Budget you want to set on the trunk, in their respective trunk parameters.

  For detailed instructions on assigning the Cost Factor and Call Budget to the trunk, see “Configuring CO Trunks”, “Configuring Mobile Trunks”, “Configuring E1 Trunks”, “Configuring T1 Trunks”,“Configuring BRI Trunks”, “Configuring SIP Trunks”.

CHAPTER 15

Features and Facilities

If you have CDMA Mobile Card installed in your system, it is recommended to avoid using features that support DTMF Detection.
The features where the caller is asked to dial digits and the system has to detect it, for example, DID, Voice Mail Auto Attendant etc might not work efficiently.

Abbreviated Dialing

What's this?

Abbreviated Dialing is the use of short codes (abbreviated numbers), to dial out long-digit numbers. It is also referred to as Memory Dialing.

Abbreviated Dialing allows you to dial quickly and easily, frequently called, long-digit numbers.

This feature requires you to store the frequently called, long-digit numbers[231] and their corresponding short codes in special lists, known as 'directories'. These directories may be 'personal' or 'global'.

SARVAM UCS supports two types of Abbreviated Dialing based on the type of directory used: Personal Abbreviated Dialing and Global Abbreviated Dialing.

Abbreviated Dialing forms the basis of two other features of the SARVAM UCS: “Dialed Number Directory” and “Quick Dial”.

Personal Abbreviated Dialing

This variation of Abbreviated Dialing makes use of the Personal Directory.

Personal Directories can be programmed and assigned to groups of extensions. The use of Personal Directories is limited to the extensions to which they are assigned.

A personal directory accommodates 25 numbers. Each number may be up to 16 digits long. A personal directory has Index numbers from 001 to 025[232] against which the frequently dialed telephone numbers are stored along with their corresponding names and trunk access code ID.

231. These may be numbers of your branch offices, your clients, as also numbers of emergency services such as fire, police.
232. For ETERNITY LENX/MENX it is 0001 to 0025.

As many as 50 different personal directories, numbered from 01 to 50 can be created and assigned to SLT, DKP and SIP extensions.

With a personal directory assigned to an extension, the extension user simply dials out the Feature Access Code for Abbreviated Dialing and the Index Number at which the desired number is stored in the personal directory.

For example: personal directory number 02 is assigned to extension 2001. The number 02652630555 is stored at Index number 16 of this directory. The user of extension 2001 can call this number by simply dialing '8' (feature access code) followed by '016' (the index number).

The system will automatically dial out the number using the trunk access code ID specified for this number in the personal directory.

- *When an extension user dials an abbreviated number from the Personal Directory, the system first checks OG Trunk Bundle Group (OGTBG) and Toll Control Level (Call Privilege) of that extension and then dials out the number.*
- *Each extension can access only the personal directory assigned to it.*
- *Personal Directory can be programmed by the System Engineer, as well as extension users. Extension users can add contacts to the Personal Directory assigned to them from their extensions phones (DKP/ SLT/ISDN phone/IP Phone).*

Global Abbreviated Dialing

This variation of Abbreviated Dialing makes use of a system-wide list of numbers stored in the memory of the SARVAM UCS, known as the Global Directory.

Being a system-wide list, the Global Directory can be accessed by any extension connected to the system.

The Global Directory has the capacity to store up to 900[233] numbers of a maximum of 16 digits each. The Global Directory is divided into three parts:

- Part 1 - contains Memory Location codes 100 to 799.
- Part 2 - contains Memory Location codes 800 to 899.
- Part 3 - contains Memory Location codes 900 to 999.

The Global Directory has Memory Location codes starting from 100 to 999. The telephone numbers along with their corresponding names are stored against Memory Location codes.

Whenever extension users of the system want to use Global Abbreviated Dialing, all they needs to do is dial the feature access code ('8' or '6') and the Memory Location code at which the desired number is stored.

For example: the number 02652630566 is stored at Memory Location 102 of the Global Directory. Now, extension users of the system can call this number by simply dialing the '8' or '6' (feature access code for Abbreviated Dialing) followed by '102' (Memory Location code at which the desired number, 02652630566, is stored).

The system will dial out the number using any of the trunks in the “OG Trunk Bundle Group” assigned to it in the Memory Location.

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233. ETERNITY LENX/MENX stores up to 2900 numbers in the Global Directory; Part 1 - 100 to 2399, Part 2 - 2400 to 2699 and Part 3 - 2700 to 2999.

Extensions can use Global Abbreviated Dialing only if this feature is included in the "Class of Service (COS)" allowed to them.

- *Further, an extension can access only that part of the Global Directory which is allowed to it in the CoS. For instance, if extension 2001 is allowed Global Directory Part 1 in its CoS, the user of extension 2001 can dial out only those numbers contained in Global Directory Part 1.*
- *So, to be able to access the entire Global Directory, extensions must be assigned all three parts of the directory in their Class of Service. By default, none of the Global Directory Parts are included in the CoS of all extensions.*
- *Global Directory can be programmed by the System Engineer, and by Digital Keyphone extension users who have Global Directory Programming allowed to them in the Class of Service.*
- *While the System Engineer can program all three parts of the Global Directory, digital keyphone extension users who are allowed Global Directory Programming in their Class of Service can configure only Global Directory Part 1.*

How to configure

For both Personal and Global Abbreviated Dialing to work, the System Engineer must:

- Program the Personal Directories and the Global Directory.
- Assign Personal Directory to the desired extensions (which may be different: SLT, DKP, ISDN Terminals, IP Phones).
- Enable Global Directory Part 1/2/3 as desired in the Class of Service (CoS) group allowed to the extensions.
- Enable 'Global Directory Part 1' and 'Global Directory Programming' in the CoS group of the digital key phone extension users, who are to be allowed to program (add, delete, edit) contacts in the Global Directory Part 1 from their digital key phones.

All the above parameters can be programmed by the System Engineer using Jeeves as well as a telephone.

Preparing Numbers Lists for Personal and Global Directories

In consultation with the extension users, you may:

- Find out the number of personal directories that need to be programmed.
- Make a list of numbers frequently dialed by the extension.
- Ask the extension users the numbers they would like to be included in the personal directory of their extension.
- Make separate lists of numbers along with their corresponding names, email id, group, trunk access codes, for each personal directory. You may draw five-column tables on paper and enter the Numbers and corresponding names, email ids, groups and trunk access codes against each Index number. For example:

Personal Directory 01

Index No.	Number	Name	Email ID (optional)	Group (optional)	TAC ID
01					
02					
:	:	:			:
:	:	:			:
25					

Personal Directory 02

Index No.	Number	Name	Email ID	Group (optional)	TAC ID (optional)
01					
02					
:	:	:			:
:	:	:			:
25					

- Compile the numbers to be included in the global directory. Numbers that are commonly dialed by all extensions can be included in the global directory.
- Draw a five-column table on paper and enter the telephone numbers along with their names, the Outgoing Trunk Bundle Group (OGTBG), email id and group at each Memory location. For example:

Global Directory

Memory Location	OGTBG	Number	Name	Email ID (optional)	Group (optional)
100					
101					
:	:	:	:		
:	:	:	:		
999					

- Prepare the Global Directory keeping in mind that it is divided into three parts: Part 1 (100 to 799), Part 2 (800 to 899), and Part 3 (900 to 999).

Uploading Personal and Global Directory Contacts

If you have personal and global directory contacts database in an excel sheet, you can convert the same into a CSV file and upload it through Jeeves.

For Personal directory contacts, the format of the CSV file must be as follow: For example 21, 9867985489, Sean Gilbert, Sean@hotmail.com, 1, 0

Where,
21 is the Index Number at which you want the entry to be stored (mandatory)
9867985489 is the Contact Number (mandatory)
Sean Gilbert is the Contact Name (optional)
Sean@hotmail.com is the Email ID (optional)
1 is the Group Index number (optional), see “SMS/Email Group”
0 is the Trunk Access Code (TAC) to route the call (optional)

To upload Personal Contacts CSV file using Jeeves, see “Upload Personal Directory CSV files” .

For Global directory contacts, the format of the CSV file must be as follow: For example 121, 9867985489, Sean Gilbert, Sean@hotmail.com, 1, 01, 000

Where,
121 is the Index Number at which you want the entry to be stored (mandatory)
9867985489 is the Contact Number (mandatory)
Sean Gilbert is the Contact Name (optional)
Sean@hormail.com is the Email ID (optional)
1 is the Group Index number (optional), see “SMS/Email Group”
01 is the Outgoing Trunk Bundle Group (OTBG) to route the call (optional)
000 is the Alternate Number Group (optional)

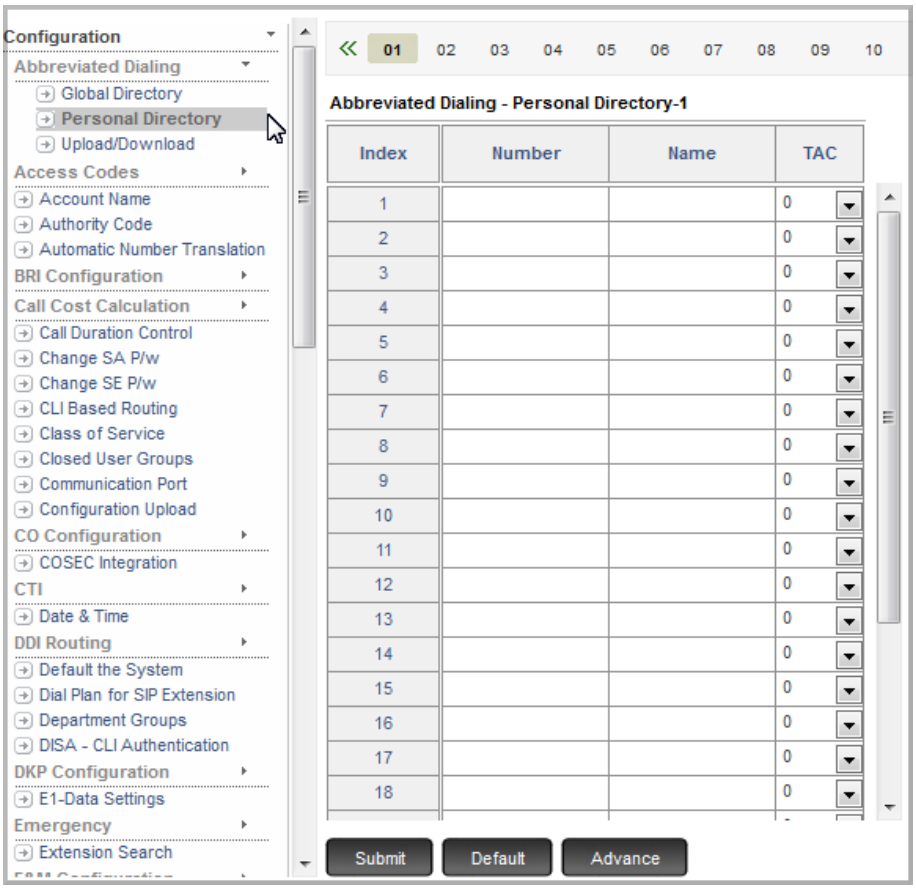
To upload Global Contacts CSV file using Jeeves, see “Upload Global Directory CSV file”.

Update Personal Directories via Email

You can update the Personal Directory contacts via Email, if you have enabled the SMS Server Application. You can Add, View, Edit and Delete contact/s via Email. For more details, see “Configuring Personal Directory via Email”.

Configuring Personal Directory using Jeeves

- Log into Jeeves as System Engineer.
- Under **Configuration**, click **Abbreviated Dialing**.
- Click **Personal Directory** to open the page.



You can configure upto 50 Personal Directories. Select the directory number by clicking the required number tab above the table.

For each directory configure the following parameters,

- Enter the **Number** you wish to store against an Index Number. and enter the contact's **Name** against the number.

  The length of the Number field is limited to 16 digits. The length of the Name field is limited to 18 alphanumeric characters. All ASCII characters except < > and " (double quote) are allowed. Ensure that the number and the name are programmed within this limit.

The Name can be configured upto 18 characters using Jeeves and CSV file only.

From the Extended Phones — SPARSH VP210 and SPARSH VP510, you will be able to view 18 characters Name configured from the Server as well as you will be able to add or edit Name of contact to a maximum of 18 characters.

From all other Extended Phones, you will be able to view 18 characters Name configured from the Server. But from these Phones you will be able to add or edit Name of contact to a maximum of 12 characters only.

Each directory has a limit of 25 entries. You may enter up to 25 Numbers and Names in each Personal Directory.

Change the **TAC** Index (TAC ID), if required.

- Click **Advance** to configure the **Email ID** and select the **Group** for the contact.
- Enter the **Email ID** of the contact you wish to store. The Email ID can be a maximum of 64 characters.
- You can assign the contact to a Group. Select the desired **Group** Type from the list. The system clubs together contacts assigned the same Group. Default: None. For details, see “SMS/Email Group”.
- Click **Submit** to save the entries.
- Repeat these steps to program each Personal Directory.
- Click **Submit** at the bottom of the page to save your entries.

Keep a print of each personal directory for your record and for the record of the extension user to whose phone the personal directory is assigned. This will also help you take care of overlaps and include some of the numbers that are dialed by all users in the Global Directory instead of the Personal Directory.

Assigning Personal Directories to Extensions

To assign Personal Directory to SLT extensions,

- Log into Jeeves as System Engineer.
- Under **Configuration**, click **SLT Configuration**.
- Click **SLT Parameters** to open the page.
- Click the **Advance** button.Go to **Personal Directory** column of the SLT to which the directory is to be assigned.
- Enter the number of the Personal Directory. For example, to assign Personal Directory No. 02 to SLT 2001 (software port 001, connected on hardware slot 03, hardware port 09), enter '02' in the 'Personal Directory' column for SLT 2001.
- Similarly, enter the Personal Directory number to be assigned to each SLT.
- Click **Submit** at the bottom of the page to save changes.

To assign Personal Directory to DKP extensions,

- Under **Configuration**, click **DKP Configuration**.
- Click **DKP Parameters** to open the page.
- Click the **Advance** button. Go to **Personal Directory** column of the DKP to which the directory is to be assigned.
- Enter the number of the Personal Directory. For example, to assign Personal Directory No. 01 to DKP 3001 (software port 001, connected on hardware slot 17, hardware port 01), enter '01' in the 'Personal Directory' column for DKP 3001.
- Similarly, enter the Personal Directory number to be assigned to each DKP.

- Click **Submit** at the bottom of the page to save changes.

To assign Personal Directory to ISDN Terminal extensions,

- Under **Configuration**, click **ISDN Terminal Parameters** to open the page.
- Click the **Advance** button. Go to **Personal Directory** column of the ISDN Terminal to which the directory is to be assigned.
- For each ISDN Terminal that is to be assigned a Personal Directory, enter the number of the directory in this column.
- Click **Submit** at the bottom of the page to save changes.

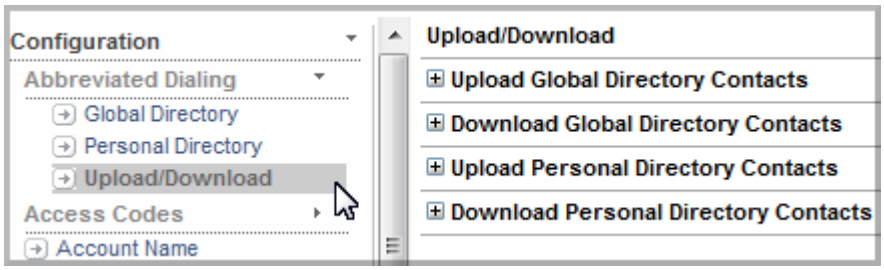
To assign Personal Directory to SIP Extensions extensions,

- Under **Configuration**, click **VoIP Configuration**.
- Click **SIP Extension Settings** to open the page.
- Select the software port number of the SIP extension you want to assign the Personal Directory.
- Click the **Advance** button.
- Scroll to Personal Directory, and select the **Personal Directory** number you want to assign to this extension.
- Click **Submit** to save.

It is possible to assign the same personal directory to multiple extensions.

Upload Personal Directory CSV files

- Log into Jeeves as System Engineer.
- Under **Configuration**, click **Abbreviated Dialing**.
- Click **Upload/Download** to open the page.

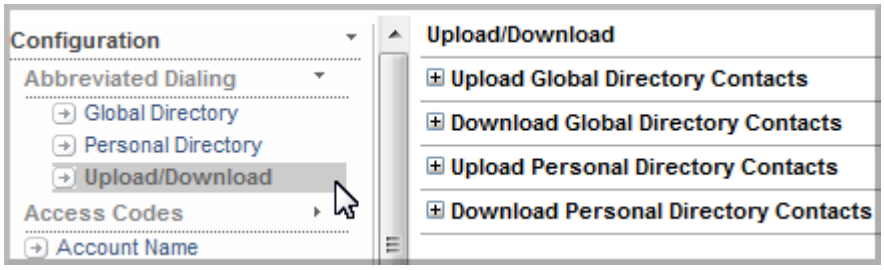


- Click **Upload Personal Directory Contacts** to expand.

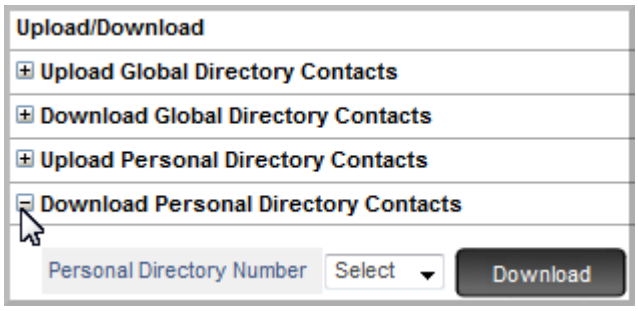
  - In **Personal Directory Number**, select the number of the personal directory in which you want to upload the contacts from the CSV file.
  - Select the **Clear all other indices of the selected Personal Directory, which are not specified in the .csv file being uploaded** check box, to overwrite the existing contacts in the Personal directory with the contacts of CSV file. Default: Disabled.
  - Click the **Browse** button to **Select the .csv file to be uploaded** from the location on the local disk.
  - Click the **Upload** button.
  - All the contacts of the CSV file will be uploaded in the selected Personal Directory. To view, click the respective Personal Directory link.

Download Personal Directory CSV files

- Log into Jeeves as System Engineer.
- Under **Configuration**, click **Abbreviated Dialing**.
- Click **Upload/Download** to open the page.

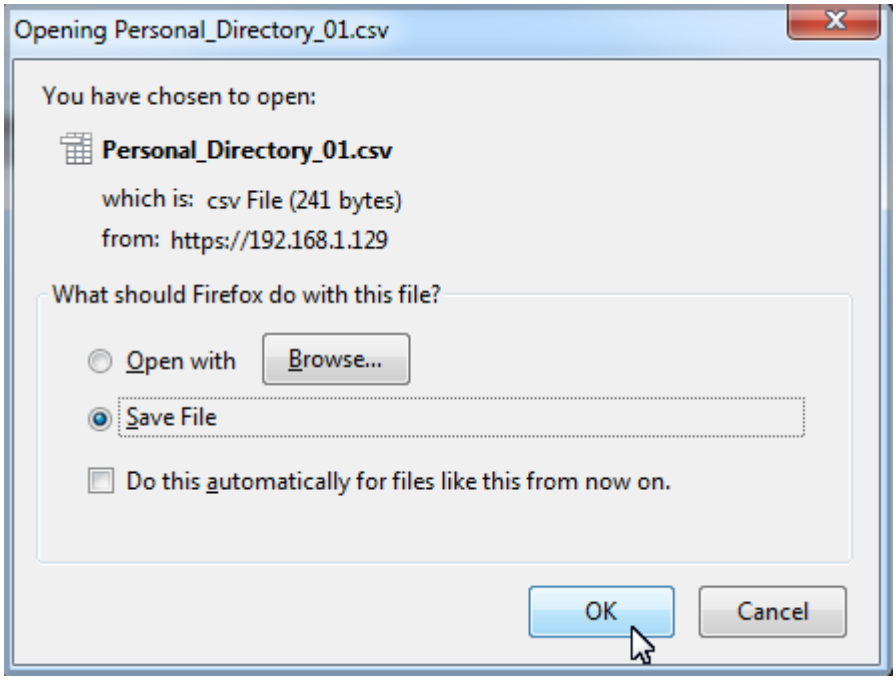


- Click **Download Personal Directory Contacts** to expand.



  - In **Personal Directory Number**, select the number of the personal directory from which you want to download the contacts.

  - Click the **Download** button.



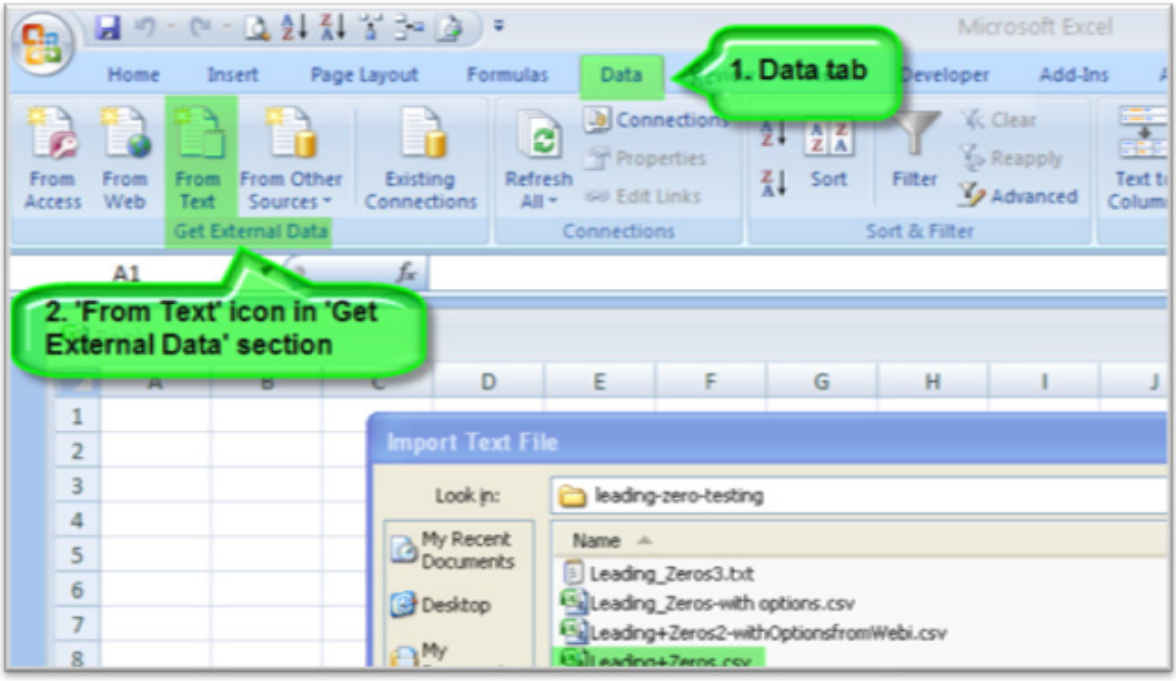
  - You will get a prompt with an option to open the **Opening Personal_Directory_01.csv** file or save the file to a location. Save the file on the local disk.

  - To Open the **Personal_Directory_01.csv** file from the location on the local disk, follow the steps given below:

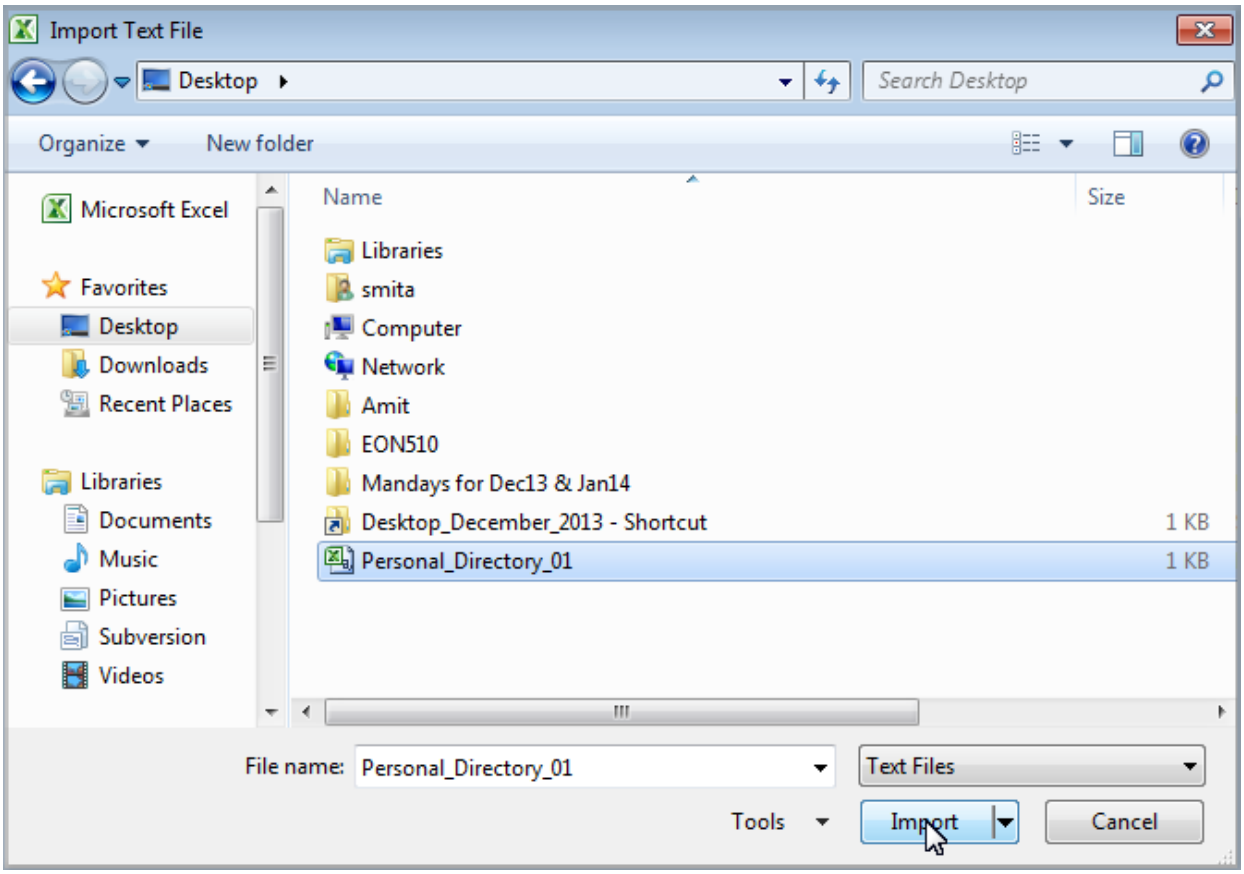
    - **DO NOT OPEN THE CSV FILE DIRECTLY WITH EXCEL**!

    - Open a **New** worksheet in Excel.

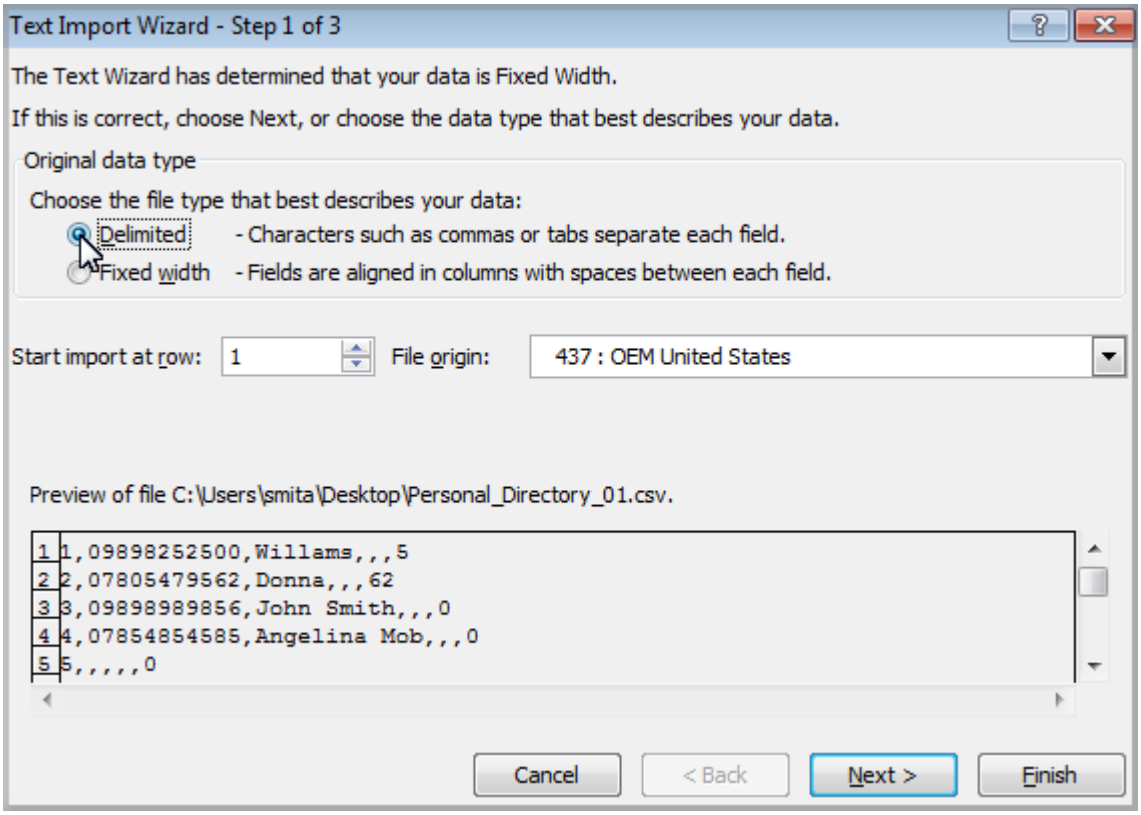
- Open the **Data** tab and select **From text button in the Get External Data**.



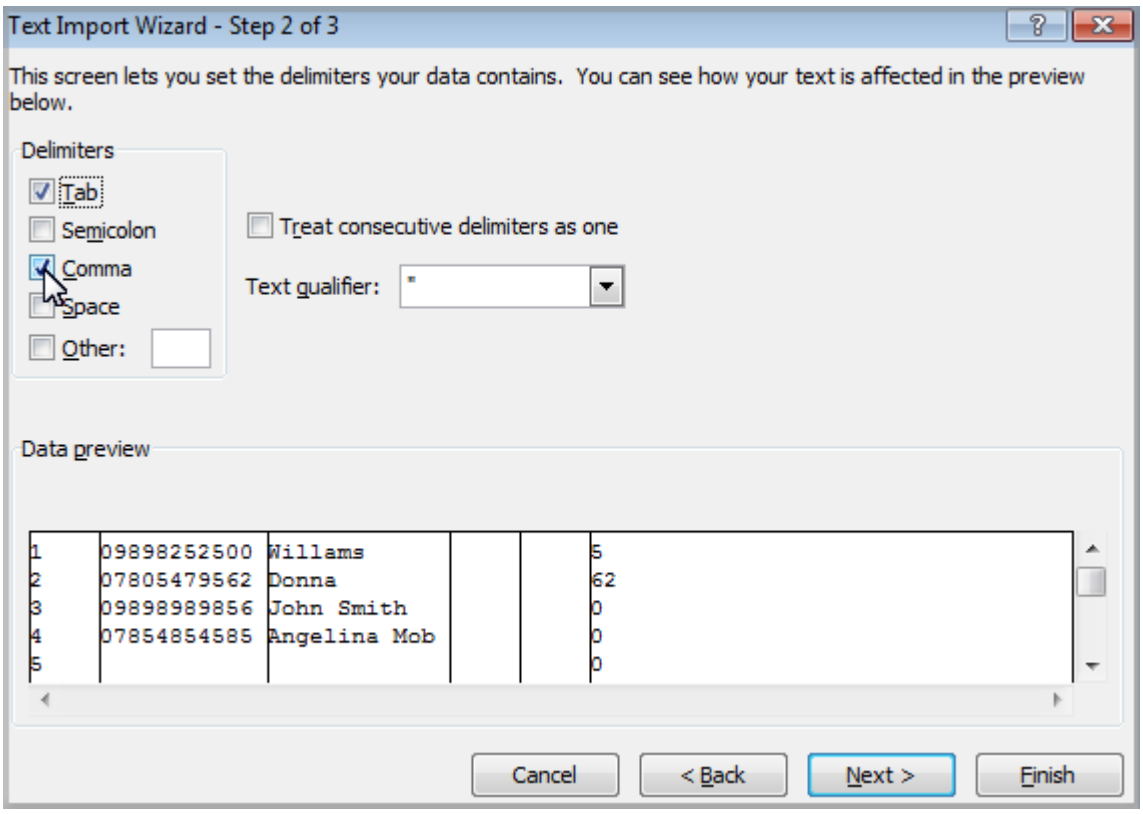
- Select your CSV file from the location on the local disk to import.



- In **Original data type** section, select **Delimited** radio button and click **Next**.

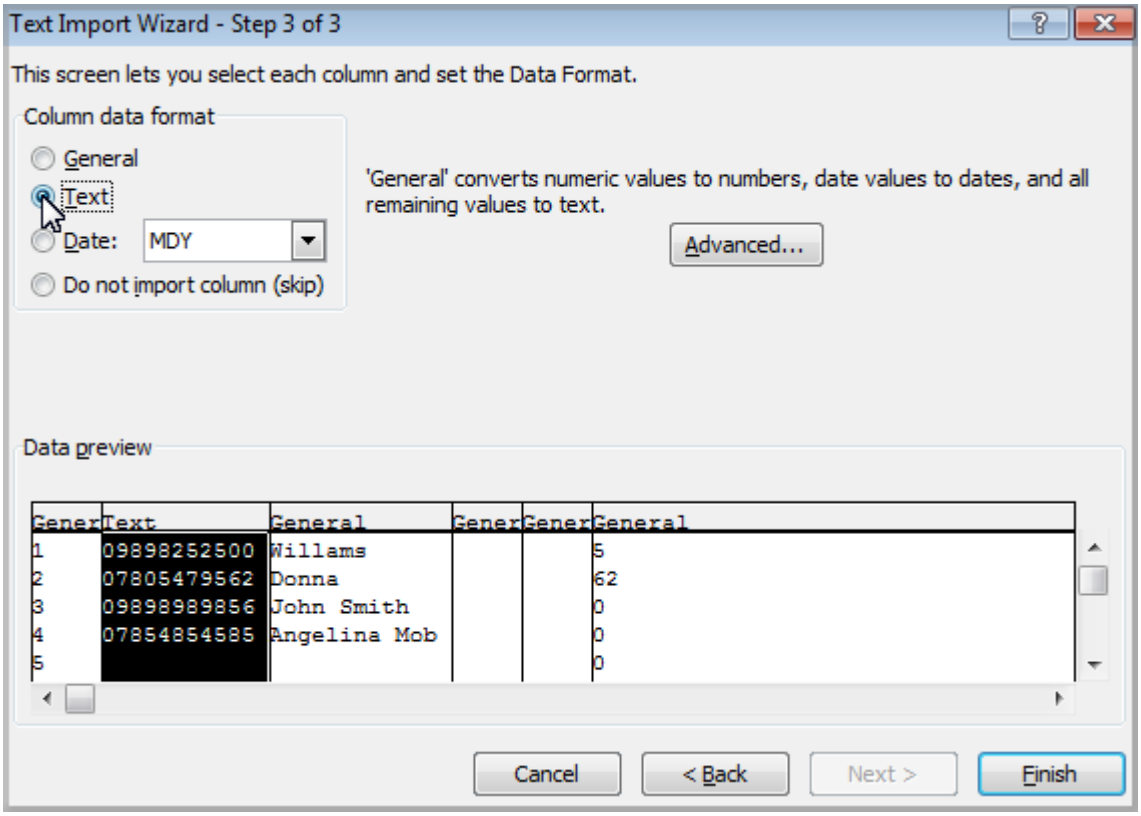


- In **Delimiters** select the **Comma** check box (column dividers will appear in preview) and click **Nex**t.



- Select the column with leading zeros and in **Column data format** select the **Text** radio button.

  You will have to do this for each column where the data contains leading zeros.



- Click **Finish**.

- The leading zeros will still be there in the new worksheet with the imported data.

Configuring Personal Directory via Email

You can configure the Personal Directory via Email, if you have enabled the SMS Server Application. To know more, see “SMS Server”.

If you have enabled SMS Server Application, you can configure contacts in the Personal Directory via Email. The SMS Server Application of SARVAM UCS allows you to Add, View, Edit and Delete contact/s via Email.

When the SMS Server receives an Email for configuring the Personal Directory, the system searches for the Email ID in the system users lists.

If a match is not found, the system sends a reply mail to the sender, with the Subject: *SMS Server Personal Directory Configuration*' and the message in the body: *Not a valid user!*.

If a match is found, the system checks for the Personal Directory assigned to the user.

If a Personal Directory is not assigned, for the extension user then the system sends a reply mail to the sender with the subject: *SMS Server Personal Directory Configuration* and the message in the body: *Personal Directory is not assigned, can't add contact Name: XXXXX Number: XXXXX*. Where name/number is as received in mail for directory configuration.

If a match is found for the user and the user is assigned a Personal Directory, the system accepts and takes the necessary action as requested by the user. The request may be for adding, viewing, deleting or editing a contact.

The following validations are applicable when you want to configure contacts via Email:
The **Number** can be a maximum of 16 digits.
The **Name** can be a maximum of 12 characters.
The **Email ID** can be a maximum of 64 characters.
The **Group** can be a maximum of 16 characters.

While configuring Personal Directory via Email, make sure you have configured the Type of Group in the "SMS/Email Group" topic under SMS Server.

Adding a contact

You can add a single/ multiple contacts via Email. To add a contact you must send an Email to the SMS Server in a specific format. Given below are the various examples and formats of adding a contact.

To add a single contact

Given below is the format of the mail to be sent to the SMS Server, to add a single contact.

To: smssever@domain.com (this is the Email ID of the SMS Server)
Subject: James Smith=+919898985400 (The name and number of the contact you wish to add)

Here,
The To field contains the Email ID of the SMS Server.
The Subject contains the details of the contacts.
The name of the contact is James Smith
The number of the contact is +919898985400

The contact is added at a first free index, between 0001 to 0025 in the Personal Directory and a confirmation mail is sent to the sender, with the Subject: *SMS Server Personal Directory Configuration* and the message in the body: *New entry is added successfully at index-XXXX*.

To add multiple contacts

To add multiple contacts via single mail, you must enter the contacts by separating them with a comma(,). Given below is the format of the mail.

To: smssever@domain.com
Subject: James Smith=+919898985400, Steve D=898954045

Here,
The To field contains the Email ID of the SMS Server.
The Subject contains the details of the contacts.
The names to be added are James Smith and Steve D and their numbers are +919898985400 and 898954045 respectively.

- *Make sure you do not enter any space before and after "=".*

The contact is added at a first free index, between 0001 to 0025 in the Personal Directory and a confirmation mail is sent to the sender, with the Subject: *SMS Server Personal Directory Configuration* and the message in the body: New entry is successfully added at index-XXXX, Name: XXXXX, Number: XXXXX. Where XXXXX=is the information received in the mail for configuration. A separate mail is sent for each contact that is added.

View details of existing contact

You can view the numbers saved against a Name or Group in the Personal Directory. Given below is the format of the mail sent to the SMS Server.

Given below is the format of the mail to be sent to the SMS Server, to get the details of the existing contacts.

To: smssever@domain.com
Subject: HDFC Customer Care? (this is the name/group)

Here,
The To field contains the Email ID of the SMS Server.
The Subject contains the Name/Group Name whose details are required by you.
The Group Name is HDFC Customer Care.

The server will send a return mail with the number(s) as given below:
HDFC Customer Care:
James: +919898985400
John: +919897894512
Steve: +91898954045

If multiple entries exist, the return mail will contain the details of all. For example, if the query is made for the name James, that is Subject: James?. If the system finds two entries (James D and James S) then reply email contains both entries.

James D: 9426712345
James S: 9426921345

Similarly, if you want the details of all the names/groups beginning with **J**, then return mail will contain all names starting with character "J" and each name will be displayed in a separate row, followed by their numbers.

James D: 9426712345
James S: 9426921345
John S :9996565123

If no match is found for the Name/Group Name, the system sends a reply mail with the Subject: *SMS Server Error Cause* and the message in the body: *X not found in Personal Directory*, where X is the actual name.

Deleting a Contact

To delete an entry from the Personal Directory, the mail to be sent to the SMS Server must be in the format as given below.

To: smssever@domain.com
Subject: HDFC Customer Care=
Here,
The To field contains the Email ID of the SMS Server.
The Subject contains the Name/Group Name that you want to delete.

In this case, there are three contacts +919898985400 (James), +919897894512 (John), +91898954045 (Steve) that will be deleted.

If the system is able to delete the entry, a reply mail with Subject: *SMS Server Personal Directory Configuration* and the message in the body: *X is deleted from Personal Directory'*, where X is the particular name which is received for deletion of the contact/group name. In this case, the reply Email will be *HDFC Customer Care is deleted from Personal Directory*.

If the SMS Server received a delete request for a contact and the same is not configured in the Personal Directory, then a reply mail is sent to the sender, with the Subject: SMS Server Error Cause and the message in the body: *X not found in Personal Directory*. Where, X is the actual name.

Modify/Edit a Contact

You cannot edit any contact in the Personal Directory via Email.

To modify any contacts details, you must first delete the existing contact from the Personal Directory. Then you can add the same contact with new contact details.

Configuring Personal Directory using a Telephone

- Enter SE mode.

  To program a telephone number in a personal directory, dial:

  - ***1902-Personal Directory-Location Code-Number***
    Where,
    Personal Directory is from 01 to 50.
    Location Code is from 01 to 25.
    Number is the telephone number, max. 16 digits. If the number has fewer than 16 digits, you must dial #* to terminate the command.

  To clear a telephone number from a location in a personal directory, dial:
  - ***1902-Personal Directory-Location Code-#****

  To program a name in the personal directory, dial:
  - ***1903-Personal Directory-Location Code-Name***
    Where,
    Personal Directory is from 01 to 50.
    Location Code is from 01 to 25.
    Name is a string of alphanumeric characters, max. 12 characters. If the name has fewer than 12 characters, you must dial #* to terminate the command.

  To clear a name from a location in a personal directory, dial:
  - ***1903-Personal Directory-Location Code-#****

  To program the TAC Index for personal directory, dial:
  - ***1904-Personal Directory-Location Code-TAC Index***
    Where,
    PM Group is from 01 to 50.
    Location Code is from 01 to 25.
    TAC Index is from 1 to 6.
    By default, the TAC Index 1 is assigned to Personal Directory.

- To assign a Personal Directory to an extension, dial the following commands:

  If the extension is an SLT, dial:
  - ***1905-1-SLT-Personal Directory*** to apply personal directory on a single extension.
  - ***1905-2-SLT-SLT-Personal Directory*** to apply the same personal directory on a range of extensions.
  - ***1905-*-Personal Directory*** to apply the same personal directory on all extensions.
    Where,
    SLT is the Software Port Number of SLT from 001 to 512.
    Personal Directory is from 01 to 50.
    By default, 00 is the Personal Directory assigned to all SLTs.

  To clear the personal directory assigned to the SLT, dial:
  - ***1905-1-SLT-00*** to clear personal directory from a single SLT.
  - ***1905-2-SLT-SLT-00*** to clear personal directory from a range of SLTs.
  - ***1905-*-00*** to clear personal directory from all SLTs.

  If the extension is a DKP, dial:
  - ***1906-1-DKP-Personal Directory*** to assign personal directory to a single DKP.
  - ***1906-2-DKP-DKP-Personal Directory*** to assign the same personal directory to a range of DKPs.
  - ***1906-*- Personal Directory*** to assign the same personal directory to all DKPs.
    Where,
    DKP is the Software Port Number of the DKP from 001 to 128.
    Personal Directory is from 01 to 50.
    By default, 00 is the Personal Directory assigned to all DKPs.

  To clear the personal directory assigned to the DKP, dial:
  - ***1906-1-DKP-00*** to clear personal directory from a single DKP.
  - ***1906-2-DKP-DKP-00*** to clear personal directory from a range of DKPs.
  - ***1906-*-00*** to clear personal directory from all DKPs.

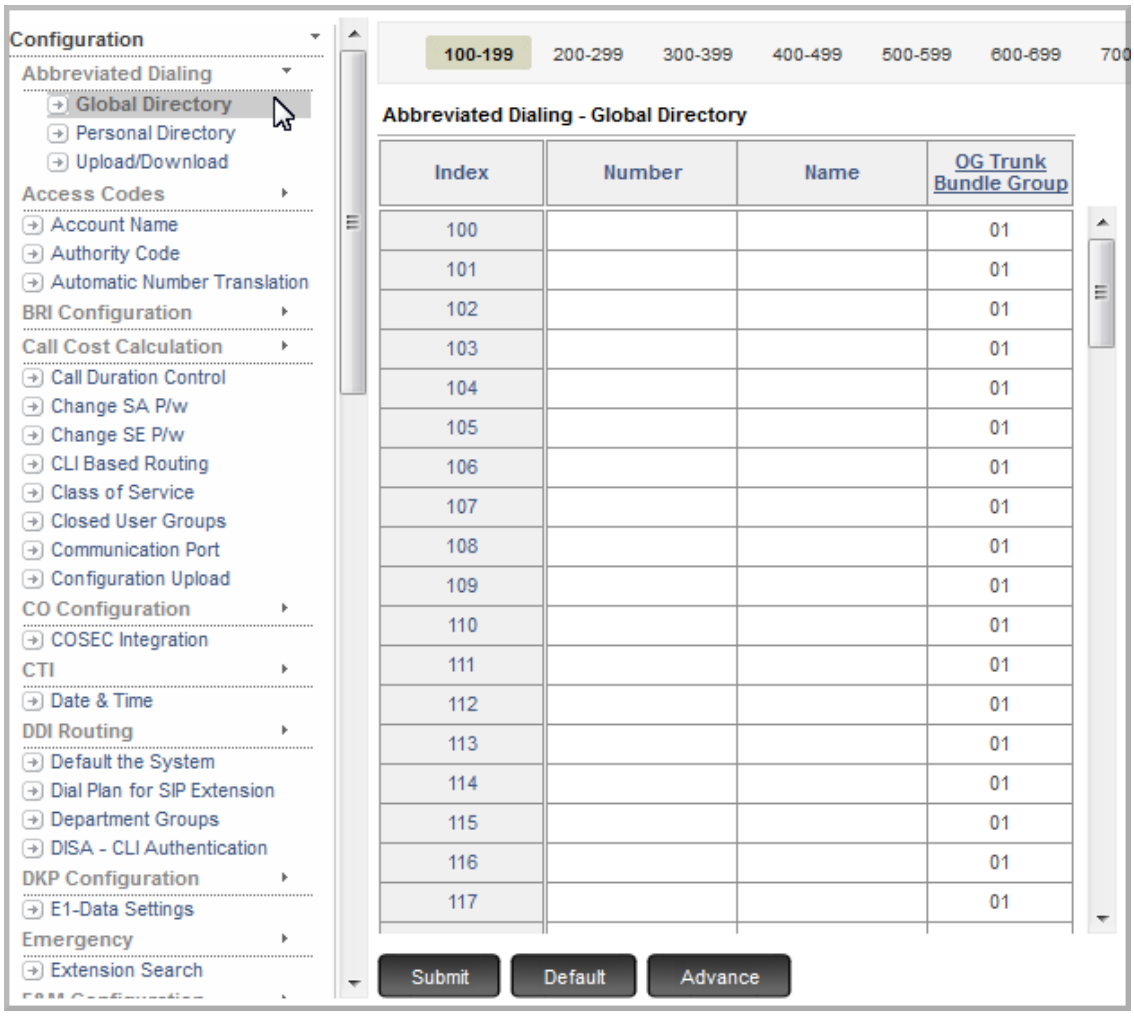
  If the extension is an ISDN Terminal, dial:
  - ***1907-1-ISDN-Personal Directory*** to assign personal directory to a single ISDN Terminal.
  - ***1907-2-ISDN-ISDN-Personal Directory*** to assign the same personal directory to a range of ISDN Terminals.
  - ***1907-*-Personal Directory*** to assign the same personal directory to all ISDN Terminals.
    Where,
    ISDN is the Software Port Number of the ISDN Terminal from 01 to 64.
    Personal Directory is from 01 to 50.
    By default, 00 is the Personal Directory assigned to all ISDN Terminals.

  To clear the personal directory assigned to an ISDN Terminal, dial:
  - ***1907-1-ISDN-00*** to clear personal directory of a single ISDN Terminal.
  - ***1907-2-ISDN-ISDN-00*** to clear personal directory of a range of ISDN Terminals.
  - ***1907-*-00*** to clear personal directory of all ISDN Terminals.
    Where,
    ISDN is the Software Port Number of the ISDN Terminal from 01 to 64.
    Personal Directory is from 01 to 50.
    By default, 00 is the Personal Directory assigned to all ISDN Terminals.

- Exit SE mode.

Configuring Global Directory using Jeeves

- Log into Jeeves as System Engineer.
- Under **Configuration**, click **Abbreviated Dialing**.
- Click **Global Directory** to open the page.



Each page of has 100 entries. To go to the next 100 entries click the links above the table '200-299'

- Enter the **Number** you wish to store against a Memory Location Code. Enter the contact's **Name** against the number.

  The length of the Number field is limited to 16 digits. The length of the Name field is limited to 18 alphanumeric characters. All ASCII characters except < > and “ (double quote) are allowed. Ensure that the number and the name are programmed within this limit.

The Name can be configured upto 18 characters using Jeeves and CSV file only.

From the Extended Phones — SPARSH VP210 and SPARSH VP510, you will be able to view 18 characters Name configured from the Server as well as you will be able to add or edit Name of contact to a maximum of 18 characters.

From all other Extended Phones, you will be able to view 18 characters Name configured from the Server. But from these Phones you will be able to add or edit Name of contact to a maximum of 12 characters only.

- Change the **OG Trunk Bundle Group,** if required. Default: 01. See “OG Trunk Bundle Group”.

*While routing the Global Directory number, the system will use the OGTBG configured in the Station Advanced Feature Template assigned to the Extension that is dialing a Global Directory number. So, if you want the system to route Global Directory numbers using OGTBG you configured here, make sure you have selected **OGTBG configured in the Global Dir.** in the **Route Global Directory Calls using** parameter of the Station Advanced Feature Template.*

- Click **Advance** to configure the **Email ID** and select the **Group** for the contact.
- Enter the **Email ID** of the contact you wish to store. The Email ID can be a maximum of 64 characters.
- You can assign the contact to a Group. Select the desired **Group** Type from the list. The system clubs together contacts assigned the same Group. Default: None. For details, see “SMS/Email Group”.
- Click **Submit** at the bottom of the page to save your entries.

Applying Global Directory to Extensions

To apply the Global Directory to extensions,

- Make sure that the feature Global Directory (desired Part1/2/3) is enabled in the CoS of the extensions to which you are assigning the Global Directory.

  If the entire directory is to be assigned to all extensions, you may simply enable Global Directory Part 1 Global Directory Part 2 and Part 3 in the default CoS group 01 in the default Station Basic Feature Template 01 assigned to the extensions.

  However, if selected extensions are to be allowed Global Directory Part 1/2/ 3, follow these steps:

  - Define a CoS group with Global Directory Part 1/Part 2/Part 3 enabled.
  - Prepare a Station Basic Feature Template with this CoS group applicable in all the “Time Zones”.
  - Assign this template to the extensions to which Global Directory Part1/Part2 /Part 3 are to be allowed.

  Refer the topics “Class of Service (COS)” and “Station Basic Feature Template” for further instructions.

- Decide which of the DKP Extension users are to be allowed ‘Global Directory Programming’ (of Global Directory Part 1) and allow this feature in their Class of Service.

  By default, Global Directory Programming is disabled in the default CoS group 01 in the default Station Basic Feature Template 01 assigned to all extensions of the system. This means none of the extensions can program Global Directory.

If you want to allow Global Directory Programming to all DKP extension users, simply enable this feature in the CoS group of the Station Basic Feature Template assigned to them.

If you want to allow Global Directory Programming to only selected extensions, then follow these steps:

- Define a CoS group with Global Directory Programming enabled.
- Make sure this CoS also has Global Directory Part 1 enabled.
- Prepare a Station Basic Feature Template with this CoS group applicable in all the "Time Zones".
- Assign this template to the DKP extensions to which Global Directory Programming is to be allowed.

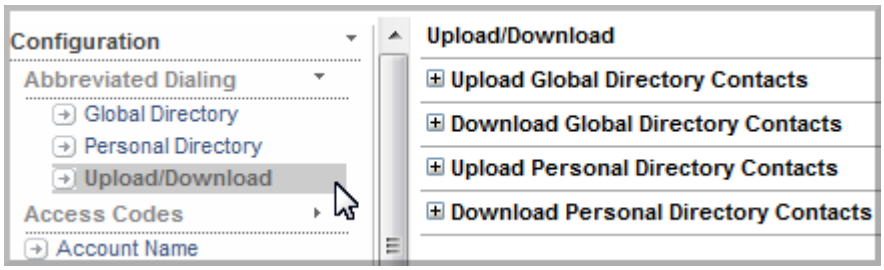
Refer the topics "Class of Service (COS)" and "Station Basic Feature Template" for programming instructions.

- *If Global Directory Part 1, 2 or 3 is assigned to an extension user, the system will not check for Toll Control.*
- *When you assign Global Directory Programming to a DKP extension user, the user can program any number in Global Directory Part 1, this includes numbers denied to the extension user in the Call Privilege defined in the Toll Control level of this extension user.*
- *Since the system does not check for Toll Control for numbers dialed out from Global Directory Part 1, there is a possibility of extension users programming numbers not allowed to them in their Toll Control level in the Global Directory Part 1, inadvertently or intentionally.*
- *Hence, the System Engineer is advised to exercise caution when allowing this feature to DKP extension users.*

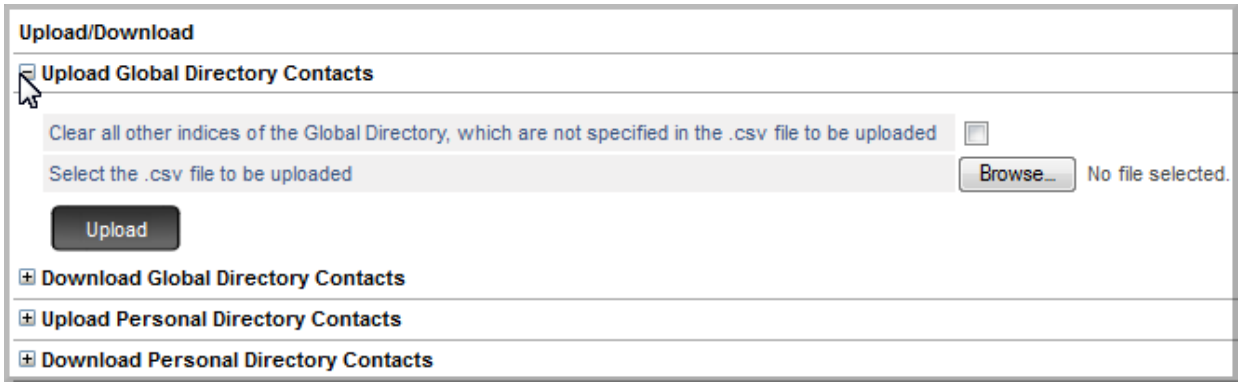
- Click **Submit** at the bottom of the pages on which you make changes to save your settings.
- If you have finished configuration, you may log out of Jeeves. Or you may continue, as required.

Upload Global Directory CSV file

- Log into Jeeves as System Engineer.
- Under **Configuration**, click **Abbreviated Dialing**.
- Click **Upload/Download** to open the page.



- Click **Upload Global Directory Contacts** to expand.

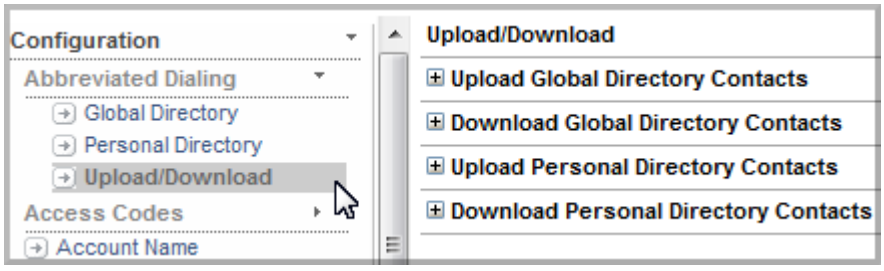


  - Select the **Clear all other indices of the Global Directory, which are not specified in the .csv file being uploaded** check box, to overwrite the existing contacts in the Global directory with the contacts of CSV file. Default: Disabled.
  - Click the **Browse** button to **Select the .csv file to be uploaded** from the location on the local disk.
  - Click the **Upload** button.
  - All the contacts of the CSV file will be uploaded in the Global Directory. To view, click the Global Directory link.

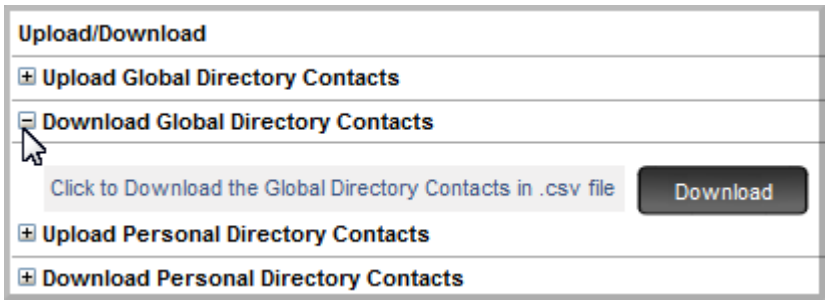
  - *If all the Global Directories are selected for LDAP, then contacts will not be uploaded through .csv file.*
  - *The contacts will be uploaded through .csv file only in the Global Directory that is not synchronized for LDAP.*

Download Global Directory CSV files

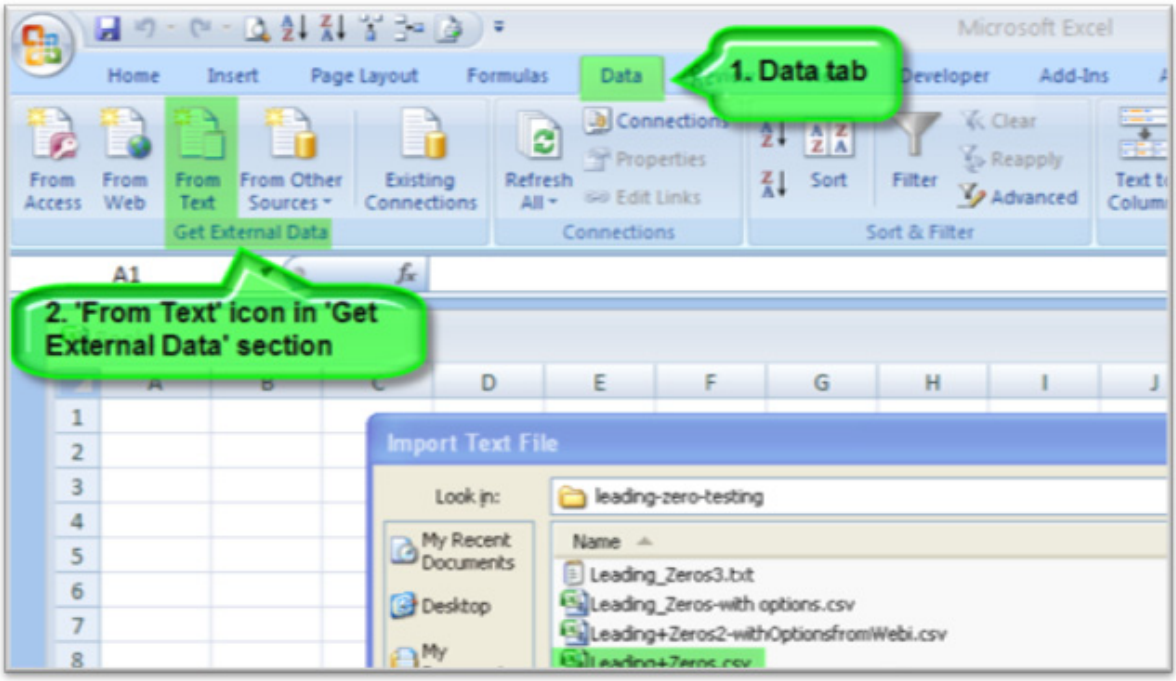
- Log into Jeeves as System Engineer.
- Under **Configuration**, click **Abbreviated Dialing**.
- Click **Upload/Download** to open the page.



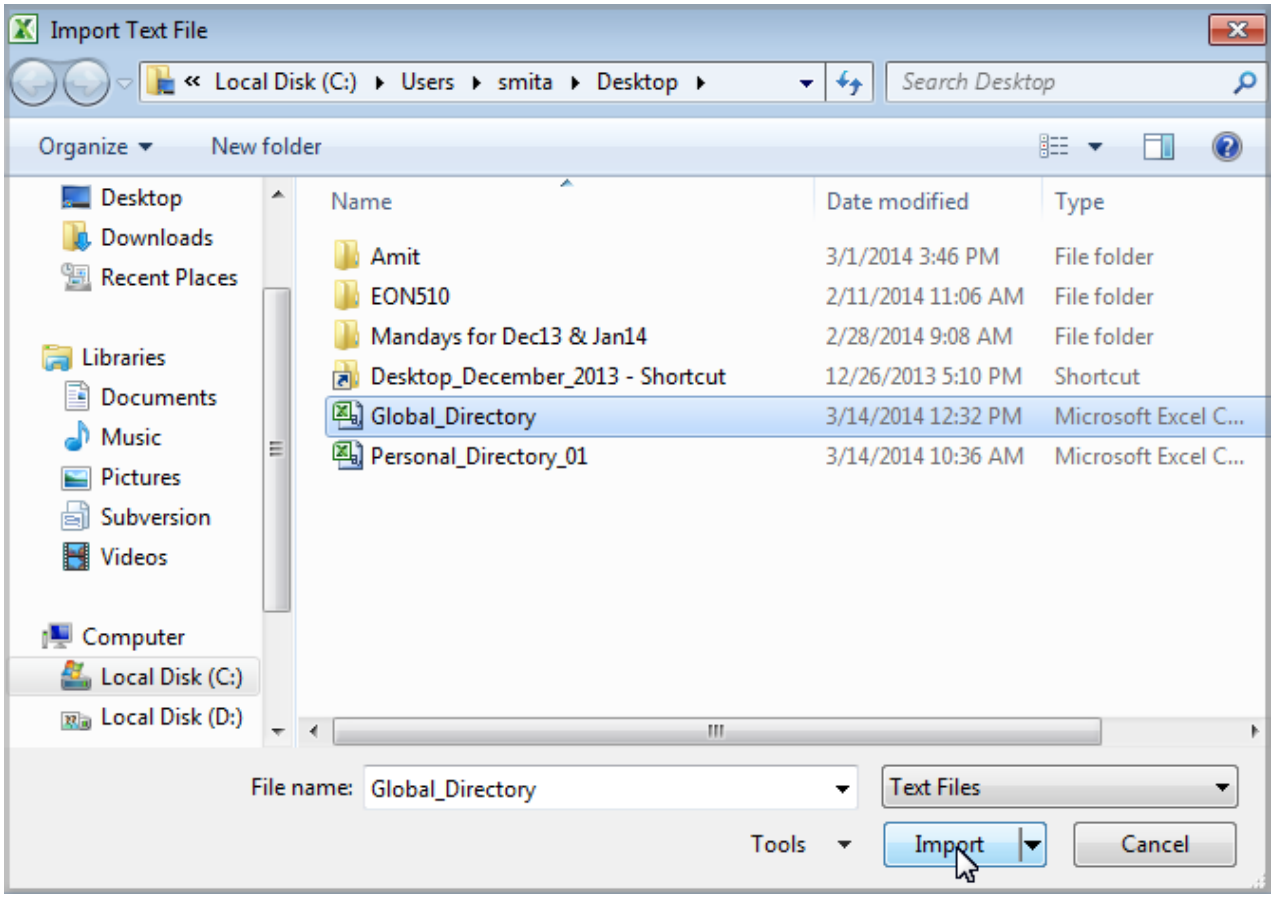
- Click **Download Global Directory Contacts** to expand.



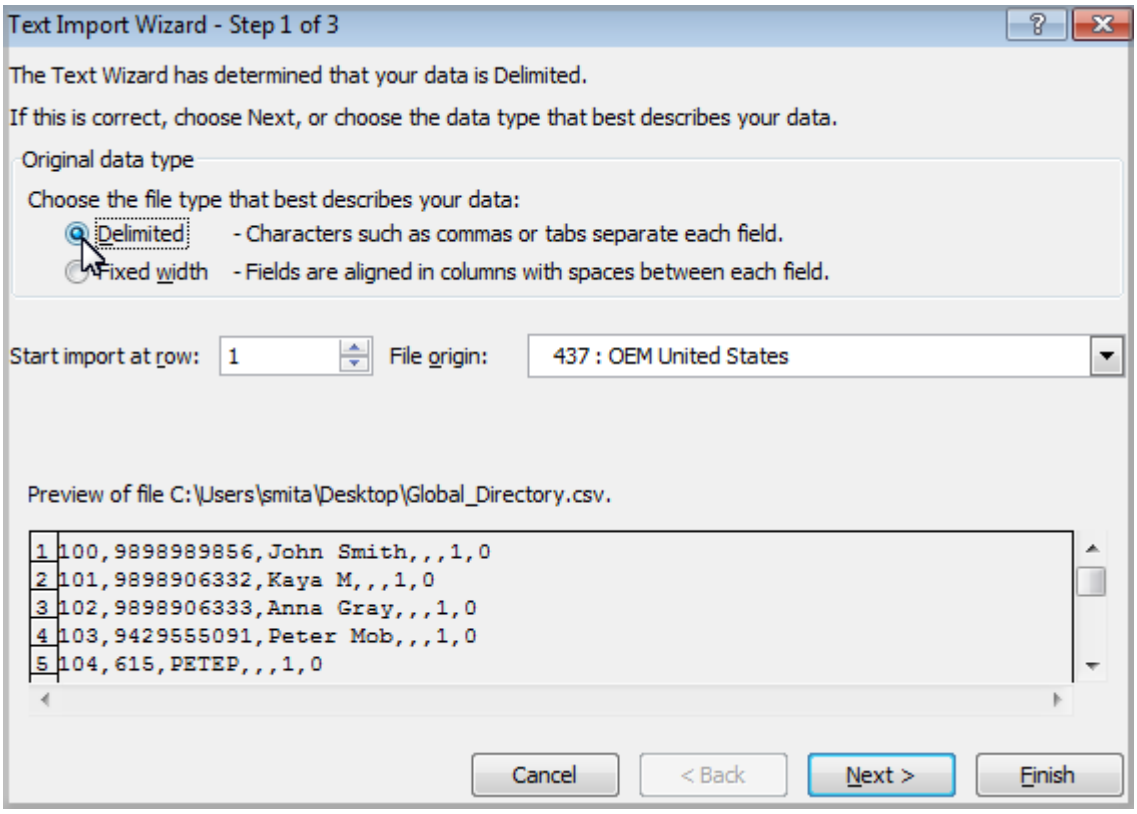
- Click the **Download** button.
- You will get a prompt with an option to open the **Opening Global_Directory.csv** file or save the file to a location. Save the file on the local disk.
- To Open the **Global_Directory.csv** file from the location on the local disk, make sure you follow the steps given below:
  - **DO NOT OPEN THE CSV FILE DIRECTLY WITH EXCEL**!
  - Open a **New** worksheet in Excel.
  - Open the **Data** tab and select **From text button in the Get External Data**.



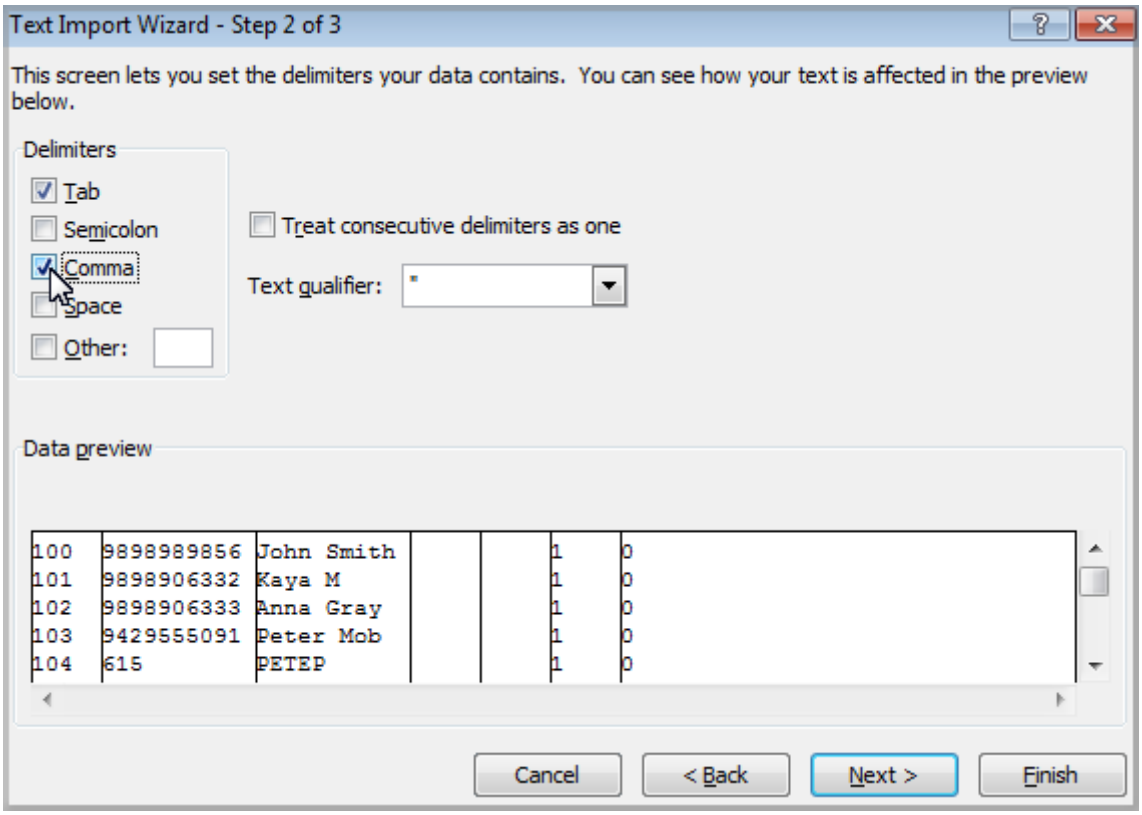
- Select your CSV file from the location on the local disk to import.



- In **Original data type** section, select **Delimited** radio button and click **Next**.

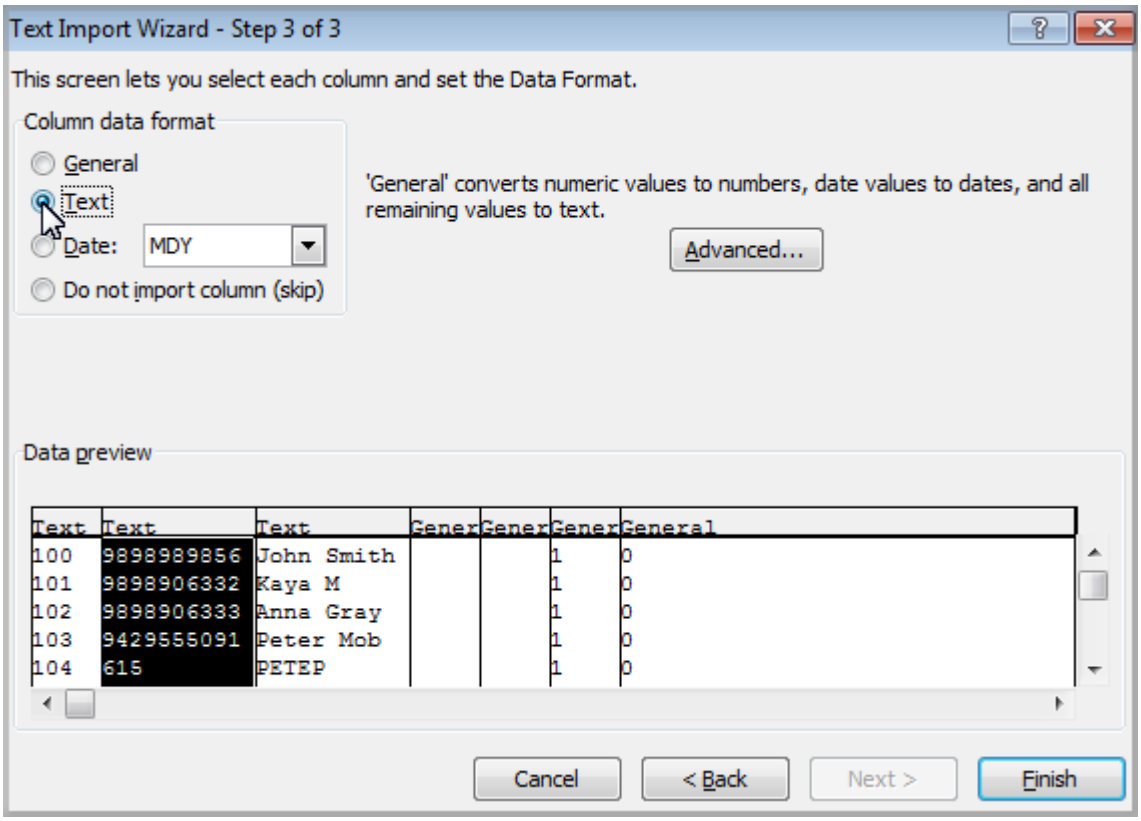


- In **Delimiters** select the **Comma** check box (column dividers will appear in preview) and click **Next.**



- Select the column with leading zeros and in **Column data format** select the **Text** radio button.

  You will have to do this for each column where the data contains leading zeros.



- Click **Finish**.

- The leading zeros will still be there in the new worksheet with the imported data.

Configuring Global Directory using a Telephone

- Enter SE mode.

  To program a telephone number in the global directory, dial:
  - ***1801-Location Code-Number***
    Where,
    Location Code is from 100 to 999.
    Number is the telephone number, max. 16 digits. If the number has fewer than 16 digits, you must dial #* to terminate the command.

  To clear a telephone number from a location in the global directory, dial:
  - ***1801-Location Code-#****

  To program a name in global directory, dial:
  - ***1802-Location Code-Name***
    Where,
    Location Code is from 100 to 999.
    Name is a string of alphanumeric characters, max. 12 characters. If the name has fewer 12 characters, you must dial #* to terminate the command.

To clear a name from a location in the global directory, dial:

- ***1802-Location Code-#****

To program the OGTBG for a contact in the global directory, dial:

- ***1803-1-Location Code-OGTBG*** to assign OGTBG to a single Location code.
- ***1803-2-Location Code-Location Code-OGTBG*** to assign OGTBG to a range of Location Codes.
- ***1803-*-OGTBG*** to assign OGTBG to all Location Codes.
  Where,
  Location Code is from 100 to 999.
  OGTBG is from 01 to 32.
  By default, the OGTBG assigned is 01.

To clear a location code in the global directory:

- ***1800-1-Location Code***
- ***1800-2-Location Code-Location Code***
- ***1800-****

- Exit SE mode.

Refer the topics "Class of Service (COS)" and "Station Basic Feature Template" for instructions on how to use SE commands to

- enable a feature (Global Directory) in the CoS Group.
- assign the CoS group to a Station Basic Feature Template
- apply the template with the newly programmed CoS group to the extensions.

Personal Directory Configuration by Extension Users

Extension users can program their own Personal Directories using their extension phones. The personal directory may be programmed using DKP or SLT.

If you are using an SLT, you will not be able to program the Name of the contact in the directory.

For EON and Extended IP Phone Users

- Press DSS Key assigned to Personal Directory (if programmed).

  OR

- Dial 1071.
- Enter Personal Memory Index (01 to 25)
- Enter Number of the contact (max. 16 digits).
- Press 'Enter' key.
- Enter Name of the contact (max. 12 characters).
- Press 'Enter' key.
- Enter Trunk Access Code.
- Press 'Enter' key.
- You get confirmation tone and the message on your phone's display.

For SLT Users

- Pick up handset.
- Dial 1071.
- Dial Personal Memory Index (01 to 25).
- Dial Number of the contact (max. 16 digits).
- Press #*.
- Dial Trunk Access Code.
- You get confirmation tone.
- Replace handset

Global Directory Configuration by Extension Users

Extension users can add, delete and edit contacts in Global Directory Part 1 using their extension phones, provided that:

- their phone is a digital key phone;
- Global Directory Part 1 is allowed to them in their Class of Service.
- Global Directory Programming is allowed to them in their Class of Service.

- *Extension users can only add, delete and edit names and numbers of contacts in Global Directory Part 1. However, they cannot program the Outgoing Trunk Bundle Group (OGTBG) for the contacts in the directory.*

- *If LDAP is enabled, then the contacts stored in the Global Directory synchronized with LDAP cannot be edited or deleted from the system.*

- *When an extension user programs Global Directory Part 1, the system will automatically assign the number and name to a free Memory Location. The system will use the OGTBG assigned to that Memory Location by the System Engineer to dial out the number added by the extension user.*

- *By default, OGTBG 01 is assigned to all Memory Location Codes in the Global Directory.*

- *To simplify configuration for both the System Engineer and the extension users, the System Engineer is recommended to assign the same OGTBG number uniformly to all Memory Location Codes, and enable Least Cost Routing on this OGTBG.*

- *If no OGTBG has been assigned to a Memory Location in the Global Directory (that is, the field is blank), and an extension user adds a contact to this Memory Location, the number will not be dialed out.*

To program Global Directory Part 1 from the Digital Key Phone or Extended IP Phone, follow these steps:

- Press Enter key to enter Phone Menu.
- Scroll to ‘Contacts’ and press Enter Key.
- You will get the following options:
  - Add
  - Edit
  - Delete

Adding a contact

To add a contact, select ‘Add’ and press Enter key.

- Enter your contact’s name on the prompt: ‘Name:’
  A maximum of 12 characters are allowed.
- Press Enter key to save name.
- Enter your contact’s number on the prompt: ‘Number:’
  A maximum of 16 digits are allowed.
- Press Enter key to save number.
- You will get the confirmation tone and the confirmatory message: “Stored at Index xxx”.

Editing a contact

To edit a contact,

- Scroll to ‘Contacts’ in the phone menu and press Enter key.
- Select ‘Edit’ and press Enter key.
- You get the prompt: 'Name:'
- Enter the initial letters of the contact's name.
- The number of matching entries that will appear at a time on your phone's display will vary according to your phone's LCD display capacity.
- Scroll with the Up/Down navigation keys to reach the desired contact's name on the list.
- Press 'Enter' key to select the name.
- The system displays the name you selected.
- To delete a character, use the Back/Forward navigation key to place the cursor under the character you want to delete.
- Press the ‘Cancel’ key to delete the character you selected with the cursor.
- To enter a character, use the Back/Forward navigation key to place the cursor in the position you want to enter the character.
- Enter the desired character by pressing the relevant digit pad keys in quick succession.
- After you have finished editing the name/ number, press Enter key.
- The number of the contact whose name you edited will be displayed.
- Repeat the same steps as you did for editing the name.
- After you have finished editing the number, press Enter key.
- You will get the confirmation tone and the confirmatory message: “Stored at Index xxx”.

Deleting a contact

To delete a contact,

- Scroll to ‘Contacts’ in the phone menu and press Enter key.
- Select ‘Delete’ and press Enter key.
- You get the prompt: 'Name'
- Enter the initial letters of the contact's name.
- The number of matching entries that will appear at a time on your phone's display will vary according to your phone's LCD display capacity.
- Scroll with the Up/Down navigation keys to reach the desired contact's name on the list.
- Press 'Enter' key to delete the name.
- You will get the confirmation tone and the confirmatory message: ’Deleted’.

How to use

Personal Abbreviated Dialing

For EON and Extended IP Phone Users

- Press DSS Key assigned 'Abbreviated Dialing' function (if programmed).

  OR
- Dial 8 (users world wide)
  OR

- Dial 6 (users in USA)
- You get the message 'Give Index'
- Enter Personal Directory Index number: 001 to 025[234].
- The desired number will be dialed out.

For SLT Users

- Pick up the handset.
- Dial 8 (users world wide)

  OR

- Dial 6 (users in USA)
- Dial Personal Directory Index number: 001 to 025[235].
- The desired number will be dialed out.

Global Abbreviated Dialing

For EON Users

- Press DSS Key assigned 'Abbreviated Dialing' function (if programmed).

  OR

- Dial 8 (users world wide)

  OR

- Dial 6 (users in USA)
- You get the message 'Give Index'
- Enter Global Directory Index number: 100 to 999[236].
- The desired number will be dialed out.

234. For ETERNITY LENX/MENX it is 0001 to 0025.
235. For ETERNITY LENX/MENX it is 0001 to 0025.
236. ETERNITY LENX/MENX stores up to 2900 numbers in the Global Directory; Part 1 - 0100 to 2399, Part 2 - 2400 to 2699 and Part 3 - 2700 to 2999.

For SLT Users

- Pick up the handset.
- Dial 8 (users world wide)

  OR

- Dial 6 (users in USA)
- Dial Global Directory Index number: 100 to 999[237].
- The desired number will be dialed out.

237. *ETERNITY LENX/MENX stores up to 2900 numbers in the Global Directory; Part 1 - 0100 to 2399, Part 2 - 2400 to 2699 and Part 3 - 2700 to 2999.*

Access Codes

What's this?

Access codes are short digit sequences dialed from an extension phone to instruct the System to perform a function such as:

- Calling an extension.
- Calling a group of extensions (“Department Call”).
- Grabbing a trunk line or any trunk line from a group of trunks (“OG Trunk Bundle Group”).
- Invoking a feature. Activating or deactivating a feature.

Accordingly Access Codes are classified into:

- **Station Codes:** Codes used for calling extensions. These codes are also commonly referred to extension numbers, phone numbers. For the purpose of this document, station codes are referred to as Flexible Numbers.

  Default station codes: the factory-set default values for SLT extensions are from 2001 to 2512; for DKP extensions from 3001 to 3128.

- **Logical Group Codes:** codes used for calling a group of stations as in a Department group, a group of trunks as in Outgoing Trunk Bundle Group.

  Default logical group codes: the factory-set codes for Department Numbers start from 3901, 3902…. Outgoing Trunk Bundle Groups from 61, 62, etc.

- **Feature Codes:** codes used for invoking a feature.

  Default feature codes: there are different feature codes for every feature/function of the SARVAM UCS, e.g.: '2' for Auto Call Back, '5' for Raid, '13' for Call Forward, etc.

  Access codes may consist of single digits or a sequence of a maximum of 6 digits.

  You can change the default access codes to codes of your choice. For example: the default Operator code '9' can be changed to '0'; the default trunk access code for dialing Trunk Group1 can be changed from '61' to '5'.

How it works

Whenever an access code is dialed from an extension, the system matches each digit in the code with the access codes programmed within the system to determine the instruction, that is, whether it is an extension it must call, or a trunk line it must grab, a port it has to activate, etc. The system processes the instruction when a match is found.

For example:

- An extension user dials 131 to set Call Forward.

- When the first digit '1' is dialed, the system finds a match. As several default access codes begin with '1' the system waits for the next digit to be dialed.
- When the second digit '3' is dialed, the system finds a match for '13'.
- As '13' is common for all Call Forward options[238], the system waits for the next digit to be dialed
- When the user dials the third digit '1', the system finds a match for '131'.
- If there is more than one access codes matching with '131', e.g. '1311', '1314', '1315' the system will wait for the next digit to be dialed.
- If no further digit is dialed on expiry of the Inter Digit Wait Timer, the system understands the instruction as 'Call Forward - Unconditional' and waits for the destination phone number to be dialed.

Access Codes are related to various phases of a call. When a call is processed by a System, it goes through a number of pre-defined phases.

Typically a call passes through the different phases as shown below:

Idle	Dial	Routing	Blocked	Placed	Matured 2-Way	Matured 3-Way	Denied
No activity.	Digits are pressed on the phone keypad/dialed from the rotary. Dial tone is played.	The system is processing the call. The call is neither placed nor blocked. Beeps are played.	The dialed extension is busy. Busy tone is played.	The dialed extension is ringing. Ring Back Tone is played.	Connected with the dialed extension. Two-way speech.	Connected with two extensions. Three-way speech.	No reply from dialed extension. Error Tone is played.

Different access codes are dialed at different call phases. Station Codes and Logical Group Codes are dialed in the 'Dial' phase.

As different features are invoked in each call phase, Feature Access Codes are dialed at different call phases. For example:

- Call Forward code is dialed at the 'Dial' phase'.
- DND Override code is dialed at the 'Routing' phase'
- Auto Call Back code is dialed at the 'Blocked' phase' as well as 'Placed' phase.
- Three-party Conference code is dialed at the 'Matured 2 way' phase.

'Idle' phase is when no code is dialed. In the 'Denied Phase' no code is allowed to be dialed.

238. Call forwarding options: Unconditional, When Busy, When No Reply, When Busy or No Reply.

Each access code in a single call phase may be of different lengths, but must be unique. For example, the same access code cannot be used for two different features like Call Forward and Redial, since both these features are invoked in the 'Dial' phase.

However, the same access code can be used for features in different call phases. For example, '4' is the default feature access code for DND Override (Routing Phase), Call Pick-Up-Group (Dial Phase) and Barge-In (Blocked Phase).

Similarly, Station and Logical Group Codes too must be unique and should not match with any of the features invoked in the 'Dial' phase. Refer the topics “Flexible Numbers” and “OG Trunk Bundle Group” to know more.

How to configure

SARVAM UCS provides default Access Codes for stations, logical groups - department and trunk groups - and features.

It also provides country-specific default Access Codes which are applied automatically when you select the 'Region' to configure the system.

The default Access Codes for India are presented in the table below. The default Access Code tables also indicate the call phase in which each feature is invoked.

Feature	Feature Number	Access Code	Call Phases					
			Dial	Routing	Blocked	Placed	Matured 2-way	Matured 3-way
Enter SE Programming Mode	1	1#91	Y				Y	
Enter SA Programming Mode	2	1#92	Y				Y	
Call Pickup - Group	3	4	Y				Y	
Call Pickup - Selective	4	12	Y				Y	
Auto Call Back - Set	5	2			Y	Y		
Auto Call Back - Cancel	6	102	Y				Y	
Redial	7	7	Y				Y	
Auto Redial - Set	8	17	Y				Y	
Auto Redial - Cancel	9	1070	Y				Y	
Personal Directory Programming	10	1071	Y				Y	
Abbreviated Dialing	11	8	Y				Y	
Operator	12	9	Y				Y	Y
Call Forward	13	13	Y				Y	
Dynamic Lock	14	14	Y				Y	
Hotline	15	15	Y				Y	
Alarm	16	161	Y				Y	

Feature	Feature Number	Access Code	Call Phases					
			Dial	Routing	Blocked	Placed	Matured 2-way	Matured 3-way
Do Not Disturb	17	18	Y				Y	
Interrupt Request	18	3			Y			
Barge-In	19	4			Y			
Raid	20	5			Y			
Trunk Reservation	21	6			Y			
Call Toggle	22	1						Y
Conference	23	*3						Y
PIN Dialing	24	*2	Y				Y	
Dial-In Conference	25	*19	Y				Y	
Call Park	26	115					Y	
Call Park - Retrieve	27	116	Y				Y	
Room Monitor	28	1073	Y				Y	
Last Caller Recall	29	1092	Y				Y	
Voice Help	30	1090	Y				Y	
Walk-In Class of Service	31	111	Y				Y	
Change User Password	32	114	Y				Y	
Paging	33	1074	Y				Y	
DISA Login	34	1079	Y					
Trunk to Trunk Call Release	35	##					Y	
Cancel all Feature Settings	36	1051	Y				Y	
Selective Port Access	37	69	Y				Y	
Flashing on Trunk	38	*	Y				Y	
User Absent/Present	39	104	Y				Y	
Account Code by Number	40	1058	Y				Y	
Account Code by Name	41	1059	Y				Y	
Meet Me Paging	43	1093	Y				Y	
Hot Desk	44	1091	Y					
Do Not Disturb Override	45	4		Y				
Presence	46	1097	Y					
Live Call Screening	47	1094	Y				Y	
Conversation Recording	48	1095					Y	
Forced Release	49	#*			Y			
Transfer	50	F					Y	
Live Call Supervision	51	1098	Y				Y	

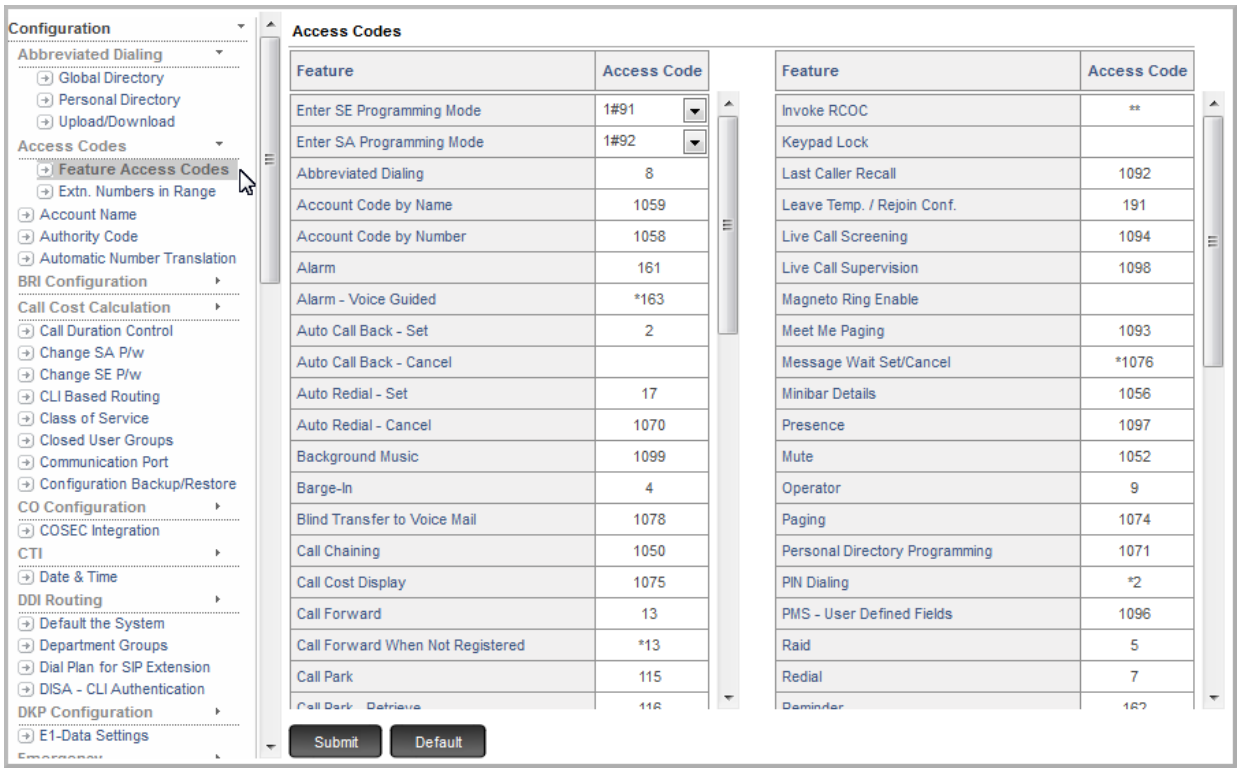
Feature	Feature Number	Access Code	Call Phases					
			Dial	Routing	Blocked	Placed	Matured 2-way	Matured 3-way
Forced Answer	52	5				Y		
Change Room Clean Status	53	1054	Y					
Guest Number Prefix	54	1055	Y					
Minibar Details	55	1056	Y					
Mute	56	1052	Y				Y	
Emergency Conference	57	1177	Y					
Self Ring Test	58	1057	Y					
Call Chaining	59	1050					Y	
SA Command Prefix	60	1072	Y				Y	
COSEC Door Open	61	*7	Y					
Floor Service	62	38	Y				Y	
Keypad Lock	63	-	Y					
CLI Restriction	64	103	Y					
Call Cost Display	65	1075	Y					
Reminder	66	162	Y				Y	
Alarm - Voice Guided	67	163	Y					
Reminder - Voice Guided	68	164	Y					
Blind Transfer to Voice Mail	69	1078					Y	
Message Wait Set/Cancel	70	1076	Y				Y	
Retrieve New Message	71	1077	Y					
PMS - User Defined Fields	72	1096	Y					
Magneto Ring Enable	77	#				Y		
Invoke RCOC	78	**	Y					
Scheduled Call Forward	79	1175	Y				Y	
E&M Manual Priority Intrusion	80	*37					Y	
E&M Forced Release Order	81	*38					Y	
Department Call Forward	82	1179	Y				Y	
General Mailbox	83	1176	Y				Y	
Intercom	84	*5	Y				Y	
Terminate Conference	86	190						Y
Leave Temp. / Rejoin Conf.	87	191						Y
Call Forward - When Not Registered	88	*13	Y					

Feature	Feature Number	Access Code	Call Phases					
			Dial	Routing	Blocked	Placed	Matured 2-way	Matured 3-way
Trunk Access Code 1(TAC1)	-	0	Y					
Trunk Access Code 2(TAC2)	-	5	Y					
Trunk Access Code 3(TAC3)	-	61	Y					
Trunk Access Code 4(TAC4)	-	62	Y					
Trunk Access Code 5(TAC5)	-	63	Y					
Trunk Access Code 6(TAC6)	-	64	Y					
Voice Mail	-	3931	Y					

You can either use the default Access Codes or change them to suit your preferences.

Changing Feature Access Codes using Jeeves

- Login as System Engineer.
- Under **Configuration**, click **Access Codes**.
- Click **Feature Access Code** to open the page.



- Change the Access Codes as per your preference. A maximum of 6 digits.
- Click **Submit** at the bottom of the page to save changes.
  If you enter a code that is already assigned to a station or a feature, the system will not accept the duplicated code. The value will remain unchanged.

- To disable an access code, delete the existing code and leave the field blank.
- Click **Default** all to default all Access Codes.

To change Station Access Codes (for extensions), Department Groups, Trunks and Trunk Groups, and General Mailbox Access Code refer the topics “Flexible Numbers”, “Department Call”,“OG Trunk Bundle Group” and “General Mailbox Settings”.

Changing Feature Access Codes using a Telephone

- Enter SE mode from an SLT/DKP.

  To program the access code for a feature, dial:
  - ***3111-1-Feature Number-Access Code-#****
    Where,
    Feature Number is from 01 to 88.
    Access Code is a string of maximum 6 digits

  - E.g.: To change the prefix of SA command '1072' to '107' dial: ***3111-1-60-107-#****
    Where,
    60 is Feature Number for SA command.
    107 is the new Access Code for SA command.
    If you try to assign a number string that is already used to access an extension or use a feature then the system will not accept the command and will play error tone.

  To default access codes, dial:
  - ***3161-1-Feature Number*** to default access codes for a single feature.
  - ***3161-2-Feature Number-Feature Number*** to default access codes of a range of features.
  - ***3161-**** to default access codes of all features
    Where,
    Feature Number is from 01 to 88.

    E.g.: To default the prefix of SA command '107' to '1072' dial: ***3111-1-60***

A default trunk access codes table is given below:

OGTBG Index	Default Trunk Access Code
1	0
2	5
3	61
4	62
5	63
6	64

To assign access code for a trunk access index, dial:
- ***3112-1-OGTBG Index-Access Code-#****
  Where,
  TAC Index is from 1 to 6.
  Access Code is maximum 6 digits (Generally access code for trunk is of two digits).

To clear the access code for a trunk access index, dial:

- ***3112-1-OGTBG Index-#****
- ***3112-2-OGTBG Index-OGTBG Index-#****
- ***3112-*-#****

To assign default access code for a TAC index, dial:

- ***3162-1-OGTBG Index***
- ***3162-2-OGTBG Index-OGTBG Index***
- ***3162-****

To program the access code for a VMS Group, dial:

- ***3114-1-VMS Group Index-Access Code-#****
- ***3114-2-VMS Group Index-VMS Group Index-Access Code-#****
- ***3114-*-Access Code-#****
  Where,
  VMS Group Index is 1.
  Access Code is maximum of 6 digits. If it is less than 6 digits, terminate it with #*
  By default, the Access Code of the VMS Group is 3931.

To default the access code for a VMS Group, dial:

- ***3164-1-VMS Group Index***
- ***3164-2-VMS Group Index-VMS Group Index***
- ***3164-****
  Where,
  VMS Group Index is 1.

To clear the access code for a VMS Group, dial:

- **3114-1-VMS Group Index-#***
- **3114-2-VMS Group Index-#***
- **3114-*-#***
  Where,
  VMS Group Index is 1.

- Exit SE mode.

Account Codes

What's this?

Account Codes are very useful feature for organizations such as business consultants, law firms, advertising and media agencies, and the like that cater to several clients, interacting with third parties on behalf of their clients. Such organizations need to keep track of calls made to and on behalf of each client.

An 'Account Code' is a unique three-digit number that an organization can assign to each of its clients. Each Account Code may be given a name and programmed in the Account Name List.

Doing so, whenever calls are made to the client or to a third party on behalf of the client,

- The extension user dials the Account Code or Name assigned to the client.
- The Account Code may be dialed,
  - before dialing the external number.
    Or
  - when in speech with the client/third party
- Details of these calls are recorded by the Account Code dialed in the Station Message Detail Recording Report (SMDR) for Outgoing Calls.
- The SMDR report can be printed using the Account Code as filter.

  This way, the organization can know the details of calls made to and on behalf of each client.
- The SARVAM UCS supports as many as 999 Account Codes.

How it works

For example, an advertising media agency makes nearly 100 calls every day to and on behalf of its clients that includes 'Midas Business Solutions', 'Jet-Set Holidays', 'Bacchus Vineyard'.

- Assign a three-digit account code to Midas Business Solutions, for instance '001' and the name code 'Midas Biz' in the Account Name List.
- Assign a three-digit account code to Jet-Set Holidays, for instance '002' and the name code 'Jet' in the Account Name List.

Case 1: Applying same Account Code

- A, a person from advertising media agency makes a call to Midas Business Solutions and talks to the secretary B. During an ongoing conversation A dials the account code 001. In between, A needs to consult the manager C. Therefore, A presses the *Transfer* key to put B on **Consultation hold** and dials the number of C. Account code 001 will be applicable to the second call made to C also.

Case 2: Applying different Account Codes

- A, a person from advertising media agency makes a call to Midas Business Solutions and talks to the secretary B. During an ongoing conversation A dials the account code 001. In between, A needs to talk to another client, say Jet-Set Holidays D. Therefore, A pressed the *Call Hold* key to put B on **Exclusive/ Global hold** and dials the number of D. While in speech with D, A dials account code 002. Here, Account Code 001 will be applicable to Midas Business Solutions and Account Code 002 will be applicable to Jet-Set Holidays.

Forced Account Code

The SARVAM UCS can be programmed to prompt extension users to dial the Account Code whenever they grab a trunk to dial out a number. Hence the feature name Forced Account Code.

To apply Forced Account Code, this feature must be enabled on the extensions and trunks from which calls using Account Codes are to be made. When Forced Account Code is enabled on an extension, and the extension user dials out a number or the Trunk Access Code to grab a trunk, the system will play an error tone. If the extension is a DKP, the system will flash a message on the phone's display that Account Code is required to make the outgoing call.

- *Account Codes are applicable for external calls only.*
- *To use Account Codes, this feature must be included in the Class of Service (CoS) group allowed to the extensions.*
- *If you want to use Account Names, you must program the Account Name List.*
- *When the Forced Account Code is enabled on an extension and trunk, the system will ask the user to enter the account code irrespective of the method of dialing: Global Abbreviated dialing, Personal abbreviated dialing, Least Cost Routing, or Selective Trunk Access.*
- *However, if Forced Account Code is enabled on the selected trunk, and the number is dialed using Selective Trunk Access, the system will dial out the number using Store and Forward dialing.*

  In the case of Abbreviated Dialing or Direct Dialing, if the extension user fails to dial the Account Code, an error message will be displayed on the extension user's DKP.

How to configure

For Account Code to work, you must:

1. Enable Account Codes feature in the Class of Service (CoS) of the extensions to which this feature is to be allowed.
2. Prepare and program the Account Name List, if it is to be used.
3. If Forced Account Code is to be used, you must enable 'Forced Account Code' flag in
   - the "Station Advanced Feature Template" applied to the extensions from which calls using account codes are to be made.
   - the "Trunk Feature Template" applied on the trunks through which calls using account codes are to be made.

All the above feature parameters can be programmed using Jeeves or by dialing commands from a Telephone.

Preparing Account Name List

In consultation with the Users, you may prepare the Account Name list. You may do the following

- Draw a two-column table on a paper.
- Write Account Codes on one column. Account codes may be any three-digit number between 001 and 999.
- Write the Account Names, that is, names of the clients on the second column, against their respective Account Codes.
- The names must not exceed 12 characters. All ASCII characters except < > and “ (double quote) are allowed. For example:

Account Code	Client Account Name
001	Midas Biz
002	Jet Set
:	
010	Bacchus

You need not follow a cardinal numbering sequence when assigning Account Codes.

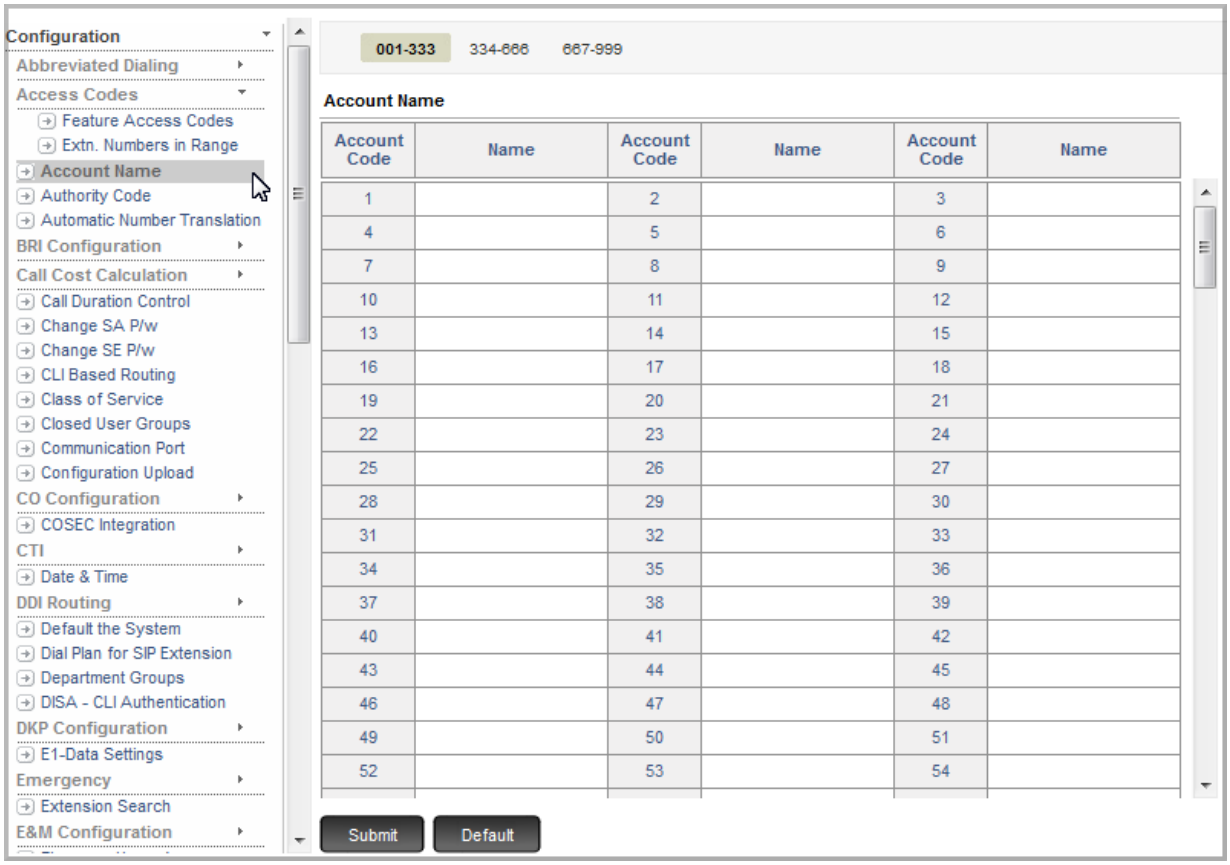
You may assign any code to any client. For instance, you can assign code '111' to Midas Business Solutions, '222' to Jet-Set Holidays, '333' to Bacchus Vineyard.

Configuring Account Codes using Jeeves

After preparing the Account Name list,

- Log in as System Engineer.

- Under **Configuration**, click **Account Name**.



- Enter the Names of the clients against the account codes you have assigned to them. Refer the paper with the two-column table you created.

- Click **Submit** to save your settings.

Now, include the feature **Account Codes** in the **Class of Service** of the extensions.

In the default factory settings, Station Basic Feature Template Number 01 is assigned to all extensions of the system. Template 01 has the feature Account Codes in the default CoS Group (Number 01). So, all extensions of the system can use this feature.

If Account Codes is to be allowed only to selected extensions, follow these steps:

1. Define a CoS group with 'Account Codes' enabled.

2. Prepare a Station Basic Feature Template with this CoS group applicable in all the "Time Zones".

3. Assign this new Template to the extensions to which Account Codes is to be allowed.

Refer the topics "Class of Service (COS)" and "Station Basic Feature Template" for detailed instructions.

If Forced Account Code is to be used, enable the 'Forced Account Code' flag in the "Station Advanced Feature Template" of the extensions and in the "Trunk Feature Template" assigned to the trunks.

When you enable 'Forced Account Code' in a template, this feature will be enabled on all DKP, SLT and SIP extensions that are assigned this template. If necessary, create a separate template with this feature and assign this template only to those extensions that are to be assigned this feature.

To enable **Forced Account Code** flag on Trunks,

- Go to the parameters page of the type of trunk you want to program. For example: **CO Parameters** page to program CO Trunk, **Mobile Parameters** page to program GSM Trunk, **SIP Parameters** page to program SIP Trunks etc.
- Click the **Trunk Feature Template** link in the column. The feature template page will open.

  By default Trunk Feature Template 01 is assigned to all Trunk Types.

  If you want to enable this feature on all trunks, enable it in Trunk Feature Template 01.

  If you want to enable this feature only on select trunks, program a different Template number with this feature.
- Select the **Forced Account Code** check box in the template number assigned to the trunk.
- Click **Submit** to save your changes.
- Return to the trunk parameters page.
- Now change the Trunk Feature Template number of the trunk you want to program. This number should be the same as the template in which you have enabled the Forced Account Code check box.
- Click **Submit** to save changes.

Repeat the above steps to configure all other trunk types.

Configuring Account Codes using a Telephone

- Enter SE mode.

  To create the Account List:
  - Dial command ***4851-1-Account Code-Account Name***
    Where,
    Account Code is 001 to 999.
    Account Name is a string of 12 characters.

    Terminate Account Name with #*, if it is less than 12 characters.
    For example: to program account code '001' with the name 'Midas Biz', dial
    4851-1-001-MIDAS BIZ-#*
    Press the digit key '1' twice to give space between characters.

  To enable Forced Account Code flag in Station Advanced Feature Template:
  - Dial command ***5602-1-Template Number-Feature Number-Flag Code***
    Where,
    Template Number is Station Advanced Feature Template from 01 to 50. Default: 01
    Feature Number for Forced Account Code is 09.
    Flag Code is

0 for Disable
1 for Enable

For example: To enable Forced Account Code Flag in Station Advanced Feature Template 02: Dial ***5602-1-02-09-1***

To apply the Station Advanced Feature Template now programmed with the Forced Account Code on stations, refer the topic “Customizing Station Advanced Feature Template using a Telephone”.

To enable Forced Account Code flag on a Trunk:

- Dial command ***5802-1-Template Number-Feature Number-Flag Code***
  Where,
  Template Number is from 01 to 50.
  Feature Number for Forced Account Code flag is 29.
  Flag Code is
  0 for Disable
  1 for Enable

  For example: To enable Forced Account Code Flag in Trunk Feature Template 01: Dial ***5802-1-01-29-1***

  To apply the Trunk Feature Template now programmed with the Forced Account Code on different types of trunks, refer the topic “Customizing Trunk Feature Template using a Telephone”.

- Exit SE mode.

How to use

Account Codes can be dialed in two ways: by Number and by Names.

Account codes, that is, number and names, can be dialed:

- before making the call,
- during the call,
- when grabbing a trunk (if Forced Account Code flag is enabled).

Print and hand out copies of the Account Code List to everyone in the organization for reference while making calls.

Dialing Account Code by Number

For EON Users and Extended IP Phone Users

To enter Account Code Number before making the call:

- Press DSS Key assigned to 'Account Code by Number'.

  OR

- Dial **1058**
- Enter Account Code
- Dial Trunk Access Code

- Dial the number of the client.

To enter Account Code Number during the call:

- Press ‘Transfer’ Key
- Press DSS Key assigned to Account Code by Number.

  OR

- Dial **1058**
- Enter Account Code
- Speech will be resumed.

To enter Account Code Number when Forced Account Code Flag is enabled:

For EON Users and Extended IP Phone Users

- Press DSS Key assigned to Account Code by Number.

  OR

- Dial **1058**
- Enter the Account Code Number.
- You get dial tone.
- Dial Trunk Access Code followed by the number of the client.

  If you dial the Trunk Access Code to grab a trunk, without dialing the Forced Account Code, you will get an error tone. Go ON-hook and then go OFF-hook. Now follow the same steps in the sequence mentioned above.

For SLT Users

- Go OFF hook.
- Dial Account Code first.
- Dial Trunk Access Code followed by the number of the client.

  If you dial the Trunk Access Code to grab a trunk, without dialing the Forced Account Code, you will get an error tone. Go ON-hook and then go OFF-hook. Now follow the steps in the sequence mentioned above.

Dialing Account Code by Name

Dialing Account Code by Names is possible only if your phone is a DKP / Extended IP Phone.

For EON Users and Extended IP Phone Users

To enter Account Code Name before making the call:

- Press DSS Key assigned to Account Code by Name.

  OR

- Dial **1059**.

- Enter the initial letter of the client's name.
  The Account Name List will be displayed on your DKP, alphabetically with the corresponding account codes.
- Scroll to select the desired client name and press Enter key.
- Dial Trunk Access Code.
- Dial the client's number.

To enter Account Code Name during the call:

- Press Hold key to put the called party on hold.
- Press DSS Key assigned to Account Code by Name.

  OR

- Dial **1059**.
- Dial the initial letter of the client's name.
  The Account Name List will be displayed on your phone, alphabetically with the corresponding account codes.
- Scroll to select the desired client name and press Enter key.
  Speech will be resumed with the called party.

Dialed the wrong account code or name?

If you have dialed the wrong account code or name while in the middle of a call, you can correct it by pressing 'Hold' again and following the steps described above. The system will override the previously dialed account code or name.

Printing Call Reports of Clients

You can print the call details of your clients using their account codes as filter.

Refer the section “Station Message Detail Recording-Report”, for more detailed instructions on printing reports using filters.

AC Impedance Test

What's this?

SARVAM UCS supports the AC Impedance Test for clear, audible and echo-free speech over the CO Trunks. This test helps you to set the most appropriate values for the CO Trunk parameters —AC Impedance, CO Termination and CO Line Type— to correct the line impedance mismatch between the AC Termination Impedance presented by the CO port of SARVAM UCS to the line and the CO Termination Impedance presented by the Central Office to the line.

Conducting AC Impedance Test

You can conduct an AC Impedance Test,

- by making an outgoing call
  or
- on an ongoing incoming call.

To conduct the AC Impedance Test you will need,

- a telephone with a valid number. You are recommended to use -a mobile phone with Mute function.
- the CO Trunk which you want to test.

  For the CO Trunk you want to test, make sure you select a CO Hardware Template number.

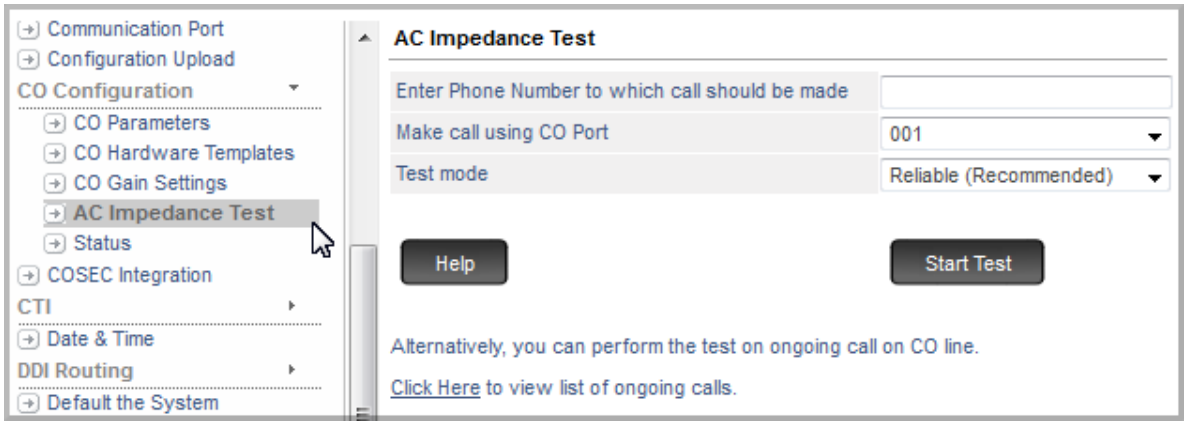
The DSS Key assigned to this CO Trunk will display status as busy to all the user, for the duration of the test.

Configuring Using Jeeves

You can conduct the test after you have set the relevant test parameters. To do this,

- Log in as System Engineer.

- Under **CO Configuration**, click **AC Impedance Test**.



To conduct an AC Impedance Test by making an outgoing call

- In **Enter Phone Number to which call should be made**, enter the phone number on which you want to make test call. The number can be a landline or a mobile number. We recommend you to use a mobile number for the test call.

If you are using a mobile phone number, be sure the handset of the configured number supports the Mute function.

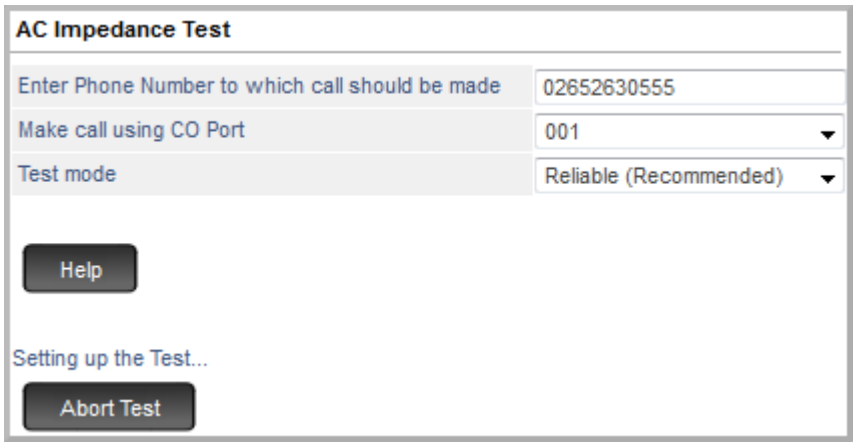
- In **Make call using CO Port**, select the CO trunk using which you want to make the test call. This must be the same CO trunk for which AC Impedance is to be set.

- Select the **Test Mode**. You may select **Reliable (Recommended)** or **Accurate**.

  The **Reliable Test** mode suggests the AC Impedance settings on the basis of most commonly used AC Impedances, CO Terminations and CO Line Types across the globe. The test using Reliable Test mode takes approximately 5 minutes to complete.

  The **Accurate Test** mode suggests the AC Impedance settings on the basis of all the possible AC Impedances, CO Terminations and CO Line Types across the globe. The test using the Accurate Test mode takes 1 hour and 20 minutes to complete.

- Click the **Start Test** button. The system calls the phone number you configured. The message '*Setting up the Test....*' appears on your screen.



While the test is being conducted, you will hear pulsating tone on all the ports of the same card. For ETERNITY PENX, you will hear this tone on all the CO ports of the system.

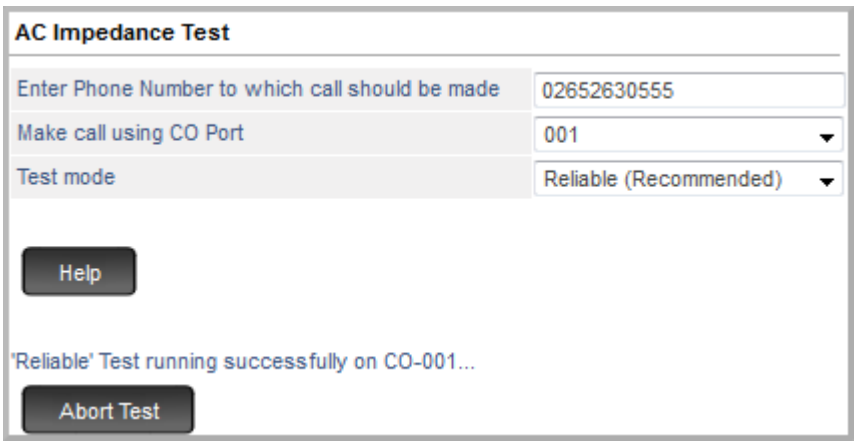
- Answer the test call from your mobile phone. You will hear the Music-on-Hold as per the type of Answer Supervision you have configured in the CO Hardware Template assigned to the CO trunk you are testing.

  By default, the Answer Supervision selected in the template is Pseudo Answer and the Pseudo Answer Supervision Timer is set to 10 seconds. If you have not changed this default setting, you will hear Music-on-Hold after 10 seconds of answering the call.

- As the Music-on-Hold begins to play, Mute the microphone of your mobile phone.

  If you are making the test call on a landline number, mute the call using the Mute key of the phone. If your phone does not have a Mute key, unplug the handset cable from the phone body. This is to prevent test signals from reflecting back into the mic of the handset.

- After 5 seconds of Music-on-Hold, you will hear the test signals being transmitted by the system for the duration of the test. The message '*'Reliable' Test running successfully...*' appears on your screen.

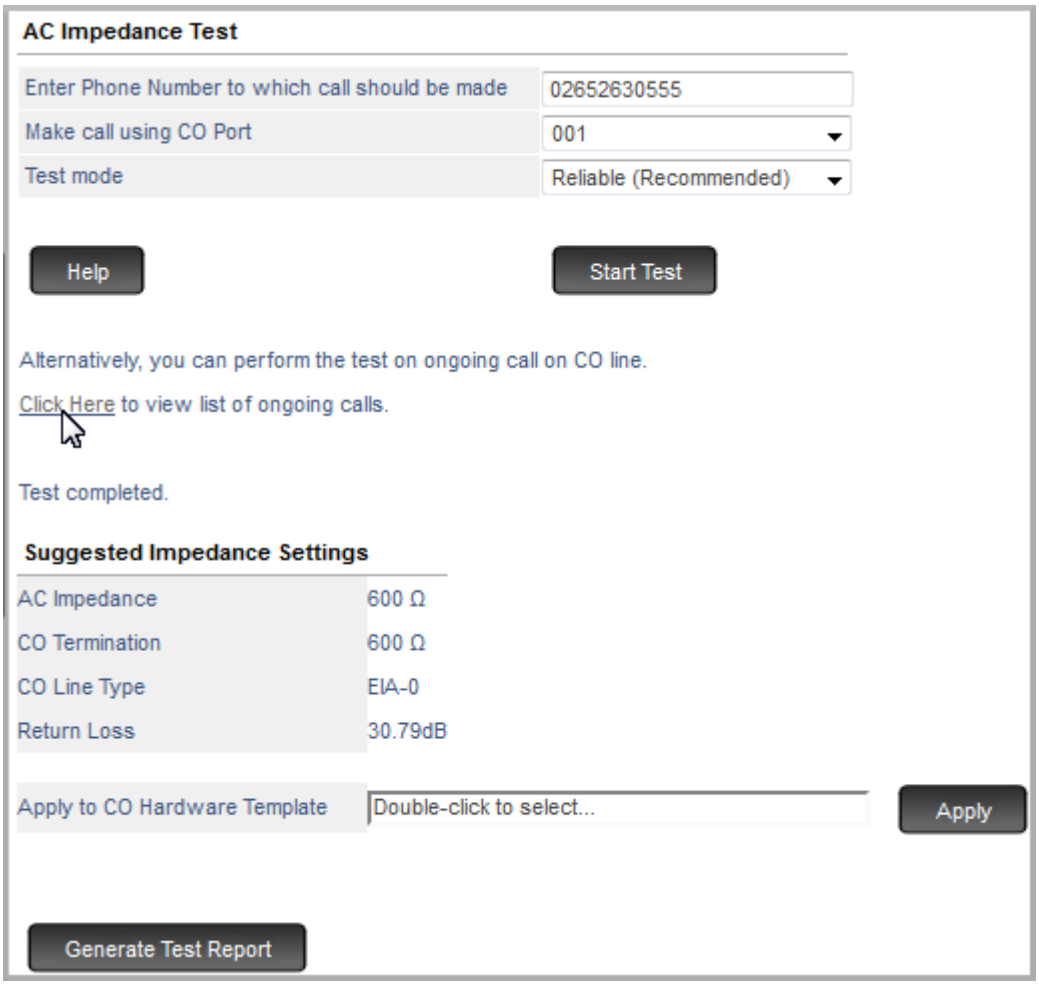


- On completion of the test, the system will automatically disconnect the call. The message '*Test completed*' appears on your screen.

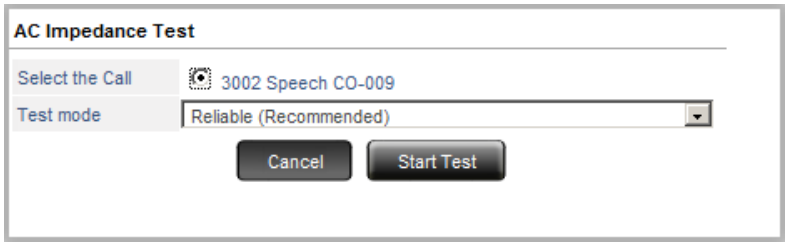
However, if you wish to abort the test midway, you may click the **Abort Test** button.

To conduct an AC Impedance Test on an ongoing incoming call

- Click the *Click Here* link on the AC Impedance Test page.



- A new window opens. A list of ongoing calls will be displayed. Select a call on which you want to perform the test.



- Select the **Test Mode**. You may select **Reliable (Recommended)** or **Accurate**.

  The **Reliable Test** mode suggests the AC Impedance settings on the basis of most commonly used AC Impedances, CO Terminations and CO Line Types across the globe. The test using Reliable Test mode takes approximately 5 minutes to complete.

- The **Accurate Test** mode suggests the AC Impedance settings on the basis of all the possible AC Impedances, CO Terminations and CO Line Types across the globe. The test using the Accurate Test mode takes 1 hour and 20 minutes to complete.
- Click the **Start Test** button. The message '*Setting up the Test....*' appears on your screen.

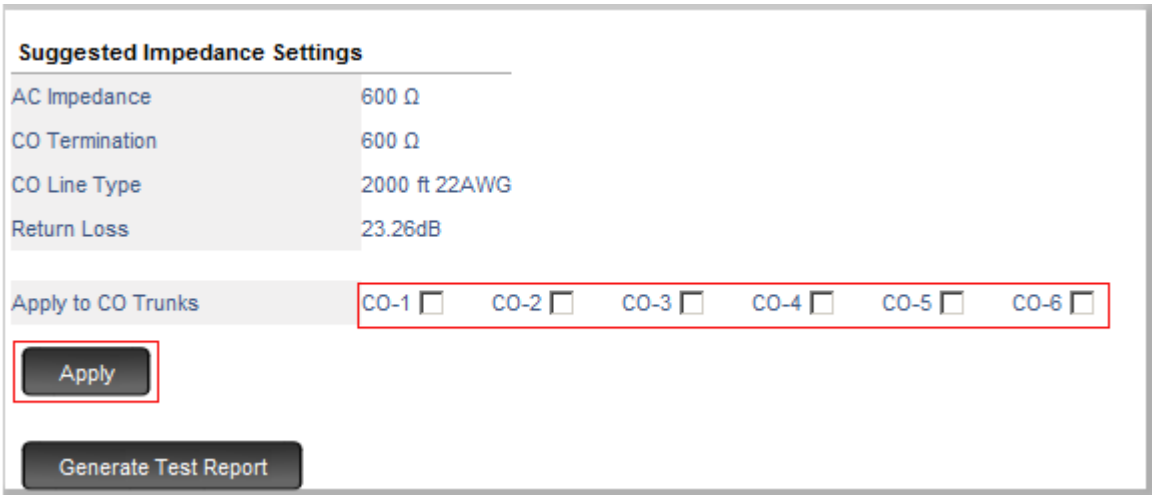
While the test is being conducted, you will hear pulsating tone on all the ports of the same card.

- After 5 seconds the message '*Test running successfully...*' appears on your screen.
- On completion of the test, the system will automatically disconnect the call. The message '*Test completed*' appears on your screen.

  However, if you wish to abort the test midway, you may click the **Abort Test** button.

Suggested Impedance Settings

- At the end of the test, the page displays the **Suggested Impedance Settings** for the AC Impedance, CO Termination, CO Line Type and Return Loss.
- You may now apply the suggested AC Impedance settings to the CO Trunk. To apply these settings, select the desired CO Trunks and click on the **Apply** button.



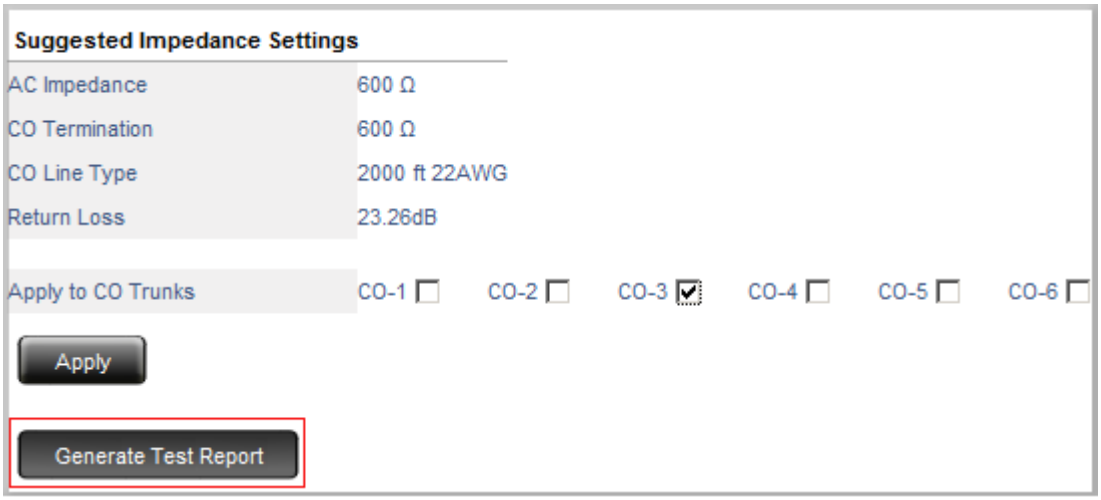
- Verify the settings by making a trial call from a DKP. There should be no echo and speech should be audible and clear.

  If you still hear echo during the trial call, you may re-run the test using the **Accurate Test** mode.

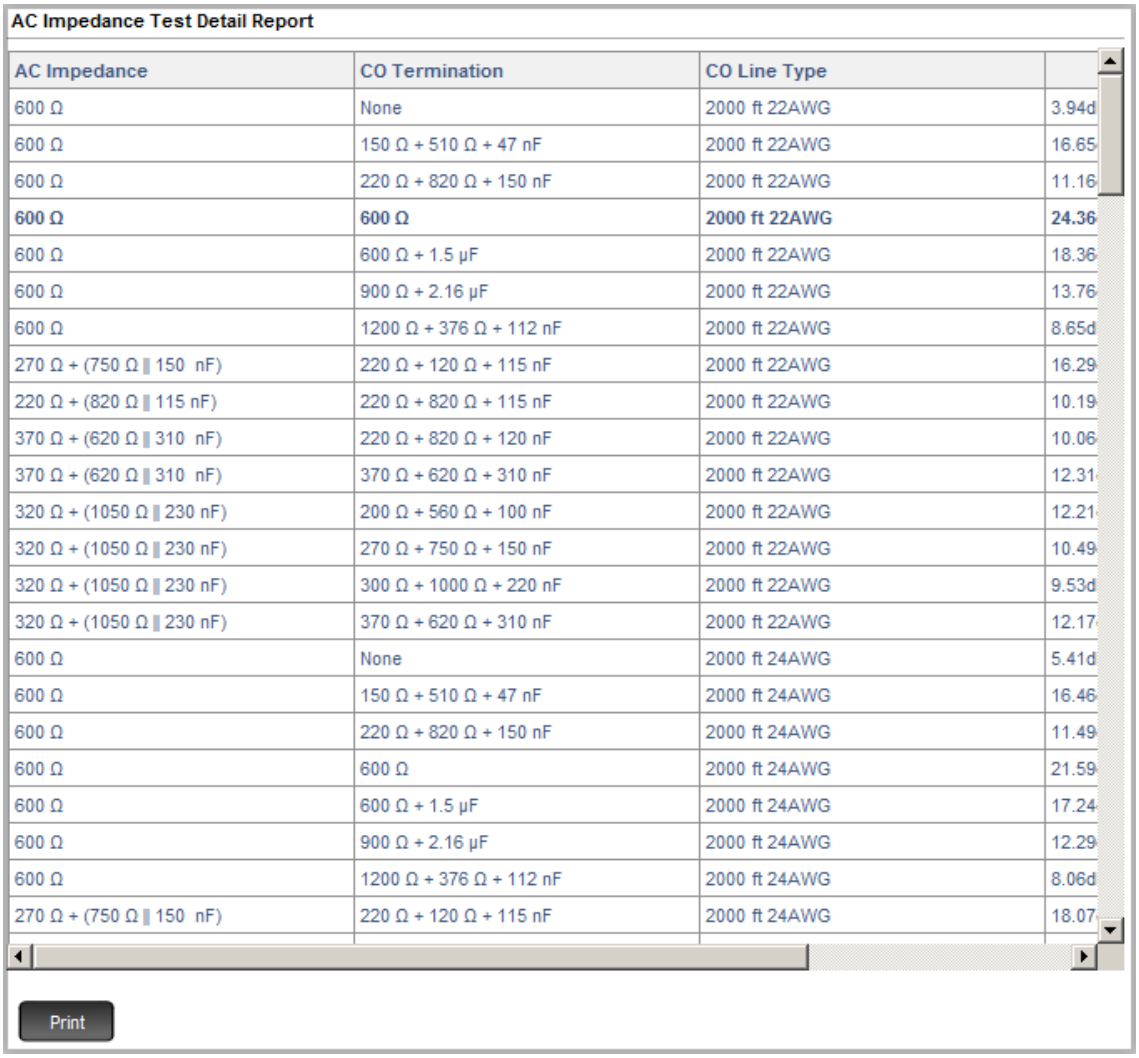
- After you have determined the best matching AC Impedance, CO Termination, CO Line Type and Return Loss by running the tests, apply the same suggested settings on the CO Trunk you are testing. You may configure the same settings to all other CO Trunks, you have subscribed from the same CO exchange.

It is possible that the CO trunks subscribed from the same exchange differ in their AC Impedance settings, in such a case, you must run the test for each CO trunk separately and configure a different CO Hardware Template for each of these trunks.

- To generate the detailed test report, click the **Generate Test Report** button.

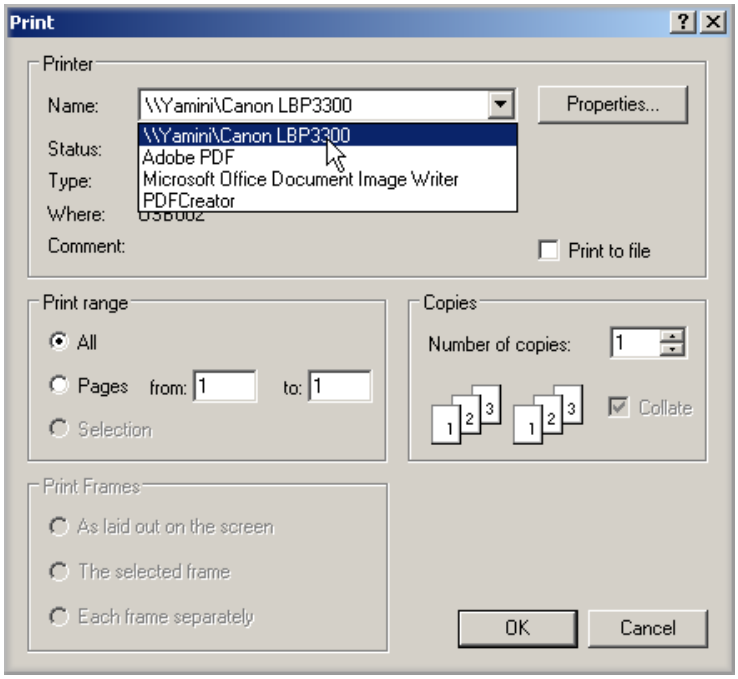


The detailed test report appears in a new window. The **Suggested Impedance Setting** will appear in bold.

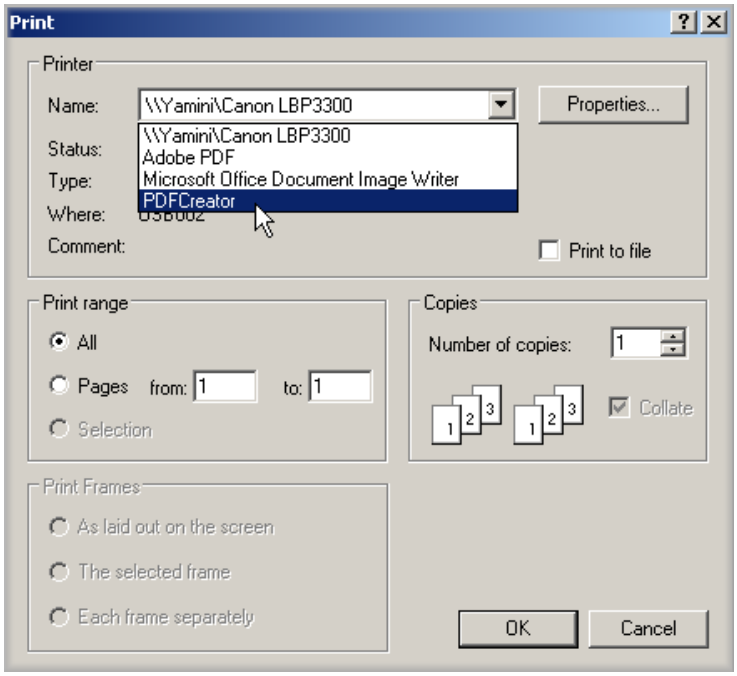


- You may print the report by clicking the **Print** in the test report window.

- Select your Printer in the Printer options.



- You can also save the report in PDF format by selecting the PDF creator in your Printer options.



- Close the window to return to the AC Impedance Test page.

- Repeat the above steps to conduct further tests as per your requirement.

Configuring Using Telephone

- Enter SE mode from a DKP/SLT.

  To run the AC Impedance Test using the Accurate Mode, dial:
  - **3361-1-CO Port Number**
    where,
    CO Port Number is 001 to 128.

  You can view the details of the test report on the port configured as the **Syslog Server IP Address: Port** on the System Debug page.

  To view the report, dial:
  - **3362**

Alarms

What's this?

Alarms are an efficient and user-friendly feature available to all extensions of the SARVAM UCS.

Alarms can be set and canceled

- by the Operator for extension users.
- by the extension users for themselves.

Alarms can be set as:

- *Once Only* - A one-time call, where the extension phone rings at the set time.
- *Daily* - A repeat call, where the extension phone rings at the set time everyday.

Alarms can be served as:

- *Personalized* - The Operator greets the extension user to serve the alarm request.
- *Automated* - The system serves the alarm request by playing a voice message or music.

How it works

Personalized Alarm

When the Alarm serving mechanism is configured as 'Personalized',

- The Operator phone rings first[239], displaying the number of the extension to which the alarm is to be served.
- When the Operator answers this call, a call is placed on the extension on which the alarm is set.
- The extension rings for the duration of the Alarm Ring Timer.
- When the extension user answers the call, the Operator greets the extension user with the time and alarm message.
- This event is recorded in the Hotel-Motel Activity Log as 'Wake-up Alarm of <HH:MM> Answered on <phone number>'.
- If the extension user does not answer the call till the Alarm Ring Timer has elapsed, the Operator phone will display a text message notifying 'No Reply' from the extension. The Alarm is now considered as served.

239. *The Operator phone rings for the duration of the Alarm Ring Timer. If the Operator does not answer the call, the system will make two more Alarm Attempts at an Alarm Attempt Interval of 5 minutes to call the Operator.*

- This event is recorded in the Hotel-Motel Activity log as 'Wake-up Alarm of <HH:MM> No Reply on <Phone Number>'.

- If the extension is busy[240], the Operator phone will display a text message notifying that the extension number is 'Busy'.

- The Operator can now choose to
  - inform the extension user about the alarm in person or send someone to do it.
  - try the busy extension again.
  - set “Auto Call Back (ACB)”.

Automated Alarm

When the Alarm serving mechanism is configured as 'Automated',

- The extension phone rings at the set time till the end of the Alarm Ring Timer. If the extension phone is from the EON series, an Alarm message will appear on its display.

- When the extension user answers the call, s/he may be played music-on-hold, or a pre-recorded voice message or be connected to a routing group, depending upon the Alarm Notification Type programmed by the System Engineer.

  The System Engineer may consult with the Enterprise to decide which of these options is to be programmed as the Alarm Notification Type.

- If the extension user does not answer the alarm call, the system makes two more attempts (in all, 3 attempts) at an interval of 5 minutes between each attempt, to call the extension. (Each attempt is recorded in the Hotel-Motel Activity log as 'Wake-up Alarm of <HH:MM> No Reply on <Phone Number>'.

- If all Alarm attempts go unanswered, the system places the call on the Operator phone. The Operator phone rings till the end of the Alarm Ring Timer. The Operator phone displays the extension number with the message 'No Reply'. The Alarm is now considered as served. (This event is recorded as "Alarm Notification to Front Desk for <Phone Number>").

- If the extension phone is busy the system will continue to make Alarm Attempts at the Alarm Interval programmed. When all Alarm Attempts go unanswered, the system will place a call on the Operator phone. The Operator phone will display the number of the extension phone with the message 'Busy'.

Snooze

The Snooze function can be added to Automated-Alarms to ensure that the extension user answers the call. Snooze is a system-wide feature; when set, this function will be added to all Automated Alarms.

When Snooze is activated,

- The extension phone rings for the Number of Alarm Attempts programmed, at set Alarm Attempt Intervals.

- The extension stops ringing, when the extension user answers the call and dials the Code '0' to acknowledge the Alarm. Please note that this Alarm Acknowledgement Code is non-programmable.

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240. An improperly placed receiver may also be the cause for the busy tone on the extension phone. In that case, the system will notify the Operator Phone with the 'OFF-Hook Alert'. This event is recorded in the Hotel-Motel Activity Log as "Alarm not Served, <phone number> is Busy".

Consider you have set an alarm with snooze enabled and Number of Alarm Attempts set as three (configurable). If this alarm call is not acknowledged by the extension user at the first alarm attempt and due to some reason, the system restarts, then the pending two alarm attempts will not be served. However, this alarm will be displayed under the pending alarm list.

The system now considers this as a new alarm and will serve the same on the next day at the same time. Also, the number attempts made by the system will be as per value configured in the parameter ***Number of Alarm Attempts****, provided it is not acknowledged by the extension user.*

Alarm Status Report

The Operator can know the status of Alarms (details of Alarms that have not been served) from the WakeUp Alarms Reports from SA mode or by pressing the DSS key assigned to Wakeup Call Log.

The WakeUp Alarms Report is useful when Operators change shifts.

Using SA Mode

The Jeeves displays the status of Alarms set by Operator as well as extension users appears on this report, with details of time (hours and minutes), type (once only, daily), and serving mechanism (personalized, automated). The Alarm Report generated by the system can be printed or sent to a computer.

Using DSS Key

The DKP / Extended IP Phone users can also view the Wakeup Call Log Status on the phone LCD using the DSS key assigned to Wakeup Call Log. For instructions to assign a DSS Key to Wakeup Call Log, see “Configuring Alarm Parameters using Jeeves”.

The Wakeup Call Log[241] is the log of:

- Unanaswered Alarm Calls - This log will display both Alarms Calls that have not been answered as well as unacknowledged Alarm Calls.
- Pending Alarm Calls - This log will display Alarm Calls which have been set for a later time and/or date.
- Served Alarm Calls - This log will display Alarm Calls which have already been served.

You can check Alarms set (pending), served and unanswered for last 25 hours. Altogether maximum 500 entries will be displayed. Each Alarm Call will display the details of time (hours and minutes), date and type (once only, daily).

The LED of the DSS key assigned to Wakeup Call Log glows in Red to indicate Unanswered Alarm calls or it glows in Blue to indicate Pending Alarm calls.

If there are both, Unanswered and Pending Alarm calls, the LED of the DSS key will glow in Red. After you view the Unanswered Alarm Calls, the LED will glow in Blue to indicate Pending Alarm Calls. The LED will glow in Blue till all the Pending Alarm calls have been served. If any Daily Alarm has been set the LED of the DSS key will glow in Blue till the alarm is canceled.

241. The logs will also display the Reminders.

To view the log from any DKP / Extended IP Phone user,

- Press the DSS Key assigned to Wakeup Call Log.
- The phone displays the logs — Unanswered Alarm Calls, Pending Alarm Calls, Served Alarm Calls.
- Select the desired log to view the details in the respective log.
- The acknowledged call is removed from the Alarms Log and is logged into the System Activity Log with the details of the extension that acknowledged the call.

- *SARVAM UCS can register as many as 960 Alarm requests set by the Operator and extension users.*
- *Multiple Alarms can be set for an extension by the Operator and/or by the extension user. For example, Daily Alarm at 09:00am is set for an extension. The extension user wants to change the alarm time to 08:30am for a day. The extension user/Operator can set another alarm, that is, a Once Only Alarm, at 08:30am without disturbing the daily alarm. Both the Alarms will ring at the set time.*
- *When multiple alarm requests have been set on an extension, if the Operator/extension user cancels an alarm set for an extension, the system cancels all alarms set for the extension. It is not possible to cancel any of these alarms selectively.*
- *It is not possible to modify an alarm request. Instead, the alarm request should be canceled and a new one should be made.*
- *The duration of Alarm Ring Timer, the Number of Alarm Attempts and the Alarm Attempt Interval are programmable.*
- *Alarms can be set for all extensions of the system, including the Operator phone also.*
- *All the Alarm events are logged in the "Hotel-Motel Activity Log".*
- *Alarm settings will be retained in the system during power down and system upgrades.*

How to configure

The following parameters play an important role in the functioning of the Alarm feature. These parameters carry default values. The default values have been selected keeping the larger user base in mind. However, these values can be changed by the System Engineer at the time of installation or afterwards as per users' requirements.

1. **Alarm Ring Timer** - The duration for which the system rings the extension to serve an Alarm call. By default, the Alarm Ring Timer is set to 45 seconds. This timer can be set between 001 to 255 seconds. This timer also signifies the duration for which the Operator phone rings to notify that an Alarm call has not been answered or the extension phone is busy.
2. **Number of Alarm Attempts** - Number of times the system attempts to place an Alarm call on the extension phone before notifying the Operator that the call is not answered or the phone is busy. By default, the Number of Alarm Attempts is set to '3'. The Number of Alarm Attempts can be set between 1 and 9.

3. **Alarm Attempt Interval** - The time period between each Alarm Call attempt. By default, the Alarm Attempt Interval is set to 5 minutes. The Alarm Attempt Interval can be set between 1 and 9.

4. **Use Alarm with Snooze** - Snooze is a functionality which forces the extension user to acknowledge the Alarm call. With snooze enabled, the system expects the user to answer the Alarm call by going OFF-Hook and dial Acknowledgement code '0'. With snooze disabled, the system considers the Alarm as answered when the extension user simply answers the alarm call by going OFF-Hook (dialing acknowledgement code is not mandatory). Users may choose whether or not to enable snooze. By default, snooze is disabled.

5. **Configurable Alarm Type** - When the Operator and extension user set an Alarm call request, the system gives them the choice of setting 'Once Only' or 'Daily' Alarm calls.

   User experience however, shows that 'Once Only' Alarm call requests are more common than 'Daily' Alarm requests. So, the system allows you the flexibility of setting 'Once Only' as the default Alarm Type, by disabling the 'Configuring Alarm Type' flag.

   When this flag is disabled the system will prompt the Operator/Extension user to enter the Time of the Alarm call and consider the Alarm Type as 'Once Only'.

   By default, this flag is disabled.

6. **Configurable Alarm Category** - When the Operator sets an Alarm call for an extension, the system prompts the Operator to select an Alarm Type (Once Only or Daily) and to select the alarm serving mechanism - 'Automated or Personalized'.

   If the Enterprise wishes to offer only 'Automated' Alarms to its extension users, the system allows the flexibility to set 'Automated' as the default Alarm call serving mechanism. This can be done by disabling the 'Configurable Alarm Category' flag.

   When this flag is disabled, the system will consider the Alarm call serving mechanism as 'Automated' and will prompt the Operator only for the Time of the Alarm call.

   By default, this flag is disabled.

- *When both flags 'Configurable Alarm Type' and 'Configurable Alarm Category' are disabled, the system will set and serve 'Once Only - Automated' alarms only.*

- *If the 'Configurable Alarm Type' flag is disabled, but the 'Configurable Alarm Category' flag is enabled, the system will set 'Once Only' alarm calls, but give the option of selecting 'Automated' or 'Personalized' as the serving mechanism.*

- *Similarly, if 'Configurable Alarm Type' is enabled, but the 'Configurable Alarm Category' flag is disabled, the system will allow both 'Once Only' and 'Daily' alarms to be set, but the serving mechanism will be 'Automated'.*

7. **Voice Guided Alarm Verification:** For Voice-guided Alarms, the VMS of the system allows you to enable/ disable the Alarm Verification for alarms and reminders, allowing extension users who want to use alarms and reminders to confirm the Time set for an alarm and Date and time set as a reminder. By default, this flag is enabled.

The flags 'Configurable Alarm Type' and 'Configurable Alarm Category' are not applicable for Voice-guided Alarms. In the case of Voice-guided Alarms, the Operator/Extension user will be prompted to select the Alarm type and serving mechanism, each time, even when both aforementioned flags are disabled.

8. **Alarm Notification Type** - This is the means of notifying the extension user about the Alarm call. The extension user can be played Music-On-Hold, Live Music, Pre-recorded Voice Message, Weather information, Date and Time, etc. The system supports four types of Alarm Notifications:

    - *Voice Message:* Selecting this option would play a recorded message recorded to the extension user when s/he answers the Alarm call.

    - *Music-On-Hold:* Selecting this option would play music-on-hold to the extension user when s/he answers the Alarm call.

    - *Routing Group:* Selecting this option would connect the extension user to the stations programmed in the Alarm Notification Group. The System Engineer may connect a device which can play customized alarm greetings with date, time, weather conditions, traffic conditions, a marketing message, etc. on the stations programmed in the Alarm Notification Group.

If Voice Mail Auto Attendant is selected as a Routing Group member, the system will place the call on the Voice Mail System.

    - *Voice Mail:* Selecting this option would connect the extension user to the Voice mail System. Use this option only if you have the VMS module installed in the system.

9. **Macros** - This is a short code for simulating the Alarm call. The SLTs with special function keys send a fixed string to the system, when each function key is pressed. The system interprets this string and translates it into a string that can be understood by the system. For example, the SLT has a special function key for Alarm calls which sends the string '51' to the system. The system can be programmed to translate '51' into the feature access code for Alarm calls, '*161'.

    All the above listed parameters can be programmed using Jeeves and a Telephone.

Configuring Alarm Parameters using Jeeves

- Log in as System Engineer.

- To configure Alarm parameters, go to "System Parameters".

- To view the Alarm Report Status on your phone LCD, you must assign a DSS Key to Wakeup Call Log. Refer the topic "DSS Keys Programming", "DSS Key Settings" in "Configuring Matrix SPARSH VP330", "DSS Key Settings" in "Configuring Matrix SPARSH VP248", "DSS Key Settings" in "Configuring Matrix SPARSH VP310" and "DSS Key Settings" in "Configuring Matrix SPARSH VP510" for instructions.

- To select Alarm Notification Type for extensions, under *Configuring Extensions*, see"Station Advanced Feature Template".

    - If you select Voice Message as Alarm Notification type, ensure that you assign a voice module to 'Alarm' voice message application. Please refer topic "Voice Message Applications" for more details.

    - If you select Music-On-Hold as Alarm Notification type, no further configuration is required.

- If you select Voice Mail as the Alarm Notification Type, make sure you have installed the Voice Mail module.

- If you have selected Routing Group as Alarm Notification Type, you must create a Routing Group and assign this Routing Group number in the Station Advanced Feature Template of the extensions. See "Routing Group" and "Station Advanced Feature Template" for instructions on applying the template to SLTs, DKPs, ISDN Terminals, Virtual Extensions.

- To program SLTs with special Alarm function key and to create macro for a DKP key, see "Macros".

- To use a customized alarm messaging device, see "Configuring Customized Alarm Messaging Devices"

Configuring the Alarm Parameters using a Telephone

- Enter SE mode from a DKP/SLT.

  To configure Alarm Ring Timer, dial:
  - ***2201-Seconds***
    Where,
    Seconds is from 001 to 255. Default: 45 seconds.

  To configure Number of Alarm Attempts, dial:
  - ***2202-Number of Alarm Attempts***
    Where,
    Number of Alarm Attempts is from 1 to 9. Default: 3.

  To configure Alarm Attempt Interval, dial:
  - ***2203-Alarm Attempt Interval***
    Where,
    Alarm Attempt Interval is from 1 to 9 minutes. Default: 5 minutes.

  To configure Snooze function, dial:
  - ***2204-Snooze***
    Where,
    Snooze is 0 or 1.
    Select '0' to disable Snooze and '1' to enable snooze.

  To disable/enable Configurable Alarm Type, dial:
  - ***2208-Flag***
    Where,
    Flag is 0 for Disable, 1 for Enable1. Default: Enabled.

  To disable/enable Configurable Alarm Category, dial:
  - ***2209-Flag***
    Where,
    Flag is 0 for Disable, 1 for Enable. Default: Enabled

To configure Alarm Notification Type, dial:

- ***5602-1-Template Number-12-Alarm Notification Type***
  Where,
  Template Number is Station Advanced Feature Template from 01 to 50.
  Default: 01.
  Alarm Notification Type is:
  1 for Music on Hold
  2 for Voice Message (Voice Modules)
  3 for Routing Group
  4 for Voice Mail
  For example: To program Routing Group as Notification Type in Station Advanced Feature Template 02: Dial ***5602-1-02 -12-3***

  To apply the Station Advanced Feature Template now programmed with the Alarm Notification Type to extension phones, refer the topic "Customizing Station Advanced Feature Template using a Telephone".

  To program Macros, dial the following commands:
- Dial command ***1810-Macro Index-Number String*** to create a macro.
  Where,
  Macro Index is from 01 to 25 (as 25 macros can be created)
  Number String is the feature access code for Alarm, in this case '*163'
  Terminate the command with #* as the number string is less than 24 digits.
  For example: to create a macro for Alarm on Index 2: dial ***1810-02-*163-#****

- Dial command ***1810-Macro Index-#**** to clear a macro.

- Dial command ***3115-1-Macro Index-Access Code*** to program Access code (that is, number sent to the system by the SLT)
  Where,
  Macro Index is from 01 to 25 Access Code is a string of 4-digits.
  If the length of the Access code is less than 4-digits terminate the command with #*
  For example: To program access code '53' for the macro for Alarm dial: ***3115-1-02-53-#****

- Dial command ***3115-1-Macro Index*** to clear the Access code for the macro.

- Exit SE mode.

Configuring Customized Alarm Messaging Devices

Alarm Messaging devices play real-time updated information like the date and time, greetings, weather information, Road highway status, specific event announcements, etc. when called upon. These devices should be connected to the system on its SLT ports.

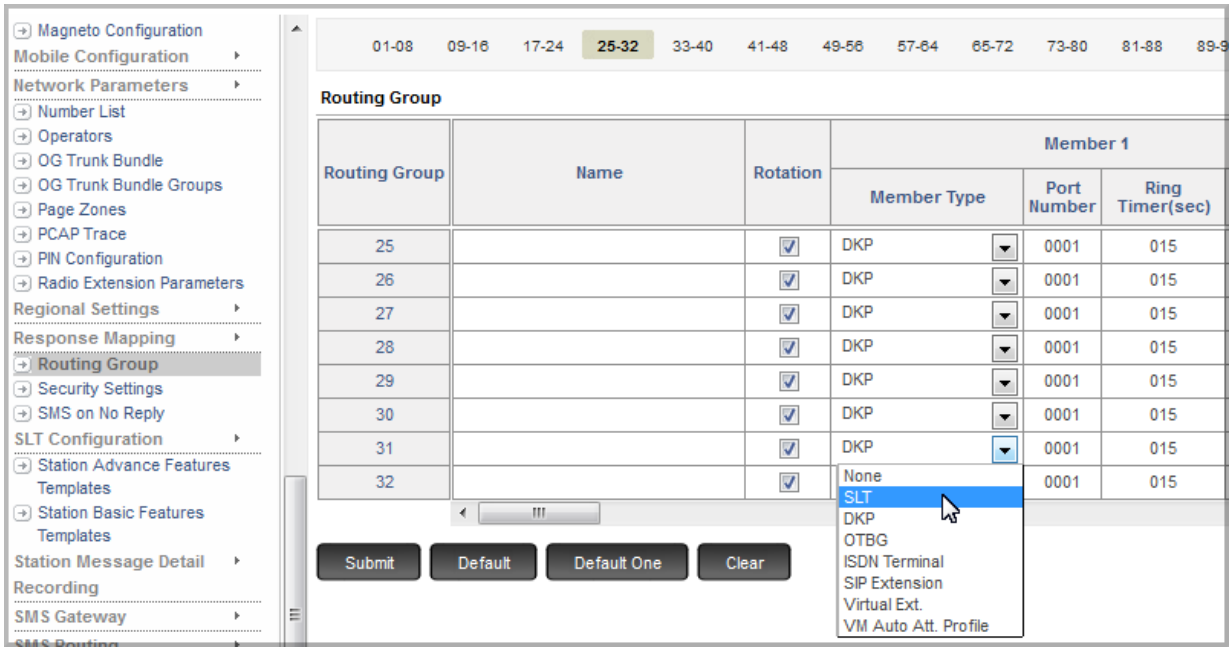
If the User wants to connect customized alarm messaging devices to the SARVAM UCS, the SE should configure the system as instructed below:

- Connect the devices for customized alarm greetings to SLT ports only.

- Log into Jeeves as System Engineer.

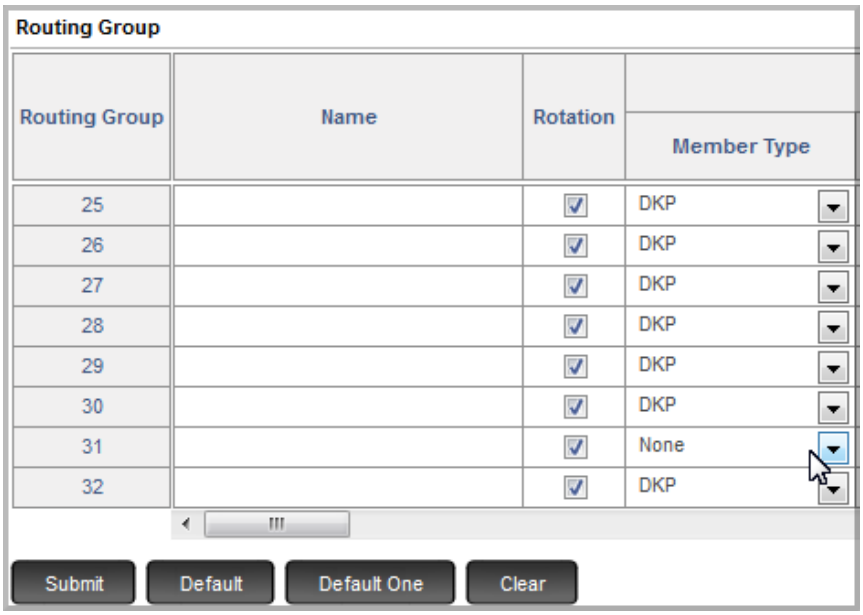
- Under **Configuration**, click **Routing Group** to open the page.

- Select the default Routing Group 31 used for Alarm Notification Group, or any other Routing Group number.



- Select **SLT** as the **Member Type** and enter the SLT **Port Number** where the device is connected. It is possible to configure 32 members in a single routing group.

  If only one device is connected, disable all other members from 02-32 by setting **Member Type** to **None**.



- Click **Submit** at the bottom of the page to save changes.

- Open the **Station Advanced Feature Template** page. By default Template Number 01 is applied to all stations. The template has **Voice Message** as default notification type. It is recommended that you program another Template.

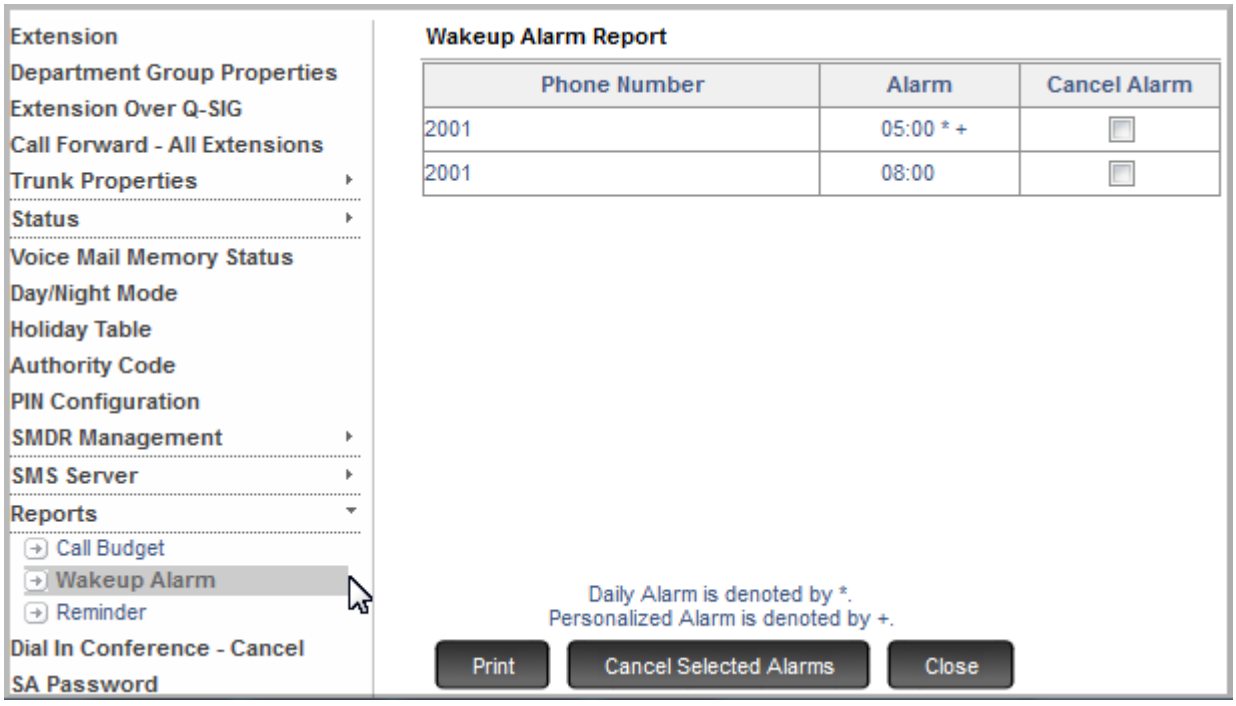
- Select **Routing Group** as the Alarm Notification Type in the template you have selected for configuration.

- Enter the number of the **Alarm Notification Routing Group** (default group: 31) in which you have programmed the device (SLT port).
- Click **Submit** at the bottom of the page to save changes.
- Apply the Advanced Feature Template now configured with Routing Group as Alarm Notification Type and the number of the Alarm Notification Routing Group to the stations.
- Refer the section "Station Advanced Feature Template" for instructions on applying this template to extension phones.

Viewing and Printing Alarm Reports

The Operator can view the status of Alarms that are yet to be served from the System Administrator mode and if required print them.

- Log in as System Administrator.
- Click the **Reports** link.
- Under **Reports**, click **Wakeup Alarm**.



- The unserved Alarm calls will appear on your screen.
- You can cancel any of the unserved Alarm calls by selecting the check-box and clicking the **Cancel Selected Alarms** button on this page.
- You can also print this page by clicking the **Print** button on this page.
- Click **Close** to exit the page.
- Click **Submit** to save changes.

How to use

Alarms can be set by the extension users by themselves. The extension users can also ask the Operator to set the alarm for them.

Alarms set/canceled by Operator

The Operator can set/cancel non-voice guided Alarms using EON, SLT and from System Administrator mode.

Operator using EON

Using DSS Key:

To set Alarm for the extension user,

- Press the key assigned the 'Remote Alarm' function.
- Enter the Extension Number.
- Enter Time in HH:MM
- Select 'Once Only' or 'Daily'.
- Press 'Enter' key.
- Select 'Personalized' or 'Automated'.
- Press 'Enter' key to set Alarm.
- You get a confirmation tone and a text message with the phone number for which the alarm is set.
- Go Idle or you get dial tone after 3 seconds.

To cancel Alarms,

- Press key assigned the 'Remote Alarm' function.
- Enter Extension Number.
- Select 'Cancel All'.
- Press 'Enter' Key.

Using Commands:

To set Alarm for the extension user,

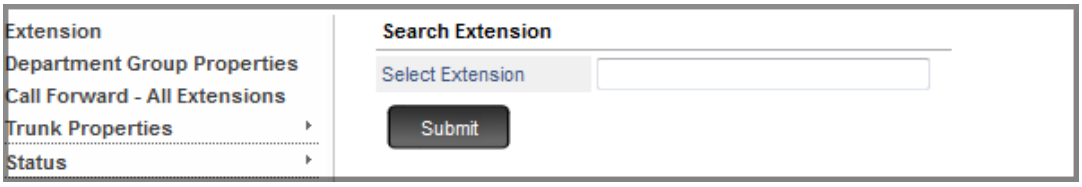
- Pick up the handset.
- Dial ***1072-003.***
- Enter Extension number.
- Enter Time in HH:MM
- Dial 1 for Once Only or Dial 2 for Daily
- Dial 1 for Personalized or Dial 2 for Automated.
- Press 'Enter' key to set Alarm.
- You get a confirmation tone and a text message with the phone number for which the alarm is set.
- Replace handset or you get dial tone after 3 seconds.

To cancel Alarms,

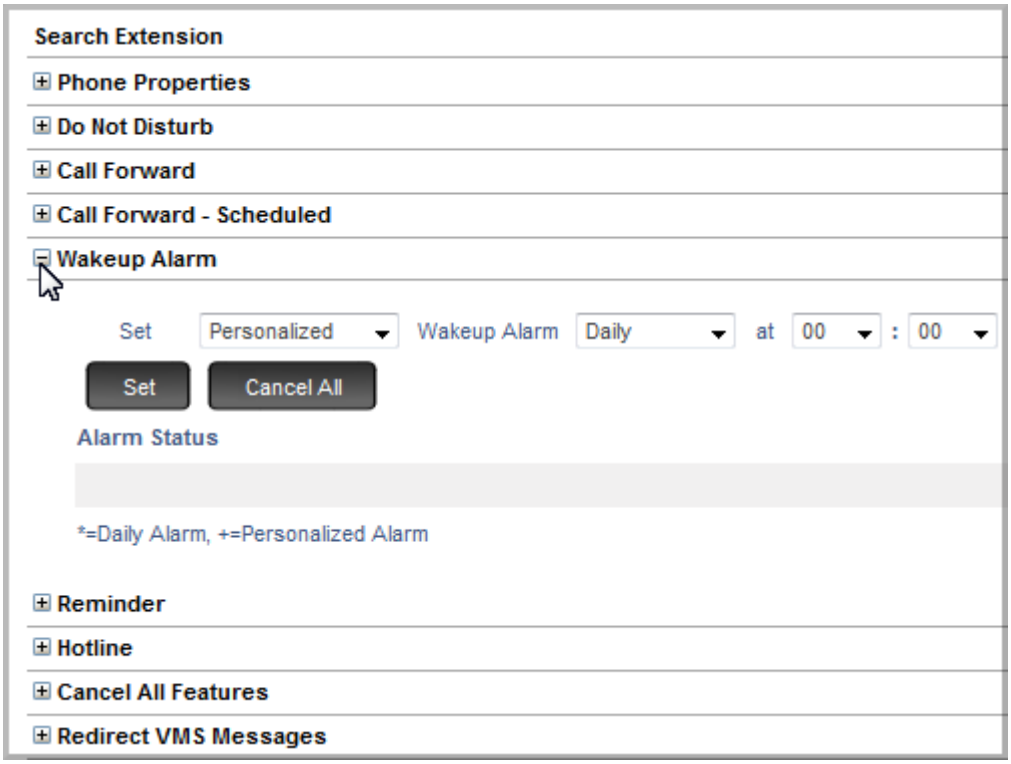
- Pick up the handset.
- Dial ***1072-003.***
- Enter Extension Number.
- Dial #.
- You get a confirmation tone and a text message with the phone number for which the alarm is canceled.
- Replace handset or you get dial tone after 3 seconds.

Using Jeeves:

- Log in as System Administrator.
- Click the **Extension**.



- In **Select Extension,** enter the Number or the Name of the extension on which you want to set the alarm.
- The page of the Extension will open.



- Now set the desired type of alarm on this extension.
- Click **Submit** to save.
- Repeat the same to set alarm on another extension number.
- Click **Logout** to exit the System Administrator mode.

Alarms set/cancel by Extension Users

Extension Users using EON / Extended IP Phone

If the extension user has EON, alarms can be set using the DSS key as well as by dialing the command.

Using DSS Key:

To set Alarm,

- Press the key assigned the 'Alarm' function.
- Enter Time in HH:MM
- Select 'Once Only' or 'Daily'.
- Press 'Enter' key.
- You get a confirmatory text message and confirmation tone.
- Go Idle or you get dial tone after 3 seconds.

To cancel Alarms,

- Press the key assigned the 'Alarm' function.
  OR
- Dial 161
- Select 'Cancel All'.
- Press 'Enter' Key.

Dialing Commands:

To set Alarm,

- Pick up the handset.
- Dial ***161.***
- Enter Time in HH:MM (24-hours format)
- Dial 1 for Once Only or Dial 2 for Daily.
- Press 'Enter' key.
- You get a confirmatory text message and confirmation tone.
- Replace handset or you get dial tone after 3 seconds

To cancel Alarms,

- Pick up the handset.
- Dial ***161.***
- Dial #.
- You get a confirmatory text message and confirmation tone.
- Replace handset or you get dial tone after 3 seconds.

Extension Users using SLT

To set Alarms,

- Pick up the handset.
- Dial ***161.***
- Dial HH:MM
- Dial 1 for Once Only or Dial 2 for Daily.
- You get confirmation tone.
- Replace the handset on the cradle.

To cancel Alarms,

- Pick up the handset.
- Dial ***161***.
- Dial #.
- You get confirmation tone.
- Replace the handset.

- *Extension users can set only automated alarms from their phones. For personalized alarms, they must request the Operator.*
- *If there are multiple alarms set, alarms cannot be canceled selectively. Only the Operator can cancel alarms selectively from SA mode.*
- *Alarms set on an extension will be served, even if DND is also set on the same extension.*

Printing Alarm Reports

When Scheduled Alarm Report is enabled and the time is set, the system will automatically print the report at the set time. The Operator can also print Alarm Status Report any time s/he using SA commands.
The Operator can issue SA commands from EON or an SLT to print Alarm Status Reports.

For EON and Extended IP Phone Users

- Press the key assigned the 'Print Alarm Report' function (if programmed).

  OR

- Dial ***1072-913.***
- You get a confirmatory text message and a confirmation tone.
- Go idle.

For SLT Users

- Pick up the handset.
- Dial ***1072-913.***
- You get confirmation tone.
- Replace the Handset on the cradle.

```
               Wakeup/Alarm Report         AS ON 05-05-2016(Thu) AT 23:49
--------------------------------------------------------------------------------
  Room#   Phone#  Wakeup D  P                    Room#   Phone#  Wakeup D  P
--------------------------------------------------------------------------------
          3001    12:18                                2017    14:12
          3001    12:21                                2017    12:14
--------------------------------------------------------------------------------
 * indicates Daily Alarm and + indicates Personal Alarm            Page : 1
                      ---End of Report---
```

Alternate Number Dialing

What's this?

Alternate Number Dialing allows you to dial different phone numbers in an attempt to reach a person whose line is busy.

Alternate Number Dialing is useful when the person or organization you are trying to reach has more than one number, where they may be reached. The system dials out different phone numbers of the same party, saving you time and effort of dialing each of these numbers manually.

How it works

This feature works as an extension of the features “Last Number Redial” and “Auto Redial”. It requires you to program Alternate Number Groups in the Global Directory first. With the alternate numbers programmed in the Global Directory, all you need to do is to use Last Number Redial or Auto Redial, every time you want the system to try Alternate Number Dialing.

For example: Midas Business Solutions has four telephone numbers: 2640459, 2631235, 2635589 and 2565590. To be able to use Alternate Number Dialing, you must first program all four numbers as Alternate Number Group in the Global Directory.

Now, when you dial one of these numbers, '2640459', and get a busy tone, you can either initiate Last Number Redial or set an Auto Redial request.

When you initiate Last Number Redial,

- The system will dial an alternative number for the dialed number.
- If the redialed number is busy, you can set Last Number Redial again.
- The system will dial a second alternative number.
- If the second alternative number is also busy, you can set Last Number Redial again.
- The system will dial a third alternative number.
- This process will be repeated each time you set Last Number Redial, until the call gets through.

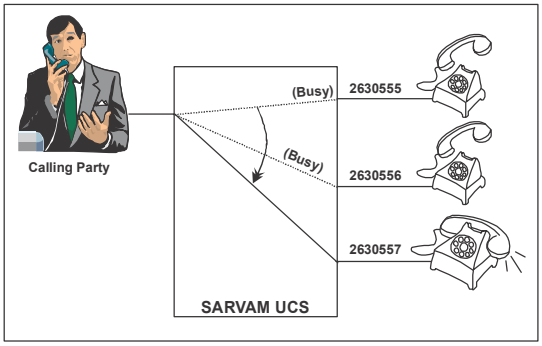
When you set an Auto Redial request on busy tone,

- The system will dial an alternative number.
- If the alternative number is busy, the system will redial another alternative number.
- The system will dial a different (alternative) number on each auto redial attempt[242], until the call gets through.

242. *The number of auto redial attempts depends on the Auto Redial Count programmed in the system. By default, the system will make 5 redial attempts if Auto Redial 'normal' is set. If Auto Redial 'Priority' is set, the system will make 20 redial attempts.*

- (for the number of redial attempts programmed), until the call gets through.
- when any of the alternate numbers gets through, the system will give a ring on your extension.

This process is illustrated in the graphic below:



- *Alternate Number Dialing will work only on extensions that are allowed the features "Last Number Redial" in their "Class of Service (COS)"*
- *Also, Alternate Number Dialing will work only for those numbers that exist in the Global Directory assigned to each extension. The Global Directory is divided into three parts[243], 100-799 (Part 1), 800-899 (Part 2), and 900-999 (Part 3). If an extension is assigned only Global Directory Part 2, Alternate Number Dialing will work only for those numbers grouped as Alternate Number Groups in Global Directory Part 2.*
- *Alternate Number Dialing will work also with "Abbreviated Dialing". For example, an extension user dials the abbreviated code 8100, and the dialed out number is busy. When the extension user sets Redial or Auto Redial, the system will try the alternate numbers related to 8100.*

How to configure

For Alternate Number Dialing to work, the System Engineer must:

1. Make a List of Alternate Numbers.
2. Create Alternate Number Groups.
3. Program Alternate Number Groups in the Global Directory.
4. Enable the features 'Last Number Redial', 'Global Directory', in the Class of Service (CoS) group of the extensions to which Alternate Number Dialing facility is to be provided. If desired, 'Auto Redial', 'Auto Redial Priority' may also be enabled in the CoS of these extensions.

All of the above parameters can be programmed using Jeeves or dialing SE Commands from a telephone.

243. *ETERNITY LENX/MENX stores up to 2900 numbers in the Global Directory; Part 1 - 100 to 2399, Part 2 - 2400 to 2699 and Part 3 - 2700 to 2999*

- *To create Alternate Number Groups, the alternate numbers must exist in the Global Directory. If any of the alternate numbers do not exist in the Global Directory, first program the numbers in the directory, before you begin creating Alternate Number Groups. Refer the topic "Abbreviated Dialing" for instructions on programming the Global Directory.*

- *As Alternate Number Dialing works only for the Alternate Number Groups in the Global Directory assigned to each extension, ensure that the relevant Global Directory with the Alternate Number Groups is allowed in the CoS of the extensions.*

Preparing Alternate Number List

In consultation with the User, you may:

- Draw a two-column table on a paper.
- Write the name of the contact on one column and the Alternate Numbers for the contact on the other column.
- Make a list of the numbers which need to be grouped as alternate numbers. For example:

Name of the Contact	Alternate Numbers
Midas Business Solutions	2640459, 2631235, 2635589, 2565590
Jet Set Holidays	022281110001, 022281110002
Bacchus Vineyard	2640075, 2640076
GoodLife Inn	2788856, 2788896

You may include as many alternate numbers as required by the User.

Creating Alternate Number Groups

- Assign the alternate numbers of each of contacts to an 'Alternate Number Group'.
- Each group must be assigned a number between 000 to 255.
- Taking the above example further, the Alternate Number Groups on the list may be numbered as follows:

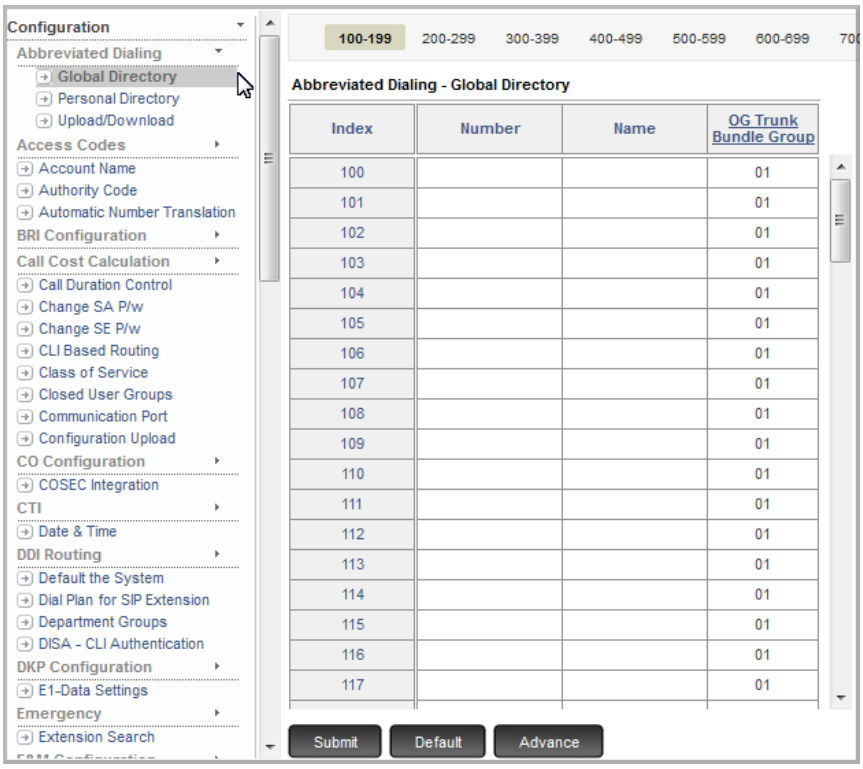
Name of the Contact	Alternate Numbers	Alternate Number Group (No.)
Midas Business Solutions	2640459, 2631235, 2635589, 2565590	001
Jet Set Holidays	022281110001, 022281110002	002
Bacchus Vineyard	2640075, 2640076	003
GoodLife Inn	2788856, 2788896	004

Programming Alternate Number Groups in Global Directory using Jeeves

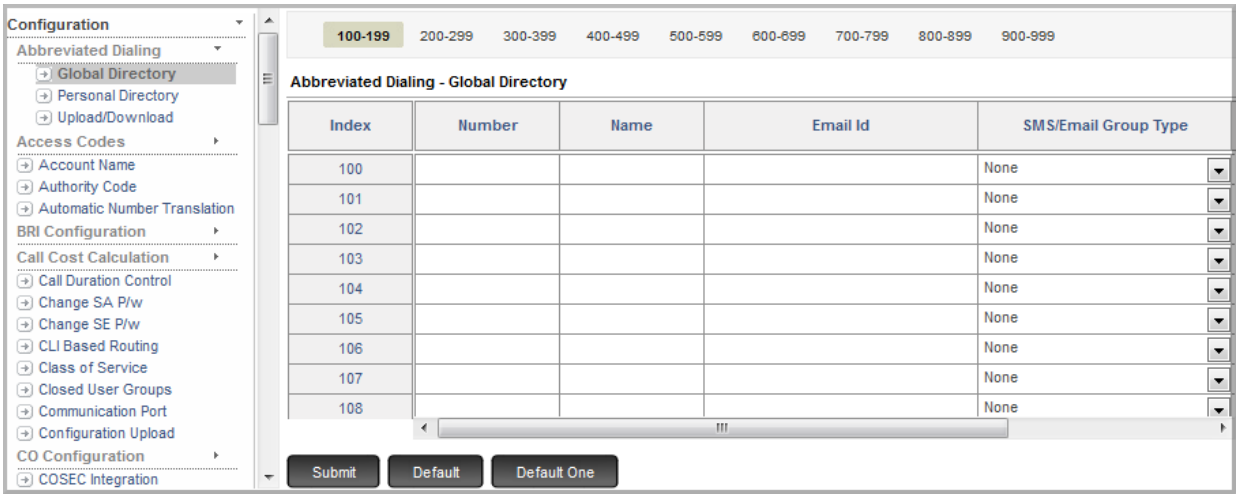
Make sure that the numbers on the list are also programmed in the Global Directory. If any of these numbers do not exist in the Global directory already, program them in the Global Directory first. Refer the topic “Abbreviated Dialing” for instructions on programming the Global Directory.

To create Alternate Number Groups and program them in the Global Directory,

- Log in as System Engineer.
- Under **Configuration**, click **Abbreviated Dialing**.
- Click **Global Directory** to open the page.



- Click the **Advance** button on the bottom of this page.



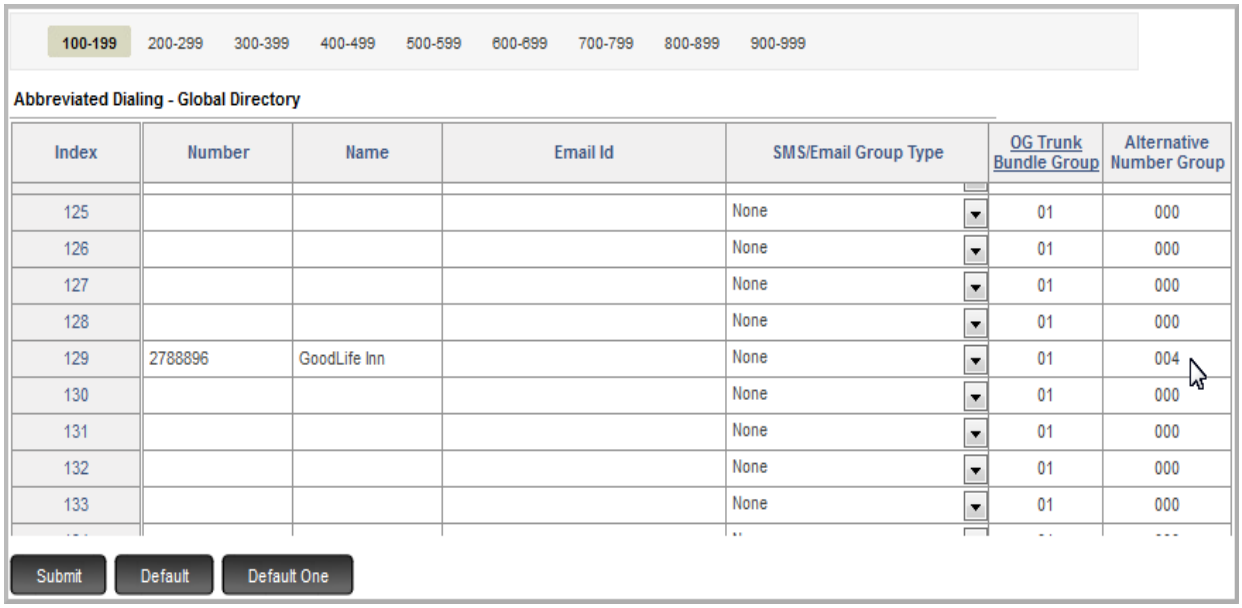
- Enter the number of the **Alternate Number Group** in the last column of the page.

  For example, you have assigned Alternate Number Group '001' to all the numbers of the contact Midas Business Solutions, enter this number against each number belonging to this contact.

  Similarly, enter Alternate Group number '004' against the numbers belonging to the 'GoodLife Inn' to which it is assigned.

Memory Location	Outgoing Trunk Bundle Group	Number	Name	Alternate Number Group
100	01	2640459	Midas Biz	001
101	01	2631235	Midas Biz	001
102	01	2635589	Midas Biz	001
103	01	2565590	Midas Biz	001
104	01	2788856	GoodLife Inn	004
105	01	022281110001	Jet Set	002
106	01	022281110002	Jet Set	002
107	01	033298765432	R. Mendez	000
108	01	2640075	Bacchus	003
109	01	2640076	Bacchus	003
:	:	:	:	:
129	01	2788896	GoodLife Inn	004

The numbers of the contacts may not necessarily appear alphabetically or in a sequence. It is possible that the numbers of the same contact may be programmed at different memory locations in the Global Directory.



In the above example, one number of the GoodLife Inn is programmed at memory location Index 104 and the other on Index 129. Since these two numbers are grouped and assigned the number alternate group

number '004', this number must be entered against the GoodLife Inn numbers at the respective memory location Index.

- After assigning Alternate Number Groups, click **Submit** at the bottom of the page to save changes.

- Enable the features **Last Number Redial** and **Global Directory**, in the Class of Service (CoS) group of the extensions to which Alternate Number Dialing facility is to be provided. If desired, **Auto Redial**, **Auto Redial Priority** may also be enabled in the CoS of these extensions.

  By default Station Basic Feature Template Number 01 is assigned to all extensions of the system. The default CoS Group 01 in this template has 'Redial' enabled in the set of 'Basic Features', so all extensions of the system can use Last Number Redial.

  However, the default CoS Group 01 has only Global Directory Part 1 enabled.

  Recall that Alternate Number Dialing will work only for those numbers that exist in the Global Directory assigned to each extension. So, the Global Directory Part containing the Alternate Number Groups must be allowed to the extensions in their Class of Service. For example, if Alternate Number Groups are programmed in Global Directory Part 2, extensions must have Global Directory Part 2 in their Class of Service.

  If all extensions are to be allowed the Alternate Number Dialing facility, simply enable the Global Directories containing Alternate Number groups in the default CoS group 01.

  However, if Alternate Number Dialing is to be allowed to select extensions only, define a new CoS group and prepare a new Station Basic Feature Template with this CoS group and apply it to the desired extensions.

  Refer the topics “Class of Service (COS)” and “Station Basic Feature Template” for detailed instructions.

The Station Basic Feature Template 01 does not have the features Auto Redial and Auto Redial Priority in the default CoS group 01. If these features are also to be allowed to the extensions, enable them in the CoS you prepare.

Programming Alternate Number Groups in Global Directory using a Telephone

- Enter SE mode from a DKP/SLT.

  To assign Alternate Group Numbers, dial:
  - ***1804-1-Memory Location Code-Alternate Group Number***
    Where,
    Memory Location Code = 100 to 999. This is the Index number, where the number and name of the contact is programmed.
    Alternate Group Number = 000 to 255.

    For example: To assign the numbers of the GoodLife Inn to an Alternate Number Group, dial:
    1804-1-104-004

    The number '2788856' stored at Index 104 will be assigned to Alternate Number Group 004, which is the number assigned to GoodLife Inn.

    Similarly, to program the second number dial: ***1804-1-129-004***

    The number '2788896' stored at Index 129 will be assigned to Alternate Number Group 004.

If you have a continuous sequence of numbers that need to be programmed in the same group, you may dial:

- ***1804-2-Memory Location Code-Memory Location Code-Alternate Group Number***
  Here you enter a sequence of Memory Location Codes, from 100 to 999, and the number of the Alternate Number Group.

  For example: To assign the numbers of Midas Biz to an Alternate Number Group, dial: ***1804-2-100-103-001***
  The numbers stored at Index 100 to 103 will be assigned to Alternate Number Group 001, which is the number assigned to Midas Biz.

To clear the Alternate Number Groups, dial the following commands:

- ***1804-1-Memory Location Code-000*** to clear group of a single number.
- ***1804-2-Memory Location Code-Memory Location Code-000*** to clear group of a continuous sequence of numbers.
- ***1804-*-000*** to clear all Alternate Number Groups

  For example: To clear the Alternate Number Group assigned to a number of GoodLife Inn, dial: ***1804-1-104-000***

  The Alternate Number Group assigned to the GoodLife Inn number '2788856' stored at Index 104 will be cleared.

  For example: To clear the Alternate Number Group assigned to the sequence of numbers of Midas Biz, dial: ***1804-2-100-103-000***

  The Alternate Number Group assigned to the Midas Biz numbers stored in a continuous sequence starting from Index 100 to 103 will be cleared.

- Exit SE mode.

How to use

Confirm with your System Engineer that

- Alternate Number Groups are programmed in the Global Directory allowed to your extension.
- 'Basic Features' (these include Redial) are enabled in the Class of Service allowed to your extension.

Now, follow the instructions for using the feature “Last Number Redial”.

Authority Codes

What's this?

Authority Code is a unique password-protected code with an associated Class of Service, Toll Control and Call Budget, which can be assigned to extension users. With Authority Codes, extension users of the system can make calls or access features from any other extension of the system as per the Class of Service, Toll Control and other features / facilities assigned to their code.

This feature is useful when you want a group to extension users to use a single extension, but at the same time you want to keep an account of the calls made by each user. If required, you can assign a call budget to each Authority code to control call cost.

To make outgoing calls or access features, extension users must 'Walk-In' from any extension port: DKP, SLT, SIP, ISDN Terminal, E&M (with Station as Orientation Type), and then dial their Authority Code and Password.

How it works

An 'Authority Code' is a unique three-digit number, protected by a four-digit password. The default Authority Password is ***1111***. To be able to use an Authority Code, the password must be changed to another value. To avoid unauthorized access, we recommend you to change the password regularly. Make sure it is strong and is kept confidential.

The SARVAM UCS supports as many as 999 Authority Codes.

Each Authority Code has an associated Class of Service and Toll Control, which is configured in the Station Basic Feature Template assigned to the Code.

To make calls using an Authority Code,

User A is assigned 222 as Authority Code and is provided a unique 4-digit Authority Password.

To access features or make calls as per the assigned Authority Code 222, User A must do the following:

- Dial the feature code for Walk-In Class of Service (default 111) from any extension of the system.
- Dial the code for Walk-In by Authority followed by the Authority Code and then the Authority Password.

  To make calls or access features according to the Authority Code dial 111, the feature code for 'Walk-In Class of Service' from any extension of the system.
- Select the option ‘Walk in by Authority Code’ or dial ‘2’.
- A dials the Authority Code assigned to him/her followed by the Password.

  If the user enters a wrong password, to prevent any attempt to misuse the Authority Code, by default, the system allows the user only three attempts to re-enter the password. If the user fails to enter the correct password at the third attempt, the system will set the password to default, 1111. With the default password the user will not be able to use the Authority Code.

The number of attempts to re-enter can be configured by setting the parameter **Retry counts for Authority Code** password in the “System Timers and Counts”.

- User A can now make outgoing calls.
- If the Walk Out mode set for A is One Call A will automatically be logged out from the extension after one call.
- If the Walk Out mode set for A is Multiple Calls, A can make as many calls as desired, and remains 'walked-in' until the A dials the feature code to 'Walk-Out', or until another extension user walks into the same extension.
- If a call budget has been assigned to A’s Authority Code, A will be able to make calls till the assigned amount is consumed.
- Details of the calls made by A are recorded by the Authority Code in the Station Message Detail Recording Report (SMDR) for Outgoing Calls.
- The SMDR report can be printed using the Authority Code as filter.

  This way, the organization can know the details of calls made by each user as well as have control over the expenses

How to configure

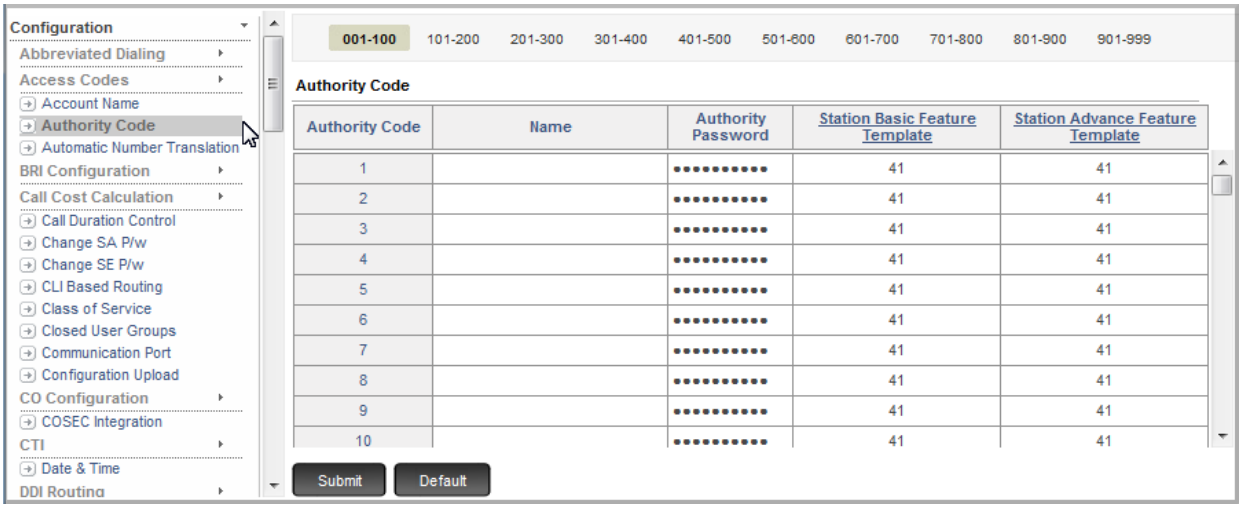
For Authority Codes to work, you must:

- Configure the Authority Code Table.
- Assign a Station Basic Feature Template to the Authority Code, with the desired Class of Service, Toll Control and features/facilities.
- Enable the parameter *Store Outgoing Calls* in the Station Basic Feature Template assigned to the Authority Code, if you want details of outgoing calls made using Authority Codes.
- Assign a Station Advanced Feature Template to the Authority Code with the desired Walk-Out Mode
- Assign each user a Call Budget, if required.
- Change the Retry Counts for the Authority Code, if required.

Configuring Authority Code Table

- Make a list of users you want to assign Authority Codes to and their respective passwords.
- Log in as System Engineer.

- Under **Configuration**, click **Authority Code**.



Against each **Authority Code**,

- Assign an **Authority Password**. The password can be a minimum of 4 digits and a maximum of upto 10 digits. Valid digits are 0 to 9, * and #. Default: 1111.

  To avoid unauthorized access, we recommend you to change the password regularly. Make sure it is strong and is kept confidential.

- Enter the **Name** of the user. The name acts as an identifier. Default: Blank.

- By default, **Station Basic Feature Template** number 41 and **Station Advanced Feature Template** number 41 are assigned to all Authority codes.

  If you want to change the calling permission, allow/deny features to users, you must customize the Class of Service, Toll Control and other features/facilties in the Station Basic Feature Template according to your requirement.

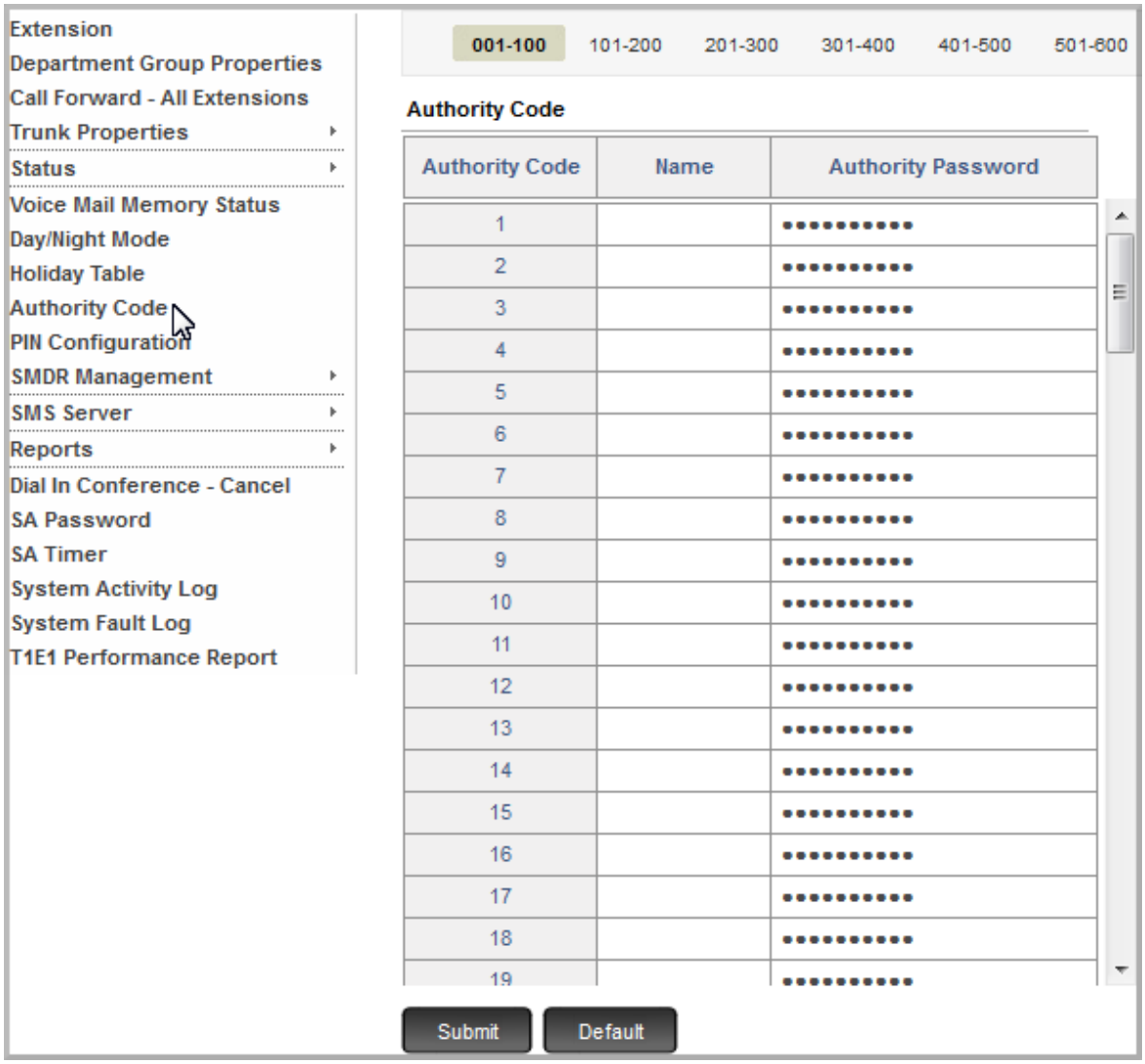
  You may set the 'Walk-Out Mode' in as One Call or Multiple Calls in the Advanced Feature Template. Refer to “Station Basic Feature Template” and “Station Advanced Feature Template” for detailed instructions.

- Click **Submit** to save your settings.

You can assign/change the Name and Password from the SA mode also. To do this,

- Log in as System Administrator.

- Click **Authority Code**.



Against each **Authority Code**,

- Assign/change **Authority Password**. The password can be a minimum of 4 digits and a maximum of upto 10 digits. Valid digits are 0 to 9, * and #. Default: 1111.

  To avoid unauthorized access, we recommend you to change the password regularly. Make sure it is strong and is kept confidential.

- Enter the **Name** of the user against the authority code you have assigned to them. The name acts as an identifier. Default: Blank.

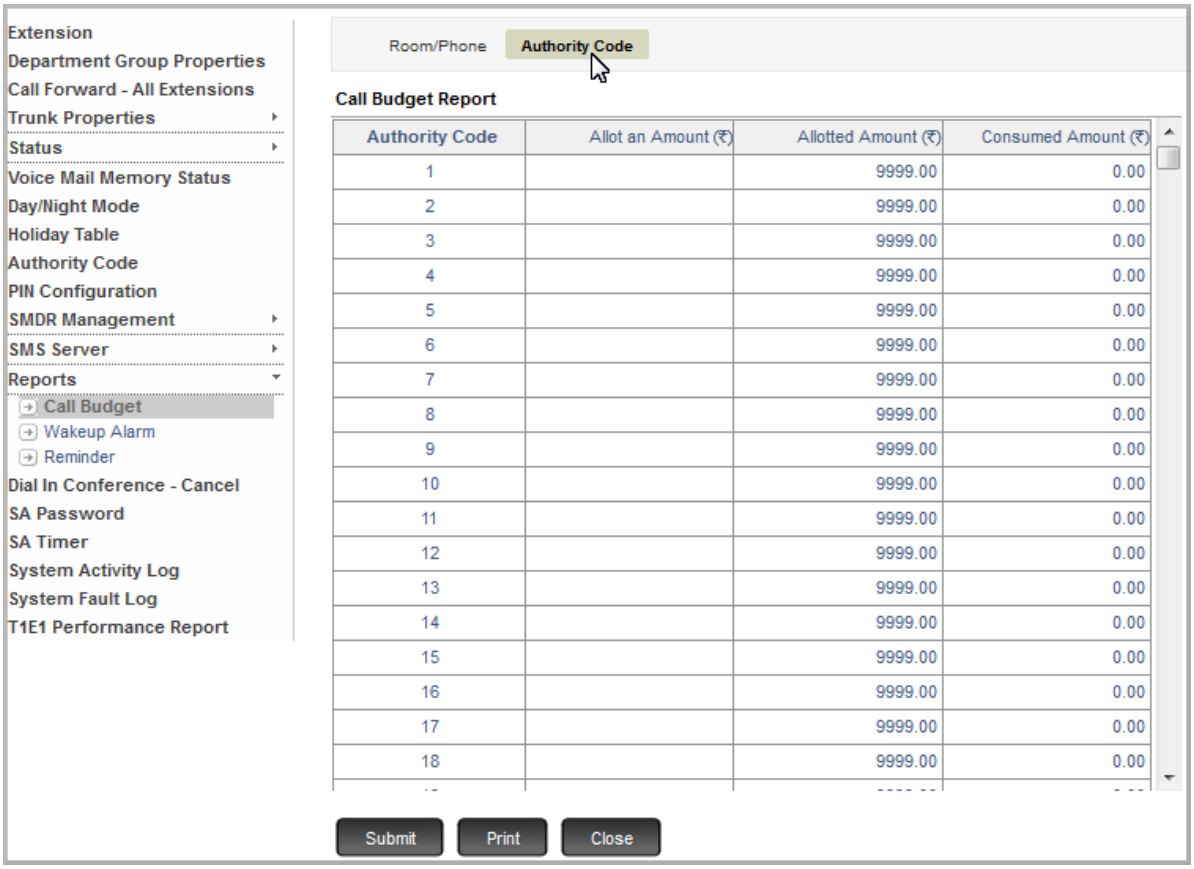
- Click **Submit** to save your settings.

Assigning a Call Budget to Authority Codes

Call Budget is a cost control feature that allows you to keep a tab on the total cost of each user. With this each user can be allotted a 'budget' limit for outgoing calls, which is automatically reloaded at the start of every month.

To assign a Call Budget to the Authority Code,

- Log in as System Administrator.
- Under **Reports**, click **Call Budget**.
- Click the **Authority Code** tab.



For each Authority Code assigned to the user,

- In the **Allot an Amount** column, enter the amount you want to assign to the user as budget limit for outgoing calls. The amount you allot here will be displayed as Allotted Amount.

*If you are re-assigning a new amount before the previous balance is consumed, make sure you add the available balance to the new amount. Enter this amount in **Allot an Amount**.*

For example, you have allotted an amount of Rs.1000 and the consumed amount is Rs.600. The available balance is Rs.400. Now, if you want to assign a new amount of Rs.500. In Allot an Amount you must enter 900 (Available Balance + New = 400 + 500).

- The **Allotted Amount** column displays the amount allotted to the user for making outgoing calls.

- The **Consumed Amount** column displays the call budget amount consumed by the user.
- Click **Submit** to save.

Setting the Retry Counts for Authority Code

By default, the system allows three attempts to re-enter the Authority Code Password. If you want to increase or decrease the number of attempts, set the parameter ***Retry Counts for Authority Code*** to the desired value. For instructions, see “System Timers and Counts”.

How to use

For EON Users and Extended IP Phone Users

To use Authority Code from any extension:

- Go OFF-Hook.
- Press DSS Key assigned to ‘Walk-in COS’.
  OR
- Dial **111**
- Select the ‘Walk-in by Authority Code’ option and press the Enter key.
- Dial the Authority Code, followed by the Password.
- You will hear the confirmation tone, followed by dial tone.
- Dial the desired number.

For SLT Users

- Go OFF-Hook.
- Dial **111**
- Press 2 to select ‘Walk-in by Authority Code
- Dial the Authority Code, followed by the Password.
- You will hear the Confirmation tone, followed by dial tone.
- Dial the desired number.

Printing Reports of Outgoing Calls made using Authority Codes

You can print call details of users who made outgoing calls using Authority Codes. For this, you will need to:

- enable **Store Outgoing Calls** in the Station Basic Feature Template of the extension user.
- set the **Calls made using Authority Code** filter in Outgoing Call Report.
- configure the **destination port** for SMDR-Outgoing Call Report.

Refer the section “Station Message Detail Recording-Report”, for detailed instructions on printing reports using filters.

Apple Push Notification Service Support

What's this?

Apple Push Notification Service (commonly referred to as Apple Notification Service or APNs) is a platform notification service created by Apple Inc. that enables third party application developers to send notification data to their applications installed on Apple devices.

Previously, VoIP applications needed to maintain a persistent connection in order to receive calls. Keeping a connection open in the background, drains the battery as well as causes all kinds of problems when the application crashes or is terminated by users.

In iOS 8 Apple has introduced PushKit as part of their effort to improve battery life, performance and stability for VoIP applications such as Skype, WhatsApp, etc. PushKit offers high-priority push notification with a large payload. The VoIP application receives the notification in the background, sets up the connection and displays a local notification to the user.

SARVAM UCS supports PushKit for VARTA AMP100 Application only. Push Notifications will be sent for calls, new messages as well as for voicemail. Push Notifications will be sent to the MATRIX VARTA AMP100 Application only if it is in the background and when there is persistent internet connection. You will receive the Push Notifications even after the you exit the application, provided the check box *Calls and Messages after exit* is enabled in the VARTA AMP100 Application.For details refer to the VARTA AMP100 User Guide.

How it works

Pre-requisites for Push Notifications:

- Make sure that the server has a persistent internet connection and there is connectivity with the APNS Server. To check the connectivity, refer “APNS Connectivity”.
- Make sure the Date and Time of the server is synchronized with the NTP Server.
- To receive IM and IM notifications make sure the application is registered at Location 1. For more details, refer “Configuring Matrix VARTA ADR100/AMP100 UC Clients”.

Let us see how the notifications will be sent by the server when MATRIX VARTA AMP100 application is registered with the server as a SIP Extension and it is in the background. There is an incoming call or message:

- You can check the status of the SIP Extension user. It will display Registered (as the device is in the background) and under the respective Contact 1, 2, 3 it will display the time remaining for the expiry of the VARTA Client Inactivity Timer. The default value of the VARTA Client Inactivity Timer is 10 days. To configure this timer, refer to “System Timers and Counts”.

- The server will send a Push Notification to the MATRIX VARTA AMP100 application (client).

- The server will wait for 15 seconds after sending the Push Notification:

    - if the client registers with the server within this time, the call will be connected or the message will be delivered. The status of the SIP Extension will display Registered and under the respective Contact 1, 2, 3 it will display the SIP ID, IP Address and the Registration Expiry Timer.

    - if the client does not register with the server within this time, the call will be disconnected or the message will be rejected.The status of the SIP Extension will display Registered and under the

respective Contact 1, 2, 3 it will display the time remaining for the expiry of the VARTA Client Inactivity Timer.

- The server maintains a configurable timer, VARTA Client Inactivity Timer which is set as 10 days. Till the expiry of the timer the server will send Push Notifications to the application.
- If for this duration, the server does not receive any registration request from the application and the timer expires, the server will consider the application as unregistered and will stop all Push Notifications to the application.The status of the SIP Extension will display Not Registered and under the respective Contact 1, 2, 3 the details will be cleared. Calls and messages will be rejected.

  The server will start sending notifications to the application in the background after the application is bought in the foreground once and a registration request is received by the server.

Feature Interactions when the Application is in the Background

Call Forward when not Registered:

If the application receives an incoming call from a QSIG caller, the Call Forward functionality will not be applicable.

To know more, see “Call Forward-When Not Registered”.

Handover and Handoff:

VARTA AMP100 users will be able to use Handover but Handoff will not be possible. To know more, see “Handover and Handoff”

System Restart

After System Restart the VARTA Client Inactivity Timer will be reset.

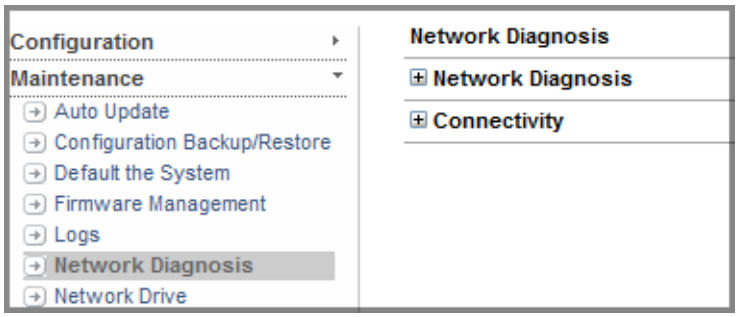
APNS Connectivity

A connectivity between the system and the APNS Server is required so that the Push notifications can be sent to the clients.

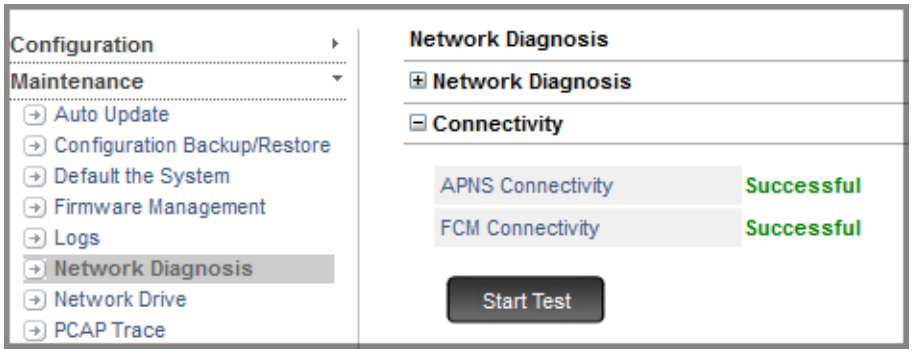
To check the APNS connectivity status,

- Log into Jeeves.
- Click the **Maintenance** link.

- Click the **Network Diagnosis** link



- Click **Connectivity** to expand.



- Click **Start Test**.

- In **Connectivity** the status is displayed as:
  - Successful - if the connectivity between the system and the APNS/FCM Server is established
  - Timeout - if there is no connectivity.

*If the **Connectivity** Test of either of the servers (APNS or FCM) with SARVAM UCS is not successful, then Push Notifications will not be sent to the Mobile Clients — MATRIX VARTA ADR100 / AMP100 application.*

Auto Attendant

What's this?

Auto Attendant allows external callers to reach an extension directly without the intervention of the Operator.

If Auto Attendant is enabled on a trunk, whenever an external call lands on that trunk, the *Built-In Auto Attendant* or the *Voice Mail Auto Attendant* (if VMS module is installed) of SARVAM UCS greets the caller and prompts the caller to dial the desired extension number. The call is then placed to the extension number dialed by the caller.

SARVAM UCS offers *Delayed Auto Attendant*, whereby incoming calls routed to the Operator or the Trunk Landing Group, can be answered by the Built-In Auto Attendant or the Voice Mail Auto Attendant, if none of the landing extensions answers the call within a certain time period.

Regular callers who know the extension numbers, can use the Auto Attendant to reach the desired extensions without Operator assistance. Thus, this reduces call traffic on the Operator extension, saves callers the time for call set-up and transfer. The Auto Attendant is particularly useful during non-working hours and holidays, and it helps project a professional image of the organization.

*Built-In Auto Attendant will not work, when the dialed extension has **Privacy from Built-In Auto Attendant** enabled in its Class of Service. So, if you want to prevent external callers from accessing certain extensions, you must enable **Privacy from Built-In Auto Attendant** in their Class of Service. To know more, see "Privacy".*

How it works

Auto Attendant can be configured on all trunk types, for the three time zones (working hours, break hours and non-working hours).

When configuring Auto Attendant on a trunk, you may choose to have calls answered by the Built-In Auto Attendant or the Voice Mail Auto Attendant.

Built-In Auto Attendant

When the Built-In Auto Attendant of SARVAM UCS is selected as the destination for incoming calls on a trunk, this is how it will work:

- A call lands on a Trunk.
- The system waits for the period of the *Built-In Auto Attendant Answer Wait Timer* (default: 05 seconds) to answer the call during this period. The caller gets Ring Back Tone from the CO.
- The system greets the caller with the pre-recorded Time-based Greetings. These are: *Built-In Auto Attendant— Morning Greetings/Afternoon Greeting/ Evening Greeting.* This is then followed by the *Built-In Auto Attendant-Welcome Greeting* for the current time zone (working hours, break hours, non-working hours). A Voice Module must be assigned for the Built-In Auto Attendant Time-based as well as the Welcome Greeting.
- The Built-In Auto Attendant Time-based and Welcome Greeting messages are played once.

If no voice module is assigned as Welcome Greeting, the system will play music-on-hold after answering the call. It will play music-on-hold until the end of the *Built-In Auto Attendant Music Timer* (default: 5 seconds).

- On the completion of the Welcome Greeting or music-on-hold at the end of the Built-In Auto Attendant Music Timer, the system plays the *Built-In Auto Attendant Dial Message* to prompt the caller to dial the desired extension number.

  The Built-In Auto Attendant Dial Message is played once and the caller gets Beeps. The system waits for the *Built-In Auto Attendant Beeps Timer* (default: 10 seconds) to expire.

- If the caller does not dial any number before the *Built-In Auto Attendant* Beeps Timer expires, the system plays the *Built-In Auto Attendant Call Transfer to Operator* message and transfers the call to the landing destination as configured by you—Extensions or Operator.

  The system waits for the duration of the *Built-In Auto Attendant Inactivity Timer* (default: 60 seconds) for the Operator to answer the call. If there is no answer at the end of this timer, the system releases the trunk.

If the caller fails to dial digits, you can have the call disconnected instead of having it routed to the Operator. For this, you need to enable the ***Disconnect Built-In Auto Attendant call, when caller does not dial any digit*** *flag in the System Parameters. When this flag is enabled, the system will play the Built-In Auto Attendant No Dial Voice message to the caller. If the caller fails to dial a digit within the Built-In Auto Attendant Beeps Timer, the system will disconnect the call.*

- If the caller dials the extension number, the system checks if the number is valid.

  If the dialed digits are invalid, the system plays the *Wrong Dial* voice message to the caller. This message is played once. The system waits for the duration for the *Built-In Auto Attendant Error Tone Timer* (default: 5 seconds).

  If the Wrong Dial Voice Message is not programmed, the system plays Error Tone to the caller for the duration of the Built-In Auto Attendant Error Tone Timer, followed by the *Built-In Auto Attendant Dial Prompt*.

- If the number dialed by the caller is valid, the system checks if the dialed extension is free.

- If the dialed extension is busy, the system plays the *Built-In Auto Attendant Busy Message* to the caller. The message is played once.

- If no *Built-In Auto Attendant Busy Message* is programmed, the caller will hear Busy Tone. The Busy Tone is played for duration of the *Built-In Auto Attendant Busy Tone Timer* (default: 15 seconds), followed by the *Built-In Auto Attendant* Dial Prompt.

To have the call disconnected if the dialed extension is busy, you may enable the ***Disconnect Built-In Auto Attendant Call, when dialed number is busy*** *flag in the System Parameters.*

- The dialed extension is free. The system calls the extension and plays *Built-In Auto Attendant Ring Back Tone Message* (if programmed) or Ring Back Tone to the caller. This message is played until the dialed extension is ringing.

- The system waits for the period of the Built-In Auto Attendant Ring Timer for the dialed extension to answer the call.

- When the dialed extension answers the call, the caller gets connected to the extension.

  If the dialed extension does not answer before the expiry of the Built-In Auto Attendant Ring Timer, the system prompts the caller to dial again with the *Built-In Auto Attendant Dial Prompt* message to the caller.

- The system diverts the call to the Operator. When the call is transferred to the Operator, the system plays the *Built-In Auto Attendant Call Transfer to Operator* voice message (if programmed) or plays Ring Back Tone to the caller.

If there is no reply from the dialed extension, you can have the call disconnected instead of having it routed it to the Operator by enabling the ***Disconnect Built-In Auto Attendant call, when dialed number is not responding*** *flag in the System Parameters.*

Voice Mail Auto Attendant

SARVAM UCS supports 64 simultaneous VMS calls[244]. These calls can be made by extension users to access their mailbox or incoming calls landing on a trunks that has Voice Mail Auto Attendant enabled.

If all channels are used by extension users, the external incoming calls on the Voice Mail Auto Attendant enabled trunks remain unanswered. To make sure that the incoming calls on the trunks are answered you can reserve channels of the VMS as per your requirement.

The extension users will not have access to the channels reserved for incoming calls on the trunks to access their mailbox.

If all the reserved channels are busy and there is an incoming call on the trunk, it will land on the unreserved channel if free.

To reserve a channel, see “Configuring VMS General Parameters”.

When the Voice Mail Auto Attendant of SARVAM UCS is selected as the destination for incoming calls on a trunk, this is how it will work:

- A call lands on a Trunk.
- The Voice Mail System (VMS) answers the call.
- The VMS greets the caller with the Welcome message and the Greeting Message selected for the current time zone (working hours, break hours and non-working hours).
- If the system detects the day as a holiday, the VMS plays the Holiday Message. To know more, see “Holiday Table”.
- The VMS plays prompts to the caller to process the call further.

Delayed Auto Attendant

You can use Delayed Auto Attendant to have incoming calls that are not answered by the landing destinations—the Operator and the Trunk Landing Group—within a certain time period, to be handled either by the Built-In or the Voice Mail Auto Attendant.

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244. For the VMS calls, the number of channels that will be supported would be as per the license you purchase.

When you use Delayed Auto Attendant,

- as a call lands on a trunk, the system checks the incoming call routing configured for the current time zone for the trunk.
- on finding *Delayed Auto Attendant* enabled, the system rings on the destination extensions (Operator and Trunk Landing Group) for the duration of time defined for ringing the extensions (default: 10 seconds).
- if no reply is received from the extensions, the system routes the call to the auto attendant you selected, which may by the Built-In Auto Attendant or the Voice Mail Auto Attendant.
- the call is processed further by the auto attendant you selected.

How to configure

To use the **Built-In Auto Attendant** on trunks, do the following:

1. Make a list of the trunks by their port type (CO, Mobile, SIP, T1E1PRI, BRI) and port number on which you want to use the Built-In Auto Attendant.
2. In the "Trunk Feature Template" assigned to these trunks, enable Auto Attendant for the desired time zones by selecting **Built-In Auto Attendant**.
3. Set the Start Time for the Morning, Afternoon and Evening Greeting Messages. Refer "Greeting Message Time" in *System Parameters* for instructions.
4. Assign *Voice Modules* for Built-In Auto Attendant Messages. To play to callers pre-recorded voice messages as Built-In Auto Attendant greetings and to play voice prompts at each stage of the call, you need to assign Voice Modules for the following Built-In Auto Attendant Messages:

    - **Built-In Auto Attendant-Time-based Greetings (Morning, Afternoon and Evening Greetings)**: These are played to the caller as soon as the call is answered by the **Built-In Auto Attendant**. Different messages can be recorded for Morning, Afternoon and Evening Hours. You can also set the time during which you want to play these greetings. For detailed instructions, see "Greeting Message Time" in *System Parameters*.
    - **Built-In Auto Attendant-Welcome Greeting**: Played to callers when answering the call. Different welcome greetings can be programmed for Working Hours, Break Hours and Non-working Hours. The Built-In Auto Attendant Welcome Greeting message is played once.
    - **Built-In Auto Attendant-Dial Prompt**: Played after the Welcome greeting message to prompt the caller to dial the desired extension number. This message is played once.
    - **Built-In Auto Attendant-Ring Back Tone:** Played after the caller has dialed the number and the system is ringing the dialed extension. This message is played continuously as the dialed extension rings.
    - **Built-In Auto Attendant-Wrong Dial message:** Played when the caller dials a wrong number or the number dialed by the caller does not match with any extension number of SARVAM UCS. This message is played once.
    - **Built-In Auto Attendant-Destination Busy:** Played when the dialed extension is busy. This message is played once.

- **Built-In Auto Attendant-Destination No Reply:** Played when the dialed extension does not respond. This message is played once.

- **Built-In Auto Attendant-No Dial:** Played when the caller has not dialed any number. This message is played once.

- **Built-In Auto Attendant-Call Transfer to Operator:** Played to the caller when the call is being transferred to the Operator. This message is played once.

The default Voice Module numbers assigned to Built-In Auto Attendant messages and the messages recorded on each module are:

Voice Module Number	Voice Message Application	Voice Message
02	Built-In Auto Attendant - Morning Greeting	Good Morning!
03	Built-In Auto Attendant - Afternoon Greeting	Good Afternoon!
04	Built-In Auto Attendant - Evening Greeting	Good Evening!
05	Built-In Auto Attendant - Welcome Greeting for Day Time (Working Hours)	Welcome!
06	Built-In Auto Attendant - Welcome Greeting for Night time (Non-working hours)	Welcome! I am sorry, we are closed.
07	Built-In Auto Attendant - Dial prompt	Please dial the desired number.
08	Built-In Auto Attendant - No Dial message	Sorry! You have not dialed any number.
09	Built-In Auto Attendant - Wrong Dial message	Sorry! The number is not valid.
10	Built-In Auto Attendant - Destination Busy message	The person you dialed is busy.
11	Built-In Auto Attendant - Destination Ringing message (Ring Back Tone)	The number you have dialed is ringing.
12	Built-In Auto Attendant - Destination No Reply message	The person you dialed is not responding.
13	Built-In Auto Attendant - Call Transfer to Operator message	Please hold, transferring your call to the Operator.

You may customize these Built-In Auto Attendant voice messages by recording messages of your choice and assigning them to the voice modules. For instructions on recording messages on the voice modules and assigning voice modules to different functions, see “Voice Message Applications”.

If you do not use any of the above voice modules, the system will play the Call Progress Tone for each call state.

To use the **Voice Mail Auto Attendant** on trunks, do the following:

1. Make a list of the trunks by their port type (CO, Mobile, SIP, BRI, T1E1PRI) and port number on which you want to use the Voice Mail Auto Attendant.

2. In the “Trunk Feature Template” assigned to these trunks,

   - enable Auto Attendant by selecting **Voice Mail Auto Attendant** for the desired time zones.

   - select the desired **Voice Mail Auto Attendant (VMAA) Menu**.

3. Complete the VMS related configuration.

   - Configure Welcome and Greeting messages. You may either use the default, pre-recorded welcome messages of the VMS, or record the custom welcome messages that meet your requirements, in .WAV file format.

   - Configure the Voice Mail Auto Attendant (VMAA) Menu parameters.

   For more information and instructions, see the “Configuring Voice Mail System”.

To use **Delayed Auto Attendant** on trunks, do the following:

1. Make a list of the trunks by their port type and port number on which you want to enable Delayed Auto Attendant.

2. In the “Trunk Feature Template” assigned to these trunks,

   - set the **Auto Attendant Delayed Timer**.

   - if you want to use the *VMS Auto Attendant for Delayed Auto Attendant*, select **Voice Mail Auto Attendant** and complete the voice mail related configuration. For more information and instructions, see the “Configuring Voice Mail System”.

   - if you want to use the *Built-In Auto Attendant for Delayed Auto Attendant*, select **Built-In Auto Attendant**, and assign the Voice Modules for Built-In Auto Attendant Messages, as described earlier.

If required, you may also change the default values of the following Built-In Auto Attendant related Timers and set them to the desired values.

- **Built-In Auto Attendant Inactivity Timer** (default: 60 seconds)
- **Built-In Auto Attendant Answer Wait Timer** (default: 5 seconds)
- **Built-In Auto Attendant Music Timer** (default:5 seconds)
- **Built-In Auto Attendant Beeps Timer** (default: 10 seconds)
- **Built-In Auto Attendant Ring Timer** (default: 30 seconds)
- **Built-In Auto Attendant Busy Tone Timer** (default: 15 seconds)
- **Built-In Auto Attendant Error Tone Timer** (default: 5 seconds)

To know more about these timers and for configuration instructions, see “System Timers and Counts”.

You may also configure the following Built-In Auto Attendant related flags, as required:

- **Disconnect Built-In Auto Attendant call, when dialed number is busy:** When this flag is enabled, if the dialed extension is found busy, the system will disconnect the call instead of routing it to the Operator. Default: disabled.

- **Disconnect Built-In Auto Attendant call, when dialed number is not responding:** When this flag is enabled, if there is no reply from the landing destination extensions, the system will disconnect the call instead of routing it to the Operator. Default: disabled.

- **Disconnect Built-In Auto Attendant call, when caller does not dial any digit:** When this flag is enabled, if the caller fails to dial a digit within the Built-In Auto Attendant Beeps Timer, the system will disconnect the call instead of routing it to the Operator. Default: disabled.

These flags may be used in Hotels that provide 'Limited Services' and do not want to receive unanswered/busy calls on the guest phones. For instructions on enabling or disabling these flags, see “System Parameters”.

Auto Answer

What's this?

Auto Answer allows incoming calls to be answered without any manual interventions by the extension users.

This feature is particularly useful for Operators in high call traffic settings, as it saves them the effort of picking up the handset or pressing the speaker key repeatedly.

This feature works on digital key phones (DKP) as well as in Extended IP Phones.

How it works

With Auto Answer set on an extension DKP/Extended IP Phone, whenever a call lands on the DKP/Extended IP Phone extension,

- the extension rings for the duration of the Auto Answer Timer[245]. This timer is programmable, and by default it is set to 1 second.
- on the expiry of the Auto Answer Timer the system plays a beep[246] to the user.
- the phone goes OFF-Hook to answer the call, without any intervention by the extension user such as picking up the handset or pressing the speaker or the headset key.
- If a headset is connected, and headset connectivity is enabled, the incoming speech audio will be diverted to the headset automatically.
- the extension user can talk to the caller.

Auto Answer works only if the phone is in idle state; the phone must not be busy with an active call or using a feature.

How to configure

For Auto Answer to work, you are required to do the following:

1. Enable Auto Answer in the DKP Parameters/Extended IP Phone Extensions.
2. Change Auto Answer Timer, if required. The range of this timer is 1 to 9 seconds. By default, the Auto Answer Timer is set to 1 second.
3. Enable Headset Connectivity, if headset is to be used for Auto Answer.

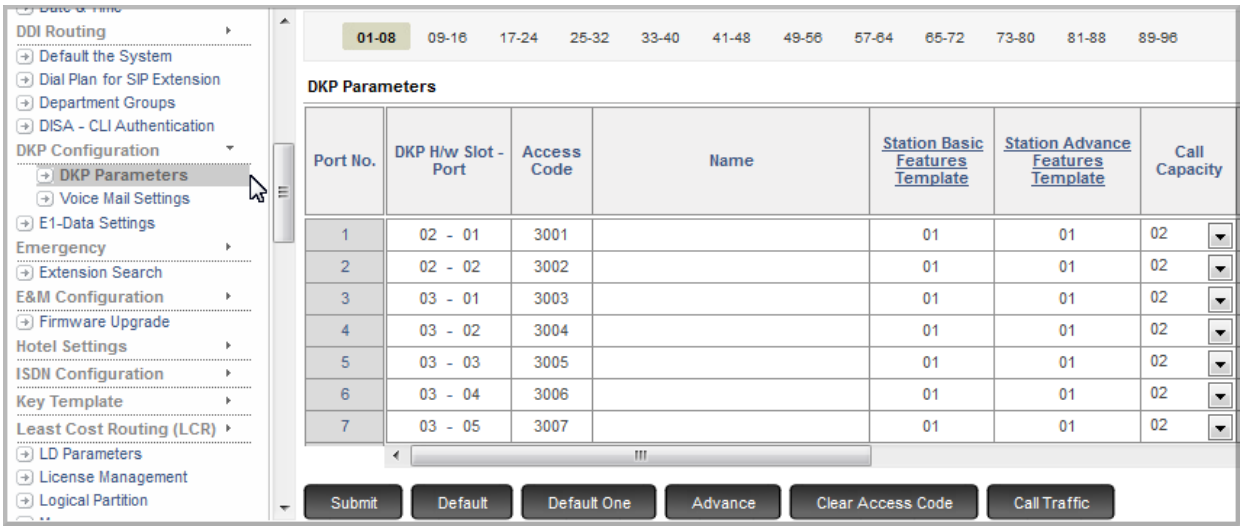
All of the above can be programmed by the System Engineer using Jeeves and a Telephone.

245. This timer defines the time in seconds that the phone should wait before going OFF-Hook to answer incoming calls.
246. Beep is not supported on IP Phones.

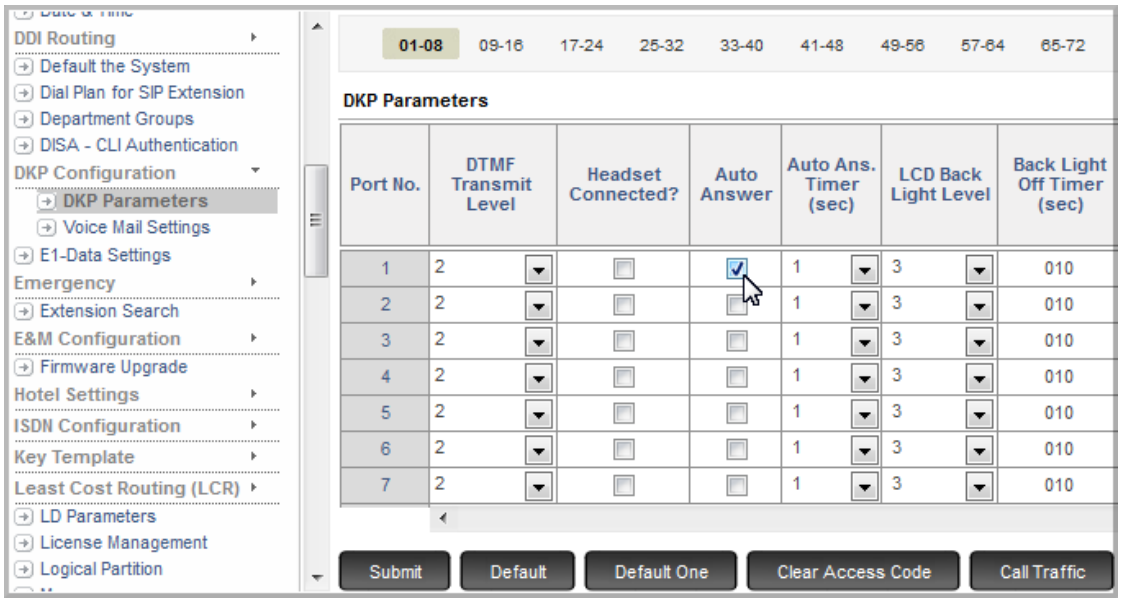
The extension users can also program the above parameters using the Phone Menu. See "How to use" Auto Answer later in this topic.

Programming Auto Answer using Jeeves

- Login as System Engineer.
- Under **Configuration**, click **DKP Configuration**.
- Click **DKP Parameters** to open the page.
- If the DKPs are already installed and configured, identify the DKP you wish to provide Auto Answer feature by its Hardware Port/Slot Number, Access Code or Name.



- Click the **Advance** button.
- Go to the column **Auto Answer** of the DKP parameters using the horizontal scroll bar. Enable the flag by selecting the check box.



To cancel Auto Answer, disable the flag by selecting the check box.

- Now, go to the column **Auto Answer Timer**, and set the Timer as required. By default the Timer is set to 1 second.
- If Headset is to be used by the DKP, go to the column **Headset Connected?** and click the check box to enable the flag.
- Repeat the above steps to program Auto Answer parameters for each DKP that is to be provided this feature.
- Click **Submit** to save changes.
- Similarly configure these parameters for Extended IP Phones, see “Configuring Matrix SPARSH VP248” and “Configuring Matrix SPARSH VP310”.

Programming Auto Answer using a Telephone

- Enter SE mode.

  To set auto answer on DKP, dial:
  - ***1214-1-DKP-Auto Call Answer Mode*** to set Auto Answer on a single DKP.
  - ***1214-2-DKP-DKP-Auto Call Answer Mode*** to set Auto Answer on a range of DKPs.
  - ***1214-*-Auto Call Answer Mode*** to set Auto Answer on all DKPs.
    Where,
    DKP is the Software port number of the DKP, from 001 to 128.
    Auto Call Answer Mode is
    0 for Manual mode
    1 for Auto Answer mode.

  To set Auto Answer Timer, dial:
  - ***1215-1-DKP-Auto Call Answer Timer*** to set Timer for a single DKP.
  - ***1215-2-DKP-DKP-Auto Call Answer Timer*** to set the same Timer for a range of DKPs.
  - ***1215-*-Auto Call Answer Timer*** to set the same Timer for all DKPs.
    Where,
    DKP is the Software port number of the DKP, from 001 to 128.
    Auto Call Answer Timer is from 1 to 9 seconds. Default: 1 second

  To enable/disable Headset Connectivity flag, dial:
  - ***1213-1-DKP-Headset Connectivity Flag*** to enable/disable the flag on a single DKP.
  - ***1213-2-DKP-DKP-Headset Connectivity Flag*** to enable/disable the flag on a range of DKPs.
  - ***1213-*-Headset Connectivity Flag*** to enable/disable the flag on all DKPs.
    Where,
    DKP is the Software port number of the DKP, from 001 to 128.
    Headset Connectivity Flag is
    0 for Disable
    1 for Enable

- Exit SE mode.

Similarly to configure these parameters for Extended IP Phones using a telephone, see “Configuring Matrix SPARSH VP248”, “Configuring Matrix SPARSH VP310”, “Configuring Matrix SPARSH VP510”, for SPARSH VP330 refer the SPARSH VP330 User Guide and for SPARSH VP210 refer the SPARSH VP210 *(Extended)* User Guide.

How to use

Extension users can set/cancel Auto Answer and enable Headset connectivity from their DKP/ Extended IP Phone or by navigating the Phone Menu.

To set Auto Answer:

- Press the DSS Key assigned to Auto Answer function[247].

  OR

- Press 'Enter' Key.
- Scroll down to select 'Phone Settings'; press Enter Key.
- Enter Your User Password (default: 1111).
- Scroll down to select 'Call Answer Type'; press Enter Key.
- You get the options:
  Manual Call Answer
  Auto Call Answer
- Scroll to select Auto Answer and press Enter key.
- Now select the Timer for Auto Answer from any of the options:
  Answer After 1 sec (default)
  Answer After 2 sec
  :
  Answer After 9 sec
- Press Enter Key.

To cancel Auto Answer:

- Repeat the above steps.
- Select 'Manual Answer' as the Call Answer Type.

It is recommended that Auto Answer Timer be set to at least 2 seconds.

To enable Headset Connectivity:

- Press the DSS Key assigned to Headset function[248].

  OR

- Press 'Enter' Key.
- Scroll down to select 'Phone Settings'; press Enter Key.
- Enter Your User Password (default: 1111).
- Scroll down to select 'Headset Connectivity'; press Enter Key.
- You get the options:
  Headset Not Connected
  Headset Connected
- Scroll to select 'Headset Connected' and press Enter key.

---

247. This function must have been programmed by the System Engineer on a DSS Key of the phone. Refer "DSS Keys Programming" for instructions.

248. This function must have been programmed by the System Engineer on a DSS Key of the phone. Refer "DSS Keys Programming" for instructions.

Auto Call Back (ACB)

What's this?

If the extension number you have dialed is busy or is not responding, you may use the Auto Call Back feature, instead of repeatedly dialing the number. Similarly, when you dial a code to access a trunk and the trunk is busy, you may set Auto Call Back. SARVAM UCS allows you to set a maximum of 300 Auto Call Backs.

How it works

When you set Auto Call Back,

- the SARVAM UCS will queue your call attempt.
- As soon as both extensions, yours and the remote extension, are available, the system will ring first on your extension for the duration of the Auto Call Back Ring Timer. This timer is set by default to 30 seconds and is programmable.
- When you go OFF-Hook, the system will ring on the remote extension (provided it is also available at that moment) for the duration of the Auto Call Back Ring Timer.
- When the remote extension user goes OFF-Hook, your call will get connected.

  However, if the remote extension gets busy before the system can ring on it, the system will continue to try again.

  Auto Call Back set for a busy trunk works the same way. As soon as the busy trunk port you are trying to access is available, the system will ring your extension. When you go OFF-Hook you will be connected to the trunk port.

- *Each extension of the SARVAM UCS can set only one Auto Call Back request at a time. If you set another Auto Call Back request, before the first one has been served, the system will override the first request and serve the second.*
- *The SARVAM UCS has the capacity to serve 300 Auto Call Back requests from its extensions at a time. The service duration for each request is 60 minutes. Requests that are not served within 60 minutes are automatically canceled by the system. Also, the system will not serve any more requests if all the 300 requests are pending. In such a case, the system will play an error tone, when an extension attempts to make a request.*

Auto Call Back request set by you will be cleared by the system if:

- it was successfully served, that is, your extension was connected to the remote extension or the trunk you were trying to reach.
- you do not answer the Auto Call Back ring, before the expiry of the Ring Timer, that is, within 30 seconds (default setting).
- the remote extension does not answer the Auto Call Back ring before the expiry of the Ring Timer.

- it has not been served within 60 minutes.

- *Auto Call Back works for internal calls and for accessing trunk ports only.*
- *Internal calls include calls between Systems that are networked using Q-SIG.*

How to configure

Auto Call Back is a Class-of-Service dependant feature. An extension user can set/cancel Auto Call Back only if it is enabled in the extension's Class of Service.

The only programming involved in this feature is enabling/disabling Auto Call Back in the Class of Service and changing the duration of the Auto Call Back Ring Timer, if required.

Both these can be programmed using Jeeves and a Telephone.

Programming Auto Call Back using Jeeves

In the default factory settings, Station Basic Feature Template Number 01 is assigned to all extensions of SARVAM UCS. Template 01 has the features 'Auto Call Back Busy' and 'Auto Call Back No Reply' enabled in the default CoS group 01. So, all extensions of the SARVAM UCS can set/cancel Auto Call Back if the called number is busy or does not reply.

However, if Auto Call Back Busy/No Reply is to be denied to any of the extensions, follow these steps:

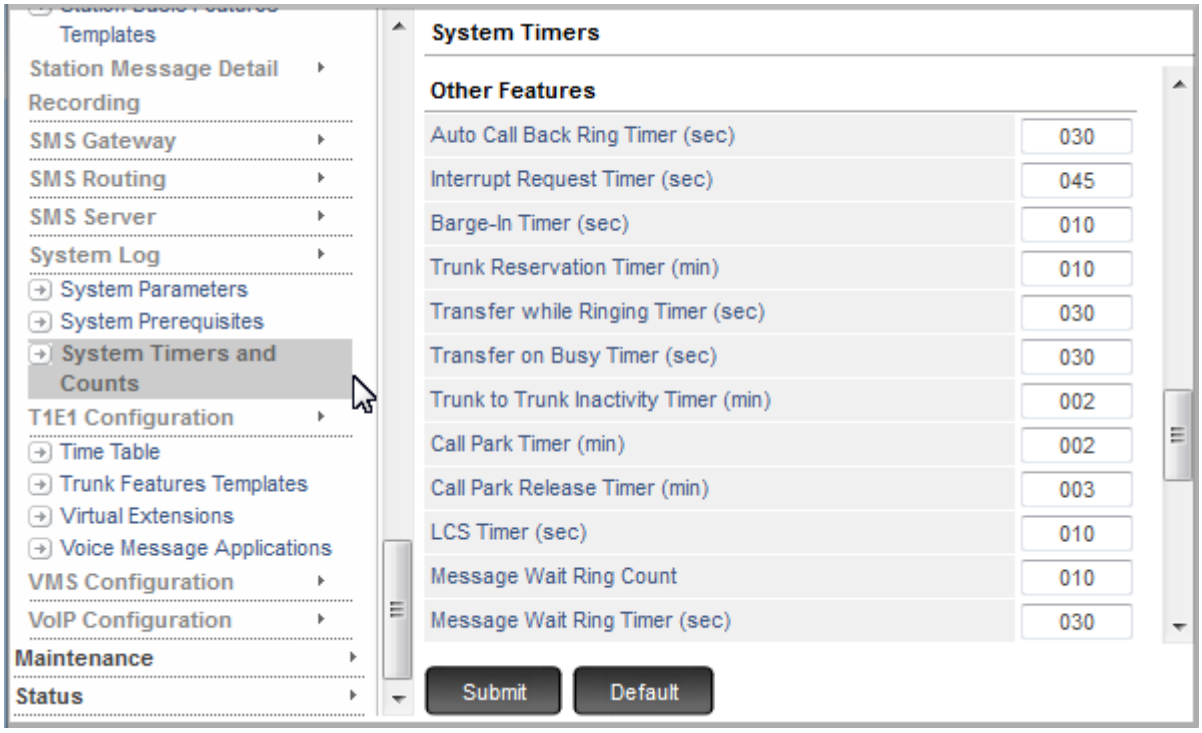
1. Define a CoS group with Auto Call Back Busy/No Reply disable.
2. Prepare a Station Basic Feature Template with this CoS group applicable in all the “Time Zones”.
3. Assign this new Template to the selected extensions to which Auto Call Back is to be denied.

Refer the topics “Class of Service (COS)” and “Station Basic Feature Template” for instructions on how to enable/disable a feature in a CoS group, how to prepare a Station Basic Feature Template with a new CoS group and assign the new template to SLT, DKP, SIP and ISDN Terminal extensions using Jeeves.

If the User wants to increase or decrease the duration of the of the Auto Call Back ring on both extensions, that is, the extension requesting Auto Call Back and the destination extension, program the 'Auto Call Back Ring Timer', according to User preference.

- Log into Jeeves as System Engineer.
- Under **Configuration**, click **System Timers and Counts** to open the page.

- Scroll to reach **Other Features**.



- Set the **Auto Call Back Ring Timer (sec)** to the desired duration.

- Click **Submit** to save your settings.

Programming Auto Call Back using a Telephone

Refer the topics "Class of Service (COS)" and "Station Basic Feature Template" for instructions on how to enable/ disable a feature in a CoS group, how to prepare a Station Basic Feature Template with a new CoS group and assign the new template to SLT and DKP extensions using SE commands.

- Enter SE mode from a DKP/SLT.

  To program Auto Call Back when Busy/No Reply in a CoS group, dial:
  - ***1302-1-COS Group-Feature Number-Code***
    Where,
    CoS group is from 01 to 20
    Feature Number for Auto Call Back when Busy is '04'
    Feature Number for Auto Call Back when No Reply is '05'.
    Code is
    0 for Disable
    1 for Enable.

    For example: To enable Auto Call Back when Busy in CoS group 02, dial ***1302-1-02-04-1***
    To enable Auto Call Back when No Reply in CoS group 02, dial ***1302-1-02-05-1***

  To assign the CoS group with Auto Call Back Busy/No Reply to a Station Basic Feature Template, dial:
  - ***5502-1-Template Number-Feature Number-Code***
    Where,
    Template Number is from 01 to 50.
    Feature Number is

03 for CoS group for Working Hours
04 for CoS group for Break Hours
05 for CoS group for Non-Working Hours
Code is CoS group number from 01 to 20.

For example: To apply CoS group 02 with Auto Call Back Busy and No Reply for each Time zone in Station Basic Feature Template 02, dial the following commands:
5502-1-02-03-02 for Working hours
5502-1-02-04-02 for Break hours
5502-1-02-05-02 for Non-working hours

- To apply the Station Basic Feature Template now programmed with Auto Call Back when Busy and No Reply, to the extensions, dial the following commands:

  If extension is an SLT, dial:
  - ***5503-1-SLT-Template Number*** to apply on a single SLT.
  - ***5503-2-SLT-SLT-Template Number*** to apply on a range of SLTs.
  - ***5503-*-Template Number*** to apply on all SLTs.
    Where,
    SLT is the Software port number of the SLT, from 001 to 512
    Template Number is the number of the Station Basic Feature Template (01 to 50) you have programmed with the Auto Call Back feature.

  If extension is a DKP, dial:
  - ***5504-1-DKP-Template Number*** to apply to a single DKP.
  - ***5504-2-DKP-DKP-Template Number*** to apply to a range of DKPs.
  - ***5504-*-Template Number*** to apply to all DKPs
    Where,
    DKP is the Software port number of the DKP, from 001 to 128.
    Template Number is from 01 to 50.

  To program the Auto Call Back Ring Timer:
  - Dial command ***3801-Seconds***
    Where,
    Seconds = 001 to 255 seconds.

- Exit SE mode.

How to use

Extension users can set two types of Auto Call Back:

- Auto Call Back on Busy - when the extension/trunk they are trying is Busy.
- Auto Call Back on No Reply - when there is no reply from the extension they are trying.

Auto Call Back on Busy

For EON and Extended IP Phone Users

Using DSS Key:

To set Auto Call Back on Busy:

- Press the 'Call Back' Key on EON48 on Busy Tone or press the DSS Key assigned to Auto Call Back on EON310 on Busy Tone.
- You get confirmatory message "Auto Call Back Set" on the phone's display. The LED of the DSS Key will be turned on.
- Go idle or you get dial tone after 3 seconds.

To cancel Auto Call Back on Busy:

- Press the 'Call Back' Key on EON48 again or press the DSS Key assigned to Auto Call Back on EON310 again.
- You get confirmatory message "Auto Call Back Canceled" on the phone's display. The LED of the DSS Key will be turned off.
- Go idle or you get dial tone after 3 seconds.

Using Command:

To set Auto Call Back on Busy:

- Dial 2 on Busy Tone.
- You get confirmatory message "Auto Call Back Set" on the phone's display. The LED of the DSS Key assigned to Auto Call Back will be turned on.
- Go idle or you get dial tone after 3 seconds.

To cancel Auto Call Back on Busy:

- Dial 102.
- You get confirmatory message 'Auto Call Back Canceled' on the phone's display. The LED of the DSS key assigned to Auto Call Back will be turned off.
- Go idle or you get dial tone after 3 seconds.

For SLT Users

To set Auto Call Back on Busy:

- On Busy Tone.
- Dial 2.
- You get confirmatory tone
- Replace handset or you get dial tone after 3 seconds.

To cancel Auto Call Back:

- Pick up the handset.
- Dial 102.
- You get confirmatory tone.
- Replace handset or you get dial tone after 3 seconds.

Auto Call Back on No Reply

For EON and Extended IP Phone Users

Using DSS Key:

To set Auto Call Back on No Reply:

- Press the 'Call Back' Key / DSS Key assigned to Auto Call Back on Ring Back Tone.
- You get confirmatory message "Auto Call Back Set" on the phone's display. The LED of the DSS Key will be turned on.
- Go idle or you get dial tone after 3 seconds.

To cancel Auto Call Back on No Reply:

- Press the 'Call Back' Key / DSS Key assigned to Auto Call Back again.
- You get confirmatory message "Auto Call Back Canceled" on the phone's display. The LED of the DSS Key will be turned off.
- Go idle or you get dial tone after 3 seconds.

Using Command:

To set Auto Call Back on No Reply:

- Dial 2 on Ring Back Tone.
- You get confirmatory message "Auto Call Back Set" on the phone's display. The LED of the DSS Key assigned to Auto Call Back will be turned on.
- Go idle or you get dial tone after 3 seconds.

To cancel Auto Call Back:

- Dial 102.
- You get confirmatory message 'Auto Call Back Canceled' on the phone's display. The LED of the DSS key assigned to Auto Call Back will be turned off.
- Go idle or you get dial tone after 3 seconds.

For SLT Users

To set Auto Call Back on No Reply:

- On Ring Back Tone
- Dial 2.
- You get confirmatory tone
- Replace handset or you get dial tone after 3 seconds.

To cancel Auto Call Back on No Reply:

- Pick up the handset.
- Dial 102.
- You get confirmatory tone.
- Replace handset or you get dial tone after 3 seconds.

If you hear an error tone while setting an Auto Call Back request, it is likely that the system already has 300 pending requests and is unable to accept yours.

Auto Redial

What's this?

The Auto Redial feature retries a call automatically if the dialed number is busy. It repeatedly checks the busy line till it is free. When the called number is no longer busy, the extension of the caller rings.

Auto Redial saves time and the effort of repeatedly dialing the entire phone number over and over until the called party gets off the phone.

The Auto Redial feature is supported for external numbers only. Maximum 50 Auto Redials can be set by extension users.

How it works

When an extension user dials a number and gets a busy tone, s/he may set Auto Redial. When Auto Redial is set,

- SARVAM UCS checks for a free trunk to dial the number.
- SARVAM UCS will dial out the requested number and will wait until the 'Ring Back Tone Wait Timer[249]' expires to sense the Ring Back Tone from the requested number. This timer is programmable and is set to 60 seconds as default.
- If the system does not detect Ring Back Tone for 60 seconds, it releases the trunk and tries again after some time. If the system detects a busy tone, it releases the trunk and redials the number automatically after some time. This process is repeated until the system detects the Ring Back Tone.
- When the SARVAM UCS detects the Ring Back Tone instead of the Busy Tone, it will ring on the extension that set Auto Redial. The extension will ring for the duration of the 'Redial Ring Timer[250]'. This timer is programmable and is set to 45 seconds as default.
- The extension must go OFF-Hook to get connected to the remote party.
- If the extension is in the middle of any activity such as dialing, ringing or speech, the SARVAM UCS will suspend Auto Redial until the extension becomes idle again. After which it dials the requested number again.

Two types of Auto Redial are supported by the SARVAM UCS - Auto Redial (normal) and Auto Redial 'Priority' - that differ from each other in terms of the number of redial attempts and the interval between attempts.

- **Auto Redial (normal):** The system is programmed by default to make 5 attempts to redial at an interval of 45 seconds (default) between each attempt. Both, the number of attempts as well as the duration of the interval can be changed match User preference, like decreasing the number of attempts to 3 and increasing the interval to 60 seconds.

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249. Time for which SARVAM UCS waits to sense the RBT from the PSTN/CO Network after dialing the requested number. This timer is particularly relevant to CO ports. Valid range of the timer: 000 to 255 seconds. Default: 060 seconds.

250. Time for which the extension that has requested Auto Redial should ring. Valid range of the timer: 000 to 255 seconds. Default: 045 seconds.

- **Auto Redial 'Priority':** the system makes a greater number of attempts to redial and the duration of the interval between each attempt is less. By default the system is programmed to make 20 redial attempts at intervals of 20 seconds. The number of attempts as well as duration of the interval are programmable; for instance, the number of attempts can be set to 30 and the interval to 15 seconds.

To change the number of redial attempts and the interval between them, the SE must Auto Redial Count and the Auto Redial Timer respectively. In addition to these, the system has three other related timers, which can be programmed to match User preference:

- Auto Redial Ring Back Tone (RBT) Wait Timer
- Auto Redial Ring Timer

- *An extension user can request Auto Redial for multiple numbers at a time from the same extension and more than one extension can attempt auto redial simultaneously.*
- *The system uses the same OG Trunk Bundle Group you used. If you dialed the number on group code 60, the system grabs one of the free trunks from group code 60 for Auto Redial.*
- *If the number was dialed the first time using selective trunk access, the system will use the same trunk to execute Auto Redial.*
- *If the extension is programmed for 'Dynamic Lock', and you have set the 'Auto Redial', the system will not check the Toll control as per dynamic lock level.*

Auto Redial may not work well on Two-wire Trunk lines, as its functioning greatly depends on line condition.

How to configure

For Auto Redial to work, the System Engineer must:

1. Enable the features 'Auto Redial' and 'Auto Redial Priority' in the Class of Service (CoS) group of the extensions to which this feature is to be allowed.
2. Change the 'Auto Redial Normal/Priority Count' and the 'Auto Redial Normal/Priority Timer' to match User preference. This will change the number of redial attempts made by the system and the interval between them.
3. If required, also change other related Timers such as Auto Redial Dial Tone Wait Timer, Auto Redial Ring Back Tone (RBT) Wait Timer, Auto Redial Ring Timer.

   All the above parameters can be programmed using Jeeves and by dialing SE commands from a telephone.

Programming Auto Redial using Jeeves

In the default factory settings, Station Basic Feature Template Number 01 is assigned to all extensions of the system. The Station Basic Feature Template 01 does not have the feature Auto Redial and Auto Redial Priority in the default CoS group 01. Thus none of the extensions of the SARVAM UCS have this feature.

If the User wants to allow all extensions the Auto Redial and /or the Auto Redial Priority feature, the you can simply enable this feature in the default CoS group 01.

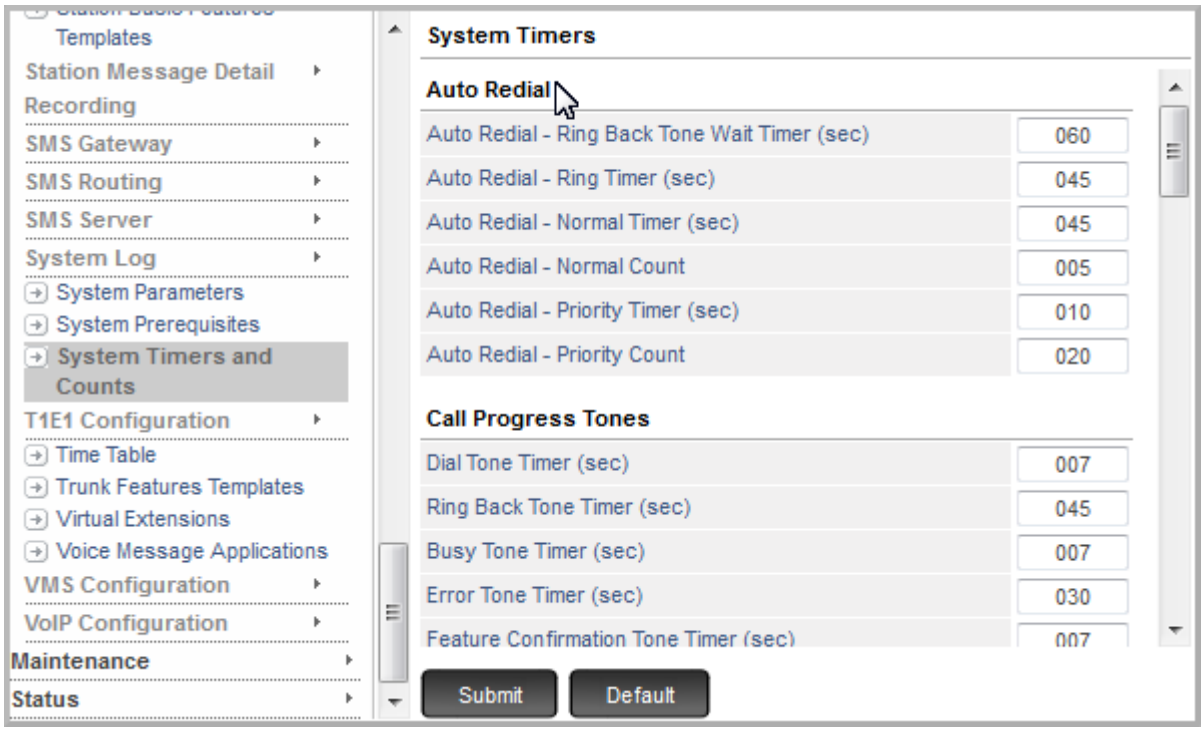
However, if Auto Redial/Auto Redial Priority is to be allowed on only select extensions, follow these steps:

1. Define a CoS group with Auto Redial/Auto Redial Priority enabled.
2. Prepare a Station Basic Feature Template with this CoS group applicable in all the "Time Zones".
3. Assign this new Template to the extensions to which Auto Redial/Auto Redial Priority is to be allowed.

   Refer the topics "Class of Service (COS)" and "Station Basic Feature Template" for detailed instructions.

To change Auto Redial Counts and Timers:

- Log in as System Engineer.
- Under **Configuration**, click **System Timers and Counts**.



- Below **Auto Redial**, change the Count and Timer of the type of Auto Redial - Normal or Priority - as per your requirement.
- You may change any of the related timers - Auto Redial Dial Tone Wait Timer, Auto Redial Ring Back Tone (RBT) Wait Timer, Auto Redial Ring Timer - as per your preferences on this page.
- Click **Submit** at the bottom of the page to save changes.

- *If the SARVAM UCS is installed in Australia, please ensure that:*
  i. *The Timer for Auto Redial Normal as well as Priority must be set to more than 5 seconds.*
  ii. *The Auto Redial Priority Count should be set to less than 15.*

- *Click Submit at the bottom of the page to save your setting.*

Programming Auto Redial using a Telephone

Refer the topics “Class of Service (COS)” and “Station Basic Feature Template” for instructions on how to use SE commands to:

- enable a feature (in this case Auto Redial and Auto Redial Priority) in the CoS group.
- assign the CoS group to a Station Basic Feature Template.
- apply the Template with the on DKP and SLT extensions, using SE commands.

To program the relevant Timers and count for Auto Redial:

- Enter SE mode from a DKP/SLT.

  To set Auto Redial Normal - Timer:
  - Dial command ***1704-Seconds***
    Where,
    Seconds is from 000 to 255.

  To set Auto Redial Normal - Count:
  - Dial command ***1705-Count***
    Where,
    Count is from 000 to 255.

  To set Auto Redial - Priority:
  - Dial command ***1706-Seconds***
    Where,
    Seconds is from 000 to 255.

  To set Auto Redial - Count:
  - Dial command ***1707-Count***
    Where,
    Count is from 000 to 255.

  To change Auto Redial RBT Wait Timer:
  - Dial command ***1702-Seconds***
    Where,
    Seconds is from 000 to 255 seconds.

Do not set this timer to less than 2 seconds.

  To change Redial Ring Timer:
  - Dial command ***1703-Seconds***
    Where,
    Seconds is from 000 to 255 seconds.

Do not set this timer to less than 2 seconds.

- Exit SE mode.

How to use

Auto Redial can be set/canceled from EON/Extended IP Phone as well as SLT.

For EON and Extended IP Phone Users

Using DSS Key

To set Auto Redial:

- When the external number you are trying is busy,
- Go ON-Hook on Busy Tone.
- Press the DSS Key assigned to 'Auto Redial' function.
- Go Idle.

To cancel Auto Redial:

- Press the DSS Key assigned to ‘Auto Redial’ function again.

Using Command:

To set Auto Redial:

- When the external number you are trying is busy,
- Go ON-Hook on Busy Tone.
- Dial 17.
- Go Idle or you get dial tone after 3 seconds.

To cancel Auto Redial:

- Pick up the handset.
- Dial 1070.
- Go Idle or you get dial tone after 3 seconds.

For SLT Users

To set Auto Redial

- When the external number you are trying is busy,
- Go ON-Hook on Busy Tone.
- Dial 17.
- Replace handset or you get dial tone after 3 seconds.

To cancel Auto Redial:

- Pick up the handset.
- Dial 1070.
- Replace handset or you get dial tone after 3 seconds.

Automatic Number Translation

What's this?

SARVAM UCS offers connectivity to different networks - PSTN, GSM, ISDN T1E1PRI, BRI, VoIP - each having a different numbering plan. For example, the GSM network requires area codes to be dialed also for local numbers, whereas the PSTN requires dialing of area codes for long distance calls.

When SARVAM UCS is connected with multiple networks, outgoing calls may be routed through any of these networks, depending on the routing pattern configured in the system. However, as extension users do not know through which telecom network their calls will be routed, they cannot be expected to dial numbers according to the numbering plan of the destination networks.

The Automatic Number Translation feature of SARVAM UCS takes care of this. It modifies/manipulates dialed numbers or part thereof to match with specific route numbering plan understood by the destination network (PSTN, GSM, VoIP). This includes adding or stripping off of country codes and area codes.

For example, when an extension user dials a local landline number, if Automatic Number Translation is so programmed, the SARVAM UCS will prefix the number with the appropriate country-area code before routing the call through the GSM network.

How it works

Automatic Number Translation makes use of the Automatic Number Translation (ANT) Table. The ANT Table consists of three columns:

- **Dialed Number:** This column contains the numbers you expect the users to dial.
- **Strip Digit:** This column contains the number of digit(s) to be stripped off by the system from the Dialed Number string before dialing it out.
- **Add Prefix:** This column contains the digit(s) which are to be added as prefix to the Dialed Number string by the system before dialing it out.

This table is applied on the desired trunk, through which outgoing calls are made.

Upto 8 different ANT tables can be configured and each table can accommodate upto 32 strings.

Here is an example of how this table is to be configured and used:

You want

- All 10-digit numbers to be dialed out after adding the prefix ‘1’.
- All 7-digit numbers, starting with 2 to be dialed out after adding the prefix ‘1315’.
- All numbers beginning with 91 to be stripped off the first 2 digits and ‘0’ to be added as prefix.

When you do not want to specify any numeric digits in the numbers to be modified, use the character **X**. This character represents any numeric digit from 0 to 9. For example, a 10-digit number (having the numeric digits from 0 to 9) can be represented using this character as **XXXXXXXXXX**.

Thus, the entries you will need to make in the ANT table will be as follows:

Dialed Numbers	Strip Digit	Add Prefix
XXXXXXXXXX	0	1
2XXXXXX	0	1315
91	2	0

- The entry for 10-digit numbers to be dialed out after adding the prefix ‘1’ will be as shown in the first row of this table. The 10-digit number is represented with the **X** character in the Dialed Numbers column. Since no digit is to be stripped off, ‘0’ is entered in the Strip Digit column. As the prefix ‘1’ is to be added, this number is entered in the Add Prefix column. The system will add 1 as prefix before dialing out numbers from 0000000000 to 9999999999.

- Similarly, the entry for 7-digit numbers starting with 2 to be dialed out after adding the prefix ‘1315’ will be as shown in the second row of this table. The system will add 1315 as prefix before dialing out numbers from 2000000 to 2999999.

- The entry for all numbers beginning with 91 to be stripped off the first 2 digits and ‘0’ to be added as prefix will be as shown in the third row of this table. When users dial numbers beginning with 91, the system will strip off the first two digits and add 0. For example, when a user dials the number 919925801882, the system will dial out 09925801882.

  Automatic Number Translation also forms the basis of “Multi-Stage Dialing”.

How to configure

To apply Automatic Number Translation on a trunk,

- Decide which of the trunks types are to be assigned the Automatic Number Translation (ANT) feature.
- Decide the number of ANT tables you need. You can program 8 different tables, with a maximum of 32 entries in each.
- Make the tables on a piece of paper by drawing three-column tables. In the first column of a the table, write the dialed numbers that need to be modified before being dialed out from the trunk. For each dialed number in the first column, enter the number of digits you want the system to strip off (if required) from this number in the second column. In the third column enter the number you want the system to add as prefix (if required) before dialing out the number.
- Configure the Automatic Number Translation Table in the system using the tables you prepared.
- Enable the Automatic Number Translation flag in the Outgoing Trunk Bundle (OGTB) of the trunk.
- Assign the Automatic Number Translation Table you configured to the OGTB of the trunk.

This can be done using Jeeves and a Telephone.

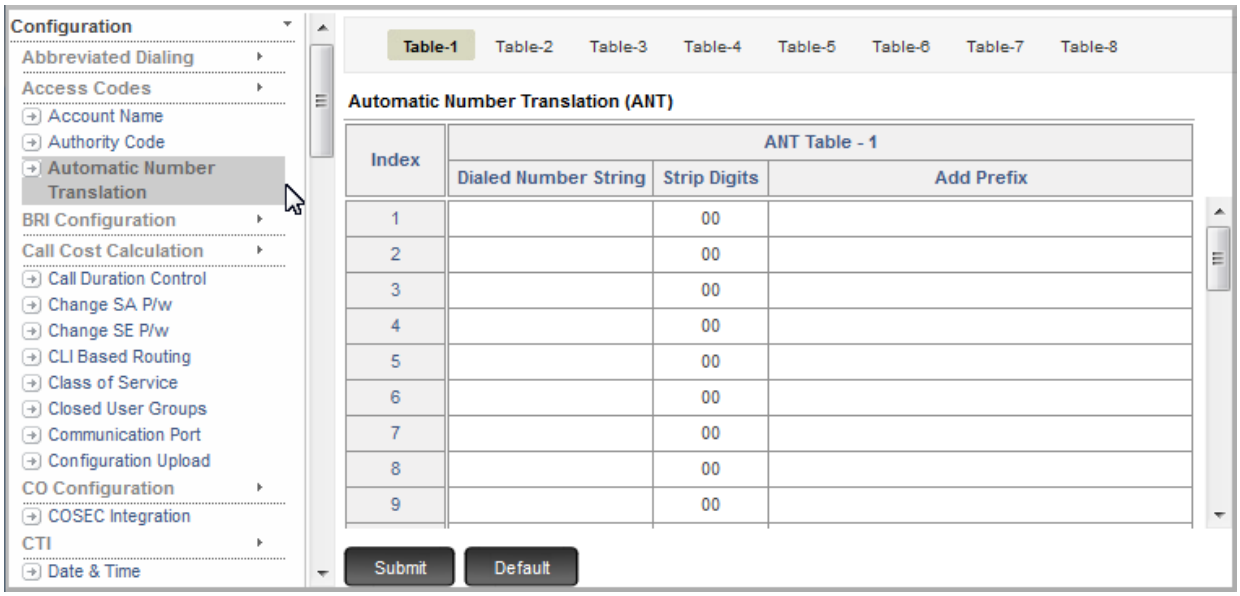
Configuring Automatic Number Translation using Jeeves

- Log in as System Engineer.

- Click **Configuration**.

Configuring the ANT Table

- Click **Automatic Number Translation**.

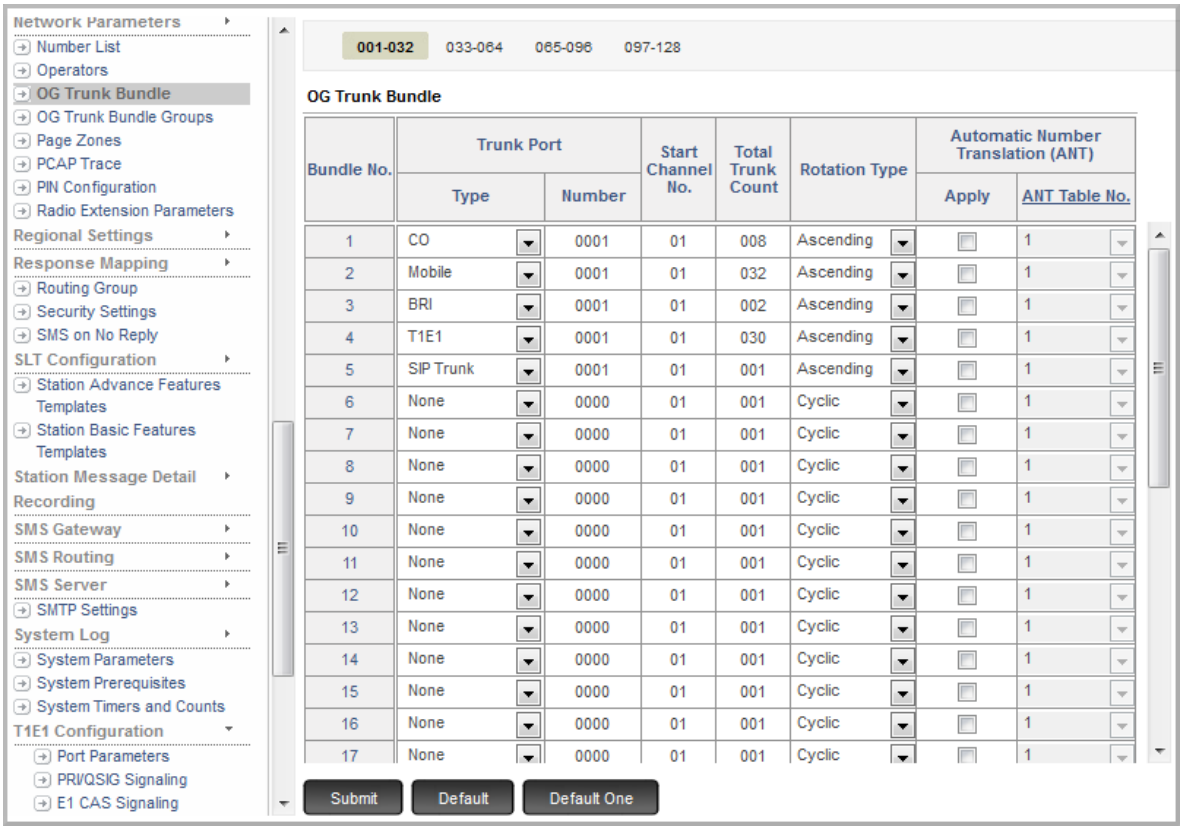


- Choose any table number from 1 to 8.
- In the **Dialed Number** column of the table you chose, configure the numbers you expect the extension users to dial. The Dialed Numbers can be a maximum of 16 characters. Default: Blank.

  When you want to specify number-length without specifying the numeric digits, use the character **X** to represent the digit. **X** represents any number from 0 to 9.
- In the **Strip Digit** column, enter the number of digits you want the system to strip off from the Dialed Number before the dialing out this number. The valid range is from 00 to 16. Default: 0.
- In the **Add Prefix** column, enter the number you want the system to add as prefix to the Dialed Number before the system dials out this number. The Prefix can be a maximum of 40 characters. Default: Blank.
- Click **Submit** to save your entries.

Enabling Automatic Number Translation on Trunks

- Under **Configuration**, click **Outgoing Trunk Bundle** to open the page.
- In the OG Trunk Bundle of the desired trunk, enable the **Automatic Number Translation (ANT) Apply** flag by selecting the check box.



- In the **ANT Table No.** list, select the table number you configured for this trunk.
- Repeat the steps to apply the feature on another OGTB.
- Click **Submit** at the bottom of the page to save changes.

Configuring Automatic Number Translation using a Telephone

- Enter SE mode from a DKP/SLT.

The ANT Table can be configured using Jeeves only.

To enable Automatic Number Translation Flag on OGTB, dial:

- ***6702-1-OG Trunk Bundle Number-Feature Number-Code*** to enable the flag in a single trunk bundle.
- ***6702-2-OG Trunk Bundle Number-OG Trunk Bundle Number-Feature Number-Code*** to enable the flag in a range of trunk bundles.
- ***6702-*-Feature Number-Code*** to enable the flag in all trunk bundles.

  Where,
  OG Trunk Bundle Number is from 001 to 128.
  Feature Number for Automatic Number Translation flag is '5'.
  Code is:
  0 for Disable
  1 for Enable
  Default: Disabled.

For example:
To enable Automatic Number Translation flag in OGTB number 1, dial: ***6702-1-001-5-1***
To enable the same flag in OGTB numbers 1 to 8, dial: ***6702-2-001-008-5-1***
To enable the same flag in all OGTBs, dial: ***6702-*-5-1***

To assign an Automatic Number Translation Table, dial:

- ***6702-1-OG Trunk Bundle Number-Feature Number-Code*** to assign a Table to a single trunk bundle.
- ***6702-2-OG Trunk Bundle Number-OG Trunk Bundle Number-Feature Number-Code*** to assign the same Table to a range of trunk bundles.
- ***6702-*-Feature Number-Code*** to assign the same Table to all trunk bundles.
  Where,
  OG Trunk Bundle Number is from 001 to 128.
  Feature Number for Automatic Number Table is '6'.
  Code is ANT Table number from 1 to 8.

  For example:
  To assign ANT Table 3 to OGTB number 1, dial: ***6702-1-001-6-3***
  To assign ANT Table 3 to OGTB numbers 1 to 8, dial: ***6702-2-001-008-6-3***
  To assign ANT Table to all OGTBs, dial: ***6702-*-6-3***

- Exit SE mode.

Barge-In

What's this?

Barge-In allows you to break into an on-going conversation between two extension users, between an extension user and an external caller as well.

Barge-In can be used by Operators to transfer Incoming calls to busy extensions. The Operator can put the caller on hold, barge into the busy extension to inform about the call, and then transfer the call.

Barge-In can be used by a Boss to interrupt the secretary's busy extension.

SARVAM UCS offers flexibility to allow/deny Barge-In feature to an extension user, that is, allow the extension user to barge into on-going conversations. It also provides the flexibility to prevent conversations of extension users from being barged in, referred to as Privacy against Barge-In.

How it works

- A, B and C are users of the system.
- A and B are talking to each other.
- C calls A.
- C gets busy tone.
- C dials Barge-In feature code.
- C gets Ring Back tone (RBT) and A gets beeps indicating a new call. If A is using EON, C's name and number appear on C’s phone display.
- C gets RBT and A gets beeps for Barge-in timer. (By default, 10 seconds)
- During the beeps, A may press 'Flash' to answer the call.
- If A does not respond till the end of the Barge-In Timer (set to 10 seconds, by default), A gets connected to C. B is put on hold and is given hold-on music.

Feature Interactions

- **Call States:**
  - Barge-In works only if the dialed extension is busy. The dialed extension may be busy with another extension or trunk (external number).
  - Barge-In cannot be used when accessed trunk is busy.
  - Barge-In works only if the user about to be barged in is in a two-way normal speech with another user or external party. However, it will not work if the conversation is being recorded.

- It will not work if the busy signal is due to the user being OFF-Hook, or in the middle of dialing, or accessing a feature of the System.

- **"Call Toggle":** Once A and C comes in speech with each other, A can toggle between B and C using Call Toggle feature.

- **Privacy against Barge-In:** If the feature 'Privacy against Barge-in is enabled for an extension, it cannot be barged into.

- **"Priority":** No Interaction with Barge-In. If 'A' has lower priority than 'B' but has Barge-In enabled; A can barge in B.

- **"Do Not Disturb (DND)":** Barge-In will not work if the called user has set DND. If 'A' has set DND. A is busy with C. B calls A. B cannot barge in A.

- **DND Override:** Barge-In will work if the calling user is allowed DND-Override. If 'A' has set DND. A is busy with C. B calls A. On busy signal, B dials the Barge-In code. Barge-In will be successful only if B has DND-Override enabled.

- **"Call Taping"**: Barge-In will not work when the two-way conversation between the users is being taped.

How to configure

The functioning of this feature is controlled by three parameters, 'Barge-In', 'Privacy against Barge-In' and 'Barge-In Timer'.

Barge-In and Privacy against Barge-In

First decide which of the extensions are to be allowed Barge-In and the extension that are to be protected against Barge-In.

In the default factory settings, Station Basic Feature Template Number 01 is assigned to all the extensions of SARVAM UCS. The Station Basic Feature Template 01 is assigned CoS group 01. The default CoS group 01 has both Barge-In and Privacy from Barge-In are disabled. So, none of the extensions of the SARVAM UCS can use these features.

If you want to allow Barge-In to the all extensions, simply enable Barge-In in the default CoS group 01.

However, if Barge-In is to be allowed on only selected extensions then follow these steps:

a. Define a CoS group with Barge-In enabled.

b. Prepare an extension Basic Feature Template with this CoS group applicable in all the "Time Zones".

c. Assign this new Template to the extensions to which Barge-In is to be allowed.

Repeat the above steps to allow 'Privacy from Barge-In' in the CoS of extensions that are to be exempted from Barge-In.

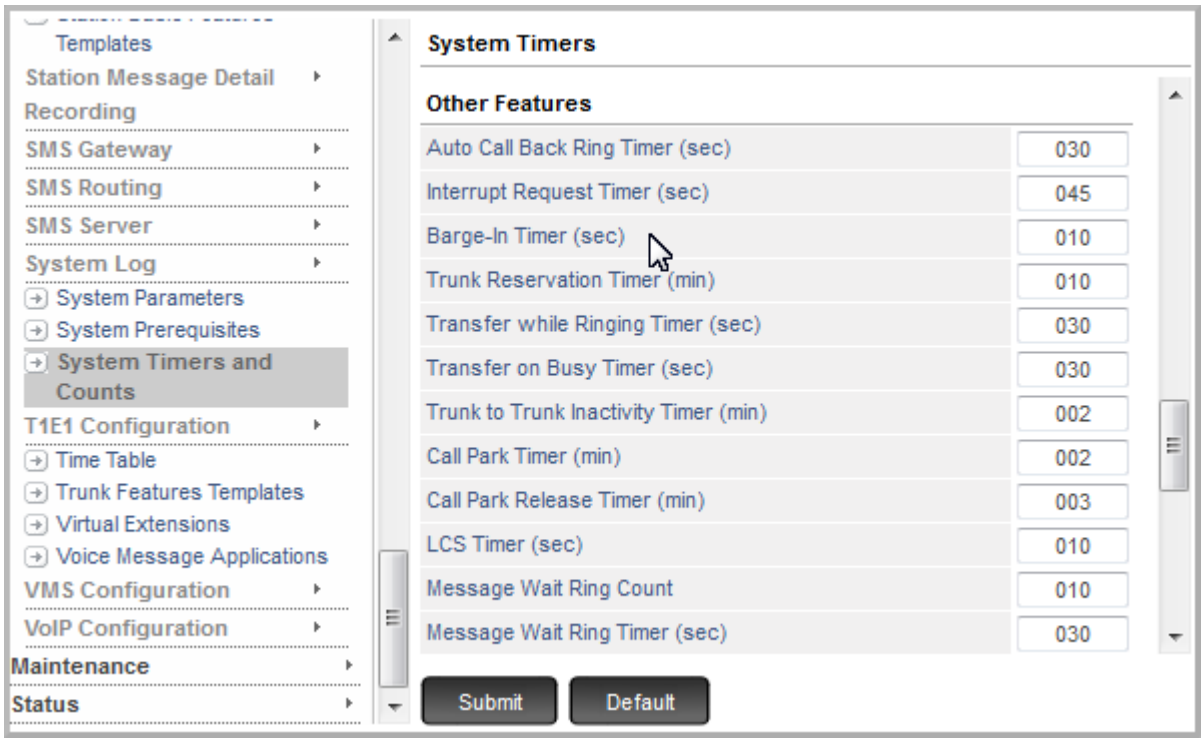
Refer the topics "Class of Service (COS)" and "Station Basic Feature Template" for detailed instructions on programming.

Barge-In Timer

Barge-In Timer is the time after which the caller gets connected to the called party. By default the Timer is set to 10 seconds.

Changing Barge-In Timer using Jeeves

- Login as System Engineer.
- Under **Configuration**, click **System Timers and Counts** to open the page.
- Scroll to reach **Other Features**.
- Go to the parameter **Barge-In Timer (sec)**.



- Set the desired value for the Timer.
- Click **Submit** at the bottom of the page to save changes.
- Log out of Jeeves or continue, as required.

Changing Barge-In Timer using a Telephone

- Enter SE mode.
- Dial command ***3803-Seconds***
  Where,
  Seconds is from 001 to 255 seconds. Default is 10 seconds.
- Exit SE mode.

How to use

For EON and Extended IP Phone Users

- Dial an extension.
- If the extension is busy, you get Busy Tone.
- Press DSS Key assigned to 'Barge-In' function.

  OR

- Dial 4[251].
- You get Ring Back Tone.
- Wait for the system to connect you to the called extension.
- Talk.
- Replace the handset after the conversation has ended.

For SLT Users

- Dial an extension.
- If the extension is busy, you get Busy Tone.
- Dial 4.
- You get Ring Back Tone.
- Wait for the system to connect you to the called extension.
- Talk.
- Replace the handset after the conversation has ended.

251. This default feature access code can be changed to suit your preference. Refer the topic "Access Codes".

BCCH Selection

What's this?

BCCH Selection feature enables you to lock the Mobile Port of SARVAM UCS to a particular cell or channel or BTS (Base Transceiver Station) for various reasons such as:

- better network availability
- minimum call drop due to bad signal/ network failure, etc.

- *This feature is not applicable if CDMA Mobile Card is installed in your system.*
- *To use this feature, make sure GSM[252] or LTE is selected as the Preferred Network Mode of the Mobile Port. For instructions, see "Mobile Port Parameters" in "Configuring Mobile Trunks".*

How it works

In the GSM network, each BTS is assigned one particular channel called as ARFCN (Absolute Radio Frequency Channel Number), which is transmitted by BTS in BCCH (Broadcast Control Channel).

Now, when SARVAM UCS is switched on, the Mobile Port gets registered with the network on a particular BTS which has the highest signal strength. However, the signal strength is not consistent. It keeps fluctuating, resulting in call drop or poor voice quality.

Therefore, to avoid this, SARVAM UCS enables you to lock the Mobile Port to a particular cell or channel manually after checking Signal Strength and Signal Quality of each cell.

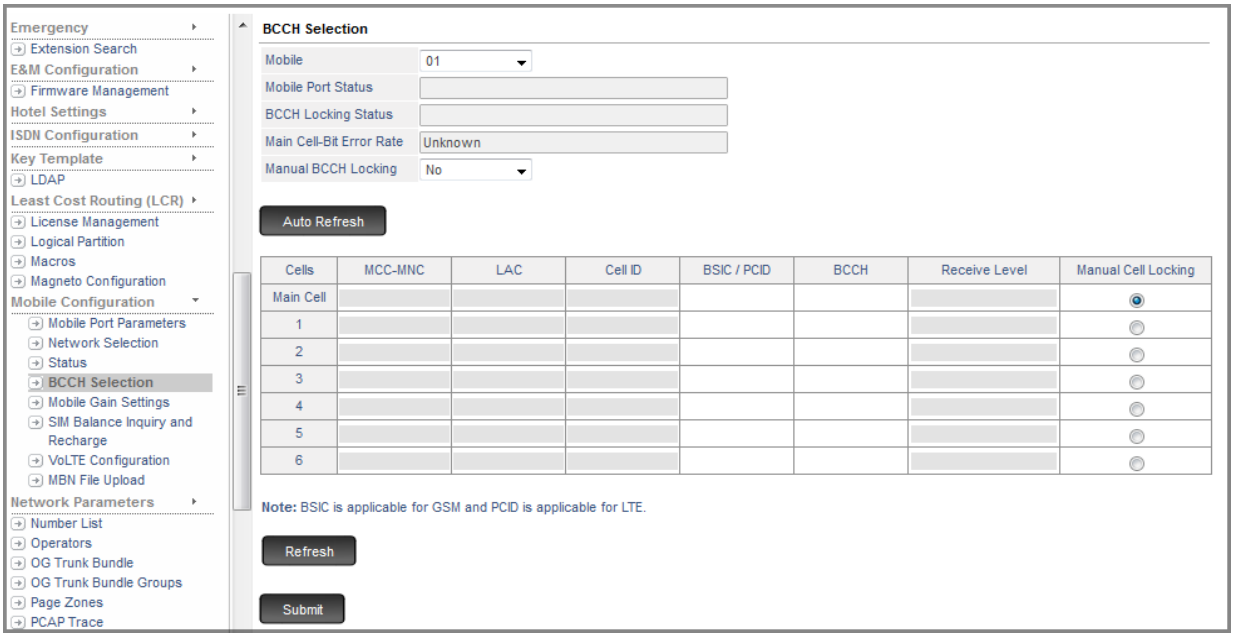
How to configure

You can lock Mobile Port to a cell or a channel only through Jeeves.

- Log in as System Engineer.
- Under **Configuration**, click **Mobile Configuration**.

252. BCCH Selection will not be supported if Quectel UC20 is installed in your system.

- Click **BCCH Selection** to open the page.



- The page displays the following parameters:
  - **Mobile Port Number:** This is number of the Mobile port for which BCCH Selection status is displayed. You can choose a different Mobile Port number from the drop down list. The page will display the BCCH Selection related parameters for the selected mobile port.
  - **Mobile Port Status:** The current state of the Mobile Port is displayed in this field. Given below is the description of the various status indication messages that will appear in this field.

STATUS	DESCRIPTION
Disabled	Displayed when Mobile Port is disabled.
GSM Initialization	Displayed when GSM module is in initialization state, that is, before SIM detection.
SIM Absent	Displayed when SIM Card is not detected by the system.
SIM PIN wrong	Displayed when wrong SIM PIN is issued.
SIM PUK required	Displayed when SIM PUK is required.
Registering	Displayed when the Mobile Port is in registration process with the Network.
Idle	Displayed when the Mobile Port is registered with the Network and it is free.
Busy	Displayed when any active call is present on the Mobile Port.

- **BCCH Locking Status:** The current BCCH Locking status of the mobile port is displayed in this field. Given below is a description of the various BCCH Locking status indication messages that will appear in this field.

STATUS	DESCRIPTION
Trying to Lock	Displayed when user selects Manual BCCH Locking as 'No' from 'Yes' and module is in initialization process after system or module restart.
Trying to lock on BCCH xxxxx, BSIC / PCID xxx	Displayed when BCCH Locking is selected as Manual and the Mobile Port is in the registration process with the Network. xxxxx is the BCCH selected by the user for locking the cell.
Manually Locked on BCCH xxxxx, BSIC / PCID xxx	Displayed when BCCH Locking is selected as Manual and Mobile Port is successfully registered with the Network. xxxxx is the BCCH selected by the user for locking the cell.
Auto Locked on BCCH xxxxx	Displayed when BCCH Locking is selected as Auto and Mobile Port is successfully registered with the Network. xxxxx is the BCCH of the Main Cell. xxxxx is updated as per the changes in the Main Cell's BCCH.

BSIC is applicable for GSM, whereas PCID is applicable for LTE.

- **Main Cell- Bit Error Rate (%):** Bit Error Rate of the Main Cell is displayed in this field. Bit Error Rate (BER) is the percentage of received bits on a digital link that are in error relative to the number of bits received. Bit Error Rate is calculated from the received signal quality.

- **Manual BCCH Locking:** This parameter allows you to lock the Mobile Port to a particular cell of your preference. By default, manual BCCH locking is set to 'No'. When manual BCCH locking is set to 'No', Mobile Port gets locked to the cell as per the highest signal strength. Select 'Yes' if you want to lock the Mobile Port to the particular cell selected by you.

- *When you apply Manual cell lock with a Service Provider, make sure while changing the SIM, the SIM from the same service provider is inserted.*

- *During Manual BCCH locking if the SIM is not getting registered with the selected cell, you must select No as the Manual BCCH Locking option.*

- **Auto Refresh:** Click this button to refresh BCCH Selection page. All parameters on this page will be downloaded automatically after every 15 seconds. By default, Auto Refresh button is enabled.

- **Stop Auto Refresh:** By clicking this button, you can stop the system from automatically refreshing the BCCH Selection page every 15 seconds. When you stop Auto Refresh, you must click 'Refresh' at the bottom of this page to refresh the page whenever you want

- **Cells:** Indicates the cells with which the Mobile Port can be locked. You can decide to lock the Mobile Port with a particular cell after considering the following cell related parameters, which appear on the page:

  - **MCC-MNC:** In this field, MCC-MNC of a cell is displayed. Mobile Country Code (MCC) is a three digit number uniquely identifying a country and Mobile Network Code (MNC) is either a two or three digit number used to identify a given network from within a specific country.

  - **LAC (Location Area Code):** In this field, LAC (Location Area Code) is displayed. LAC uniquely identifies a location area within a GSM PLMN (Public Land Mobile Network). The maximum length of

LAC is 16 bits ranging from 0 to 65535. LAC is displayed in hexadecimal characters for SIMCOM-2G and Wavecom-2G engines which ranges from 0000 to FFFF. For SIMCOM-3G engine, LAC is displayed in decimal digits which ranges from 00000 to 65535.

- **Cell ID:** In this field, Cell ID is displayed. It is a 16-bit identifier that identifies the cell. Cell ID is displayed in hexadecimal characters for SIMCOM-2G and Wavecom-2G engines which ranges from 0000 to FFFF. For SIMCOM-3G engine, Cell ID is displayed in decimal digits which ranges from 00000 to 65535.

- **BSIC/PCID:** In this field, BSIC (Base Station Identification Code) is applicable for GSM whereas PCID (Physical Cell Identity) is applicable for LTE. BSIC/PCID allows a mobile extension to distinguish between different neighboring base stations. BSIC/PCID is a three-digit value ranging from 0 to 255.

- **BCCH (Broadcast Control Channel):** In this field, the BCCH value of the cell is displayed. BCCH defines the frequency channel number.

- **Receive Level:** In this field, the Receive Signal Strength level of the cell is displayed. It is the average Receive Signal Strength of the cell. Its value ranges from -110 dBm to -47 dBm.

- **Manual Cell Locking:** This radio button is for locking a Mobile Port to a selected cell manually.

Now, to lock a Mobile Port to a particular cell,

- Select the desired Mobile Port Number from the drop down list.
- Set the parameter **Manual BCCH Locking** to **Yes**.
- Go to the Cell to which you want to lock the Mobile Port you selected.
- Select the radio button **Manual Cell Locking** of that Cell.
- Click **Submit** at the bottom of the page.
- The BCCH Locking for the selected Mobile Port will appear on this page, if Auto Refresh is enabled.
- If you have stopped Auto Refresh, click 'Refresh' at the bottom of the page to refresh the page and view the current BCCH Locking settings of the selected Mobile port.
- You may now log out of Jeeves.

Example:
Consider the following example when using this feature:

Problem:
- SARVAM UCS is installed in roaming area, where more than one network is available, say A and B.
- Mobile Network Selection is set to 'Manual' mode and the first priority is programmed as network A and the second priority is programmed as network B.
- The Mobile Port gets registered with A network. After registration, the user locks the Mobile Port to one of the cells of A network.
- After registration, if the module or the system restarts or gets de-registered from the network, module starts registration process again.
- While re-registering, SARVAM UCS tries to lock the Mobile Port to the last selected cell of network A.
- If network A is unavailable then the Mobile Port will not get registered with the network.

Solution:
- In this situation, user should set Manual BCCH locking mode to 'No' to register Mobile Port with the suitable network automatically.
- Later, the user can set the Manual BCCH locking mode to 'Yes' and lock the Mobile Port to the desired cell after assessing the cell information.

Behind the System Application

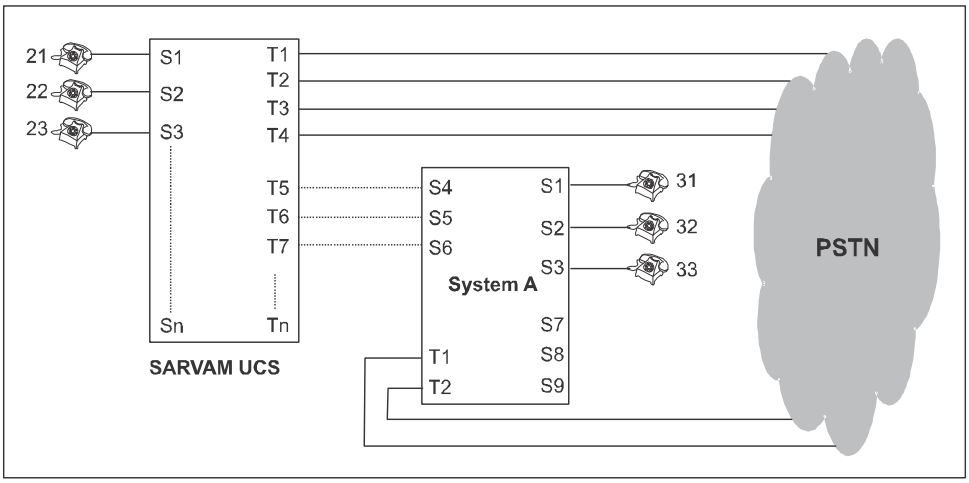
What's this?

It is common for small and medium Systems to be connected to larger System systems, where the trunks of the larger System are connected to the stations of the smaller system. This is usually done for the purpose of expanding the capacity of the large System already in use.

Such configurations are referred to as 'Behind the System Application'.

How it works

Consider the following illustration.



System-A is connected behind SARVAM UCS. In this 'Behind the System' configuration, the Trunk Lines T5, T6, T7 of SARVAM UCS are connected to the Stations (SLT) S4, S5, S6 of System-A.

However, Trunk lines T1 and T2 of System-A are connected directly to the PSTN.

In such application scenarios, implementing toll control restrictions for the trunks is a difficult task for SARVAM UCS.

For example: Extension number 21 of SARVAM UCS in the above illustration is not allowed the facility of long distance dialing. It has access to all the CO trunks.

When the user of Extension 21 wants to access T1, T2 or T3 (which are direct trunks from the PSTN to SARVAM UCS) the user dials '0' (Trunk Access Code programmed), gets PSTN dial tone. When the user dials the number, SARVAM UCS applies Toll Control.

When the user of Extension 21 tries to grab a trunk T5, T6 or T7 (which are connected to stations of System-A) by dialing Trunk Access Code, for example, '0', the user gets the dial tone of System A. This means, the user of Extension 21 must dial '0' again to grab PSTN dial tone of the T1/T2 connected to System-A.

But when the user dials '0' again, SARVAM UCS plays an Error Tone, because SARVAM UCS has applied Toll Control and since Extension 21 is not allowed long distance dialing, SARVAM UCS rejects dialing on trunk and plays error tone.

This would not have been a problem if Extension 21 were allowed long distance dialing. Since Extension 21 cannot be allowed long distance dialing, SARVAM UCS provides a solution for this in the form of a programmable Pre-PSTN Digit Count (PPDC) for each CO trunk.

The Pre-PSTN Digit Count defines the number of digits to be dialed to reach the PSTN. The system will apply Toll Control check for the extension only after the programmed PPDC.

PPDC is to be programmed only for trunks that are connected to another System, and not for Trunks connected directly to the PSTN. To take the above illustration further, PPDC must be programmed only for T5, T6, and T7.

PPDC count programmed should have the same number of digits as the Trunk Access Codes programmed for System-A. For example, if the Trunk Access Code is a single digit number, such as '0', the PPDC will be '1'. If Trunk Access Code is a two-digit number, such as 61, the PPDC will be '2'.

Since PPDC is not applicable on trunks directly connected to the PSTN, it must be programmed as '0' for T1, T2, T3, T4 of SARVAM UCS.

How to configure

The 'Pre-PSTN Digit Count' (PPDC) is to be programmed in the "CO Hardware Template" applied to the CO trunks of the System that are connected to station ports of the other System as well as to CO trunks that are directly connected to the PSTN.

- For CO Trunks that are directly connected to the PSTN, PPDC must be programmed as '0'.
- For CO Trunks that are connected to the stations of another System, PPDC must be programmed as per the number of digits in the Trunk Access Codes defined for the second System.

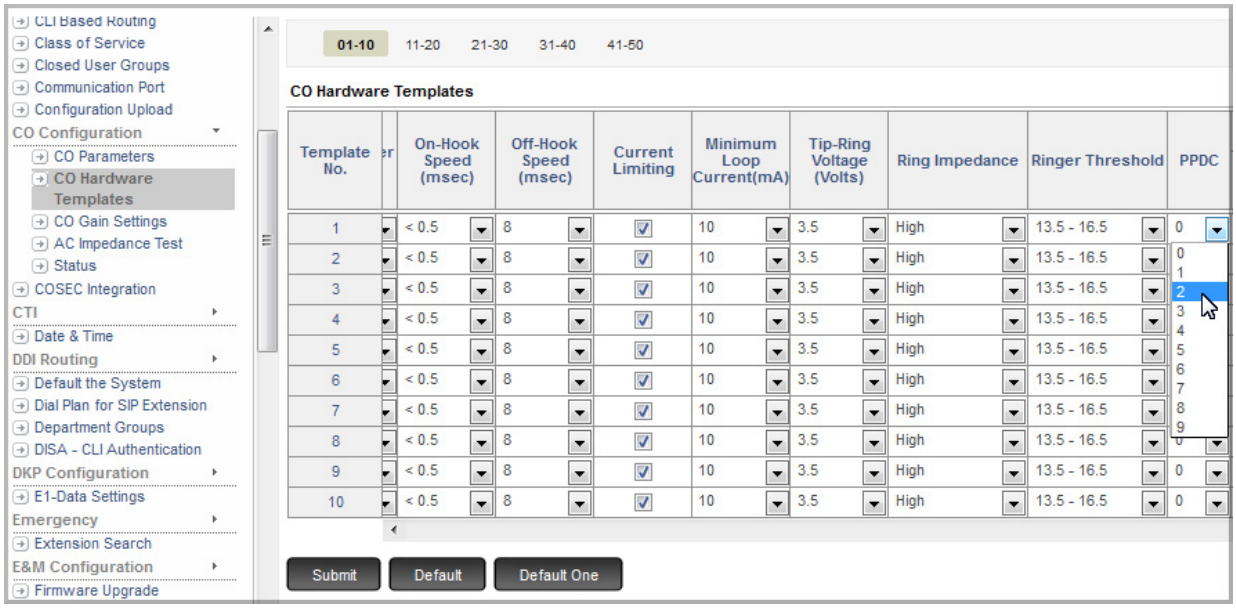
This can be done using Jeeves as well as from a Telephone.

The PPDC should be programmed only for 'Behind the System Applications'. For all normal applications, this count must be set to '0' for all the trunks. Otherwise, external number dialing may be hampered. Features like Least Cost Routing and Station Message Detail Recording will also be affected.

Configuring Pre-PSTN Digit Count (PPDC) using Jeeves

- Log in as System Engineer.
- Under **Configuration**, click **CO Configuration**.
- Click **CO Hardware Template** to open the page.

- Scroll with the horizontal scroll bar to reach the **PPDC** column of the template.



- By default CO Hardware Template Number 01 is assigned to all trunks. The default 'PPDC' in this template is '0'.
- For all trunks that are to be assigned PPDC '0' (that is, trunks connected directly to the PSTN), you may retain this template.
- For trunks that are to be assigned a PPDC count from 1 to 6 (that is, trunks connected to the stations of another System), prepare another CO Hardware Template by selecting another template number, for instance Template 02.
- From the drop down list, select the appropriate value. This would depend on the number of digits in the Trunk Access Code defined for the trunks in the other System. If the TAC is single digit, select '1'. If TAC is double or triple digit, select '2' or '3' as applicable as the PPDC.
- Click **Submit** at the bottom of the page to save your setting.
- Apply the CO Hardware Template to the CO ports. To do this,
  - Click **CO Parameters** to open the page.
  - Enter the number of the template you prepared (Template 02) in the field **CO Hardware Template** for each port you want to assign this template.
  - For Trunks to be assigned **PPDC** Count **0**, retain CO Hardware Template Number 01.
  - Click **Submit** at the bottom of the page to save your settings.
  - Log out or continue programming.

Configuring Pre-PSTN Digit Count (PPDC) using Telephone

- Enter SE mode from a DKP/SLT.

  To program PPDC Count in CO Hardware Template:
  - Dial command ***5902-1-Template Number-08-Code***
    Where,
    Template Number is the CO Hardware Template, from 01 to 50. Default: 01
    38 is the Feature Number for PPDC
    Code is the PPDC Count, from 0 to 6.

    For instance: To program PPDC
    '1' in Template Number 02, dial: ***5902-1-02-38-1***
    '2' in Template Number 03, dial: ***5902-1-03-38-2***

  To assign the CO Hardware Template now programmed with the PPDC to trunks, dial:
  - ***5903-1-CO-Template Number*** to apply the template on a single CO trunk port.
  - ***5903-2-CO-CO-Template Number*** to apply the same template on a range of CO trunk ports.
  - ***5903-*-Template Number*** to apply the same template on all CO trunk ports.
    Where,
    CO is the number of the Software port of CO Trunks, from 001 to 128.
    Template Number is CO Hardware Template number programmed with the PPDC, from 01 to 50.

    For instance: To apply Template Number 02 to CO 003 to 005, dial ***5903-2-003-005-02.***

- Exit SE Mode.

Also, refer the topic “CO Hardware Template” to know more.

Building Intercom

What's this?

SARVAM UCS offers the Building Intercom solution by integrating Telecom and Security systems, for commercial and residential buildings, such as malls, shopping complexes, residential apartment blocks and gated-communities.

With the Building Intercom solution, you can also connect private networks, that is few Systems can be connected to each other using VoIP Network.

To use the Building Intercom solution you can either purchase the ILC cards or the SLT cards. For details, refer to *"Installing ETERNITY MENX", "Installing ETERNITY LENX", "Installing ETERNITY GENX"* and *"Installing ETERNITY PENX"*.

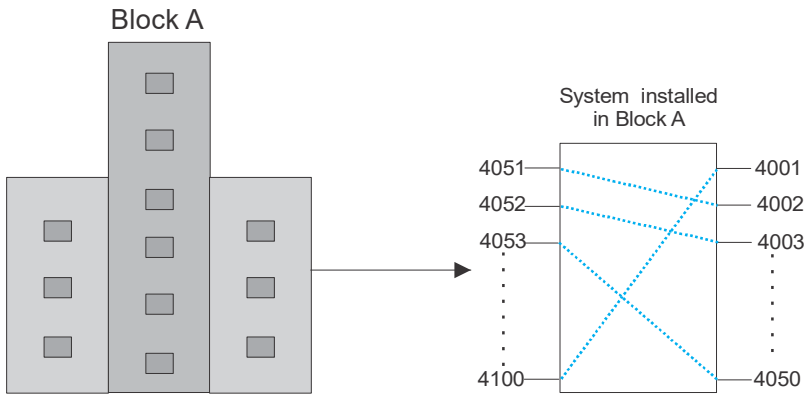
How to use

Building Intercom has two applications:

- Standalone Application
- Extended Application

Standalone Application[253]

In this case the system is installed at the site where internal communication is required between internal members in a same complex only.



Here Block A is a residential complex with 100 flats. SARVAM UCS with ILC Cards is installed for internal communication between the residents.

All the extensions 4001 to 4100 can communicate with each other.

Extended Application

Let us understand how to use the Building Intercom Application with the following illustration:

253. SARVAM SMB (PENX) does not support Standalone Application.

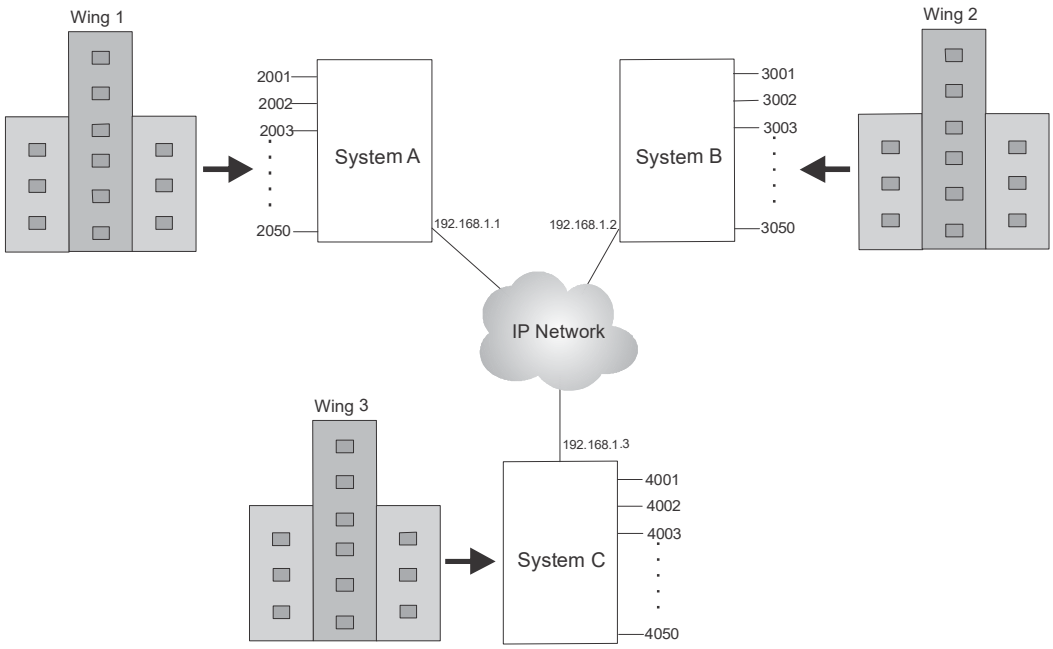
A residential society has three buildings. Each building has 50 flats. In each building a System (having Intercom Line Cards) is installed. The residents in the same building only, can communicate with each other.
The residents of all the three buildings want to communication with each other.

To overcome this, you must install a Vocoder Module in each system. All the three buildings will be connected using the IP network as shown in the figure below.

For installing the Vocoder Module in:

- ETERNITY LENX, see “Installing the VOCODER Module”.
- ETERNITY MENX, see “Installing the VOCODER Module”.
- ETERNITY GENX, see “Installing the VOCODER Module”.
- ETERNITY PENX, see “Installing the VOCODER Module”.

In this case, System A, System B and System C will be connected to form a single group.The residents in Wing 1, 2 and 3 will be able to call each other using extension numbers.



Intercom calls can be made between Wing 1, 2 and 3 with suitable configurations of the System.

- **At each location**, you need to do the following configuration in the System:
  - Select a SIP trunk to be used for this application and enable it. For example, SIP Trunk 1.
  - Set the **SIP Trunk Mode** of this trunk to **Peer-to-Peer**.
  - In **Treat Incoming call as** select **Station**.
  - Keep the **SIP ID** field of this SIP trunk **blank**.

  For detailed instruction, see “Configuring SIP Extensions”.

- In **System-A**, to route the calls configure the following:
  - The CUG table

- The Peer-to-Peer table.

To configure the CUG table in System A,

- In the **Route Code** field of the CUG table, enter the Number that will be dialed to call the users of System B and C. In this case, 3 and 4 respectively. As the system uses the best match logic to match number strings in the CUG table, you may configure only the prefix of the number to be dialed, instead of configuring the complete number string.
- For the number you entered, in the **Dialed Digit Count** field, enter 4.
- Select the **OG Trunk Bundle Group**. Configure SIP Trunk 1 as the only member in this group. The calls will be routed through this SIP Trunk only.
- Configure the **Strip Digit Count** as 0 and clear the **Self Route** check box.

  The CUG table you configure in System A would look like this:

Index	Route Code	OG Trunk Bundle Group	Strip Digit Count	Self Route	Dialed Digit Count
1	3	01	0		4
2	4	01	0		4

  For detailed instructions, see “Closed User Group (CUG)”.

- To configure the Peer-to-Peer table in System A,
  - In the **Number** field of the Peer-to-Peer table, enter the numbers of the extension users of System B and C. In this case, 3 and 4, respectively. As the system uses the best match logic to match number strings in the Peer-to-Peer table, you may configure only the prefix of the number to be dialed, instead of configuring the complete number string.
  - For the number 3001 enter the **Domain Address**, enter the Domain Name/IP Address of System B. In this case, 192.168.1.2 and for 4001 the Domain Address is 192.168.1.3
  - You can assign a **Name** to the numbers you have configured.
  - The SIP messages can be transported using UDP, TCP or TLS. Select the **Default Transport for Outgoing Message** as per your requirement for each index.

    The Peer-to-Peer table you configure in System A would look like this:

Index	Number	Domain Address	Name	Default Transport for Outgoing Message
1	No Match Found			

Index	Number	Domain Address	Name	Default Transport for Outgoing Message
2	3	192.168.1.2	System B	TCP
3	4	192.168.1.3	System C	TCP

- When 2001 from System A dials 3001, the system compares it with the CUG table configured in System A. When a match is found in the CUG table, the system uses SIP Trunk 1 to route the call.
- SIP Trunk 1 is configured as a Peer-to-Peer trunk, hence the system will check the Peer-to-Peer table.
- As 3001 is configured in the Peer-to-Peer table, the system fetches Destination address and transports the SIP messages using the protocol selected as the Default Transport for Outgoing Message.
- The system places the call on System B, using SIP Trunk 1.
- When there is an incoming call on System B, the system checks the CUG table first and as 3001 is not programmed in the CUG table, it checks the flexible number of the extensions.
- As 3001 is found in the flexible number list, the call is routed to the extension 3001.

When 2001 dials 4001, the call will be routed as per the above logic.

Similarly you must configure the parameters in System B and C.

In certain cases different building may have the same extension numbers, in these cases you must configure the CUG Table with Exchange ID. For detailed information, see “Closed User Group-With Exchange ID”.

Busy Lamp Field for Trunks

What is this?

On SIP extensions, SARVAM UCS supports the feature Busy Lamp Field (BLF) Subscription for Trunks, enabling SIP extension users to monitor the status of desired trunks.

Using BLF, SIP extension users can monitor the status of different Trunk types, namely CO, BRI, T1E1, Mobile and SIP. In the case of T1E1 and BRI Trunks, the state of calls on each channel can be monitored using BLF.

To provide SIP extension users this feature, you must enable this feature on their SIP phones and configure the trunk to be monitored on the BLF key of their SIP phones. The number of trunks that can be monitored will depend on the number of BLF keys supported on the SIP Phones.

With BLF subscription and BLF key configured, whenever, there is a change in the state of the monitored trunk, SARVAM UCS sends a NOTIFY message to the SIP Extension. The NOTIFY message contains the Call State. On receiving the NOTIFY message, the SIP Extension updates the LED indication of the BLF key on the SIP phone.

The SIP extensions will indicate the following calls states for the outgoing and incoming calls on the monitored trunks:

Outgoing Calls	
Call State	Description
Trying	When an outgoing call is made through the monitored trunk.
Confirmed	When the external party answers the call and speech is established with the extension user, that is, the call is matured.
Hold	When the call on the trunk has been put on hold by the extension.
Available/Idle	When the SIP EXtension user disconnects the call.

Incoming Calls	
Call State	Description
Early	When an indication is received from the Network that the external party is ringing.
Confirmed	When the incoming call is placed on the SIP Extension as the destination and speech is established with the extension user, that is, the call is matured.
Hold	When the call on the trunk has been put on hold by the SIP extension.
Available/Idle	When the SIP Extension user disconnects the call

SIP phones may differ in the BLF indication (LED color and cadence, text message display) they provide for the Call States. Refer to the manufacturer's documentation for BLF Indication supported on the SIP phones.

How it works

- An Standard SIP Phone is registered as a SIP Extension, with the extension number 3301.
- As the user of extension 3301 wants to monitor the trunk CO-001, BLF subscription is enabled on extension 3301 and CO-001 is assigned to the BLF key on the SIP phone.
- Extension 3301 makes an outgoing call to an external number 2630555. The BLF key will indicate the current call state of the CO-001 Trunk as “Trying” according to the LED indication supported by the SIP phone for this call state.

  If the SIP phone supports text message display for call states, each call state will be displayed on the phone.
- When the external party answers the call, the call between the CO-001 and SIP Extension 3301 gets matured. The BLF key will indicate the current call state of the CO Trunk as “Confirmed” according to the LED indication supported by the SIP phone for this call state.
- When the SIP Extension 3301 disconnects the call, the SARVAM UCS will disconnect the call of external number and the BLF key will display the call state of the CO as “Terminated” according to the LED indication supported by the SIP phone for this call state.

  Similarly, the BLF key configured on the SIP Extension 3301 will display the call states of the CO-001 trunk for incoming calls from external numbers.

Since multiple calls can be made through a single trunk, the BLF key will indicate the status of the first call detected by the system. When the first call is terminated, the status of the second call (if ongoing) will be indicated. Similarly the status of all subsequent calls will indicated after the previous call is terminated.

How to configure

To provide BLF to SIP extension users, you must do the following:

- Enable **Busy Lamp Field Subscription** on the SIP Extensions you want to provide this feature. For instructions, see “Configuring SIP Extension using Jeeves” under “Configuring SIP Extensions”.
- Assign a BLF Key for the trunk to be monitored on the SIP Phones registered as extensions. For instructions refer to the manufacturer’s documentation (Installation Guide/User Guide) for the respective SIP Phones.
- To monitor the trunks, configure the BLF Key as per the table given below:

Trunk	User ID part in SUBSCRIBE	Remarks
CO	COxxx	Here, xxx should be a valid CO port number, that is, from 001 to 128.
BRI	BRIxxCHyy	Here, xx should be a valid BRI port number, that is, from 01 to 32 and the yy should be a valid channel no., that is, 01 and 02
T1E1	T1E1xxCHyy	Here, xx should be a valid T1E1 port number, that is, from 01 to 08 and the yy should be a valid channel number, that is, 01 to 30.

Trunk	User ID part in SUBSCRIBE	Remarks
Mobile	MOBxxx	Here, xxx should be a valid Mobile port number, that is, from 001 to 064.
SIP	SIPxx	Here, xx should be a valid SIP Trunk number, that is, from 01 to 32.
E&M	ENMxxx	Here, xxx should be a valid ENM port number, that is, from 001 to 128.

Call Back on Trunk Ports

What's this?

The feature Call Back on Trunk Ports is used to respond to missed calls from particular numbers on the different trunk ports of SARVAM UCS: SIP Trunks, T1E1PRI trunks, BRI trunks, Mobile trunks and CO Trunks.

When Call Back feature is enabled on a trunk port, and there is a missed call on that trunk port, the SARVAM UCS determines if the calling number is eligible for a call back or not. It calls back the same number or an alternative number programmed for that number, either from the port on which it was received or from a different port, depending on the programming. SARVAM UCS can be programmed to choose the most cost effective line to call back the missed call numbers.

Employees at remote locations can use this feature to have the SARVAM UCS installed in their office call them back, thereby saving on charges (for example, roaming charges on mobile calls), where applicable.

How it works

For this feature to work:

- Call Back must be enabled on the desired Trunk Ports.
- The CLI of those callers whom the system should call back must be programmed in the ‘Call Back Incoming Number List’.
- The ‘Call Back Timer’ may be programmed. When the caller disconnects within the Call Back Timer, the Call Back will be applied for that number.
- You must define ‘Call Back on’, that is, you must select whether the number which must be called back should be the same CLI number which the call was received or an alternative number.
- The number on which call back is to be made must be programmed in the ‘Call Back Outgoing Number List’, if it is not the same CLI number or if it is an alternative number.
- You must select whether the call back should be made using the same trunk port on which the call was received or an Outgoing Trunk Bundle Group (OGTBG). If you select OGTBG, you must also program the OGTBG.
- You may enable Least Cost Routing (LCR) on the OGTB if you want the system to select the least cost trunk for calling back the missed call number. Program LCR accordingly.
- Select a ‘Call Back Mode’, that is, how the call should be routed when the call back is answered by the remote party; whether it should be routed through Built-In Auto Attendant, DISA or Operator.

Following is an example of a Call Back on a mobile port, when the above parameters are programmed.

- Caller A calls mobile port 01.
- The system checks if the Call Back flag is enabled on mobile port 01.
- The flag is enabled.

- The system matches the CLI of A with the Call Back Incoming Number List assigned to mobile port 01 to determine if the calling number is eligible for a call back.
- A match is found on Index 15 of the Call Back Incoming Number List.[254]
- The system waits for the period of the Call Back Timer (programmable, default: 10 seconds).
- A must disconnect before the expiry of the Call Back Timer so that the system can treat it as a Missed Call.
- If A disconnects within the Call Back Timer, the system applies Call Back for A's number.
- The system checks the 'Call Back on' parameter, whether it has to call back the same number or an alternative number.
- If an alternative number is programmed as 'Call Back on', the system checks the Outgoing Call Back Number List for the alternative number. As the CLI of A matches with the number on Index 15 of the Call Back *Incoming* Number List, the system checks Index 15 of the Call Back *Outgoing* Number List for the corresponding alternative number to this number.
- The system checks if the number is to be called from the same port or an OGTBG.
- If the same port is programmed, the system will make a call to the number using mobile port 01.
- If OGTBG is programmed, the system will check if Least Cost Routing is enabled in the OGTBG and make the call back accordingly.
- When A answers the call,
- The system checks the type of Call Back Mode enabled on mobile port 01 (the port on which the call back request was made).

  Four scenarios are possible:

  1. "Auto Attendant" is enabled as Call Back Mode on mobile port 01.
     - A gets dial tone of SARVAM UCS.
     - A can now use Built-In Auto Attendant feature.
  2. 'Pin Authentication - Multiple Calls' or 'CLI Authentication - Multiple Calls' is enabled as Call Back Mode on mobile port 01.
     - A gets dial tone of SARVAM UCS.
     - A can now reach any station or trunk of SARVAM UCS from DISA Mode.
  3. 'CLI Authentication - Single Call Answer Signaling' is enabled as Call Back Mode on mobile port 01.
     - A gets dial tone of another trunk of SARVAM UCS.

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254. If the system does not find a match for the CLI of the caller in the Call Back Incoming Number List, the 'Call Back' feature will not be applicable and the call will be processed according to the normal incoming call logic.

- A can make calls from the trunk.

4. ‘Operator’ is enabled as Call Back Mode on mobile port 01.

   - A gets Ring Back Tone.

   - The system lands the call on the Operator extension assigned to mobile port 01.

Read the topics “Auto Attendant”, “Direct Inward System Access (DISA)” and “Configuring 'Operator'” to know more about the call respective call logic.

- *Since this feature is essentially for callers, they must be aware of its functioning to be able to use it, that is, disconnect the call within the Call Back Timer. If the caller does not disconnect within the Call Back Timer, the call will be processed according to the normal incoming call logic.*

- *SARVAM UCS supports only one call back request at a time, for one trunk port. The second incoming call on that trunk port will be processed by the system as per normal incoming call routing.*

- *For call back requests made from an OGTBG, if any of its trunks is busy, SARVAM UCS will support only the last call back request in the OGTBG. Previous requests will be processed as per the normal incoming call management logic.*

How to configure

For this feature to function, you must program the following parameters on each Trunk port type (CO, BRI, T1, E1, Mobile, SIP) on which you want to use this feature:

- **Enable Call Back:** This flag must be enabled on the desired trunk port on which you want to activate the Call Back on Trunk Port feature. By default, this flag is disabled on all trunk port types.

- **Call Back Timer:** This is the duration for which the system waits for the caller to disconnect the call after the system has found a matching number for the caller’s CLI in the Call Back Incoming Number List.

  When the caller disconnects within Call Back Timer, the system applies Call Back on the port. If the caller does not disconnect within the Call Back Timer, the incoming call management logic is applied for the call on the trunk port.

  The range of this timer is from 01 to 99 seconds. By default, it is set to 10 seconds.

- Call Back **Incoming Number List:** This is the list of numbers that are eligible for Call Back. The system checks the CLI of the caller with this list to determine if the caller is eligible for a call back.

  The system compares the number string programmed in the Call Back Incoming List with the number string received as CLI.

  Number string programmed in the 'Call Back Incoming Number List' shall be compared with the actual received CLI.

  The number string programmed in the Call Back Incoming Number List may be shorter than the number string received as CLI, but only if the programmed number string completely matches with the received CLI from the right towards left, the system will consider it as a complete match.

For example, if the programmed string is 263055 and the number string received in the CLI is 2652630555, the system will consider it a complete match. If the received CLI 912652630555, the system will consider this caller too as eligible for a call back. Thus any CLI received with 263055 as the last 7 digits will be considered as match found.

By default, 'Number List' 15 is assigned to all trunk port types as Call Back Incoming Number List. You may program this list for all port types, or you may program another Number List and assign it to the particular trunk port type.

Refer the topic "Number Lists" for instructions on how to configure the Number List.

- **Call Back on:** For each Trunk port type you have set the Call Back feature, you must define 'Call Back on', that is, you must select whether the number which must be called back should be the same number from which the call was received or a different number.

  When missed call is eligible for call back (matches with Incoming Number list), the 'Call Back on' parameter determines the number on which the call back is to be made, that is, whether on the same number from which the missed call is received or on a different number.

  In countries where CLI received on trunks can be dialed out without any modification, you may select 'CLI Number' as 'Call Back on' option.

  In countries where CLI received on trunks can be dialed only after appropriate modification, you may select "Alternate Number' as the 'Call Back on' option. You may also select 'Alternate Number' as Call Back on when you want the call back to be made to a different number.

- Call Back **Outgoing Number List:** When the system finds a missed call eligible for a call back, it will make the call back on the basis of the Call Back on option you selected and the Outgoing Number List you programmed.

  If you selected 'CLI Number' as "Call Back on' option, you do not need to program the corresponding outgoing number for the CLI received.

  However, if the CLI received needs to be modified before being dialed out, then program the modified CLI in the Outgoing List as the corresponding outgoing number for the CLI received.

  The modified CLI or the Alternate number should be programmed at the same index number as the index number at with the received CLI is programmed in the Call Back Incoming Number List. For example, for the received CLI number string programmed at Index 15 in the Call Back Incoming Number List, the corresponding modified CLI/Alternate number string should be programmed at the same Index, 15, in the Call Back Outgoing Number List.

  When the CLI received matches with the number string programmed at Index 15 of the 'Call Back Incoming Number List', the call back will be made using the (modified/Alternate) number programmed at Index 15 of the 'Call Back Outgoing Number List'.

  By default, 'Number List' 16 is assigned to all trunk port types as Call Back Outgoing Number List. You may program this list for all port types, or you may program another Number List and assign it to the particular trunk port type.

  Refer the topic "Number Lists" for instructions on how to configure the Number List.

If you have selected 'Alternate Number' as 'Call Back on' option, but do not want to provide alternative numbers to call back particular callers (that is, CLI received), in such a case, program the CLI of these callers in the Incoming Number List but keep the corresponding index numbers in the Outgoing Number Lists blank.

- **Call Back from:** This parameter determines the trunk port to be used to make call back.The call back can be made using the same port or an Outgoing Trunk Bundle Group (OTGTBG). Select 'Same port' if you want the call back to be made using the same port on which the missed call was received. If you select OGTBG, the call back will be made using the OGTBG, which you have defined.

- **OGTB Group** for Call Back**:** If you selected OGTBG for making the call back in the previous parameter, you must assign the OGTBG that must be used in this parameter.

  By default, OGTBG 01 is selected for Call Back.

  If you want the system to select the lowest cost trunk for making the call back, enable Least Cost Routing on the OGTBG that you define here for Call Back.

- **Call Back Mode:** Select from the following options how a 'Call Back' call answered by the remote party should be routed:

  - **Built-In Auto Attendant:** The system will process the call as per the Built-In Auto Attendant call logic - give a dial tone to the remote party, who can now call any extension. Refer the feature description for "Auto Attendant".

  - **PIN Authentication-Multiple Calls:** The system will process the call as per DISA call logic - allow remote party to enter DISA mode with PIN-Authentication. On successful authentication (DISA Login) the user is allowed to make calls or use features as allowed to him/her.

  - **CLI Authentication-Multiple Calls:** The system will process the call as per DISA call logic, allowing the remote party to enter DISA mode with CLI Authentication-Multiple calls as authentication method and level of access.

  - **CLI Authentication-Single Call:** The system will process the call as per DISA call logic, allowing the remote party to enter DISA mode with CLI Authentication-Single call as authentication method and level of access. Refer the feature description for "Direct Inward System Access (DISA)".

  - **Operator:** When the remote party answers the Call Back call, the system will route the call to the Operator[255].

  By default, Operator is selected as the Call Back Mode.

All these parameters may be programmed using Jeeves or by dialing SE commands from a telephone.

To program Call Back on different port types, refer the relevant topics mentioned below:

- For Call Back on SIP Trunks, refer the topic "Configuring SIP Trunks".
- For Call Back on T1E1 Ports, refer the topic 'Call Back on T1E1 Trunk Ports', under "Configuring E1 Trunks" and "Configuring T1 Trunks".
- For Call Back on BRI Ports, refer the topic 'BRI Parameters' under "Configuring BRI Trunks".

255. 'Operator' is the station which is assigned to the Mobile port in the Trunk Feature Template. Refer "Trunk Feature Template" to know more.

- For Call Back on Mobile Ports, refer the topic “Configuring Mobile Trunks”.
- For Call Back on CO Ports, refer the topic “Configuring CO Trunks”.

Call Budget on Extension

What's this?

Call Budget is a cost control feature that allows you to keep a tab on the total cost of phone call made by extension users.

With this feature, each extension can be allotted a 'budget' limit for outgoing calls, which is automatically reloaded at the start of every month.

Long distance calls form a major part of the increased cost of telephone calls. Though excessive use or misuse of long distance dialing can be restricted using Toll Control, there may be extension users whose nature of work requires them to make long distance calls. Instead of denying them the facility, their telephone bill can be limited to a certain amount using Call Budget.

With a Call Budget allotted to the extension, the user is free to make calls as long as s/he does not cross the budget limit. Once the user exceeds the budget limit, the extension can be denied access to long distance dialing.

The extension user can be assigned a fresh budget, after which s/he can resume making long distance calls.

Call Budget can be enabled on all the extensions as well as on selected extensions. Each extension can be assigned a different amount depending on user requirement.

How it works

When an extension allotted Call Budget makes a call,

- The system checks the current call budget amount of the extension.
- If the consumed amount is within the budget limit allotted to the extension,
    - The system allows the extension to make the call as per the “Toll Control Levels” assigned to it.
    - After the call ends, the system calculates and adds the call amount to the extension's account. Thus it calculates and updates the total cost of calls made from the phone.
- If the consumed amount exceeds the budget limit allotted to the extension,
    - The system considers this as Call Budget exhausted.
    - The system allows the extension to make the call as per the Toll Control-Call Budget Consumed assigned to the extension.
    - After the call ends, the system calculates and adds the call amount to the extension's account.
- Until a new Call Budget is allocated to the extension user, the extension user can make calls only as per Toll Control assigned for the Call Budget Consumed state.
- Once a new Call Budget is allocated, the extension user can make calls as per the “Toll Control” assigned to the extension.

- If the budget exceeds anytime during the month, and if no fresh budget amount is allotted, the system allows calls to be made as per the Allowed and Denied List of Toll Control-Call Budget Consumed till the end of the month. From the 1st day of the following month, the system automatically reloads the budget amount. The extension can now make calls.

- The Call Budget allotted to extension is valid for one month. The system automatically reloads the budget at the start of every month.

- The budget amount can be changed or allotted afresh to extensions from the System Administrator (SA) mode, at any time. The Call Budget allotted by the SA will be reloaded in the following month.

- *Call Budget is not based on real time (online) call cost calculation. The SARVAM UCS calculates the call cost only after the call has ended.*

- *So, if the Call Budget allotted to an extension user gets exhausted in the middle of a call, the call will not get disconnected, though the budget exceeds. To prevent this from occurring, the System Engineer may program the "Call Duration Control (CDC)" feature.*

- *Call Budget is dependent on precise Call Cost Calculation. So, SMDR parameters and long distance codes must be programmed properly to prevent errors in calculation.*

- *This feature works independent of any Call Accounting Software (CAS) installed with the SARVAM UCS.*

- *The SARVAM UCS will calculate cost of phone calls made by extension phones even when no call budget is allocated[256].*

How to configure

The working of this feature is controlled by three parameters: **Call Budget** flag, **Toll Control-Call Budget Consumed** and **Preset Call Budget Amount** (this parameter is applicable only for the Hotel Mode).

These parameters can be programmed using Jeeves and Telephone.

Call Budget flag and Toll Control-Call Budget Consumed

To enable Call Budget feature on an extension, the System Engineer must enable the Call Budget flag and define the Toll Control-Call Budget Consumed in the Station Basic Feature Template assigned to the extensions.

In the default Station Basic Feature Template 01 assigned to all extensions of the SARVAM UCS, the Call Budget flag is disabled and the Toll Control-Call Budget Consumed is set to 'No Calls'.

If Call Budget is to be allowed to all extensions, simply enable the flag in the default Station Basic Feature Template 01 and select the Toll Control for Call Budget Consumed state.

However, if Call Budget is to be allowed to selected extensions, then prepare a separate Station Basic Feature Template with the Call Budget flag enabled and the Toll Control-Call Budget Consumed set. Now, apply this template on extensions that are to be allowed this feature.

Refer the topic "Station Basic Feature Template" for detailed instructions for programming a feature in the template and assigning templates to extensions.

256. Based on the feature 'Call Cost Calculation'.

Preset Call Budget Amount[257]

The amount of '9999' is set as default Call Budget in the system. This value may be changed by the System Engineer according to the organization's practices. For example, if the organization wants to allocate a fixed amount of $10 to all extension users, the Call Budget value can be set to '10'.

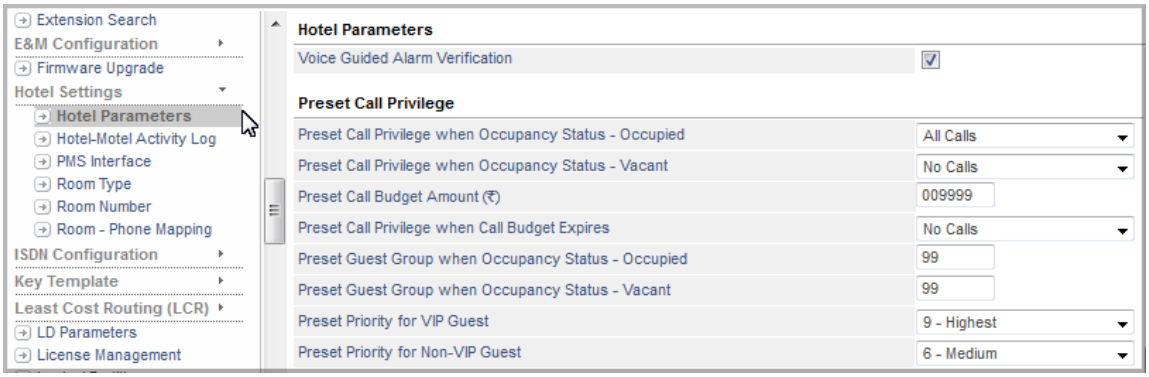
The new Call Budget set by the System Engineer will be considered as the Preset Call Budget amount. This amount will be allocated at the start of every month to all extensions having Call Budget feature in their Station Basic Feature Template.

Further, the System Administrator (SA) can override the Preset Call Budget amount set by the System Engineer, and allot call budgets on an extension-by-extension basis. For example: allotting higher amount to extensions of senior managers, Marketing, Sales, Exports departments, and lower amount to extensions that are less likely to make long distance calls frequently.

The amount may be greater or lesser than the default amount set by the System Engineer. The Call Budget amount allotted by the SA will be reloaded at the start of every month on the extension. For instructions refer 'How to Use' later in this section.

Changing the Preset Call Budget Amount using Jeeves

- Log in as System Engineer.
- Under **Configuration**, click **Hotel Settings**.
- Click **Hotel Parameters**.
- Scroll to **Preset Call Privilege** and change the **Preset Call Budget Amount** to the required value.



- Click **Submit** to save changes.

Changing Preset Call Budget Amount using a Telephone

- Enter SE mode from a DKP/SLT.
- Dial command ***3710-Preset Call Budget Amount***
  Where,
  Preset Call Budget Amount is any amount of 6-digits.
  If the amount is less than 6 digits, use leading zeros.
  For example: To assign Call Budget = $10, dial ***3710-000010***
- Exit SE mode.

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257. Applicable only for the Hotel Mode.

- Dial '***00***' to exit from an SLT
- Dial '***00***' and press '***Enter***' key to exit from a DKP.

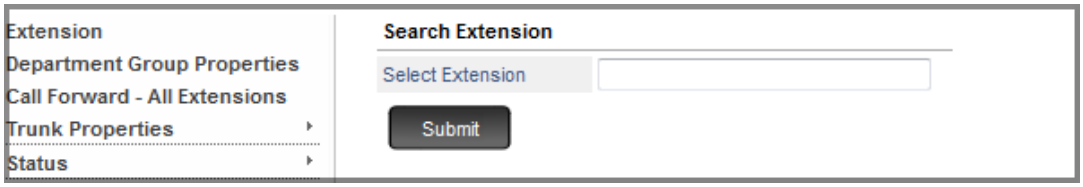
- *The amount programmed as Preset Call Budget is to be considered as the local currency.*
- *At the time of installation, when the SE selects the Region Code (country code) and defaults the system, the related Currency Code is applied.*
- *The currency symbol will not be displayed on the Operator's phone, on account of the limited number of characters that can be displayed.*
- *The local currency symbol will appear at the relevant places in the outgoing SMDR reports.*

How to use

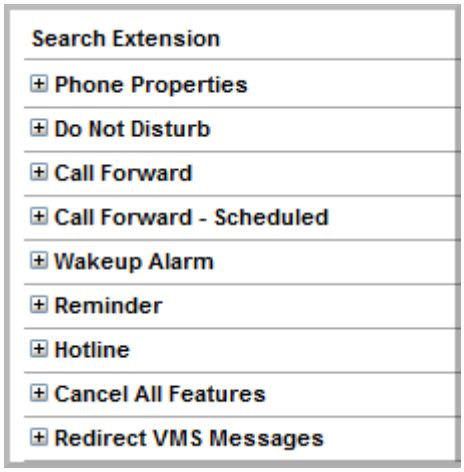
Call Budget amount can be allotted to extensions from the System Administrator mode, using Jeeves or by dialing SA commands from an extension phone.

Assigning Call Budget to Extensions using Jeeves

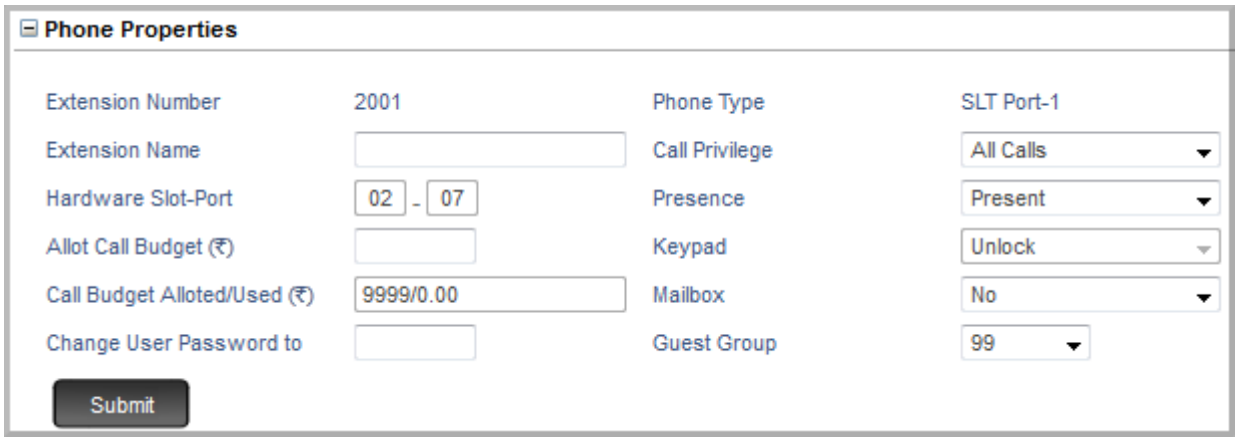
- Log into Jeeves as System Administrator.
- Click **Extension**.



- In **Select Extension,** enter the Number or the Name of the extension on which you want to set this feature
- Click **Submit**.
- The searched extension user details appear on your screen.



- Click **Phone Properties** to expand.



  - In **Allot Call Budget**, enter the amount you want to assign to the user as budget limit for outgoing calls.

    To re-assigning a new amount before the previous balance is consumed, make sure you add the available balance to the new amount. Enter this amount in Allot a Call Budget.

    For example, if you have allotted an amount is Rs.1000 and the consumed amount is Rs.600. The available balance is Rs.400. Now, if you want to assign a new amount of Rs.500. In Allot a Call Budget you must enter 900 (Balance + New = 400 + 500).

  - The **Allotted Amount/Used** displays the amount allotted to the user as well as the call budget amount consumed by the user for making outgoing calls.

  - Click **Submit** button to save.

  - To allot call budget to another extension, follow the same instructions as above.

For EON and Extended IP Phone Users

To assign Call Budget to an Extension:

- Press DSS Key assigned to ‘Set Call Budget for Remote Extension' function.

  OR

- Dial ***1072-004*** (from SA mode).
- Enter Extension Number that is to the assigned a budget.
- Enter Call Budget Amount.
  Use leading zero if amount is fewer than 6 digits.
- You get confirmatory message.
- Go idle.

To view Call Budget assigned to an Extension:

- Press DSS Key assigned to the ‘Call Budget Extension’ function.

  OR

- Dial ***1072-011*** (from SA mode).

- Enter Extension Number
- Call Budget assigned to the extension appears on your phone display.

For SLT Users

- Lift the handset.
- Dial ***1072-004*** (from SA mode).
- Dial Extension Number.
- Dial Call Budget Amount.
  Use leading zero if amount is less than 6 digits.
- You get confirmation tone.
- Replace handset.

Call Budget cannot be viewed on SLT.

Call Budget on Trunk

What's this?

Call Budget on Trunks is an expense control feature of SARVAM UCS that allows you to keep track of the cost of phone calls made from the different Trunk ports of SARVAM UCS.

With this feature, each trunk can be allotted a 'budget' limit for outgoing calls. This budget limit can be programmed to be reloaded manually each time it is exceeded or at a scheduled date, either daily or at a particular date of the month.

There are three types of Call Budget limit that can be set on the trunks:

- **Amount:** In this type of Call Budget, a fixed amount is assigned to the trunk. By default the amount of 999999 (to be considered in the local currency) is set as Call Budget Amount on trunks. With Amount-based Call Budget you can control the actual expense incurred on making calls from a trunk.

- **Minutes:** In this type of Call Budget, a fixed number of Minutes are assigned to the trunk. By default, 999999 minutes are assigned as Call Budget Minutes on trunks. This type of Call Budget is useful when the Service Provider offers 'Free' minutes. For example, the Service Provider allows the customer to make calls for the first 1000 minutes every month. This offer can be availed of by programming Minutes-based Call Budget on the trunk port.

- **Number of Calls:** In this type of Call Budget, you can define the maximum number of calls that can be made from a trunk. By default, the maximum number of Call Budget - Calls is set to 9999 calls on the trunks. This type of Call Budget is useful when the Service Provider offers a certain number of free calls or a certain number of free calls for a fixed period. For instance, the Service Provider offers 150 free calls per month.

With a Call Budget allotted to a trunk, the users can make calls from the trunk as long as the budget limit set for the trunk—Amount or Minutes or Maximum number of Calls—is not crossed. Once the budget limit exceeds calls will not be routed through this trunk.

The consumed Budget can be reset, after which the trunk becomes functional again and allows outgoing calls to be made. The consumed Call Budget can be reset manually, that is, anytime, as required/desired, or on a scheduled date either daily or on a particular date of the month.

By default, *Call Budget Type - Minutes* is enabled on all the trunk port types - SIP, T1E1PRI, BRI, Mobile, CO for *300 Minutes*. You can change the configurations as per your requirement, refer “How to configure” and “Outgoing Call Routing”.

How it works

Call Budget can be enabled on trunk port types - SIP, T1E1PRI, BRI, Mobile, CO - all at once or on selected trunk port types from among them. Each trunk can be assigned a different Call Budget, depending on the requirement of the users.

When Call Budget is enabled on a trunk port, for each outgoing call,

- The system checks the type of Call Budget set on the trunk - Amount, Minutes or number of Calls.

- It checks the Call Budget consumed.

Call Budget- Amount

- When Amount-based Call Budget is selected, the Amount should be assigned to the trunk.
- At the end of each outgoing call made from the trunk, the system will calculate the cost of the call on the basis of the Pulse Rate Type programmed. The system will thus calculate the total amount consumed after the end of each call. Refer the topic “Call Cost Calculation (CCC)” to know more.

Call Budget - Minutes

- When Minutes-based Call Budget is set, the total minutes for which calls will be allowed from the trunk port must be defined.
- With the number of Minutes defined, at the end of each call, the system will calculate the duration of the call on the basis of the units programmed in the Pulse Rate. The system will calculate the consumed minute on the basis of the duration of the call. Refer the topic “Call Cost Calculation (CCC)” to know more.

Call Budget - Number of Calls

- When the Call Budget is based on 'Number of Calls', the maximum number of calls to be allowed from the trunk port is to be defined.
- With the number of calls programmed, the system will maintain a count for the number of matured outgoing calls made from that trunk port.
- Thus for each matured call, the Number of Calls-Count is incremented, irrespective of the actual duration of the matured call.

When the assigned 'cost' or 'minutes' or 'number of calls' assigned to trunk is exhausted, SARVAM UCS will:

- print 'system activity log'.
- bar or limit outgoing calls from such trunks.
- play an Error Tone to the extension users who attempt to access such trunks using Selective Trunk Access.
- However, incoming calls will remain unaffected, and will be allowed on these trunks.

The consumed Call Budget Amount/Minutes/Calls can be reset manually at any time from the System Administrator mode or the System Engineer mode or can be programmed to be automatically reset either daily or on a particular date of the month.

The current Call Budget Amount/Minutes/Calls limit can be changed from the System Administrator (SA) mode, at any time. If scheduled reset of consumed Call Budget is programmed, then the Call Budget allotted by the SA will be reloaded on the scheduled date.

Once a new Call Budget is allocated to the trunk, outgoing call facility is resumed on the trunk.

- *Call Budget on Trunks is not based on real time (online) call cost calculation. The SARVAM UCS calculates the call cost only after the call has ended.*
- *If the Call Budget allotted to a Trunk Port gets exhausted in the middle of a call, the call will not disconnected, though the budget is exceeded.*

- *Call Budget on Trunks is dependent on precise Call Cost Calculation. So, SMDR parameters and long distance codes must be programmed properly to prevent errors in calculation.*
- *This feature works independent of any Call Accounting Software (CAS) installed with the SARVAM UCS.*

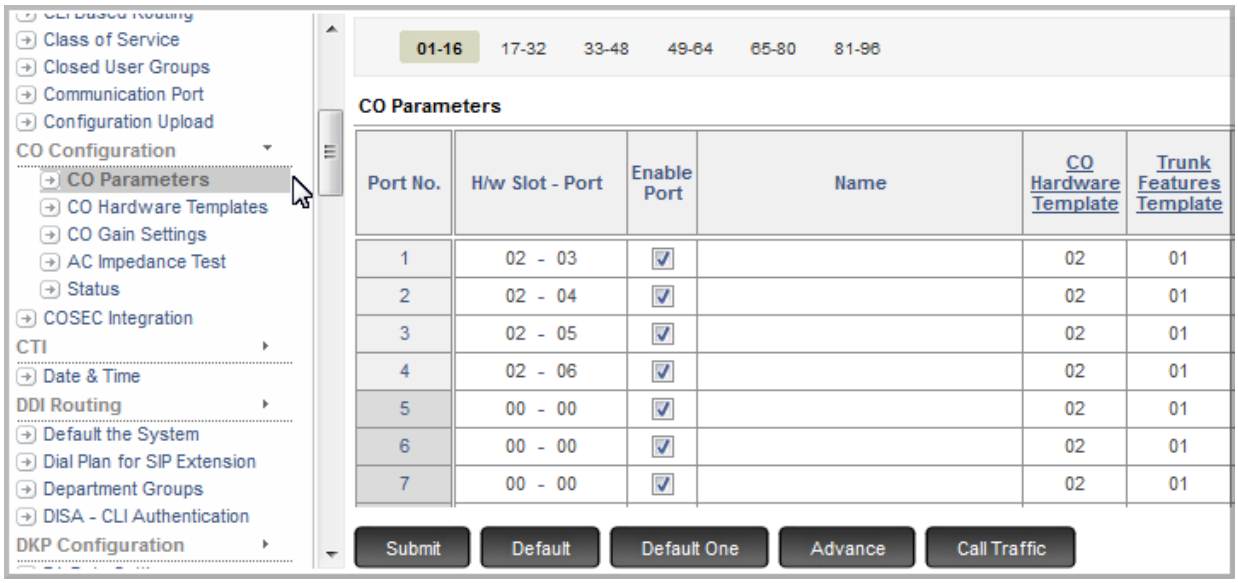
- *The SARVAM UCS will calculate cost of phone calls made by the trunks even when no call budget is allocated[258].*

How to configure

Call Budget on Trunks is to be programmed in the Trunk Port Parameters of the trunk type on which you want to enable this feature. This can be done using Jeeves as well as a telephone

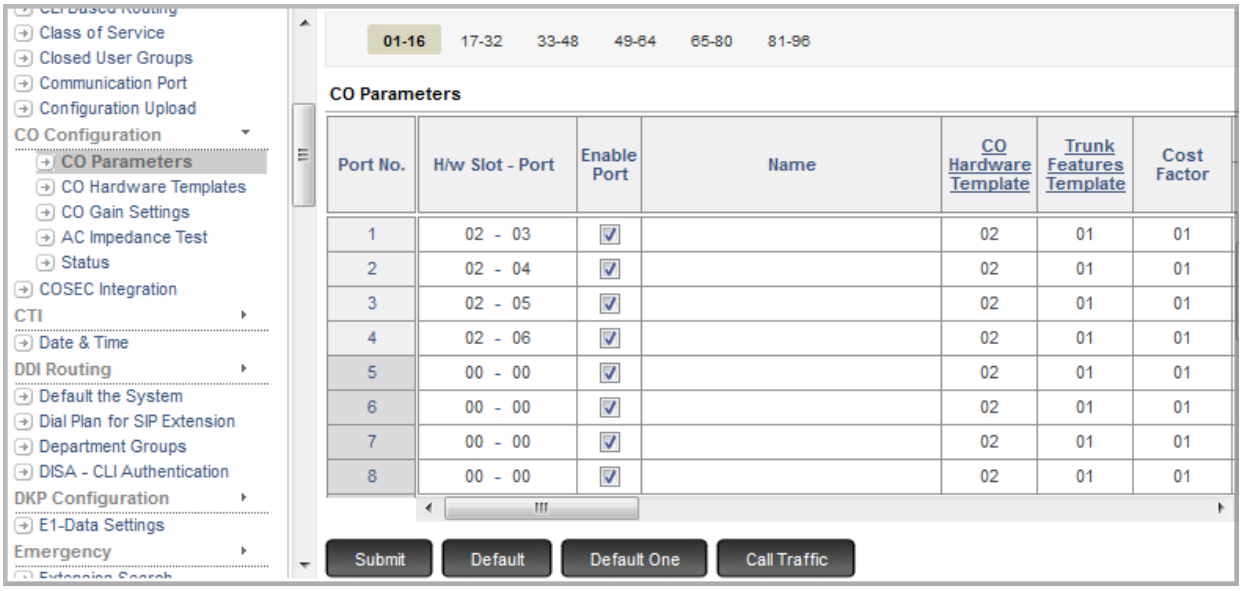
Programming Call Budget on Trunks using Jeeves

- Log in as System Engineer.
- **Call Budget** parameters must be programmed in the **Port Parameters** page of the trunk type you want to configure. For instance, to configure Call Budget on a CO Trunk,
- Under **Configuration**, click **CO Configuration** link.
- Click **CO Parameters** to open the page.



258. *Based on the feature 'Call Cost Calculation'.*

- Click the **Advance** button and configure the following parameters:



- **Call Budget:** By default, Call Budget is enabled on the trunk. If you wish to change the default configuration or disable it, configure the parameters as per your requirement:
  - **Type:** Select the type of Call Budget on Trunk, that is, Amount or Minutes or Calls to be applied on this CO trunk port. By default, Minutes is selected as the Call Budget type.To disable select Type as None.
  - **Amount:** If you selected 'Amount' as the Call Budget Type, enter the Budget Amount in this field. By default the Amount is set to 999999.
  - **Minutes:** If you selected 'Minutes' as the Call Budget Type, enter the number of Minutes in this field. By default the number of minutes is set as 000300.
  - **Calls:** If you selected 'Calls' as the Call Budget Type, enter the number of Calls in this field. By default the number of calls is set to 9999.
  - **Scheduled Reset:** Enable this flag if you want the Call Budget Amount/Minutes/Number of Calls to be reset on a particular date of every month.
  - **Scheduled (Date):** Select the date of the month (Daily or 1-31) on which you want the Call Budget Amount/Minutes/Number of Calls to be reset every month. You may select 'Daily' if your plan suggests so.
- Click **Submit** at the bottom of the page to save your settings.
- You may program the same Call Budget parameters as listed above for other trunk types:
  - Click **VoIP Configuration**, click **SIP Trunk Parameters** to program Call Budget on SIP trunks. Click the **Advance** button on this page to reach Call Budget parameters.
  - Click **Port Parameters** under **T1E1 Configuration** to program Call Budget on T1E1PRI trunks. Click the **Advance** button on this page to reach Call Budget parameters.

- Click **BRI Parameters** under **BRI Configuration** to program Call Budget on BRI trunks. Click the **Advance** button on this page to reach Call Budget parameters.

- Click **Mobile Port Parameters** under **Mobile Configuration t**o program Call Budget on Mobile Ports. Click the **Advance** button on this page to reach Call Budget parameters.

- You may log out of Jeeves.

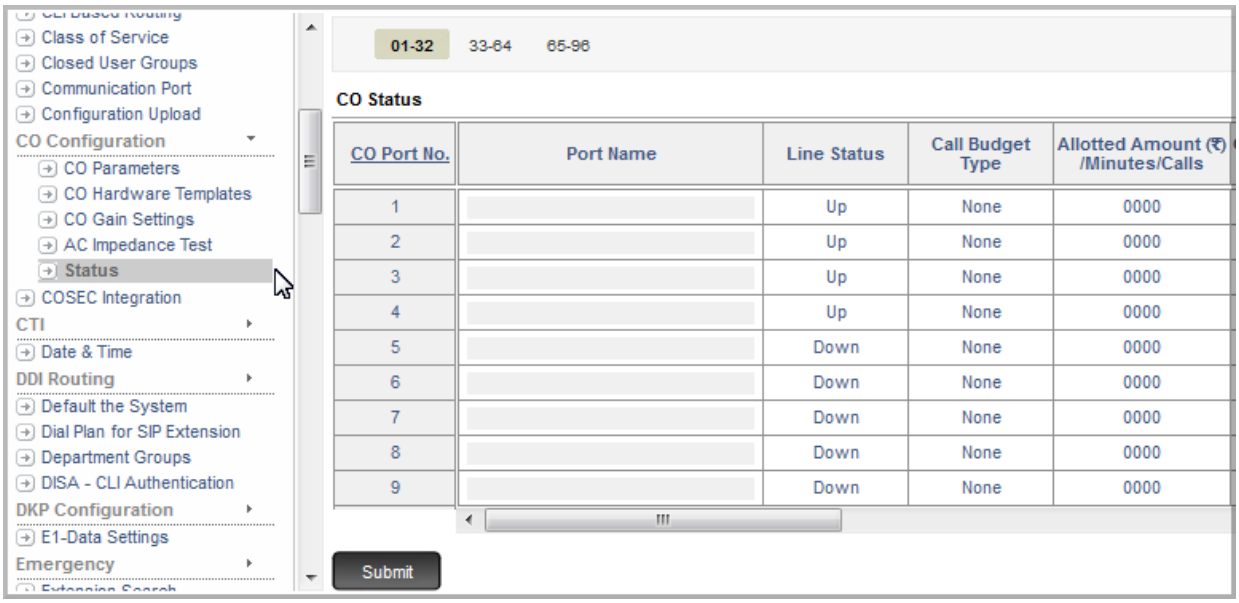
- *The consumed Call Budget on trunk can be reset from the System Engineer mode as well as the System Administrator mode manually at any time, referred to as Manual Reset.*

- *Manual Reset of Call Budget on Trunks by the System Engineer can be done either from Jeeves or using a Telephone.*

Manual Reset of Call Budget on Trunk using Jeeves

- You may perform manual reset of the consumed Call Budget on the above listed trunk types from the **Status** page of each of these trunk types.

  - Click the **Status** page under **CO Configuration**. Select the **Reset Consumed Amount/Minutes/Calls** check box of the CO port for which you want to reset the consumed Call Budget.



- Similarly, click **Status** page under **VoIP Configuration** and enable the parameter **Reset Consumed Amount/Minutes/Calls** to manually reset consumed Call Budget on the desired **SIP trunks**.

- Click **Status** page under **T1E1 Configuration** to enable the same parameter **Reset Consumed Amount/Minutes/Calls** of the T1E1 port for which you want to reset the consumed Call Budget.

- Click **Status** page under **BRI Configuration**, and enable the parameter **Reset Consumed Amount/ Minutes/Calls** for the BRI port for which you want to reset the consumed Call Budget.

- Click **Status** page under **Mobile Configuration** to enable the same parameter **Reset Consumed Amount/Minutes/Calls** of the Mobile port for which you want to reset the consumed Call Budget.

Similarly, you can reset Call Budget on Trunks from the SA mode.

Programming Call Budget on Trunks using a Telephone

- Enter SE mode.

Call Budget on CO Trunks

To program Call Budget Type on CO, dial:

- ***3301-1-CO-Budget Type*** to select Call Budget Type for a single trunk port.
- ***3301-2-CO-CO-Budget Type*** to select the same Call Budget Type for a range of trunk ports.
- ***3301-*-Budget Type*** to select the same Call Budget Type for all trunk ports.
  Where,
  CO is the Software Port number of the CO port from 001 to 128.
  Budget Type
  0 for None
  1 for Amount
  2 for Minutes
  3 for Number of Calls
  By default, Budget Type is None.

To program Call Budget Amount on CO, dial:

- ***3302-1-CO-Budget Amount*** to program amount for a single trunk port.
- ***3302-2-CO-CO-Budget Amount*** to program the same amount for a range of trunk ports.
- ***3302-*-Budget Amount*** to program the same amount for all trunk ports.
  Where,
  CO is the Software Port number of the CO port from 001 to 128.
  Budget Amount is of 6 digits max. Use leading zeros if amount to be programmed has fewer than 6 digits.
  By default Budget Amount is 999999.

To program Call Budget Minutes on CO, dial:

- ***3303-1-CO-Minutes*** to program minutes for a single trunk port.
- ***3303-2-CO-CO-Minutes*** to program the same minutes for a range of trunk ports.
- ***3303-*-Minutes*** to program the same minutes for all trunk ports.
  Where,
  CO is the Software Port number of the CO port from 001 to 128.
  'Minutes' is of 6 digits max. Use leading zeros if Minutes to be programmed has less than 6 digits.
  By default, Minutes is 999999.

To program Call Budget - Number of Calls on CO, dial:

- ***3309-1-CO-Number of calls*** to program number of calls for a single trunk port.
- ***3309-2-CO-CO-Number of calls*** to program the same number of calls for a range of trunk ports.
- ***3309-*-Number of calls*** to program the same number of calls for all trunk ports.
  Where,
  CO is the Software Port number of the CO port from 001 to 128.
  Number of Calls is of 4 digits from 0001 to 9999. Use leading zeros if number of calls to be programmed has fewer than 4 digits.
  By default, Number of calls is 9999.

To program Call Budget Reset Mode for CO, dial:

- ***3304-1-CO-Call Budget Reset Mode*** to program reset mode for a single trunk port.
- ***3304-2-CO-CO-Call Budget Reset Mode*** to program the same reset mode for a range of trunk ports.
- ***3304-*-Call Budget Reset Mode*** to program the same reset mode for all trunk ports.

Where,
CO is the Software Port number of the CO port from 001 to 128.
Reset Mode is
1 for Scheduled reset
2 for Manual reset
By default, Call Budget Reset Mode is Scheduled.

To program the Date for Scheduled Reset mode, dial:

- ***3305-1-CO-Date*** to program date for a single trunk port.
- ***3305-2-CO-CO-Date*** to program the same date for a range of trunk ports.
- ***3305-*-Date*** to program the same date for all trunk ports
  Where,
  CO is the Software Port number of the CO port from 001 to 128.
  Date is
  01 to 31 for Scheduled date to reset every month.
  00 for Scheduled reset Daily.
  By default, Reset date is 1st. of every month.

Call Budget on Mobile Trunks

To program Call Budget Type on Mobile trunk port, dial:

- ***8019-1-Mobile-Budget Type*** to program budget type for a single trunk port.
- ***8019-2-Mobile-Mobile-Budget Type*** to program the same budget type for a range of trunk ports.
- ***8019-*-Budget Type*** to program the same budget type for all trunk ports.
  Where,
  Mobile is the number of the Mobile software port from 001 to 064.
  Budget Type
  0 for None
  1 for Amount
  2 for Minutes
  3 for Number of Calls
  By default, Budget Type is None.

To program Call Budget Amount on Mobile trunk port, dial:

- ***8020-1-Mobile-Budget Amount*** to program amount for a single trunk port.
- ***8020-2-Mobile-Mobile-Budget Amount*** to program the same amount for a range of trunk ports.
- ***8020-*-Budget Amount*** to program the same amount for all trunk ports.
  Where,
  Mobile is the number of the Mobile software port from 001 to 064.
  Budget Amount is of max. 6 digits. Use leading zeros when programming an amount with fewer than 6 digits.
  By default, Budget Amount is 999999.

To program Call Budget Minutes on Mobile trunk port, dial:

- ***8021-1-Mobile-Minutes*** to program minutes for a single trunk port.
- ***8021-2-Mobile-Mobile-Minutes*** to program the same minutes for a range of trunk ports.
- ***8021-*-Minutes*** to program the same minutes for all trunk ports.
  Where,
  Mobile is the number of the Mobile software port from 001 to 064.
  Minutes is of max. 6 digits. Use leading zeros if Minutes to be programmed has fewer than 6 digits.
  By default, Free Minutes is 999999.

To program Call Budget Number of Calls on Mobile trunk port, dial:

- ***8033-1-Mobile-Number of calls*** to program number of calls for a single trunk port.
- ***8033-2-Mobile-Mobile-Number of calls*** to program the same number of calls for a range of trunk ports.
- ***8033-*- Number of calls*** to program the same number of calls for all trunk ports.
  Where,
  Mobile is the number of the Mobile software port from 001 to 064.
  Number of Calls is of 4 digits from 0001 to 9999. Use leading zeros if number of calls to be programmed has fewer than 4 digits.
  By default, Number of calls is 9999.

To program Call Budget Reset Mode for Mobile trunk port, dial:

- ***8022-1-Mobile-Call Budget Reset Mode*** to program reset mode for a single trunk port.
- ***8022-2-Mobile-Mobile-Call Budget Reset Mode*** to program the same reset mode for a range of trunk ports.
- ***8022-*-Call Budget Reset Mode*** to program the same reset mode for all trunk ports.
  Where,
  Mobile is the number of the Mobile software port from 001 to 064.
  1 for Scheduled reset
  2 for Manual reset
  By default, Call Budget Reset Mode is Scheduled.

To program the Date for Scheduled Reset mode for Mobile trunk port, dial:

- ***8023-1-Mobile-Date*** to program reset date for a single trunk port.
- ***8023-2-Mobile-Mobile-Date*** to program the same reset date for a range of trunk ports.
- ***8023-*-Date*** to program the same reset date for all trunk ports.
  Where,
  Mobile is the number of the Mobile software port from 001 to 064.
  Date is
  01 to 31 for Scheduled date to reset every month.
  00 for Scheduled reset Daily.
  By default, Reset date is 1st. of every month.

Call Budget on BRI Trunks

To program Call Budget Type on BRI, dial:

- ***6214-1-BRI-Budget Type*** to program budget type for a single trunk.
- ***6214-2-BRI-BRI-Budget Type*** to program the same budget type for a range of trunks.
- ***6214-*-Budget Type*** to program the same budget type for all trunks.
  Where,
  BRI is the number of the BRI software port from 01 to 32.
  Budget Type
  0 for None
  1 for Amount
  2 for Minutes
  3 for Number of Calls
  By default, Budget Type is None.

To program Call Budget Amount on BRI, dial

- ***6215-1-BRI-Budget Amount*** to program amount for a single trunk.
- ***6215-2-BRI-BRI-Budget Amount*** to program the same amount for a range of trunks.
- ***6215-*-Budget Amount*** to program the same amount for all trunks.
  Where,
  BRI is the number of the BRI software port from 01 to 32.

Budget Amount is of max. 6 digits. Use leading zeros when programming an amount with fewer than 6 digits.
By default, Budget Amount is 999999.

To program Call Budget Minutes on BRI, dial:

- ***6216-1-BRI-Minutes*** to program minutes for a single trunk.
- ***6216-2-BRI-BRI-Minutes*** to program the same minutes for a range of trunks.
- ***6216-*-Minutes*** to program the same minutes for all trunks.
  Where,
  BRI is the number of the BRI software port from 01 to 32.
  Minutes is of max. 6 digits. Use leading zeros when programming an amount with fewer than 6 digits.
  By default, Minutes is 999999.

To program Call Budget Number of Calls on BRI, dial:

- ***6205-1-BRI-Number of calls*** to program number of calls for a single trunk.
- ***6205-2-BRI-BRI-Number of calls*** to program the same number of calls for a range of trunks.
- ***6205-*-Number of calls*** to program the same number of calls for all trunks.
  Where,
  BRI is the number of the BRI software port from 01 to 32.
  Number of Calls is of 4 digits from 0001 to 9999. Use leading zeros if number of calls to be programmed has fewer than 4 digits.
  By default, Number of calls is 9999.

To program Call Budget Reset Mode for BRI, dial:

- ***6217-1-BRI-Reset Mode*** to program reset mode for a single trunk.
- ***6217-2-BRI-BRI-Reset Mode*** to program reset mode for a range of trunks.
- ***6217-*-Reset Mode*** to program reset mode for all trunks.
  Where,
  BRI is the number of the BRI software port from 01 to 32.
  1 for Scheduled reset
  2 for Manual reset
  By default, Call Budget Reset Mode is Scheduled.

To program the Date for Scheduled Reset mode for BRI, dial:

- ***6218-1-BRI-Date*** to program date for a single trunk.
- ***6218-2-BRI-BRI-Date*** to program the same date for a range or trunks.
- ***6218-*-Date*** to program the same date for all trunks.
  Where,
  BRI is the number of the BRI software port from 01 to 32.
  Date is
  01 to 31 for Scheduled date to reset every month.
  00 for Scheduled reset Daily.
  By default, Reset date is 1st. of every month.

Call Budget on T1E1 Trunks

To program Call Budget Type on T1E1 Port, dial:

- ***6122-1-T1E1-Budget Type*** to program call budget type for a single trunk port.
- ***6122-2-T1E1-T1E1-Budget Type*** to program the same call budget type for a range of trunk ports.
- ***6122-*-Budget Type*** to program the same call budget type for all trunk ports.
  Where,
  T1E1is the number of the T1E1 software port from 01 to 08.
  Budget Type is
  0 for None

1 for Amount
2 for Minutes
3 for Number of Calls
By default, Budget Type is None.

To program Call Budget Amount on T1E1 Port, dial:

- ***6123-1-T1E1-Budget Amount*** to program call budget amount for a single trunk port.
- ***6123-2-T1E1-T1E1-Budget Amount*** to program the same amount for a range of trunk ports.
- ***6123-*-Budget Amount*** to program the same amount for all trunk ports.
  Where,
  T1E1is the number of the T1E1 software port from 01 to 08.
  Budget Amount is of 6 digits max. Use leading zeros if amount to be programmed has fewer than 6 digits.
  By default Budget Amount is 999999.

To program Call Budget Minutes on T1E1 Port:

- ***6124-1-T1E1-Minutes*** to program minutes for a single trunk port.
- ***6124-2-T1E1-T1E1-Minutes*** to program the same minutes for a range of trunk ports.
- ***6124-*-Minutes*** to program the same minutes for all trunk ports.
  Where,
  T1E1is the number of the T1E1 software port from 01 to 08.
  Minutes is of 6 digits max. Use leading zeros if Minutes to be programmed has less than 6 digits.
  By default, Minutes is 999999.

To program Call Budget Number of Calls on T1E1 Port, dial:

- ***6125-1-T1E1-Number of calls*** to program number of calls for a single trunk port.
- ***6125-2-T1E1-T1E1-Number of calls*** to program the same number of calls for a range of trunk ports.
- ***6125-*-Number of calls*** to program the same number of calls for all trunk ports.
  Where,
  T1E1is the number of the T1E1 software port from 01 to 08.
  Number of Calls is of 4 digits from 0001 to 9999. Use leading zeros if number of calls to be programmed has fewer than 4 digits.
  By default, Number of calls is 9999.

To program Call Budget Reset Mode for T1E1, dial:

- ***6138-1-T1E1-Call Budget Reset Mode*** to program reset mode for a single trunk port.
- ***6138-2-T1E1-T1E1-Call Budget Reset Mode*** to program the same reset mode for a range of trunk ports.
- ***6138-*-Call Budget Reset Mode*** to program the same reset mode for all trunk ports.
  Where,
  T1E1is the number of the T1E1 software port from 01 to 08.
  1 for Scheduled reset
  2 for Manual reset
  By default, Call Budget Reset Mode is Scheduled.

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To program the Date for Scheduled Reset mode for T1E1, dial:

- ***6139-1-T1E1-Date*** to program date for a single trunk port.
- ***6139-2-T1E1-T1E1-Date*** to program the date for a range of trunk ports.
- ***6139-*-Date*** to program the date for all trunk ports.
  Where,
  T1E1is the number of the T1E1 software port from 01 to 08.
  Date is
  01 to 31 for Scheduled date to reset every month.

00 for Scheduled reset Daily.
By default, Reset date is 1st. of every month.

- Exit SE mode.

Resetting Consumed Call Budget on Trunks using a Telephone

- Enter SE mode.

  To manually reset consumed Call Budget of CO trunks, dial:
  - ***3306-1-CO*** to manually reset a single trunk port.
  - ***3306-2-CO-CO*** to manually reset a range of trunk ports.
  - ***3306-**** to manually reset all trunk ports.
    Where,
    CO is the Software Port number of the CO port from 001 to 128.

  To manually reset consumed Call Budget of Mobile trunks, dial:
  - ***8024-1-Mobile*** to reset a single trunk port.
  - ***8024-2-Mobile-Mobile*** to reset a range of trunk ports.
  - ***8024-**** to reset all trunk ports.
    Where,
    Mobile is the number of the Mobile software port from 001 to 064.

  To manually reset consumed Call Budget of BRI trunks, dial:
  - ***6219-1-BRI*** to reset a single trunk.
  - ***6219-2-BRI-BRI*** to reset a range of trunks.
  - ***6219-**** to reset all trunks.
    Where,
    BRI is the number of the BRI software port from 01 to 32.

  To manually reset consumed Call Budget of T1E1 trunks, dial:
  - ***6140-1-T1E1*** to reset a single trunk port.
  - ***6140-2-T1E1-T1E1*** to reset a range of trunk ports.
  - ***6140-**** to reset all trunk ports.
    Where,
    T1E1is the number of the T1E1 software port from 01 to 08.

- Exit SE mode.

Call Chaining

What's this?

Call Chaining is when an external/internal call transferred by the Operator to another extension or external number is made to return to the Operator's extension after the conversation between the caller and the extension/external number to which it is transferred has ended.

Call Chaining is useful situations where the Operator intervention is required after the transferred call has ended. For instance:

- The caller needs to take an appointment or requires some information from the Operator after talking to the desired extension.

- A marketing executive who calls his supervisor to consult on a technical problem needs to be informed about his travel itinerary and ticket booking by the Operator. The Operator can transfer the call to the supervisor, and use Call Chaining to retrieve the call once the conversation has ended to give the information to the executive.

Call Chaining can be set for multiple calls.

Also refer “Call Transfer”, “Call Park”.

How it works

- A is an External Caller
- B is an extension.

- A calls a Trunk of SARVAM UCS.
- The Operator answers the call.
- The Operator sets Call Chaining and transfers the call to B.
- B disconnects the call with A, but A is still connected.
- The call comes back to the Operator, if the Operator is free.
- Now A is in speech with the Operator.

- If A disconnects the call with B, the call will be released. It will not return to the Operator.

- If the Operator is busy, A will be played music on hold for the duration of the Call Park Release Timer.
- If the Operator is busy and the Timer elapses, the call will be released.

Call Chaining can be performed when call is transferred from any extension to another extension or external number.

How to configure

The only programming involved in the functioning of this feature is assigning this feature to a DSS key which has LED and programming, if necessary, the Call Park Release Timer.

Refer the topic “DSS Keys Programming” and “Call Park” for instructions.

How to use

For EON & Extended IP Phone Users

- Go OFF-Hook to answer the incoming call.
- The caller requests for an extension/trunk.
- Press DSS Key assigned to Call Chaining.

  OR

- Press the ‘Transfer Key’ and dial 1050.

  You get confirmation tone and the message 'Party Chained' on your phone's display. If DSS Key is used, the LED of the key will glow.

- Dial the requested extension/external number.

- You get Ring Back Tone.
- The called party answers.
- Go ON Hook to transfer the call.
- When extension/trunk disconnects, your extension rings.
- Go OFF Hook. You get connected with the caller.
- Go Idle after the conversation ends, or you get dial tone after 3 seconds.

For SLT Users

- Answer the incoming call.
- The caller requests for an extension/trunk.
- Dial Flash-1050.
- Dial the requested extension/external number.
- You get Ring Back Tone.
- The called party answers.
- Go On-Hook to transfer call.
- When extension/trunk disconnects, your extension rings.
- Lift handset to answer. You get connected with the caller.
- Go Idle after the conversation ends.

Call Cost Calculation (CCC)

What's this?

SARVAM UCS can calculate the cost in amount for the external calls made by the extension users.

The call cost is calculated according to the tariffs offered by the Service Providers. Different types of tariffs are provided by different Service Providers. These may be:

- Time Based Tariffs, for example, free calling from 11.00 p.m. to 6.00 a.m.
- Unit Based Tariffs, for example, first 60 seconds at the rate of 50 paise and additional at the rate of 10 paise.
- Special Day Tariffs, for example, subsidized calls rates on August 15.
- Tariffs according to the Geographical Distances, for example, calling rates for UAE differ from calling rates to Canada.

Using the Call Cost Calculation feature, the system calculates the cost of the call according to the duration of the call interpreted in terms of units and the charge applicable for the duration as per the service provider.

How it works

For this feature to work,

- you must get the tariff details from the Service Providers and configure the same.
- determine the outgoing trunk for the calls according to the type of calls, namely, local calls, national calls or international calls.

SARVAM UCS will calculate the cost of the call as follows:

- When the call is made from a trunk, the system checks the **Call Cost Calculation Pulse Rate Option,** 1 to 4 assigned to the trunk, on the basis of the **Call Cost Calculation Time Schedule** configured for the outgoing call.

  Each Call Cost Calculation Pulse Rate option contains a **Pulse Rate Type for the Pulse Rate**, which is assigned in the Area Code Table.

- The system matches the Number dialed by the extension user with the **Area Code Table** configured in the system. When the area code matches with an entry in the table, the system obtains the **Pulse rate type** configured for the Call Cost Calculation Pulse Rate option assigned to the trunk.

- This **Pulse Rate type** obtained from the **Area Code Table** is checked in the **Pulse Rate table** to obtain the corresponding **duration** and **cost** to be applied for the call duration. The Pulse Rate Table may be the **Normal Pulse Rate Table** or **Discounted Pulse Rate Table**, depending on the day of the call. The system uses the built-in RTC to determine the day.

- The **Pulse Rate Type** applied (duration and cost) is divided into two parts for each time zone:
  - First unit.

  - Additional units.

- Number of Units is derived from the pulse rate at the time of the call and duration of the call. System acquires the pulse rate type and call duration with the help of in-built RTC.

  Total Units = First Unit + Additional Unit.

  If the call duration is less than the pulse rate of the first unit then additional unit is zero.

  Call Units = (Call duration in seconds)/(Pulse rate in seconds).

- If the duration of the call is less than or equal to one unit,

  Cost of Call = Cost of First Unit + Service Charge as applied.

- If the duration of the call is more than one unit,

  *Cost of Call = [Cost of First Unit + (Number of Additional units x Cost for Additional units)] + Service Charge a*s applied.

- The system applies the rates as configured in the Normal Pulse Rate Table for all the days, except when it detects a day configured in the Discounted Pulse Rate Schedule. These are special days when special Tariffs are offered.

  For the days configured in the Discounted Pulse Rate Schedule as special days, the system checks the the duration and cost of the First unit and the Additional units configured in the Discounted Pulse Rate Table.

SARVAM UCS uses the **Cost of the Call** for SMDR. This cost is deducted from the Call Budget (Amount), if allotted to the trunk and also from the Call Budget, if assigned to the extension users.

The logic for call cost calculation is explained with the help of an example:

- An Outgoing Call is made using trunk, CO-1.
- On CO-1, Call Cost Calculation is configured as follows:
  - Call Cost Calculation Pulse Rate option = 1
  - Call Cost Calculation Schedule
    - Time Zone 1 = Start Time: 00:00, End Time: 22:00
    - Time Zone 2 = Start Time: 22:01, End Time: 23:59
  - Assign Trunk Feature Template 1 to CO-1. Configuring the Call Cost Calculation parameters in the Trunk Feature Template as follows:

Template No.	Call Cost Calculation Pulse Rate Option	Call Cost Calculation Time Schedule											
		T1				T2				T3			
		Start Time		End Time		Start Time		End Time		Start Time		End Time	
		HH	MM	HH	MM	HH	MM	HH	MM	HH	MM	HH	MM
001	1	00	00	22	00	22	01	23	59				

- Area Code Table is configured as:

Index	Area Code	Name	Ignore Digit Count	Pulse Rate Type for Pulse Rate			
				Option - 1	Option - 2	Option - 3	Option - 4
001	26	Local		03	06	09	10
002	09	Mobile		05	03	07	08

- The Normal Pulse Rate Table is configured as:

Pulse Rate Type		Time Zone T1		Time Zone T2		Time Zone T3		Time Zone T4	
		First Unit	Add. Unit	First Unit	Add. Unit	First Unit	Add. Unit	First Unit	Add. Unit
01	Duration (sec)	180	180	60	30	90	30	120	60
	Cost	002.00	002.00	002.00	002.00	002.00	002.00	002.00	002.00
02	Duration (sec)	300	300	300	300	300	300	300	300
	Cost	001.00	001.00	001.00	001.00	001.00	001.00	001.00	001.00
03	Durations)	30	30	30	30	30	30	30	30
	Cost	001.00	001.00	001.00	001.00	001.00	001.00	001.00	001.00
04	Duration (sec)	45	45	45	45	45	45	45	45
	Cost	001.00	001.00	001.00	001.00	001.00	001.00	001.00	001.00
05	Duration (sec)	180	180	180	180	180	180	180	180
	Cost	003.00	003.00	003.00	003.00	003.00	003.00	003.00	003.00
:	Durations)	:	:	:	:	:	:	:	:
	Cost	:	:	:	:	:	:	:	:
32	Duration (sec)								
	Cost								

- With this configuration, SARVAM UCS will calculate the call cost as follows:

  - An Outgoing Call is made by an extension user, to the number ‘2630555’ through the trunk, CO-1 at 20:10 hours. SARVAM UCS will check the Call Cost Calculation parameters assigned on the trunk and determine the Time Zone as per time of the call.The system will also check the corresponding Pulse Rate Option configured on the trunk.

    In this example, Time Zone for CO-1 at 20:10 Hours would be Time Zone 1, and the Pulse Rate Option for CO-1 is ‘1’.

  - SARVAM UCS will match the dialed number ‘2630555’ in the Area Code table. A best match is found with the entry configured at index 001 in the Area Code Table.

    As per the Area Code Table, the Pulse Rate Type ‘03’ is programmed in ‘Pulse Rate Option 1’ for the matching entry (at Index 001).

(However, if CO-1 would have been assigned Pulse Rate Option ‘2’, the Pulse Rate type ‘06’ would have been selected as shown in Area Code Table)

- Finally, for Pulse Rate Type ‘03’ SARVAM UCS will check the Normal Pulse Rate Table. SARVAM UCS will consider the Cost for the First Unit as 001.00 (As ` or $ as per applicable currency) for the duration of 30 seconds and for the additional unit also, the cost will be considered as 001.00 for the duration of 30 seconds. This data will be used for calculating the total cost of call based on the total duration of the call.

Similarly, when there are Special Tariffs offered on certain days, the system will check the Discounted Pulse Rate Schedule and the Discounted Pulse Rate Table.

The days on which the special rates are to be applied must be configured in the Discounted Pulse Rate Schedule. The duration and cost of the First unit and the Additional units must be configured in the Discounted Pulse Rate Table according to the Time Zones.

How to configure

To be able to use Call Cost Calculation, you must do the following:

- Define the Unit and Service Charge on the basis of which call cost is to be calculated.
- Assign Call Cost Calculation Pulse Rate Option and the Call Cost Calculation Schedule on the trunks on which you want to apply this feature.
- Configure the Pulse Rate Types for the Pulse Rate Option you assign to the trunk.
- You may configure different Pulse Rate Types:
  - For Normal Days configure the pulse rate in the Normal Pulse Rate Table.
  - For Special days configure the pulse rate in the Discounted Rate Table. If you configure the Discounted Pulse Rate Table, you must also configure the Discounted Pulse Rate schedule.
- Configure the Area Code Table.

Configuring Call Cost Calculation using Jeeves

To configure call cost calculation parameters,

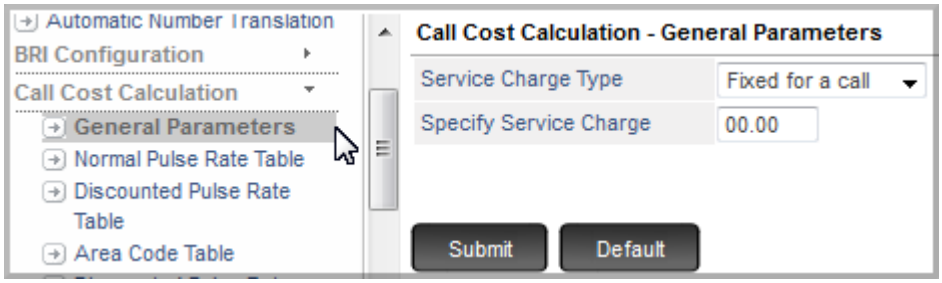
- Log in as System Engineer.
- Under **Configuration** click the **Call Cost Calculation** link.

Service Charge

By default, no Service Charge is applied on call cost by the system. Service Charge on call cost is generally applied in Hotels and other organization which charge users for the calls made by them.

If you want to apply Service Charge,

- Click **General Parameters** to open the page.



- In the **Service Charge Type** field, select the type of service charge you want to apply from the options:
  - **Fixed for a call:** A fixed amount is added as service charge to every call regardless of the cost of that call.

    If you select this option, you must define the Amount to be added as service charge in the **Specify Service Charge** field.
  - **per Unit:** service charge is added to each unit of the call. For example, if a call worth 10 units was made, the service charge will be applied on each of the 10 units, instead of the one time service charge as in the case of Fixed service charge.

    If you select this option, you must define the amount to be added as service charge on each unit in the **Specify Service Charge** field.
  - **Percentage of call cost:** A percentage of the cost of the call is added as a service charge for that call.

    If you select this option, you must define the percentage in the **Specify Service Charge in %** field which appears.
- Click **Submit** to save changes.

Applying Call Cost Calculation on Trunks

- To assign **Call Cost Calculation Pulse Rate Option** and configure the **Call Cost Calculation Schedule** on the trunks, go to *Configuration*, and configure these in the "Trunk Feature Template" assigned to the different trunk port types. See "Configuring CO Trunks", "Configuring Mobile Trunks", "Configuring BRI Trunks", "Configuring E1 Trunks", "Configuring T1 Trunks", "Configuring Trunks" and "Configuring SIP Trunks" under *Configuring SARVAM UCS* for instructions.

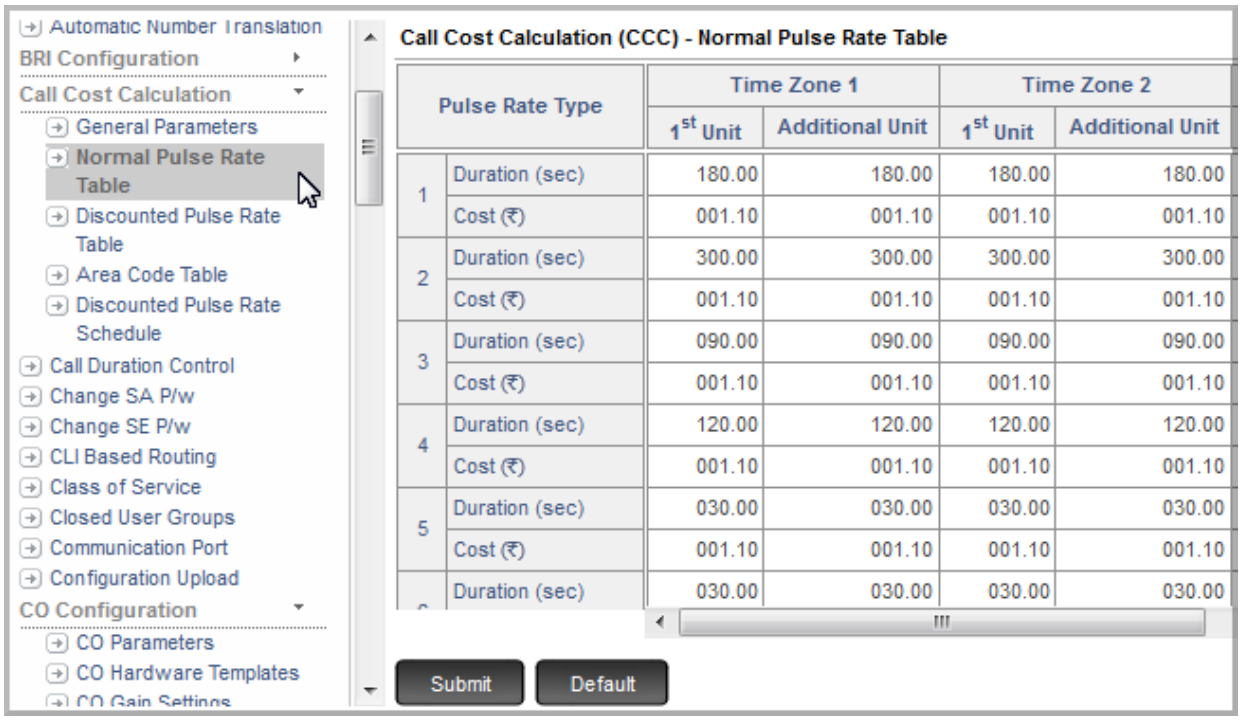
Configuring Pulse Rate Types

Generally service providers offer discounted call rates for special days. To take care of this, SARVAM UCS provides two types of Pulse Rate Tables: Normal and Discounted.
If your service providers do not offer any special rates, you may skip configuring the Discounted Pulse Rate table.

To configure the Normal Pulse Rate table,

- Click **Call Cost Calculation** under **Configuration**.

- Click the **Normal Pulse Rate Table** link.



- Configure the **Pulse Rate Type** with rates for the **First Unit** and the **Additional Unit**.

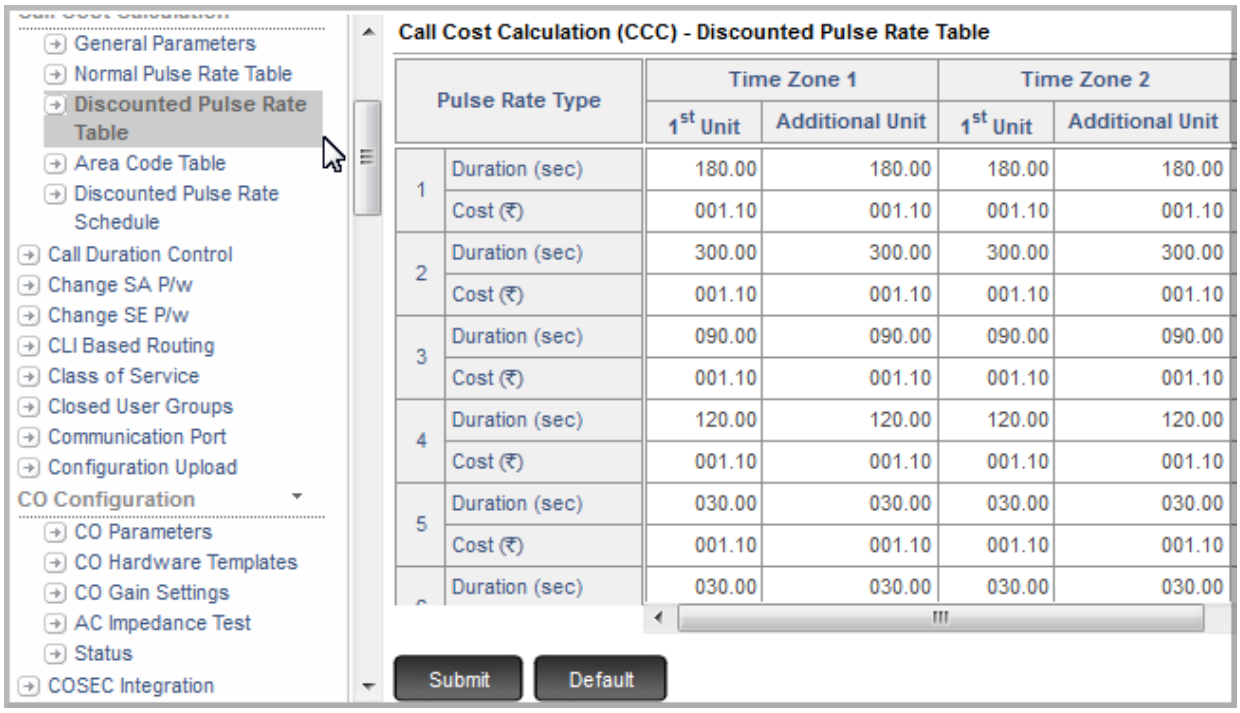
  Generally service providers offer different call rates for different types of calls, for example: local, national, international. You can configure different Pulse Rate Types for different types of calls. Thus, each Pulse Rate Type can have different rates for the First and the Additional unit.

  The Pulse Rates offered by service providers may vary according to the time of the day. In such cases, you will need to configure the **Call Cost Calculation Schedule** for the trunk, by dividing the day into Time Zones, from 1 to 4, as required, to match the time of the pulse rates offered by your service provider.
  If you have configured Time Zones for the Call Cost Schedule on a trunk, you may define the different Pulse Rate Types for each Time Zone.

- Click **Submit** to save changes.

To configure the Discounted Pulse Rate table,

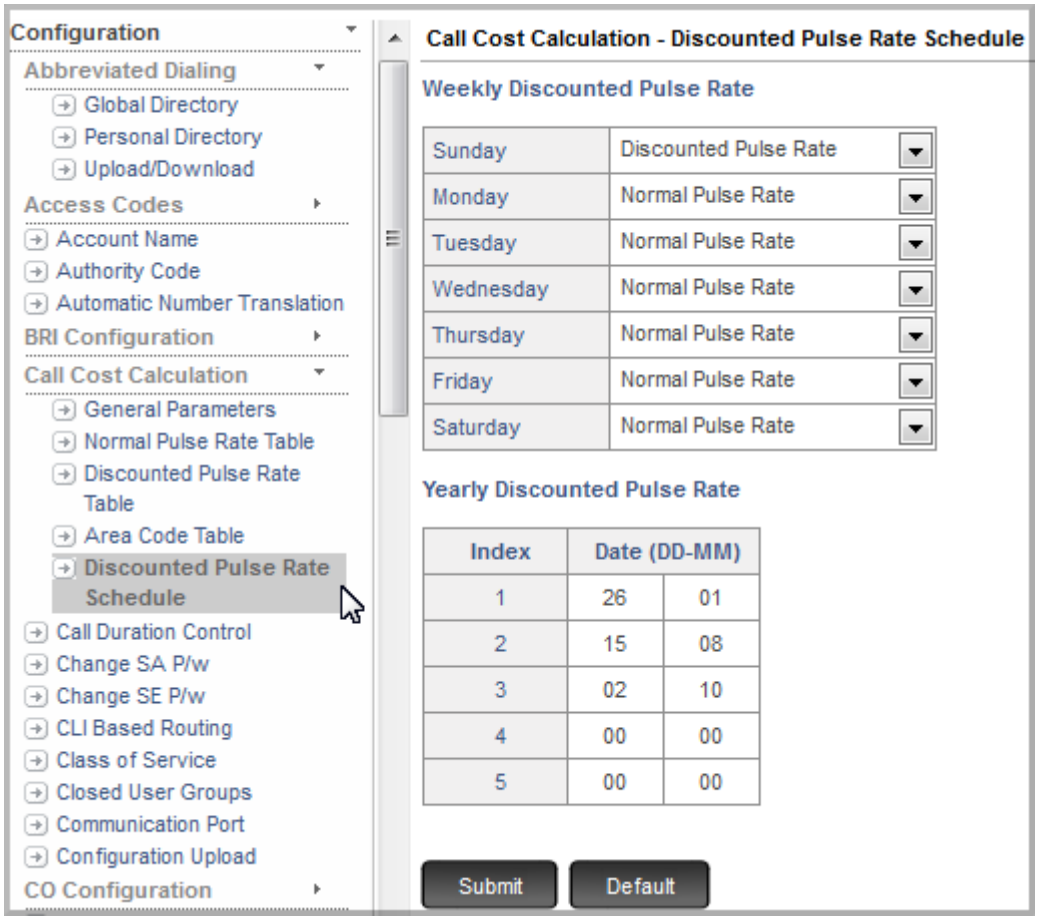
- Click the **Discounted Pulse Rate Table** link.



- Configure the **Discounted Pulse Rate Type** with rates for the **First Unit** and the **Additional Unit**, as required.

- Click **Submit** to save changes.

- Click the **Discounted Pulse Rate Schedule** link.



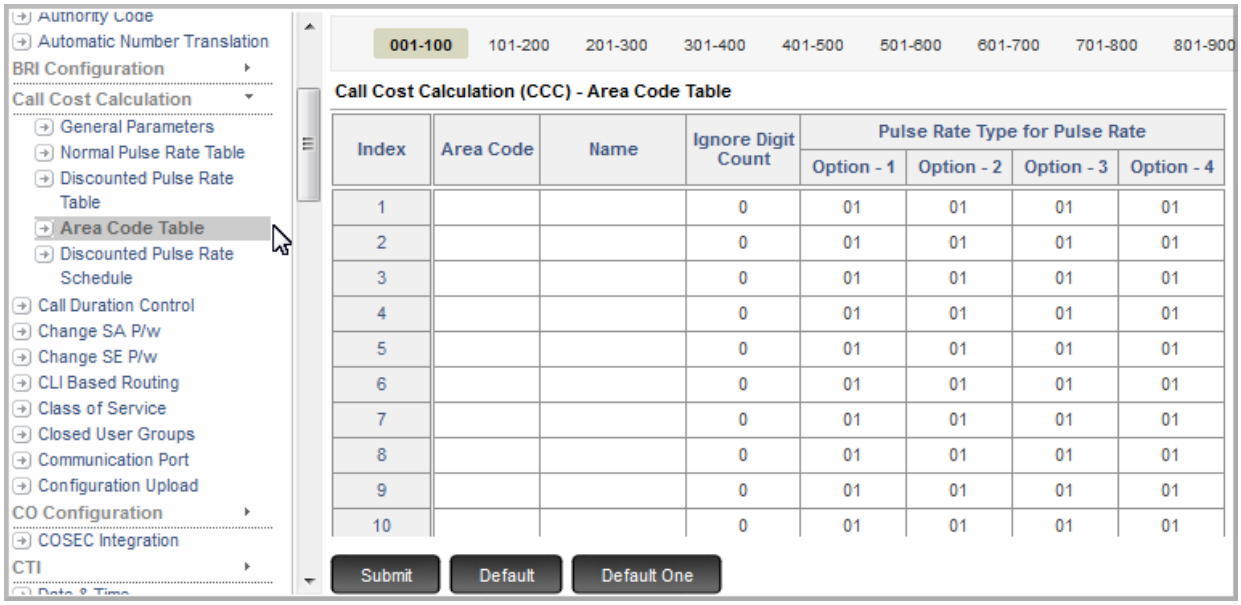
- Define **Weekly** and **Yearly** special days on this page.
- Click **Submit** to save.

Configuring Area Code Table

The pulse rate of a call depends on the destination number dialed. Generally, pulse rate varies according to the type of the call: Local, national, international. SARVAM UCS enables you to configure Pulse Rates according to Area codes.

To configure Pulse Rates for Area Codes,

- Click the **Area Code** link.



- In the **Area Code** column, enter the number strings (prefix) of the Area Codes, country codes, local numbers. You can configure as many as 999 area codes in the table.
- You may enter a **Name** to tag each Area Code.
- Do not configure the Ignore Digit Count. This parameter is relevant only for Service Provider Based Least Cost Routing.
- Different service providers offer different pulse rates for the same type of calls. To take care of this, SARVAM UCS allows you to assign different Pulse Rate Options for each area code.
- For each area code, configure the Pulse Rate to be followed for the desired Pulse Rate Options in the **Pulse Rate Type for Pulse Rate (Option 1 to 4)** column.
- Click **Submit** to save changes.

For “Default Area Code Table for the Region-USA”’ at end of the chapter.

Configuring Call Cost using Telephone

Unit Charge

When call cost is to be calculated on the basis of 16 KHz metering pulses, user can select different unit charge for first unit and different unit charge for the additional units.

To program the unit charge for first unit when 16 KHz metering is used, dial:

- ***2600-Unit Charge for First Unit***
  Where,
  Unit charge is the amount in XXX.XX format in any currency.
  By default, unit charge for first unit is Rs.1.10.

To program the unit charge for additional units when 12/16 KHz metering is used, dial:

- ***2601-Unit Charge for Additional Unit***
  Where,
  Unit charge is the amount in XXX.XX format in any currency.
  By default, Unit charge for additional unit is Rs.1.10.

Example: To program unit change for first unit to Rs. 1.50, dial ***2600-0150***
To program unit charge for additional unit to US$0.75, dial ***2601-0075***

Service Charge

To select service charge type, dial:

- ***2602-Service Charge Type***
  Where,

Service Charge Type	Meaning
0	Fixed Service Charge
1	Unit wise Service Charge
2	Percentage wise Service Charge

By default, the service charge type is fixed.

To program the service charges for Fixed or Unit wise Service Charge, dial:

- ***2603-Service Charge***
  Where,
  Service Charge is the amount in XXX.XX format in any currency.
  By default, Service Charge is Rs.00.00.

Example: Let us program the service charge to Rs.2.00, dial ***2603-0200***
Let us program the service charge to US$1.75, dial **2603-0175**

To set the percentage, if the service charge is selected based on the percentage, dial:

- ***2604-Percentage***
  Where,
  Percentage is from 000 to 100.
  By default, percentage is 000.

Example: To set the percentage to 10% of the cost of each call, **dial *2604-010***

Number of Units

- Number of Units is derived from the pulse rate at the time of the call and duration of the call. System acquires the pulse rate type and call duration with the help of in-built RTC.

  Total Units = First Unit + Additional Unit.

- If the call duration is less than the pulse rate of the first unit then additional unit is zero.

  Call Units = (Call duration in seconds)/(Pulse rate in seconds).

Assign parameters; 'CCC Pulse Rate Option' and program 'CCC Time Schedule' for the Trunk Feature Template which is assigned to the specific trunk used for OG calls.

- Assign 'CCC Pulse Rate Option' from 1 to 4.
- Program four 'CCC Time Schedule', T1, T2,T3 and T4. Program Start Time and End Time for each.
- Refer chapter “Trunk Feature Template” for more details.

Pulse Rate Types

Pulse rate can differ for normal and holidays. Maximum 32 entries can be made in the pulse rate type. Each pulse rate type can have different rate and different cost for first and additional unit. The table below displays the format of the pulse rate for normal day.

Pulse Rate Type		Time Zone 1		Time Zone 2		Time Zone 3		Time Zone 4	
		First Unit	Add. Unit	First Unit	Add. Unit	First Unit	Add. Unit	First Unit	Add. Unit
01	Duration	180.00	180.00	180.00	180.00	180.00	180.00	180.00	180.00
	Cost	001.10	001.10	001.10	001.10	001.10	001.10	001.10	001.10
02	Duration	:	:	:	:	:	:	:	:
	Cost	:	:	:	:	:	:	:	:
:	:	:	:	:	:	:	:	:	:
32	Duration	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00
	Cost	001.10	001.10	001.10	001.10	001.10	001.10	001.10	001.10

To program duration of first unit for a pulse rate type on normal days, dial:

- ***2607-Pulse Rate Type-Time Zone-Duration of First Unit***
  Where,
  Pulse Rate Type is from 01 to 32.
  Time Zone is from 1 to 4.
  Duration of First Unit is from 000.00 to 999.99.

To load default normal pulse rate type, dial:

- ***2606***

To program duration of additional unit for a pulse rate type on normal days, dial:

- ***2608-Pulse Rate Type-Time Zone-Duration of Additional Unit***
  Where,
  Pulse Rate Type is from 01 to 32.
  Time Zone is from 1 to 4.

Duration of Additional Unit is from 000.00 to 999.99.

Pulse Rate Type		Time Zone 1		Time Zone 2		Time Zone 3		Time Zone 4	
		First Unit	Add. Unit	First Unit	Add. Unit	First Unit	Add. Unit	First Unit	Add. Unit
01	Duration	180.00	180.00	180.00	180.00	180.00	180.00	180.00	180.00
	Cost	001.10	001.10	001.10	001.10	001.10	001.10	001.10	001.10
02	Duration	300.00	300.00	300.00	300.00	300.00	300.00	300.00	300.00
	Cost	001.10	001.10	001.10	001.10	001.10	001.10	001.10	001.10
03	Duration	90.00	90.00	90.00	90.00	90.00	90.00	90.00	90.00
	Cost	001.10	001.10	001.10	001.10	001.10	001.10	001.10	001.10
04	Duration	120.00	120.00	120.00	120.00	120.00	120.00	120.00	120.00
	Cost	001.10	001.10	001.10	001.10	001.10	001.10	001.10	001.10
05	Duration	30.00	30.00	30.00	30.00	30.00	30.00	30.00	30.00
	Cost	001.10	001.10	001.10	001.10	001.10	001.10	001.10	001.10
06	Duration	30.00	30.00	30.00	30.00	30.00	30.00	30.00	30.00
	Cost	001.10	001.10	001.10	001.10	001.10	001.10	001.10	001.10
07	Duration	16.00	16.00	16.00	16.00	16.00	16.00	16.00	16.00
	Cost	001.10	001.10	001.10	001.10	001.10	001.10	001.10	001.10
08	Duration	3.30	3.30	3.30	3.30	3.30	3.30	3.30	3.30
	Cost	001.10	001.10	001.10	001.10	001.10	001.10	001.10	001.10
09	Duration	2.10	2.10	2.10	2.10	2.10	2.10	2.10	2.10
	Cost	001.10	001.10	001.10	001.10	001.10	001.10	001.10	001.10
10	Duration	1.70	1.70	1.70	1.70	1.70	1.70	1.70	1.70
	Cost	001.10	001.10	001.10	001.10	001.10	001.10	001.10	001.10
11	Duration	2.10	2.10	2.10	2.10	2.10	2.10	2.10	2.10
	Cost	001.10	001.10	001.10	001.10	001.10	001.10	001.10	001.10
12	Duration	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00
	Cost	001.10	001.10	001.10	001.10	001.10	001.10	001.10	001.10
:	Duration	:	:	:	:	:	:	:	:
	Cost	:	:	:	:	:	:	:	:
32	Duration	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00
	Cost	001.10	001.10	001.10	001.10	001.10	001.10	001.10	001.10

To program the cost of first unit of a pulse rate type for normal days, dial:

- ***2609-Pulse Rate Type-Time Zone-Cost of First Unit***
  Where,
  Pulse Rate Type is from 01 to 32.

Time Zone is from 1 to 4.
Cost of first Unit is from XXX.XX.

To program the cost of additional unit of a pulse rate type for normal days, dial:

- ***2610-Pulse Rate Type-Time Zone-Cost of Additional Unit***
  Where,
  Pulse Rate Type is from 01 to 32.
  Time Zone is from 1 to 4.
  Cost of Additional Unit is from XXX.XX.

Discounted Pulse Rate Schedule

To load default Discounted Pulse Rate type:

- ***2611***

CCC Pulse Rate Type		Time Zone 1		Time Zone 2		Time Zone 3		Time Zone 4	
		First Unit	Add. Unit	First Unit	Add. Unit	First Unit	Add. Unit	First Unit	Add. Unit
01	Duration	180.00	180.00	180.00	180.00	180.00	180.00	180.00	180.00
	Cost	001.10	001.10	001.10	001.10	001.10	001.10	001.10	001.10
02	Duration	300.00	300.00	300.00	300.00	300.00	300.00	300.00	300.00
	Cost	001.10	001.10	001.10	001.10	001.10	001.10	001.10	001.10
03	Duration	90.00	90.00	90.00	90.00	90.00	90.00	90.00	90.00
	Cost	001.10	001.10	001.10	001.10	001.10	001.10	001.10	001.10
04	Duration	120.00	120.00	120.00	120.00	120.00	120.00	120.00	120.00
	Cost	001.10	001.10	001.10	001.10	001.10	001.10	001.10	001.10
05	Duration	30.00	30.00	30.00	30.00	30.00	30.00	30.00	30.00
	Cost	001.10	001.10	001.10	001.10	001.10	001.10	001.10	001.10
06	Duration	30.00	30.00	30.00	30.00	30.00	30.00	30.00	30.00
	Cost	001.10	001.10	001.10	001.10	001.10	001.10	001.10	001.10
07	Duration	16.00	16.00	16.00	16.00	16.00	16.00	16.00	16.00
	Cost	001.10	001.10	001.10	001.10	001.10	001.10	001.10	001.10
08	Duration	3.30	3.30	3.30	3.30	3.30	3.30	3.30	3.30
	Cost	001.10	001.10	001.10	001.10	001.10	001.10	001.10	001.10
09	Duration	2.10	2.10	2.10	2.10	2.10	2.10	2.10	2.10
	Cost	001.10	001.10	001.10	001.10	001.10	001.10	001.10	001.10
10	Duration	1.50	1.50	1.50	1.50	1.50	1.50	1.50	1.50
	Cost	001.10	001.10	001.10	001.10	001.10	001.10	001.10	001.10
11	Duration	2.10	2.10	2.10	2.10	2.10	2.10	2.10	2.10
	Cost	001.10	001.10	001.10	001.10	001.10	001.10	001.10	001.10

CCC Pulse Rate Type		Time Zone 1		Time Zone 2		Time Zone 3		Time Zone 4	
		First Unit	Add. Unit	First Unit	Add. Unit	First Unit	Add. Unit	First Unit	Add. Unit
12	Duration	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00
	Cost	001.10	001.10	001.10	001.10	001.10	001.10	001.10	001.10
:	Duration	:	:	:	:	:	:	:	:
	Cost	:	:	:	:	:	:	:	:
16	Duration	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00
	Cost	001.10	001.10	001.10	001.10	001.10	001.10	001.10	001.10

To program duration for a first unit of a pulse rate type on Special Days, dial:

- **2612-Pulse Rate Type-Time Zone-Duration of First Unit**
  Where,
  Pulse rate type is from 01 to 32.
  Time Zone from 1 to 4.
  Duration of First Unit is from 000.00 to 999.99.

To program duration for additional unit for a pulse rate type on Special Days, dial:

- ***2613-Pulse Rate Type-Time Zone-Duration of Additional Unit***
  Where,
  Pulse rate type is from 01 to 32.
  Time Zone from 1 to 4.
  Duration of Additional Unit is from 000.00 to 999.99.

To program the cost of first unit of a pulse rate type for holidays, dial:

- ***2614-Pulse Rate Type-Time Zone-Cost of First Unit***
  Where,
  Pulse Rate Type is from 01 to 32.
  Time Zone is from 1 to 4.
  Cost of First Unit is from XXX.XX.

To program the cost of additional unit of a pulse rate type for Special Days, dial:

- ***2615-Pulse Rate Type-Time Zone-Cost of Additional Unit***
  Where,
  Pulse Rate Type is from 01 to 32.
  Time Zone is from 1 to 4.
  Cost of Additional Unit is from XXX.XX.

Area Code Table

To program an area code, dial:

- ***2620-Area Code Index-Area Code-#****
  Where,
  Area Code Index is from 001 to 999.
  Area Code is a number string of maximum of 4 digits.

To clear the area code for an index, dial:

- ***2620-Area Code Index-#****

Example: Program area code 022 for Mumbai at area code index 001, dial ***2620-001-022-#****

To program Pulse Rate Type for Pulse Rate Option of area code index, dial:

- ***2621-Area Code Index-Pulse Rate Option-Pulse Rate Type***
  Where,
  Area Code Index is from 001 to 999.
  Pulse Rate Option is from 1 to 4.
  Pulse Rate Type is from 01 to 32.

To delete the complete Area Code Table, dial:

- ***2622-Reverse SE Password***

Example: To delete the area code table, dial ***2622-4321*** (The SE password is assumed to be 1234).

There is no command to delete a single entry from the area code table. However, these can be cleared or overwritten.

To program ignore digit count when Service Provider based LCR is to be used, dial:

- ***2623-Area Code Index-Ignore Digit Count***
  Where,
  Area Code Index is from 001 to 999. Refer "Default Area Code Table for the Region-USA" at end of the chapter.
  Ignore Digit Count is from 0 to 9.
  By default, Ignore digit is 0.
  Please refer topic "Configuring LCR" for more details on Service Provider base LCR.

Special Days

To assign a Pulse Rate Table to any day of the week, dial:

- 2630-Day-Code
  Where,

Index	1	2	3	4	5	6	7
Day	Sun	Mon	Tue	Wed	Thu	Fri	Sat

Code	Meaning
0	Normal Pulse Rate
1	Discounted Pulse Rate

Example: To program Discounted Pulse Rate for Tuesday, dial ***2630-3-1***

To program a special date, dial:

- ***2631-Special Date Index-Date-Month***
  Where,
  Special Date Index is from 1 to 5 (Five dates can be programmed).
  Date is from 01 to 31.
  Month is from 01 to 12.
  By default, Special is shown below:

Special Date Index	Day
1	26-01
2	15-08

Special Date Index	Day
3	02-10
4	Blank
5	Blank

To clear a Special date index, dial:

- ***2632-Special Date Index***

Example: To program 1st May as a Special Day, dial **2631-1-01-05**

To program Area code name, dial:

- ***2633-Area Code Index-Name***
  Where,
  Area Code Index is from 001 to 999.
  Name is character string is of maximum 12 characters.

Use following command to clear area code name, dial:

- ***2633-Area Code Index-#****

Default Area Code Table for the Region-USA

Index	Area Code	Area Name	Pulse Rate Type	Ignore Digit Count
1	1201	NJ	2	0
2	1202	DC	2	0
3	1203	CT	2	0
4	1204	Manitoba	2	0
5	1205	AL	2	0
6	1206	WA	2	0
7	1207	ME	2	0
8	1208	ID	2	0
9	1209	CA	2	0
10	1210	TX	2	0
11	1212	NY	2	0
12	1213	CA	2	0
13	1214	TX	2	0
14	1215	PA	2	0
15	1216	OH	2	0
16	1217	IL	2	0
17	1218	MN	2	0
18	1219	IN	2	0
19	1224	IL	2	0

Index	Area Code	Area Name	Pulse Rate Type	Ignore Digit Count
20	1225	LA	2	0
21	1226	Ontario	2	0
22	1228	MS	2	0
23	1229	GA	2	0
24	1231	MI	2	0
25	1234	OH	2	0
26	1239	FL	2	0
27	1240	MD	2	0
28	1242	Bahamas	2	0
29	1246	Barbados	2	0
30	1248	MI	2	0
31	1250	BC	2	0
32	1251	AL	2	0
33	1252	NC	2	0
34	1253	WA	2	0
35	1254	TX	2	0
36	1256	AL	2	0
37	1260	IN	2	0
38	1262	WI	2	0
39	1264	Anguilla	2	0
40	1267	PA	2	0
41	1268	Antigua	2	0
42	1269	MI	2	0
43	1270	KY	2	0
44	1276	VA	2	0
45	1281	TX	2	0
46	1284	BVI	2	0
47	1289	Ontario	2	0
48	1301	MD	2	0
49	1302	DE	2	0
50	1303	CO	2	0
51	1304	WV	2	0
52	1305	FL	2	0
53	1306	Saskatchewan	2	0

Index	Area Code	Area Name	Pulse Rate Type	Ignore Digit Count
54	1307	WY	2	0
55	1308	NE	2	0
56	1309	IL	2	0
57	1310	CA	2	0
58	1312	IL	2	0
59	1313	MI	2	0
60	1314	MO	2	0
61	1315	NY	2	0
62	1316	KS	2	0
63	1317	IN	2	0
64	1318	LA	2	0
65	1319	IA	2	0
66	1320	MN	2	0
67	1321	FL	2	0
68	1323	CA	2	0
69	1325	TX	2	0
70	1330	OH	2	0
71	1331	IL	2	0
72	1334	AL	2	0
73	1336	NC	2	0
74	1337	LA	2	0
75	1339	MA	2	0
76	1340	USVI	2	0
77	1345	Cayman	2	0
78	1347	NY	2	0
79	1351	MA	2	0
80	1352	FL	2	0
81	1360	WA	2	0
82	1361	TX	2	0
83	1386	FL	2	0
84	1401	RI	2	0
85	1402	NE	2	0
86	1403	Alberta	2	0
87	1404	GA	2	0

Index	Area Code	Area Name	Pulse Rate Type	Ignore Digit Count
88	1405	OK	2	0
89	1406	MT	2	0
90	1407	FL	2	0
91	1408	CA	2	0
92	1409	TX	2	0
93	1410	MD	2	0
94	1412	PA	2	0
95	1413	MA	2	0
96	1414	WI	2	0
97	1415	CA	2	0
98	1416	Ontario	2	0
99	1417	MO	2	0
100	1418	Quebec	2	0
101	1419	OH	2	0
102	1423	TN	2	0
103	1424	CA	2	0
104	1425	WA	2	0
105	1430	TX	2	0
106	1432	TX	2	0
107	1434	VA	2	0
108	1435	UT	2	0
109	1438	Quebec	2	0
110	1440	OH	2	0
111	1441	Bermuda	2	0
112	1443	MD	2	0
113	1450	Quebec	2	0
114	1456	NANParea	2	0
115	1469	TX	2	0
116	1473	Grenada	2	0
117	1478	GA	2	0
118	1479	AR	2	0
119	1480	AZ	2	0
120	1484	PA	2	0
121	1500	NANParea	2	0

Index	Area Code	Area Name	Pulse Rate Type	Ignore Digit Count
122	1501	AR	2	0
123	1502	KY	2	0
124	1503	OR	2	0
125	1504	LA	2	0
126	1505	NM	2	0
127	1506	NewBrunswick	2	0
128	1507	MN	2	0
129	1508	MA	2	0
130	1509	WA	2	0
131	1510	CA	2	0
132	1512	TX	2	0
133	1513	OH	2	0
134	1514	Quebec	2	0
135	1515	IA	2	0
136	1516	NY	2	0
137	1517	MI	2	0
138	1518	NY	2	0
139	1519	Ontario	2	0
140	1520	AZ	2	0
141	1530	CA	2	0
142	1540	VA	2	0
143	1541	OR	2	0
144	1551	NJ	2	0
145	1559	CA	2	0
146	1561	FL	2	0
147	1562	CA	2	0
148	1563	IA	2	0
149	1567	OH	2	0
150	1570	PA	2	0
151	1571	VA	2	0
152	1573	MO	2	0
153	1574	IN	2	0
154	1575	NM	2	0
155	1580	OK	2	0

Index	Area Code	Area Name	Pulse Rate Type	Ignore Digit Count
156	1585	NY	2	0
157	1586	MI	2	0
158	1600	Canada	2	0
159	1601	MS	2	0
160	1602	AZ	2	0
161	1603	NH	2	0
162	1604	BC	2	0
163	1605	SD	2	0
164	1606	KY	2	0
165	1607	NY	2	0
166	1608	WI	2	0
167	1609	NJ	2	0
168	1610	PA	2	0
169	1612	MN	2	0
170	1613	Ontario	2	0
171	1614	OH	2	0
172	1615	TN	2	0
173	1616	MI	2	0
174	1617	MA	2	0
175	1618	IL	2	0
176	1619	CA	2	0
177	1620	KS	2	0
178	1623	AZ	2	0
179	1626	CA	2	0
180	1630	IL	2	0
181	1631	NY	2	0
182	1636	MO	2	0
183	1641	IA	2	0
184	1646	NY	2	0
185	1647	Ontario	2	0
186	1649	T&CIsland	2	0
187	1650	CA	2	0
188	1651	MN	2	0
189	1660	MO	2	0

Index	Area Code	Area Name	Pulse Rate Type	Ignore Digit Count
190	1661	CA	2	0
191	1662	MS	2	0
192	1664	Montsrat	2	0
193	1670	CNMI	2	0
194	1671	GU	2	0
195	1678	GA	2	0
196	1682	TX	2	0
197	1684	AS	2	0
198	1700	NANParea	2	0
199	1701	ND	2	0
200	1702	NV	2	0
201	1703	VA	2	0
202	1704	NC	2	0
203	1705	Ontario	2	0
204	1706	GA	2	0
205	1707	CA	2	0
206	1708	IL	2	0
207	1709	Newfoundland	2	0
208	1710	US	2	0
209	1712	IA	2	0
210	1713	TX	2	0
211	1714	CA	2	0
212	1715	WI	2	0
213	1716	NY	2	0
214	1717	PA	2	0
215	1718	NY	2	0
216	1719	CO	2	0
217	1720	CO	2	0
218	1724	PA	2	0
219	1727	FL	2	0
220	1731	TN	2	0
221	1732	NJ	2	0
222	1734	MI	2	0
223	1740	OH	2	0

Index	Area Code	Area Name	Pulse Rate Type	Ignore Digit Count
224	1754	FL	2	0
225	1757	VA	2	0
226	1758	St.Lucia	2	0
227	1760	CA	2	0
228	1762	GA	2	0
229	1763	MN	2	0
230	1765	IN	2	0
231	1767	Dominica	2	0
232	1769	MS	2	0
233	1770	GA	2	0
234	1772	FL	2	0
235	1773	IL	2	0
236	1774	MA	2	0
237	1775	NV	2	0
238	1778	BC	2	0
239	1779	IL	2	0
240	1780	Alberta	2	0
241	1781	MA	2	0
242	1784	St. V&G	2	0
243	1785	KS	2	0
244	1786	FL	2	0
245	1787	PrtoRico	2	0
246	1800	NANParea	2	0
247	1801	UT	2	0
248	1802	VT	2	0
249	1803	SC	2	0
250	1804	VA	2	0
251	1805	CA	2	0
252	1806	TX	2	0
253	1807	Ontario	2	0
254	1808	HI	2	0
255	1809	DomRepub	2	0
256	1810	MI	2	0
257	1812	IN	2	0

Index	Area Code	Area Name	Pulse Rate Type	Ignore Digit Count
258	1813	FL	2	0
259	1814	PA	2	0
260	1815	IL	2	0
261	1816	MO	2	0
262	1817	TX	2	0
263	1818	CA	2	0
264	1819	Quebec	2	0
265	1828	NC	2	0
266	1829	DomRepub	2	0
267	1830	TX	2	0
268	1831	CA	2	0
269	1832	TX	2	0
270	1843	SC	2	0
271	1845	NY	2	0
272	1847	IL	2	0
273	1848	NJ	2	0
274	1850	FL	2	0
275	1856	NJ	2	0
276	1857	MA	2	0
277	1858	CA	2	0
278	1859	KY	2	0
279	1860	CT	2	0
280	1862	NJ	2	0
281	1863	FL	2	0
282	1864	SC	2	0
283	1865	TN	2	0
284	1866	NANParea	2	0
285	1867	Yukon	2	0
286	1868	Tri&Tob	2	0
287	1869	St. K&N	2	0
288	1870	AR	2	0
289	1876	Jamaica	2	0
290	1877	NANParea	2	0
291	1878	PA	2	0

Index	Area Code	Area Name	Pulse Rate Type	Ignore Digit Count
292	1888	NANParea	2	0
293	1900	NANParea	2	0
294	1901	TN	2	0
295	1902	N Scotia	2	0
296	1903	TX	2	0
297	1904	FL	2	0
298	1905	Ontario	2	0
299	1906	MI	2	0
300	1907	AK	2	0
301	1908	NJ	2	0
302	1909	CA	2	0
303	1910	NC	2	0
304	1912	GA	2	0
305	1913	KS	2	0
306	1914	NY	2	0
307	1915	TX	2	0
308	1916	CA	2	0
309	1917	NY	2	0
310	1918	OK	2	0
311	1919	NC	2	0
312	1920	WI	2	0
313	1925	CA	2	0
314	1928	AZ	2	0
315	1931	TN	2	0
316	1936	TX	2	0
317	1937	OH	2	0
318	1939	PrtoRico	2	0
319	1940	TX	2	0
320	1941	FL	2	0
321	1947	MI	2	0
322	1949	CA	2	0
323	1951	CA	2	0
324	1952	MN	2	0
325	1954	FL	2	0

Index	Area Code	Area Name	Pulse Rate Type	Ignore Digit Count
326	1956	TX	2	0
327	1970	CO	2	0
328	1971	OR	2	0
329	1972	TX	2	0
330	1973	NJ	2	0
331	1978	MA	2	0
332	1979	TX	2	0
333	1980	NC	2	0
334	1985	LA	2	0
335	1989	MI	2	0
336	0117	Kazkhstan	3	0
337	01120	Egypt	3	0
338	01127	South Africa	3	0
339	01130	Greece	3	0
340	01131	Netherlands	3	0
341	01132	Belgium	3	0
342	01133	France	3	0
343	01134	Spain	3	0
344	01136	Hungary	3	0
345	01139	VaticanCity	3	0
346	01140	Romania	3	0
347	01141	Switzerland	3	0
348	01143	Austria	3	0
349	01144	UK	3	0
350	01145	Denmark	3	0
351	01146	Sweden	3	0
352	01147	Norway	3	0
353	01148	Poland	3	0
354	01149	Germany	3	0
355	01151	Peru	3	0
356	01152	Mexico	3	0
357	01153	Cuba	3	0
358	01154	Argentine	3	0
359	01155	Brazil	3	0

Index	Area Code	Area Name	Pulse Rate Type	Ignore Digit Count
360	01156	Chile	3	0
361	01157	Colombia	3	0
362	01158	Venezuela	3	0
363	01160	Malaysia	3	0
364	01161	Australia	3	0
365	01162	Indonesia	3	0
366	01163	Philippines	3	0
367	01164	NZ	3	0
368	01165	Singapore	3	0
369	01166	Thailand	3	0
370	01181	Japan	3	0
371	01182	Korea	3	0
372	01184	VietNam	3	0
373	01186	China	3	0
374	01190	Turkey	3	0
375	01191	India	3	0
376	01192	Pakistan	3	0
377	01193	Afghanistan	3	0
378	01194	Sri Lanka	3	0
379	01195	Myanmar	3	0
380	01198	Iran	3	0
381	011212	Morocco	3	0
382	011213	Algeria	3	0
383	011216	Tunisia	3	0
384	011218	Libya	3	0
385	011220	Gambia	3	0
386	011221	Senegal	3	0
387	011222	Mauritania	3	0
388	011223	Mali	3	0
389	011224	Guinea	3	0
390	011225	IvoryCoast	3	0
391	011226	BurkinaFaso	3	0
392	011227	Niger	3	0
393	011228	Togolese	3	0

Index	Area Code	Area Name	Pulse Rate Type	Ignore Digit Count
394	011229	Benin	3	0
395	011230	Mauritius	3	0
396	011231	Liberia	3	0
397	011232	SierraLeone	3	0
398	011233	Ghana	3	0
399	011234	Nigeria	3	0
400	011235	Chad	3	0
401	011236	CenAfrica	3	0
402	011237	Cameroon	3	0
403	011238	CapeVerde	3	0
404	011239	SaoTome	3	0
405	011240	Equtl_Guinea	3	0
406	011241	Gabonese	3	0
407	011242	Congo	3	0
408	011243	CongoDem	3	0
409	011244	Angola	3	0
410	011245	GuineaBissa	3	0
411	011246	DiegoGarcia	3	0
412	011247	Ascension	3	0
413	011248	Seychelles	3	0
414	011249	Sudan	3	0
415	011250	Rwandese	3	0
416	011251	Ethiopia	3	0
417	011252	SomalianRep	3	0
418	011253	Djibouti	3	0
419	011254	Kenya	3	0
420	011255	Tanzania	3	0
421	011256	Uganda	3	0
422	011257	Burundi	3	0
423	011258	Mozambique	3	0
424	011260	Zambia	3	0
425	011261	Madagascar	3	0
426	011262	Reunion	3	0
427	011263	Zimbabwe	3	0

Index	Area Code	Area Name	Pulse Rate Type	Ignore Digit Count
428	011264	Namibia	3	0
429	011265	Malawi	3	0
430	011266	Lesotho	3	0
431	011267	Botswana	3	0
432	011268	Swaziland	3	0
433	011269	Comoros	3	0
434	011290	StHelena	3	0
435	011291	Eritrea	3	0
436	011297	Aruba	3	0
437	011298	FaroeIsland	3	0
438	011299	Greenland	3	0
439	011350	Gibraltar	3	0
440	011351	Portugal	3	0
441	011352	Luxembourg	3	0
442	011353	Ireland	3	0
443	011354	Iceland	3	0
444	011355	Albania	3	0
445	011356	Malta	3	0
446	011357	Cyprus	3	0
447	011358	Finland	3	0
448	011359	Bulgaria	3	0
449	011370	Lithuania	3	0
450	011371	Latvia	3	0
451	011372	Estonia	3	0
452	011373	Moldova	3	0
453	011374	Armenia	3	0
454	011375	Belarus	3	0
455	011376	Andorra	3	0
456	011377	Monaco	3	0
457	011378	SanMarino	3	0
458	011379	VaticanCity	3	0
459	011380	Ukraine	3	0
460	011381	Yugoslavia	3	0
461	011385	Croatia	3	0

Index	Area Code	Area Name	Pulse Rate Type	Ignore Digit Count
462	011386	Slovenia	3	0
463	011387	Bosnia	3	0
464	011389	Macedonia	3	0
465	011420	Czech Repub	3	0
466	011421	Slovakia	3	0
467	011423	Liechtenstein	3	0
468	011500	Falklands	3	0
469	011501	Belize	3	0
470	011502	Guatemala	3	0
471	011503	El Salvador	3	0
472	011504	Honduras	3	0
473	011505	Nicaragua	3	0
474	011506	CostaRica	3	0
475	011507	Panama	3	0
476	011508	St.Pierre	3	0
477	011509	Haiti	3	0
478	011590	Guadeloupe	3	0
479	011591	Bolivia	3	0
480	011592	Guyana	3	0
481	011593	Ecuador	3	0
482	011594	FrenchGuyana	3	0
483	011595	Paraguay	3	0
484	011596	Martinique	3	0
485	011597	Suriname	3	0
486	011598	Uruguay	3	0
487	011599	NethAntilles	3	0
488	011670	East Timor	3	0
489	011672	Antarctic	3	0
490	011673	Brunei	3	0
491	011674	Nauru	3	0
492	011675	PapuaNewGuin	3	0
493	011676	Tonga	3	0
494	011677	SolomonIslnd	3	0
495	011678	Vanuatu	3	0

Index	Area Code	Area Name	Pulse Rate Type	Ignore Digit Count
496	011679	Fiji	3	0
497	011680	Palau	3	0
498	011681	Wallis Island	3	0
499	011682	Cook Islands	3	0
500	011683	Niuel Island	3	0
501	011684	AmerSamoa	3	0
502	011685	WSamoa	3	0
503	011686	Kiribati	3	0
504	011687	NewCaledonia	3	0
505	011688	Tuvalu	3	0
506	011689	FrenchPolyne	3	0
507	011690	Tokelau	3	0
508	011691	Micronesia	3	0
509	011692	MarshalIslnd	3	0
510	011850	Korea North	3	0
511	011852	Hong Kong	3	0
512	011853	Macau	3	0
513	011855	Cambodia	3	0
514	011856	Laos	3	0
515	011870	SatIndlOcn	3	0
516	011871	SatEastAtl	3	0
517	011872	SatPacific	3	0
518	011873	SatIndianOcn	3	0
519	011874	SatWestAtl	3	0
520	011880	Bangladesh	3	0
521	011960	Maldives	3	0
522	011961	Lebanon	3	0
523	011962	Jordan	3	0
524	011963	SyrianArab	3	0
525	011964	Iraq	3	0
526	011965	Kuwait	3	0
527	011966	SaudiArabia	3	0
528	011967	Yemen	3	0
529	011968	Oman	3	0

Index	Area Code	Area Name	Pulse Rate Type	Ignore Digit Count
530	011971	UAE	3	0
531	011972	Israel	3	0
532	011973	Bahrain	3	0
533	011974	Qatar	3	0
534	011975	Bhutan	3	0
535	011976	Mongolia	3	0
536	011977	Nepal	3	0
537	011992	Tajikistan	3	0
538	011993	Turkmenistan	3	0
539	011994	Azerbaijani	3	0
540	011995	Georgia	3	0
541	011996	Kyrgyzstan	3	0
542	011998	Uzbekistan	3	0
543				
544				
545				
546				
547				
:				
998				
999				

Call Cost Display

What is this?

With Call Cost Display, you can view the cost of the last 10 external calls made from your extension. These external calls may have been made from Trunk Ports and on the Tie line network.

The system will display the dialed numbers and the call cost for each number that it has calculated on the LCD of EON / Extended IP Phone.

How to configure

For this feature to work, it must be enabled on the extension by the System Administrator (SA).

To enable 'Call Cost Display' for an extension:

- Enter SA mode.
- Dial command ***1072-181-Station-Flag***
  Where,
  Station is Number of the Extension
  Flag is
  0 for Disable
  1 for Enable
  By default Call Cost Display is enabled.
- Exit SA mode.

How to use

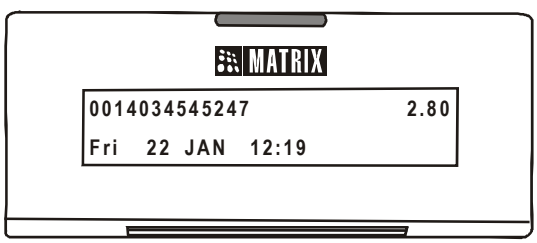
For EON and Extended IP Phone Users Only

- Press DSS Key assigned to Call Cost Display feature.

  OR

- Dial 1075.
- Scroll with the up/down navigation keys to view the cost of the last 10 calls.
- The display shows the last 10 dialed numbers and their corresponding call cost.

  For example: If the call charge is for the dialed number 0014034545247 is $2 and 80 cents' then the display will show:



- The display is retained till the extension remains OFF-Hook.
- Go Idle or you get dial tone after 3 seconds.

Call Duration Control (CDC)

What's this?

Call Duration Control (CDC) allows a maximum time limit to be set on internal and external (both incoming and outgoing) telephone calls. When the maximum call duration is reached, the calls are disconnected, after a warning tone indicating to the user that the calls in progress will be disconnected.

By limiting the duration of the conversations, CDC helps increase availability of trunks for making outgoing calls and for receiving incoming calls, which is important in high call traffic situations. Besides increasing trunk availability, CDC curbs unrelated and unproductive conversations.

By default this feature is enabled on the trunks and extensions, hence calls will be disconnected after the expiry of the timer. You can change the configurations as per your requirement, refer “How to configure” and “Outgoing Call Routing”.

How it works

- A is an extension user. B is an external number.

External-Outgoing Calls

- A dials B's number.
- The system checks the CDC Table applied to A.
- SARVAM UCS will check whether the CDC is applicable on Extension of A as well as the Trunk used for making the outgoing call.

To check whether CDC is applicable on Extension A

- The system checks the **CDC Table** selected in the Station Advanced Feature Template assigned to A. In the Table,
  - It checks whether the check box, **Apply CDC for outgoing calls made from trunk**, is enabled.
  - It matches B's number with the entries on the **Apply CDC for calls matching with numbers** list and the **Do Not Apply CDC for calls matching with numbers** list in the CDC table.

To check whether CDC is applicable on the Trunk

- The system checks the **Call Duration Control - For OG Calls** check box in the Trunk Features Template assigned to the trunk.

The following results are possible:

- In the CDC Table, the **Apply CDC for outgoing calls made from trunk** check box is enabled and a match is found for the number in the **Apply CDC for calls matching with numbers** List. The **Call Duration Control - For OG Calls** check box is enabled in the Trunk Features Template assigned to the trunk. CDC is applied on the call.

- In the CDC Table, the **Apply CDC for outgoing calls made from trunk** check box is enabled and a match is found for the number in the **Apply CDC for calls matching with numbers** List. The **Call Duration Control - For OG Calls** check box is disabled in the Trunk Features Template assigned to the trunk. CDC is not applied on the call.

- In the CDC Table, the **Apply CDC for outgoing calls made from trunk** check box is enabled and a match is found in the **Do Not Apply CDC for calls matching with numbers** list. The **Call Duration Control - For OG Calls** check box is enabled in the Trunk Features Template assigned to the trunk. CDC is not applied on the call.

- In the CDC Table, the **Apply CDC for outgoing calls made from trunk** check box is enabled and a match is found in the **Do Not Apply CDC for calls matching with numbers** list. The **Call Duration Control - For OG Calls** check box is disabled in the Trunk Features Template assigned to the trunk. CDC is not applied on the call.

- In the CDC Table, the **Apply CDC for outgoing calls made from trunk** check box is enabled and a match is found in both the number lists. The system gives precedence to the Do Not Apply CDC for calls matching with numbers list. The **Call Duration Control - For OG Calls** check box is enabled in the Trunk Features Template assigned to the trunk. CDC is not applied on the call.

- In the CDC Table, the **Apply CDC for outgoing calls made from trunk** check box is enabled and a match is found in both the number lists. The system gives precedence to the Do Not Apply CDC for calls matching with numbers list. The **Call Duration Control - For OG Calls** check box is disabled in the Trunk Features Template assigned to the trunk. CDC is not applied on the call.

- When CDC is applied to the call, the CDC Timer starts as soon as B has answered the call. This timer is set to 160 seconds as default, but can be programmed to the desired time limit.

- At the end of the default/programmed time limit of the CDC Timer, the Beep Timer starts (5 seconds; non-programmable) then the CDC Goodbye Timer starts. This Goodbye timer provides a grace period of 20 seconds for the user to finish the call. This Timer is non-programmable.

- At the end of the Goodbye Timer, the call is disconnected, if the **Disconnect CDC after Time** check box is enabled.

- If this flag has not been programmed, the call will not be disconnected.

- Instead, the **CDC Timer** will be loaded again for the default/programmed duration. The user can know how long s/he has been talking.

- A is played Warning Beeps. B cannot hear the beeps. This continues until either party disconnects.

External-Incoming Calls

CDC works similarly for incoming calls.

- B calls A.

- The system checks whether the following conditions are fulfilled,

  - the **Apply CDC for incoming calls received from trunk** check box is enabled in the CDC Table assigned to A.

- the **Disconnect Call after CDC Timer** check box is enabled in the CDC Table assigned to A.

- a match is found for the B's number in the **Apply CDC for calls matching with numbers** List.
- the **Call Duration Control - For IC Calls** check box is enabled in the Trunk Features Template assigned to the trunk on which the call is received.

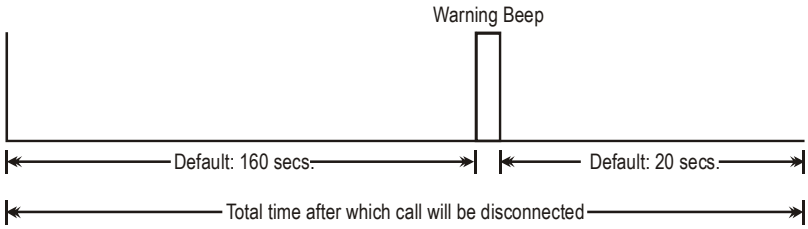
  CDC is applied on the call.

- A will get beeps.

Internal Calls

A and B are extension users.

- A calls B.
- The system checks whether the check box, **Apply CDC to Internal Calls**, is enabled in the CDC Table.
- If the check box is enabled, CDC is applied on the call.
- Warning beeps are played to A, who made the call.



In this case the total time after which the call is disconnected will be 160 seconds (default CDC Timer) + 5 seconds (Beep Timer; non-programmable) + 20 seconds (Goodbye Timer; non-programmable).

Feature Interactions:

- **Call Transfer:** In case of transferred call, the CDC timer gets reset and starts again afresh on the transferred extension.
- **Conference, Conversation Recording** and **Call Taping:** CDC is treated as turned OFF.
- **Call Park and Call Hold (Exclusive and Global):** CDC is treated as turned ON.
- **Interrupt Request, Barge-In:** CDC is treated as turned ON.
- **Raid:** CDC is not applicable when you raid an extension as the conversation is converted into a 3 party conference.
- **Emergency Number Dialing:** Emergency calls are not affected by this feature, i.e. CDC will not be applied on the dialing of Emergency Numbers.

  For Inter PINX or Intra PINX calls (QSIG Calls), the CDC will work only if it is enabled on the source port (calling extension) irrespective of whether CDC is enabled or disabled on the called extension.

How to configure

By default this feature is enabled on the Trunks and Extensions, that is:

- **Call Duration Control Table 1** is assigned to CDC for Trunks and Extensions.
- In the **Call Duration Control Table 1,** the check boxes **Call Duration Control For IC Calls**, **Call Duration Control For OG Calls**, **Apply CDC to Internal Calls** and **Disconnect Call after CDC Timer** all are enabled.
- In the **Call Duration Control Table 1, List Number 2** is assigned to **Apply CDC for calls matching with numbers.**
- The **CDC Timer** is set as 300 seconds.
- In the default **Trunk Feature Template 1** assigned to all the Trunks, the check boxes **Call Duration Control For IC Calls** and **Call Duration Control For OG Calls** are enabled.
- In the default **Station Advanced Feature Template 1** assigned to all the extensions, **CDC Table 1** is assigned.

If you wish to change the configurations, follow the steps as mentioned below:

- Configure Call Duration Control Table. You may configure up to 8 different Tables.
- Assign a CDC Table in the Station Advanced Feature Template of those extensions on which Call Duration Control is to be applied.
- Enable CDC for incoming and outgoing calls from the Trunks on which you want to apply this feature.

To program the Call Duration Control Table,

- decide the types of calls - Outgoing, Incoming and Internal - on which CDC is to be enabled.
- make a list of numbers on which CDC is to be applied, that is, the Apply CDC to Numbers List.
- Make a list of numbers on which CDC is not to be applied, the Do Not Apply CDC to Number List.

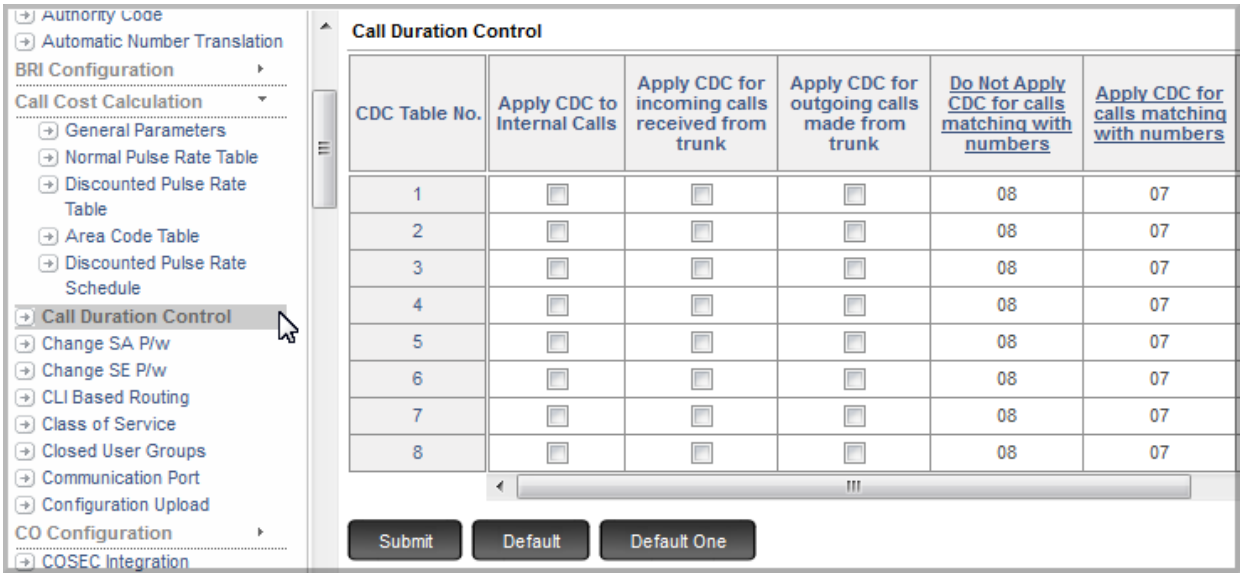
The Call Duration Control Table can be programmed using Jeeves and a Telephone.

Configuring CDC using Jeeves

CDC Table

- Log in as System Engineer.

- Under **Configuration,** click the **Call Duration Control** to open the page.



- The CDC Table will open. There are 8 CDC Tables. By default CDC Table No. 1 is assigned to all extensions of SARVAM UCS. If the same CDC is to be assigned to all extensions, program this table.

  If different CDC is to be applied to different extensions, program separate CDC tables for these extensions.

- Now program the following parameters in the table you have selected:

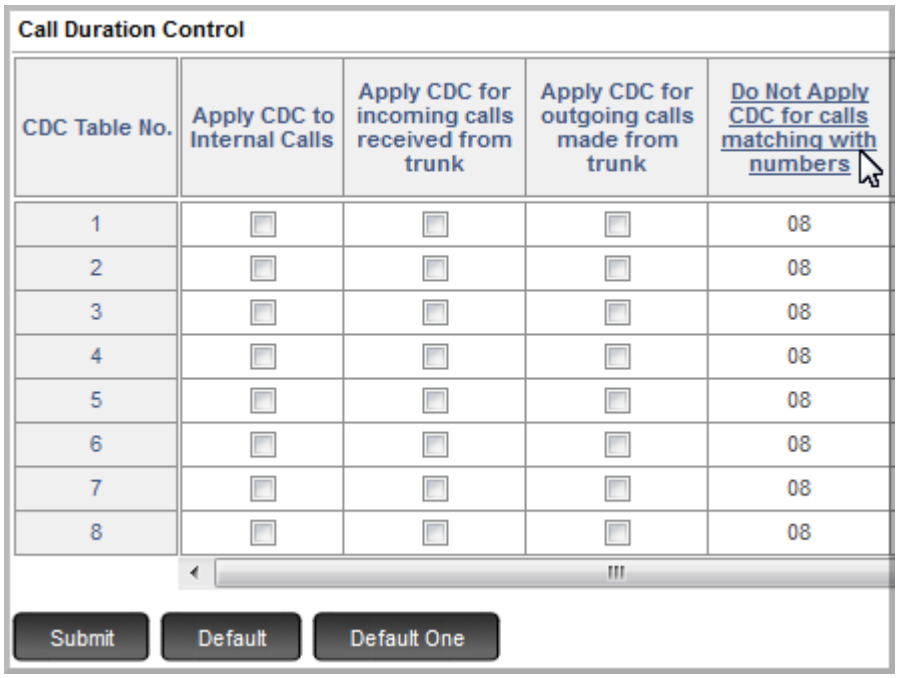
  - **Apply CDC to Internal Calls:** By default, this check box is enabled, that is, CDC will be applied on internal calls. Clear the check box to disable.

  - **Apply CDC to Incoming Calls received from Trunk:** By default, this check box is enabled, that is, CDC will be applied to incoming external calls. Clear the check box to disable.

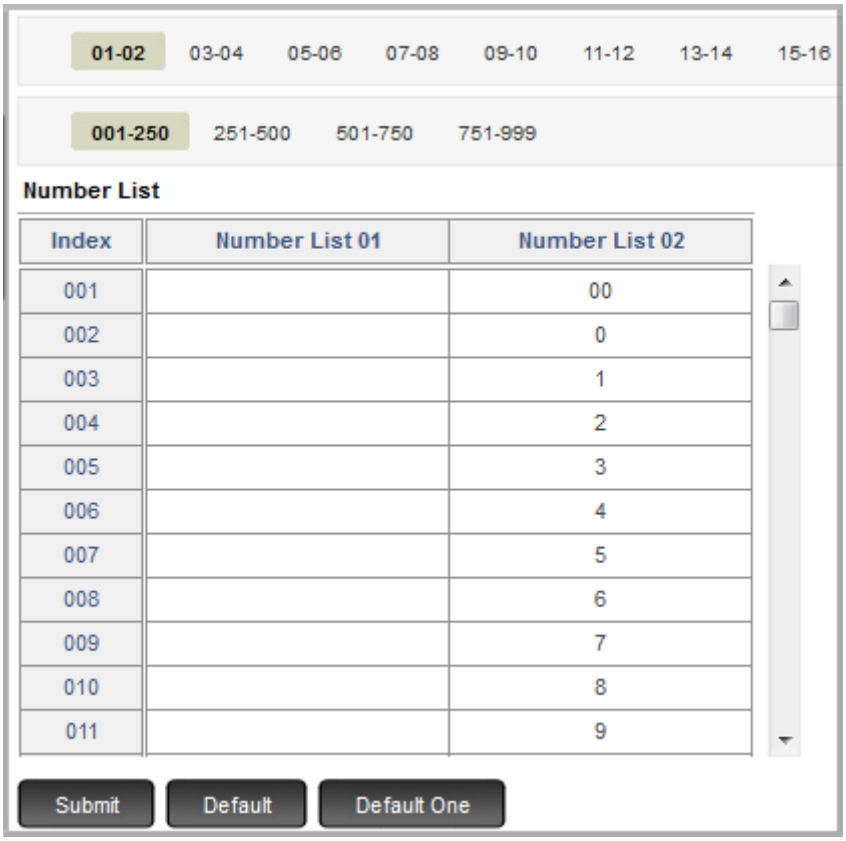
  - **Apply CDC to Outgoing Calls made from Trunks:** By default, this check box is enabled, that is, CDC will be applied to outgoing external calls. Clear the check box to disable.

  - **Do Not Apply CDC for calls matching with numbers:** This is the list of numbers on which CDC is not to be applied. By default, Number List 08 is assigned to this parameter. You must program this list with numbers which you want to be exempt from CDC.

    To program the list, click **Do Not Apply CDC for calls matching with numbers**.



This will lead you to the "Number Lists" page. Click '001-250' of Number list 07-08.



By default, Number List 08 is assigned to this parameter. You can also program any other Number List you want. Enter the list of numbers on which CDC is not be applied (refer to the list you prepared).

Click **Submit** at the bottom of the page to save your list.

Return to the **Call Duration Control** page. If you have prepared a Number List other than the default 08, then enter the number of that list in the Do Not Apply CDC to Number List column.

- **Apply CDC for calls matching with numbers:** This is the list of numbers on which CDC is to be applied. By default, Number List 02 is assigned to this parameter. You must program this list with numbers on which you want CDC to be applied.

  To program the list, click **Apply CDC for calls matching with numbers**. The 'Number Lists' page opens.

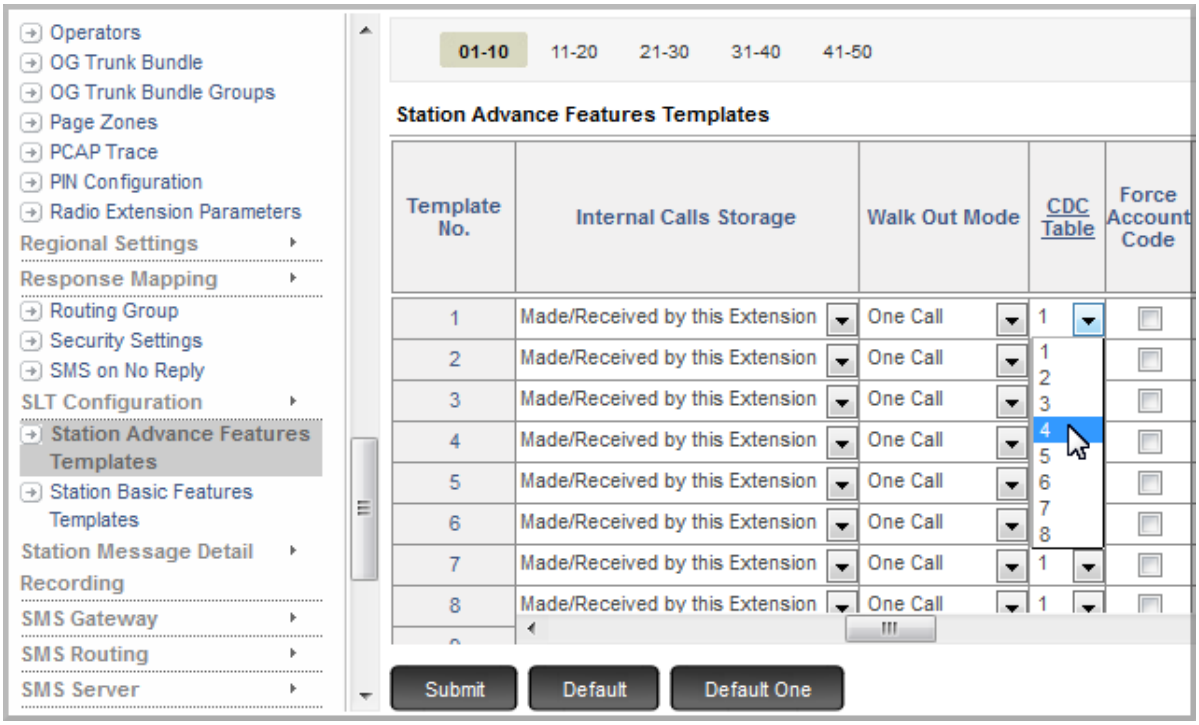
  Click '001-250' of Number list 01-02. Follow the same steps as described above for programming the Do Not Apply CDC Number List.

- **CDC Timer:** This is the time for which the warning beeps are to be played before the system disconnects the call. The range of the timer is 0001 to 9999 seconds. By default this Timer is set to 300 seconds. Set the CDC Timer to the desired time limit.

- **Disconnect Call after CDC Timer:** By default, this check box enabled, that is calls will automatically disconnect on the expiry of the CDC Timer. Clear the check box to disable, that is, if you do not want calls to disconnect on expiry of the CDC Timer.

- Click **Submit** at the bottom of the page to save changes.

- Under **Configuration**, click **Station Advanced Feature Template**. Ensure that the **CDC Table** Number (in this case 01) you have programmed is assigned in the Template you want to apply to the extensions.



- Click **Submit** at the bottom of the page to save changes.

- To assign the Station Advanced Feature Template with the CDC Table on SLT, DKP and ISDN Terminal extensions, SIP Extensions go to the respective pages **SLT Parameters** under **SLT Configuration**, **DKP**

Parameters under **DKP Configuration** and **ISDN Terminal Parameters** under **ISDN Configuration** and **SIP Extensions Settings** under **VoIP Configuration**.

Refer "Station Advanced Feature Template" for instructions on customizing the templates and assigning them to extensions.

- If selected extensions are to be allowed CDC or if different CDC parameters are to be allowed to selected extensions (for example, 160 seconds duration timer for a few extensions, 360 duration timer for some other extensions), then follow these steps:

  a. Define a new CDC table.

  b. Program the different CDC parameters in this table, as required for the extensions.

  c. Apply this CDC table on a separate Station Advanced Feature Template.

  d. Apply the new Station Advanced Feature Template now programmed with a different CDC table on the selected extensions which are to be allowed this feature.

CDC on Trunk

- By default, CDC is enabled in the Trunk Feature Template assigned to the trunks. For instructions, see "Configuring Trunks".

- Log out of Jeeves or continue programming if required.

Configuring CDC using a Telephone

CDC Table

- Enter SE mode.

  To enable CDC for Outgoing Calls in a CDC Table, dial:
  - ***4202-1-CDC Table-Code*** to enable the flag in a single table.
  - ***4202-2-CDC Table-CDC Table-Code*** to enable the flag in a range of tables.
  - ***4202-*-Code*** to enable the flag in all tables.
    Where,
    CDC Table is from 1 to 8.
    Code is
    0 for Disable
    1 for Enable.

  To enable CDC flag for Incoming Calls in a CDC Table, dial:
  - ***4203-1-CDC Table-Code*** to enable the flag in a single table.
  - ***4203-2-CDC Table-CDC Table-Code*** to enable the flag in a range of tables.
  - ***4203-*-Code*** to enable the flag in all tables.
    Where,
    CDC Table is from 1 to 8.
    Code is
    0 for Disable.
    1 for Enable.

To enable CDC flag for Internal Call, dial:

- ***4204-1-CDC Table-Code*** to enable the flag in a single table.
- ***4204-2-CDC Table-CDC Table-Code*** to enable the flag in a range of tables.
- ***4204-*-Code*** to enable the flag in all tables.
  Where,
  CDC Table is from 1 to 8.
  Code is
  0 for Disable.
  1 for Enable.

To assign a number list to Apply CDC to Number List, dial:

- ***4205-1-CDC Table-Number List*** to enable the flag in a single table.
- ***4205-2-CDC Table-CDC Table-Number List*** to enable the flag in a range of tables.
- ***4205-*-Number List*** to enable the flag in all tables.
  Where,
  CDC Table is from 1 to 8.
  Number List is from 01 to 16.
  Default Number List is 07.

To assign a number list to Do Not Apply CDC to Number List, dial:

- ***4206-1-CDC Table- Number List*** to assign Number List to a single table.
- ***4206-2-CDC Table-CDC Table- Number List*** to assign Number List to a range of tables.
- ***4206-*-Number List*** to assign Number List to all tables.
  Where,
  CDC Table is from 1 to 8.
  Number List is from 01 to 16.
  Default, Number List is 08.

You must also program the Number Lists (default: 07 and 08) before assigning them to a CDC table. Refer the topic 'Number Lists' for programming instructions.

To change CDC timer in a CDC table:

- ***4207-1-CDC Table-CDC Timer*** to set Timer value in a single table.
- ***4207-2-CDC Table-CDC Table-CDC Timer*** to set the same Timer value in a range of table
- ***4207-*-CDC Timer*** to set the same Timer value in all tables.
  Where,
  CDC Table is from 1 to 8.
  CDC Timer range is from 0001 to 9999 seconds.
  Default CDC Timer value is 160 seconds.

To enable the Disconnection Flag in a CDC table:

- ***4208-1-CDC Table-Disconnection Flag*** to enable the flag in a single table.
- ***4208-2-CDC Table-CDC Table-Disconnection Flag*** to enable the flag in a range of tables.
- ***4208-*-Disconnection Flag*** to enable the flag in all tables.
  Where,
  CDC Table is from 1 to 8.
  Disconnection Flag is
  0 for Disable.
  1 for Enable.

To default a CDC table[259] (see Table with Default Values):

259. Refer Call Duration Control Table (Default Values).

- ***4201-1-CDC Table*** to default a single table.
- ***4201-2-CDC Table-CDC Table*** to default a range of tables.
- ***4201-**** to default all tables.
  Where,
  CDC Table is 1 to 8.

Example: Apply Call Duration Control on SLT 202 (connected at software port number 001) to disconnect all calls starting with '0' after 240 seconds, except calls starting with '022'.

Solution: Since only one SLT is to be programmed, it is recommended that you use a CDC table other than the default (CDC Table No. 1). So, select another CDC table to be assigned to SLT extension 202, for example CDC Table No. 5. Follow these steps:

1. First, program the Apply CDC to Number List and Do Not Apply CDC to Number List. For example, take Number List 04 as the Apply CDC to Number List and program the number '0' in this list. Take Number List 05 as the Do Not Apply CDC to Number List and program '022' in this list. Refer the topic “Number Lists” for programming instructions.

2. Enable CDC for Outgoing call in the table. If using SE Commands, dial ***4202-1-5-1.***

3. Assign Number List 04 as allowed list and Number List 05 as denied list in table 5. If using SE commands dial ***4205-1-5-04*** to assign List 04 and dial ***4206-1-5-05*** to assign List 05.

4. Change the CDC timer to 240 seconds. If using SE Commands, dial ***4207-1-5-240.***

5. Enable CDC disconnection Flag in CDC Table 5. If using SE Commands, dial ***4208-1-5-1.***

6. Assign CDC Table 5 to SLT 202. To do this, change the CDC number in a Station Advanced Feature Template of SLT 202.

7. Program a different Station Advanced Feature Template for SLT 202. Ensure that all other features and parameters in the template are relevant for SLT 202.

8. Change the CDC Table number in the Template to 5.

9. Assign the Station Advanced Feature Template now programmed with CDC Table No. 5 to SLT 202. Refer the topic “Station Advanced Feature Template” for programming instructions.

Call Duration Display

What's this?

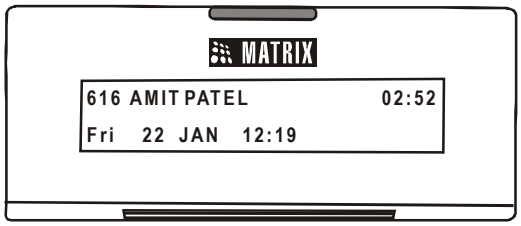
By invoking Call Duration Display, extension users can view the duration of the current call instantly.

The system displays the duration of the current call on the LCD display of the phone.

The current call may be incoming or outgoing, internal or external call.

How it works

- The user goes OFF-Hook
- Dials an external number.
- The remote party answers the call.
- The dialed external number with duration (5-digits in the format of MM:SS) is displayed on the LCD of the phone, when the call is answered.



Call Forward

What's this?

During a typical workday, it is common for people in an organization to move from one place to another. For instance, a manager might go on the production floor or remain in the conference room for a few hours; a field engineer may spend half of the day on site. So, they need to be able to attend their calls even when they are not present at their desks. The 'Call Forward' feature of SARVAM UCS ensures this.

Using this feature, calls landing on an extension can be forwarded to another extension, an external number, Voice Mail, or a Department Group. This way, extension users can ensure that callers can reach them and that they do not miss calls when they are not present at their extension.

You can also set Call Forward for all the extensions from the SA Mode. See “Settings Call Forward for All Extensions using SA Jeeves”.

The Call Forward feature of SARVAM UCS offers the following forwarding options:

- **Unconditionally** - calls are forwarded to the destination phone number automatically without waiting for a response from the called party's phone.
- **If Busy** - calls are forwarded to the destination phone number only when the called party's phone is busy.
- **If No Reply** - calls are forwarded to the destination phone number only when the called party does not answer the phone. Each extension can set a different time after which the call should be forwarded, in case of no reply. The default time is 30 seconds for all extensions and can be changed by programming the Call Forward No-Reply Timer.
- **If Busy or No Reply** - calls are forwarded to the destination phone number when the called party's phone is either busy or does not reply.
- **Dual Ring**[260] - when calls are forwarded to another phone number. Both phones, that is, the source phone (whose calls are forwarded) as well as the destination phone (on which call is forwarded) will ring and the user can answer from either extension.

  Dual Ring is useful to users who may have to be present frequently at two different places. As it is cumbersome to forward the calls from one extension to another and cancel it repeatedly, extensions users can set Dual Ring, so they can attend to their calls at either place they are present.

If you do not want to set/cancel Call Forward manually, you can set Preset Call Forward. Call will be forwarded automatically to the selected destination according to the type of Preset Call Forward set. You can set a different type of forward and destination for each time zone. *To know more, see “Preset Call Forward”.*

260. *This feature is not supported in Q-Signaling.*

How it works

A has set Call Forward to extension B unconditionally.

- The system forwards all calls for A to B, without checking for Busy Tone and without waiting for the Call Forward No-Reply Timer to expire.

A has set Call Forward to external number on No-Reply.

- The system waits for the Call Forward No-Reply Timer to expire and forwards all external incoming calls to the external number.

A has set Call Forward to extension B on Busy.

- The system forwards the call for A to B on detecting Busy signal from A.

B belongs to a Department Group and has set Call Forward-If Busy to C within the Department Group.

- If the system detects Busy signal on B, it forwards the call for B to C in the Department Group.
- However, if the caller has called the Department Group instead of calling B directly, the call will land in the sequence on all Department group extensions. When it is B's turn, the call will not be forwarded to C, B will ring instead.

C belongs to a Department Group and has set Call Forward-No Reply to D within the Department Group.

- The system waits for the Call Forward No-Reply Timer to expire, and forwards the call for C to D in the Department Group.

D has set Call Forward to Voice Mail on Busy or No Reply.

- Whenever there is a call for D, if the system does not detect a busy signal from D, it waits for the Call Forward No-Reply timer to expire.
- The system forwards the call to the Voice Mail System.

E has set Call Forward Dual Ring on extension F.

- When there is a call for E, the system rings on both E and the destination F.
- E can answer the call at E or at F.

Call Forward set by member extensions in a routing group will be ignored by the system if, the Ignore call forward set by member extension, when call is routed on Routing/Dept. Group flag is enabled. See "System Parameters" for more information.

When an incoming call is routed to the Routing Group and if any member has set Call Forward to the Department Group, then only the first extension of that Department Group will ring.

Call Forward when set/canceled from the SA mode, will not depend on the assigned CoS.

Feature Interaction:

- **Do Not Disturb (DND):**

  When DND or DND with Intercept Destination is set along with Call Forward-Unconditional on an extension, Call Forward is given priority.

  If any other type of Call Forward and DND are set on an extension, DND is given priority. However, DND with Intercept Destination will not work.

  If an extension has set both Call Forward and DND, then Feature Tone will be played to the extension user.

- *You can select the types of calls, that is, internal calls only, or trunk calls, or both, to be forwarded to external numbers. You can program the system to forward internal calls only, or trunk calls only or both trunk calls and internal calls, to the external number. For this, the parameter 'Allow External Call Forward for' must be programmed in the "Station Advanced Feature Template" of the extensions that are allowed Call Forward.*

- *The system supports only single-point Call Forward, which means, if the destination extension is also forwarded, the call will not follow the forwarding path. For example: Calls for extension A are set to be forwarded to extension B. Call Forward is also set on extension B with C as the destination number. Calls for A will land on B only and calls for B will land on C only.*

- *Only one Call Forward Type can be set from an extension. Every new Call Forward Type set overrides the previous one.*

- *When the calls are forwarded the extension user gets the feature tone on lifting the handset to indicate that Call Forward is set on his/her extension.*

How to configure

The functioning of this feature is controlled by three parameters: 'Class of Service' and 'Call Forward No-Reply Timer' and ‘Allow External Call Forward for’.

Call Forward must be enabled in the Class of Service (COS) group of the extensions to which this feature is to be allowed.

When Call Forward No-Reply is set, if required the Call Forward No-Reply Timer needs to be programmed.

You may select the types of calls, that is, internal, external, both internal and external calls to be forwarded by programming the ‘Allow External Call Forward for’ parameter.

You can set Call Forward for All the Extensions from the SA mode only, see “Settings Call Forward for All Extensions using SA Jeeves”.

Call Forward in Class of Service

In the default factory settings, Station Basic Feature Template Number 01 is assigned to all the extensions of SARVAM UCS. Template 01 is assigned COS group 01 in which Call Forward is enabled. So, all extensions of SARVAM UCS can use Call Forward.

If you want to deny Call Forward to certain extensions, follow these steps:

a. Define a CoS group with Call Forward disabled.

b. Prepare a Station Basic Feature Template with this CoS group applicable in all the “Time Zones”.

c. Assign this new Template to the extensions to which Call Forward is to be denied.

Refer the topics Class of Service and Station Basic Feature Template for programming instructions.

Call Forward No Reply Timer

When using Call Forward -No Reply, each extension can set a different Time after which the incoming call on the extension should get forwarded when there is no reply from the extension. For this, the Call Forward No-Reply Timer must be programmed. By default, this timer is set to 30 seconds.

The Call Forward No-Reply Timer is to be programmed in the “Station Advanced Feature Template” applied on the extensions which are allowed Call Forward in their COS.

If you want to set this timer to the same duration for all extensions, simply set the Call Forward No-Reply Timer in the default Station Advanced Feature Template 01 which is assigned to all extensions.

If you want to set different Timer duration for different extensions, then prepare separate Station Advanced Feature Templates with the desired Timer durations and assign different Templates (with different Timer durations) to the extensions as desired.

Allow External Call Forward for

The types of calls to be forwarded to the external number may be selected in the parameter "*Allow External Call Forward for"* in the “Station Advanced Feature Template” applied on the extensions which are allowed Call Forward.

You may select from 'Internal Calls', 'Trunk Calls' and 'Internal + Trunk Calls'. By default, only trunk calls are forwarded to external numbers in the default Station Advanced Feature Template 01 which is assigned to all extensions.

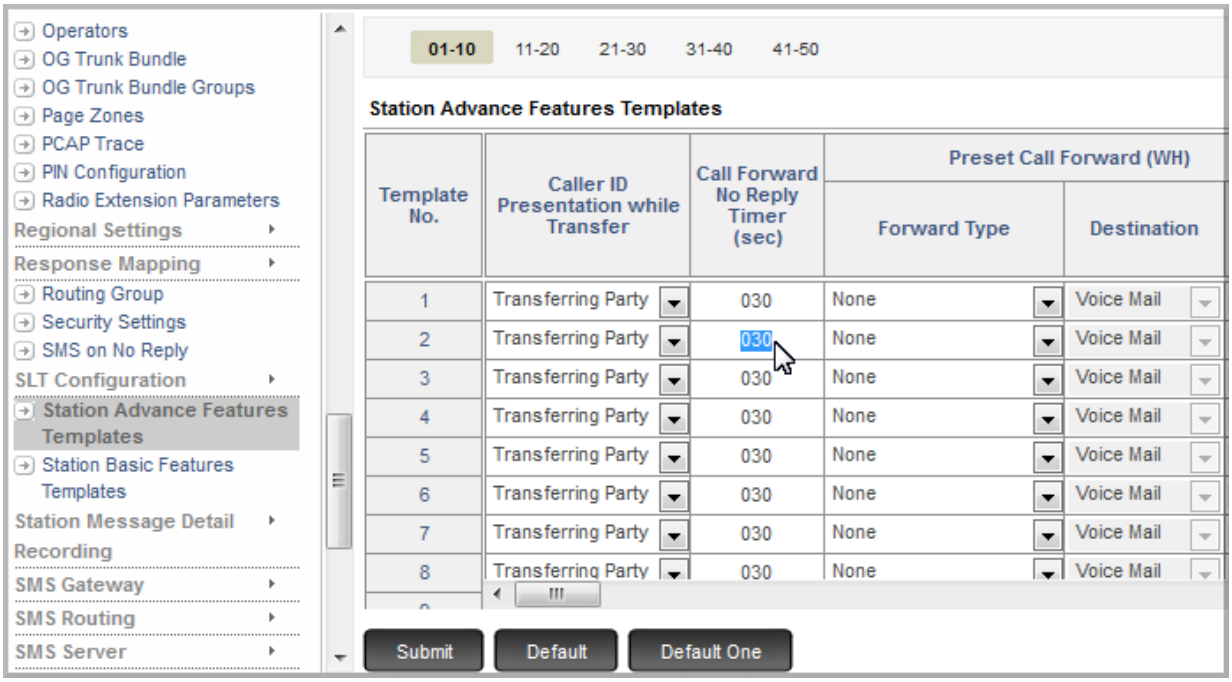
If you want to set different call types for different extensions, then prepare separate Station Advanced Feature Templates with the desired Call Types and assign these different Templates to the extensions as appropriate.

Refer the topic “Station Advanced Feature Template” for instructions on customizing the template and applying the template to extensions using Jeeves and from a Telephone.

Changing Call Forward No-Reply Timer using Jeeves

- Log in as System Engineer.
- Under **Configuration**, click **Station Advanced Feature Template** to open the page.

- Select an Advanced Feature Template number. (by default Template 01 is assigned to all extensions)



- Go to the column **Call Forward No-Reply Timer (Sec.)**.
- Change to the desired value.
- Click **Submit** at the bottom of the page to save changes.
- Apply the Template now configured with the Call Forward No-Reply Timer to the extensions.

Changing Call Forward No-Reply Timer using a Telephone

- Enter SE mode from a DKP/SLT.

- Dial command ***5602-1-Station Advanced Feature Template Number-02-Call Forward No-Reply Timer***
  Where,
  Station Advanced Feature Template is from 01 to 50. Default: 50.
  Timer is from 001 to 255 seconds.
  02 is the parameter number for "Call Forward No-Reply Timer" in the Template.

  For example: To program Call Forward No-Reply Timer as '60' secs.' in Template number 02, dial ***5602-1-02-02-060***

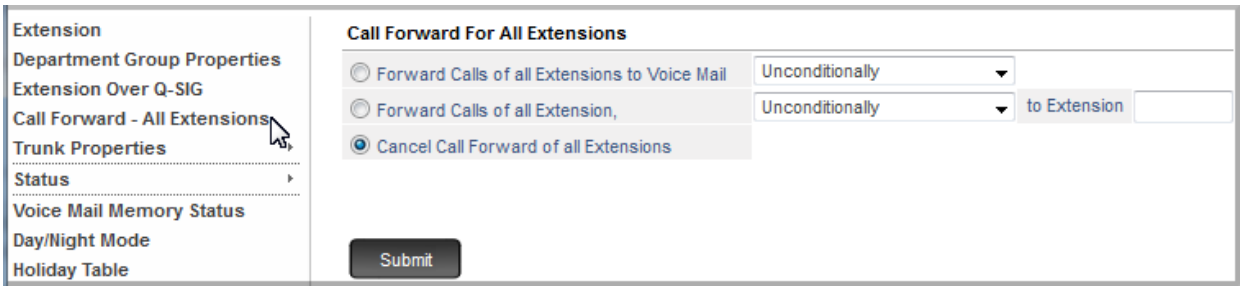
- Exit SE mode.

Refer the topic "Station Advanced Feature Template" for instructions on applying the template to extensions using Jeeves and from a Telephone.

When Call Forward No-Reply is set on a phone that is programmed in a Trunk Landing Group, the calls will be forwarded on expiry of 'Call Forward No-Reply Timer' programmed in the routing group for this member phone. Call Forward No-Reply Timer, programmed in Station Advanced Feature Template will not be applied in this case.

Settings Call Forward for All Extensions using SA Jeeves

- Log in to Jeeves as System Administrator.
- Click **Call Forward- All Extension**.



  - To set Call Forward of all Extensions to the Voice Mail, select **Forward Calls of all Extensions to Voice Mail**. Select the type of Call Forward—Unconditionally, When Busy, When No Reply, When Busy or No Reply you want to set.
  - To set Call Forward of all Extensions to an Extension, select **Forward Calls of all Extensions to Extension** Configure the **Extension** Number to which the calls are to be forwarded. Select the type of Call Forward—Unconditionally, When Busy, When No Reply, When Busy or No Reply you want to set.
  - To Cancel Call Forward, select **Cancel Call Forward of all Extensions.**
- Click **Submit** to save the option you select.

How to use

Call Forward can be set/canceled by extension users who are allowed this feature. It can be set/canceled by an extension user for another extension (refer “Call Forward-Remote” to know more).

For EON and Extended IP Phone Users

To set Call Forward,

- Press the 'Forward' key.
  OR
- Dial 13.
- Scroll to select the desired Call Forward Type.
- Press 'Enter' key.
- Enter destination Phone Number/Voice Mail System/ Department Group Number.
- You get a confirmatory text message and confirmation tone.
- Go Idle or you get dial tone after 3 seconds.

To set Call Forward-Dual Ring,

- First, set the desired Call Forward type.
- Press the 'Forward' key.
  OR
- Dial 13.
- Scroll to select Dual Ring.

- Press 'Enter' key.
- Select Dual Ring ON
- Press ‘Enter’ key.
- You get a confirmatory text message and confirmation tone.
- Go Idle or you get dial tone after 3 seconds.

- *If the call is to be forwarded to an extension, dial the extension number.*
- *If the call is to be forwarded to an external number, dial Trunk Access Code, then the external phone number and terminate the command with #*.*
  - *For users world wide, Trunk Access Code (TAC) for dialing external numbers are: 0, 5, 61, 62, 63, 64.*
  - *For users in USA, TAC for dialing external numbers are: 9, 5, 81, 82, 83, 84.*
- *If call is to be forwarded on voice mail, dial the Access Code for the Voice Mail System. The default Access Code is 3931. Verify with the System Engineer if the default VMS Access Code has been changed and use the new code to dial the VMS.*

To cancel Call Forward,

- Press 'Forward' Key again.
  OR
- Dial 13.
- Select 'Cancel'.
- You get a confirmatory text message and confirmation tone.
- Replace Handset on the cradle or you get dial tone after 3 seconds.

For SLT Users

To set Call Forward,

- Lift the handset.
- Dial 131 for Call Forward - All Calls
- Dial 132 for Call Forward - If Busy
- Dial 133 for Call Forward -If No Reply
- Dial 134 for Call Forward -If Busy or No Reply
- Dial destination Phone Number/Voice Mail System/Department Number.
- You get confirmation tone.
- Replace handset.

To set Call Forward-Dual Ring,

- Set the desired Call Forward type.
- Dial 136-1 for Call Forward -Dual Ring

To cancel Call Forward,

- Lift the handset.
- Dial 130.
- You get confirmation tone.
- Replace the handset.

To disable Dual Ring,

- Lift the handset.
- Dial 136.
- Dial 0.
- Replace Handset.

Call Forward-Remote

What's this?

An extension user can set Call Forward for another ('remote') extension from his/her own extension. Thus, Call Forward set for an extension from another extension is called 'Call Forward-Remote'.

This feature can be used by the Operator or the Receptionist to forward the calls for the Managers and other extension users to the destinations where they will be available.

This feature is also useful in Hotels, where the Front Desk can set Call Forward for guests. Refer the SARVAM UCS Hospitality System Manual to know how this feature can be used in hotels.

- *Call Forward-Remote is possible only from the System Administration (SA) mode.*
- *Call Forward-Remote is not supported by Q-Signaling (QSIG).*

How it works

This feature works in the same way as Call Forward. The only difference is that it is set by one extension user for another extension.

For example:

- A and B are extension users.
- A needs to forward calls for B's extension to another extension 'C' or an external number or a Voice Mail System or a Department Number.
- A dials the Call Forward-Remote feature code followed by B's extension number, the destination number where the calls for B should land.
- The system routes all incoming calls for B to the destination number.

How to configure

As Call Forward-Remote can be invoked only from the SA mode, either the feature 'SA Mode' or 'SA Extension' must be enabled in the Class of Service of extensions that are to be allowed this feature.

It can also be set from Jeeves by the SA.

The feature 'SA Mode' requires a password to be dialed. Users must be provided a password to use this feature from their extensions. The feature 'SA Extension' allows entry into SA mode, without a password.

In the default factory settings, Station Basic Feature Template Number 01 is assigned to all the extensions of SARVAM UCS. This Template is assigned CoS group 01 by default. The default CoS group 01 has both 'SA Mode' and SA Extension disabled.

You may decide which extensions should be allowed Call Forward-Remote feature. In general practice only very few extensions are allowed this feature.

So, to allow this feature to a few extensions only:

a. Define a CoS group with either 'SA Mode' or 'SA Extension' enabled. Recall that the facility 'SA Mode' is password protected, so the extensions allowed access to this feature must also be provided an SA Password.

b. Prepare a Station Basic Feature Template with this CoS group applicable in all the "Time Zones".

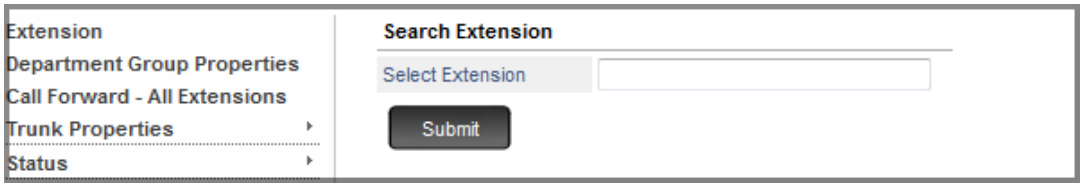
c. Assign this new Template to the extensions to which Call Forward-Remote is to be allowed.

Refer the topics "Class of Service (COS)" and "Station Basic Feature Template" for detailed instructions and programming.

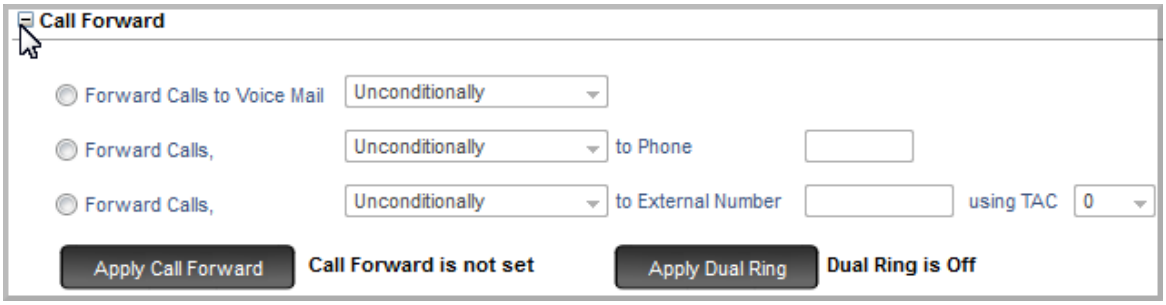
How to use

Settings Call Forward-Remote using SA Jeeves

- Log in to Jeeves as System Administrator.
- Click **Extension**.



- In **Select Extension,** enter the Number or the Name of the extension on which you want to set this feature.
- Click **Submit**.
- The searched extension users details appear on your screen.
- Click **Call Forward** to expand.



- Select the type of Call Forward you want to set for the extension:
  - To forward calls to voice mail, select **Forward Calls to Voice Mail** and the type of call forward. Default: Unconditionally.

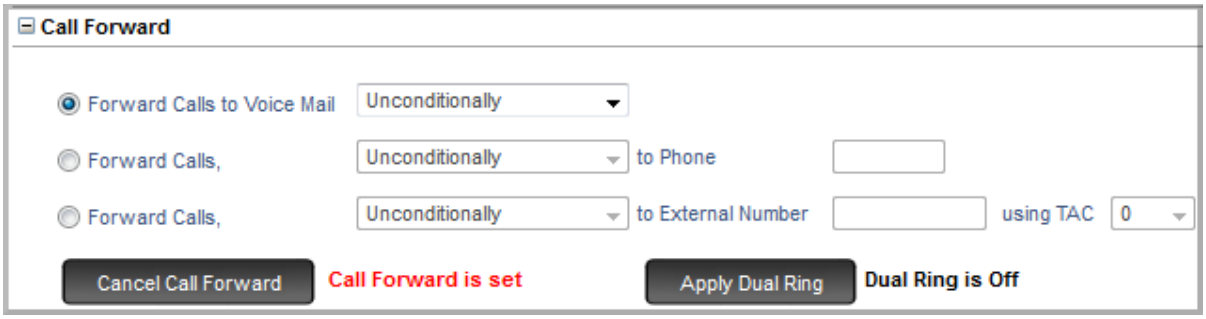
- To forward calls of this extension to another extension, select **Forward Calls to Phone** and the type of call forward. Default: Unconditionally.

  Enter the extension number to which calls must be forwarded.

- To forward calls of this extension to an external number, select **Forward Calls to External Number a**nd select the type of call forward. Default: Unconditionally.

  Enter the external number to which calls must be forwarded.

- Click the **Call Forward** button to set Call Forward.

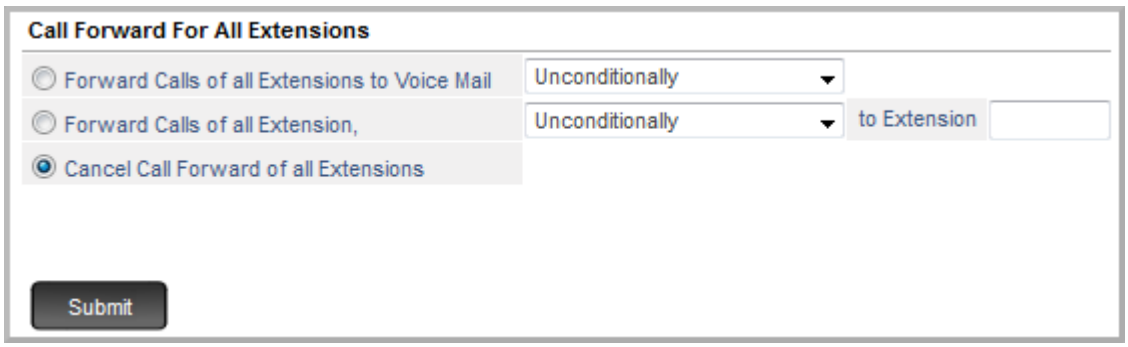


The message “Call Forward is set” appears.

- Click the **Dual Ring** button to set Call-Forward Remote with Dual Ring.

  The message “Dual ring is On” appears.

- To set Call Forward on another extension, follow the same instructions as above.
- You can also forward the calls of all extensions at one go to the same destination. To do this,
- Click **Call Forward-All Extensions**.



- Select the **Forward Calls of all Extensions** and select the Call Forward type.
- In **to Extension**, enter the Call Forward destination number.
- Click **Submit** to save.
- To cancel, click **Cancel Call Forward of All extensions**.

- Click **Submit** to save.
- You may log out of Jeeves.

For EON and Extended IP Phone Users

To set Call Forward-Remote:

- Press the DSS Key assigned to 'Call Forward-Remote'.

  OR

- Dial ***1072-006.***
- Enter the Destination Phone Number.
- Scroll to select the desired Call Forward Type:
  - All Calls.
  - If Busy.
  - If No Reply.
  - If Busy or No Reply.
  - Dual Ring.
- Press 'Enter' key.
- Enter Destination Phone Number[261]/Voice Mail System[262]/Department Group.
- You get a confirmation tone and a text message for the Call Forward type set.
- Go Idle or you get dial tone after 3 seconds.

To cancel Call Forward set for an extension:

- Press the DSS Key assigned to 'Call Forward-Remote'.

  OR

- Dial ***1072-006.***
- Enter Extension Number.
- Scroll to select 'Cancel'.
- Press 'Enter' key.
- You get a confirmation tone and text message for Call Forward canceled.
- Go Idle or you get dial tone after 3 seconds.

For SLT Users

To set Call Forward-Remote:

- Lift handset.
- Dial ***1072-006.***
- Enter Extension Number.
  - Dial 1 for All Calls
  - Dial 2 for If Busy

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261. If call is to be forwarded to an extension of the SARVAM UCS, dial the extension number. If call is to be forwarded on an external number, dial Trunk Access Code, then dial the external phone number and terminate the command with #.*

For users world wide, Trunk Access Code (TAC) for dialing external numbers are: 0, 5, 61, 62, 63, 64. For users in USA, TAC for dialing external numbers are: 9, 5, 81, 82, 83, 84.

262. If call is to be forwarded on voice mail, dial the Access Code for the Voice Mail System. The default Access Code is 3931.

    - Dial 3 for If No Reply
    - Dial 4 for If Busy or No Reply
    - Dial 5 for Dual Ring
- Dial destination Phone Number/Voice Mail System.
- You get confirmation tone.
- Replace handset.

To cancel Call Forward Remote:

- Lift the handset.
- Dial ***1072-006.***
- Dial the Extension Number.
- Dial 0.
- You get confirmation tone.
- Replace handset.

Call Forward-Scheduled

What's this?

Extension users may want their calls to be automatically forwarded to a desired destination number during working hours or non-working hours. To cite an example, a Support Technician spends working hours on the field and wants all incoming calls on his extension in the office to be forwarded to his cell phone during working hours. During non-working hours, he wants call calls to be forwarded to his voice mail.

Remembering to set and cancel Call Forward and changing the destination number for each Time Zone, that is, working hours, non-working hours, break hours, every day proves to be cumbersome for such extension users.

In addition to “Call Forward”, SARVAM UCS supports 'Call Forward - Scheduled', which allows extension users to set call forward for desired Time Zones at one time, and the system automatically forwards the calls to the destination defined for each Time Zone.

How it works

Call Forward-Scheduled supports all the forwarding options as Call Forward: Unconditionally, If Busy, If No Reply, If Busy or No Reply, Dual Ring.

Any of these options can be set for the three Time Zones: working hours, break hours and non-working hours.

The destination for Call Forward-Scheduled can be an internal (extension) number or an external number.

Both 'Call Forward' and Call Forward-Scheduled can be set on the same extension. In this case, priority is given to 'Call Forward' over Call Forward-Scheduled.

The logic for forwarding calls to the destination number remains the same as described in the topic “Call Forward”, illustrated in the following example.

- Extension user A sets Call Forward-Unconditionally to extension B for Non-working hours.
- When there is a call on extension A, the system first checks if there is any 'Call Forward' type (that is, Unconditional, Busy, No Reply, Busy/No Reply, Call Follow Me) set on extension A.
- If 'Call Forward' is set on extension A, the system will follow the logic described in 'How it works' under the topic 'Call Forward".
- If no 'Call Forward' is set on extension A, the system will check if Call Forward-Scheduled is set on A.
- Since Call Forward-Scheduled is set on extension A, the system will compare the Time Zone for which the Call Forward is scheduled with the current Time Zone of extension A.
- If the current Time Zone of extension A is the same as the Time Zone set for Call Forward Scheduled, that is, non-working hours, the call will be forwarded to extension B as per the call forward type set.
- As the Call Forward Type set by A is Unconditional, the system will forward the call to B, without checking for the Busy Tone and without waiting for the Call Forward No-Reply Timer to expire.

- If the current Time Zone of extension A is not the same as Time Zone set for Call Forward-Scheduled, the call will not be forwarded. The system will consider that no call forward has been set.

- *Call Forward - Scheduled can be set simultaneously for more than one Time Zone from the same extension. For example, extension A can set Call Forward-Scheduled for working hours, then again set Call Forward-Scheduled for non-working hours, and again for break hours.*

- *A different Call Forward Type can be set for a different Time Zone. For example, extension A can set Call Forward -Unconditional for non-working hours, and Call Forward -Busy for working hours. Also, a different destination number can be set for forwarding calls in each Time Zone. For example, extension A can set Call Forward-Unconditional for non-working hours to a mobile number and set extension B as destination number for working hours.*

- *When more than one Call Forward type is set on the same extension for the same Time Zone, the latest Call Forward type set for the Time Zone will override the previous Call Forward type set for that Time Zone. For example, extension A sets Call Forward -Busy for working hours, then sets Call Forward Busy or No Reply for working hours, the latter will override the former. The system will consider the latest, that is, Busy or No Reply as the Call Forward type for forwarding calls during working hours.*

- *Call Forward-Scheduled can be cancelled individually for a desired Time Zone or all at once for all Time Zones.*

- *Call Forward-Scheduled can be set by extension users as well as for extension users from the System Administrator mode.*

- *It is also possible to select the types of calls, that is, internal calls only, or trunk calls, or both, to be forwarded to external numbers. You can program the system to forward internal calls only, or trunk calls only or both trunk calls and internal calls to the external number. For this, the parameter 'Allow External Call Forward for' must be programmed in the "Station Advanced Feature Template" of the extensions that want to use Call Forward-Scheduled.*

- *Call Forward-Scheduled when set/canceled from the SA mode, will not depend on the assigned CoS.*

How to configure

The programming of this feature involves the same parameters as in "Call Forward".

'Call Forward' must be enabled in the Class of Service (CoS) group of the extensions to which this feature is to be allowed. Refer the topic "Call Forward".

If Call Forward No-Reply is to be set, and if required, the Call Forward No-Reply Timer may be programmed in the "Station Advanced Feature Template" applied on the extensions which are to be allowed this feature. Refer the topic "Call Forward".

The types of calls to be forwarded to the external number may be selected in the parameter "Allow External Call Forward for" in the "Station Advanced Feature Template" applied on the extensions which are allowed Call Forward-Scheduled. You may select from 'Internal Calls', 'Trunk Calls' and 'Internal + Trunk Calls'. By default, only trunk calls are forwarded to external numbers.

Extensions that are to be allowed to set Call Forward-Scheduled for other extensions must be allowed either the feature 'SA Mode' or 'SA Extension' in their COS. Refer the topic "Call Forward-Remote".

How to use

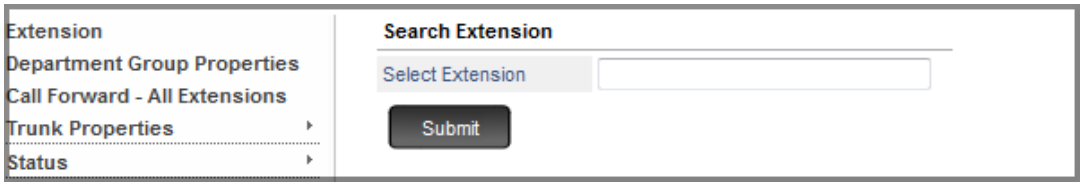
Call Forward-Scheduled can be set/canceled by users for their own extension, or for any other extension from the SA mode.

Setting Call Forward-Scheduled for Extension Users

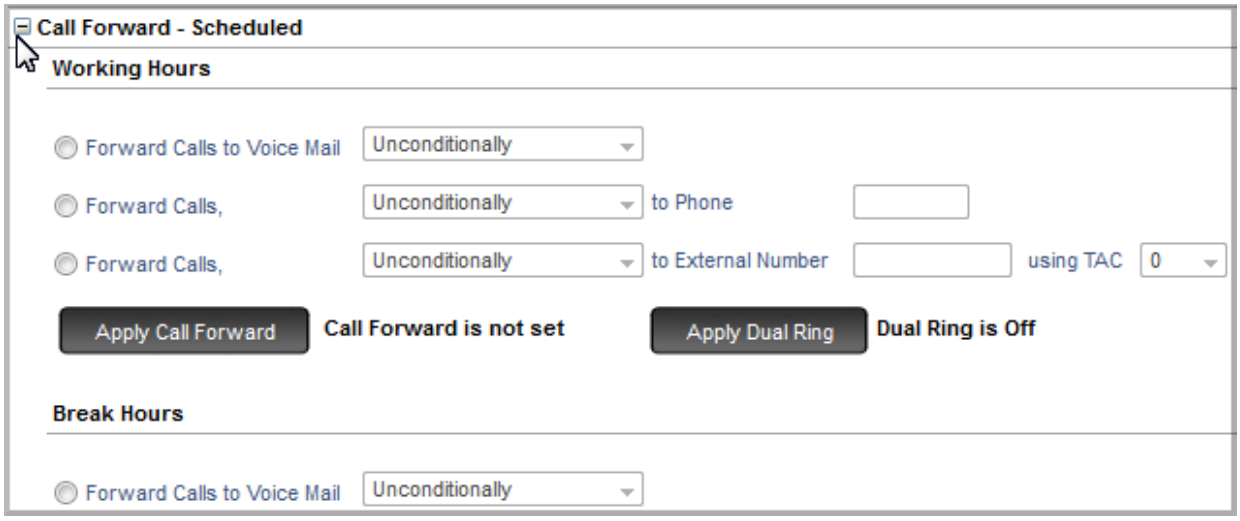
The Operator or any extension user having access to System Administrator mode can set or cancel Call Forward-Scheduled for other extension users using Jeeves or by dialing SA commands from a telephone.

Setting Call Forward-Scheduled using Jeeves

- Log in to Jeeves as System Administrator.
- Click **Extension**.



- In **Select Extension,** enter the Number or the Name of the extension on which you want to set this feature.
- Click **Submit**.
- The searched extension users details appear on your screen.
- Click **Call Forward - Scheduled** to expand.



- You can set **Call Forward - Scheduled** for Working Hours, Break Hours as well as for Non-working Hours

  To set Call Forward for the extension for the Working Hours, under **Working Hours,** select the type of Call Forward:

  - To forward calls to voice mail, select the radio button **Forward Calls to Voice Mail** and the type of call forward from the drop down list. Default: Unconditionally.

- To forward calls of this extension to another extension, select **Forward Calls to Phone** radio button and the type of call forward for this option from the drop down list. Default: Unconditionally.

  Enter the extension number to which calls must be forwarded in the empty field provided for this option.

- To forward calls of this extension to an external number, select **Forward Calls to External Number** radio button, and select the type of call forward for this option from the drop down list. Default: Unconditionally.

  Enter the external number to which calls must be forwarded in the empty field provided for this option.

- Click the **Apply Call Forward** button to set Call Forward -Scheduled.

  The message “Call Forward is set” appears.

- Click the **Dual Ring** button to set Call Forward-Scheduled with Dual Ring.

  The message “Dual ring is On” appears.

- To set call forward for Break Hours and Non-working Hours, follow the same instructions as above.

- To set Call Forward - Scheduled for another extension, follow the same instructions as above.

Setting Call Forward-Scheduled using a Telephone

- *The destination number for forwarding calls can be a maximum of 16 digits. Terminate the command with #* if destination number has fewer than 16 digits.*

- *If the destination number is an external number, enter the Trunk Access Code followed by the destination number.*

For EON and Extended IP Phone Users

Call Forward-Scheduled by Extension Users

To set Call Forward-Scheduled:

- Press DSS key assigned to Call Forward-Scheduled.
  OR
- Dial ***1175.***
- Scroll to the desired Time Zone.
- Press Enter key to select Time Zone.
- Scroll to the desired Call Forward type for the selected Time Zone.
- Press Enter key to select Call Forward type.
- Enter Destination Number on prompt.
- You get confirmation tone and message showing extension to which Call Forward is set.

To cancel Call Forward-Scheduled for a Time Zone:

- Press DSS key assigned to Call Forward-Scheduled.
  OR
- Dial ***1175.***
- Scroll to the desired Time Zone.
- Press Enter key to select Time Zone.
- Scroll to select Cancel.
- Press Enter key.
- You get confirmation tone and message.

To cancel Call Forward-Scheduled in all Time Zones:

- Press DSS key assigned to Call Forward-Scheduled.
  OR
- Dial ***1175.***
- Scroll to 'Cancel Call Forward'.
- Press Enter key.
- You get confirmation tone and message.

Call Forward-Scheduled from SA mode

To set Call Forward-Scheduled:

- Press DSS key assigned to Call Forward-Scheduled - Remote.
  OR
- Dial ***1072-223.***
- Enter extension number (from which calls are to be forwarded)
- Scroll to the desired Time Zone.
- Press Enter key to select Time Zone.
- Scroll to the desired Call Forward type for the selected Time Zone.
- Press Enter key to select Call Forward type.
- Enter Destination Number on prompt.
- You get confirmation tone and message showing extension to which Call Forward is set.

To cancel Call Forward-Scheduled for a Time Zone:

- Press DSS key assigned to Call Forward-Scheduled - Remote.
  OR
- Dial ***1072-223***
- Enter extension number (for which it is to be canceled)
- Scroll to the desired Time Zone.
- Press Enter key to select Time Zone.
- Scroll to select Cancel.
- Press Enter key.
- You get confirmation tone and message.

To cancel Call Forward-Scheduled in all Time Zones:

- Press DSS key assigned to Call Forward-Scheduled - Remote.
  OR
- Dial ***1072-223***
- Enter extension number (for which it is to be canceled)
- Scroll to 'Cancel Call Forward'.
- Press Enter key.
- You get confirmation tone and message.

For SLT Users

Call Forward (CF)-Scheduled by Extension Users

To set/cancel CF-Scheduled for working hours:

- Lift handset.
- Dial ***1175-1-1-Destination Number*** for CF-Scheduled-Unconditional.
- Dial ***1175-1-2-Destination Number*** for CF-Scheduled -Busy.
- Dial ***1175-1-3-Destination Number*** for CF-Scheduled -No Reply.
- Dial ***1175-1-4-Destination Number*** for CF-Scheduled-Busy/No Reply.
- Dial ***1175-1-5-1*** for CF-Scheduled -Dual Ring.
- Dial ***1175-1-5-0*** to cancel CF-Scheduled -Dual Ring.
- Dial ***1175-1-0*** to cancel CF-Scheduled for Working Hours.
- Replace handset.

To set/cancel CF-Scheduled for break hours:

- Lift handset.
- Dial ***1175-2-1-Destination Number*** for CF-Scheduled -Unconditional.
- Dial ***1175-2-2-Destination Number*** for CF-Scheduled -Busy.
- Dial ***1175-2-3-Destination Number*** for CF-Scheduled -No Reply.
- Dial ***1175-2-4-Destination Number*** for CF-Scheduled -Busy/No Reply.
- Dial ***1175-2-5-1*** for CF-Scheduled -Dual Ring.
- Dial ***1175-2-5-0*** to cancel CF-Scheduled -Dual Ring.
- Dial ***1175-2-0*** to cancel CF-Scheduled for Break Hours.
- Replace handset.

To set/cancel CF-Scheduled for non-working hours:

- Lift handset.
- Dial ***1175-3-1-Destination Number*** for CF-Scheduled -Unconditional.
- Dial ***1175-3-2-Destination Number*** for CF-Scheduled -Busy.
- Dial ***1175-3-3-Destination Number*** for CF-Scheduled -No Reply.
- Dial ***1175-3-4-Destination Number*** for CF-Scheduled -Busy/No Reply.
- Dial ***1175-3-5-1*** for CF-Scheduled -Dual Ring.
- Dial ***1175-3-5-0*** to cancel CF-Scheduled -Dual Ring.
- Dial ***1175-3-0*** to cancel CF-Scheduled for Non-working Hours.
- Replace handset.

To cancel CF-Scheduled for all Time Zones:

- Lift handset.
- Dial ***1175-0.***
- Replace handset.

Call Forward (CF)-Scheduled from SA mode

To set CF-Scheduled for working hours:

- Lift handset.
- Dial ***1072-223-Extension number-1-1-Destination Number*** for CF-Scheduled -Unconditional.
- Dial ***1072-223-Extension number-1-2-Destination Number*** for CF-Scheduled -Busy.
- Dial ***1072-223-Extension number-1-3-Destination Number*** for CF-Scheduled -No Reply.
- Dial ***1072-223-Extension number-1-4-Destination Number*** for CF-Scheduled -Busy/No Reply.
- Dial ***1072-223-Extension number-1-5-1*** for CF-Scheduled -Dual Ring.

- Dial ***1072-223-Extension number-1-5-0*** to cancel CF-Scheduled -Dual Ring.
- Dial ***1072-223-Extension number-1-0*** to cancel CF-Scheduled -for working hours.
- Replace handset.

To set Call Forward-Scheduled for break hours:

- Lift handset.
- Dial ***1072-223-Extension number-2-1-Destination Number*** for CF-Scheduled -Unconditional.
- Dial ***1072-223-Extension number-2-2-Destination Number*** for CF-Scheduled -Busy.
- Dial ***1072-223-Extension number-2-3-Destination Number*** for CF-Scheduled -No Reply.
- Dial ***1072-223-Extension number-2-4-Destination Number*** for CF-Scheduled -Busy/No Reply.
- Dial ***1072-223-Extension number-2-5-1*** for CF-Scheduled -Dual Ring.
- Dial ***1072-223-Extension number-2-5-0*** to cancel CF-Scheduled -Dual Ring.
- Dial ***1072-223-Extension number-2-0*** to cancel CF-Scheduled -for break hours.
- Replace handset.

To set Call Forward-Scheduled for non-working hours:

- Lift handset.
- Dial ***1072-223-Extension number-3-1-Destination Number*** for CF-Scheduled -Unconditional.
- Dial ***1072-223-Extension number-3-2-Destination Number*** for CF-Scheduled -Busy.
- Dial ***1072-223-Extension number-3-3-Destination Number*** for CF-Scheduled -No Reply.
- Dial ***1072-223-Extension number-3-4-Destination Number*** for CF-Scheduled -Busy/No Reply.
- Dial ***1072-223-Extension number-3-5-1*** for CF-Scheduled-Dual Ring.
- Dial ***1072-223-Extension number-3-5-0*** to cancel CF-Scheduled-Dual Ring.
- Dial ***1072-223-Extension number-3-0*** to cancel CF-Scheduled - for break hours.
- Replace handset.

To cancel Call Forward-Scheduled for all Time Zones:

- Lift handset.
- Dial ***1072-223-Extension Number-0***
- Replace handset.

Call Forward-When Not Registered

What's this?

SIP Phones connected as extensions may fail to register with SARVAM UCS when the network link is down or when there is power failure. Using the Call Forward-When Not Registered feature, the extension users can have their calls forwarded even when their extension phone is not registered with SARVAM UCS.

The destination for ‘Call Forward-When Not Registered’ can be an internal number, an external number or the Voice Mail.

It is also possible to select the types of calls—internal calls only, or trunk calls, or both—to be forwarded to external numbers.

Call Forward-When Not Registered can be set/canceled by,

- the System Administrator mode.
- SIP phone users from their phones.

Call Forward- When Not Registered can also be set for each Time Zone—Working Hours, Break Hours, Non-working Hours, by setting *Call Forward-When Not Registered - Scheduled*.

Call Forward - When Not Registered-Scheduled can be set for more than one Time Zone at a time on the same SIP phone. It can be canceled individually for a desired Time Zone, or all at once for all Time Zones. A different destination number can be set for forwarding calls in each Time Zone. For example, the destination number for non-working hours can be a mobile number and the destination number for working hours can be another extension number.

When the VARTA AMP100 / ADR100 application is in background, and is a member in a Routing Group or Department Group, then Call Forward functionality will not be achieved.

Feature Interaction:

- If ‘Call Forward-Unconditional’ and ‘Call Forward-When Not Registered’, have been set on the same SIP phone. ‘Call Forward-Unconditional’ will have priority over ‘Call Forward-When Not Registered’.

- If ‘Call Forward-Scheduled-Unconditional’ and ‘Call Forward-When Not Registered-Scheduled’, have been set on the same SIP phone. ‘Call Forward - Scheduled - Unconditional’ will have priority over ‘Call Forward-When Not Registered-Scheduled’.

How to configure

The Call Forward-When Not Registered feature does not require any specific programming except:

- ensuring that 'Call Forward' in the “Class of Service (COS)” group in the “Station Basic Feature Template Parameters” applied to the SIP phones.

- if required, selecting the types of calls to be forwarded to the external number. By default, only trunk calls are forwarded to external numbers. If you want to select a different type of call, configure the parameter “Allow External Call Forward for” in the *Station Advanced Feature Template* applied to the SIP phones. Refer the sub-topic “Station Advanced Feature Template”, under *Configuring Extensions*.

- If you want to allow Call Forward-When Not Registered to be set only by the System Administrator (SA) for the extension users, the System Engineer (SE) must disable 'Call Forward' feature in the Class of Service (CoS) group in the Station Basic Feature Template applied to the SIP phones.

If you disable 'Call Forward' in the CoS of a SIP phone, the user will not be able to set any other type of Call Forward.

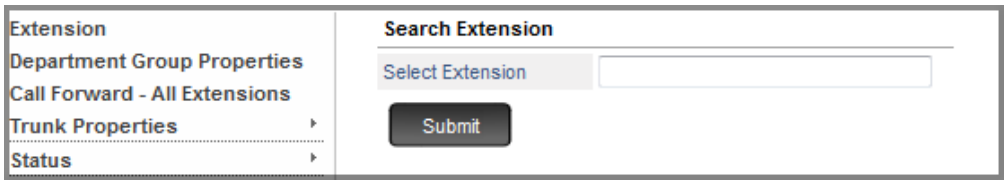
Setting Call Forward-When Not Registered

Call Forward-When Not Registered can be set from

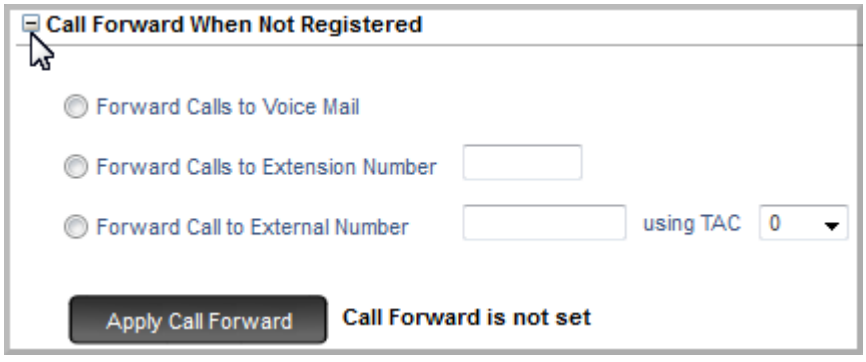
- the SA mode from Jeeves.
- the SIP phones connected as extensions.

Call Forward-When Not Registered set by SA

- Log in to Jeeves as System Administrator.
- Click **Extension**.



- In **Select Extension,** enter the Number or the Name of the extension on which you want to set this feature
- Click **Submit**.
- The searched extension user details appear on your screen.
- Click **Call Forward When Not Registered**.

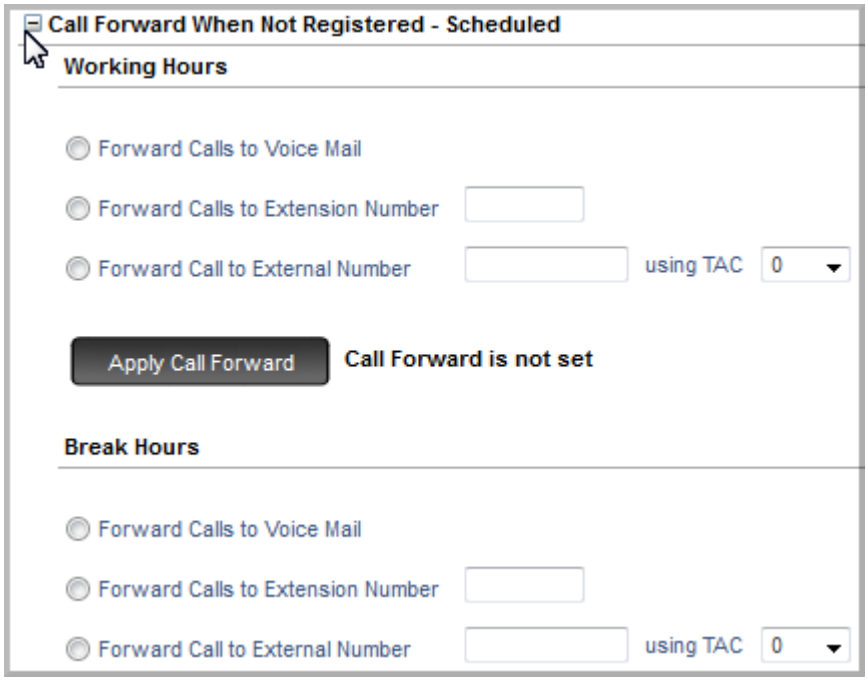


- Select the destination for forwarding calls when the SIP Extension fails to register from the following:
  - **Forward Calls to Voice Mail**.
  - **Forward Calls to Extension Number**. If you select this option, you must enter the desired Extension Number in the corresponding box.

- **Forward Calls to External Number**. If you select this option, you must enter the desired external number in the corresponding box. Also, assign a trunk to route the call by selecting the Trunk Access Code from the **using TAC** list.

- Click the **Apply Call Forward** button. The message “Call Forward is set” appears.

- To set time-zone based Call Forward - When Not Registered, click **Call Forward When Not Registered-Scheduled** to expand.



- To set Call Forward When Not Registered for working hours, under **Working Hours,** select the desired destination from the following options:

  - **Forward Calls to Voice Mail**.

  - **Forward Calls to Extension Number**. If you select this option, you must enter the desired Extension Number in the corresponding box.

  - **Forward Calls to External Number**. If you select this option, you must enter the desired number in the corresponding box, and assign a trunk to route the call by selecting the Trunk Access Code in the **using TAC** list.

- Click the **Apply Call Forward** button. The message “Call Forward is set” appears.

- To set call forward for Break Hours and Non-working Hours, follow the same instructions as above.

- To set Call Forward When Not Registered - Scheduled for another extension, follow the same instructions as above.

Call Forward-When Not Registered set/canceled by SIP Phone Users

SIP extension users can set/cancel Call Forward-When Not Registered from their SIP phones. The SIP phone may be a Matrix Extended IP Phone or any Standard SIP phone.

Using Matrix Extended IP Phone

- Lift handset.
- Press DSS key assigned to Call Forward-When Not Registered (if programmed).
  OR
- Dial ***13**.
- Scroll to the desired option.

To set Call Forward - When Not Registered regardless of time-zone,
- Select ‘Always’ and press 'Enter' key.
- Select ‘Set’ and press ‘Enter’ key.

To set Call Forward When Not Registered - Scheduled,
- Select ‘Working Hours’/’Break Hours’/’Non-Working Hours’, and press ‘Enter’ key.
- Select ‘Set’ and press ‘Enter’ key.

- On the prompt, ‘Forward to Number’, enter the Destination Number—Extension Number/External Number/ Voice Mail System.

- *The destination number for forwarding calls can be a maximum of 16 digits. Terminate the command with #* if destination number has fewer than 16 digits.*
- *If the you want to route the calls to the Voice Mail, enter the VMS Access Code as the destination number.*
- *If the destination number is an external number, enter the Trunk Access Code followed by the destination number and #*.*

- You get confirmation tone and message.

To cancel Call Forward - When Not Registered,
- Lift handset.
- Press DSS key assigned to Call Forward-When Not Registered (if programmed).
  OR
- Dial ***13**.
- Select ‘Always’ and press 'Enter' key.
- Select ‘Cancel’ and press ‘Enter’ key.

To cancel Call Forward When Not Registered - Scheduled for each Time Zone,
- Lift handset.
- Press DSS key assigned to Call Forward-When Not Registered (if programmed).
  OR
- Dial ***13**.
- Select the desired time-zone ‘Working Hours’/’Break Hours’/’Non-Working Hours’, and press ‘Enter’ key.
- Select ‘Cancel’ and press ‘Enter’ key.

To cancel All Call Forward When Not Registered,
- Press DSS key assigned to Call Forward-When Not Registered (if programmed).
  OR
- Dial ***13**.
- Select ‘Cancel Call Forward’ and press ‘Enter’ key.

Using Standard IP Phone

- Lift handset.

To set Call Forward - When Not Registered regardless of time-zone,
- Dial ****13-1-1-Destination Number***

- *The destination number for forwarding calls can be a maximum of 16 digits. Terminate the command with #* if destination number has fewer than 16 digits.*
- *If the you want to route the calls to the Voice Mail, enter the VMS Access Code as the destination number.*
- *If the destination number is an external number, enter the Trunk Access Code followed by the destination number and #*.*

To set Call Forward - When Not Registered - Scheduled,
- Dial ****13-2-1-Destination Number*** for working hours.
- Dial ****13-3-1-Destination Number*** for break hours.
- Dial ****13-4-1-Destination Number*** for non-working hours.
- Replace handset.

To cancel Call Forward - When Not Registered,
- Lift handset.
- Dial ****13-1-0***.

To cancel Call Forward When Not Registered - Scheduled,
- Dial ****13-2-0*** for working hours
- Dial ****13-3-0*** for break hours.
- Dial ****13-4-0*** for non-working hours.
- Replace handset.

To cancel All Call Forward-When Not Registered,
- Lift handset.
- Dial ****13-0.***
- Replace handset.

Call Hold

What's this?

Call Hold enables you to put an on-going conversation (with an internal or external number) on hold. SARVAM UCS offers three types of Call Hold:

- **Exclusive Hold:** An on-going conversation is put on hold from a DKP/Extended IP Phone and is retrieved from the same DKP/Extended IP Phone that put it on hold.
- **Global Hold:** An on-going conversation is put on hold from a DKP/Extended IP Phone and is retrieved from any DKP/Extended IP Phone connected to SARVAM UCS.
- **Consultation Hold:** An on-going conversation is put on hold in order to perform any further activity, such as Call Transfer, Conference, Call Toggle.

Exclusive Hold and Global Hold are supported on SIP and DKP extensions.

Consultation Hold is supported on the SIP, DKP and SLT extensions.

SARVAM UCS supports interoperability with the Polycom IP Phones. When any extension of SARVAM UCS puts a SIP Extension on hold (Exclusive, Global or Consultation Hold), SARVAM UCS will send Re-Invite message to the SIP Extension put on hold.

Call Hold is supported using RFC 2543 as well as RFC 3264. You can select the required RFC as the Call Hold Method, as per your requirement. For details, refer "SIP Hardware Template Parameters".

How it works

Exclusive Hold using the Hold Feature Key

When a call is put on Exclusive Hold,

- SARVAM UCS starts the *Exclusive Hold Retrieval Timer* (programmable; default: 2 minutes).
- The call remains on hold for the duration of this timer.
- The extension user can retrieve the call within this timer.
- If the call is not retrieved before the expiry of this timer, it will return back to the DKP/Extended IP Phone that has put the call on hold. The DKP/Extended IP Phone rings and the user may answer the call.
- The returned call is disconnected, if the DKP/Extended IP Phone is not in the idle state, or if the call is not answered by the DKP/Extended IP Phone user.

A call placed on Exclusive Hold can be retrieved in the following ways:

- Pressing the Hold key again (when the DKP/Extended IP Phone is idle).

- Pressing the Call Appearance key of the call put on hold (when the DKP/Extended IP Phone is busy).
- Pressing DSS key assigned to the Trunk/extension you put on hold.
- Answering the call, when it returns at the end of the Exclusive Hold Retrieval Timer.

To be able to place calls on Exclusive Hold, you must select **Exclusive Hold** as the **Default Call Hold Type** in the System Parameters. See “System Parameters” for instructions.

If multiple calls have been put on Exclusive Hold and you press the Hold key, then the last call that was put on hold will be retrieved.

Exclusive Hold using DSS Key

DKP and/or Extended IP Phone users can configure upto 8 DSS keys for Exclusive Hold, namely Exclusive Hold1 to 8. Using DSS key assigned to Exclusive Hold calls can be put on Exclusive Hold only. The functioning of the DSS key does not depend on the Default Call Hold Type you select.

For instructions to assign a DSS Key to Exclusive Hold, see “DSS Keys Programming”.

When a call is put on Exclusive Hold using DSS key assigned to Exclusive Hold 1,

- SARVAM UCS starts the *Exclusive Hold Retrieval Timer* (programmable; default: 2 minutes).
- The call remains on hold for the duration of this timer.
- The extension user can retrieve the call within this timer.
- If the call is not retrieved before the expiry of this timer, it will return back to the DKP/Extended IP Phone that has put the call on hold. The DKP/Extended IP Phone rings and the user may answer the call. The LCD displays the message ‘Held X Recall’, where X is the hold position number.
- The returned call is disconnected, if the DKP/Extended IP Phone is not in idle state, or if the call is not answered by DKP/Extended IP Phone user.

A call placed on Exclusive Hold can be retrieved in the following ways:

- Pressing the DSS key (when the DKP/Extended IP Phone is idle).
- Pressing the Call Appearance key of the call put on hold (when the DKP/Extended IP Phone is busy).
- Pressing DSS key assigned to the Trunk/extension you put on hold.
- Answering the call, when it returns at the end of the Exclusive Hold Retrieval Timer.

Global Hold

When a call is placed on Global Hold,

- The call remains connected in the system. The call remains on hold for the duration of the *Global Hold Retrieval Timer* (programmable; default: 60 seconds).

- Any DKP/Extended IP Phone connected to the SARVAM UCS can pick up the call put on Global hold by:
    - Pressing DSS key assigned to the Trunk put on Global Hold.
    - Pressing the DSS key assigned to the extension put on Global Hold.
- If this call is not retrieved before the expiry of the Global Hold Retrieval Timer, the call is returned to the DKP/Extended IP Phone which put it on hold. The DKP/Extended IP Phone rings and the user may answer the call.
- The returned call is disconnected, if the DKP/Extended IP Phone is not in idle state, or if the call is not answered by DKP/Extended IP Phone.

To be able to place calls on Global Hold, you must select 'Global Hold' as the Default Call Hold Type in the System Parameters of SARVAM UCS. The DKP/Extended IP Phone (which picks up the call) must have a DSS Key to access the Trunk or the Extension which is put on hold. See “System Parameters” for instructions.

SARVAM UCS provides the flexibility to use Exclusive Hold and Global Hold at the same time. You can put calls on Exclusive Hold even when Global Hold is enabled in the system using the Hold Feature key only.

SARVAM UCS does not support Global Hold on SIP Trunks.

You must first retrieve the call that is put on Exclusive or Global Hold, if it is to be transferred or included in a Conference.

Consultation Hold

During an on-going conversation, any SLT, DKP or Extended IP Phone can place a call on Consultation Hold to perform any of the following:

- “Call Transfer”
- “Call Toggle”
- “Conference-3 Party”, “Conference-Multiparty”, “Conference Dial-In”
- “Call Park”
- “Mute”
- “Call Chaining”
- “Conversation Recording”
- “Flashing on Trunks (Continued Dialing)”
- “Priority Calls in E&M MFCR2 Signaling”

The call is released from the held state once the operation has been performed or canceled.

For instructions on using the above mentioned features, refer *How to Use* of the respective feature.

How to configure

For *Exclusive and Global Hold*, you must configure the following parameters:

- **Class of Service:** Call Hold must be enabled in the Class of Service (CoS) of the DKPs/Extended IP Phones you want to allow this feature.

  In the default Station Basic Feature Template 01 assigned to all extensions of SARVAM UCS, Call Hold is included in the 'Basic Features' assigned to all Class of Service groups, including the default CoS group 01. So, all extensions of SARVAM UCS can use this feature.

Refer the topics “Class of Service (COS)” and “Station Basic Feature Template” to know more.

- **Call Hold Type:** Enable the desired option, that is, Exclusive Hold or Global Hold in the “System Parameters”.

- **Send Re-INVITE over SIP Trunk on Hold:** When an external call over a SIP Trunk is put on hold by any extension, and you want SARVAM UCS to send Re-INVITE message over SIP Trunk to the remote end, you must enable this flag on the SIP Trunk. See “Configuring SIP Trunks” to know more.

- **DSS Keys:** Program DSS Keys for Trunks and Stations on the DKPs which are allowed to retrieve calls on Global Hold. Program the DSS Keys for Exclusive Hold, if required. Refer the topic “DSS Keys Programming”, “DSS Key Settings” in “Configuring Matrix SPARSH VP330”, “DSS Key Settings” in “Configuring Matrix SPARSH VP248”, “DSS Key Settings” in “Configuring Matrix SPARSH VP310”, “DSS Key Settings” in “Configuring Matrix SPARSH VP210” and “DSS Key Settings” in “Configuring Matrix SPARSH VP510” for instructions.

- **Global Hold Retrieval Timer:** Change the default setting of this timer to the desired duration, if required.

- **Exclusive Hold Retrieval Timer:** Change the default setting of this timer to the desired duration, if required.

  For instructions to change the Timers, see “System Timers and Counts”.

For *Consultation Hold* to work, Call Hold must be enabled in the “Class of Service (COS)” of the SLTs, DKPs/ Extended IP Phones you want to allow this feature.

How to use

Exclusive Hold

For EON and Extended IP Phone Users Only

To put a call on Exclusive Hold, when Exclusive Hold is selected as the Default Call Hold Type:

- You are in speech with a Trunk/with an extension.
- Press ‘Hold’ Key.
- Go idle.
- Call with Trunk/extension is put on 'Exclusive Hold'.

To retrieve the call:

- Press ‘Hold’ Key again.
- Press Call Appearance key of your DKP/Extended IP Phone
- Press DSS Key of the Trunk/extension you put on hold from your DKP/Extended IP Phone.

To put a call on Exclusive Hold, when Global Hold is enabled:

- You are in speech with a Trunk/with an extension.
- Press ‘Hold’ Key twice in quick succession within 2 seconds.
- Go idle.
- Call with Trunk/extension is put on 'Exclusive Hold'.

To retrieve the call:

- Press ‘Hold’ Key again.
- Press Call Appearance key of your DKP/Extended IP Phone
- Press DSS Key of the Trunk/extension you put on hold from your DKP/Extended IP Phone.

To put a call on Exclusive Hold using DSS key assigned to Exclusive Hold:

- You are in speech with on a Trunk/with an extension.
- Press DSS key assigned to Exclusive Hold.
- Go idle.
- Call with Trunk/extension is put on 'Exclusive Hold'.

To retrieve the call put on hold using DSS key:

- Press DSS Key again.
- Press Call Appearance key of your DKP/Extended IP Phone
- Press DSS Key of the Trunk/extension you put on hold from your DKP/Extended IP Phone.

Using DSS key assigned to Exclusive Hold calls can be put on Exclusive Hold only. The functioning of the DSS key does not depend on the Default Call Hold Type you select in the System Parameters.

Global Hold

For EON and Extended IP Phone Users Only

To put a call on Global Hold, when Global Hold is selected as the Default Call Hold Type:

- You are in speech with a Trunk/with an extension.
- Press ‘Hold’ Key.
- Go idle.
- Call with Trunk/extension is put on 'Global Hold'.

To retrieve a call on Global Hold:

- From any DKP/Extended IP Phone, press the DSS Key of the Trunk/extension put on Global Hold.

To put a call on Global Hold, when Exclusive Hold is enabled:

- You are in speech with a Trunk/with an extension.
- Press ‘Hold’ Key twice in quick succession within 2 seconds.
- Go idle.
- Call with Trunk/extension is put on 'Global Hold'.

To retrieve a call on Global Hold:

- From any DKP/Extended IP Phone,
- Press the DSS Key of the Trunk/extension put on Global hold.

Consultation Hold

For EON and Extended IP Phone Users

To put a call on Consultation Hold:

- Press the DSS Key assigned to the feature (if configured).
  OR
- Press Transfer key.

For SLT Users

To put a call on Consultation Hold:

- Press Flash.
  OR
- Tap the Hook switch of your phone.

Call Park

What is this?

Call Park allows you to place a call on Hold, so it can be retrieved from the same or another extension of the system.

A call is 'parked' when the extension user temporarily places the call into a location in the system called 'Orbit'. The user can attend to other calls. The parked call can be retrieved on completion of the current call by dialing the Orbit number.

Call Parking is useful in offices housed in different parts of a building or multi-storied offices. It is useful in situations like:

- the person who picked up the call is not the desired called party or the desired party is at an unknown location. The person who picked up the call can then either go to find the desired called party or call other numbers to find him/her. When found, the desired called party can pick up the call from the same or any extension by dialing the Orbit number.

- the person who picked up the call may have to go to another part of the office to look up a file or consult a colleague. The person can park the call and continue the conversation from the other part of the office.

- the person who picked up the call is an Operator. The Operator needs to handle many calls on a daily basis. It becomes difficult to know the available free orbit or to remember the orbit number after parking the call. In such cases, the Operator can assign a separate DSS key for each General Orbit. The availability of the orbit will be indicated by the LED of the assigned DSS key. Thus, the Operator can now easily park or retrieve the call by pressing the DSS key.

The SARVAM UCS offers two types of Call Park facility:

- **Call Park-General Orbit:** The calls can be parked either manually or automatically in the General Orbit. The calls parked in the General Orbit can be retrieved from any extension including your own extension. General Orbit number can vary from 2 to 9.

    - **Manual Call Park:** The extension user can park calls in any of the 8 General Orbits, which are like fictional extensions located in the system. The calls parked in the General Orbit can be picked up from any extension by dialing the General Orbit Number. At a time, only one call can be parked in each General Orbit.

      The extension user can assign a separate DSS key for each General Obit, this will help the user to know the status of the Orbit - available, occupied. During an ongoing call, the user can park the call by pressing the DSS key of the available orbit depending on the LED indication. When the call is parked in the orbit, the LED blinks in blue and when it is free, the LED is off.

    - **Auto Call Park:** To park the call automatically in the free General Orbit, SARVAM supports General Call Park - Auto. When the DSS key assigned to General Call Park - Auto is pressed, the system searches for a free General Orbit (2 to 9) and automatically parks the call in the free orbit. The Orbit number is then displayed on the phone’s LCD. At a time, only one call can be parked in each General Orbit.

- **Call Park-Personal Orbit:** Each telephone instrument (EON/SLT/IP Phone) connected as extension has one Personal Orbit. Calls parked in personal orbit can be picked up only from where the call is parked. So, no other person can pick up this call. Multiple calls can be parked in the Personal Orbit at a time.

Extension users can park the call either in the General Orbit or the Personal Orbit by dialing an Orbit Number from 1 to 9, where:

- 1 is the Personal Orbit Number.
- 2 to 9 are General Orbit Numbers.

After parking a call, the extension user can continue to make and answer other calls and use other system features.

- *For Standard SIP phones, SARVAM UCS supports Call Park and Retrieve using REFER Message. For a list of IP phones on which this feature has been tested, see "SARVAM UCS Features tested on IP Phones of different Brands"in the Appendix.*
- *General Call Park - Auto feature is not supported on SLT and Standard SIP Phones.*

How it works

A and B are extension users. C, D and E are callers.

Parking Calls in General Orbit

- C calls B.
- A picks up the call.
- As B is not present at his extension, A parks the call in General Orbit Number 2 by dialing the Access Code or by pressing the DSS key of the free General Orbit.
- C is played music on-hold.
- A tries to find B (by either calling several numbers or by going in person or sending someone).
- The parked call remains in orbit for the duration of the Call Park Timer, which is set to 2 minutes by default.
- A finds B.
- B retrieves the call from another extension by dialing the feature access code for retrieving Call Park and Orbit number 2.
- If A cannot locate B or if B cannot attend the call, A can also retrieve the call from his extension.

However,

- If neither A nor B retrieves the parked call within the Call Park Timer, the system will hunt for the extension that parked the call (A) on the expiry of the Call Park Timer.
- Meanwhile, if A is busy, the system again keeps the call parked in orbit number 2 for the period of the Call Park Timer. This process continues for the duration of the Call Park Release Timer, which is set to 3 minutes by default.
- If A is free, the system will ring on A's phone. A gets connected to C again.
- If A does not retrieve the parked call till the end of the Call Park Release Timer, C gets disconnected.

Automatically Parking Calls in General Orbit

- C calls B.
- A picks up the call.
- As B is not present at his extension, A presses the DSS key assigned to General Call Park - Auto.

  OR

  Press the Transfer key and dial access code for Call Park+0.

- System internally checks for the available free General Orbit and parks the held call there.
- Once the call is parked automatically in the free orbit, a confirmation message appears on the LCD of A's phone 'Call Parked in Orbit X' (where value of X varies from 2 to 9).
- C is played music on-hold.
- A tries to find B (by either calling several numbers or by going in person or sending someone).
- The parked call remains in orbit for the duration of the Call Park Timer, which is set to 2 minutes by default.
- A finds B.
- B retrieves the call from another extension by dialing the feature access code for retrieving Call Park and Orbit number X.
- If A cannot locate B or if B cannot attend the call, A can also retrieve the call from his extension.
- If the parked call is not retrieved before the expiry of the Call Park Release Timer, C gets disconnected.

Parking Calls in Personal Orbit

Parking calls in the Personal Orbit works the same way as in General Orbit. The only difference is that A can park multiple calls by dialing the Personal Orbit Number. But calls can be retrieved from A's phone only.

When there are multiple calls to be retrieved from the Personal Orbit, they are retrieved one by one, without following any particular sequence like FIFO or LIFO.

To be able to use 'Call Park', this feature must be enabled in the COS of the requesting extension. However, for retrieving parked calls, the system does not check COS. So any extension can retrieve parked calls.

How to configure

To provide this feature to extensions,

- Enable the Call Park in the "Class of Service (COS)" of the "Station Basic Feature Template" of the extensions. By default, this feature is enabled in the CoS of all extension types for all the time zones.

- Assign a DSS key for 'General Call Park - Auto'.
  OR
  Assign a separate DSS key for each General Orbit (2-9).

  For instructions refer to "DSS Keys Programming".

- If required, you may change the duration of the Call Park Timer and the Call Park Release timer. See "System Timers and Counts" for instructions.

How to use

For EON and Extended IP Phone Users

To park a call:

- You are in speech with extension/external caller.
- Press DSS Key assigned to 'Call Park'.
  OR
  Press Transfer Key and dial 115.
- Enter Orbit Number (1-9)
  (Personal Orbit:1; General: 2-9).

To park a call manually in the available General Orbit (indicated through LED):

- You are in speech with extension/external caller.
- Press DSS Key assigned to the specific General Orbit (2-9).
  Call is parked.

To park a call automatically in the General Orbit:

- You are in speech with extension/external caller.
- Press DSS Key/CSF Key[263] assigned to ‘General Call Park - Auto[264]’
  OR
  Press Transfer Key and dial 115-0.
- The system checks for the available free General Orbit (2-9) and automatically parks the held call in it.
  The General Orbit number is displayed on the phone’s LCD.

To retrieve a parked call from your phone, when your phone is in idle state:

- Press DSS Key assigned to 'Call Park - Retrieve'.
  OR
  Dial 116
- Enter Orbit Number where you parked the call (1-9)
  (Personal Orbit:1; General: 2-9).
- You are in speech with the extension/external caller.

To retrieve a parked call from your phone, when you are in speech with someone:

- Press Transfer Key.
- Press DSS Key assigned to 'Call Park - Retrieve'.
  OR
  Dial 116
- Enter Orbit Number where you parked the call (1-9)
  (Personal Orbit:1; General: 2-9).

For SLT Users

To park a call:

263. This key is applicable only for EON 510 AND SPARSH VP510.
264. This feature is not applicable for SPARSH VP330 and SPARSH VP210.

- You are in speech with extension/external caller.
- Dial-Flash-115-Orbit Number
  (Personal Orbit:1; General: 2-9).
- Call is parked.

To retrieve a parked call from your phone, when your phone is in idle state:

- Lift the handset.
- Dial 116-Orbit Number
  (Personal Orbit:1; General: 2-9).
- You are in speech with the extension/external caller.

To retrieve a parked call from your phone, when you are in speech with someone:

- Dial Flash -116-Orbit Number
  (Personal Orbit:1; General: 2-9).
- You are in speech with the extension/external caller.

Call Logs

What's this?

SARVAM UCS stores the details of 20 each, of the following types of calls:

- **Missed calls:** incoming calls that were not answered by extension users.
- **Answered calls:** incoming calls answered by extension users.
- **Dialed calls:** calls made by extension users.

The call history of each of the above types of calls is stored by Name, Number, and Date-Time of the Call.

If there is no name in the CLI of the above types of calls, the system stores and displays the Number and the Date-Time. In case there is no number in the CLI, the system will display the Port number on/from which the call was received/made.

The Call Logs contain details of both internal as well as external calls made or received by the extension users.

The Call Logs feature is supported on EON and Extended IP Phones.

Using call logs you can:

- **view call history:** you can see the calls you missed, answered or dialed.
- **make calls:** you can call any number that you have missed, answered, or dialed.
- **edit the numbers:** you can change or modify the number in the call log. This is useful when the CLI received and stored in the call log is not in the same format that is to be used to make calls.
- **save the numbers:** you can store the external numbers in your call logs in the "Personal Directory" and use them for “Personal Abbreviated Dialing”.

The maximum number of calls that can be stored under each Call Log type is 20. The logs will be cleared automatically using the First-In, First-Out method, that is, the latest call detail will replace the record of the oldest call detail.

Given the limited Call Log capacity, the system also allows you to choose if you want internal calls to be displayed or not in the Missed, Answered and Dialed Call Logs. And accordingly it will store internal calls in the logs.

The system stores each Missed, Answered and Dialed call individually even if the same number is received multiple times.

How to configure

This feature does not require any specific programming, except:

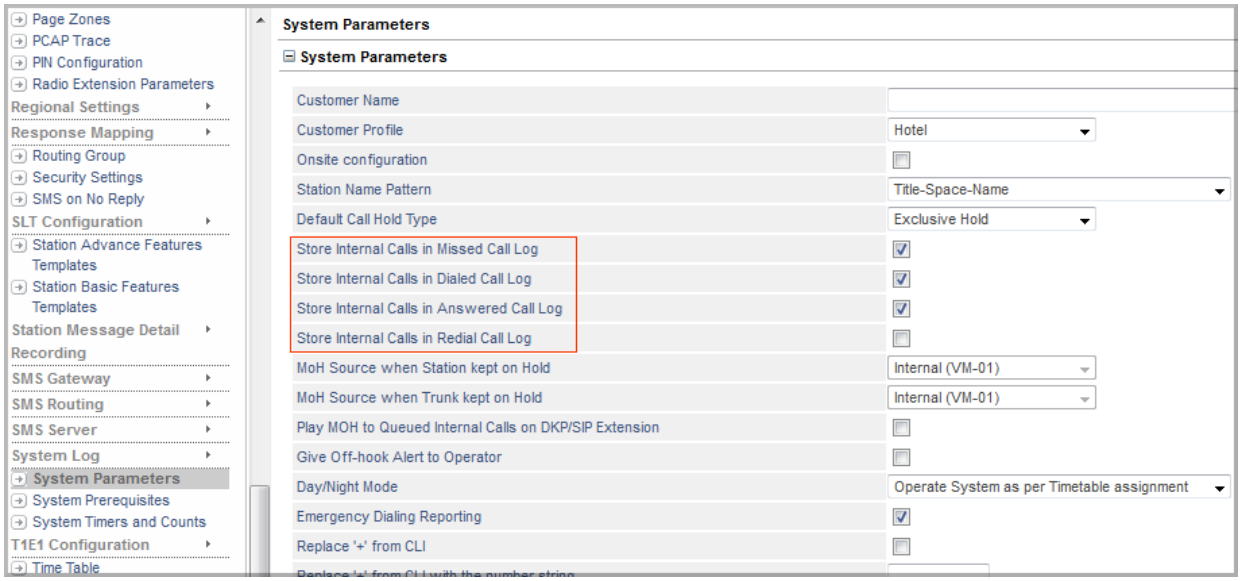
- Selecting whether internal calls should be logged in the Missed, Answered and Dialed Call Logs. This can be done on the 'System Parameters' page of Jeeves or by using a Telephone.

- Programming of a DSS key for the Call Logs feature. For instructions please refer the topic “Configuring DKP Extensions”, “DSS Keys Programming”, “DSS Key Settings” in “Configuring Matrix SPARSH VP330”, “DSS Key Settings” in “Configuring Matrix SPARSH VP248”, “DSS Key Settings” in “Configuring Matrix

SPARSH VP310", "DSS Key Settings" in "Configuring Matrix SPARSH VP210" and "DSS Key Settings" in "Configuring Matrix SPARSH VP510" for instructions.

Programming Internal Call Logging using Jeeves

- Login as System Engineer.
- Under **Configuration**, click the **System Parameters** to open the page.



- You may enable any or all of the following flags by selecting the respective check box:
  - Store Internal Calls in Missed Call Log
  - Store Internal Calls in Answered Call Log
  - Store Internal Calls in Dialed Call Log
  - Store Internal Calls in Redial Call Log
- Click **Submit** at the bottom of the page to save your settings.
- Log out of Jeeves or continue, as required.

Programming Internal Call Logging using a Telephone

- Enter SE mode.

  To enable/disable Log Internal Calls in Missed Calls, dial:
  - ***5361-Code***

    Where,

    Code is

    0 for Disable (Do not store internal calls in "Missed Calls" log)

    1 for Enable (Store internal calls in "Missed Calls" log)

    By default, Internal Call Logs in Missed Calls is enabled.

  To enable/disable Log Internal Calls in Answered Calls, dial:
  - ***5362-Code***

    Where,

    Code is

    0 for Disable (Do not store internal calls in "Answered" log)

1 for Enable (Store internal calls in "Answered Calls" log)
By default, Internal Call Logs in Answered Calls is enabled.

To enable/disable Log Internal Calls in Dialed Calls, dial:

- ***5363-Code***
  Where,
  Code is
  0 for Disable (Do not store internal calls in "Dialed Calls" log)
  1 for Enable (Store internal calls in "Dialed Calls" log)
  By default, Internal Call Logs in Dialed Calls is enabled.

- Exit SE mode.

How to use

The Call Logs feature allows you to view calls and edit numbers, make calls to any number logged, and store numbers.

For EON & Extended IP Phone Users

Viewing Call Logs

There are two ways to view call logs:

- From the Phone Menu
- Using the Feature Key assigned to Call Logs.

If you are using a DSS Key for the Call Logs feature, whenever there is a missed call, the LED of the DSS key will glow. If you press the Call Logs key, the system will display the last missed call details.

To view Call Logs from Phone Menu:

- Press Enter key when the phone is idle.
- Place your cursor on Call Logs option, press Enter key.
- Scroll with the Up/Down Navigation Key to reach the desired Call Log: Missed, Answered, Dialed.
- Press Enter key to select the desired Call Log.
- The phone will display the call log details.
- To view another call log, scroll with the Back Navigation key to return to the previous option.
- You may exit the Phone Menu by going ON-Hook or pressing the Cancel key.

- *If there is no name in the CLI, the Call Log will only display the number.*
- *If you press the 'Enter' key, the system will dial out the number you just viewed.*

To view Call Logs using DSS Key:

- Press DSS Key programmed for Call Logs, when the phone is idle.
- Scroll to select the desired Call Log: Missed, Answered, Dialed.
- The phone will display the call log details.
- To view another call log, scroll with the Back Navigation key to return to the previous option.
- You may exit the Phone Menu by going ON-Hook or pressing the Cancel key.

OR

- Press the DSS Key assigned the Call Logs feature, when it glows.
- The Phone will display the Call Logs: Missed, Answered, Dialed.
- Press Enter key to select the Missed Call Log.
- The phone will display the call log details.
- To view another call log, scroll with the Back Navigation key to return to the previous option.
- You may exit the Phone Menu by going ON-Hook or pressing the Cancel key.
- The LED of the Call Logs DSS key will be turned off once you have viewed the missed call.

Editing Numbers in Call Logs

- Go to Call Logs from the Phone Menu or by pressing Call Logs DSS Key. (see instructions given above).
- Scroll with the Up/Down Navigation Key to reach the desired Call Log: Missed, Answered, Dialed.
- Press Enter key to select the desired Call Log.
- The phone will display the call log details.
- Scroll with the Up/Down Navigation Key to reach the desired number.
- To edit the number, move the cursor with the Forward (>) navigation key.
- Place the cursor under the digit you want to delete.
- Press 'Cancel'' key to delete a digit.
- To insert a digit, place the cursor where you want to insert the digit, and enter the digit using the dial pad. The digit will be inserted in the number string accordingly.
- Repeat the same to delete/insert another digit.
- After editing the number, you may store it in the Personal Directory or dial the edited number by pressing the Enter key.

The original number (you now changed) will remain unaffected in the Log. However, if you make a call to the new number (you changed), it will be logged in the "Dialed" call log and the Last Number Redial list.

Making calls using Call Logs

- Go to Call Logs from the Phone Menu or by pressing Call Logs DSS Key.
- Scroll with the Up/Down Navigation Key to reach the desired Call Log: Missed, Answered, Dialed.
- Press Enter key to select the desired Call Log.
- The phone will display the call log details.
- Scroll with the Up/Down Navigation Key to reach the desired number.
- Press Enter key.
- The system will dial out the selected number using the Outgoing Trunks assigned for dialing '0'.
- The dialed number will be logged in the "Dialed" call log and the Last Number Redial List.

Storing numbers of Call Logs in the Personal Memory of the Phone

To store any external call record (trunk call) from Received, Missed and Dialed call logs to Personal Directory,

- Go to Call Logs from the Phone Menu or by pressing Call Logs DSS Key.
- Scroll with the Up/Down Navigation Key to reach the desired Call Log: Missed, Answered, Dialed.
- Press Enter key to select the desired Call Log.
- The phone will display the call log details.
- Scroll with the Up/Down Navigation Key to reach the desired number.
- Edit the number (following instructions given above), if required.
- Press 'v' (Down Navigation key).
- You will get the prompt: "Enter Name"[265].
- Enter the name of the contact.
- Press Enter key.

265. Only if there is a free Memory Index in the Personal Directory.

- You will get the confirmation message: "Stored at Memory: <XXX>".

- *When you store the number in the Personal Directory, the system will automatically assign Trunk Access Code "0".*

- *If all 25 Location Index Numbers of the Personal Directory are already programmed, the message "Memory Full" will appear on your phone's display and you will get an Error Tone. Refer the topic "Abbreviated Dialing" to know more.*

Call Pick Up

What's this?

Call Pick-Up allows extension users to answer calls ringing on other extensions from their own extension; without physically going to the ringing extensions.

Extension users can 'pick-up' both internal and trunk calls ringing on other extensions.

As extension users can answer calls of their colleagues or co-workers without physically going to their extensions, this feature ensures that all incoming calls are answered.

Call Pick-Up Notification will be displayed if you have SPARSH VP510. Make sure you have enabled the **Call Pick-up Notification (Only for SPARSH VP510)** in “Configuring Matrix SPARSH VP510”. For details also refer to the EON510_SPARSH VP510 V2 User Guide.

SARVAM UCS offers two types of Call Pick-Up:

- **Call Pick Up-Group** - extensions are assigned to Pick-Up Groups. Any extension in a Pick-Up Group can answer calls ringing on other extensions within the same group only.
- **Call Pick-Up Selective** - calls ringing on any extension of the system can be answered.

On SIP extensions, SARVAM UCS supports Call Pickup-Selective and Call Pickup-Group using Temporary Subscription. For a list of IP phones on which this feature has been tested, see “SARVAM UCS Features tested on IP Phones of different Brands” in the Appendix.

SARVAM UCS will send only first 3 ringing call's information in NOTIFY message to the SIP Extension, which has requested Group Call Pickup. This feature has been supported in SIP Phones of CISCO and POLYCOM.

SIP Extension which has subscribed for BLF of DKP / SIP Extension of SARVAM UCS, the SARVAM UCS will send information for the call present in the first call loop only.

How it works

Call Pick-Up Group

- Extensions must be assigned to Call Pick-Up Groups. The extensions in a Call Pick-Up group may be SLT, DKP and ISDN Terminal.
- As many as 99 such groups may be formed.
- Each group is assigned a number 01 to 99
- For example, extensions 2007, 2008, 2009, 2010. 2011, 2012, 2013 are assigned to Pick-Up Group number 03.
- When an extension in this group rings, any extension in the group can pick up the call by dialing the feature access code for “Call Pick-Up Group” (default: 4).

- The ringing extension should be in the same Pick-Up Group.

Call Pick-Up Selective

- Extensions need not be in Call Pick-Up Groups.
- Whenever an extension in the system rings, the call can be picked up by any extension of the system by dialing the feature access code and the number of ringing extension.

When more than one extension in a Pick-Up Group is ringing, you can choose which one to answer first, using Call Pick-Up Selective.

Feature Interactions:

- **Call States:** Call Pick-Up will fail if the ringing extension goes into idle state just when you are dialing the pick-up access code.
- **Auto Call Back:** Call Pick-Up will fail if the call ringing on the extension is an Auto Call Back request.
- **Alarms:** Call Pick-Up will fail if the call ringing on the extension is an Alarm Call.

How to configure

For this feature to function, **Call Pick-Up** should be enabled in the **Class of Service** of extension that are to be allowed this feature.

Call Pick-Up Groups

On a sheet of paper, list the extensions that are to be grouped into a Call Pick-Up Group. Make as many Call Pick-Up Groups as required. Assign each group a number.

Call Pick-Up Group Number	SLT Extensions	DKP Extensions	ISDN Terminals
01	2002, 2003, 2006, 2014	3003, 3004, 3005,	3201, 3201, 3203
02		3019, 3020, 3021, 3022, 3023, 3024	
03	2007, 2008, 2009, 2010. 2011, 2012, 2013		3205, 3206
:			
99			

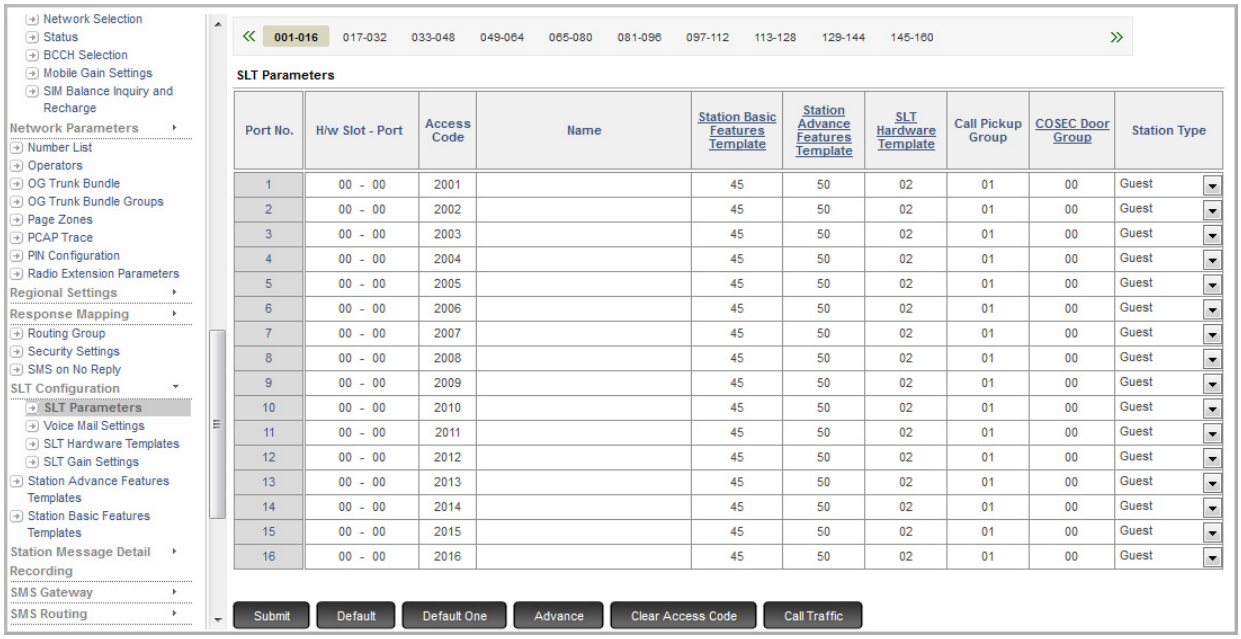
The numbering of Call Pick-Up Groups must start from 01 and end at 99.

Do not assign '00' as Call Pick-Up Group. '00' is the command to de-assign from a Call Pick-Up Group.

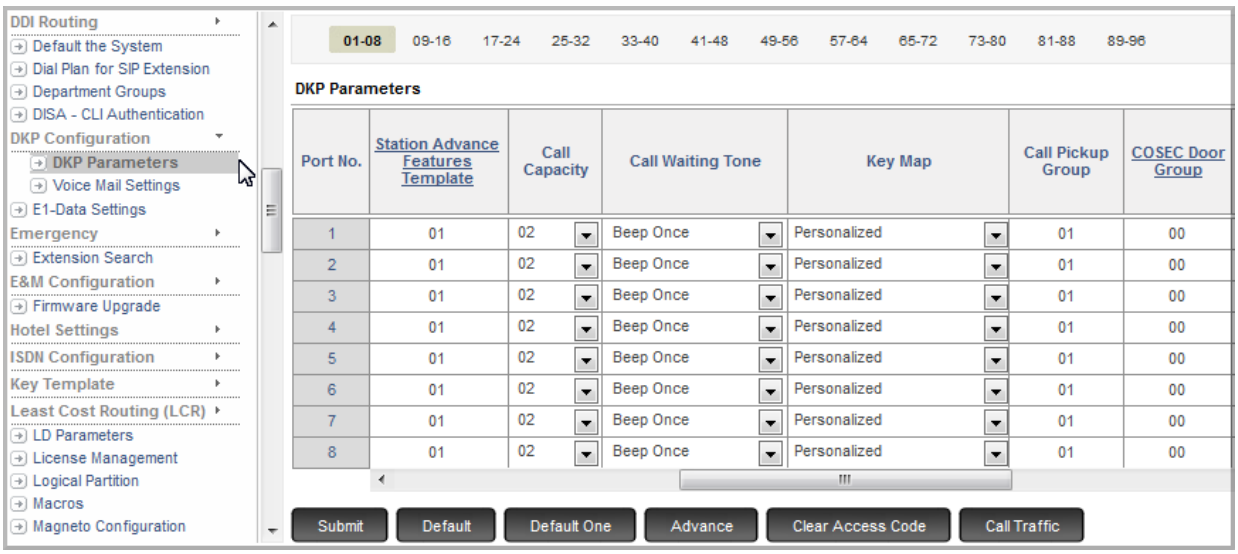
To program these groups, you may use Jeeves or issuing SE commands from a telephone.

Assigning Extensions to Call Pick-Up Groups using Jeeves

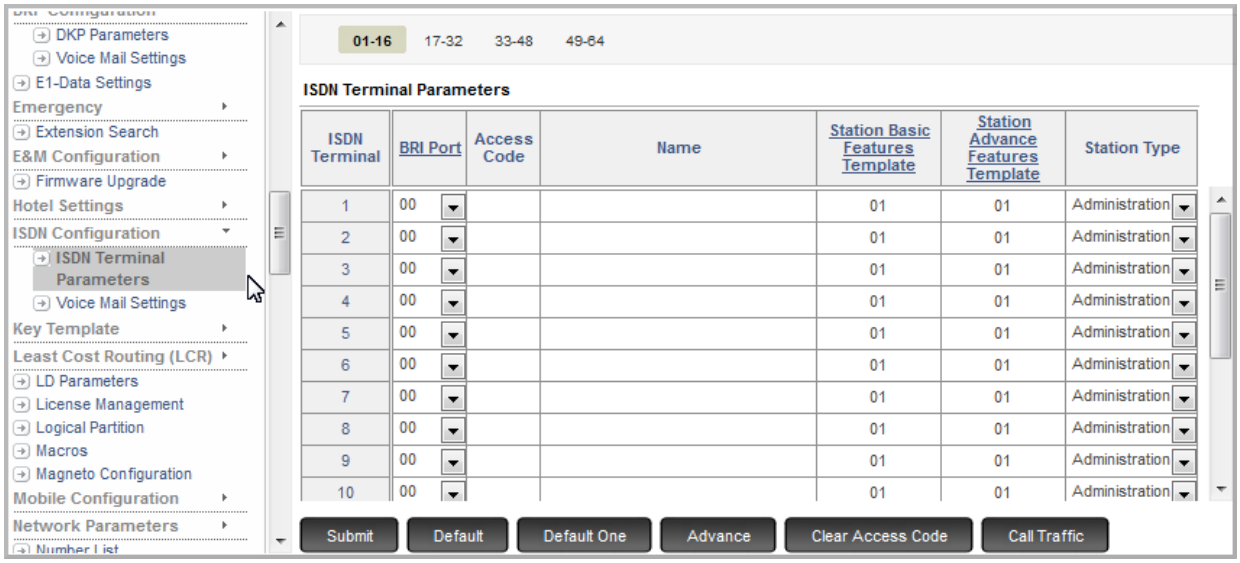
- Log in as System Engineer.
- Under **Configuration**, click **SLT Configuration**.
- Click **SLT Parameters** to open the page.



- In the column **Call Pick-up Group**, assign the group number for SLT extensions. Refer to the sheet of paper you prepared.
- Click **Submit** at the bottom of the page to save changes.
- Under **Configuration**, click **DKP Configuration**.
- Click **DKP Parameters** to open the page.



- Assign **Call Pick-Up Group** number to DKP extensions. Refer to the sheet of paper you prepared.
- Click **Submit** at the bottom of the page to save changes.
- Under **Configuration**, click **ISDN Terminal Parameters** to open the page.
- Click the **Advance** button on the bottom of the page.



- Assign **Call Pick-Up Group** numbers to ISDN Terminals. Refer to the sheet of paper you prepared.
- Click **Submit** at the bottom of the page to save changes.

Assigning Extensions to Call Pick-Up Groups using a Telephone

- Enter SE mode from DKP/SLT.

  To assign SLT extension to a group, dial:
  - ***3901-1-SLT-Call Pick-Up Group*** to assign a single SLT to a group.
  - ***3901-2-SLT-SLT-Call Pick-Up Group*** to assign a range of SLTs to the same group.
  - ***3901-*-Call Pick-Up Group*** to assign all SLTs to the same group.

    Where,

    SLT is the Software Port of the SLT, from 001 to 240.

    Call Pick-Up Group is from 00 to 99.

    By default, Call Pickup Group for SLTs is 99.

  To remove SLT extension from a group, dial:
  - ***3901-1-SLT-00*** to remove a single SLT extension.
  - ***3901-2-SLT-SLT-00*** to remove a range of SLT extensions.
  - ***3901-**** to remove all SLT extensions from a group.

  To assign DKP extension to a group, dial:
  - ***3902-1-DKP-Call Pick-Up Group*** to assign a single DKP to a group.
  - ***3902-2-DKP-DKP-Call Pick-Up Group*** to assign a range of DKPs to the same group.
  - ***3902-*-Call Pick-Up Group*** to assign all DKPs to the same group.

    Where,

    DKP is the Software Port of the DKP, from 001 to 96.

Call Pick-Up Group is from 00 to 99.
By default, Call Pickup Group for DKPs is 99.

To remove DKP extension from a group, dial:
- ***3902-1-DKP-00*** to remove a single DKP extension.
- ***3902-2-DKP-DKP-00*** to remove a range of DKP extensions.
- ***3902-**** to remove all DKP extensions from a group.

To assign an ISDN Terminal to a group, dial:
- ***3903-1-ISDN Terminal-Call Pick-Up Group*** to assign a single ISDN Terminal to a group.
- ***3903-2-ISDN Terminal-ISDN Terminal-Call Pick-Up Group*** to assign a range of ISDN Terminals to the same group.
- ***3903-*-Call Pick-Up Group*** to assign all ISDN Terminals to the same group.
  Where,
  ISDN is the Software Port of the ISDN Terminal, from 01 to 64.
  Call Pickup Group is from 00 to 99.
  By default, Call Pickup Group for ISDN Terminals is 99.

To remove an ISDN Terminal from a group, dial:
- ***3903-1-ISDN Terminal-00*** to remove a single terminal.
- ***3903-2-ISDN Terminal-ISDN Terminal-00*** to remove a range of terminals.
- ***3903-**** to remove all terminals.

- Exit SE mode.

Call Pick-Up in Class of Service

Decide which extensions are to be assigned to Call Pick-Up groups. Ensure that the feature Call Pick-Up is enabled in the Class of Service (COS) of these extensions.

In the default factory settings, Station Basic Feature Template Number 01 is assigned to all the extensions of SARVAM UCS. The Station Basic Feature Template 01 is assigned COS group 01 which has Call Pick-Up feature enabled. Thus, all the extensions of the system can use Call Pick-Up by default.

If you want to deny Call Pick-up feature to all extensions, you can simply disable Call Pick-Up in the default CoS group 01.

However, if Call Pick-Up is to be denied on only selected extensions then:

1. Define a CoS group with Call Pick-Up disabled.

2. Prepare a Station Basic Feature Template with this CoS group applicable in all the “Time Zones”.

3. Assign this new Template to the extensions to which Call Pick-Up is to be denied.

Refer the topics “Class of Service (COS)” and “Station Basic Feature Template” for detailed instructions and programming.

How to use

For EON and Extended IP Phone Users

To pick up a ringing extension in your Group:

- Press DSS Key assigned to Call Pick-Up Group.

  OR

- Dial 4.
- Talk.
- Go idle.

To pick up any one of several ringing extensions ringing or the extension that is not in your group:

- Press DSS Key assigned to Call Pick-Up Selective.

  OR

- Dial 12.
- Dial number of the Extension you want to pick up.
- Talk.
- Go idle.

For SLT Users

To pick up a ringing extension in your Group:

- Lift the handset.
- Dial 4.
- Talk.
- Replace handset.

To pick up any one of several ringing extensions ringing or the extension that is not in your group:

- Lift the handset.
- Dial 12.
- Dial number of the Extension you want to pick up.
- Talk.
- Replace handset.

Call Progress Tones

What's this?

Call Progress Tones (CPT) are audible tones sent from switching systems such as PSTN or PBX to calling parties to show the status of phone calls, like dial tone, error tone, ringing error in number dialed, ringing called party, busy line, etc.

Each CPT has a distinctive tone frequency and cadence assigned to it, for which some standards have been established by the International Telecommunication Union (ITU).

On the basis of specific frequency, modulating frequency and cadence, the CPTs generated by SARVAM UCS are categorized as:

CPT	Event	Sound	Duration	Timer
Dial Tone 1	Played on lifting the handset.	Tooooooooooo	Played for 7 seconds. After which Error Tone starts	Dial Tone Timer
Dial Tone 2	Played on lifting the handset, when 'Store and Forward Dialing[a]' is done.	Tooooooooooo	Played for 7 seconds. After which Error Tone starts	Dial Tone Timer
Ring Back Tone	Played when the internal number you have dialed is free.	Turroo... Turrroo	Played for 45 seconds	Ring Back Tone Timer
Busy Tone (Engaged Tone)	High pitch beeps with equal ON and OFF periods, played when the dialed extension is busy. Busy tone continues for 7 seconds. This Busy Tone Timer is programmable.	Toooooo..... Toooooooo	Played for 7 seconds.	Busy Tone Timer
Error Tone (Congestion/ Refusal Tone as per ITU	Fast beeps, played on a wrong operation being performed or a feature invoked without access.	Too…Too…Too …Too	Played for 30 seconds	Error Tone Timer
Internal Call Waiting Tone (Intrusion Tone as per ITU)	Short beep followed by longer OFF duration repeated every second; played to the busy extension when another extension attempts Interrupt Request/ Barge-In	Beep…… Beep	Played for duration of the Interrupt Request Timer or the Barge-In Timer.	Interrupt Request Timer, Barge-In Timer

CPT	Event	Sound	Duration	Timer
External Call Waiting Tone (Call Waiting Tone as per ITU)	Two ticks followed by a longer OFF time of approx. 3 seconds; played to a busy extension when there is a new incoming Trunk call.	Beep...Beep… …….....Beep... Beep	Played for the duration of the Transfer-On Busy Timer.	Transfer-On Busy Timer.
Confirmation Tone (Acceptance Tone as per ITU)	Continuous, fast beeps, played to confirm successful use of features.	Beep... Beep... Beep	Played for 7 seconds.	Confirmation Tone Timer
Feature Tone	Short beep followed by a longer off duration repeated every second; played when dialing feature access codes	Beep..... Beep	Played until user goes ON-Hook or dials a feature code.	
Programming Tone	Short beep followed by a longer off duration repeated every second; played to prompt entering of fresh commands during programming.	Beep..... Beep	Played until user goes ON-Hook or dials a command.	
Programming Confirmation Tone	Continuous, fast beeps; played to indicate that system has received a valid command and is processing it.	Beep... Beep... Beep	Played for 3 seconds.	Programming Confirmation Timer
Programming Error Tone	Fast beeps, played on a wrong programming command being dialed.	Too…Too…Too …Too	Played for 3 seconds.	Programming Error Tone Timer

a. In Store and Forward dialing, the digits are first stored in a memory location and then these are dialed on the trunk. For example: When Least Cost Routing (LCR) is enabled, the system will store the dialed digits first, check the trunk through which the call is to be routed and then dials the number on the appropriate trunk.

Tone standards vary with the country of application. For example, as per ITU standard, the Dial Tone for India consists of 400Hz modulated by 25Hz, whereas it is 350+440Hz, without modulation, for USA/Canada. Further, many countries use different frequencies and cadences for the same tone. For example, in the US, five different frequency and cadence are used for Dial Tone.

SARVAM UCS offers the flexibility of setting the Call Progress Tone Generation (CPTG) type to match the country-specific CPT standards established by ITU.

India being the default 'Region' for SARVAM UCS, the CTPG for India is set as default in the system.

How it works

At the time of installation, when you select the **Region** (according to the geographical location of the site where the SARVAM UCS is installed), SARVAM UCS sets the country-specific CPTG type defined for the selected Region. To see default CPTG types applicable for each region, see "CPTG Region Codes".

For countries that use different frequencies and cadences for the same tone, e.g. USA, only one frequency/cadence among the group is considered. See "Default CPTG Type" at the end of this topic.

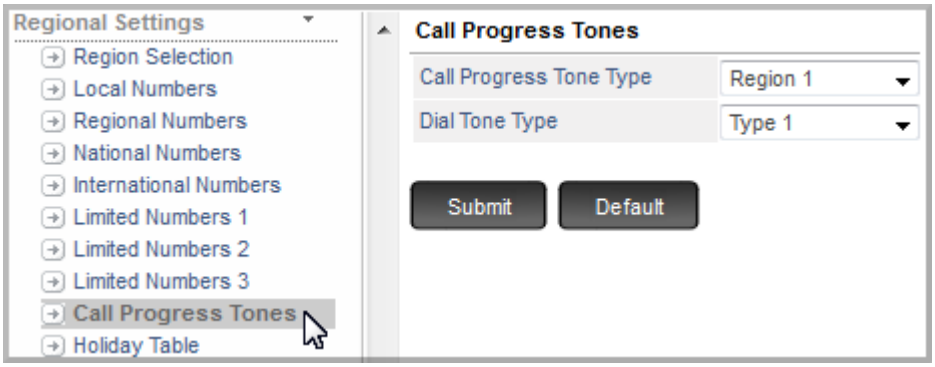
How to configure

Programming of Call Progress Tones involves configuration of three parameters: CPTG Type (Region), CPT related Timers, and Dial Tone Type.

The country-specific CPTG type is set automatically by the system when the 'Region' is selected. However, if required, the System Engineer can change the CTPG type set by the system.

Programming CPTG Parameters using Jeeves

- Log in as System Engineer.
- Under **Configuration**, click **Regional Settings**.
- Click **Call Progress Tones** to open the page.

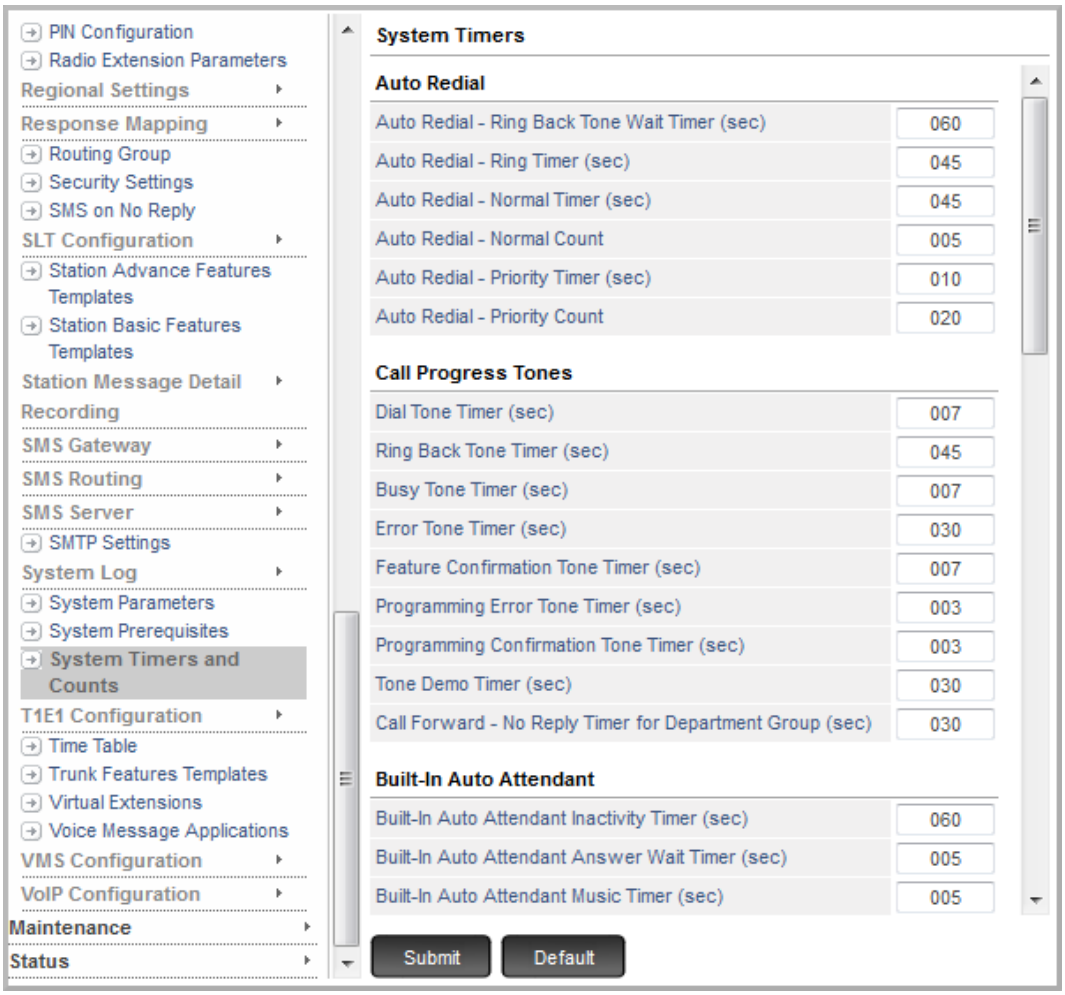


- To set the **Call Progress Tone Type**, select the desired **Region** from the list.
- Select the desired **Dial Tone Type** from Type 1 or Type 2.
- Click **Submit** to save changes.

To change CPT-related Timers

- Under **Configuration**, click **System Timers and Counts** to open the page.

- Go to **Call Progress Tones**.



- Change the values of the CPT-related Timers as desired.
- Click **Submit** to save changes.

Programming CPTG Parameters using a Telephone

- Enter SE mode from a DKP/SLT.

  To select the Region, dial:
  - ***3501-Region Code***
    For Region Code, see “CPTG Region Codes”.

  To select the Dial Tone Type, dial:
  - ***5307-Flag***
    Where,
    Flag is
    1 for Dial Tone 1
    2 for Dial Tone 2

  To change CPT-related Timers, dial:
  - ***3502-Seconds*** to program the Dial Tone Timer.
    Where,

Seconds is from 002 to 255.
Default: 007 seconds

- ***3503-Seconds*** to program the Ring Back Tone Timer.
  Where,
  Seconds is from 001 to 255 seconds.
  Default: 045 seconds.

- ***3504-Seconds*** to program the Busy Tone Timer.
  Where,
  Seconds is from 001 to 255 seconds.
  Default: 007 seconds.

- ***3505-Seconds*** to program the Error Tone Timer.
  Where,
  Seconds is from 001 to 255 seconds.
  Default: 030 seconds.

- ***3506-Seconds*** to program the Feature Confirmation Tone Timer.
  Where,
  Seconds is from 001 to 255 seconds.
  Default: 007 seconds.

- ***3509-Seconds*** to program the Programming Confirmation Tone Timer.
  Where,
  Seconds is from 001 to 255 seconds.
  Default: 003 seconds.

- ***3508-Seconds*** to program the Programming Error Tone Timer.
  Where,
  Seconds is from 001 to 255 seconds.
  Default: 003 seconds.

- ***3542-Seconds*** to change Tone Demo Timer[266].
  Where,
  Seconds is from 001 to 255.

- Exit SE mode.

For SE commands to change Interrupt Request Timer and the Barge-In Timer, Transfer-On Busy Timer, refer the relevant topics: "Interrupt Request (IR)", "Barge-In" and "Call Transfer".

How to use

It is important that users of SARVAM UCS also get acquainted with the different Call Progress Tones played by the system, so that they understand the meaning of the terms used for various tones. Therefore, SARVAM UCS makes it possible for users to listen to the various Call Progress Tones.

266. Time for which the system demonstrates the tone/ring to the user.

Demonstration of Tones

It is possible to demonstrate Call Progress Tones to users by dialing the SE commands from EON or an SLT.

By default, the system will play each tone as demonstration for 30 seconds. The duration of demonstration can be changed by setting the 'Tone Demo Timer' to match user preference (see "Changing CPT-related Timers using a Telephone" above).

To demonstrate call progress tones:

- Enter SE mode from a DKP/SLT.
- Dial command ***3541-Code.***
  Where,
  Code is
  01 Dial Tone 1
  02 Dial Tone 2
  03 Ring Back Tone
  04 Busy Tone
  05 Error Tone
  06 Confirmation Tone
  07 Feature Tone
  08 Routing Tone
  09 Programming Tone
  10 Intrusion Tone (ICWT)
  11 External Call Waiting Tone (CCWT)

- Exit SE mode.

CPTG Region Codes

CPTG Region Code	Region	Dial tone 1		Dial Tone 2		Ring Back Tone		Busy Tone	
		Freq.	Cadence (sec)	Freq.	Cadence (sec)	Freq.	Cadence (sec)	Freq.	Cadence (sec)
1	Region1	440	Continuous	350+440	Continuous	350+440	0.4on 0.2off 0.4on 2.0off	440	0.75on 0.75off
2	Region2	400	Continuous	400	Continuous	400	0.6on 0.2off 0.2on 2.0off	400	0.5on 0.5off
3	Region3	350+440	Continuous	350+440	Continuous	440+480	2.0on 4.0off	480+620	0.5on 0.5off
4	Argentina	425	Continuous	425	Continuous	425	1.0on 4.0 off	425	0.3on 0.2off
5	Australia	425*25	Continuous	425*25	Continuous	400*25	.4on .2off .4on 2.0off	425	0.375on 0.375off
6	Brazil	425	Continuous	425	Continuous	425	1.0on 4.0 off	425	0.25on 0.25off
7	Canada	350+440	Continuous	350+440	Continuous	440+480	2.0on 4.0off	480+620	0.5on 0.5off
8	China	450	Continuous	450	Continuous	450	1.0on 4.0off	450	0.35 on 0.36off
9	Egypt	425*50	Continuous	425*50	Continuous	425*50	2.0on 1.0off	425*50	1.0on 4.0off
10	France	440	Continuous	440	Continuous	440	1.5on 3.5off	440	0.5on 0.5off
11	Germany	425	Continuous	425	Continuous	425	1.0on 4.0off	425	0.48on 0.48off
12	Greece	425	0.2on 0.3off 0.7on 0.8off	425	0.2on 0.3off 0.7on 0.8off	425	1.0on 4.0off	425	0.3on 0.3off
13	India1	400*25	Continuous	400*25	Continuous	400*25	.4on .2off .4on 2.0off	400	0.75on 0.75off
14	Indonesia	425	Continuous	425	Continuous	425	1.0on 4.0off	425	0.5on 0.5off

CPTG Region Code	Region	Dial tone 1		Dial Tone 2		Ring Back Tone		Busy Tone	
		Freq.	Cadence (sec)	Freq.	Cadence (sec)	Freq.	Cadence (sec)	Freq.	Cadence (sec)
15	Iran	425	Continuous	425	Continuous	425	1.0on 4.0off	425	0.5on 0.5off
16	Iraq	400	0.4on 0.2off 0.4on 1.5off	400	0.4on 0.2off 0.4on 1.5off	400	Continuous	400	1.0on 1.0off
17	Israel	400	Continuous	400	Continuous	400	1.0on 3.0off	400	0.5on 0.5off
18	Italy1	425	Continuous	425	0.2on 0.2off 0.6on 1.0off	425	1.0on 4.0off	425	0.5on 0.5off
19	Japan	400	Continuous	400	Continuous	400*25	1.0on 2.0off	400	.5on .5off
20	Kenya	425	Continuous	425	Continuous	425	0.67on 3.0off 1.5on 5.0off	425	0.2on 0.6off 0.2on 0.6off
21	Korea	350+440	Continuous	350+440	Continuous	440+480	1.0on 2.0off	480+620	0.5on 0.5off
22	Malaysia	425	Continuous	425	Continuous	425	0.4on 0.2off 0.4on 2.0off	425	0.5on 0.5off
23	Mexico	425	Continuous	425	Continuous	425	1.0on 4.0off	425	0.25on 0.25off
24	New Zealand	400	Continuous	400	Continuous	400+450	0.4on 0.2off 0.4on 2.0off	400	0.5on 0.5off
25	Phillippines	425	Continuous	425	Continuous	425+480	1.0on 4.0off	480+620	0.5on 0.5off
26	Poland	425	Continuous	425	Continuous	425	1.0on 4.0off	425	0.5on 0.5off
27	Portugal	425	Continuous	425	Continuous	425	1.0on 5.0off	425	0.5on 0.5off
28	Russia	425	Continuous	425	Continuous	425	0.8on 3.2off	425	0.4on 0.4off
29	Saudi Arabia	425	Continuous	425	Continuous	425	1.2on 4.6off	425	0.5on 0.5off
30	Singapore	425	Continuous	425	Continuous	425*24	0.4on 0.2off 0.4on 2.0off	425	.75on.75off
31	South Africa	400*33	Continuous	400*33	Continuous	400*33	0.4on 0.2off 0.4on 2.0off	400	.5on.5off
32	Spain	425	Continuous	425	Continuous	425	1.5on 3.0off	425	0.2on 0.2off
33	Thailand	400*50	Continuous	400*50	Continuous	400	1.0on 4.0off	400	0.5on 0.5off
34	Turkey	450	Continuous	450	Continuous	450	2.0on 4.0off	450	0.5on 0.5off
35	UAE	350+440	Continuous	350+440	Continuous	400+450	0.4on 0.2off 0.4on 2.0off	400	0.375on 0.375off
36	UK	350+440	Continuous	350+440	Continuous	400+450	0.4on 0.2off 0.4on 2.0off	400	0.375on 0.375off
37	USA	350+440	Continuous	350+440	Continuous	440+480	2.0on 4.0off	480+620	0.5on 0.5off
38	Italy2	400	Continuous	400	Continuous	400	1.0on 2.0off	400	0.5on 0.5off
39	Belgium	425	Continuous	425	1.0on 0.25off	425	1.0on 3.0off	425	0.5on 0.5off
40	India2	350+440	Continuous	350+440	Continuous	350+440	0.4on 0.2off 0.4on 2.0off	400	0.75on 0.75off

CPTG Region Code	Region	Error Tone		Confirmation Tone		Feature Tone		CCWT		ICWT	
		Freq.	Cadence (sec)	Freq.	Cadence (sec)	Freq.	Cadence (sec)	Freq.	Cadence (sec)	Freq.	Cadence (sec)
1	Region1	440	0.25on 0.25 off	350+440	0.1on 0.1off	350+ 440	0.1on 0.9off	350+440	0.1on 0.1off 0.1on 2.7off	440	0.1on 2.9off
2	Region2	400	0.25on 0.25 off	400	0.1on 0.1off	400	1.5on 0.1off	400	0.2on 4.8off	400	0.2on 4.8off

CPTG Region Code	Region	Error Tone		Confirmation Tone		Feature Tone		CCWT		ICWT	
		Freq.	Cadence (sec)	Freq.	Cadence (sec)	Freq.	Cadence (sec)	Freq.	Cadence (sec)	Freq.	Cadence (sec)
3	Region3	440	0.25on 0.25 off	350+440	0.1on 0.1off	350+ 440	0.1on 0.9off	440+480	0.1on 0.1off 0.1on 2.7off	440	0.1on 2.9off
4	Argentina	425	0.3on 0.4off	425	0.1on 0.1off	425	0.1on 0.9off	425	0.3on 10.0off	425	0.1on 2.9off
5	Australia	425	0.375on 0.375off	425*25	0.1on 0.1off	425* 25	0.1on 0.9off	425	0.2on 0.2off 0.2on 4.4off	425	Continuous
6	Brazil	425	0.25on 0.25 off	425	0.1on 0.1off	425	0.1on 0.9off	425	0.05on 1.0off	425	0.1on 2.9off
7	Canada	480+ 620	0.25on 0.25off	350+440	0.1on 0.1off	350+ 440	0.1on 0.9off	440	0.3on 10.0off	480+ 620	0.5on 0.5off
8	China	450	0.7on 0.7off	450	0.1on 0.1off	450	0.1on 0.9off	450	0.4 on 4.0off	450	0.2on 0.2off 0.2on 0.6off
9	Egypt	450	0.5on 0.5off	425*50	0.1on 0.1off	425* 50	0.1on 0.9off	425*50	0.1on 0.1off 0.1on 2.7off	450	0.5on 0.5off
10	France	440	0.25on 0.25off	440	0.1on 0.1off	440	0.1on 0.9off	440	0.3on 10.0off	440	0.1on 2.9off
11	Germany	425	0.24on 0.24off	425	0.1on 0.1off	425	0.1on 0.9off	425	0.2on .2off .2on 5.0off	425	0.1on 2.9off
12	Greece	425	0.15on 0.15off	425	0.1on 0.1off	425	0.1on 0.9off	425	0.3on 10.0off 0.3on 10.0off	425	0.15on 0.25off 0.15on 1.45off
13	India1	400	0.25on 0.25off	400	1.0on 4.0off	400* 25	0.1on 0.9off	400	0.2on 0.1off 0.2on 7.5off	400	0.15on 4.85off
14	Indonesia	425	0.25on 0.25off	425	0.1on 0.1off	425	0.1on 0.9off	425	0.15on 0.15off 0.15on 10.0off	425	0.1on 2.9off
15	Iran	425	0.25on 0.25off	425	0.1on 0.1off	425	0.1on 0.9off	425	0.2on 0.2off 0.2on 10.0off	425	0.1on 2.9off
16	Iraq	400	0.25on 0.25off	400	0.1on 0.1off	400	0.1on 0.9off	400	0.1on 0.1off 0.1on 2.7off	400	0.1on 2.9off
17	Israel	400	0.25on 0.25off	400	0.17on 0.14off 0.34on 5.0off	400	0.1on 0.9off	400	0.5on 10.0off	400	0.1on 2.9off
18	Italy1	425	0.2on 0.2off	425	0.1on 0.1off	425	0.1on 0.9off	425	0.4on 0.1off 0.25on 0.1off 0.15on 5.0off	425	0.1on 2.9off

CPTG Region Code	Region	Error Tone		Confirmation Tone		Feature Tone		CCWT		ICWT	
		Freq.	Cadence (sec)	Freq.	Cadence (sec)	Freq.	Cadence (sec)	Freq.	Cadence (sec)	Freq.	Cadence (sec)
19	Japan	400	0.25on 0.25off	400	0.1on 0.1off	400	0.1on 0.9off	400*25	0.5on 2.0off 0.05on 0.45off 0.05on 3.45off	400* 25	0.1on 2.9off
20	Kenya	425	0.2on 0.6off	425	0.1on 0.1off	425	0.1on 0.9off	425	0.1on 0.1off 0.1on 2.7off	425	0.1on 2.9off
21	Korea	480+ 620	0.3on 0.2off	350+440	0.1on 0.1off	350+ 440	0.1on 0.9off	350+440	0.25on 0.25off 0.25on 3.25off	350+ 440	0.1on 2.9off
22	Malaysia	425	2.5on 0.5off	425	0.1on 0.1off	425	0.1on 0.9off	425	0.2on 0.2off 0.2on 5.0off	425	0.1on 2.9off
23	Mexico	425	0.25on 0.25off	425	0.1on 0.1off	425	0.1on 0.9off	425	0.1on 0.1off 0.1on 2.7off	425	0.1on 2.9off
24	New Zealand	400	0.25on 0.25off	400	0.1on 0.1off	400	0.1on 0.9off	400	0.2on 3.0off 0.2on 5.0off	425	0.1on 2.9off
25	Phillippines	480+ 620	0.25on 0.25off	425	0.1on 0.1off	425	0.1on 0.9off	440	0.3on 10.0off	440	0.1on 2.9off
26	Poland	425	0.5on 0.5off	425	0.1on 0.1off	425	0.1on 0.9off	425	0.15on 0.15off 0.15on 4.0off	425	0.1on 2.9off
27	Portugal	450	0.33on 1.0off	425	1.0on 0.2off	425	0.1on 0.9off	425	0.2on 0.2off 0.2on 5.0off	425	0.2on 1.4off
28	Russia	425	0.25on 0.25off	425	0.1on 0.1off	425	0.1on 0.9off	950	0.333on 1.0off	425	0.1on 2.9off
29	Saudi Arabia	425	0.25on 0.25off	425	0.1on 0.1off	425	0.1on 0.9off	425	0.15on 0.2off 0.15on 10.0off	425	0.1on 2.9off
30	Singapore	425	0.25on 0.25off	425	0.125on 0.125off	425	0.1on 0.9off	425	0.3on 0.2off 0.3on 3.2off	425	0.25on 2.0off
31	South Africa	400	0.25on 0.25off	400*33	0.1on 0.1off	400* 33	0.1on 0.9off	400*33	0.4on 4.0off	400	0.15on 0.25off 0.15on 1.45off
32	Spain	425	0.25on 0.25off	425	0.1on 0.1off	425	0.1on 0.9off	425	0.175on 0.175off 0.175on 3.5off	425	0.1on 2.9off

CPTG Region Code	Region	Error Tone		Confirmation Tone		Feature Tone		CCWT		ICWT	
		Freq.	Cadence (sec)	Freq.	Cadence (sec)	Freq.	Cadence (sec)	Freq.	Cadence (sec)	Freq.	Cadence (sec)
33	Thailand	400	0.3on 0.3off	400*50	0.1on 0.1off	400* 50	0.1on 0.9off	400	0.1on 0.1off 0.1on 2.7off	400	0.1on 2.9off
34	Turkey	450	0.2on 0.2off .6on .2off	450	0.04on 0.04off	450	0.1on 0.9off	450	.2on .6off .2on 8.0off	450	0.1on 2.9off
35	UAE	400	0.4on 0.35off 0.225on 0.525off	350+440	0.1on 0.1off	350+ 440	0.1on 0.9off	350+440	0.1on 0.1off 0.1on 2.7off	350+ 440	0.1on 2.9off
36	UK	400	0.4on 0.35off 0.225on 0.525off	350+440	0.1on 0.1off	350+ 440	0.1on 0.9off	350+440	0.1on 0.1off 0.1on 2.7off	400	0.2on 4.8off
37	USA	480+ 620	0.25on 0.25off	350+440	0.1on 0.1off	350+ 440	0.1on 0.9off	440	0.3on 10.0off	480+ 620	0.5on 0.5off
38	Italy2	400	0.25on 0.25 off	400	0.1on 0.1off	400	1.75on 0.1off	400	0.2on 2.5off	400	0.2on 0.2off 0.2on 2.5off
39	Belgium	425	0.167on 0.167 off	425	0.1on 0.1off	425	0.1on 0.9off	1400	0.175on 0.175off 0.175on 3.5off	440	0.1on 2.9off
40	India2	400	0.25on 0.25 off	350+440	0.1on 0.1off	350+ 440	0.1on 0.9off	350+440	0.1on 0.1off 0.1on 2.7off	350+ 440	0.5on 0.5off 1.0on 5.0off

Stuttered Dial Tone

Frequency: Same as Dial Tone 1 (Region wise)

Cadence: 400 ms On - 100 ms Off, 400 ms On - 100 ms Off, 400 ms On - 100 ms Off, 400 ms On - 100 ms Off, 400 ms On - 100 ms Off, 400 ms On - 100 ms Off (same for all Regions)

The meaning of frequency notation is as follows:

- ***f1*f2:*** *f1 is modulated by f2.*
- ***f1+f2:*** *The juxtaposition of two frequencies f1 and f2 without modulation.*

Default CPTG Type

Region Code	Meaning	CPTG Region Code
001	Afghanistan	
002	Algeria	
003	Antigua and Barbuda	
004	Argentina	04
005	Australia	05
006	Austria	
007	Bahamas	
008	Bahrain	
009	Bangladesh	
010	Belarus	
011	Belgium	
012	Bhutan	
013	Bolivia	
014	Bosnia	
015	Botswana	
016	Brunei	
017	Brazil	06
018	Bulgaria	
019	Cambodia	
020	Cameroon	
021	Canada	03
022	Chile	
023	China	08
024	Colombia	
025	Costa Rica	
026	Croatia	
027	Cuba	
028	Cyprus	
029	Czech Republic	
030	Denmark	
031	Egypt	09
032	Fiji	
033	Finland	
034	France	10
035	Germany	11

Region Code	Meaning	CPTG Region Code
036	Greece	12
037	Guyana	
038	Holland	
039	Hong kong	
040	Hungary	
041	India	01
042	Indonesia	14
043	Iran	15
044	Iraq	16
045	Ireland	
046	Israel	17
047	Italy	18
048	Japan	19
049	Jordan	
050	Kazakhstan	
051	Kenya	20
052	Korea-North	21
053	Korea-South	21
054	Kuwait	
055	Kyrgyzstan	
056	Lebanon	
057	Libya	
058	Malaysia	22
059	Maldives	
060	Mauritius	
061	Mexico	03
062	Mongolia	
063	Mozambique	
064	Myanmar	
065	Namibia	03
066	Nepal	
067	Netherlands	
068	New Zealand	24
069	Nigeria	
070	Norway	

Region Code	Meaning	CPTG Region Code
071	Oman	
072	Pakistan	
073	Paraguay	
074	Peru	
075	Philippines	25
076	Poland	26
077	Portugal	27
078	Qatar	
079	Romania	
080	Russia	28
081	Singapore	30
082	Slovakia	
083	South Africa	31
084	Spain	32
085	Sri Lanka	
086	Sudan	
087	Sweden	
088	Switzerland	
089	Syria	
090	Taiwan	
091	Tajikistan	
092	Thailand	33
093	Turkey	34
094	Uganda	
095	Ukraine	
096	United Arab Emirates	35
097	United Kingdom	02
098	United States	03
099	Uzbekistan	
100	Venezuela	
101	Vietnam	
102	Yemen	
103	Yugoslavia	
104	Zambia	
105	Zimbabwe	

Call Restriction based on IP Address

What's this?

When the CPU Card of SARVAM UCS is connected to a public IP network, it is may be necessary to allow traffic from particular IP address only.

With the feature 'Call Restriction based on IP Address', SARVAM UCS makes it possible to entertain requests on its LAN/WAN Ports from predefined IP Addresses only.

How it works

For this feature to work,

- the *Trusted IP Address/es* table must be configured for each SIP Trunk.
- with this table configured, incoming call traffic from all IP Addresses, other than those programmed in the Trusted IP Address/es table, will be blocked.

All incoming traffic on the SIP Trunk will be rejected if the Trusted IP Address/es Table of that SIP Trunk is blank.

How to configure

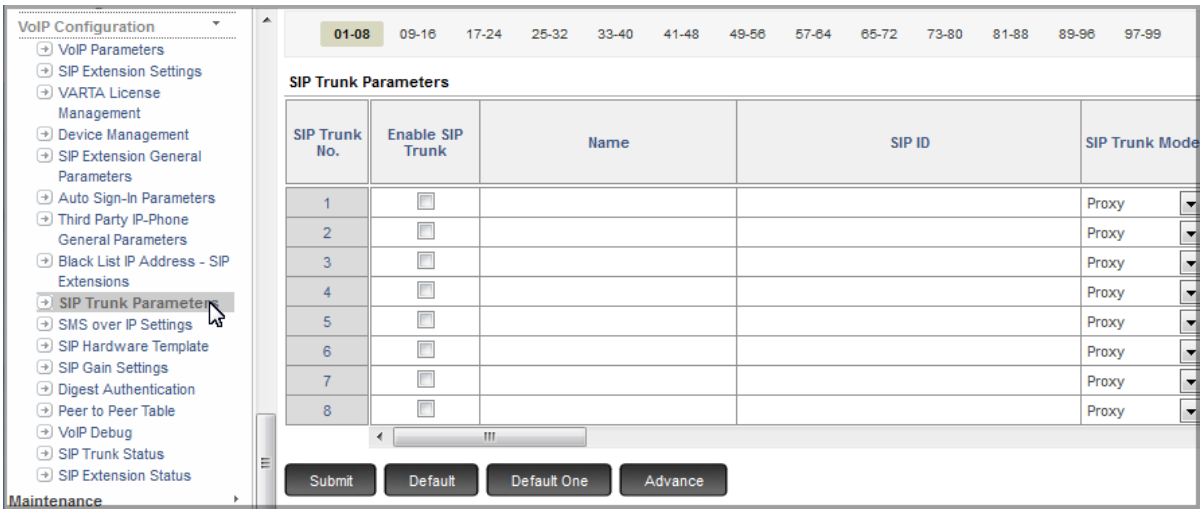
For each SIP Trunk, make a list of IP Addresses:Port from which you want to allow traffic. If you want to allow incoming calls from all ports for a particular IP Address, configure only the IP Address.

You are allowed to configure a maximum of 10 IP Addresses.

Programming IP Address Based Call Traffic Restriction using Jeeves

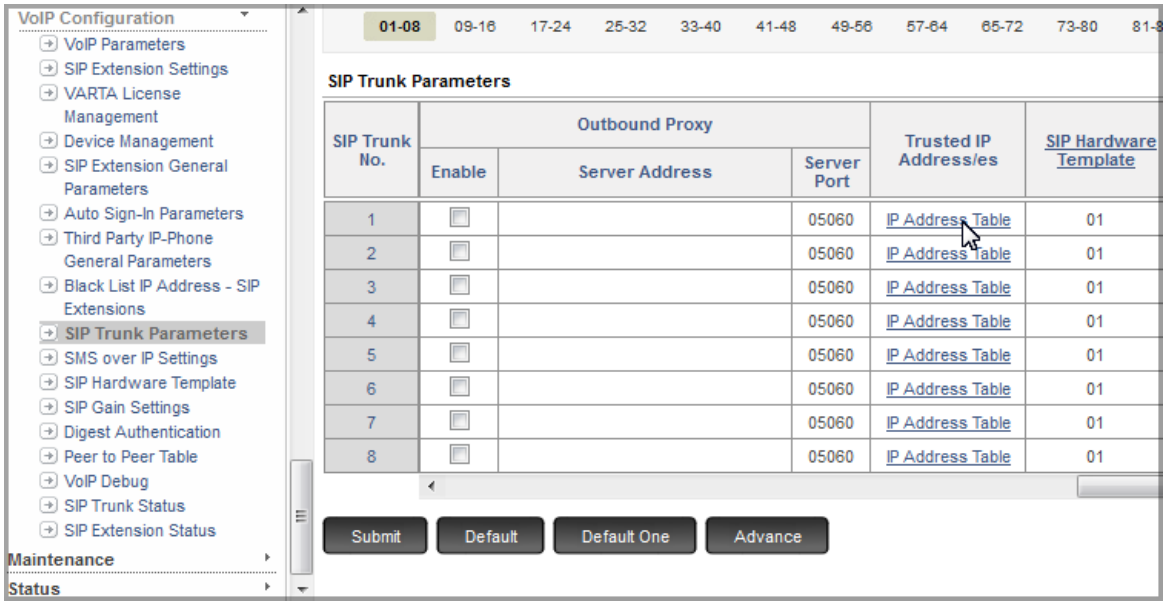
- Login to Jeeves as System Engineer.
- Under **Configuration**, click **VoIP Configuration**.

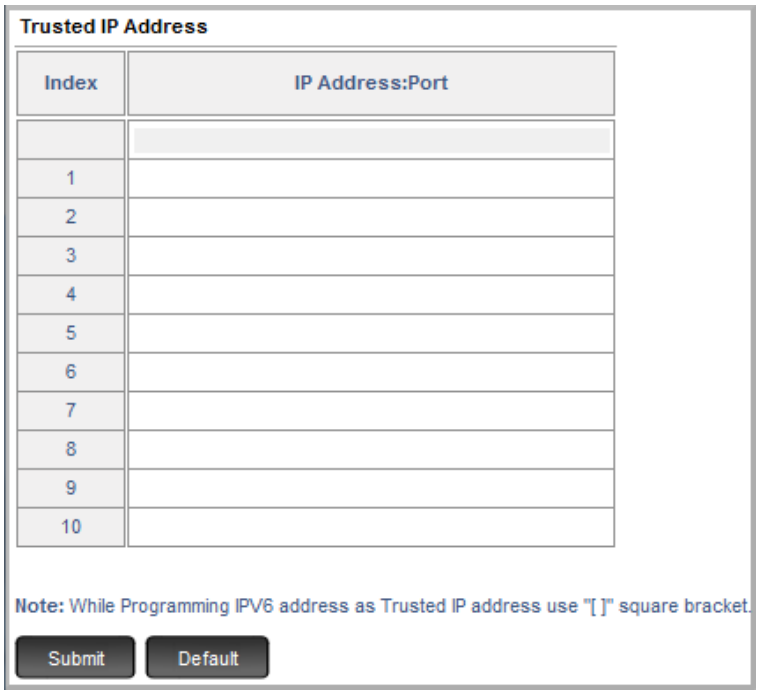
- Click **SIP Trunk Parameters**.



- Scroll to the **Trusted IP Address/es** parameter.

- Click **IP Address Table**.





The first entry in the table will display the *Proxy/Registrar Server Address:Port* or *Outbound Proxy Address: Port* as configured for the “Configuring SIP Trunks”. For the Index numbers 1 to 10,

- Enter the **IP Address** and the corresponding **Port** from which you want to allow incoming calls.

  Do not configure the port, if you want to allow incoming calls from all the ports for a particular IP Address.

- Click **Submit** to save your entries.

Call Taping

What's this?

Call Taping allows extension users to record the telephone conversations they have with other extensions or external numbers, without the opposite party coming to know about it.

Feature is useful for keeping records of important conversations. For this feature to work, make sure the VMS Module is installed.

Call Taping can be done for:

- Incoming and outgoing external calls.
- Incoming and outgoing internal calls.

SARVAM UCS supports taping of 20 calls[267] simultaneously in a 3-Party Conference. Calls can be taped either in the extension user's personal mailbox or a common mailbox assigned to this feature. The taped calls are stored along with the call details, that is, the time and date of the call, the calling number and the called number. If calls are taped in a common mailbox, only the extension users having access to the common mailbox can retrieve and listen to the recorded conversations.

To be able to record external incoming and outgoing calls,

- a list of phone numbers (both incoming and outgoing) must have been programmed in **Number List- Incoming Calls** and **Number List- Outgoing Calls** respectively, in the Station Advanced Feature Template.

- **Call Taping** must be enabled in the Trunk Feature Template assigned to the trunk used for making/ receiving external calls.

Incoming calls without Calling Line Identification (CLI) can also be taped. For this, the check box **Tape calls coming without CLI** must be enabled in the Station Advanced Feature Template assigned to the extensions.

To be able to record internal calls, the **Call Taping for Internal Calls** check box must be enabled must be enabled in the Station Advanced Feature Template assigned to the extension.

- *Use this feature in accordance with the local privacy laws.*

- *Matrix Comsec is not responsible for any mis-/abuse of this feature by users.*

How it works

A and B are extensions. Both have Call Taping parameters configured in the Station Advanced Feature Template assigned to them. Also, the trunks assigned to both the extensions for making outgoing calls have Call Taping enabled.

C and D are external parties.

E is the mailbox extension where the taped calls are recorded.

---

267. ETERNITY PENX supports 16 calls simultaneously in a 3-Party Conference.

A calls C

- The system matches the dialed number with the numbers in the **Number List - Outgoing Calls**. The system finds a match.
- When speech is established, the system starts recording the conversation between A and C automatically in E's mailbox.

- Call Taping Beeps will be played to A and C, if **Play Beep when Raid/Call Taping/Conversation Recording starts** is enabled in the System Parameters.

D calls B

- The system matches the incoming number with the numbers in the **Number List-Incoming Calls**.

- On finding a match, when speech is established, system records the speech between D and B in E's mailbox.

- Call Taping Beeps will be played to D and B only if **Play Beep when Raid/Call Taping/Conversation Recording starts** is enabled in the System Parameters.

- *If an incoming call does not have any CLI, the system checks the check box **Tape calls coming without CLI** under the Call Taping in Station Advanced Feature Template assigned to the extension.*

- *If the check box is enabled, all calls without CLI are taped.*

A calls B

- The system checks if the **Call Taping for Internal Calls** is enabled under the Call Taping in Station Advanced Feature Template assigned to A.

- If the check box is enabled, the system records the speech between A and B in E's mailbox.

- Call Taping Beeps will be played to A and B only if **Play Beep when Raid/Call Taping/Conversation Recording starts** is enabled in the System Parameters.

- If the check box is disabled, the speech between A and B will not be recorded.

- The same is done when B calls A. The speech will be recorded in E's mailbox.

To listen to the conversation, A and B must have access to the mailbox of E.

During Call Taping, if the user puts the call on Consultation Hold and access the feature — Call Transfer/ Conference/ Call Park/ Call Toggle, then in this case, the conversation after the call is put on consultation hold will not be taped. However, the system will start tapping this call again once it is transferred to the respective user.

For example, consider F and G are in a two-party speech, and during the ongoing conversation, F wants to speak to H. So, when G puts the call of F on consultation hold and speaks with H, the conversation between G and H will not be taped. Call taping will be resumed again, after the call is transfered to H, that is, the conversation between F and H will be taped.

You can save the taped conversation either in Personal Mailbox or in a Common Mailbox. If you select Personal Mailbox, the taped conversations will be saved in each extension user’s Personal Mailbox.

In you select Common Mailbox, you can save the taped conversation either in a single file or in individual files as per your requirement.

If the call is not transferred successfully and returns back to the transferor, then in this case, the system will save the taped conversation in two separate files.

Feature Interaction:

- **Conversation Recording:** If Call Taping and "Conversation Recording" both are enabled for an extension, then priority is given to Call Taping.

How to configure

The functioning of this feature requires the following parameters to be programmed:

- **Save Call Taping Files in**: Specify the location to save Call Taping files. You can select **Common Mailbox** or **Personal Mailbox**.
- **Common Mailbox for Call Taping (Enter Extension Number):** You must program the extension number of the user in whose mailbox the calls are to be taped.
- **Save Call Tapping Files as:** If you select common mailbox, select the type of file you want the system to generate for saving the tapped conversation. You may select — *Individual File before and after call transfe*r or *Single File before and after call transfer* as per your requirement**.**
- **Call Taping on Trunks:** Call Taping must be enabled in the Trunk Feature Template assigned to the Trunks. Make sure these trunks are used by extension users for making/receiving external calls.
- **Tape calls coming without CLI check box:** This check box must be enabled if you want calls without CLI to be taped.
- **Number Lists for Incoming and Outgoing Calls:** The Call Taping Number List-Incoming Calls and Call Taping Number List-Outgoing Calls are to be programmed in the Station Advanced Feature Template assigned to the extension users so that the system can match the phone numbers of the incoming and outgoing calls and initiate the recording of the speech.

  On a sheet of paper, prepare the Call Taping List Incoming and Call Taping List Outgoing.

  You can add as many as 999 numbers to each list. Each entry on these Lists is stored in a serial order against a 'Location Number'. So, draw three columns and enter the numbers against a location number from 01 to 999.

Location	Number List-Incoming Calls	Number List-Outgoing Calls
001		
002		
:		
:		
999		

  Use this table to program the Number lists. By default Number List 09 is assigned for numbers of incoming calls, and Number List 10 is assigned to numbers of outgoing calls.

- **Call Taping for Internal Calls check box:** This check box is to be enabled in the Station Advanced Feature Template applied on those extensions that are to be allowed Call Taping of internal calls, that is, calls made or received by them to or from other extensions.

- **Play Beep when Raid/Call Taping/Conversation Recording starts:** This check box is to be enabled if Call Taping Beeps are to be played to the two parties in speech. Enable Call Taping Beeps only when you want indication of speech recording to the two parties in speech. By default, this check box is enabled.

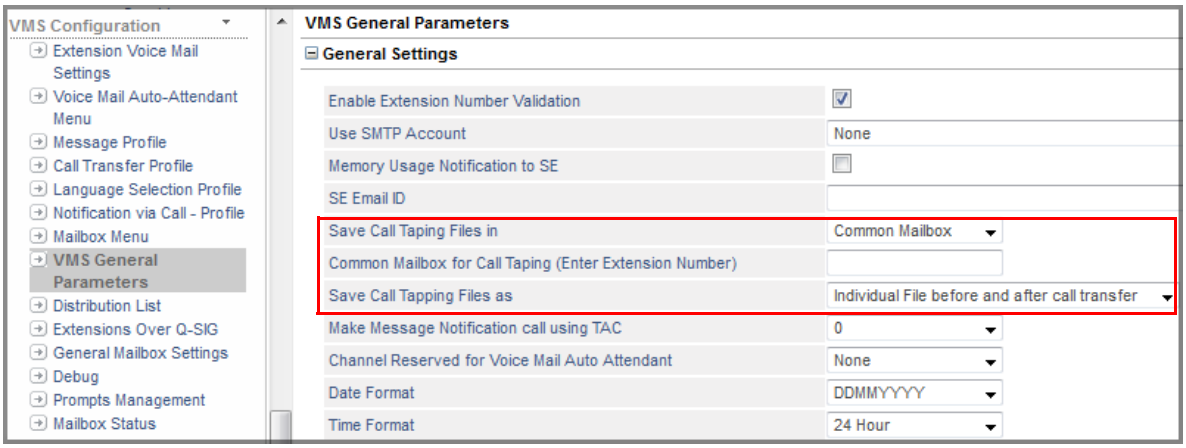
Configuring Call Taping Parameters using Jeeves

- Log in as System Engineer.

- Under **Configuration**.

For Call Taping Mailbox,

- Click **VMS General Parameters** to open the page.



- Under **General Settings**,

- Specify the location where you want to **Save Call Taping Files in**. You can save the Call Taping files either in the Common Mailbox or in your Personal Mailbox. By default, Call Taping files are saved in the Common Mailbox.

- If you choose to save Call Taping files in the Common Mailbox,

  - enter the access code of any SLT, DKP, SIP Extension, Department Group or General Mailbox, whose mailbox you want to assign for Call Taping in **Common Mailbox for Call Taping (Enter Extension Number)**.

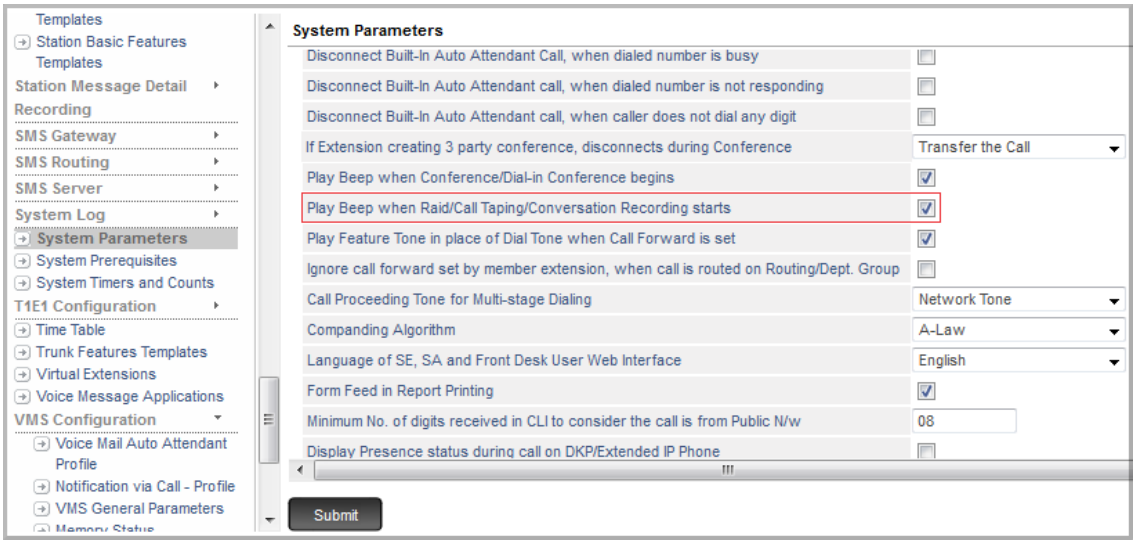
  - select the type of file you want the system to generate for saving the tapped conversation in **Save Call Tapping Files as**. You may select — *Individual File before and after call transfe*r or *Single File before and after call transfer* as per your requirement.

    If you select *Individual File before and after call transfe*r, the system will generate two separate files for saving the tapped conversation, that is, one file containing the conversation tapped before the call is transferred and another file containing the conversation tapped after the call is transferred. However, if you select *Single File before and after call transfer,* the system will generate one single file for saving the conversation tapped before and after the call is transferred.

- Click **Submit** to save changes.

To enable/disable Call Taping Beeps,

- On the same **System Parameters** page, go to **Play Beep when Call Taping/Conversation Recording Starts** and enable/disable beeps by selecting/clearing the check box.



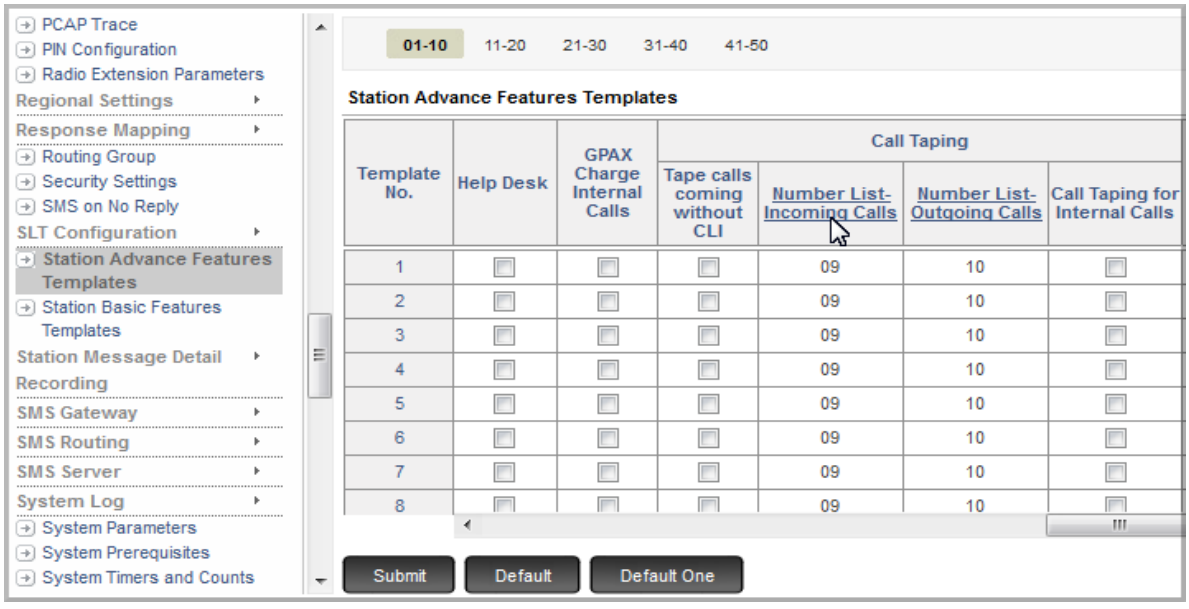
- Click **Submit** at the bottom of the page to save changes.

To enable **Call Taping on Trunks**,

- Select the Call Taping check box in the Trunk Feature Template assigned to the Trunks. For detailed instructions, see “Configuring Trunks”.

To configure the Call Taping parameters and Number Lists,

- Click **Station Advanced Feature Template** to open the page.



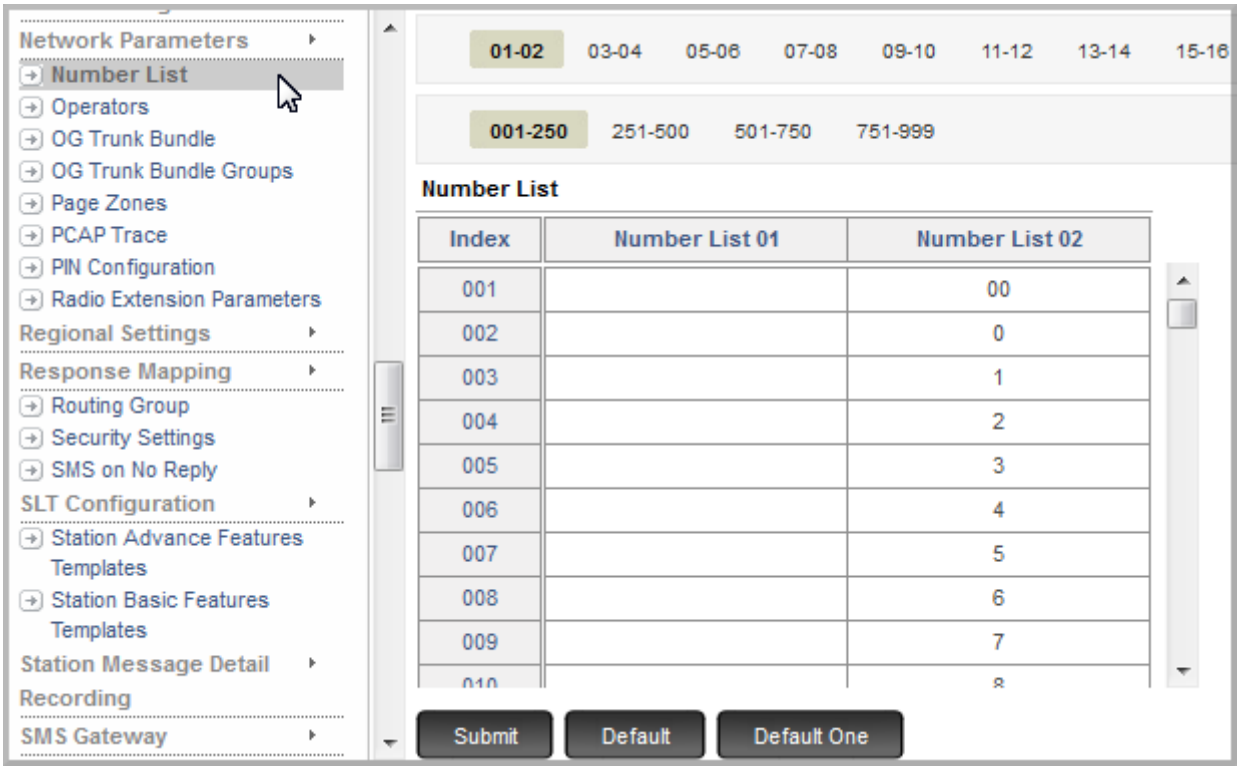
By default, Station Advanced Feature Template 01 is assigned to all extensions of the SARVAM UCS. If you want to assign Call Taping facility to all extensions, then program the Call Taping related parameters and Number Lists, in Template 01.

However, if only selected extensions are to be assigned this feature, then:

1. Prepare a separate Station Advanced Feature Template.
2. Set the Call Taping Parameters in this template.
3. Apply this new template to desired extensions that are to be allowed this feature.

- Scroll with the horizontal bar to reach the **Call Taping** column of the Template Number assigned to the extensions.
- If you want calls without CLI to be taped, select the **Tape calls coming without CLI** check box.
- To program the list of numbers of incoming calls, click the link **Number List- Incoming Calls**.

  The Number List page will open.



- Click the default number list **9-10** link assigned to Call Taping, then click the **001-255** link of the default.

  The Number Lists 9 and 10 will open.

- *If the same incoming and outgoing numbers are to be programmed for all extensions, you may simply program the default Number lists 09 and 10.*
- *If different incoming and outgoing numbers are to be programmed for different extensions, then prepare different number lists.*

- Enter the List of Incoming Numbers that the system should match in List No. 09.
- Enter the List of Outgoing Numbers that the system should match in List No. 10.
  You can program as many as 999 numbers in each list. Each entry on these Lists is stored in a serial order against a 'Location Index, starting from 001-999'. There are 250 Location Index on each page on your screen. To go to the next set of Location Index, for instance, 251-500, click the link under 09-10.
- Click **Submit** at the bottom of the page to save your number lists.
- Follow the same steps to program a different Call Taping number list. But ensure that the different List number you programmed is entered in the Station Advanced Feature Template applied to the extensions.
- If you want calls between extensions to be taped, click the **Call Taping for Internal Calls** check box.
- Click **Submit** at the bottom of the page to save changes to the template.
- Now, apply the programmed template to the desired extensions to which you want to provide the Call Taping facility. Refer the topic "Station Advanced Feature Template" for programming instructions.

Programming the Station Software Port using a Telephone

- Enter SE mode from a DKP/SLT.

  To enable/disable Call Taping Beeps, dial:
  - ***5332-Code***
    Where,
    Code is
    0 for Disable
    1 for Enable

  To program a number in a List:
  - ***4302-List Number-Location Index-Number-#****
    Where,
    List Number is from 01 to 16. In this case, 09 for Incoming Number List and 10 for Outgoing Number List.
    Location Index is from 001 to 999.
    Number is a number string of maximum 16 digit long terminated with #*.
    The digits used are as given in Table below for digits: 0-9, #, *, A, B, C, D, F, P, +.

Special Digit	Code
Flash (F)	#2
Pause (P)	#3
A	#4
B	#5
C	#6
D	#7
+	#8
Dot (.)	#9

Special Digit	Code
#	##
*	**

To clear a number from a Location Index:

- ***4302-List Number-Location Index-#****
  Where,
  List Number is from 01 to 16. In this case, 09 for Incoming Number List and 10 for Outgoing Number List.
  Location Index is from 001 to 999, where the number you want to clear is stored.

To program Number List - Incoming in Station Advanced Feature Template:

- ***5602-1-Template Number-Feature Number-Code***
  Where,
  Template Number is from 01 to 50.
  Feature Number is
  18 for Number List-Incoming.
  19 for Number List-Outgoing.
  Code is the Number List from 01 to 16.
  In this case, 09 for Incoming Number List and 10 for Outgoing Number List.

To enable Call Taping Internal Flag in a Station Advanced Feature Template, dial:

- ***5602-1-Template Number-Feature Number-Code***
  Where,
  Template Number is from 01 to 50.
  Feature Number for Call Taping Internal Flag is 20.
  Code is
  0 for Disable
  1 for Enable

To enable Tape calls coming without CLI Flag in a Station Advanced Feature Template, dial:

- ***5602-1-Template Number-Feature Number-Code***
  Where,
  Template Number is from 01 to 50.
  Feature Number for Tape calls without CLI is 17.
  Code is
  0 for Disable
  1 for Enable

- Exit SE mode.

For SE commands for applying the programmed template to DKP and SLT extensions, refer the topic “Customizing Station Advanced Feature Template using a Telephone”.

Also refer the topic “Number Lists” to know more.

How to use

This feature works automatically on the extensions which have the Call Taping parameters configured.

Call Taping conversations can be recorded either in a common mailbox or in the extension user’s personal mailbox.

- If calls are taped in a common mailbox, only the extension users having access to the common mailbox can retrieve and listen to the recorded conversations.

- If calls are taped in a personal mailbox, the extension user can listen to the recorded conversation by accessing their personal mailbox.

Accessing Personal Mailbox

If you are an EON/Extended IP Phone user

- Press ‘Voice Mail’ Key.
  OR
- Dial ***3931-Your Mailbox Password***[268]
- Follow Voice Mail Prompts to listen to new messages.

If you are an SLT user

- Lift the handset.
- Dial ***3931-Your Mailbox Password***
- Follow Voice Mail Prompts to listen to new messages.
- Replace handset.

268. Only if the mailbox is password protected, you will be prompted to enter the password.

Call Toggle

What's this?

Call Toggle allows you to have two simultaneous telephone conversations, talking to two persons alternately.

Call Toggle is also referred to as Hold-Consult or Call Splitting,

You can toggle between:

- Two internal calls (that is, two extensions).
- An internal Call and an External Call (extension and trunk).
- Two external calls (two trunks).

How it works

- A, B, and C are extensions.
- D and E are trunks.

Toggling between two internal calls

- A is in speech with B and C is on Consultation Hold.
- To talk with C, A dials Flash-1. Speech with C.
- To talk with B, A dials Flash-1. Speech with B.
- A can toggle back to C by dialing Flash-1.

Toggling between internal call and external call

- A is in speech with B and D (external call) is on Consultation Hold.
- To talk with D, A dials Flash-1. Speech with D.
- To talk with B, A dials Flash-1. Speech with B.
- A can toggle back to D by dialing Flash-1.

Toggling between two external calls

- A is in speech with D and E is on Consultation Hold.
- To talk with E, A dials Flash-1. Speech with E.
- To talk with D, A dials Flash-1. Speech with D.

- A can toggle back to E by dialing Flash-1.

- *The party put on Consultation Hold during Call Toggle cannot hear the conversation between the other two parties.*
- *You can also toggle between an incoming internal/external call (indicated by call waiting tone) and an internal/external call you are currently in speech with.*
- *You can also answer an incoming 'Interrupt Request' call and toggle between the interrupting extension and the extension you were in speech with.*
- *You can convert a Call Toggle into a three-party conference by dialing Flash-*3.*
- *You can transfer the call you are currently in speech with to another extension.*
- *You can park the call you are currently in speech with.*

How to configure

Call Toggle is a Class of Service (CoS) dependant feature.

In the default Station Basic Feature Template 01 assigned to all extensions of SARVAM UCS, Call Toggle is included in the 'Basic Features' assigned to all CoS groups, including the default CoS group 01. So, all extensions of SARVAM UCS can use this feature.

As Call Toggle is a part of the set of 'Basic Features', you cannot disable this feature selectively in the COS of extensions, without disabling the entire set of features.

No specific programming is required for this feature, except for programming a DSS key for Call Toggle, if required. Refer the topic “DSS Keys Programming” for instructions.

How to use

For EON and Extended IP Phone Users

Call Toggle between two internal calls:

- Speech with Extension 1.
- Extension 2 on Consultation Hold.
- To talk with Extension 2, press DSS key assigned to Call Toggle.
- Speech with extension 2.
- Press DSS key assigned to Call Toggle again.
- Speech with Extension 1.

Call Toggle between an Internal Call and an External Call:

- Speech with extension.
- External party on Consultation Hold.
- To talk with the external party, press DSS key assigned to Call Toggle.
- Speech with external party.
- Press DSS key assigned to Call Toggle again.
- Speech with Extension.

Call Toggle between two External Calls:

- Speech with external party 1.
- External party 2 on Consultation Hold.
- Press DSS Key assigned to Call Toggle.
- Speech with External party 2.
- Press DSS key assigned to Call Toggle again.
- Speech with External party 1.

For SLT Users

Call Toggle between two internal calls:

- Speech with extension 1.
- Extension 2 on Consultation Hold.
- To talk with extension 2, dial Flash-1.
- Speech with extension 2.
- Dial Flash-1again.
- Speech with extension 1.

Call Toggle between an Internal Call and an External Call:

- Speech with extension.
- External party on Consultation Hold.
- To talk with external party, dial Flash-1.
- Speech with external party.
- Dial Flash-1 again.
- Speech with extension.

Call Toggle between two External Calls:

- Speech with external party 1.
- External party 2 on trunk 2 on Consultation Hold.
- To talk with external party 2, dial Flash-1.
- Speech with external party 2.
- Dial Flash-1 again.
- Speech with external party 1.

Call Traffic

What is this?

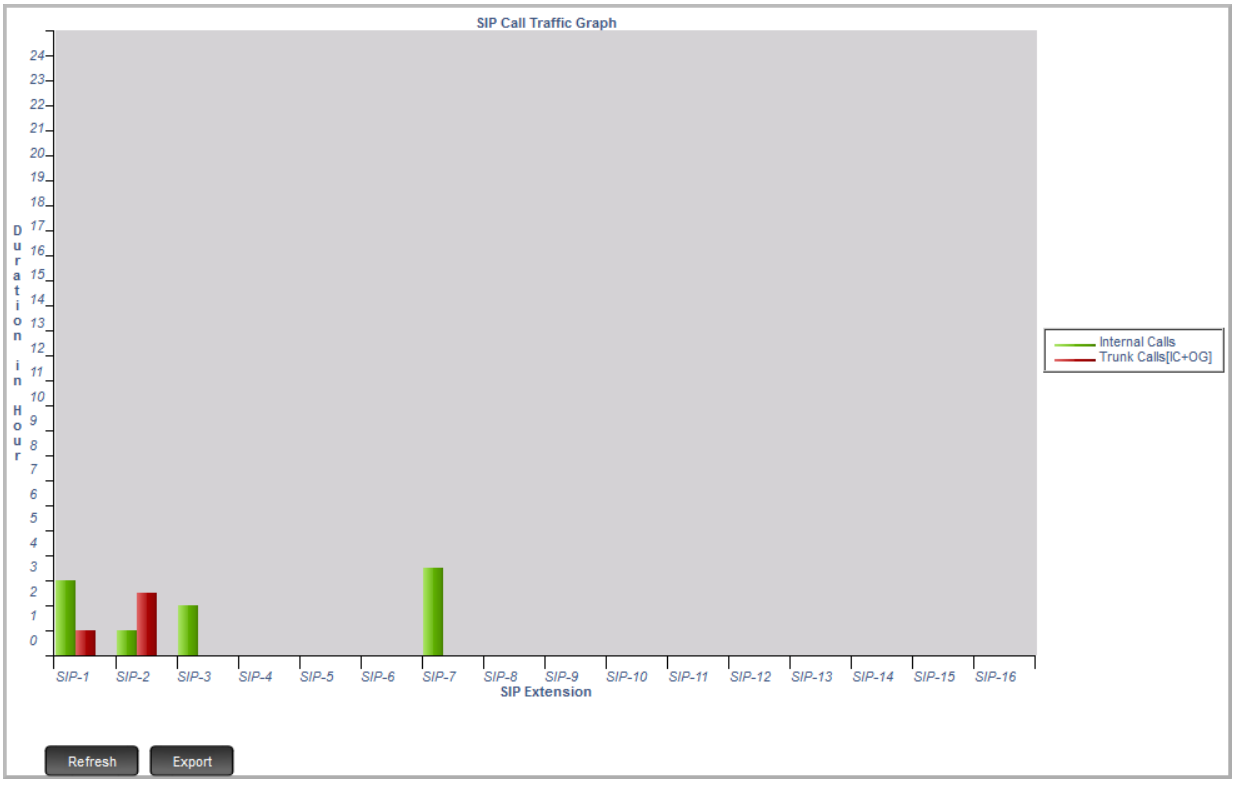
Call traffic measurement feature of SARVAM UCS gives a graphical representation of the time duration for which extensions and trunk ports remained Off-hook. The data is represented in a bar graph format for the SLT, DKP, Magneto, ISDN Terminals, CO, E&M, T1E1 and Mobile ports as well as SIP Extensions.

In the graph, the duration is shown in Hours along the Y-axis while the extension(s)/ trunk(s) port names are shown along the X-axis. The following call traffic measurement data (for each Extension and Trunk port separately) is displayed:

- Time for which each extension remained in speech for making/receiving Internal calls.
- Time for which each extension remained in speech for making/receiving Trunk calls (Incoming + Outgoing).
- Time for which each trunk port remained in speech for Incoming calls.
- Time for which each trunk port remained in speech for Outgoing calls.

This time is measured in terms of the number of hours, and the traffic is measured for last 24 hours.

You can view this traffic information in graphical format on the Jeeves; see the illustration below for Call Traffic information generated for SIP extension.



The two-color bars distinguish Internal calls (green) and Trunk calls (red)

You can also export this information in a database readable format like Microsoft Excel. The call traffic information files can be saved on a local disk.

How to use

To view Call Traffic in graphical format, you need to log into the Jeeves as System Engineer.

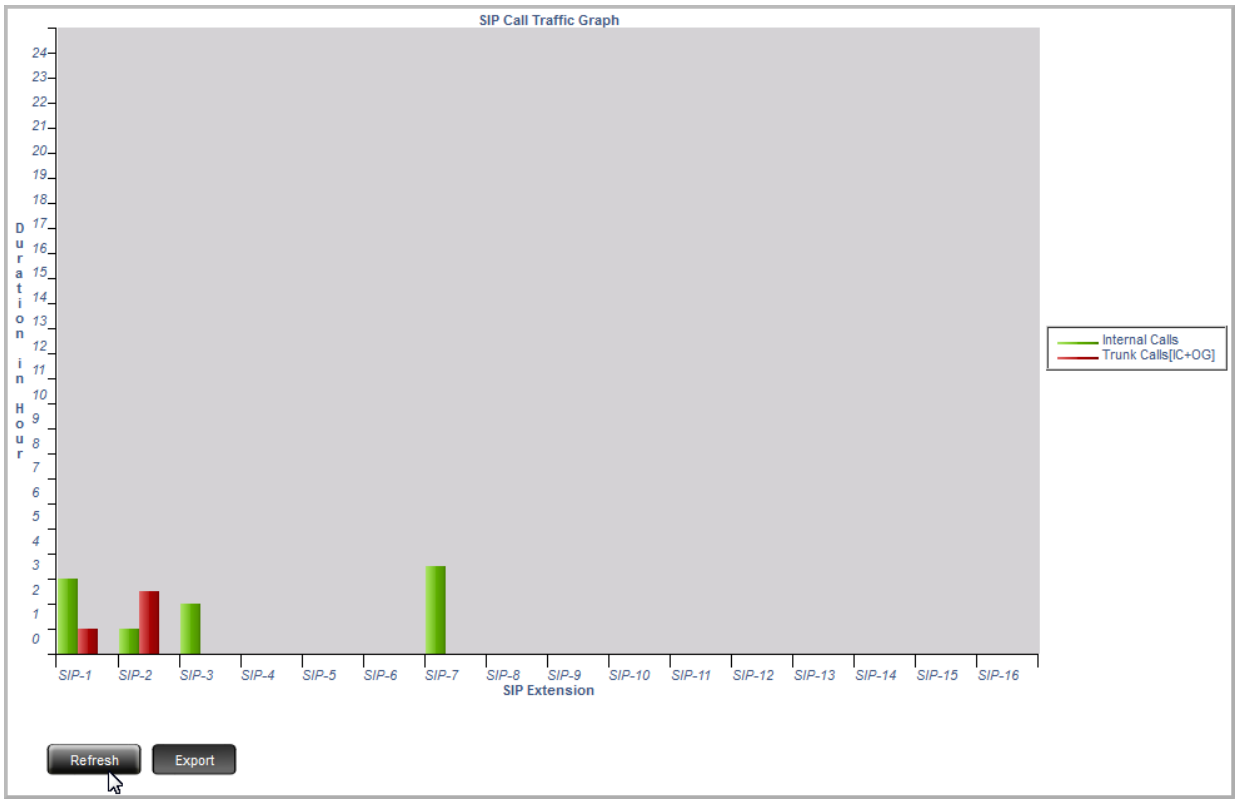
Go to the configuration page of the desired trunk or extension port for which you want to view Call Traffic data.

On the page, click the **Call Traffic** button.

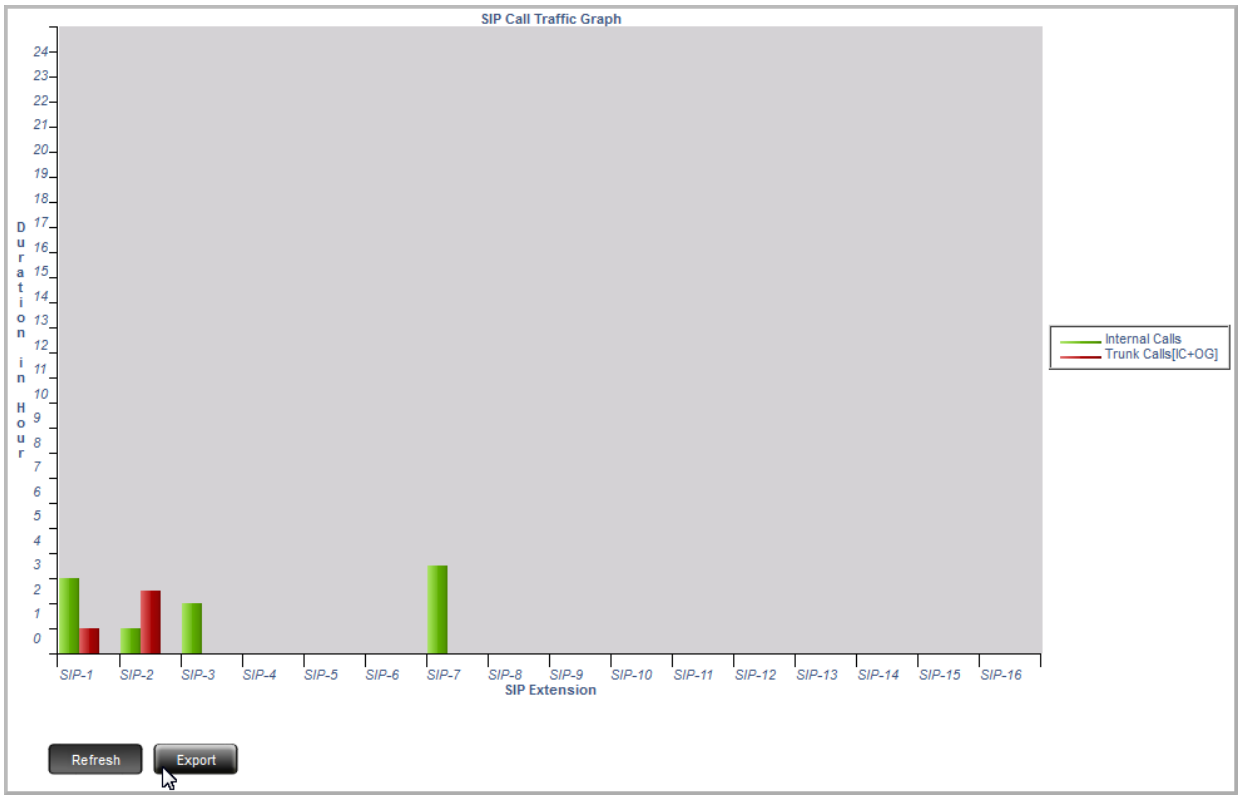
The call traffic will be generated as a graph.

If there is no call traffic usage data is available for the extension/ trunk you are currently viewing, the page will not show any bars.

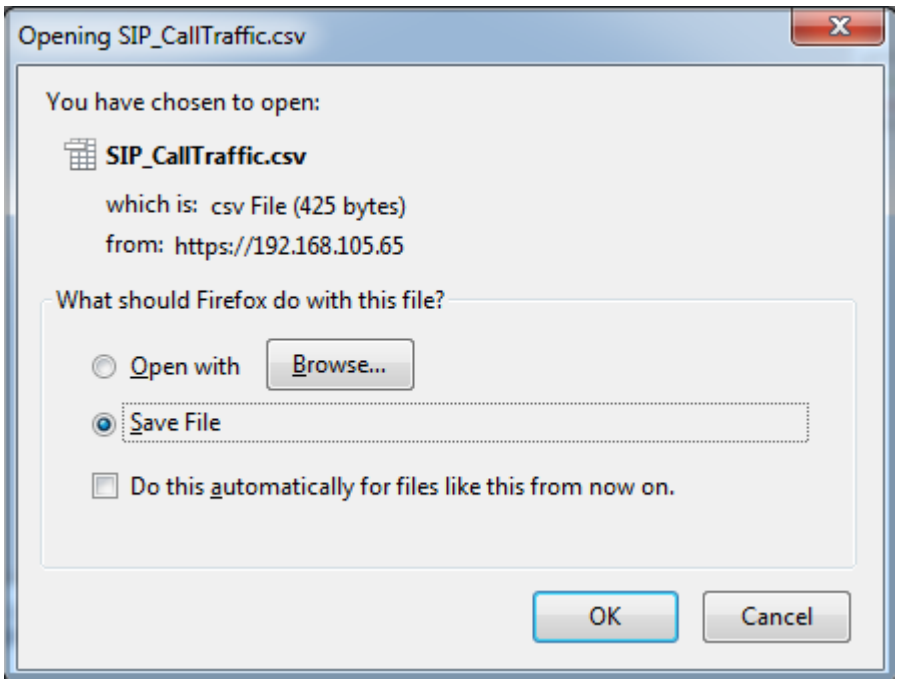
Click the **Refresh** button to get the latest 24hrs call traffic statistics. The application filters the SMDR records for last 24hrs and graph is generated accordingly.



- Click the **Export** button to export the data you are currently viewing.

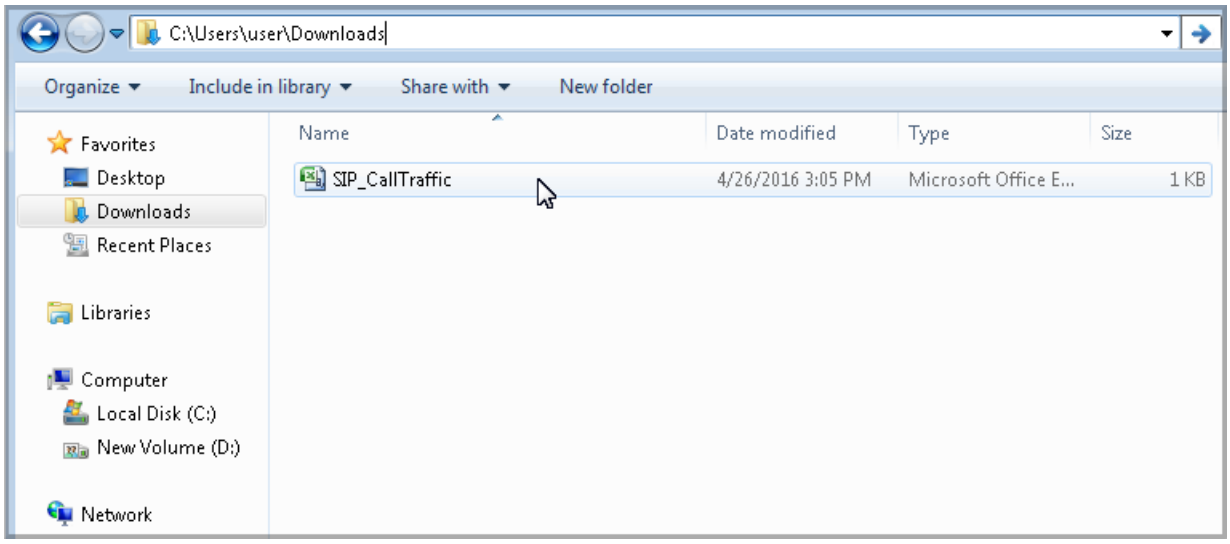


- The **Opening SIP_CallTraffic.csv** window will open.

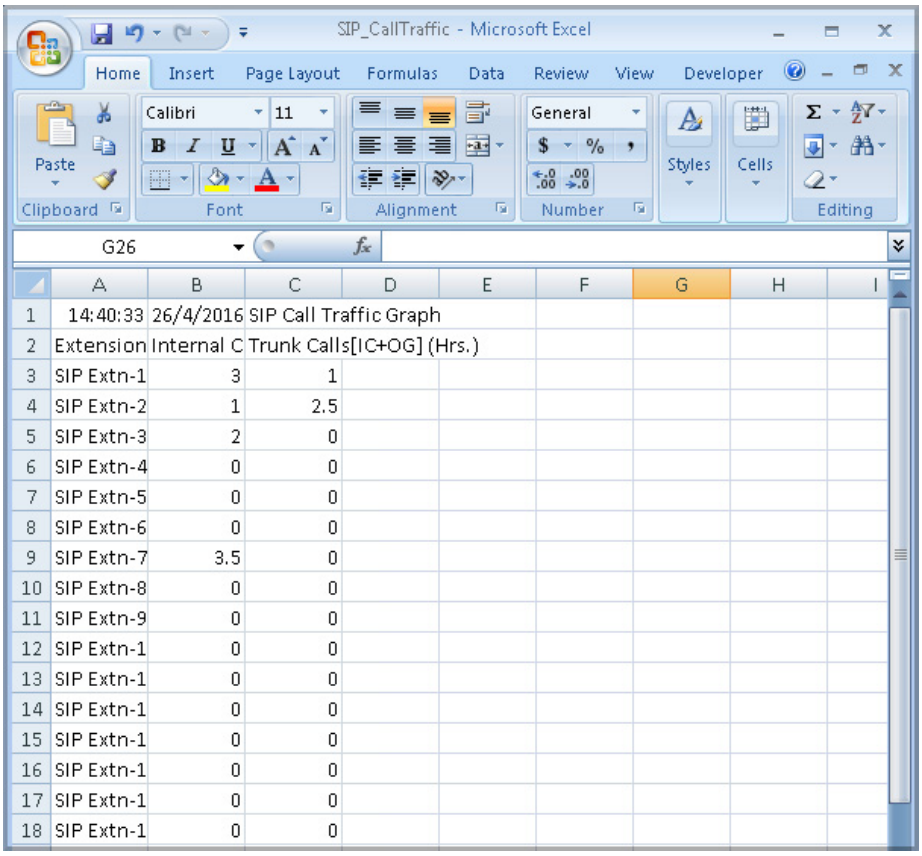


You can choose whether to open this file or save it in .xls format as per your requirement.

- Save the file on the local disk.



- If you open this file in MS Excel then it may look as shown in the following image.



The file contains the date and time when the call traffic data is calculated and the names of the extensions/ trunks for which it is calculated.

- You can save this file on your local disk from Excel window also.
- After you are done, you may log out of Jeeves.

Call Transfer

What is this?

Call Transfer enables you to relocate an existing call from an extension or trunk to another extension or to an external number. Calls can be transferred after notifying the other extension/external number about the impending transfer or can be transferred directly without notification.

The types of Call Transfer SARVAM UCS offers are:

- **Call Transfer - Screened:** The Operator puts the caller on Consultation Hold, dials the desired party's extension, and informs the desired party of the impending transfer. If the desired party chooses to accept the call, the call is transferred over to them.

- **Call Transfer - While Ringing:** The Operator puts the caller on Consultation Hold, dials the desired party's number and transfers the call when the desired party's extension starts ringing.

  This feature is used when there are several other calls to be attended and the Operator cannot wait for the desired party to answer.

- **Call Transfer - On Busy:** The Operator puts the caller on Consultation Hold, dials the desired party's number and transfers the call even when the desired party is busy in speech with another person. The busy extension gets intrusion tone and can choose to answer the intruding (transferred) call.

- **Call Transfer - Trunk-to-Trunk:** An external call is transferred on to another trunk line. The Operator puts the external caller on Consultation Hold, dials the desired party's external number, and transfers the call after or without notifying the desired party of the impending transfer.

  Trunk-to-Trunk call transfer may be used to transfer incoming calls for out-of-office extension users to their cell phones, or to connect personnel at remote or distant locations. For instance: an out-of-office executive who does not have long distance dialing permission can call the office and request the operator to connect him to the desired party on a trunk line.

- **Blind Transfer to VMS:** The Operator puts the caller on Consultation Hold, dials the feature access code for Blind Transfer to VMS, dials the desired party's number, and transfers the call. The call is transferred to the mailbox assigned to the desired party. The caller may leave a message in the mailbox.

- *Call Transfer is not exclusively an Operator feature, though it is used mostly by Operators. Calls can be transferred by any extension to another extension or external number, if "Basic Features" (this will allow you to access all types of Call Transfer except Trunk-to-Trunk Transfer) are allowed in Class of Service of the transferring extension.*

- *To access Trunk-to-Trunk Transfer make sure Trunk-to-Trunk Transfer feature is enabled in the Class of Service of the transferring extension.*

- *SARVAM UCS enables SIP extensions to resume a transferred call before it has been answered by the transfer target (which may be an extension or an external number). For a list of IP phones on which this feature has been tested, see "SARVAM UCS Features tested on IP Phones of different Brands"in the Appendix.*

- *SARVAM UCS allows Semi-attended Transfer and Transfer on Conference Hangup on SIP Trunks. For a list of IP phones on which this feature has been tested, see "SARVAM UCS Features tested on IP Phones of different Brands"in the Appendix.*

How it works

A and B are extension users.
C is an external caller.
D is an external number.

Call Transfer from Trunk/Extension to Extension

Three scenarios are possible:

1. Screened Transfer:
   - C calls a Trunk of SARVAM UCS.
   - The Operator answers the call.
   - Operator puts C on Consultation Hold.
   - Operator dials B's extension.
   - When B answers the call, the Operator informs B about the call.
   - If B accepts the call, the Operator transfers the call to B.
   - Now, B and C are in speech.

   - *If B does not accept the call, Operator may dial Flash to retrieve the call and speak to C.*
   - *The Operator can also abort call transfer while B's phone is ringing by dialing Flash. The Operator gets connected to C.*

2. Transfer While Ringing:
   - C calls a Trunk of SARVAM UCS.
   - The Operator answers the call.
   - The Operator puts C on Consultation Hold and dials B's extension.
   - The Operator transfers the call when B's phone starts ringing.
   - If B answers the call, B gets connected to C.
   - If B does not answer the call at the end of the Transfer-While Ringing Timer (programmable; default: 30 seconds), the call is returned to the Operator. C gets Ring Back Tone.
   - The Operator answers the call and is in speech with C.
   - However, if the Operator is busy at the time of call return, the system waits for the Operator to become free. When the Operator is free, C gets Ring Back Tone.
   - The Operator answers the call and is in speech with C.

3. Transfer On Busy:
   - C calls a Trunk of SARVAM UCS.
   - The Operator answers the call.
   - The Operator puts C on Consultation Hold and dials B's extension.
   - The Operator gets busy tone from B's extension. B is busy with A
   - The Operator transfers the call to B on Busy tone.
   - B gets intrusion beeps. The beeps are played for the duration of the Transfer on Busy Timer (programmable; default: 30 seconds)
   - B may dial Flash to answer the call. A is put on hold.
   - B is now connected to C.

- *If B does not answer the intrusion beeps at the end of the Transfer on Busy Timer, the call is returned to the Operator. C gets ring back tone.*
- *If the Operator too is busy at the time of call return, C gets busy tone.*

Call Transfer - Trunk to Trunk

- C calls a Trunk of the SARVAM UCS.
- The Operator answers the call.
- B is out of office, but is available at external number D.
- The Operator puts C on Consultation Hold.
- The Operator dials trunk access code and calls the external number D.
- The Operator may:
  Wait for D to answer the call, transfer the call if D accepts the call. (screened transfer)

  OR

  Transfer the call as soon as D's phone starts ringing. (transfer while ringing)
- C and D are now in speech for the duration of the Trunk-to-Trunk Inactivity Timer[269].
- A warning tone is given at the end of the Trunk-to-Trunk Inactivity timer (programmable; default: 2 minutes). On expiry of this timer, the call is disconnected.
- To extend the call, either C or D must dial any digit in tone (DTMF), except '##'.

Dialing '##' to extend the call will result in Call Disconnection.

In the case of Trunk-to-Trunk transfer, both parties in speech on trunk lines must be informed that their call would be disconnected at the end of the Trunk-to-Trunk Inactivity Timer and that they must dial any digit, except "##" to extend the call.

Call Transfer - Blind Transfer to VMS

- C calls a Trunk of SARVAM UCS. C wants to talk to A.
- The Operator attends the call.
- The Operator puts C on Consultation Hold.
- The Operator dials Feature Access Code for Blind Transfer to VMS and A's extension number.
- The system hunts for the mailbox assigned to A's extension.
- When the mailbox is found, the Operator gets confirmation tone. C gets connected to A's mailbox.
- C gets voice prompts. C may follow the prompts to leave a message.

- *If A does not have a mailbox assigned, the Operator will get an error tone while transferring the call.*
- *The Operator may retrieve C's call by pressing Transfer Key/Flash /Call Appearance key.*

Feature Interactions:

269. The process of Trunk-to-Trunk transfer takes place outside of the System. So, the System will not know which of the two trunks have gone ON-Hook. Hence the call is automatically disconnected when the Trunk-to-Trunk Inactivity Timer expires.

- **CLIP and Caller ID Presentation while Transfer:** SARVAM UCS provides the flexibility to display either the extension number that is transferring the call or the held party's number, that is, the number of the party that is about to be transferred. Refer “Calling Line Identification and Presentation (CLIP)”.

- **Privacy:** Call Transfer-On Busy will not work if the busy extension has Call Privacy from intrusion Tone in its Class of Service.

- **DND:** Call Transfer will not work if the destination extension has set DND.

- **Call Hold:** You must retrieve a held call, to Transfer it.

How to configure

To be able to use Call Transfer, this feature must be enabled in the Class of Service group of the extensions to be allowed this feature. The default values of the related Timers may be changed, if required.

To be able to use Blind Transfer to VMS, the extensions must be assigned a mailbox. For instructions, refer to “Configuring Voice Mail System” .

Call Transfer in Class of Service

In the default Station Basic Feature Template Number 01 is assigned to all the extensions of SARVAM UCS, the default CoS group 01 in this template has Call Transfer included in the 'Basic Features'. So, all extensions of SARVAM UCS can use this feature.

You cannot disable 'Call Transfer' selectively without disabling the entire set of 'Basic Features'.

Refer the topics “Class of Service (COS)” and “Station Basic Feature Template” to know more.

Call Transfer Related Timers

- **Transfer While Ringing Timer:** This timer is related to Call Transfer - While Ringing. It is the time for which the system rings the extension. By default it is set to 30 seconds. At the end of the timer the call is returned to the transferring extension.

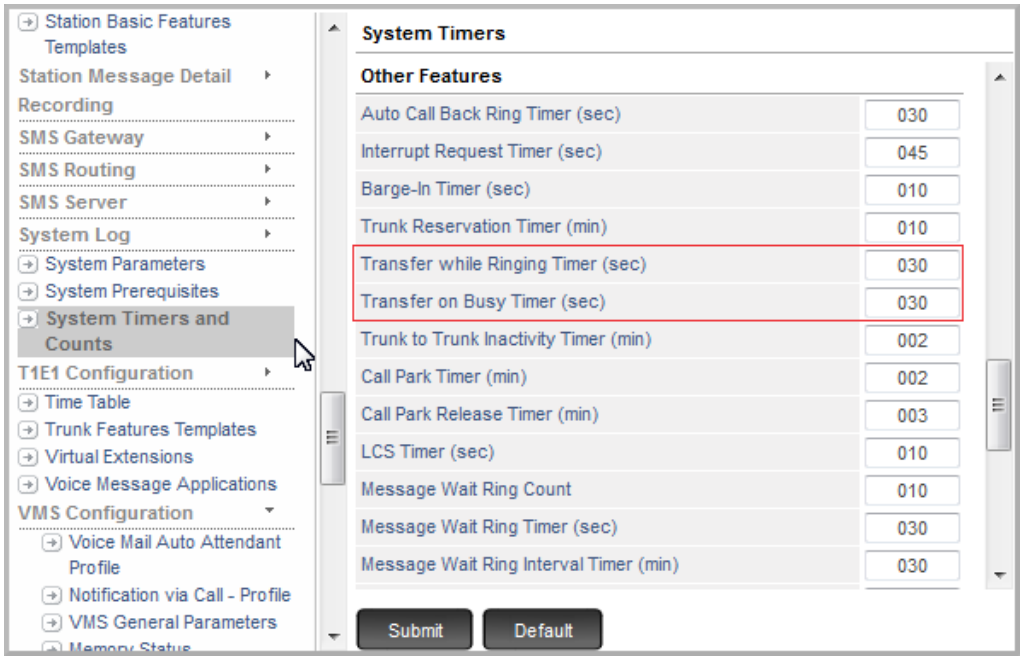
- **Transfer on Busy Timer:** This timer is related to Call-Transfer on Busy. It is the time for which the system waits for the busy extension to respond to the intrusion tone. By default the timer is set to 30 seconds. At the end of the timer the call is returned to the transferring extension.

- **Trunk to Trunk Inactivity Timer:** This is the time duration after which the system disconnects the call transferred from one trunk line to another. By default it is set to 2 minutes. At the end of the timer the call is disconnected, if either party does not dial digits to extend the call. This Timer is relevant for CO to CO and CO to E&M calls only.

Changing Call Transfer Related Timers using Jeeves

- Log in as System Engineer.

- Under **Configuration**, click **System Timers and Counts** to open the page.



- Scroll to reach **Other Features** and change the values as required for Call Transfer related timers.
- Click **Submit** at the bottom of the page to save changes.
- Log out of Jeeves or continue with programming tasks, as required.

Changing Call Transfer Related Timers using a Telephone

- Enter SE mode.

  To program Transfer While Ringing Timer:
  - Dial command ***3806-Seconds***
    Where,
    Seconds is from 001 to 255 seconds.
    Default: 030

  To program Transfer on Busy Timer:
  - Dial command ***3807-Seconds***
    Where,
    Seconds is from 001 to 255 seconds.
    Default: 030

  To program the Trunk-to-Trunk Inactivity Timer:
  - Dial command ***3808-Minutes***
    Where,
    Seconds is from 001 to 255 minutes.
    Default: 2 minutes

- Exit SE mode.

How to use

For EON and Extended IP Phone Users

Extension to Extension:

- Speech with extension.
- Press DSS Key assigned to desired party extension.
- Go ON-Hook or press ‘Transfer’ Key.

OR

- Speech with extension.
- Press ‘Transfer’ Key. You get feature tone.
- Dial desired party's extension number.
- Go ON-Hook or press ‘Transfer’ Key.

Extension to Trunk:

- Speech with extension.
- Press DSS Key assigned to Trunk.
- Dial External Number (transfer target)
- Go ON-Hook or press ‘Transfer’ Key.

OR

- Speech with extension.
- Press ‘Transfer’ Key. You get feature tone.
- Dial-Trunk Access Code[270]- External Number.
- Go ON-Hook or press ‘Transfer’ Key.

Trunk to Extension Transfer:

- Speech with Trunk.
- Press DSS Key assigned to desired party extension.
- Go ON-Hook or press ‘Transfer’ Key.

OR

- Speech with Trunk
- Press ‘Transfer’ Key and dial desired party's extension number
- Go ON-Hook or press ‘Transfer’ Key.

Trunk to Trunk Transfer:

- Speech with Trunk.
- Press DSS Key assigned to Trunk.
- Dial External Number.
- Speech with External Number.
- Go ON-Hook or press ‘Transfer’ Key.

270.Trunk Access Code: users worldwide may dial a code from 0, 5, 61, 62, 63, and 64. Users in USA may dial a code from 0, 9, 81, 82, 83, and 84.

OR

- Speech with Trunk.
- Press ‘Transfer’ Key and dial Trunk Access Code[271]-External Number.
- Speech with the External Number.
- Go ON-Hook or press ‘Transfer’ Key.

Blind Transfer to VMS:

- Speech with extension.
- Press DSS Key assigned to Blind Transfer to VMS.
- Dial desired party’s extension number.
- Go ON-Hook or press ‘Transfer’ Key.

OR

- Speech with extension.
- Press Transfer key.
- Dial 1078.
- Dial desired party's extension number.
- Go ON-Hook or press ‘Transfer’ Key.

For SLT Users

Extension to Extension Transfer:

- Speech with extension.
- Press Flash.
- Dial desired party's extension number.
- Replace handset.

Extension to External Number:

- Speech with extension.
- Press Flash.
- Dial Trunk Access Code.
- Dial External Number.
- Replace handset.

External Number to Extension Transfer:

- Speech with External Number.
- Press Flash.
- Dial desired party's extension number.
- Replace handset.

Trunk to Trunk Transfer:

- Speech with External Number 1.
- Dial Flash.
- Dial Trunk Access Code

271.Trunk Access Code: users worldwide may dial a code from 0, 5, 61, 62, 63, and 64. Users in USA may dial a code from 0, 9, 81, 82, 83, and 84.

- Dial External Number 2.
- Speech with the External Number 2.
- Replace Handset.

Blind Transfer to VMS

- Speech with Trunk/Extension.
- Press Flash.
- Dial 1078.
- Dial desired party's extension number.
- Replace handset.

Calling Line Identification and Presentation (CLIP)

What's this?

The SARVAM UCS provides the facility of detecting the caller's number and presenting it on the display of the called extension phone. This feature is called Calling Line Identification and Presentation (CLIP).

The calling number can be presented on ISDN Terminals, EON and also on SLTs that support CLI protocols.

The signaling protocols for CLI supported by SARVAM UCS are: DTMF, FSK V.23, and FSK-Bellcore.

These protocols are supported on trunks as well as extensions. Any type of trunk line and supporting DTMF or FSK signaling can be interfaced with the SARVAM UCS.

Similarly, any type of telephone instrument supporting DTMF or FSK signaling protocol can be connected to the SLT port.

How it works

When CLIP is enabled on a trunk,

- SARVAM UCS senses the digits/codes sent by PSTN.
- It sends this information to the landing extension/Operator along with the ringing signal.
- In case of, Internal calls the calling extension's name and number both are presented to the called extension.
- In the case of External calls, only the number will be displayed on the landing/Operator extension.
- When the landing extension/Operator transfers the incoming call to an extension, putting the external caller on hold, the system sends this information to the extension to which the call is transferred.
- During the transfer, the number of the landing extension/Operator will be displayed on the transfer destination extension.
- On successful call transfer, the caller's number will be displayed on the transfer destination extension.

In the case of Call Transfer, the system also provides the option of displaying to the destination extension either the number of the party that is put on hold to be transferred, that is, the Held Party OR the number of the Transferring Party, while the call transfer is taking place. This feature is called Caller ID Presentation while Transfer.

It is also possible to remove and replace the '+' character received as CLI on telephones that do not support CLIP starting with this character.

For example, the GSM network sends the calling party number with '+' as the prefix. If the telephone connected as extension does not support this, it will not present the CLI of the caller. To overcome this, SARVAM UCS provides you the option of replacing '+' with an appropriate number string which these telephones can display.

Feature Interactions:

- **CLIR:** CLIP and Caller ID Presentation while Transfer will work only if CLIR is not enabled on the extension that has transferred the call. Refer the topic “Calling Line Identification Restriction (CLIR)”.

- **Q-Sig:** When two Systems - System A and System B are networked using Q-Sig, and an extension of one System, for instance, System A transfers a call to an extension of System B by putting the caller on hold, the CLI presented on the extension of System B will be according to the type of Caller ID Presentation while Transfer set on the transferring extension of System A.

How to configure

The functioning of this feature is controlled by two parameters: **CLIP Type** and **Caller ID Presentation while Transfer**.

If you want to replace '+' characters received as CLI on telephones that do not support CLI prefixed with this character, you must program the relevant flag and the desired number string in the 'System Parameters'.

All these parameters can be programmed using Jeeves and a Telephone.

CLIP Type

If SLTs supporting CLI are connected to the SARVAM UCS, the System Engineer must select a signaling protocol for CLI in the SLT Hardware Template applied on the SLT extensions. By default SLT Hardware Template 01 is assigned to all SLT extensions. The default CLIP Type in Template 01 is 'DTMF'.

There is no need to select a CLIP Type in the default Hardware Template 01, if all the SLTs support DTMF protocol.

If all SLTs support a different CLIP Type say FSK-Bellcore, you may simply select this CLIP Type in the default Hardware Template applied on all SLT extensions.

However, if certain SLTs support a particular CLIP type and some support a different CLIP type, then create separate SLT Hardware Templates with different CLIP types and apply them to the appropriate SLTs.

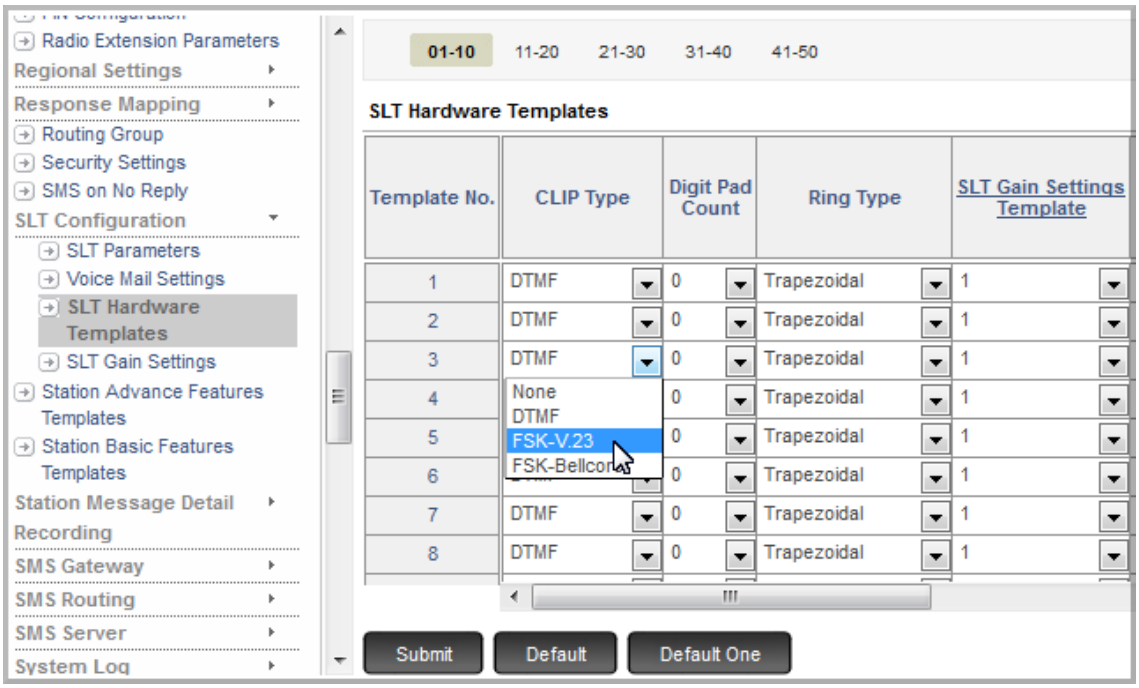
For example, you may select SLT Hardware Template 02 with FSK V.23 and SLT Hardware Template 03 with FSK Bellcore, and Template 04 with DTMF as the CLIP Type and apply each template to the SLTs as per the CLIP Type they support.

Selecting CLIP Type using Jeeves

- Log in as System Engineer.

- Under **Configuration**, click **SLT Configuration**.

- Click **SLT Hardware Templates** to open the page.



- Select the **CLIP Type** supported by the SLT.
- Click **Submit** at the bottom of the page to save changes.

Selecting CLIP Type using Telephone

- Enter SE mode from a DKP/SLT.
- Dial command ***5702-1-Template Number-Parameter Number-Code***
  Where,
  Template Number is SLT Hardware Template Number from 01 to 50
  Parameter Number is 01 for CLIP Type.
  Code is:
  0 for None
  1 for DTMF
  2 for FSK-V23
  3 for FSK Bellcore
  For instance: To select FSK-V23 in SLT Hardware Number 01, dial 5702-1-01-01-2
- Exit SE mode.

Caller ID Presentation while Transfer

The Caller ID Presentation while Transfer gives you the choice of displaying to the destination extension, the CLI of either the party put on hold to be transferred, that is, the Held Party or the Transferring Party, that is, the party transferring the call, during a call transfer.

This feature is to be programmed in the Station Advanced Feature Template applied on the extension.

In the Station Advanced Feature Template 01 assigned to all extensions by default, Caller ID of the Held Party is selected.

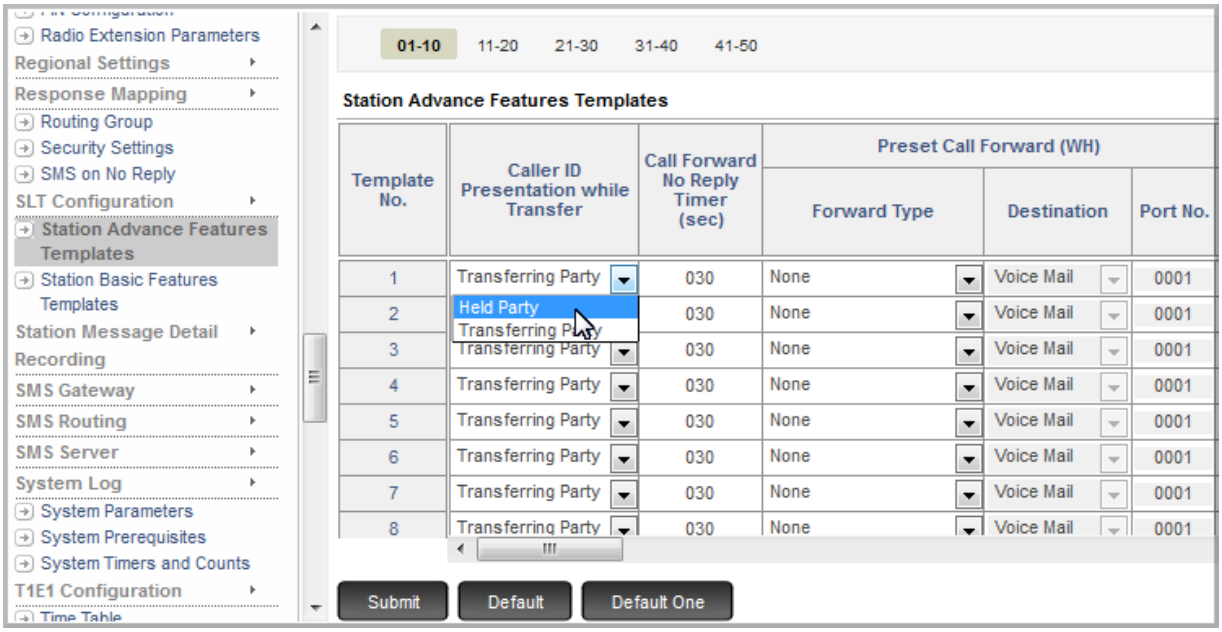
But if you want to show the Caller ID of the Transferring party, select this option in the default Station Advanced Feature Template 01.

If all certain extensions are to be provided Caller ID of the Held Party and others the Caller ID of the Transferring Party,

1. Create separate Station Advanced Feature Templates.
2. Select 'Held Party' or 'Transferring Party' as desired in each Template.
3. Apply this Template on the extensions.

Configuring Caller ID Presentation while Transfer using Jeeves

- Log in as System Engineer.
- Under **Configuration**, click **Station Advanced Feature Templates** to open the page.
- In **Caller ID Presentation while Transfer** field, select the desired option: **Held Party** or **Transferring Party**.



- Click **Submit** at the bottom of the page to save your setting.
- Log out of Jeeves or continue programming.

Configuring Caller ID Presentation while Transfer using a Telephone

- Enter SE mode from a DKP/SLT.
  - Dial command ***5602-1-Template Number-Feature Number-Code***
    Where,
    Template Number is Station Advanced Feature Template Number from 01 to 50

Parameter Number is 01 for Caller ID Presentation while Transfer.
Code is:
1 for Transferring Party
2 for Held Party
For instance: To select Transferring Party in Template Number 02, dial 5602-1-02-01-1

- Exit SE mode

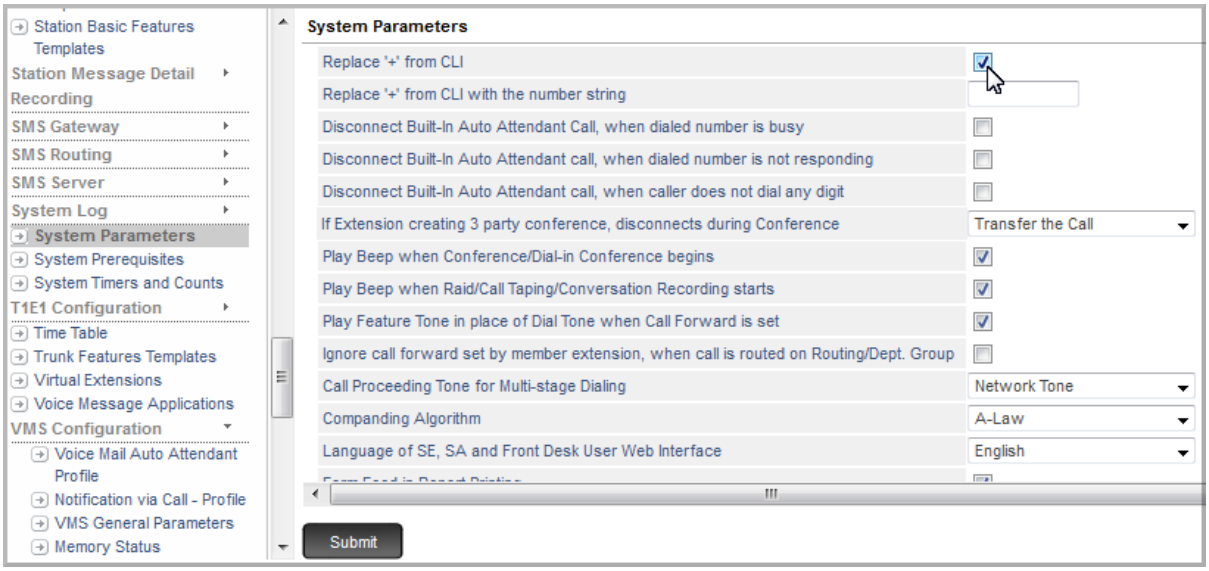
Refer the topic “Station Advanced Feature Template” for instructions assigning Templates to extensions.

Replacing '+' in CLI

To replace '+' characters received as CLI on telephones that do not support CLI prefixed with this character, you must program the flag 'Replace '+' from CLI?' and the desired number string that should replace this character.

Configuring Replace '+' in received CLI using Jeeves

- Log in as System Engineer.
- Under **Configuration**, click **System Parameters** to open the page.



- Enable the **Replace '+' from CLI** flag by selecting the check box.
- Enter the desired number string in the field **Replace '+' from CLI with the number string**.
- Click **Submit** at the bottom of the page to save your setting.
- Log out of Jeeves or continue programming.

Configuring Replace '+' using Telephone

- Enter SE mode from a DKP/SLT.

  To replace '+', dial:
  - ***5334-Code***

Where,
Code is
0 for Disable (no modification in CLI)
1 for Enable (replace '+' in CLI with the substitute number string programmed)
Default: 0

To program Replacement string, dial:

- ***5335-Replacement String-#****
  Where,
  Replacement string is any string with digits from 0 to 9. A maximum of 6 digits are permitted. Terminate the command with #* if the replacement string is fewer than 6 digits.

To remove the replacement string, dial:

- ***5335-#****

- Exit SE mode.

Calling Line Identification Restriction (CLIR)

This feature is not applicable if CDMA Mobile Card is installed in your system.

What's this?

The SARVAM UCS allows extension users to suppress their identity, that is, extension number and name, when they call other extensions. This feature is called Calling Line Identification Restriction (CLIR).

Extensions that have 'CLIR Override' facility can view the CLI of those that have suppressed it with CLIR.

This is a feature of the system and not the PSTN. It is applicable for internal calls only.

This feature will work only on the Matrix Extended IP Phone, VARTA UC Clients, proprietary digital key phone EON and SLTs that support Caller Line Identification (CLI).

How it works

- Extension A has CLIR enabled.
- Extension B does not have CLIR Override enabled.
- Extension C has CLIR Override enabled.
- When A calls B, B cannot view the extension name and number of A.
- When A calls C, C can view A's extension name and number.

Now,

- Extension D calls extension E.
- A picks up the call.
- D will be able to view A's name and extension only if it has CLIR Override enabled and is a digital key phone, EON.

Feature Interactions:

- **CLIP and Caller ID Presentation while Transfer:** Both these features will not work if CLIR is enabled.

How to configure

CLIR and **CLIR Override** are Class-of-Service-dependent features. Extensions that are to be allowed these features, must have them enabled in their **Class of Service (CoS)** group.

Decide which extensions should be allowed CLIR and which should be allowed CLIR Override.

In the default factory settings, Station Basic Feature Template Number 01 is assigned to all the extensions of SARVAM UCS. Template 01 is assigned CoS group 01 in which both CLIR and CLIR Override are disabled. Thus, none of the extensions of the SARVAM UCS can suppress their CLI or force any other extension to display its CLI.

If you want to enable both features on all extensions, simply enable CLIR and CLIR Override in the default CoS group 01.

If you want to allow CLIR to all extensions, but not allow CLIR Override to any extension, simply enable CLIR in the default CoS group 01.

If you want to allow CLIR and/or CLIR Override to selected extensions, only, then follow these steps:

1. Define a new CoS group with CLIR/CLIR Override enabled.
2. Prepare a Station Basic Feature Template with this CoS group applicable in all the “Time Zones”.
3. Assign this new Template to the extensions to which CLIR/CLIR Override is to be allowed.

   Refer the topics “Class of Service (COS)” and “Station Basic Feature Template” for detailed instructions and programming.

How to use

For EON and Extended IP Phone Users

To enable CLIR

- Press DSS Key assigned to CLIR[272].
  OR
- Dial 1031 to enable CLIR.
- You get confirmatory tone and message on the phone's display.
- Go idle or you get dial tone after 3 seconds.

To disable CLIR:

- Press DSS Key assigned to CLIR again[273].
  OR
- Dial 1030.
- You get confirmatory tone and message on the phone's display.
- Go idle or you get dial tone after 3 seconds.

For SLT Users

To enable CLIR:

- Lift handset.
- Dial 103-1.
- You get confirmation tone.
- Replace handset.

To disable CLIR:

- Lift handset.
- Dial 103-0.
- You get confirmation tone.

---

272. *System Engineer is recommended to assign a DSS Key with LED to this feature. When the assigned DSS key is pressed, it will glow red indicating that CLIR is enabled.*
273. *If a DSS key with LED has been assigned, when you press the key again, the LED will be turned off indicating CLIR is now canceled.*

- Replace handset.

Cancel All Station Features

What's this?

For each feature of the SARVAM UCS that extension users have set/enabled on their extension, the system provides access code for cancellation of the feature.

Extension users can also cancel *all* features set on their extension by dialing a feature access code. These set of features can also be canceled for any extension user from the SA Jeeves.

When an extension user dials the '*Cancel All Station Features*' access code, the following features, if set, are canceled from the extension:

- Auto Answer
- Auto Call Back
- Call Forward
- Do Not Disturb
- Hot Line
- Trunk Reservation
- Walk-In Class of Service

How to use

The extension users can cancel *all* features set on their extension from their own extension or these can be canceled from SA Jeeves for any extension user.

Cancel All Features by Extension Users

For EON and Extended IP Phone Users

- Press DSS Key assigned to 'Cancel All Features' (if programmed).

  OR

- Dial 1051.
- You get confirmation tone and confirmatory message on your phone display.
- Go idle or wait for dial tone.

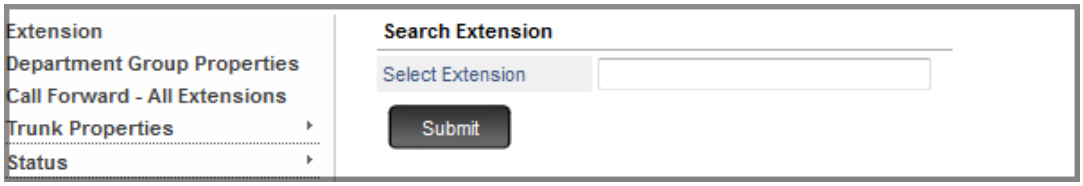
For SLT Users

- Lift handset.
- Dial 1051.
- You get confirmation tone.
- Replace the handset.

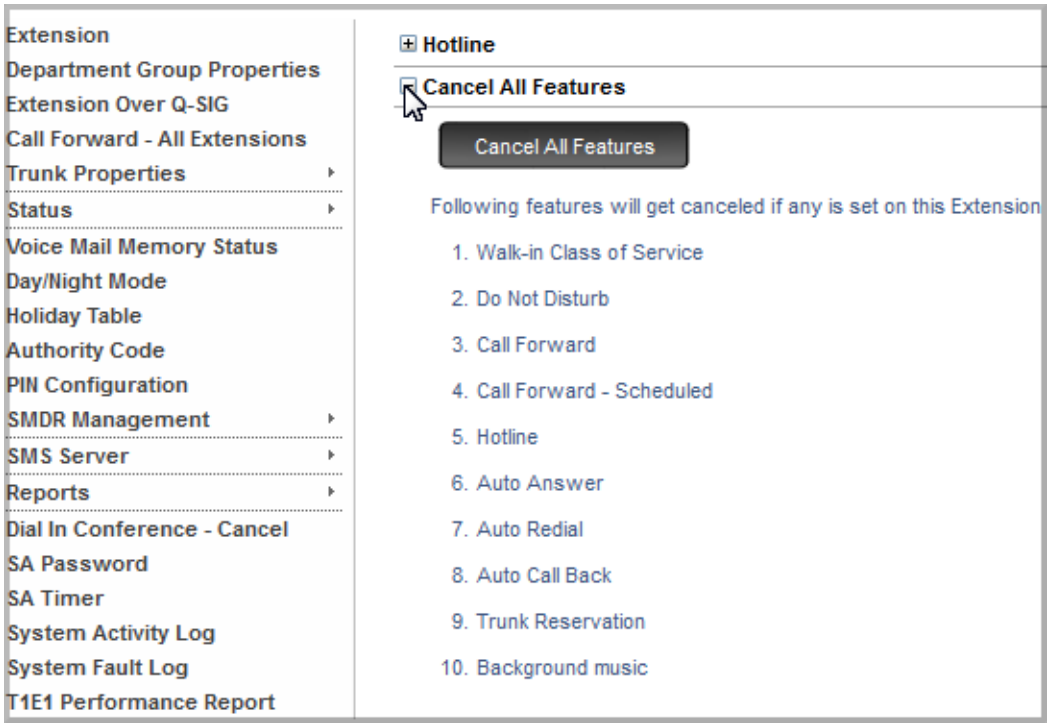
Cancel All Features for Extension Users

The Operator or any extension user having access to System Administrator mode can cancel all features for extension users using Jeeves.

- Log in to Jeeves as System Administrator.
- Click **Extension**.



- **Select Extension:** Enter the Extension Number or the Extension Name of the extension you want to search.
- Click **Submit**.
- The searched extension users details appear on your screen.
- Click **Cancel All Features** to expand.



- Click the **Cancel All Features** button. The following features will get canceled if any is set on this extension user:
  - Walk-in Class of Service
  - Do Not Disturb
  - Call Forward
  - Call Forward - Scheduled
  - Hotline

- Auto Answer
- Auto Redial
- Auto Call Back
- Trunk Reservation
- Background music[274]

274. ETERNITY PENX does not support Background Music.

Class of Service (COS)

What's this?

Class of Service (CoS) defines the permission an extension will have on a System. It defines the set features of the system that the extension is to be allowed access to.

Feature requirements vary among users and with time. Certain groups of extension users may have a need for voice mail, while another group may need the ability to forward calls to a cell phone, and still others may have no need to make calls outside the office.

Similarly, certain features that are required during working hours may not be required during break or non-working hours.

SARVAM UCS offers the flexibility to allow or deny extension users access to features of the System, on the basis of their requirement and according to time of the day. For users, access to various features from their extensions is their CoS.

How it works

The list of all features allowed to an extension is referred to as 'CoS group'. There are 20 CoS groups numbered from 01 to 20.

In each CoS group there are 60 features, which are identified by 2-digit numbers, from 01 to 60. These are referred to as the 'CoS Feature Numbers'.

Each extension port of the System has an associated CoS group that indicates which features of the System the port is allowed to access.

The CoS group of an extension port is defined in the “Station Basic Feature Template” applied to that extension port. It is defined for each "Time Zone", namely, working hours, break hours, and non-working hours, in the Template.

A feature can be allowed or denied to an extension by enabling or disabling it in the CoS group of the Station Basic Feature Template applied to that extension.

The same CoS group uniformly to all extensions ports for all Time Zones. Doing so, all extensions can access the same set of features in all time zones. For example: CoS group 03 is assigned to all extensions for Working, Break and Non-Working hours.

A different CoS group for each Time Zone can be assigned to all extension ports. Doing so, all extensions can access only those features allowed for the particular Time Zone.

For example: All extensions are assigned CoS group 03 for Working, CoS group 04 for Break hours and CoS group 05 for Non-Working Hours.

Different CoS groups can be assigned to different extension ports, for all or for different Time Zones. Doing so, each extension can access a different set of features in each Time Zone.

For example: extensions 3001 to 3010 are allowed CoS group 03 for all Time Zones, while extensions 3011 to 3015 are assigned CoS group 03 during Working Hours, and for the Non-Working and Break Hours, they are assigned CoS group 04 and 05 respectively.

The following features can be set/cancelled on extensions from the SA mode, regardless of whether these features are allowed or denied in the CoS assigned to the extensions:

- *Call Forward*
- *DND*
- *Dynamic Lock and Timer*
- *Hotline*

Basic Features

A set of features including Internal Call, Call Hold, Call Toggle, Call Transfer, Department Call, Operator Access, Redial, and Call Mute defined as Basic Features and allowed in all CoS groups.

It is not possible to enable or disable selectively any of the features included in "Basic Features".

How to configure

The table below presents the CoS groups from 01 to 20 with the list of 01 to 60 features supported on the extensions.

Default CoS Groups

Feature Number	Feature Name	Class of Service Group																			
		01	02	03	04	05	06	07	08	09	10	11	12	13	14	15	16	17	18	19	20
01	Account Code	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓
02	SA Mode																				
03	SA Extension																				
04	Auto Call Back-Busy	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓
05	Auto Call Back-No Reply	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓
06	Auto Redial																				
07	Auto Redial Priority																				
08	Barge-In																				
09	Call Forward	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓
10	Call Park	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓
11	Call Pickup	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓
12	Change Room Clean Status																				
13	Global Directory Programming																				
14	CLIR																				
15	CLIR Override																				
16	Trunk Call Waiting																				
17	Conference	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓
18	Continued Dialing	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓
19	Conversation Recording																				
20	Decrement Dynamic Lock Timer for Internal Calls																				
21	DISA																				
22	Do Not Disturb	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓
23	Do Not Disturb - Override																				
24	Dynamic Lock	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓
27	Forced Answer																				

Feature Number	Feature Name	Class of Service Group																			
		01	02	03	04	05	06	07	08	09	10	11	12	13	14	15	16	17	18	19	20
28	Forced Release																				
29	Forced Release Order																				
30	Global Directory Part-1																				
31	Global Directory Part-2																				
32	Global Directory Part-3																				
33	Hot Desk																				
34	Hotline																				
35	Interrupt Request	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓
36	Live Call Screening	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓
37	Live Call Supervision																				
38	Manual Priority Intrusion																				
39	Message Wait (Set/Cancel)	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓
40	Mini Bar Details																				
41	Paging	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓
42	Privacy from Built-In Auto Attendant																				
43	Privacy from Interrupt Request, Barge-In and DND-Override																				
44	Privacy from Raid	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓
45	Raid																				
46	Return Call to Original Caller (RCOC)	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓
47	Room Monitor																				
48	Selective Port Access																				
49	Trunk Reservation																				
50	Trunk-Trunk Transfer																				
51	Basic Features[a]	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓
52	General MailBox	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓
53	Dept. Group-Call Forward	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓
54	Emergency Conference	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓
55	Intercom																				
56	DSS Call Pickup-Station																				
57	DSS Call Pickup-Trunk																				
58	Closed User Group (CUG)																				
59	PIN Dialing																				

a. Basic Features includes: Internal Call, Call Hold, Call Toggle, Call Transfer, Department Call, Operator Access, Redial, Mute.

- *CoS group number 01 is assigned for all Time Zones in the default Station Basic Feature Template 01 assigned to all extensions of the SARVAM UCS.*
- *CoS group number 19 and 20 are assigned when the Hospitality Application of SARVAM UCS is used. When this application is being used in CoS group number 20, the SA Mode will be enabled. See SARVAM UCS Hospitality System Manual.*

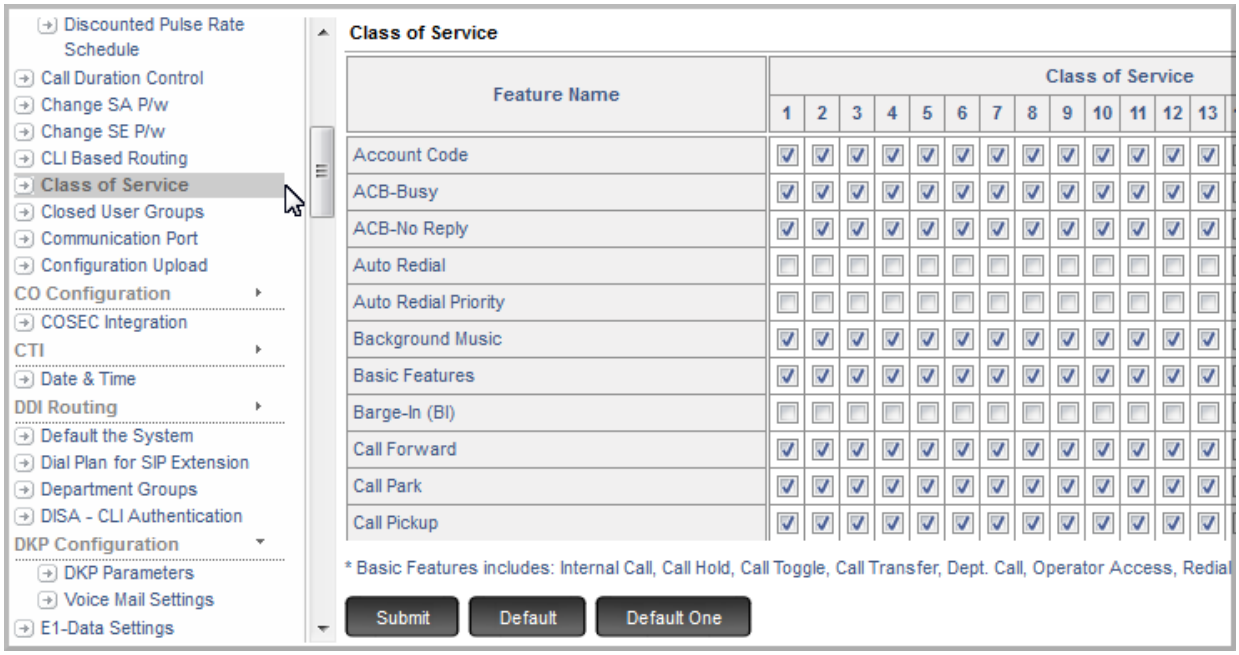
Creating CoS Groups

- Take a pen and a paper pad.
- Prepare a list of extensions.

- Read the list of features supported on the extensions (see above Table 'Default CoS Groups for Features').
- Against each extension name on the list, write the features needed for each Time Zone. You will notice that the features needed by many extensions are identical.
- List the common features to be allowed to and features to be denied to all extensions. Assign a CoS Group Number to this list.
- Are there any other features, in addition to those on the common list, which you want to allow to selected extensions?
- If yes, extend the common list you prepared by adding the features to be allowed to selected extensions. Assign a CoS Group Number to this extended list.
- You can prepare different CoS Groups for different Time Zones and assign a number to each group.
- For example, you may end up creating five different CoS groups. The First group may contain none of the features. The Second group may contain the most common features like Call Forward, Call Transfer, Internal Dialing, etc. The Third group may contain more advanced features, and the Fourth group may contain even more advanced features. The Fifth group may contain all the features.
- When you are finished preparing the CoS groups you need, program the CoS groups using Jeeves or by issuing SE commands from a Telephone (DKP or SLT). See below for instructions.
- Now, the CoS groups to be assigned to extensions must be programmed in the Station Basic Feature Template applied to the extensions. This can be done using Jeeves or by issuing SE commands from a Telephone (DKP or SLT). See below for instructions.

Enabling a feature in a CoS Group using Jeeves

- Log in as System Engineer.
- Under **Configuration**, Click the link **Class of Service (CoS)**.



- The default CoS groups from 01 to 20 appear. The check boxes selected under each CoS group column indicate that the feature is enabled in that CoS group. The default CoS groups meet the requirements of most extension users. Check the default CoS groups whether the features you want to allow are enabled and features you want to deny are disabled.
- To enable a feature in a CoS group, select corresponding check box in the CoS group. To disable a feature simply clear the check box. For example: to enable DND-Override in CoS group 01, select the check box against DND-Override in CoS group 01. To disable clear the check box again.
- Click 'Submit' to save changes.

Enabling a feature in a CoS Group using a Telephone

- Enter SE mode from a DKP/SLT.

  To enable a feature in a CoS group, dial:
  - **1302-1-CoS Group-Feature Number-Code** to enable a feature in a single CoS group.
  - ***1302-2-CoS Group-CoS Group-Feature Number-Code*** to enable the same feature in a range of CoS groups.
  - ***1302-*-Feature Number-Code*** to enable the same feature in all CoS groups.

    Where,

    CoS Group from 01 to 20.

    Feature Number from 01 to 60. Refer the Table “Default CoS Groups” for Feature Numbers.

    Code is

    1 for Enable

    0 for Disable

To default a CoS group, dial:

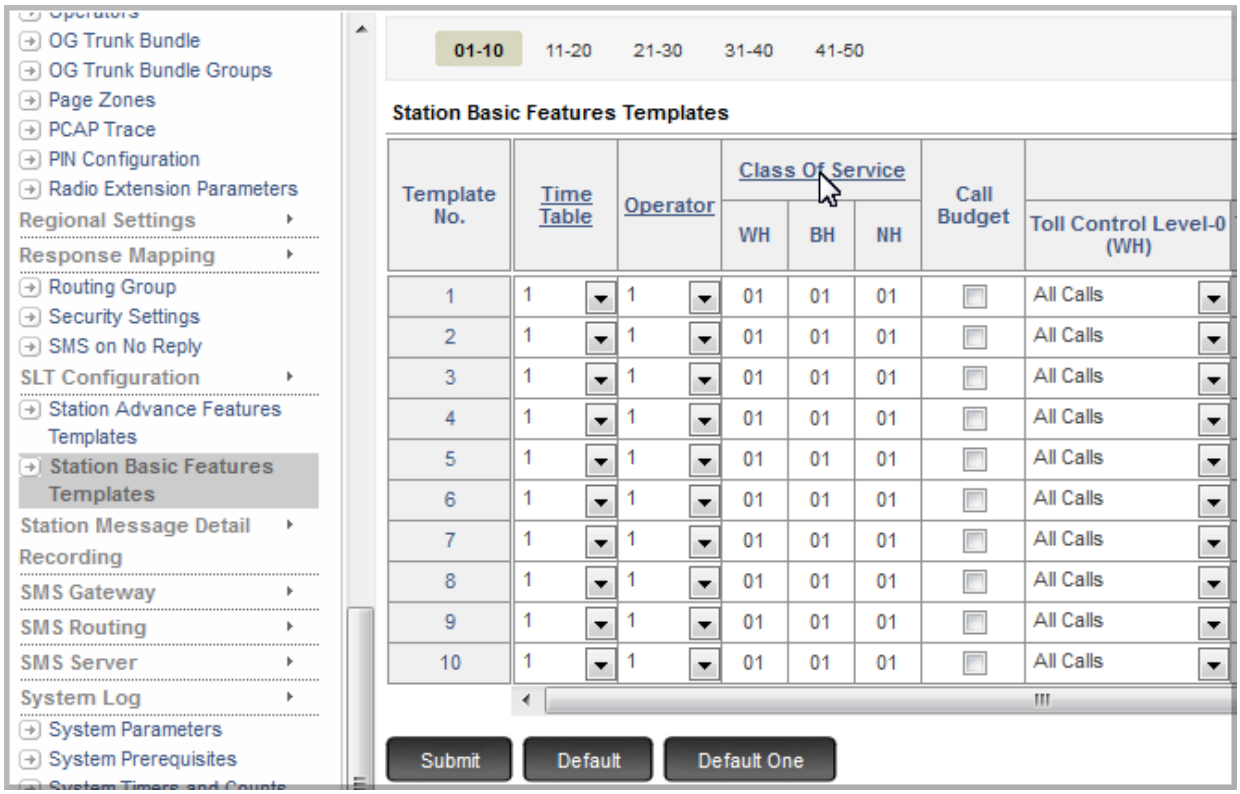
- ***1301-1-CoS Group*** to default a single CoS group.
- ***1301-2-CoS Group-CoS Group*** to default a range of CoS groups.
- ***1301-**** to default all CoS groups.
  Where,
  CoS Group is from 01 to 20.

- Exit SE mode.

Assigning a CoS group in a Station Basic Feature Template using Jeeves

- Log in as System Engineer, if not already logged in.

- Under **Configuration**, click **Station Basic Feature Template**.



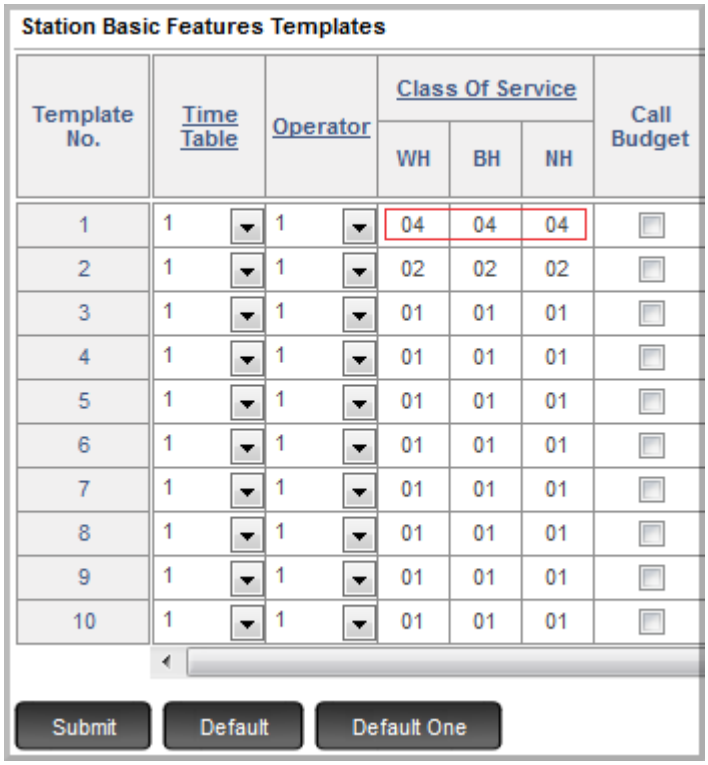
- The default CoS group assigned to each time zone, that is, working hour (WH), non-working hour (NH) and break hour (BH), appears under Class of Service in each Template.

- To assign a CoS group to a Station Basic Feature Template, enter the CoS group number for each time zone under Class of Service.

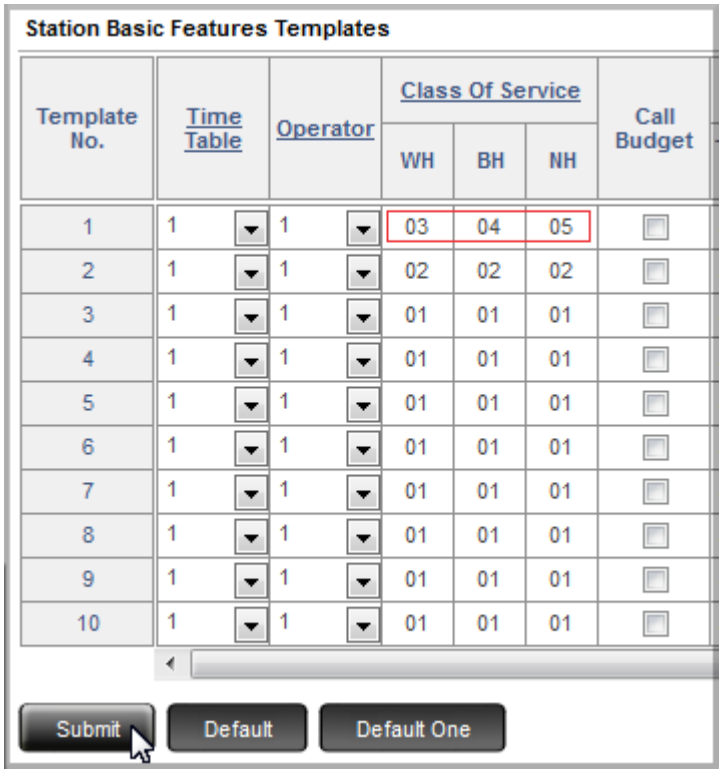
- Click **Submit** to save changes.

  By default, Station Basic Feature Template 01 is applied on all extensions and CoS group 01 is the assigned by default to this template in all time zones.

  If all extensions to be allowed the same set of features during working hours, break hours, non-working hours, enter the same CoS group number in all time zones in the Template Number applied to all extensions. For example: to assign CoS group 04 to all time zones in Template Number 01; enter 04 under WH, NH and BH.

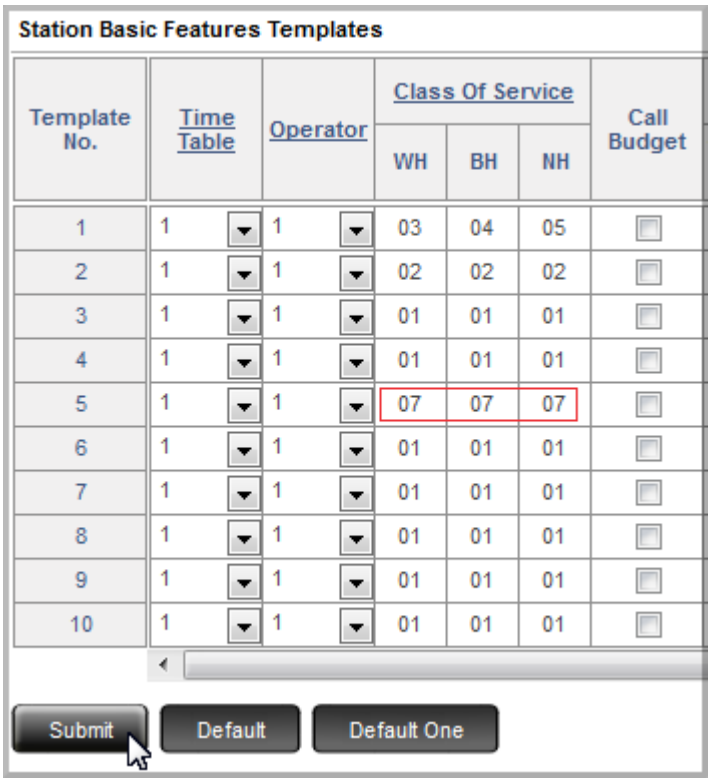


If all extensions are to be allowed a different set of features in each Time Zone, enter the CoS group for each Time Zone. For example: to assign CoS group 03 in working hours, 04 in Break Hours and 05 in Non-Working hours in Template 01, enter 03, 04, 05 under WH, BH and NH respectively.



- If a set of features is to be allowed to select extensions only, assign the CoS group with these features enabled to a separate Station Basic Feature Template. Apply this template to the select extensions which are to be allowed this CoS.

  For example: To assign all features to extensions, create a CoS group with all features enabled, CoS group 07. Select a different Station Basic Feature Template, for example 05. Enter CoS Group 07 in all Time Zones in Template 05. Apply Template 05 to the software ports of the extensions that are to be assigned all features.



- Remember to click **Submit** to save the changes you make on every page.

Assigning CoS group to Station Basic Feature Template using a Telephone

- Enter SE mode from a DKP/SLT.

  To assign CoS group to a template, dial:
  - ***5502-1-Template Number-Feature Number-Code***
    Where,
    Template Number is Station Basic Feature Template Number, from 01 to 50.
    Feature Number is
    03 for CoS group for Working Hours
    04 for CoS group for Break Hours
    05 for CoS group for Non-Working Hours
    Code is CoS group number from 01 to 20.

    For Example: To program CoS group 04 in Station Basic Feature Template Number 06 for all Time Zones, dial:
    - ***5502-1-06-03-04*** to program CoS group 04 for Working Hours.
    - ***5502-1-06-04-04*** to program CoS group 04 for Break Hours.
    - ***5502-1-06-05-04*** to program CoS group 04 for Non-Working Hours.

Similarly, to program CoS Group 03 in working hours, 04 in Break Hours and 05 in Non-Working hours, dial:

- ***5502-1-06-03-03*** to program CoS group 03 for Working Hours.
- ***5502-1-06-04-04*** to program CoS group 04 for Break Hours.
- ***5502-1-06-05-05*** to program CoS group 05 for Non-Working Hours.

- Exit SE mode.

After you have programmed the CoS group in the Station Basic Feature Template, you must assign this template to the stations. Refer the topic “Station Basic Feature Template” for instructions on applying templates on SLT, DKP, ISDN Terminal and SIP extensions.

Finally, test the CoS programmed for each extension by invoking the features from each extension.

CLI Based Routing

What is this?

SARVAM UCS offers the facility to detect the calling party's number and name. This is known as Calling Line Identification.

On the basis of CLI, it is possible to land calls from a particular telephone number on a particular extension or group of extensions. This is known as CLI Based Routing.

How it works

A, B, C are extensions. D and E are external callers.
Calls made by D are to be landed on A.
Calls made by E are to be landed on B and C.

The CLI of D and E and their corresponding landing destinations should be entered in the CLI Based Routing Table.

CLI Based Routing should be enabled on the desired trunks for each Time Zone (working hours, break hours and non-working hours).

The system can match the incoming call CLI with the numbers configured in the CLI table in two ways, that is, Match from last digit of CLI or Match from first digit of CLI.

- If you select Match from last digit of CLI, this is how the call will be routed:
  - D calls on a trunk of SARVAM UCS.
  - The system checks if CLI Based Routing is enabled on the trunk for the current time zone.
  - If CLI Based Routing is enabled on the trunk, the system checks the numbers stored in the CLI Based Routing table.
  - D's number is found in the CLI Based Routing Table.
  - The system checks the destination number stored against D's CLI.
  - A's number is found as the destination extension.
  - The system lands the call on A.

- If you select Match from first digit of CLI, this is how the call will be routed:
  - D calls on a trunk of SARVAM UCS.
  - The system checks if CLI Based Routing is enabled on the trunk for the current time zone.
  - Then the system checks if the parameter **Replace '+' from CLI**, is enabled.
  - If enabled, the system will replace **+** sign received in the incoming CLI with the number string configured in the **Replace '+' from CLI with the number string**. To know more, see “System Parameters”.
  - Now the system checks if **Incoming CLI Modification** is enabled.
  - If enabled the system modifies the incoming number according to the parameters configured in the Incoming CLI Modification. To know more, see “Incoming CLI Modification” in “System Parameters”.
  - The system matches the modified number string with the numbers stored in the CLI Based Routing table.
  - D's number is found in the CLI Based Routing Table.
  - The system checks the destination number stored against D's CLI.
  - A's number is found as the destination extension.
  - The system lands the call on A.

If D's number does not exist in the CLI Based Routing Table, the call will be routed according to the incoming call management logic.

How to configure

For this feature to work, you must do the following:

- enter the numbers of the calling parties and the numbers of the corresponding destination extensions in the CLI Based Routing Table. You can store up to 2000 numbers in the CLI Routing Table.
- enable CLI Based Routing on the desired trunks according to time zones in their “Trunk Feature Template”.

Creating CLI Routing Table

To apply this feature,

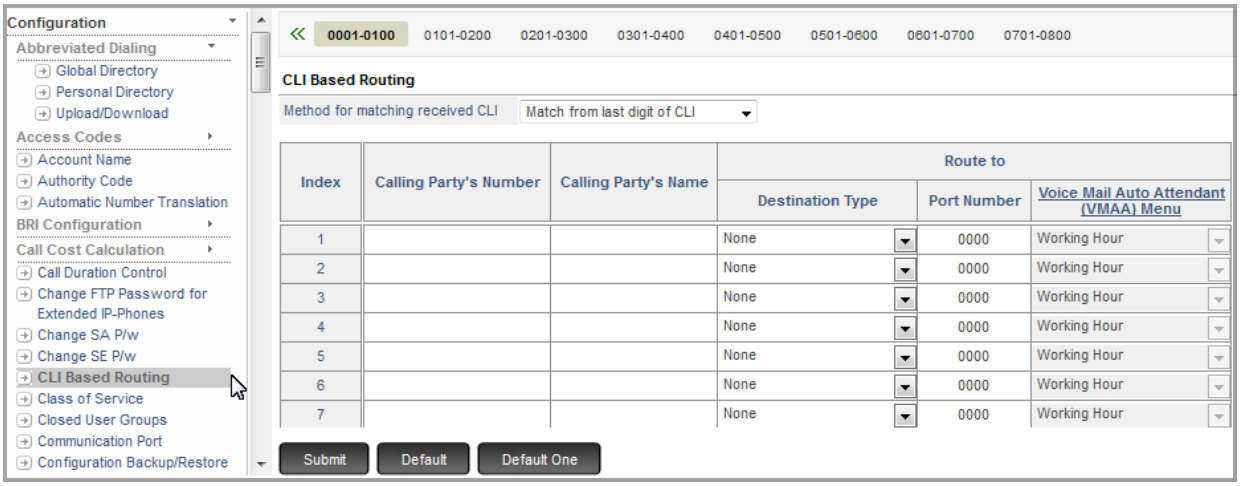
- On a sheet of paper, create a 5-column table, as illustrated below. Each calling party number in the CLI table is stored a location index in the system. Enter the telephone numbers and names of the calling parties and the corresponding landing destinations, that is, the Port Type and Port Number. The Port Type may be SLT, DKP, ISDN Terminal, SIP Extension, a Routing Group or Virtual Extension or a Voice Mail Auto Attendant.
- The 'Name' field is for identifying the entry. When placing a call on the destination extensions, both the number and the 'Name' are presented in the CLI.
- Determine the method which the system should use to match the incoming CLI with the numbers in the table, that is, Match from last digit of CLI or Match from first digit of CLI Based Routing Table.

Index	Telephone Number	Name	Port Type	Port Number
1	2640459	MidasBiz	Routing Group	02
2	022281110001	Jet Set	SLT	004
:				:
10	2640075	Bacchus	Routing Group	03

Configuring CLI Based Routing using Jeeves

- Log in as System Engineer.

- Under **Configuration**, click **CLI Based Routing**.



- In **Method for matching received CLI**, select the method according to which you want the system to match the received CLI with the numbers stored in the CLI table. You can select:
  - Match from last digit of CLI
  - Match from first digit of the CLI

  The method you select will be applicable to all the numbers configured in the CLI Based Routing Table.

- In the CLI table each number is to be stored at a Location Index numbered from 0001 to 2000.

  There are 100 entries on each page. To go to the next 100 Index numbers, click the tabs 0101-0200, 0201-0300, 0301-0400 , 0301-0400.....1901-2000.

- At each Location Index, enter the information for the following parameters:

  - **Calling Party's Number:** enter the number of the calling party, not exceeding 16 digits. You can also enter '+' in the number string.

  - **Calling Party's Name:** enter the name of the calling party. You can enter a maximum of 8 characters in this field.

  - **Destination Type:** select the landing destination extension. It may be an SLT, a DKP, an ISDN Terminal, SIP extension, a Routing Group, a Virtual Extension or Voice Mail Auto Attendant.

  - **Port Number:** enter the port number (software port or routing group number) to which the landing destination is connected. This is applicable for Destination Types — SLT, DKP, ISDN Terminal, SIP Extension or Routing Group.

    If you have selected *Virtual Extension* as the Destination Type, enter the software port of landing destination of the Virtual Extension.

    If you have selected a *Routing Group* as the Destination Type, enter the number of the Routing Group (01 to 96) in this field.

  - **Voice Mail Auto Attendant (VMAA) Menu:** if you have selected the *Voice Mail Auto Attendant* as the Destination Type, select the VMAA Menu to be assigned to the respective calling party.

You may click the *Voice Mail Auto Attendant (VMAA) Menu* link to edit the parameters of desired VMAA Menu. For details, see “Voice Mail Auto-Attendant Menu”.

- Click **Submit** to save entries.

- Enable CLI Based Routing for the desired Trunks in their “Trunk Feature Template”. Refer the topic “Customizing Trunk Feature Templates” for instructions.

Configuring CLI Based Routing Table using a Telephone

- Enter SE mode from a DKP/SLT.

  To add the calling party telephone number in the CLI Based Routing table, dial:
  - ***4101-Index-Telephone Number-#****
    Where,
    Index is from 0001 to 2000.
    Telephone Number is the calling party's telephone number (Max. 16 digits).
    Terminate the command with #*, if the number is less than 16 digits.
    To enter '+' in the number string, dial #8

    For Example: to enter 2640459 at Index Location 001, dial ***4101-001-2640459-#****

  To clear a telephone number from the CLI Based Routing table, dial:
  - ***4101-Index-#****

  To enter the name of the calling party corresponding to the calling party's telephone number, dial:
  - ***4102-Index-Name-#****
    Where, Index is from 0001 to 2000.
    Name can be a maximum of 8 characters.
    Terminate the name with #*, if less than 8 characters.

    For Example: to enter the name Midas Biz for the number 2640459 at Index Location 001, dial ***4102-001-MidasBiz***

  To clear the name stored at a location index in the CLI table, dial:
  - ***4102-Index-#****

  To assign the landing destination extension in the CLI table, dial:
  - ***4103-Index-Port Type-Port Number***
    Where,
    Index is from 0001 to 2000.
    Port Type is
    00 for Null
    01 for SLT
    02 for DKP
    20 for Routing Group
    28 for ISDN Terminal
    34 for SIP Extension
    36 for Virtual Extension

    Port Number is Software Port number of SLT/DKP/ISDN Terminal/SIP Extension.
    Port Number of SLT is from 001 to 240.
    Port Number of DKP is from 001 to 96.
    Port Number of ISDN Terminal is 01 to 64.

Port Number of Routing Group is from 01 to 96.
Port Number of SIP Extension is from 001 to 999.
Port Number of Virtual Extension is from 01 to 64.

For Example: to assign an SLT connected at Software Port number 004 as the landing destination for calls from Midas Biz 2640459, dial ***4103-001-01-004***

To clear a location index in the CLI Based Routing table, dial:

- **4104-Index** to clear a single location Index.
  Where,
  Index is from 0001 to 2000.

  By using these commands, the telephone number, name and the Port type-Port number will be cleared from the location index.

- Exit SE mode.

Clock Synchronization

What's this?

When data is transmitted from the SARVAM UCS to external lines or when SARVAM UCS receives data from the external lines, it is necessary that the transmitter and receiver be properly synchronized. If not clock slips can occur. A clock slip can generate a loss or addition of data to the data stream.

How it works

- This can be done in three ways viz. using the data clock or using the external clock (clock is sent by the network on a dedicated cable pair) or using the internal clock. SARVAM UCS does not support external clock. When the SARVAM UCS is connected to the PSTN, then it is recommended to extract the clock from the incoming data whereas if the SARVAM UCS is used to form a private network, you are recommended to use the internal clock. For example, if a private network is formed by connecting three SARVAM UCS systems, then one system should be programmed as master clock whereas other two should be programmed in the slave mode.

- If two or more T1E1 Ports are connected to the PSTN (or a Private Network) then in such case, clock will be extracted from the first T1E1 Software port whereas the transmit data on all other ports whether connected to PSTN or private network will be clocked as per received.

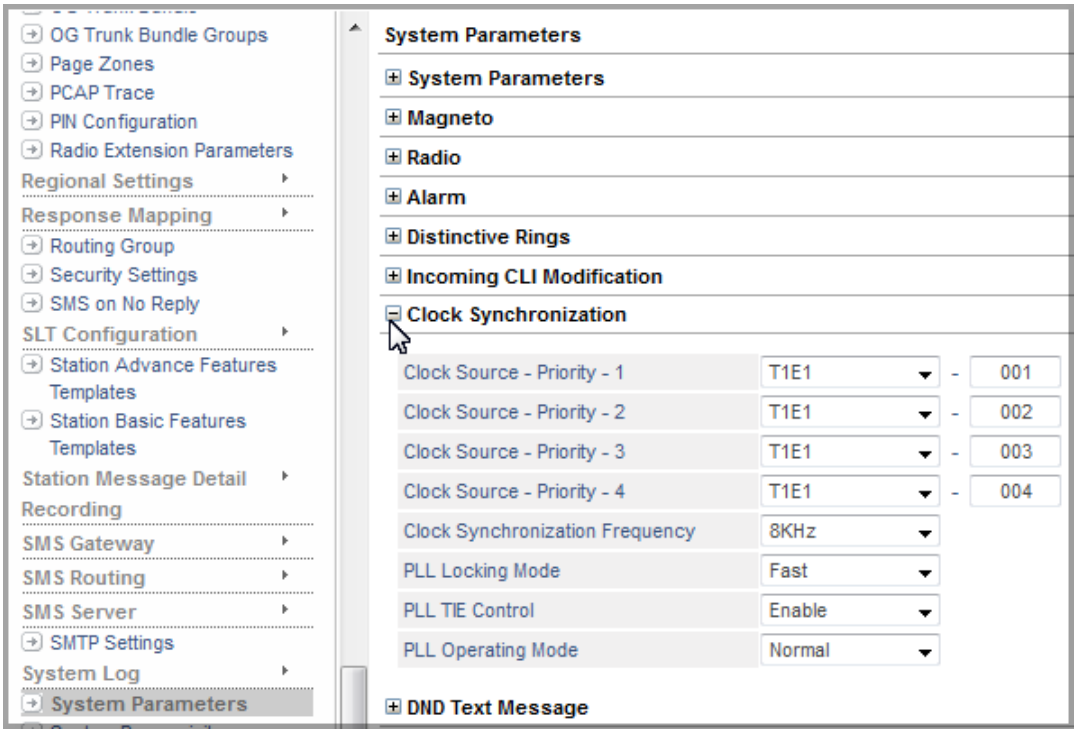
How to configure

Configuring Clock Synchronization using Jeeves

- Log in as System Engineer.

- Under **Configuration**, click **System Parameters**.

- Click **Clock Synchronization** to expand.



- **Clock Source - Priority 1to 4:** In the Priority levels, Clock Source Priority-1 to Clock Source Priority-4, select the Clock Source option as per your preference.

  By Default: Priority 1 is T1E1-001, Priority 2 is T1E1-002, Priority 3 is T1E1-003 and Priority 4 is T1E1-004.

- **Clock Synchronization Frequency:** Select Clock Synchronization Frequency in this field. You can select from the following options:
  - 8 KHz Derived
  - 8 KHz
  - 2.048 MHz
  - 1.54 MHz

  By Default, Clock Synchronization Frequency is set to8 KHz.

The system will restart when the frequency is changed.

- **PLL Locking Mode:** Depending upon the speed required for clock synchronization, select the speed for PLL Locking Mode. You can set PLL Locking Mode to either fast or slow. By default, it is slow.

- **PLL TIE Control:** You can enable or disable PLL TIE Control. By default, it is disabled.

- **PLL Operating Mode:** Select the PLL Operating Mode in this field. You can select from the following options:
  - Normal
  - Hold Over
  - Free Run

  By default, PLL Operating Mode is Normal.

You can program PLL TIE Control and PLL Operating Mode only using Jeeves.

Configuring Clock Synchronization using a Telephone

Clock Source for 'Master Clock Synchronization'

The SARVAM UCS supports 4 clock sources which can be programmed for the specific port.

Use the following command to configure the clock sources:
5341-Clock Source Index-Port Type-Port Offset
Where,
Port Offset is 01 to 32.
Clock Source Index is from 1 to 4.

Port Type	Meaning	Port Offset
05	T1E1	01-08
04	BRI	01-32
00	Null	000

0 is a valid port to follow internal clock.

Default Table for the clock source, programmed in the system

Clock Source Index	Port Type-Port Offset
1	T1E1-1
2	T1E1-2
3	T1E1-3
4	T1E1-4

The system checks this table for a master clock. If none of the ports is synchronized out of this table, the system gives priority to the internal clock. If any one port is synchronized, the system selects that port as a system clock master. Here index is given priority, that is, if the second port of this table is selected as clock master and suddenly first port is synchronized, then the system changes its master from 2nd port to first port. Now if first port has lost its synchronization then in this case again, the second port is selected as master clock of the system.

Clock Synchronization Frequency

Use following command to select Clock Synchronization Frequency:
5342-Clock Synchronization Frequency
Where,
Clock Synchronization Frequency is from 1 to 4.

System Clock Synchronization	Meaning
1	8 KHz Derived

System Clock Synchronization	Meaning
2	8 KHz
3	2.048 MHz
4	1.54 MHz

By Default, 'System Clock Synchronization' is 2.048 MHz for India and other countries except USA.
For USA, default 'System Clock Synchronization' is 1.54 MHz.

PLL Locking Mode

This parameter is used to select the suitable speed of the PLL locking in the system.

Use following command to select the 'PLL Locking Mode':
5343-PLL Locking Mode
Where,
PLL Locking Mode is from 1 to 2.

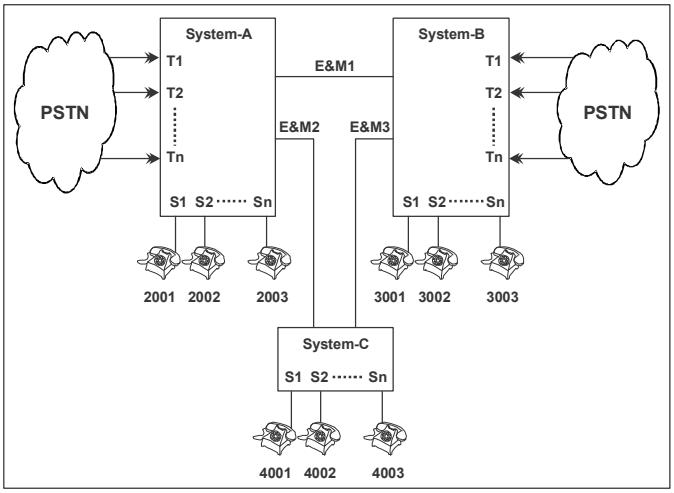
PLL Locking Mode	Meaning
1	Fast
2	Slow

By default, PLL Locking Mode is 'Slow'.

Closed User Group (CUG)

What's this?

To have private networks, few systems can be connected to each other using E&M, T1/E1, QSIG, as well as over the IP Network. The requirement demands that the systems connected to each other forming the network behave as a single group. The users need not dial a separate code to access an extension user of other system. The entire network should behave as a single unit. Users will not know whether they are dialing an extension number of their own system or of the other system. This is called Closed User Group.



In the above figure, 3 Systems are connected through E&M connectivity.

- T1 to Tn are trunk lines from the local central office (CO).
- S1 to Sn are extensions.
- E&M1 to E&M3 are E&M lines between the three Systems.

How it works

For Closed User Group, you must:

- have unique extension number in all the systems, that is, one cannot have extension number 2001 in System-A as well as in System-B or System-C.
- enable Closed User Group (CUG) in the Class of Service assigned to the extension users.

Few new words have been used to explain this application, each of these words have been explained below:

- **Closed User Group** Table**:** This table has the following parameters viz. Route Index, Route Code, OG Trunk Bundle Group, Strip Digit Count, Self Route check box, Apply Toll Control check box and Apply Call Cost check box. The closed user group programming works according to this table.

Index	Route Code	OGTBG	Strip Digit Count	Self Route	Dialed Digit Count	Apply Toll Control	Apply Call Cost
001							
002							
003							
:							
250							

- **Index:** Maximum 250 different routes (001-250) can be programmed.

- **Route Code:** Route code could be of maximum sixteen digits. Valid Digits: 0 to 9, * # A B C D F P, where P is Pause, F is Flash, A to D is DTMF Digits. Generally route code will be a truncated number of the extension numbers. For example in the figure given above, route code for System-B can be defined as ‘3’ and that for System-C can be defined as ‘4’.

  If System-B were having extension numbers from 3100 to 3199 and System-C were having extension numbers from 3200 to 3299 then route code for System-B can be defined as ‘31’ and that for System-C can be defined as ‘32’. If System-B were having extension numbers from 301 to 399 and 401 to 499 then two route codes can be defined for System-B viz. ‘3’ and ‘4’. Likewise for System-C.

- **OG Trunk Bundle Group:** An OG Trunk Bundle Group (OGTBG) is assigned to each route code. Whenever a call is to be made on that route, a free trunk from the OGTBG is selected and the extension number is dialed on it. The same logic of Rotation On/Off for trunk selection from the OGTBG is used. If rotation is OFF then always the first trunk in the OGTBG is selected. If it is busy then the next trunk in the group is selected. This helps to select an alternate route. Whereas if Rotation is ON then the trunks in the OGTBG are selected in round robin fashion.

- **Strip Digit Count:** For the Closed User Group application, configure this parameter as 0. This parameter is relevant when you are configuring “Closed User Group-With Exchange ID”.

- **Self-Route:** For the Closed User Group application make sure this is disabled. This parameter is relevant when you are configuring “Closed User Group-With Exchange ID”.

- **Dialed Digit Count:** When digits are dialed on the trunk, the system waits for inter digit timer after the last digit is dialed. In order to avoid this timer and number of digits dialed to be routed without further delay, count for the number of digits to be programmed in this field. If the number of digits received are equal to the parameters programmed then the number is dialed out immediately without waiting for the inter digit timer. If the number of digits dialed by the user are not equal to the digits programmed, the number is dialed after inter digit timer.

- **Apply Toll Control**: When Self Route check box is disabled, system will check this parameter.

  By default, the Apply Toll Control check box is enabled. The system will apply toll control to all the outgoing calls. Disable this check box, if you do not want to apply toll control to the CUG numbers dialed by you.

For detailed information, see “Toll Control”.

Apply Call Cost: By default, this check box is enabled and the system will calculate the cost of each call.

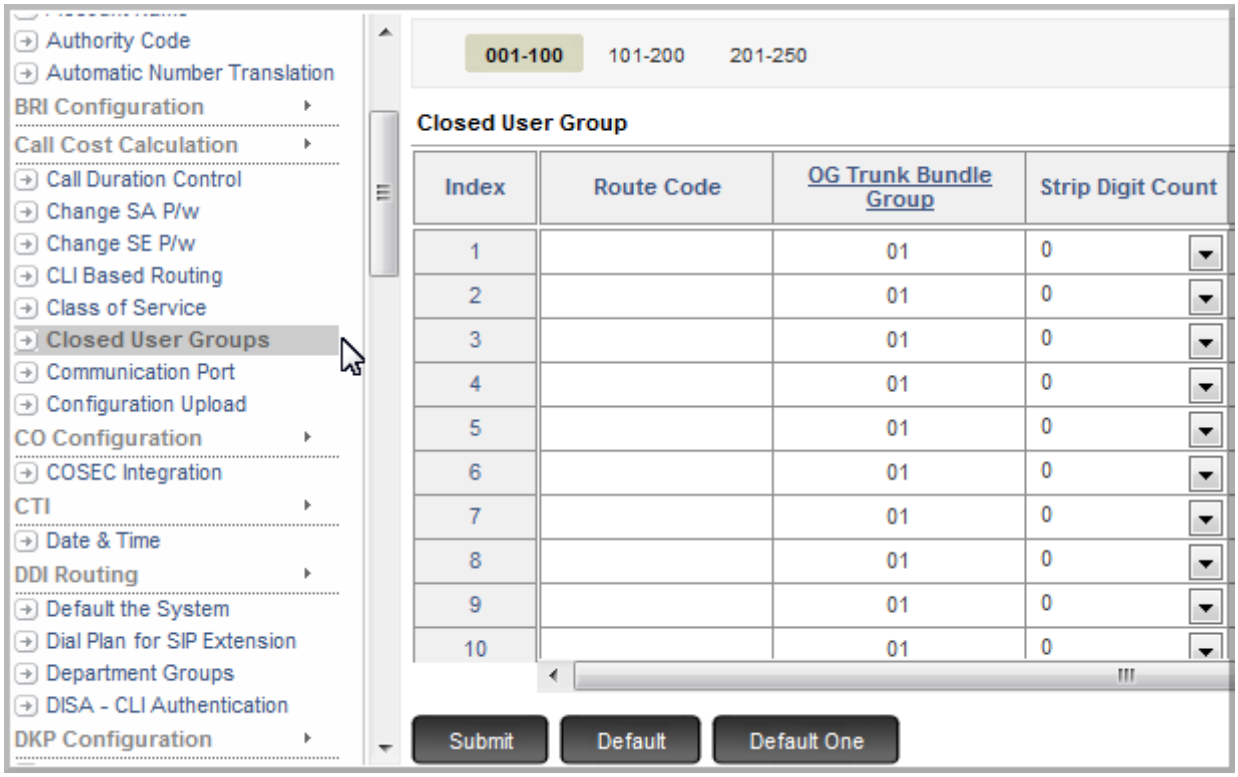
For certain calls (internal calls) you do not require the call cost calculation, clear the check box corresponding to these entries. You can also set the filter **Calls with units more than** to generate a report according to the Call Cost. For details, see “Station Message Detail Recording-Report”.

The SARVAM UCS has only one routing table. The same table is used for Closed User Group and Close User Group-With Exchange ID applications. Hence the table has to be programmed keeping the application in mind.

Configuring using Jeeves

To configure the Closed User Groups,

- Log in as System Engineer.
- Under **Configuration**, click **Closed User Groups**.



Against each Index configure the following parameters:

- **Route Code:** Route code could be of maximum sixteen digits. Valid Digits: 0 to 9, * # A B C D F P, where P is Pause, F is Flash, A to D is DTMF Digits. Generally route code will be a truncated number of the extension numbers. For example in the figure given above, route code for System-B can be defined as ‘3’ and that for System-C can be defined as ‘4’.

  If System-B were having extension numbers from 3100 to 3199 and System-C were having extension numbers from 3200 to 3299 then route code for System-B can be defined as ‘31’ and that for System-C

can be defined as ‘32’. If System-B were having extension numbers from 301 to 399 and 401 to 499 then two route codes can be defined for System-B viz. ‘3’ and ‘4’. Likewise for System-C.

- **OG Trunk Bundle Group:** An OG Trunk Bundle Group (OGTBG) is assigned to each route code. Whenever a call is to be made on that route, a free trunk from the OGTBG is selected and the extension number is dialed on it. The same logic of Rotation On/Off for trunk selection from the OGTBG is used. If rotation is OFF then always the first trunk in the OGTBG is selected. If it is busy then the next trunk in the group is selected. This helps to select an alternate route. Whereas if Rotation is ON then the trunks in the OGTBG are selected in round robin fashion.

- **Strip Digit Count:** It has no significance for Closed User Group application. But it has to be programmed as 0.

- **Self-Route:** It has no significance for Closed User Group application. But it has to be programmed as 0.

- **Dialed Digit Count:** When digits are dialed on the trunk, the system waits for inter digit timer after the last digit is dialed. In order to avoid this timer and number of digits dialed to be routed without further delay, count for the number of digits to be programmed in this field. If the number of digits received are equal to the parameters programmed then the number is dialed out immediately without waiting for the inter digit timer. If the number of digits dialed by the user are not equal to the digits programmed, the number is dialed after inter digit timer.

- **Apply Toll Control**: By default, this check box is enabled. The system will apply toll control to all the outgoing calls. Disable this check box, if you do not want to apply toll control to the CUG numbers dialed by you.

- **Apply Call Cost:** By default, this check box is enabled and the system will calculate the cost of each call.

  For certain calls (internal calls) you do not require the call cost calculation, clear the check box corresponding to these entries.

- Click **Submit** to save entries.

Configuring using a Telephone

The commands explained below should be referred as:
To configure a single port: ***XXXX-1***
To configure a range of ports: ***XXXX-2***
To configure all the ports: ***XXXX-****

Step 1
Use following command to configure route code:
4502-1-Route Index-Route Code-#*
4502-2-Route Index-Route Index-Route Code-#*
4502-*-Route Code-#*
Where,
Route Index is from 001 to 250.
Route Code is a sixteen digits string of numbers.

Use following command to clear a particular route code:
4502-1-Route Index-#*
4502-2-Route Index-Route Index-#*
4502-*-#*
Where,
Route Index is from 001 to 250.
By default, Program Route Code is Blank.

Step 2
Use following command to assign OG Trunk Bundle Group to the route code:
4503-1-Route Index-OG Trunk Bundle Group
4503-2-Route Index-Route Index-OGTBG
4503-*-OG Trunk Bundle Group
Where,
Route Index is from 001 to 250.
OG Trunk Bundle Group is from 01 to 32.
By default, OG Trunk Bundle Group is 01.

Step 3
Use following command to configure strip digit count for a route:
4504-1-Route Index-Strip Digit Count
4504-2-Route Index-Route Index-Strip Digit Count
4504-*-Strip Digit Count
Where,
Router Index is from 001 to 250.
Strip Digit Count is from 0 to 9.
By default, Strip Digit Count is 0.

Step 4
Use following command to configure self-route check box for a route:
4505-1-Route Index-Code
4505-2-Route Index-Route Index-Code
4505-*-Code
Where,
Route Index is from 001 to 250.

Code	Meaning
0	Disable Self-route check box
1	Enable Self-route check box

By default, Self-route check box is disabled.

Step 5
Use following command to configure maximum dialed digits to select router for a route code:
4506-1-Route Index-Dialed Digit Count
4506-2-Route Index-Route Index-Dialed Digit Count
4506-*-Dialed Digit Count
Where,
Route Index is from 001 to 250.
Maximum dialed digits is from 00 to 99.

Step 6

Use following command to clear an entry in a routing table:

4501-1-Route Index

4501-2-Route Index-Route Index

4501-*

Where,

Route Index is from 001 to 250.

Closed User Group-With Exchange ID

What's this?

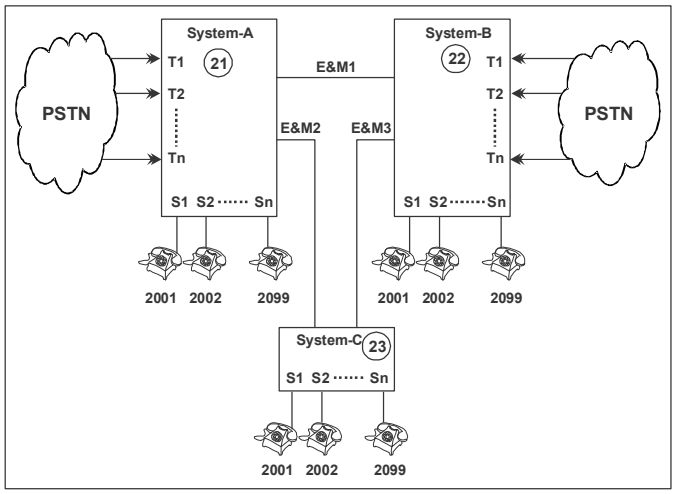
To have private networks, few systems can be connected to each other using E&M, T1/E1, QSIG, as well as over the IP Network. The requirement demands that the systems connected to each other forming the network behave as a single group. The users need not dial a separate code to access an extension user of other System. The entire network should behave as a single unit. The extension users will not know whether they are dialing an extension number of their own system or other system. This is called Closed User Group. However, it is possible that the Systems connected to form a network may have same extension numbers. Also, all the exchanges within the network may have their own identity (called Exchange ID). In such cases, the routing scheme (the routing table) has to be programmed keeping the Exchange ID (SID) in mind. This is known as Closed User Group-With Exchange ID.

This facility is generally used in PLCC Applications wherein new power extensions (and hence Systems) are added in the network. It is not feasible to have unique extension numbers throughout the network. In such cases, an Exchange ID is assigned to the newly added System and a routing table is programmed in the exchange. Also, the routing tables of other exchanges are modified to include the newly added exchange in the network.

Figure 1: Open User Group Application

In the below figure, 3 Systems are connected through E&M connectivity.

- T1 to Tn are trunk lines from the local central office (CO).
- S1 to Sn are extensions.
- E&M1 to E&M3 are E&M lines between the three Systems.



How it works

In this application, it is possible to have same extension numbers in two or more Systems of the network, but you must enable Closed User Group (CUG) in the Class of Service assigned to the extension users.

Few new words have been used to explain Closed User Group-With Exchange ID application, each of these words have been explained above.

- **Closed User Group Routing** Table**:** This table has following parameters viz. Route Index, Route Code, OG Trunk Bundle Group, Strip Digit Count, Self Route check box, Apply Toll Control check box and Apply Call Cost check box. The Closed User Group-With Exchange ID programming works according to this table.

Index	Route Code	OGTBG	Strip Digit Count	Self Route	Dialed Digit Count	Apply Toll Control	Apply Call Cost
001							
:							
250							

- **Index:** Maximum 250 different routes (001-250) can be programmed.

- **Route Code:** Route code could be of maximum sixteen digits. Valid Digits: 0 to 9, * # A B C D F P, where P is Pause, F is Flash, A to D is DTMF Digits. Generally, route code will be a unique number. The route code should not clash with any of the extension numbers of same System. For example in the figure given above, route code for System-A can be defined as ‘21’, route code for System-B can be defined as ‘22’ and that for System-C can be defined as ‘23’. This means that no extension in System-A can start with ‘22’ or ‘23’. Similarly, no extension in System-B can start with ‘21’ or ‘23’ and no extension in System-C start with ‘21’ and ‘22’.

- **OG Trunk Bundle Group:** An OG Trunk Bundle Group (OGTBG) is assigned to each route code. Whenever a call is to be made on that route, a free trunk from the OGTBG is selected and the extension number is dialed on it. The same logic of rotation On/Off for trunk selection from the OGTBG is used. If rotation is OFF then always the first trunk in the OGTBG is selected. If it is busy then the next trunk in the group is selected. This helps to select an alternate route. Whereas if rotation is ON then the trunks in the OGTBG are selected in round robin fashion.

- **Strip Digit Count:** This count signifies the number of digits to be stripped off while dialing/decoding a number. To elaborate: Consider figure 1. The requirement is that if extension 2001 of System-B dials 212002 and if E&M 1 is busy then the call should reach extension 2002 of System-A through alternate route. In this case the strip digit count of System-A should be programmed as 2 and that of System-B and System-C should be programmed as 0. Doing so, when extension 2001 of System-B dials 212002 and if E&M1 is busy then the call is routed through System-C. In this case, System-B dials 212002 on E&M3, System-C receive this code and dials out the same code, that is, 212002 on E&M2 without striping of any digit. On receiving 212002, System-A strips of two digits as per the programming and routes the call to extension 2002.

- **Self-Route:** This flag signifies that the digits being dialed are for the same System and are not to be dialed on the E&M trunk.

- **Dialed Digit Count:** When digits are dialed on the trunk, the system waits for inter digit timer after the last digit is dialed. In order to avoid this timer and number of digits dialed to be routed without further delay, count for the number of digits to be programmed in this field. If the number of digits received are equal to the parameters programmed then the number is dialed out immediately without waiting for the inter digit timer. If the number of digits dialed by the user are not equal to the digits programmed, the number is dialed after inter digit timer.

- **Apply Toll Control:** This parameter is not relevant as Self Route flag is enabled. This parameter is relevant when you are configuring “Closed User Group (CUG)”.

- **Apply Call Cost:** By default, this check box is enabled and the system will calculate the cost of each call.

  For certain calls (internal calls) you do not require the call cost calculation, clear the check box corresponding to these entries. You can also set the filter **Calls with units more than** to generate a report according to the Call Cost. For details, see “Station Message Detail Recording-Report”.

  Please note that the SARVAM UCS has only one routing table. The same table is used for Closed User Group and Closed User Group-With Exchange ID. Hence the table has to be programmed keeping the application in mind.

How to configure

- To configure the Closed User Groups with Exchange ID, refer “Closed User Group (CUG)”.
- To configure the OG Trunk Bundle Group (OGTBG), refer “OG Trunk Bundle Group”.
- To configure the E&M template parameters, refer “E&M Feature Template”

Communication Ports

What's this?

SARVAM UCS supports two communication ports for ETERNITY GENX platform. The first COM Port is inbuilt on the CPU Card and the other can be used by connecting the USB to COM[275] converter in the "External USB Port (Device Port) 3.0" of the CPU Card.

SARVAM UCS supports only one communication port for ETERNITY LENX and MENX platform. This can be used by connecting the USB to COM converter in the "External USB Port (Device Port) 3.0" of the CPU Card.

SARVAM UCS supports serial, asynchronous, DB-9 connector for the Communication Ports.

A Communication Port (COM or USB to COM) is used for the following facilities:

- Programming SARVAM UCS using a PC
- PMS Interface (see SARVAM UCS Hospitality System Manual)
- SMDR Reports
- SMDR Online
- SMDR Posting
- System Activity Log
- System Fault Log
- Hotel-Motel Activity Log (see SARVAM UCS Hospitality System Manual)

A Communication Port is necessary for programming SARVAM UCS using a PC, whereas for other above listed facilities, Communication Port may or may not be used[276].

How to configure

In order for each of the above listed facilities to work, a Communication Port (COM or USB to COM Port) must be assigned first as the 'Destination Port' and the attributes of the COM or USB to COM Port of SARVAM UCS and the Communication Port of the PC to which it is connected must programmed to match.

Communication Port as Destination Port

COM or USB to COM Port can be assigned as Destination Port using Jeeves as well as by dialing SE commands from a Telephone. For instructions, refer the individual topics:

- PMS Interface
- *"Station Message Detail Recording-Report"*
- *"Station Message Detail Recording-Online"*
- *"Station Message Detail Recording-Posting"*
- *"System Activity Log"*
- *"System Fault Log"*
- Hotel-Motel Activity Log
- Print Check-out Report
- *"System Debug"*
- *"Loop Back Tests"*

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275. ETERNITY PENX supports 1 COM Port. This can be used by connecting the USB to COM Converter to the External USB Port.
276. PMS can be interfaced on the LAN/WAN Port. For SMDR-Posting LAN/WAN Port can be used.

You may assign the destination port as Communication Port (COM Port or USB to COM) to multiple facilities as mentioned above, but at a time only one facility will be served by each Communication Port. The facility assigned first will be served first.

Communication Port Attributes

The functioning of the COM or USB to COM Port is controlled by the following attributes:

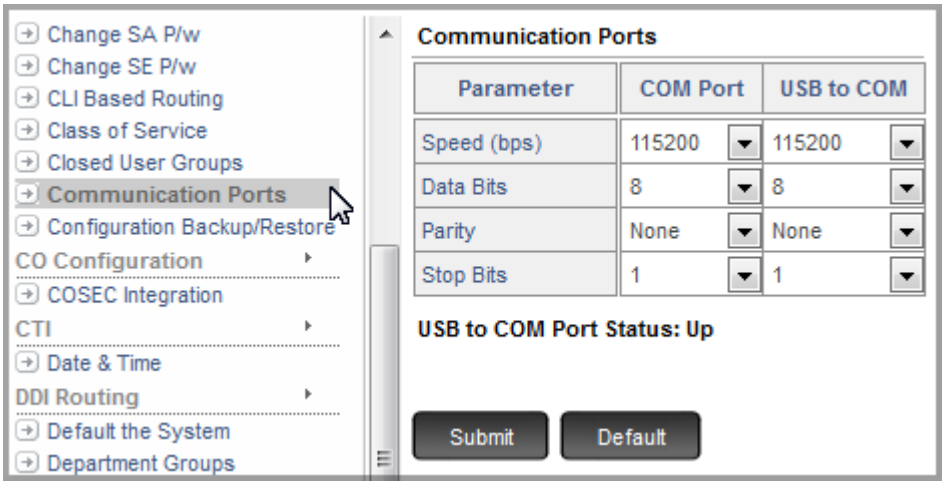
- Speed in bps.
- Number of data bits.
- Number of stop bits.
- Parity

These attributes must be programmed keeping in mind the application for which the communication port is used (for instance, Programming through PC, generating SMDR Reports, etc.)

The Communication Port attributes can be changed using Jeeves and dialing SE commands from a telephone.

Changing the Communication Port attributes using Jeeves

- Log in as System Engineer.
- Under **Configuration**, click **Communication Ports** to open the page.



- Set the desired values for COM Port and USB to COM Port:
  - Speed (Bps)
  - Data Bits
  - Parity
  - Stop Bits
- Click **Submit** to save changes.

When the USB to COM converter is connected to the External USB Port, the same will be notified on this page.

Any one of the following USB to COM status will be displayed:

Status	Description
Up	This will be displayed when the USB to COM converter is connected, the port is available and functioning.
Down	This will be displayed when the USB to COM converter is not connected or not detected. Try re-plugging the device. The status will be updated accordingly.
Fail	This will be displayed when the USB to COM converter is connected but the internal registration fails. You may change any of the above parameter values or set the page to default to re-initiate the registration process. The status will be updated accordingly.

Changing Communication Port attributes using a Telephone

The SE Commands mentioned below are applicable only for COM Port and not for USB to COM Port.

- Enter SE mode from a DKP/SLT.

To set Data Transfer rate for COM Port, dial:

- ***3201-Port-Speed***
  Where,
  Port is
  1 for COM Port
  Speed is
  0 for 1200 bps
  1 for 2400 bps
  2 for 4800 bps
  3 for 9600 bps
  4 for 19200 bps
  5 for 38400 bps
  6 for 57600 bps
  7 for 115200 bps

  By default, Data Transfer Speed is 115200 bps.

To set Data Bits for COM Port, dial:

- ***3202-Port-Data Bits***
  Where,
  Port is
  1 for COM Port

  Data Bits are
  0 for 7 data bits
  1 for 8 data bits

By default, Data Bits are 8.

To set Parity for COM Port, dial:

- ***3203-Port-Parity***
  Where,
  Port is
  1 for COM Port
  Parity is
  0 for None
  1 for Odd
  2 for Even
  3 for Mark
  4 for Space

  By default, Parity is set as 'None'.

To set Stop Bits for COM Port, dial:

- ***3204-Port-Stop Bits***
  Where,
  Port is
  1 for COM Port

  Stop Bits are
  0 for 1 stop bit
  1 for 2 stop bit

  By default, Stop Bits are 1.

To assign default parameters to COM Port, dial:

- ***3210-Port***
  Where,
  Port is
  1 for COM Port

- Exit SE Mode.

Example 1: Configure COM Port parameters with these values:

- Speed = 9600 bps.
- Data Bits = 8 bits.
- Parity = None.
- Stop Bits = 1.

To do this, dial the following SE commands:

3201-1-3
3202-1-1
3203-1-0
3204-1-0

For connecting SARVAM UCS to a Computer/programming SARVAM UCS through a computer set the Communication Port parameters to the following values (recommended):
Speed = 9600 bps.

Data Bits = 8.
Parity = None.
Stop Bits = 1.

How to use

For SARVAM UCS to communicate with a PC through the Communication Port, it must be connected with the Communication Port of the PC.

Connecting SARVAM UCS with a Computer over COM/USB to COM Port

To connect the SARVAM UCS with a PC over COM or USB to COM Port, use the communication cable supplied by Matrix[277]. This communication cable is provided with DB-9 female connectors on both ends.

You may connect any end to the SARVAM UCS and the other end to the PC. If the PC supports only USB connectivity, use a USB-to-DB-9 converter of any standard make.

Refer the following Table for pin-out details of the COM Port.

Pin No.	Signal Name
1	NC
2	Receive Data (RXD)
3	Transmit Data (TXD)
4	NC
5	Ground (GND)
6	NC
7	NC
8	NC
9	NC

277. This cable is supplied as an optional item. Contact your Matrix Dealer or the company to obtain this cable.

Conference-3 Party

What's this?

SARVAM UCS offers three types of conference calls: Conference-3 Party, “Conference Dial-In”, and “Conference-Multiparty”.

Conference-3 Party (also referred to as Three-Way Calling) is a telephone call, in which the calling party can have two other persons participate in the call.

A 3-Party Conference is initiated by dialing the number of the first person one wishes to talk to. The first person is informed about the conference and put on Consultation Hold. The number of the second person one wishes to talk to is dialed. When the second person answers, s/he is informed about the conference. Three-way speech is established by pressing Flash-*3.

An already connected two-way speech can be converted into a conference by adding a second person, without disconnecting the call with the first person.

Thus, a 3-Party Conference may be planned or conducted on the spur of the moment.

A 3-Party Conference can be conducted with extensions of SARVAM UCS and between extensions and external numbers.

It is also possible to conduct an Unsupervised 3-Party Conference, wherein the operator (assistant) connects two trunks through the system and withdraws from the three-way speech. For confidential discussions where the parties need to know that the operator has withdrawn, the system provides facility to play beeps.

Beeps will be played:

- when the operator is present in the conference at regular intervals
- when the operator has left the conference

The maximum number of simultaneous 3-Party conferences supported by each model is mentioned in the table below:

Model	Max. number of simultaneous 3-Party conferences supported
ETERNITY LENX	15
ETERNITY MENX	15
ETERNITY GENX	20
ETERNITY PENX	16

How it works

A, B, C are extensions.
D and E are external numbers.

3-Party Conference between extensions

- A is in speech with B.
- A and B want to include C in their conversation.
- A presses the ‘Conference’ Key. B is put on Consultation Hold.
- A gets feature tone. B gets on-hold music.
- A dials C's extension number. A gets ring back tone.
- A is in speech with C. B cannot hear their conversation.
- A presses the ‘Conference’ key a three-way speech is established.
- A, B, and C are now in speech.

- Any of them can disconnect to withdraw from the conference.
- If C disconnects, A and B will be in two-way speech.
- A and B can carry on the conversation or can have a conference with another trunk (external number) or with another extension.

3-Party Conference between two extensions and a trunk

- A is in speech with B.
- A and B want to include D in their conversation.
- A presses the ‘Conference’ Key. B is put on Consultation Hold.
- A gets feature tone. B gets on-hold music.
- A grabs a Trunk and dials D's extension number. A gets ring back tone.
- A is in speech with D. B cannot hear their conversation.
- A presses the ‘Conference’ Key to enable three-way speech.
- A, B, and D are now in speech.

- Any of them can disconnect to withdraw from the conference.
- If B disconnects, A and D will be in two-way speech.
- A can now conduct a conference with another extension or trunk.

3-Party Conference between an extension and two trunks

- A is in speech with D.
- A and D want to include E in their conversation.
- A presses the ‘Conference’ Key. D is put on Consultation Hold.
- A gets feature tone. D gets on-hold music.
- A grabs a Trunk and dials E's number.
- A is in speech with E. D cannot hear their conversation.
- A presses the ‘Conference’ Key to enable three-way speech.
- A, D, and E are now in speech.

- Any of them can disconnect to withdraw from the conference.
- If A disconnects, D and E are now in two-way speech.

3-Party Confidential Conference established by Operator

- F is in speech with D.
- F wants to include E in their conversation.
- F presses the ‘Conference’ Key. D is put on Consultation Hold.
- F gets feature tone. D gets on-hold music.
- F grabs a Trunk and dials E's number.
- F is in speech with E. D cannot hear their conversation.
- F presses the ‘Conference’ Key to enable three-way speech.

- F, D, and E are now in speech.
- D and E will hear Assistant Present beeps[278] as long as F is present in the conference.
- For confidential conversation between D and E to be initiated, F must leave the conference.
- D and E will hear Assistant Leave beeps as soon as F leaves the conference.
- D and E are now in two-way speech.

- *A, B and C are in speech. When A disconnects, either B and C are also disconnected or speech is established between them depending on the option you select in **If the Extension creating 3 party conference, disconnects during Conference** in the System Parameters.*
- *The Conference can be broken only by the master DKP/Extended IP Phone that has initiated the Conference.*
- *If a call put on hold is to be included in a Conference, it must be retrieved first.*
- *If all the parties to the conference are SIP Extensions/Trunks and if the initiator of the Conference goes on-Hook during the conference, the other parties will still remain in conversation. This is known as Transfer on Conference Hangup.*

How to configure

For this feature to work, the feature 'Conference' must be enabled in the Class of Service group of the extensions that are to be allowed this feature.

For confidential conference to be established by Operator(Assistant), you must:

- Enable the **Play beeps when Assistant present in Multiparty Conference** check box. For instructions, see “System Parameters”.
- Enable the **Play beeps when Assistant leaves the Conference** check box. For instructions, see “System Parameters”.
- Configure the desired interval time in **Conference – Assistant Present Beep Interval (sec).** For instructions, see “System Timers and Counts”.
- Make sure you select **Station Type** as **Assistant** for the Operator Extension. For instructions, see “Configuring SLT Extensions”, “Configuring DKP Extensions”, “Configuring Matrix SPARSH VP248”, “Configuring Matrix SPARSH VP310”, “Configuring Matrix SPARSH VP330”, “Configuring Matrix SPARSH VP210”, “Configuring Matrix SPARSH VP510”, “Configuring Matrix VARTA ADR100/AMP100 UC Clients” and “Configuring Matrix VARTA WIN200 UC Client”.

If extension users at remote locations are to be allowed to initiate the 3-party conference, “Direct Inward System Access (DISA)” or “Auto Attendant” must be enabled on the trunk on which their call lands.

3-Party Conference in Class of Service

In the default factory settings, Station Basic Feature Template Number 01 is assigned to all the extensions of the SARVAM UCS. The Station Basic Feature Template 01 is assigned CoS group 01 which has 'Conference' enabled. So, all extensions of the SARVAM UCS can make Conference calls.

If you want to deny 3-Party Conference to selected extensions, follow these steps:

1. Define a CoS group with 'Conference' disabled.
2. Prepare a Station Basic Feature Template with this CoS group applicable in all the “Time Zones”.

278. The total Beep Time = Beep Cadence + Wait Timer. The Wait Timer(non-programmable) is fixed as 2 seconds by the system.

3. Assign this new Template to the extensions to which Conference is to be denied.

Refer the topics “Class of Service (COS)” and “Station Basic Feature Template” for detailed instructions and programming.

The feature 'Conference' in the Class of Service also includes Dial-In and Multi-party Conference. Extensions that are denied 'Conference' in their Class of Service will not be allowed all three types of conferences - 3-Party, Dial-In and Multi-party Conference.

How to use

For EON and Extended IP Phone Users

- Speech with Party 1 on trunk/extension.
- Press ‘Conference’ Key. Party 1 put on Consultation Hold.
- Dial the number of Party 2.

If Party 2 is a trunk,

- Dial Trunk Access Code, to grab a trunk.
- You get Trunk dial tone.
- Dial telephone number of Party 2. You get ring back tone.
- Speech with Party 2.
- Press the 'Conference' Key.
- Three-way speech is established.

When the Conference is established, the Conference Key LED will glow continuously and you will get a message “Conference” on your phone display.

For SLT Users

- Speech with Party 1 on trunk/extension.
- Dial Flash. You get Feature tone. Party 1 put on Consultation Hold.
- Dial the number of Party 2.

If Party 2 is a trunk,

- Dial Trunk Access Code to grab a trunk. You get Trunk dial tone.
- Dial telephone number of Party 2. You get ring back tone.
- Speech with Party 2.

- Dial Flash-*3.
- Three-way speech is established.

Conference-Multiparty

What's this?

Like the Dial-In Conference, a Multi-party conference allows speech between more than three participants.

The key difference between Dial-In and Multi-party conference is that in a Dial-In conference participants can include themselves in the conference by dialing into it without assistance, whereas in a Multi-party Conference the party initiating the conference must include the participants by dialing their numbers.

In a 3-party Conference, when you add the fourth participant, a Multiparty Conference is initiated.

A Multiparty conference may be

- between extensions
- between extensions and trunks, that is, external numbers.

Any participant in a Multiparty Conference can Include a party, Remove a party, Leave a conference temporarily or can Cancel a conference. When any participant is included in the conference, the system plays a beep to indicate the inclusion.

A conference may also include the Operator(Assistant). For confidential discussions, where the parties need to know that the operator has withdrawn, the system provides facility to play beeps.
Beeps[279] will be played:

- when the operator is present in the conference at regular intervals
- when the operator has left the conference

External callers can initiate multiparty conference using "Direct Inward System Access (DISA)".

Depending on the model you are using, refer below table for the details regarding the multiparty conferences supported.

Model	Maximum conference participants	Maximum simultaneous conferences (If all the conferences involve 3 parties)	Maximum parties included in a conference (If all the parties are in a single conference)
ETERNITY LENX	45	15	21
ETERNITY MENX	45	15	21
ETERNITY GENX	62	20	21
ETERNITY PENX	48	16	15

How it works

A, B, C, and D are extension users.
E and F are external numbers.
A decides to hold a teleconference with B, C, D, E and F.
G is the Operator.

279. The total Beep Time = Beep Cadence + Wait Timer. The Wait Timer(non-programmable) is fixed as 2 seconds by the system.

Initiating a Multiparty Conference

- A has initiated a 3-party conference with B and C.
- If A dials the D's number followed by the 3-party conference code. D is included in the conference.
- If A dials the G's number followed by the 3-party conference code. G is included in the conference. The system plays a beep to indicate the inclusion.

  Now, the 3-party Conference is converted into a Multiparty Conference.
- The system will play beeps at regular intervals indicating that one of the participants present in the conference is the Operator (Assistant).

After the Conference has been initiated, conference participants can:

- **Include a Party in an on-going Multiparty Conference**.

  Internal as well as external callers can be included in an ongoing conference by the any of the participants.

  To include external callers, in this case, E and F. A must dial the Trunk Access Code followed by E's number and the 3-party conference code, Similarly F can be included.

- **Remove a Participant from an on-going Multiparty Conference**.

  Only DKP or Extended IP Phone users can remove another participant from the Multiparty Conference. The DKP/Extended IP phone displays the numbers of all the participants, select the number of the participant to be removed from this list.

  If the last Operator (Assistant) in the conference leaves the group, a beep will be played.

- **Temporarily Leave a Multiparty Conference**.

  Any participant in a conference can dial the Temporarily Leave Conference code to leave the conference for a short time period.

If all participants, Temporarily Leave the conference one-by-one, the system will start the 'Release Conference if idle for more than (Minutes) Timer'. This Timer is programmable, and by default it is set to 002 Minutes. If no participant rejoins the conference before the expiry of this Timer, the system will free the resource occupied by the conference on the conferencing circuit.

- **Rejoin a Multiparty Conference**.

  The participants who leave temporarily can rejoin the conference at a later stage by going Off-Hook and dialing the Rejoin Conference conference code.

- **Permanently Leave from a Multiparty Conference**.

  Participants in a conference can exclude themselves by going ON-Hook. Once any participant goes ON-Hook, he/she cannot rejoin the conference.

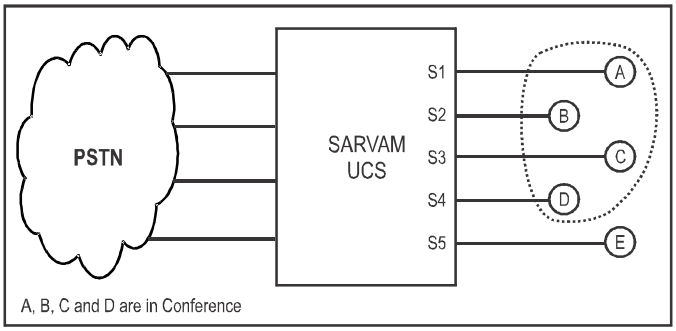
- **Cancel a Multiparty Conference**.

  Any participant in a conference can dial the Cancel conference code to end the conference.

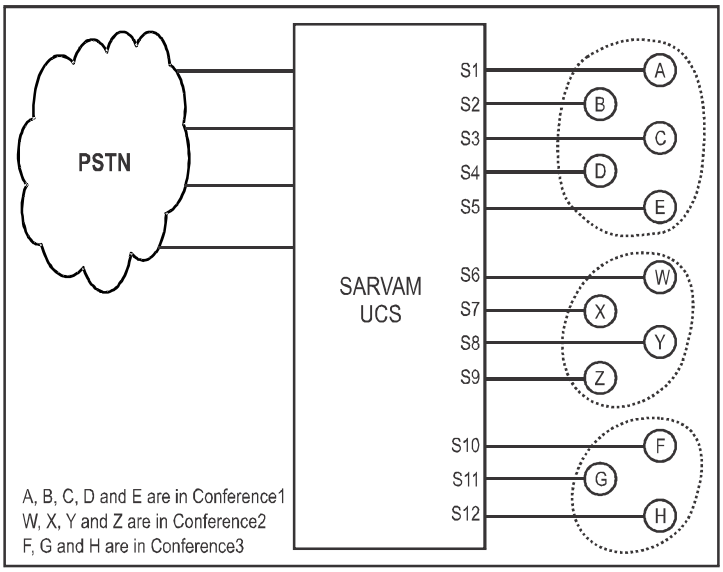
All participants will get Error Tone and the system resource occupied by the conference will be freed.

Examples of Multi-party Conferences:

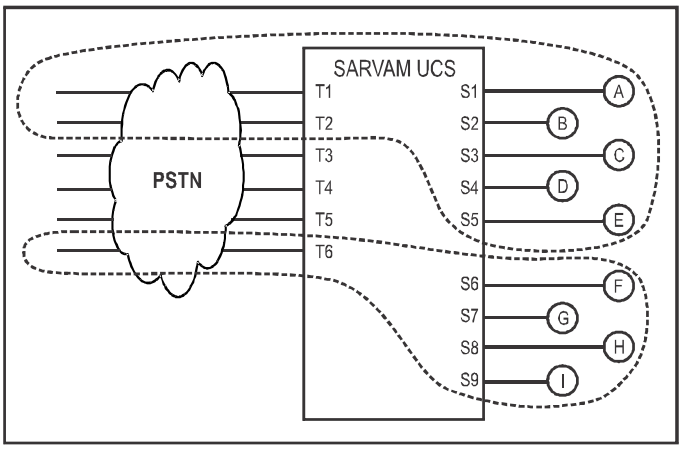
A Multiparty conference between extensions.



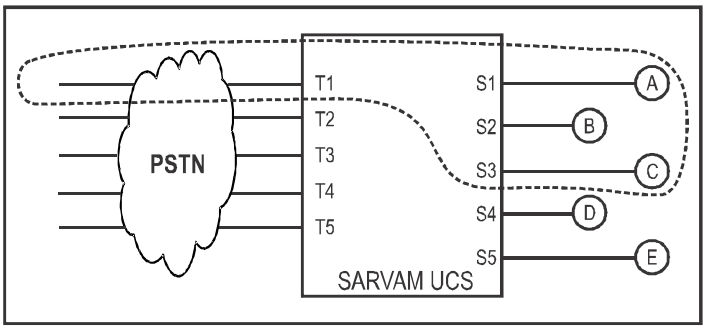
Simultaneous Multi Party conferences between extensions.



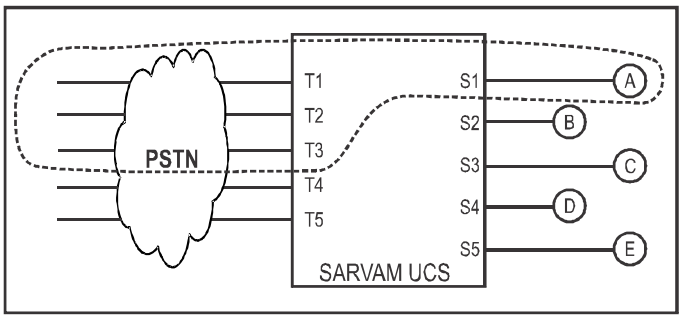
Multiple Multiparty Conferences between trunks and extensions.



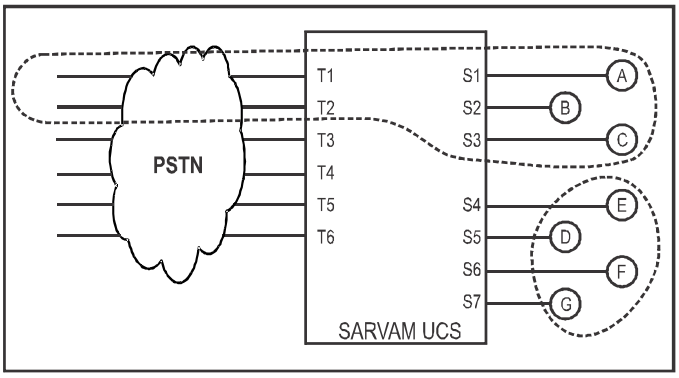
Multiparty Conferences between one trunk and a few extensions.



Multiparty Conference between a few trunks and one extension.



Simultaneous Multiparty Conference between a few trunks and extensions and between extensions.



How to configure

To provide this feature to extensions,

- You must enable the feature 'Conference' in the"Class of Service (COS)" of the extensions in their "Station Basic Feature Template". By default, this feature is enabled on all extensions, so all extensions can use this feature.

The feature 'Conference' in the Class of Service also includes 3-Party and Multi-party Conference. Extensions that are denied 'Conference' in their Class of Service will not be allowed Dial-In as well as 3-Party and Multi-party Conference.

For confidential conferences, you must:

- Enable the **Play beeps when Assistant present in Multiparty Conference** check box. For instructions, see "System Parameters".
- Enable the **Play beeps when Assistant leaves the Conference** check box. For instructions, see "System Parameters".
- Configure the desired interval time in **Conference – Assistant Present Beep Interval (sec).** For instructions, see "System Timers and Counts".
- Make sure you select **Station Type** as **Assistant** for the Operator Extension. For instructions, see "Configuring SLT Extensions", "Configuring DKP Extensions", "Configuring Matrix SPARSH VP248", "Configuring Matrix SPARSH VP310", "Configuring Matrix SPARSH VP330", "Configuring Matrix SPARSH VP510", "Configuring Matrix SPARSH VP210", "Configuring Matrix VARTA ADR100/AMP100 UC Clients" and "Configuring Matrix VARTA WIN200 UC Client".

- If desired, you may also change default value of the ***Release Conference if Idle for more than (min.) Timer***. See "System Timers and Counts".

- If external parties are to be allowed to initiate or join the Conference, "Direct Inward System Access (DISA)" must be enabled on the trunk on which they call.

- You can program a DSS key for Terminating a Conference, Temporarily Leave/Rejoining a Conference, if required. Refer the topic "DSS Keys Programming" for instructions.

How to use

For EON and Extended IP Phone Users

To initiate multiparty conference:

- Dial the number of party 1.
- When you are in speech with party 1, press ‘Conference’ Key.
- Party 1 is put on hold and gets on-hold music.
- You get feature tone. Dial number of party 2.
- When you are in speech with party 2, press the ‘Conference’ Key.
- Party 2 is included in the conference. A 3-way speech is established.
- Press the ‘Conference’ Key. Dial the number of party 3.
- When in speech with party 3, press the ‘Conference’ Key.
- A Multiparty Conference is initiated. Press the ‘Conference’ Key.
- The Multiparty Conference menu appears on the LCD.
- Select the option Include Party and dial the number.
- When you are in speech with the party, press the ‘Conference’ Key.
- Repeat the above steps to add new participants in the conference (max. 21).

To include a party in a multiparty conference:

- After the Multiparty Conference has been initiated. Press the ‘Conference’ Key.
- The Multiparty Conference menu appears on the LCD.
- Select the option Include Party and dial the number.
- When you are in speech with the party, press the ‘Conference’ Key.
- Repeat the above steps to add new participants in the conference (max. 21).

To temporarily leave multiparty conference:

- While in Conference, press the ‘Conference’ Key.
- The Multiparty Conference menu appears on the LCD.
- Select the option ‘Leave Temporary’ and press the Enter Key.

  OR

- Press the DSS key assigned to Temporary Leave Conference/Rejoin Conference.

To rejoin multiparty conference:

- Press the ‘Conference’ Key.

  OR

- Press the DSS key assigned to ‘Temporary Leave’ Conference/Rejoin Conference.

To remove a party from multiparty conference:

- While in Conference, press the ‘Conference’ Key.
- The Multiparty Conference menu appears on the LCD.
- Select the option ‘Remove Party from Conf’
- The LCD displays the numbers of all the participants.
- Select the number of the participant you want to remove and press the Enter Key.

- The selected participant is disconnected.

To permanently leave from the multiparty conference:

- While in Conference, go ON-Hook.

To cancel multiparty conference:

- While in Conference, press the ‘Conference’ Key.
- The Multiparty Conference menu appears on the LCD.
- Select the option ‘Terminate Conference’ and press the Enter Key.

  OR

- Press the DSS key assigned to Terminate Conference.
- All the participants will get an Error Tone and the system resource occupied by the conference will be freed.

For SLT Users

To initiate multiparty conference:

- Lift handset.
- Dial number of party 1.
- Speech with party 1.
- Dial Flash.
- Dial number of party 2.
- Speech with party 2. Dial Flash-*3. A 3-way speech is established.
- Dial Flash.
- Dial number of party 3.
- Speech with party 3. Dial Flash-*3. A multiparty conference is established.

To include a party in a multiparty conference:

- Lift handset.
- Dial number of party 1.
- Speech with party 1.
- Dial Flash.
- Dial number of party 2.
- Speech with party 2. Dial Flash-*3. A 3-way speech is established.
- Dial Flash.
- Dial number of party 3.
- Speech with party 3. Dial Flash-*3.
- Repeat the steps to include the desired number of parties (max. 21).

To temporarily leave from the multiparty conference:

- While you are all in speech, dial Flash-191.

To rejoin the multiparty conference:

- Go OFF-Hook.
- Dial 191.

To permanently leave from the multiparty conference:

- Go ON-Hook.

To cancel multiparty conference:

- While you are all in speech, dial Flash-190.

SLT users cannot Remove any participant from the Multiparty Conference.

To use Multiparty Conference from DISA mode, you may see the instructions “Dial-In Conference using DISA” under “Conference Dial-In”.

Conference Dial-In

What's this?

Dial-In Conference is a multi-party conference held at a pre-defined time. Extension users can schedule a Dial-In Conference and inform other participants to join in the conference at the scheduled time. Dial-In Conference can be used to conduct client meetings or sales presentations, project meetings and updates, regular team meetings, and to communicate with coworkers who operate in different locations. Thus, this feature helps to increase productivity by saving time and cost of travel for out-of-office meetings.

A conference may also include the Operator (Assistant). For confidential discussions, where the parties need to know that the operator has withdrawn, the system provides facility to play beeps.
Beeps[280] will be played:

- when the operator is present in the conference at regular intervals
- when the operator has left the conference

Depending on the model you are using, refer below table for the details regarding the Dial-In conferences supported.

Model	Maximum conference participants	Maximum simultaneous conferences (If all the conferences involve 3 parties)	Maximum parties included in a conference (If all the parties are in a single conference)
ETERNITY LENX	45	15	21
ETERNITY MENX	45	15	21
ETERNITY GENX	62	20	21
ETRENITY PENX	48	16	15

How it works

- A Dial-In Conference can be scheduled by dialing the access code for Dial-In conference followed by the conference number and a password.
    - The **Conference Number** can be: **01** to **20**.

      The Conference Number must correspond with the number of simultaneous Dial-In Conferences supported by the ETERNITY GENX (01 to 20), ETERNITY LENX/MENX (01 to 15) and ETERNITY PENX (01 to 16). If a user dials a conference number other than this, system will play an Error Tone.
    - The **Conference Password** is a four digit number string. The default conference password is **1111** and must be changed before using this feature.

      To avoid unauthorized access, make sure the password is strong and is provided to the participants only.
- All the other participants must be informed about the conference.

280. The total Beep Time = Beep Cadence + Wait Timer. The Wait Timer (non-programmable) is fixed as 2 seconds by the system.

You can also have external callers join the conference by providing them the DISA login.

Let us understand how Dial-In Conference works with the following example.

- Extension user A wants to schedule a Dial-In Conference at 4:30 p.m. with B, C, D, E and F.
- B and C are extension users. C is an extension user who has been provided a DISA login to access an extension of SARVAM UCS.
- D, E and F are external parties.
- Any extension user can initiate the conference, in this case A initiates the conference.

Scheduling a Dial-In Conference

- A schedules a Dial-In Conference for 4.30 pm.
- A informs B, C, D, E and F about the conference and provides them conference number, for example, '1' and the password, '4040'.

If C wants to schedule a conference, C must log into his extension from DISA mode.

Initiating a Dial-In Conference

- Any extension user can initiate the Dial-In conference by dialing the feature access code for Dial-In conference followed by the two-digit conference number and the password, for instance: '01' and '4040'. In this case, A initiates the conference, the system plays a beep.

After the Conference has been initiated, conference participants can:

- **Join a Dial-In Conference**.

  In this, example, B can join the conference by dialing the feature access code to join the conference followed by the two-digit conference number and password. When B joins a beep will be played.

  C can join the conference from the DISA mode.

  When a new party joins the conference, the system plays beeps to the existing participants, to inform them of the new inclusion. Beeps are programmable (default: enabled).

- **Include a Party in an on-going Dial-In Conference**.

  Internal as well as external callers can be included in an ongoing conference by the any of the participants.

  To include external callers, in this case D, E and F, A must dial the Trunk Access Code followed by the their numbers.

- **Remove a Participant from an on-going Dial-In Conference**.

  Only DKP or Extended IP Phone users can remove another participant from the Dial-In conference. The DKP/Extended IP phone displays the numbers of all the participants, select the number of the participant to be removed from this list.

- **Temporarily Leave a Dial-In Conference**.

  Any participant in a conference can dial the Temporarily Leave Conference code to leave the conference for a short time period.

If all participants, Temporarily Leave the conference one-by-one, the system will start the 'Release Conference if idle for more than (Minutes) Timer'. This Timer is programmable, and by default it is set to 002 Minutes. If no participant rejoins the conference before the expiry of this Timer, the system will free the resource occupied by the conference on the conferencing circuit.

- **Rejoin a Dial-In Conference**.
  The participants who leave temporarily can rejoin the conference at a later stage by going Off-Hook and dialing the Rejoin Conference conference code.

- **Permanently Leave from a Dial-In Conference**.
  Participants in a conference can exclude themselves by going ON-Hook. Once any participant goes ON-Hook, he/she cannot rejoin the conference.

- **Cancel a Dial-In Conference**.

  Any participant in a conference can dial the Cancel conference code to end the conference.

  All participants will get Error Tone and the system resource occupied by the conference will be freed.

  The conference can also be can canceled by logging into the SA mode.

How to configure

To provide this feature to extensions,

- You must enable the feature 'Conference' in the"Class of Service (COS)" of the extensions in their "Station Basic Feature Template". By default, this feature is enabled on all extensions, so all extensions can use this feature.

The feature 'Conference' in the Class of Service also includes 3-Party and Multi-party Conference. Extensions that are denied 'Conference' in their Class of Service will not be allowed Dial-In as well as 3-Party and Multi-party Conference.

- If you want beeps to be played when any one joins the conference, enable ***Play Beep when Raid/ Conference/Dial-in Conference begins***. See "System Parameters", for instructions.

For confidential conferences, you must:

- Enable the **Play beeps when Assistant present in Multiparty Conference** check box. For instructions, see "System Parameters".
- Enable the **Play beeps when Assistant leaves the Conference** check box. For instructions, see "System Parameters".
- Configure the desired interval time in **Conference – Assistant Present Beep Interval (sec).** For instructions, see "System Timers and Counts".
- Make sure you select **Station Type** as **Assistant** for the Operator Extension. For instructions, see "Configuring SLT Extensions", "Configuring DKP Extensions", "Configuring Matrix SPARSH VP248", "Configuring Matrix SPARSH VP310", "Configuring Matrix SPARSH VP330", "Configuring Matrix SPARSH VP510", "Configuring Matrix SPARSH VP210", "Configuring Matrix VARTA ADR100/AMP100 UC Clients" and "Configuring Matrix VARTA WIN200 UC Client".

- If desired, you may also change default value of the ***Release Conference if Idle for more than (min.) Timer***. See "System Timers and Counts".

- If external parties are to be allowed to initiate or join the Conference, “Direct Inward System Access (DISA)” must be enabled on the trunk on which they call.

- You can program a DSS key for Dial-In Conference, Terminating a Conference, Temporarily Leave/ Rejoining a Conference, if required. Refer the topic “DSS Keys Programming” for instructions.

How to use

For EON and Extended IP Phone Users

To Schedule a Dial-In Conference:

- Go OFF-Hook.
- Press the DSS key assigned for Dial-In Conference.
  OR
- Dial *19.
- The Dial-In Conference menu appears on the LCD.
- Select the option ‘Schedule a Conf’ and press the Enter Key.
- Enter Conference Number on the prompt.
- Enter Conference Password on the prompt.
- You get confirmation tone and the message 'Conf <number> Scheduled' on your phone's display.
- Go ON-Hook.
- Call all participants and inform them of the time of the Dial-In conference, the Conference Number and Password.

To join the Dial-In Conference:

- Go OFF-Hook.
- Press the DSS key assigned for Dial-In Conference.
  OR
- Dial *19.
- The Dial-In Conference menu appears on the LCD.
- Select the option ‘Include in Schd Conf’ and press the Enter Key.
- Enter the Conference Number on the prompt.
- Enter the Conference Password on the prompt. If you enter the wrong password, you get the message 'Check Conf P/w' on your phone's display.
- Conference is initiated.
- You will hear speech if any other participant has joined it. You will hear silence if no other participant has joined it.
- If you hear silence, wait for others to join in.
  OR
- Include the other participants in the conference, if you initiated the conference.

To include a party in the Dial-In conference:

- After initiating the conference and while you are all in speech, press the ‘Conference’ Key.
- The Multiparty Conference menu appears on the LCD.
- Select the option ‘Include Party in Conf’ and dial the desired number to be included in the conference. You get Ring Back Tone.
- When you are in speech with the party, press the ‘Conference’ Key The party is included in the conference.

To remove a participant from the Dial-In conference:

- While in Conference, press the ‘Conference’ Key.
- The Multiparty Conference menu appears on the LCD.

- Select the option ‘Remove Party from Conf’.
- The numbers of the participants appear on your phone's display.
- Scroll to select the participant you want to remove.
- Press 'Enter' key.
- You get confirmation tone. The extension user is now excluded from the conference.

To temporarily leave Dial-In conference:

- While in Conference, press the ‘Conference’ Key.
- The Multiparty Conference menu appears on the LCD.
- Select the option ‘Leave Temporary’ and press the Enter Key.
  OR
- Press the DSS key assigned to Temporary Leave Conference/Rejoin Conference.

To rejoin Dial-In conference:

- Press the ‘Conference’ Key.
  OR
- Press the DSS key assigned to ‘Temporary Leave /Rejoin Conference’.

If you have configured a DSS key for Temporary Leave and you leave the Conference by pressing the DSS key, to Rejoin the Conference press the DSS key again.

To permanently leave from the Dial-In conference:

While in Conference, go ON-Hook.

To cancel Dial-In conference:

- While in Conference, press the ‘Conference’ Key.
- The Multiparty Conference menu appears on the LCD.
- Select the option ‘Terminate Conference’ and press the Enter Key.
  OR
- Press the DSS key assigned to Terminate Conference.
- All the participants will get an Error Tone and the system resource occupied by the conference will be freed.

To cancel a Dial-In Conference from the **System Administrator (SA) Mode**:

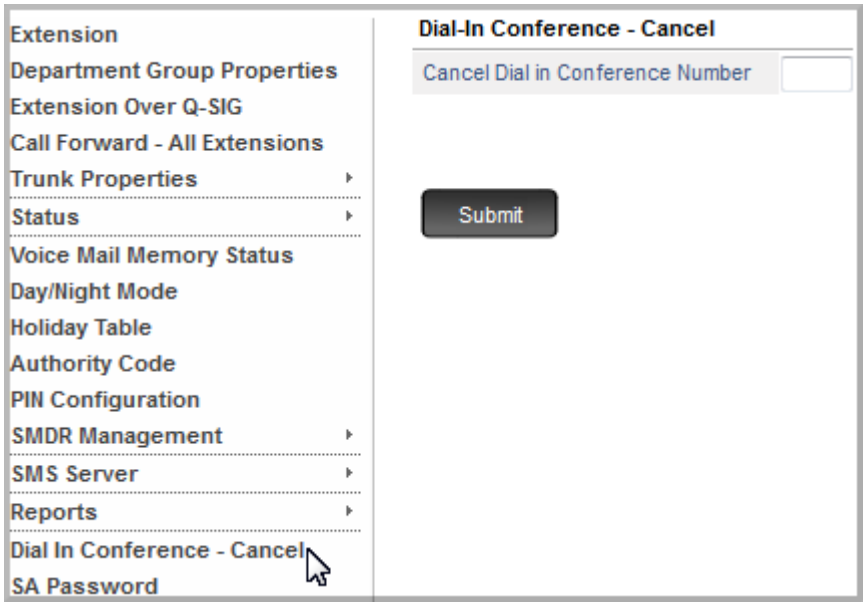
- Enter SA mode from a DKP/SLT/Extended IP Phone.
- Dial 1072-026-Conference Number.
- You get confirmation, the conference is released.

You can also cancel a Dial-In Conference from System Administrator (SA) Mode using Jeeves. To do this,

- Open Jeeves.

- Log in as System Administrator.

- Click the **Dial-In Conference - Cancel** link.



- Enter the two-digit conference number (01 to 20) which you want to cancel in the **Cancel Dial-In Conference Number** field.
- Click **Submit** button.
- Log out of Jeeves.

For SLT Users

- Lift the handset.
- You get dial tone.

Make sure you enter the two-digit Conference Number — 01 to 20.

To schedule a Dial-In conference:

- Dial *191-Conference Number-Conference Password.
- You get confirmation tone.
- Replace Handset.
- Inform all intended participants about the time, conference number and password.

To initiate the Dial-In conference:

- Dial *192-Conference Number-Conference Password.
- Talk, if you hear speech, or wait for others to join in.
  OR
- Include any participant in the conference.

To join the Dial-In conference:

- Dial *192-Conference Number-Conference Password.
- Talk, if you hear speech, or wait for others to join in.
  OR
- Include any participant in the conference.

To include a participant in Dial-In conference:

- After initiating the conference, or when in speech,
- Dial Flash.
- Dial the number of the desired party.
- Speech with party, dial Flash-*3.

To temporarily leave from the Dial-In conference:

- While you are all in speech, dial Flash-191.

To rejoin the Dial-In conference:

- Go OFF-Hook.
- Dial 191.

To permanently leave from the Dial-In conference:

- Go ON-Hook.

To cancel a Dial-In conference:

- While you are all in speech, dial Flash-190.

SLT users cannot Remove any participant from the Dial-In Conference.

Dial-In Conference using DISA

To schedule a Dial-In Conference:

- Dial a DISA enabled Trunk
- Dial DISA Login Code 1079-Extension Number-User Password, if PIN Authentication is required.
- After DISA Login beeps, dial *191-Conference Number-Conference Password.

To initiate or join a Dial-In Conference:

- After DISA Login beeps, dial *192-Conference Number-Conference Password.

To include a party midway of the Conference:

- Dial #2.
- You get dial tone.
- Dial the extension number to be included in the conference. You get Ring Back Tone.
- When you are in speech with the extension user, dial #2.
- You get feature tone, and the extension user is played music on hold.
- Dial *3. Both of you are now included in the conference.

To temporarily leave the Conference:

- Dial Flash-191.

To rejoin the Conference

- Go Off-Hook Dial 191.

To permanently leave the Conference:

- Dial #0 to go ON-Hook.
- Dial #0#9 to end DISA session.

When you enter DISA mode, you get beeps, dial digits before the DISA Inactivity Timer elapses.

Never dial 'Flash' when in DISA mode, you will get disconnected.

Keep dialing any digit to continue the conference.

See "Direct Inward System Access (DISA)" to know more.

Join Dial-In Conference using VMAA

Users can also Join a Dial-In Conference using the trunks on which the Voice Mail Auto Attendant is enabled. To know more refer to "Join Conference Dial-In using VMS". After joining the conference users can Temporary Leave, Rejoin or Permanently Leave the Conference also.

Conflict Dialing

What's this?

You may recall that “Access Codes” are dialed at different call phases. No two Access Codes must be the same in the same call phase.

For example, the same access code cannot be used for two different features like Call Forward and Redial, since both these features are invoked in the 'Dial' phase. Similarly, Station (Extension) and Logical Group Codes too must be unique and should not match with any of the features invoked in the 'Dial' phase.

However, SARVAM UCS allows overlaps within Feature Codes and “Flexible Numbers” (Station Codes). One Feature Access Code can be a part of (subset) another code, for example, 4, 41, 412; Flexible Numbers of extensions can be 201, 2011 etc.

So, when such overlapping access codes are dialed, the system matches the first digit. On finding more than one Access code starting with the same digit, the system will not know how to interpret the instruction and act accordingly.

Conflict Dialing feature resolves this confusion. When an access code that is a subset of any other access code is dialed, the system waits for some time for the extension user to dial the next digit. If the user does not dial any digit within that time, the system interprets it as the smaller Access Code, and invokes the associated feature.

The time for which the system waits for the next digit to be dialed before resolving the Access Codes is called "Conflict Dialing Timer". This timer is set to 2 seconds and is programmable.

Refer the topics “Access Codes” to know more.

How it works

You may set,

- The Access code of Call Pick Up as '4'.
- The Access code for Alarms as '41'.
- The Access code for Department Group 01 as '412'.

- Extension user A dials '4'.
- The system finds three access codes starting with '4' (4, 41, 412).
- So, it waits for 2 seconds, which is the default duration of the Conflict Dialing Timer, for the next digit to be dialed.
- If A does not dial any other digit before the Timer expires, the system interprets the code as '4' and invokes Call Pick-Up.

- If A dials '1' before the Timer elapses,
  - The system interprets it as '41'.
  - The system detects another access code starting with '41'.
  - So it waits for 2 seconds again for the next digit to be dialed.

- If A does not dial any other digit before the Timer elapses, the system interprets the code as '41' and invokes the Alarm feature.

- If A dials '2' before the Timer elapses,
  - The system interprets the code as '412' and invokes Department Call to Group 01, provided there are no other access codes like 4121, 4123, etc.

- If such access codes exist, the system again waits for the duration of the Conflict Dialing Timer for another digit to be dialed.

- Thus, only when the conflict in the access codes is resolved will the system respond accordingly.

How to configure

The working of this feature is controlled by the Conflict Dialing Timer, which is set by default to 2 seconds and can be changed as desired.

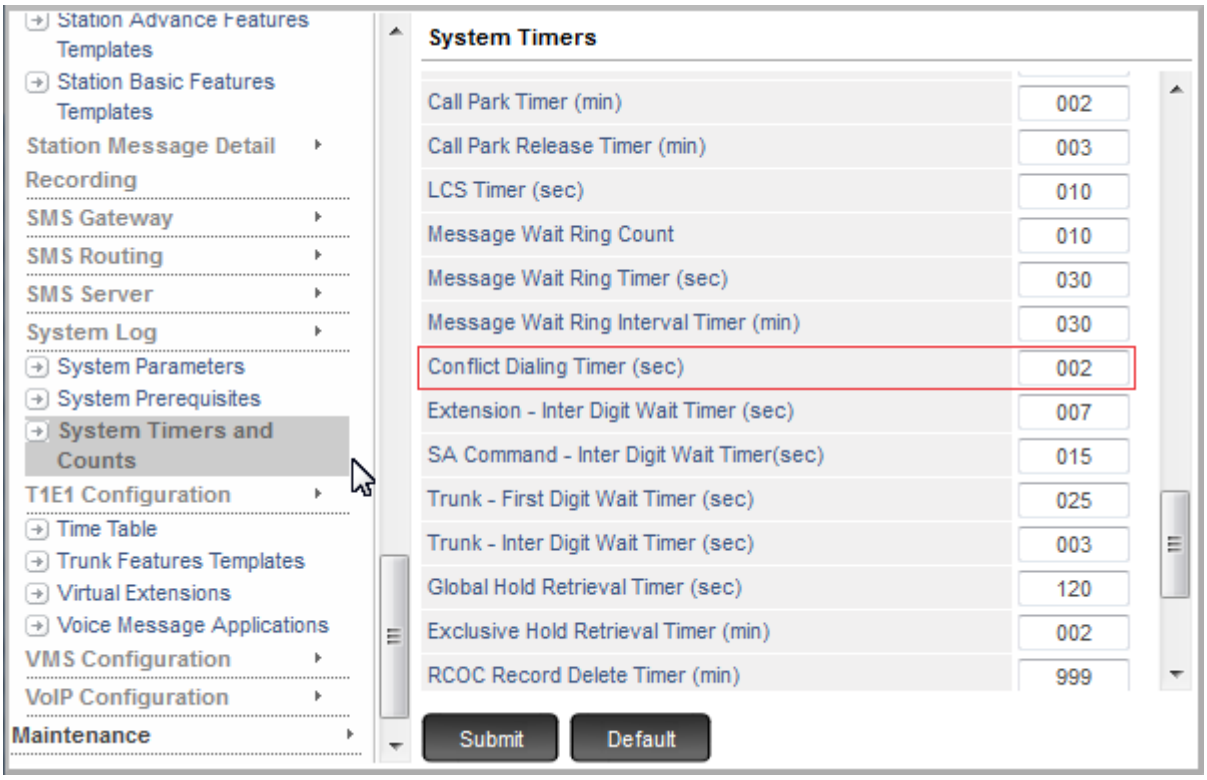
If the duration of the Conflict Dialing Timer is long, it may cause delay in the system's response to the feature. If the duration is less, the system may misinterpret the access codes. Ensure that the value of the Timer is programmed optimally (that is, at least the default value).

Changing Conflict Dialing Wait Timer using Jeeves

- Log in as System Engineer.

- Under **Configuration**, click **System Timers and Counts**.

- Scroll to **Other Features** and set the time in seconds as desired for the **Conflict Dialing Timer**.



- Click **Submit** at the bottom of the page to save changes.

- Log out of Jeeves or continue programming further.

Changing Conflict Dialing Wait Timer using a Telephone

- Enter SE mode from a DKP/SLT.

    - Dial command ***5351-Seconds***
      Where,
      Seconds is from 001 to 255 seconds.
      Default: 002 seconds.

- Exit SE mode.

Conversation Recording

What's this?

Conversation Recording allows extension users to record their talk with other extension users or external parties, after or without informing the opposite party.

This feature can be used to record verbal agreements, important discussions, instructions, interviews, client requirements, take or place orders, etc.

Extensions must have a mailbox assigned to them for recording conversations. So, a VMS Module must be installed on the CPU Card of the system for this feature to work.

SARVAM UCS supports recording of 20 calls[281] simultaneously.

- *Use this feature in accordance with the local laws.*
- *Matrix Comsec is not responsible for any mis-/abuse of this feature by users.*

On SIP extensions, SARVAM UCS supports Conversation Recording using INFO Message. For a list of IP phones on which this feature has been tested, see "SARVAM UCS Features tested on IP Phones of different Brands"in the Appendix.

It is not possible to pause the Conversation Recording in SIP Extension. So, when SIP Extension puts the call on hold and then retrieves the call then for the hold duration, silence will be recorded in the recorded file.

How it works

A and B are extensions. Both are assigned a mailbox each.
C and D are external parties.

- A calls C.
- C answers the call.
- A presses the Transfer Key.
- C is put on Consultation Hold.
- A dials the command for Conversation Recording.
- The system sends a string of digits to the Voice Mail System to initiate Conversation Recording.
- A and C are in speech again.
- The conversation recording starts in A's mailbox. The system plays beeps, if Conversation Recording Beeps are enabled.
- A or C disconnects the call.
- Conversation recording ends.
- A can listen to the recorded conversation by invoking the Voice mail feature.

The same is repeated when B calls A. As both have mailboxes assigned, both can record the conversation.

281. ETERNITY PENX supports 16 calls simultaneously.

How to configure

The functioning of this feature is controlled by three parameters: 'Class of Service', 'Mailbox', and 'Conversation Recording Beeps'. These parameters can be programmed using Jeeves or by dialing SE Commands from a telephone.

Conversation Recording in Class of Service

Conversation Recording must be enabled in the COS group of the extensions to which this feature is to be allowed.

In the default Station Basic Feature Template Number 01 is assigned to all the extensions of SARVAM UCS, CoS group 01 is assigned as default. Conversation Recording is disabled in CoS group 01. So, none of the extensions of the SARVAM UCS can record conversations.

Decide which extensions should be allowed Conversation Recording.

If you want to allow this feature to all extensions, simply enable Conversation Recording in the default CoS group 01.

If you want to allow Conversation Recording to selected extensions,

1. Define a CoS group with Conversation Recording enabled.
2. Prepare a Station Basic Feature Template with this CoS group applicable in all the "Time Zones".
3. Assign this new Template to the extensions to which Conversation Recording is to be allowed.

Refer the topics "Class of Service (COS)" and "Station Basic Feature Template" for detailed instructions and programming.

Mailbox

Extensions that are to be allowed Conversation Recording must also have a mailbox. Refer "Configuring SLT Parameters using a Telephone", "Configuring DKP Extensions using a Telephone", "Configuring ISDN Terminals using a Telephone", "Viewing SIP Extension Status" for more information and programming instructions.

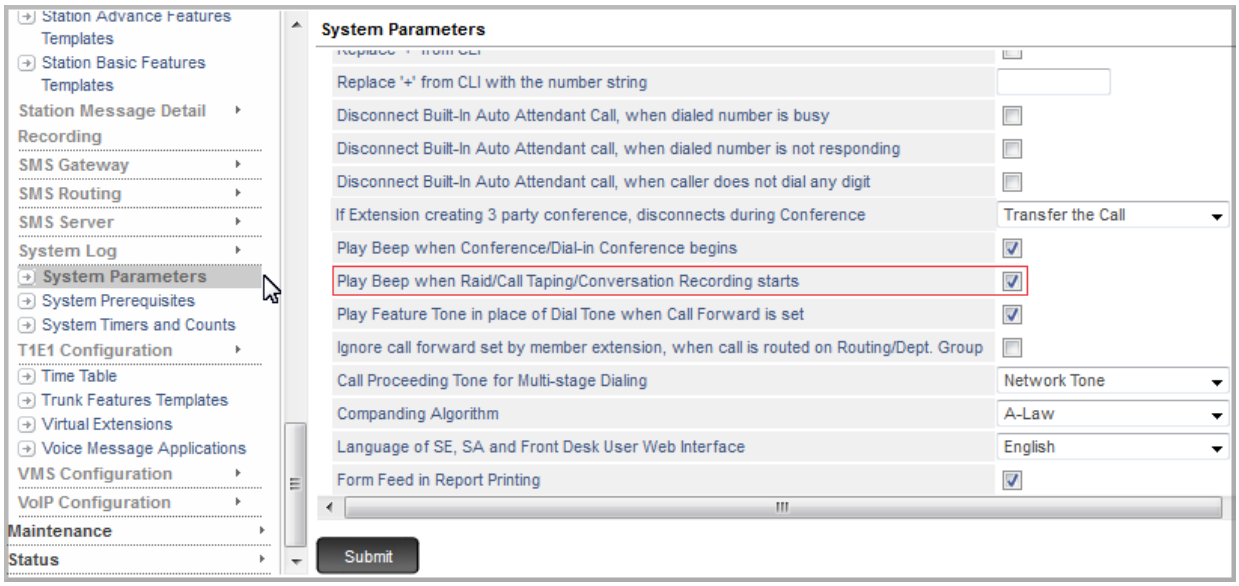
Conversation Recording Beeps

Decide whether you want Beeps to be played during Conversation Recording. Follow the instructions given below.

Programming Conversation Recording Beeps using Jeeves

- Log in as System Engineer.

- Under **Configuration**, click **System Parameters** to open the page.



- Go to **Play Beep when Call Taping/Conversation Recording starts**. Click the check box to enable or clear the check box to disable this feature.
- Click **Submit** at the bottom of the page to save changes.
- Log out of Jeeves or continue with further programming.

Programming Conversation Recording Beeps using a Telephone

- Enter SE mode from a DKP/SLT.
  - Dial command ***5332-Code.***
    Where,
    Code is
    0 for Disable
    1 for Enable
- Exit SE mode.

How to use

For EON and Extended IP Phone Users

To record a conversation:

- You are in speech with another extension/external number.
- Press DSS Key assigned to Conversation Recording (if programmed).

  OR

  Press Transfer Key and Dial ***1095.***
- You get beeps (if enabled).

- Speech with party re-established.
- Recording starts.
- Go ON-Hook, after conversation ends.

To listen to a recorded conversation:

- Press ‘Voice Mail’ Key

  OR

  Dial ***3931***[282]

- Follow Voice Mail Prompts.
- Go Idle or you get dial tone after 3 seconds.

Conversations are recorded as New Messages. So, follow the voice mail prompts for listening to new messages.

For SLT Users

To record a conversation:

- You are in speech with another extension/external number.
- Dial ***Flash-1095***
- You get beeps (if enabled).
- Speech with party reestablished.
- Recording starts.
- Replace handset when conversation ends.

To listen to a recorded conversation:

- Lift the handset
- Dial ***3931***
- Follow Voice Mail Prompts.
- Replace handset.

282. This is the default Voice Mail Feature Access Code. Verify with you System Engineer if this has been changed and use the new code.

Customer Name

What's this?

Customer Name is the name of the organization/enterprise that has deployed SARVAM UCS. As the User, you can enter the name of your company/organization in the system.

When Customer Name is assigned in the system, this name will appear as header on the various System Reports generated and printed by the SARVAM UCS like SMDR Incoming, Outgoing and Internal Call Reports, T1E1PRI Performance reports, Alarm Status reports, etc.

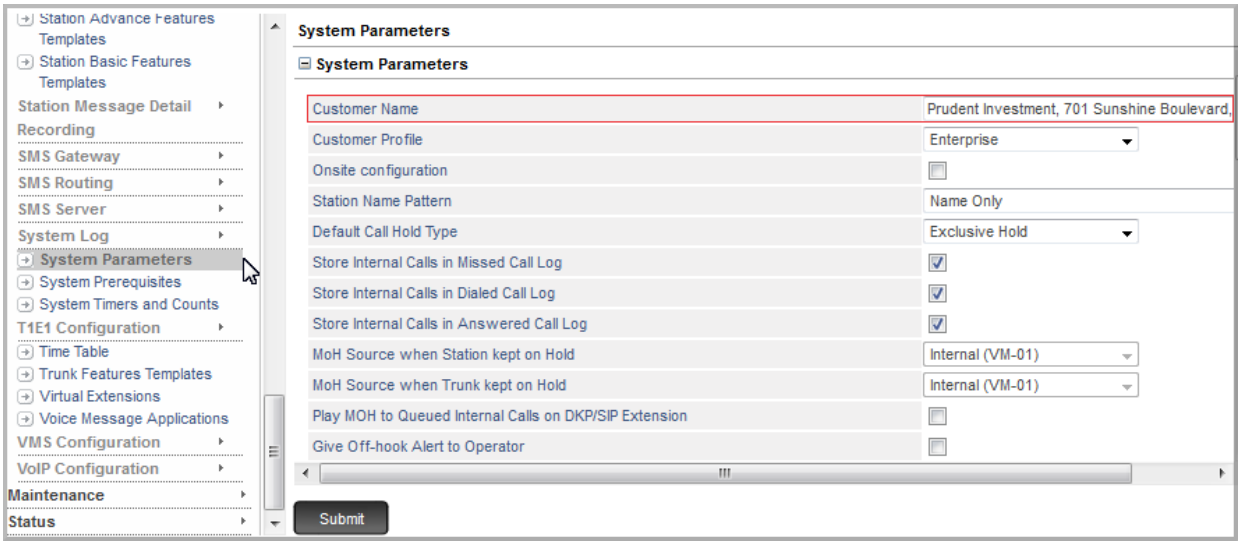
The Customer Name may consist of a maximum of 80 alphanumeric characters, including punctuation marks. So, you can enter the organization's address along with the Customer Name.

How to configure

Customer Name can be programmed using Jeeves and dialing SE commands from a Telephone at the time of installation, or any time thereafter. It can also be corrected or changed any time.

Programming Customer Name using Jeeves

- Login as System Engineer.
- Under **Configuration**, click **System Parameters** to open the page.



- Enter the name (and address, if desired) of the organization/enterprise in the field **Customer Name**. For example: Prudent Investment, 701 Sunshine Boulevard, Bannerghatta, Bangalore.
- Click **Submit** to save changes.

Programming Customer Name using a Telephone

- Enter SE mode from a DKP.

  To enter Customer Name, dial:
  - **5401-Customer Name-#***

  To clear the Customer Name, dial:
  - ***5401-#****

- Exit SE mode

- *Use EON to assign Customer Name, as SLT does not support alphanumeric dialing.*
- *The method of entering Customer Name from EON is similar to typing text messages from the mobile phone.*

COSEC Integration

SARVAM UCS supports integration with COSEC for “Building Intercom” solutions, that is, few Systems can be connected to each other using VoIP and these are also integrated with the Matrix COSEC Door Controllers. However, COSEC Integration can also be used in VIP Apartments and Villa’s, wherein ETERNITY LENX/MENX/GENX/PENX is installed.

With this integration the users — SLT, DKP and Extended IP— of SARVAM UCS can unlock the COSEC Door Controller.

Matrix COSEC is an enterprise-grade people mobility management solution for organizations covering Time-Attendance, Access Control, Visitor Management, Employee Self Service Portal, Roster Management, Contract Workers Management and Cafeteria Management. To know more about Matrix COSEC, refer to COSEC documentation or visit our website: www.MatrixComSec.com.

How it works

- Group together extensions of SARVAM UCS that need to access the same COSEC Door Controller. Assign a Group ID to each group. You can create 50 groups. Each group must have a unique Group ID.
- In SARVAM UCS for integration a User Name and Password is configured. Make sure the same User Name and Password is configured in each COSEC Door Controller you have installed. You can integrate 50 such Door Controllers with SARVAM UCS.
- Each COSEC Door Controller has a unique IP Address and port.
- Each Group ID is mapped to the IP Address and Port of the COSEC Door Controller. The extensions in the same Group ID can only unlock the COSEC Door mapped with their Group ID.
- The extension users can open the door by dialing *7 (access code to unlock the COSEC Door, programmable, see “Access Codes”)or by pressing the DSS key assigned to COSEC Door Open.

*Standard SIP Phone users can put a call on hold by dialing #2. These users can also unlock the COSEC Door by dialing *7.*

Let us understand how this feature works with the help of an example.

- ABC Limited has SARVAM UCS with Building Intercom Application installed, where there are 10 residential towers. Each tower has around 100 flats. Each flat has an SLT extension.
- The users in Tower 1 are grouped together, that is extensions from 200 to 300. They are assigned a Group ID 01.
- Each tower has its own gate with COSEC Door Controller installed. At each gate there is an extension for visitors.
- Group 01 is mapped to the COSEC Door Controller IP Address 192.168.101.200:8080

When there is a visitor at the main gate the security guides the visitor to the Tower 1. The visitor makes a call to the flat owner, extension 205. After confirming the identity of the visitor, the tower door needs to be opened.

The flat owner puts the caller on hold and dial *7 (access code to unlock the door) or press the DSS key assigned to COSEC Door Open.

The system checks the Group ID of the flat owners extension and the corresponding Door Controller mapped to it. The system sends the request along with the User Name and Password to the COSEC Door Controller.

If within 10 seconds (Response Timer, not programmable) a success response is received from the COSEC Door, the door is unlocked. If a response is not received or if failure response is received the user will get an error tone.

*The COSEC Door access code can be dialed in Idle state also, that is, if the flat owner has received the call on his/her mobile from the visitor and needs to open the door, the flat owner can lift the handset and dial *7.*

How to configure

For this feature to work, you must:

- On a piece of paper make four columns and enter the details as given below (taking the above example further):

User Name	Extension Number	Group ID	COSEC Door they need to access	COSEC Door IP Address and Port
James William	200	01	Tower 1	192.168.101.200:8080
Jessica James	201	01	Tower 1	192.168.101.200:8080
Kelly	302	02	Tower 2	192.168.101.205:8080
. . . .	. . . .	. . . .	. . . .	
William	505	05	Tower 5	192.168.101.250:8080

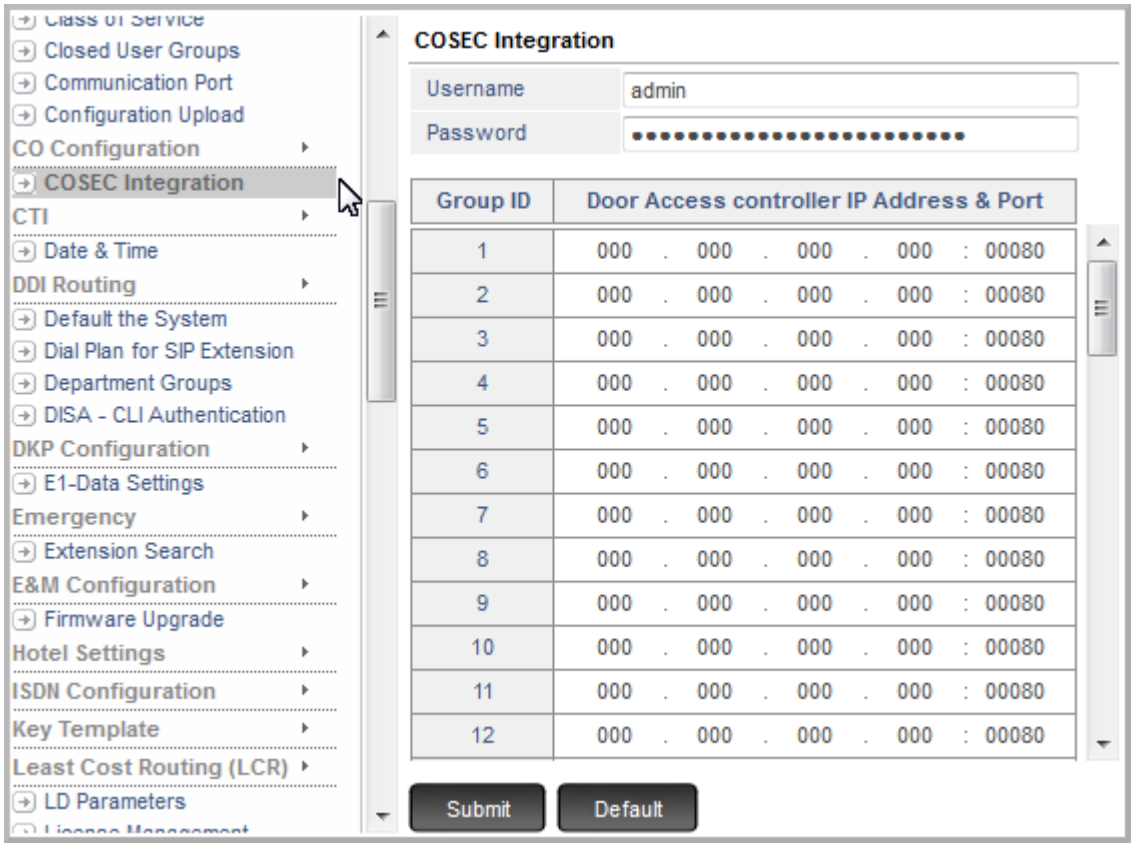
  You can create 50 groups. Assign each group a Group ID. Range of the Group ID is from 00 to 50.

- Configure the COSEC Integration table.
- Assign each user the Group ID.

To configure the COSEC Integration table,

- Login as System Engineer.

- Under **Configuration**, click **COSEC Integration** to open the page.



- For COSEC Integration, enter the **User Name** and **Password**. Default User Name: admin, Password: 1234.

  The User Name can be a maximum of 24 characters. Valid characters: 0 - 9, a - z, A - Z.

  The Password can be a maximum of 24 characters.Valid characters are 0 - 9, a - z, A - Z, !,@,*,(,),-,.+,/ and comma

  Make sure the same User Name and Password is configured in the COSEC Door Controllers you have installed.

- Against each **Group ID**, enter the **COSEC Door Controller IP Address** and **Port**.

  The IP Address can be a maximum of 15 characters (only IPv4 Addresses are supported).

  The Port can be a maximum of 5 digits. Valid range:1025 - 65535 or 80.

The same COSEC Door Controller IP Address and Port cannot be assigned to different Group ID's.

- Click **Submit**.

To assign Group IDs to SLT users, see parameter COSEC Door Group in "Configuring SLT Extensions".

To assign Group IDs to DKP users, see parameter COSEC Door Group in "Configuring DKP Extensions".

To assign Group IDs to SIP users, see parameter COSEC Door Group in “Configuring Matrix SPARSH VP248”, “Configuring Matrix SPARSH VP310”, “Configuring Matrix SPARSH VP330”, “Configuring Matrix VARTA ADR100/ AMP100 UC Clients”, “Configuring Matrix SPARSH VP510”, “Configuring Matrix SPARSH VP210”, “Configuring Matrix VARTA WIN200 UC Client” and “Configuring Standard SIP Phones”

To assign a DSS key to COSEC Door Open, see “DSS Keys Programming”.

Day Night Mode

What's this?

Certain features of the SARVAM UCS like Operator, Class of Service, Toll Control, Outgoing Trunk Bundle Access Groups, Trunk Landing Group, Auto Attendant, Direct Inward System Access (DISA), Security Alarms, etc, require extensions and trunks to behave differently according to the working hours, non-working hours and break hours, which are referred to as Time Zones.

These Time Zone-dependant features and facilities are operated automatically according to the Time Tables programmed in the system. In a Time Table, the Time Zones - Working Hours, Non-Working Hours, Break Hours - are defined for the entire week. Time Table is assigned to trunks, extensions and other time-zone dependant features. The system executes the Time-Zone dependent features and facilities automatically according to the Time Table.

To know more refer the topic “Time Tables”.

Day/Night Mode allows you to manually change the Time Zone of the system at any point in time, by issuing a command or by pressing the DSS key on the phone. For example, the office is to be closed on account of an unplanned holiday or emergency. So, the Time Zones of all extensions and trunks must be set to Non-working hours to route outgoing calls and land incoming calls from/to the appropriate destination. You can set the SARVAM UCS to Night Mode until the office remains closed and set it back to operate as per the Time Table, when work is resumed.

To cite another example, the office must work for extended hours. You can set the SARVAM UCS to Day Mode and set it back to operate as per the Time Table.

When you set the system in Day/Night Mode, the system overrides the Time Tables assigned to Trunks, Extensions and Operator. According to the mode you selected, it applies Working Hours/Non-Working Hours to run all the Time-Zone dependent features of the system.

When the system is set to Day Mode, it applies Working Hours as the Time Zone for all extensions, trunks and time zone dependant features and facilities. When the system is set to Night Mode, it applies Non-Working Hours as the Time Zone on Time-Zone dependent features of the system.

Thus, Day/Night Mode forces the system to work in a particular Time Zone, until it is changed again, manually.

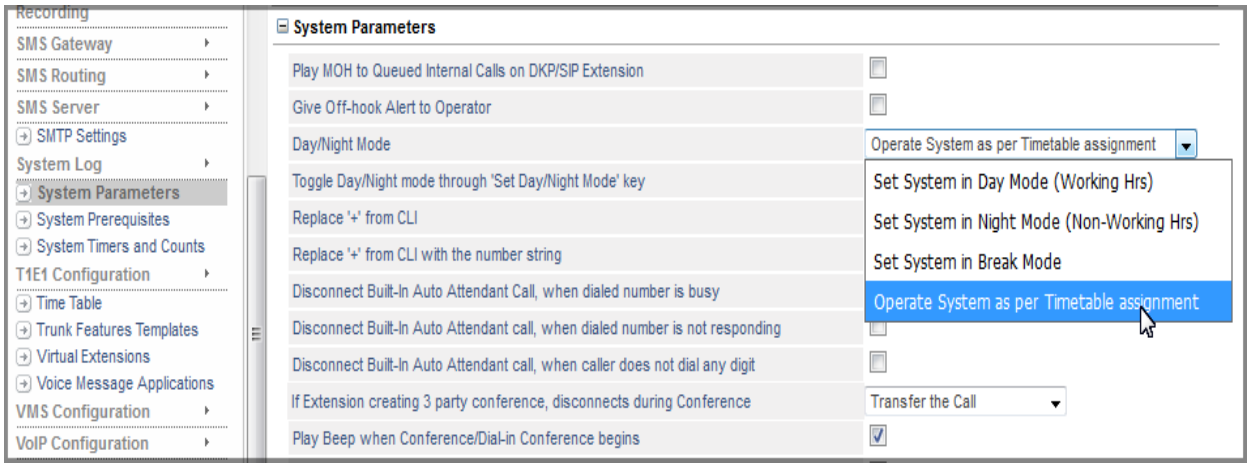
Day/Night mode can be set by the System Engineer (SE mode) as well as by the System Administrator (SA Mode). It can be done using Jeeves or by dialing a command from a Telephone.

How to configure

Setting Day/Night Mode from SE mode using Jeeves

- Log into Jeeves as System Engineer.

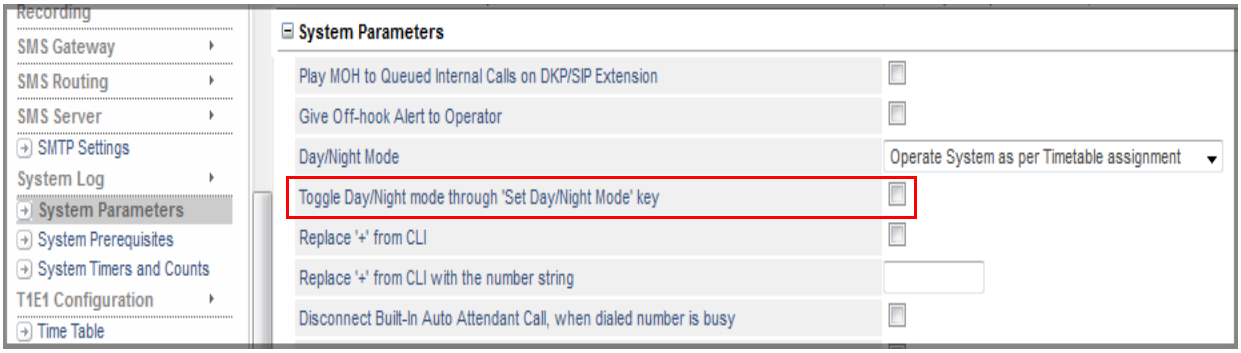
- Under **Configuration**, click **System Parameters** to open the page.



- Click **System Parameters** to expand.

- Go to **Day/ Night Mode**. Select the desired option:

  - Set System in Day Mode (Working Hrs)
  - Set System in Night Mode (Non-Working Hrs)
  - Set System in Break Hours Mode
  - Operate System as per Timetable assignment (default).



- Select the **Toggle Day/Night mode through 'Set Day/Night Mode' key** check box to switch to Day Mode (Working Hours) or Night Mode (Non-Working Hours) on pressing the DSS Key. Default: Disabled.

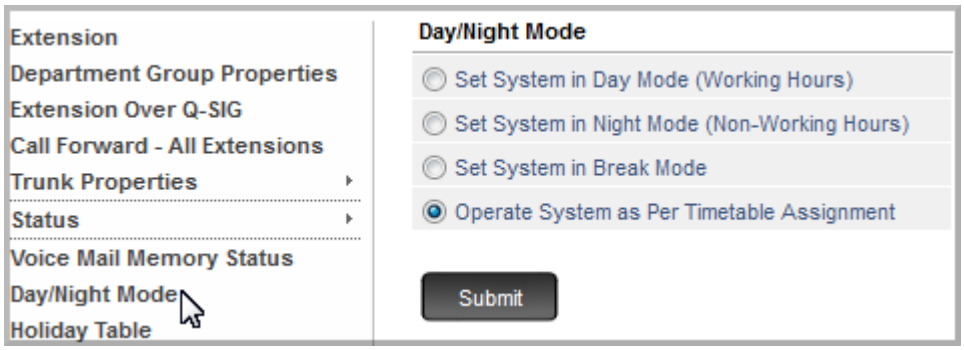
- Click **Submit** at the bottom of the page.

- Log out of Jeeves or continue, as desired.

Setting Day/Night Mode from SA mode using Jeeves

- Log into Jeeves as System Administrator.

- Click **Day/Night Mode** to open the page.



- Select the desired option:
  - Set System in Day Mode (Working Hours)
  - Set System in Night Mode (Non-Working Hours)
  - Set System in Break Mode
  - Operate System as per Timetable assignment (default).
- Click **Submit** at the bottom of the page.
- Log out of Jeeves or continue, as desired.

Setting Day/Night Mode using DSS key

SARVAM UCS offers to the extension users the flexibility to manually switch to **Day/Night Mode,** at any point in time, by pressing the DSS key.

How to configure

- Log into Jeeves as System Engineer.
- Assign a DSS Key for **Day/Night Mode** to the required extension. To know more about assigning a DSS Key for a specific feature, refer to “DSS Keys Programming”.
- Enable the toggle functionality for the DSS Key by selecting the check box for the parameter **Toggle Day/ Night mode through ‘Set Day/Night Mode Key’**. To know more, refer to “Setting Day/Night Mode from SE mode using Jeeves”.

  When you enable this check-box, you can switch to Day Mode (Working Hrs) or Night Mode (Non-Working Hrs) as per your requirement. By default, this is disabled.

You cannot switch to Break Hours or As per Timetable assignment using the DSS key. This can be achieved only by logging in Jeeves or through telephone.

How to use

For EON and Extended IP Phone Users

- Press DSS Key assigned to Day/Night Mode.

When you press the DSS Key of extension, the system overrides the Time Table assigned to that extension. According to the current Day/Night Mode, it switches to Working Hours/Non-Working Hours.

If you are setting Day/Night Mode from a DKP/IP Phone using a DSS key with Toggle functionality disabled, refer the LED indication in table below.

LED Indication on DSS Key assigned to Day/Night Mode (without Toggle)

Model	Event	Color	Cadence
EON48/EON310/ EON510/SPARSH VP248/SPARSH VP310/SPARSH VP510	Day Mode Set	Blue	Continuous ON
	Night Mode Set	Red	Continuous ON
	Break Hours Mode	Violet	Continuous ON
	System set to work as per Time Table	--	OFF

If you are setting Day/Night Mode from a DKP/IP Phone using a DSS key with Toggle functionality enabled, refer the LED indication in table below.

LED Indication on DSS Key assigned to Day/Night Mode (with Toggle enabled)

Model	Present Mode	Next mode when user presses Day/Night Mode DSS Key	LED Color after toggle	Cadence
EON48/EON310/ EON510/ SPARSH VP248/ SPARSH VP310/ SPARSH VP510	As per Time Table (Day Mode)	Night Mode	Red	Continuous ON
	As per Time Table (Night Mode)	Day Mode	Blue	Continuous ON
	As per Time Table (Break Hour Mode)	Day Mode	Blue	Continuous ON
	Day Mode	Night Mode	Red	Continuous ON
	Night Mode	Day Mode	Blue	Continuous ON
	Break Hour Mode	Day Mode	Blue	Continuous ON

Setting Day/Night Mode from SE mode using a Telephone

- Enter SE mode from a DKP/SLT.

  To set the system in Day/Night mode, dial:
  - ***4801-Code***
    Where,
    Code is from 1 to 4
    1 is for Day Mode
    2 is for Night Mode
    3 is for Operate system as per Time Table.
    4 is for Break Hours

- Exit SE mode.

Setting Day/Night Mode from SA mode using a Telephone

- Enter SA mode from a DKP/SLT.

    - Dial ***1072-018-Code***
      Where,
      Code is from 1 to 4
      1 is for Day Mode
      2 is for Night Mode
      3 is for Operate system as per Time Table.
      4 is for Break Hours

- Exit SA mode.

Daylight Saving Time (DST)

What's this?

Daylight Saving Time (DST) is the practice of advancing clocks so that afternoons have more daylight and mornings have less. Typically clocks are adjusted forward one hour near the start of spring and are adjusted backward in autumn.

Many countries of the world use DST, though the start and end dates of DST vary with location and year. Even within countries, uniform DST may not be observed. For example the states of Arizona and Hawaii do not observe DST. Certain countries may observe DST in certain years, for instance Guatemala, while in most countries of Asia and Africa, and in certain countries of South America, DST is not observed at all.

When SARVAM UCS is installed in a country/region where DST is used, it is necessary to synchronize the Real Time Clock of SARVAM UCS with the local time.

So, if you are installing SARVAM UCS in a country where DST is used, find out the DST convention currently in use in that country, and adjust DST accordingly.

How it works

The forward and backward adjustment of clocks can be Scheduled or Manual.

- **Scheduled DST Adjustment:** The Real Time Clock of the SARVAM UCS is advanced and set backward automatically according to the DST convention of the country/region where the SARVAM UCS is installed.

  Scheduled DST Adjustment is useful in countries/regions where DST Time is fixed, such as in Europe, USA and Canada, without yearly variations.

  The table below gives describes the DST conventions followed in the different countries for which SARVAM UCS will automatically adjust DST.

  SARVAM UCS supports 18 DST Types for Scheduled DST Adjustment.

DST Type	DST Timings		Applicable in Countries
	Start Time	End Time	
01	Last Sun MAR From 01:59 to 03:00	Last Sun OCT From 02:59 to 02:00	Austria, Poland, Russia, Spain
02	Last Sun OCT From 01:59 to 03:00	Last Sun MAR From 02:59 to 02:00	Australia, Australia-Tasmania, Belgium, France, Germany, Greece, Hungary, Italy, Sweden, Switzerland
03	Second Sunday MAR From 01:59 to 03:00	First Sunday NOV From 01:59 to 01:00	Bahrain, Mexico, Turkey, United States
04	First Sun NOV From 23:59 to 01:00	Third Sun FEB From 23:59 to 23:00	Brazil

DST Type	DST Timings		Applicable in Countries
	Start Time	End Time	
05	Second Sunday MAR From 01:59 to 03:00	First Sunday NOV From 01:59 to 01:00	Canada
06	Second Sat OCT From 23:59 to 01:00	Second Sat MAR From 23:59 to 23:00	Chile
07	Last Sun MAR 00:59 02:00	Last Sun OCT 01:59 01:00	Denmark, Ireland, Portugal, United Kingdom
08	Last Sun MAR 02:59 04:00	Last Sun OCT 03:59 03:00	Finland
09	First APR 02:59 04:00	First OCT 03:59 03:00	Iraq
10	Last Sun MAR 02:29 03:30	Last Sun OCT 02:29 01:30	Kyrgyzstan
11	Last Fri APRIL 23:59 01:00	Last Thu SEP 23:59 23:00	Egypt
12	Last Sun MAR 23:59 01:00	Last Sun OCT 23:59 23:00	Lebanon
13	First Sun SEP 01:59 03:00	First Sun APRIL 01:59 01:00	Namibia
14	Last Sun SEP 01:59 03:00	First Sun APR 02:59 02:00	New Zealand
15	Last Sun MAR 01:59 03:00	Last Sun OCT 02:59 02:00	Norway
16	First Sun OCT 23:59 01:00	First Sun APRIL 23:59 23:00	Paraguay
17	First APRIL 23:59 01:00	First OCT 23:59 23:00	Syria
18	First APRIL 23:59 01:00	Last Sun OCT 23:59 23:00	Cuba

The DST Type is to be selected according to the country/region where the system is installed.

When DST Mode is set to 'Scheduled' and the DST Type is selected, the system will automatically adjust DST at the preset dates and time for the country/region where the system is installed.

For example, if SARVAM UCS is installed Spain, the DST Type 01 applicable to this country should be programmed as Scheduled DST. The system will automatically advance the clock on the last Sunday of March at 01.59.03:00 am every year (the start date of DST) and set the clock backward on the last Sunday of October at 02.59.02:00 am of the same year.

- **Manual DST Adjustment:** The Real Time Clock of the SARVAM UCS is advanced and set backward manually according to the DST convention of the country/region where the SARVAM UCS is installed.

Manual DST Adjustment is to be used in regions/countries that have no fixed DST Convention and where yearly variations in DST practices are likely.

When DST Mode is set as 'Manual', you must set the start and the end time, that is, the time at which the clock is to be advanced and the time at which the clock is to be delayed.

There are two ways to adjust DST manually:

1. The 'Day of Month' method, which specifies a day of the month DST will start or end. For example: starting on the 2nd Sunday of March and ending on 1st Sunday of November.

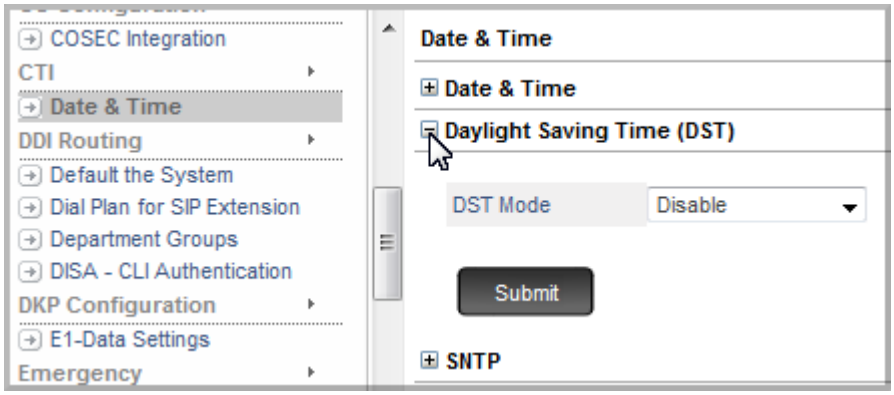
2. The 'Date and Month' method, which specifies a date of the month that DST will start or end. For example: starting on March 11 and ending on November 4.

DST is not applicable in certain regions/countries, like Asia and South America. In such cases, the DST Mode is to be 'Disabled'.

How to configure

Adjusting DST using Jeeves

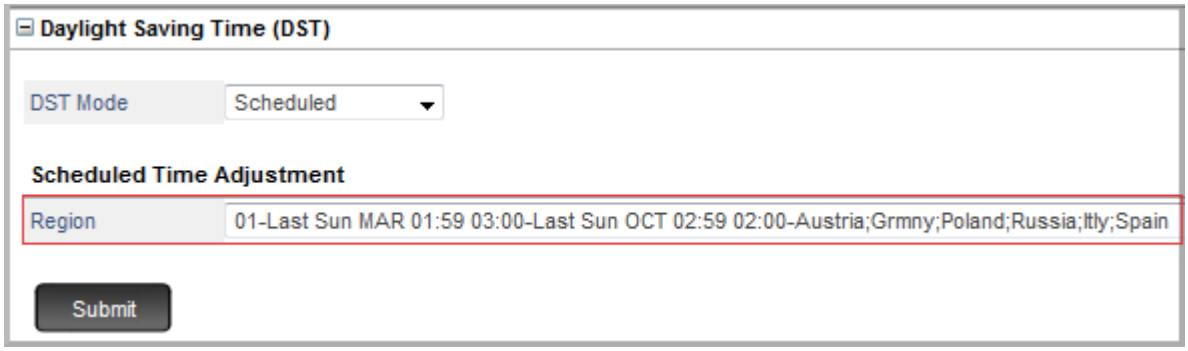
- Log in as System Engineer.
- Under **Configuration**, click **Date and Time**.
- Click **Day Light Saving Time** to expand.



- Set the **DST Mode** to **Manual** or **Scheduled** as per your requirement.

Scheduled DST Adjustment

- If you have selected **Scheduled** as DST mode, in the **Region** list, select the name of the country/region where your system is installed.

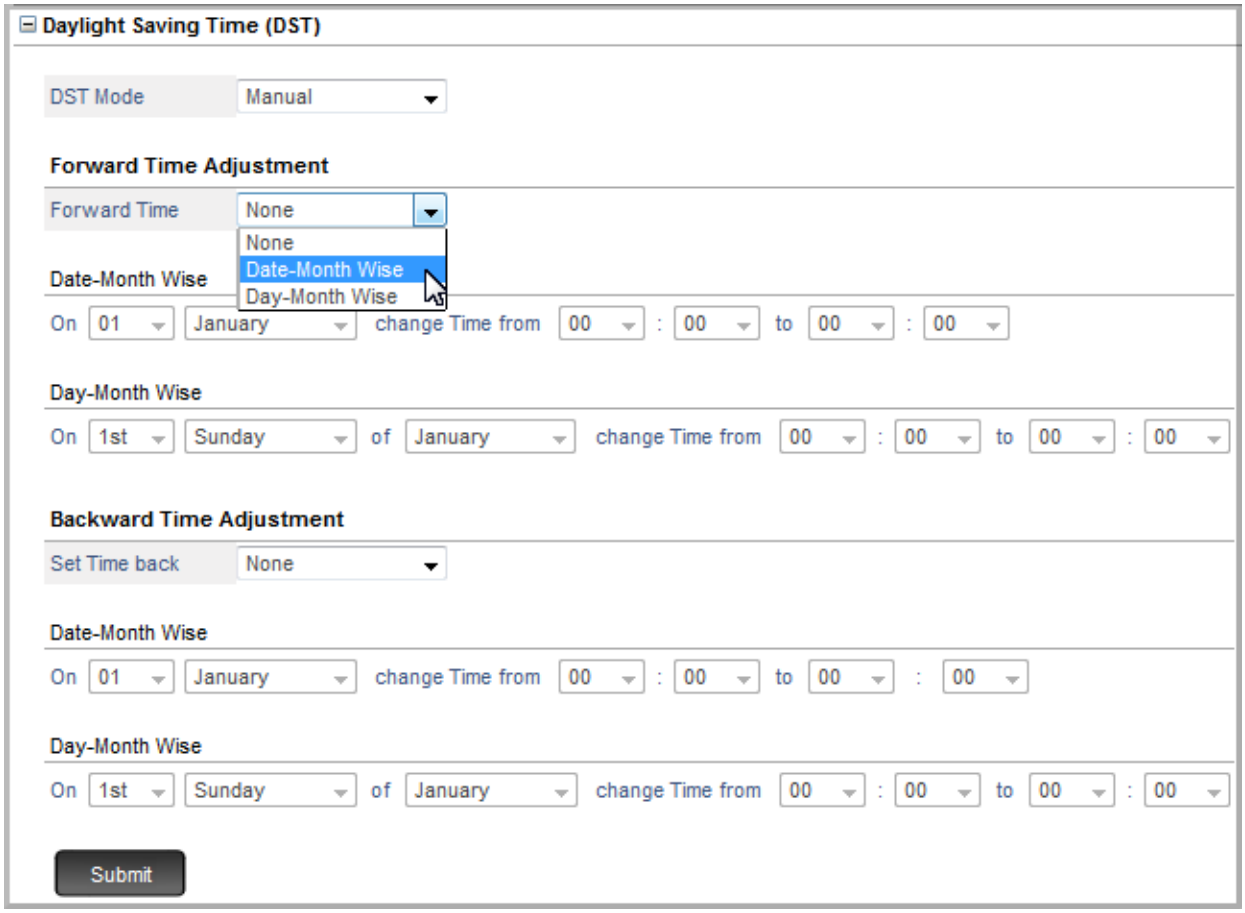


- Click **Submit** at the bottom of the page to save your DST setting.
- If you do not find your region on this list, you are recommended to set DST Mode to 'Manual' and adjust DST manually.

Manual DST Adjustment

- If you have selected **Manual** as DST Mode, set the **Forward and Backward Time Adjustments**.
- Go to the option **Forward Time Adjustment** to advance the time when DST starts.

- To **Forward Time**, select the desired option:



  - **Day-Month Wise** to specify the day of the month DST will start.

    OR

  - **Date-Month Wise** to specify the date of the month DST will start.

If you select 'Day-Month Wise' option, the 'Date-Month Wise' option will be disabled, and vice versa.

Day-Month Wise

- If you select the 'Day-Month Wise' option, you should now select the desired options in each of the following:

  - **Ordinal number:** Select the Ordinal number of the day of the month, that is, the 1st, 2nd, 3rd, 4th, 5th day, when DST begins.

  - **Day:** Select the day of the month - Sunday, Monday, Tuesday, Wednesday, Thursday, Friday, Saturday - when DST begins.

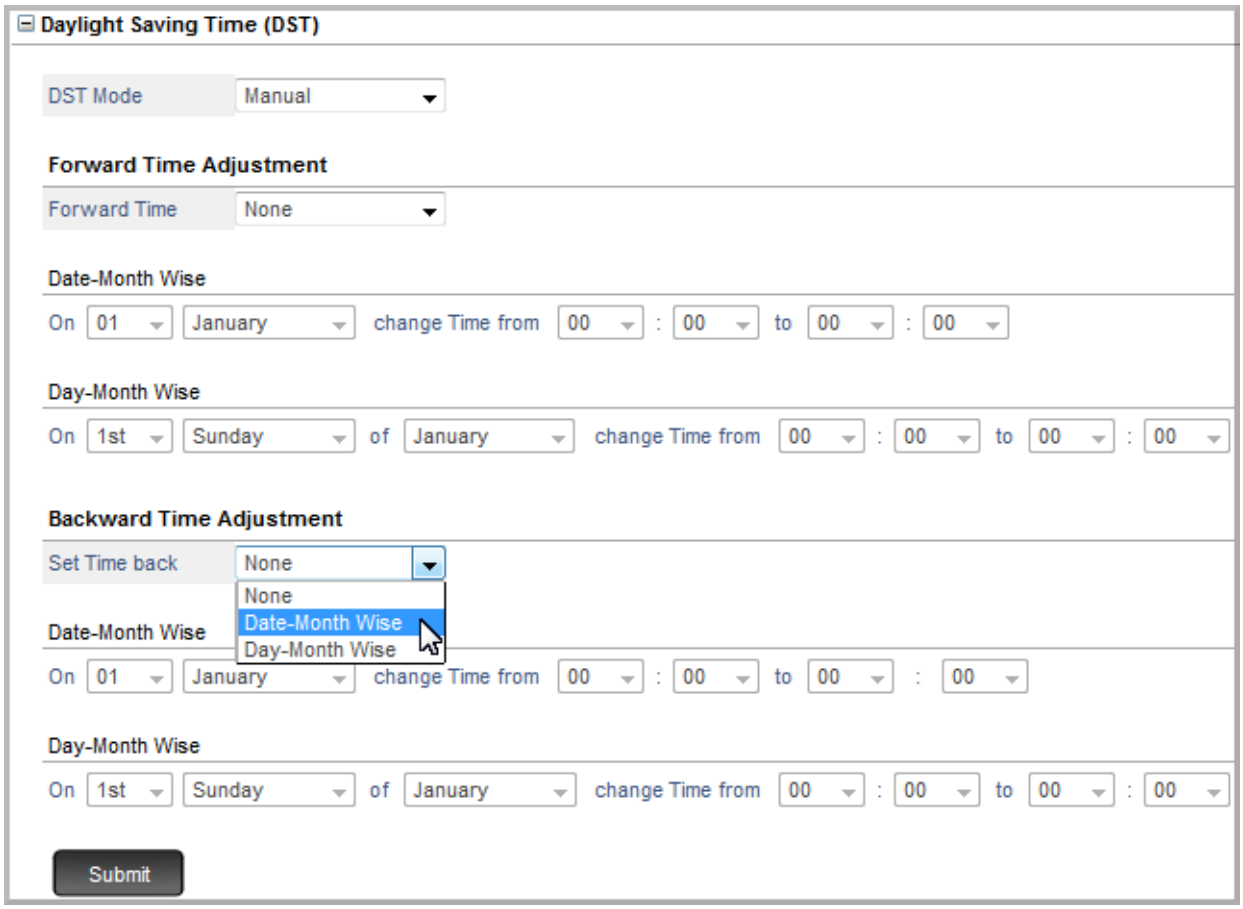
  - **Month:** Select the month when DST beings (January-December).

  - **Change Time From:** Select the time when DST will begin to change. The time mode is 24 hours, with options from 00 to 23 hours and 00 to 59 minutes.

- **To:** Select the time to which the DST is advanced. The time mode is 24 hours, with options from 00 to 23 hours and 00 to 59 minutes.

Date-Month Wise

- If you select 'Date-Month Wise' option, you should now select the desired options in each of the following:
  - **Date:** The date on which DST begins (1-31).
  - **Month:** The name of the month when DST begins (January-December).
  - **Change Time From:** The time when DST will begin to change. The time mode is 24 hours, with options from 00 to 23 hours and 00 to 59 minutes.
  - **To:** The time to which the DST is advanced. The time mode is 24 hours, with options from 00 to 23 hours and 00 to 59 minutes.
- Now, go to the option **Backward Time Adjustments** to set the time back (that is, end DST and begin standard time).



- Follow the same steps described above (step no. 4 to 6) to set the day/date, month, hours and minutes except, here you must set these parameters according to the time when DST ends.
- Click **Submit** at the bottom of the page to save your DST settings.

- *SARVAM UCS gives you the flexibility to set the 'Forward DST Adjustment' according to Date-Month, while the Backward DST Adjustment according to Day-Month. Similarly, the reverse is also possible, that is, Forward DST may be set according to Day-Month, while the Backward DST may be set as Date-Month. This flexibility is particularly useful for setting DST of countries where the start of DST is defined by date and month, like the First of April, but the end of DST is defined by Day and Month, such as the last Sunday of October (as observed in Cuba).*

- *When the DST of a particular country starts or ends on the Last Sunday or any other day, for example, the last Tuesday, last Friday of the month, always set the Ordinal Number as '5th'.*

- *Wherever time adjustments are made at 00:00 hours, use the previous date and set DST start time (that is, "from" time) at 23:59 hrs.*

- *If you synchronize the RTC with the SNTP Server and the Date and Time changes, the DST will be applicable as per the new Date and Time.*

Adjusting DST using a Telephone

- Enter SE mode.

  To set the DST Mode, dial:
  - ***1010-DST Mode***
    Where,
    DST Mode is
    0 for Disabled
    1 for Manual
    2 for Scheduled

  If DST Mode is selected as 'Manual'.

  To set the start time of DST, dial:
  - ***1011-Date-Month-Current Time-Advance Time***
    Where,
    Date is from 01 to 31. Use leading zero in case of single digit date (Default 01).
    Month is from 01 to 12. Use leading zero in case of single digit month (Default 01).

    Current Time[283] is in HH:MM format,
    HH from 00 to 23 (use leading zero)
    MM from 00 to 59 (use leading zero) (Default 00:00).

    Advance Time[284] is in HH:MM format,
    HH is from 00 to 23 (use leading zero)
    MM from 00 to 59 (use leading zero) (Default 00:00).

    Please note that the advance time will be greater than the current time.

  To set the end time of DST, dial:

283. The current time is the time that is presently followed by the system.
284. This is the time to which the Real Time Clock should be advanced.

- ***1012-Date-Month-Current Time-Delay Time***
  Where,
  Date is from 01 to 31 (use leading zero for single digit date) (Default 01).
  Month is from 01 to 12 (use leading zero for single digit month) (Default 01).

  Current Time is in HH:MM format (use leading zero),
  HH is from 00 to 23
  MM is from 00 to 59 (Default 00:00).
  Delay Time[285] is in HH:MM format (use leading zero must)
  HH is from 00 to 23
  MM is from 00 to 59 (Default 00:00).

  Please note that the delay time will be less than the current time.

If DST Mode is selected as 'Scheduled'

To select the DST Type, dial:

- ***1013-DST Type***
  Where,

DST Type	Applicable in Countries
01	Austria, Poland, Russia, Spain
02	Australia, Australia-Tasmania, Belgium, Cote d'Ivoire, France, Germany, Greece, Hungary, Italy, Sweden, Switzerland
03	Bahrain, Mexico, Turkey, United States
04	Brazil
05	Canada
06	Chile
07	Denmark, Ireland, Portugal, United Kingdom
08	Finland
09	Iraq
10	Kyrgyzstan
11	Egypt
12	Lebanon
13	Namibia
14	New Zealand
15	Norway
16	Paraguay
17	Syria
18	Cuba

For example: to select DST Type for Brazil, dial: ***1013-04.***

285. This is the time to which the Real Time Clock should be set back to.

- Exit SE mode.

Department Call

What's this?

Department Call enables you to group together extensions of a particular department so that callers can reach anyone in the department by dialing a common access code assigned to the department.

Calls made to such groups of extensions are called Department Calls and the access code used to make department calls is called Department Number.

This feature is useful in situations where any member of a department may interact with callers, as for instance in a information counter, a customer care cell, a technical support team, etc.

Callers can also reach individual extensions in a Department group by dialing the extension number.

SARVAM UCS supports the formation of 24 department groups[286]. The *member* extensions of a department group may be single line telephones (SLT), digital key phones (DKP), SIP Extensions, ISDN Terminals or Virtual Extensions.

Each Department Group can also be assigned a mailbox for voice mail, which any member extension can access.

Each Department Group can forward its calls to an extension or to its voice mail, or to another Department Group.

How it works

Extensions A, B, C, D are grouped as a Department with the access code 3901.

Internal Calls

- Extension E dials 3901 to call the Department.
- The system checks E's Class of Service for the Department Call feature.
- The feature is enabled. The system checks if Rotation is enabled in the routing group assigned to the Department.
- The Rotation flag is enabled. The system lands the call on the extension which is set to ring first.
- Extension A, configured as the first landing destination rings for the duration of the Ring Timer (configurable; default: 15 seconds).
- A answers the call. Speech established between A and E.

- If A does not answer, the system hunts for the next extension in the group to land the call, say B.
- B starts ringing for the duration of the Ring Timer.
- If Continuous Ring is enabled on A, A will continue to ring even as B is ringing.
- If B does not answer the call at the end of the timer, the system hunts for the next extension, C.
- If B has Continuous Ring enabled, B will continue to ring even as C is ringing.
- If the call is not answered even after hunting the last extension, the system will loop back and start from the first extension once again.

286. ETERNITY LENX /MENX supports 32 Department Groups and ETERNITY PENX supports 16 Department Groups.

External Calls

Department Calls can be made using Auto Attendant - Built-In Auto Attendant or VMS Auto Attendant. For example, a company may use the Built-In Auto Attendant to have callers who want information only to dial the Information Department instead of waiting for the Operator.

- An external caller places a call to Department 3901 using the Built-In Auto Attendant.
- The system checks if Rotation is enabled in the routing group assigned to the Department.
- As the Rotation flag is enabled, and the first call was landed on A, the system lands the call on the next extension B.
- Extension B rings for the duration of the Ring Timer (configurable; default: 15 seconds). If the Continuous Ring flag is enabled for B, it will continue to ring, even as the system hunts for another extension in the group to land the call.

- A third call internal/external made to Department 3901.
- The same process as described above will be repeated.
- But the system will land the call on extension C first, because Rotation flag is enabled on this routing group.
- The subsequent incoming calls will land on the extension which is next to the one that received the last call. So the next call to the Department will land on extension D, the one thereafter on A, and so forth.

Thus for each call, the system will hunt for a landing extension as per the Rotation set for the routing group. The extensions will ring for the duration of the Ring Timer, either continuously or one-by-one (as per the Continuous Flag configured), and according to the sequence in which the extensions in the group are arranged.

Rotation ensures equal distribution of call traffic. If Rotation is disabled, the fresh call will always land on first extension of the Department group.

Department Group and ISDN Terminal: *When more than two ISDN terminals connected to the same BRI port are configured as members of a Department group, if a call is made to this group using the department group access code, only two ISDN terminals connected to the BRI port will ring. This limitation is because of the BRI protocol.*

Voice Mail

A Department Group can be assigned a common mailbox for Voice Mail, called the *Department Group Mailbox*. For this a VMS module must be installed in the system. This common mailbox for the group is called Department Group Mailbox. You can assign Department Group Mailbox to selected extensions or to *all* extensions in the Department.

To take the example of Extensions A, B, C, D with the Department Access Code 3901 further,

- Extensions A, B, C and D are all members of Department Group 1 with the Access Code 3901.

- Department Group Mailbox is assigned to all the four extensions.

- When there is a new message in the Group Mailbox, all four extensions - A, B, C, D - will get the Message Wait Notification.

- The message wait indication may be a Stuttered Dial Tone or a Voice Message when the extension user goes OFF-Hook, or blinking of the LED Lamp on the extension, or a Ring.[287]

287. This will depend on the type of Message Wait Indication configured for the Extension's Voice Mail Settings.

- To the first extension that answers the notification call, for example, Extension A, the Voice Mail System informs about the new message(s) waiting in the Department Group Mailbox and in the Personal Mailbox. *"You have <x> new Message in your Personal Mail Box. You have <y> new Messages in your Department Group Mail Box."*

    - If there is no new message in both mailboxes, the VMS will play the message: *"You have Zero new Message."*

    - If there is a new message in the Department Group Mailbox, but none in the Personal Mailbox, the VMS will play the message:*"You have <x> new Message in your Department Group Mailbox."*

    - If there is no new message in the Department Group Mailbox, but new message in the Personal Mailbox, the VMS will play the message: *"You have <x> new Message in your Personal Mailbox."*

- The VMS prompts Extension A to access the Group mailbox: *"To go to Personal Mailbox, press 1. To go to Department Group Mailbox, press 2."*

- The user of Extension A presses 2, and is taken to the Department Group Mailbox,

- VMS prompts A: "Enter your mailbox password". Enter your department group mailbox password.

- The VMS checks the utilized mailbox memory,

    - if 80% of the mailbox memory has been consumed, the VMS prompts the caller: “Your Mailbox is 80% Full. Please Delete few messages.”.

    - if 100% of the mailbox memory has been consumed, the VMS prompts the caller: ”Your Mailbox is Full. Please Delete few messages.”.

- VMS prompts: "To listen to new messages press '1', to listen to old messages press '2', to send a message press '3', to change your mailbox settings press '4'."

- Extension A presses 1.
- VMS plays the new messages.
- After playing the new messages, the VMS cancels Message Wait Notification set for extensions B, C and D.

Call Forward

Just as calls can be forwarded to a Department Group, a Department Group can also forward its calls to:

- an extension
- its own Department Group Mailbox
- another Department Group

For Department Groups, SARVAM UCS does not support Call Forward to an external destination number.
You can set Call Forward for Department Group from the SA Mode only.

SARVAM UCS supports the following Call Forward options for Department Groups:

- **Call Forward - unconditionally:** calls are forwarded to the destination number, without checking the status or waiting for a response from the Department Group.

- **Call Forward- if Busy:** calls are placed on the Department Group as per the Rotation configured for it and are forwarded to the set destination, only when all the member extensions of the Department Group are found to be busy.

- **Call Forward- if No Reply:** when a call is made to the Department group, SARVAM UCS will place the call as per the Rotation configured for the Department Group for the duration of the '*Call Forward No Reply Timer for Department*' (default: 30sec). If none of the member extensions answers the call before the expiry of this timer, the call is forwarded to the destination.

  If you select this option, you may set the *Call Forward No Reply Timer for Department* to the desired value. This Timer is commonly applied on all Department Groups which set Call Forward No Reply.

- **Call Forward - if Busy/No Reply:** calls made to the Department Group will be routed to the destination, if all members of the Department Group are busy or when none of the member extensions answered the call within the Call Forward No Reply Timer for the Department.

- *Call Forward - Dual Ring is not supported for Department Groups.*

- *Member extensions of a Department Group can set Call Forward on their extensions. However, Call Forward set for the Department Group will have precedence over Call Forward set by individual member extensions.*

- *Call Forward set by member extensions in a routing group will be ignored by the system if, the Ignore call forward set by member extension, when call is routed on Routing/Dept. Group flag is enabled. See "System Parameters" for more information.*

- *Call Forward for a Department Group can also be set from SA Mode. See "Setting Call Forward for Department Group".*

Again, taking the above example of Department Group 1 further, here's how call forward will work:

- Extensions A, B, C, D of Department Group 1 are allowed Call Forward Department Group in their Class of Service.

- Any of them can set Call Forward for Department Group 1. Extensions A, B, C and D can also set Call Forward on their own extensions.

- When any extension or an external caller (also using Auto Attendant or Direct Inward System Access) dials the Access Code 3901 to call Department Group1, SARVAM UCS will check the Call Forward option set for the Department Group and route the call accordingly.

- If Call Forward - unconditionally is set, the call will be routed to the destination number, if the *Ignore call forward set by member extension, when call is routed on Routing/Dept. Group is enabled*.

- If Call Forward - Busy is set, and the first extension in the Department Group is busy, the system will hunt for the next free extension in the group. It will continue to hunt for a free extension. If all extensions in the group are busy, the call will be forwarded to the destination number.

  Call Forward unconditional, busy, or busy/No reply set by any member extension will not work.

- If Call Forward - No Reply is set, the system will start the Call Forward No Reply Timer Department Group and place the call as per the Rotation set for the Department Group. If the call is not answered by any of the extensions before the timer expires, the call will be forwarded to the destination number.

If a member extension that is offered the call has set Call Forward-Unconditional, and the Call Forward No Reply Timer Department Group has not expired, the call forward set by the extension will be applied. If the timer expires, the Call Forward No Reply set for the Department Group will be applied.

If a member extension that is offered the call has set Call Forward-No Reply, or No-Reply/Busy, the Call Forward No Reply Timer (for individual extension) will start simultaneously with the Call Forward No-Reply Timer Department Group. If the No Reply Timer for the extension expires first, the call will be forwarded to the destination set for the extension. If the No Reply Timer of the Department Group expires first, before the call is answered, the call will be forwarded to the destination set for the Department Group.

Call Forward-No Reply Timer can be set from the SE Mode only.

How to configure

The functioning of this feature requires you to do the following:

- create Department Groups.
- configure "Routing Group" (each routing group consisting of extensions related to a Department) and assign the routing groups and appropriate access codes to the department groups.

If you want to provide voice mail facility to the Department Group, you must:

- assign a Mailbox to the Department Group.
- allow member extensions access to the Department Group Mailbox.

If you want to enable Call Forward to the Department Group, you must:

- enable 'Department Group Call Forward' in the Class of Service (CoS) of the member extensions.
- change, if required the default value of the *Call Forward No Reply Timer for Department Group*.

Creating Department Groups

- On a sheet of paper, draw a table.
- Decide the number of department groups you want to create, e.g.: 4 groups.
- Group all the extensions you want to put in each department group. You cannot group more than 32 extensions in a single department group.
- Decide in what sequence the extensions in each group should ring, that is, which extensions should ring first, second, third, and so forth.
- Decide the access code you want to assign to each department group.
- You may also assign a name to the department group.
- Your table may look like this:

Department Group Index	Access Code to be assigned	Name	Extensions to be included as members			
			SLT	DKP	SIP Extension	ISDN Terminal
1	3901		2001, 2002, 2003	3010, 3011, 3012	3301, 3302	

Department Group Index	Access Code to be assigned	Name	Extensions to be included as members			
			SLT	DKP	SIP Extension	ISDN Terminal
2	3902		2015, 2020, 2021	3015, 3020	3305	
3	3903		2017, 2028	3021	3305	
4	3904		2023, 2024, 2025	3025	3315	

The access codes for the department groups and extensions in this table are default access codes.

- Now, with this information ready, you may configure the department groups using Jeeves or dialing the relevant SE commands from a Telephone.

Enabling Call Forward for Department Call in Class of Service

To be able to set Call Forward for the Department Group, member extensions must have this feature enabled in their COS.

In the default Station Basic Feature Template Number 01 is assigned to all the extensions of SARVAM UCS, the default COS group 01 in the template has ‘Department Group Call Forward’ enabled. So, all extensions of SARVAM UCS can set Call Forward Department Group.

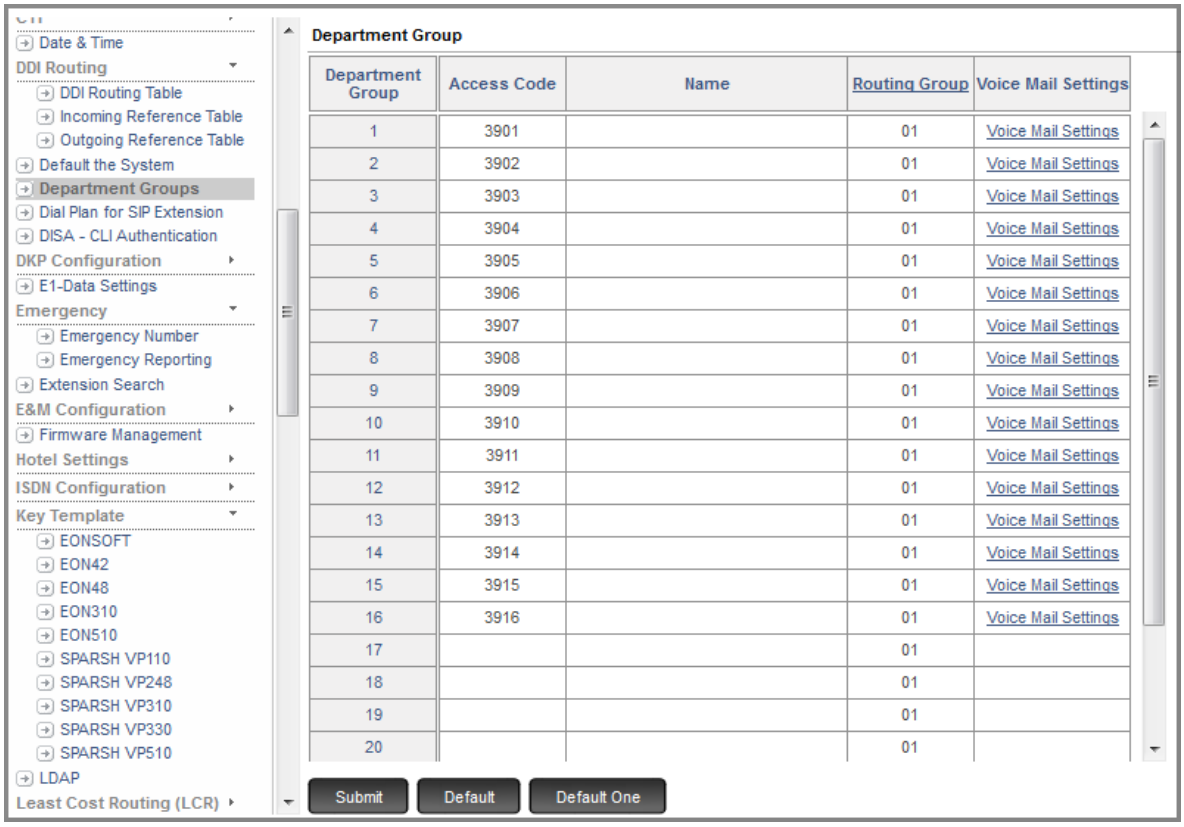
If you wish to allow this feature to member extensions only, retain this feature ‘Department Group Call Forward’ in the COS group of member extensions, and disable Call Forward feature in the COS of all the extensions. For this, you may create separate templates for member extensions and other extensions.

Refer the topic “Class of Service (COS)” and “Station Basic Feature Template” for instructions.

Configuring Department Groups using Jeeves

- Log in as System Engineer.

- Under **Configuration**, click **Department Groups**.



Creating Department Groups

- To create a Department group, assign an **Access Code** for the department group against the Index Number.

  By default, the Access Codes assigned to Department groups for Index Numbers 1 to 16 are 3901 to 3916 and for Index Numbers 17 to 24 is blank.

  If you decide not to use the default access codes, ensure that the access code you assign to each department group is unique and does not match with any SLT, DKP, ISDN, SIP access code or any feature access code of the Dial Phase. Refer the topic “Access Codes” to know more.

  To assign Station Access Codes according to your preference and requirement to a range of Department Groups, see “Assigning Access Codes to a Range of Extensions”.

- You may also assign a **Name** to the department group to facilitate identification. This name will appear in the Dial by Name directory along with the department group number. The Name can be a maximum of 18 characters.

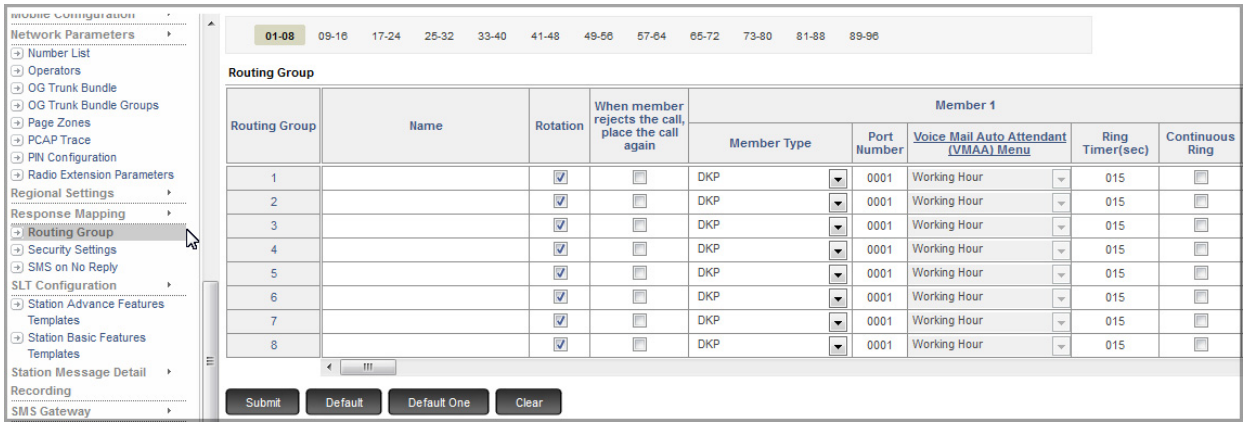
- Now, enter the **Routing Group**, that is, the number of the group you created for this department. Where multiple departments exist, you must create separate routing groups for each department group.

  This can be done in two ways:

  - create the routing groups first and simply enter the relevant routing group number against the Department Group Index (to which you have assigned the access code).

OR

- in Routing Group, click the **Routing Group** link. The Routing Group page opens. Now create the Routing Group as per your requirement. For details, see “Routing Group”.



- In **Voice Mail Settings**, click the Voice Mail Settings link. The respective Extension Voice Mail Settings window will open. You may edit the parameters. For details, see “Extension Voice Mail Settings”.

The Voice Mail Settings link will be visible only if you have configured the respective access codes.

Click **Close** to close the window.

- Click **Submit** to save your settings.

For example:

The Customer Care Department of a company has four extensions: 201, 202, 203 and 204 (on software ports 001, 002, 003, and 004 respectively), which needs to be grouped for Department Calls.

Extensions 201 and 202 are SLTs. Extensions 203 and 204 are DKPs.

Requirement: The company wants the following:

1. Call traffic should be distributed equally on all four extensions.
2. Ring sequence should be 201, 202, 203, and 204 (first to last).
3. First 201 should ring for 20 seconds.
4. If no reply, 201 should continue to ring and 202 should ring for 10 seconds.
5. If still no reply, 201 should continue to ring and 203 should ring for 15 seconds.
6. If still no reply, 201 should continue to ring and 204 should ring for 20 seconds.
7. 51 should be the access code for this department group.

Solution: Select a routing group e.g. 03, and configure as follows:

1. 201 as member 01, with member type SLT, and Port number 001.
2. 202 as member 02, with member type SLT, and Port number 002.
3. 203 as member 03, with member type DKP, and Port number 003
4. 204 as member 04, with member type DKP, and Port number 004.

Set ‘member type’ of members 5 to 32 in Routing Group 3 to 'None'.

1. Enable the 'Rotation Flag' on routing group number 03 to distribute call traffic.
2. Enable the 'Continuous Ring Flag' for member 01 (201) and set the 'Ring Timer' to '20 seconds.
3. Set the Ring Timer of member 02 (202) to 10 seconds. Disable 'Continuous Ring' flag.
4. Retain the Ring Timer of member 03 (203) as default 15 seconds. Disable 'Continuous Ring' flag.
5. Set the Ring Timer of member 04 (204) to 20 seconds. Disable 'Continuous Ring' flag.
6. Assign 51 as access code and routing group 03 to the Customer Care Department.

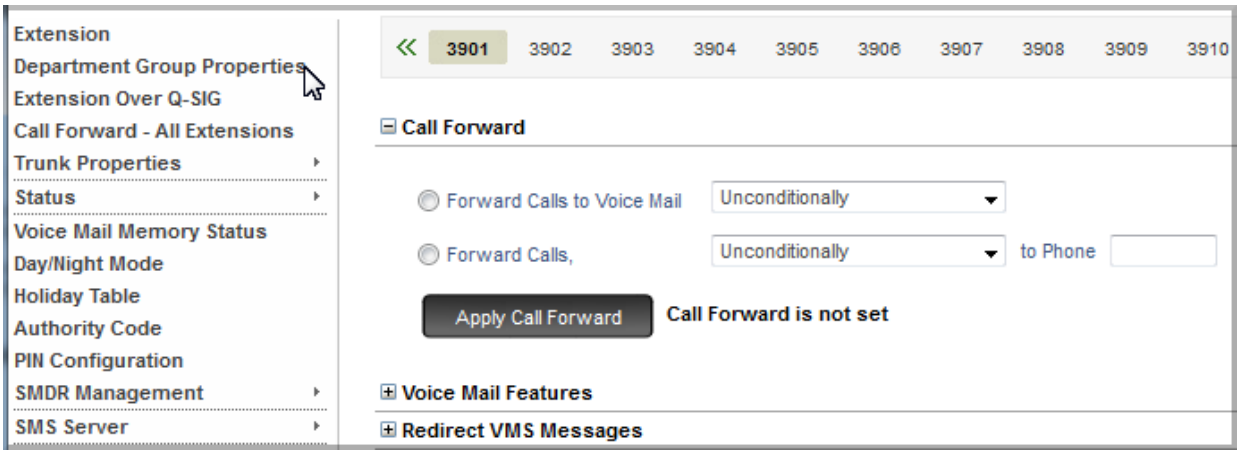
   Remember to click **Submit** to save your settings.

Setting Call Forward for Department Group

You can set Call Forward for Department Group from SA Mode only.

To set Call Forward,

- Log in to Jeeves as System Administrator.
- Click **Department Group Properties**.



- Click the desired Department Group Number tab for which you want to set Call Forward.
- Click **Call Forward** to expand.
- You have two options for Call Forward:
  - To forward all department calls to Voice Mail, select the **Forward Calls to Voice Mail** and select the Call Forward type from the combo box.
  - To forward all department calls to a specific number, select **Forward Calls-to Phone** and enter an extension number where the call is to be forwarded.
- Click the **Apply Call Forward** button.

  The color of the text indicating that Call Forward is set will change to red.

To cancel Call Forward,

- Click the **Cancel Call Forward** button.

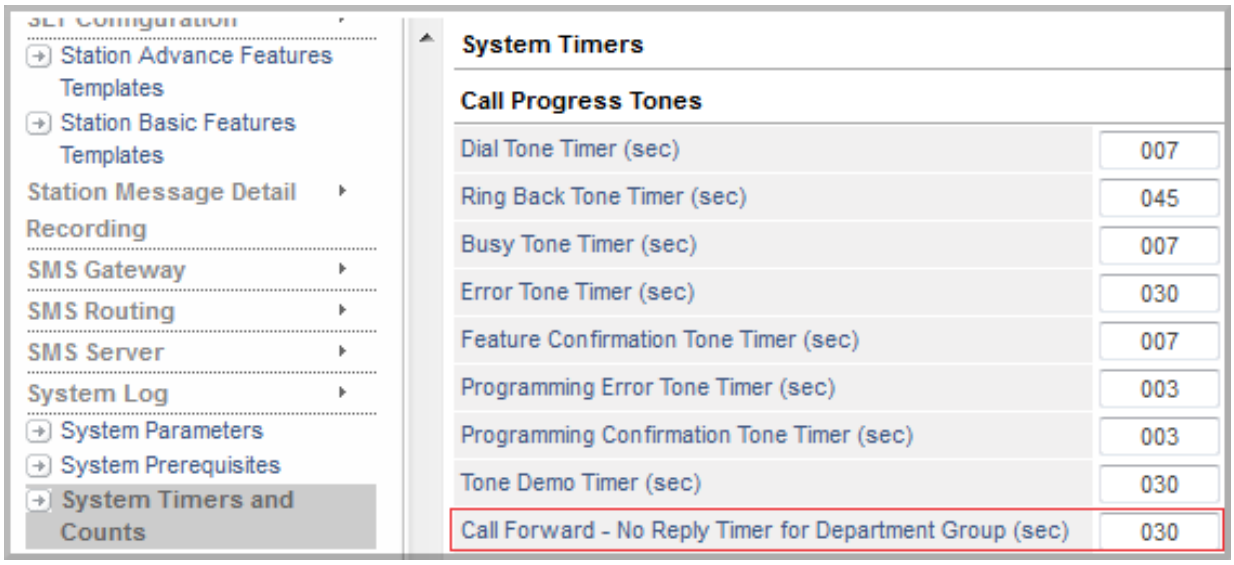
  The color of the text indicating that Call Forward is not set will change to black.

Setting Call Forward No Reply Timer for Department Group

If you have enabled Call Forward Department Group, you may change, if required, the Call Forward No Reply Timer for Department Group. You can change this timer only from SE Mode. By default the Timer is set to 30 seconds.

To change the Call Forward No Reply Timer,

- Log in as System Engineer.
- Under **Configuration**, click **System Timers and Count**.



- Scroll with the vertical bar to **Call Forward No-Reply Timer for Department Group (sec)**.
- Set the timer to the desired value. The range of this timer is 1 to 255 seconds.
- Click **Submit** to save your timer setting.

Configuring Department Groups using a Telephone

- Enter SE mode from a DKP/SLT.

  To create a routing group with members, dial:
  - ***6502-1-Routing Group-Destination Index-Port Type-Port Number***

    Where,
    Routing Group is the number of the Routing Group 01 to 96.
    Destination Index is from 01 to 32.
    Port Type is the 'Member type':
    00 for None
    01 for SLT
    02 for DKP
    28 for ISDN terminal
    34 for SIP Extension
    36 for Virtual Extension
    16 for OTBG

Port Number is the Software port number on which the member extension SLT, DKP, SIP Extension, ISDN Terminal is attached.
Software port number of the SLT, from 001 to 240.
Software port number of the DKP, from 001 to 096.
Software port number of the ISDN Terminal, from 01 to 64.
Software port number of the SIP extension, from 001 to 999.
Virtual Extension is from 01 to 64.
OGTBG Number is from 01 to 32.
E.g.: To include the four extensions in the above example in Routing Group 03, dial:

- ***6502-1-03-01-01-001*** to include extension 201.
- ***6502-1-03-02-01-002*** to include extension 202.
- ***6502-1-03-03-02-003*** to include extension 203.
- ***6502-1-03-04-02-004*** to include extension 204.

To set the Ring Timer for each member extension in the routing group, dial:

- ***6503-1-Routing Group-Destination Index-Ring Timer***
  Where,
  Routing Group is the number of the Routing Group 01 to 96.
  Destination Index is number of the member extension in the routing group from 01 to 32.
  Ring Timer is from 000 to 255 seconds. (Default: 015 seconds)

  E.g.: To set the Ring Timer for the individual extensions of the Routing Group 03 in the above example, dial:

  - ***6503-1-03-01-020*** to make extension 201 ring for 20 seconds.
  - ***6503-1-03-02-010*** to make extension 202 ring for 10 seconds.
  - ***6503-1-03-03-015*** to make extension 203 ring for 15 seconds.
  - ***6503-1-03-04-020*** to make extension 204 ring for 20 seconds.

To set the Continuous Ring Flag for extensions in the routing group, dial:

- ***6504-1-Routing Group-Destination Index-Flag***
  Where,
  Routing Group is the number of the Routing Group 01 to 96.
  Destination Index is the number of the member extension in the routing group from 01 to 32.

  Continuous Ring Flag is:
  0 for disable continuous ring (each member extension in the group will ring for the duration of the 'Ring Timer' for the group).

  1 for enable continuous ring (the first extension in the group will ring till the call is answered).

  E.g.: To enable/disable the Continuous Ring Flag for the individual extensions of the Routing Group 03 in the above example, dial:

  - ***6503-1-03-01-1*** to enable the flag on extension 201.
  - ***6503-1-03-02-0*** to disable the flag on extension 202.
  - ***6503-1-03-03-0*** to disable the flag on extension 203.
  - ***6503-1-03-04-0*** to disable the flag on extension 204.

To turn Rotation ON for the Routing Group, dial:

- ***6505-1-Routing Group-Rotation Flag*** to enable rotation on a single routing group.

- ***6505-2-Routing Group-Routing Group-Rotation Flag*** to enable rotation on a range of routing groups.
- ***6505-*-Rotation Flag*** to enable rotation on all routing groups.
  Where,
  Routing Group is the number of the Routing Group 01 to 96.
  Rotation Flag is:
  0 for Disable (fresh call lands on the first extension within the group).
  1 for Enable (fresh call lands on the extension next to the one that received the previous calls).
  E.g.: To enable Rotation Flag for Routing Group 03 as in the above example, dial ***6505-1-03-1.***

To assign routing group to the department group, dial:

- ***2001-1-Department Group Index-Routing Group***
  Where,
  Department Group Index is from 01 to 24.
  Routing Group is from 01 to 96. Default: 01.

  E.g.: To assign Routing Group 03 to Department Group 01 (Customer Care) as in the above example, dial ***2001-1-01-03.***

To clear the routing group assigned to department group, dial:

- ***2001-1-Department Group Index-00*** to clear the routing group of a single department group.
- ***2001-2-Department Group Index-Department Group Index-00*** to clear the routing groups of a range of department groups.
- ***2001-*-00*** to clear the routing groups of all department groups.

To assign an access code (that is, Department Number) to a department group, dial:

- ***3113-1-Department Group Index-Access Code***
- ***3113-2-Department Group Index-Department Group Index-Access Code***
- ***3113-*-Access Code***
  Where,
  Department Group Index is from 01 to 24.
  Access Code is a string of a maximum of 6 digits.
  Terminate the command string with #* if the Access Code has fewer than 6 digits.

  E.g.: To assign '51' as Access Code for the Department Group 01 as in the above example, dial ***3113-1-01-51-#****

To clear the access code assigned to a department group, dial:

- ***3113-1-Department Group Index-#**** to clear access code assigned to a single department group.
- ***3113-2-Department Group Index-Department Group Index-#**** to clear access codes assigned to a range of department groups.
- ***3113-*-#**** to clear access codes of all department groups.

To default the access codes assigned to Department groups, dial:

- ***3163-1-Department Group Index*** to default the access code of a single department group.
- ***3163-2-Department Group Index-Department Group Index*** to default the access codes of a range of department groups.
- ***3163-**** to default the access codes of all department groups.

- Exit SE mode.

How to use

Making a Department Call

Making a department call is the same as calling another extension.

For EON and Extended IP Phone Users

- Press DSS Key assigned to the desired Department Group.
  OR
- Dial the desired Department Group Number (default: 3901-3916).
- You get Ring Back Tone as the call lands on an extension within the department group.
- Talk when the call is answered.
- Go Idle or you get dial tone after 3 seconds.

For SLT Users

- Lift handset
- Dial the desired Department Group Number (default: 3901-3916).
- You get Ring Back Tone as the call lands on an extension within the department group.
- Talk when the call is answered.
- Replace handset.

Accessing Department Group Voice Mail

For EON and Extended IP Phone Users

When LED of Voice Mail Key is turned on to indicate new message,

- Press Voice Mail key.
- VMS informs you about new message(s) in your Department Group mailbox and Personal Mailbox.
- Follow Voice Mail prompts to access Department Group mailbox.
- Press 2 to go to Department Group mailbox.
- Press 1 to listen to new messages.
- Go ON-Hook or follow voice prompts for the desired option.

For SLT Users

New messages in your mailbox will be notified according to the type of Message Wait Notification set on your extension phone.

When the LED Lamp of the Message Waiting Key glows,
OR
When your phone plays Message Wait Ring[288],

- Lift the handset.
- Dial ***3931***.
- Follow voice prompts.
- Press ***2*** for Department Group mailbox.
- Press ***1*** to listen to new messages.
- Replace handset after listening to messages. Or you may follow voice prompts.

---

288. Refer "Distinctive Rings" for description.

If you hear Stuttered Dial Tone/Voice Message when you go OFF-Hook,

- Dial ***3931***.
- Follow voice prompts.
- Press ***2*** for Department Group mailbox.
- Press ***1*** to listen to new messages.
- Replace handset after listening to messages or follow voice prompts to continue.

Setting Call Forward for Department Group

For EON and Extended IP Phone Users

- Press DSS Key assigned to the Call Forward Department Group.
  OR
- Dial ***1179*** (users worldwide). Users in the Philippines, dial ***1108***.
- Enter the Department Group Number whose calls are to be forwarded.
- Scroll to select Call Forward Type from the following options:
  - Cancel Call Forward
  - Forward Unconditionally
  - Forward when Busy
  - Forward when No Reply
  - Forward when Busy/No Reply
- Press Enter key.
- Enter Destination number (extension number or voice mail)
- You get confirmation tone and message on your phone's LCD.
- Go Idle.

For SLT Users

- Lift handset.
- Dial ***1179*** (users worldwide). Users in the Philippines, dial ***1108***.
- Dial the Department Group Number whose calls are to be forwarded.
- Dial Call Forward Type:
  - ***1*** for Forward Unconditionally
  - ***2*** for Forward when Busy
  - ***3*** for Forward when No Reply
  - ***4*** for Forward when Busy/No Reply
  - ***0*** to Cancel Call Forward
- Dial Destination number (number of the desired extension or voice mail)
- You get confirmation tone.
- Replace handset.

Dial By Name

What's this?

Dial By Name enables extension users to call another extension or an external party by dialing the name of the person, instead of dialing their telephone number.

This feature is accessible only to users of the proprietary Digital Key Phones and the Extended IP phones of Matrix.

With Dial By Name users need not remember the desired party's telephone number or short codes, that is, “Abbreviated Dialing” codes.

For each extension, the database for names used in Dial by Name is drawn from:

- the **Personal Directory**, which is assigned to each extension, wherein up to 25 external party numbers along with their names may stored. The system uses the Personal Directory to dial external parties by their names. See “Abbreviated Dialing” to know more.

- the **Global Directory**, which is assigned to the extension in its “Class of Service (COS)”. The Global Directory is a system-wide list of external party numbers and names. Up to 999 numbers can be stored in this directory, and parts of the Global Directory (Part 1, 2, 3) can be assigned to each extension in its Class of Service. See “Abbreviated Dialing” to know more.

- **Names of Extensions**, which are names of users/departments groups. Their names are assigned to SLT, DKP and SIP extensions to identify the extension users. Names of Extensions are necessary for making internal calls using the Dial By Name feature.

How it works

- Extension user presses the DSS Key assigned to 'Dial By Name' feature.
- On EON48 models and on SPARSH VP248, press the 'Names' key. On EON310/SPARSH VP310, press the ‘Contacts’ key.
- The prompt <Name: > appears on the phone display.
- User enters the name of the desired party[289].
- For example, user wants to call Midas Biz, and enters the letter 'M' using the keypad.
- The system displays in alphabetical order, all names starting with 'M'. These numbers are drawn from the Personal and Global Directories assigned to the extension and the Extension Names programmed in the system.
- User scrolls the list using the Up/Down navigation keys to reach the desired contact's name.

  OR

  Instead of scrolling the entire list, the user enters more than one initial letter of the contact's name. The search is narrowed down to more accurate matches. The phone displays the matching entries in the directory.

- The user must select the desired name by pressing 'Enter' Key.

289. *The process of entering the names is the same as when writing text messages (SMS) from a cell phone. The keys must be pressed multiple times in quick succession to enter the desired alphabet.*

- The system dials out the number stored under the selected name. The name and number are displayed on the user's phone.

How to configure

For this feature to work, the following must be programmed:

1. **DSS Key:** A direct station selection (DSS) key must be programmed for the Dial by Name feature. Without the DSS Key this feature will not be accessible.

   The factory-default key map of EON48 and SPARSH VP248, phones have the DSS Key labeled as 'Names'. EON310/SPARSH VP310 have a fixed feature key ‘Contacts'

2. **Global Directory:** The names of the external parties must be programmed against their respective telephone numbers in the directory. Refer the topic “Abbreviated Dialing” for instructions on programming the Global Directories.

3. **Personal Directory:** The names of the external parties must be programmed against their respective telephone numbers in the Personal Directory. Refer the topic “Abbreviated Dialing” for instructions on programming the Personal Directories.

4. **Extension Names:** Extensions may be SLTs, DKPs, ISDN Terminals or SIP extensions. Refer the topics related to configuration of extensions[290].

5. **Class of Service:** Dial By Name is allowed to all DKP users. However, the use of this feature is related to the following features, which must be enabled in the Class of Service of the DKP and SIP extension users:

   - Internal Calls - This is a part of the Basic Features. By default these are enabled.
   - Global Directory Part 1
   - Global Directory Part 2
   - Global Directory Part 3.

   If you want the names to be drawn from Global Directory Part 1, Part 2 and Part 3, provided these are programmed, you must enable these directories in the CoS of the DKP and SIP extensions.

   Refer “Abbreviated Dialing”, for instructions on programming the Global Directory.

   Refer “Class of Service (COS)” and “Station Basic Feature Template” for programming instructions on how to enable a feature in the CoS and how to apply it on extensions.

The system will display the names exactly as they have been programmed in the Personal and Global Directories and the SLT/DKP/ISDN Parameters.

290. *You may also refer the instructions provided under the topic Configuring Extensions: “Configuring SLT Extensions”, “Configuring DKP Extensions”, “Configuring ISDN Terminals” , “Configuring Matrix SPARSH VP330”, “Configuring Matrix SPARSH VP248” , “Configuring Matrix SPARSH VP310” , “Configuring Matrix SPARSH VP510”,* “Configuring Matrix SPARSH VP210”

How to use

For EON48 and Extended IP Phone Users

- Press 'Names' key.
- You get the prompt: 'Name'.
- Enter the initial letters for the contact's name.
- The number of matching entries that will appear at a time on your phone's display will vary according to your phone's LCD display capacity.
- Scroll with the Up/Down navigation keys to reach the desired contact's name on the list.
- Press 'Enter' key to select the name.
- The system displays the name and number being dialed out.
- You get Ring Back Tone or Busy Tone.

Entered the wrong alphabet?

- Go ON-Hook.
- Go OFF-Hook.
- Press the DSS Key labeled 'Names' again.
- Enter the name/initial letters of the contact's name.

Dialed Number Directory

What's this?

Dialed Number Directory feature is available only to the users of the proprietary DKP, EON and Extended IP Phones.

It is the list of numbers dialed out from the phone, similar to the call history of recently dialed calls on a cell phone.

SARVAM UCS retains up to 16 numbers dialed out from a phone in a directory.

These numbers may have been dialed out using features like Abbreviated Dialing, Quick Dial, Redial, Walk-In Class of Service, or may be a simple outgoing call made by directly dialing the external number.

How it works

- When a DKP/IP Phone extension user makes an outgoing call, the number is stored in the Redial Number List.

By default the system stores only the external numbers in the Last Number Redial List. If you want the system to store internal calls in this list, make sure you enable the ***Store Internal Calls in Redial Call Log*** *check box in the "System Parameters".*

- The list has a capacity of storing a maximum of 16 recently dialed numbers.
- The list is updated using the First-In First-Out logic, whereby the earliest dialed number is replaced with the most recently dialed number.
- To use this feature, the phone user must invoke the "Last Number Redial" feature.
- Doing so, the Redial Number List will appear on the phone display.
- The user may now navigate the list, select the number to be dialed out.
- The system will dial out the selected number using the same Outgoing Trunk Bundle Group used to place this call earlier.
- If the number had been dialed earlier using Abbreviated Dialing, the system will check for Toll Control when dialing out the number again from the dialed number directory[291].

291. Recall that the system does not check for Toll Control when Abbreviated Dialing is used.

How to configure

No specific programming required.

How to use

For EON and Extended IP Phone Users Only

- Press the ‘Redial’ Key.

  OR
  Dial 7

- The list of last dialed calls appear on your phone's display.
- Scroll with up/down navigation key to reach the desired number.
- Press 'Enter' key.
- The desired number is dialed out and appears on your phone's display.
- You get ring back tone.
- Talk when speech is established.
- Go ON-Hook after the conversation has ended.

Dial Plan for SIP Extension

What's this?

SARVAM UCS supports 8 Dial Plans with total 32 entries in each table. The Dial Plan contains a series of digits and/or wild card characters.

When a user dials a number, it is compared with the Rule configured in the Dial Plan. If a match is found, the IP Phone routes the call immediately without waiting for End of Dialing and if a match is not found, the IP Phone will wait for the End of Dialing and then route the call.

How to configure

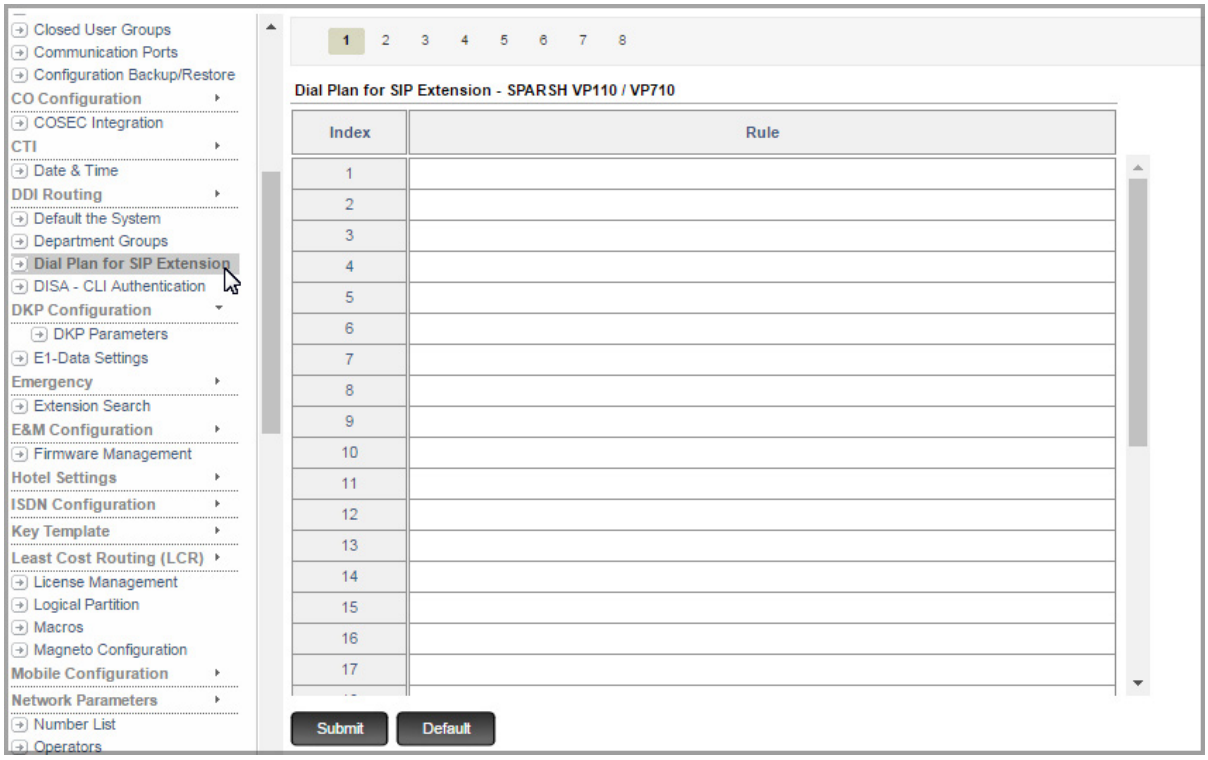
Programming the Dial Plan involves the following steps:

- Selecting a Dial Plan and configuring the rules in the Dial Plan.
- Assigning the Dial Plan to the desired SIP Extensions.

Programming Dial Plan using Jeeves

- Log in as System Engineer.

- Under **Configuration**, click **Dial Plan for SIP Extension**.



- Click the desired Dial Plan number. The **Dial Plan for SIP Extension - VP110/VP710** page opens.

- Against each **Index** configure the **Rule** according to which you want the system to process the call. You can configure a maximum of 32 Rules in each Dial Plan.

  For example, if you want that users should be able to dial extension numbers from 3000 to 3999 without any delay, configure the Rule as 3XXX.
  For more details to configure the rules, refer to the topic *Dial Plan* in the *SPARSH VP110 User Guide*.

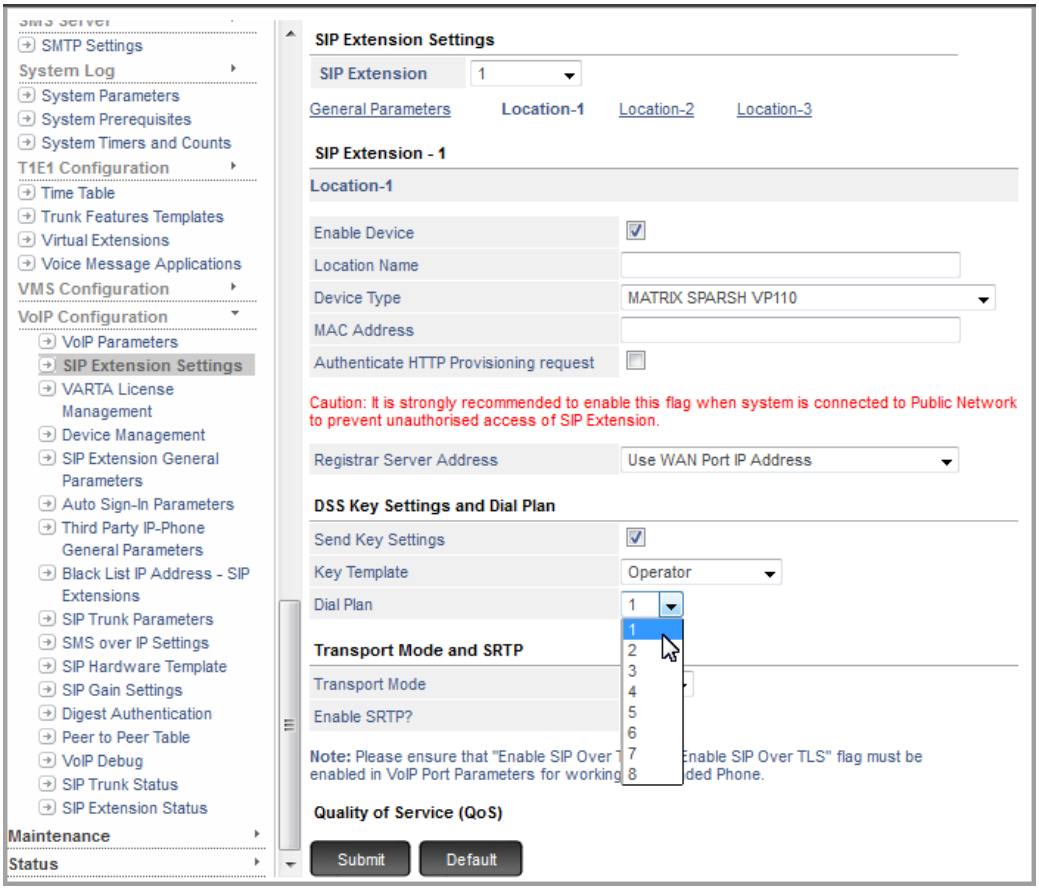
- Click **Submit**.

Assigning Dial Plan to SIP Extensions

To assign the Dial Plan you configured for VP110,

- Under **VoIP Configuration**, click **SIP Extension Settings**.

- Click the location at which you have registered SPARSH VP110, for example, **Location 1**.



- Scroll to **DSS Key Settings and Dial Plan.**

- Select the **Dial Plan** number you configured.

- Click **Submit**.

To assign the Dial Plan you configured for VP710,

- Under **VoIP Configuration**, click **SIP Extension Settings**.

- Click the location at which you have registered SPARSH VP710, for example, **Location 1**.
- Scroll to **Dial Plan.**
- Select the **Dial Plan** number you configured.
- Click **Submit**.

Digest Authentication

What's this?

Digest Authentication is a challenge-based authentication service of SIP to authenticate the identity of the originator of SIP request in the INVITE message. The recipient of the request can ascertain whether or not the originator of the request is authorised to make the request. When the digest credentials of the originator—User Name and Password—in the INVITE message are authenticated and accepted by the recipient, the originator and the recipient are connected.

SARVAM UCS supports Digest Authentication. You may use Digest Authentication to

- restrict access to SARVAM UCS to specific callers.
- prevent unwanted or malicious calls.

How it works

The Digest Authentication feature works on the basis of the Digest Authentication Table, in which the credentials, namely the User Name and Passwords of trusted/authorised calling party SIP devices are stored. You must configure this table. The Digest Authentication Table is common for all SIP trunks on which this feature is enabled.

When you enable this feature on a SIP trunk, for all incoming calls (SIP requests),

- SARVAM UCS will challenge the identity of the calling party (the SIP device initiating the request) to send its digest credentials.
- When the calling party sends its credentials, SARVAM UCS authenticates the credentials by matching it with its Digest Authentication Table.
- If a match is found, the calling party will be authenticated and the call will be allowed on the SIP trunk.
- If no match is found, SARVAM UCS will consider it as invalid authentication information and reject the call.

How to configure

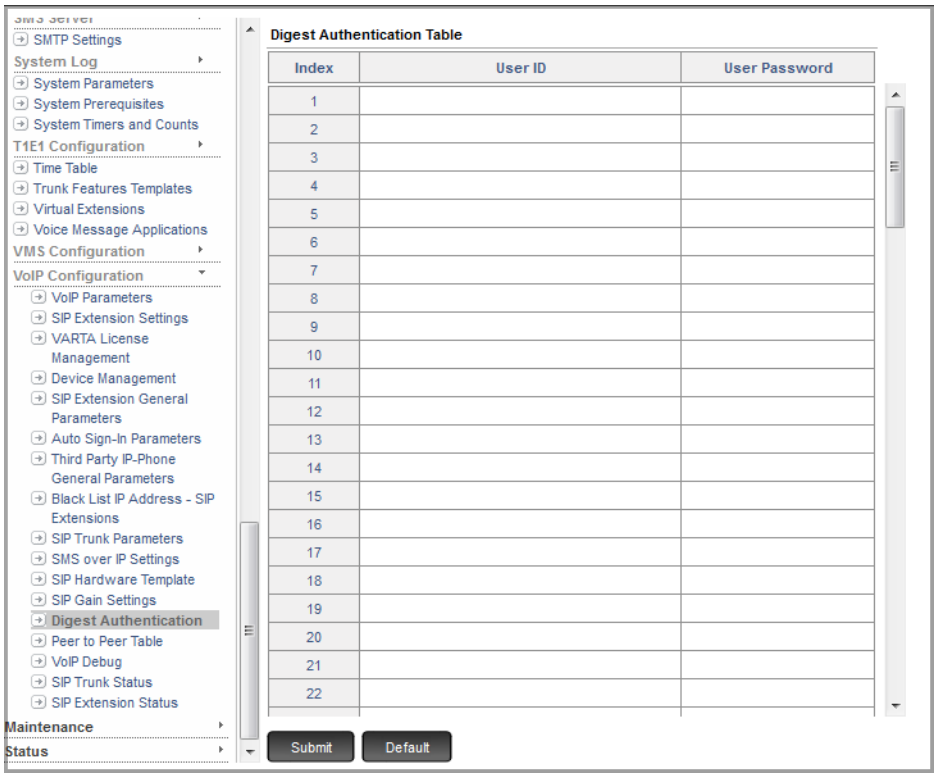
To use this feature on SIP Trunks, you must do the following:

- Enable Digest Authentication on the SIP trunks you want to use this feature.
- Configure the Digest Authentication Table.

You can configure the Digest Authentication Table using Jeeves and from a telephone.

Configuring Digest Authentication using Jeeves

- Log into Jeeves as System Engineer.
- Under **Configuration**, click the **VoIP Configuration** link.
- Click **Digest Authentication**.



The Digest Authentication table opens.

- In the **User ID** field, enter the User ID to be authenticated. The User ID must be within 40 characters.
- In the **User Password** field, enter the corresponding Password. The Password must be within 16 characters.

  To avoid unauthorized access, we recommend you to change the Password regularly. Make sure it is strong and is kept confidential.
- Click **Submit** to save entries.
- Now, enable Digest Authentication on the desired SIP trunks. For instructions, see “Configuring SIP Trunks”.

Configuring Digest Authentication using a Telephone

- Enter SE mode from a DKP/SLT.

To configure the User Password for the User ID, dial:

- ***4119-Index-User Password***
  Where,
  Index is from 01 to 99. The User Password should be configured at the same index as the User ID.
  User Password may have a maximum for 16 characters. If the User Password has fewer than 16 characters, terminate the command string with **#*** if using an SLT or press the 'Enter' key if using a DKP.
  Default: Blank.
- Exit SE mode.

For SE Command for enabling Digest Authentication on SIP trunks, see “Configuring SIP Trunks”.

Digital Key Phone-Operation

EON - the Proprietary Digital Key Phone

Matrix offers 'EON', the proprietary digital key phone (DKP). Matrix offers various models of EON. Each of these is explained later. These are powerful stations, supporting a host of phone and System features, as listed below.

DKP Features

- Status of other ports (Tri-color LED indication)
- Programmable Direct Station Selection (DSS) Keys and Feature keys
- LCD notification messages
- Ringer Tune selection
- Adjustable Speech level
- Adjustable Ringer Volume
- Adjustable Backlight and Contrast levels
- Hands-free operation - Speaker key and headset connectivity.
- Call Logs - last 20 Missed, Answered and Dialed Calls.
- Operator, Executive, Hotel Attendant and Guest Functionality
- Message Paging
- Menu based operation of System features
- Multiple Language support.

System Features

Listed below are the features of SARVAM UCS that require a Digital Key Phone:

- Abbreviated Dialing
- Auto Answer
- Call Chaining
- Call Cost Display
- Call Duration Display
- Call Mute
- Dialed Number Directory
- Directory Dialing by Name
- Dynamic Lock
- Forced Answer
- Keypad Lock
- Live Call Screening
- Message Paging
- Off-Hook Alert
- Room Monitor
- Text Message Reply
- Time Zone Display
- User Status (Presence)

Models of EON at a Glance

Feature	Models			
	EON48S	EON48P	EON310	EON510
Total number of keys	48	48	42	49
Number of programmable keys	29	29	21	22
Capsense keys	Yes	Yes	No	No
Context Sensitive keys	No	No	No	Yes
Feature keys	12	12	9	8
DSS keys	16	16	12	16
LCD display	2 lines x 24 characters	6 lines x 24 characters	2 lines x 24 characters	240 x 64 Pixel Graphical LCD display
LCD with backlight	Yes	Yes	Yes	Yes
Headset Interface	Yes	Yes	Yes	Yes
Ringer Lamp (LED)	Yes	Yes	Yes	Yes
Speaker Phone	Full duplex	Full duplex	Full duplex	Full duplex

EON48

EON48S

2 lines and 24 characters LCD display, full duplex, capsense feature keys

EON48P

6 lines and 24 characters LCD display, full duplex, capsense feature keys.

LCD Display

The LCD display of EON48 is backlit and can be tilted at a convenient angle for a clear view of the text/characters displayed.

The LCD backlight can be turned on and off as well as adjusted for contrast and brightness from the *"Phone Settings"* of the DKP Phone Menu.

Ringer LED

The Ringer LED indicates incoming internal and external calls. The LED Cadence will match with the Ring Cadence of the incoming internal/external call.

The Ringer LED cadence changes according to the type of call, as described in the table below.

Type of Call	Cadence
Internal Call	Short, very slow (750ms ON, 2250ms OFF)
External Call	Double (400ms ON, 200ms OFF, 400ms ON, 2000ms OFF)
Alarm	Long, fast (1500ms ON, 500ms OFF)
Auto Redial Call	Long, very slow (2000ms ON, 4000ms OFF)
Auto Call Back Call	Short, slow (750ms ON, 2250ms OFF)
Priority	Triple (400ms ON, 200ms OFF, 400ms ON, 200ms OFF, 400ms ON, 2000ms OFF)
Programming mode	Continuous

Navigation Keys

These are 5 capsense keys. The functions of each are described briefly below.

- **Enter Key:** To enter the Menu; when the phone is in the idle state (without any incoming or outgoing call being made), if you tap the 'Enter' key, you will enter into the 'Menu'.

  Enter key is also used to make a selection from the Menu/sub-menu options or to complete an action.

- **Up Key:** To scroll upwards while navigating the Menu/sub-menu.

- **Down Key:** To scroll downwards while navigating the Menu/sub-menu.

- **Forward Key:** To move forward when dialing a number.

- **Back Key:** To move backwards when dialing a number; to go back one level in the Menu.

Feature Keys

These are 12 capsense keys assigned to important or frequently accessed features of SARVAM UCS. Refer to the table given below:

Sr.No.	Description	LED
1.	Voice Mail	Single Color - Blue
2.	Call Back	Single Color - Blue
3.	Cancel	No
4.	Mute	Single Color - Blue
5.	Conference	No
6.	Transfer	No
7.	Forward	Single Color - Blue
8.	DND	Single Color - Blue
9.	Names	No
10.	Redial	No
11.	Release	No
12.	Hold	No

These Feature keys are programmable. Refer the “DSS Keys Programming” topic for instructions on programming these keys. You cannot change the labels of these keys and therefore it is recommended that you avoid programming these keys.

A few of these Feature keys are equipped with an LED to indicate the status of the feature assigned to it. You may re-assign other features to these keys. We recommend you to assign those features that require LED indication to the Feature keys equipped with an LED.

Not all features require LED indication. For example, the feature Call Pick-Up does not require an LED. So when you assign this feature to a feature key, the LED of the key remains inactive, when Call Pick-Up is accessed.

Feature like Auto Redial requires an LED to show that it has been set or canceled. So, the LED of the Feature key to which the Auto Redial feature has been assigned will glow Blue, when Auto-Redial is set, and the LED is turned off when the feature is canceled.

Direct Station Selection (DSS) Keys

These are 16 programmable keys that can be assigned to Stations and Trunks and important or frequently accessed features of SARVAM UCS.

Refer the "DSS Keys Programming" topic for instructions on programming these keys.

DSS Key LEDs

Each DSS key is equipped with an LED which glows to indicate the status of the Trunk/Station or Feature assigned to it.

- **Status of Stations and Trunks:** The LED of DSS keys assigned to Stations/Trunks glow in three colors to indicate status of the call event on the Stations/Trunks and on the DKP.

  Thus, the status of the DKP user's own Station as well as that of the other Stations and the status of Trunk lines are indicated by the LED of the DSS keys assigned to those Stations and Trunks on the DKP.

  The following table shows the relationship between the color of the LED and various events:

LED Color	LED Mode		
	Continuously ON	Slow Blink	Fast Blink
Blue	The key assigned to the Station you are in speech with.	The key assigned to the Station you have kept on hold.	The key assigned to the Station you are calling or from which you are being called.
Red	The key assigned to the Station that is now busy with another Station/Trunk.	The key assigned to the Station which has put another Station/Trunk on hold.	The key assigned to the Station/Trunk that is called or being called by another.
Violet	You are talking on a Trunk (external call)	You have held a Trunk (external call)	You have an incoming call on the Trunk (external call)

  - **Blue** indicates the state of the station/trunk you access. For example, when you make a call to another Station 203, the LED of the DSS key assigned to Station 203 blinks Blue to indicate ringing at the Station. If you have successfully established speech with Station 203 the LED glows Blue continuously.

  - **Red** indicates the state of other Stations/Trunks. For example, if the LED of the DSS key assigned to Station 201 is glowing Red continuously, it means Station 201 is busy with another Station or Trunk.

  - **Violet** indicates the state of the trunk you are in speech with. For example, when you are in speech on an outgoing call on Trunk 1 the LED of the DSS Key assigned to Trunk 1 will be continuously ON. When you put the call on hold, the LED will blink slowly.

- **Status of Features:** The LED of a DSS key is activated when the feature assigned to this key is used.

  Not all features require LED indication. For example, the feature Call Pick-Up does not require an LED. So when you assign this feature to a DSS key, the LED of the key remains inactive, when Call Pick-Up is accessed.

Feature like Auto Redial requires an LED to show that it has been set or canceled. So, the LED of the DSS key to which the Auto Redial feature has been assigned will glow Red, when Auto-Redial is set, and the LED is turned off when the feature is canceled.

The LEDs of the Call Appearance (CA) Keys will function in the same manner as the DSS Key LEDs.

Dial Pad

The dial pad consists of 12 fixed keys for the digits 0, 1-9, and the characters * and #. The dial pad is used for dialing numbers of stations, external parties, and for dialing the programming and feature access codes.

Speaker Key

The speaker key sets the phone in 'Speaker mode' for hands-free operation. The Speaker key is programmable, you can assign any other feature/function on this key.

Speaker Key LED

The Speaker Key on EON48 is equipped with a single color LED which glows Blue when pressed for the speaker mode and is turned off, when you exit the speaker mode.

Volume Keys

- **"+" (plus)**: This is the increase key, to raise the volume of speech while talking and to increase the Ringer volume, when the phone is ringing.

- **"-" (minus)**: This is the decrease key, to lower the volume of speech while talking and to decrease the Ringer volume when the phone is ringing.

Headset Connectivity

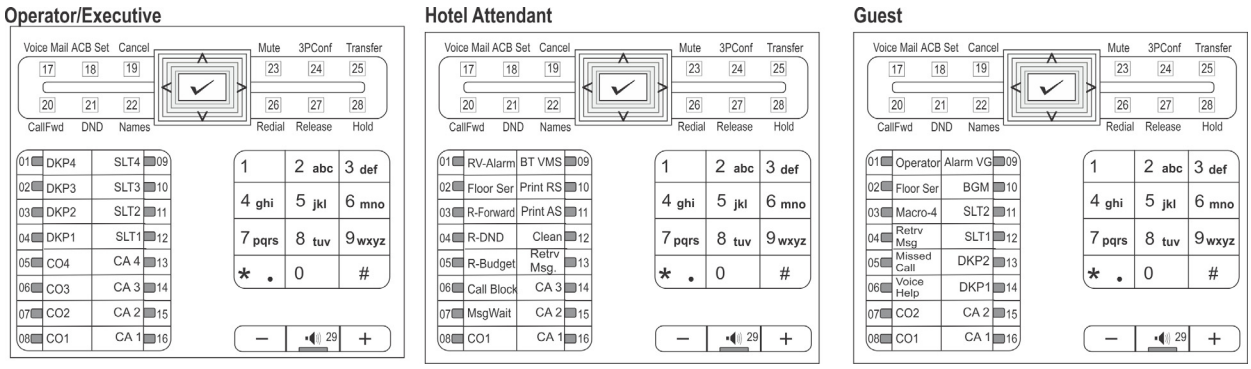
The EON48 provides two Headset interfaces: A 2.5mm Audio Jack and an RJ9 connector at the bottom of the phone body.

So you can use any stereo headset of standard make with a 2.5 mm single connector or a headset with an RJ9 connector.

You can also assign Headset key function to any of the DSS keys. Refer the topic “DSS Keys Programming” for instructions.

Key Maps

As EON48 may be the extension of the Operator/s and Executives in an enterprise, and the extension of the Front Desk Attendant and Guest in hotels, to meet the varied requirements of each user group, SARVAM UCS provides Key Maps for Operator, Executive1, Executive2, Executive3, Hotel Attendant and Hotel Guest.



The key maps of the Operator and Executive 1, 2, 3 are the same.

These key maps can be customized to match the exact requirement of individual users. Refer the topic “DSS Keys Programming” for instructions on customizing the Key Maps.

Phone Menu

You can access the following System and phone features from the Menu of EON48:

Menu option	Description
Call Logs	To view call history of internal and external Missed, Answered and Dialed calls. You can also edit numbers in the call logs and store them in the Personal Directory.
Call Forward	To set and cancel Call Forward-Busy, Call-Forward No Reply, Call-Forward-Unconditional, and Follow Me.
Dynamic Lock	To change the Toll Control level of the phone.
User Status	To set User Present or User Absent.
Keypad Lock	To lock the keypad of the phone.
Do Not Disturb	To set/cancel Do Not Disturb on the phone, that is, block incoming internal and external calls.
Call Cost Display	To view the cost of calls made from the phone.
Hotline	To set/cancel Hotline and Delayed Hotline.
Alarm	To set/cancel Personalized and Automated Alarms.
Change User Password	To change User Password.
One Touch Transfer	To configure the fixed destination number for One Touch Transfer.
Phone Settings	To customize settings of the phone such as Speech and Ringer Controls, LCD Display settings (Brightness and Contrast, Backlight ON/OFF), Headset Connectivity, Call Answering Mode (manual/auto answer).

Navigating the Phone Menu

To navigate the menu,

- Tap on Enter key when the phone is idle.
- Scroll by tapping the Up/Down Navigation Key to reach the desired Menu option.
- Tap on Enter key to select the desired Menu option.
- Scroll by tapping on the Up/Down Navigation Key to reach the desired sub-menu option.
- Tap on Enter key to select the desired sub-menu option.

To exit menu,

- Press Cancel key.

  Or

- Go ON-Hook.

Call Waiting Indication

For EON48 V2 and earlier

During an on-going call, if there is another incoming call, an indication will be provided to you for the waiting call.

- The call waiting indication depends on the **Call Waiting Tone** option you select — Beep Once, Beep until answered or Off.
- the LCD will display the Caller ID of the extension/trunk.

For EON48 V3 and later

During an on-going call, if there is another incoming call, an indication will be provided to you for the waiting call.

Ringer Mode — Ring if Idle, Ring Immediately or Ring After Delay

If you are in speech using the handset/headset and there is a waiting call for you,

- the ring will be played on the speaker.
- the LCD will display the Caller ID of the extension/trunk.

If you are in speech on a speaker and there is a waiting call for you,

- the Call Waiting Tone will be played as per the option you select — Beep Once, Beep until answered or Off.
- the LCD will display the Caller ID of the extension/trunk.

Ringer Mode — Silent

If you are in speech using the handset/headset/speaker and there is a waiting call for you,

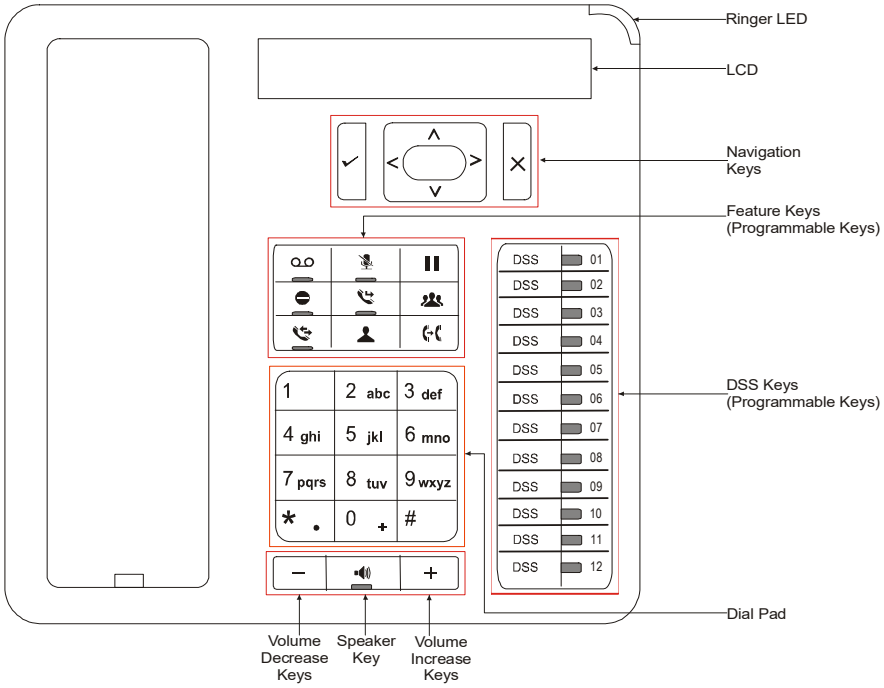
- the LCD will display the Caller ID of the extension/trunk.

Operating EON48

Please refer the EON48_310_SPARSH VP248_310 User Guide for instructions on operating the features of SARVAM UCS using EON.

EON310

2 lines and 24 characters LCD display, full duplex, fixed feature keys.



LCD Display

The LCD display of EON310 is backlit for a clear view of the text/characters displayed.

The LCD backlight can be turned on and off as well as adjusted for contrast and brightness from the "Phone Settings" of the DKP Phone Menu.

Ringer LED

The Ringer LED will glow in Blue to indicate incoming internal and external calls.

Navigation Keys

These are 6 keys. The functions of each are described briefly below.

- **Enter Key:** To enter the Menu; when the phone is in the idle state (without any incoming or outgoing call being made), if you tap the 'Enter' key, you will enter into the 'Menu'.

  Enter key is also used to make a selection from the Menu/sub-menu options or to complete an action.

- **Cancel Key:** To exit a menu; To abort a function/process; To delete a digit.
- **Up Key:** To scroll upwards while navigating the Menu/sub-menu.
- **Down Key:** To scroll downwards while navigating the Menu/sub-menu.
- **Forward Key:** To move forward when dialing a number.
- **Back Key:** To move backwards when dialing a number; to go back one level in the Menu.

Feature Keys

Sr.No.	Feature Icon	Assigned Feature	LED
1.		Voice Mail	Single Color - Blue
2.		Mute	Single Color - Blue
3.		Hold	No
4.		Do Not Disturb (DND)	Single Color - Blue
5.		Forward	Single Color - Blue
6.		Conference	No
7.		Logs	Single Color - Blue
8.		Contacts	No
9.		Transfer	No

These Feature keys are programmable. Refer the topic "DSS Keys Programming" for instructions on programming these keys. You cannot change the labels of these keys and therefore it is recommended that you avoid programming these keys.

A few of these Feature keys are equipped with an LED to indicate the status of the feature assigned to it. You may re-assign other features to these keys. We recommend you to assign those features that require LED indication to the Feature keys equipped with an LED.

Not all features require LED indication. For example, the feature Call Pick-Up does not require an LED. So when you assign this feature to a feature key, the LED of the key remains inactive, when Call Pick-Up is accessed. Feature like Auto Redial requires an LED to show that it has been set or canceled. So, the LED of the Feature key to which the Auto Redial feature has been assigned will glow Blue, when Auto-Redial is set, and the LED is turned off when the feature is canceled.

Direct Station Selection (DSS) Keys

These are 6 programmable keys that can be assigned to Stations and Trunks and important or frequently accessed features of SARVAM UCS.

Refer the topic "DSS Keys Programming" for instructions on programming these keys.

DSS Key LEDs

Each DSS key is equipped with an LED which glows to indicate the status of the Trunk/Station or Feature assigned to it.

- **Status of Stations and Trunks:** The LED of DSS keys assigned to Stations/Trunks glow in three colors to indicate status of the call event on the Stations/Trunks and on the DKP.

  Thus, the status of the DKP user's own Station as well as that of the other Stations and the status of Trunk lines are indicated by the LED of the DSS keys assigned to those Stations and Trunks on the DKP.

  The following table shows the relationship between the color of the LED and various events:

LED Color	LED Mode		
	Continuously ON	Slow Blink	Fast Blink
Blue	The key assigned to the Station you are in speech with.	The key assigned to the Station you have kept on hold.	The key assigned to the Station you are calling or from which you are being called.
Red	The key assigned to the Station that is now busy with another Station/Trunk.	The key assigned to the Station which has put another Station/Trunk on hold.	The key assigned to the Station/Trunk that is called or being called by another.
Violet	You are talking on a Trunk (external call)	You have held a Trunk (external call)	You have an incoming call on the Trunk (external call)

  - **Blue** indicates the state of the station/trunk you access. For example, when you make a call to another Station 203, the LED of the DSS key assigned to Station 203 blinks Blue to indicate ringing at the Station. If you have successfully established speech with Station 203 the LED glows Blue continuously.

  - **Red** indicates the state of other Stations/Trunks. For example, if the LED of the DSS key assigned to Station 201 is glowing Red continuously, it means Station 201 is busy with another Station or Trunk.

- **Violet** indicates the state of the trunk you are in speech with. For example, when you are in speech on an outgoing call on Trunk 1 the LED of the DSS Key assigned to Trunk 1 will be continuously ON. When you put the call on hold, the LED will blink slowly.

- **Status of Features:** The LED of a DSS key is activated when the feature assigned to this key is used.

  Not all features require LED indication. For example, the feature Call Pick-Up does not require an LED. So when you assign this feature to a DSS key, the LED of the key remains inactive, when Call Pick-Up is accessed.
  Feature like Auto Redial requires an LED to show that it has been set or canceled. So, the LED of the DSS key to which the Auto Redial feature has been assigned will glow Red, when Auto-Redial is set, and the LED is turned off when the feature is canceled.

The LEDs of the Call Appearance (CA) Keys will function in the same manner as the DSS Key LEDs.

Dial Pad

The dial pad consists of 12 fixed keys for the digits 0, 1-9, and the characters * and #. The dial pad is used for dialing numbers of stations, external parties, and for dialing the programming and feature access codes.

Speaker Key

The speaker key sets the phone in 'Speaker mode' for hands-free operation.

Speaker Key LED

The Speaker Key on EON310 is equipped with a single color LED which glows Blue when pressed for the speaker mode and is turned off, when you exit the speaker mode.

Volume Keys

- **"+" (plus)**: This is the increase key, to raise the volume of speech while talking and to increase the Ringer volume, when the phone is ringing.

- **"-" (minus)**: This is the decrease key, to lower the volume of speech while talking and to decrease the Ringer volume when the phone is ringing.

Headset Connectivity

The EON310 provides two Headset interfaces: A 3.5mm Audio Jack and an RJ12 connector at the bottom of the phone body.

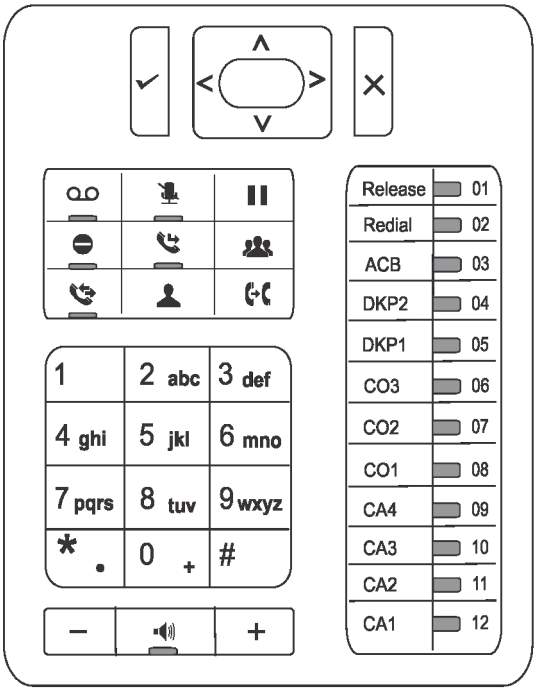
So you can use any headset of standard make with a 3.5 mm single connector or a stereo headset with an RJ9 connector.

You can also assign Headset key function to any of the DSS keys. Refer the topic “DSS Keys Programming” for instructions.

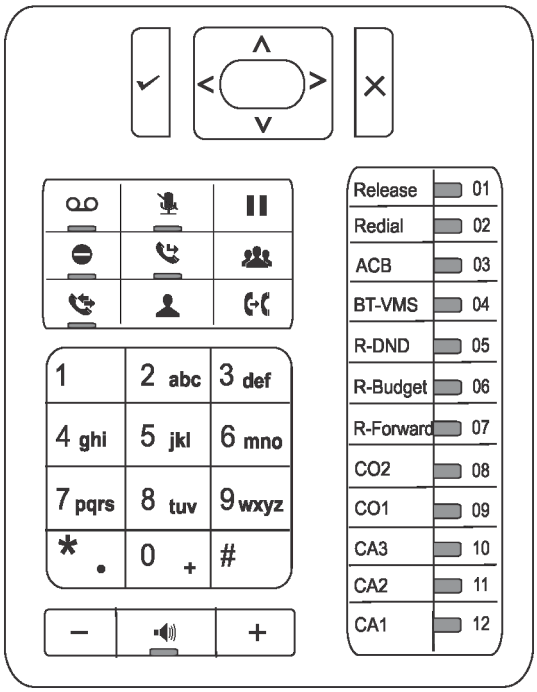
Key Maps

As EON310 may be the extension of the Operator/s and Executives in an enterprise, and the extension of the Front Desk Attendant and Guest in hotels, to meet the varied requirements of each user group, SARVAM UCS provides Key Maps for Operator, Executive1, Executive2, Executive3, Hotel Attendant, and Hotel Guest.

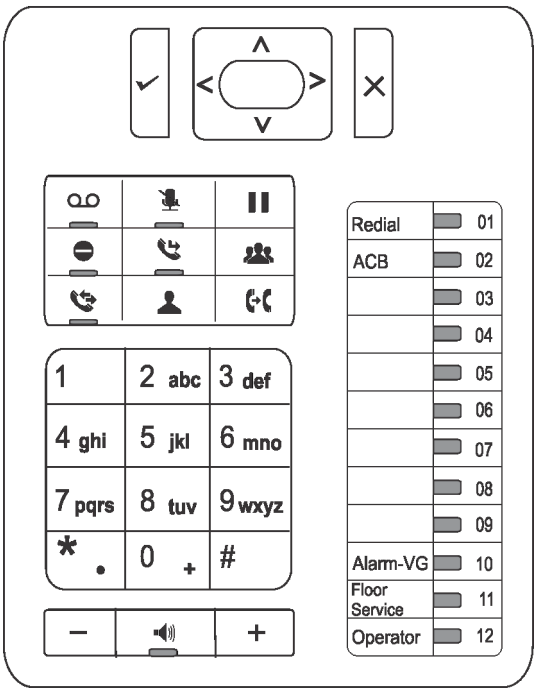
Operator\Executive



Hotel Attendant



Guest



The key maps of Executive 1, 2, 3 are the same.

These key maps can be customized to match the exact requirement of individual users. Refer the topic "DSS Keys Programming" for instructions on customizing the Key Maps.

Phone Menu

You can access the following System and phone features from the Menu of EON310:

Menu option	Description
Call Logs	To view call history of internal and external Missed, Answered and Dialed calls. You can also edit numbers in the call logs and store them in the Personal Directory.
Contacts	To add, edit, delete names and numbers of contacts in the Global Directory Part 1.
Call Forward	To set and cancel Call Forward-Busy, Call-Forward No Reply, Call-Forward-Unconditional, and Follow Me.
Dynamic Lock	To change the Toll Control level of the phone.
User Status	To set User Present or User Absent.
Keypad Lock	To lock the keypad of the phone.
Do Not Disturb	To set/cancel Do Not Disturb on the phone, that is, block incoming internal and external calls.
Call Cost Display	To view the cost of calls made from the phone.
Hotline	To set/cancel Hotline and Delayed Hotline.
Alarm	To set/cancel Personalized and Automated Alarms.
Change User Password	To change User Password.
One Touch Transfer	To configure the fixed destination number for One Touch Transfer.
Phone Settings	To customize settings of the phone such as Speech and Ringer Controls, LCD Display settings (Brightness and Contrast, Backlight ON/OFF), Headset Connectivity, Call Answering Mode (manual/auto answer).

Navigating the Phone Menu

To navigate the menu,

- Tap on Enter key when the phone is idle.
- Scroll by tapping the Up/Down Navigation Key to reach the desired Menu option.
- Tap on Enter key to select the desired Menu option.
- Scroll by tapping on the Up/Down Navigation Key to reach the desired sub-menu option.
- Tap on Enter key to select the desired sub-menu option.

To exit menu,

- Press Cancel key.
  Or
- Go ON-Hook.

Call Waiting Indication

During an on-going call, if there is another incoming call, an indication will be provided to you for the waiting call.

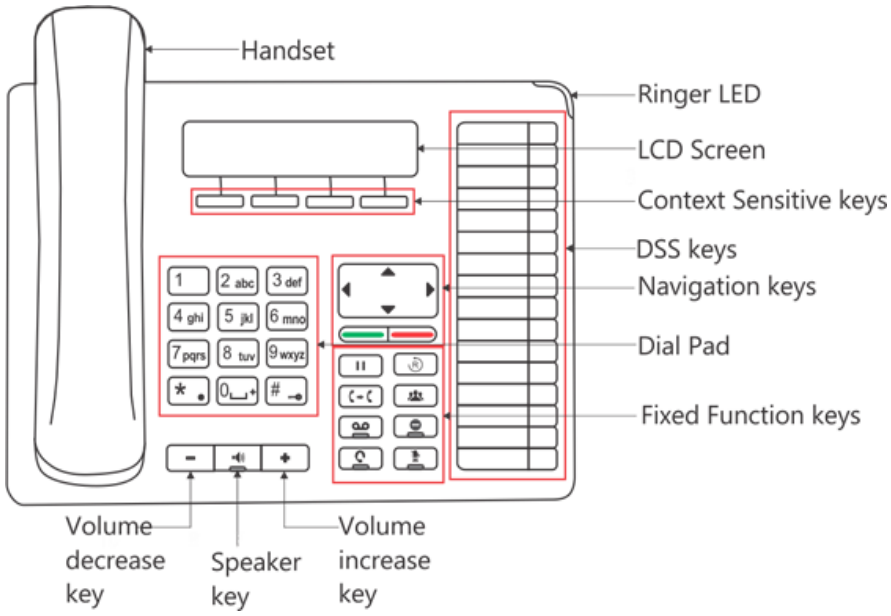
The call waiting indication depends on the Call Waiting Tone option you select — Beep Once, Beep until answered or Off.

Operating EON310

Please refer the EON48_310_SPARSH VP248_310 User Guide for instructions on operating the features of SARVAM UCS using EON.

EON510

240 x 64 Pixel Graphical LCD display, full duplex, fixed feature keys.



LCD Display

The LCD display of the phone is backlit. The LCD backlight can be turned on and off as well as adjusted for contrast and brightness from the Phone Menu.

Ringer LED

The Ringer LED indicates incoming internal and external calls. The LED Cadence will match with the Ring Cadence of the incoming internal/external call.

The Ringer LED cadence changes according to the type of call, as described in the table below.

Type of Call	Cadence
Internal Call	Short, very slow (750ms ON, 2250ms OFF)
External Call	Double (400ms ON, 200ms OFF, 400ms ON, 2000ms OFF)
Alarm	Long, fast (1500ms ON, 500ms OFF)
Auto Redial Call	Long, very slow (2000ms ON, 4000ms OFF)
Auto Call Back Call	Short, slow (750ms ON, 2250ms OFF)
Priority	Triple (400ms ON, 200ms OFF, 400ms ON, 200ms OFF, 400ms ON, 2000ms OFF)
Programming mode	Continuous

Navigation Keys

The function of each Navigation key is described below.

- **Up Key:** To scroll upwards while navigating the Menu/sub-menu.
- **Down Key:** To scroll downwards while navigating the Menu/sub-menu.

- **Forward Key:** To move forward when dialing a number or scroll to view the Context Sensitive Key options.
- **Back Key:** To move backwards when dialing a number, to go back one level in the Menu or scroll backwards to view the Context Key options.
- **Menu or Select / OK Key** : To enter the Menu; when the phone is in the idle state (without any incoming or outgoing call being made).

  Menu Key functions as the **Select / OK** Key to make a selection from the Menu/sub-menu options or to complete an action. When there is an incoming call it functions as the **Answer** Key.
- **Cancel Key** : To Cancel all features set by you or exit the Menu/sub-menu.

Feature Keys

EON510 supports following feature keys. Default features assigned to these keys are as follows.

Feature icon	Assigned Feature	LED	Programmable
	Hold	No	Yes
	Redial	No	Yes
	Transfer	No	Yes
	Conference	No	Yes
	Voicemail	Single Color - Blue	Yes
	Do Not Disturb	Single Color - Blue	Yes
	Headset	Single Color - Blue	No
	Mute	Single Color - Blue	No

All Feature keys are programmable except Headset and Mute key. Refer the topic “DSS Keys Programming” for instructions on programming these keys. You cannot change the labels of these keys and therefore it is recommended that you avoid programming these keys.

A few of these Feature keys are equipped with an LED to indicate the status of the feature assigned to it. You may re-assign other features to these keys. We recommend you to assign those features that require LED indication to the Feature keys equipped with an LED.

Not all features require LED indication. For example, the feature Call Pick-Up does not require an LED. So when you assign this feature to a feature key, the LED of the key remains inactive, when Call Pick-Up is accessed.

Feature like Auto Redial requires an LED to show that it has been set or canceled. So, the LED of the Feature key to which the Auto Redial feature has been assigned will glow Blue, when Auto-Redial is set, and the LED is turned off when the feature is canceled.

Direct Station Selection (DSS) Keys

There are 16 programmable keys that can be assigned to Extensions and Trunks and important or frequently accessed features of SARVAM UCS.

Refer the topic “DSS Keys Programming” for instructions on programming these keys.

DSS Key LEDs

Each DSS Key is equipped with an LED which glows to indicate the status of the Trunk/Extension or Feature assigned to it.

- **Status of Extensions and Trunks:** The LED of DSS keys assigned to Extensions/Trunks glow in three colors to indicate status of the call event on the Extensions/Trunks and on the phone.

  Thus, the status of the phone user's own Extension as well as that of the other Extensions and the status of Trunk lines are indicated by the LED of the DSS keys assigned to those Extensions and Trunks on the phone.

  The following table shows the relationship between the color of the LED and various events:

LED Color	**LED Mode**		
	Continuously ON	Slow Blink	Fast Blink
Blue	The key assigned to the Extension you are in speech with.	The key assigned to the Extension you have kept on hold.	The key assigned to the Extension you are calling or from which you are being called.
Red	The key assigned to the Extension that is now busy with another Extension/Trunk.	The key assigned to the Extension which has put another Extension/Trunk on hold.	The key assigned to the Extension/Trunk that is called or being called by another.
Violet	You are talking on a Trunk (external call)	You have held a Trunk (external call)	You have an incoming call on the Trunk (external call)

  - **Blue** indicates the state of the extension/trunk you access. For example, when you make a call to another Extension 203, the LED of the DSS key assigned to Extension 203 blinks Blue to indicate ringing at the Extension. If you have successfully established speech with Extension 203 the LED glows Blue continuously.

  - **Red** indicates the state of other Extension/Trunks. For example, if the LED of the DSS key assigned to Extension 201 is glowing Red continuously, it means Extension 201 is busy with another Extension or Trunk.

  - **Violet** indicates the state of the trunk you are in speech with. For example, when you are in speech on an outgoing call on Trunk 1 the LED of the DSS Key assigned to Trunk 1 will be continuously ON. When you put the call on hold, the LED will blink slowly.

- **Status of Features:** The LED of a DSS key is activated when the feature assigned to this key is used.

Not all features require LED indication. For example, the feature Call Pick-Up does not require an LED. So when you assign this feature to a DSS key, the LED of the key remains inactive, when Call Pick-Up is accessed.

Feature like Auto Redial requires an LED to show that it has been set or canceled. So, the LED of the DSS key to which the Auto Redial feature has been assigned will glow Red, when Auto-Redial is set, and the LED is turned off when the feature is canceled.

The LEDs of the Call Appearance (CA) Keys will function in the same manner as the DSS Key LEDs.

Dial Pad

The dial pad consists of 12 fixed keys for the digits 0, 1-9, and the characters * , #, Lock. The dial pad is used for dialing numbers of extensions, external parties, and for dialing the programming and feature access codes.

Speaker Key

The speaker key sets the phone in 'Speaker mode' for hands-free operation.

Speaker Key LED

The Speaker Key on the phone is equipped with a single color LED which glows Blue when pressed for the speaker mode and is turned off, when you exit the speaker mode.

Volume Keys

- **"+" (plus)**: This is the increase key, to raise the volume of speech while talking and to increase the Ringer volume, when the phone is ringing.

- **"-" (minus)**: This is the decrease key, to lower the volume of speech while talking and to decrease the Ringer volume when the phone is ringing.

Headset Connectivity

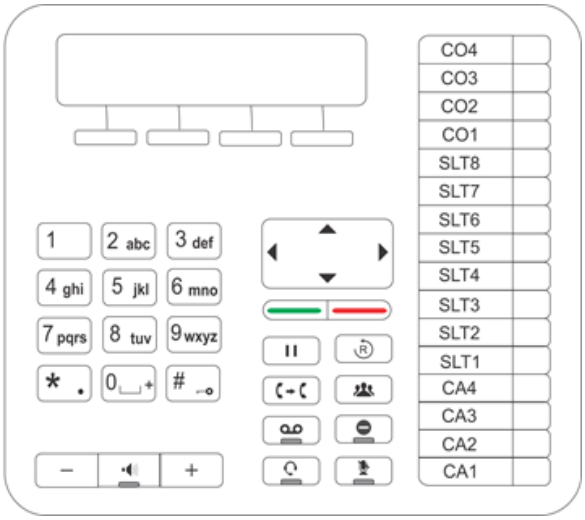
The phone provides two Headset interfaces: A 3.5mm Audio Jack and an RJ9 connector at the bottom of the phone body.

So, you can use any stereo headset of standard make with a 3.5 mm single connector or a stereo headset with an RJ9 connector.

To use the Headset, a Headset Key is assigned on the phone. The Headset Key on the phone is equipped with a single color LED which glows Blue when pressed for the headset mode and is turned off, when you exit the headset mode.

Key Maps

As EON510 may be the extension of the Operator/s and Executives in an enterprise, to meet the varied requirements of each user group, SARVAM UCS provides Key Maps for Operator and Executives. These key maps can be customized to match the exact requirement of individual users. All the key maps of the EON510 are the same.



Refer the topic “DSS Keys Programming” for instructions on programming these keys.

Phone Menu

You can access the following System and phone features from the Menu of the phone:

Menu option	Description
Call Logs	To view call history of internal and external Missed, Answered and Dialed calls. You can also edit numbers in the call logs and store them in the Personal Directory.
Call Forward	To set and cancel Call Forward-Busy, Call-Forward No Reply, Call-Forward-Unconditional, and Follow Me.
Dynamic Lock	To change the Toll Control level of the phone.
User Status	To set User Present or User Absent.
Keypad Lock	To lock the keypad of the phone.
Do Not Disturb	To set/cancel Do Not Disturb on the phone, that is, block incoming internal and external calls.
Call Cost Display	To view the cost of calls made from the phone.
Hotline	To set/cancel Hotline and Delayed Hotline.
Alarm	To set/cancel Personalized and Automated Alarms.
Change User Password	To change User Password.
One Touch Transfer	To configure the fixed destination number for One Touch Transfer.

Menu option	Description
Phone Settings	To customize settings of the phone such as Speech and Ringer Controls, LCD Display settings (Brightness and Contrast, Backlight ON/OFF), Headset Connectivity, Call Answering Mode (manual/auto answer).

Navigating the Phone Menu

To navigate the menu,

- Press the Menu Key when the phone is idle.
- Scroll by pressing the Up/Down Navigation Key to reach the desired Menu option.
- Press the Select or OK Key to select the desired Menu option.
- Scroll by pressing the Up/Down Navigation Key to reach the desired sub-menu option.
- Press the Select or OK Key to select the desired sub-menu option.

To exit menu,

- Press Cancel Key.
  or
  Go ON-Hook.

Call Waiting Indication

During an on-going call, if there is another incoming call, an indication will be provided to you for the waiting call.

The call waiting indication depends on the Call Waiting Tone option you select — Beep Once, Beep until answered or Off.

Operating EON510

Please refer the *EON510_SPARSH VP510 User Guide* for instructions on operating the features of SARVAM UCS using EON.

EONSOFT

The EONSOFT is a PC-based Digital Key Phone. Based on a graphic user Interface (GUI), the EONSOFT offers all the features of EON48, making it a substitute for the Digital Key Phone. Its integration with the SARVAM UCS obviates the need for a separate telephone instrument.

The EONSOFT can be installed on any personal computer with Windows or NT operating system.

Two PC-based DSS64 Consoles are available to be used with the EONSOFT. You can use either one or both DSS64 Consoles.

The EONSOFT occupies only a single port, even when both PC-based DSS64 Consoles are used. Thus it supports all the features of the Digital Key Phone and the DSS64 Console on a single window.

DKP Port

The DKP port connects EONSOFT to the DKP port of SARVAM UCS's DKP Card.

Handset Port

The Handset port connects the Receiver of the phone, to be used for speech.

The EONSOFT has the provision for attaching a Handset. A handset with spring cord is supplied by Matrix and is to be connected to the handset jack (RJ12) on the Dongle.

Headset connectivity

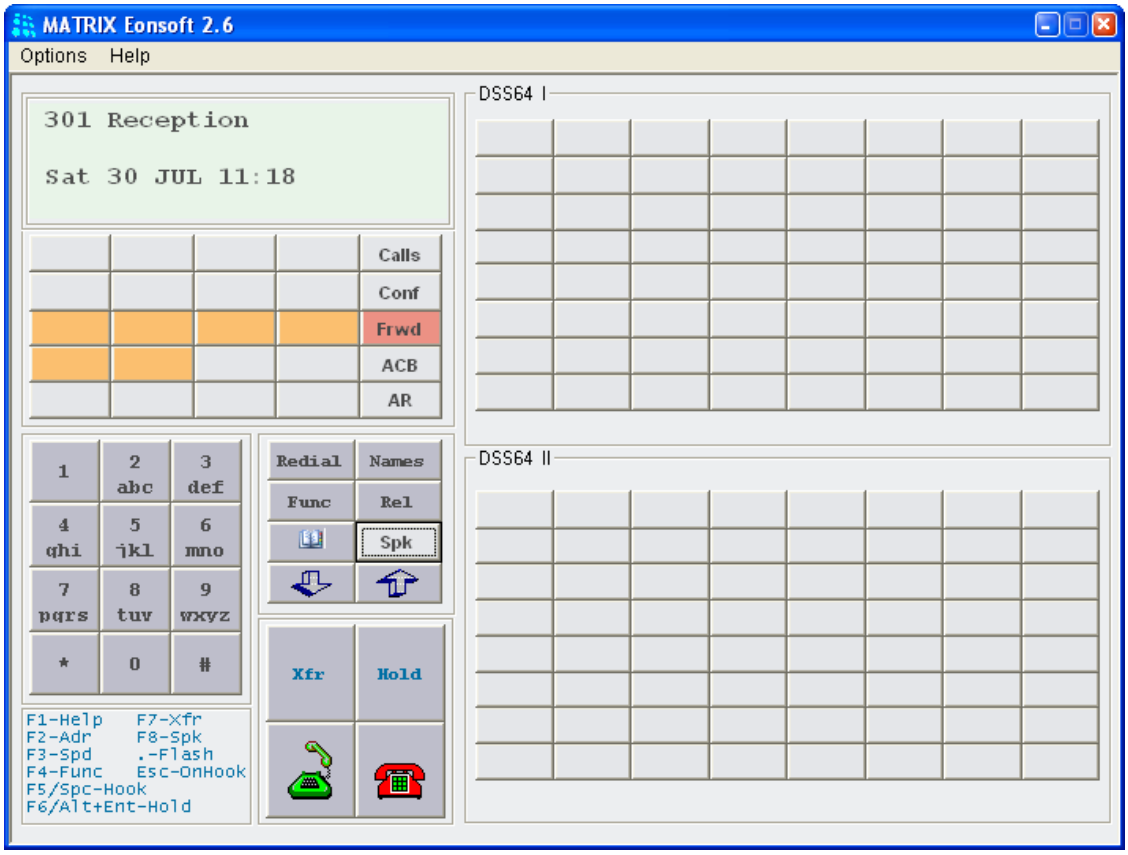
EONSOFT supports headset connectivity, providing a MIC and a Speaker interface. Any stereo Headset of standard make, with dual connectors can be connected to the MIC and the Speaker on the Dongle.

COM Port

The COM port connects EONSOFT to a PC (COM Port).

After EONSOFT has been successfully installed on a PC and the DKP parameters have been configured, each time you open EONSOFT, the display and keypad of the phone will appear on your PC screen.

The illustration below shows EONSOFT with two DSS64 Consoles attached to it.



Phone Display

The EONSOFT has a 2-line and 24-character display. In the ON-Hook or idle condition, the first line displays the Station Number and the Station name. The second line displays the Day, Date and Time.

When there is an incoming call, the calling party's number is displayed on Line 2 of the LCD[292].

The LCD messages for various call events (dial, transfer, forward, hold, etc.), for prompts, alerts, confirmation, errors, text messages, are displayed.

Direct Station Selection (DSS) Keys

These are a set of 25 keys programmable keys, allowing you to assign Stations, Trunks or Features to them, so that you can access any of these by pressing the respective key.

Refer the topic “DSS Keys Programming” for instructions on assigning stations, trunks, features to keys.

DSS Key LEDs

Each DSS key is equipped with an LED which changes the color to indicate the status of the Trunk/Station or Feature assigned to it.

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292. Only if the Station, to which EONSOFT is connected, has been allowed CLIP facility in its Class of Service.

- **Status of Stations and Trunks:** The LED of DSS keys assigned to Stations/Trunks may change in three different colors to indicate status of the call event on the Stations/Trunks on EONSOFT.

  Thus the status of the Stations and the Trunk lines are indicated by the LED of the DSS keys assigned to those Stations and Trunks on EONSOFT.

  The following table shows the relationship between the color of the LED and various events:

LED Color	LED Mode		
	Continuously ON	Slow Blink	Fast Blink
Green	The key assigned to the Station you are in speech with.	The key assigned to the Station you have kept on hold.	The key assigned to the Station you are calling or from which you are being called.
Red	The key assigned to the Station that is now busy with another Station/ Trunk.	The key assigned to the Station which has put another Station/Trunk on hold.	The key assigned to the Station/ Trunk that is called or being called by another.
Orange	You are talking on a Trunk (external call)	You have held a Trunk (external call)	You have an incoming call on the Trunk (external call)

  - **Green** indicates the state of the station/trunk you access. For example, when you make a call to another Station 203, the LED of the DSS key assigned to Station 203 blinks Green to indicate ringing at the Station. If you have successfully established speech with Station 203 the LED is Green continuously.

  - **Red** indicates the state of other Stations/Trunks. For example, if the LED of the DSS key assigned to Station 201 is Red continuously, it means Station 201 is busy with another Station or Trunk.

  - **Orange** indicates the state of the trunk you are in speech with. For example, when you are in speech on an outgoing call on Trunk 1 the LED of the DSS Key assigned to Trunk 1 will be continuously ON. When you put the call on hold, the LED will blink slowly.

  The LEDs of DSS Keys that are designated as Call Appearance (CA) Keys will function as follows:

LED Color	LED Mode		
	Continuously ON	Slow Blink	Fast Blink
Green	When you are in speech with a Station (internal call)	When you have put a Station on hold (internal call)	When any Station is calling (internal call)
Orange	When you are in speech with Trunk (external call)	When you have put a Trunk on hold (external call)	When any Trunk is calling (external call)

- **Status of Features:** The LED of a DSS key is activated when the feature assigned to this key is used.

  The LED of DSS keys assigned to Stations/Trunks turn Red to indicate status of the call event on the Stations/Trunks.

- *Not all features require LED indication. Hence the LED on a DSS Key is activated only if the feature assigned to that key requires LED.*
- *For example, Call Pick-Up; this feature does not require an LED. So when a DSS key is assigned to this feature, the LED of the key remains inactive, when Call Pick-Up is accessed.*
- *A feature like Auto Redial requires an LED to show that it has been set or canceled. So, the LED of the DSS key to which the Auto Redial feature has been assigned will turn Red, when Auto-Redial is set, and the LED is turned off when the feature is canceled.*
- *Thus the LEDs of the DSS keys function only if the LED is relevant for the feature/ function assigned to the keys, otherwise it remains inactive.*
- *The LEDs of DSS keys to which features like Raid, Interrupt Request, Barge-In, Last Caller Recall are assigned, will not change color.*

Dial Pad

The dial pad consists of 12 keys (non-programmable), which include the digit keys for 0, 1-9, and character keys for * and #.

Function Keys

These are non-programmable keys on the keypad of EONSOFT which have fixed functions.

- **Redial:** This key is used for redialing the last external number.
- **Func Key:** This key is used for accessing the Phone menu.
- **Names:** This key is used to dial by Name.
- **:** This key is used for accessing the Address Book. The EONSOFT provides the facility of an Address Book that is integrated with the Standard Windows Address Book, for storing the numbers and addresses of contacts. So, when a call is to be made, you can select and dial the desired number from the directory.
- **Rel:** This key is used to disconnect an on-going call.
- **Xfr:** This key is used for transferring calls to another destination
- **Hold:** This key is used for putting the caller on hold. This key is also used to make a selection in the Phone Menu.
- **:** This key is used for going OFF-Hook. It simulates lifting of the handset, pressing of the speaker key to make or receive calls.
- **:** This key is used for going ON-Hook. It simulates replacing of the handset, pressing of the speaker key to disconnect.

Volume Control Keys

The ↑ and ↓ keys are used for increasing or decreasing the phone volume. Volume can be increased by clicking ↑ and decreased by clicking ↓.

Navigation Keys

The following keys are used for navigating the phone menu:

- **'Func' key:** This key is used for entering the Phone menu and to go back one level in the menu.
- **Up and Down keys:** The ↑ and ↓ keys function as the Up and Down keys to scroll the Menu and sub-menu options. You can scroll up down the menu by clicking ↑ and scroll up the menu by clicking ↓.
- **'Hold' key:** This key is used to make a selection in the Menu.

Speaker key

The 'Spk' key sets the phone in 'Speaker mode' for hands-free operation. The Speaker key is programmable; you can assign any other feature/function to this key.

Shortcut keys

You can use the Keyboard of the PC to operate EONSOFT, with the help of "shortcut keys'. The following table describes the functions performed when shortcut keys on the keyboard are pressed:

Short Cut Key Label	Description
F1	Help
F2	Windows Address Book
F3	Spd
F4	Func
F5	Space Bar - Hook
F6	Alt+Enter - Hold
F7	Xfr
F8	Spk
Esc	ON-Hook
. (dot/period)	Flash
↑ (Up Arrow Key)	Volume key. To increase volume of ringer and speech
↓ (Down Arrow Key)	Volume key. To decrease volume of ringer and speech
Ctrl+C	To open COM Port Settings Dialog Box
Ctrl+T	To open Trunk Access Code Dialog Box
Tab → (Tab Forward)	Forward shifting of selection
Tab ← (Tab Backward)	Backward shifting of selection
Shift + Enter ↵	To press the button with selection

Short Cut Key Label	Description
Num Pad	Dial out a number

Key Maps

EONSOFT can function as a Station for the Operator, Executive, and Hotel Attendant, also Guest (though unlikely to be used by guests).

Phone Menu

The Phone menu is the same as EON48.

Navigating the Phone Menu

To navigate the menu,

- Click 'Func' key when the phone is idle.
- Scroll by clicking the Up/Down Key to reach the desired Menu option.
- Click 'Hold' key to select the desired Menu option.
- Scroll by clicking the with Up/Down Key to reach the desired sub-menu option.
- Click 'Hold' key to select the desired sub-menu option.

To exit menu,

- Press the 'Func' key repeatedly to go back one level in the menu, till you reach 'Menu'.

  Or

- Press Cancel key, if programmed.

If you want to use the Keyboard, press the Shortcut key for the desired function.

Tool Tips

You can assign labels and tool tips for the DSS keys, which are displayed to the user on mouse over. You can assign the function of each key as Tool Tip, to help user in intuitive operation of EONSOFT.

To assign a tool tip,

- right click the desired DSS key.
- The dialog box will show the Key Number.
- Enter the label for the key.
- Enter the Tool Tip for the key.
- Click 'OK'.
- Repeat the same steps to configure Tool Tip on another key.

Each time you move the mouse on the key, the Tool Tip you programmed for this key will be displayed.

Call Indication

Incoming Calls are indicated by:

- Popping up of the EONSOFT window, when the window is minimized.
- Flashing of the Title Bar of the EONSOFT window, when the window is maximized.

In order for the EONSOFT window to pop up, you must have enabled the 'PopUp When Ring' option. When this option is enabled and the EONSOFT window is minimized a new incoming call causes the window to pop up to its full size notifying the user about the new call. When this option is enabled and the EONSOFT window is maximized, a new incoming call is indicated by the flashing of the Title bar of the window.

When the 'PopUp When Ring' option is disabled and the EONSOFT window is minimized, a new incoming call is indicated by the flashing of the EONSOFT Title at the bottom bar of the PC screen.

By default, the 'PopUp When Ring' option is enabled.

Second Call Indication

When you are in an on-going conversation and there is a second incoming call, EONSOFT will indicate the second incoming call. The second call indication depends on the type of Ringer Mode selected by you.

The tables below describes the second call indications for each Ringer Mode.

Ringer Mode - Ring if Idle

Second call indication, when the first call is on the Speaker	Second call indication, when the first call is on the Handset	Second call indication, when the first call is on the Headset
Call Waiting Tone is played and the LCD displays the Caller ID of the station/trunk.	Call Waiting Tone is played and the LCD displays the Caller ID of the station/trunk.	NA

Ringer Mode - Ring Immediately

Second call indication, when the first call is on the Speaker	Second call indication, when the first call is on the Handset	Second call indication, when the first call is on the Headset
Call Waiting Tone is played and the LCD displays the Caller ID of the station/trunk.	Ring on the speaker and the LCD displays the Caller ID of the station/ trunk.	NA

Ringer Mode - Ring After Delay

Second call indication, when the first call is on the Speaker	Second call indication, when the first call is on the Handset	Second call indication, when the first call is on the Headset
The LCD displays the Caller ID of the station/trunk and the Call Waiting Tone is played after the expiry of the Ring Delay Timer.	The LCD displays the Caller ID of the station/trunk and there is a Ring on the speaker after the expiry of the Ring Delay Timer.	NA

Ringer Mode - Silent

The LCD will display the Caller ID of the station/trunk..

Operating EONSOFT

EONSOFT can be operated using the keyboard and the mouse.

Making calls

To make calls,

- Click the desired digits on the dial pad.
- The number will be dialed out.
- Talk.
- Click ON-Hook key to disconnect.

To make calls from Headset mode,

- Click OFF-Hook key.
- Click the desired digits on the dial pad.
- The number will be dialed out.

To make calls from Speaker mode,

- Click 'Spk' key.
- Click the desired digits on the dial pad.
- The number will be dialed out.

- *A headset must be connected and 'Headset Connectivity' must be enabled in the 'DKP Parameters'. Refer "DSS Keys Programming" for instructions.*

- *If you are using the keyboard instead of the mouse, press the appropriate Shortcut Keys listed above and use the Number pad on the keyboard to dial digits.*

Receiving calls

- When window pops up to indicate a call,
- Click 'Spk' key or 'OFF-Hook' Key.
- Talk.
- Click 'ON-Hook' key to disconnect.

Direct Dialing-In (DDI)

What's this?

DDI is the service feature provided to the customers by the ISDN exchanges / ITSPs over digital trunks / SIP Trunks. DDIs are Direct Dial-In numbers. These are additional numbers that can be associated with the main number assigned to the ISDN line/SIP Trunk. They are normally used with the system to provide separate numbers to extension users.

When there is an incoming call on the a specific DDI number it is routed through to the required extension without operator intervention.

Before we understand how DDI routing works, we must understand the following terms:

- **Pilot Number/ISDN Number:** It is a single number assigned by the service provider to a single ISDN/IP trunk. This is the combination of MSN Number and the first DDI Number. The Pilot Number is of maximum 16 digits. This is also known as ISDN Installation Number/ISDN Number. The MSN is a fixed series and is given by the service provider whereas the DDI Numbers can be selected by the user and these series vary. For example, if the Pilot Number is 2630555, the MSN is 2630 and the DDI Number is 555. However the number of digits to be used as DDI Numbers is informed by the service provider as they may differ for each service provider.

- **Multiple Subscriber Number (MSN):** In India, the MSN are the first four digits of the Pilot Number, which remain fixed and are provided by the service provider. The number of fixed digits may vary depending on the service provider.

- **DDI Numbers:** These are additional numbers, which can be selected by you but are allotted by the service provider. These are sequential numbers. In the DDI number only the last 3 or 4 digits vary. These are known a DDI numbers. However the number of DDI digits which vary are provided by the service provider. These numbers are normally used with a PBX and are assigned to the extension user.

How it works

Let us understand this with the help of an example:

- A BRI line has been connected to the PBX.

- 50 DDI numbers have been given by the service provider.

- Incoming calls on the DDI numbers can be routed to different landing destinations as extension users, department groups or other trunks within the organization.

- To assign DDI numbers to the extension users and to route calls as per the DDI numbers, you need to configure the following:
  - Incoming Reference Table
  - Outgoing Reference Table
  - DDI Routing Table

Handling Incoming Calls

Pre-requisites to route incoming calls as per the DDI logic,

- Assign an Incoming (IC) Reference ID to the BRI port. This Incoming (IC) Reference ID acts as a link between the port settings and the Incoming Reference Table.

- In the Incoming Reference Table against the same Incoming (IC) Reference ID, configure the Start Channel No., Total Channel Count, DDI Routing Ref. ID, Route on First Destination, Ring Timer (sec), When No Reply, When Busy, Trunk Features Template as per your requirement.

- To route and place the call on the final destination, the system refers to the DDI Routing Ref. ID configured in the Incoming Reference Table. As per this ID the system checks the DDI Routing Table. The DDI Routing Ref. ID acts as a link between the Incoming Reference Table and the DDI Routing Table. The system places the call on the final destination as configured against the DDI Routing Reference ID in the DDI Routing Table.

When there is an incoming call on the BRI port,

- As per the time zone, the system checks the Incoming (IC) Reference ID assigned to the BRI port

- If the Incoming (IC) Reference ID has been configured, the system then hunts for the same ID in the Incoming Reference Table.

  If no channel match is found in the Incoming Reference Table, the call is routed as per the Trunk Feature Template assigned to the BRI port.

  If a match is found, the system verifies the channels on which the routing logic is to be applied. These are defined in the parameters Start Channel Number and the Total Channel Count. Hence the system also matches the channel number on which the incoming call landed and the channel configured.

- For the matching entry, the system checks the corresponding DDI Routing Reference ID. The DDI Routing Reference ID acts as a link between the Incoming Reference Table and the DDI Routing Table.

- The system hunts for the same DDI Routing Reference ID in the DDI Routing Table. In the DDI Routing Table, a range of DDI numbers and their respective landing destinations are configured. You can configure as many as 224 DDI Reference ID’s with different DDI numbers and their corresponding landing destinations. From this table the system fetches the extension number configured as the landing destination.

- When a match is found in the DDI Routing Table for the incoming call on the respective DDI number, the system places the call on the corresponding landing destination configured by you.

  When the call is not answered by the landing destination or if the landing destination is busy, the system again checks the Incoming Reference Table for the options you have selected for the parameters, When No Reply and When busy. You may select from the following options - Disconnect, Route to a TLG, Greet and Disconnect, Greet and Route to TLG or Route to VMS.

- If no match is found for the DDI number in the DDI Routing Table, the call is routed as per the Trunk Feature Template assigned in the Incoming Reference Table.

Handling Outgoing Calls

Reverse DDI is when you want the DDI numbers assigned to the extension users to be displayed as the CLI to the called party. For this you must configure the DDI Routing Table and the Outgoing Reference Table.

When an outgoing call is made by the extension user, the DDI numbers will be sent as CLI to the called parties only if, the service provider supports this facility. If this facility is not supported by the service provider and if SARVAM UCS sends the DDI Number, this number will be swapped by with the Pilot number by the service provider and then the call will be routed further.

Pre-requisites to route outgoing calls as per the DDI logic,

- Assign an Outgoing Reference ID to the BRI port. This Outgoing Reference ID acts as a link between the port settings and the Outgoing Reference Table.
- In the Outgoing Reference Table against the same OG Reference ID, configure the Start Channel No., Total Channel Count, DDI Routing Ref. ID and the ISDN Number as per your requirement.
- To fetch the DDI number the system refers to the DDI Routing Ref. ID configured in the Outgoing Reference Table. As per this ID the system checks the DDI Routing Table. The DDI Routing Ref. ID acts as a link between the Outgoing Reference Table and the DDI Routing Table.

When an outgoing call is made using the BRI port,

- The extensions user dials a trunk access code to grab the BRI line to out-dial a number.
- As per the time zone, the system checks for the Outgoing (OG) Reference ID assigned to the BRI port.
- If the OG Reference ID has been configured, the system then hunts for the same ID in the Outgoing Reference Table.

  If a match is found, the system verifies the channels assigned to the user for making an outgoing call. These are defined in the parameters Start Channel Number and the Total Channel Count. Hence the system also matches the channel assigned for making the outgoing call with the channel configured in the table.
- The system the checks the corresponding DDI Routing Reference ID. The DDI Routing Reference ID acts as a link between the Outgoing Reference Table and the DDI Routing Table.
- The system hunts for the same DDI Routing Reference ID in the DDI Routing Table. From this table the system fetches the DDI number assigned to the extension user. You can configure as many as 224 DDI Reference ID’s with different DDI numbers.
- When a match is found in the DDI Routing Table, the system sends the MSN Number + DDI number of the extension in the calling party field to the called party.
- If no match is found in the DDI table, the ISDN Number, that is the MSN Number + the first DDI number is sent in the calling party field to called party.

Configuring using Jeeves

To route incoming calls as per the DDI logic you must configure the following:

- Assign the Incoming Reference ID on the respective port.
- Incoming Reference Table
- DDI Routing Table

To configure the Incoming Reference Table,

- Determine the ISDN ports/SIP trunk on which you want to apply the DDI logic, for example, BRI port.
- On the BRI port, assign an Incoming Reference ID number. To do this,
  - Log into Jeeves as System Engineer.
  - Under **Configuration**, click **BRI Configuration**.
  - Click **BRI Parameters**.
  - Scroll to **Incoming (IC) Reference ID**.
  - Assign an **Incoming (IC) Reference ID** number for each time zone.
- Under **Configuration**, click **DDI Routing**.
- Click **Incoming Reference Table** and configure the following parameters:
  - **IC Reference ID**: You must configure the parameters to apply DDI Routing against the same ID as assigned on the BRI port. This ID is a link between the port settings and the Incoming Reference Table.
  - **Start Channel Number**: The system starts applying the routing logic from this channel of the trunk.
  - **Total Channel Count**: This is the total number of channels of the trunk on which the system applies the routing logic.
  - **DDI Routing Reference ID**: Enter the DDI Routing Reference ID number. As per the this reference number the system checks the DDI Routing Table and determines the final landing destination for placing the call.

    This ID is a link between the Incoming Reference Table and the DDI Routing Table.
  - **Route on First Destination**: When the final destination extension is identified as per the DDI number table, the system checks if this check box is enabled. If it is enabled, every incoming call will always be placed on the first extension configured in the DDI Table. If the check box is clear, the call is routed to the respective extension number.
  - **Ring Timer**: This timer signifies the time for which the extension on which the incoming call is placed must ring. On expiry of this timer if the call is not answered the call is routed as per the option you configure for the parameter **When No Reply**.
  - **When No Reply**: When the call is not answered by the DDI extension, the system checks the option you have selected for this parameter and processes the call further. You may select any option from the following:

- Disconnect the call
- Route the call to Trunk Landing Group
- Answer the call automatically, greet the caller with a voice message, and on completion of message disconnect the call.
- Answer the call, greet the caller with a voice message, and on completion of the message route the call to Trunk Landing Group.
- Route the call to Voice Mail, that is to the Mail box assigned to the DDI extension. For this, a mail box must be assigned to the extensions. If the extension is not assigned mail box, the caller will hear the welcome message of the VMS, but will not be able to access the mail box.

- **When Busy**: If the DDI extension is busy, the system checks the option you have selected for this parameter and processes the call further. You may select any option from the following:
  - Disconnect the call.
  - Route the call to Trunk Landing Group.
  - Answer the call automatically, greet the caller with a voice message, and on completion of message disconnect the call.
  - Answer the call, greet the caller with a voice message, and on completion of the message route the call to Trunk Landing Group.
  - Route the call to Voice Mail, that is to the Mail box of the DDI extension. For this, a mail box must be assigned to the extensions. If the extension is not assigned mail box, the caller will hear the welcome message of the VMS, but will not be able to access the mail box.

- **Trunk Feature Template**: A Trunk feature template is assigned to each Incoming Reference ID. You can enable different features in the Trunk Feature Templates. A different Trunk Feature Template can be assigned to each channel or a range of channels, such as set Auto Answer Time Zone wise, allow DID according to the time zone. Hence incoming calls can be answered and routed in different ways. For more details refer the topic “Trunk Feature Template”.

- Click **Submit** to save the settings.

- Similarly, to assign IC Reference ID to T1E1 ports, under **Configuration**, click **T1E1 Configuration**. Click **Port Parameters**.

- To assign IC Reference ID to SIP Trunks, under **Configuration**, click **VoIP Configuration**. Click **SIP Trunk Parameters**.

- After the Incoming Reference Table has been configured you must configure the DDI Routing Table.

  DDI numbers can be assigned to extensions users, Department Groups, Routing Groups or to different port types, depending on the organizations requirement.

  You can configure as many as 224 DDI Reference ID’s with different DDI numbers and landing destinations. To configure the DDI Routing Table,

- Under **Configuration**, click **DDI Routing**.

- Click **DDI Routing Table** and configure the following parameters:

  - **DDI Routing Reference ID**: You must configure the parameters to apply DDI Routing against the same ID as assigned in the Incoming Reference Table. This ID is a link between the Incoming Reference Table and the DDI Routing Table.

DDI Routing Ref. ID's in the DDI Routing Table act as identifiers. This Reference ID is assigned in the Incoming Reference Table.

When there is an incoming call the system checks the Incoming Reference ID assigned to the port/ trunk. The system tracks this ID from the Incoming Reference Table.

- **Start DDI Number**: This is the first DDI Number from where the DDI range starts. You must only enter the DDI number. For example, if the Pilot Number is 2630555. If 10 DDI numbers have been taken, then the DDI Number range is from 555 to 564. The Start DDI Number in this case is 555.

- **Total DDI Numbers**: The total number of DDI numbers for you want to set the same landing destination. For example, if the Start DDI Number is 555, Total DDI Number is 1, then the landing destination selected will only be for the DDI Number 555. If Start DDI number is 556 and the Total DDI Numbers is configured as 10, then the landing destination will be for the entire range of numbers, that is from 556 to 565.

- **DDI Number of Digit**: The number of digits in a DDI number. Suppose 100 DDI numbers are supported on an ISDN Trunk, then the Number of Digits for that Trunk should be configured as 3. Suppose 10 DDI numbers are supported on another ISDN Trunk, then the Number of Digits for that Trunk should be configured as 2.

- **Route to Destination**: The incoming calls on the DDI numbers can be placed on extensions users, Department Groups, Routing Groups or to different port types, depending on the organizations requirement.

  Configure the following for the parameters for the landing destination:

  - **Port Type**: Select the port type to place the DDI call from the following:
    - BRI
    - T1E1
    - E&M
    - AOP
    - Department Group
    - Quick Dial
    - Routing Group
    - Voice Mail Auto Attendant
    - Mobile
    - SIP Trunk
    - Flexible Number
    - Virtual Extension

    For example, to use SARVAM UCS as a gateway select Port Type as Mobile, BRI, T1E1 etc.

    To place incoming calls on the extensions, select Flexible Number as the Port Type.

  - **Port Number**: For the Port Type you have selected, enter the Port Number/Group Number. The Port Number range depends on the Port Type you have selected.

  - **Start DDI Flexible Number**: If you have selected the Port Type as Flexible Number, enter the first extension number to which you want to assign the first DDI Number. As per the Total DDI numbers configured, the system automatically maps the DDI numbers with the extension numbers. Both the DDI numbers and the extensions numbers are mapped sequentially. For example, if the Start DDI

number is 2630555, the Total DDI Numbers are 10 and Start DDI Flexible Number 2001, the system maps the DDI numbers with the extensions numbers as follows:

DDI Number	Start Flexible Number
555	2001
556 564	2002 2010

Hence the call for the respective DDI Number is placed on the respective extension number.

- **Voice Mail Auto Attendant (VMAA) Menu:** if you have selected the *Voice Mail Auto Attendant* as the Port Type, select the VMAA Menu to be assigned to the respective DDI Routing Table entry.

  You may click the *Voice Mail Auto Attendant (VMAA) Menu* link to edit the parameters of desired VMAA Menu. For details, see “Voice Mail Auto-Attendant Menu”.

- Click **Submit** to save the settings.

*DDI Routing is not supported on T1/E1 trunk line if you have selected **E&M** as the **Signal Type**.*

If you want to apply Reverse DDI, that is if you want DDI numbers to be displayed as the CLI to the called party, you must configure the Outgoing Reference Table. The system uses the same DDI Routing Table to trace the extension users and their respective DDI numbers when an outgoing call is made.

To assign the Outgoing Reference ID and to configure the Outgoing Reference Table you must,

- Determine the ISDN ports/SIP trunk on which you want to apply the Reverse DDI logic, for example, BRI port.
- On the BRI port, assign an Outgoing Reference ID number. To do this,
  - Log into Jeeves as System Engineer.
  - Under **Configuration**, click **BRI Configuration**.
  - Click **BRI Parameters**.
  - Scroll to **Outgoing (OG) Reference ID**.
  - Assign an **Outgoing (OG) Reference ID** number for each time zone.
- Under **Configuration**, click **DDI Routing**.
- Click **Outgoing Reference Table** and configure the following parameters:

- **Outgoing Reference ID**: To apply DDI Routing configure the parameters given below against the same ID as assigned on the BRI port. This ID is a link between the port settings and the Outgoing Reference Table.

- **Start Channel Number**: The system starts applying the routing logic from this channel of the trunk.

- **Channel Count**: This is the total number of channels of the trunk on which the system applies the routing logic.

- **ISDN Number**: Each ISDN Trunk is given an Pilot Number by the Service Provider. This is the combination of MSN Number and the first DDI Number. The Number is of maximum 16 digits. This is also known as ISDN Installation Number/ISDN Number. The MSN number is given by the service provider whereas the DDI Numbers can be selected by the user. However the number of digits to be used for the DDI Number is informed by the service provider.

- **DDI Routing Reference ID**: Enter the DDI Routing Reference ID number. As per the this reference number the system checks the DDI Routing Table and determines the DDI number assigned to the extension user. The system then swaps this DDI number with the ISDN number when an outgoing call is made. This DDI number is displayed as the CLI to the called party.

The DDI numbers will be displayed as CLI to the called party only if the service provider supports this facility over the ISDN Trunk/SIP Trunk.

- Click **Submit** to save the settings.

- Similarly, to assign OG Reference ID to T1E1 ports, under **Configuration**, click **T1E1 Configuration**. Click **Port Parameters**.

- To assign OG Reference ID to SIP Trunks, under **Configuration**, click **VoIP Configuration**. Click **SIP Trunk Parameters**.

Configuring IC Reference Table using a Telephone

- Enter SE mode from a DKP/SLT.

- To program IC Reference Table Index, dial:
  - ***6302-1-IC Reference Table Index-Feature Number-Value***
  - ***6302-2-IC Ref. Table Index-IC Ref. Table Index-Feature Number-Value***
  - ***6302-*-Feature Number-Value***
    Where,
    IC Reference Table Index is from 01 to 64.
    Feature Number is from 01 to 09.

    See the table at the end of this topic for Feature numbers and the codes.

- To restore default IC Reference Table Index, dial:
  - ***6301-1-IC Reference Table Index*** to restore default of a single index.
  - ***6301-2-IC Ref. Table Index-IC Ref. Table Index*** to restore defaults of a range of index.
  - ***6301-**** to restore defaults of the entire table.
    Where,
    IC Reference Table Index is from 01 to 64.

- Exit SE mode.

Default values of the IC Reference Table

Feature No.	01	02	03	04	05	06	07	08	09
Feature Name/Table Index	IC Ref. ID	Start Channel No.	Total Channel Count	DDI Routing Ref. ID	Route on First Dest.	DDI Ring Timer (Sec.)	When No-Reply	When Busy	Trunk Feature Template
01	00	01	00	00	Disable	045	Disconnect	Disconnect	01
02	00	01	00	00	Disable	045	Disconnect	Disconnect	01
:	:	:	:	:	:	:	:	:	:
64	00	01	00	00	Disable	045	Disconnect	Disconnect	01

Parameters Value:

Code	0	00-99	01-30	00-30	00-99	Disable	001-255	Disconnect	Disconnect	01-50
	1					Enable		Route to TLG	Route to TLG	
	2							Greet and Disconnect	Greet and Disconnect	
	3							Greet and Route to TLG	Greet and Route to TLG	
	4							Route to Voice Mail Auto Attendant	Route to Voice Mail Auto Attendant	

Configuring OG Reference Table using a Telephone

- Enter SE mode from a DKP/SLT.
- To program OG Reference Table Index, dial:
  - **6312-1-OG Reference Table Index-Parameter Number-Value**
  - **6312-2-OG Reference Table Index-OG Reference Table Index-Parameter Number-Value**
  - **6312-*-Parameter Number-Value**
    Where,
    OG Reference Table Index is from 01 to 64.
    Parameter Number is from 1 to 5.
- To default an OG Reference Table:
  - **6311-1-OG Reference Table Index**
  - **6311-2-OG Reference Table Index-OG Reference Table Index**
  - **6311-***
    Where,
    OG Reference Table Index is from 01 to 64.
- Exit SE mode.

Default values of OG reference table as given below:

Parameter No.	1	2	3	4	5
Parameter Name/ Index	OG Ref. ID	Start Channel No.	Channel Count	ISDN Number	DDI Routing Ref. ID
01	00	01	00	Blank	000
02	00	01	00	Blank	000
03-63	Same as 02				
64	00	01	00	Blank	000

Parameter Value:

Code	01-99	01-02	01-30	16 digits	001-128
		01-30			

Configuring DDI Routing Table using a Telephone

- Enter SE mode from a DKP/SLT.

- To configure the feature in a DDI Routing Table:
  - **6322-1-DDI Routing Table ID-Parameter Number-Value**
  - **6322-2-DDI Routing Table ID-DDI Routing Table ID-Parameter Number-Value**
  - **6322-*-Parameter Number-Value**
    Where,
    DDI Routing Table ID is from 001 to 224.
    Parameter Value is from 01 to 06. Refer the DDI Routing Table for Parameter Values and their codes.

- To default a DDI Routing Table:
  - **6321-1-DDI Routing Table ID**
  - **6321-2-DDI Routing Table ID-DDI Routing Table ID**
  - **6321-***
    Where,
    DDI Routing Table ID is from 001 to 224.

- Exit SE mode.

Following table shows default DDI Routing Table

Para. No./ Table ID	01	02	03	04	05		06
	DDI Routing Ref. ID	Start DDI Numbers	Total DDI Numbers	DDI Number of Digit	Port Type	Port Number	Start Flexible Number
001	00	000000	000000	0	None	Blank	Blank
002	00	000000	000000	0	None	Blank	Blank
003-223	Same as 224						
224	00	000000	000000	0	None	Blank	Blank

Parameter Value:

Code	00	00-99	000001-999999	000000-999999	0-4	None	--	Max. 6-digits
	04					BRI	01-32	
	05					T1E1PRI	1-8	
	06					E&M	01-32	
	11					Dept Grp	01-24	
	12					Quick Dial	001-999	
	20					Routing Group	01-32	
	25					Mobile	01-40	
	26					SIP	01-32	
	27					Flex. No.	4-digits	
	36					Virtual Extension	01-64	

- Refer to above table. Whenever Flexible number is selected as the port type, the range concept of Start flexible number becomes relevant. In all other cases, range concept is not relevant.
- Flexible Number includes SIP Extension Numbers.
- When a call is placed on SLT/DKP port, the calling party number is displayed on the terminal.

When the call is placed on BRI-NT or PRI-NT, the calling party number and the called party number both are sent to the NT port. Doing so, the system connected to the NT port can resolve the DDI number and place the call on the programmed extension.

Direct Inward System Access (DISA)

What's this?

With Direct Inward System Access (DISA) remote users can access and use the system's features and facilities using Trunks, on which this feature is enabled.

Using DISA, remote users can:

- call any extension.
- make external calls.
- use features and facilities of the system.
- configure features and facilities of the system and administer the system.

All these can be done as if being done from a local extension of the SARVAM UCS.

DISA Variants

SARVAM UCS offers three types of DISA, each with a different method of authentication and level of access:

- PIN Authentication-Multiple Calls
- DISA with CLI Authentication-Multiple Calls
- DISA with CLI Authentication-Single Calls

PIN Authentication - Multiple Calls

Callers can access an extension of SARVAM UCS by dialing the DISA Login Code that consists of:

- the DISA Feature Access Code.
- the extension number they want to access.
- the User Password of that extension.

Callers are authenticated and allowed to use the extension on which they are logged in.

The callers must dial special digits or codes to go On-hook, Off-hook. They are allowed to make as many trunk calls and internal calls for as long as they remain logged in to the DISA mode.

To end the DISA login session, callers must dial the Termination code or disconnect from the remote end.

Callers can access an extension to use DISA PIN Authentication-Multiple Calls only if the extension has DISA feature enabled in its Class of Service.

DISA with CLI Authentication - Multiple Calls

The system authenticates the caller by matching the caller's CLI with the entries of the *DISA-CLI Authentication Table* and logs the caller in to the extension configured as 'Auto Login' extension for the CLI.

Callers are not required to dial any DISA Login Code or any password.

When a caller is authenticated on the basis of CLI, the system plays the ('internal' system) Dial Tone to the caller.

The callers must dial special digits or codes to go On-hook, Off-hook. They are allowed to make as many trunk calls and internal calls for as long as they remain logged in to the DISA mode.
To end the DISA login session, callers must dial the Termination code or disconnect from the remote end.

For this type of DISA, the DISA CLI Authentication Table must be configured first.

DISA with CLI Authentication - One Call

This type of DISA is similar to the previous one. The system authenticates callers by matching the callers' CLI with the entries of the DISA-CLI Authentication Table and logs the callers in to the extension designated as 'Auto Login' extension for the CLI.

When the caller is authenticated on the basis of CLI, the system gives the caller direct access to the Outgoing Trunks selected for TAC-1 for the current time zone (working hours, break hours, non-working hours)' in the "Station Basic Feature Template" assigned to the Auto Login extension. It plays the dial tone.

Callers are allowed to make a single external call. The system ends the DISA session on the completion of the call by the caller or by the other remote party.

For this type of DISA, the DISA CLI Authentication Table must be configured first.

To make another call, the caller must enter the DISA mode again, by calling the SARVAM UCS from the remote location.

DISA with CLI Authentication - One Call Answer Signaling is generally used when SARVAM UCS is deployed in the Gateway Mode, where SARVAM UCS is configured to send an answer signal to the caller/calling device, receive the DTMF digits dialed by the caller/calling device and dial out the digits dialed by the caller/calling device. To know more, see "Gateway Application-Answer Signaling".

How it works

For this feature to work, you must enable the desired DISA variant on the desired trunks: CO, Mobile, SIP, T1E1PRI, BRI.

- A call lands on a DISA enabled Trunk.
- The system checks if a DISA variant is enabled on the trunk for the current time zone, that is, working hours, break-hours and non-working hours.
- If a DISA variant is enabled on the trunk, the system processes the call according to the DISA variant enabled on the trunk.
- If **DISA with PIN Authentication - Multiple Calls** is enabled,
  - The system plays Welcome Greeting message to the caller[293].
  - The caller must dial the DISA Login Code consisting of:
    - the DISA Feature Access Code.
    - the number of the extension the caller wants to access.
    - the user password of the extension.
  - On successful login, the system starts the *DISA Idle State Timer* (configurable; default: 20 seconds). The system waits for the caller to go Off-hook[294].

293. If no voice message is recorded, the system plays music-on-hold to the caller.
294. If the caller does not go Off-hook within this timer, the system releases the call.

- When the caller goes Off-hook by dialing the Off-hook code #1, the system plays the internal dial tone and waits for the caller to dial digits.
- If the caller dials an external number using a CO trunk, the system starts the *DISA Inactivity Timer* (configurable; default: 2 minutes)[295].
- The system waits for the caller to dial digits within the DISA Inactivity Timer.
- The system reloads this timer each time it receives digits from the caller. If the caller fails to dial any digit within this timer, the system plays beeps for the duration of the *DISA Warning Beeps Timer* (fixed; 15 seconds). If no digit is received at the end of the Warning Beeps, the system terminates the DISA session. If digits are received before the end of the Warning Beeps, the system reloads the DISA Inactivity Timer.
- The caller can make as many trunk calls and internal calls as the caller wants.
- The caller can terminate the DISA login session either by disconnecting from the remote end or by dialing the Termination Code *#9*.

- If **DISA with CLI Authentication - Multiple Calls** is enabled,
  - The system compares the CLI of the caller with the *Calling Party Numbers* configured in the CLI Authentication Table.
  - If the CLI matches with any of the Calling Party Numbers in the Table, the system provides access to the extension configured as *Auto Login* extension for this Calling Party Number in the Table[296].
  - The caller gets logged into the Auto Login extension and gets the dial tone of SARVAM UCS.
  - At the end of the call, the caller dials the On-hook code *#0* to go On-hook. To make another call, the caller dials Off-hook code #1 and dials the desired number. Thus the caller dials the On-hook and Off-hook codes to make as many trunk and internal calls as desired.
  - If the caller dials an external number using a CO trunk, the system starts the *DISA Inactivity Timer* (configurable; default: 2 minutes)[297]. The system waits for the caller to dial digits within the DISA Inactivity Timer.
  - The system reloads this timer each time it receives digits from the caller. If the caller fails to dial any digit within this timer, the system plays beeps for the duration of the *DISA Warning Beeps Timer* (fixed; 15 seconds). If no digit is received at the end of the Warning Beeps, the system terminates the DISA session. If digits are received before the end of the Warning Beeps, the system reloads the DISA Inactivity Timer.
- If **DISA with CLI Authentication - One Call Answer Signaling** is enabled,
  - The system compares the CLI of the caller with the *Calling Party Numbers* configured in the CLI Authentication Table.

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295. DISA Inactivity Timer is not applicable for T1E1PRI lines, BRI lines, SIP and Mobile trunks.
296. If no match is found for the CLI of the caller in the Table, the call will be routed as per the Incoming Call Routing configured in SARVAM UCS.
297. DISA Inactivity Timer is not applicable for T1E1PRI line, BRI, SIP and Mobile trunks.

- If the CLI matches with any of the Calling Party Numbers in the Table, the system provides access to the extension configured as *Auto Login* extension for this Calling Party Number in the Table[298].
- The caller gets logged into the Auto Login extension and gets dial tone of the outgoing trunks selected for TAC-1 for the current Time Zone (working hours, break hours, non-working hours).
- If the caller dials an external number using a CO trunk, the system starts the *DISA Inactivity Timer* (configured; default: 2 minutes).
- The system waits for the caller to dial digits within the DISA Inactivity Timer. If the caller fails to dial any digit within this timer, the system plays beeps for the duration of the *DISA Warning Beeps Timer* (fixed; 15 seconds). If no digit is received at the end of the Warning Beeps, the system terminates the DISA session. If digits are received before the end of the Warning Beeps, the system reloads the DISA Inactivity Timer.
- After the external call is completed, that is, the caller disconnects from the remote end or the other remote called party has disconnected, the caller is logged out.
- To make another external call, the caller must call the DISA enabled trunk of SARVAM UCS again.

In all the variants of DISA, the caller can use all the features allowed in the "Class of Service (COS)" of the extension the caller is logged in to (using PIN Authentication or CLI Authentication).

- *DISA calls in the SMDR report are marked as "O" in the remarks column. See "Station Message Detail Recording-Report".*
- *If DISA is disabled, SARVAM UCS will route the call by Auto Attendant logic, if Auto Attendant is enabled.*
- *If DISA and Auto Attendant both are disabled, the incoming call will be routed as per the incoming call routing configured. To know more, see "Auto Attendant".*

***WARNING!** This feature allows access to system resources to remote users, and therefore has serious implications for your system's security. There is a risk of fraudulent calls being made from your system, if a third party comes to know the authentication PIN or the User Password of an extension number. The cost of such fraudulent calls will have to be borne by the owner of SARVAM UCS.*

To avoid unauthorized access, we recommend you to change the PIN regularly. Make sure it is strong and is provided to users who need to access the system using DISA only.

Feature Interaction:

- If both, Built-In Auto Attendant and DISA are enabled on the trunk, SARVAM UCS supports all types of DISA.
- If both, VMS Auto Attendant and DISA are enabled on the trunk, SARVAM UCS supports only PIN Authentication-Multiple Calls. To know how the VMS handles a DISA call, see "VMS DISA Login".

298. *If no match is found for the CLI of the caller in the Table, the call will be routed as per the Incoming Call Routing configured in the SARVAM UCS.*

How to configure

To provide DISA to remote users you need to do the following configuration:

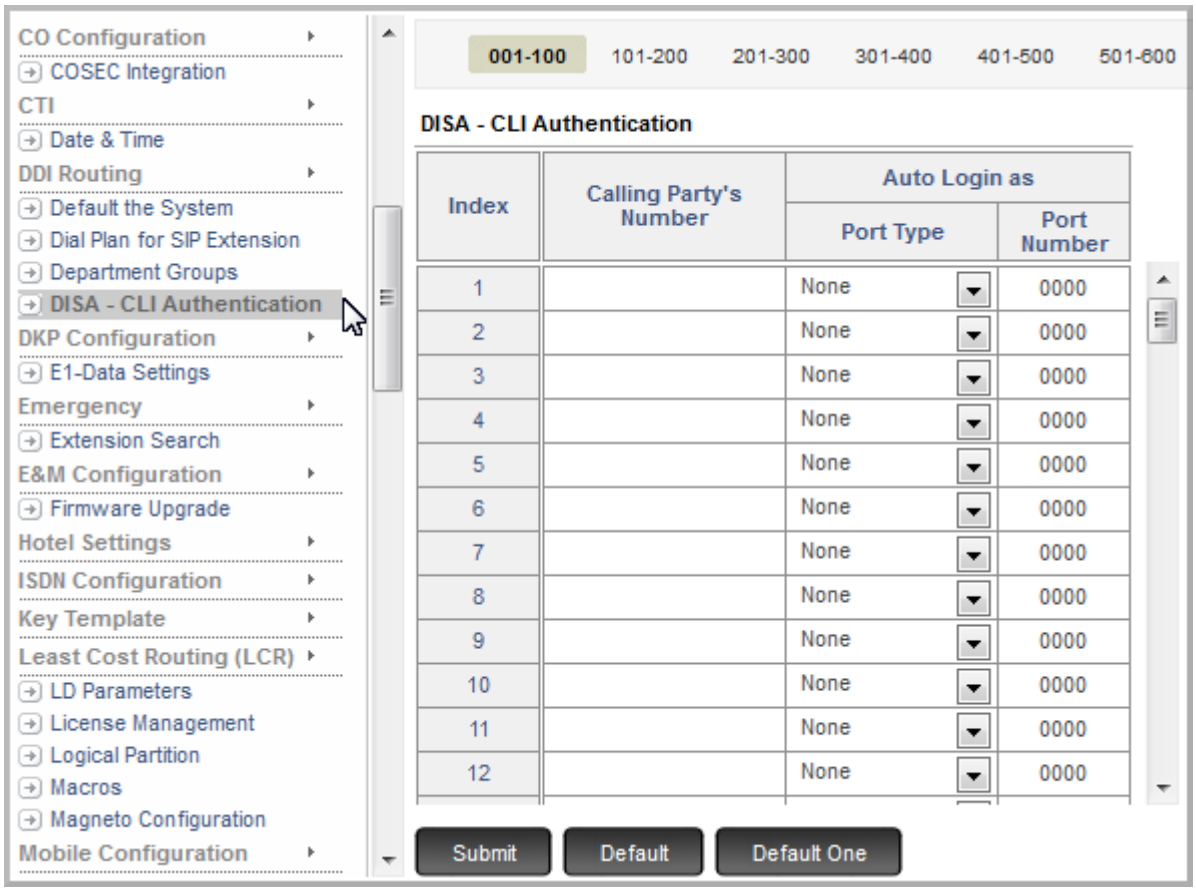
- Select the DISA variant for the Trunks on which you want to apply this feature in their “Trunk Feature Template”.
- Enable DISA in the “Class of Service (COS)” of the extensions which you want to allow callers to access using DISA. This is applicable for DISA PIN Authentication-Multiple Calls only. DISA CLI Authentication-Single Call/Multiple Calls is not dependant on COS.
- Change the User Password of the DISA extensions, if you selected DISA PIN Authentication-Multiple Calls. If you selected *DISA PIN Authentication-Multiple Calls* on a trunk, the default User Password (1111) will not work. See “User Password” and “System Security” more information and instructions.
- Configure the related timers, *DISA Idle State Timer* and *DISA Inactivity Timer*, if required. See “System Timers and Counts” for instructions.
- If you have selected the *DISA CLI Authentication-Multiple Calls* or *CLI Authentication-One Call Answer Signaling* on a trunk, you must configure the **CLI Authentication Table**.

Configuring DISA CLI Authentication Table using Jeeves

To configure the DISA CLI Authentication Table,

- Make a list of remote users and their numbers whom you want to allow DISA.
- For each remote user’s number on your list, write the Extension number of the SARVAM UCS you want to allow this extension user to log in.
- Open Jeeves.
- Log in as System Engineer.

- Under **Configuration**, click **DISA - CLI Authentication**. The CLI Authentication Table page opens.



- You can configure as many as 999 numbers in this table, by clicking the tabs of the index on the top of the table.
- Refer to the list of remote user numbers and the corresponding SARVAM UCS extension numbers you made.
- In the **Calling Party's Number** column, enter the number of the remote users whom you want to allow access to DISA using CLI Authentication. The system will match the CLI of the callers with the numbers you store here.
- For each Calling Party Number, in the **Auto Login as** field, select the extension **Port Type** (SLT, DKP, SIP Extension, ISDN Terminal, Virtual Extension) and **Port Number** you want to allow access to after the Calling Party Number is authenticated.
- Click **Submit** button to save your entries.
- You may log out of Jeeves.

Configuring DISA CLI Authentication Table using a Telephone

- Enter SE mode from a DKP/SLT.

To enter Calling Number in the CLI Authentication Table, dial:

- ***4111-Index-Calling Number-#****

Where,
Index is from 001 to 999.
Calling Number may contain a maximum of 16 digits.The allowed digits are 0-9, #, *, A, B, C, D, +. Use following codes to enter these digits:

Special Digit	Code
A	#4
B	#5
C	#6
D	#7
+	#8
*	**
#	##

To assign Port Type and Port Number, dial:

- ***4112-Index-Port Type-Port Number***
  Where,
  Index is from 001 to 999.
  Port Type and Port Number are:

Port Type	Port Number	Meaning
00	000	None
01	001 to 512	SLT
02	001 to 128	DKP
28	01 to 64	ISDN Terminal
34	001 to 999	SIP Extension
36	01 to 64	Virtual Extension

  For example, to configure extension '3001', which is a DKP with port number 001, as auto login station in Index 001 of the Table, you must dial ***4112-001-02-001***.

- Exit SE mode.

How to use

If you are a Remote user, to be able to use DISA, you must know:

- the number of the Trunk on which DISA is enabled and the variant of DISA enabled on this trunk.
- the number of the extension and the user password which you want to access, if using DISA with PIN Authentication.

- the duration of the DISA related Timers: The *DISA Idle State Timer* and the *DISA Inactivity Timer*, so that you may dial digits accordingly, without delay.

- the special digits to be dialed during a DISA login session.

Dialing Special Digits

After successful login, you will be required to go on-hook, go off-hook, use 'flash', use 'pause' or dial characters like A, B, C, D, from your remote device to use the features and facilities of SARVAM UCS.

However, SARVAM UCS will not be able to understand the conventional way of dialing 'flash' key or going on-hook with momentary make/break of loop current. Therefore, SARVAM UCS supports specific codes for specific activities. If these codes are received during a DISA session, SARVAM UCS interprets it and performs the associated activity.

When you are in DISA mode, use the following codes to indicate an activity:

Special Digit/activity	Code to be dialed
on-hook	#0
off-hook	#1
Flash	#2
Pause	#3
A	#4
B	#5
C	#6
D	#7
+	#8
To Terminate the DISA	#9
#	##
End of String	#*
To program # when in SE Mode	####

To use DISA,

- Dial the number of the Trunk on which DISA is enabled for the current time zone, Working, Break, Non-working hours.

- SARVAM UCS answers the call. You will get music or Built-In Auto Attendant Voice Message, if configured.

If **DISA PIN Authentication-Multiple Calls** is enabled,
- You will get beeps at the end of music/Built-In Auto Attendant Voice Message.
- Dial DISA Login Code 1079 during the beeps.
- Dial the extension number. You will get beeps.
- Dial the User Password for the extension during the beeps.

- You get feature tone on successful login.
- Dial the special digits **#0** to go On-hook and then **#1** to go Off-hook, you get the dial tone. Dial the number to make your calls like any local extension. For example, making an external call, dial Trunk Access Code '0' to grab a trunk and dial the number.
- To terminate DISA session dial **#9** or disconnect the call from your (remote) end.

If **DISA CLI Authentication-Multiple Calls** is enabled,

- You will get system dial tone.
- Dial the special digits **#1** to go Off-hook, **#0** to go On-hook, and make your calls like any local extension.
- To terminate DISA session dial **#9** or disconnect the call from your (remote) end.

If **DISA CLI Authentication - One Call Answer Signaling** is enabled,

- You will get Trunk dial tone.
- Dial the external number, without the Trunk Access Code.
- The DISA session will be terminated when you or the remote called party disconnects.
- To make another call, you must dial the number of the DISA enabled trunk again.

The features listed below are not supported in the DISA mode.

- *Auto Call Back*
- *Auto Redial*
- *Call Park*
- *Call Chaining*
- *Self Ring Test*
- *Trunk Reservation*
- *Walk-In Class of Service*
- *Live Call Supervision*

Direct Station Selection Console

The Direct Station Selection (DSS) Console is an add-on module with tri-color LEDs for the Digital Key Phones (DKP) and SPARSH VP510. It provides you quick access to Stations, Trunks, Features/Functions of SARVAM UCS; also making calling operations easy.

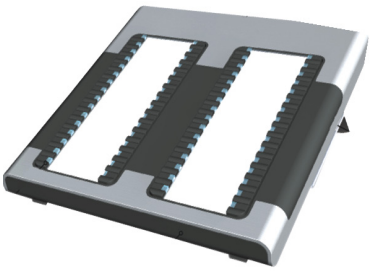
While the DSS Console is generally used by the Operator/receptionist in an organization, it is also meant to be used by anyone who needs to access various features of SARVAM UCS at a touch of a single key.

Matrix offers two types of DSS Consoles:

- **DSS64:** This can be attached only with EON48 and EON310.
- **DSS532**: This can be attached only with EON510 and SPARSH VP510.

Maximum 32 DSS Consoles (DSS64 or DSS532)[299] are supported by SARVAM UCS.

DSS64



A maximum of two DSS64 can be attached to a single DKP. Each DSS Console occupies a Digital Key Phone Port. For example, if you attach two DSS consoles to a single DKP, three DKP ports would be occupied.The DSS Console can be attached with SARVAM UCS in the same way as the DKP and is programmed as an attachment of the DKP.

Each DSS Console that is attached to a DKP occupies a DKP port. Hence, the more DSS Consoles you attach to DKPs, the lesser number of DKP ports will be available on SARVAM UCS.

When a single DSS64 is attached with a DKP, the DSS keys of the DKP as well as all the 64 keys of the DSS64 can be used. Similarly, if two DSS64 are attached to a DKP, 128 additional keys are at your disposal to be used as DSS keys.

To install and configure the DSS Consoles, refer “Installing DSS64” and “Programming DSS Console Keys”.

299. ETERNITY PENX supports maximum of 6 DSS Consoles.

DSS532

A maximum of four DSS532 can be attached to EON510 or SPARSH VP510. Unlike DSS64, the DSS532 does not occupy a DKP port. Although DSS64 and DSS532 work in a similar manner.

With each DSS532, 32 additional keys are at your disposal to be used as DSS keys. When all four DSS532 are connected, you get 128 additional DSS keys at your disposal to be used.

For instructions:

- to install the DSS532 with SPARSH VP510, see “Installing DSS532 with SPARSH VP510”.
- to install the DSS532 with EON510, see “Installing DSS532 with EON510”.
- to configure the DSS keys of the Console, see “Programming DSS Console Keys”.

DSS Keys

You can assign Station numbers or features/functions to the keys on the DSS Console, so that they can be accessed easily simply by pressing a single key.

LEDs

Each DSS Key is equipped with an LED which glows to indicate the status of the Trunk/Extension or Feature assigned to it.

Status of Extensions and Trunks

The LED of DSS keys assigned to Extensions/Trunks glow in three colors to indicate status of the call event on the Extensions/Trunks and on the phone.

Thus, the status of user's own Extension, status of other Extensions and status of the trunk lines are indicated by the LED of the DSS keys assigned to those Extensions and Trunks on the phone.

The following table shows the relationship between the color of the LED and various events:

LED Color	**LED Mode**		
	Continuously ON	Slow Blink	Fast Blink
Blue	The key assigned to the Extension you are in speech with.	The key assigned to the Extension you have kept on hold.	The key assigned to the Extension you are calling or from which you are being called.

LED Color	LED Mode		
	Continuously ON	Slow Blink	Fast Blink
Red	The key assigned to the Extension that is now busy with another Extension/Trunk.	The key assigned to the Extension which has put another Extension/Trunk on hold.	The key assigned to the Extension/Trunk that is called or being called by another.
Violet	You are talking on a Trunk (external call)	You have held a Trunk (external call)	You have an incoming call on the Trunk (external call)

- **Blue** indicates the state of the extension/trunk you access. For example, when you make a call to another Extension 203, the LED of the DSS key assigned to Extension 203 blinks Blue to indicate ringing at the Extension. If you have successfully established speech with Extension 203 the LED glows Blue continuously.

- **Red** indicates the state of other Extension/Trunks. For example, if the LED of the DSS key assigned to Extension 201 is glowing Red continuously, it means Extension 201 is busy with another Extension or Trunk.

- **Violet** indicates the state of the trunk you are in speech with. For example, when you are in speech on an outgoing call on Trunk 1, the LED of the DSS Key assigned to Trunk 1 will be continuously ON. When you put the call on hold, the LED will blink slowly.

Status of Features

The LED of a DSS key is activated when the feature assigned to this key is used.

Not all features require LED indication. For example, the feature Call Pick-Up does not require an LED. So when you assign this feature to a DSS key, the LED of the key remains inactive, when Call Pick-Up is accessed.

Feature like Auto Redial requires an LED to show that it has been set or canceled. So, the LED of the DSS key to which the Auto Redial feature has been assigned will glow Red, when Auto-Redial is set, and the LED is turned off when the feature is canceled.

Distinctive Rings

What's this?

Distinctive Rings are ringing patterns used for distinguishing between different types of call events.

SARVAM UCS supports the following types of call events:

1. Internal Call
2. Priority Internal Call
3. External Call
4. Alarm Call
5. Auto Call Back Call
6. Auto Redial Call
7. Message Wait Call
8. SE Mode (Programming Ring)
9. Operator Alarm
10. Emergency
11. Self Ring
12. Call Supervision
13. Presence
14. Emergency Conference
15. Conference

With Distinctive Rings, it is possible to use ring cadence of user's choice for each of these call events. For instance, Triple ring can be set for 'Priority Internal Calls' and long rings can be set for 'Alarm Calls'.

A set of ring types is called Distinctive Ring type. The default Distinctive Ring Types are:

Call Event	Ring Type - **T1**	Ring Type - **T2**	Ring Type- **T3**
Internal Call	Short Very Slow	Double	Double
Trunk Call	Double	Long Slow	Long Slow
Auto Call Back	Short Slow	Short Slow	Short Slow
Auto Redial	Long Very Slow	Long Very Slow	Long Very Slow
Self Alarm	Long Fast	Long Fast	Long Fast
Emergency	Long Fast	Long Fast	Long Fast
Operator Alarm	Long Fast	Long Fast	Long Fast
Message Wait Call	Short Fast	Short Fast	Short Fast
Programming Ring	Continuous	Continuous	Continuous
Self Ring	Short Slow	Short Slow	Short Slow
Priority	Triple	Triple	Triple
Call Supervision	Continuous	Continuous	Continuous

Call Event	Ring Type - **T1**	Ring Type - **T2**	Ring Type- **T3**
Presence	Continuous	Continuous	Continuous
Emergency Conference	Triple	Triple	Triple
Conference	Triple	Triple	Triple

These ring types have the following ring cadence:

Ring Pattern	Cadence (in milliseconds)
Short Fast	750-750
Short Long	500-1500
Short Very Slow	750-2250
Long Fast	1500-500
Long Slow	1000-4000
Very Long Slow	2000-4000
Double	400-200-400-2000
Triple	400-200-400-200-400-2000

Ring cadence is not programmable.

Distinctive Rings on SIP Extensions

On SIP Extensions, SARVAM UCS supports Distinctive Rings using Alert-INFO field in the INVITE message. To indicate the different call events, SARVAM UCS sends the Ring Text for the respective Ring Type.

The default Distinctive Ring Types and their corresponding Ring Texts are given below.

Call Events	Ring Type - **T1**	Ring Text
Internal Call	Short Very Slow	internal
Trunk Call	Double	external
Auto Call Back (ACB)	Short Slow	acb
Auto Redial (AR)	Long Very Slow	autord
Self Alarm	Long Fast	selfalram
Emergency	Long Fast	emergency
Operator Alarm	Long Fast	opratoralarm
Message Wait	Short Fast	msgwait
Programming Ring	Continuous	prog
Ring Test	Short Slow	test

Call Events	Ring Type - **T1**	Ring Text
Priority	Triple	priority
Call Supervision	Continuous	callsup
Presence	Continuous	presence
Emergency Conference	Triple	emergencyconf
Conference	Triple	conf

The Ring Text is sent in the Alert-INFO field of the INVITE message and the corresponding Ring Type is played on terminal registered as the SIP Extension, if the terminal supports Distinctive Rings.

The Ring Text is programmable. You can change the Ring Text, if required.

How to configure

At the time of installation, when you select the "Region" (as per the geographical location of the system), and set the system to default, SARVAM UCS loads the country-specific Distinctive Ring Type defined for the selected Region.

Refer the topic "Default Settings" for the default Distinctive Ring Type applied to your country/region.

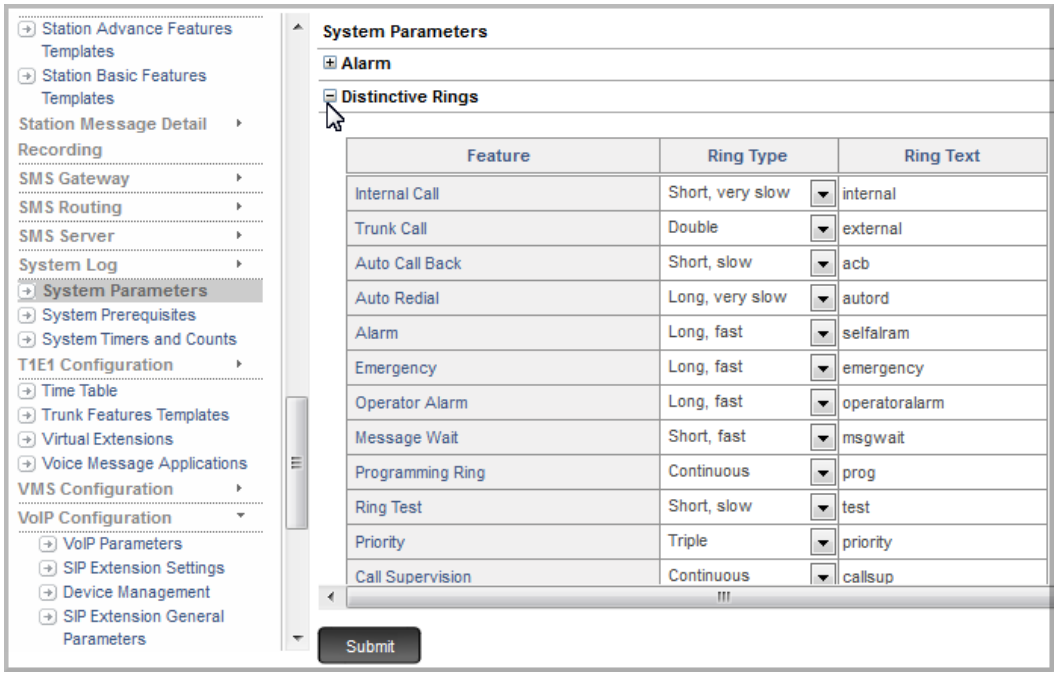
However, if required, you can change the default Ring Pattern and the Ring Text (for SIP Extensions) loaded by the system.

When you change the Ring Texts for the Ring Types in SARVAM UCS, you must configure the same Ring Text in the SIP phones.

Programming Distinctive Rings using Jeeves

- Log in as System Engineer.
- Under **Configuration**, click **System Parameters**.

- Click **Distinctive Rings** to expand.



- Select the desired Ring Pattern (Ring Type) for each call event that you want to customize.
- You can also customize the Ring Text of each call event on SIP extensions. The text can be a maximum of 20 alphanumerical characters.
- Click **Submit** to save changes.
- Log out of Jeeves or continue programming as required.

Programming Distinctive Rings using a Telephone

- Enter SE mode from a DKP/SLT.

  To change Ring Pattern for the call event, dial:
  - ***4002-Event-Ring Pattern***

    Where,

    Event is:

    01 for Internal Call
    02 for Trunk Call
    03 for Auto Call Back Call
    04 for Auto Redial Call
    05 for Self Alarm Call
    06 for Emergency
    07 for Operator Alarm
    08 for Message Wait Call
    09 for Programming Ring
    10 for Self Ring
    11 for Priority Call
    12 for Call Supervision
    14 for Presence

15 for Emergency Conference
16 for Conference

Ring Pattern is:

00 for OFF
01 for Continuous
02 for Short Fast
03 Short Slow
04 Short Very Slow
05 Long Fast
06 Long Slow
07 Long Very Slow
08 Double
09 Triple

To default ring patterns of events:

- Dial command ***4001***

- Exit SE mode.

Demonstration of rings

It is possible to demonstrate Ring Types to users by dialing the SE commands on DKP, Extended IP Phone and SLT extensions of SARVAM UCS.

By default, the system will play each Ring Type as demonstration for 30 seconds.

Users of SARVAM UCS may be acquainted with the different Distinctive Rings played by the system so that they can associate the terms used to describe the rings with the sound emitted by the system for each ring.

For EON and Extended IP Phone Users

- Enter SE mode.

  - Dial command ***4003-Ring Pattern***
    Where,
    Ring Pattern is

    01 for Continuous
    02 for Short Fast
    03 for Short Slow
    04 for Short Very Slow
    05 for Long Fast
    06 for Long Slow
    07 for Long Very Slow
    08 for Double
    09 for Triple

  - After dialing the desired Ring Pattern Code,

  - Press ‘Enter' key.

- You get the prompt 'Go Idle for Ring' on your phone display.
- Go Idle.
- The Ring Pattern you selected will be played.
- You may repeat the SE commands to demonstrate another Ring Pattern.

- Exit SE mode.

For SLT Users

- Enter SE mode.
  - Dial command ***4003-Ring Pattern***
    Where,
    Ring Pattern is

    01 for Continuous
    02 for Short Fast
    03 for Short Slow
    04 for Short Very Slow
    05 for Long Fast
    06 for Long Slow
    07 for Long Very Slow
    08 for Double
    09 for Triple
  - Go Idle, after dialing the desired Ring Pattern Code.
  - The Ring Pattern you selected will be played.
  - You may repeat the SE commands to demonstrate another Ring Pattern.
- Exit SE mode.

Do Not Disturb (DND)

What's this?

Extension users may restrict calls to their extensions in order to work uninterrupted by frequent phone calls. The feature, Do Not Disturb, enables users accomplish this. This feature is useful to extension users who are in the middle of a meeting or any important work that requires their undivided attention.

Using DND users can restrict—all calls, internal calls or external calls. However even if DND is set, users can route their incoming calls to an Intercept Destination. This destination can be the extension users own mailbox or another extension. In this way, extension users can ensure that they do not miss any important calls.

If required, when DND is set, a Stuttered Dial Tone can be played to the user for notification.

DND can be set and canceled by

- Extension users
- Operator for extension users, referred to as DND-Remote.

Doing so, calls — all, internal or external— will be barred. However, the extension user would continue to receive:

- Alarm calls.
- Reminder calls.
- Auto Call Back calls (Auto Callback as well as Auto Redial)
- Emergency Reporting Calls.

Also, the extension user can:

- use all the features of the System.
- make Outgoing calls and
- make Internal calls to other extensions.

DND has two supplementary features— DND-Override and Privacy from DND-Override.

The 'Do Not Disturb' feature bars calls to the phone on which DND is set. The 'DND-Override' feature breaks this bar and allowing the calls to land on the phone on which DND is set. Protection is also given to the phone on which DND-Override is attempted. If the phone on which DND-Override is attempted has 'Privacy from DND-Override' enabled, the calling phone shall not be able to Override the DND.

When a caller calls a phone on which DND is set, he/she gets Routing tone (Feature tone). The caller can dial DND-Override code. On dialing DND-Override code the call is placed on the called phone and the called phone starts ringing.

The DND-override feature works only if the calling phone has 'DND-Override' feature enabled in its CoS group.

DND-Override will not work if the called phone has ‘Privacy from DND-Override’ enabled in its Class of Service or if the called extension has opted for intercept routing.

So, using DND-Override feature, the users can be reached in case of some Emergency despite the DND set on the phone.

- *DND when set/canceled from the SA mode, will not depend on the assigned CoS.*
- *The system supports only single-point DND with Intercept Destination, which means, if the destination extension has also set DND with Intercept Destination, the call will not follow the forwarding path.*

How it works

For this feature to work,

- you must select the "DND Call Type".
- you may select the "Intercept Destination for DND".
- you may select the "DND Text Message" as per your requirement.
- you may assign a voice module for DND Notification. See "Voice Message for DND Notification".

DND Call Type

The extension user/Operator can select the type of calls to be restricted while enabling DND. They can select either All, Internal or External Calls.

Intercept Destination for DND

If the user wants that the calls are attended to even if DND is set, System Engineer must configure the Intercept Destination for the user. Incoming calls landing on the extension that has set DND will be routed to the Intercept Destination. This destination can be the users own mailbox or another extension (SLT, DKP, SIP).

DND Text Message

A DND Message is a short Text Message such as 'In a Meeting', 'In a Conference', 'On Vacation'.

When setting DND (also DND-Remote), the extension user/Operator can select an appropriate text message to be displayed to the calling extension.

This DND text message is displayed on the calling extension, only if the calling extension is the proprietary digital key phone, EON or an Extended IP Phone.

The SARVAM UCS supports 9 different DND Text Messages, out of which 8 messages can be changed as per user requirement by the System Engineer. User can select and set on their phones any of the DND messages programmed by the System Engineer.

Voice Message for DND Notification

Using this feature, a pre-recorded Voice Message can be played to the caller informing him/her about the DND set on the called extension. For example, "The dialed extension has activated Do Not Disturb".

When DND is set on an extension, callers who try to reach that extension will be played an error tone. Callers who are using EON/Extended IP Phones are displayed the DND Text Message set by the called extension, and thus come to know the cause of the error tone. Such a facility is not available to callers who are using SLTs, who can hear only the error tone and have no way of knowing the cause of the error tone.

Using Voice Message for DND Notification, a pre-recorded Voice Message can be played to the callers to notify them of the DND set on the called extension.

You must record and assign a Voice Module to play the pre-recorded voice messages as DND Notification to the callers.

Let us understand this feature with an example:

A, B and C are extension users.
B has EON, while C has an SLT.
B has DND-Override in his Class of Service, C does not have this feature.
DND Text messages as well as Voice Message Notification for DND have been programmed by the System Engineer.

- A has set DND on his extension with the DND Text message 'In Meeting'[300].
- B calls A.
- As B has DND-Override, the Voice Message for DND Notification is played to B once, and the DND message 'In Meeting' set by A appears on B's phone display. B gets routing Beeps.
- To exercise DND-Override, B must dial '4' the feature access code for 'DND-Override' during either during the Voice Message or during the routing Beeps.
- B gets Ring Back Tone, if A's extension is free.
- B gets Busy Tone, if A's extension is busy.
- However, if A has Privacy from DND Override, B will get error tone and the DND message set by A appears on B's phone.

If B fails to dial the DND-Override code before the end of the routing beeps, error tone will be played to him.

Now, to take another example,

- C calls A.
- As C has an SLT, C will get only the Error tone.
- But as Voice Message for DND Notification is programmed in the system, C will be played the pre-recorded message once.
- Since C is not allowed 'DND-Override' in his Class of Service, he cannot exercise this feature during the Voice Message.
- At the end of the voice message, C will be played error tone.

In the above examples, if A has set E’s extension as the Intercept Destination, then calls from B and C will be routed to E’s extension.

Feature Interaction:

Call Forward: When DND and Call Forward-Unconditional are set on an extension, Call Forward is given priority. If any other type of Call Forward and DND are set on an extension, DND is given priority. However, if DND with Intercept Destination is set, it will not work. If an extension has set both Call Forward and DND, then Feature Tone will be played to the extension user.

If an extension A has set DND with Intercept Destination as E, incoming calls on A’s extension will be routed to E’s extension.

If E sets Call Forward or DND with Intercept Destination as B, incoming calls from A’s extension will be still routed to E’s extension. Only incoming calls on E’s extension will be routed to the B’s extension.

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300. While DND and DND Text Message can be set from any phone, DND Text Message can be viewed on EON /Extended IP Phone only.

How to configure

For this feature to work, 'DND' must be enabled in the Class of Service of the group of the extensions which is to be allowed this feature.

Also, 'DND-Override' and 'Privacy from DND-Override' can be enabled in the Class of Service of the extensions to whom this features are to be provided.

Besides these, the System Engineer may program the DND Text Message, Stuttered Dial Tone when DND is set, Intercept Destination for DND and the Voice Message for DND Notification, as per user requirements.

The user can then select the type of calls for which he wants to set DND.

Programming DND in Class of Service

In the default Station Basic Feature Template 01 assigned to all extensions of the SARVAM UCS, the default CoS group 01 has DND enabled. So, all extensions of SARVAM UCS can set and cancel DND. DND-Override and Privacy from DND-Override are disabled in the default CoS group 01. So, none of the extensions can use DND-Override, or be exempt from DND.

While it makes sense to offer all extensions DND, providing DND-Override and Privacy from DND also to all extensions will not serve the purpose of DND.

Decide which extensions are to be allowed 'DND', which are to be allowed 'DND-Override', and which are to be allowed 'Privacy from DND-Override'. Generally, DND-Override is allowed to the Operator extension. It may be allowed to extensions of persons in senior positions in the organization. Similarly, Privacy from DND-Override may be allowed to persons in senior positions in the organization.

If you want to allow DND to all extensions, retain the default CoS group 01 in Station Basic Feature Template 01. However, if you want to allow DND only to selected extensions, disable this feature in the default CoS group 1.

Now, to assign DND to selected extensions, follow these steps:

1. Define a CoS group with DND enabled.
2. Prepare a Station Basic Template with this CoS group applicable in all the time zones.
3. Assign this newly prepared Station Basic Feature Template to the extensions on which 'DND' is to be enabled.

Similarly, if 'DND-Override' is to be to be allowed to the Operator and a few other extensions, follow these steps:

1. Define a CoS group with DND-Override enabled. If DND is also to be allowed, enable both DND-Override and DND in this CoS group.
2. Prepare a Station Basic Template with this CoS group applicable in all the time zones.
3. Assign this newly prepared Station Basic Feature Template to the Operator extension on which 'DND-Override' is to be enabled.

Repeat the same steps to allow 'Privacy from DND-Override' to selected extensions. For extensions that are to be allowed 'DND' as well as 'Privacy from DND-Override', enable both features in the CoS group in the Station Basic Feature Template applied on these extensions.

Similarly, for extensions that are to be allowed 'DND', 'DND-Override' and 'Privacy from DND-Override', enable all three features in the CoS group that you prepare for these extensions.

Refer the topics “Class of Service (COS)” and “Station Basic Feature Template” for detailed programming instructions on how to prepare a CoS in the Station Basic Feature Template and how to apply this template on extensions.

Programming DND Text Messages

By default, 9 DND Text Messages are programmed in the SARVAM UCS as listed below:

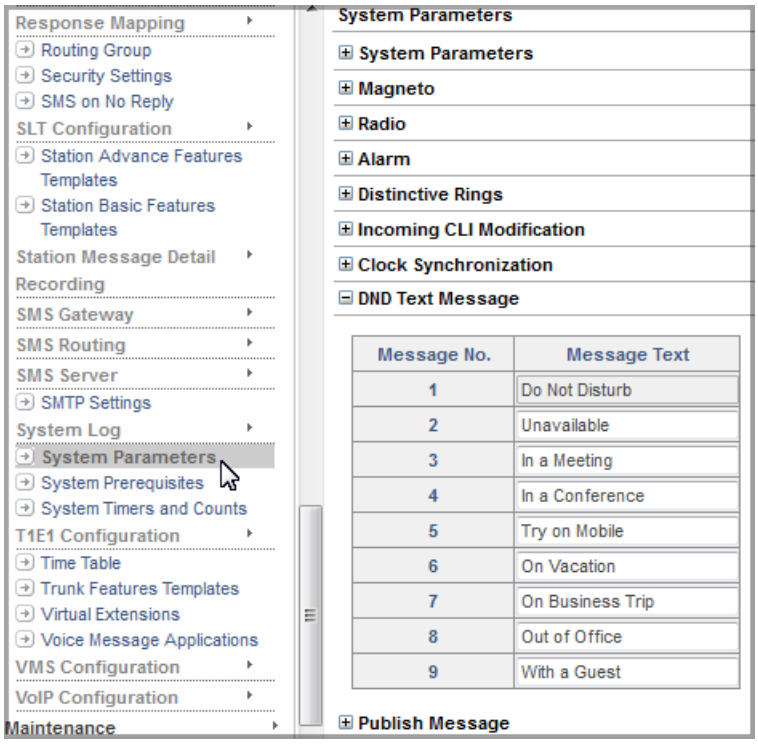
Message #	DND Message
1	Do Not Disturb
2	Unavailable
3	In a Meeting
4	In a Conference
5	Try on Mobile
6	On Vacation
7	On Business Trip
8	Out of Office
9	With a Guest

You can use these default message options or program messages from 2 to 9 as per user preferences.

Programming DND Messages using Jeeves

- Login as System Engineer.
- Under **Configuration**, click **System Parameters** to open the page.

- Click **DND Text Message** to expand.



- All the default text messages appear in the DND message field. You may change the DND text messages 2 to 9 as per your requirement. Click the field and enter your custom DND text message.

- Click **Submit** to save the changes.

Programming Stuttered Dial Tone

If required by users, then the system will play Stuttered Dial Tone on the extension when DND is set.

For instructions, see “System Parameters”.

Programming Intercept Destination for DND

By default, the Station Advanced Feature Template 01 is assigned to all extensions of the SARVAM UCS and in this template the Intercept Destination is configured as None.

You can either configure the Intercept Destination in this template or you may select another template and customize it as per your requirement. For instructions see “Station Advanced Feature Template”.

Programming Voice Message for DND Notification

To be able to play a voice message to callers for DND notification, you must first record a Voice Module with the desired message.

ETERNITY LENX/MENX supports 16 Voice Modules, ETERNITY GENX supports 16 Voice Modules and ETERNITY PENX supports 16 Voice Modules. The duration of each module can be 16 seconds.

Record a Voice Module with the message “The dialed extension has activated Do Not Disturb” (recommended). To assign the Voice Module for 'DND Notification'. Refer the topic “Voice Message Applications” for programming instructions.

How to use

DND set/canceled by Extension Users

For EON and Extended IP Phone Users

To set DND:

- Press the 'DND' Key.

  OR

- Dial 18
- Scroll to select the type of call:
  - All calls
  - Internal calls
  - External calls

- Press 'Enter' key.
- You get a text message 'DND Set' on the phone's display and confirmation tone.

To select a DND Message

- Press the 'DND' Key.

  OR

- Dial 18
- Scroll to select the Set DND Message option.
- The list of DND messages appear on the phone's display:

  - Do Not Disturb
  - Unavailable
  - In a Meeting
  - In a Conference
  - Try on Mobile
  - On Vacation
  - On Business Trip
  - Out of Office
  - With a Guest
- Scroll to the desired option and press 'Enter' key.
- You get a text message 'DND Set' on the phone's display and confirmation tone.
- Go Idle or you get dial tone after the confirmation tone.

To cancel DND:

- Press the DND Key again.
- The following options appear on the phone's display
  - All calls
  - Internal calls
  - External calls
  - Cancel DND

- Select Cancel DND and press 'Enter' key.

  OR

- Dial 18-0
- You get a text message 'DND Canceled' on the phone's display and confirmation tone.

For SLT Users

To set DND:

- Lift handset.
- Dial 18-1 to set DND for All calls
- Dial 18-2 to set DND for Internal calls
- Dial 18-3 to set DND for External calls
- Replace handset.

To select DND Message:

- Lift handset.
- Dial 18-4-1 for 'Do Not Disturb'
- Dial 18-4-2 for 'Unavailable'
- Dial 18-4-3 for 'In a Meeting'
- Dial 18-4-4 for 'In a Conference'
- Dial 18-4-5 for 'Try on Mobile'
- Dial 18-4-6 for 'On Vacation'
- Dial 18-4-7 for 'On Business Trip'
- Dial 18-4-8 for 'Out of Office'
- Dial 18-4-9 for 'With a Guest'
- Replace handset.

To cancel DND:

- Lift handset.
- Dial 18-0
- Replace handset.

DND-Remote

For EON and Extended IP Phone Users

To set DND for an extension user:

- Press the DSS Key assigned to DND-Remote.
  OR
  Dial 1072-001.
- Enter the extension number.
- Scroll to select the type of call:
  - All calls
  - Internal calls
  - External calls

- Press 'Enter' key.
- You get a text message 'DND Set on <Extension Number>' and confirmation tone.
- Go Idle or you get dial tone after confirmation tone.

To select a DND Message:

- Press the DSS Key assigned to DND-Remote.
  OR
  Dial 1072-001.
- Enter the extension number.
- Scroll to select the option Set DND Message
- The list of DND messages appear on the phone's display:
  - Do Not Disturb
  - Unavailable
  - In a Meeting
  - In a Conference
  - Try on Mobile
  - On Vacation
  - On Business Trip
  - Out of Office
  - With a Guest

- Scroll to the desired option and press 'Enter' key.
- You get a text message 'DND Set' on the phone's display and confirmation tone.
- Go Idle or you get dial tone after the confirmation tone.

To cancel DND-Remote:

- Press the key assigned Remote-DND function.
  OR
  Dial 1072-001.
- Enter the Extension Number.
- Scroll to select the message 'Cancel DND'.
- Press 'Enter' key.
- You get a text message 'DND Canceled on <Extension number>' and confirmation tone.
- Go Idle or you get dial tone after confirmation tone.

For SLT Users

To set DND for an extension user:

- Lift the handset.
- Dial 1072-001. You get feature tone.
- Dial Extension number. You get feature tone.
- Dial 1 for All Calls
- Dial 2 for Internal Calls
- Dial 3 for External Calls
- You get confirmation tone.
- Replace handset.

To select DND Message:

- Lift the handset.
- Dial 1072-001. You get feature tone.
  - Dial Extension number. You get feature tone.
  - Dial 4-1 for 'Do Not Disturb'
  - Dial 4-2 for 'Unavailable'
  - Dial 4-3 for 'In a Meeting'
  - Dial 4-4 for 'In a Conference'
  - Dial 4-5 for 'Try on Mobile'
  - Dial 4-6 for 'On Vacation'
  - Dial 4-7 for 'On Business Trip'
  - Dial 4-8 for 'Out of Office'

    - Dial 4-9 for 'With a Guest'
- Replace handset.

To cancel DND set for an extension user:

- Pick up the handset.
- Dial 1072-001. You get feature tone.
- Dial Extension Number. You get feature tone.
- Dial '0' to cancel DND.
- You get confirmation tone.
- Replace handset.

DND-Override

For EON, Extended IP Phone Users and SLT Users

- Dial an extension number.
- You get routing beeps and a DND Notification message, if programmed (and a DND Text message, if using EON/Extended IP Phone)
- Dial '4', the DND-Override Code, during the message or the routing beeps.
- The called extension will start ringing.
- You will get Ring Back tone.
- If the dialed phone is busy, you will get busy tone.

DSS Call Pick-Up

What's this?

DSS Call Pick-Up allows DKP/Extended IP Phone users to answer calls ringing on other extensions or incoming calls on trunks by pressing the DSS Keys assigned to those extensions/trunks on their phones.

There are two types of DSS Call-Up:

- **DSS Call Pick-Up-Station** - internal or external calls ringing on any extension, can be picked-up by pressing the DSS Key assigned to that extension on your phone.

- **DSS Call Pick-Up-Trunk** - incoming calls on any trunk for any extension can be picked-up by pressing the key assigned to that trunk on your phone.

For T1E1 and BRI Trunks, DSS keys can be assigned for each channel. Similarly for SIP Trunks, DSS keys can be assigned for each Call Appearance.

If you have assigned a DSS key to All the Channels for T1E1/BRI Trunks or All the Call Appearances for SIP Trunks, you will only be able to grab the trunk to make outgoing calls. You will not be able to pick up incoming calls on these trunks.

How it works

For this feature to work, you must:

- enable the desired Call Pick-up in the COS of the extension user.
- assign DSS Keys with LED to the desired extensions/trunks on their DKP/Extended IP Phone.

This is how DSS Call Pick-Up works:

- Extension user A has configured a DSS Keys for extension 2007 and CO trunk 1 on his/her DKP/Extended IP Phone.

- When a call lands on extension 2007 and it rings, the DSS Key assigned to 2007 blinks fast in **Blue** color to indicate that the extension is ringing. A presses the DSS Key to pick-up the call ringing on extension 2007.

  If DSS Call Pick-Up-Station is not enabled in the COS assigned to extension user A, the DSS Key blinks fast in **Red** color to indicate that the extension is ringing. However, A will not be able to pick-up the call ringing on extension 2007.

- Similarly when there is an incoming call on CO trunk 1, the DSS Key assigned to CO trunk 1 blinks fast in **Violet** color to indicate that there is an incoming call on the trunk. A presses the DSS Key to answer the call on the trunk.

  If DSS Call Pick-Up-Trunk is not enabled in the COS assigned to extension user A, the DSS Key blinks fast in **Red** color to indicate that the there is an incoming ringing call. However, A will not be able to pick-up the incoming ringing call on that trunk.

Feature Interactions:

- **Call States:** DSS Call Pick-Up-Station and DSS Call Pick-Up-Trunk are possible only when the calls are in ringing state.

- **Priority**: If multiple calls are ringing on an extension, when you press the DSS Key assigned to that extension, you will be connected to the ringing call with the highest priority. To know more, see “Priority”.

How to configure

To provide this feature to extension users, you must

- enable these features in their Class of Service to be assigned to the users. For instructions, see “Class of Service (COS)” and “Station Basic Feature Template”.

- configure the DSS Keys for the desired extensions and trunks on their DKP/Extended IP Phone. To know more about assigning DSS Keys, see “DSS Keys Programming”.

How to use

For EON & Extended IP Phone Users

To use DSS Call Pick-Up-Station:

- When the DSS key assigned to the station blinks fast in blue color to indicate that the station is ringing, press the DSS Key.
- You are in speech with the calling party.
- You may talk.

To use DSS Call Pick-Up-Trunk:

- When the DSS key assigned to the trunk blinks fast in violet color to indicate that there is an incoming call on the trunk, press the DSS Key.
- You are in speech with the calling party.
- You may talk.

Dynamic Lock

What's this?

Dynamic Lock allows extension users to change the Toll Control Levels (Calling Permissions) of their extensions on their own by dialing a code.

The System Administrator/Operator can also change the Toll Control Levels of extensions using Dynamic Lock.

With this feature, extension users can prevent misuse of outgoing call facility from their extensions, especially in their absence.

Dynamic Lock also forms the basis of 'Call Privilege', which is feature of the Hotel Application of SARVAM UCS. Refer the *SARVAM UCS Hospitality System Manual* to know more.

There are four types of Toll Control Levels, starting from Level 0 to Level 3 that can be set for extension phones.

For each Toll Control Level from 0 to 3, a 'Call Privilege'[301] is to be assigned and corresponding numbers strings to be allowed and number strings to be denied for each Call Privilege are to be programmed.

- **Toll Control - Level 0** is Time Zone based, wherein the Call Privilege Type must be defined for each Time Zone, that is, Working Hours, Break Hours and Non-Working Hours. For instance, you may define 'All Calls' as Call Privilege for Working Hours, 'Local Calls' as Call Privilege for Break Hours and 'No Calls' as Call Privilege for 'Non-Working' Hours.

  By default, Call Privilege 'No Calls' is selected for all three Time Zones.

- **Toll Control - Level 1** is not based on Time Zones. By default, the Call Privilege Type for this level is 'No Calls'.

- **Toll Control - Level 2** is not based on Time Zones. By default, the Call Privilege type set for this level is 'No Calls'.

- **Toll Control - Level 3** is not based on Time Zones. By default, Call Privilege 'No Calls' is selected for this level.

  The Call Privilege for each of the above Toll Control Levels can be redefined according to user requirements. For example, Toll Control Level 3 can be programmed for allowing all types of calls by selecting 'All Calls' as Call Privilege Type and Level 0 can be programmed to allow only Local Calls, by programming the strings of 'Local Numbers'.

  Refer the feature description for “Toll Control” to know more.

  Extension users who are allowed the Dynamic Lock feature in their Class of Service, can set the Toll Control Level in two ways:

  - **Manually:** the extension user changes the Toll Control Level of the extension whenever s/he wants by dialing the feature access code.

301. The Call Privilege types are: No Calls, Local Calls, Regional Calls, National Calls, International Calls, All Calls and Limited Calls.

For example, an extension user having Toll Control Level 2 (National calls) can restrict long distance dialing on his/her extension by setting the Toll Control Level to 1 (Local calls) before leaving the workplace. On return, the user can restore the previous Toll Control Level, by setting it back to Level 2.

Thus the extension user sets Dynamic Lock s/he manually selects the desired Toll Control Level for his/her extension and restores the original Toll Control Level assigned to the extension.

- **Automatically:** the extension user changes the Toll Control Level of the extension using the Dynamic Lock Timer. The user sets the Timer to the desired number of minutes. On the expiry of this Timer, the system restores the original Toll Control Level assigned to the extension.

  For example, an organization has defined Toll Control Level 0 as Local Calls, and Level 3 as All Calls. An extension user of this organization is assigned Level 0. When this extension user wants to make international calls, he sets the Dynamic Lock Timer and selects Toll Control Level 3. At the end of the timer, Level 3 gets locked and Toll Control Level 0 is reapplied on the extension phone.

- *The changing of Toll Control level requires the user to dial the 4-digit User Password. The system will not accept the default User Password (1111). The extension user must first change the default User Password.*
- *The Dynamic Lock Timer can be set to a maximum of 99 minutes.*
- *The Dynamic Lock Timer must be set to '00' when using Manual Dynamic Lock.*
- *Dynamic Lock when set/canceled from the SA mode, will not depend on the assigned CoS.*

How it works

The Pre-requisites

- The Toll Control Levels 0 to 3 are programmed in the Station Basic Feature Template applied on the extension.
- Dynamic Lock is allowed to the extension in its Class of Service.

The Process

For Dynamic Lock - Manual

- The user of extension A sets the Dynamic Lock manually by entering the User Password and selecting the desired Toll Control Level.

  OR

- The Operator sets Dynamic Lock manually for an extension by entering the extension number and selecting the Toll Control Level.

For Dynamic Lock - Automatic

- The user of extension A sets the Dynamic Lock by entering the User Password, setting the Dynamic Lock Timer, and selecting the desired Toll Control Level.
  OR

- The Operator sets Dynamic Lock for an extension by entering the extension number, setting the Dynamic Lock Timer, and selecting the Toll Control Level.
- Now, whenever a call is made from extension A, the system checks for Toll Control Level.
- The system then checks the associated Lists of allowed and denied numbers.
    - If the Toll Control Level is 0, then Toll control is time zone based, that is, working hours, break hours and non-working hours. The outgoing call is allowed/denied as per the Call Privilege and the corresponding Allowed and Denied Number List programmed for that time of the day by the System Engineer.
    - If the Toll Control Level is 1, 2, 3 the outgoing call is allowed/denied as per the Call Privilege and the corresponding number list programmed for each level.
- If Dynamic Lock - Automatic has been set by user/Operator, the system waits for the duration of the Dynamic Lock Timer set for the extension. At the end of each outgoing call made during the period of this Timer, the system will restart the Timer again. The system will change the Toll Control back to the previous Level when no outgoing call is made till the expiry of this Timer.
- If Dynamic Lock - Automatic has been set by user/Operator, and an internal call is made during the period of the Dynamic Lock Timer, the system will check for the 'Decrement Dynamic Lock Timer Internal Calls' feature in the Class of Service of allowed to the extension. If this feature is enabled, the system will start the decrement of the Dynamic Lock Timer. The system will change the Toll Control back to the previous level on the expiry of this Timer. However, if the 'Decrement Dynamic Lock Timer' feature is disabled in the Class of Service, the system will reset the Toll Control as described in the previous step.
- If Dynamic Lock - Manual has been set, the extension user/Operator must set the Toll Control Level back to the previous Level.

Feature Interactions

- **Redial and Auto Redial:** The system will check for Toll Control Level when an extension on which Dynamic Lock is set, attempts Redial. The system will not check the same for Auto Redial.
- **Emergency Number Dialing:** All extensions will be able to dial Emergency numbers always, regardless of the Toll Control set on them.

  SARVAM UCS provides for separate programming of Emergency Numbers, which remain unaffected by Dynamic Lock set on the phones. Refer the topic “Emergency Dialing” to know more about this feature.

How to configure

For this feature to work, it must be enabled in the Class of Service of the extensions; Toll Control Level must be programmed in the Station Basic Feature Template of the extensions. The user must change the default User Password.

Dynamic Lock in Class of Service

In the default Station Basic Feature Template 01 assigned to all extensions of the SARVAM UCS, the default COS group 01 has Dynamic Lock enabled. So, all extensions can set Dynamic Lock.
In the default COS group 01, 'Decrement Dynamic Lock Timer for Internal Calls' is disabled.

Retain the default template, if you want to allow this feature to all extensions and keep the Decrement Timer disabled.

If you want to deny Dynamic Lock to all extension, simply disable this feature in the default COS group 01 of Station Basic Feature Template 01.

If you want to allow Dynamic Lock and/or the Decrement Dynamic Lock Timer for Internal Calls only to selected extensions, then follow these steps:

a. Define a new CoS group with Dynamic Lock and the Decrement Dynamic Lock Timer for Internal Calls enabled.

b. Prepare a Station Basic Template with this CoS group applicable in all the time zones.

c. Assign this newly prepared Station Basic Feature Template to the extension on which 'Dynamic Lock' and 'Decrement Dynamic Lock Timer' for Internal Call is to be allowed.

Similarly, if you want to deny Dynamic Lock/Decrement Dynamic Lock Timer to selected extension, prepare a new Station Basic Feature Template with this feature disabled in the CoS group. Assign this feature to those extensions which are to be denied this feature.

Refer the topics "Class of Service (COS)" and "Station Basic Feature Template" for programming instructions.

Toll Control Levels in Station Basic Feature Template

Program the Toll Control Levels in the Station Basic Feature Template that are assigned to the extensions which are to be allowed the Dynamic Lock feature. Refer the topic "Toll Control" for instructions.

How to use

Dynamic Lock, Manual and Automatic, can be set by extension users as well as from their own extensions or from the SA mode by the Operator.

The extension user/Operator must first set the Dynamic Lock Timer and then change the Dynamic Lock Level.

To set Dynamic Lock-Manual, the extension user/Operation must set the Dynamic Lock Timer to ***00.***

Recall that

- *When the Dynamic Lock-Manual is set (Timer set to 00), the extension user/Operator must dial the feature access code to restore the previous Toll Control Level.*
- *When Dynamic Lock-Automatic is set (Timer set to desired number of minutes), the system will restore the previous Toll Control Level at the end of the Timer.*
- *The extension user must change the default User Password to be able to set the Dynamic Lock on his/her extension. Refer the topic "User Password" for instructions on changing the password.*

Changing Dynamic Lock by Extension Users

For EON and Extended IP Phone Users

To set Dynamic Lock-Manual:

- Press DSS Key assigned to Dynamic Lock.

  OR

- Dial 142.

  OR

- Enter Phone Menu by pressing 'Enter' key.
- You get the prompt <Enter User Password>
- Enter your User Password.
- You get the prompt: <Lock Timer = XX Minutes>.
- Enter 00
- You get the Text message <Lock Timer = 00 Minutes> and confirmation tone.

To set Dynamic Lock-Automatic:

- Press DSS Key assigned to Dynamic Lock.

  OR

- Dial 142.

  OR

- Enter Phone Menu by pressing 'Enter' key.
- You get the prompt <Enter User Password>
- Enter your User Password.
- You get the prompt: <Lock Timer = XX Minutes>.
- Enter the desired number of minutes (max. 99 minutes).
- You get the Text message <Lock Timer = xx Minutes> and confirmation tone.

To set Dynamic Lock Level:

- Press DSS Key assigned to Dynamic Lock.

  OR

- Dial 141.

  OR

- Enter Phone Menu by pressing 'Enter' key.
- You get the prompt: <Enter User Password>.
- Enter your User Password.
- Select the desired Toll Control Level:

- Toll Control Level 0
- Toll Control Level 1
- Toll Control Level 2
- Toll Control Level 3

- Press Enter key.
- You get the text message 'OK!' and a confirmation tone.

If you are still working with the default User Password, the system will prompt you to 'Change User Password' when you attempt to set Dynamic Lock. Change your User Password first, before you use this feature.

For SLT Users

To set Dynamic Lock-Manual:

- Lift the handset.
- Dial 142-User Password-00.
- You get Confirmation tone.
- Replace handset.

To set Dynamic Lock-Automatic:

- Lift the handset.
- Dial 142-User Password-Minutes (max. 99 minutes)
- You get Confirmation Tone.
- Replace handset.

To set Dynamic Lock Level:

- Lift the handset.
- Dial 141-User Password-Toll Control Level.
- Dial 0 Level 0.
- Dial 1 for Level 1.
- Dial 2 for Level 2.
- Dial 3 Level 3.
- You get Confirmation tone.
- Replace handset.

Changing Toll Control Level for an Extension (SA mode)

For EON and Extended IP Phone Users

To set Dynamic Lock-Manual:

- Press DSS key assigned to Dynamic Lock.

  OR

- Dial 1072-002[302].

302. *Ensure that the feature 'SA Extension' is enabled in the "Class of Service (COS)" allowed to the extension from which this code is being dialed.*

OR

- Enter Phone Menu.
- You get a text message 'Enter Room/Phone Number'.
- Enter the extension number for which you want to set Dynamic Lock.
- Select the option 'Change Lock Timer'.
- You get the following prompt: 'XX Minutes'.
- Enter 00
- You get the text message <Lock Timer = 00 Minutes> and confirmation tone.

To set Dynamic Lock-Automatic:

- Press DSS key assigned to Dynamic Lock.

  OR

- Dial 1072-002[303].

  OR

- Enter Phone Menu.
- You get a text message 'Enter Room/Phone Number'.
- Enter the extension number for which you want to set Dynamic Lock.
- Select the option 'Change Lock Timer'.
- You get the following prompt: 'XX Minutes'.
- Enter the desired number of minutes (max. 99)
- You get the text message <Lock Timer = xx Minutes> and confirmation tone.

To set Dynamic Lock Level:

- Press DSS key assigned to Dynamic Lock.

  OR

- Dial 1072-002[304].

  OR

- Enter Phone Menu.
- You get a text message 'Enter Room/Phone Number'.
- Enter the extension number for which you want to set Dynamic Lock.
- Select the option 'Change Toll Ctrl. Level'
- Press Enter key.
- Scroll to select the desired Toll Control Level:
  - Toll Control Level 0
  - Toll Control Level 1
  - Toll Control Level 2
  - Toll Control Level 3

303. *Ensure that the feature 'SA Extension' is enabled in the "Class of Service (COS)" allowed to the extension from which this code is being dialed.*
304. *Ensure that the feature 'SA Extension' is enabled in the "Class of Service (COS)" allowed to the extension from which this code is being dialed.*

- Press 'Enter' key.
- You get the confirmatory text message 'OK!' and confirmation tone.

For SLT Users

To set Dynamic Lock-Manual:

- Pick up the handset.
- Dial 1072-002, you get feature tone.
- Dial Extension Number, you get feature tone.
- Dial 2, the code for 'Change Lock Timer'.
- Dial 00.
- You get confirmation tone.
- Replace the Handset or you get dial tone after 3 seconds.

To set Dynamic Lock-Automatic:

- Pick up the handset.
- Dial 1072-002, you get feature tone.
- Dial Extension Number, you get feature tone.
- Dial 2, the code for 'Change Lock Timer'.
- Dial the number of Minutes you want to set the Timer.
- You get confirmation tone.
- Replace the Handset or you get dial tone after 3 seconds.

To change Dynamic Lock Level:

- Pick up the handset.
- Dial 1072-002, you get feature tone.
- Dial Extension Number, you get feature tone.
- Dial 1, the code for 'Change Toll Control Level'.
- Dial 0 for Level 0
- Dial 1 for Level 1
- Dial 2 for Level 2
- Dial 3 for Level 3
- You get confirmation tone.
- Replace the Handset or you get dial tone after 3 seconds.

E1/T1 Maintenance

What's this?

The E1 Maintenance consists of Error Counts (Performance Statistics), Alarms and Loop Back Tests. This is as per standards like G.704, G.706 and G.732. G.775 is also considered for detection of defect conditions like Loss of Signal (LOS), Loss of Frame (LOF), Alarm Indication Signal (AIS), etc.

To elaborate, the Digital line can have transmission errors. All the errors will not generate an Alarm. Few severe errors generate Alarms. However, all the errors are logged in the System Fault Log.

The SNIIC (Subscriber Network Interface Integrated Circuit), is used to interface E1 line to SARVAM UCS. It supports error counters listed in the table given below. See "Error Counts (Performance Statistics)".

Each error detected by the T1E1 port is sent to the CPU Card in the form of an event. The system counts these errors and prepares a statistical record if the condition matches. For example, Severely Errored Seconds Count is incremented when one OOF (Out of Frame) event detected by the system or more than 320 framing errors are detected by the system. This statistical record is updated and maintained by the system.

Facility-Associated and Non-Facility Associated Signaling:

Signaling on ISDN PRI trunks consists of messages transported over the D-Channel, which is channel 24 on T1 interface or channel 16 on E1 interface. This signaling can be provided by two methods:

- **FAS:** The D channel can provide signaling for the other B channels on the same interface. This is called 'Facility Associated Signaling' (FAS).

- **NFAS:** The D channel can provide signaling for the other B channels on more than one interface. This is called 'No facility Associated Signaling' (NFAS). The signaling arrangements, the capability is supported to designate a D channel on one interface to be a backup to a D channel on another interface in case of failure. This is called D channel backup.

Error Counts (Performance Statistics)

Error Counters supported by SNIIC in E1 Mode.

Errored Frame Alignment Signal	This counter is incremented on receipt of each errored FAS.
E-bit	This counter is incremented when either E1 or E2 bit is set in the transmit frame.
CRC-4 Error	This counter is incremented when the received frame has CRC-4 errors.
Line Code violation Error	This counter is incremented when a line code violation error occurs.
Excessive Zeros Error	This counter is incremented when excessive zeros are received or Line code violation error occurs.
Positive Slip Buffer	This counter is incremented every time a positive slip occurs.
Negative Slip Buffer	This counter is incremented every time a negative slip occurs.

Following parameters form the statistical record. This can be generated in the form of a report as shown below:

Performance Parameter	Seconds/Count
Error Seconds	000 to 255
Bursty Errored Seconds	000 to 255
Severely Errored Seconds	000 to 255
Severely Errored Framing Seconds	000 to 255
Unavailable Seconds	000 to 255
Positive Slip Seconds	000 to 255
Negative Slip Seconds	000 to 255
Loss of Frame Count	000 to 255
Line Errored Seconds	00000 to 65535
Excessive Zeroes Error Count	000 to 255
CRC-4 Error Count	00000 to 65535

Out Of Frame (OOF) - Out of Frame is the occurrence of a particular density of framing error events. OOF is declared when three consecutive frame alignment signals have been received with an error. OOF ends when;

- In frame N, the FAS is correct.
- In frame N+1, the FAS is absent.
- In frame N+2, the FAS is present and is correct.

Errored Seconds - It is defined as a second with one of the following:

- One FAS Errors.
- One or more OOF defects.
- One or more Slip events.
- A detected AIS defect.

Bursty Errored Seconds

Without CRC - It is a second with:

- More than one but less than 320 Errored FAS.
- No OOF.
- No SES.
- No AIS.

With CRC - It is a second with:

- More than one but less than 320 CRC errors.
- No OOF.
- No SES.
- No AIS.

Severely Errored Seconds

With CRC - It is a second with one of the following:

- 832 or more CRC error events.
- one or more OOF defects.

Without CRC - It is a second with one of the following:

- 2048 or framing errors.
- Slips are not included. This is not incremented during Unavailable seconds.

Severely Errored Framing Seconds (SEFS) - It is a second with either one or more OOF defects or a detected AIS defect.

Unavailable Seconds - It is defined as a second in which E1 service is unavailable. An unavailable state is declared at the onset of 10 consecutive severely errored seconds and is cleared on onset of 10 consecutive seconds with no severely errored seconds.

Positive Slip Seconds - It is defined as a second in which a frame is repeated to account for frequency drift between ET2 and the network.

Negative Slip Seconds - It is defined as a second in which a frame is deleted to account for frequency drift between ET2 and the network.

Loss of frame count - Loss of Frame is declared after 2.5 seconds of continuous loss of signal or OOF. LOF is cleared after 10 seconds of continuous no loss of signal or OOF.

Line Errored Seconds - It is a second in which one or more than one line code violation error occurs.

Excessive Zeroes Error Count - This counter is incremented when excessive zeroes are received on the line or when line code violation error occurs.

CRC-4 Error Count - This counter is incremented when a CRC-4 Error is detected. This is applicable for E1 ports only.

CRC-6 Error Count - This counter is incremented when a CRC-6 Error is detected. This is applicable for T1 ports only.

Parameters of Facility Data Link (FDL)

This parameter is applicable for T1 Ports only.

FDL is used for communicating general maintenance information or for transmitting user defined information within the T1 link. General maintenance information is in the form of Performance Message Report which is generated by the T1E1PRI Card and depending upon the T1 FDL Protocol, the Performance Message Report is sent every second or on request.

How to configure

The commands explained below should be referred as:
To program a single port: ***XXXX-1***
To program a range of ports: ***XXXX-2***
To program all the ports: ***XXXX-****

T1 FDL

T1 FDL can be enabled/disabled. This parameter is applicable only if Framing = ESF. If the Network (Public or Private) to which the SARVAM UCS is connected does not support FDL then T1 FDL will be disabled.

Use following command to enable/disable T1 FDL on a T1E1PRI port:
6164-1-T1E1PRI-T1 FDL
6164-2-T1E1PRI-T1E1PRI-T1 FDL
6164-*-T1 FDL
Where,
T1E1PRI is from 01 to 08.

T1 FDL	Meaning
0	Disable
1	Enable

By default, the T1 FDL is disabled.

T1 FDL Protocol

SARVAM UCS will support both the protocols of reporting the performance monitoring. This parameter is applicable only if T1 FDL is enabled and Framing = ESF. This parameter will match the protocol expected by the other end of the link.

Use following command to program the T1 FDL Protocol for a T1E1PRI port:
6165-1-T1E1PRI-T1 FDL Protocol
6165-2-T1E1PRI-T1E1PRI-T1 FDL Protocol
6165-*-T1 FDL Protocol
Where,
T1E1PRI is from 01 to 08.

T1 FDL Protocol	Meaning
0	Disable
1	AT&T 54016
2	ANSI T1.403

By default, the T1 FDL Protocol is ANSI T1.403.

ANSI T1.403

As per this standard, the receiving equipment transmits a performance report message (PRM) each second over the FDL. This PRM is not sent to any specific remote location, but is broadcast so that any PRM receiving device on the T1 line can intercept the message. The PRM contains error information pertaining to only the previous 4 seconds.

It is the responsibility of the PRM receiver to accumulate the information and store it for 24 hours or the time desired. This method allows performance monitoring points at different locations along the T1 network so that error localization is determined.

AT&T 54016

As per this standard, the receiving equipment collects the data but does not transmit it on its own based on time as done by ANSI. Instead, the transmitting end sends a request to the receiving end to transmit the performance data.

The Performance Message Report Format is as per ANSI T1.403.

ALARMS

Alarms are indicated on the LEDs of the T1E1PRI Card. The T1E1PRI Card has four LEDs, namely L1 to L4. L1 and L2 indicate alarms for T1E1 PRI Port1 whereas L3 and L4 indicate alarms for T1E1 PRI Port2. During normal conditions, the LED blinks green (1 sec ON, 1 sec. OFF).

RED Alarm

- This alarm is generated if Loss of Signal persists for 2.5 seconds.
- Received signal is more than 20 dB or 40 dB below nominal for at least 1ms.
- 32 consecutive zeroes are received.
- Loss of frame alignment occurs.
- This is indicated by flashing the LED Red (500ms ON, 500ms OFF).
- The system logs this event in the System Fault Log as RED Alarm <Slot No.> <Port No.> at HH:MM:SS.
- When RED Alarm is declared, Yellow Alarm is sent to the far end within 12ms of detection of LOS.
- RED Alarm is cleared when the signal is acquired back and persists for 10 seconds.
- The LED is turned OFF. The system logs this event in the System Fault Log as RED Alarm Cleared <Slot No.> <Port No.> at HH:MM:SS.

YELLOW Alarm

- This Alarm is also known as Remote Alarm Indication or Distant Alarm.
- This Alarm is generated when Yellow Alarm is sent by the far end (Yellow Alarm is sent by the far end to indicate that it has lost the incoming signal).
- Yellow Alarm is declared when the signal corresponding to Yellow Alarm persists for 0.5 seconds.
- This is indicated by flashing the LED Orange (500ms ON, 500ms OFF).
- The system logs this event in the System Fault Log as YELLOW Alarm <Slot No.> <Port No.> at HH:MM:SS.
- This alarm is cleared when No Yellow Alarm signal persists for 0.5 seconds.
- The LED is turned OFF. The system logs this event in the System Fault Log as YELLOW Alarm Cleared <Slot No.> <Port No.> at HH:MM:SS.
- Yellow Alarm is declared if Bit 3 of the receive NFAS is 1 on two consecutive occasions.
- Yellow Alarm is cleared if Bit 3 of the receive NFAS is 0.

If equipment is connected in downstream (Drop and insert mode, that is, NT mode) then on receipt of Yellow Alarm, a Blue Alarm will be sent on the port, which is configured in NT mode.

BLUE Alarm

- It is also known as Alarm Indication Signal (AIS).
- This alarm indicates that the upstream equipment connected to the SARVAM UCS has lost receiving its incoming signal.
- This alarm is generated when AIS persists for 2.5 seconds.
- This is indicated by flashing the LED Red (1 sec. ON, 1 sec. OFF).
- The system logs this event in the System Fault Log as BLUE Alarm <Slot No.> <Port No.> at HH:MM:SS.
- This alarm is cleared when clearance of AIS is detected for continuous 10 seconds.
- The LED is turned OFF. The system logs this event in the System Fault Log as BLUE Alarm Cleared <Slot No.> <Port No.> at HH:MM:SS.
- Blue Alarm (AIS) is declared if unframed all ones signal is received as a string of 512 bits containing fewer than three zero bits.
- When BLUE Alarm is declared, Yellow Alarm is sent to the downstream equipment connected to the SARVAM UCS.

Port1 Status

LED	Card Status	Port Status
L1 Green Flashing @1 Sec.	Card Heart Bit	Port 1 Layer is established
L1 Red Flashing @1 Sec.	Card Heart Bit	Port 1 Layer is not established LOS-Red alarm detected.
L1 Red Flashing @500 msec.	Card Heart Bit	Port 1 Layer is not established LBFA -Loss of Basic Frame Alignment.
L1 Red Flashing @100 msec.	Card Heart Bit	Port 1 Layer is not established- LMFA loss of multi-frame alignment.
L1 Yellow Flashing @1 Sec.	Card Heart Bit	Port 1 Layer is established -RAI (Yellow Alarm) is detected.
L2 Green		Port 1 achieves CRC4 synchronization.
L2 Yellow		Port1 does not achieve CRC4 synchronization.
L2 Green Flashing @1 Sec.		Port 1 achieves CRC4 synchronization-AIS (Blue Alarm) is detected.
L2 Yellow flashing @1 Sec.		Port 1 doesn’t achieves CRC4 synchronization-AIS (Blue Alarm) is detected.

Port2 Status

LED	Card Status	Port Status
L2 Green Steady		Port2 Layer 1 is established.
L3 Red Flashing @1 Sec.		Port 2 Layer is not established-LOS-Red alarm detected.
L3 Red Flashing @500 msec.		Port 2 Layer is not established-LBFA-Loss of Multi Frame Alignment.
L3 Red Flashing @100 msec.		Port 2 Layer is not established-LMFA Loss of Multi Frame Alignment.
L3 Yellow Flashing @1 Sec.		Port 2 Layer is established-RAI (Yellow Alarm) is detected.
L4 Green		Port 2 achieves CRC4 synchronization-AIS (Blue alarm) is detected.

LED	Card Status	Port Status
L4 Yellow		Port2 doesn’t achieves CRC4 synchronization-AIS (Blue Alarm) is detected.

Refer chapter “Loop Back Tests” for more details.

E&M Connectivity

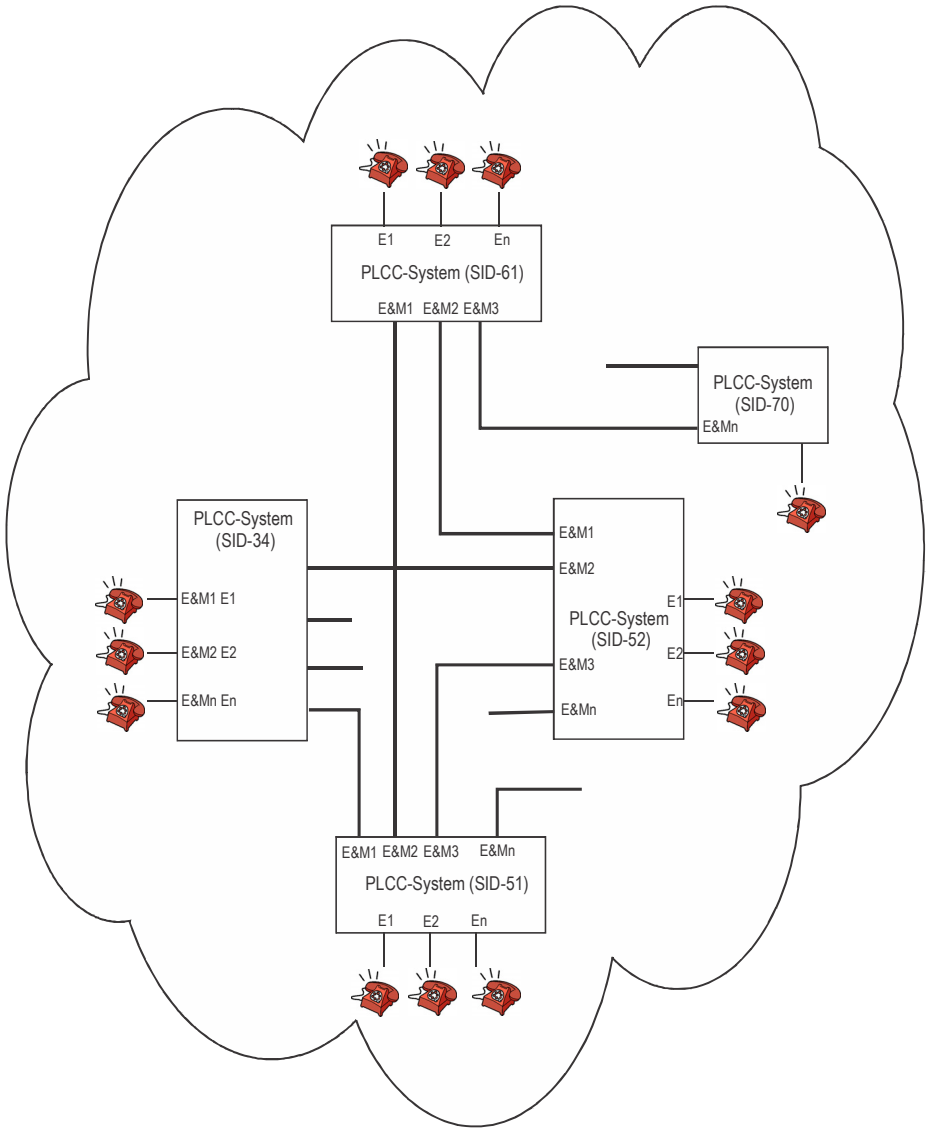
What’s this?

E&M connectivity feature of SARVAM UCS offers seamless connectivity in PLCC network and also in between various communication products like PBX, Router, Lease Line. E&M interface is widely used interface to connect such diverse equipment. For example, in a PLCC network, number of PLCC System needs to be connected. As shown in the figure 1 of PLCC network, number of Systems are connected with each other through E&M tie lines.

Say, an existing System capacity needs to be expanded beyond the configuration limit of a System. Installing one more System and connecting both the Systems through E&M interfaces can get us the desired expansion.

Four different applications of E&M connectivity are shown below:

1. E&M Connectivity on PLCC-Network



How it works in PLCC Network

E&M interface is achieved using the E&M Card. An E&M port of the System has dual personality, both of a subscriber and a trunk. An E&M port works like a subscriber interface for any incoming call to it and works like a trunk interface when any subscriber makes an outgoing call through it. However, please note that a trunk line cannot be connected to an E&M port. Also a subscriber cannot be connected to an E&M port.

Incoming call on an E&M port in PLCC network

When any subscriber of PLCC System with SID-51 grabs an E&M2 port, s/he gets dial tone of SID-61. Now he can call any subscriber of SID-61 and use features like priority, conference, just like a normal subscriber would do. However, this E&M port needs to be programmed for the same.

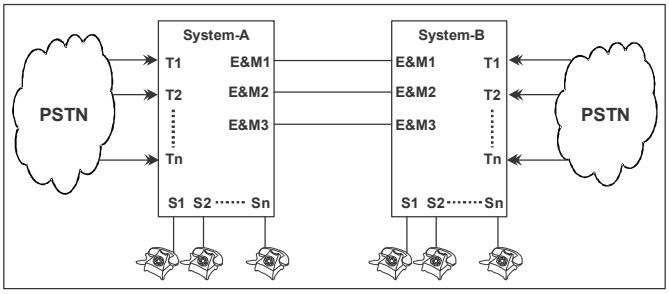
Outgoing call using E&M port in PLCC network

When any subscriber of PLCC System-34 dials network access code (as per programming) and dials SID say 52, an E&M1 port is grabbed (as it is programmed for the same). Now he can call any subscriber of SID-52 and establish speech. However, this E&M port needs to be programmed as any other normal trunk like enabling the port, programming the dial type.

All E&M ports can be put together in one single group depending on the requirement. A separate access code can be assigned to it. This makes the operations easy. Now the subscribers have at least two different access codes for making outgoing calls using trunk lines and other for making outgoing calls using E&M tie lines.

2. System EXPANSION (2-SYSTEMS)

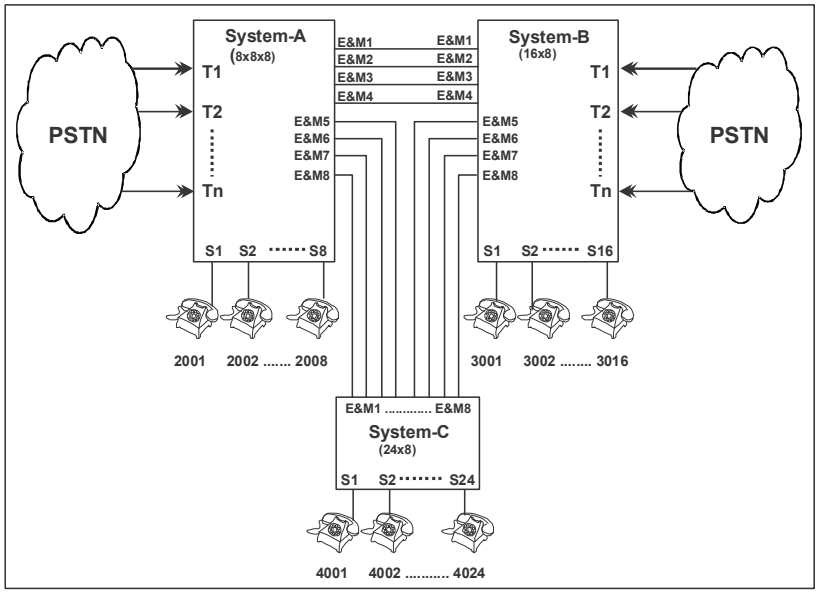
Two Systems are connected using E&M interface.



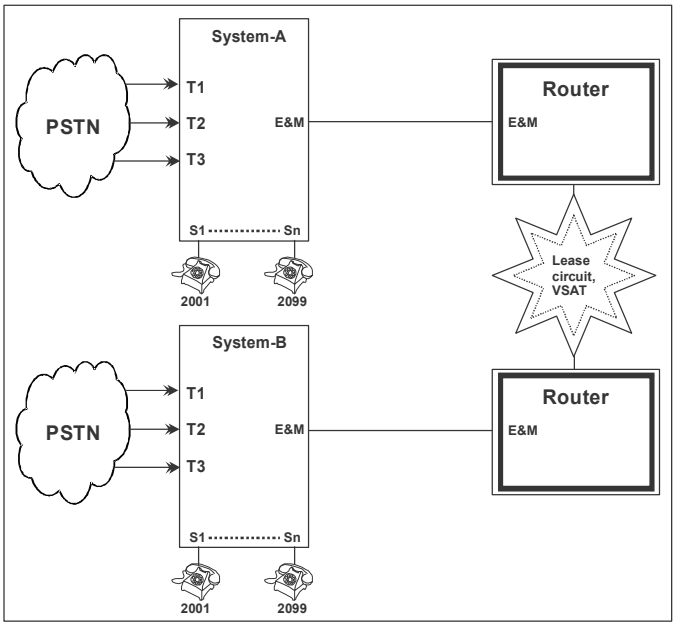
In the above figure,

- T1 to Tn are trunk lines from the local central office (CO).
- S1 to Sn are stations.
- E&M1 to E&Mn are E&M lines between two Systems.

3. Systems are connected through E&M connectivity



4. Two Systems located far from each other connected to each other using E&M connectivity.



How it works

E&M interface is achieved using the E&M Card. An E&M port has dual personality: both of a station and a trunk. An E&M port works like a station interface for any incoming call to it and works like a trunk interface when any station makes an outgoing call through it. However, please note that a trunk line cannot be connected to an E&M port. Also a SLT or DKP cannot be connected to an E&M port.

Incoming call on an E&M port

When any station of System B grabs an E&M port, he gets dial tone of System A. This port of System A behaves as a SLT port. Now he can call another station, grab a trunk, or use a feature like raid, just like a normal station would do. However, this E&M port needs to be programmed as any other normal station. Features like Class of Service, Toll Control, will apply to this port, if programmed.

Outgoing call using E&M port

When any station from System A makes an outgoing call through an E&M port, s/he dials trunk access code. The E&M port behaves as trunk interface and gives the station, the dial tone of System B. Now s/he can dial any station of System B or grab a trunk line of System B to make a call or use any feature depending on the station programming done by the SE of System B. Hence the E&M port of System A should be programmed as any other normal trunk by enabling the port, programming the dial type, etc.

All E&M ports can be put together in one single group or few groups depending on the requirement. A separate access code can be assigned to it. This makes the operations easy. In such case the stations can have at least two different access codes for making outgoing calls, one for making outgoing calls using trunk lines and other for making outgoing calls using E&M lines. Generally, E&M connectivity is used to expand the System capacity or connect two or more remotely located Systems. This forms a network of Systems. The requirement is that, the so formed network should work as one Group. This is commonly known as Closed User Group. Please refer “Closed User Group (CUG)” and “Closed User Group-With Exchange ID” for more details.

How to configure

Please refer “Configuring E&M Lines”, “Station Basic Feature Template”, “Station Advanced Feature Template”, “E&M Feature Template” for more details.

Emergency Calls (911) - Reporting to PSAP

What's this?

Make sure:
- *you have selected the **Region** as USA or Canada.*
- *in the Emergency Table for the Number 911, you have assigned the outgoing trunk bundle group with only T1 PRI lines. For instructions, see "Configuring Emergency Number Dialing".*

If you are using the system in the Hospitality mode, to dial the Emergency Number 911, you must purchase the E911 license. For details, refer "License Management".

The SARVAM UCS allows you to assign Customer Emergency Services Identification (CESID) number to each extension user. When the user dials the Emergency number 911, the system sends the caller's CESID number to the network and the call is routed to the Public Safety Answering Point (PSAP).

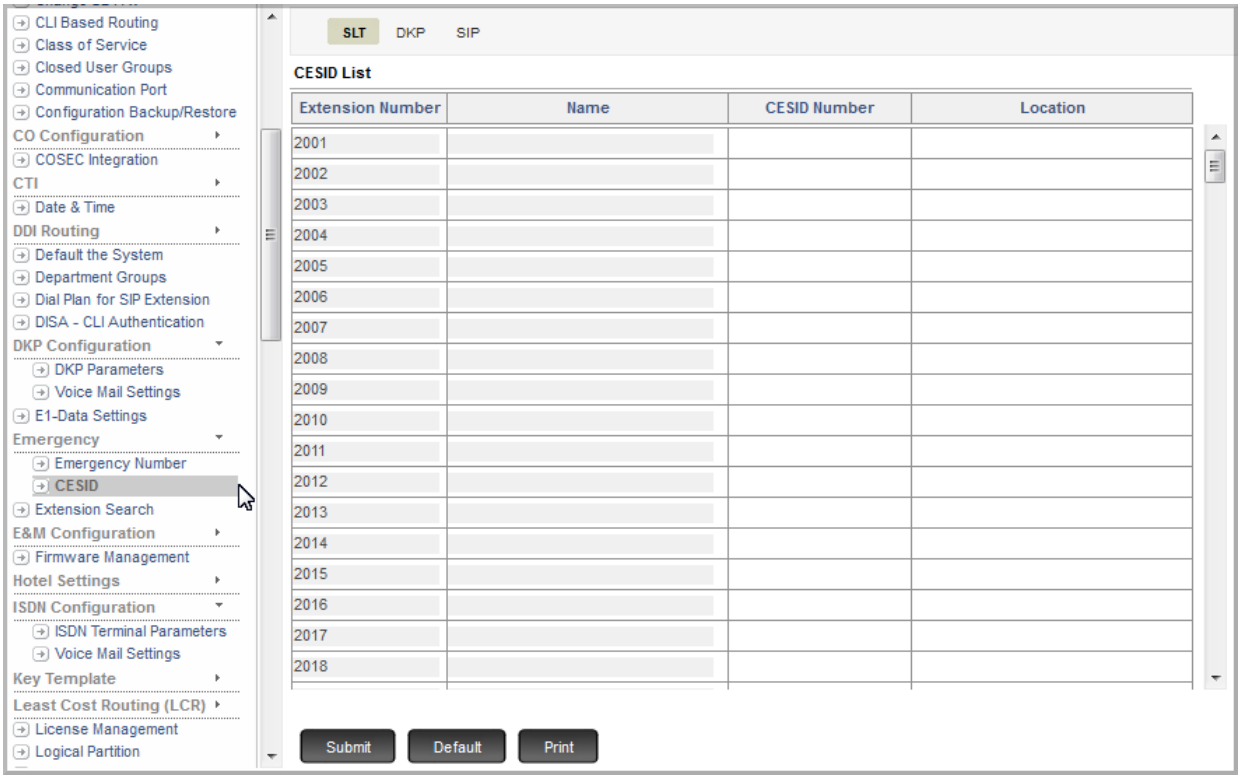
The PSAP stores information, such as name, location, and CESID number of the users in an Automatic Line Information (ALI) database. The PSAP uses the CESID number that you have assigned to an extension to locate the user's information in the ALI database. Hence, the PSAP can provide user's the desired service without delay.

The CESID numbers are provided by the Service Provider. In case these are not provided, you can assign DDI numbers to extension users. See "Direct Dialing-In (DDI)" to know more.

Configuring CESID List using Jeeves

- Log in as System Engineer.
- After you have selected the **Region** and configured the **Emergency Number** table.
- Under **Configuration**, click **Emergency**.

- Click **CESID** to open the page.



- Click the desired tab —**SLT**, **DKP**, **SIP**.
- The list of **Extension Number** and Extension **Name** of the users appear as configured by you.
- For each Extension Number assign the **CESID Number** and enter his/her **Location**.

  The CESID Number can a maximum of upto 12 digits and the Location can be a maximum of upto 18 alphanumeric characters.

- Click **Submit**.

Emergency Conference

What's this?

Emergency Conference enables you to establish a Conference between a pre-defined group of extensions using a feature access code.

This feature can be used to call and consult with a group of people in emergency situations.

The number of parties that can be included in an Emergency Conference group depends on the Multiparty Conference capacity of the SARVAM UCS. For details, see "Conference-Multiparty".

How it works

For this feature to work, you must do the following:

- First decide the key persons in the organization who should be parties to the Emergency Conference.
- Form a Department Group with the extensions of these key persons as members. A single Department Group can have up to 32 extensions. For more information on forming Department Groups, see the topic "Department Call".

For example, you have formed a Department Group for Emergency Conference, with the extensions A to G as members. The Access Code assigned to the Department Group is 3901. H is the initiator of the conference as Emergency Conference is enabled in the Class of Service assigned to his/her extension.

Another Emergency Conference is formed with Department Group 3902, having extensions J to P as members and J as the initiator of the Conference as Emergency Conference is enabled in the Class of Service assigned to his/her extension.

Now, extension H wants to initiate an Emergency Conference.

This is how the feature will work:

Initiating an Emergency Conference

- H dials the feature access code for Emergency Conference, followed by access code of the Department Group (3901).
- All extensions in the Department Group (extensions A to G) which are free will start ringing. The system will play *Emergency Conference* ring (default: Triple Ring) on the Extensions. Extensions that are busy will not be included in the call.

  If there are DKP/Extended IP Phone extensions in the group, and these phones have a Call Appearance free, the system will ring these extensions on the free Call Appearance, but will not wait for the extensions to become free.

When the Emergency Conference is initiated, the system will ring on the extensions that are not occupied in any other single multi-party or 3-party conference.

For more information regarding the number of single multi-party or 3-party conferences supported, refer "Technical Specifications - SARVAM UCS".

- Extension A goes Off-Hook to answer the call first. A gets connected to the initiator of the conference, extension H.
- Two-way speech is established with extension A and H. All other extensions continue to ring.
- When another extension, B goes Off-Hook to answer the call, A and H get a beep, and three-way speech is established between A, B, H.
- Thus, whenever a new member joins the conference, all other extensions already in conference will get a beep, if the flag *Play Beep when Conference/Dial-In Conference Starts* is enabled in the "System Parameters".
- If the conference initiator, extension H, goes idle, all other extensions in the conference will still be in conversation.

Merging Emergency Conferences

- Only the initiator of the conference, extension H, can merge the conference using the phone menu.
  - While in Conference, H presses the 'Conference' Key.
  - From the Multiparty Conference Menu, H selects Merge Conference to merge with another ongoing Emergency Conference.
  - H selects 3902 from the list of ongoing Emergency Conferences. Both the Emergency Conferences (3901 and 3902) are merged.

Cancelling an Emergency Conference

- Only the initiator of the conference, extension H, can cancel the conference. The initiator of the conference can cancel the conference at two stages:
  - When speech is established with one or more member extensions of the Emergency Conference department group.

    Or
  - During Ring Back Tone, as the system rings on the extensions of the group, after the initiator of the conference has dialed the feature access code.
- To cancel the Emergency Conference, extension H must dial the feature access code for Cancel Conference, 190 (default).

How to configure

To provide this feature to extensions,

- You must enable the feature **Emergency Conference** in the"Class of Service (COS)" of the extensions in their "Station Basic Feature Template". By default, this feature is enabled on all extensions, so all extensions can use this feature.

- If the extension you are providing this feature is a DKP or an Extended IP phone, you may program a DSS key on the phone with this feature.

- You must also create a Department Group as Emergency Conference group. For instructions, see "Department Call".

- By default, the system plays a beep when the Emergency Conference starts. If you do not want the beep to be played, you must disable the flag *Play Beep when Conference/Dial-In Conference Starts*. For instructions, see "System Parameters".

  This flag is common for other features like "Conference-Multiparty", "Conference Dial-In" and "Raid".

- By default, the system plays *Triple Ring* as Ring Type for Emergency Conference. If necessary, you may configure a different ring type. For more information and for instructions, see "Distinctive Rings".

How to use

For EON and Extended IP Phone Users

To initiate an Emergency Conference:

- Press DSS Key assigned to Emergency Conference.
  Or
- Dial ***1177***
- Dial Department Group Number.
- All free extensions in the group will ring.
- You get connected to the extensions that answer.

If no resources are free, you will get the 'Conf. Resource full' message on your LCD.

You can initiate an Emergency Conference also using "Direct Inward System Access (DISA)".

To merge Emergency Conferences:

- While in Conference, press the 'Conference' Key.
- The Multiparty Conference menu appears on the LCD.
- Select the option 'Merge Conferences'
- The LCD displays list of Department Number and Group Name of ongoing Emergency Conferences.
- Select the desired Department Group number and press the Enter Key.
- Both the conferences will be merged.

Emergency Conferences can be merged by DKP or Extended IP Phone users only through the Phone Menu.

To cancel an Emergency Conference while in speech with one or more extensions:

- While in Conference, press the ‘Conference’ Key.
- The Multiparty Conference menu appears on the LCD.
- Select the option ‘Terminate Conference’ and press the Enter Key.

  OR

- Press the DSS key assigned to Terminate Conference.
- All the participants will get an Error Tone and the system resource occupied by the conference will be freed.

To cancel an Emergency Conference while the extensions are ringing:

- Go ON-Hook during Ring Back Tone.
- All extensions in the conference group will stop ringing.

You can cancel the conference only if you have initiated it.

For SLT Users

To initiate an Emergency Conference:

- Dial ***1177***
- Dial Department Group Number.

To cancel an Emergency Conference while in speech with one or more extensions:

- Go ON-Hook and then go OFF-Hook.
- Dial ***190***

To cancel an Emergency Conference while the extensions are ringing:

- Go ON-Hook during Ring Back Tone.

Emergency Detection and Reporting

If you are using the system in the Hospitality mode, to dial the Emergency Number 911, you must purchase the E911 license. For details, refer "License Management".

What's this?

When an emergency call is made from an extension, the system dials out the number using any of the free trunks selected for routing the Emergency Numbers. Since the number is dialed out by the System, the Emergency Service that attends to the call will be able to locate the System, but *not* the extension that made the call.

Similarly, the Operator too has no way of knowing which of the extensions made the call, thus making it difficult to quickly reach and provide help to the extension that made the emergency call.

With the Emergency Detection and Reporting feature, the Operator can know from which extension the emergency call is being made. Whenever an Emergency call is made by an extension user, the system detects and reports it to the Operator.

The system provides you an option to select the landing destination for reporting calls. You can:

- select the Operator as the landing destination
- define a separate Emergency Reporting Group

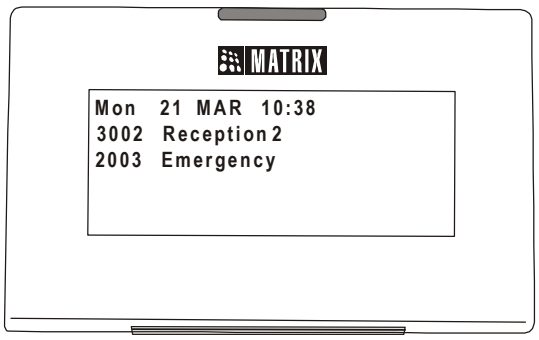
How it works

When an extension of SARVAM UCS makes an emergency call by dialing an Emergency Number,

- the system hunts for a free trunk in the OGTBG selected for routing the emergency number, and dials out the number from a free trunk.

- simultaneously, the system informs the Emergency Reporting Group/Operator by ringing on the desired extensions for the duration of the Emergency Reporting Call-Ring Timer (configurable; default: 10 minutes).

- If Emergency Reporting Group/Operator is a DKP or an Extended IP Phone, it will ring continuously, and an emergency message will be displayed on the LCD.



The emergency message shows the number of the extension which has made the emergency call, in this case, extension 2003.

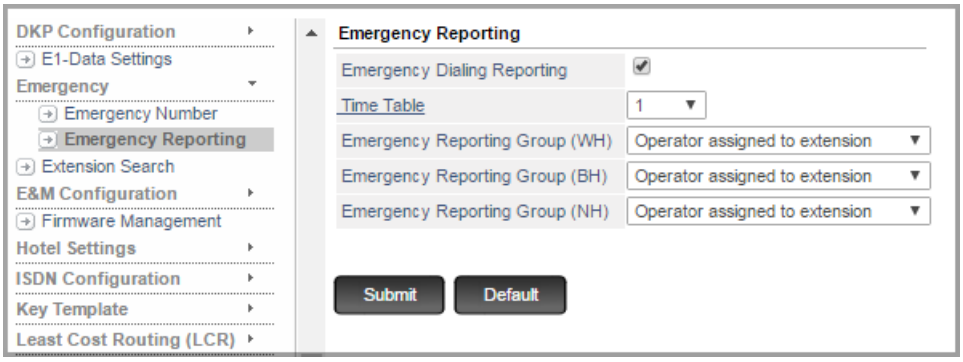
- To acknowledge the Emergency call the Emergency Reporting Group/Operator must press the enter key. The acknowledged Emergency calls are logged into the System Activity Log.
- If the Emergency Call is not acknowledged by the Emergency Reporting Group/Operator, the emergency call is logged into the Emergency Alarms Log. To know more about the Emergency Alarms Log, see "Emergency Alarms Log" at the end of the topic.

Also see the topics "Configuring Emergency Number Dialing" and "Emergency Dialing".

How to configure

Configuring Emergency Reporting using the Jeeves

- Log in to the Jeeves as System Engineer.
- Under **Configuration**, click **Emergency**.
- Click **Emergency Reporting**.



- **Emergency Dialing Reporting:** By default this check box is enabled. The system will detect the extension that makes an emergency call.
- **Time Table:** A Time Table is a schedule of the three Time Zones, namely: Working Hours, Break Hours, Non-Working hours for a week.

  You can define and select the Time Table for the Emergency Reporting Group as per you requirement.

  There are 8 different Time Table templates to select from. By default, the Time Table 1 is selected. In Time Table 1, six days of the week - Monday to Saturday -have working hours from 9:00-18:00, break hours from 13:00-14:00 hours and non-working hours from 18:00 to 09:00. Sunday is a holiday, with all three Time Zones set to 00:00 hours.

  You may also customize the default Time Table 1 OR customize and assign a different Time Table. Refer to "Time Tables" for more details.
- **Emergency Reporting Group:** By defaut **Operator assigned to extension** is selected as the landing destination group for placing emergency reporting calls, for all the time zones.

  You can either customize the default group OR customize and assign a different group.

You can select a different group for each time zone, that is Working Hours (WH), Break Hours (BH) and Non-working Hours (NH).

- Click **Submit**.

If required, you can change the Emergency Reporting Call-Ring Timer, see “System Timers and Counts”.

Emergency Alarms Log

Emergency Alarms Log is the log of unacknowledged Emergency Calls.

When an Emergency call is made by any extension user and is not acknowledged, that call is logged into the Emergency Alarms Log.

The log can be viewed by any DKP / Extended IP Phone user, using the DSS Key assigned to Emergency Alarms Log only. When an Emergency call is made by any extension user, the LED of the DSS Key glows continuous RED.

To view the log from any DKP / Extended IP Phone user,

- Press the DSS Key assigned to Emergency Alarms Log.
- A list of the last 20 unacknowledged Emergency calls appears with the following details:
  - Extension number from which the Emergency call was made.
  - Date and Time when the Emergency call was initiated from that Extension.
- Press the enter key to acknowledge the Emergency Call. The message "Emergency Acknowledged" appears on the screen.
- The system plays the Confirmation Tone followed by the Dial Tone.
- The acknowledged Emergency call is removed from the Emergency Alarms Log and is logged into the System Activity Log with the details of the extension that acknowledged the call.

Emergency Dialing

If you are using the system in the Hospitality mode, to dial the Emergency Number 911, you must purchase the E911 license. For details, refer "License Management".

What's this?

The SARVAM UCS supports dialing of Emergency number immediately without any blocking.

When an extension user dials an Emergency number, the system will hunt for a free trunk from the outgoing trunk bundle group selected for the emergency number. See "Configuring Emergency Number Dialing".

The system will not apply any of the following on the extension dialing the Emergency number:

- Toll Control (Allowed Denied Numbers, Dynamic Lock)
- Call Budget (even when call budget is consumed)
- Call Duration Control
- Automatic Number Translation

The system will allow the extension to dial the Emergency number even in the following conditions:

- the extension is in Off-Hook state.
- the extension is in Standby Mode.
- the extension has grabbed the trunk line (using Trunk access code or selective access)
- the call state is in any state: Ringing, Busy, Error, Confirmation.
- SIM card is not present in the Mobile port.
- Mobile port is not registered with the network.
- SIM PIN is not valid.
- the keypad of the extension phone is locked.

Emergency Numbers will always be out dialed through the OGTBG you have selected for the numbers.

Emergency Number will not be out dialed in the following cases:

- If the trunk port from which number is to be routed (CO, Mobile, T1E1PRI, BRI, SIP) is disabled.
- If the hardware related to dialing of Emergency number is not present.
- If the Emergency number is dialed from the SE Programming mode.

Emergency dialing will not work if Mains Power to the SARVAM UCS fails.

How to configure

The Emergency numbers are fixed as per the Region where SARVAM UCS is installed, you can add emergency numbers, as required. For instructions, see "Configuring Emergency Number Dialing".

How to use

To dial an Emergency number,

- Go Off-Hook
- Dial the Emergency Number
  OR
- Dial Trunk Access Code-Emergency Number
  For example:
  Dial ***0-112***

- *Wherever the Trunk Access Code conflicts with the Emergency Number, the emergency number should be dialed after dialing the Trunk Access Code.*

- *Let us take the example of Australia, where the emergency number is 000 and the trunk access code is 0. Now, when an extension user of SARVAM UCS located in Australia dials '0' of the emergency number, the system will consider it as trunk access code and will apply the trunk access code logic. Therefore, in such cases, the extension user must first dial the Trunk Access Code and then the Emergency Number. In this case, the extension user must dial 0-000 for emergency number dialing, so that the system will not wait for the Conflict Timer to apply the Trunk access code logic.*

Extended IP Phone/VARTA UC Client - Operation

Matrix offers the following proprietary Extended IP Phones/Mobile UC Clients:

- SPARSH VP248, the High-Definition Edge to your IP Communication. For Detailed description, see “Matrix SPARSH VP248”.
- SPARSH VP310, the Executive IP Phone. For Detailed description, see “Matrix SPARSH VP310”.
- SPARSH VP330, the Intuitive Touchscreen IP Phone. For detailed description, see “Matrix SPARSH VP330”.
- SPARSH VP510, the Premium IP Phone. For detailed description, see “Matrix SPARSH VP510”.
- Extended SPARSH VP710, the Smart Video IP Phone. For detailed description, see “Matrix Extended SPARSH VP710”
- SPARSH VP210, the Entry Level IP Phone. For detailed description, see “Matrix SPARSH VP210”
- Mobile UC Clients - VARTA ADR100, Mobile UC Client for Android Smart Phones and VARTA AMP100, Mobile UC Client for iPhones. See “Matrix VARTA ADR100 UC Client” and “Matrix VARTA AMP100 UC Client”
- MATRIX VARTA WIN200, UC Client for Windows. For detailed description, see “MATRIX VARTA WIN200”.

To know the list of featured supported, refer to “SARVAM UCS Features Supported in Terminals”.

Matrix SPARSH VP248

The Matrix SPARSH VP248, the proprietary Extended IP Phone is a feature-rich, VoIP (Voice over Internet Protocol) phone, providing voice communication over IP network. It looks and works like any normal phone, having all the traditional phone features such as redial, speed dial, call transfer, call hold, call forward, conference, and so forth. It allows you to make and receive calls using the handset, the headset (if connected) and the speaker.

The models of SPARSH VP248 are:

- **SPARSH VP248S** - the standard model, with a 2-line x 24-character LCD display.
- **SPARSH VP248P** - the premium model, with a 6-line x 24-character LCD display.

It is a powerful extension, supporting a host of phone and SARVAM UCS features, as listed below.

IP Phone Features

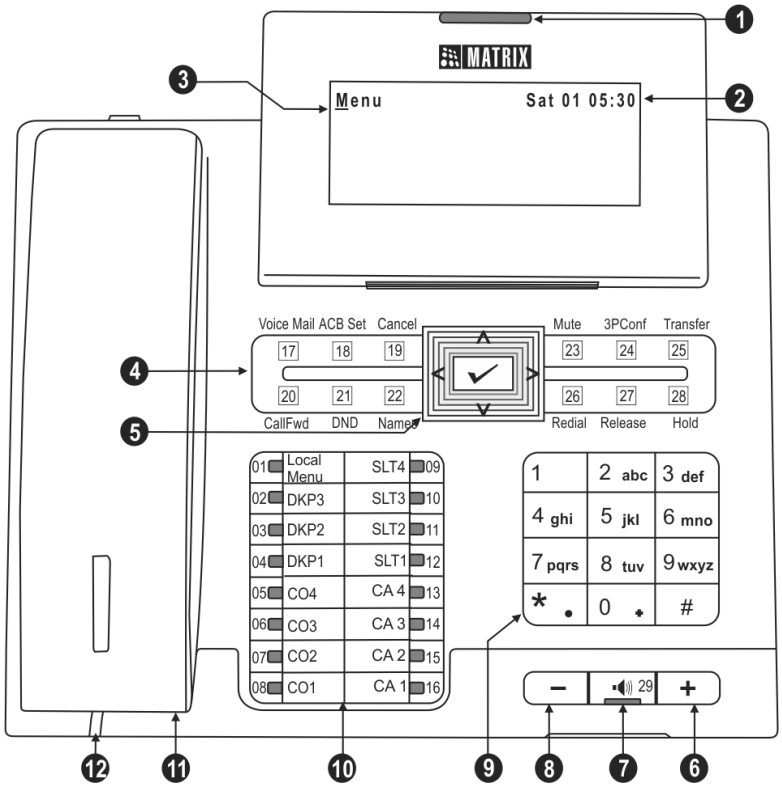
- Status of other ports (Tri-color LED indication)
- Programmable Direct Station Selection (DSS) Keys and Feature keys
- LCD notification messages
- Ringer Tune selection
- Adjustable Speech level
- Adjustable Ringer Volume
- Adjustable Backlight and Contrast levels
- Hands-free operation - Speaker key and headset connectivity.
- Call Logs - last 20 Missed, Answered and Dialed Calls.
- Message Paging
- Menu based operation of SARVAM UCS features
- Multiple Language support.

SARVAM UCS Features

The SPARSH VP248 supports SARVAM UCS features. A few of these are listed below:

- Abbreviated Dialing
- Auto Answer
- Call Chaining
- Call Cost Display
- Call Duration Display
- Call Mute
- Dialed Number Directory
- Directory Dialing by Name
- Dynamic Lock
- Forced Answer
- Keypad Lock
- Message Paging
- Off-Hook Alert
- Room Monitor
- User Status (Presence)

SPARSH VP248, Front View



1. Ringer LED
2. Date and Time
3. Cursor
4. Touch sense feature keys
5. Touch sense navigation keys
6. Volume increase key
7. Speaker key with LED
8. Volume decrease key
9. Dial Pad
10. Programmable feature keys
11. Handset
12. 4P4C Spring Cord

Models of SPARSH VP248 at a Glance

Feature	Model	
	SPARSH VP248S	SPARSH VP248P
Total number of keys	48	48
Number of programmable keys	29	29
Capsense keys	Yes	Yes
LCD display capacity	2 lines x 24 characters	6 lines x 24 characters
LCD with backlight	Yes	Yes
Headset Interface	Yes	Yes
Ringer Lamp (LED)	Yes	Yes
Speaker Phone	Full duplex	Full duplex

SPARSH VP248S

2 lines and 24 characters LCD display, full duplex, capsense feature keys

SPARSH VP248P

6 lines and 24 characters LCD display, full duplex, capsense feature keys.

LCD Display

The LCD display of SPARSH VP248 is backlit and can be tilted at a convenient angle for a clear view of the text/ characters displayed.

The LCD backlight can be turned on and off as well as adjusted for contrast and brightness from the "*Phone Settings*" of the SPARSH VP248 Phone Menu.

Ringer LED

The Ringer LED indicates incoming internal and external calls. The LED Cadence will match with the Ring Cadence of the incoming internal/external call.

The Ringer LED cadence changes according to the type of call, as described in the table below.

Type of Call	Cadence
Internal Call	Short, very slow (750ms ON, 2250ms OFF)
External Call	Double (400ms ON, 200ms OFF, 400ms ON, 2000ms OFF)
Alarm	Long, fast (1500ms ON, 500ms OFF)
Auto Redial Call	Long, very slow (2000ms ON, 4000ms OFF)
Auto Call Back Call	Short, slow (750ms ON, 2250ms OFF)
Priority	Triple (400ms ON, 200ms OFF, 400ms ON, 200ms OFF, 400ms ON, 2000ms OFF)
Programming mode	Continuous

Navigation Keys

The phone has 5 touch sense navigation keys to be used to move the cursor and scroll through Menu options.

✓ is the Enter key, used to make a selection or to complete an action.

^ is the Up key, used to scroll upwards while navigating the 'Menu'.

v is the Down key, used to scroll downwards while navigating the 'Menu'.

> is the Forward key, used to move the cursor.

< is the Back key, used to move the cursor, return from the Sub-menu to the Main Menu.

Feature Keys

These are 12 capsense keys assigned to important or frequently accessed features of SARVAM UCS. Refer to the table given below:

Sr.No.	Description	LED
1.	Voice Mail	Single Color - Blue
2.	Call Back	Single Color - Blue
3.	Cancel	No
4.	Mute	Single Color - Blue
5.	Conference	No
6.	Transfer	No
7.	Forward	Single Color - Blue
8.	DND	Single Color - Blue
9.	Names	No
10.	Redial	No
11.	Release	No
12.	Hold	No

These keys are programmable. However, as you cannot change the labels, avoid programming these keys.

For instructions on programming these keys, see “DSS Key Settings” under “Configuring Matrix SPARSH VP248”.

A few of these Feature keys are equipped with an LED to indicate the status of the feature assigned to it. You may re-assign other features to these keys. We recommend you to assign those features that require LED indication to the Feature keys equipped with an LED.

Not all features require LED indication. For example, the feature Call Pick-Up does not require an LED. So when you assign this feature to a feature key, the LED of the key remains inactive, when Call Pick-Up is accessed.

Feature like Auto Redial requires an LED to show that it has been set or canceled. So, the LED of the Feature key to which the Auto Redial feature has been assigned will glow Blue, when Auto-Redial is set, and the LED is turned off when the feature is canceled.

Direct Station Selection (DSS) Keys

These are 16 programmable keys that can be assigned to Stations and Trunks and important or frequently accessed features of SARVAM UCS.

For instructions on programming these keys, see “DSS Key Settings” under “Configuring Matrix SPARSH VP248”.

DSS Key LEDs

Each DSS key is equipped with an LED which glows to indicate the status of the Trunk/Extension or Feature assigned to it.

- **Status of Extensions and Trunks:** The LED of DSS keys assigned to Extensions/Trunks glow in three colors to indicate status of the call event on the Extensions/Trunks and on the Extended IP Phone.

  Thus, the status of the Extended IP Phone user's own Extension as well as that of the other Extensions (i.e. Extended IP Phones and SLTs) and the status of Trunk lines are indicated by the LED of the DSS keys assigned to those Extensions and Trunks on the Extended IP Phone.

  The following table shows the relationship between the color of the LED and various events:

LED Color	LED Mode		
	Continuously ON	Slow Blink	Fast Blink
Blue	The key assigned to the Extension you are in speech with.	The key assigned to the Extension you have kept on hold.	The key assigned to the Extension you are calling or from which you are being called.
Red	The key assigned to the Extension that is now busy with another Extension/ Trunk.	The key assigned to the Extension which has put another Extension/Trunk on hold.	The key assigned to the Extension/Trunk that is called or being called by another.
Violet	You are talking on a Trunk (external call)	You have held a Trunk (external call)	You have an incoming call on the Trunk (external call)

  - **Blue** indicates the state of the extension/trunk you access. For example, when you make a call to another Extension 203, the LED of the DSS key assigned to Extension 203 blinks Blue to indicate ringing at the Extension. If you have successfully established speech with Extension 203 the LED glows Blue continuously.

  - **Red** indicates the state of other Extensions/Trunks. For example, if the LED of the DSS key assigned to Extension 201 is glowing Red continuously, it means Extension 201 is busy with another Extension or Trunk.

  - **Violet** indicates the state of the trunk you are in speech with. For example, when you are in speech on an outgoing call on Trunk 1 the LED of the DSS Key assigned to Trunk 1 will be continuously ON. When you put the call on hold, the LED will blink slowly.

- **Status of Features:** The LED of a DSS key is activated when the feature assigned to this key is used.

Not all features require LED indication. For example, the feature Call Pick-Up does not require an LED. So when you assign this feature to a DSS key, the LED of the key remains inactive, when Call Pick-Up is accessed.

Feature like Auto Redial requires an LED to show that it has been set or canceled. So, the LED of the DSS key to which the Auto Redial feature has been assigned will glow Red, when Auto-Redial is set, and the LED is turned off when the feature is canceled.

The LEDs of the Call Appearance (CA) Keys will function in the same manner as the DSS Key LEDs.

Dial Pad

The dial pad consists of 12 fixed keys for the digits 0, 1-9, and the characters * and #. The dial pad is used for dialing numbers of extensions, external parties, and for dialing the programming and feature access codes.

Speaker Key

The speaker key sets the phone in 'Speaker mode' for hands-free operation. The Speaker key is programmable, you can program any other feature/function on this key.

Speaker Key LED

The Speaker Key on SPARSH VP248 is equipped with a single color LED which glows Blue when pressed for the speaker mode and is turned off, when you exit the speaker mode.

Volume Keys

- **"+" (plus)**: This is the increase key, to raise the volume of speech while talking and to increase the Ringer volume, when the phone is ringing.

- **"-" (minus)**: This is the decrease key, to lower the volume of speech while talking and to decrease the Ringer volume when the phone is ringing.

Headset Connectivity

The SPARSH VP248 provides two Headset interfaces: a 2.5mm Audio Jack and an RJ9 connector at the bottom of the phone body.

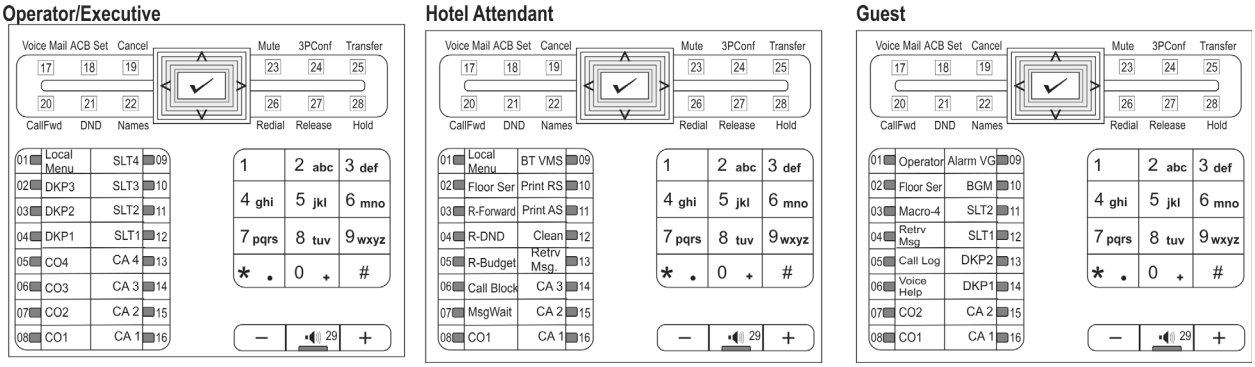
So you can use any stereo headset of standard make with a 2.5 mm single connector or a headset with an RJ9 connector.

You can also program any of the DSS keys to function as the Headset key. For instructions on programming the key, see “DSS Key Settings” under “Configuring Matrix SPARSH VP248”.

Key Maps

As SPARSH VP248 may be the extension of the Operators and Executives in an enterprise to meet the varied requirements of each user group, these key maps can be customized to match the exact requirement of individual users. For instructions on customizing the Key Maps, see “Customizing Extended IP Phone Templates using Jeeves” . You can also personalize the key maps for each location, for instructions, see “DSS Key Settings” under “Configuring Matrix SPARSH VP248”.

Matrix Extended IP Phone, SPARSH VP248 Key Template (default)



The key maps of the Operator and Executive 1, 2, 3 are the same.

By using Key Templates you can prepare and assign common key maps to all or as many Extended IP Phones as you want, at one go.

SARVAM UCS also offers the flexibility to personalize the Key Maps of each Extended IP Phone, instead of using the Key Templates. For example, if you have assigned a common Executive Key Template to 12 Extended IP Phones, but you want to reassign some of the keys on two of these Extended IP Phones, SARVAM UCS allows you to selectively personalize the key maps of these two Extended IP Phones.

Phone Menu

You can access the following SARVAM UCS and phone features from the Menu of SPARSH VP248:

Menu option	Description
Call Logs	To view call history of internal and external Missed, Answered and Dialed calls. You can also edit numbers in the call logs and store them in the Personal Directory.
Contacts	To add, edit, delete names and numbers of contacts in the Global Directory Part 1.
Call Forward	To set and cancel Call Forward-Busy, Call-Forward No Reply, Call-Forward-Unconditional, and Follow Me.
Dynamic Lock	To change the Toll Control level of the phone.
User Status	To set User Present or User Absent.
Keypad Lock	To lock the keypad of the phone.
Do Not Disturb	To set/cancel Do Not Disturb on the phone, i.e. block incoming internal and external calls.
Call Cost Display	To view the cost of calls made from the phone.
Hotline	To set/cancel Hotline and Delayed Hotline.
Alarm	To set/cancel Personalized and Automated Alarms.
Change User Password	To change User Password.
One Touch Transfer	To set/clear Transfer Number.

Menu option	Description
Phone Settings	To customize settings of the phone such as Speech and Ringer Controls, LCD Display settings (Brightness and Contrast, Backlight ON/OFF), Headset Connectivity, Call Answering Mode (manual/auto answer).

Navigating the Phone Menu

To navigate the menu,

- Tap on Enter key when the phone is idle.
- Scroll by tapping the Up/Down Navigation Key to reach the desired Menu option.
- Tap on Enter key to select the desired Menu option.
- Scroll by tapping on the Up/Down Navigation Key to reach the desired sub-menu option.
- Tap on Enter key to select the desired sub-menu option.

To exit menu,

- Press Cancel key.
  Or,
- Go ON-Hook.

Call Waiting Indication

During an on-going call, if there is another incoming call, an indication will be provided to you for the waiting call.

The call waiting indication depends on the **Call Waiting Tone (for SPARSH VP248/VP310/VP510)** option you select in General Parameters under SIP Extensions. See *"Configuring Matrix SPARSH VP248"* for instructions.

Connecting SPARSH VP248

For detailed instructions to connect SPARSH VP248, see *"Connecting SPARSH VP248 as Extended SIP Extension"* in SIP Extensions for ETERNITY LENX, *"Connecting SPARSH VP248 as Extended SIP Extension"* in SIP Extensions for ETERNITY MENX, *"Connecting SPARSH VP248 as Extended SIP Extension"* in SIP Extensions for ETERNITY GENX and *"Connecting SPARSH VP248 as Extended SIP Extension"* in SIP Extensions for ETERNITY PENX.

Configuring SPARSH VP248

For detailed instructions on how to configure SPARSH VP248, see *"Configuring Matrix SPARSH VP248"*.

Operating SPARSH VP248

Please refer the EON48_310_SPARSH VP248_310 User Guide for instructions on operating the features of SARVAM UCS.

Matrix SPARSH VP310

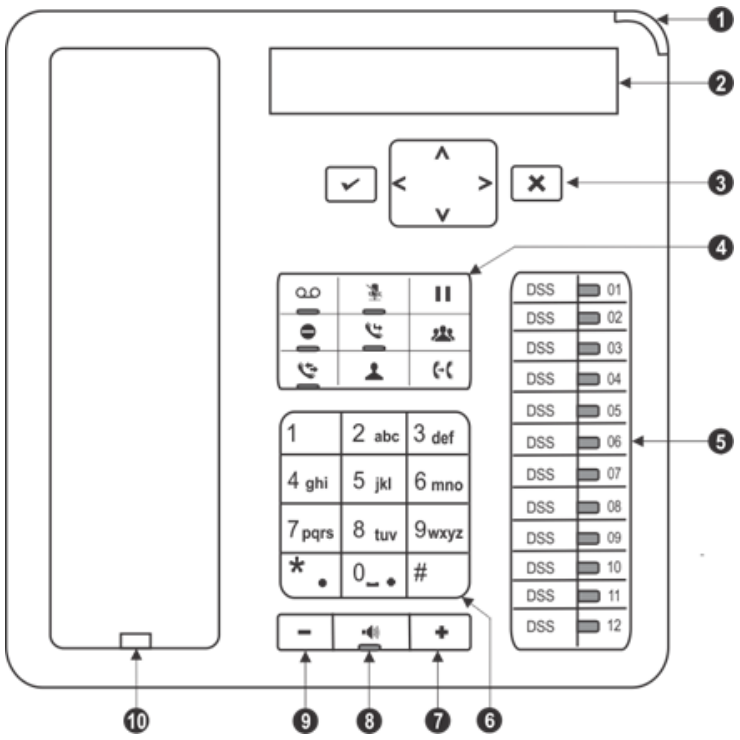
SPARSH VP310, The Executive IP Phone is engineered to offer a contemporary design with crystal-clear audio and feature-rich capabilities at economical price. Elegant design, built-in programmable DSS Keys and plug-n-play connectivity makes SPARSH VP310 an easy to use phone for executives. SPARSH VP310 works in tight integration with SARVAM UCS for speed of operations and better workforce collaboration.

Key Features

- 2 Line LCD with Backlit
- Fixed Function Keys (With LED) - Voice Mail, Mute, Do Not Disturb, Forward, Logs, Speaker
- Fixed Function Keys (Without LED) - Hold, Conference, Contacts, Transfer
- Superior Voice Quality with HD Audio
- Full Duplex Speaker Phone
- PC and LAN Ethernet Ports
- Power over Ethernet (IEEE 802.3af)
- 12 DSS/BLF Keys for Feature, Line, Extension
- Message Wait and Ringer Lamp
- 3.5mm Headset Connectivity
- Adjustable Desk Stand
- Plug & Play

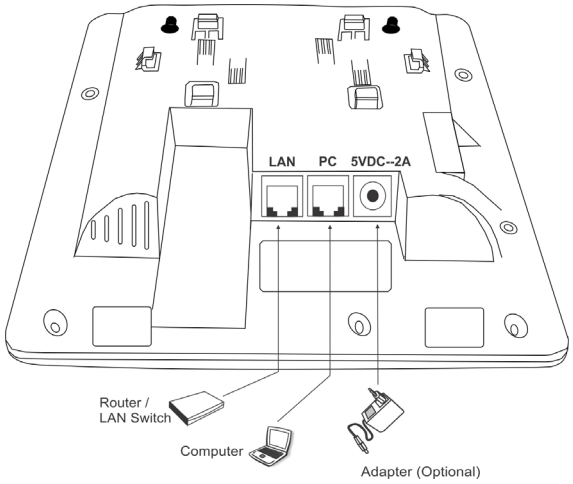
If any SPARSH VP310 extension is Off-Hook, the system will not be able to detect it.However, if any SLT/ DKP is Off-Hook, Off-Hook Alert will be provided to SPARSH VP310 if it is the Operator phone.

SPARSH VP310, Front View



1	Ringer LED
2	LCD Screen
3	Navigation Keys
4	Fixed Function Keys
5	DSS (Direct Station Selection) Keys
6	Digit Keys/ Dial Pad Keys
7	Volume Increase Key/ "+" Key
8	Speaker Key
9	Volume Decrease Key/ "-" Key
10	Handset

Bottom View



LCD Display

The LCD display of SPARSH VP310 is backlit. The LCD backlight can be turned on and off as well as adjusted for contrast and brightness from the *Phone Settings* of the SPARSH VP310 Phone Menu.

Ringer LED

The Ringer LED indicates incoming internal and external calls. The LED Cadence will match with the Ring Cadence of the incoming internal/external call.

The Ringer LED changes according to the type of call, as described in the table below.

Type of Call	Cadence
Internal Call	Short, very slow (750ms ON, 2250ms OFF)
External Call	Double (400ms ON, 200ms OFF, 400ms ON, 2000ms OFF)
Alarm	Long, fast (1500ms ON, 500ms OFF)
Auto Redial Call	Long, very slow (2000ms ON, 4000ms OFF)
Auto Call Back Call	Short, slow (750ms ON, 2250ms OFF)
Priority	Triple (400ms ON, 200ms OFF, 400ms ON, 200ms OFF, 400ms ON, 2000ms OFF)
Programming mode	Continuous

Navigation Keys

The phone has the following navigation keys which are used to move the cursor and scroll through Menu options.

✓ is the Enter key, used to make a selection or to complete an action.

^ is the Up key, used to scroll upwards while navigating the Menu.

v is the Down key, used to scroll downwards while navigating the Menu.

> is the Forward key, used to move the cursor.

< is the Back key, used to move the cursor, return from the Sub-menu to the Main Menu.

✕ is the Cancel key, used to exit a menu.

Feature Keys

There are 9 Feature keys assigned to important or frequently accessed features of SARVAM UCS.

Sr.No.	Feature icon	Description	LED
1.		Voice Mail	Single Color - Blue
2.		Mute	Single Color - Blue
3.		Hold	No
4.		DND	Single Color - Blue
5.		Call Forward	Single Color - Blue
6.		Conference	No
7.		Call Logs	Single Color - Blue
8.		Names	No
9.		Transfer	No

These keys are programmable. However, as you cannot change the labels avoid programming these keys.

For instructions on programming these keys, see “DSS Key Settings” under “Configuring Matrix SPARSH VP310”.

A few of these Feature keys are equipped with an LED to indicate the status of the feature assigned to it. You may re-assign other features to these keys. We recommend you to assign those features that require LED indication to the Feature keys equipped with an LED.

Not all features require LED indication. For example, the feature Call Pick-Up does not require an LED. So when you assign this feature to a feature key, the LED of the key remains inactive, when Call Pick-Up is accessed.

Feature like Auto Redial requires an LED to show that it has been set or canceled. So, the LED of the Feature key to which the Auto Redial feature has been assigned will glow Blue, when Auto-Redial is set, and the LED is turned off when the feature is canceled.

Direct Station Selection (DSS) Keys

These are 12 programmable keys that can be assigned to Stations and Trunks and important or frequently accessed features of SARVAM UCS.

For instructions on programming these keys, see “DSS Key Settings” under “Configuring Matrix SPARSH VP310”.

DSS Key LEDs

Each DSS key is equipped with an LED which glows to indicate the status of the Trunk/Extension or Feature assigned to it.

- **Status of Extensions and Trunks:** The LED of DSS keys assigned to Extensions/Trunks glow in three colors to indicate status of the call event on the Extensions/Trunks and on the Extended IP Phone.

  Thus, the status of the Extended IP Phone user's own Extension as well as that of the other Extensions (i.e. Extended IP Phones, SLTs) and the status of Trunk lines are indicated by the LED of the DSS keys assigned to those Extensions and Trunks on the Extended IP Phone.

  The following table shows the relationship between the color of the LED and various events:

LED Color	LED Mode		
	Continuously ON	Slow Blink	Fast Blink
Blue	The key assigned to the Extension you are in speech with.	The key assigned to the Extension you have kept on hold.	The key assigned to the Extension you are calling or from which you are being called.
Red	The key assigned to the Extension that is now busy with another Extension/ Trunk.	The key assigned to the Extension which has put another Extension/Trunk on hold.	The key assigned to the Extension/Trunk that is called or being called by another.
Violet	You are talking on a Trunk (external call)	You have held a Trunk (external call)	You have an incoming call on the Trunk (external call)

  - **Blue** indicates the state of the extension/trunk you access. For example, when you make a call to another Extension 203, the LED of the DSS key assigned to Extension 203 blinks Blue to indicate ringing at the Extension. If you have successfully established speech with Extension 203 the LED glows Blue continuously.

  - **Red** indicates the state of other Extensions/Trunks. For example, if the LED of the DSS key assigned to Extension 201 is glowing Red continuously, it means Extension 201 is busy with another Extension or Trunk.

  - **Violet** indicates the state of the trunk you are in speech with. For example, when you are in speech on an outgoing call on Trunk 1 the LED of the DSS Key assigned to Trunk 1 will be continuously ON. When you put the call on hold, the LED will blink slowly.

- **Status of Features:** The LED of a DSS key is activated when the feature assigned to this key is used.

  Not all features require LED indication. For example, the feature Call Pick-Up does not require an LED. So when you assign this feature to a DSS key, the LED of the key remains inactive, when Call Pick-Up is accessed.

  Feature like Auto Redial requires an LED to show that it has been set or canceled. So, the LED of the DSS key to which the Auto Redial feature has been assigned will glow Red, when Auto Redial is set, and the LED is turned off when the feature is canceled.

The LEDs of the Call Appearance (CA) Keys will function in the same manner as the DSS Key LEDs.

Dial Pad

The dial pad consists of 12 fixed keys for the digits 0-9, and the characters * and #. The dial pad is used for dialing numbers of extensions, external parties, and for dialing the programming and feature access codes.

Speaker Key

The speaker key sets the phone in 'Speaker mode' for hands-free operation.

Speaker Key LED

The Speaker Key on SPARSH VP310 is equipped with a single color LED which glows Blue when pressed for the speaker mode and is turned off, when you exit the speaker mode.

Volume Keys

- **"+" (plus)**: This is the increase key, to raise the volume of speech while talking and to increase the Ringer volume, when the phone is ringing.

- **"-" (minus)**: This is the decrease key, to lower the volume of speech while talking and to decrease the Ringer volume when the phone is ringing.

Headset Connectivity

The SPARSH VP310 provides two Headset interfaces: a 3.5mm Audio Jack and an RJ9 connector on the left side panel of the phone.

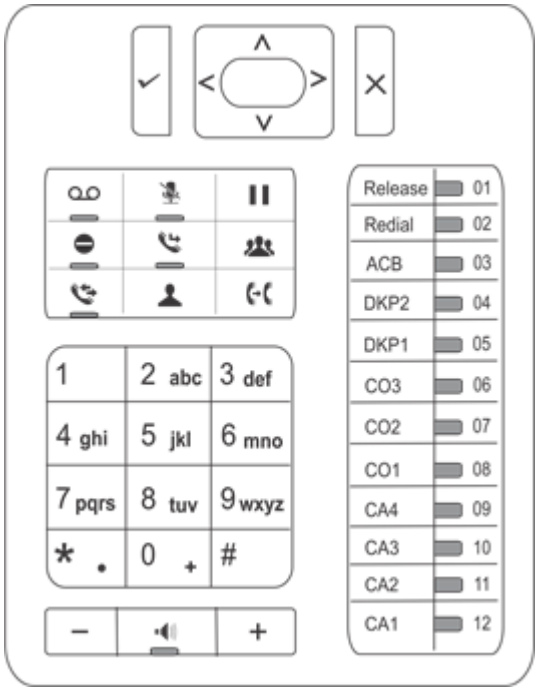
So you can use any stereo headset of standard make with a 3.5 mm single connector or a headset with an RJ9 connector.

You can also program any of the DSS keys to function as the Headset key. For instructions on programming the key, see “DSS Key Settings” under “Configuring Matrix SPARSH VP310”.

Key Maps

As SPARSH VP310 may be the extension of the Operators and Executives in an enterprise to meet the varied requirements of each user group, these key maps can be customized to match the exact requirement of individual users. For instructions on customizing the Key Maps, see “Customizing Extended IP Phone Templates using Jeeves” . You can also personalize the key maps for each location, for instructions, see “DSS Key Settings” under “Configuring Matrix SPARSH VP310”.

Matrix Extended IP Phone, SPARSH VP310 Key Template (default)



The key maps of the Operator and Executive 1, 2, 3 are the same.

By using Key Templates you can prepare and assign common key maps to all or as many Extended IP Phones as you want, at one go.

SARVAM UCS also offers the flexibility to personalize the Key Maps of each Extended IP Phone, instead of using the Key Templates. For example, if you have assigned a common Executive Key Template to 12 Extended IP Phones, but you want to reassign some of the keys on two of these Extended IP Phones, SARVAM UCS allows you to selectively personalize the key maps of these two Extended IP Phones.

Phone Menu

Press the Enter key ✓ to access the Phone Menu.

You can access the following SARVAM UCS and phone features from the Menu of SPARSH VP310.

Menu option	Description
Call Logs	To view call history of internal and external Missed, Answered and Dialed calls. You can also edit numbers in the call logs and store them in the Personal Directory.
Call Forward	To set and cancel Call Forward-Busy, Call-Forward No Reply, Call-Forward-Unconditional, and Follow Me.
Dynamic Lock	To change the Toll Control level of the phone.
User Status	To set User Present or User Absent.
Keypad Lock	To lock the keypad of the phone. (when the keypad is locked, the features Call Log, Contact, Call Forward, Dynamic Lock, User Status, DND, Call Cost Display, Hotline, Alarm, Change User Password will not be accessible.)

Menu option	Description
Do Not Disturb	To set/cancel Do Not Disturb on the phone, i.e. block incoming internal and external calls.
Call Cost Display	To view the cost of calls made from the phone.
Hotline	To set/cancel Hotline and Delayed Hotline.
Alarm	To set/cancel Personalized and Automated Alarms.
Change User Password	To change User Password.
One Touch Transfer	To set/clear Transfer Number.
Phone Settings	To customize settings of the phone such as Speech and Ringer Controls, LCD Display settings (Brightness and Contrast, Backlight ON/OFF), Headset Connectivity, Call Answering Mode (manual/auto answer).

When the phone is in idle state,

- Press the Down key ⋁ to access the Network Settings.
- Press the Up key ⋀ , if you wish to change the Ringtone and Play Key Tone.

Navigating the Phone Menu

To navigate the menu,

- Press on Enter key when the phone is idle.
- Scroll by tapping the Up/Down Navigation Key to reach the desired Menu option.
- Press on Enter key to select the desired Menu option.
- Scroll by tapping on the Up/Down Navigation Key to reach the desired sub-menu option.
- Press on Enter key to select the desired sub-menu option.

To exit menu,

- Press Cancel key.
  Or
- Go ON-Hook.

Call Waiting Indication

During an on-going call, if there is another incoming call, an indication will be provided to you for the waiting call.

The call waiting indication depends on the **Call Waiting Tone (for SPARSH VP248/VP310/VP510)** option you select in General Parameters under SIP Extensions. See “Configuring Matrix SPARSH VP310” for instructions.

Connecting SPARSH VP310

For detailed instructions to connect SPARSH VP310, see *“Connecting SPARSH VP310 as Extended SIP Extension”* in SIP Extensions for ETERNITY LENX, *“Connecting SPARSH VP310 as Extended SIP Extension”* in SIP Extensions for ETERNITY MENX, *“Connecting SPARSH VP310 as Extended SIP Extension”* in SIP Extensions for ETERNITY GENX and *“Connecting SPARSH VP310 as Extended SIP Extension”* in SIP Extensions for ETERNITY PENX.

Configuring SPARSH VP310

For detailed instructions on how to configure SPARSH VP310, see “Configuring Matrix SPARSH VP310”.

Operating SPARSH VP310

Please refer the EON48_310_SPARSH VP248_310 User Guide for instructions on operating the features of SARVAM UCS.

Matrix SPARSH VP330

SPARSH VP330, the next generation feature rich SIP phone of Matrix with an intuitive GUI (Graphical User Interface) based touch-screen. It provides you an easy way of managing your communication needs to meet the day to day business requirements. Designed to change the way you communicate and collaborate, SPARSH VP330 delivers a seamless communication experience that is convenient and ready to use in real time, so that you can focus on the task in hand. Once it is registered with the SARVAM UCS, you can start operating the phone.

Key Features

- **Enhanced Call Management:** Dedicated one-touch feature keys and intuitive user interface provides quick access to full range of System call management features including Call Hold, Call Park, Call Transfer, Conference and Voicemail.
- **Access to Corporate Directory:** Easy integration with the enterprise’s Corporate Directory (Global Directory) which allows you to easily locate and dial corporate contacts at one click.
- **Presence:** Provides intuitive Presence status display and supports changing your Presence status which is viewable to other extension users. You can also view the Presence Status of remote users.

- **Busy Lamp Field (BLF) Indication:** You can monitor the status of the extensions and trunks who are assigned DSS Soft keys. You can also pickup ringing extensions/trunks using DSS Soft keys.

- **Plug & Play:** Integrated Plug & Play feature that enables to power up the phone and start using it. On the other hand, it helps in mass deployment of the phones in your organization without requiring a lot of manual intervention to configure each of the phones separately.

- **Multiple Language Support:** The phone can be operated in six different languages including English, French, German, Spanish, Portuguese and Italian.

- **Capacitive Touch Screen:** 4.3 inch Capacitive Touch Screen LCD (Liquid Crystal Display) that delivers easy access to advanced features and a unique experience beyond traditional desk phones. Supports adjustable Brightness controls from the touch screen to suit your customized LCD requirement.

- **Easy-to-Use Hard Keys:** Supports following hard keys and LEDs:
  - 6 Fixed Feature Keys
  - 12 Alphanumeric Digit Keys (Dial pad Keys)
  - 1 Speaker Key.
  - 12 DSS (Direct Station Selection) keys.
  - 1 Ringer LED.

- **Improved Audio:** High Definition (HD) Audio output that delivers crystal clear voice and life like conversations over HD handset and hands-free speaker.

- **High Speed Ethernet connectivity:** Dual switched 10/100 Base-T auto-sensing Ethernet LAN connectivity allows unconstrained bandwidth from the network to the phone. LAN port is used to connect the phone to a Switch or a Hub or a Router or an xDSL Modem. You can connect your PC to the PC port of the phone.

- **Power over Ethernet (PoE):** Integrated IEEE 802.3f Power over Ethernet allows easy deployment with centralized powering without a need for external power adapter.

- **Wi-Fi Support:** Supports Wi-Fi (WLAN) connectivity using which the phone provides seamless connectivity to the corporate Wi-Fi network and offers flexibility to work from anywhere in the office. If your installation setup does not meet the requirements of suitable wired Ethernet connectivity due to any reason, then you can connect the compatible Wi-Fi Adapter supplied by Matrix to the USB port to register and use the phone.

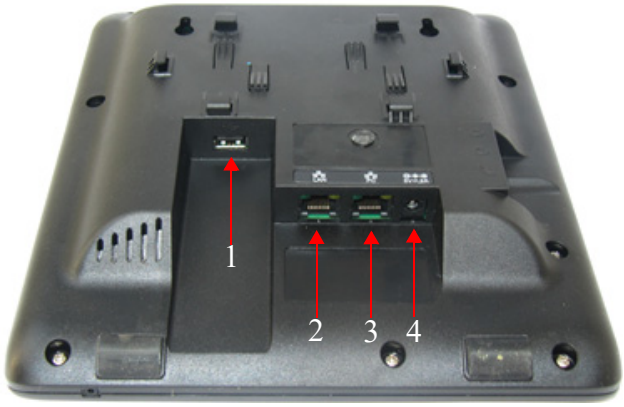
- **Improved Audio:** High Definition (HD) Audio output that delivers crystal clear voice and life like conversations over HD handset and hands-free speaker.

- **Full-duplex Hands-free Speaker:** High quality speaker with acoustic echo cancellation to deliver natural and clear speech even for hands free operation without any distortion.

Front View

1	Ringer LED
2	LCD Screen
3	Fixed Function Keys
4	DSS (Direct Station Selection) Keys
5	Digit Keys/ Dial pad Keys
6	Volume Up Key/ "+" Key
7	Speaker Key
8	Volume Down Key/ "-" Key
9	Handset

Bottom View

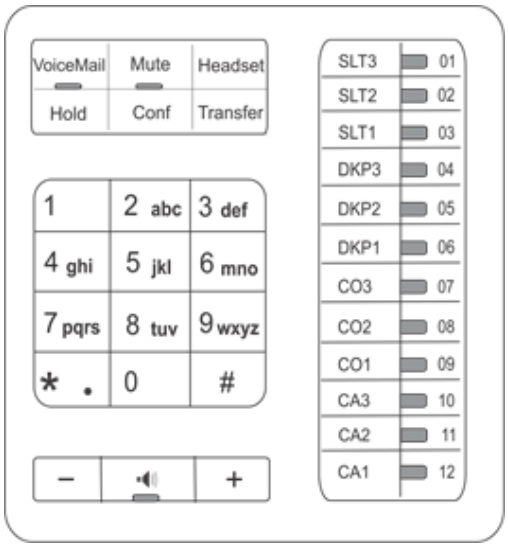


1	USB Port for Wi-Fi Adapter
2	LAN Port for a LAN Switch / Router
3	PC Port for a PC / Computer
4	Power Port for the Power Adapter (Optional)

Left Side View

1	RJ11 Headset Port
2	Headset Port
3	Handset Port

SPARSH VP330 Key Template (default)



All the key maps of the SPARSH VP330 are the same.

Feature Keys and DSS Keys

Feature Keys

There are 6 Feature Keys. Each Feature Key is accompanied by a feature icon that describes its function. Default features assigned to these keys are as follows.

SR. No.	Feature icon	Assigned Feature	LED
1		Voicemail	Single Color - Blue
2		Mute	Single Color - Blue
3		Headset	Single Color - Blue
4		Hold	No
5		Conference	No
6		Transfer	No

These Feature keys are programmable. You cannot change the labels of these keys and therefore it is recommended that you avoid programming these keys. However, if you still decide to reprogram features on these keys, keep a note of the changes you have made.

A few of these Feature keys are equipped with an LED to indicate the status of the feature assigned to it. You may re-assign other features to these keys. We recommend you to assign those features that require LED indication to the Feature keys equipped with an LED.

Not all features require LED indication. For example, the feature Call Pick-Up does not require an LED. So when you assign this feature to a feature key, the LED of the key remains inactive, when Call Pick-Up is accessed.

Feature like Auto Redial requires an LED to show that it has been set or canceled. So, the LED of the Feature key to which the Auto Redial feature has been assigned will glow Blue, when Auto-Redial is set, and the LED is turned off when the feature is canceled.

DSS (Direct Station Selection) Keys

There are 12 DSS Keys which can be used for quick access to different features and facilities. For example, to dial an extension number at one key press, just press the DSS key assigned to that extension and the call will be placed automatically. These Keys have dual color LEDs - blue and red. Features/facilities assigned to these keys can be changed by the System Engineer.

As SPARSH VP330 may be the extension of the Operators and Executives in an enterprise to meet the varied requirements of each user group, these key maps can be customized to match the exact requirement of individual users. For instructions on customizing the Key Maps, see “Customizing Extended IP Phone Templates using

Jeeves" . You can also personalize the key maps for each location, for instructions, see, "DSS Key Settings" under "Configuring Matrix SPARSH VP330".

For detailed information about SPARSH VP330, refer to the *SPARSH VP330 User Guide*.

Connecting SPARSH VP330

For detailed instructions to connect SPARSH VP330, see *"Connecting SPARSH VP330 as Extended SIP Extension"* in SIP Extensions for ETERNITY LENX, *"Connecting SPARSH VP330 as Extended SIP Extension"* in SIP Extensions for ETERNITY MENX, *"Connecting SPARSH VP330 as Extended SIP Extension"* in SIP Extensions for ETERNITY GENX and *"Connecting SPARSH VP330 as Extended SIP Extension"* in SIP Extensions for ETERNITY PENX.

Configuring SPARSH VP330

For detailed instructions on how to configure SPARSH VP330, see *"Configuring Matrix SPARSH VP330"*.

Operating SPARSH VP330

Refer the *SPARSH VP330 User Guide* for instructions on operating the features of SARVAM UCS.

Matrix SPARSH VP510

SPARSH VP510, the Premium IP Phone sets the benchmark for quality performance with elegant design and crystal-clear voice. SPARSH VP510 features a Vivid LCD Graphical Display, Direct Station Selection (DSS) Keys, 3.5mm Headset connectivity, High Quality speakerphone and high definition audio quality.

Engineered to deliver full feature access of SARVAM UCS, SPARSH VP510 acts as face of your communication system covering wide array of business environments.

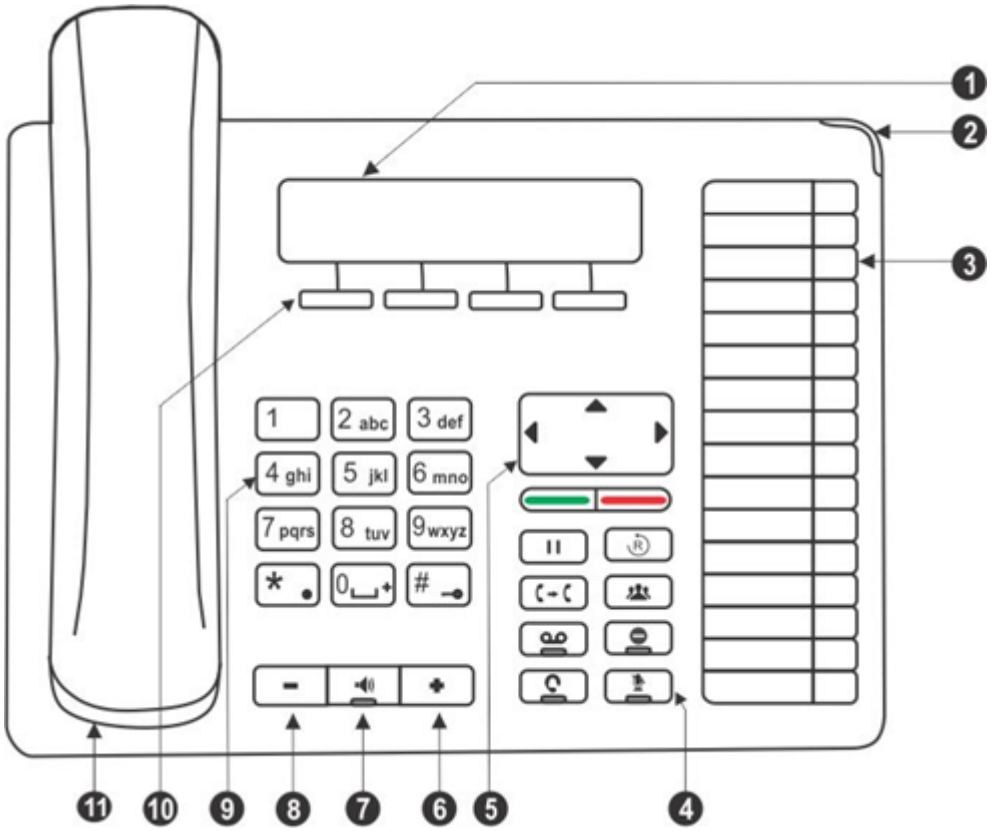
The State-of-the-art Deskphone is best suited to deployment for Reception, Supervisors, Managers, Executives, Call Center Agents and Office professionals. These phones offer flexibility to streamline communications and attain higher return over investment.

Key Features

- Minimum 240*64 pixel and above graphical LCD with Back light
- Message Wait and Ringer Lamp
- Built-in 16 DSS Keys for Feature, Line, Extension
- Add -on DSS module facility
- HD Voice,HD Handset, HD Speaker
- 4 Context Sensitive Keys
- 5 Features Keys (With LED) -Voice mail ,Headset ,Mute, DND, Speaker
- Fixed Function Keys (Without LED) - Hold , Conference, Redial, Transfer
- Tight integration with Server over SIP protocol /Proprietary Protocol
- HTTP Auto Provisioning
- Dual Color LED illuminated LED for line status
- One Touch Transfer
- Call logs
- Ringtone selection
- Wideband Codec : G722
- Narrowband Codec: G.711,G.723,G.729ab,GSM
- VAD,CNG,AEC,AJB,AGC
- Full Duplex speaker phone with AEC ,VAD ,CNG
- IP Assignment : Static /DHCP
- TCP/DNS-SRV
- AEC encryption for config file
- IEEE802.1x
- 3.5 mm / RJ9 headset port
- Dual port 10/100 Mbps Ethernet
- Stand with 2 adjustable angles
- PoE (IEEE 802.af) class2

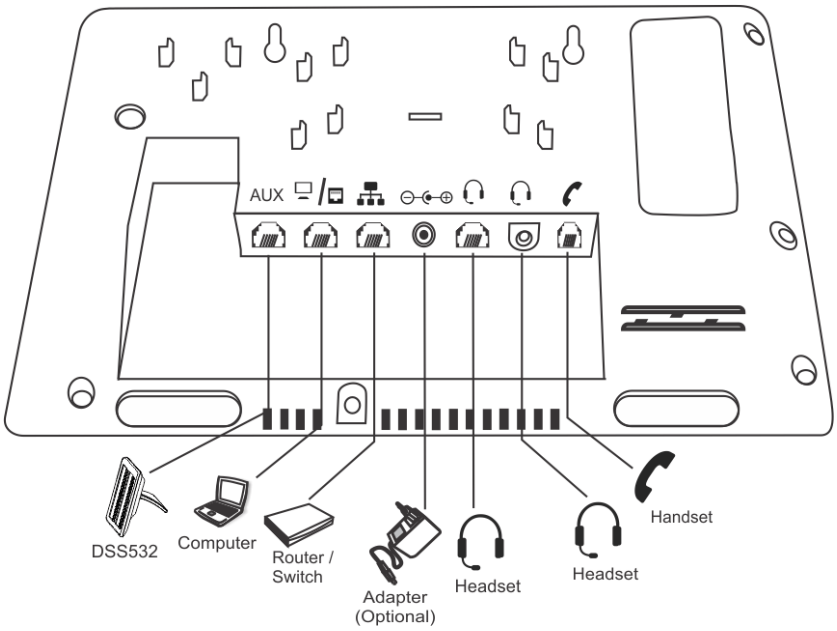
If any SPARSH VP510 extension is Off-Hook, the system will not be able to detect it. However, if any SLT/ DKP is Off-Hook, Off-Hook Alert will be provided to SPARSH VP510 if it is the Operator phone.

SPARSH VP510, Front View



1	LCD Screen
2	Ringer LED
3	DSS (Direct Station Selection) Keys
4	Fixed Function Keys
5	Navigation Keys
6	Volume Increase Key
7	Speaker Key
8	Volume Decrease Key
9	Dial Pad/ Key Pad Keys
10	Context Sensitive Keys
11	Handset

Bottom View



LCD Display

The LCD display of the phone is backlit. The LCD backlight can be turned on and off as well as adjusted for contrast and brightness from the *Display Settings* of the Phone Menu.

Ringer LED

The Ringer LED will glow in Blue (1 sec ON – 500 msec OFF) to indicate incoming internal and external calls.

Navigation Keys

The functions of each are described briefly below.

- **Up Key:** To scroll upwards while navigating the Menu/sub-menu or to access Phone Settings and set the Ringtone (when phone is in the idle state).
- **Down Key:** To scroll downwards while navigating the Menu/sub-menu.
- **Forward Key:** To move forward when dialing a number or scroll to view the Context Sensitive Key options.
- **Back Key:** To move backwards when dialing a number, to go back one level in the Menu or scroll backwards to view the Context Key options.
- **Menu or Select / OK Key** : To enter the Menu; when the phone is in the idle state (without any incoming or outgoing call being made).

  Menu Key functions as the **Select / OK** Key to make a selection from the Menu/sub-menu options or to complete an action. When there is an incoming call it functions as the **Answer** Key.
- **Cancel Key** : To Cancel all features set by you or exit the Menu/sub-menu.

Feature Keys

There are 8 Feature keys assigned to important or frequently accessed features of SARVAM UCS.

Feature icon	Assigned Feature	LED	Programmable
	Hold	No	Yes
	Redial	No	Yes
	Transfer	No	Yes
	Conference	No	Yes
	Voicemail	Single Color - Blue	Yes
	Do Not Disturb	Single Color - Blue	Yes
	Headset	Single Color - Blue	No
	Mute	Single Color - Blue	No

These keys are programmable. However, as you cannot change the labels avoid programming these keys.

For instructions on programming these keys, see “DSS Key Settings” under “Configuring Matrix SPARSH VP510”.

A few of these Feature keys are equipped with an LED to indicate the status of the feature assigned to it. You may re-assign other features to these keys. We recommend you to assign those features that require LED indication to the Feature keys equipped with an LED.

Not all features require LED indication. For example, the feature Call Pick-Up does not require an LED. So when you assign this feature to a feature key, the LED of the key remains inactive, when Call Pick-Up is accessed.

Feature like Auto Redial requires an LED to show that it has been set or canceled. So, the LED of the Feature key to which the Auto Redial feature has been assigned will glow Blue, when Auto-Redial is set, and the LED is turned off when the feature is canceled.

Direct Station Selection (DSS) Keys

These are 16 programmable keys that can be assigned to Stations and Trunks and important or frequently accessed features of SARVAM UCS.

For instructions on programming these keys, see “DSS Key Settings” under “Configuring Matrix SPARSH VP510”.

DSS Key LEDs

Each DSS key is equipped with an LED which glows to indicate the status of the Trunk/Extension or Feature assigned to it.

- **Status of Extensions and Trunks:** The LED of DSS keys assigned to Extensions/Trunks glow in three colors to indicate status of the call event on the Extensions/Trunks and on the Extended IP Phone. Thus, the status of the Extended IP Phone user's own Extension as well as that of the other Extensions (i.e. Extended IP Phones, SLTs) and the status of Trunk lines are indicated by the LED of the DSS keys assigned to those Extensions and Trunks on the Extended IP Phone.

  The following table shows the relationship between the color of the LED and various events:

LED Color	LED Mode		
	Continuously ON	Slow Blink	Fast Blink
Blue	The key assigned to the Extension you are in speech with.	The key assigned to the Extension you have kept on hold.	The key assigned to the Extension you are calling or from which you are being called.
Red	The key assigned to the Extension that is now busy with another Extension/ Trunk.	The key assigned to the Extension which has put another Extension/Trunk on hold.	The key assigned to the Extension/Trunk that is called or being called by another.
Violet	You are talking on a Trunk (external call)	You have held a Trunk (external call)	You have an incoming call on the Trunk (external call)

  - **Blue** indicates the state of the extension/trunk you access. For example, when you make a call to another Extension 203, the LED of the DSS key assigned to Extension 203 blinks Blue to indicate ringing at the Extension. If you have successfully established speech with Extension 203 the LED glows Blue continuously.

  - **Red** indicates the state of other Extensions/Trunks. For example, if the LED of the DSS key assigned to Extension 201 is glowing Red continuously, it means Extension 201 is busy with another Extension or Trunk.

  - **Violet** indicates the state of the trunk you are in speech with. For example, when you are in speech on an outgoing call on Trunk 1 the LED of the DSS Key assigned to Trunk 1 will be continuously ON. When you put the call on hold, the LED will blink slowly.

- **Status of Features:** The LED of a DSS key is activated when the feature assigned to this key is used.

  Not all features require LED indication. For example, the feature Call Pick-Up does not require an LED. So when you assign this feature to a DSS key, the LED of the key remains inactive, when Call Pick-Up is accessed.

  Feature like Auto Redial requires an LED to show that it has been set or canceled. So, the LED of the DSS key to which the Auto Redial feature has been assigned will glow Red, when Auto Redial is set, and the LED is turned off when the feature is canceled.

The LEDs of the Call Appearance (CA) Keys will function in the same manner as the DSS Key LEDs.

Dial Pad

The dial pad consists of 12 fixed keys for the digits 0, 1-9, and the characters Star (*), Hash (#), Lock ( ), Plus (+) and Dot. The dial pad is used for dialing numbers of stations, external parties, and for dialing the programming and feature access codes.

Speaker Key

The speaker key sets the phone in 'Speaker mode' for hands-free operation.

Speaker Key LED

The Speaker Key on the phone is equipped with a single color LED which glows Blue when pressed for the speaker mode and is turned off, when you exit the speaker mode.

Volume Keys

- **"+" (plus)**: This is the increase key, to raise the volume of speech while talking and to increase the Ringer volume, when the phone is ringing.
- **"-" (minus)**: This is the decrease key, to lower the volume of speech while talking and to decrease the Ringer volume when the phone is ringing.

Headset Connectivity

The phone provides two Headset interfaces: A 3.5mm Audio Jack and an RJ9 connector at the bottom of the phone body.

So you can use any stereo headset of standard make with a 3.5 mm single connector or a headset with an RJ9 connector.

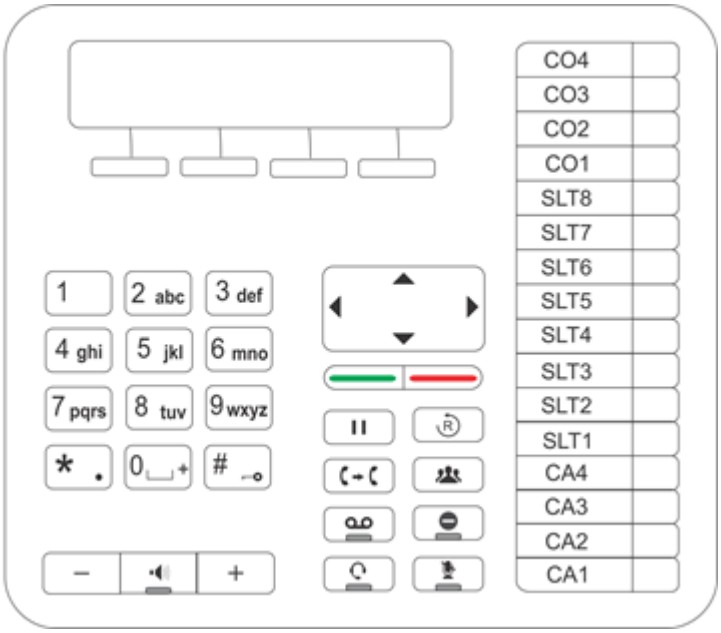
To use the Headset, a Headset Key is assigned on the phone. The Headset Key on the phone is equipped with a single color LED which glows Blue when pressed to indicate that the Headset mode is turned on and is turned off, when you press it again to indicate that you have exit the headset mode.

You can also program any of the DSS keys to function as the Headset key. For instructions on programming the key, see “DSS Key Settings” under “Configuring Matrix SPARSH VP510”.

Key Maps

As SPARSH VP510 may be the extension of the Operators and Executives in an enterprise to meet the varied requirements of each user group, these key maps can be customized to match the exact requirement of individual users. For instructions on customizing the Key Maps, see “Customizing Extended IP Phone Templates using Jeeves”. You can also personalize the key maps for each location, for instructions, see “DSS Key Settings” under “Configuring Matrix SPARSH VP510”.

Matrix Extended IP Phone, SPARSH VP510 Key Template (default)



The key maps of the Operator and Executive 1, 2, 3 are the same.

By using Key Templates you can prepare and assign common key maps to all or as many Extended IP Phones as you want, at one go.

SARVAM UCS also offers the flexibility to personalize the Key Maps of each Extended IP Phone, instead of using the Key Templates. For example, if you have assigned a common Executive Key Template to 12 Extended IP Phones, but you want to reassign some of the keys on two of these Extended IP Phones, SARVAM UCS allows you to selectively personalize the key maps of these two Extended IP Phones.

Phone Menu

To access the Phone Menu, press the Enter Key. You can access the following System and phone features from the phone:

Menu option	Description
Call Logs	To view call history of internal and external Missed, Answered and Dialed calls. You can also edit numbers in the call logs and store them in the Personal Directory.
Contacts	To add, edit, delete names and numbers of contacts in the Global Directory Part 1.
Call Forward	To set and cancel Call Forward-Unconditional, Call Forward-Busy, Call Forward No Reply, Forward On Busy/No Reply and Follow Me.
Dynamic Lock	To change the Toll Control level of the phone.
User Status	To set User Present or User Absent and Presence Status.
Keypad Lock	To lock the keypad of the phone.
Do Not Disturb	To set/cancel Do Not Disturb on the phone, that is, block incoming internal and external calls.
Call Cost Display	To view the cost of calls made from the phone.

Menu option	Description
Hotline	To set/cancel Hotline and Delayed Hotline.
Alarm	To set/cancel Personalized and Automated Alarms.
Change User Password	To change User Password.
One Touch Transfer	To set/clear the fixed destination number for One Touch Transfer.
Phone Settings	To customize settings of the phone.

When the phone is in idle state,

- Press the Down key ⋁ to access the Network Settings.
- Press the Up key ⋀ , if you wish to change the Ringtone and Play Key Tone.

Navigating the Phone Menu

To navigate the menu,

- Press the Menu Key when the phone is idle.
- Scroll by pressing the Up/Down Navigation Key to reach the desired Menu option.
- Press the Select / OK Key to select the desired Menu option.
- Scroll by pressing the Up/Down Navigation Key to reach the desired sub-menu option.
- Press the Select / OK Key to select the desired sub-menu option.

To exit menu,

- Press Cancel Key.
  or
  Go ON-Hook.

Call Waiting Indication

During an on-going call, if there is another incoming call, an indication will be provided to you for the waiting call.

The call waiting indication depends on the **Call Waiting Tone (for SPARSH VP248/VP310/VP510)** option you select in General Parameters under SIP Extensions. See "Configuring Matrix SPARSH VP510" for instructions.

Connecting SPARSH VP510

For detailed instructions to connect SPARSH VP510, see *"Connecting SPARSH VP510 as Extended SIP Extension"* in SIP Extensions for ETERNITY LENX, *"Connecting SPARSH VP510 as Extended SIP Extension"* in SIP Extensions for ETERNITY MENX, *"Connecting SPARSH VP510 as Extended SIP Extension"* in SIP Extensions for ETERNITY GENX and *"Connecting SPARSH VP510 as Extended SIP Extension"* in SIP Extensions for ETERNITY PENX.

For detailed instructions on how to configure SPARSH VP510, see *"Configuring Matrix SPARSH VP510"*.

Operating SPARSH VP510

Please refer the *EON510_SPARSH VP510 User Guide* for instructions on operating the features of SARVAM UCS.

Matrix Extended SPARSH VP710

Extended SPARSH VP710, the Smart Video IP Deskphone is engineered to deliver a seamless communication solution to the user with experience of an android touch screen. Extended SPARSH VP710 is an integration of SPARSH VP710, an android based deskphone with VARTA ADR100 application. This tight integration of the UC Client, VARTA ADR100 with SPARSH VP710 offers advance calling capabilities.

The IP Phone provides an easy way of managing the modern communication needs for meeting the day to day business requirements. With sophisticated looks and innovative features, it is the next step towards collaboration for offering flexibility and convenience in day to day communication. Deskphone is the perfect client to extract the capability of server and with a smarter deskphone you can optimize the return over investment by utilizing the UC features of the IP Phone. Once it is registered with the SARVAM UCS, you can start operating the IP Phone.

Key Features

- **Enhanced Call Management:** Dedicated one touch feature keys and intuitive user interface provides quick access to full range of PBX call management features including Call Hold, Call Park, Call Transfer, Conference and Voicemail.

  The IP Phone also provides an easy way for businesses to integrate their enterprises’ voice solutions within the Android OS family.

- **System Extension:** The IP Phone becomes an extension of the System. It provides users with a quicker and more user-friendly access to phone facilities, helping businesses optimize their employees' productivity.

- **Dial by Extension:** Flexibility to communicate with colleagues by dialing their respective extension numbers.

- **Smart Directory Access:** Provides you with the easy and quick way to access the extensions and other contacts through Smart Directory.

- **Presence:** You can set your presence status and view other extension users' presence statuses.

- **Voicemail Access:** Access to the corporate Voicemail System from any location ensures no opportunity is lost.

- **Multiple Call Support:** Easy handling of multiple incoming calls by keeping the ongoing call on hold and attending the higher priority call first. It also supports merging of calls to initiate a conference or splitting the conference to attend the calls separately.

- **Video Calling:** Video calling provides you the facility to make video calls to anyone, anywhere in the world. This makes it easier to conduct business meetings, discussions, demonstrations and presentations between people working at different locations.

- **Handover and vice-versa:** Using handover you can automatically move an active call from the IP Phone to your cellular number on the cellular network and vice-versa, without disconnecting the call and/or having to redial.

- **Busy Lamp Field (BLF)**: Using BLF you can monitor the status of another extension or trunk and confirm whether it is available, busy, ringing or on hold.

- **IM and SMS:** Using this feature, you can send/receive IMs and SMSs to/from remote users.

- **One Touch Transfer:** You can transfer the ongoing call to a fixed extension without entering the number of that extension and without putting the call on hold. Similarly, you can also transfer a call from the fixed extension to your IP Phone.

- **Better Voice Quality:** Using customized codec settings, enhanced voice output is available. If you are aware of the bandwidth and the network criteria of your location, you can select the appropriate codec to get high quality voice output.

- **Standard Phone Features:** Provides intuitive access to Keypad, Contacts, Call Logs and more based on the Native Android design. One-touch access to call feature options during VoIP (Voice over IP) calls including Adding a New Call, Mute, Hold, Transfer and Speaker-phone.

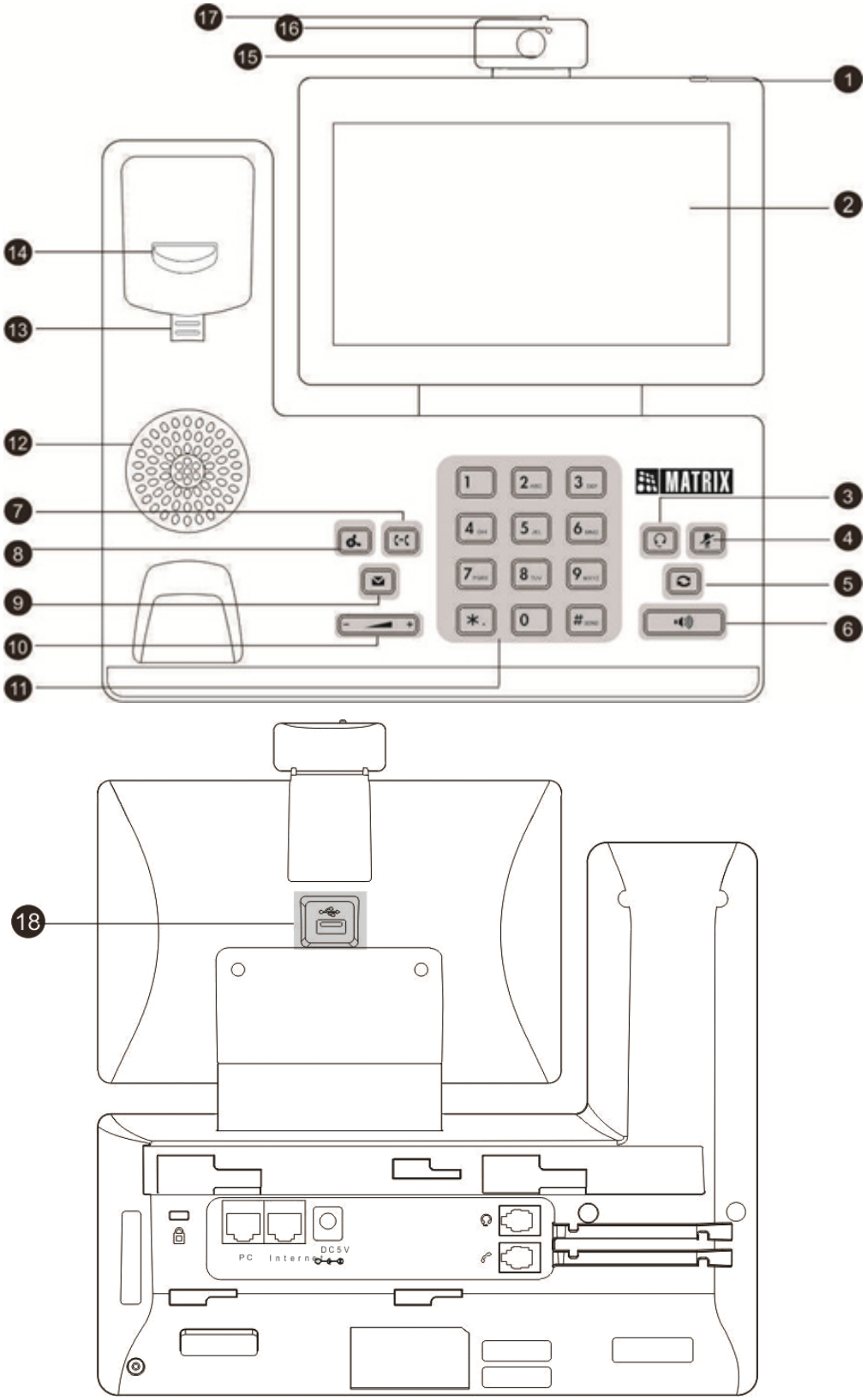
- **Cost Effective Calling:** If you are using the enterprise Wi-Fi or Ethernet network to register the IP Phone with the System, calls made will be almost free.

- **Wi-Fi Support:** Supports Wi-Fi (WLAN) connectivity using which the IP Phone provides seamless connectivity to the corporate Wi-Fi network and offers flexibility to work from anywhere in the office. If your installation setup does not meet the requirements of suitable wired Ethernet connectivity due to any reason, then you can register the IP Phone through the Wi-Fi Network.

- **Advanced Call Capabilities:** Provides access to the features such as Callback, Dial-in Conference, Conversation Recording and many more.

If any Extended SPARSH VP710 extension is Off-Hook, the system will not be able to detect it. However, if any SLT/DKP is Off-Hook, Off-Hook Alert will be provided to Extended SPARSH VP710 if it is the Operator phone.

Extended SPARSH VP710, Front View and Bottom View



Key Label	Item	Description
1	Power Indicator LED	Indicates the status of calls, messages and voicemails. Also displays the registration status of the IP Phone.
2	Touch Screen	7 inch (1024 x 600) capacitive (5 point) touch screen. Tap to select and highlight screen items.
3	Headset Key	Toggles and indicates the headset mode. The key LED illuminates solid green when you activate the headset mode.
4	Mute Key	Toggles and indicates mute feature. The key LED illuminates solid red when you mute a call.
5	Call Log Key	Displays all the missed, received and dialed calls.
6	Speaker Key	Toggles and indicates the speaker mode. The key LED illuminates solid green when you activate the speaker mode.
7	Transfer Key	Transfers a call to another party.
8	Hold Key	Places a call on hold or resumes a held call.
9	Voicemail Key	Accesses voice mails.
10	Volume Key	Adjusts the volume of the handset, headset, speaker, ringer or media.
11	Keypad	Provides the digits and special characters.
12	Speaker	Provides speaker audio output.
13	Hookswitch Tab	Secures the handset in the handset cradle when the IP Phone is mounted vertically.
14	Hookswitch	Picking up the handset from the handset cradle, the hookswitch bounces and the phone connects to the line. Laying the handset down on the handset cradle, the phone disconnects from the line.
15	Camera Lens	2 Mega-pixel camera. Provides near-site video. The better distance between camera and images you want to capture should be in the range of 0.35 meters (1 foot) to 2 meters (6 feet).
16	Camera Indicator LED	Indicates the status of camera and video calls.
17	Shutter Switch	Covers and uncovers the camera. When the camera is switched off, the video image is black.
18	USB2.0 Port	Allows you to connect the USB flash drive/USB headset to the phone.

LED Indications

Power Indicator LED

LED Status	Description
Solid Red	When the Phone is not registered.

LED Status	Description
Fast Flashing Red	When the Phone is in ringing state.
Slow Flashing Red	When the Phone receives a missed call, message or voice mail.
Off	When the Phone is powered off. When the Phone is in busy state. When the Phone is idle. When the call is placed on hold. When the call is muted.

Camera Indicator LED

LED Status	Description
Solid Green	When the Phone is powered on and the camera is connected properly. When the camera is idle. When the phone receives an audio call.
Solid Red	When the Phone receives a video call. When there is an active video call. When the video call is muted. When the video call is placed on hold.
Off	When the Phone is powered off. When the camera is not connected properly. When the camera shutter switch is closed.

Connecting Extended SPARSH VP710

For detailed instructions to connect Extended SPARSH VP710, see *"Connecting Extended SPARSH VP710 as Extended SIP Extension"* in SIP Extensions for ETERNITY LENX, *"Connecting Extended SPARSH VP710 as Extended SIP Extension"* in SIP Extensions for ETERNITY MENX, *"Connecting Extended SPARSH VP710 as Extended SIP Extension"* in SIP Extensions for ETERNITY GENX and *"Connecting Extended SPARSH VP710 as Extended SIP Extension"* in SIP Extensions for ETERNITY PENX.

Configuring Extended SPARSH VP710

For detailed instructions on how to configure Extended SPARSH VP710, see *"Configuring Matrix Extended SPARSH VP710"*.

Operating Extended SPARSH VP710

Please refer the *EXTENDED SPARSH VP710 User Guide* for instructions on operating the features of SARVAM UCS.

Matrix SPARSH VP210

SPARSH VP210, the Entry Level IP Phone sets the benchmark for quality performance with elegant design and crystal-clear voice. SPARSH VP210 features a 3.1” Graphical LCD Display, SIP Line Keys, High Quality speaker-phone and high definition audio quality.

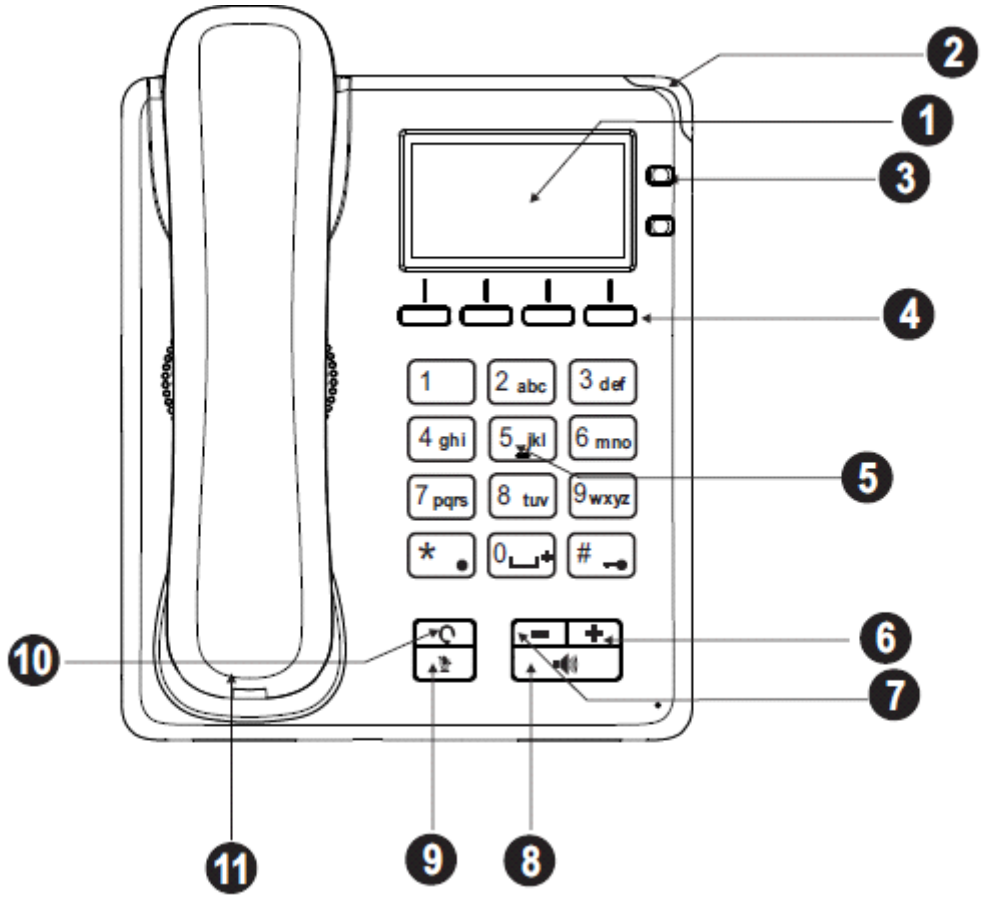
The State-of-the-art Deskphone is best suited for usage in lobbies, cafeterias, conference centers wherein the basic level endpoint security is sufficient. It can also be used by Administrative Staff, Hospitality guest rooms, knowledge workers etc.These phones offer flexibility to streamline communication and attain higher return over investment.

Key Features

- 128 x 64 Graphical LCD
- LED for Call and Message Wait Indication
- HD Voice, HD Handset, HD Speaker
- 4 Context Sensitive Keys
- 3 feature keys: Headset, Mute, Hands-free speaker phone
- Fixed Function Keys (Without LED) — Hold, Conference, Redial, Transfer
- Tight integration with Server over SIP protocol /Proprietary Protocol
- HTTP Auto Provisioning
- Dual Color illuminated LED for line status
- One Touch Transfer
- Call logs
- Ringtone selection
- Wideband Codec : G722
- Narrowband Codec: G.711(A/μ), G.729, G.726, G.723
- VAD, CNG, AEC, AJB, AGC
- Full Duplex speaker phone with AEC

- IP Assignment : Static / DHCP
- TCP/ DNS-SRV
- AEC encryption for config file
- IEEE802.1x
- RJ9 headset port
- Dual port 10/100 Mbps Ethernet
- Stand with 2 adjustable angles
- PoE (IEEE 802.af) class2

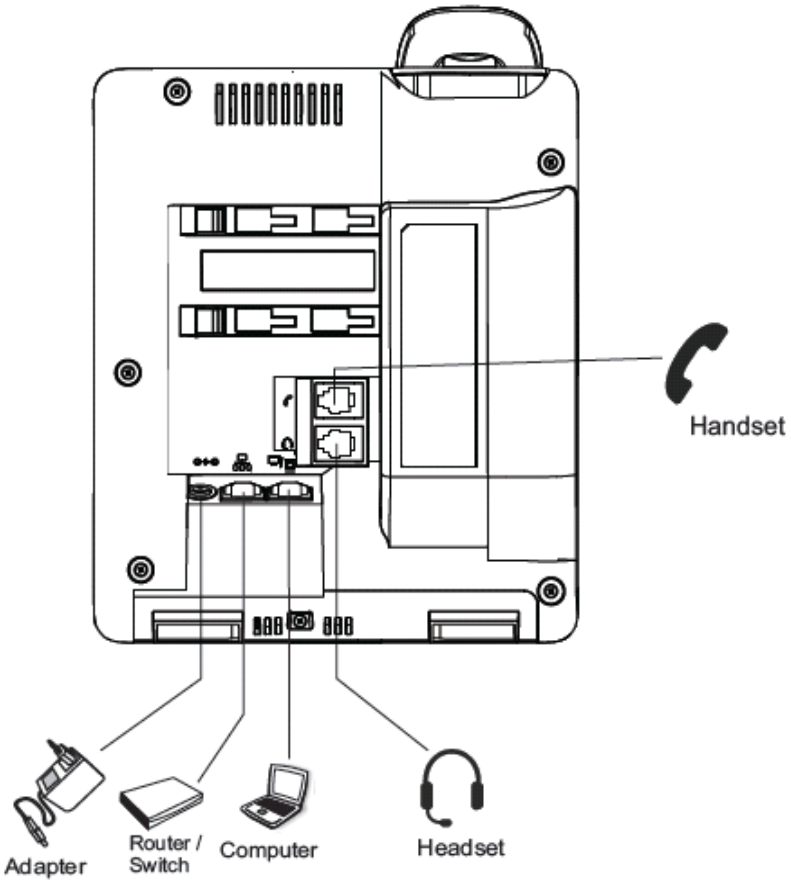
Front View



1	LCD Screen
2	Ringer LED
3	Navigation/Notification Keys
4	Context Sensitive Key
5	Dial Pad
6	Volume Increase Key
7	Volume Decrease Key
8	Speaker Key
9	Mute

10	Headset Key
11	Handset

Bottom View



It is a PoE enabled phone and can be powered over Ethernet by connecting it to a PoE enabled LAN Switch (IEEE 802.3af Compliant). If you do not want to use PoE, plug in the connector of the Adapter into the power jack (DC Jack) at the bottom of the phone. The IP phone should be used with Matrix original power adapter (5V/0.6A) only.

LCD Display

The LCD display of the phone is Dot Matrix Graphic LCD. The LCD backlight can be turned on and off as well as adjusted for contrast and brightness from the Phone Menu.

Ringer LED

The Ringer LED will glow in Blue (1 sec ON – 500 msec OFF) to indicate incoming internal and external calls.

Feature Keys

here are 3 Feature Keys. Each Feature Key is accompanied by a feature icon that describes its function. Default features assigned to these keys are as follows.

Feature icon	Assigned Feature	LED	Programmable
	Headset	No	No
	Mute	No	No
	Speaker	No	No

Navigation/Notification Keys

There are 2 Navigation Keys, Up/Down Keys.

When the phone is in idle state these keys are used for accessing the Notifications - Call Back, Auto Redial, Trunk Reservation, Contact Sync.

You can navigate sideways using the context keys, that is, **Left Navigation <** Key or **Right Navigation >** Key.

Dial Pad/Key Pad

The dial pad consists of 12 fixed keys for the digits 0, 1-9, and the characters Star (*), Hash (#), Lock ( ), Plus (+) and Dot. The dial pad is used for dialing numbers of stations or external parties.

Speaker Key

The speaker key sets the phone in 'Speaker mode' for hands-free operation.

Volume Keys

- **"+" (plus)**: This is the increase key, to raise the volume of speech while talking and to increase the Ringer volume, when the phone is ringing.
- **"-" (minus)**: This is the decrease key, to lower the volume of speech while talking and to decrease the Ringer volume when the phone is ringing.

Headset Connectivity

The phone provides an RJ9 connector at the bottom of the phone body to connect a headset.

To use the Headset, a Headset Key is assigned on the phone. Make sure you have enabled the **Use Headset** option.

Phone Menu

You can access the following PBX and phone features from the Menu of the phone:

Menu option	Description
Call Logs	To view call history of internal and external Missed, Answered and Dialed calls.
Contacts	To add, edit, delete names and numbers of contacts in the Global Directory Part 1.
Call Forward	To set and cancel Call Forward - Unconditional, Call Forward - Busy, Call Forward - No Reply, Call Forward - Busy/No Reply, Call Forward - Not Registered.
Dynamic Lock	To change the Toll Control level of the phone.
User Status	To set User Present or Absent.
Keypad Lock	To lock the keypad of the phone.
Do Not Disturb	To set/cancel Do Not Disturb on the phone, that is, block incoming internal and external calls.
Hotline	To set/cancel Hotline and Delayed Hotline.
Alarm & Reminder	To set/cancel Personalized and Automated Alarms/Reminders.
One Touch Transfer	To set/clear the fixed destination number for One Touch Transfer.
Pickup	To configure as well as access Group/Selective Call Pick-up
Voicemail	To access your Mailbox.
Dial-In Conference	To schedule as well as establish the Conference.
Call Retrieve	To retrieve a call parked in the Personal or General Orbit.
CLIR	To set/cancel CLIR.
Call Supervision	To configure as well as access Call Supervision.
Message Wait	To set/cancel Message Wait.
Paging	To configure the Page Zone and make the announcement.
Meet Me Paging	To access Meet be Paging.
Room Monitor	To configure and access Room Monitor.
Intercom	To configure and access Intercom.
Follow Me	To set Follow Me.
Walk-in	To set/cancel Walk-in.
PIN Dialing	To make calls using PIN.
Department Group Call Forward	To set/cancel Department Group Call Forward.
Open Cosec Door	To open the Cosec Door Lock.
Settings	To change the following settings: • User Password: To change User Password. • Phone Settings: To customize settings of the phone. • Network Settings: To change Network Settings. • PCAP: To Start/Stop PCAP
Phone Info	Displays the phone information.

Navigating the Phone Menu

To navigate the menu,

- Press the **Menu** Key when the phone is idle.
- Scroll by pressing the **Up/Down Navigation** Key to reach the desired Menu option.
- Press the **Select** Key to select the desired Menu option.
- Scroll by pressing the **Up/Down Navigation** Key to reach the desired sub-menu option.
- Press the **Select** Key to select the desired sub-menu option.

To exit menu,

- Press **Back** Key.
  or
  Go ON-Hook.

  To scroll Up or Down you need to use the **Up/Down Navigation** Keys. To scroll sideways, you need to use the **Left Navigation <** Context Key or **Right Navigation >** Key.

Context Specific Keys (CSK)

SPARSH VP210 has the provision to program the four Context Keys. These keys enable you to access the most frequently used functions/features at the press of a single button.

You can configure these Keys from SARVAM UCS Jeeves only.

The screens — Idle Screen, Ringing Screen, Busy Screen, Call Screen, Conversation Recording Screen, all have different set of features that can be accessed. SPRASH VP210, enables you to customize these by allowing you to set the priorities of the features in each type of screen as per your preference. You can assign the features to the Context Keys depending on the state of the call.

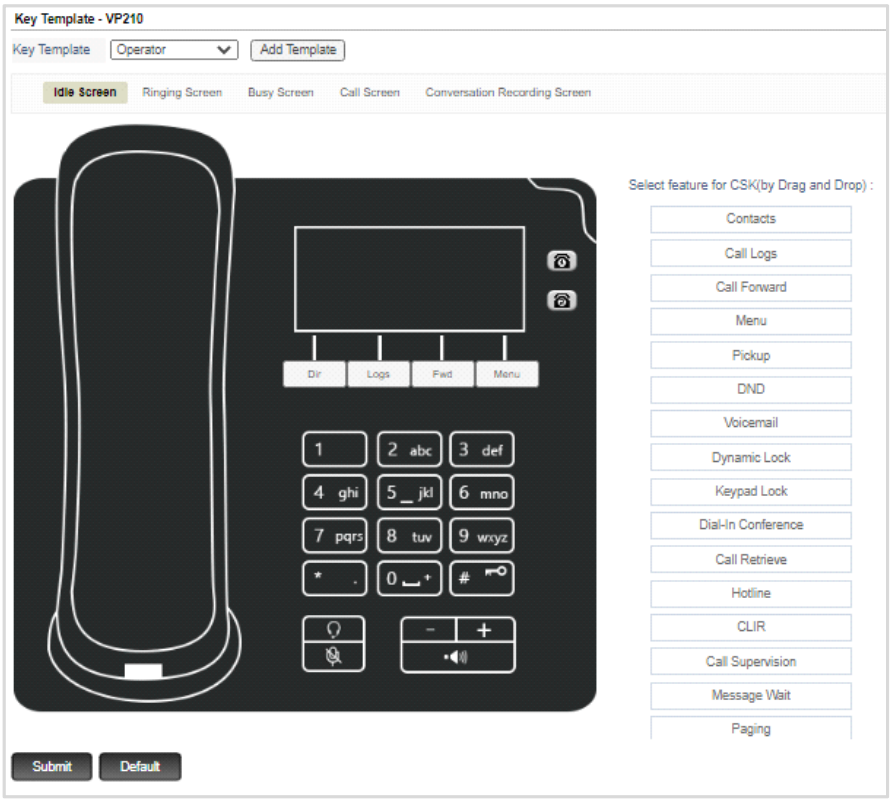
- In the Idle Screen you can assign the desired feature/function to the Context Keys as well as set their priorities as per your requirement.
- In the other Screens you can only set the priorities of the features.

Refer to “Key Maps”, “Customizing Extended IP Phone Templates using Jeeves” as well as “DSS Key Settings” in “Configuring Matrix SPARSH VP210”.

Key Maps

As SPARSH VP210 may be the extension of the Operators and Executives in an enterprise to meet the varied requirements of each user group, these key maps can be customized to match the exact requirement of individual users. For instructions on customizing the Key Maps, see “Customizing Extended IP Phone Templates using Jeeves”. You can also personalize the key maps for each location, for instructions, see “DSS Key Settings” in “Configuring Matrix SPARSH VP210”.

Matrix Extended IP Phone, SPARSH VP210 Key Template (default)



The key maps of the Operator and Executive 1, 2, 3, Hotel Attendant as well as Guest are the same.

By using Key Templates you can prepare and assign common key maps to all or as many Extended IP Phones as you want, at one go.

SARVAM UCS also offers the flexibility to personalize the Key Maps of each Extended IP Phone, instead of using the Key Templates. For example, if you have assigned a common Executive Key Template to 12 Extended IP Phones, but you want to reassign some of the keys on two of these Extended IP Phones, SARVAM UCS allows you to selectively personalize the key maps of these two Extended IP Phones.

Connecting SPARSH VP210

For detailed instructions to connect SPARSH VP210, see “Connecting SPARSH VP210 as Extended SIP Extension” in SIP Extensions for ETERNITY LENX, “Connecting SPARSH VP210 as Extended SIP Extension” in SIP Extensions for ETERNITY MENX, “Connecting SPARSH VP210 as Extended SIP Extension” in SIP Extensions for ETERNITY GENX and “Connecting SPARSH VP210 as Extended SIP Extension” in SIP Extensions for ETERNITY PENX.

Configuring SPARSH VP210

For detailed instructions on how to configure SPARSH VP210, see “Configuring Matrix SPARSH VP210”.

Operating SPARSH VP210

Refer the *SPARSH VP210 (Extended) User Guide* for instructions on operating the features of SARVAM UCS.

Matrix VARTA ADR100 UC Client

Matrix VARTA ADR100 is a proprietary SIP (Session Initiation Protocol) based UC Client running on Android Phones/Tablets, delivering full-array of Matrix SARVAM UCS features to the user on-the-go along with an added advantage of video calling. Through tight integration with the enterprise mobility features of the SARVAM UCS, Matrix VARTA ADR100 provides advance call capabilities including Conferencing, Corporate Directory Access (Global Directory), Call Logs and Conversation Recording with one-touch access. Other than these you can take the advantage of using UC features like Presence, IM, IM to SMS, Corporate VMS access to enhance your overall mobile experience.

Mobile workers can use any Wi-Fi or cellular data networks to stay connected with business communications while working from office, home or traveling to any location. An innovative and easy to understand user interface delivers all productivity features at fingertips that enhance speed of communication and collaboration with office users and customers.

Make sure the phone/tablet in which you install Matrix VARTA ADR100, runs on Android V5.0 or later.

To use MATRIX VARTA ADR100 make sure you have:

- Purchased and activated the VARTA Essential or VARTA Professional or VARTA Collaboration license. For more details, see “License Management”.
- Assigned the desired license to the SIP Extension. For more details, see “VARTA License Management”.

Key Features

- **System Extension:** Matrix VARTA ADR100 becomes a mobile extension of the SARVAM UCS. As an increased number of business professionals are using the collaborative tools found in smartphone devices to help them in their work activities, this application offers businesses an easy way to integrate their enterprises’ voice solutions within the Android OS family.

- **Advanced Call Capabilities:** Access to features such as Callback, Dial-in Conference, Conversation Recording and many more.

- **Mobility:** Matrix VARTA ADR100 provides you the mobility that you need in today's highly competitive business environment; with the ability to access SARVAM UCS features easily once you are connected to

either Wi-Fi or 3G network. Considering the case of roaming users, one can register Matrix VARTA ADR100 with the SARVAM UCS using the enterprise Wi-Fi network when working within the office (that is within the organization's dedicated Wi-Fi coverage area). While working out of the office (where Wi-Fi network may not be available), one can register the application using the Mobile Data (3G) network.

- **Single Number Reach:** Retains the identity of the corporate phone system while working away from the office; so enhances business collaboration and lowers communication delays.
- **Dial by Extension:** Flexibility to reach to office users with direct extension number dialing.
- **Corporate Directory Access:** Enhance business collaboration with one-touch access to the Corporate Directory contacts using the SARVAM UCS Global Directory.
- **Video Support:** The application offers the added advantage of Video Calling.
- **Presence:** Supports changing your Presence status as well as you can view the Presence status of other extension users.
- **Busy Lamp Field (BLF):** Using BLF you can monitor the status of another extension or trunk and confirm whether it is available or busy or ringing or on hold.
- **IM and SMS:** The application allows you to send IM and SMS to remote users.
- **One Touch Transfer:** You can transfer an ongoing call to a fixed extension without entering the number of that extension and without putting the call on hold. Similarly, you can also transfer a call from the fixed extension to your application.
- **Voicemail Access:** Access to the corporate Voicemail System from any location ensures no opportunity is lost.
- **Multiple Call Support:** With multiple call support, you can easily handle multiple incoming calls, merge and split calls apart, and place users on hold with a simple tap. With this Android Application, it’s like taking your deskphone on the road.
- **Wi-Fi to Cellular Handover and vise versa:** The application can automatically move an active call from the application to your cellular number on the cellular network and vise versa, without disconnecting the call and having to redial.
- **Better Voice Quality:** Using customized codec settings, enhanced voice output is available. If you are aware of the bandwidth and the network criteria of your location, you can select the proper codec from the application to get high quality voice output.
- **Standard Telephone Features:** Provides intuitive access to Keypad, Contacts, Call Logs and more, based on the Native Android design. One-touch access to call feature options during VoIP (Voice over IP) calls including Adding a New Call, Mute, Hold, Transfer and Speakerphone. Also provides DTMF support to enter numbers using an Auto Attendant.
- **Cost Effective Calling:** If you are using the enterprise Wi-Fi network to register Matrix VARTA ADR100 with the SARVAM UCS; calls made from the application will be almost free. Even if you are using the application via 3G network during roaming, external calls can be made using the SARVAM UCS trunks and thus reducing mobile calling and roaming charges.

- **Multiple Language Support:** The application can be viewed in six different languages including English, French, German, Spanish, Portuguese and Italian.

- **Application diagnostics:** Supports logging and sending of log files to concerned recipients by e-mail. These logs are used by the system engineer and/ or Matrix support engineers for troubleshooting.

Installing VARTA ADR100

For detailed instruction to install VARTA ADR100, refer to the *Matrix VARTA ADR100 User Guide*.

Configuring VARTA ADR100

For detailed instructions on how to configure VARTA ADR100, see “Configuring Matrix VARTA ADR100/AMP100 UC Clients”.

Operating VARTA ADR100

Refer to *Matrix VARTA ADR100 User Guide* for instructions on operating the features of SARVAM UCS.

Matrix VARTA AMP100 UC Client

Matrix VARTA AMP100 is a proprietary SIP (Session Initiation Protocol) based UC Client running on iPhones, delivering full-array of Matrix SARVAM UCS features to the user on-the-go along with an added advantage of video calling. Through tight integration with the enterprise mobility features of the SARVAM UCS, Matrix VARTA AMP100 provides advanced call capabilities including Conferencing, Corporate Directory Access (Global Directory), Call Logs and Conversation Recording with one-touch access. Other than these, you can take the advantage of using UC features like Presence, IM, IM to SMS, Corporate VMS access to enhance your overall mobile experience.

Mobile workers can use any Wi-Fi or cellular data networks to stay connected with business communications while working from office, home or travelling to any location. An innovative and easy to understand user interface delivers all productivity features at fingertips that enhances speed of communication and collaboration with office users and customers.

Make sure the phone in which you install Matrix VARTA AMP100, runs on **iOS9**.

Key Features

- **System Extension:** Matrix VARTA AMP100 becomes a mobile extension of the SARVAM UCS. As an increased number of business professionals are using the collaborative tools found in smartphone devices to help them in their work activities, this application offers businesses an easy way to integrate their enterprises' voice solutions within the iOS family.

- **Advanced Call Capabilities:** Access to features such as Callback, Dial-in Conference, Conversation Recording and many more.

- **Mobility:** Matrix VARTA AMP100 provides you the mobility that you need in today's highly competitive business environment; with the ability to access SARVAM UCS features easily once you are connected to either Wi-Fi or 3G network. Considering the case of roaming users, one can register Matrix VARTA AMP100 with the SARVAM UCS using the enterprise Wi-Fi network when working within the office (that is within the organization's dedicated Wi-Fi coverage area). While working out of the office (where Wi-Fi network may not be available), one can register the application using the Mobile Data (3G) network.

- **Single Number Reach:** Retains the identity of the corporate phone system while working away from the office; so enhances business collaboration and lowers communication delays.

- **Dial by Extension:** Flexibility to reach to office users with direct extension number dialing.

- **Corporate Directory Access:** Enhances business collaboration with one-touch access to the Corporate Directory contacts using the SARVAM UCS's Global Directory.

- **Video Support:** The application offers the added advantage of Video Calling.

- **Presence:** Supports changing your Presence status as well as viewing Presence status of other extension users.

- **Busy Lamp Field (BLF):** Using BLF you can monitor the status of another extension or trunk and confirm whether it is available or busy or ringing or on hold.

- **IM and SMS:** The application allows you to send IM and SMS to remote users. It also supports the Emoji keyboard to add Emoticons (Smileys) in your messages.

- **One Touch Transfer:** You can transfer an ongoing call to a fixed extension without entering the number of that extension and without putting the call on hold. Similarly, you can also transfer a call from the fixed extension to your application.

- **Voicemail Access:** Access to the corporate Voicemail from any location ensures no opportunity is lost.

- **Multiple Call Support:** With multiple call support, you can easily handle multiple incoming calls, merge and split calls apart, and place users on hold with a simple tap. With this iPhone application, it's like taking your deskphone on the road.

- **Cellular to Wi-Fi Handover and vice-versa:** You can move an active call from the Cellular number (on the Cellular network) to your application (registered using Wi-Fi network) without disconnecting or redialing the number. Similarly Wi-Fi to Cellular Handover is also possible where you can manually handover your call from the Wi-Fi to the Cellular network.

- **Better Voice and Video Quality:** Using customized codec settings and video quality preferences, enhanced voice output and video rendering are available. If you are aware of the bandwidth and the network criteria of your location, you can select the proper codec and video quality option from the application to get optimum audio and/or video output.

- **Standard Telephone Features:** Provides intuitive access to Keypad, Contacts, Call Logs and more. One touch access to call feature options during VoIP (Voice over IP) calls including Adding a New Call, Mute, Hold, Transfer and Speakerphone. Also provides DTMF support to enter numbers using an Auto Attendant.

- **Cost Effective Calling:** If you are using the enterprise Wi-Fi network to register Matrix VARTA AMP100 with the SARVAM UCS; calls made from the application will be almost free. Even if you are using the application via 3G network during roaming, external calls can be made using the SARVAM UCS trunks which reduces calling and roaming charges to a significant amount.

- **Multiple Language Support:** The application can be viewed in six different languages including English, French, German, Spanish, Portuguese and Italian.

- **Application Diagnostics:** Supports logging and sending of log files to concerned recipients by e-mail. These logs are used by the system engineer and/or the Matrix support engineers for troubleshooting.

Installing VARTA AMP100

For detailed instruction to install VARTA AMP100, refer to the *Matrix VARTA AMP100 User Guide*.

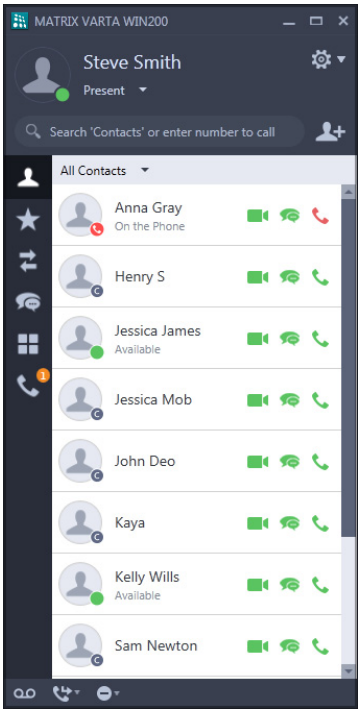
Configuring VARTA AMP100

For detailed instructions on how to configure VARTA AMP100, see “Configuring Matrix VARTA ADR100/AMP100 UC Clients”.

Operating VARTA AMP100

Refer to *Matrix VARTA AMP100 User Guide* for instructions on operating the features of SARVAM UCS.

MATRIX VARTA WIN200



MATRIX VARTA WIN200, is a SIP (Session Initiation Protocol) based Unified Communication Client running on Windows OS, delivering full-array of the System features to the user on-the-go along with an added advantage of video calling. Through tight integration with the enterprise features of the System, UC Client provides advance call capabilities including Conferencing, Corporate Directory Access (Global Directory), Call Logs and Conversation Recording with one-touch access. Other than these you can take the advantage of using UC features like Presence, IM, IM to SMS, Corporate VMS access to enhance your overall mobile experience.

To use MATRIX VARTA WIN200 make sure you have:

- Purchased and activated the VARTA Essential or VARTA Professional or VARTA Collaboration license. For more details, see “License Management”.
- Assigned the desired license to the SIP Extension. For more details, see “VARTA License Management”.

Installing MATRIX VARTA WIN200

For detailed instruction to install MATRIX VARTA WIN200, refer to the *MATRIX VARTA WIN200 User Guide*.

Configuring MATRIX VARTA WIN200

For detailed instructions on how to configure MATRIX VARTA WIN200, see “Configuring Matrix VARTA WIN200 UC Client”.

Operating MATRIX VARTA WIN200

Refer to *MATRIX VARTA WIN200 User Guide* for instructions on operating the features of SARVAM UCS.

Firebase Cloud Messaging (FCM) Support

What's this?

Firebase Cloud Messaging (commonly referred to as Android Push Notification or FCM) is a platform notification service created by Google LLC that enables third party application developers to send notification data to their applications installed on Android devices.

Previously, VoIP applications needed to maintain a persistent connection in order to receive calls. Keeping a connection open in the background, drains the battery as well as causes all kinds of problems when the application crashes or is terminated by users.

In Android 4.1 and above, Google has introduced FCM as part of their effort to improve battery life, performance and stability for VoIP applications such as Skype, WhatsApp, etc. FCM offers high-priority push notification with a large payload. The VoIP application receives the notification in the background, sets up the connection and displays a local notification to the user.

SARVAM UCS supports FCM for VARTA ADR100 Application only. Push Notifications will be sent for calls, new messages as well as for voicemail. Push Notifications will be sent to the MATRIX VARTA ADR100 Application only if it is in the background and when there is persistent internet connection. You will receive the Push Notifications even after you exit the application provided the check box *Calls and Messages after exit* is enabled in the VARTA ADR100 Application. For details refer to the VARTA ADR100 User Guide.

How it works

Pre-requisites for Push Notifications:

- Make sure that the server has a persistent internet connection and there is connectivity with the FCM Server. To check the connectivity, refer “FCM Connectivity”.
- Make sure the Date and Time of the server is synchronized with the NTP Server.
- To receive IM and IM notifications make sure the application is registered at Location 1. For more details, refer “Configuring Matrix VARTA ADR100/AMP100 UC Clients”.

Let us see how the notifications will be sent by the server when MATRIX VARTA ADR100 application is registered with the server as a SIP Extension and it is in the background. There is an incoming call or message:

- You can check the status of the SIP Extension user. It will display Registered (as the device is in the background) and under the respective Contact 1, 2, 3, it will display the time remaining for the expiry of the VARTA Client Inactivity Timer. The default value of the VARTA Client Inactivity Timer is 10 days. To configure this timer, refer to “System Timers and Counts”.

- The server will send a Push Notification to the MATRIX VARTA ADR100 application (client).

- The server will wait for 15 seconds after sending the Push Notification:

    - if the client registers with the server within this time, the call will be connected or the message will be delivered. The status of the SIP Extension will display Registered and under the respective Contact 1, 2, 3 it will display the SIP ID, IP Address and the Registration Expiry Timer.

    - if the client does not register with the server within this time, the call will be disconnected or the message will be rejected.The status of the SIP Extension will display Registered and under the

respective Contact 1, 2, 3 it will display the time remaining for the expiry of the VARTA Client Inactivity Timer.

- The server maintains a configurable timer, VARTA Client Inactivity Timer which is set as 10 days. Till the expiry of the timer the server will send Push Notifications to the application.
- If for this duration, the server does not receive any registration request from the application and the timer expires, the server will consider the application as unregistered and will stop all Push Notifications to the application.The status of the SIP Extension will display Not Registered and under the respective Contact 1, 2, 3 the details will be cleared. Calls and messages will be rejected.

  The server will start sending notifications to the application in the background after the application is bought in the foreground once and a registration request is received by the server.

Feature Interactions when the Application is in the Background

Call Forward when not Registered:

If the application receives an incoming call from a QSIG caller, the Call Forward functionality will not be applicable.

To know more, see “Call Forward-When Not Registered”.

Handover and Handoff:

VARTA ADR100 users will be able to use Handover but Handoff will not be possible. To know more, see “Handover and Handoff”

System Restart

After System Restart the VARTA Client Inactivity Timer will be reset.

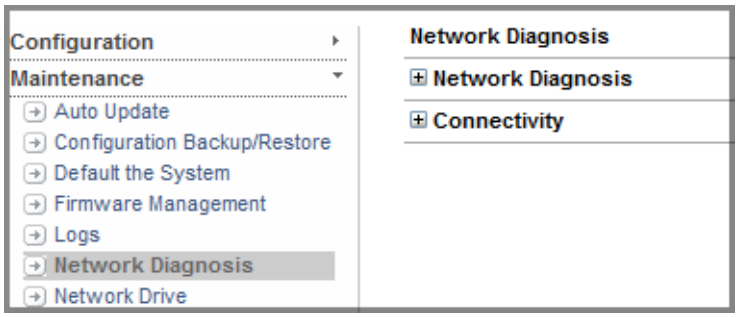
FCM Connectivity

A connectivity between the system and the FCM Server is required so that the Push notifications can be sent to the clients.

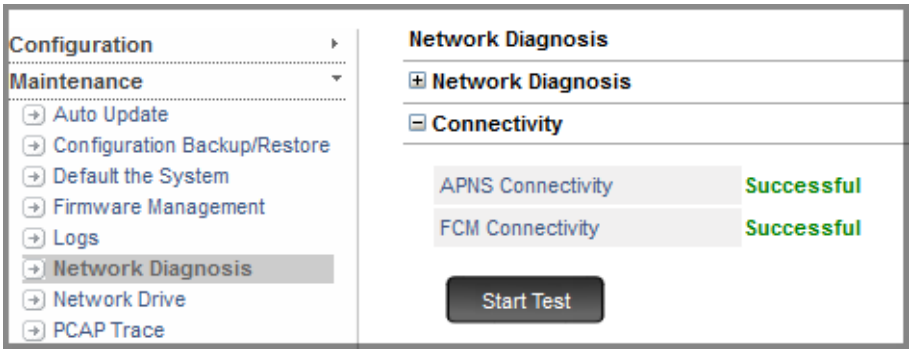
To check the FCM connectivity status,

- Log into Jeeves.
- Click the **Maintenance** link.

- Click the **Network Diagnosis** link



- Click **Connectivity** to expand.



- Click **Start Test**.

- In **Connectivity**, the status is displayed as:
  - Successful - if the connectivity between the system and the APNS/FCM Server is established
  - Timeout - if there is no connectivity.

*If the **Connectivity** Test of either of the servers (APNS or FCM) with SARVAM UCS is not successful, then Push Notifications will not be sent to the Mobile Clients — MATRIX VARTA ADR100 / AMP100 application.*

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Flash Timer

What is Flash?

Pulse dialing is a type of signaling in which codes (digits) are dialed in pulses. A hook switch or a Flash key is generally used to dial this code. Technically, Flash is breaking the loop current for 200 milliseconds to 900 ms. Please note that since this code is not simulated in standard DTMF convention, one cannot dial it in DTMF mode.

Flash timer signifies the time period for which the loop current breaks. Flash timer is programmable. Flash timer ranges from 083 ms to 999 ms. *By default, Flash Timer is 600 ms.*

Where is it used?

Extensions dial flash to use few System features and also to use few PSTN features. Flash is used in following cases:

- While transferring a call from one extension to another.
- While making conference between few extensions.
- While toggling call between two extensions.
- While using call-waiting facility available from PSTN exchange.

Now a days, more number of basic service providers and different types of advanced electronic telephone exchanges are prevailing. It is possible that one service provider interprets breaking of loop current for 300 ms as flash, other service provider interprets breaking of loop current for 900 ms as flash and system interprets breaking of loop current for 600 ms as flash. Hence if the system engineer sets the flash timer to 600 ms then he might not be able to use features provided by the service provider interpreting 900 ms for flash. To take care of this situation, SARVAM UCS offers Flexibility to program different flash timers for both extensions and trunks.

How to configure

To set the timer to the desired value:

- for the SLT ports, refer “SLT Hardware Template”.
- for CO Trunks, refer “CO Hardware Template”.
- for E&M ports, refer “E&M Feature Template”.

- *Many times it happens that while transferring the call, the call either gets disconnected or is not transferred.*
    - *This happens due to mismatch of time for which the hook switch is pressed, if used for transferring the call hence it is advisable to use Flash Key of the telephone instrument, instead of hook switch.*
    - *This problem may occur with Flash Key also, if the timer for the Flash Key on the telephone instrument and the flash timer of the system are not set properly.*
    - *Few telephone instruments have flash timers set to 800 ms. In such case the call does not get transferred because the flash timer of all extensions is set at 600 ms by default. In such cases the flash timer of the extension where the phone is connected should be increased to 800 ms.*

Flashing on Trunks (Continued Dialing)

What's this?

Public exchanges support features like call waiting, call forward, etc. To be able to use these features, users need to dial certain codes during speech.

When a system is connected between the user and the central office, the codes for dialing the features of the central office may clash with the codes for accessing the features of the system, making it difficult for users to access the features of the central office while in speech.

To overcome this, SARVAM UCS supports Flashing on Trunks (Continued Dialing), which informs the system about the codes dialed for the features of the central office on trunks by extension users.

How to configure

To be able to use this feature, *Continued Dialing* must be allowed to the extension in its "Class of Service (COS)" .

How to use

For EON and Extended IP Phone Users

While in speech on trunk,

- Press 'Transfer' Key, dial * and the Desired Service Provider Code.
  Or
- Press DSS Key assigned to Flashing on Trunks (if programmed).
- Dial the Desired Service Provider Code.

For SLT Users

While in speech on trunk,

- Press 'Flash'.
- Dial *
- Dial the Desired Service Provider Code.

Example:
To use Call Waiting facility of service provider exchange from an SLT extension, follow the steps given below:

1	Dial ***Flash-*.***	This informs the system to pass following code.
2	Dial ***Flash-1.***	Speech with second call.
3	Dial ***Flash-*.***	This informs the system to pass following code.
4	Dial ***Flash-1.***	Speech with first call.
5	Dial ***Flash-*.***	This informs the system to pass following code.
6	Dial ***Flash-1.***	Speech with second call.

Flexible Numbers

What's this?

SARVAM UCS offers Flexibility to assign a code of your choice to access an extension. This code is called Flexible number. For example, to access first SLT having software port 001, one has to dial 2001. It is possible to change this code to any other number of your choice.

SARVAM UCS offers the following types of extensions, namely SLT, DKP, Magneto, ISDN Terminals, SIP Extensions, Radio Extensions, Virtual Extensions and Department Groups. The system loads default access codes to all extensions on first power ON. Later on the extensions can be assigned default Flexible numbers using a command.

The Default Access Codes for the Extensions are given below:

Software Port	Default Access Codes (Flexible Numbers)
001 to 240	2001 to 2240 for SLT Extensions
01 to 96	3001 to 3096 for DKP Extensions
01 to 16	Blank for Magneto Ports
01 to 64	Blank for ISDN Terminals
001 to 999	Blank for SIP Extensions (This is the SIP ID)
01 to 16	Bank for Radio Extensions
01 to 64	Blank for Virtual Extensions
01 to 16 17 to 24	3901 to 3916 for Department Groups Blank

SARVAM UCS also allows you to assign a Flexible Number to each extension individually or to a range of extensions simultaneously.

Configuring Flexible Numbers (Access Codes)

Assigning Access Codes to Extensions

You can configure the Flexible Numbers using Jeeves as well as commands. For instructions, see "Configuring SLT Extensions", "Configuring DKP Extensions", "Configuring ISDN Terminals", "Configuring SIP Extension Settings as per the Extended Phone Type", "Virtual Extension", "Configuring Radio Interface", "Configuring Magneto Interface"and "Department Call".

- *It is possible to have maximum of 6 digit flexible numbers.*
- *It is possible to clear the flexible number of a extension, range of extension and all extensions.*
- *Flexible numbers are the codes dialed from dial phase to call another extension. These flexible numbers should be unique and should not match with either other SLT extensions or DKP extensions or any of the features available from the dial phase.*

- *Flexible number having common digits can be assigned to another extension. Please refer "Conflict Dialing" for more details.*
- *Same flexible number cannot be assigned to two different extensions.*
- *Use flexible numbers for all the features used from User mode and SA mode. Software port numbers are to be used only from the SE mode.*
- *When the access code of a extension is cleared; its flexible number becomes null or void.*
- *If access code of a extension is cleared, one cannot call that extension. However the extension with NULL flexible number can make calls as usual.*

Assigning Access Codes to a Range of Extensions

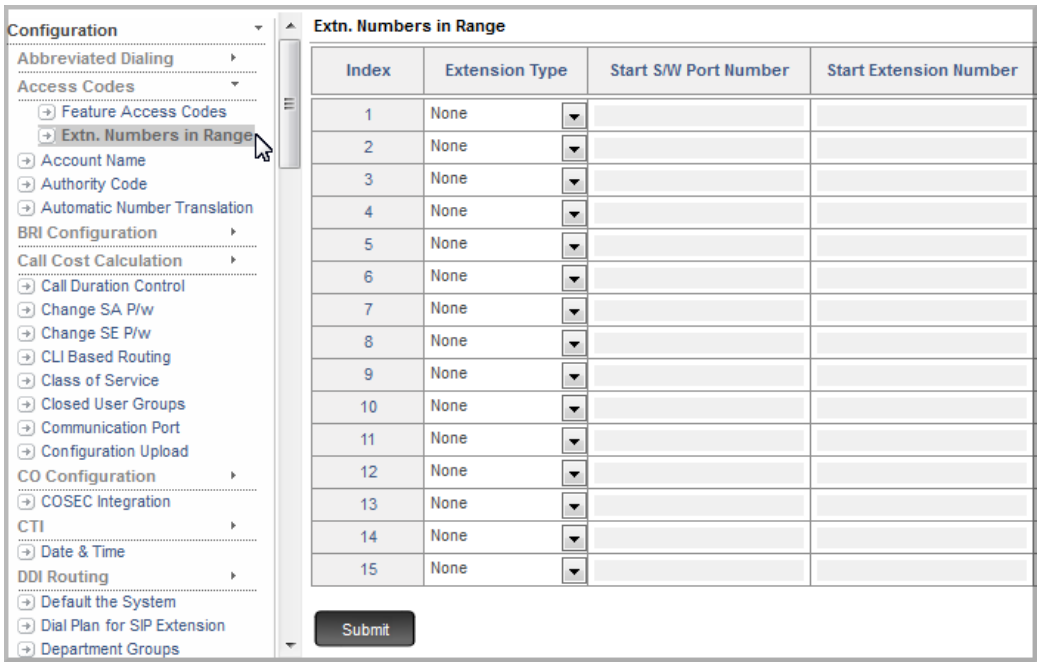
You can assign Access Codes to a range of Extensions using Jeeves only.

If you are assigning a range of Extension Numbers (Access Codes) to the desired ports, and a match is found for the same extension numbers, the system will clear these extension numbers from the existing database. The new extension numbers will be assigned according to the given range.

Assigning Extension Numbers through **Extn. Numbers Range** has a priority over Extension Numbers assigned on individual ports.

To assign a range of Extension Numbers,

- Login as System Engineer.
- Under **Configuration**, click **Access Codes.**
- Click **Extn Numbers in Range** to open the page.



- Against each Index configure the following:

    - **Extension Type:** Select the Extension Type. You can select SLT,DKP, SIP, ISDN Terminal, Magneto, Radio, Department Group or Virtual Extension.

    - **Start S/W Port Number:** Enter the Software Port Number from which you want the system to start assigning the desired extension numbers.

    - Define the range of Station Access Codes(Extension Number/Flexible Number/Access Code) that you wish to assign to the Extension Type you selected in **Start Extension Number** and **End Extension Number**.

      For the given range of extension numbers, the system will assign extension numbers from the software port number specified in Start S/W Port Number for this particular entry. The range of extension numbers will be assigned in ascending order of the Software Port Number.

      For example:
      Extension Type you selected is SLT
      Start Software Port Number is 1
      Start Extension Number is 2001
      End Extension Number is 2100

      The system will assign Extension Number 2001 to Software Port Number 1, 2002 to Software Port Number 2 and so on. The system will assign the last Extension Number 2100 to Software Port Number 100.

If you want to assign access codes starting with **#** or ***** to a range of extensions, make sure both **Start** and **End extension numbers** begin with **#** or *****.

- Click **Submit**.

To check the Extension Numbers (Access Codes) assigned by the system,

- Click the Extension Type, in this case, under **Configuration** click **SLT Configuration**.

- Click **SLT Parameters**.

- Similarly you can assign the Access Codes (Flexible Numbers) to DKP, SIP, ISDN Terminal, Magneto, Radio, Department Group or Virtual Extension.

Floor Service

What's this?

The Floor Service feature allows you to provide a common access code to extension users which they can dial to call floor service.

Essentially a hospitality feature, Floor Service is also useful in offices. Floor service can be any administration or service department in the building, such as a stationery room, back office, backroom, photocopy/ mail room, secretarial assistance, concierge/janitor, Storeroom.

Just as all extension users can reach the Operator by dialing the common access code '9', they can reach the floor service by dialing a common access code, '38'. This is the default Floor Service access code, for all geographical regions where SARVAM UCS is installed.

This feature can be used in:

- Multi-storied buildings, which have floor service (pantry, mail sorting, house keeping, janitor, coffee room, refreshment area) for each floor. The SARVAM UCS can be programmed to land calls made by extension users dialing the common access code '38' on the floor service extensions of their respective floors.
- Offices that have a centralized floor service, instead of one on each floor. The SARVAM UCS can be programmed to land calls made from all extension phones by dialing '38' on the common floor service extensions.

This feature requires a license. To use this feature you must purchase the license for Hospitality. Refer the topic "License Management" to know more.

How it works

For example, Midas Towers houses different departments on each floor. Each floor has Floor Service.

Extensions 2001 to 2010 are on the first floor, 2011 to 2020 on the second floor, and 3001 to 3010 on the third floor. The floor service extensions are numbered as 2012 on the first floor, 2022 on the second floor and 3012 on the third floor.

With the Floor Service programmed for each floor, when the extension user 2001 dials '38', the call will land on the service extension 2012, assigned to room service on the first floor. Similarly, when the extension user 3008 on the third floor dials '38', the call will land on the service extension 3012 on the third floor.

If Midas Towers had a single floor service extension 2012 for all floors, with Floor service programmed, calls made from all extensions by dialing '38' would land on extension 2012 only.

How to configure

Programming the Floor Service feature involves the following steps:

1. Creating a routing group for each floor. Include Floor service extensions of a floor in a routing group prepared for that floor.

2. Assigning a routing group (number) in the Floor Service feature in the Station Advanced Feature Template. Prepare a different Station Advance Feature Template for each floor.

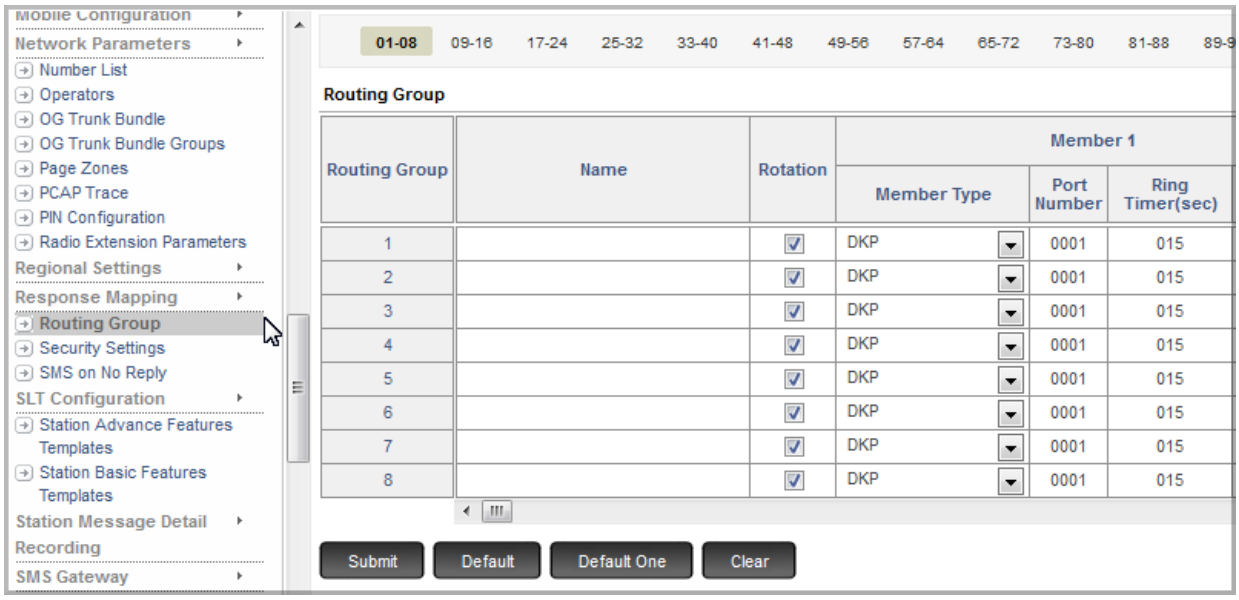
3. Applying the Station Advanced Feature Template (with the Floor service group programmed) to the extension. This will assign the extensions to the routing group programmed in the Template.

If the Enterprise/Building has centralized floor service, you only need to create a single Routing Group with service extensions, as required. This routing group number can be programmed on a common Station Advanced Feature Template which will be applied to all extensions.

Floor Service parameters can be programmed using Jeeves and a Telephone.

Programming Floor Service using Jeeves

- Log in as System Engineer.
- Under **Configuration**, click **Routing Group**.
- Choose the Routing Group number (01-96) you want to use as floor service group.



You can program different routing groups for different floors. In each routing group you can program maximum 32 service extensions as 'members'.

- For routing group to be used as floor service, program the following parameters:
  - **Rotation Flag:** With this flag, you can enable or disable the rotation of calls in the routing group which has multiple 'member' extensions. When enabled, each fresh call will land on the extension which is next to the one that received the last call. This ensures equal distribution of incoming calls to all the destinations within the routing group. The flag has no relevance if the routing group has only one member extension.
  - **When member rejects the call, place the call again:** If any SIP/DKP member extension rejects an incoming call and the system again checks the routing group sequence, you can allow or restrict placing the same call on this extension. Select the When member rejects the call, place the call again check box, if you want the system to place the call again on the extension.

If this check box is cleared (disabled) and you have selected the Continuous check box, the extension that rejects the call stops ringing while the system hunts for other extensions in the routing group to land the call.

- **Member Type:** Select the 'Member Type'. You can select SLT, DKP, SIP, Virtual Extensions, ISDN Terminal, OGTBG or the Voice Mail Auto Attendant.

  Program only as many extensions as you want in the routing group and set the remaining Member Types to 'None'.

  For example: if you want to program only one extension in the routing group, set the Member Type in the remaining columns (Member 02-Member 32) to 'None.'

- **Port Number:** Enter the software port number on which the SLT/DKP/SIP/Virtual Extension/ISDN Terminal is connected.

  If you have selected OTBG then enter the OTBG number here.

- **Voice Mail Auto Attendant (VMAA) Menu:** if you have selected the *Voice Mail Auto Attendant* as the Port Type, select the VMAA Menu to assign to the respective routing group.

  You may click the *Voice Mail Auto Attendant (VMAA) Menu* link to edit the parameters of desired VMAA Menu. For details, see “Voice Mail Auto-Attendant Menu”.

- **Ring Timer(s):** This timer defines the time for which the extension, on which the call lands, should ring. By default, the ring timer is set to 015 seconds and can be changed.

- **Continuous Ring Flag:** With this flag, you can set an extension to ring continuously until the call is answered. The first extension will continue to ring even as the system hunts for other extensions in the routing group to land the call. If the call still remains unanswered, the system will return the call to the first extension once again. This flag is of no relevance, if there is only one member extension in a routing group.

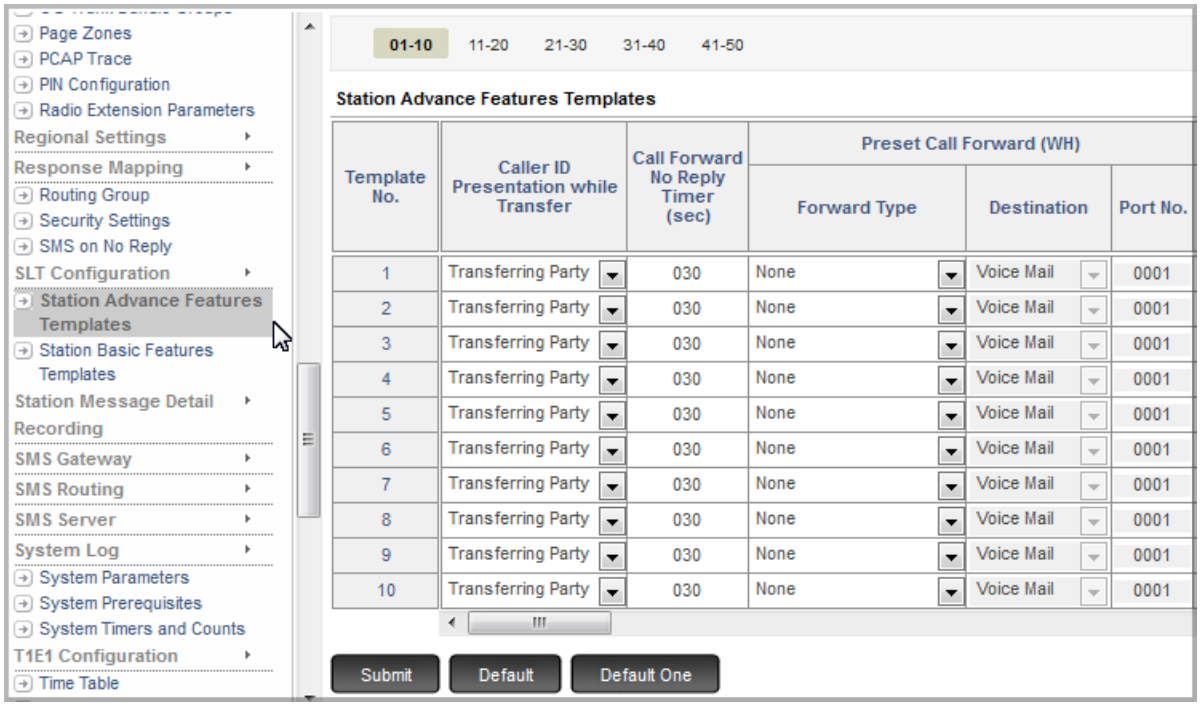
- Repeat the above steps to include other floor service extensions in the routing group.

- Click **Submit** at the bottom of the page to save your settings.

- Click **Station Advanced Feature Template** to open the page.

All extensions are assigned the Advanced Feature Template 01, by default. If the enterprise requires separate floor-service group for each floor, program a separate Station Advanced Feature Template for the extensions of each floor.

- Select a Station Advance Feature Template number to be assigned to the user extensions of a floor. For example: Template number 02 for extensions 2001 to 2010 are on the first floor, Template number 03 for extensions, 2011 to 2020 on the second floor, and Template number 04 for extensions 3001 to 3010 on the third floor.



- Scroll with the horizontal bar to reach the column **Floor Service** of the selected Templates. Enter the Routing Group number you want to use as floor service group for that particular Station Advance Feature Template.

- Click **Submit** at the bottom of the page to save your settings.

- Now, apply the Station Advanced Feature Templates (with floor service routing groups programmed) to the extensions of the respective floors. For example: Template number 02 on extensions 2001 to 2010 are on the first floor, Template number 03 on extensions, 2011 to 2020 on the second floor, and Template number 04 on extensions 3001 to 3010 on the third floor.

  If extensions are SLT, assign the Template on the **SLT Parameters** page.

  If extensions are DKP, assign the Template on the **DKP Parameters** page.

  If extensions are ISDN Terminals, assign the Template on the **ISDN Parameters** page.

  If extensions are Virtual Extensions, assign the Template on the **Virtual Extensions** page.

  Refer the topic “Station Advanced Feature Template” for instructions.

Programming Floor Service using a Telephone

- Enter SE mode.

  To program a routing group with member extensions, dial:
  - ***6502-1-Routing Group-Destination Index-Port Type-Port Number***
    Where,
    Routing Group is the number of the Routing Group 01 to 96.
    Destination Index is from 01 to 32

Port Type is the 'Member type':
00 for None
01 for SLT
02 for DKP
28 for ISDN terminal
34 for SIP Extension
36 for Virtual Extension

Port Number is the Software port number[305] on which the floor service member extension SLT, DKP, ISDN Terminal is attached.
Software port number of the SLT, from 001 to 512.
Software port number of the DKP, from 001 to 128.
Software port number of the ISDN Terminal, from 01 to 64.
Software port number of the SIP Extension, from 001 to 999.
Software port number of the Virtual Extension, from 01 to 64.

To program the Ring Timer for the routing group, dial:

- ***6503-1-Routing Group-Destination Index-Ring Timer***
  Where,
  Routing Group is the number of the Routing Group 01 to 96
  Destination Index is from 01 to 32
  Ring Timer is from 000 to 255 seconds (default: 015 seconds).

To program the Continuous Ring Flag for the routing group, dial:

- ***6504-1-Routing Group-Destination Index-Flag***
  Where,
  Routing Group is the number of the Routing Group 01 to 96.
  Destination Index is from 01 to 32
  Continuous Ring Flag is
  0 for disable continuous ring (each member extension in the group will ring for the programmed 'Ring Timer' for the group)
  1 for enable continuous ring (the first extension in the group will ring till the call is answered)

To program the routing group in a Station Advanced Feature Template, dial:

- ***5602-1-Template Number-11-Routing Group***
  Where,
  Template Number is from 01 to 50
  11 is the feature code for Floor Service
  Routing Group is from 01 to 96[306]

  To apply the Station Advanced Feature Template now programmed with the Routing Group to SLTs, DKPs, ISDN Terminals, SIP extensions, refer the topic “Customizing Station Advanced Feature Template using a Telephone”.

- Exit SE mode.

305. Refer the topic 'Software Port and Hardware ID' .
306. Enter the number of the routing group you programmed as Floor Service group.

How to use

To be able to use floor service, extension users may dial the default Access Code defined for Floor Service: 38.

Check with your System Engineer if this access code has been changed and dial the new access code obtained from the System Engineer.

For EON and Extended IP Phone Users

- Go OFF-Hook.
- Press DSS Key assigned to Floor Service (if programmed)

  OR

- Dial Access Code '38'
- Talk
- Go ON-Hook

For SLT Users

- Lift handset
- Dial, Access Code '38'
- Talk
- Replace Handset.

Follow Me

What's this?

Using this feature, extension users can make your calls follow you wherever you go. Extension users can receive their calls on another extension, whenever they want.

How it works

- A's extension number is 2001.
- B's extension number is 2003.
- A is currently at B's extension.
- A wants to receive calls from extension 2001 on extension 2003.
- A sets Call Follow Me on extension 2003.
- All calls landing on A's extension 2001 will be forwarded to extension 2003.
- When A returns to extension 2001, A cancels Call Follow Me.

- *The extensions dial tone changes to feature tone if its calls are forwarded.*
- *Multiple users can use 'Follow Me' from the same extension.*
- *Follow Me can be overwritten. Extension A sets Follow-Me on extension B. After a period of time; goes to extension C. A can receive calls on extension C by setting Follow Me on extension C. Follow Me set by A on extension B will be cancelled.*
- *Follow Me cannot be chained. If extension A sets Follow Me to extension B. And extension B sets Follow Me on extension C, Calls for A will land on B only and calls for B will land on C only.*
- *DND is given priority over Call Follow Me feature.*

Also see "Call Forward", "Class of Service (COS)" and "Do Not Disturb (DND)"

How to configure

To be able to use Follow Me, extension users must have Call Forward feature enabled in their Class of Service for the time zone. For instructions, see "Class of Service (COS)" and "Station Basic Feature Template".

How to use

For EON & Extended IP Phone Users

To set Follow Me from another extension,

- Press 'Forward' Key of the other extension.
  OR
  Dial 135.
- Enter your extension number.
- Enter your user password.

To cancel Follow me, from your extension,

- Press ‘Forward’ Key of your extension phone.
- Select ‘Cancel’
  OR
  Dial 130.

For SLT Users

To set follow me from another extension,

- Lift the handset of the other extension.
- Dial 135
- Dial your extension number.
- Dial your user password.
- Replace handset.

To cancel Follow me,

- Lift the handset of your extension.
- Dial 130
- Replace handset.

Forced Answer

What's this?

Extension users can force other extension users to answer their calls when there is no response from the called extensions.

How it works

Forced Answer can be requested by the calling extension. The calling extension may be an SLT, a DKP or an Extended IP Phone. However, the called extension (being forced to answer) must be either a DKP or an Extended IP Phone.

- Extension user A (SLT) calls extension user B (DKP/Extended IP Phone).
- B's phone is ringing, but B does not answer.
- A dials Forced Answer feature code.
- The speaker of B's phone's is turned on (goes OFF-Hook).
- A may now talk to B.

Forced Answer can be used when the called extension idle or ringing.

How to configure

To be able to use Forced Answer, extension users must have this feature enabled in their Class of Service for the time zone. For instructions, see "Class of Service (COS)" and "Station Basic Feature Template".

How to use

For EON & Extended IP Phone Users

To use forced answer on an extension:

- Dial the desired extension number.
- Press the DSS Key assigned to Forced Answer on Ring Back tone.
  OR
  Dial **5** on Ring Back Tone.
- The Ring Back Tone stops.
- The called extension's speaker is turned on.
- You are in speech with the called extension.
- You may talk.

For SLT Users

To use forced answer on an extension:

- Dial the desired extension number.
- Dial **5** on Ring Back tone.
- The Ring Back Tone stops.
- The called extension phone's speaker is turned on.

- You are in speech with the called extension.
- You may talk.

You can also dial '5', the feature code for Forced Answer, immediately after dialing the desired extension number, instead of dialing it during the Ring Back Tone. This way, you can talk to the desired extension user without waiting for the called extension user to answer your call.

Forced Call Disconnection

What's this?

Forced Call Disconnection enables extension users to disconnect a busy extension or a trunk at will, and free the system resources (access to extension and trunk) for themselves.

How it works

Forced Call Disconnection of an Extension:

- A, B and C are extensions.
- A and B are in speech.
- C calls B and finds it busy.
- C uses Forced Call Disconnection by dialing the feature command.
- C gets confirmation tone, while A and B get error tone.

Forced Call Disconnection of a Trunk:

- A and B are extensions. C is the external party.
- A is in speech with C on Trunk 1.
- B grabs Trunk 1 using *Selective Port Access*, but gets busy tone.
- B uses Forced Call Disconnection by dialing the feature command.
- B gets confirmation tone. A gets disconnected and gets error tone.
- B must grab Trunk 1 again to get the dial tone of the network.

To be able to use Forced Call Disconnection, the extension user must have a higher "Priority" than the extension user whom he/she tries to forcibly disconnect.

To be able to use Forced Call Disconnection on a busy trunk, the extension user must have grabbed that trunk using "Selective Port Access". If the extension user has grabbed the trunk using a Trunk Access Code, the feature code to dial Forced Call Disconnection will not work.

In PLCC applications, Forced Call Disconnection can be used in a chain to reach the last Exchange through many tandem exchanges in between.

Forced Call Disconnection is not supported on SIP Trunks, BRI and T1E1 lines.

You are advised to restrict access to this feature only to important extension users. Extension Users who are allowed this feature are advised to use it judiciously.

How to configure

To be able to use Forced Call Disconnection, the extension must have:

- **Forced Release** feature enabled in the Class of Service. For instructions see "Class of Service (COS)" and "Station Basic Feature Template".

- As Forced Call Disconnection on a busy trunk is possible only if the extension user has grabbed that trunk using **Selective Port Access**, this feature must be enabled in the Class of Service of the extension. For instructions see "Class of Service (COS)" and "Station Basic Feature Template".

- Higher **Priority** assigned than other extensions. See “Priority” for instructions.

How to use

For EON & Extended IP Phone Users

To forcibly disconnect a busy extension/trunk[307]:

- Press the DSS Key assigned to Forced Call Disconnection on Busy tone.

  OR

  Dial **#*** on Busy tone.
- You get confirmation tone.
- You may now dial the extension number/grab the trunk.

For SLT Users

To forcibly disconnect a busy extension/trunk:

- Dial **#*** on Busy tone.
- You get confirmation tone.
- You may now dial the extension number/grab the trunk.

307. Only if you have grabbed this trunk using Selective Port Access.

Gain Settings

What's this?

To avoid noise or echo during speech, you must set the speech volume levels on the ports. SARVAM UCS allows you to set the speech volume levels for the following port types—CO, Mobile, SIP and SLT.

The speech volume levels can be adjusted by increasing or decreasing the Gain Settings provided on each port type.

How it works

A call received on the CO port can be placed on any of the following ports—DKP, SLT, Mobile, T1E1, SIP, BRI, E&M. The speech volume levels differ according to the port type. Hence, on the CO port you must set the speech volume levels for each of these port types.

In this case, let us assume that the call on the CO Trunk is to be placed on the SLT Port.

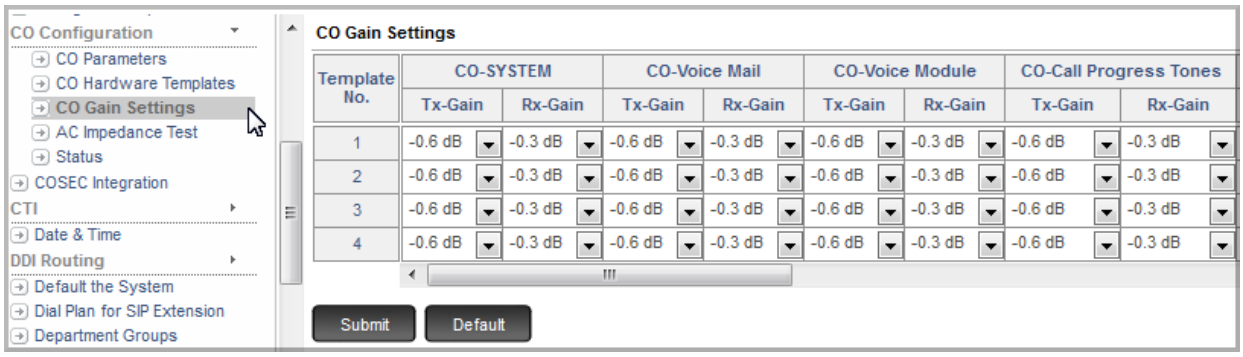
Before placing the call on the SLT Port, the system applies the CO to SLT Gain Setting (Receive and Transmit Gain settings) configured on the CO Trunk to adjust the speech volume level.

When the call is placed on the SLT Port, the SLT to CO Gain Settings (Receive and Transmit Gain settings) on the SLT Port are applied to adjust the speech volume level.

Hence, you can set different speech volume levels for each port type and the system automatically detects and applies these gain settings for each port type.

How to configure

- Log into Jeeves as System Engineer.
- Under **Configuration**, click **CO Configuration**.
- Click **CO Gain Settings** to open the page.



- Select the CO Gain Settings Template Number you want to assign in the CO Hardware Template and configure the following Transmit and Receive Gain Settings:

- **CO - System (Tx-Gain and Rx-Gain)**: Configure the Gain Settings that you want the system to apply on the CO port with respect to the system (for example Call Conference). These will not be applicable for any other port type. Valid Range for Tx gain: +10dB to -15dB and Rx Gain +10dB to -15dB.

- **CO - Voice Mail (Tx-Gain and Rx-Gain)**: Configure the Gain Setting that you want the system to apply on the CO port when the incoming calls on the CO port are being answered by the VMS. Valid Range for Tx gain: +10dB to -15dB and Rx Gain +10dB to -15dB.

- **CO - Voice Module (Tx-Gain and Rx-Gain)**: Configure the Gain Settings that you want the system to apply on the CO port when incoming calls are answered using Auto Attendant or Trunk Auto Answer. Valid Range for Tx gain: +10dB to -15dB and Rx Gain +10dB to -15dB.

- **CO - Call Progress Tones (Tx-Gain and Rx-Gain)**: Configure the Gain Setting that you want the system to apply on the CO port while playing Call Progress Tones. Valid Range for Tx gain: +10dB to -15dB and Rx Gain +10dB to -15dB.

- **CO - Radio (Tx-Gain and Rx-Gain)**: Configure the Gain Setting that you want the system to apply on the CO port when the CO port is connected to any Radio Port of the system during an incoming or outgoing call. Valid Range for Tx gain: +10dB to -15dB and Rx Gain +10dB to -15dB.

- **CO - SLT (Tx-Gain and Rx-Gain)**: Configure the Gain Setting that you want the system to apply on the CO port when the CO port is connected to any FXS Port of the system during an incoming or outgoing call. Valid Range for Tx gain: +10dB to -15dB and Rx Gain +10dB to -15dB.

- **CO - CO (Tx-Gain and Rx-Gain)**: Configure the Gain setting that you want the system to apply on the CO port when the CO port is connected to another CO Port of the system during an incoming or outgoing call. Valid Range for Tx gain: +10dB to -15dB and Rx Gain +10dB to -15dB.

- **CO - DKP (Tx-Gain and Rx-Gain)**: Configure the Gain Setting that you want the system to apply on the CO port when the CO port is connected to any DKP Port of the system during an incoming or outgoing call. Valid Range for Tx gain: +10dB to -15dB and Rx Gain +10dB to -15dB.

- **CO - SIP (Tx-Gain and Rx-Gain)**: Configure the Gain Setting that you want the system to apply on the CO port when the CO port is connected to any SIP Trunk or SIP Extension of the system during an incoming or outgoing call. Valid Range for Tx gain: +10dB to -15dB and Rx Gain +10dB to -15dB.

- **CO - Mobile (Tx-Gain and Rx-Gain)**: Configure the Gain Setting that you want the system to apply on the CO port when the CO port is connected to any Mobile Port of the system during an incoming or outgoing call. Valid Range for Tx gain: +10dB to -15dB and Rx Gain +10dB to -15dB.

- **CO - T1E1/BRI (Tx-Gain and Rx-Gain)**: Configure the Gain Setting that you want the system to apply on the CO port when the CO port is connected to any T1E1/BRI Port of the system during an incoming or outgoing call. Valid Range for Tx gain: +10dB to -15dB and Rx Gain +10dB to -15dB.

- **CO - E&M (Tx-Gain and Rx-Gain)**: Configure the Gain Setting that you want the system to apply on the CO port when the CO port is connected to any E&M Port of the system during an incoming or outgoing call. Valid Range for Tx gain: +10dB to -15dB and Rx Gain +10dB to -15dB.

- Click **Submit** to save the settings.

- Similarly, to configure the Gain settings for the Mobile Port, click **Mobile Gain Settings** under **Mobile Configuration**.

- For SIP Trunk / Extension, click **SIP Gain Settings** under **VoIP Configuration**.
- For SLT Port, click **SLT Gain Settings** under **SLT Configuration**.

Assigning the Gain Settings Templates

To apply the Gain Settings you configured for each port type, you must assign these templates on the respective ports

- Assign the CO Gain Settings Template you configured in the CO Hardware Template, see “CO Hardware Template”.
- Assign the Mobile Gain Settings Template you configured in the Mobile Port Parameters, see “Mobile Port Parameters”.
- Assign the SIP Gain Settings Template you configured in the SIP Hardware Template, see “SIP Hardware Template”.
- Assign the SLT Gain Settings Template you configured in the SLT Hardware Template, see “SLT Hardware Template”.

Gateway Application-Answer Signaling

What's this?

When SARVAM UCS acts as a Gateway, this feature is used to convey the call maturity information on source port, when call made using destination port gets matured. This information can be useful for billing equipment connected at calling party end.

When the system acts as a Gateway, the Source Port is the port on which call originates and the Destination Port is the port on which the call terminates.

How it works

- This feature of answer signaling is applicable only for DISA option selected as 'CLI Authentication-one call-Ans. Sig.' in the DISA-CLI Authentication Table. Refer chapter “Direct Inward System Access (DISA)”.
- The SARVAM UCS works as gateway for routing the call to the destination.
- If the calling party's number is programmed in Table - 'DISA-CLI Authentication', the SARVAM UCS will consider the calling party as successfully logged in as station which is programmed as "Auto Login station". The call will get answered by the SARVAM UCS. The caller will get dial tone.
- Caller can dial the desired number.
- SARVAM UCS will route the call from the destination port.
- When called party answers the call (that is, call on destination port gets matured), the Answer Signaling will be done on source port, if enabled.
- Answer Signaling will be done in form of DTMF digit string as programmed on the source port.
- If Answer Signaling is disabled on the source port, the DTMF digit string will not be dialed out, even if it is programmed.
- On receiving these digits, Billing equipment/System with which the calling party is connected, can consider the call is matured and start billing.

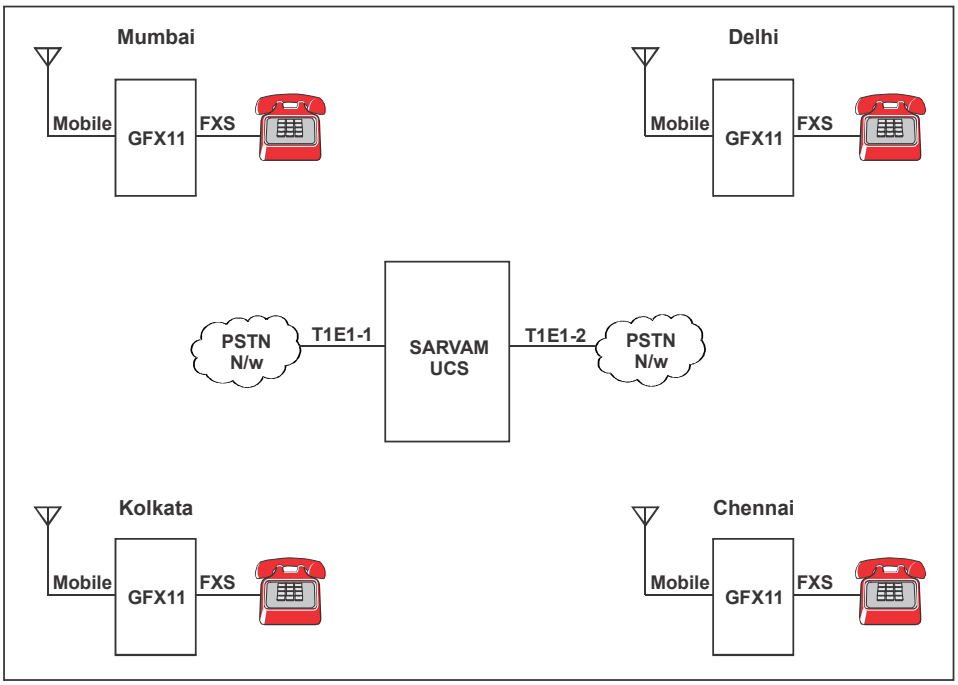
How to configure

To configure 'Gateway Application-Answer Signaling' flag and 'DTMF String' on desired trunk refer following chapters:

- For CO Trunk, Refer chapter “CO Hardware Template”
- For E&M Trunk, Refer chapter “E&M Feature Template”
- For Mobile Trunk, Refer chapter “Configuring Mobile Trunks”
- For T1E1 Trunk, Refer chapter “Configuring PRI Trunks”
- For BRI Trunk, Refer chapter “Configuring BRI Trunks”
- For SIP Trunk, Refer chapter “Configuring SIP Trunks”

By default, 'Gateway Application-Answer Signaling' is disabled for all ports. By default, DTMF Digit String' is programmed as 'CCC' on all ports.

Application:



- Multiple GFX11s are installed at different places for PCO Application.
- One SARVAM UCS is installed at central place.
- On FXS port of each GFX11, 'Pay Phone' is connected.
- On GSM port of each GFX11, SIM card is installed.
- GFX11s are configured for multi stage dialing, in which when ever caller dials the number using pay phone, GFX11 will store the number, it will first make a call to SARVAM UCS's T1E1 line on which DISA - 'CLI Auth- One Call - Ans. Sig.' is enabled.
- GFX11's SIM Numbers are programmed in 'DISA - CLI Auth.' table in SARVAM UCS.
- SARVAM UCS compares the Calling Number received on T1E1 Port with DISA-CLI Auth. Table.
- As the number is programmed in the table, SARVAM UCS answers the call and offers the trunk assigned in Station Basic Feature Template of the station used as 'Auto Login'.
- When SARVAM UCS answers the call, the GFX11 sends the stored called number (which is actually dialed by the caller) in DTMF digits.
- SARVAM UCS routes the call on this number using the offered trunk.
- When called party answers the call, SARVAM UCS will send 'DTMF Digit Strings' programmed on the T1E1 Port (Source Port, on which call originated) as a Gateway Answer Signaling'.

- The Pay Phone connected with FXS Port of the GFX11 is also configured to understand the same DTMF string as call maturity.
- Pay Phone will start billing only on receipt of the desired DTMF digits.

GPAX Application

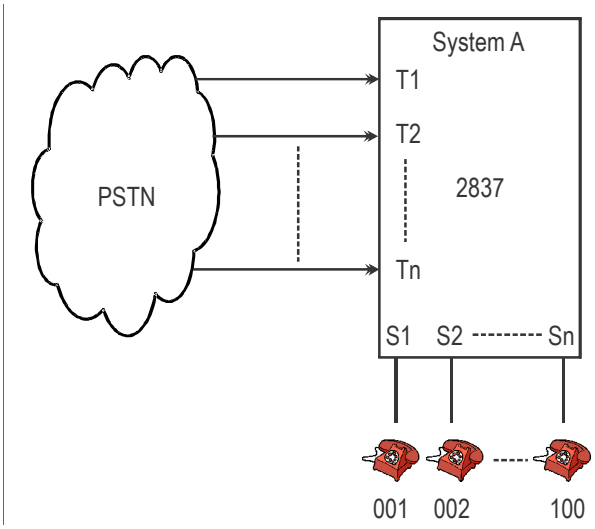
What's this?

GPAX application is one of the applications provided by SARVAM UCS and is used in commercial establishment/ Society/Organization, etc. Group Systems are installed, operated and maintained by organizations/agencies. The owner will be treated as the main hirer. The private exchange (PSTN)/service provider will provide junctions to the System and the owner will pay the rental of the junctions, and the call charges. The number of junctions provided to the System should be adequate to carry the traffic. The owner/agencies of the Systems will be responsible for the payment of all charges to the private exchange/service provider.

In this application say a System 'A' is connected to PSTN. System 'A' is given an exchange ID (say 2837). A number of stations (say 001-100) can be connected to System 'A'. When a station 015 dials 2837025, System 'A' interpret this number to be dialed for the same system. However, for dialing a station number belonging to same system, it is not necessary to prefix the station number with exchange ID. When a station 020 dials 2834537, System 'A' does not interpret this to be dialed for the same system and hence will dial the digits on the trunk.

When a station user picks up the handset and dials any digit except the one programmed in the routing table will be dialed on the trunk. If the user dials digit that is programmed in the routing table with Self-flag enabled, the system will not dial the digits on the trunk since it would interpret these to be dialed for the same system.

For dialing the digits on the trunk it is required to program the routing table carefully. Route code should be specified in route code column of routing table in association with Self Route Flag disable. This will make the call to be routed on trunk which is specified in OG Trunk Bundle Group. For details refer "Closed User Group (CUG)" and "Closed User Group-With Exchange ID".



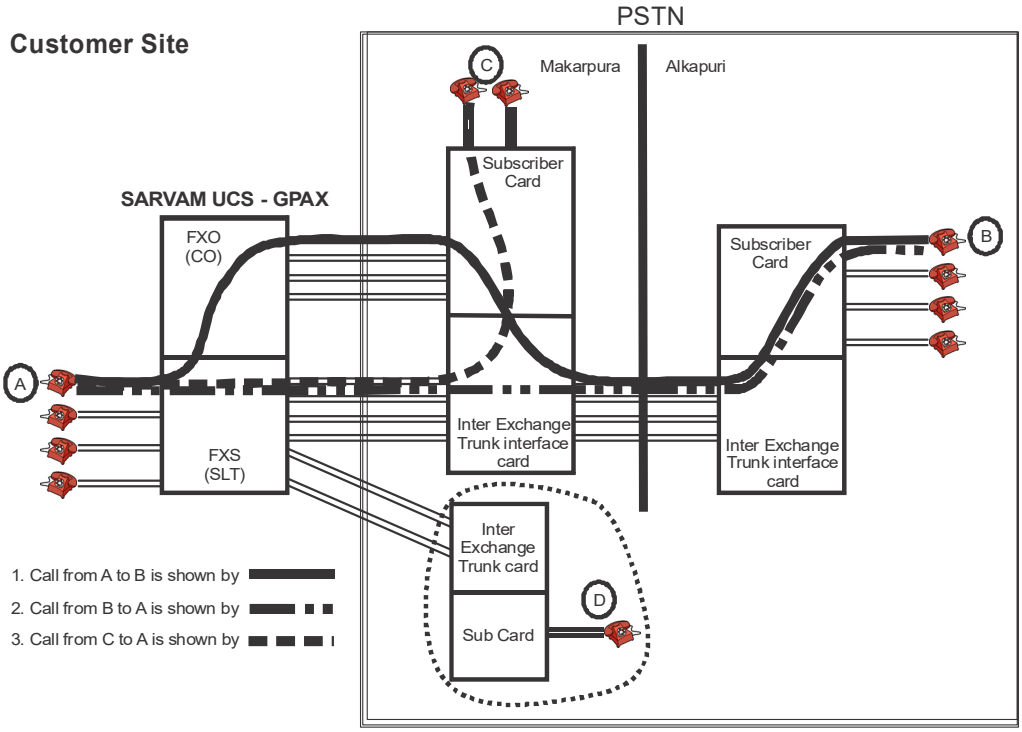
In order to make internal calls, user is advised to program Routing Table such that one of the entries in routing table should have route code as '#', Strip digit count as '1' and Self route flag as '1' (enable). Doing so, when user picks up handset and dials the required number with prefix '#', the system interpret this number as an activity for the internal users and waits for relevant access code.

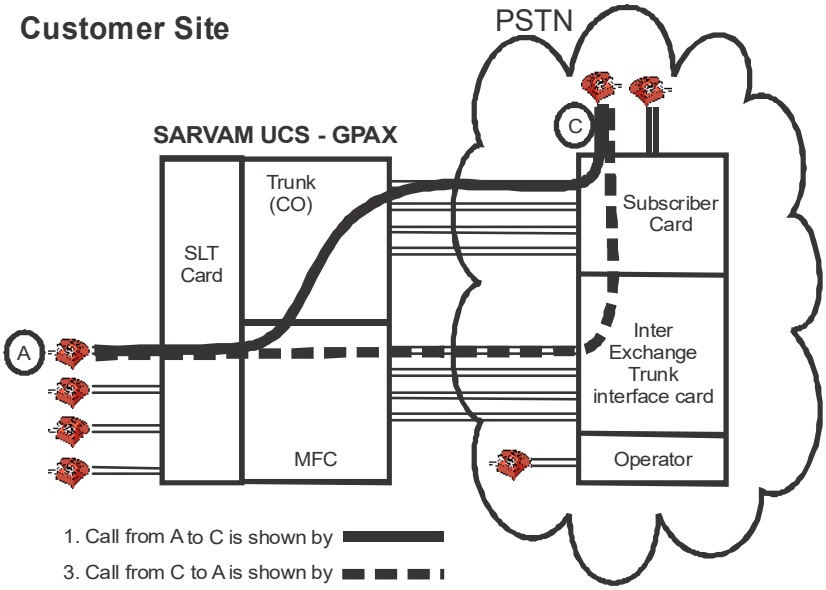
Because of Strip digit count=1, the first digit dialed by the user (that is, # in this case) will be ignored and next digit will be processed which could be a feature code or a station number.

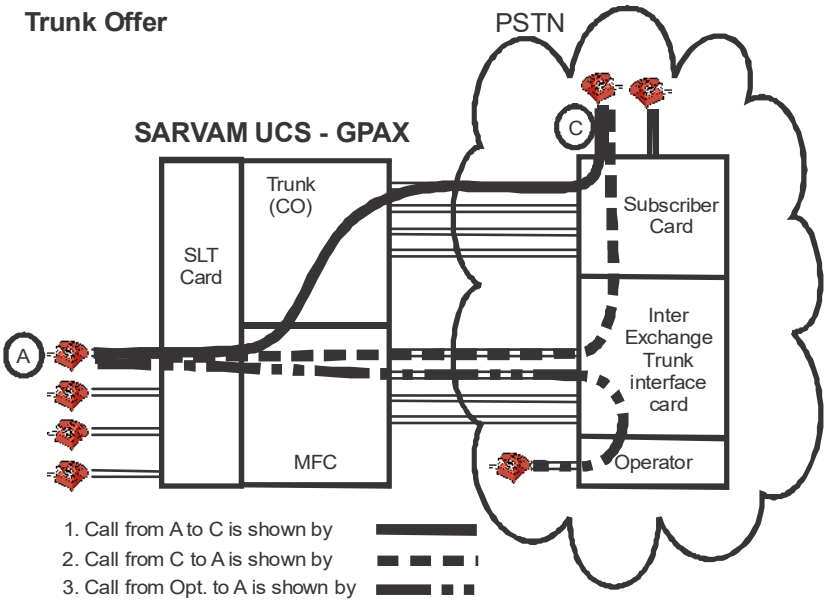
In order to use system features, the feature codes should be prefixed by '#'. Whenever the user dials '#' on picking the handset, the System assumes it to be an activity for the internal users and waits for relevant access code.

In GPAX application, Answer Signaling on SLT and Answer Supervision on trunk shall be programmed properly. Also the Disconnect Supervision parameter shall be programmed as appropriate for proper billing of calls.

Following figures shows how call is established between users using PSTN and SARVAM UCS-GPAX.






Trunk Offer
PSTN
SARVAM UCS - GPAX
C
Trunk
(CO)
SLT
Card
Subscriber
Card
Inter
Exchange
Trunk
interface
card
A
MFC
Operator
1. Call from A to C is shown by
2. Call from C to A is shown by
3. Call from Opt. to A is shown by

GPAX Billing

What's this?

In commercial establishments/large societies/organizations etc. staff make calls from their rooms. It is required that the cost of these calls is calculated so that amount of calls made by a staff member can be paid. GPAX provides a facility which if enabled can calculate cost of each call if programmed properly.

GPAX has a dedicated memory space (commonly called buffer) to store details of each call. These calls are retained in the buffer even during power failure. Once the buffer is 100% full, the new call overwrites the oldest one, that is First In First Out (FIFO) method.

You can also generate various reports and the same can be routed on the computer from this buffer. It is recommended that printing of various reports should be regularized on fixed dates. This should be done regardless of whether the buffer is full or not.This will prevent spilling and subsequent loss of data.

You can enable or disable call logging for individual trunk. This gives flexibility of monitoring only important (STD) trunks. This also prevents frequent spilling of the buffer, as new local calls will not be recorded.

How to configure

To calculate the total cost of a call, you must configure the relevant parameters in "Call Cost Calculation (CCC)", "Call Duration Control (CDC)", "Call Budget on Extension".

In order to bill internal calls between GPAX users, GPAX charge internal calls option must be enabled. For instructions see, "Station Advanced Feature Template". If GPAX charge internal calls option is enabled, this call will be recorded in the Station message detail recording-outgoing buffer. If GPAX charge internal calls option is disabled, the call made to an internal station will not be billed and will be recorded in the SMDR-Internal buffer as normal internal call.

Handover and Handoff

What is this?

Handover allows you to move an active VARTA Mobile UC Client call from the Wi-Fi network to the cellular number (in the cellular network). This is useful when you have an ongoing call and you leave the Wi-Fi network, or if there are voice quality issues over the Wi-Fi network.

Handoff is when you are back into the Wi-Fi network you can move the call from the cellular number to the VARTA Mobile UC Client. The call is moved without being disconnected and redialing the number.

SARVAM UCS will serve the Handover request only if:

- *the Cellular Number is configured in the Application Settings in the Mobile Client. Refer to the respective Mobile Client User Guide for details.*

  or

- *the Mobile Number is configured in the system. See "Configuring Matrix VARTA ADR100/AMP100 UC Clients".*

When the VARTA ADR100/ VARTA AMP100 application is in the background, Handoff will not be possible. Refer VARTA ADR100 User Guide and VARTA AMP100 User Guide to know more.

How to configure

To use this feature, make sure the *Basic Features* are enabled in the Class of Service assigned to you. For instructions, see "Class of Service (COS)" and "Station Basic Feature Template".

Help Desk

What's this?

An organization may have a Centralized Information Office which provides information related to different departments such as HR, IT, or General information. For each department in the organization, an extension number can be defined as a Help Desk.

How it works

- Extension 2002 is defined as Help Desk for HR policies and general rules.
- Extension 2016 calls the Help Desk extension 2002.
- If the Help Desk extension is busy/not responding, an Auto Callback request is set automatically on the Help Desk extension.
- The system will serve the auto callback request as soon as the Help Desk extension is available.
- The Help Desk extension calls back extension 2016.

Also see "Auto Call Back (ACB)"

How to configure

You can define an extension as 'Help Desk' by enabling the 'Help Desk' flag in its "Station Advanced Feature Template".

Holiday Table

What's this?

The Holiday Table feature of SARVAM UCS enables you to configure incoming call management for holidays.

Using the Holiday Table feature of SARVAM UCS you can,

- define the landing destination for incoming calls on trunks on holidays.
- greet callers with customized holiday messages.
- determine the way extensions must work on the holidays.

You can configure a list of holidays in a single table.

How it works

In the Holiday Table, you need define the following:

- The Start and the End Dates and the Time to be considered as Holiday.

- The Time Zone to be considered for operating the time-zone based trunk and extension features[308] during the Date and Time configured as Holiday. The Time Zone for Holiday can be defined as *Non-Working Hours*, *Working Hours*, or *As per Time Table*.

If you have Voice Mail System and are using the Voice Mail Auto Attendant as the landing destination for calls, on holidays, you can play customized greeting messages to callers. For each holiday, you can play a different message.

For example, A company, ABC Ltd., has the following requirements:

- December 23 to December 31, all the employees will be on a holiday. The callers must be greeted with a holiday message.
- January 1 to January 4, few employees will be attending the office.

In this case you must define the following in the Holiday Table:

At Index 1,

- In **Start**, enter the starting date and time of the holiday in DD-MMM-HH-MM format, that is 23 DEC, 00:00 and in **End** enter the last date and time of the holiday, that is 31 DEC, 23:59.

- In **Time Zone for Holiday**, select Non-working hours.

- In **Holiday Message** select the customized holiday message number. The system will greet the callers with this message.

308. Trunk Landing Group, Auto-Attendant, DISA are time-zone based features of Trunks configured in the Trunk Feature Template assigned to trunks. Class of Service, Toll Control, and OG Trunk Bundle Group, are time-zone based features of extensions that are configured in the Station Basic Feature Template assigned to extensions.

At Index 2,

- In **Start**, enter the starting date and time of the holiday in DD-MMM-HH-MM format, that is 31 DEC, 00:00 and in **End** enter the last date and time of the holiday, that is 4 JAN 23:59.
- In **Time Zone for Holiday**, select As per Time Table.
- In **Holiday Message** select **None**.

Index	Holiday								Name	Time Zone for Holiday	Holiday Message
	Start (DD-MMM-HH-MM)				End (DD-MMM-HH-MM)						
1	23	DEC	00	00	31	DEC	23	59	Christmas Holidays	Non-Working Hours	01
2	31	DEC	00	00	04	JAN	23	59	Christmas Holidays	As per Time Table	

After you have defined the above parameters, this is how the feature Holiday Table works,

- On the set date and time, when SARVAM UCS detects a day as a holiday, it checks the configured Time Zone for Holiday and whether Holiday Message is configured.
- When the Time Zone for Holiday is defined as Non-working hours,
  - the system checks for and routes the incoming call on the trunk as per the Trunk Landing Group and the Auto Attendant defined for the trunk for Non-Working hours in the Trunk Feature Template.
  - customized Holiday Message, if configured, is played to the caller. The Holiday message is played in place of the Welcome message for Non-Working hours.
  - Extensions work according to the Class of Service, Toll Control and Outgoing Trunk Bundle Group assigned to them for Non-working hours in the Station Basic Feature Template.

Similarly, if the Time Zone for Holiday is defined as Working Hours, the system will operate the time-zone based features of the trunks and extensions according to Working Hours.

- When Time Zone for Holiday is As per Timetable,
  - the system routes the call as per the Trunk Landing Group and Auto Attendant defined for the current time zone in the Trunk Feature Template.
  - customized Holiday message, if configured, is played to the caller in place of the Welcome message for the current time zone.
  - extensions work according to the Class of Service, Toll Control and Outgoing Trunk Bundle Group assigned to them for the current time zone in the Station Basic Feature Template.

Feature Interaction:

Day/Night Mode: You can set the system in Day/Night Mode, even when you have configured the Holiday Table. In that case, the mode you select, Day (Working Hours) or Night (Non-working Hours) will override the Time Zone you have selected for Holiday.

In order for the Time Zone for Holiday to come into effect, you must set the Day/Night Mode in System Parameters to ***Operate System as per Timetable Assignment***.

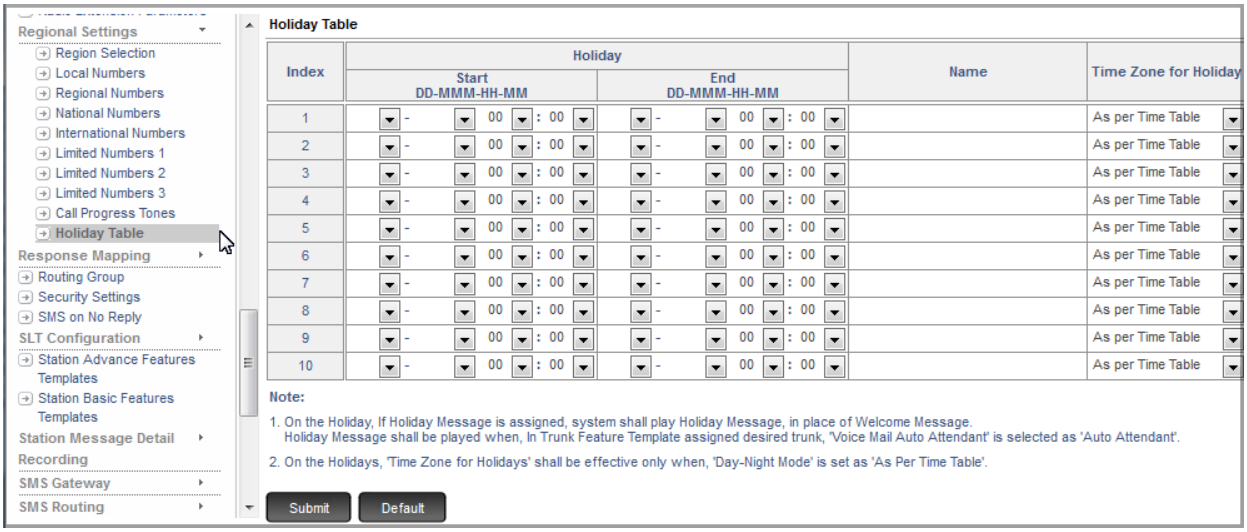
See "Day Night Mode" and "System Parameters" to know more about this feature.

How to configure

You can configure the Holiday Table from the SE as well as the SA mode.

Configuring Holiday Table

- Log in as System Engineer.
- Under **Configuration**, click **Regional Settings**.
- Click **Holiday Table**.



Against each Index configure the following parameters:

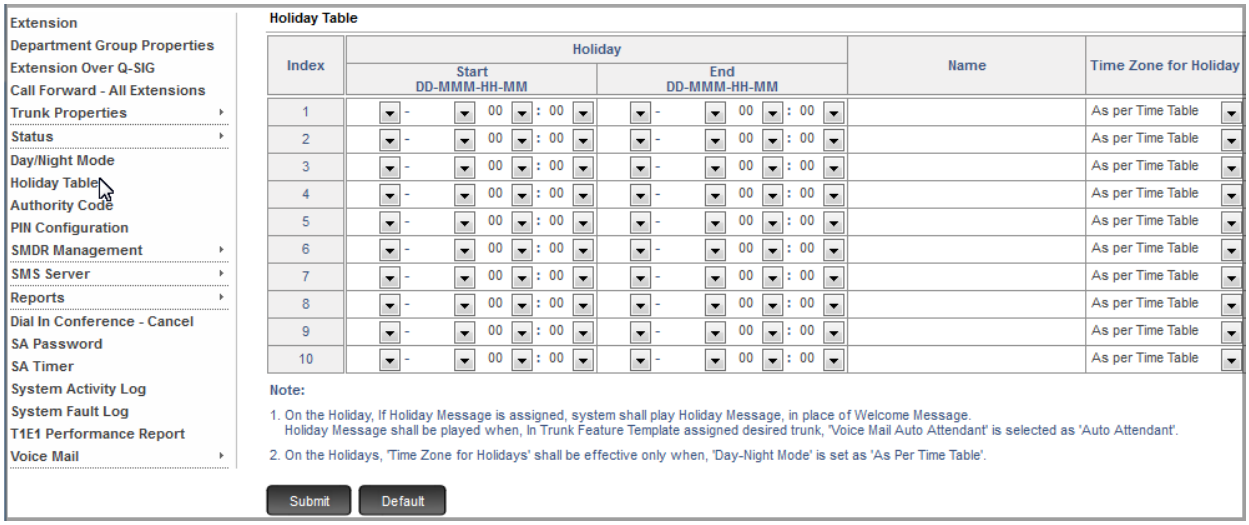
- In **Holiday** configure the **Start** Date, Month and Time of the holiday and **End** Date, Month and Time of the holiday.

  Default, Start and End Date are Blank. Valid Range is from 01 to 31.
  Start and End Month are Blank. Valid Range is from January to December.
  Start and End Time is 00:00 (Hours:Minutes). Valid Range is from 00:00 to 23:59

- You can assign a **Name** to each time period you have defined as Holiday. For example, Christmas, Independence Day, Thanksgiving. Default: Blank
- As the **Time Zone for Holiday** select the time zone for the period you have defined as Holiday: Working Hours, Non-working Hours or As Per Time Table. Default: As Per Time Table.
- If the incoming calls are routed to the VMS Auto Attendant, select the **Holiday Message** number that you want the system to play to the callers. Default: 01.
- Click **Submit** to save changes.

You can also configure the Holiday Table from the SA mode also. To do this,

- Log in as System Administrator.
- Click **Holiday Table**.



- Follow the same steps as given above to configure the Holiday Table.
- Click **Submit** to save your settings.

Configuring Holiday Messages

To greet callers with customized holiday messages, you need to do the following configuration:

- Ensure that Voice Mail Auto Attendant is selected as the Auto Attendant in the Trunk Feature Template. For detailed information, see “Trunk Feature Template”.
- You can either use the default Holiday messages or customize your messages by recording messages of your preference. To know more about the default Holiday Messages and how to record customized Holiday Messages, see “Recording Voice Messages”.

Hot Desking

What's this?

Hot Desking enables extension users to use all the properties of their own extension from another extension.

Hot Desking is useful for people who are often away from their own desks and must work from another. Hot Desking allows them to use all the features and facilities of their own extension from another.

How it works

This feature is supported on DKP and SLT extensions only.

Hot Desking is possible only between extensions of the same type: SLT to SLT and DKP to DKP extensions.

The User Password of both extensions involved in Hot Desking must not be 1111.

Hot Desking can be performed only when both the extensions are idle.

To perform Hot Desking two extensions are required:

- The Host Extension - the extension whose user performs the Hot Desking.
- The Hot Desk Extension - the extension on which Hot Desking is performed.
- When Hot Desk is performed from the Hot Desking extension, all the properties of the Host Extension are copied to the Hot Desk Extension.
- On the Host Extension, the user cannot perform any activity except Cancel Hot Desking.
- You must cancel Hot Desk from both the Hot Desk Extension and the Host Extension.
- After cancelling Hot Desk, the Host Extension and the Hot Desk Extension acquire their original properties.

How to configure

For this feature to work, the feature 'Hot Desk' must be enabled in the Class of Service of the Host Extension and the Hot Desk Extension. See "Class of Service (COS)" and "Station Basic Feature Template" for instructions.

How to use

For EON users

To perform Hot Desk:

- Go to the EON extension (Hot Desk Extension) with which you want to swap your EON extension (Host Extension) properties.
- Press DSS Key assigned to Hot Desk.
  OR

- Dial 1091
- Enter Host Extension number
- Enter Host Extension User Password
- Go ON-Hook.

The User password of both the extensions involved cannot be default password.

To cancel Hot Desk:

- On the Host Extension, press DSS Key assigned to Hot Desk.
  OR
- Dial 1091
- Enter own extension number
- Enter own extension User Password
- Go ON-Hook.

- On the Hot Desk Extension, press DSS Key assigned to Hot Desk.
  OR
- Dial 1091
- Enter own extension number
- Enter own extension User Password
- You get confirmation, Hot Desk cleared
- Go ON-Hook.

For SLT Users

To perform Hot Desk:

- Go to the SLT extension (Hot Desk Extension) with which you want to swap your SLT extension (Host Extension) properties.
- Lift the handset of the Hot Desk extension.
- Dial 1091
- Dial Host Extension number
- Dial Host Extension User Password
- Replace handset.

To cancel Hot Desk:

- On the Host Extension, dial 1091
- Dial own extension number
- Dial own extension User Password
- Go ON-Hook.

- On the Hot Desk Extension, dial 1091
- Dial own extension number
- Dial own extension User Password
- You get confirmation, Hot Desk cleared
- Go ON-Hook.

Hotline

What's this?

The Hotline feature connects the extension user immediately to a particular number or trunk, whenever the extension user goes OFF-Hook.

You can set Hotline to connect immediately to another extension, to a Department Group, to an external number or to an outgoing trunk.

Hotline set for external numbers and outgoing trunks is referred to as *Hot Outward Dialing*.

SARVAM UCS offers two types of Hotline/Hot Outward Dialing:

- **Immediate:** As soon as the extension user goes Off-Hook, the user gets connected to the desired hotline extension number, department group, external number, or outgoing trunk. For this the *Hotline Timer* must be set to '00' seconds (default: 3 seconds).

- **Delayed:** When the extension user goes OFF-Hook, the system plays Dial Tone to the extension user and waits for the *Hotline Timer* (default: 3 seconds). On the expiry of this timer, it connects the extension user to the desired hotline extension number, department group, external number or outgoing trunk.

How it works

- Hotline/Hot Outward Dialing can be set from an SLT, DKP or Extended IP Phone extension, if the extension has Hotline in its Class of Service.

- To be able to use Hotline/Hot Outward Dialing, extension users must do the following:

  - Select the type of Hotline they want to set on their extension; whether to an internal Extension Number, a Department Group, or an External Number or Outgoing Trunk.

  - Configure the *Hotline Timer*. For *Immediate Hotline*, extension users must set the Hotline Timer to '00' seconds. For *Delayed Hotline*, extension users can set the Timer as per their requirement.

- Here is an example of how Hotline works:

  A frequently dials the number of B. So, A sets Hotline for B's number and also sets the Hotline Timer to 5 seconds (Delayed Hotline).
  - A goes Off-Hook
  - SARVAM UCS plays dial tone and waits for 5 seconds
  - If A dials a number within the Hotline Timer, SARVAM UCS outdials the number dialed by A.
  - If A *does not* dial any digit within this time, SARVAM UCS dials B's number.
  - A gets connected to B.

If 'Dial Tone' timer of the system is less than Hotline Timer, the Hotline Timer will override the 'Dial Tone' timer. To know more about these timers, see "System Timers and Counts".

  - If A had set the *Hotline Timer* to '00' seconds (Immediate Hotline), A would be connected to B as soon as A goes Off-Hook.

- If A sets delayed Hot Outward Dialing for a Trunk or an External Number, the system will play dial tone to A and wait for the duration of the Hotline Timer for A to dial digits. If A does not dial any digits within this timer, the system connects A to the Trunk/External Number.

- If A sets immediate Hot Outward Dialing (Hotline Timer set to '00' seconds), A will be connected to the Trunk/External number as soon as A goes Off-Hook.

- Delayed Hotline/Hot Outward Dialing allows extension users to dial out other numbers or grab another trunk, without having to cancel the Hotline/Hot Outward Dialing they have set for a particular number or trunk.

How to configure

To be able to use Hotline, extension users must have this feature enabled in their "Class of Service (COS)" for the time zone, as required.

How to use

Hotline can be set/canceled by users for their own extension, or for any other extension from the SA mode.

Hotline when set/canceled from the SA mode, will not depend on the assigned CoS.

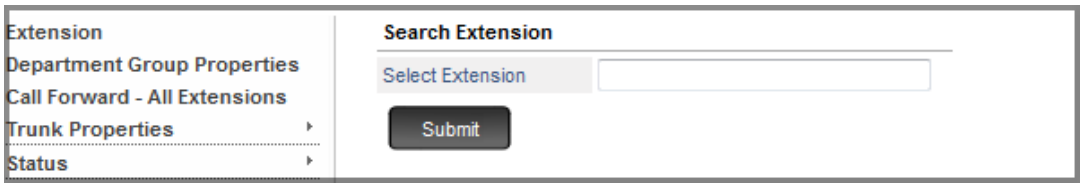
Set/Cancel Hotline for Extension Users

The Operator or any extension user having access to System Administrator mode can set or cancel Hotline for other extension users.

Set Hotline using SA Jeeves

- Log in to Jeeves as System Administrator.

- Click **Extension**.

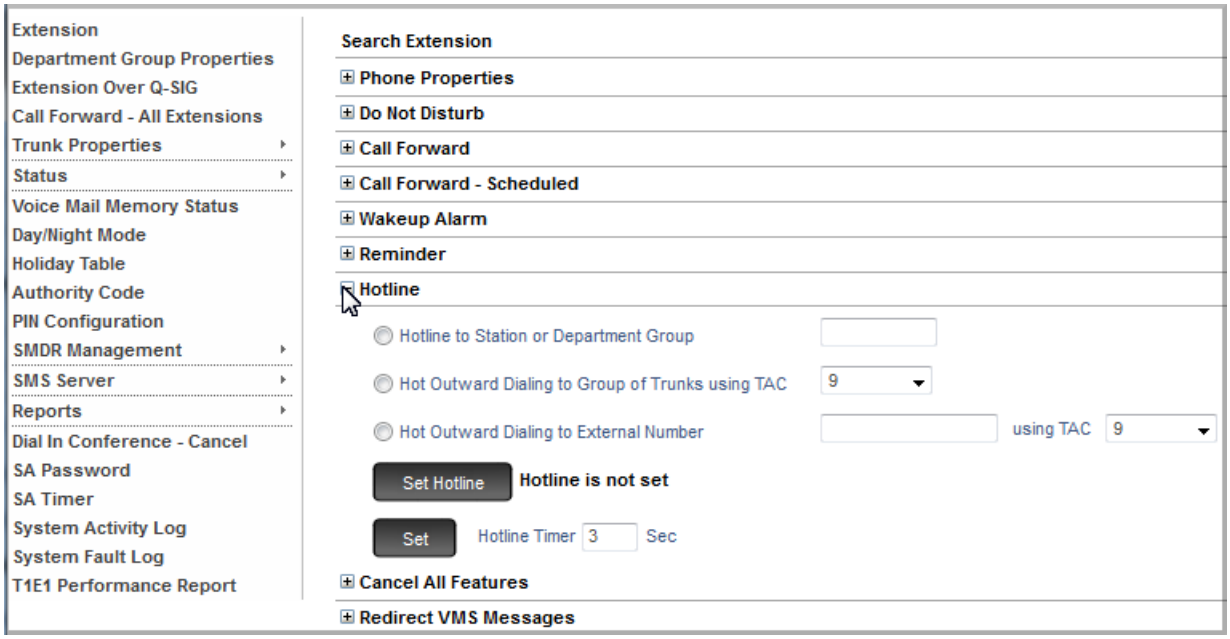


- In **Select Extension,** enter the Number or the Name of the extension on which you want to set this feature

- Click **Submit**.

- The searched extension users details appear on your screen.

- Click **Hotline** to expand.



- Select the type of Hotline you want to set for the extension user from the following:
  - To set Hotline for an Extension or Department Group, select the radio button **Hotline to Station or Department Group**. Enter the Extension number or the Department Group Number in the corresponding box. Default: Blank.
  - To set Hotline for a group of trunks, select the radio button **Hot Outward Dialing to Group of Trunks using TAC** and select the Trunk Access Code from the corresponding drop down list.

    Each Trunk Access Code has a group of trunks for which Hotline will be set.
  - To set Hotline for an external number, select the radio button **Hot Outward Dialing to External Number.** Enter the external number in the corresponding box and in **Using TAC** select a TAC from the drop down list. Using a free trunk from this TAC the external number will be dialled out by the system.
- Click the **Set Hotline** button to set Hotline.

  The message “Hotline is set” appears.
- To set Delayed Hotline, in **Hotline Timer** enter the desired time in seconds.

  On the expiry of this timer, it connects the extension user to the desired hotline extension number, department group, external number or outgoing trunk.
- Click the **Set** button to set Delayed Hotline.
- To cancel Hotline, click the **Cancel Hotline** button.

Set/Cancel Hotline by Extension Users

For EON & Extended IP Phone Users

To set Hotline on an Extension / Department Group,

- Press DSS key assigned to Hotline.
- Scroll to select ‘Set Hotline: Stn/ Dept’, press enter key
- Enter Extension / Department Group Number
  OR
- Dial 151 and enter Extension Number / Department Group Number

To set Hot Outward Dialing for Trunk,

- Press DSS key assigned to Hotline.
- Scroll to select ‘Set Hotline: OG Trunk’, press enter key
- Enter TAC
  OR
- Dial 152-TAC

To set Hot Outward Dialing to External Number,

- Press DSS key assigned to Hotline.
- Scroll to select ‘Set Hotline: Ext Num’, press enter key
- Enter TAC
- Enter External Number #*
  OR
- Dial 153-TAC-External Number#*

You cannot set Hotline and Hot Outward Dialing on the same extension at the same time.

To set Hotline Timer,

- Press DSS key assigned to Hotline
- Scroll to select ‘Set Hotline Timer’, press enter key
- Enter Hotline Timer:000-255 seconds
  OR
- Dial 154-seconds (000-255)

  The default value of Hot Line Timer is 3 seconds.

To Cancel Hotline / Hot Outward Dialing,

- Press DSS key assigned to Hotline
- Scroll to select ‘Cancel Hotline’, press enter key.
  OR
- Dial 150

The cancellation code must be dialed from the dial tone. You have to be very quick in dialing the cancellation code, if the delay in the Hotline Timer is set to 1 or 2 seconds.

For SLT Users

To set Hotline on an Extension / Department Group,

- Dial 151-Extension Number / Department Group Number

To set Hot Outward Dialing for Trunk,

- Dial 152-TAC

To set Hot Outward Dialing for External Number,

- Dial 153-TAC-External Number#*

To set Hotline Timer,

- Dial 154-seconds (000-255)

To cancel Hotline / Hot Outward Dialing,

- Dial 150

When you set the Hotline Timer to '00' seconds (for immediate Hotline), you will not be able to dial any digits, not even the feature code to Cancel Hotline.

If you have set Immediate Hot Outward Dialing for a Trunk or External Number, you will not be allowed to dial any feature code, not even the feature code to cancel Hot Outward Dialing. However, if you need to cancel, you must follow the steps described below.

- Go OFF-Hook.
  You get the CO network Dial Tone.

- Dial by Digit.
  You will hear Pause/Silence.

- Wait for the duration of the Trunk Inter Digit Timer
  You will hear Pause/Silence.

- Press Flash.
  You will hear the Feature Tone.

- Dial the code to change the Hot Outward Dialing Timer (154) and change the duration of the timer
  –or–
  Dial the access code to cancel the Hot Outward Dialing (150).

  You get Confirmation Tone.

- Go ON-hook.
  You get the return ring of the trunk.

- Go OFF-Hook again.
  You get connected to the held trunk.

- Go ON-Hook.

Incoming CLI Modification

What's this?

For System users in countries, where the Calling Line Identification (CLI) received must be suitably modified before it can be used to dial out the number, SARVAM UCS offers the feature 'Incoming CLI Modification'.

The Incoming CLI received with the Country or Area Code, or both. However, the dialing pattern of the public network may require the received CLI to be prefixed with additional digits, to dial out the same number. Or the dialing pattern of the public network may require the CLI to be stripped off the prefixed digits to dial out the same number.

With the feature 'Incoming CLI Modification' programmed, the SARVAM UCS detects whether the incoming CLI is a local number, a national, or an international number. It modifies the incoming CLI accordingly, by adding or stripping off the prefixed digits so that the number can be dialed out as per the dialing pattern supported by the public network.

The modified CLI is presented to the extension phones and is stored in the "Call Logs", and SMDR (see "Station Message Detail Recording (SMDR)"). Extension users can call a number in the Call Logs without having to modify the CLI manually.

How it works

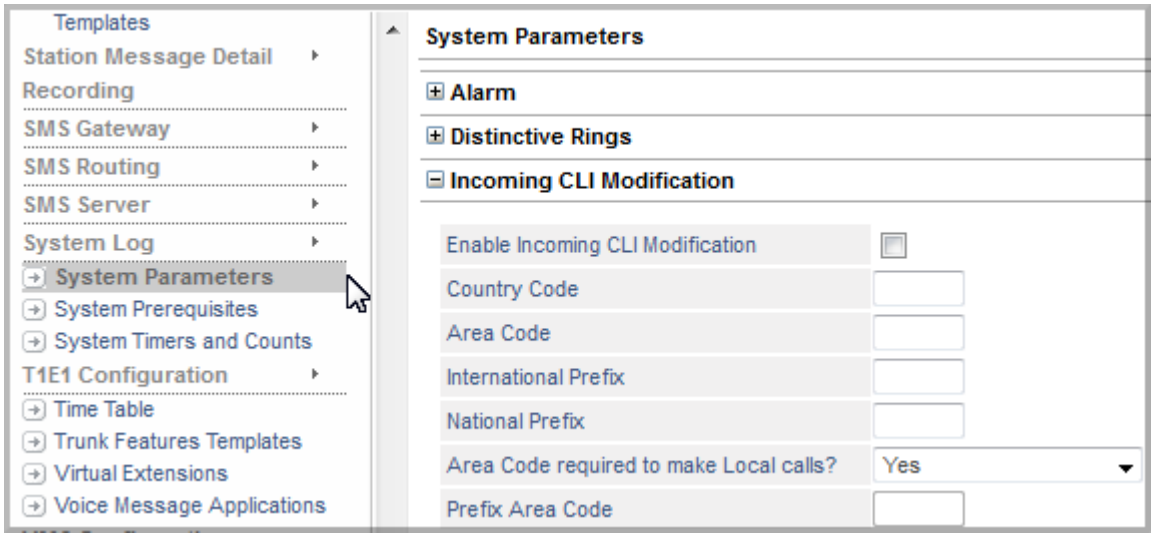
- Incoming CLI Modification parameters must be programmed in the system considering the dialing pattern supported by the local public network.
- Accordingly, SARVAM UCS matches the CLI received with the programmed parameters.
- It detects whether it is an international, national or local number.
- It modifies the CLI according as per the Modification parameters programmed.
- It presents the modified CLI to the extension; stores the modified CLI in the SMDR and in the Call Logs of the extension, provided it is a digital key phone.
- When the received CLI is dialed out by the extension user from Call Log, SARVAM UCS dials out the same number.

How to configure

For this feature to work, the **Incoming CLI Modification** check box in 'System Parameters' and the **Allow Incoming CLI Modification** check box in the respective trunk must be enabled. This can be done from Jeeves or by dialing SE commands from a Telephone. To know more, refer to "Configuring BRI Trunks", "Configuring CO Trunks", "Configuring E&M Lines", "Configuring Mobile Trunks", "Configuring SIP Trunks", "Configuring T1 Trunks" and "Configuring E1 Trunks".

Programming Incoming CLI Modification using Jeeves

- Log in as System Engineer.
- Under **Configuration**, click **System Parameters** to open the page.
- Click **Incoming CLI Modification** to expand.



Now, program these parameters:

- **Enable Incoming CLI Modification:** Enable this flag if you want to use the Incoming CLI Modification feature. By default, this flag is disabled.

If you receive CLI in dialable format, there is no need to use this feature. In such case, keep the flag disabled. You do not need to program any of the CLI modification parameters.

- **Country Code:** Enter the Country Code of the country where SARVAM UCS is installed. The Country Code helps SARVAM UCS detect whether the Incoming CLI received is a national or an international number. Do not enter any prefix for the Country Code. For example, if your SARVAM UCS is installed in USA, enter only '1' as the Country Code. Do not enter '+' or "00' as prefix to the country code '1'. By default the Country Code is '91' (India).

- **Area Code:** Enter the Area Code of the place where the SARVAM UCS is installed. The Area Code helps SARVAM UCS detect whether the Incoming CLI received is a local number. Do not enter any prefix for the Area Code. For example, if you want to enter Area Code for Mumbai, enter only '22'. Do not enter the prefix '0' to the area code. By default, Area Code is '265' (Vadodara city).

- **International Prefix:** Enter the digits that are required as Prefix for dialing International Numbers. The prefix may be up to 5 digits, with numbers from 00000 to 99999. By default, '00' is set as the prefix for dialing International numbers.

- **National Prefix**: Enter the digits that are required as Prefix for dialing long distance, National (within the country) numbers. The prefix may be up to 5 digits, with numbers from 00000 to 99999. By default, '0' is set as prefix for dialing national numbers.

- **Area Code required to make local calls?:** Depending on the dialing pattern of your local public telephone network, you may choose from the following options:

  - **No (Area Code not required)**: select this option if your public telephone network does not require the dialing of Area Code for local numbers.

  - **Yes (Area Code is required)**: select this option if your public telephone network requires you to dial the Area Code for local numbers.

- **Yes (Area Code with Prefix required):** select this option if you public telephone network requires you to dial Area Code with a particular Prefix for local numbers. If you select this option, you must also program the Prefix digits for the Area Code.

  By default, the option, 'No (Area Code not required) is selected.

- **Prefix Area Code:** If you have enabled the 'Area Code required to make local calls' flag in the previous parameter, enter the prefix digits for the area code for local calls in this field.

- Click **Submit** at the bottom of the page to save changes.

Programming CLI Based Routing Table using a Telephone

- Enter SE mode from a DKP/SLT.

  To enable/disable Incoming CLI Modification, dial:
  - ***5367-Flag***
    Where,
    Flag is
    0 for Disable
    1 for Enable
    Default: Disabled

  To program the Country Code, dial:
  - ***5368-Country Code***
    Where,
    Country Code is a number string up to 5 digits. The number string may consist of numbers from 00000 and 99999.
    Default: 91 (India)

    Program the Country Code without any prefix. For example, if you want to program USA as Country Code, enter '1' only (without the prefix '00' or '+')

  To program the Area Code, dial:
  - ***5369-Area Code***
    Where,
    Area Code is a number string up to 5 digits. The number string may consist of numbers from 00000 and 99999.
    Default: 265 (Vadodara)

    Program the Area Code without any prefix. For example, if you want to program Mumbai as Area Code, enter '22' only (without the prefix '0')

  To program International Call Prefix, dial:
  - ***5370-International Call Prefix***
    Where,
    International Call Prefix is a number string up to 5 digits. The number string may consist of numbers from 00000 and 99999.
    Default: 00

  To program National Call Prefix, dial:
  - ***5371-National Call Prefix***
    Where,

National Call Prefix is a number string up to 5 digits. The number string may consist of numbers from 00000 and 99999.
Default: 0

To enable/disable ‘Area Code required to make local calls?’, dial:

- ***5372-Code***
  Where,
  Code is
  1 for No (Area Code is not required)
  2 for Yes (Area Code is required)
  3 for Yes (Area Code with Prefix required)
  Default: 1

To program Prefix Digits to Area Code for Local Calls, dial:

- ***5373-Prefix Digits***
  Where,
  Prefix Digits to Area Code is a number string up to 5 digits. The number string may consist of numbers from 00000 and 99999.
  Default: Blank

- Exit SE mode.

Intercom

What is this?

The Intercom feature of SARVAM UCS enables extension users to connect quickly with any desired extension, without waiting for the called extension to answer.

- *SARVAM UCS will serve an Intercom call made by an extension only if:*
  - *the called extension is a DKP or a SIP Extension (Matrix Extended IP Phone or Standard SIP Phone).*
  - *the called extension is in idle state.*
  - *the called extension is able to identify the incoming call as an intercom call (applicable in the case of Standard SIP Phones).*
  - *the calling extension has Intercom in its Class of Service.*
  - *the Priority of the calling extension is higher than that of the called extension.*

- *On SIP extensions, SARVAM UCS supports Intercom using Call-INFO / Alert-INFO Message. For a list of IP phones on which this feature has been tested, see "SARVAM UCS Features tested on IP Phones of different Brands" in the Appendix.*

How it works

- A's extension number is 3001 with Priority Level '7' and 'Intercom' feature enabled in the Class of Service.
- B's extension number is 3003 with Priority Level '5'.
- A wants to quickly connect to B. A dials the Intercom feature code ***5** followed by B's number, 3003.
- SARVAM UCS places the Intercom call on B.
- B's extension is idle at the time of the call, and the speaker of B's phone goes OFF-Hook, creating a speech path between A and B.
- A can now talk with B.

Feature Interactions

- **Do Not Disturb (DND):** If the called extension has set DND, SARVAM UCS will not place the intercom call on the called extension.
- **Privacy from DND Override:** If DND as well as Privacy from DND Override is enabled in the Class of Service of the called extension, SARVAM UCS will reject the Intercom call.
- **Call Forward-Unconditional:** If the called extension has set Call Forward-Unconditional, SARVAM UCS will forward the intercom call to the forwarded destination number. The call placed on the forwarded destination will not be an Intercom call.
- **Call Forward-No-Reply:** If the called extension has set Call Forward-No-Reply, SARVAM UCS will not forward the intercom call to the forwarded destination number on the expiry of the No-Reply Timer.

- **Call Forward-Busy:** If the called extension has set Call Forward-Busy, SARVAM UCS will place the call on the forwarded destination number. However, this call (placed on the forwarded destination) will not be an Intercom call.

- **User Absent/Present:** SARVAM UCS will place the Intercom call on the called extension only if the status of the called extension is 'Present'.

- **Auto Call Back:** When the Intercom call is generated and the called extension is busy, the calling extension can set Auto Call Back on the called extension. When the called extension is free, SARVAM UCS will serve the Auto Call Back request set by the calling extension. The ACB call placed on the called extension will be a normal call.

.

- **DISA:** An Intercom call can be generated also from DISA mode.

- **Priority**: The calling extension must have a higher Priority level than the called extension.

When the Intercom call is generated on SIP Extension having multiple call appearance and already a call is present on the SIP Extension then the SARVAM UCS will place the Intercom call as normal call on the SIP Extension.

How to configure

To provide this feature to extension users, you must enable this feature in their Class of Service. For instructions, see "Class of Service (COS)" and "Station Basic Feature Template".

How to use

For EON & Extended IP Phone Users

To use intercom to call an extension:

- Press the DSS Key assigned to Intercom.
  OR
  Dial ***5**
- Dial the desired extension number.
- The called extension's speaker goes OFF-Hook.
- You are in speech with the called extension.
- You may talk.

For SLT Users

To use intercom to call an extension:

- Dial ***5** followed by the desired extension number.
- The called extension's speaker goes OFF-Hook.
- You are in speech with the called extension.
- You may talk.

Internal Call Restriction

What’s this?

Using this feature, the operator will be able to allow/restrict internal calls. The operator has flexibility to allow calls during day time and restrict calls during night time.

In SARVAM UCS this feature is implemented in the following ways:

- Assigning 'Guest Group' to the station
- Enable/disable Internal Call Barring (Call Block)

Assigning 'Guest Group' to the station

- Privacy of the user is very important in Enterprises and hospitals. In general, most of the users need to communicate only with reception, operator, pantry and such other service stations. He is not required to dial other user numbers. For example, in hospitals the patients need to call the nurse and not the doctor or other patients.

- Sometimes, a group of users occupy multiple rooms in the Enterprise. In such cases members of the group would like to communicate only among themselves and service stations.

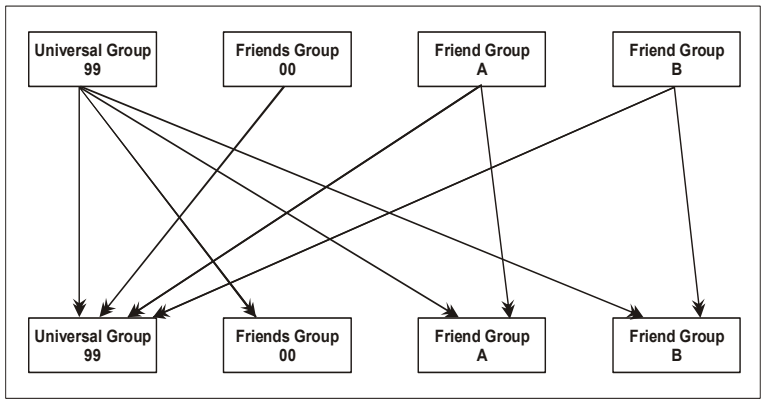
Essentially we need following logic:

- Certain service stations should be able to dial any other service station or any guest (for example Nurses, Operator).
- Certain stations should be able to dial only service stations (for example: patients, single user in the office).
- A group of station users need to dial amongst each other and the service stations (for example: doctors).

Above requirements are implemented as follows:

- Solo Group (for example patients, single user of the station). This is classified in group 00.
- Universal Group (Nurses and other services staff for example). This is classified in group 99.
- Friend's Group (for example, Group of visitors). Such a group of station users can be assigned to any group number from 01 to 98.

Following diagram depicts this logic:



How to use

From SA mode:

Dial, ***1072-904-Flexible Number-Guest Group***
OR
Press DSS key for 'Guest Group'
Where,
Flexible Number is the code given to access a station.
Guest Group is from 00 to 99.
By default, the guest group assigned is 99.

- The user can call any body in his 'guest group' number '99'.
- The station users in one 'guest group', '99' can call each other within the group as well as the station users in other groups.
- The default Guest group of all the stations = 99.

Enable/Disable Internal Call Barring (Call Block)

SARVAM UCS supports a feature, to allow or bar the Internal Calls for the users of SARVAM UCS. This is called Call Blocking.

How to use

From SA mode:

Dial, ***1072-045-Code***
OR
Press DSS key for 'Enable/Disable Internal Call Block'.
Where,

Code	Meaning
0	Disable Call Block, (that is, Internal Calls allowed)
1	Enable Call Block (that is, Internal Calls barred)

- When this feature is enabled, the LED on the DSS key assigned to 'Call Block' will glow Red and when disabled, the LED will be turned Off.
- When the operator issues "Call Block”- Enable command, all the station users will be assigned guest group = 00.
- When the operator issues "Call Block”- Disable command, all the station users will be assigned guest group = 99.

- *Even after issuing SA command for 'Call Block', the operator can use the SA command **1072-904** to change the guest group of any station user.*
- *Please refer the Hospitality Manual for more details.*

Interrupt Request (IR)

What's this?

Interrupt Request allows you to break into an on-going conversation after intimating the extension user about the interruption.

In case of an important or urgent trunk call the operator can put the call on hold, interrupt the busy extension user to inform about the urgent call and then transfer the urgent call.

How it works

- A, B and C are extension users.
- A and B are talking to each other.
- C calls A.
- C gets busy tone.
- C dials Interrupt Request feature code.
- C gets Ring Back tone (RBT) and A gets beeps indicating a new call.
- To answer C's call, A must dial **Flash** before the expiry of the Interrupt Request Timer. A will be in speech with C. B will be put on hold and will get music on hold.
- If A does not dial Flash before expiry of the Interrupt Request Timer, C's call will be disconnected.

Feature Interactions

- **Call States:**
  - Interrupt Request works only if the dialed extension is busy. The dialed extension may be busy with another extension or trunk (external number).
  - Interrupt Request works only if the user about to be interrupted in is in a two-way normal speech with another user or external party. However, it will not work if the conversation is being recorded.
  - It will not work if the busy signal is due to the user being Off-hook, or in the middle of dialing, or accessing a feature of the SARVAM UCS.
- **"Call Toggle":** Once A and C comes in speech with each other, A can toggle between B and C using Call Toggle feature.
- **Privacy against Interrupt Request:** If the feature 'Privacy against Interrupt Request is enabled for an extension, it cannot be interrupted. See "Privacy".
- **"Priority":** No Interaction with Interrupt Request. If 'A' has lower priority than 'B' but has Interrupt Request enabled; A can interrupt B.

- **“Call Taping”**: Interrupt Request will not work when the two-way conversation between the users is being taped.

How to configure

To be able to use Interrupt Request, extension users must have this feature enabled in their “Class of Service (COS)” in their “Station Basic Feature Template”.

For instructions on configuring the different extension port types:

- “Configuring SLT Extensions”
- “Configuring DKP Extensions”
- “Configuring ISDN Terminals”
- “Configuring SIP Extensions”

If required you may also change the default value of the *Interrupt Request Timer.* For instructions see “System Timers and Counts”.

How to use

For EON & Extended IP Phone Users

When dialed extension is busy,

- Press DSS Key assigned to Interrupt Request
  OR
- Dial 3 on Busy Tone

For SLT Users

When dialed extension is busy,

- Dial 3 on Busy Tone.

Last Caller Recall

What's this?

SARVAM UCS offers a facility—Last Caller Recall— to trace the extension that last made the call to your extension.

How it works

- A's extension number is 2001.
- A wants to know who made the last call to extension 2001.
- A dials Last Caller Recall feature access code (Default:1092).
- The extension that last called 2001 rings.
- When the called extension answers, speech is established between A and the called extension user.

On SIP extensions, SARVAM UCS supports Last Caller Recall using text (lcr) as access code.For a list of IP phones on which this feature has been tested, see "SARVAM UCS Features tested on IP Phones of different Brands"in the Appendix.

How to use

For EON & Extended IP Phone Users

- Press DSS Key assigned to Last Caller Recall.
  OR
- Go OFF-Hook
- Dial **1092**
  The system dials out the extension number that last called your extension.

For SLT Users

- Lift the handset.
- Dial **1092**
  The system dials out the extension number that last called your extension.

Last Number Redial

What's this?

The system redials the last number string (external/internal) dialed from an extension. By default the system stores only the external numbers in the Last Number Redial List. If you want the system to store internal calls in this list, make sure you enable the **Store Internal Calls in Redial Call Log** check box in the "System Parameters".

How it works

- Extension A dials the feature access code for 'Redial'.
- If Extension A is a SLT, the system dials the last number dialed from Extension A using the same trunk access code used for dialing that number. However, the system will not place the call, if the last number dialed is an internal number.
- If Extension A is a DKP or an Extended IP Phone, last 16 numbers dialed by Extension A are displayed on the phone's LCD.
- Extension A may select the number to be dialed out. The system will dial out this number using the same trunk access code used for dialing this number.

If Extension A has 'Dynamic Lock' set and uses Redial feature, the system will check for Toll Control as per the Lock Level set for Extension A before dialing out the number.

How to configure

No particular configuration is required for this feature to work. Redial is included in the Basic Features allowed to all extensions by default in their "Class of Service (COS)". So, all extensions can use the Redial feature.

How to use

For EON & Extended IP Phone Users

- Press Redial Key.
  Or
- Dial 7
  A List of last 16 numbers dialed will appear on your phone's display.
- Scroll to select the desired number.
- Press Enter key.
- The system dials out the number.

For SLT Users

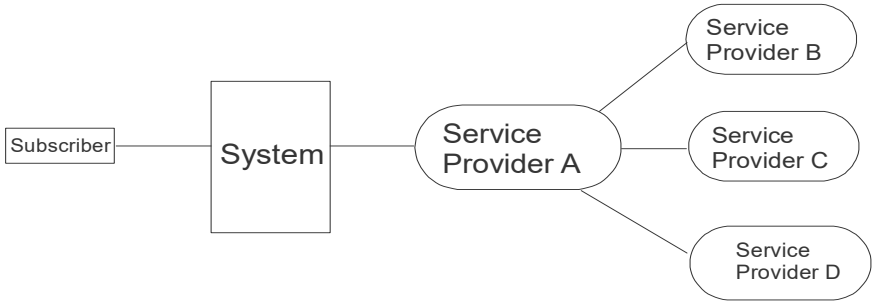
- Lift Handset
- Dial 7
- The system dials out the number last dialed from your extension.

Least Cost Routing-Carrier Pre-Selection

What's this?

This type of Least Cost Routing is used in countries where the same service provider offers local call and long distance calling services. These service providers allow subscribers to select the service provider or Carrier for long distance calling.

For example,



The subscriber of Service Provider A must grab trunk lines of Service Provider A to call other subscribers in the local area.

However, when the subscriber of Service Provider A wants to make a long distance call, the subscriber must dial a prefix to select the a carrier (trunk) of the desired long distance, Service Provider B, C and D. Thus, the subscriber accesses a secondary service provider by dialing a short code or prefix for long distance calling.

This feature works on the basis of "Automatic Number Translation". Using Automatic Number Translation, SARVAM UCS adds the code of the appropriate secondary Service Provider to the number string dialed by the extension user to route the call to the desired secondary Service Provider.

How to configure

To use this feature, you must do the following:

- Select an Automatic Number Translation Table Number from 1 to 8.
- Configure the Automatic Number Translation Table. In the Automatic Number Translation Table, in the Dialed Number String column, enter the long distance numbers that extension users will dial. In the Add Prefix column, enter the digits which are to be added as prefix to the Dialed Number string by the system before dialing it out and in the Strip Digits column enter the number of digit(s) to be stripped off by the system from the Dialed Number string before dialing it out. For example, the code '961' for Service Provider B must be prefixed to the number '2630555' dialed by extension users, you must enter '2630555' in the Dialed Number String column, '961' in the Add Prefix column and '0' in the Strip Digits column.
- Enable"Automatic Number Translation" feature on the trunk (CO/Mobile/SIP/T1E1/BRI) of your Primary Service Provider in the "Outgoing Trunk Bundle" and apply the ANT Table number you configured to this OG Trunk Bundle.
- Create "OG Trunk Bundle Group" that includes the OG Trunk Bundle you created with Automatic Number Translation as member.

- Assign the OG Trunk Bundle Group (containing the OG Trunk Bundle with ANT) to the extensions for the Time Zones, that is, Working Hours, Non-Working Hours, Break Hours, in the “Station Basic Feature Template” of the extensions.

License Management

What's this?

The application SARVAM UCS and the features that it supports, require the purchase of licenses. When you buy the ETERNITY GENX Platform, you get a unique default license key for the platform.

SARVAM UCS Application License[309]

To use ETERNITY GENX as the Unified Communication Server, you need to purchase the **SARVAM UCS SME** Application software license.

To use ETERNITY LENX/MENX as the Unified Communication Server, you need to purchase the **SARVAM UCS ENT** Application software license.

To use ETERNITY PENX as the Unified Communication Server, you need to purchase the **SARVAM UCS SMB** Application software license.

If you do not have the license for the SARVAM UCS SME/ENT Application and you do not start the Demo Period, the system will disconnect all the connected calls[310] (internal or external, incoming or outgoing) from any port after 60 seconds[311]. For details, refer "Demo Provision".

Described below are the features of SARVAM UCS for which you require licenses. To know the name of the license you need to purchase, see *"Supported Licenses"*.

IP Subscribers (For SIP Extensions)

SARVAM UCS supports upto 999 SIP Extensions. With a license, you can register SIP-enabled devices with the SARVAM UCS. Without a license you cannot register any SIP Extension.

MATRIX VARTA User Licenses

SARVAM UCS supports three types of user licenses for VARTA Users — VARTA Essential Users, VARTA Professional Users and VARTA Collaboration Users.

Following table lists the features which will be supported in MATRIX VARTA WIN200 / VARTA ADR100 / VARTA AMP100 when you activate the respective license.

Features	VARTA Essential Users			VARTA Professional Users			VARTA Collaboration Users		
	WIN 200	ADR 100	AMP 100	WIN 200	ADR 100	AMP 100	WIN 200	ADR 100	AMP 100
Making Calls	✓	✓	✓	✓	✓	✓	✓	✓	✓
Receiving Calls	✓	✓	✓	✓	✓	✓	✓	✓	✓

309. Refer *"Pre-activated Licenses"*.
310. *Connected calls means where speech is connected between calling party and called party even if the called party port is not matured.*
311. *Make sure the NX DBM VOCODER64 module is installed for SIP calls.*

Hold	✓	✓	✓	✓	✓	✓	✓	✓	✓
Transfer	✓	✓	✓	✓	✓	✓	✓	✓	✓
Blind Transfer	✓	✓	✓	✓	✓	✓	✓	✓	✓
One Touch Transfer	✓	✓	✓	✓	✓	✓	✓	✓	✓
3-Party Audio Conference	✓	✓	✓	✓	✓	✓	✓	✓	✓
Video Call	✓	✓	✓	✓	✓	✓	✓	✓	✓
Intercom	✓	✓	✓	✓	✓	✓	✓	✓	✓
Voicemail	✓	✓	✓	✓	✓	✓	✓	✓	✓
Call Forward	✓	✓	✓	✓	✓	✓	✓	✓	✓
Do Not Disturb	✓	✓	✓	✓	✓	✓	✓	✓	✓
Presence	✓	✓	✓	✓	✓	✓	✓	✓	✓
IM and SMS	✓	✓	✓	✓	✓	✓	✓	✓	✓
Favorites	✓	✓	✓	✓	✓	✓	✓	✓	✓
Global Directory Access	✓	✓	✓	✓	✓	✓	✓	✓	✓
All Menu Features	✓	✓	✓	✓	✓	✓	✓	✓	✓
All Call Features	✓	✓	✓	✓	✓	✓	✓	✓	✓
Hotkeys	✓	×	×	✓	×	×	✓	×	×
Multiparty Audio Conference				✓	✓	✓	✓	✓	✓
Handover				✓	✓	✓	✓	✓	✓
Drag and Drop Transfer				✓	×	×	✓	×	×
Drag and Drop Conference				✓	×	×	✓	×	×
Contact Grouping				✓	×	×	✓	×	×
BLF Subscription				✓	✓	✓	✓	✓	✓
DSS Soft Keys				✓	×	×	✓	×	×
DSS Soft Keys for Mobile users				×	✓	✓	×	✓	✓
Call Transfer to other user's Voicemail (Blind Transfer to VMS)				✓	×	×	✓	×	×
Click to Call				✓	×	×	✓	×	×
Outlook Integration							✓	×	×
Presence Contact Card Integration							✓	×	×
Calendar Integration							✓	×	×

To know more about Matrix VARTA WIN200 refer to the *MATRIX VARTA WIN200* User Guide and for VARTA UC Clients for Mobile refer to the respective User Guide — *Matrix ADR100* User Guide and *Matrix AMP100* User Guide.

Expansion Slots[312]

Expansion Slots License is required to expand the number of functional universal slots in the system. This license is applicable only for Firmware version V1R3 and later.

If you have upgraded the system firmware to V1R3 and later in the old ETERNITY LENX/MENX/GENX system, the Expansion Slots license will be applicable for the universal slots. No universal slots will be functional by default. You must purchase the license to activate the universal slots as required.

If you have purchased the new ETERNITY LENX/MENX/GENX system with the firmware V1R3 and later, the Expansion Slots license will be applicable for the universal slots.

- For ETERNITY LENX, the first eight universal slots after the power supply card in the first rack will be functional by default.
- For ETERNITY MENX, the first eight universal slots after the power supply card will be functional by default.
- For ETERNITY GENX, the first four universal slots after the power supply card will be functional by default.

  If you require more functional universal slots, you must purchase the Expansion Slots License to activate the same.

Matrix provides two Expansion Slots licenses:

- SARVAM EXP4 SME for ETERNITY GENX system
- SARVAM EXP4 ENT for ETERNITY MENX and LENX system.

  Each SARVAM EXP4 SME/ENT license will provide the activation for next four universal slots in the sequence.

VOCODER Channels

VOCODER Channels license is required for SIP calls. The maximum number of VOCODER channels that will be supported would be as per the license you purchase. SIP calls uses the VOCODER Channels, if you select RTP Mode for routing SIP calls. Refer “RTP Mode” for details of VOCODER Channels usage during SIP calls.

The system supports two[313] NX DBM VOCODER64 Modules. You must purchase the module separately. Each NX DBM VOCODER64 module supports a maximum of 64 VOCODER channels.

The system provides 4 pre-activated VOCODER channels by default. To use these channels make sure you have installed atleast one NX DBM VOCODER64 module. If you require more channels, you can purchase the channel licenses according to your requirement.

If you require more than 64 VOCODER channels, you must install another NX DBM VOCODER64 Module. During a SIP call, VOCODER channel is required and if no free channel is available, the system will reject the call.

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312. This License is not applicable to ETERNITY PENX.
313. ETERNITY PENX supports only one NX DBM VOCODER64 Module.

VMS Channels

VMS Channels license is required for allowing the VMS calls. The system allows simultaneous VMS calls as per the license only.

The NX DBM VMS64 Module is an optional module. If required, you may purchase it separately. The system supports a maximum of 64 channels out of which 4 channels are provided by default. If you require more channels, you can purchase the channel licenses according to your requirement.

If VMS call request is received and no free VMS channels are available, the system will reject the call.

CTI

CTI stands for Computer Telephony Integration. CTI is a technology that integrates a telephone and a computer. With the CTI license, the Matrix TAPI Service Provider (TSP) acts as a link to integrate the interactions between your telephones and your computers. The computer in which you install this application functions as a Client and SARVAM UCS functions as a Server. In an organization, it is possible that the data is stored in different computers. In this scenario, the Matrix TAPI Service Provider can be installed in three (maximum) different computers, if required.

Q-Sig

When SARVAM UCS is networked with another SARVAM UCS or with any other ISDN-System, *Q-Sig* or Q-Signaling is supported to facilitate feature transparency between the systems in the network. Q-Signaling will be activated when you buy the license for the same.

Hospitality Management System

This functional module contains a set of special telephone and guest/patient management features for hospitality and accommodation establishments like hotels and hospitals, which SARVAM UCS supports when it is deployed in a hotel or hospital. When you buy the license for this module, the following features will be activated:

- Room Shift
- Check-In, Check-Out
- Change Room Occupancy Status
- Floor Service
- Change Room Clean Status
- Front Desk User GUI (web interface)

Property Management System (PMS)

SARVAM UCS supports interface for PMS, the application software commonly used by hotels to manage their administrative functions. The PMS Interface supports proprietary PMS protocols of Matrix (Matrix PMS Type 1 and Type 2), Micros Opera and Softbrands Extended Starlight. To be able to select any of these PMS Protocols, you must buy a license.

License for the Hospitality module is a pre-requisite for PMS.

E911

SARVAM UCS supports the E911 license. You will be able to dial the Emergency number 911 from the system only if you buy this license.

This license is applicable in new systems dispatched with the Firmware V1R5.2 and later only.

Gateway

When SARVAM UCS is to be used with Gateway functionality, a license is required to activate this functionality.

PLCC[314]

This functional module contains a set of special features supported by the SARVAM UCS when it is deployed in a Power Line Carrier Communication Network of electric utilities. When you buy the license for this module, the following features will be enabled:

- CCS Signaling when End Point and Transit.
- Express Signaling.
- Seizure Pulse and Release Pulse Signaling.

SMS Server

With the SMS Server license, you can:

- Send/ receive SMS to/from individuals or groups using the Mobile Port of SARVAM UCS.
- Forward SMS received on Mobile Port as Emails to users through the Email Client.
- Forward Email of the users as SMS to the Mobile users through the Mobile Port.

The SMS Server application works as an intermediary between the GSM Short Message Service and the SARVAM UCS. The Server supports multipart, 7 bit text messages as well as UNICODE messages.

The Server functions as an SMTP Client to send emails and as a POP3 Client to receive emails. SMS Server supports three types of Emails—Plain Text, HTML and MIME— from its mail clients.

SMS Gateway

With the SMS Gateway license, you can send/receive messages to/from individuals, selective groups or masses using the Mobile Port of SARVAM UCS.

SARVAM UCS allows you to register multiple SMPP Clients (Software Applications used for sending/receiving messages) with SARVAM UCS. SARVAM UCS functions as an SMPP Server. These Clients can send/receive messages using the Mobile port/s of SARVAM UCS.

Redundancy License

With Redundancy license, you can increase the system reliability by providing uninterrupted communication.

You must activate this license in case you want Hardware and Application redundancy in ETERNITY LENX/MENX. To know in detail about redundancy, refer to “Redundancy”.

For redundancy, you must install two CPU cards in the system - CPU Card 1 and CPU Card 2.

This redundancy license allows the CPU Card 1 (Active) to share its licenses with the CPU Card 2 (Standby) during redundancy. In other words, if you activate redundancy license along with the required licenses in the CPU Card 1

314. ETERNITY PENX does not support PLCC

(Active) and activate no licenses in CPU Card 2 (Standby), you will still be allowed to access all the licensed features activated in CPU Card 1, even when this CPU Card 1 fails. However, you can use these licenses for a limited period of time.

You can activate redundancy license either in CPU Card 1 or CPU Card 2 as per your requirement. However, it is recommended to activate this license in the CPU Card which has maximum number of licenses activated. This is important as during redundancy, the system will allow you to use the set of features and facilities as per the licenses activated in that CPU Card.

Supported Licenses

Refer to the table below to know the name of the respective licenses you need to activate for each feature.

License Name	Description
SARVAM UCS SME	Unified Communication Server License for Small-Medium Enterprise. This is a UC Server License for ETERNITY GENX.
SARVAM UCS ENT	Unified Communication Server License for an Enterprise. This is a common UC Server License for ETERNITY LENX and ETERNITY MENX.
SARVAM UCS SMB	Unified Communication Server License for an Enterprise. This is a UC Server License for ETERNITY PENX.
SARVAM EXP4 SME	License for SARVAM UCS SME to activate four Expansion Slots.
SARVAM EXP4 ENT	License for SARVAM UCS ENT to activate four Expansion Slots.
SARVAM IPSUB5	License for 5 IP Subscribers to create 5 SIP Users.
SARVAM IPSUB10	License for 10 IP Subscribers to create 10 SIP Users.
SARVAM IPSUB50	License for 50 IP Subscribers to create 50 SIP Users.
SARVAM IPSUB100	License for 100 IP Subscribers to create 100 SIP Users.
SARVAM IPSUB500	License for 500 IP Subscribers to create 500 SIP Users..
SARVAM VARTA USER5E	License for 5 VARTA UC Users with Essential features. This license is required to register Android/iOS/Windows UC Clients.
SARVAM VARTA USER10E	License for 10 VARTA UC Users with Essential features. This license is required to register Android/iOS/Windows UC Clients.
SARVAM VARTA USER50E	License for 50 VARTA UC Users with Essential features. This license is required to register Android/iOS/Windows UC Clients.
SARVAM VARTA USER100E	License for 100 VARTA UC Users with Essential features. This license is required to register Android/iOS/Windows UC Clients.
SARVAM VARTA USER500E	License for 500 VARTA UC Users with Essential features. This license is required to register Android/iOS/Windows UC Clients.
SARVAM VARTA USER5P	License for 5 VARTA UC Users with Professional features. This license is required to register Android/iOS/Windows UC Clients.
SARVAM VARTA USER10P	License for 10 VARTA UC Users with Professional features. This license is required to register Android/iOS/Windows UC Clients.
SARVAM VARTA USER50P	License for 50 VARTA UC Users with Professional features. This license is required to register Android/iOS/Windows UC Clients.

License Name	Description
SARVAM VARTA USER100P	License for 100 VARTA UC Users with Professional features. This license is required to register Android/iOS/Windows UC Clients.
SARVAM VARTA USER500P	License for 500 VARTA UC Users with Professional features. This license is required to register Android/iOS/Windows UC Clients.
SARVAM VARTA USER5C	License for 5 VARTA UC Users with Collaboration features. This license is required to register Android/iOS/Windows UC Clients.
SARVAM VARTA USER10C	License for 10 VARTA UC Users with Collaboration features. This license is required to register Android/iOS/Windows UC Clients.
SARVAM VARTA USER50C	License for 50 VARTA UC Users with Collaboration features. This license is required to register Android/iOS/Windows UC Clients.
SARVAM VARTA USER100C	License for 100 VARTA UC Users with Collaboration features. This license is required to register Android/iOS/Windows UC Clients.
SARVAM VARTA USER500C	License for 500 VARTA UC Users with Collaboration features. This license is required to register Android/iOS/Windows UC Clients.
SARVAM VOCODER CHNL4	License for activating 4 additional VOCODER Channels.
SARVAM VOCODER CHNL16	License for activating 16 additional VOCODER Channels.
SARVAM VMS CHNL4	License for activating 4 additional VMS Channels.
SARVAM VMS CHNL16	License for activating 16 additional VMS Channels.
SARVAM CTI SME	License for SARVAM UCS SME to enable CTI functionality to connect to third party CTI application.
SARVAM QSIG SME	License for QSIG interface for SARVAM UCS SME to contact with other Matrix or third party System for seamless calling and inter-working of certain System features.
SARVAM HOSPITALITY SME	License for Hospitality functions suite for SARVAM UCS SME to support hospitality functions and features used in Hotel-Motel.
SARVAM PMS SME	License for Property Management System interface for SARVAM UCS SME to contact with third party PMS used in Hotel-Motel.
SARVAM GATEWAY SME	License for SARVAM UCS SME to support Gateway functions in addition to UCS functions.
SARVAM SMS SERVER SME	License for SARVAM UCS SME to enable SMS Server functionality to connect to any Email Client and send/receive Email to SMS and vice versa over GSM SIM installed on GSM Interface Card.
SARVAM SMS GATEWAY SME	License for SARVAM UCS SME to enable SMS Gateway functionality to connect to third party SMS Gateway Clients and send/receive SMS over GSM SIM installed on GSM Interface Card.
SARVAM CTI ENT	License for SARVAM UCS ENT to enable CTI functionality to connect to third party CTI application.
SARVAM QSIG ENT	License for QSIG interface for SARVAM UCS ENT to contact with other Matrix or third party System for seamless calling and inter-working of certain System features.
SARVAM HOSPITALITY ENT	License for Hospitality functions suite for SARVAM UCS ENT to support hospitality functions and features used in Hotel-Motel.

License Name	Description
SARVAM PMS ENT	License for Property Management System interface for SARVAM UCS ENT to contact with third party PMS used in Hotel-Motel.
SARVAM GATEWAY ENT	License for SARVAM UCS ENT to support Gateway functions in addition to UCS functions.
SARVAM SMS SERVER ENT	License for SARVAM UCS ENT to enable SMS Server functionality to connect to any Email Client and send/receive Email to SMS and vice versa over GSM SIM installed on GSM Interface Card.
SARVAM SMS GATEWAY ENT	License for SARVAM UCS ENT to enable SMS Gateway functionality to connect to third party SMS Gateway Clients and send/receive SMS over GSM SIM installed on GSM Interface Card.
SARVAM HOSPITALITY E911 SME	License for SARVAM UCS SME to enable dialing of emergency number 911.
SARVAM HOSPITALITY E911 ENT	License for SARVAM UCS ENT to enable dialing of emergency number 911.
SARVAM REDUNDANCY ENT	License for SARVAM UCS ENT to support Hardware and Application Redundancy.
SARVAM CTI SMB	License for SARVAM UCS SMB to enable CTI functionality to connect to third party CTI application.
SARVAM QSIG SMB	License for QSIG interface for SARVAM UCS SMB to contact with other Matrix or third party System for seamless calling and inter-working of certain System features.
SARVAM HOSPITALITY SMB	License for Hospitality functions suite for SARVAM UCS SMB to support hospitality functions and features used in Hotel-Motel.
SARVAM PMS SMB	License for Property Management System interface for SARVAM UCS SMB to contact with third party PMS used in Hotel-Motel.
SARVAM GATEWAY SMB	License for SARVAM UCS SMB to support Gateway functions in addition to UCS functions.
SARVAM SMS SERVER SMB	License for SARVAM UCS SMB to enable SMS Server functionality to connect to any Email Client and send/receive Email to SMS and vice versa over GSM SIM installed on GSM Interface Card.
SARVAM SMS GATEWAY SMB	License for SARVAM UCS SMB to enable SMS Gateway functionality to connect to third party SMS Gateway Clients and send/receive SMS over GSM SIM installed on GSM Interface Card.
SARVAM HOSPITALITY E911 SMB	License for SARVAM UCS SMB to enable dialing of emergency number 911.

Pre-activated Licenses

Pre-activated Licenses	**Systems purchased on January 1, 2021 and later**	**Systems purchased before January 1, 2021**
SARVAM UCS SME	Yes	No
SARVAM UCS ENT	Yes	No
SARVAM UCS SMB	Yes	No

Pre-activated Licenses	Systems purchased on January 1, 2021 and later	Systems purchased before January 1, 2021
IP SUBSCRIBERS (SIP EXTENSIONS)	5 (SME,ENT & SMB)	5 (SME,ENT & SMB)
VOCODER CHANNELS	4 (SME,ENT & SMB)	4 (SME,ENT & SMB)
VMS CHANNELS	4 (SME,ENT & SMB)	4 (SME,ENT & SMB)
VARTA USERS	0 (SME,ENT & SMB)	0 (SME,ENT & SMB)
EXPANSION SLOT	4 - SME 8 - ENT NA - SMB	4 - SME 8 - ENT NA - SMB

Other feature licenses mentioned in “Supported Licenses” need to be purchased.

Demo Provision

Demo provision for licensed features is useful for the following scenarios:

- when the customer’s system having licensed features cannot be repaired on-site and a standby system needs to be installed.
- when end user demands to use all the licensed features on trial basis before actually purchasing the license.

Demo Provision enables you to use the SARVAM UCS Application as well as all the licensed features it supports, free of cost for a period of 60 days. In this case, the application starts even if you have not purchased the SARVAM UCS SME/ENT/SMB license.

All the Universal Slots will be functional during the Demo period irrespective of the number of activated SARVAM EXP4 SME/ENT licenses.

The Demo Provision lets you access and use all the licensed features and functionalities[315] supported by the application.

When Demo Mode is activated, VARTA Collaboration Users license is by default assigned to all SIP Extensions internally. You do not need to configure it manually.

To avail this facility,

- Open Jeeves.
- Log in as System Engineer.

315. For VoIP and VMS, the number of channels that will be supported in demo period will be equal to the total number of channels available in the respective hardware module/s installed in the System.

Under **Configuration**, click **License Management**. The License Management page opens.

- Under **Demo Period**, click the **Start** button. The demo period for using licensed feature in your system starts and the Start button will change to Pause. The demo period is of 60 days.
- Once the Demo period starts, you can click on **View Profile** button to view the list of all the features and functionalities supported.

  You may Pause the Demo Period, if required. When you pause the demo period, all licensed features will work as per the license key installed in the system.

  If you want to use licensed feature after the expiry of the demo period, you must purchase the license key and activate it in your system.

When you default the SARVAM UCS, the demo period will not be reset.

How to activate your License

For the functional modules and features described above, you would need to activate a valid License Key.

Instructions for Matrix Channel Partners

Your license voucher may be a paper or a PDF (protected) file.

You may activate your License Online. For this, keep the following items ready:

- The License Voucher containing the 16-digit PIN.
- A valid, unique User ID and Password from the Matrix License Support Centre.
- Access to Internet.
- Current License Key of the system.

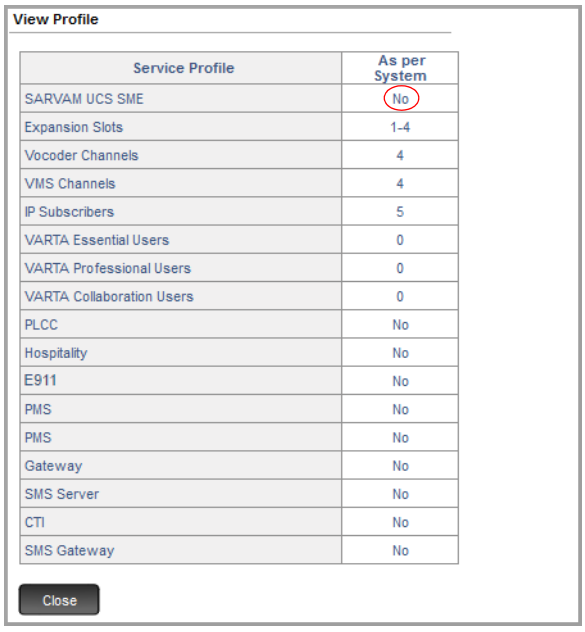
To activate the License Key *online*,

- Open Jeeves.
- Log in as System Engineer.
- Under **Configuration**, click **License Management**.

The License Management page opens.

- Note down or copy the current **License Key** on this page.

  If you wish to view the features and functions currently available on your system, click the **View Profile** button.



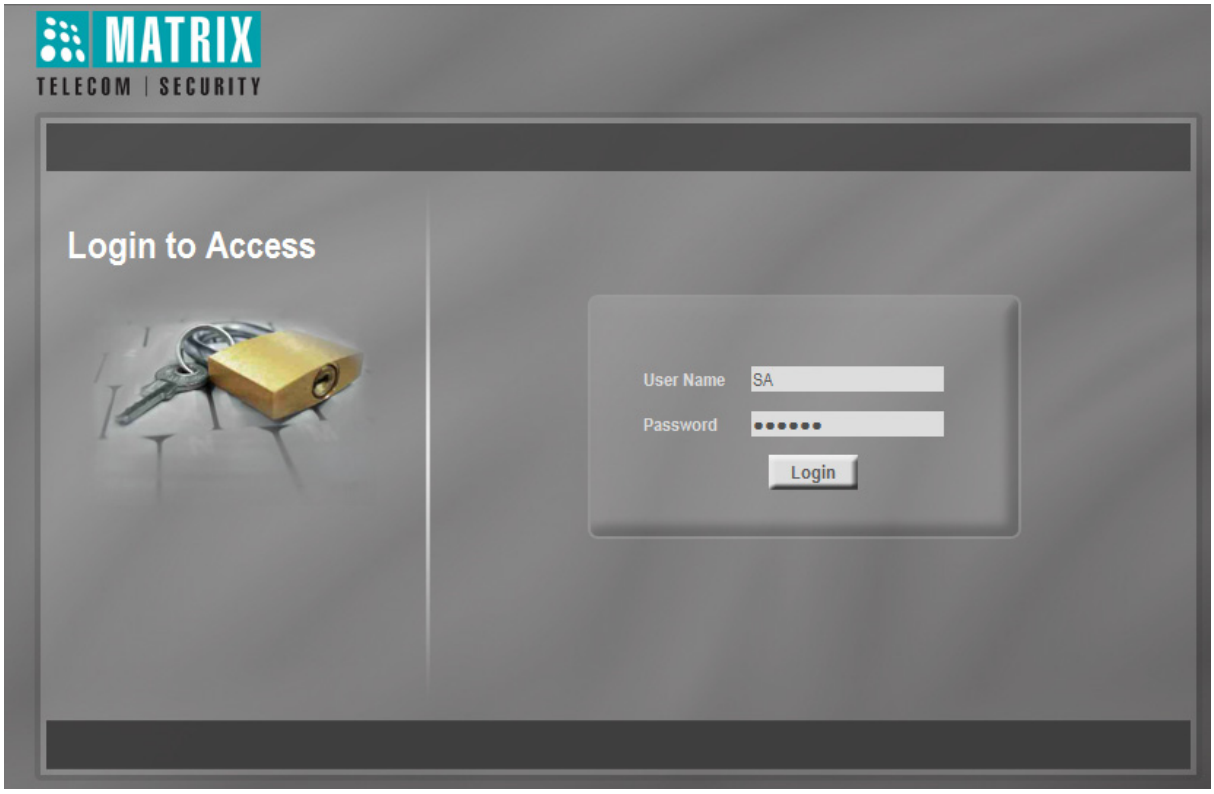
  A new window opens which displays the features and functionalities that are currently available to you.

- Keep your Current License Key and the License Voucher (paper or PDF) ready.
- Open a new window on your browser.

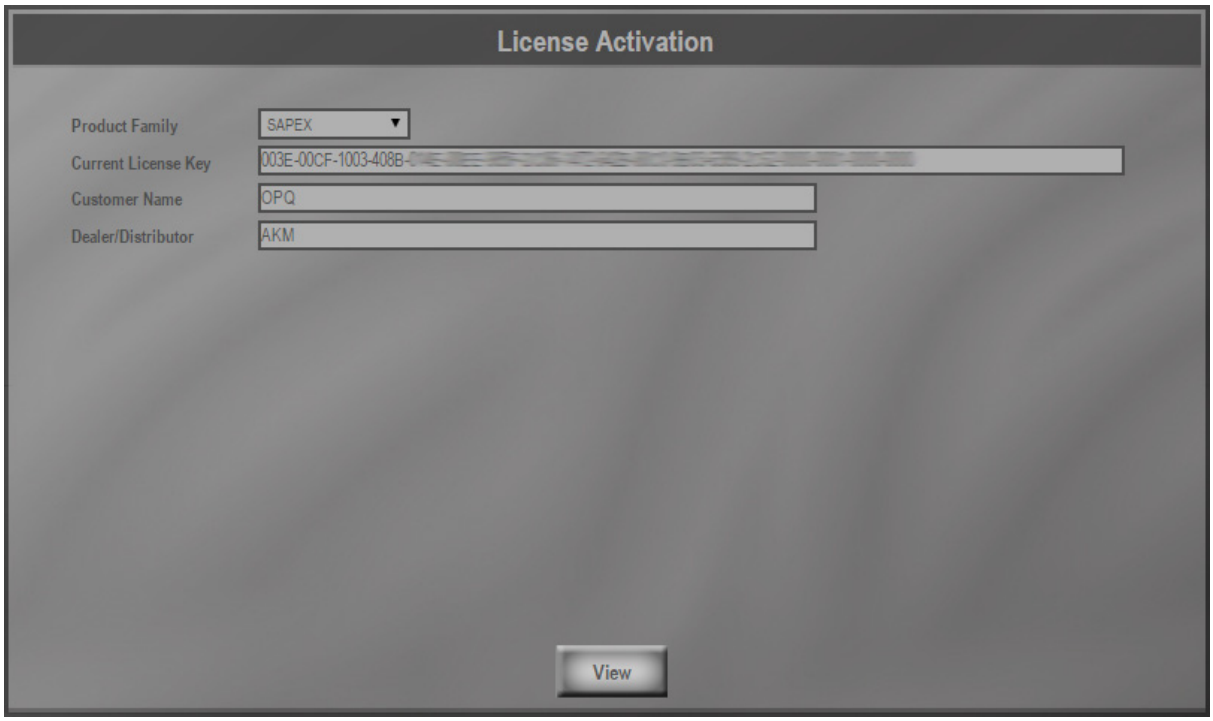
  Enter **http://www.matrixcomsec.com/support/license-activation/** in the address bar.

The Login to Access page will open.

- Enter your **User Name** and **Password** provided by Matrix and click the **Login** button.

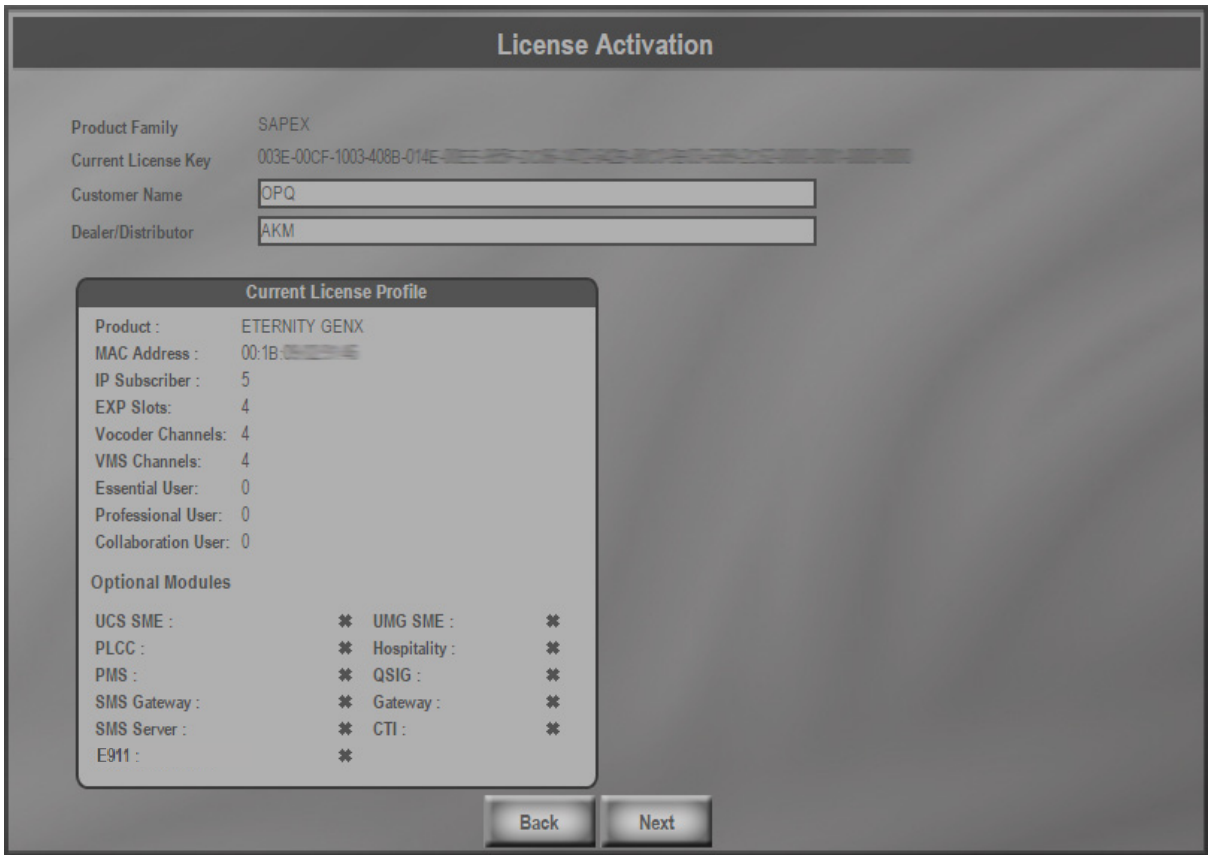


On successful login, the **License Activation** page will open.



- As **Product Family**, select the option **SAPEX**.
- In the **Current License Key** field, type the current product license key you noted or paste the key you copied from the *License Management* page of Jeeves.

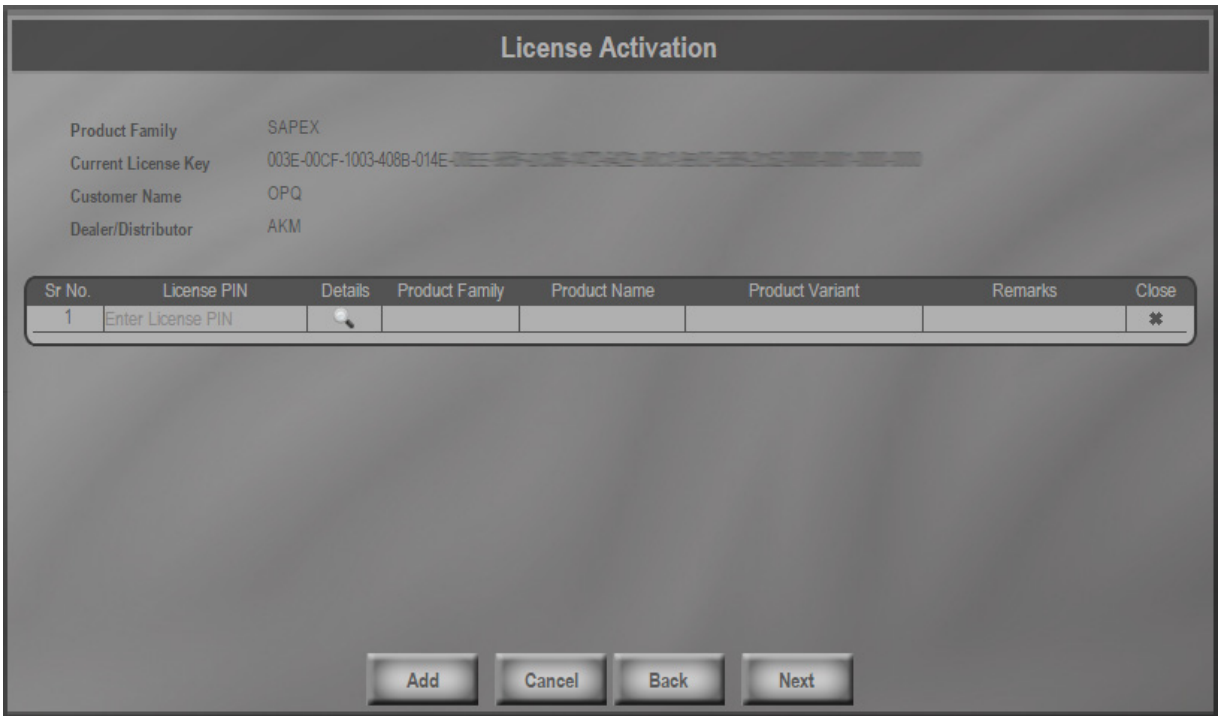
- Click the **View** button.



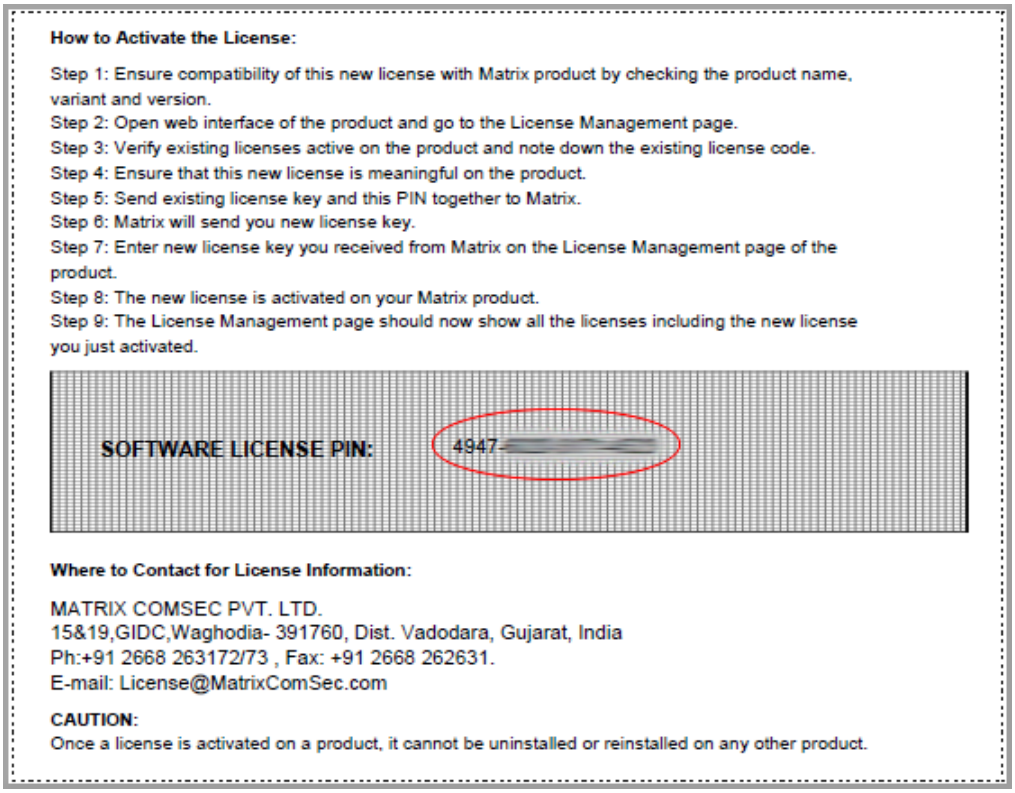
- The page displays the current License Profile[316] on ETERNITY GENX. Click the **Next** button to continue.

316. *When ETERNITY GENX is used as the Unified Communication Server, all the licenses except UMG are applicable. UMG License is applicable when you run the ETERNITY GENX as the Universal Media Gateway.*

The **License Activation** page opens.



In the **License PIN** field on this page, enter the 16-digit License PIN from the Voucher.



How to Activate the License:

Step 1: Ensure compatibility of this new license with Matrix product by checking the product name, variant and version.
Step 2: Open web interface of the product and go to the License Management page.
Step 3: Verify existing licenses active on the product and note down the existing license code.
Step 4: Ensure that this new license is meaningful on the product.
Step 5: Send existing license key and this PIN together to Matrix.
Step 6: Matrix will send you new license key.
Step 7: Enter new license key you received from Matrix on the License Management page of the product.
Step 8: The new license is activated on your Matrix product.
Step 9: The License Management page should now show all the licenses including the new license you just activated.

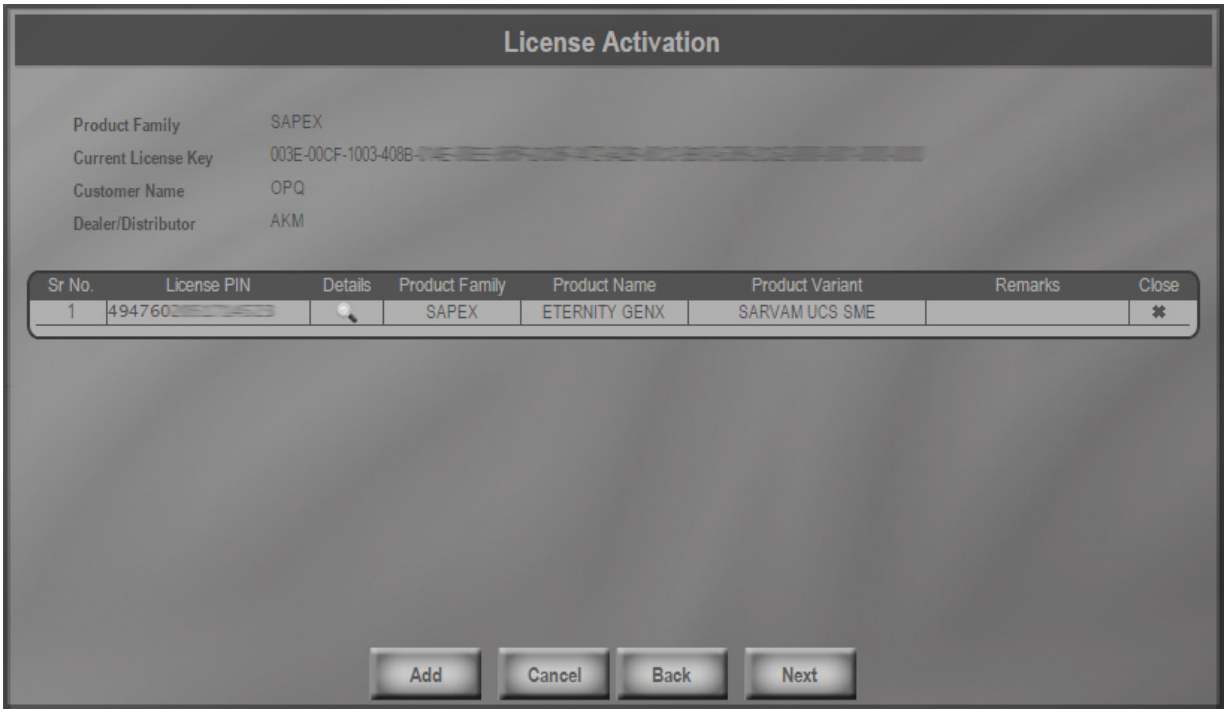
SOFTWARE LICENSE PIN: 4947-

Where to Contact for License Information:

MATRIX COMSEC PVT. LTD.
15&19,GIDC,Waghodia- 391760, Dist. Vadodara, Gujarat, India
Ph:+91 2668 263172/73 , Fax: +91 2668 262631.
E-mail: License@MatrixComSec.com

CAUTION:
Once a license is activated on a product, it cannot be uninstalled or reinstalled on any other product.

- Click **Details**. The details appear in the fields **Product Family, Product Name**, **Product Variant**.



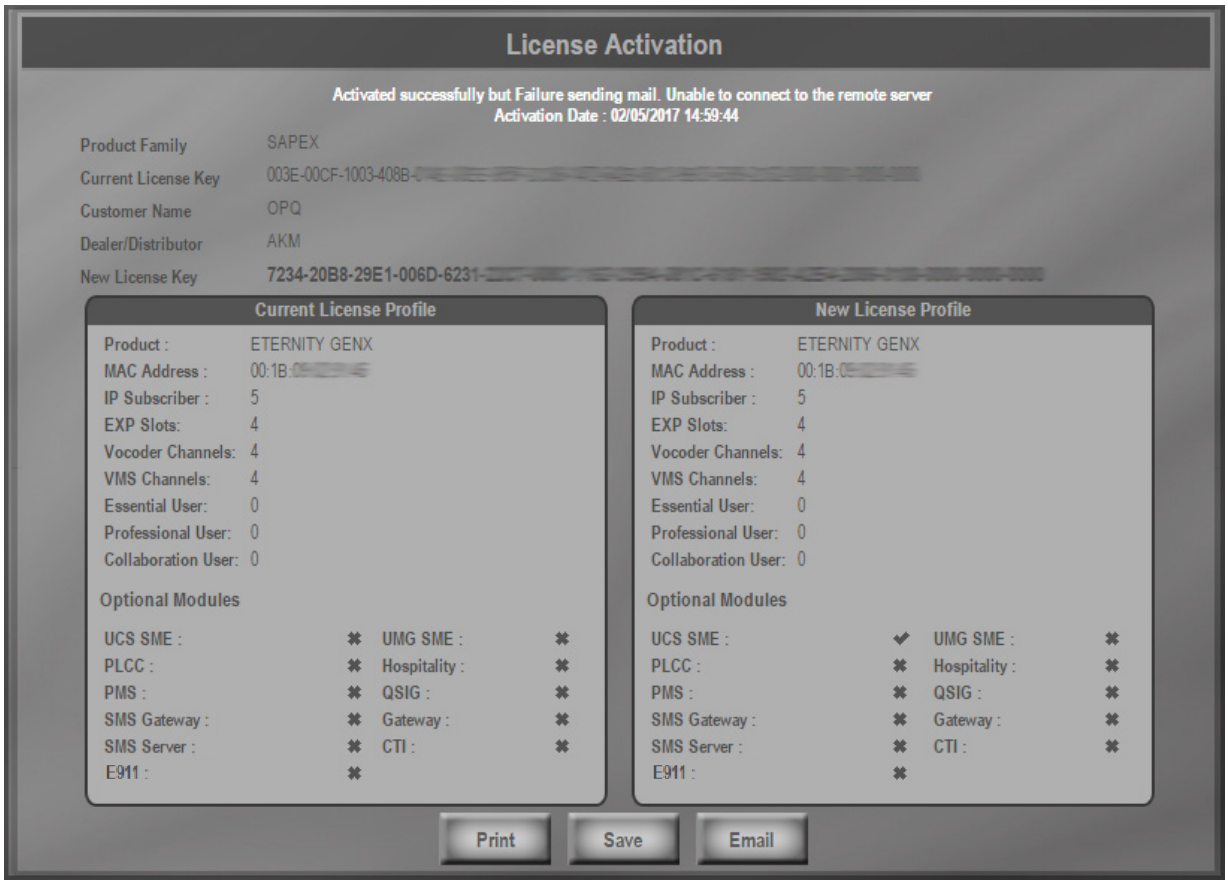
- Click the **Next** button. Your **Current License Profile** and your **New License Profile** will appear on this page.

- Click the **Activate** button and wait for a few seconds, as the activation is initiated.

On successful activation, the confirmation message will appear on your screen along with the activation date and time.

*The screens for **Current License Profile** and the **New License Profile** may differ according to the licenses purchased by you.*

A confirmation mail will also be sent to your e-mail ID (registered with Matrix).



You may **Save**, **Print**, or **Email** this information for your records, by clicking the relevant button.

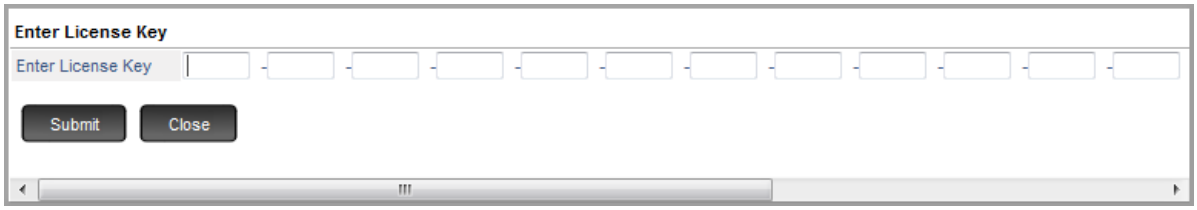
- Note down or copy the New License Key generated on this page.
- Go back to the Jeeves window (or log in as System Engineer again, if your session has ended).

  Under Configuration, click the **License Management** link again.

- Click the **Enter License Key** button.

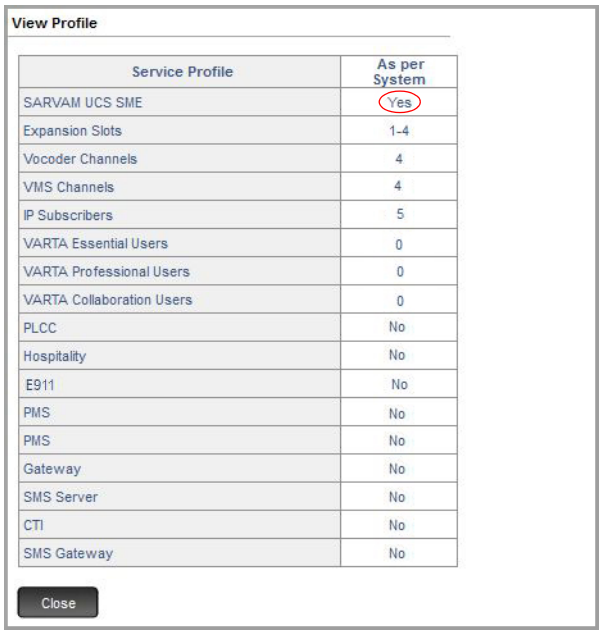
A new window opens.

- In **Enter License Key**, paste or enter the new License Key generated



- Click the **Submit** button.

  To view the status of the license activated by you, click the **View Profile** button again**.**



  A new window opens which displays the updated service profile.

- To log off, click the **Logout** button.

If you are unable to use Online Activation of the License Key or have no internet access, contact the Matrix License Support Centre for assistance in generating the new License key.

Instructions for Customers

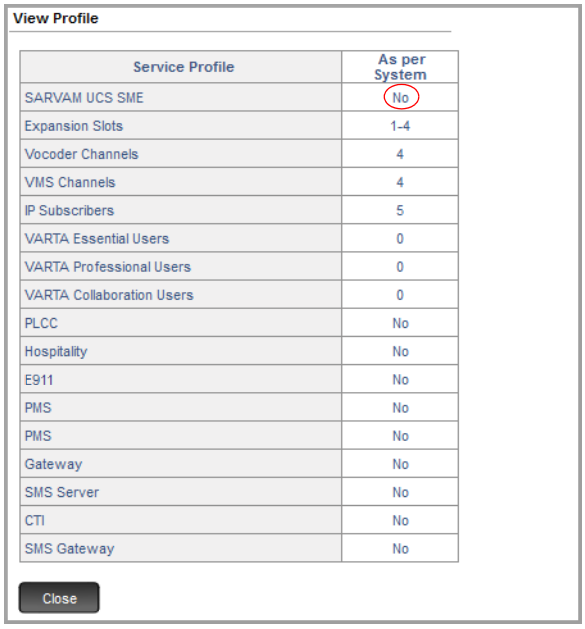
To activate your License, you would need the License Voucher containing the 16-digit License PIN. Contact your Dealer/Distributor in this regard. Your License Voucher may be a paper or a protected PDF file.

- Open Jeeves.
- Log in as System Engineer.

- Under **Configuration**, click the **License Management** link.

- Note down the current **License Key** on this page.

  If you wish to view the features and functions currently available on your system, click the **View Profile** button.



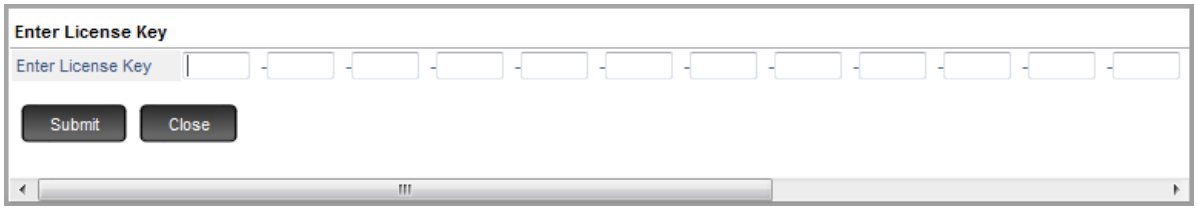
  A new window opens which displays the features and functionalities that are currently available to you.

- Now, send your Current License Key and the License PIN (on the Voucher) to the Matrix License Support Centre.
- You will receive a new License Key.
- Open Jeeves again.
- Log in as System Engineer.

- Under **Configuration**, click the **License Management** link.

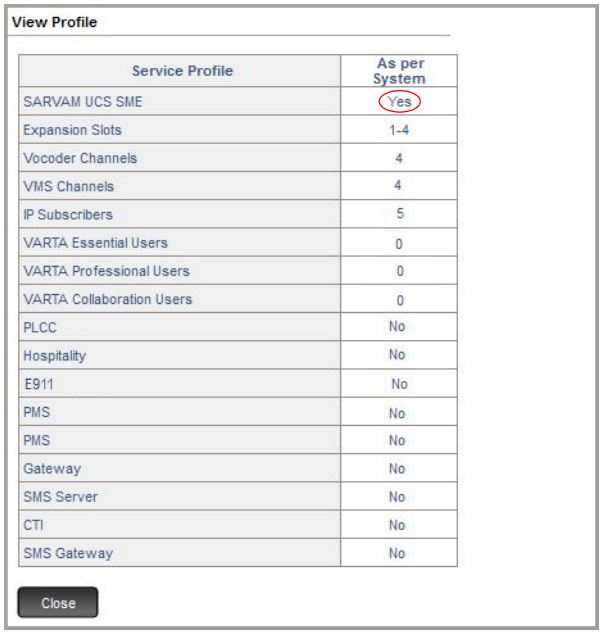
- Click **Enter License Key** button.

  A new window opens.



- In **Enter License Key**, enter the New License Key you obtained from Matrix.
- Click the **Submit** button.

  To view the status of the license activated by you, click the **View Profile** button again.



  A new window opens which displays the updated service profile.

- To log off, click **Logout**.

The current License Key and Service Profile will remain unchanged when the system is set to default or the firmware is upgraded.

Lightweight Directory Access Protocol (LDAP)

What's this?

LDAP (Lightweight Directory Access Protocol) is an application protocol used by the system for accessing and maintaining information services for the distributed directory over an IP network.

The advantage for LDAP is that you can access the central LDAP directory of the organization using your system. Therefore, you do not have to maintain the local directory. The LDAP server differentiates the data in its entries based on specific attributes. SARVAM UCS functions as the LDAP client and once it synchronizes with the LDAP server, it saves the fetched LDAP entries to the Global directory. You can search and dial out the desired contact from the Global directory.

This feature is beneficial for organizations where multiple servers are deployed at various locations and all these servers need to maintain a centralized directory to facilitate ease of inter-department communication.

How it works

Pre-requisites

To be able to use this feature,

- Make sure that the third-party LDAP server supports LDAP Protocol Version 3.

  The following LDAP servers have been tested successfully:

  - Microsoft Active Directory
  - Open LDAP Directory Server

- The required details of the contacts are stored in the LDAP server directory.
- Make sure the LDAP parameters have been properly configured. For details, refer to “Configuring LDAP using Jeeves”.
- Manually synchronize your system with the LDAP server.

Once you have fulfilled the above pre-requisites, the sequence of the events that occur are as follows:

- When you manually synchronize, the system sends request to the LDAP server.
- The LDAP server asks for Authentication ID and Password depending upon the authentication settings in the server. Consult your SE for more details.
- After successful authentication, the LDAP server sends response to the system with the required contacts and their details as per the search criteria configured.
- The fetched results are stored in the Global Directory. The total number of search results stored depends upon the number of Global Directories synchronized with the LDAP server.
- Now, any extension user can easily dial out the desired contact from the Global Directory. To know more about making an outgoing call from the Global Directory, refer to “Global Abbreviated Dialing”.

- *Once the required contacts are synchronized with the LDAP server and loaded in the Global directory, these cannot be edited or deleted unless the LDAP option is disabled.*

- *Extension users can edit a number before making an outgoing call, but the changes will not be applied to the details stored in the LDAP directory and in the Global Directory which is synchronized with LDAP.*

LDAP Attributes

LDAP database consists of various details — first name, last name, common name, multiple phone numbers etc. To distinguish it, each type of detail is assigned a specific attribute in LDAP database. Contact your LDAP server Administrator to know the detailed list of attributes maintained.

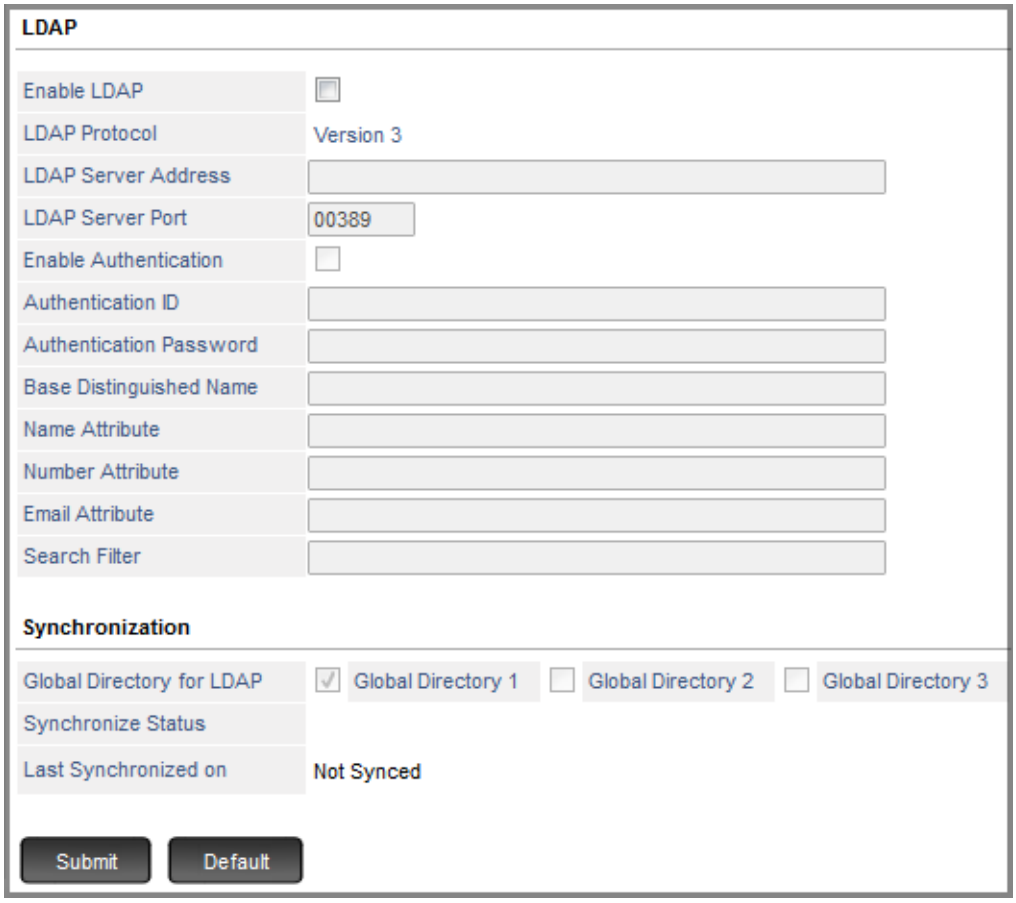
A few examples are listed below.

Abbreviation	Name	Description
gn	givenName	First name
cn	commonName	Full name
sn	surname	Last name or family name
dn	distinguishedName	Unique identifier for each entry
dc	dc	Domain component
-	company	Company or organization name
-	telephoneNumber	Office phone number.
mobile	mobilephoneNumber	Mobile or cellular phone number
mail	emailid	Email address

How to configure

Configuring LDAP using Jeeves

- Log in as System Engineer.
- Under **Configuration**, click **LDAP** to open the page.



LDAP

Make sure you Contact your LDAP Server Directory Administrator for the LDAP Server configuration details.

- **Enable LDAP:** Enable this check box if you wish to synchronize contacts from the LDAP server. When this check box is enabled, SARVAM UCS behaves as the LDAP client. By default, it is disabled.
- **LDAP Protocol:** Displays the LDAP Protocol Version supported. By default, it is Version 3.
- **LDAP Server Address:** Configure the IP address or Domain name of the third-party LDAP server here.
- **LDAP Server Port:** Configure listening port of LDAP Server here. By default, the value is 00389.
- **Enable Authentication:** This parameter enables you to set permission for validating the credentials (Authentication ID and password) of the system by the LDAP server. By default, it is disabled.

- **Authentication ID:** Enter the ID for authentication. The Authentication ID entered here should match with the Authentication ID configured in the LDAP server. You cannot keep this field blank. The ID may be a string of maximum 40 characters. Default: Blank.

- **Authentication Password:** Enter the Password for authentication. The Authentication Password entered here should match with the Authentication Password configured in the LDAP server. You cannot keep this field blank. The Password may be a string of maximum 24 characters. Default: Blank.

*Authentication ID and Password can only be configured when **Enable Authentication** check box is enabled. Contact your SE for more details.*

- **Base Distinguished Name:** Enter location in the LDAP Server from where the search should begin. You can also specify the subtree as the base entry. You cannot keep this field blank. Default: Blank.

- **Name Attribute:** You can define the Name attribute to be fetched from the LDAP server directory here.You cannot keep this field blank. Default: Blank.

- **Number Attribute:** You can define the Number attribute to be fetched from the LDAP server directory here. You can define only one attribute and hence the system will fetch only one number per contact from the search results. You cannot keep this field blank. Default: Blank.

- **Email Attribute:** You can define the Email Attribute to be fetched from the LDAP server directory here. This needs to be configured only if you need the Email-id of the required contacts.

- **Search Filter:** You can use the Search filter when specific entries from a particular Distinguished Name are required. Enter the details after contacting you LDAP Server Administrator.

Synchronization

The Synchronization with the LDAP server will function only if the parameters mentioned under LDAP have been configured correctly.

You can now proceed further with configuring the Synchronization parameters:

- **Global Directory for LDAP:** Select the Global directory in which the search results obtained from the LDAP server should be stored. You can select either one or combination of — Global Directory 1, Global Directory 2, Global Directory 3. By default, Global Directory 1 is selected.

- **Synchronize Status:** Displays the synchronization status (In Progress, Fail or Pass) of the system with LDAP server.

- **Last Synchronized on:** Displays the date & time when the system last synchronized with LDAP server.

- Click on **Sync Now** button to manually synchronize the system with the LDAP Server.

- **Test:** You can test the parameters configured in the system for LDAP synchronization. To do so,
  - click the **Test** button. A new window for Test is displayed.
  - Select the desired attribute and enter the — Name, Number or Email-id.
  - Click the **Test** button.
  - If the connection has been successfully established then search results fetched from the LDAP server will be displayed.

- ***Test** button will be displayed only when the mandatory fields in LDAP have been configured.*
- *When configuration backup is restored then the synchronized LDAP contacts in the Global directory will not be restored. Hence, you must manually synchronize the Global directory contacts with LDAP server again (if LDAP contacts are required).*

- Click **Submit** to save your settings.

How to use

To make an outgoing call to an LDAP contact from the synchronized Global Directory, refer to “Global Abbreviated Dialing”

Live Call Screening

What's this?

Live Call Screening enables extension users to screen callers before attending their calls on their extensions.

How it works

To be able to use this feature,

- the VMS module must be present in the SARVAM UCS.
- the extension users who are to be provided this feature must have a Personal Mailbox assigned, and have this feature allowed to them in their Class of Service.
- the extension users who want to use this feature must set *Call Forward to Voice Mail System* on their extension.
- the extension on which this feature is used must be a DKP.

With the above pre-requisites fulfilled, this is how Live Call Screening will work:

- Extension A has activated Live Call Screening.
- B calls A, and is transferred to A's mailbox (as Call Forward to Voice Mail System is set).
- The VMS Auto Attendant offers B the option to leave a message in the mailbox of extension A.
- As B starts to record a message in A's mailbox, the speaker of the DKP of A gets turned ON for the duration of the *Live Call Screening Timer* (configurable; default: 10 seconds).
- A hears the message being recorded in the mailbox.
- If A wants to answer the call, A can go Off-Hook. A gets connected to the caller and the system stops recording the message in the mailbox.
- If A does not answer the call, the speaker is turned Off automatically on the expiry of the *Live Call Screening Timer*.
- Also, after listening to some part of the message, if A finds that the call is not important, A can ignore the caller by dialing any digit. When A dials any digit, the speaker of the DKP is turned OFF, but the message recording in the mailbox continues.

LCS works only if your extension is idle.

How it configure

To be able to use Live Call Screening, extension users must have:

- A mailbox assigned to them.
- Live Call Screening enabled in the “Class of Service (COS)” for the time zone in their “Station Basic Feature Template”, as required.

If required, you may also change the duration of the Live Call Screening Timer. See “System Timers and Counts” for instructions.

How to use

For EON Users only

To activate Live Call Screening,

- Press DSS key assigned to Live Call Screening.
  OR
- Dial 1094-1

To answer a caller who is recording a message in your mailbox,

- Go Off-Hook by pressing the speaker key or lifting the handset.
- Speak to the caller.

To ignore the message of the caller,

- dial any digit.

To deactivate Live Call Screening,

- Press DSS key assigned to Live Call Screening, again.
  OR
- Dial 1094-0

Live Call Supervision

What's this?

Using Live Call Supervision, any extension can know the last external number dialed by another extension, even when that extension is in speech with an external party.

This feature is useful for supervisors who want to know where their subordinates are calling.

This feature is supported on DKP and SIP extensions, and on SLT extension which have CLI phone.

How it works

- A is the supervisor of B.
- A wants to know where B is calling, A can use Live Call Supervision.
- When B dials an external number it is stored in the system's memory.
- When A requests Live Call Supervision for B's extension, the system retrieves the last external number dialed by B and presents it on the display of A's phone.
- If B has not dialed any external number, A will get error tone with the message 'No Calls to Supervise' displayed on the LCD.
- If the B has dialed internal as well as external numbers, the last external number dialed by B will be displayed on LCD of A's phone.

Live Call Supervision can be used also when the extension being supervised is in speech with an external party.

How to configure

To be able to use Live Call Supervision, extension users must have this feature enabled in their "Class of Service (COS)" in their "Station Basic Feature Template" for the required time zones.

How to use

For EON & Extended IP Phone Users

- Press DSS key assigned to Live Call Supervision
  OR
- Dial 1098
- Enter the Extension number to be supervised

For SLT Users

- Dial 1098-Extension number to be supervised.

Logical Partition

What's this?

Logical Partitioning is used to restrict the flow of call traffic between PSTN and Private Networks, between PSTN and VoIP networks as well as VoIP to VoIP networks.

This feature may be used in countries where such restrictions are mandated by telecom regulations. For example, in certain countries, calls from VoIP to Public Networks (PSTN, Public Land Mobile Network) are not allowed.

Thus, local telecom regulations may either disallow termination of lines from both networks on the same equipment or may allow lines from both networks to be terminated on the same equipment, provided the equipment is designed to restrict flow of call traffic from these networks. For example, the Telecom Regulatory Authority of India allows termination of lines from the PSTN and VoIP Networks in the same equipment, only if these lines are logically partitioned. Termination of lines from both these networks in the same equipment without a logical partition constitutes an offence.

SARVAM UCS supports Logical Partitioning for this purpose.

How it works

Logical Partition is applied on the Trunk ports. Trunk ports are assigned to any of the following categories according to their installation scenario:

- **Category 1:** Trunk ports interfaced with PSTN /PLMN (Public Land Mobile Network) are assigned this category.
- **Category 2:** Leased lines terminated in the trunk ports are assigned this category.
- **Category 3:** Trunk ports used to interconnect two Systems are assigned this category. For example, QSIG used on T1E1 port or the CO of SARVAM UCS is interfaced with FXS of other System for expanding configuration.
- **Category 4:** SIP Trunks interfaced with ISP/ITSP are assigned this category.
- **SIP Extensions:** Internal calls using SIP are assigned this category.

- *These are default Categories. You have the flexibility to define each category according to suit your preference. For example, you may, if you so prefer, define Leased lines to Category 3 and Trunk ports connecting two Systems to Category 1.*
- *As calls from Category 4 (SIP trunks) to Public network trunks are always to be restricted.*

For each of the above categories, the System Engineer can program the calling permission, that is, whether to 'allow' or 'restrict' the calls across and within these categories. Similarly, the calling permission can also be programmed for the SIP Extensions.

You can change the settings as per the regulations in your country.

Depending on the calling permission programmed between the trunk categories, the system will allow or deny the calls on the trunks.

If within the office premises, you want to remove the logical partition for local SIP Extensions allowing the Internal Extension (SLT, DKP, Local SIP Extension and Remote Extension) calls and External (Trunk) calls, then you can configure the table defining the IP address range of those local SIP Extensions. These local SIP Extensions will be considered similar to the System extensions such as a DKP or SLT.

The SIP Extensions other than those configured in the table will be considered as External/Remote SIP Extensions. Though Remote SIP Extensions can make internal calls to the DKP/SLT/Local SIP Extensions, but Logical Partition will be applied for the external SIP extension calls and trunk calls.

Matrix recommends you to enable the logical partition in order to avoid any toll bypass wherever necessary. Matrix will not be responsible in case of any discrepancy related to this.

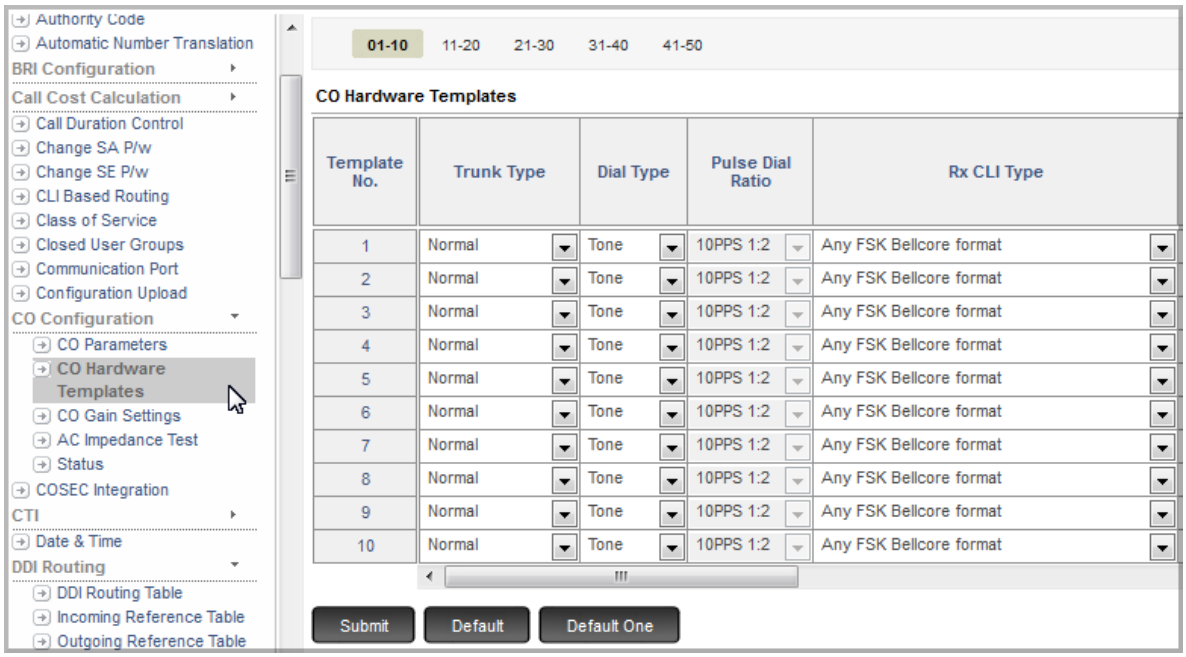
How to configure

To able to use this feature you must first assign the trunk port - CO, ISDN BRI, T1E1PRI, Mobile, VoIP to the appropriate Category and then define the calling permission - allow or restrict calls - in Category 1 to 4 and SIP Extension (as described above).

Configuring Logical Partition using Jeeves

- Log into Jeeves as System Engineer.
- Now, assign the categories to the different Trunk Type, by programming the parameter 'Category (Logical Partition)' in the Trunk Port Parameters of the respective trunk types: CO, BRI, T1E1, E&M and Mobile Port.
    - For CO trunks, assign Category in the “CO Hardware Template”.
    - For E&M ports, assign Category in the “E&M Feature Template”.
    - For Mobile ports, assign Category in the “Mobile Port Parameters”.
    - For T1E1PRI ports, assign Category in the “Configuring E1 Trunks” and “Configuring T1 Trunks”.
    - For BRI ports, assign Category in the “Configuring BRI Trunks”.

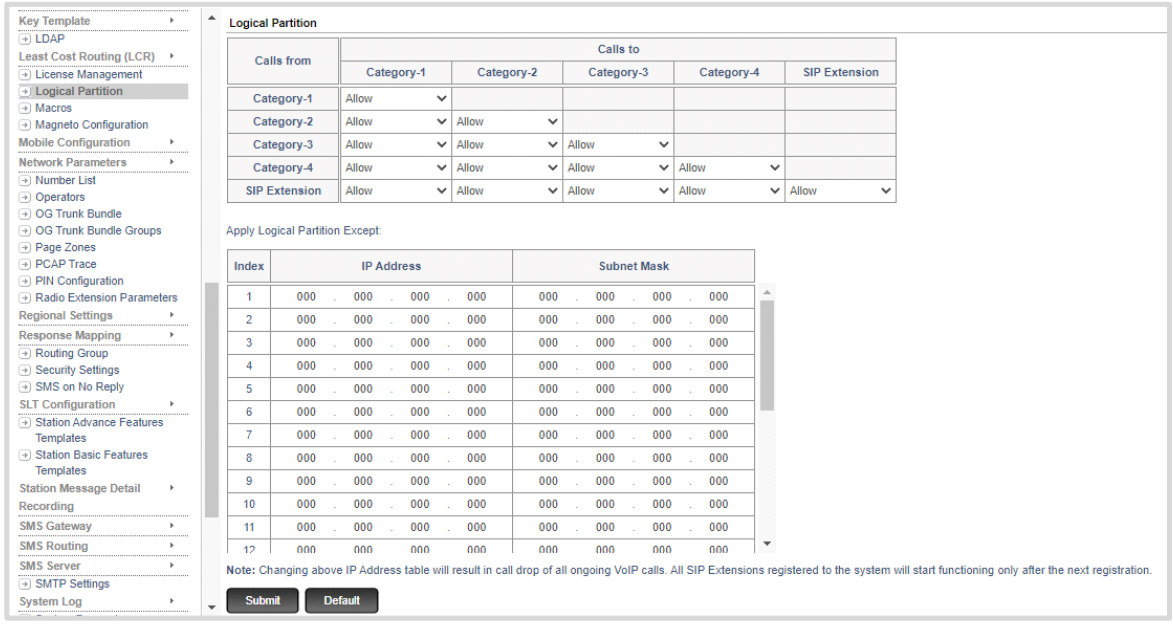
- Open the desired Trunk Port Parameters page by clicking the link. For example, to assign a Category to CO ports, open the CO Hardware Template page.



- Scroll with the horizontal bar to reach the column 'Category (Logical Partition).
- Select a Category number 1, 2, 3 or 4 to assign Category to CO ports.
- Click 'Submit' at the bottom of the page to save your settings.
- Repeat the same steps to assign Categories to other Trunk ports.

If you have completed assignment of Category to the Trunk Type,

- Under **Configuration**, click **Logical Partitioning** to open the page.



- Now, define the call permission for each category, that is, 'allow' or 'restrict' calls. For example, allow/ restrict calls from Category 1 to Category 1, from Category 1 to Category 2, from Category 1 to Category 3 and so forth.

- In the table, configure the IP addresses and subnet masks of those SIP Extensions/SIP Trunks for which you want to allow the Internal Extension and External (Trunk) calls without checking the logical partition You can change the settings as per the regulations in your country.. These local SIP Extensions will be considered similar to the System extensions such as a DKP or SLT. You can configure upto 25 entries in this table.

- Click **Submit** to save changes.

- Log out of Jeeves or continue with the configuration.

Configuring Logical Partition using a Telephone

- Enter SE mode from a DKP/SLT.

- First, program the parameter 'Category (Logical Partition)' in the Trunk Port Parameters of the respective trunk types. For SE Commands, refer the topics:

  - "CO Hardware Template"
  - "E&M Feature Template"
  - "Mobile Port Parameters"
  - "Configuring E1 Trunks" and "Configuring T1 Trunks"
  - BRI Parameters in "Configuring BRI Trunks"

  To define call permission across and between categories:

  - Dial ***5317-Category-Category-Flag***
    Where,
    Category is
    1 for Category 1 trunks
    2 for Category 2 trunks
    3 for Category 3 trunks
    4 for Category 4 trunks

    Flag is
    0 for restrict calls
    1 for allow calls
    Default: 0 (that is, restrict calls) for Category 1 to 3 and 1 (allow calls) for Category 4.

- Exit SE mode.

- *If you select India as the Region, by default, calls are restricted for all categories, except within SIP Extensions.*

  - *When call permission is restricted between two categories of trunks, following feature interactions will apply:*

    - ***Call Transfer:*** *Trunk to Trunk Transfer between restricted categories of trunk will not be allowed. If the user attempts trunk-to-trunk transfer between restricted trunks, Error Tone will be played.*

- ***Raid:*** *If a user using DISA attempts to Raid a conversation of an extension with a trunk to which call permission is restricted, the Raid attempt will fail and the user will get an Error Tone.*

- ***Conference:*** *An extension user will not be able to include restricted trunks in a 3 party or multi-party conference. An Error Tone will be played when s/he attempts it.*

- ***Dial-In Conference:*** *Participation in a Dial-In Conference from trunks with restricted call permission is not allowed.*

- ***External Call Forward:*** *In the case of Auto Attendant, DISA or when transferring a trunk call to an extension, if the extension has set call forward to an external number, the system will allow the call only if the call permission between the source and destination trunk is allowed. Otherwise, an Error Tone will be played to the user.*

- ***Hotline:*** *When a user has logged into DISA and the extension being used for the DISA login has the Hotline - Trunk or Hot outward dialing (HOD) feature enabled, the system will allow the call between the source trunk (from where the DISA login is made) and the destination trunk (which is used as Hotline Trunk) only if calling is permitted between them. Otherwise an Error Tone will be played to the DISA caller on the expiry of the Hotline Timer.*

Loop Back Tests

What’s this?

SARVAM UCS supports following types of Loop Back Tests for T1 and E1 lines:

- Near End Loop Back: These are of two type viz.
  - Line Loop Back test
  - Payload Loop Back test

- Far End Loop Back Test: These are of two type viz.
  - Line Loop Back test
  - Payload Loop Back test

  These tests are conducted when the T1E1 port is in NT mode and is connected to other System and wants to test the line between itself and the far end. In this mode, SARVAM UCS acts as a network.

Near End Loop Back

Line Loop Back

This can be implemented by shorting of signaling wires 'RTIP' and 'RRING' to 'TTIP' and 'TRING' respectively at the line side.

This is as good as shorting the **Rx pair to Tx pair**. In this case you do not need to sync the T1E1 port with the network.

Payload Loop Back

This is effectively a physical connection of **payload** channels (Timeslot) with **framing** generated by T1E1 port itself for the transmit signal. In this case you must sync the T1E1 port with the network.

When the other end connected with the T1E1 port of the SARVAM UCS wants to perform the Loop Back tests, the T1E1 port will form the Loop Back depending upon the type of test the other end wants to perform (that is, Line Loop Back or Payload Loop Back).

Loop Back Process

Method of forming loop back

Method of forming loop back at the T1E1 port side is different for the E1 and T1 carrier. These are explained in the following.

When line type is configured as ISDN_E1_PRI or ISDN_E1_CAS on the T1E1 port,

- The protocol doesn't support the facility that the remote end can close/open the loop at the T1E1 port side automatically.

- Hence when the remote end wants to perform the loop back test, the SE must be informed, to form the desired type of loop back (that is, Line Loop Back or Payload Loop Back) on the T1E1 port.

- When the SE forms the loop back at T1E1 port side (by issuing appropriate SE Command), the remote end can start the test.
- On completion of the testing, the remote end must inform the SE to release/open the loop back formed on the T1E1 port side.
- On receipt of the request from the remote end, the SE will issue SE command to open the loops on the T1E1 port.

When line type is configured as ISDN_T1_PRI or ISDN_T1_RBS on the T1E1 port,

- The protocol supports loop back Activation and deactivation message, whereby the remote end can send the loop activation code to the T1E1 port and the T1E1 port decodes the message and forms the loop back automatically.
- On completion of the testing, the remote end can send the loop deactivation code and the T1E1 port can open the already formed loop back.
- So in this case the SE's intervention is not required to form and release the loop back.
- Incase the remote end doesn't support the facility to automatically form/release the loop back for the T1E1 port though the carrier is T1, the SE can use the commands (6141-1-T1E1-Type of Loop Back) on request of the remote end to form and release the loop backs.

Loop back Activation

When the system receives the SE command to form Line/Payload loop back (in case of E1):

- It will inform the SARVAM UCS Card T1E1 about the received command SARVAM UCS Card T1E1 will release all the calls supported by the T1E1 Port under test.
- The SARVAM UCS Card T1E1 will form the required type of loop back.
- System will put the T1E1 port in maintenance mode. It will release all active calls supported by T1E1 port and restrict the usage of T1E1 port for IC/OG calls.

When the loop back activation code is received from far end on the T1E1 port, (incase of T1 carrier):

- SARVAM UCS Card T1E1 will inform the system about the received information and the T1E1 port number.
- SARVAM UCS Card T1E1 will release all the calls supported by the T1E1 Port under test.
- System will put the T1E1 port in maintenance mode. It will release all active calls supported by T1E1 port and restrict the usage of T1E1 port for IC/OG calls.

Release Loop Back

On receiving the loop back release code or receiving the command to release the loop back (either Payload or Line), the system will take the T1E1 port out from the maintenance mode and now the T1E1 port will function normally.

Far End Loop Back Test

Far End Loop Back test gives facility to SARVAM UCS to check the health of the line towards the other end connected on its T1E1 port using loop back tests.

The types of the far end loop back test are:

- Line Loop Back
- Payload Loop Back

Line Loop Back

This can be implemented by shorting of signaling wires 'RTIP' and 'RRING' to 'TTIP' and 'TRING' respectively at the line side.

This is as good as shorting the **Rx pair to Tx pair**. In this case you do not need to sync the T1E1 port with the network.

Payload Loop Back

This is effectively a physical connection of **payload** channels (Timeslot) with **framing** generated by T1E1 port itself for the transmit signal. In this case you must sync the T1E1 port with the network.

When the other end connected with the T1E1 port of the SARVAM UCS wants to perform the loop back tests, the T1E1 port will form the loop back depending upon the type of test the other end wants to perform (that is, line loop back or payload loop back).

The option 'Release All Loop Backs' is required when the far end requires the universal loop back release code to release the Loop Back.

Loop Back Process

Method of forming loop back

Method of forming loop back at the T1E1 port side is different for the E1 and T1 carrier. These are explained below.

When line type is configured as ISDN_E1_PRI or ISDN_E1_CAS on the T1E1 port,

- The protocol doesn't support the facility that the SARVAM UCS can perform the loop back tests automatically.
- The SE of the SARVAM UCS will inform the far end connected with the T1E1 port, to form the loop back as required.
- Once the far end has formed the desired loop back the SE can issue command to start the Far end loop back test.

Start Loop Back Test:

- On receiving the command to start Far End loop back test (either the Line Loop Back or Payload Loop Back test) the system will put this T1E1 port in maintenance mode and informs the T1E1 Card about the test to being performed on the T1E1 port.
- All the calls supported by the T1E1 Port under test will be released.

- The T1E1 Card will start the PRBS generator and counter.
- The T1E1 Card will send the PRBS count every 1 second to the system.
- The T1E1 Card will increment PRBS Counter for every error encountered during the test every one second.
- The T1E1 Card will reset the PRBS counter to zero, after sending the PRBS Counter to the system, every second.
- The system will store the received PRBS count (received every second) in Performance Report, which can be captured from Serial port/Ethernet port.

End Loop Back Test/Release All Loop Backs

- When the SE wants to end the loop back test he will issue the command to end the Far End Loop Back Test, for the T1E1 port under the test.
- SE will inform the other end's person that loop back test is finished and now the remote end can open the loop formed.
- On receiving the command to end loop back test for the T1E1 port, the system will take the T1E1 port out of maintenance state and inform the T1E1 port about the received command.
- Normal functioning of the T1E1 port will resume.

When the line type is configured as "ISDN_T1_PRI" or "ISDN_T1_RBS" on the T1E1 port,

- Loop back activation/deactivation is automatic.
- The activation methods are different for the D4 and ESF framing.
  - D4 framing supports only Line loop back
  - ESF framing supports both Line loop back and Payload loop back.
- **When D4 is selected as Framing,**

  Start Line Loop Back Test

  - The test will start for the T1E1 port when SE issues the command to start the far end line loop back test. When SE command (6142-1-T1E1-1) is issued the system will inform the T1E1 Card about the command and this T1E1 port will be in maintenance mode.
  - All the calls supported by the T1E1 Port under test will be released.
  - The T1E1 Card will send the line loop back "Activation Code" and will start the PRBS generator and counter.
  - The T1E1 Card will send the PRBS count every 1 second to CPU Card.
  - The T1E1 Card will increment PRBS Counter for every error encountered during the test every one second.

- The T1E1 Card will reset the PRBS counter to zero, after sending the PRBS Counter to the system, every second.

- The system will store the received PRBS count (received every second) in Performance report, which can be captured from Serial port/Ethernet port.

Stop Loop Back Test

- When SE command (6142-T1E1-2) or (6142-T1E1-5) to end the Far end loop back test is issued the system will inform the T1E1 Card about the received command and will remove the T1E1 port from the maintenance mode.

- The T1E1 Card will send the inband "Deactivation Code".

- **When ESF is selected as Framing,**

Start Loop Back Test

- Depending upon the SE command issued to start the type of Loop Back (Line or Payload) test the system will put the T1E1 port under test in Maintenance mode and will inform the T1E1 Port about the received command.

- All the calls supported by the T1E1 Port under test will be released.

- The T1E1 Card will send the Loop back "Activation Message" for the line/payload loop back as informed by the system.

- The T1E1 Card will start the PRBS generator and counter.

- The T1E1 Card will send the PRBS count every 1 second to CPU Card.

- The T1E1 Card will increment PRBS Counter for every error encountered during the test every one second.

- The T1E1 Card will reset the PRBS counter to zero, after sending the PRBS Counter to Master, every second.

- The system will store the received PRBS count (received every second) in Performance report, which can be captured from Serial port/Ethernet port.

Stop Loop Back Test/Release All Loop Backs

- When SE issues command (6142-T1E1-2) or (6142-T1E1-4) or (6142-T1E1-5) to end the Far end loop back test the system will inform the T1E1 Card about the received command and will remove the T1E1 port from the maintenance mode.

- The T1E1 Card will send the "Deactivation Message" for the line/payload loop back test as required.

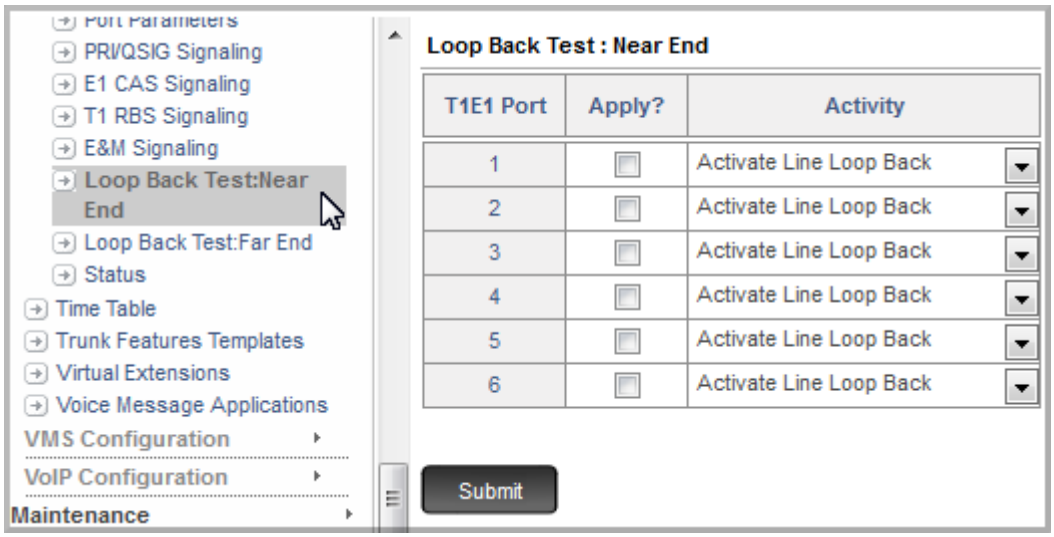
Performance Report

A Performance Report of the tests can be generated. It contains the log of all the errors. The Performance Reports stores upto 50 entries on FIFO basis.

Configuring Loop Back Test using Jeeves

To configure the Near End Loop Back Test parameters,

- Log into Jeeves as System Engineer.
- Under **Configuration**, click **T1E1 Configuration**.
- Click **Loop Back Test: Near End**.

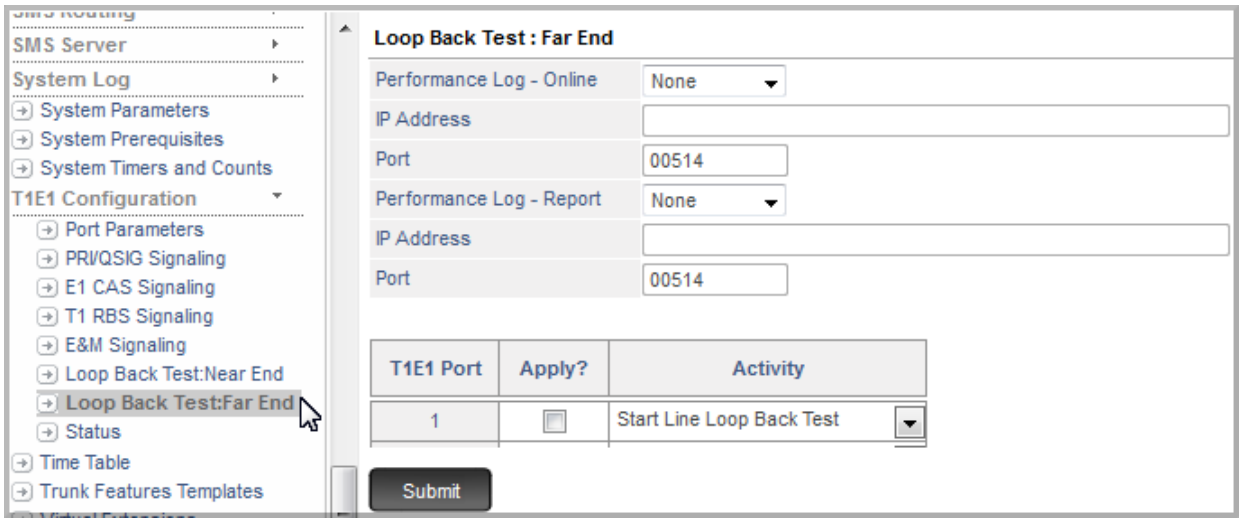


- For each T1E1 port,
  - Select the **Apply?** check box to enable loop back.
  - From the **Activity** list, you can select to Activate or Release the loop back test.
  - Click **Submit** to save the settings.

To configure the Far End Loop Back Test parameters,

- Log into Jeeves as System Engineer.
- Under **Configuration**, click **T1E1 Configuration**.

- Click **Loop Back Test: Far End**.



  - Select the Communication Port/Ethernet Port/ USB to COM to be assigned as destination port for **Online** and **Report** logs.

  - If you select Ethernet Port as the destination, enter the **IP Address** of the Ethernet Port and the **Port** (Listening Port for the packets) for the **Online** and **Report** logs respectively. Both IPv4 and IPv6 addresses are supported. Valid port range: 1025 to 65535;514.

- For each T1E1 port,

  - Select the **Apply?** check box to enable loop back.

  - From the **Activity** list, you can select to Start, End or Release the loop back test.

  - Click **Submit** to save the settings.

Configuring Loop Back Test using Telephone

The commands explained below should be referred as:
• To program a single port: *XXXX-1*
• To program a range of ports: *XXXX-2*
• To program all the ports: *XXXX-**

Near End Loop Back

Use the following command to activate/release Near End Loop Back for T1E1:
6141-1-T1E1-Loop back
Where,
T1E1 is from 01 to 08.

Loop back	Meaning
1	Activate Line Loop back

Loop back	Meaning
2	Activate Payload Loop back
3	Release Loop back

Far End Loop Back

Use the following command to start/stop far end loop back test for T1E1:
6142-1-T1E1-Code
Where,
T1E1 is from 01 to 08.

Code	Meaning
1	Start Line Loop Back Test
2	End Line Loop Back Test
3	Start Payload Loop Back Test
4	End Payload Loop Back Test
5	Release All Loop Backs

Performance Reports

Online Performance Report Printing

Use following command to assign the port for online Performance Report Printing:
6143-Port
Where,

Port	Meaning
0	None
1	COM Port
2	Ethernet Port
3	USB to COM Port

Default, None.

Offline Performance Report Printing
Use following command to assign the port for offline Performance Report Printing:
6144-Port
Where,

Port	Meaning
0	None
1	COM Port
2	Ethernet Port

Port	Meaning
3	USB to COM Port

Default: None.

Performance Reports From SA Mode

Online Performance Reports

Use the following command to start/abort online printing of T1E1 performance report:
1072-030-Flag

Where,

Flag	Meaning
0	Disable
1	Enable

By default, Flag is 0.

Offline Performance Reports

User the following command to start/abort offline printing of T1E1 performance report:
1072-031-Flag

Where,

Flag	Meaning
0	Disable
1	Enable

By default, Flag is 0.

- *PRBS counter = 0 indicates the 'healthy' condition.*
- *During loop back test the PRBS counter may be greater than zero at initial stage of the loop back stage, but it will be zero afterwards consistently for the healthy condition.*
- *PRBS counter = greater than zero, indicates the 'faulty' condition for the loop back test.*

Macros

What's this?

Extension users often have to dial access codes for specific functions like dialing a feature code, making an internal call, making an external call, etc.

SARVAM UCS supports Macros, using which, you can abbreviate long number strings for regularly used functions in to macros and assign them to a DSS key on a DKP/Extended IP Phone extension.

You can also assign Macros on SLTs that have special keys.

How it works

- Extension 2001, frequently sets *Call Forward-All Calls* to an external number 26550333.
- To do this, each time, Extension 2001 must dial ***131-Trunk Access Code-26550333-#****.
- Instead of having to dial this lengthy number string, a Macro can be created for *Call Forward-All Calls* to External number.
- If Extension 2001 is an Extended SIP Phone or a DKP, the Macro can be assigned to a DSS key on the phone.
- Instead of dialing this number string, the user of Extension 2001 can simply press the DSS key on which this Macro is assigned.
- Thus when the DSS key on which a Macro is assigned is pressed, the corresponding access code is executed.
- The system sets call forward to the external number automatically.

SARVAM UCS also supports Macros for SLT which have special keys. When each of these keys is pressed, a special number string, which you can program is dialed.

For example, an SLT instrument has 5 special purpose keys. When these keys are pressed, the strings *50, *51, *52, *53, *54 programmed on these keys are dialed out.

You can create Macros for the strings dialed out using the special keys, whereby the string dialed by each of these special keys is associated with a particular function. For example, the special key for dialing *50 is associated with *Call Forward -All Calls* to an external number. So, when the extension user presses *50, the system receives this string and takes appropriate action, that is, interprets it as call forward to the external number, and sets call forward.

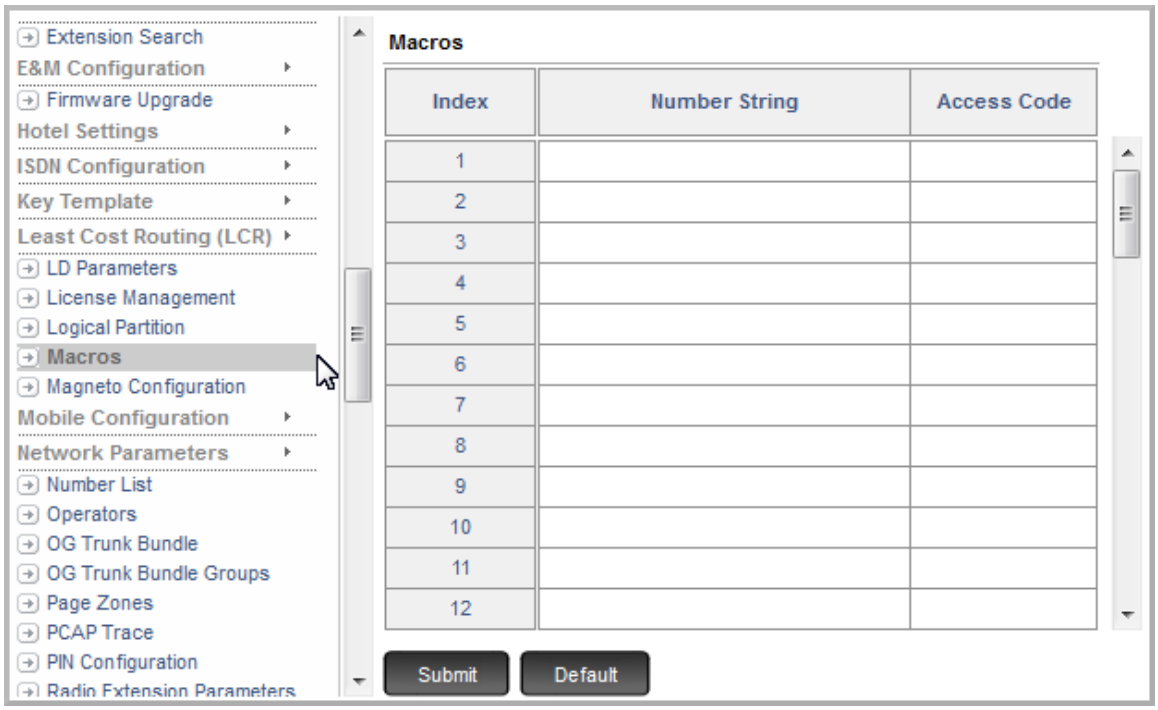
Thus, each time the extension user presses the special key *50, the system considers that the extension user has dialed *131-Trunk Access Code-26550333-#**.

How to configure

You can create as many as 50 Macros using Jeeves and by dialing system commands from a telephone connected to the SARVAM UCS.

Creating Macros using Jeeves

- Log into Jeeves as System Engineer.
- Under **Configuration**, click **Macros**.



The Macros page opens. Each macro is stored against an index number. By default the Macros Number String and Access Code are blank when the system is operated in the **Enterprise Mode**.

- When creating a Macro for a **DKP/Extended IP Phone**,
  - In the **Number String** field, enter the strings the system should consider as command when the DSS Key on the DKP/Extended IP Phone.
  - Click **Submit** to save.

*When SARVAM UCS is operated in the **Hotel** Mode (see **Customer Profile** under "System Parameters"), Number Strings and Access Codes are assigned for features such as Front Desk, Room Service, Voice Guided Alarm, Reservation Desk, Voice Mail System, Retrieve Message. These Macros will appear on this page. To know more, see the topic Customer Profile in the SARVAM UCS Hospitality System Manual.*

  - Now, assign this Macro to a DSS Key on the DKP/Extended IP Phone.
  - When you assign a Macro to a DSS Key on a DKP or Extended IP Phone,
    - you must select **Macro** as **Function Type**

- As **Offset**, you must select the Macro Index number (1-50) at which the number string is stored. In this example, it would be Macro Index 1.

For detailed instructions on assigning a macro to a DSS key of a DKP, see “DSS Keys Programming”.

For instructions on assigning a macro to a DSS key of the Extended IP Phone, see “DSS Key Settings” in “Configuring Matrix SPARSH VP330”, “DSS Key Settings” in “Configuring Matrix SPARSH VP248”, “DSS Key Settings” in “Configuring Matrix SPARSH VP310” and “DSS Key Settings” in “Configuring Matrix SPARSH VP510” for instructions.

- When creating a Macro for a special Key on an **SLT**,
  - In the **Number String** field, enter the strings the system should consider as command when the special key on the SLT is pressed.
  - In the **Access Codes** field, enter the strings sent by the SLT on pressing the special function key.

    For example, if the SLT sends the string ‘*53’ to the SARVAM UCS when the function key for Alarms is pressed, enter the string **163** (the feature access code for Voice-guided Alarms) in the **Number String** field, and enter the string ***53** in the corresponding **Access code** field.

    The Access Code that you assign here in Macros must not conflict with any other Access Codes in the Dial Phase. See “Access Codes”.
- Click **Submit** to save.
- Log out of Jeeves.

Creating Macros using a Telephone

- Enter SE mode from an SLT/DKP.

  To create a Macro, dial:
  - ***1810-Macro Index-Number String***

    Where,
    Macro Index is from 01 to 25.
    Number String is a string of 24-digits.
    If the length of the Number String is less than 24-digits terminate it with #*.

  To clear a Macro, dial:
  - ***1810-Macro Index-#****

*You can program a special digit for 'Pause' in the Key Board Macro Number string using the code **#3**.*

To assign an Access Code for the Macro, dial:
- ***3115-1-Macro Index-Access Code***

  Where,
  Macro Index is from 01 to 25.
  Access Code is a string of 4-digits.
  Access Code is maximum of 6 digits. If it is less than 6 digits, terminate it with #*.

To clear Access Code assigned to a Macro, dial:

- ***3115-1-Macro Index***
  Where,
  Macro Index is from 01 to 25.

To default the access code for the Macro, dial:

- ***3165-1-Macro Index***
  Where,
  Macro Index is from 01 to 25.

*When SARVAM UCS is operated in **Enterprise** mode, the default Macros Number String and Access Codes are blank. So, when you dial this command, the values in the Macros will become blank.*

*When SARVAM UCS is operated in the **Hotel** Mode, the Number Strings and Access Codes assigned by default will be restored when you dial this command. See the topic Customer Profile in the SARVAM UCS Hospitality System Manual to know more.*

- Exit SE mode.

Meet Me Paging

What's this?

While a Paging announcement is being made, any extension user of SARVAM UCS can get connected to the Paging extension, by dialing the Meet Me Paging feature code and the number of the Paging extension.

This feature is useful to Operators. Using this feature they can locate extension users who are away from their desks and get connected to them at their current location.

How it works

- A calls B's extension, but B is away from the desk.
- A uses Paging and makes an announcement asking B to call A's extension.
- To get connected to A's extension, B may use Meet Me Paging from any extension, while the announcement is being made.
- B dials Meet Me Paging code and A's number during the announcement.
- B gets connected to A.

- *Paging is an announcement made to a group of extensions within a Page zone. Extension users, (including those who are outside the Page zone) who want to use Meet Me Paging to answer the Paging call, will need to know the extension number they must call. Therefore, extension users who are paging are advised to announce their extension number.*

- *Meet Me Paging can be used only if the Paging call is active. Therefore, extension users who are paging must keep their call active, if they want their call to be answered using Meet Me Paging.*

How to configure

No configuration is required for Meet Me Paging. However, extension users who are using Paging to make their announcement, must have the feature "Paging" allowed in their Class of Service.

How to use

You can use Meet Me Paging to answer a Paging call from any extension, if you know the number of the Paging extension, and if the call is still active.

For EON & Extended IP Phone Users

To answer a Paging call from any extension other than the Paged extensions,

- While the announcement is being made,
- Press the DSS key assigned to Meet Me Paging.
  OR
- Dial 1093
- Dial the number of the paging extension.
- You get connected to the Paging extension user.

To answer a Paging call from the same extension that is paged,

- While the announcement is being made,
- Go ON-Hook and then go OFF-Hook.
- Press the DSS key assigned to Meet Me Paging.
  OR
- Dial 1093
- Dial the number of the paging extension.
- You get connected to the Paging extension user.

For SLT Users

To answer a Paging call from any extension other than the Paged extensions,

- While the announcement is being made,
- Lift the handset.
- Dial 1093-Paging Extension Number.
- You get connected to the Paging extension user.

To answer a Paging call from the same extension that is paged,

- While the announcement is being made,
- Go ON-Hook and then go OFF-Hook.
- Dial 1093-Paging Extension Number.
- You get connected to the Paging extension user.

Message Wait

What's this?

The Message Wait feature of SARVAM UCS enables extension users/Operator to set Message Wait on other extensions to deliver important messages.

If the extension user has a mailbox assigned, the Message Wait feature indicates to the extension user, the arrival of new messages in the user's mailbox.

Thus, Message Wait can be set by extension users as well as by the Voice Mail System.

You can set multiple Message Wait, but on different extension users. However, only one Message Wait can be set on one extension. A Maximum of 4 Message Wait can be set on an extension.

How it works

Message Wait set by Extensions/Operator

The Operator/any extension user can set Message Wait on another extension.

- The Operator calls Extension A.
- Extension A is not at the desk to attend the call.
- The Operator has an important message to communicate. So, the Operator sets Message Wait on Extension A, using the Message Wait key (if configured) or by dialing the feature access code.
- Extension B tries to reach Extension A, and sets Message Wait on Extension A, using the Message Wait key (if configured) or by dialing the feature access code.
- Message Wait will be indicated to Extension A according to the *Type of Message Wait Notification* set for Extension A. This may be in the form of a Stuttered Dial Tone, a Voice Message, Ring, or LED Lamp.
- If Extension A is a DKP or an Extended IP Phone and has DSS key assigned for Retrieve New Message, the LED of this key will glow to indicate new message wait.
- Now, Extension A can dial the feature access code to retrieve Message Wait, or press the Retrieve Message Wait Key, if assigned.
- The system will call the extension that first set Message Wait on Extension A. In this case, the Operator. If the Operator is busy, the system will place the call on Extension B. The system will try to call the extensions that set Message Wait until the call is answered.
- The extension that set Message Wait on A gets the CLI of A as Message Wait. A can now deliver the message.
- The LED of the Retrieve Message Wait key, if assigned, on Extension A will be turned off after all message wait set by other extensions on Extension A have been served.

Message Wait set by the Voice Mail System

The Voice Mail System (VMS) sets Message Wait on the extension, whenever a new message arrives in its personal Mailbox.

- Extension A is assigned a Personal Mailbox.
- There is a new message in A's Mailbox. The VMS indicates this to Extension A as per the Type of *Message Wait Indication* set for Extension A. This may be in the form of a Stuttered Dial Tone, a Voice Message, Ring or LED Lamp.
- If Extension A is a DKP or an Extended IP Phone, the Voice Mail key on the phone will also glow to indicate the arrival of a new message.
- If the Retrieve Message Wait key is assigned on the DKP/Extended IP Phone of Extension A, the LED of this key will also glow simultaneously to indicate arrival of the new voice mail.
- To listen to the new message, Extension A can
  - press the Voice Mail key
    or
  - the Retrieve New Message key (if assigned)
    or
  - dial the feature access code for Retrieve New Message.

  The VMS answers the call. After Extension A has listened to the new messages, the LED of the Voice Mail key is turned off.

  The LED of the Retrieve Message Wait key, if assigned, will also be turned off.

Types of Message Wait Indications

The system gives Message Wait indication to extensions according to the type of *Message Wait Indication* selected for the extension. The types of Message Wait Indications offered by SARVAM UCS are:

Stuttered Dial Tone/Voice Message

- When the extension user goes OFF-Hook, the user will hear a voice message, if a pre-recorded Voice Module has been assigned for Message Wait Notification. If no voice module is recorded and assigned, the extension user will hear a stuttered dial tone instead.
- If you want voice message to be played as message wait notification, record and assign a Voice Module. Refer "Voice Message Applications" for instructions.

SARVAM UCS can play only 9 Voice Modules simultaneously. The Voice Module for Message Wait Notification will not be played if there are already 9 being played simultaneously. In this case, Stuttered Dial Tone will be played for Message Wait Notification, when the extension user goes OFF-Hook.

LED Lamp

- When the extension user has an SLT with 'Message Wait' lamp, you can set this type of Message Wait Indication. When Message Wait is set, the lamp will blink continuously either using High Voltage or Polarity Reversal. The lamp will be turned off when the extension user has retrieved all the waiting messages.

Ring

- When a new Message Wait is set on the extension, the system will play *Message Wait Ring* (Short, Fast) on the extension. See “Distinctive Rings”.
- The extension will ring for the duration of the Message Wait Ring Timer (configurable; default: 30 seconds). If the call is not answered within this timer, the system will ring on the extension again for as many times as the Message Wait Ring Count (configurable; default: 10 times), and at the interval set as the Message Wait Ring Timer Interval (configurable; default: 30 minutes).
- When the extension user answers the call, the user gets connected to the VMS or the extension that set Message Wait.

Message Wait Signal generated by SARVAM UCS

Particulars	Value
Peak Voltage	82-85 V
ON Time	100ms
OFF Time	150ms
Frequency	4Hz
DC Offset	48V

How to configure

To provide this feature to extensions, you must do the following configuration on the extensions:

- Enable the Message Wait feature in the “Class of Service (COS)” of the “Station Basic Feature Template” of the extensions. This allows the extensions to set and cancel Message Wait on other extensions. Only those extensions that have this feature in their COS can set or cancel Message Wait on other extensions. By default, this feature is enabled in the COS of all extension types for all the time zones.
- Select the desired **Message Wait Indication** in the “Extension Voice Mail Settings”of the SLT, DKP, ISDN Terminal, SIP extensions and Department Groups.
- If you selected **Voice Message** as Message Wait Indication Type for an extension, you must also record the desired Voice Message in a Voice Module and assign it to the Message Wait application. See “Voice Message Applications” for instructions.
- If you selected **Ring** as Voice mail/Message Wait Indication Type for an extension, you may configure the following Ring Parameters:
  - Message Wait Ring Timer (default: 30 seconds)
  - Message Wait Ring Count (default: 10 attempts)
  - Message Wait Ring Timer Interval (default: 30 minutes).

  See “System Timers and Counts” for instructions.
- You may also assign DSS Keys to ‘Message Wait’ and ‘Retrieve Message Wait’ on DKP and Extended IP Phone extensions.

For instructions on assigning these features to DSS keys of a DKP, see “DSS Keys Programming”.

For instructions on assigning these features to DSS keys of the Extended IP Phone, see “DSS Key Settings” in “Configuring Matrix SPARSH VP330”, “DSS Key Settings” in “Configuring Matrix SPARSH VP248”, “DSS Key Settings” in “Configuring Matrix SPARSH VP310”, “DSS Key Settings” in “Configuring Matrix SPARSH VP210” and “DSS Key Settings” in “Configuring Matrix SPARSH VP510”

How to use

For EON & Extended IP Phones

To set Message Wait:

- Press DSS Key assigned to 'Message Wait'.
  OR
- Dial 1076
- Dial the extension number on which you want to set Message Wait.
- Select the option ‘Set Message Wait’
- Press Enter Key.

To cancel Message Wait:

- Press DSS Key assigned to 'Message Wait'.
  OR
- Dial 1076
- Dial the extension number on which you want to cancel Message Wait.
- Select the option ‘Cancel Message Wait’
- Press Enter Key.

To retrieve Message Wait:

- Press DSS Key assigned to 'Retrieve Message Wait' or the Voice mail Key when the LED glows.
  OR
- Dial 1077

For SLT

To set Message Wait:

- Pick up the handset.
- Dial 1076
- Dial extension number on which you want to set Message Wait.
- Dial 1
- Replace the handset.

To cancel Message Wait:

- Pick up the handset.
- Dial 1076
- Dial extension number on which you want to cancel Message Wait.
- Dial 0
- Replace the handset.

To Retrieve Message Wait:

- Pick up the handset.
- Dial 1077

Mobility Extension

What's this?

SARVAM UCS offers mobility to its extension users whose nature of work keeps them away from their desks frequently and for longer durations.

Using mobility extensions, the extension users of SARVAM UCS can make and receive their calls from their current (remote) location, placing calls through the system, and can access the system just as any other normal extension of the SARVAM UCS.

SARVAM UCS offers mobility using:

- VARTA UC Clients — VARTA ADR100 or VARTA AMP100. For details refer to the respective User Guide.
- DISA. For details, refer to the explanation given below.

How it works

SARVAM UCS supports two types of users:

- **Station Users:** they are extension users of SARVAM UCS to whom a dedicated physical station is assigned on their desk.

- **Virtual Users:** they are extension users of SARVAM UCS who share a physical station, or may not have any physical station allotted to them.

  The facility of Mobility Extension is provided to both virtual and normal extension users using the features "Direct Inward System Access (DISA)" and "Call Forward".

How to configure

To provide Mobility Extension to users, the follow the steps described below.

- List out the Station Users and Virtual Users and configure them first. The software ports of SLT, DKP and ISDN Terminals which are not assigned a physical slot - port, can be used as Virtual stations.

- Make sure that stations which are to be provided Mobility Extensions have the features Call Forward and DISA enabled in their "Class of Service (COS)"[317]. Class of Service is to be programmed in the "Station Basic Feature Template" assigned to the Mobility Extensions.

- Make a list of External numbers to which the Mobility Station users will forward their calls. Program these numbers in the 'Allowed List' of Local, Regional, National and International Numbers, as appropriate.

- The Toll Control assigned to the station will be applied when a call is forwarded to an external number. Make sure that the stations which are to be provided Mobility Extension have the required "Toll Control" level for Call Forward to the External Numbers (the numbers you have programmed in the Allowed List).

317. *It is possible to map the virtual user's flexible number (extension number) to a physical station. With this mapping, whenever a virtual user's number is dialed, the call is placed to the assigned physical station. The physical station can be assigned by defining it as the 'Landing Destination for Virtual Users' in the Station Advanced Feature Template assigned to the virtual user.*

Toll Control level is to be programmed in the “Station Basic Feature Template” assigned to the Mobility Extensions.

- Program the parameter 'External Call Forward for' in the “Station Advanced Feature Template” assigned to the Mobility Extensions. This parameter defines the types of call for which the External Call Forwarding is to be applied. Select any one of the options, Internal Calls Only, Trunk Calls Only, Internal + Trunk Calls as required.

- Program the parameter 'DISA' in the “Trunk Feature Template” of the trunk lines which Mobility Extensions users are to be provided access to. Make sure you select the option 'CLI Auth. Multiple Calls' in the 'DISA' parameter of the 'Trunk Feature Template'.

- Make a list of numbers which the Mobility Extension users will use to access the SARVAM UCS from DISA mode. Program this list of numbers in the "DISA-CLI Authentication Table".

  Program this list of numbers in the 'Calling Number' field of the Authentication Table. Program the 'Port Type' and 'Port Number' of the Station assigned to Mobility Extension Users in the 'Auto Login As' field for the respective 'Calling Number' field. Refer the topic “Direct Inward System Access (DISA)” to know more.

How to use

The Mobility Extension Users of SARVAM UCS can use the features of SARVAM UCS from a remote location as described below.

Receiving calls

To receive calls, the Mobility Extension User must set Call Forward on his station with an external number (mobile number, landline number, etc.) as the destination number.

To make calls ring on the station and the external number simultaneously, the Mobility Extension User must activate the Call Forward-Dual Ring feature on his station.

The Mobility Extension User can also choose where he wants to receive the calls during a particular time of the day. For example, he can receive calls during a particular time of the day, that is, Time Zone on his external number and have his calls received by his Voice mail or the Operator or any other number during another Time Zone. To do this, he must set **“Call Forward-Scheduled"** on his station. Dual Ring can also be set for Call Forward-Scheduled.

Making calls

The Mobility Extension User should make a call on the DISA enabled trunk of SARVAM UCS from the external number and the system will provide the dial tone to the user after authenticating the external number with the help of the DISA-CLI Authentication table.

On getting the dial tone, the Mobility Extension User can make internal as well as external calls as per the “Toll Control” and “Class of Service (COS)” assigned to his Station.

The Mobility Extension User can also dial codes of the Personal directory and Global directory numbers to use the feature Abbreviated Dialing.

Accessing Features

The Mobility Extension User can access the system features by dialing specific codes after making calls on the DISA enabled trunk of SARVAM UCS, or after answering the calls received on his external number.

These codes are listed below.

Activity	Code to be dialed
On-Hook	***#0***
Off-Hook	***#1***
Flash	***#2***
Pause	***#3***
Terminate the call	***#9***

Described below are instructions for Mobility Extension users on using different call management features.

Call Hold

To put a call on hold,

- When in speech with Party A, Mobility Extension User dials ***#2.***
- The call with Party A is put on Hold.
- Press Flash (***#2***) to retrieve the held call and talk to Party A again.

Call Transfer

To conduct a screened Call Transfer,

- When in speech with Party A, Mobility Extension User dials ***#2.***
- The Call of Party A is put on Hold.
- Mobility Extension User gets Feature Tone.
- Dial Trunk Access Code (TAC) followed by the number of Party B. [To make external call]

  OR

- Dial Internal station number [to make call on a station]
- When in speech, dial ***#0*** to go on hook.
- Party A and Party B will get connected.

To perform Call Transfer-While Ringing,

- Mobility Extension User is in speech with Party A.
- During speech, Mobility Extension User dials ***#2.***
- The Call of Party A is put on Hold.
- Mobility Extension User will get Feature Tone.
- Dial TAC - number of Party B. [To make external call]

  OR

- Dial Internal station number [to make call on station]
- Dial ***#0*** on receiving Ring Back Tone (RBT).

To perform Call Transfer-On Busy Station,

- Mobility Extension User is in speech with Party A.
- During speech, Mobility Extension User dials ***#2.***
- The Call of Party A is put on Hold.
- Mobility Extension User will get Feature Tone.
- Dial number of Station user.
- Dial ***#0*** on receiving Busy Tone.

To perform a Call Transfer-Trunk-to-Trunk,

- Mobility Extension User is in speech with Party A.
- During speech, Mobility Extension User dials ***#2.***
- The Call of Party A is put on Hold.
- Mobility Extension User will get Feature Tone.
- Dial Trunk Access Code to grab the Trunk
- After receiving the Dial Tone of the trunk, dial number of Party B.
- After Party B answers the call, dial ***#0.***
- Party A will get connected with Party B.

Making a Second Call

To make a second call by putting the current call on hold,

- Mobility Extension User is in speech with Party A.
- During speech, Mobility Extension User dials ***#2.***
- The call of Party A is put on hold.
- Mobility Extension will get Feature Tone.
- To make an external call: dial TAC - number of Party B.

  OR

- To make an internal call: dial extension number.

Call Toggle (Call Splitting)

To toggle between two calls,

- Mobility Extension User is in speech with Party A.
- During speech, Mobility Extension User dials ***#2.***
- The call of Party A is put on hold.
- Mobility Extension will get Feature Tone.
- Dial number of Party B.
- Make speech with Party B.
- Dial ***#2-1*** to put Party B in hold and speech with Party A.
- Again dial ***#2-1*** to put Party A in hold and speech with Party B.

Call Pick Up

To pick up the call of same Call Pick Up-Group,

- Make a call on DISA enabled Trunk from Mobility Extension User.
- After receiving dial tone, dial ***4.***

- The call ringing on the station of the same Call Pick up-Group will get connected with Mobility Extension User.

To pick up the selective call,

- Make a call on DISA enabled Trunk from Mobility Extension User.
- After receiving dial tone, dial ***12-Station Number.***
- The call ringing on the dialed station of will get connected with Mobility Extension User.

3-Party Conference

To conduct a 3-Party Conference,

- Make the first call.
- Dial ***#2.***
- Make the second call.
- Dial ***#2-*3*** to establish conference.

Multiparty Conference

To create a Multiparty Conference,

- Mobility Extension User has generated three-party conference with B and C.
- Dial ***#2*** from Mobility Extension user.
- Dial number of Party D.
- After speech, dial ***#2-*3.***
- Mobility Extension User, B, C and D are in Multi-party Conference.
- Dial number of Party E.
- Dial ***#2*** from Mobility Extension user.
- After speech, dial ***#2-*3.***
- Mobility Extension User, B, C, D and E are in Multi-party Conference.
- Dial ***#2*** from Mobility Extension user.
- Dial number of Party F.
- After speech, dial ***#2-*3.***
- Mobility Extension User, B, C, D, E and F are in Multi-party Conference.
- All Parties will get connected with each other.

To terminate the Multiparty Conference,

- When in the middle of a Multiparty Conference,
- Dial ***#2***.
- Dial ***190*** after getting feature tone.

To temporarily leave the Multiparty Conference,

- When in the middle of a Multiparty Conference,
- Dial ***#2***.
- Dial ***191*** after getting feature tone.

To rejoin the Multiparty Conference,

- Dial **#1** to go Off Hook.
- Dial ***191***.

To permanently leave the Multiparty Conference,

- While in multi-party conference, dial **#0** to go Off Hook.

When Dialed Station is busy

To make a call on an internal station which is busy,

- During busy tone
- Dial ***3*** for interrupt request.
- Dial ***4*** to Barge-In
- Dial ***5*** to Raid.

Call Forward-Unconditional

To set Call Forward-Unconditionally,

- Make a call on DISA enabled Trunk from Mobility Extension User.
- After getting Dial Tone, dial ***131-Station/Department Group/VMS.***
- The system will give confirmation tone.

To set Call Forward-Unconditionally on external number,

- Make a call on DISA enabled Trunk from Mobility Extension User.
- After getting Dial Tone, dial ***131-TAC-external number.***
- The system will give confirmation tone.

Call Forward-Busy

To set Call Forward-Busy,

- Make a call on DISA enabled Trunk from Mobility Extension User.
- After getting Dial Tone, dial ***132-Station/Department Group/VMS.***
- The system will give confirmation tone.

To set Call Forward-Busy on external number,

- Make a call on DISA enabled Trunk from Mobility Extension User.
- After getting Dial Tone, dial ***132-TAC-external number.***
- The system will give confirmation tone.

Call Forward-No Reply

To set Call Forward-No Reply,

- Make a call on DISA enabled Trunk from Mobility Extension User.
- After getting Dial Tone, dial ***133-Station/Department Group/VMS.***
- The system will give confirmation tone.

To set Call Forward-No Reply on external number,

- Make a call on DISA enabled Trunk from Mobility Extension User.
- After getting Dial Tone, dial ***133-TAC-external number.***
- The system will give confirmation tone.

Call Forward-Busy/No Reply

To set Call Forward-Busy/No Reply,

- Make a call on DISA enabled Trunk from Mobility Extension User.
- After the Dial Tone, dial ***134-Station/Department Group/VMS.***
- The system will give confirmation tone.

To set Call Forward-Busy/No Reply on external number,

- Make a call on DISA enabled Trunk from Mobility Extension User.
- After the Dial Tone, dial ***134-TAC-external number.***
- The system will give confirmation tone.

To cancel Call Forward,

- Make a call on DISA enabled Trunk from Mobility Extension User.
- After the Dial Tone, dial ***130.***
- The system will give confirmation tone.

Call Forward-Dual Ring

To set Call Forward-Dual Ring,

- Make a call on DISA enabled Trunk from Mobility Extension User.
- After the Dial Tone, dial ***136-1.***
- The system will give confirmation tone.

To cancel Call Forward-Dual Ring,

- Make a call on DISA enabled Trunk from Mobility Extension User.
- After the Dial Tone, dial ***136-1.***
- The system will give confirmation tone.

Call Forward-Scheduled

To set/cancel Call Forward-Scheduled,

- Make a call on DISA enabled Trunk from Mobility Extension User.
- After the Dial Tone, dial ***1175-Time Zone-Call Forward Type-Number.***
- The system will give confirmation tone.

- *Mobility Extension users can have Call Forward and Call Forward-Scheduled set on their extension by the Operator or by another extension user.*

- *Using "Call Forward-Remote" and by setting "Call Forward-Scheduled" from the SA mode (using SA commands or Jeeves), the extension user/Operator can set Call Forward and Call Forward-Scheduled for any Mobility extension user. Refer the respective topics to know more.*

Multi-Stage Dialing

What's this?

The Multi-Stage Dialing feature of SARVAM UCS is typically used in applications like Calling Card, where extension users are required to dial digits in stages when making a call using the calling card.

The Multi-Stage feature enables extension users to directly dial the number they want to call, and the system dials out the number at different stages of the call by suitably modifying the number.

How it works

A typical example of Multi-Stage Dialing is the use of prepaid Calling Cards. Here, the person using a calling card must dial a fixed number string before dialing the actual number. When using a calling card,

- Users must first dial the number of the Calling Card server, for example: 1602233 (7 digits).
- After the call is answered by the Calling Card server, users must dial the PIN provided by the calling card service provider, for example 1132121234.
- After dialing the PIN number, users can dial the number they want to call, for example 0014125126508.

Thus, when using a Calling Card, users must dial a very lengthy number string, each time they need to make a call using the Calling Card.

The use of Multi-Stage Dialing saves the time and effort of dialing out lengthy digits in stages.

The Multi-Stage Dialing makes use of the “Automatic Number Translation” table. This table must be configured on the trunk from which extension users will make calls using Calling Cards.

To take the above example further,

- Say, extension users are allowed to make international calls using Calling Card from the trunk CO1. “Automatic Number Translation” table must be configured on CO1 trunk.
- The Automatic Number Translation table consists of Dialed Number Strings, Strip Digit and Add Prefix Number strings.
- In Dialed Number, you must configure ‘00’, the prefix for international numbers.
- In Add Prefix you must configure the Calling Card server number and the PIN Number.
- Keep Strip Digits as 00.
- As the system must wait for the Calling Card server to answer before dialing the PIN, you must configure Wait for Answer between the Calling Card server number and the PIN number.

  You must insert a delay by configuring the Pause Timer after the PIN number and the destination number.

- The Automatic Number Translation table would look like this:

Index	Dialed Number String	Strip Digits	Add Prefix
1	00	00	1602233**W**1132121234**P**
2			
3			
4			
5			
6			
:			
32			

- When the Automatic Number Translation table is configured, the Extension user can simply dial the Trunk Access code and the destination number: 0/5/61/62/63/64 - 0014125126508.

- The system matches the dialed number with the Dialed Number String of the ANT table, the number matches with the entry '00' stored in the table.

- The system dials the Add Prefix Number string 1602233 (number of the calling card server). It waits for the calling card server to answer the call.

- When the call is matured, i.e. the calling card server has answered the call, the system dials the PIN number 1132121234 and waits for the Pause Timer before dialing the destination.

Thus, the extension user directly dials the desired destination number and the system substitutes this number by adding the Calling Card server number and PIN number and dials these numbers in two stages.

How to configure

Configuring Multi-Stage Dialing using Jeeves

To be able to use Multi-Stage Dialing, you must configure the following:

- Automatic Number Translation table on the trunk you want to use this feature. For instructions on configuring ANT on different Trunk types, see "Automatic Number Translation" and to assign the ANT Table number see "Outgoing Trunk Bundle".

- Configure the **DTMF Out Dial** on the Trunk. Set the **DTMF ON Time** and the **DTMF Inter Digit Pause Timer** and **Pause Timer** to the required values.

  For instructions on configuring DTMF Out Dial on different Trunk types, see "Configuring CO Trunks", "Configuring E1 Trunks", "Configuring T1 Trunks", "Configuring BRI Trunks", "Configuring Mobile Trunks", "Configuring SIP Trunks", "Configuring E&M Lines".

- If required you may configure the **Call Proceeding Tone for Multi-stage Dialing** as **Network tone**, **Pseudo Tone**, or **Silent**. For instructions, see "System Parameters".

Configuring Multi-Stage Dialing using a Telephone

Notes for programming Number String:

- It is recommended to SE, to not to program Pause character "P" before "W" character for the number string to be out dialed from Mobile port. Else, the GSM Module may get restart while out dialing the DTMF digits without call maturity signal.

- Use following Codes to program the special digits:

Special Digits	Code for Programming through Command
Flash (F)	#2
Pause (P)	#3
A	#4
B	#5
C	#6
D	#7
+	#8
. (dot)	#9
#	##
*	**
W	*1

Enable 'ANT' for the trunk port using command 6702. Refer chapter "Outgoing Trunk Bundle" for more details.

Assign the ANT Table to the trunk port using command 6702. Refer chapter "Outgoing Trunk Bundle" for more details.

Refer chapter "CO Hardware Template" for command ***'5902'*** to program Pause Timer on CO Trunk.
Refer chapter "Configuring PRI Trunks" for command ***'6109'*** to program Pause Timer on T1E1 Trunk.
Refer chapter "Configuring BRI Trunks" for command ***'6209'*** to program Pause Timer on BRI port.
Refer chapter "Configuring Mobile Trunks" for command ***'8014'*** to program Pause Timer on Mobile port.
Refer chapter "E&M Feature Template" for command ***'6002'*** to program feature Pause Timer on E&M trunk.

Refer chapter "CO Hardware Template" for command ***'5902'*** to program DTMF ON Time on CO Trunk
Refer chapter "Configuring PRI Trunks" for command ***'6117'*** to program DTMF ON Time on T1E1 Trunk.
Refer chapter "Configuring BRI Trunks" for command ***'6210'*** to program DTMF ON Time on BRI port.
Refer chapter "Configuring Mobile Trunks" for command ***'8015'*** to program DTMF ON Time on Mobile port.

Refer chapter "CO Hardware Template" for command ***'5902'*** to program Inter Digit Pause Time on CO Trunk
Refer chapter "Configuring PRI Trunks" for command ***'6118'*** to program Inter Digit Pause Timer on T1E1 Trunk.
Refer chapter "Configuring BRI Trunks" for command ***'6211'*** to program Inter Digit Pause Timer on BRI port.

Refer System Parameters for command ***'5311'*** for programming of Call Proceeding Tone.

Music on Hold (MOH)

What's this?

The music played to extension users and external callers who are put on hold is called Music on Hold (MoH).

How it works

SARVAM UCS supports Music on Hold from an Internal Music Source.

A Voice Module assigned the Music on Hold application serves as the source of music for this feature. By default, Voice Module 1 is assigned for Music on Hold.

When a caller is put on hold, SARVAM UCS plays the music recorded in Voice Module 1.

You can play a voice message of your choice instead of music to the callers. The message may contain any promotional information about your company or services provided by your organization, etc. For this, you must record a Voice Module with the custom message and assign the Voice Module to the Music on Hold application.

*If the option **Routing Group** is selected as the **Alarm Notification Type** for an extension, when the extension goes Off-hook to answer an alarm call, and the extensions in the Routing Group for Alarm Notification are busy, Music-on-Hold will also be played to the extension answering the alarm call.*

*If your DKP/SIP Extension supports multiple call appearance and you want the system to play MOH to internal callers, when your extension is busy, enable the **Play MOH to Queued Internal Calls on DKP/ Extended IP Phone** check box in "System Parameters". As soon as your extension is free, Ring Back Tone will be played to the caller.*

How to configure

You can record a piece of music or a message of maximum 16 seconds duration and assign it to Voice Module 01, which is reserved for Music-on-Hold.

Refer the topic "Voice Message Applications" for instructions on recording and assigning voice modules.

Mute

What's this?

This feature helps the extension user to disconnect the speech transmission path in the middle of a conversation. The extension user can still listen to the opposite party because the receiving path remains connected. Mute is useful when you want to consult someone in the middle of a conversation, but do not want the opposite party to listen to your discussion. You can Mute a call before making a call or during speech.

How it works

- A is in speech with B.
- A wants to consult to C in the room, but does not want B to hear their conversation.
- A presses the Mute Key.
- The transmit speech path from A to B is disconnected. The receive path remains connected.
- So, A will be able to hear B, but B will not be able to hear the conversation between A and C.
- When A has finished consulting C, to resume speech with B, A presses the Mute key again.
- The transmit speech path from A to B is restored. A and B are in speech again.

How to use

For EON & Extended IP Phone Users

To mute a call before making the call:

- Press the 'Mute' Key
  The LED of the key glows.
- Dial a number on confirmation tone.
  OR
- Dial 1052
- Dial desired number.

To mute a call during speech:

- Press the 'Mute' Key to silence outgoing speech.
  OR
- Press Transfer Key.
- Dial 1052

To resume outgoing speech:

- Press the 'Mute' Key.
  The LED of the key is turned off.
  OR
- Press Transfer Key.
- Dial 1052

For SLT Users

To mute a call before making the call:

- Dial 1052
- Dial desired number.

To mute a call during speech:

- Press Flash Key.
- Dial 1052

To resume outgoing speech:

- Press Flash Key.
- Dial 1052

Number Lists

What's this?

A 'Number List' is a group of number strings. SARVAM UCS uses Number Lists to support different features as Toll Control, Call Duration Control, Call Taping, Call Back on Trunk Ports, Station Message Detail Recording (SMDR).

SARVAM UCS supports 16 Number Lists. Each Number List can contain upto 999 number strings. Each number string consist of a maximum of 16 characters.

The number strings are stored against Location Index numbers in the Number List. The Location Index numbers start from 001 to 999.

The default values of the Number Lists in the system are shown below:

Default Number Lists

Location Index	List Number															
	01	02	03	04	05	06	07	08	09	10	11	12	13	14	15	16
001		00	0	00	*	*	*	*	*	*	*	*	*	*		
002		0	*	*	#	#	#	#	#	#	#	#	#	#		
003		1	#	#	F	F	F	F	F	F	F	F	F	F		
004		2	F	F												
005		3														
006		4														
007		5														
008		6														
009		7														
010		8														
011		9														
012		*														
013		#														
014		F														
015		+														
016																
:	:	:	:	:	:	:	:	:	:	:	:	:	:	:	:	:
999																

Use of Number Lists in various Features of SARVAM UCS

Call Back on Trunk Ports

When Call Back on set on any of the Trunk port types (CO, Mobile, BRI, T1E1, SIP), you must program two lists:

- Call Back Incoming Number List: this list defines the numbers which are eligible for a call back.
- Call Back Outgoing Number List: this list define the number to which the call back is to be made.

By default, Number List 15 is assigned to Call Back Incoming Number List, and Number List 16 is assigned to Call Back Outgoing Number List.

Call Duration Control

To set Call Duration Control (CDC) feature on an extension, you must program the Call Duration Control Table, in which you must program the numbers on which CDC must be applied as well as the numbers on which CDC must not be applied.

This requires you to program and assign two Number Lists: the Apply CDC to Number List and the Do Not Apply CDC to Number List.

By default, Number List 02 (For CDC Table 1, for others it is 7) is assigned to Apply CDC Number List, and Number List 08 is assigned to Do Not Apply CDC Number List.

Refer “Call Duration Control (CDC)” to know more.

Call Taping

Call Taping allows you to record conversations of incoming and outgoing internal and external calls. When you set Call Taping on an extension for external calls, the SARVAM UCS uses two Lists: Number List - Incoming Calls: this list has the list of numbers of external callers whose conversation is to be recorded.

Number List - Outgoing Calls: this list has phone numbers of external called parties whose conversation is to be recorded. The system matches the incoming and outgoing numbers with the respective lists to apply Call Taping.

By default, Number List 09 is assigned to Incoming Calls Number List, and Number List 10 is assigned to Outgoing Calls Number List.

Refer the feature description “Call Taping” to know more.

Toll Control

The Toll Control Call Privilege Type “Limited Calls” allows and restricts dialing of telephone numbers starting with a particular digit or a particular area code or certain telephone numbers only. To apply Call Privilege type ‘Limited Calls’ you must program an Allowed Number List with numbers that are to be allowed, and a ‘Denied List’ with numbers that are to be restricted.

Refer the feature description for “Toll Control” to know more.

Station Message Detail Recording (SMDR)

SARVAM UCS supports ‘Destination wise’ storage of outgoing calls as an SMDR Filter, using which it is possible to store calls made from and received on selected destination numbers. To set Destination wise storage of calls in the SMDR Buffer, you must program and assign a Number List containing the phone numbers whose call details need to be tracked and stored. SMDR of an outgoing call will be tracked and stored only if it matches with an entry in the Number List assigned.

Similarly, SMDR Reports of Incoming and Outgoing Calls can be generated and printed for selected numbers by programming and assigning a Number List.

By default, Number List 02 is assigned to Destination Wise storage of SMDR and SMDR Report Printing.

Refer the topics “Station Message Detail Recording (SMDR)”, “Station Message Detail Recording-Storage”, “Station Message Detail Recording-Report”.

SMS on No Reply

SMS on No Reply feature is used to send an SMS to the mobile user only, when an extension user calls the mobile user and the mobile user doesn't answer the call. The SMS that is sent to the mobile user contains the extension user’s contact information and the message to be conveyed in brief, so that the mobile user can call back if necessary. You must enter only the first digit of the number string, or a part of the string, or the complete number string in the Number list for sending SMS. The system compares the dialed numbers with the numbers in this list to confirm if the outdailed numbers are mobile numbers. Only if a match is found the SMS is sent.

By default, Number List 16 is assigned for SMS on No Reply.

Refer the feature description for “SMS on No Reply” to know more.

How to configure

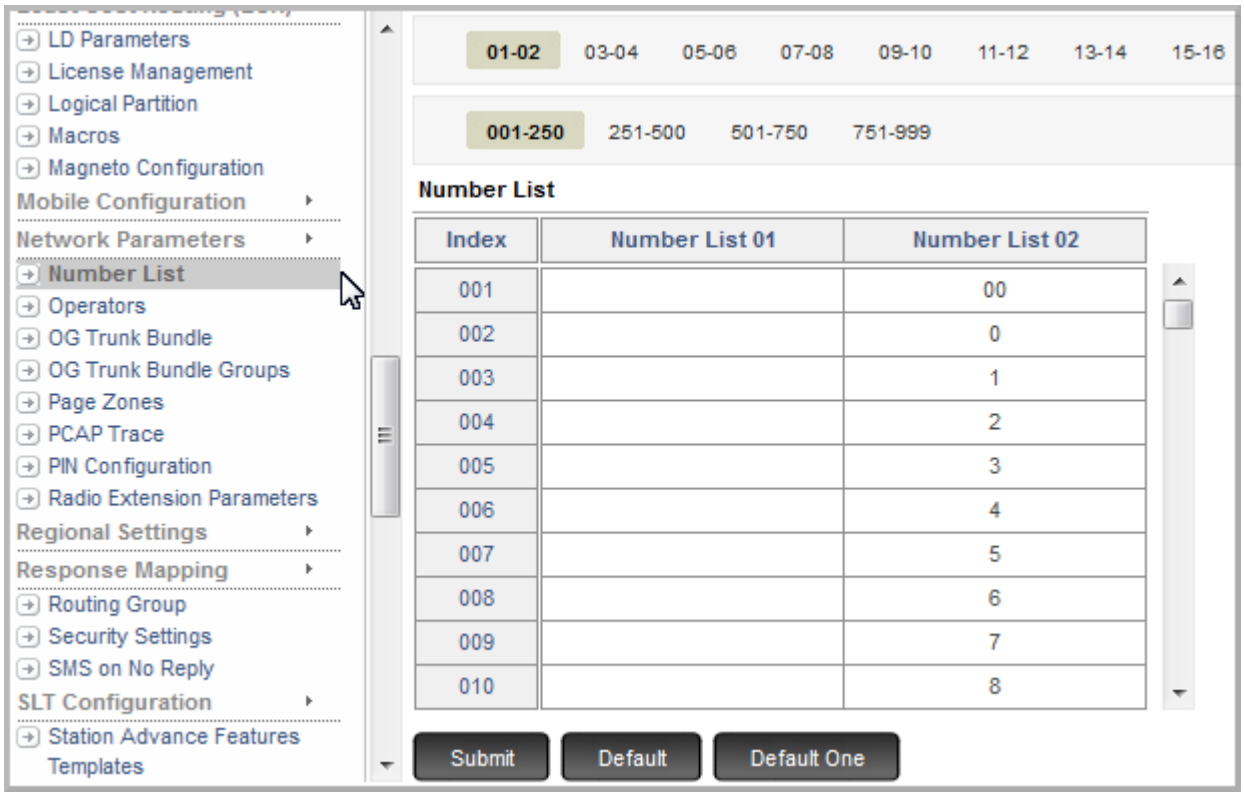
Take a pen and a paper. Decide which of the above-mentioned seven features are to be used. Number List according to the feature for which it is to be used.

Number Lists can be programmed from Jeeves or by dialing SE commands from a telephone.

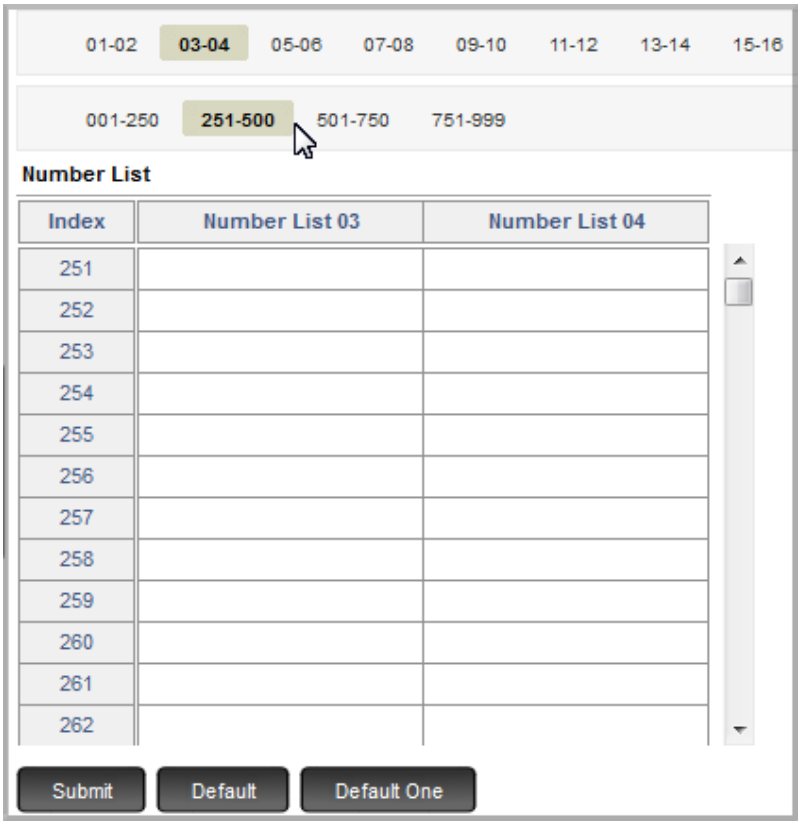
Programming Number List using Jeeves

- Log into Jeeves as System Engineer.

- Under **Configuration**, click **Number List** to open the page.



- Select the desired List Number. For example, Number List 03-04. Now click '001-250' of Number list 03-04. The location Index 001 to 250 will appear for both lists on the page.



- Enter each number string against a Location Index (refer to the list you prepared).
- Click **Submit** at the bottom of the page to save your list.
- Assign the Number List you prepared to the relevant feature.
- Log out of Jeeves if you have no other configuration tasks.

Programming Number List using a Telephone

- Enter SE mode.

  To program a number in a Number List, dial:
  - ***4302-Number List-Location Index-Number-#****
    Where,
    Where
    List is from 01 to 16.
    Location Index is from 001 to 999.
    Number is a number string of maximum 16 digits. The digits allowed are: 0-9, #, *, A, B, C, D, F, P, +.
    Refer the table below for codes for dialing the special digits. Terminate the number string with **#***

Special Digit	Code
Flash (F)	#2
Pause (P)	#3
A	#4
B	#5
C	#6
D	#7
+	#8
Dot (.)	#9
#	##
*	**

  To clear the number programmed at a Location index in a Number list, dial:
  - ***4302-Number List-Location Index-#****
    Where,
    Number List is from 01 to 16.
    Location Index is from 001 to 999.

  To default Number Lists, dial:
  - ***4301-1-Number List*** to default a single number list.
  - ***4301-2-Number List-Number List*** to default a range of number lists.
  - ***4301-**** to default all number lists.
    Where,
    Number List is from 01 to 16.

- Exit SE mode.

OFF-Hook Alert

What's this?

When the handset of an extension is not placed correctly, it will not be possible for the Operator or any other extension to call the extension. Also, incoming calls will not reach the extension, Alarms and Reminders cannot be placed on that extension.

To avoid this inconvenience, the SARVAM UCS supports the feature 'OFF-Hook Alert', whereby the system detects and informs the Operator of the extension phone that remains OFF-Hook accidentally.

How it works

To give the Operator an OFF-Hook Alert,

- SARVAM UCS places a call on the Operator's phone.
- It displays a message on the Operator's phone “<extension number> Stand-By”
- When the Operator answers the call, s/he is played a confirmation tone, the text message “Hangup <extension number> Properly” is displayed.
- The Operator can send someone to inform the extension user to place the handset of the phone correctly.
- If the extension phone is an SLT, OFF-Hook Alert will be given to the Operator phone only. The Operator phone can be an EON or a SLT with CLI support.
- If the extension phone that is OFF-Hook is EON, the SARVAM UCS will activate 'OFF-Hook Alert' on the extension phone, by playing the Error Tone continuously, on speaker to draw the attention of the extension user.
- While the Error Tone for OFF-Hook Alert is being played on the extension phone, if the user presses the Speaker Key, the Error Tone will continue to be played on the handset until it is replaced correctly.

How to configure

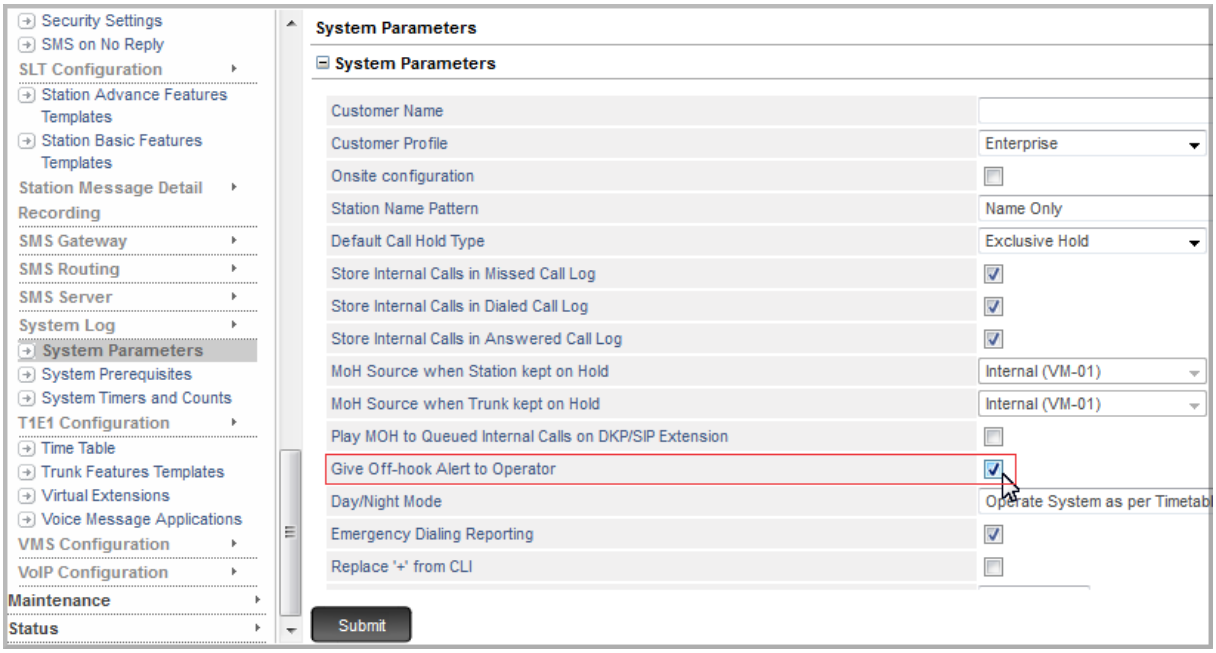
For this feature to work,

- the 'OFF-Hook Alert to Operator' flag must be enabled by the System Engineer in the 'System General Parameters'.
- the Operator phone can be EON or SLT with CLI support, the extension phones may be EON or SLT.

Programming 'OFF-Hook Alert to Operator' using Jeeves

- Log into Jeeves as System Engineer.
- Under **Configuration**, click **System Parameters** to open the page.

- Go to the parameter **Give Off-hook Alert to Operator**, select the check box to enable the flag.



- Click **Submit** to save the change.

- Log out of Jeeves or continue programming as required.

Programming 'OFF-Hook Alert to Operator' using a Telephone

- Enter SE mode from a DKP/SLT.
- Dial ***5333-1*** to enable the flag.
- To disable the flag, dial ***5333-0.***
- Exit SE mode.

One Touch Transfer

What is this?

SARVAM supports the UC feature, One Touch Transfer. Using this you can transfer an ongoing call from one extension to another mobile/fixed extension without putting the call on hold or dialing the destination extension number.

SARVAM UCS will serve the One Touch Transfer request made by a DKP or a SIP Extension (Matrix Extended IP Phones or Mobile Clients) only. The fixed destination extension number can be a SLT, DKP, SIP Extension or an ISDN Terminal.

How it works

Consider the scenario, wherein you have the following registered as your extensions:

1. VARTA Mobile UC Client is registered as SIP Extension 1 at location 1
2. Extended IP phone is registered as SIP Extension 1 at location 2.

- You are in speech with A using your Extended IP Phone and you want to move away from your desk.
- In this case, you can transfer the ongoing call on your VARTA Mobile UC Client using One Touch Transfer.
- Press the DSS key assigned to One Touch Transfer (make sure the extension number of SIP Extension 1 is pre-configured as destination number for One Touch Transfer).
- You will receive the call on your VARTA Mobile UC Client.
- Answer the call from the VARTA UC Client. Speech is established and the call on your Extended IP Phone disconnects.
- You can now move away from your desk but your call can continue.

Similarly, you can transfer a call to another fixed extensions also.

You can access One Touch Transfer only during an ongoing 2-way speech. A held or a waiting call cannot be transferred using One Touch Transfer.

For using One Touch Transfer between Extended IP Phone/VARTA UC Clients registered on different locations of the same SIP Extension, make sure the Call Appearance of the SIP Extension is more than 1.

How to configure

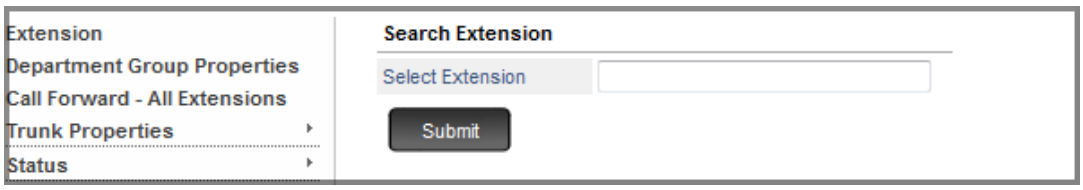
To access this feature,

- Make sure the *Basic Features* are enabled in the Class of Service assigned to you. For instructions, see “Class of Service (COS)” and “Station Basic Feature Template”.
- Make sure you have configured a DSS Key for One Touch Transfer. For instructions on configuring DSS Key, see the topic “DSS Keys Programming”.

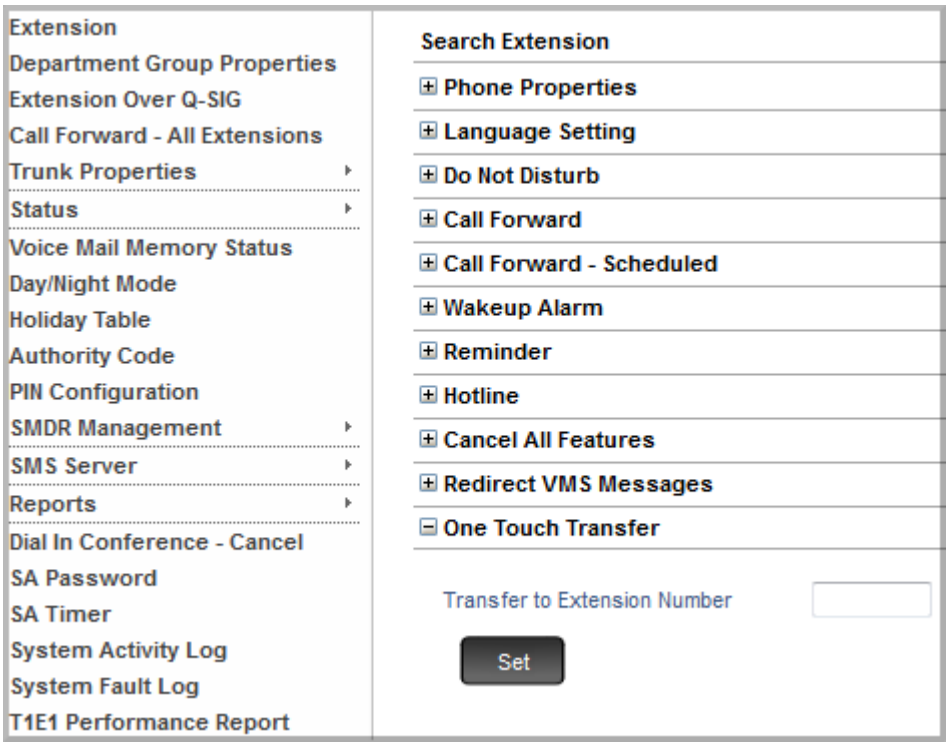
- Configure the destination extension number for One Touch Transfer. This number can be configured by extension users themselves from the Phone Menu or for any other extension from the SA mode.

Configuring One Touch Transfer using SA Jeeves

- Log in to Jeeves as System Administrator.
- Click **Extension**.



- In **Select Extension,** enter the Number or the Name of the extension on which you want to set this feature.
- Click **Submit**.
- The searched extension users details appear on your screen.
- Click **One Touch Transfer** to expand.



- In **Transfer to Extension Number**, enter the destination extension number on which you want the call to be transferred, when the feature One Touch Transfer is accessed.

Configuring One Touch Transfer by Extension Users using Phone Menu

To configure the Destination Extension Number,

- Scroll down the Local Menu of the phone.
- Select **One Touch Transfer** by pressing the Enter Key.
- Select **Set Transfer Number** by pressing the Enter Key.
- Enter the destination extension number on which you want the call to be transferred.
- You hear Confirmation tone.

How to use

For EON & Extended IP Phone Users

To use One Touch Transfer:

- During an ongoing call, press the DSS Key assigned to One Touch Transfer.
- The destination extension number configured for One Touch Transfer starts ringing.
- Answer the ringing call.
- You will be in speech with the remote party.

Outgoing Trunk Bundle

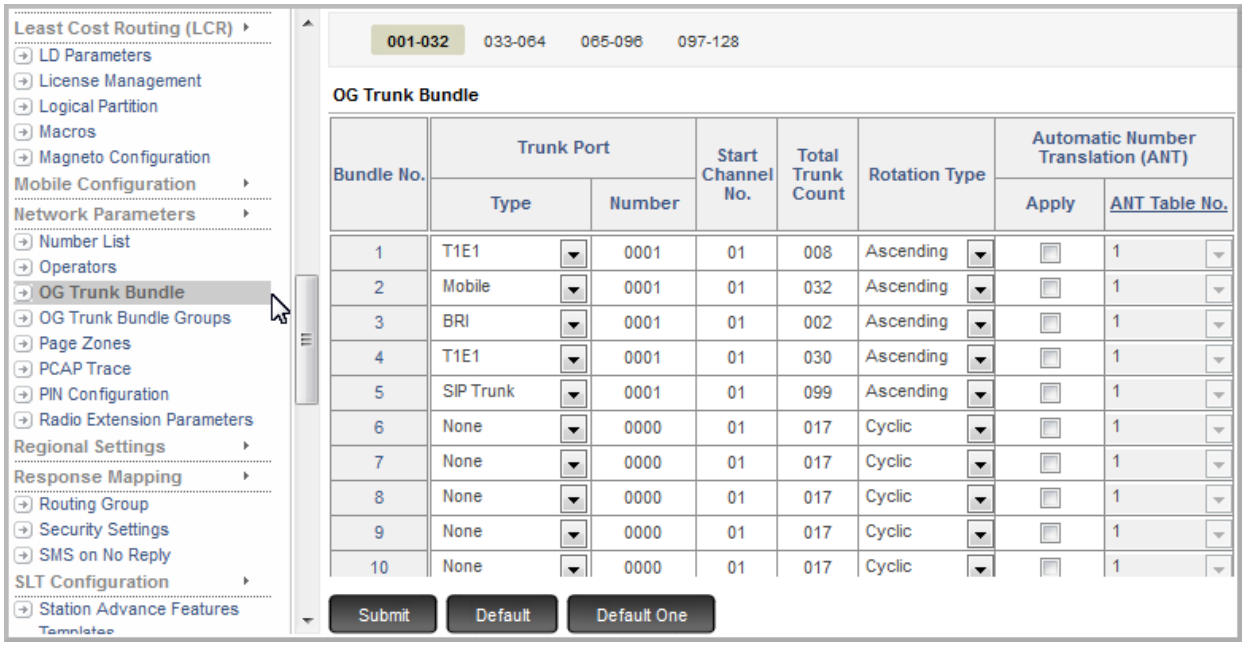
What's this?

The Outgoing Trunk (OG) Trunk Bundle is set of parameters that completely define the grouping of similar channels. The word channel refers to a speech path. By this definition, a channel can be a CO trunk, an E&M trunk, a Mobile port. Each speech path of the T1/E1 line is a channel. Bundles of similar trunks are formed. These Bundles are used to form an OGTBG. The SARVAM UCS supports 128 OG Trunk Bundles.

With the default OGTB assigned in the OGTBG outgoing calls will not be possible. Refer to the instructions below to configure the same to ensure calls are routed. Also refer "Outgoing Call Routing".

Configuring using Jeeves

- Log into Jeeves as System Engineer.
- Under **Configuration**, click **OG Trunk Bundle**.



Configure the following parameters for each Bundle Number:

- **Trunk Type:** Specific trunk bundle consists of some of these ports. The different types of ports are CO, BRI, T1E1PRI, E&M, Mobile and SIP.
- **Trunk Number:** Once the type of port is identified in the bundle, it is required to identify the number of that port, because there can be more than one port for the given type.
- **Start Channel Number:** This is applicable to BRI and T1E1 Trunks.
- **Total Trunk Count:** This is the number of trunks to be kept in the same bundle. For CO and E&M this could be 128. This value for the BRI and T1E1 Trunks is counted from the start channel. For example, if the port type is CO, port number is 002 and channel count is 025, then CO channels 002 to 027 would be

grouped together. Consider a second example where channels 15 to 25 of T1E1PRI port are to be programmed in one channel group, then the port type will be T1E1PRI, port number would be 5, start channel number will be 15 and channel count will be 11.

- **Rotation Type:** This parameter shows which channel should be selected when the next call lands on that port. For example, if ascending order is selected, the system checks 001-128, for first free channel and if descending order is selected the system checks from 128-001.

  - Ascending Order:
    - 001 to 128 (for CO/E&M)
    - 01 to 30 (for T1E1PRI)
    - 01 to 02 (for BRI)
    - 01 to 32 (for SIP)

  - Descending Order:
    - 128-001 (for CO/E&M)
    - 30 to 01 (for T1E1PRI)
    - 02 to 01 (for BRI)
    - 32 to 01 (for SIP)

  - Cyclic:
    - Always the next channel is picked for a new OG call.

- **ANT Apply:** Select from enable/disable as required. To use ANT feature, it should be enabled for the trunk port from which the number is to be dialed out.

- **ANT Table No.:** This is a Table number in which you have configured the Dialed Numbers and their corresponding Strip Digits and/or Add Prefixes. This table number is assigned to the specific trunk port from which the number is to be dialed. Refer chapter “Automatic Number Translation” for more details.

Configuring using a Telephone

Use following command to program the feature in an OG Trunk Bundle Number:
6702-1-OG Trunk Bundle Number-Feature Number-Code
6702-2-OG Trunk Bundle Number-OG Trunk Bundle Number-Feature Number-Code
6702-*-Feature Number-Code
Where,
OG Trunk Bundle Number is from 001 to 128.
Feature Number is from 1 to 6.

Use following command to set default values for OG Trunk Bundle:
6701-1-OG Trunk Bundle Number
6701-2-OG Trunk Bundle Number-OG Trunk Bundle Number
6701-*
Where,
OG Trunk Bundle Number is from 001 to 128.

Default values of OG Trunk Bundle

Bundle No.	Feature Number						
	1		2	3	4	5	6
	Trunk Port		Start Channel No.	Total Trunk Count	Rotation Type	Alternate Number Translation (ANT)	
	Type	Number				Apply	ANT Table No.
001	None	000	01	008	Ascending	No	1
002	None	000	01	032	Ascending	No	1
003	None	000	01	002	Ascending	No	1
004	None	000	01	032	Ascending	No	1
005	None	000	01	032	Ascending	No	1
006	None	000	01	017	Cyclic	No	1
007	None	000	01	017	Cyclic	No	1
:	None	000	01	017	Cyclic	No	1
060	None	000	01	017	Cyclic	No	1
061	None	000	01	008	Ascending	No	1
062	None	000	01	002	Ascending	No	1
063	None	000	01	030	Ascending	No	1
064	None	000	01	032	Ascending	No	1
065	None	000	01	017	Cyclic	No	1
:	None	000	01	017	Cyclic	No	1
128	None	000	01	017	Cyclic	No	1

Parameter Values:

Code	Port Type							
0	None	None	000				No	
1	CO	03	001-128	01-30	001-128	Ascending	Yes	1-8
2	BRI	04	001-032			Descending		
3	T1E1	05	001-008			Cyclic		
4	E&M	06	001-128					
5	Mobile	25	001-064					
6	SIP Trunks	26	001-032					

OG Trunk Bundle Group

What's this?

OG Trunk Bundle Group provides efficient allocation of trunks to different stations.

All the trunks connected to the system can be bunched in different groups called OG Trunk Bundle Group. Maximum 8 Trunk Bundles can be put in one OG Trunk Bundle Group and 32 such OG Trunk Bundle Groups can be formed.These OG Trunk Bundle Groups can be allotted to each individual extension. An Extension can be allotted different OG Trunk Bundle Group during different timings of the day.

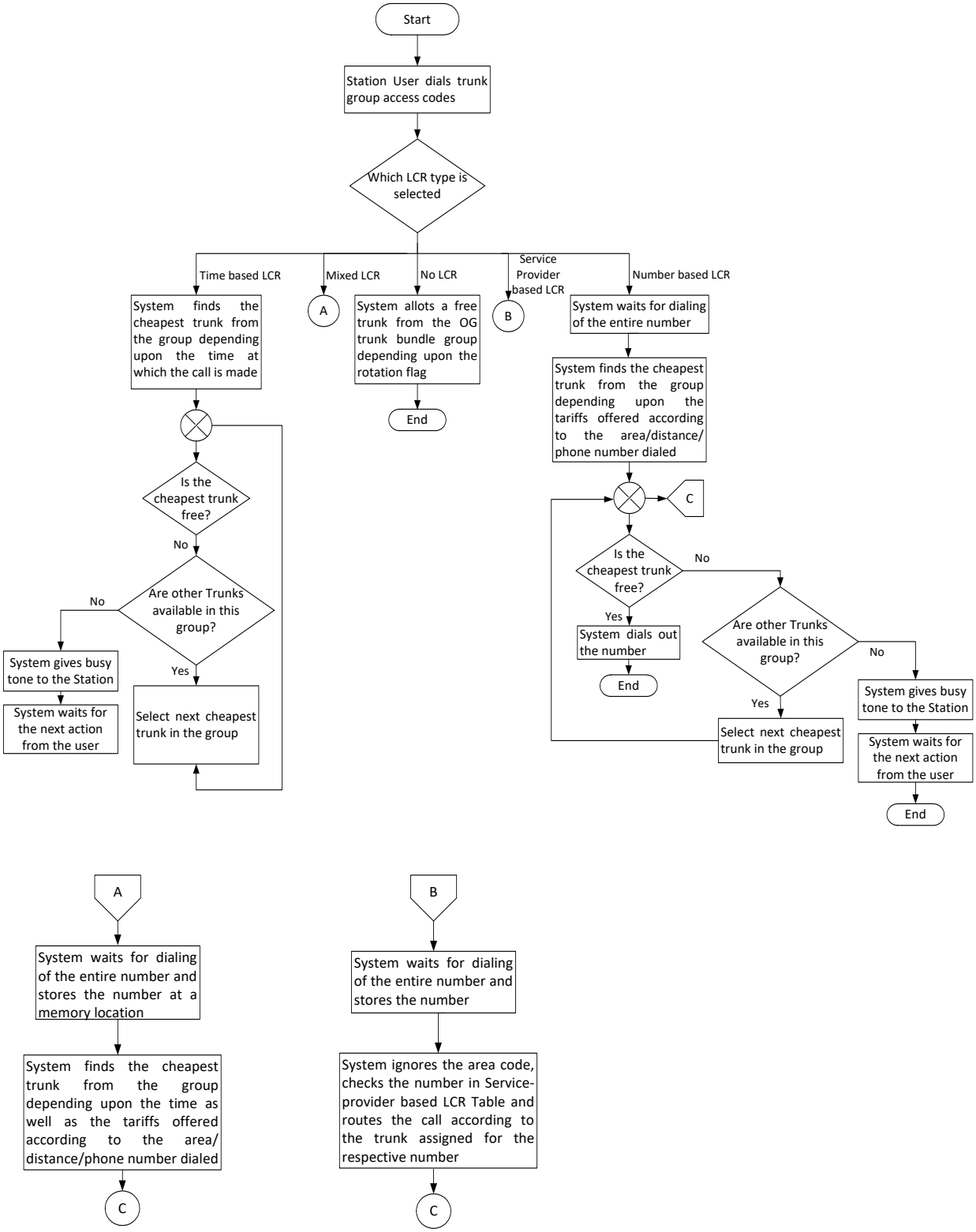
With the default OGTB assigned in the OGTBG outgoing calls will not be possible. Refer to the instructions below to configure the same to ensure calls are routed. Also refer "Outgoing Call Routing".

How it works

System uses two methods while selecting a trunk from the OG Trunk Bundle Group: 'Remember last trunk' and 'Don't Remember last trunk'.

In 'Remember last trunk' method, the system remembers the last trunk used and allots next trunk in the group to the extension. In 'Don't remember last trunk' method, the system searches for a first free trunk from the group.

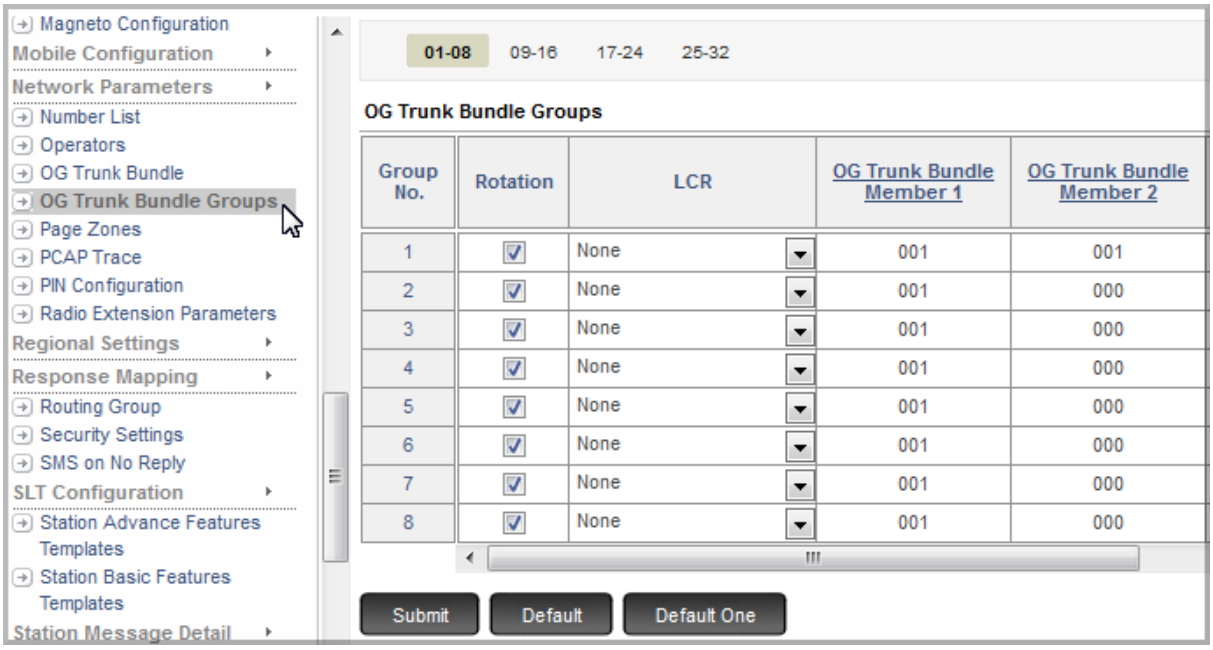
Following flow chart depicts the chronology of events when an extension grabs a trunk.



Configuring using Jeeves

- Log into Jeeves as System Engineer.

- Under **Configuration**, click **OG Trunk Bundle Groups**.



Choose the OG Trunk Bundle Group number (01-08) you want to use. In each group you can program maximum 8 'members'.

- Now program the following parameters for the selected OG Trunk Bundle Group:
  - **Rotation:** Select the Rotation check box to enable rotation for routing outgoing calls in the group which has multiple 'member' trunks. When enabled, each outgoing call will be routed through the next member to the one that routed the previous call. This ensures equal distribution in routing outgoing calls. The flag has no relevance if the group has only one member. For detailed instructions, see “Outgoing Trunk Bundle”
  - **Least Cost Routing:** If required select the desired method — Time-based LCR, Number based LCR, Time and Number based LCR or Service Provider-based LCR — as per your requirement. For details, see “Configuring LCR”.
  - **OG Trunk Bundle Member1 to 8:** Select the desired OG Trunk Bundle numbers you have already created. If you have not created the OG Trunk Bundles, for instructions see “Outgoing Trunk Bundle”.

    Configure only as many OG Trunk Bundles as members in the group as you want and set the remaining Members as '000'.
  - Click **Submit** to save.
- Repeat the same steps to create another OG Trunk Bundle Group.

Configuring using Telephone

The commands explained below should be referred as:
To program a single port: ***XXXX-1***
To program a range of ports: ***XXXX-2***

To program all the ports: ***XXXX-****

Step 1
Use following command to make default OG Trunk Bundle Group:
1401-1-OGTBG Number
1401-2-OGTBG Number-OGTBG Number
1401-*
Where,
OGTBG Number is from 01 to 32.

Step 2
Use following command to set OG Trunk Bundle:
1402-1-OGTBG Number-Destination Index-OG Trunk Bundle
1402-2-OGTBG Number-OGTBG Number-Destination Index-OG Trunk Bundle
1402-*-Destination Index-OG Trunk Bundle
Where,
OGTBG Number is from 01 to 32.
Destination Index is from 1 to 8.
OG Trunk Bundle is from 001 to 128.

For Defaults refer Table1.

Step 3
Use following command to set Rotation Flag. For explanation on how to use, refer important point at the end of chapter:
1403-1-OGTBG Number-Flag
1403-2-OGTBG Number-OGTBG Number-Flag
1403-*-Flag
Where,
OGTBG Number is from 01 to 32.

Flag	Meaning
0	Rotation OFF
1	Rotation ON

For Defaults refer Table1.

Step 4
For LCR Type-refer chapter “Configuring LCR”.

Step 5
For CPS refer chapter “Least Cost Routing-Carrier Pre-Selection”.

Step 6
Refer the topic “Station Basic Feature Template” for details on assigning OGTBG to extensions.

Step 7
How to assign an access code to OGTBG?

There are maximum 6 trunk access codes. These are Flexible access codes. Trunk access codes are common for all the users. They cannot be different for different extension.

A default trunk access codes table is given below:

OGTBG Index	Default Trunk Access Code
1	0
2	5
3	61
4	62
5	63
6	64

Use following command to program the desirable access code for a trunk access index (TAC):
3112-1-OGTBG Index-Access Code-#*/Press <Hold>
3112-2-OGTBG Index-OGTBG Index-Access Code-#*/Press <Hold>
3112-*-Access Code-#*/Press <Hold>
Where,
OGTBG Index is from 1 to 6.
Access Code is maximum 6 digits (Generally access code for trunk is of two digits).

Use following command to clear the access code for a OGTBG index:
3112-1-OGTBG Index-#*
3112-2-OGTBG Index-OGTBG Index-#*
3112-*-#*

Use following command to assign default access code for a OGTBG index:
3162-1-OGTBG Index
3162-2-OGTBG Index-OGTBG Index
3162-*

How to use

1	Lift the handset.	Dial tone
2	Dial ***0/5/61 to 64.***	Dial tone of the trunk.
3	Dial desired external number.	

- To grab OG Trunk Bundle Group 3, the user should dial ‘61’.
- To grab OG Trunk Bundle Group 1, the user should dial ‘0’.

How the Rotation flags work?

- Eight OG Trunk Bundles can be programmed as members of each OGTBG.
- OGTBG has “Rotation” flag, and each OG trunk bundle has Rotation type “Cyclic/Descending/Ascending”.
- These two flags don’t have any relation with each other, and so, will work in isolation.

If Rotation is ON in the OGTBG:

- The first call will get routed using the “OG Trunk bundle Member1”. The trunk port from the OG Trunk Bundle member 1 will get selected using the rotation type programmed for the OG trunk bundle programmed as member 1.

- When there is a second call, it will get routed using the “OG trunk bundle Member 2”, and the trunk port from the OG trunk bundle member 2 will get selected according to the rotation type programmed in the OG Trunk Bundle programmed as member2.

- Accordingly, the member of the OGTBG will be accessible to the station user accessing the OGTBG in sequence.

If the Rotation is OFF in the OGTBG:

- The calls will always get routed from the “OG trunk Bundle member 1” if any trunk/channel in it is free. If all the trunks/channels of the “OG trunk bundle member 1” are busy, then the call will get routed using the “OG trunk bundle member 2”.

- Now the trunk/channel to route the call will get selected as per the rotation type programmed for the OG trunk bundle used as member 2. When the trunk ports of OG trunk bundle programmed in member 1 and member 2 all are busy, the OG trunk bundle member 3 will be used to route the call.

- Thus the Rotation flag of OGTBG will be used to select the OG trunk bundle member1 to member 8 as per call basis while the rotation type flag associated with the OG trunk bundle will decide the rotation mechanism to select the trunk port from the particular OG trunk bundle.

Default OG Trunk Bundle Group Table (Table-1):

Group No.	Rotation	LCR	OGTB Member 1	OGTB Member 2	OGTB Member 3	OGTB Member 4	OGTB Member 5	OGTB Member 6	OGTB Member 7	OGTB Member 8
01	√	None	00	00	00	00	00	00	00	00
02	√	None	00	00	00	00	00	00	00	00
03	√	None	00	00	00	00	00	00	00	00
04	√	None	00	00	00	00	00	00	00	00
05	√	None	00	00	00	00	00	00	00	00
::	√	None	00	00	00	00	00	00	00	00
21	√	None	00	00	00	00	00	00	00	00
22	√	None	00	00	00	00	00	00	00	00
23	√	None	00	00	00	00	00	00	00	00
24	√	None	00	00	00	00	00	00	00	00
25	√	None	00	00	00	00	00	00	00	00
26	√	None	00	00	00	00	00	00	00	00
27	√	None	00	00	00	00	00	00	00	00
28	√	None	00	00	00	00	00	00	00	00

29	√	None	00	00	00	00	00	00	00	00
30		None	00	00	00	00	00	00	00	00
31		None	00	00	00	00	00	00	00	00
32		None	00	00	00	00	00	00	00	00

Paging

What's this?

Paging allows you to make announcements to groups of extension users and to make public announcements over a public address system. You can deliver a message to a mass of people at once by just lifting the handset of your phone and dialing a code.

This feature is useful when you want to call several people at once; for example, to inform them about a meeting you have scheduled. If the persons you want to call have Digital Key Phones (DKP) or the Matrix Extended IP Phones, Radio devices or Standard SIP Phones as their extensions, you can use paging instead of calling them up one by one.

SARVAM UCS supports paging where announcements are made on DKP/SIP extensions /Radio Ports.

The extensions which are to be paged must be included in 'Page Zones'.

- *You can start paging **from** an SLT, a DKP or any SIP Extension. However, the paged extensions must be DKPs, Extended IP Phones, Radio device or Standard SIP Phones. The Standard SIP Phones on which you are paging must support Call-Info or Alert-Info header for Paging.*
- *When the Paging call is generated in SIP Extension having multiple call appearance and already a call is present on the SIP Extension then the SARVAM UCS will place the Paging call as normal call on the SIP Extension (as headers required in INVITE for paging call and intercom call are same).*
- *Paging is a one-way communication. As the mic of the paged extensions is muted during Paging, the users of the paged extensions cannot speak to the paging extension.*

How it works

The Pre-requisites

- Page Zones must be created. Each Page Zone accommodates up to 32 DKP/SIP extensions/Radio devices. You can create 12 different Page Zones of 32 DKP/SIP extensions/Radio devices.
- Paging must be enabled in the Class of Service allowed to the DKP/SIP/SLT/Radio extension from which this feature is to be used.

The Process

- A user of a DKP/SLT/SIP Extension having Paging in its Class of Service, dials the Access code for Paging and the Number of the Page Zone to which the user wants to make the announcement.
- The system activates the speakers of the DKP/SIP Extensions programmed in the Page Zone number. The system activates the speakers only of those DKP/SIP/Radio extensions in the Page Zone that are free.
- The calling DKP/SLT/SIP Extension user makes the announcement.

- All DKPs/SIP/Radio Extensions in the Page Zone can hear the announcement. But as the mic of their phones is muted, their speech will not be heard by the calling DKP/SLT/SIP extension user.
- To answer the Paging call the desired extension user must use Meet Me Paging while the Paging call is active. For details refer, “Meet Me Paging”.
- If no reply is received via Meet Me Paging, the calling DKP/SLT/SIP extension goes ON-Hook after the announcement.
- The system deactivates the speakers of the DKP/SIP/Radio Extensions.

How to configure

For this feature to work, you must create Page Zones and enable this feature in the Class of Service of the extensions which are to be allowed this feature.

Allowing Paging in Class of Service

In the default factory settings, Station Basic Feature Template Number 01 is assigned to all the stations of SARVAM UCS. Template 01 is assigned CoS group 01 which has Paging enabled. So, all stations of SARVAM UCS can page.

If you want to allow Paging to all stations, retain CoS group 01 in Template 01.

However, if Paging is to be disallowed to some stations then follow these steps:

a. Define a CoS group with Paging disallowed.

b. Prepare a Station Basic Feature Template with this CoS group applicable in all the “Time Zones”.

c. Assign this new Template to the stations to which Paging is to be disallowed.

Refer the topics “Class of Service (COS)” and “Station Basic Feature Template” for detailed instructions and programming.

Creating Page Zones

Decide how many Page Zones you want to create. You can create as many as 12 different Page Zones. You can include upto 32 DKP/SIP/Radio extensions.

For each Page Zone number, decide and assign the DKP/SIP/Radio extensions.

On a sheet of paper create a three-column table for each page zone, as shown below. Enter the Type of extension, whether DKP, SIP or Radio, and the Software Port number of the extensions you want to include in the page zone.

Page Zone 1

Member Number	Type of Extension	DKP/SIP Extension Port Number
1	DKP	002

Member Number	Type of Extension	DKP/SIP Extension Port Number
2	DKP	003
3	DKP	008
4	SIP	009
:		
:		
32		

Page Zone 2

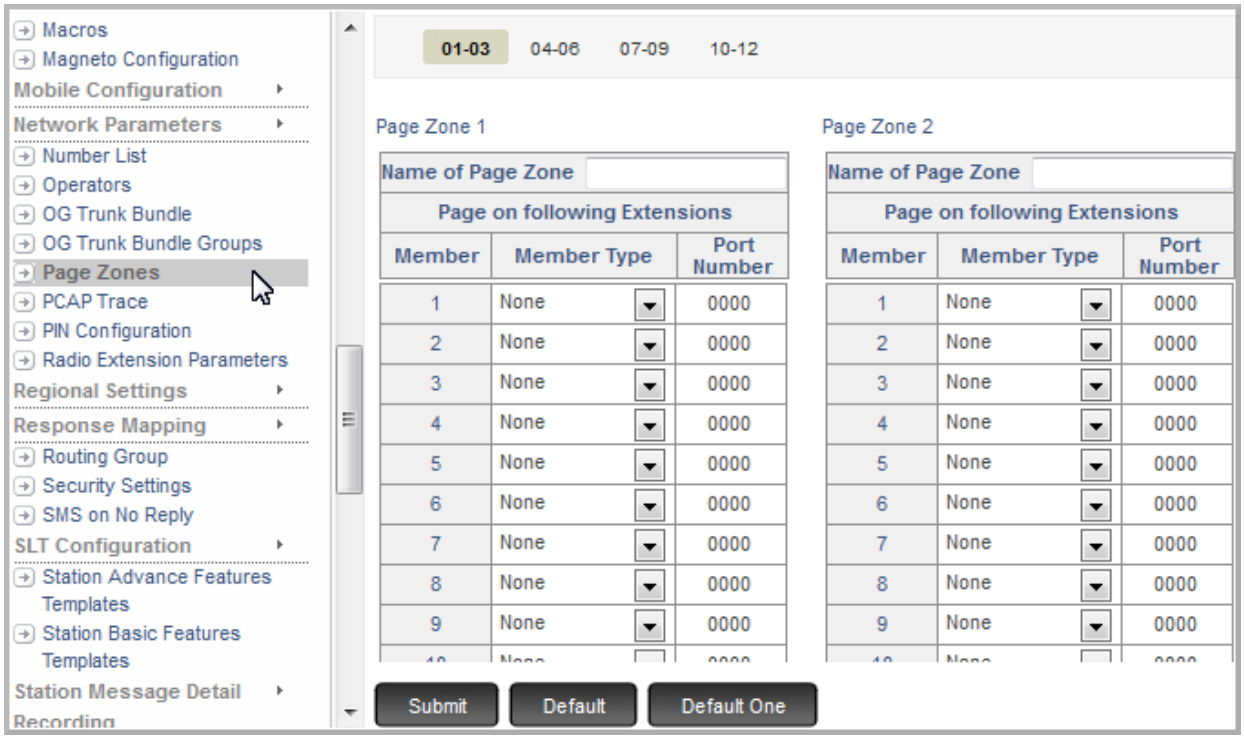
Member Number	Type of Extension	DKP/SIP Extension Port Number
1	SIP	007
2	SIP	010
3	SIP	012
4	DKP	013
:		
:		
32		

Now, program Page Zones using Jeeves or by dialing SE commands from a Telephone.

Programming Page Zones using Jeeves

- Login as System Engineer.
- Under **Configuration**, click **Page Zones** to open the page.
- The first three page zones appear on this page.
- Refer to the Page Zones you created on paper and program the following parameters in each Page Zone:
  - **Name of Page Zone:** Enter the name you want to assign to the Page Zone.
  - **Member Type:** Select the type of extension you want to include in the Page Zone: DKP, Radio or SIP extension.

- **Port Number:** Enter the number of the Software Port of the extension you selected in the Port Type.



- Click **Submit** at the bottom of the page to save your Page Zone settings.

- To go to other Page Zones, you may click the hyperlinked page zone numbers on the top of the Jeeves screen.

- If you have completed programming Page Zones, you may log out of Jeeves.

Programming Page Zones using a Telephone

- Enter SE mode from a DKP/SLT.

  To program a DKP/SIP Extension in a Page Zone, dial:
  - ***2302-1-Page Zone-Member Index- Port Type-Port Number*** to program a DKP/SIP extension in a single Page Zone.
  - ***2302-2-Page Zone-Member Index- Port Type-Port Number*** to program the same DKP/SIP Extension in a range of Page Zones.
  - ***2302-*-Member Index- Port Type-Port Number*** to program the same DKP/SIP Extension in all Page Zones.
    Where,
    Page Zone is from 01 to 12.
    Member Index is from 01 to 32.
    Port Type is 02 for DKP, 34 for SIP Extension, 40 for Radio Extension.
    Port Number is the software port number of the extension to be included in the Page Zone. Port Number is
    001 to 128 for DKP.
    001 to 999 for SIP Extensions.
    01 to 16 for Radio Extensions

  To include/exclude a member DKP, SIP or Radio Extension in/from a Page Zone, dial:

- ***2303-1-Page Zone-Member Index-Flag*** to include/exclude an extension in/from a single Page Zone.
- ***2303-2-Page Zone-Page Zone-Member Index-Flag*** to include/exclude the same extension in/from a range of Page Zones.
- ***2303-*-Member Index-Flag*** to include/exclude the same Extension in/from all Page Zones.
  Where,
  Page Zone is from 01 to 12.
  Member Index is from 01 to 32.
  Flag is
  0 for Exclude member extension from Page Zone.
  1 for Include member extension in Page Zone.

- Exit SE mode.

How to use

It is possible to page from a DKP, a SIP Phone and an SLT.

For EON and Extended IP Phone Users

- Press DSS Key assigned to Paging.
  OR
- Dial 1074
- Enter Page Zone Number on the prompt on your phone's display.
- You get the prompt on your phone's display: Start Paging <Page Zone Number>.
- Make your announcement.
- Go ON-Hook at the end of your announcement.

For SLT Users

- Lift the handset.
- You get dial tone.
- Dial 1074-Page Zone Number.
- Start your announcement.
- Replace the handset after completing your announcement.

Peer-to-Peer Calling

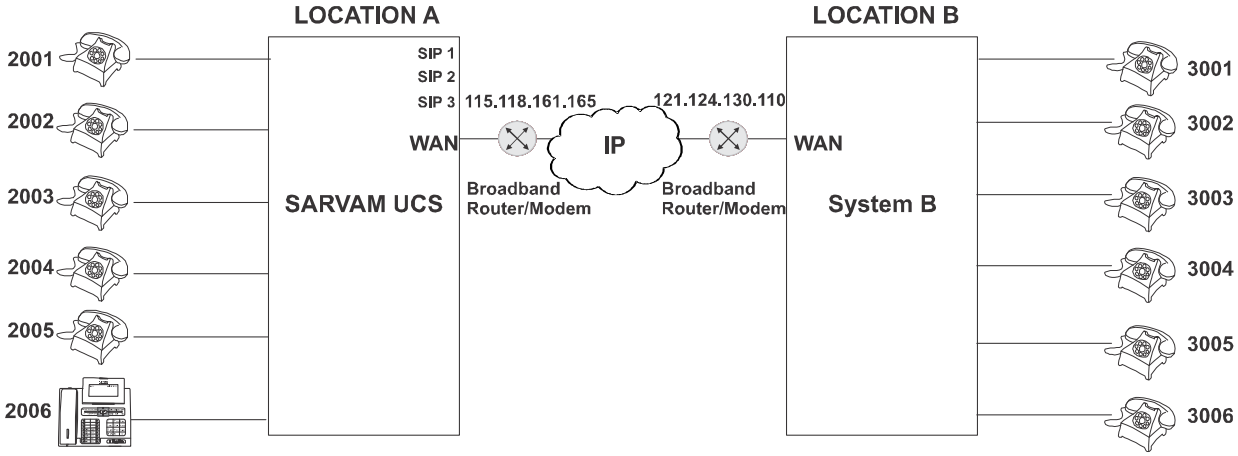
What's this?

Making an IP call without the intervention of a proxy server is called Peer-to-Peer Calling. As Peer-to-Peer calling does not require a proxy server, voice communication using this application can be done virtually free of cost. The major cost savings offered by this application makes it a very attractive mode of inter-branch or intra-office voice communication.

For dialing a number use '' in-place of '.'*

How it works

Let us understand how to use Peer-to-Peer Calling with the following illustration:



- Two offices are directly connected over the IP network.
- SARVAM UCS is installed at Location A.
- The System at Location B may also be SARVAM UCS.
- Peer-to-Peer calls can be made between the two locations with suitable configuration of SARVAM UCS and the System.
- **At Location A**, you need to do the following configuration in the SARVAM UCS:
  - Select a SIP Trunk to be used for this application and enable it. For example, SIP1.
  - Set the **SIP Trunk Mode** of this trunk to **Peer-to-Peer**.
  - Keep the **SIP ID** field of the SIP trunk blank.
  - Set the **Treat Incoming call as** option on the SIP trunk to **Station**.

  For detailed instruction, see "Configuring SIP Trunks".

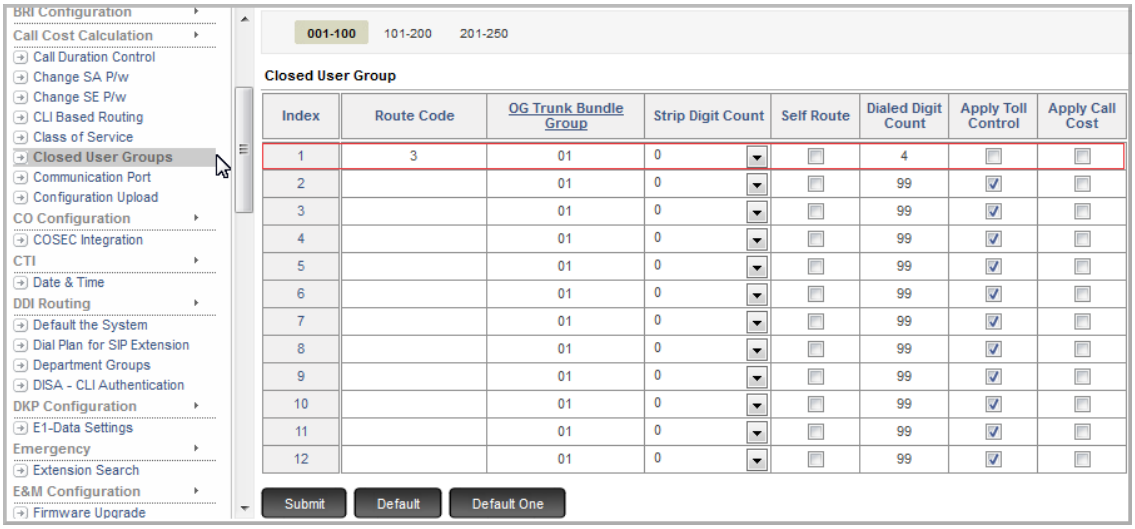
In the Router, you must configure the same SIP and RTP Ports as configured in the SARVAM UCS. In other words, you must configure Port Forwarding for SIP and RTP on the Router.

- You must also configure the **Trusted IP Address/es** table for this SIP Trunk to receive incoming calls. If you do not configure this table, all incoming calls on this SIP Trunk will be rejected. For instructions, see "Configuring SIP Trunks".

- Set the **Send CLI Option** on the SIP trunk as **Calling Party Wise**. See "Configuring SIP Trunks" for instructions.

- You may also configure the **Closed User Group (CUG) Table** to avoid dialing the Trunk Access Code for the outgoing calls made from this SIP Trunk, i.e. SIP 1.

  At Location A, you may configure the CUG table as follows:



  - In **Route Code**, enter the extension numbers of the System at Location B. Instead of entire number strings, you can configure a single digit, the starting digit of the extension numbers as Route Code. In this case, you may configure Route Code as '3', as all extensions at Location B start with '3'.

  - Keep the **Strip Digit Count** as '**0**'.

  - Keep the **Self Route** check box disabled.

  - In **Dialed Digit Count**, enter the digit length of the extension numbers at Location B. In this case, '**4**'.

  - In **OG Trunk Bundle Group**, select 01. Configure SIP Trunk1 as the only member in this group. The calls will be routed through this SIP Trunk only.

  - When Self Route check box is disabled, system will check **Apply Toll Control** parameter. By default, it is enabled. The system will apply toll control to all the outgoing calls.

    Disable this check box, if you do not want to apply toll control to the CUG numbers dialed by you.

  - By default, **Apply Call Cost** check box is enabled. Select this check box for Peer to Peer calls if you require the call cost calculation.

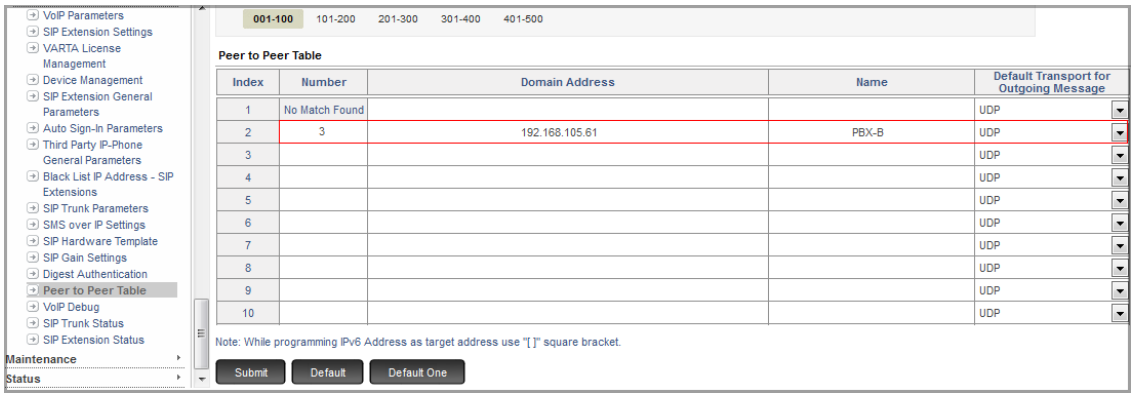
For detailed instructions, see “Closed User Group (CUG)”.

If the System’s at both the offices have the same extension numbers, the users need to dial the Exchange ID of the System along with the extension numbers. For detailed instructions, see “Closed User Group-With Exchange ID”.

- Now, configure the **Peer-to-Peer Table**.

  The Peer-to-Peer table stores up to 500 entries. Each entry consists of the parameters Number, Domain Address, Name and Default Transport for Outgoing Message.

  At location A, you would have to configure the Peer-to-Peer table as follows:



  - In **Number**, enter the digit you configured as **Route Code** in the CUG Table for calling the extension numbers of the System at Location B. In this case ‘3’.
  - In **Domain Address**, enter the **IP Address** of the WAN Port of the Router at Location B.
  - In **Name**, enter ‘Location B’ for identification.
  - Keep **Default Transport for Outgoing Message** as ‘UDP’.

- At Location B, you may do a suitable configuration of the System.
- When an extension user 2001 of SARVAM UCS at Location A dials 3001, an extension number of System B, it checks the CUG table to match the dialed digits with the Route Code and the Dialed Digit Count. As a match is found, it selects the SIP trunk defined for routing the Route Code, i.e. SIP 1.
- As SIP 1 is set to Peer-to-Peer mode, the system checks the Peer-to-Peer Table configured. It finds a match for the digit ‘3’ and will route the call to the IP Address configured for this number. In this case, to the IP Address of the Router at Location B (121.124.130.110).
- Further, the Router will forward the call to System B. With suitable configuration done in System B, the call will be routed to the desired destination i.e. extension 3001.

Configuring the Peer-to-Peer table using Jeeves

Peer to Peer Table is used, for deciding the destination IP Address for routing calls using non-proxy SIP trunks (Registrar server address shall be programmed as blank for these trunks). The Peer-to-Peer table stores upto 500 entries. Both IPv4 and IPv6 addresses are supported. This table is common for all the SIP trunks.

For instructions on configuring the SIP Trunk for the Peer-to-Peer application—SIP Trunk Mode, SIP ID, Treat incoming calls as Station, Trusted IP Address/es table—see “Configuring SIP Trunks”.

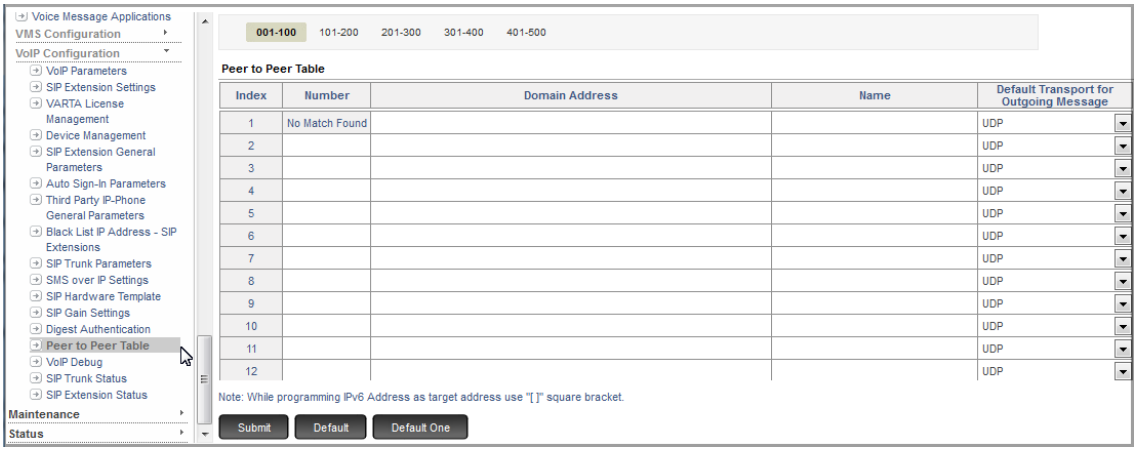
For instructions on configuring the CUG Table, see “Closed User Group (CUG)”.

To configure the Peer-to-Peer table using Jeeves,

- Log in as System Engineer.

.

- Under **VoIP Configuration**, click **Peer-to-Peer Table**. The Peer-to-Peer table opens.



The first entry is reserved for No Match Found.

- In **Number**, enter the peer-to-peer number string—prefix or entire number—that will be dialed. The number string must not exceed 8 digits. Default: Blank.

- In **Domain Address,** enter the domain name or IP Address to where the call is to be placed. Both IPv4 and IPv6 addresses are supported. The Domain Address may consist of up to 48 characters (maximum). Default: Blank.

  The Destination Address can also be in the form of Address: Port number.

- In **Name**, enter a name to identify the number string you configured. It may be the name of your contact or any name you wish to assign to the number string. The name may consist of 24 characters (maximum). Default: Blank.
  The name you configure here will not be used in SIP signaling.

- In the **Default for Outgoing Message** field, select the option for transporting outgoing SIP messages. You can select UDP, TCP or TLS.

The parameter Default for Outgoing Message can be configured through Jeeves only.

- Click **Submit** to save your entries.

Configure using Telephone

Enter the SE mode.

For programming parameters using SLT/DKP use '#9' in-place of '.'

For dialing a number use '' in-place of '.'*

To configure a Number, dial:

- **7801-1-Index-Number String-#*** to configure the number for a single index.
- **7801-2-Index-Index-Number String-#*** to configure the numbers for a range of indexes.
- **7801-*-Number String-#*** to configure the numbers for all the indexes.
  Where,
  Index is from 001-500.
  Number String can be a complete telephone number, truncated telephone number or an area code.
  Number string is of maximum 8 digits with (0-9, #. *).
  By default, Blank for all indices.

To clear a Number, dial:

- **7801-1-Index-#*** to clear the number for a single index.
- **7801-2-Index-Index-#*** to clear the numbers for a range of indexes.
- **7801-*-#*** to clear the numbers for all the indexes.

To configure the Destination Address, dial:

- **7802-1-Index-Destination Address-#*** to configure the Destination Address for a single index.
- **7802-2-Index-Index-Destination Address-#*** to configure the Destination Address for a range of indexes.
- **7802-*-Destination Address-#*** to configure the Destination Address for all the indexes.
  Where,
  Index is from 001-500.
  Destination Address is of maximum 40 char. with ASCII.
  By default, Blank for all indices.

IPv6 address can be configured using Jeeves only.

To clear the destination address, dial:

- **7802-1-Index-#*** to clear the Destination Address for a single index.
- **7802-2-Index-Index-#*** to clear the Destination Address for a range of indexes.
- **7802-*-#*** to clear the Destination Address for all the indexes.

To configure the Name, dial:

- **7803-1-Index-Name-#*** to configure the Name for a single index.
- **7803-2-Index-Index-Name-#*** to configure the Name for a range of indexes.
- **7803-*-Name-#*** to configure the Name for all the indexes.
  Where,
  Index is from 001-500.
  Name is a string of alphanumeric characters of maximum 24 characters.
  Use first 25 keys and the redial key to enter the name. Terminate with #*.
  By default, Blank for all indices.

  Display Name field is not sent in the SIP message. It is just a tag for the entry.

To clear the Name, dial:

- **7803-1-Index-#*** to clear the Name for a single index.
- **7803-2-Index-Index-#*** to clear the Name for a range of indexes.
- **7803-*-#*** to clear the Name for all the indexes.

PIN Dialing

PIN is a unique four digit code with an associated Class of Service and Toll Control, which can be assigned to the extension user.

PIN Dialing allows an extension user to make outgoing calls from any extension according to the toll control assigned to his/her PIN. PIN Dialing must be enabled in the Class of Service of the extension from where the outgoing call using PIN is to be made. SARVAM UCS supports maximum 500 PINs for each type of toll control.

Calls made using PIN can be logged in the SARVAM UCS and you can print the report online or later as and when required.

How it works

Let us understand how this feature works with the help of an example.

User A is denied dialing of International Numbers from his/her extension but has PIN Dialing enabled in his/her CoS. A is assigned a PIN 1234 with toll control level as international call. Now, User A can make international calls in two ways:

1. User A must dial the feature access code to access PIN Dialing i.e. *2 (programmable, see "Access Codes") followed by his/her PIN. As soon as SARVAM UCS detects a valid feature access code and PIN, it gives trunk dial tone to A. Now user A can dial the desired international number and talk.

2. User A can directly dial an international number from any extension that is assigned a "Station Basic Feature Template" with PIN Dialing enabled in the Class of Service and LCR enabled in the OGTB group.

   SARVAM UCS will prompt A to dial his/her PIN either by playing feature tone or a voice message. Voice message will be played, if recorded. See "Voice Message Applications" for more details. As soon as s/he dials a valid PIN, SARVAM UCS will dial out the number and connect A to the called party.

   This method of calling is useful while making calls through Standard SIP Phones, as it prevents the PIN from being stored in the call logs of the phone.

- *You cannot Redial or set Auto Redial for the numbers dialed out using PIN.*
- *While dialing the Access Code and PIN from the desired extension, the **Extension - Inter Digit Wait Timer (sec)** will be applicable. See "System Timers and Counts" for more details.*

How to configure

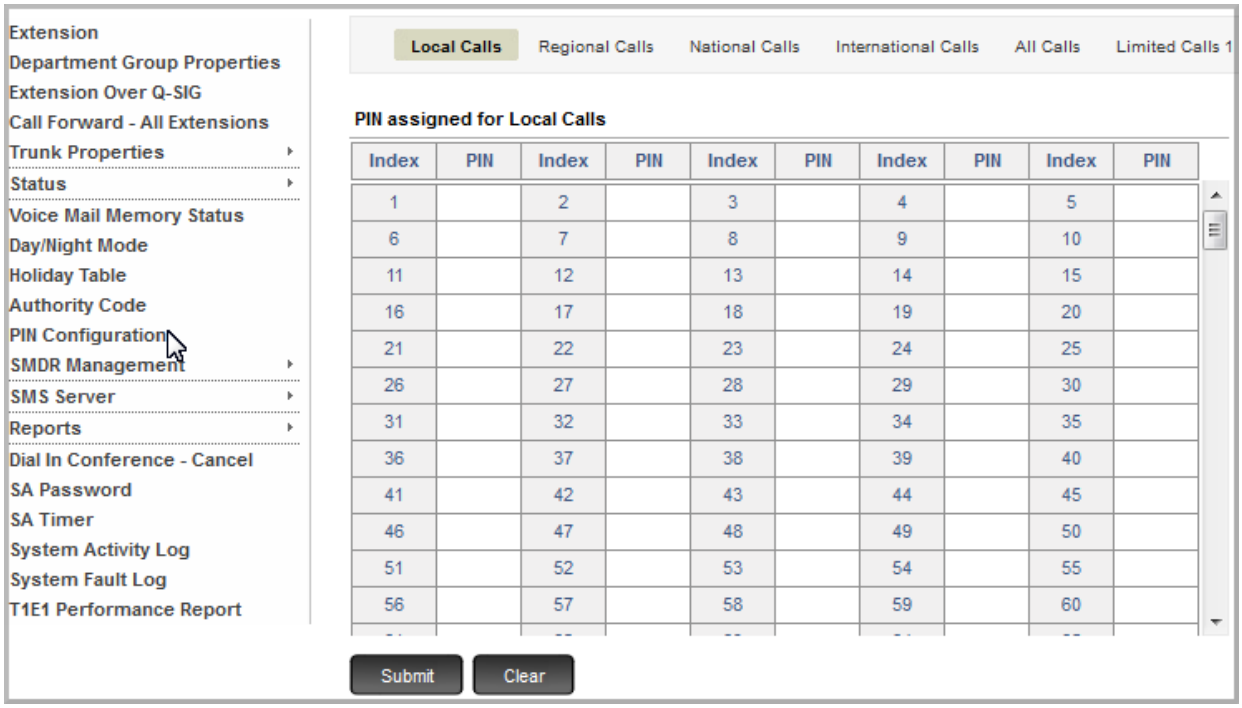
To be able to use this feature,

- Make sure, PIN Dialing is enabled in the "Class of Service (COS)" of the Extensions from where you want to make outgoing calls using PIN.
- Enable the parameter *Store Outgoing Calls* in the "Station Basic Feature Template" assigned to the extensions from where you want to make outgoing calls using PIN.
- Assign LCR enabled OGTB Group in the "Station Basic Feature Template" of the Extensions from where you want to make outgoing calls using PIN.

- Configure the PIN table for each type of Toll Control level from the SA Mode. See “Configuring PIN Table”.
- Select the OGTB Group and type of LCR for making outgoing calls using PINs, for each Toll Control Level from the SE Mode. See “Configuring OGTB Group and LCR”.

Configuring PIN Table

- Make a list of users to whom you want to assign PINs.
- Log in as **System Administrator**.
- Click **PIN Configuration**.

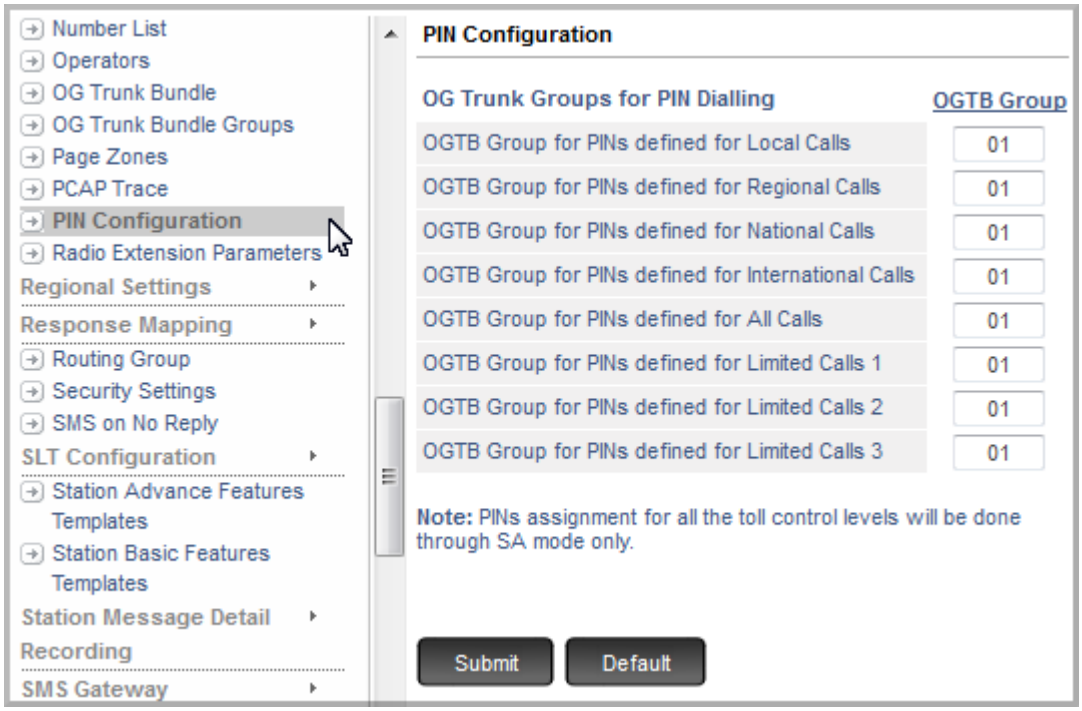


- Select the desired Toll Control level. Each Toll Control level includes the Allowed and Denied numbers as configured by SE. See “Toll Control” for more details.
- Configure the PINs at a desired index’s. The PIN must be of 4 digits. Valid digits are 0 to 9.

- *Each PIN must be unique. You cannot assign same PIN to two different index within the same Toll Control level.*
- *If you re-assign the same PIN to a different Toll Control Level, it will not be valid for the previously assigned Toll Control Level.*
- *To avoid unauthorized access, we recommend you to change the PIN regularly. Make sure it is strong and is kept confidential.*

Configuring OGTB Group and LCR

- Log in as **System Engineer**.
- Under Configuration, click **PIN Configuration**.



- In **OG Trunk Groups for PIN Dialling**, enter the desired **OGTB Group** number for each Toll Control level.
- To select the desired trunks in the OGTB Group and the type of LCR, see “OG Trunk Bundle Group” for instructions.
- To configure the LCR Table, see “Configuring LCR” for instructions.

How to use

For EON Users and Extended IP Phone Users

To use PIN Dialing from any extension:

- Go OFF-Hook.
- Press DSS Key assigned to ‘PIN Dialing’.
  OR
- Dial ***2**
- Dial the PIN.
- You will hear the trunk dial tone.
- Dial the desired number.

OR

- Go OFF-Hook.
- Dial TAC.
- Dial the desired number.
- You get Feature tone/ Voice message.

- Dial the PIN.
- Speech with the Called Party.

For SLT Users

- Go OFF-Hook.
- Dial ***2**
- Dial the PIN.
- You will hear the trunk dial tone.
- Dial the desired number.

OR

- Go OFF-Hook.
- Dial TAC.
- Dial the desired number.
- You get Feature tone/ Voice message.
- Dial the PIN.
- Speech with the Called Party.

Printing Reports of Outgoing Calls made using PIN

You can print call details of users who made outgoing calls using the PIN. For this, you will need to:

- enable **Store Outgoing Calls** in the Station Basic Feature Template of the extension from where you want to make outgoing calls using PIN.
- enable **Print Calls made using PIN** and set the PIN range in **Calls made using PIN** under Calls filter in Outgoing Call Report.
- configure the **destination port** for SMDR-Outgoing Call Report.

Refer “Station Message Detail Recording-Report”, for detailed instructions on printing reports using filters.

You may also print online report of calls made using PIN, when the SARVAM UCS is interfaced with a third party Call Accounting Software (CAS). For this, you must set the parameter PIN in SMDR - Posting. See “Station Message Detail Recording-Posting” for more details.

PLCC-An Introduction

What's this?

SARVAM UCS-PLCC Server is a digital System. It uses a digital switch and hence is a 100% non-blocking system. In PLCC network, number of PLCC Servers need to be connected.

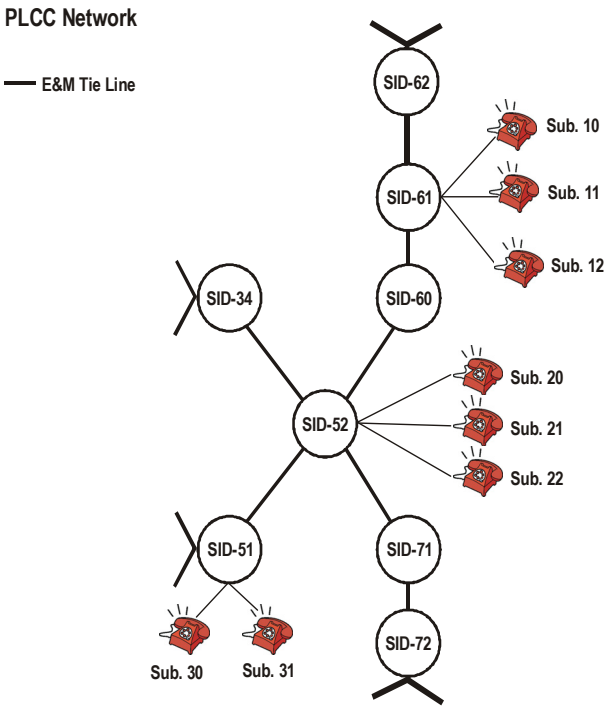
Refer to diagram showing cluster of exchanges in a PLCC network. Each exchange in PLCC network is assigned with Exchange Identity (SID) in order to get identified by other exchanges in the network. Thus each exchange is identified by Exchange Identity (SID). As shown, an exchange is connected to the other exchange through an E&M tie line. Also, it is possible that an exchange may not be directly connected to all other exchanges in a PLCC network through E&M tie lines.

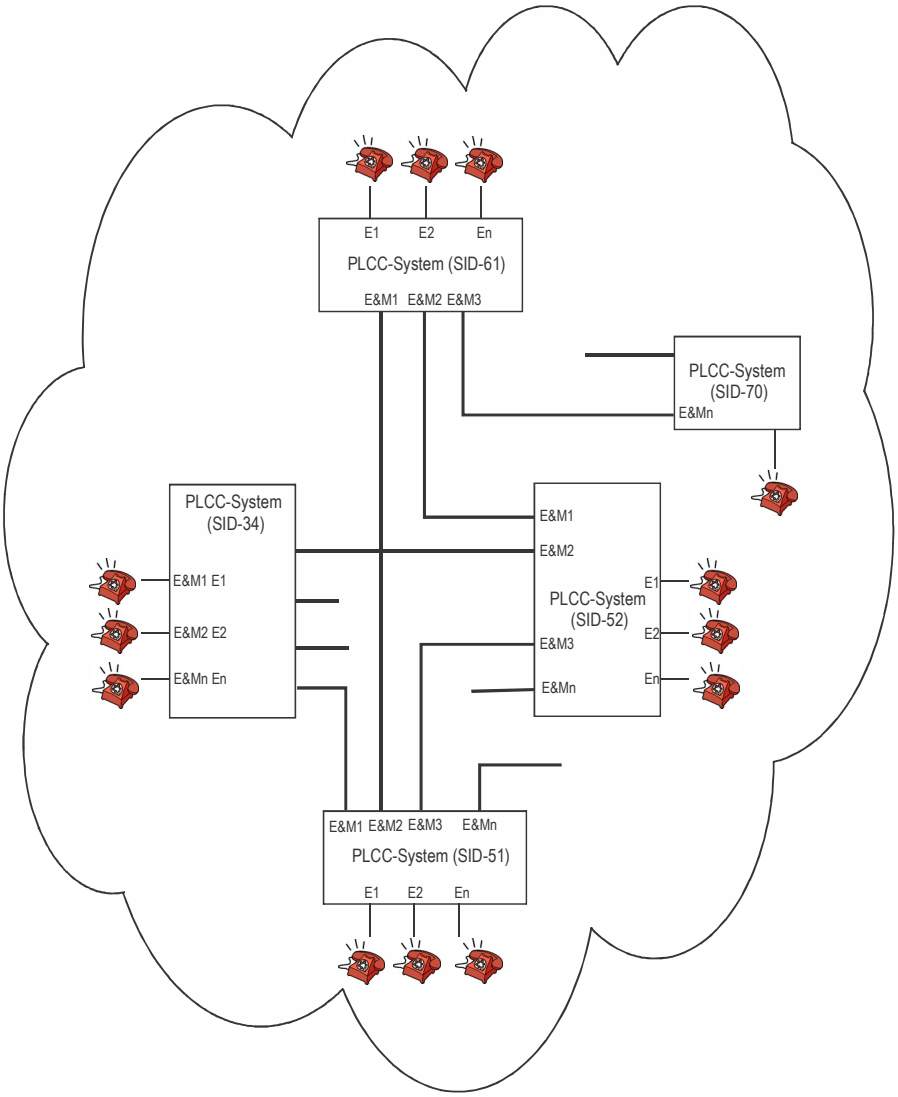
For example, SID-52 exchange is connected to SID-51, SID-34, SID-71 and SID-60 exchanges through direct E&M tie lines. However, SID-52 exchange is not directly connected to SID-61, SID-62 and SID-72 exchanges through E&M tie lines.

Following example shows how a call is established.

- Consider, an example where subscriber 20 of SID-52 needs to call subscriber 30 of SID-51. After dialing trunk access code, dial 51 (SID number of the exchange) and then dial 30 (subscriber number of SID-51). This is how a call is completed, as both exchanges; SID-51 and 52 are connected through a direct E&M tie line.

- However, consider another example, where subscriber 20 of SID-52 needs to call subscriber 10 of SID-61. Even though, here both exchanges are not connected through direct E&M tie line, it is possible by dialing trunk access code with SID number of the exchange (here 61) followed by subscriber number (here 10). However, the call will proceed through SID-60 and then reach SID-61. This is called Transit Call. Here, the subscriber will not be able to know that call has proceeded to the required exchange through transit facility.





PLCC-Priority

All callers do not have same hierarchical position in an organization. ‘Priority’ can be assigned to each caller to make sure precedence is given to certain extensions over others in being answered.

The SARVAM UCS supports flexible priority assignment for different users. Each port can be assigned a priority level between “1” to “9”. Higher the priority level, more important the caller is. Accordingly 1 has the least priority and 9 has the highest priority.

Using this feature the users can free the system resources (Trunk/Extension).

The PLCC functional module requires a license. You must purchase a license to activate CCS Signaling when End Point and Transit, Express Signaling, and Seizure Pulse and Release Pulse Signaling. Refer the topic “License Management”.

How it works

Using Priority on a Station:

- Station A and Station B are talking to each other.
- Station C calls Station B and finds it to be busy. If s/he uses Priority, s/he gets connected to Station B.
- Station A gets disconnected and gets error tone. However, for this to happen the priority of Station C should be higher than that of Station B and Station A.

Using Priority on a Trunk:

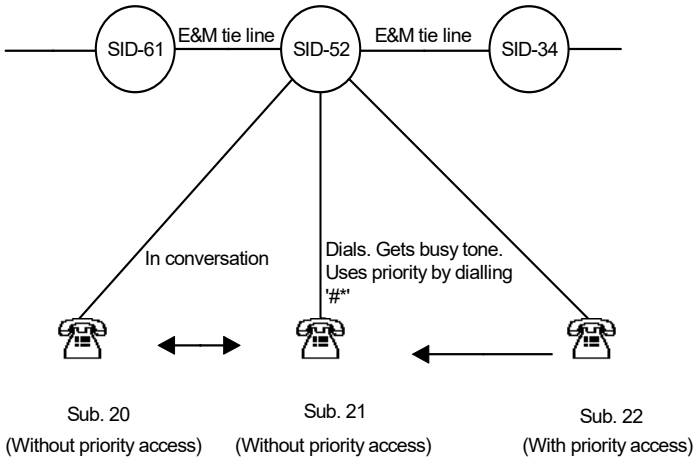
- Station A is talking to an external party through trunk 1.
- Station B tries to grab trunk 1. As the trunk is busy, s/he uses Priority. Doing so, s/he gets connected to trunk 1 and gets PSNT dial tone.
- Station A gets disconnected and gets error tone. However, for this to happen priority of Station B should be higher than Station A. In this case, priority of the trunk is not considered.
- Similarly, if station A is talking to an external party through trunk 1. Station B calls station A and finds it busy. S/he uses Priority. Doing so, station B gets dial tone whereas, the trunk 1 gets disconnected.

How to use

- Lift the handset. You hear the Dial tone.
- Dial Station/Trunk Access Code. You get Busy tone.
- Dial ***#*.*** The called station/trunk gets disconnected. You get dial tone after the confirmation tone.
- Dial ***Station/External Number.*** *You get* Ring Back tone.

Priority-Within the Exchange

Under this facility any subscriber having priority access can enter in to conversation with two conversing subscribers within the exchange.

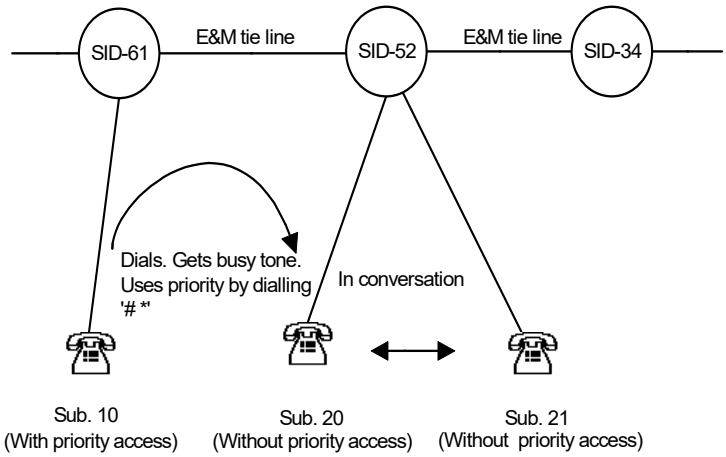


How to use

Please refer above diagram. If any two subscribers, say 20 and 21 of an exchange with SID-52 (without priority access) are in conversation and a third subscriber, say 22 (of same exchange) has priority access and wants to talk with subscriber 21. But, when subscriber 22 dials for subscriber 21, it gets busy tone. Subscriber 22 can use its priority by dialing code '#*' in such case and can come in conference with subscribers 20 and 21.

Priority-In Other Exchange

Under this facility any subscriber having priority access can enter in to conversation with two conversing subscribers in two different exchange.

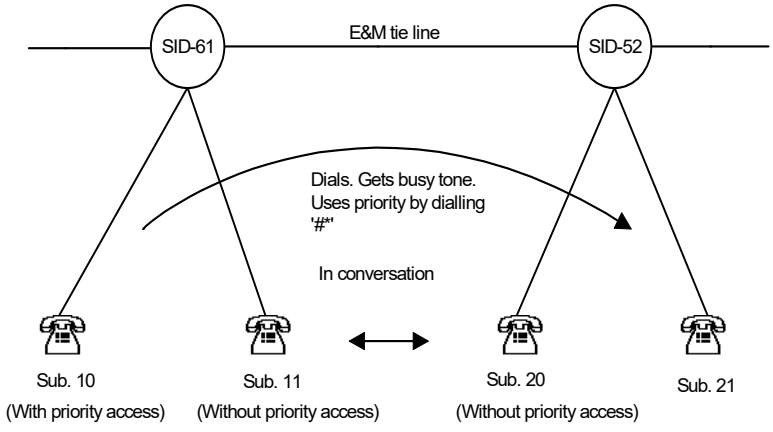


How to use

Please refer above diagram. If any two subscribers, say 20 and 21 of an exchange with SID-52 are in conversation and a subscriber, say 10 of an exchange with SID-61 (from the same network with priority access) want to talk with subscriber 20 of SID-52. After dialing the required subscriber 20 of SID-52, he gets busy tone. Subscriber 10 of SID-61 can now press **'#*'** and can enter into conference with both subscriber 20 and 21 of SID-52.

Priority-Over Busy E&M Tie Line

Under this facility any subscriber having priority access can use priority to free a busy E&M tie line.

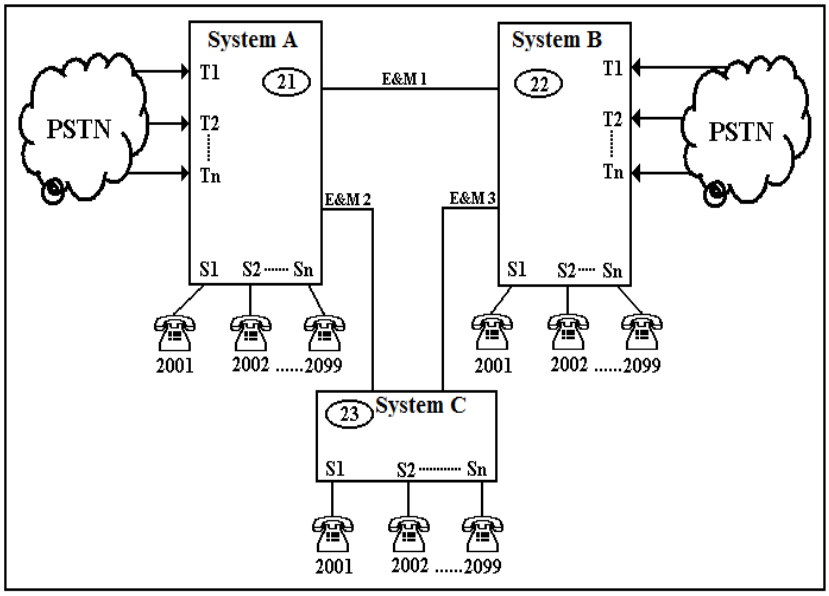


How to use

Please refer diagram, if two subscribers, say 20 of SID-52 (without priority access) and 11 of SID-61 (without priority access) are in conversation on an E&M tie line. Both the exchanges SID-52 and SID-61 are connected with each other through a single E&M tie line. Now, if subscriber, say 10 (with priority access) of SID-61 wants to talk with subscriber 21 of SID-52, then he will get busy tone. Subscriber 10 of SID-61 can now use its priority by pressing **'#*'** on his telephone and can get into conference with both subscriber 11 of SID-61 and subscriber 20 of SID-52. After termination of the call by subscribers 11 and 20, subscriber 10 gets dial tone and can make a call using the E&M tie line to subscriber 21 of SID-52.

PLCC-Routing Table

In PLCC network, number of PLCC-Servers need to be connected. The entire network should behave as a single unit or one group. It is not feasible to have unique station numbers throughout the network. In such cases, an Exchange ID is assigned to the Servers and a routing table is programmed in the exchange. In case of newly added exchange in the network, the routing tables of other exchanges need to be modified.



In the above figure, 3 systems are connected through E&M connectivity.

- T1 to Tn are trunk lines from the local central office (CO).
- S1 to Sn are stations.
- E&M1 to E&M3 are E&M lines between the three Servers.

How it works

In this application, it is possible to have same station number in two or more Servers of the network. Few new words have been used to explain PLCC routing table, each of these words have been explained below:

OG Call with Trunk Access Code:

In PLCC applications, SLT numbers are duplicated. Internal and External dialing is differentiated. Internal dialing happens by dialing SLT numbers directly. External dialing starts with dialing the trunk access code (usually '0') followed by SID and remote SLT number. Suppose the exchange ID of System-A is 21 and that of System-B is 22. Both Systems have same station numbers, say 2001,2002, etc. Suppose a station of System-A wants to call 2001

of System-B. He has two options. He can dial 222001 or 0222001 i.e one with route code 22 and other with route code 022. Hence we need to program two entries in the routing table, one with route code 22 and other with route code 022.

The SARVAM UCS has only one routing table. The same table is used for Closed User Group and Closed User Group-With Exchange ID. Hence the table has to be programmed keeping the application in mind.

- **Routing Table:** This table has following parameters viz. Route Index, Route Code, OG Trunk Bundle Group, Strip Digit Count, Self Route check box, Apply Toll Control check box and Apply Call Cost check box. The Closed User Group-With Exchange ID programming works according to this table.

Index	Route Code	OGTBG	Strip Digit Count	Self Route	Dialed Digit Count	Apply Toll Control	Apply Call Cost
001							
:							
250							

- **Index:** Maximum 250 different routes (001-250) can be programmed.

- **Route Code:** Route code could be of maximum sixteen digits. Valid Digits: 0 to 9, * # A B C D F P, where P is Pause, F is Flash, A to D is DTMF Digits. Generally, route code will be a unique number. The route code should not clash with any of the extension numbers of same System. For example in the figure given above, route code for System-A can be defined as ‘21’, route code for System-B can be defined as ‘22’ and that for System-C can be defined as ‘23’. This means that no extension in System-A can start with ‘22’ or ‘23’. Similarly, no extension in System-B can start with ‘21’ or ‘23’ and no extension in System-C start with ‘21’ and ‘22’.

- **OG Trunk Bundle Group:** An OG Trunk Bundle Group (OGTBG) is assigned to each route code. Whenever a call is to be made on that route, a free trunk from the OGTBG is selected and the extension number is dialed on it. The same logic of rotation On/Off for trunk selection from the OGTBG is used. If rotation is OFF then always the first trunk in the OGTBG is selected. If it is busy then the next trunk in the group is selected. This helps to select an alternate route. Whereas if rotation is ON then the trunks in the OGTBG are selected in round robin fashion.

- **Strip Digit Count:** This count signifies the number of digits to be stripped off while dialing/decoding a number. To elaborate: Consider figure 1. The requirement is that if extension 2001 of System-B dials 212002 and if E&M 1 is busy then the call should reach extension 2002 of System-A through alternate route. In this case the strip digit count of System-A should be programmed as 2 and that of System-B and System-C should be programmed as 0. Doing so, when extension 2001 of System-B dials 212002 and if E&M1 is busy then the call is routed through System-C. In this case, System-B dials 212002 on E&M3, System-C receive this code and dials out the same code, that is, 212002 on E&M2 without striping of any digit. On receiving 212002, System-A strips of two digits as per the programming and routes the call to extension 2002.

- **Self-Route:** This flag signifies that the digits being dialed are for the same System and are not to be dialed on the E&M trunk.

- **Dialed Digit Count:** When digits are dialed on the trunk, the system waits for inter digit timer after the last digit is dialed. In order to avoid this timer and number of digits dialed to be routed without further delay, count for the number of digits to be programmed in this field. If the number of digits received are equal to the parameters programmed then the number is dialed out immediately without waiting for the inter digit

timer. If the number of digits dialed by the user are not equal to the digits programmed, the number is dialed after inter digit timer.

- **Apply Toll Control:** This parameter is not relevant as Self Route flag is enabled. This parameter is relevant when you are configuring “Closed User Group (CUG)”.

- **Apply Call Cost:** By default, this check box is enabled and the system will calculate the cost of each call.

  For certain calls (internal calls) you do not require the call cost calculation, clear the check box corresponding to these entries. You can also set the filter **Calls with units more than** to generate a report according to the Call Cost. For details, see “Station Message Detail Recording-Report”.

How to configure

For more details on above steps, please refer topic “Closed User Group (CUG)” and “Closed User Group-With Exchange ID”.

PLCC-Express Line Network

SARVAM UCS-PLCC Express Line Communication Server is a digital system. It uses a digital switch and hence is a 100% non-blocking system. Refer to diagram in ‘Introduction to PLCC Network’ topic showing cluster of substations in a PLCC network. A substation in a PLCC network is connected to any other substation through a dedicated PLCC express line (that is, 4/6 wire E&M tie line) as shown in the diagram.

How to configure

PLCC Express Line Communication Server is one of the applications of the SARVAM UCS. We recommend the user to read relevant topics before programming it for PLCC Express Line application.

Step 1
Refer chapter “E&M Feature Template” to program the feature in an E&M Feature Template.

Step 2
Refer chapter “E&M Feature Template” to assign default values to an E&M Feature Template.

Step 3
Refer chapter “E&M Feature Template” to assign an E&M Feature Template to an E&M.

Step 4
Refer chapter “Configuring DKP Extensions” to program a name for E&M trunk.

Step 5
Hardware ID is an attribute of a software port. Hardware ID of a software port decides where the port is physically located. To derive hardware ID of a software port, we need slot number and port number of the card. Hence, all the programming is done for the software port and not for the hardware ID. Accordingly, the software port number is used for all the programming. Please refer “Software Port and Hardware ID” for more details, to assign hardware ID to an E&M software port:

Step 6
Refer chapter “Station Advanced Feature Template” to assign a Station Advanced Feature Template to an E&M.

Step 7
Refer chapter “Station Advanced Feature Template” to assign a Station Basic Feature Template to an E&M.

Step 8
Please refer “Time Tables” chapter to program the time zone of a trunk.

Step 9
To program a feature in a Trunk Feature Template, please refer “Trunk Feature Template” for more details.

Step 10
To assign a Trunk Feature Template to an E&M, please refer “Configuring the E&M using Jeeves” topic.

Step 11
Refer chapter “DSS Keys Programming” to assign a function to a key of DSS-64 assigned to a DKP. In this commands function type and function number can be programmed as explained below.

Function:
A function should be assigned to these Keys, which they should perform. There are 2 different types of functions, which can be assigned to a key for PLCC express line application. Each function that can be assigned to a key is given unique number. Following table list the function types available:

Function Type	Meaning	Function Number/Port
00	Null	--
06	Access a trunk	001 to 128

Step 12
Refer chapter “DSS Keys Programming” to assign a function to a DKP key. In this command refer above explanation to program ‘Function Type’.

Step 13
Define dialing property of E&M port. The dialing properties are based on the applications. If the dialing type for a E&M is programmed as pulse type then Pulse-Dialing ratio should be defined. The E&M support six different Pulse Dialing Ratio’s.

Value	Meaning/Ratio
1	10PPS, 1:2
2	10PPS, 2:3
3	10PPS, 1:1
4	20PPS, 1:2
5	20PPS, 2:3
6	20PPS, 1:1

For assigning Dial Type and Pulse Dial Ratio, please refer “E&M Feature Template” topic.

Presence

What's this?

Presence is an important UC feature as it helps you to know the availability status of other users. Depending on the status of users you can decide whether to initiate a conversation or find an alternative way to contact the desired user.

For example, an extension user may want to leave his desk for an indefinite period, but does not want to use Call Forward or set Do Not Disturb. He wants to indicate to callers about his absence. Similarly, extension users who are present at their desk may want to hide their presence from other users; or they may want to show their current activity to the other extension users like they are Busy, or are away from their desks, or on the phone with someone on another call, etc.

With the Presence feature of SARVAM UCS, extension users, including the Operator, can 'publish' their presence to callers from other extensions. By doing so, they can indicate to the other extensions about their availability.

In the same way, the Presence feature allows extension users to view the 'Presence' status (availability) of the extensions that they want to call, before making the call or when their call is not answered.

How it works

Publishing Presence

Any SLT, DKP, ISDN Terminal, and SIP Extension User can 'publish' their presence by setting any of the messages listed in the following on their phone, by dialing the access code for this feature.

SIP Extension users who want to publish their presence have two options:

- *Using the PUBLISH feature supported by the SIP Client.*
- *Using the feature access code for Publish Presence supported by SARVAM UCS.*

The first option requires the parameter 'PUBLISH' to be enabled in the SIP Extension Settings. Refer "Configuring SIP Extensions". By default, this parameter is disabled.

Publishing Presence Messages

1. **Absent:** When an extension user sets 'Absent' as the message, all incoming internal as well as external calls will be blocked from landing on his/her extension.

   When any other DKP/SIP extension user calls this extension, the text message 'User Absent' will appear on the caller's phone display.

   If the extension phone that has set 'Absent' is a DKP/SIP, the letter 'A' appears on the phone's display to indicate absence.

   The letter 'A' disappears when the extension user sets a presence message other than 'Absent'.

- *When an SLT extension user calls the extension which has set 'Absent', an error tone will be played. However, it is possible for the SLT extension user to find out the presence status of the called extension. Refer "Viewing Presence" later in this topic.*
- *External callers who call the extension, on which 'Absent' is set, will get an error tone only.*
- *Outgoing calls can be made from the extension which has set 'Absent'. Only incoming calls are restricted.*
- *If more than one extension is configured as "Operator" (routing group), incoming calls will be blocked only on the Operator extension which has set User Absent.*

2. **Present:**

   When an extension user sets 'Present', all incoming calls will be received as normal on this extension.

   If previously set as 'Absent', when a DKP/SIP extension user sets 'Present' the letter 'A' will disappear from the phone's display.

   When any other DKP/SIP extension user calls this extension, the name of the extension user will be displayed on the caller's phone display, when the called extension is ringing.

3. **Auto Detect:** When an extension user sets 'Auto Detect', the system will detect the state of the phone; depending on the call state, it will publish the presence message to the other extensions. Three types Publish Presence messages are possible, with Auto Detect:

   a. ***Idle:*** When the system detects the extension phone to be ON-Hook, it indicates the status of the phone to other extensions 'idle'.

   b. ***On the Phone:*** When the system detects the phone to be OFF-Hook, or in speech with another party or if it detects an incoming call placed on the phone, it will indicate to the other extensions that this extension user is 'On the Phone' with another party.

   c. ***DND Text message:*** When the system detects that the extension phone has Do Not Disturb (DND) set on it with a DND Text message, it will display to the calling extension, the DND message set by the called party (this may be the default DND message or the DND Text message set by the called extension).

3. **Away:** When an extension user sets 'Away', the system will display this message to the other extensions.

4. **On the Phone:** When an extension user sets 'On the Phone', the system will display this message to the other extensions.

5. **Do Not Disturb:** The extension user can set this message to be published to other extensions, if s/he wants to work uninterrupted.

   Unlike the DND Feature, the extension user who has set this message will continue to receive calls both internal as well as external calls, as the system considers this extension as 'present'.

6. **I am Mobile:** The extension user can set this message to be displayed to other extensions, when s/he is not at the desk.

7. **In Meeting:** The extension user can set this message to be displayed to the callers, if s/he is busy in a discussion or meeting.

8. **Out for Meal:** The extension user can set this message to be displayed to other extensions when going on a lunch break.

9. **Out of Office:** The extension user can set this message to be displayed to the callers when s/he leaves the office temporarily.

When an extension user sets any Publish Presence message other than 'Absent', the system will consider the user as 'Present'. All incoming and outgoing calls will be allowed on this extension.

It is possible to program another message in place of Publish Presence messages listed from 6 to 9: I am Mobile, In a Meeting, Out for Meal, Out of Office.

Publish Presence messages can be set or changed for any extension from the System Administrator (SA) mode.

Viewing Presence

- Extension users can know the status of another extension user before calling or when the extension user does not answer the call.

- Generally, when DKP extension users call another extension, the name of the called extension is displayed on the calling DKP extension. Now, if the flag 'Display User Status during Call' is enabled in the System Parameters, when DKP extension users call another extension, the calling DKP extensions will be displayed the 'presence' status message published by the called extension[318].

- SLT extension users, whose phone is equipped with a CLI display, can see the status of another extension by dialing a feature access code, then going ON-Hook. The system will ring back the SLT and send the Presence status of the desired extension as CLI.

- SIP extension users can use the Presence feature of SARVAM UCS to view the presence status of other extensions. For this, they must dial the feature access code and the number of the desired extension.

- SIP extension users who want to view the status of other extensions using the feature supported by their SIP Client, must have 'Presence Subscription' enabled in their SIP Extension Settings. Refer “Configuring SIP Extensions”.

How to configure

This feature involves the programming of the following parameters:

- **'Display User Status during Call' flag:** DKP extension users will be able to view the presence status for the called extension only if this flag is enabled in the System Parameters.

- **PUBLISH:** SIP extension users who want to publish their presence using the feature supported by their SIP client will be able to publish their presence status only if this feature is enabled in their SIP Extension Settings. This parameter is not necessary, if they want to publish presence using the feature of SARVAM UCS.

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318. DKP users can also dial a feature access code and the number of the extension to see the status of that extension on their DKP. But this would not be required, if the 'Display User Status during Call' flag is enabled in the System Parameters.

- **Presence Subscription:** SIP extension users who want to view the presence of other extensions using the feature supported by their SIP client must have this feature enabled in their SIP Extension Settings. This parameter is not necessary, if they want to view presence using the feature of SARVAM UCS.

- **Publish Messages:** It is possible to customize the Publish Messages listed above from 6 to 9 viz.: 'I am Mobile', 'In Meeting', 'Out for Meal', 'Out of Office'.

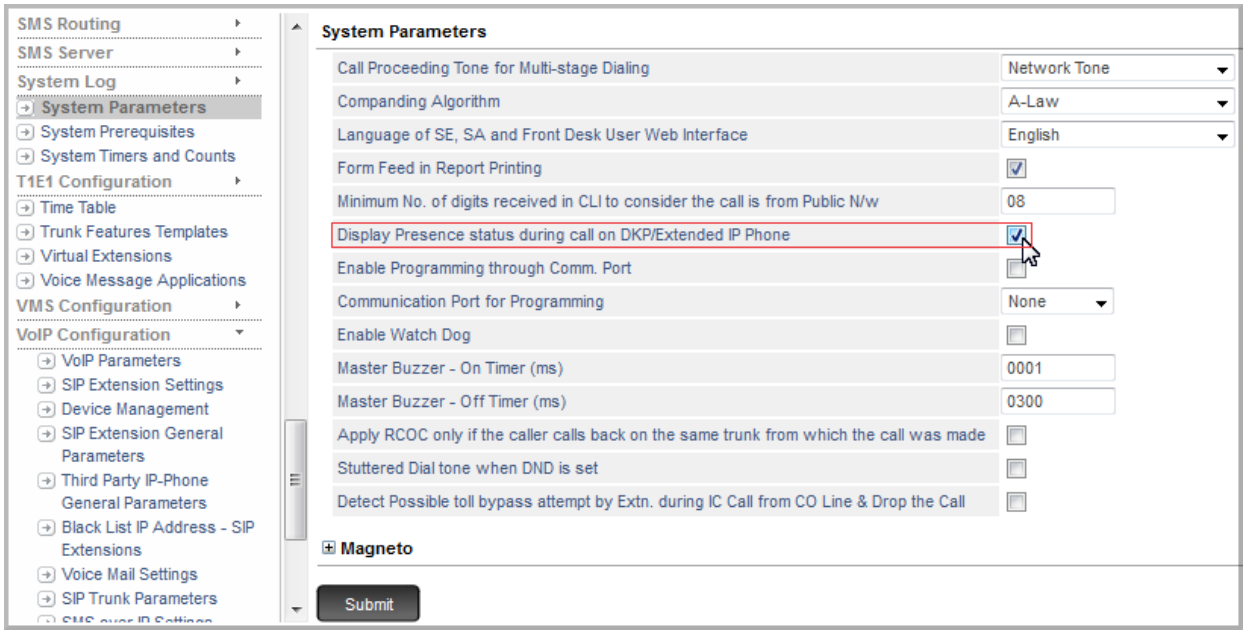
The above parameters, with the exception of 'Publish Messages', can be programmed using both Jeeves and a Telephone. You can program 'Publish Messages' using Jeeves only.

Programming Presence using Jeeves

- Log into Jeeves as System Engineer.

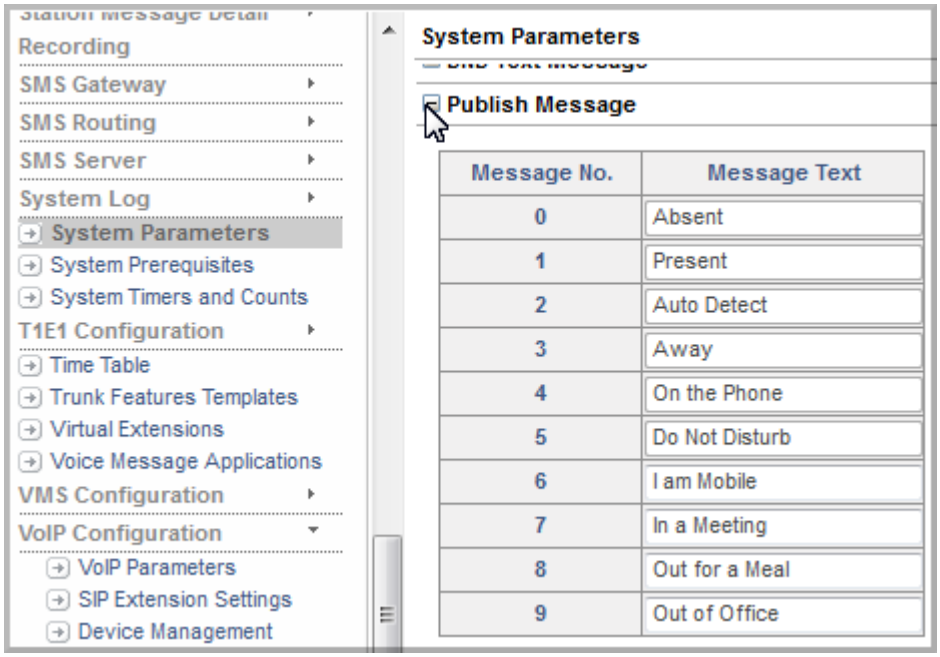
- Under **Configuration**, click **System Parameters** to open the page.

- Go to **Display Presence Status during Call on DKP**. Click to enable the flag.



- Click **Submit** at the bottom of the page.

- Click **Publish Message** to expand.



- You can change message number 6 to 9 as desired. The string may consist of a maximum of 16 characters. All ASCII characters except < > and “ (double quote) are allowed.
- Click **Submit** at the bottom of the page to save your settings.
- Under **Configuration**, click **VoIP Configuration**.
- Click **SIP Extension Settings** to open the page.
- Now, go to the desired SIP Extension number for which you want to enable the features PUBLISH and Presence Subscription. By default both features are disabled. Click the respective check boxes to enable the features.
- Click **Submit** at the bottom of the page to save your settings.
- Log out of Jeeves or continue, as desired.

Programming Presence using a Telephone

- Enter SE mode from a DKP/SLT.

  To enable/disable 'Display Presence Status during Call on DKP', dial:
  - ***5320-Flag***
    Where,
    Flag is
    0 for Disable
    1 for Enable
    By default, flag is disabled.
- Exit SE mode.

How to use

This feature requires you to dial your User Password. The default User Password 1111 is not accepted. Please change the User Password first.

Publish Presence can be set for an extension also from the System Administrator mode.

For EON and Extended IP Phone Users

Publishing Presence by Extension User

- Press DSS Key assigned to PUBLISH presence.

  OR

- Dial ***104***
- Enter User Password on the prompt.
- Scroll to the desired Publish message from the menu:
  - Absent
  - Present
  - Auto Detect
  - Away
  - On the Phone
  - Do Not Disturb
  - I am Mobile
  - In Meeting
  - Out for Meal
  - Out of Office
- Press Enter key to select message.
- You get the confirmatory tone.

Publishing Presence from SA Mode

- Press DSS Key assigned to PUBLISH presence.

  OR

- Dial ***1072-014***
- Enter Destination Number, that is, the number of the extension Publish Presence is to be set.
- Scroll to the desired Publish message from the menu:
  - Absent
  - Present
  - Auto Detect
  - Away
  - On the Phone
  - Do Not Disturb
  - I am Mobile
  - In Meeting
  - Out for Meal
  - Out of Office
- Press Enter key to select message.
- You get the confirmatory tone.

To view Presence Status

- Press DSS Key assigned to Display Presence Status.

  OR

- Dial ***1097.***
- Enter Extension number
- The status of the extension number you dialed will be displayed on your phone's LCD.
- Go ON-Hook.

For SLT Users

Publishing Presence by Extension User

- Lift the handset.
- Dial ***104-Password-Index Number***

Index No.	Meaning
0	Absent
1	Present
2	Auto Detect
3	Away
4	On the Phone
5	Do Not Disturb
6	I am Mobile
7	In Meeting
8	Out for Meal
9	Out of Office

- Replace handset.

Publishing Presence from SA Mode

- Lift the handset.
- Dial ***1072-014-Extension Number-Index Number***

Index No.	Meaning
0	Absent
1	Present
2	Auto Detect
3	Away
4	On the Phone
5	Do Not Disturb
6	I am Mobile

Index No.	Meaning
7	In Meeting
8	Out for Meal
9	Out of Office

- Replace handset.

To view Presence Status

You can view Presence Status of another extension only if your SLT has a CLI display.

- Lift handset.
- Dial ***1097-Extension Number.***
- You get confirmation tone.
- Go ON-Hook during confirmation tone.
- Your phone will ring and the status of the extension number you dialed will be displayed on your phone as CLI.

Preset Call Forward

What's this?

SARVAM UCS supports the Preset Call Forward. This feature is useful when Call Forward is not set by users, as their calls will automatically be forwarded to the selected destination. This feature is independent of the Class of Service assigned to the extension users.

Preset Call Forward options can be configured for each time zone by the SE only. The calls will be forwarded to the selected destination— Voicemail, Extension (SLT, DKP, SIP) or Department Group as per the Preset Call Forward type selected.

If users set Call Forward from their extensions, it will have a priority over Preset Call Forward. When the users cancel Call Forward from their extensions, the Preset Call Forward option will be applicable automatically.

The Preset Call Forward feature of SARVAM UCS offers the following forwarding options:

- **When Busy** - calls are forwarded to the destination phone number only when the called party's phone is busy.
- **When No Reply** - calls are forwarded to the destination phone number only when the called party does not answer the phone. The default time is 30 seconds for all extensions and can be changed by programming the Call Forward No-Reply Timer.
- **When Busy or No Reply** - calls are forwarded to the destination phone number when the called party's phone is either busy or does not reply.

How it works

A has set Preset Call Forward When No Reply to the Voicemail.

- The system waits for the Call Forward No-Reply Timer to expire and forwards all incoming calls to A’s Voicemail.

A has set Preset Call Forward When Busy to B’s extension.

- The system forwards the call for A to B on detecting Busy signal from A.

B has set Preset Call Forward-No Reply on A and A belongs to a Department Group.

- If the Call Forward-No ReplyTimer set is less than the Ringer Timer of the Department Group, the Preset Call Forward request will be served and the call will land on A.
- If the Call Forward-No Reply Timer set is greater than the Ringer Timer of the Department Group, the Preset Call Forward request will not be served.

If the Ignore call forward set by member extension, when call is routed on Routing/Dept. Group option is enabled in System Parameters, then Preset Call Forward request will not be served. See “System Parameters” for more information.

A has set Preset Call Forward to Department Group.

- The system forwards the call for A to the Department Group. The free member in the group answers the call.

A has set Preset Call Forward When Busy or No Reply to the Voicemail.

- Whenever there is a call for A, if the system does not detect a busy signal from A, it waits for the Call Forward No-Reply timer to expire.
- The system forwards the call to the Voice Mail System.

If A wants to change the Call Forward destination temporarily, then A must set Call Forward from his/her extension. For detailed instructions, see “Call Forward”. In this case, the Preset option will not be applicable. But as soon as A cancels Call Forward from his/her extension, the Preset option will be applicable.

- *Preset Call Forward cannot be cancelled by the users.*
- *The system supports only single-point Preset Call Forward, which means, if the destination extension has also forwarded its calls, the call will not follow the forwarding path. For example: Calls for extension A are forwarded to extension B. Preset Call Forward is also set on extension B with C as the destination number. In this case, Calls for A will land on B and calls for B will land on C. Calls for A will not land on C.*
- *Only one Preset Call Forward Type can be set for each Time Zone. Every new Preset Call Forward Type set overrides the previous one.*

How to configure

The functioning of this feature is controlled by the following parameters: Preset Call Forward configured in the 'Station Advanced Feature Template' assigned to the extension user and 'Call Forward No-Reply Timer'.

When Preset Call Forward No-Reply is set, if required the Call Forward No-Reply Timer needs to be programmed.

Configuring Preset Call Forward in the Station Advanced Feature Template

The default Station Advanced Feature Template 01 is assigned to all extension users of the SARVAM UCS. In this template Preset Call Forward is disabled.

Decide which extensions are to be allowed 'Preset Call Forward'. If you want to allow Preset Call Forward to all extensions, retain the default Station Advanced Feature Template 01 and configure the Preset parameters. However, if you want to allow Preset Call Forward only to selected extensions, select another Station Advanced Feature Template and configure the Preset parameters in that template.

Now, to assign the template to selected extensions, follow these steps:

1. Prepare a Station Advanced Feature Template with Preset Call Forward configured for each time zone.
2. Assign this newly prepared template to the desired extensions.

Refer the topics “Station Advanced Feature Template” for detailed programming instructions on how to customize a Station Advanced Feature Template, configure the Preset Call Forward parameters and how to apply this template on extensions.

Call Forward No Reply Timer

When using Call Forward -No Reply, each extension can set a different Time after which the incoming call on the extension should get forwarded when there is no reply from the extension. For this, the Call Forward No-Reply Timer must be programmed. By default, this timer is set to 30 seconds.

The Call Forward No-Reply Timer is to be programmed in the “Station Advanced Feature Template” applied on the extensions which are allowed Call Forward in their COS.

If you want to set this timer to the same duration for all extensions, simply set the Call Forward No-Reply Timer in the default Station Advanced Feature Template 01 which is assigned to all extensions.

If you want to set different Timer duration for different extensions, then prepare separate Station Advanced Feature Templates with the desired Timer durations and assign different Templates (with different Timer durations) to the extensions as desired.

Refer the topic “Station Advanced Feature Template” for instructions on customizing the template and applying the template to extensions.

Priority Calls in E&M MFCR2 Signaling

What's this?

In a network of Systems connected over E&M trunks, callers may be unable to reach the desired station, as the called station may be busy with another station.

The problem becomes more acute when important decision makers and persons involved in critical tasks in an organization are unable to get through to the desired extension of the networked System, because it is busy.

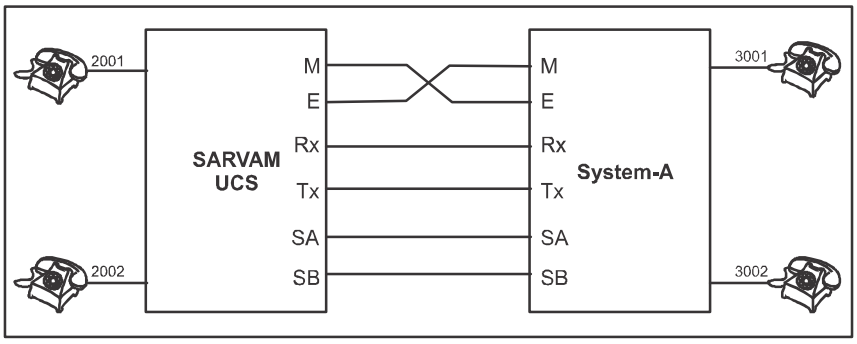
SARVAM UCS supports Priority Call in E&M MFCR2 Signaling to overcome this. This feature makes it possible to land a call on a busy extension and establish speech with the desired extension user. If required, you can also disconnect the remote party in speech with the called extension.

E&M MFCR2 Signaling is currently supported on ETERNITY GE E&M Card only. This feature is currently available on ETERNITY GENX only.

How it works

- The extensions which are to be allowed Priority Call feature must be defined as 'Priority Subscriber' in their Station Advanced Feature Templates.
- The extensions on which Priority Calls should be allowed to land must be 'Non-Priority Subscribers' (that is, defined as 'Ordinary Subscriber' in the Station Advanced Feature Template).
- In the following illustration, SARVAM UCS is networked with System-A over E&M Lines with MFCR2 Signaling.

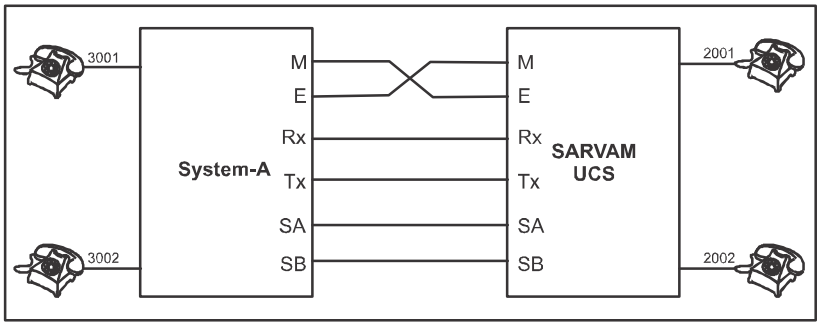
  Priority Call from SARVAM UCS to Networked System



- SARVAM UCS is the Originating System and System-A is the Terminating System.
- Extension 2001 of SARVAM UCS is a 'Priority Subscriber'. Extension 2002 is an 'Ordinary Subscriber' (Non-Priority).
- Extension 3001 of System-A is a 'Non-Priority Subscriber'. Extension 3002 of System-A is also a 'Non-Priority Subscriber'.
- When Extension 2001 calls Extension 3001, depending upon its implementation, System-A understands it as a priority call.

- If Extension 3001 is busy with another extension, 3002, and since both are non-priority subscribers, System-A will automatically intrude the conversation between 3001 and 3002 and establish 3-way speech between Extensions 2001, 3001 and 3002.
- Now, if Extension 2001 (the Priority caller) does not want Extension 3002 to be part of the conversation, s/he can ask 3002 to go idle OR s/he can use the feature Forced Release Order to disconnect 3002 from the conversation. 2001 can use Forced Release Order only if it is allowed to it in its “Class of Service (COS)”.
- When Extension 2001 dials the Feature Access Code for Forced Release Order, Extension 3002 will be disconnected and two-way speech will be established between 2001 and 3001.

Priority Call from Networked System to SARVAM UCS



- System-A is the Originating System and SARVAM UCS is the Terminating System.
- Extension 3002 of System-A is a Priority Subscriber. Extension 3001 of System-A is Non-Priority Subscriber.
- Extension 2001 and 2002 of SARVAM UCS are Non-Priority Subscribers
- Extension 3002, a Priority Subscriber, calls Extension 2002 of SARVAM UCS.
- SARVAM UCS treats the call as priority call.
- Extension 2002 is busy with Extension 2001. SARVAM UCS checks the priority of both extensions. Since both are non-priority subscribers, SARVAM UCS gives priority to Extension 3002 and plays beeps to 2002 and 2001 before establishing 3-way speech.
- SARVAM UCS will not treat a call as priority in the following cases:
  - If the busy call is itself a priority call, that is, at least one of the two parties in the busy call is a priority subscriber.
  - If any party involved in a matured call is Priority Extension or Trunk, the call will be considered as priority call and hence no incoming priority call will be allowed to intrude this priority call.
  - If the call is routed to data equipment, for example, fax machine.
- SARVAM UCS allows non-priority subscribers to intrude on a busy call using the feature Manual Priority Intrusion. For this, the intruding extension must have the feature 'Manual Priority Intrusion' in its Class of Service.

- When a non-priority extension requests Manual Priority Intrusion, SARVAM UCS receives an 80msec pulse signal on the E wire of the E&M port, on detecting this signal SARVAM UCS will treat it as a Priority Call and check whether the called extension that is busy can be intruded or not.

- If intrusion is possible SARVAM UCS creates a conference call, after playing beeps to notify both parties of the 3-way conference call. This beep will be played only if Conference beeps are enabled.

How to configure

For this feature to work, the extensions of the networked Systems which are to be allowed Priority Call feature must be defined as 'Priority Subscribers'.

Extensions on which Priority Calls must be allowed to land may be defined as 'Non-Priority Subscribers'.

If non-priority subscribers are to be allowed to Priority Calls when necessary, Manual Priority Intrusion feature must be enabled in their Class of Service.

If the Priority Subscriber extension users are to be allowed to disconnect the second party (other than the desired party) from the conversation during a Priority Call, the feature Forced Release Order must be enabled in their Class of Service.

For Stations on which data terminals are connected (like Fax machine), it is recommended that the 'Caller Category' of these stations be programmed as 'Data Transmission' (in their Station Advanced Feature Template).

Programming Extensions as Priority Subscribers

To define extensions of SARVAM UCS as 'Priority Subscribers' or 'Non-Priority Subscribers', you must set the parameter "Caller Category" in the Station Advanced Feature Template assigned to these extensions.

In the default Station Advanced Feature Template 01 assigned to all extensions of the SARVAM UCS, the Caller Category is set to 'Ordinary Subscriber', which means all extensions are by default, 'Non-Priority Subscribers'.

Since all extensions cannot be defined as 'Priority Subscribers', prepare a new Station Advanced Feature Template, select the option 'Priority Subscriber' as Caller Category in this template and apply it on the selected extension that are to be defined as Priority Subscribers.

Refer the topic “Station Advanced Feature Template” for detailed instructions.

Programming Manual Priority Intrusion and Forced Release Order in Class of Service

In the default the default CoS group 01 in Station Basic Feature Template Number 01 assigned by default to all extensions of SARVAM UCS, 'Manual Priority Intrusion' and 'Forced Release Order' are disabled.

If you want to allow all extensions these features, simply enable them in the default CoS group 01.

However, if either or both these features are to be allowed only to select extensions, follow these steps:

1. Define a CoS group with Manual Priority Intrusion/Forced Release Order enabled.

2. Prepare a Station Basic Feature Template with this CoS group applicable in all the “Time Zones”.

3. Assign this new Template to the extensions to which Manual Priority Intrusion/Forced Release Order is to be allowed.

Refer the topics “Class of Service (COS)” and “Station Basic Feature Template” for detailed instructions.

How to use

Manual Priority Intrusion

For EON and Extended IP Phone Users

- Dial the desired extension number.
- When you get the Busy Tone,
- Press DSS Key assigned to Manual Priority Intrusion (if programmed)
  OR
- Press Transfer Key and dial ***37.***
- You are in 3-way speech.

For SLT Users

- Dial the desired extension number.
- On Busy Tone, press Flash to put call on Consultation Hold.
- Dial ***37.***
- You are in 3-way speech.

Forced Release Order

For EON and Extended IP Phone Users

- Dial the desired extension number.
- When you are in 3-way speech,
- Press DSS Key assigned to Forced Release Order (if programmed)
  OR
- Press Transfer Key and dial ***38.***
- You are connected to the desired party. The second party is disconnected.

For SLT Users

- Dial the desired extension number.
- When you are in 3-way speech,
- Press Flash and dial ***38.***
- You are connected to the desired party. The second party is disconnected.

Priority

What's this?

'Priority' is the precedence given to certain trunks and extensions over others in being answered by the destination extension.

When 'Priority' is assigned to trunks, whenever there are incoming calls on multiple trunks at the same time, the call on the trunk with highest priority will be answered by the landing destination extension/Operator first.

When Priority is assigned to Extensions, calls from extensions with highest priority will have precedence in landing on the destination extension.

You can set priority levels from 1 to 9 as given in the table below.

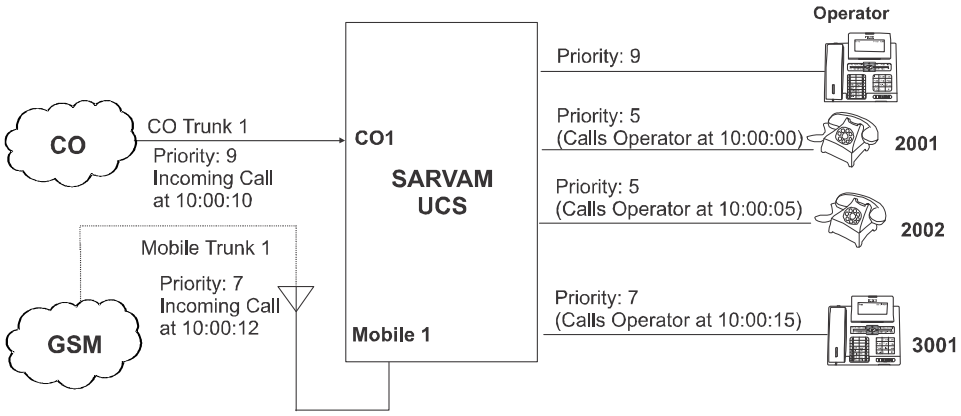
Priority Level	Meaning
1	None
2	Lowest
3	Lower
4	Low
5	Normal
6	Medium
7	High
8	Higher
9	Highest

Highest Priority can be assigned to Extensions of important or higher ranking persons in an organization; for example, calls from senior managers or top executives in an organization can be allowed to be answered first by the destination extension.

Highest Priority can be assigned to particular Trunks, such as special or private trunk lines, trunk lines dedicated as help lines or emergency trunks, or trunks designated as hotlines, so that when there are incoming calls on different trunks at the same time, the call on these trunks gets answered first by the destination extension.

Priority can be assigned to all Trunk types (CO, Mobile, SIP, T1E1PRI, BRI) and Extension types (SLT, DKP, ISDN terminal, SIP, E&M as Station, Radio, Magneto port).

To understand how this feature works, consider this illustration:



Here,

- There are two incoming calls, one on the Analog Trunk, CO 1 and the Mobile Trunk 1 at the same time.
- CO Trunk 1 has priority, ‘9’, the Mobile Trunk 1 is assigned priority ‘7’.
- Three extensions, 2001, 2002 and 3001 are calling the Operator. Extension 3001 has priority ‘7’, while extensions 2001 and 2002 have the same priority, ‘5’.
- Now, on the Operator extension, which is the landing destination, the incoming calls from the trunks and the extensions will land in the following chronological order:

Caller	Time of the Call	Priority
SLT 2001	10:00:00	5
SLT 2002	10:00:05	5
CO Trunk 1	10:00:10	9
Mobile Trunk 1	10:00:12	7
DKP 3001	10:00:15	7

- These incoming calls, however, will appear on the Display of Operator’s phone (DKP or Extended IP Phone) in the order of priority:
  - CO 1
  - Mobile Trunk 1
  - DKP 3001
  - SLT 2001
  - SLT 2002
- Now, when the Operator goes Off-hook (pressing speaker key or picking up the handset), the call on CO 1 will be answered first, as CO Trunk 1 has the highest priority.
- The Operator goes On-hook and then Off-hook, the call on Mobile Trunk 1 will be answered. Though Mobile Trunk 1 and DKP 3001 have the same priority, ‘7’, Mobile Trunk 1 will be answered first, following the chronological order.
- When the Operator goes On-hook after answering the call on Mobile Trunk 1, the call from DKP 3001 will be placed on the Operator phone with a *Priority Ring* (configurable; default: Triple Ring).

- When the Operator goes Off-hook, the call from DKP 3001 is answered.
- When the Operator goes On-hook and then Off-hook after answering the call from DKP 3001, the call from SLT 2001 will get answered first, though both 2001 and 2002 have the same priority, ‘5’. In this case, *Priority Ring* will not be played.
- Thus, calls from trunks and extensions are answered by the landing destination in the order of priority. Where priority is the same, calls are answered in chronological order. Calls from extensions with higher priority are indicated by a *Priority Ring* on the landing destination.

Priority is relevant only when there is more than one call on the destination.

You can assign Priority to SLT extensions. However, Priority is not relevant when the SLT is a landing destination, because there cannot be more than one call ringing on an SLT extension at a time.

An “Intercom” call will be placed at the destination only if the caller has a higher priority.

How to configure

To assign Priority to Trunks, you must set the priority in their “Trunk Feature Template”. See “Configuring Trunks”.

To assign Priority to Extensions, see instructions for configuring the respective Extension port type:

- “Configuring SLT Extensions”
- “Configuring DKP Extensions”
- “Configuring ISDN Terminals”
- “Configuring SIP Extensions”
- “Configuring E&M Lines”
- “Configuring Radio Interface”
- “Configuring Magneto Interface”
- “Virtual Extension”

If required, you may change the Ring Pattern of the *Priority Ring*. See “Distinctive Rings” for instructions.

Privacy

What's this?

Extensions of SARVAM UCS can be protected from the intrusions by other extensions or from trunk calls by activating Privacy.

How it works

Intrusions can occur on an extension when another extension invokes the following features:

- DND Override
- "Interrupt Request (IR)"
- "Barge-In"
- "Raid"

Intrusions can also occur,

- When an external caller uses "Auto Attendant" to reach an extension.
- When there is a call from another trunk line when you are in speech.

To prevent such intrusions, SARVAM UCS enables you to set the following types of Privacy:

- **Privacy from Interrupt Request, Barge-In, DND Override:** This type of Privacy protects an extension from intrusions by other extensions using Interrupt Request, Barge-In or DND Override.

  For example: Extension A has Privacy from Interrupt Request, Barge-In and DND Override.

  Extension A and B are in speech, Extension C attempts to intrude the conversation Interrupt Request or Barge-In. Extension C's call will be blocked and C will get error tone.

  Now, Extension A has set DND and Extension B attempts to override it using DND Override. Since A has Privacy from DND Override, B's call will be blocked and B will get error tone.

- **Privacy from Raid:** This type of Privacy protects an extension from intrusions by other extensions using Raid.

  For example: This type of Privacy is set on Extension A. Extension A and B are in speech, Extension C uses Raid to intrude the conversation. Extension C's call will be blocked and C will get error tone.

- **Privacy from Trunk call intrusion**: This type of Privacy prevents the extensions in the Trunk Landing Groups that are busy from being intruded by another waiting call. For this type of Privacy to work, the feature "Trunk Call Waiting" must be disabled on the extension.

  For example: Extension A is the first trunk landing extension for calls on Trunk 1. Extension A and B are in speech. A new call lands on Trunk 1. If A has Trunk Call Waiting beeps disabled, A will not hear the intrusion beeps. The system will land the call on the next extension in the trunk landing group for calls on Trunk 1.

- **Privacy from Built-In Auto Attendant call**: This type of Privacy protects the extension from being accessed by external callers using “Auto Attendant”.

  For example: This type of Privacy is set on Extension A. Extension A and B are in speech, external caller C uses Built-In Auto Attendant to call extension A. C’s call will be blocked and C will get error tone.

How to configure

To provide Interrupt Request to extension users, you must enable this feature in the “Class of Service (COS)” assigned to them for the time zones in their “Station Basic Feature Template”.

By default, Privacy from Raid is enabled in the Class of Service of all Extension types: SLT, DKP, SIP, ISDN. So, none of the extensions can raid the other. You may disable this feature in the Class of Service of extensions, which you want to protect from Raid.

By default, Privacy from Interrupt Request, Barge-In and DND Override are disabled in the Class of Service of all Extension types. You may enable this feature on extensions which you want to protect from intrusions using any of these features.

By default, Privacy from Built-In Auto Attendant is disabled on all Extension types. You may enable this feature on extensions which you do not want external callers to reach.

By default, Trunk Call Waiting is disabled on all Extension types. You may keep this feature disabled on extensions which you want to provide Privacy from Trunk Call intrusion beeps.

For instructions, see “Class of Service (COS)”, “Station Basic Feature Template”. Also see,

- “Configuring SLT Extensions”
- “Configuring DKP Extensions”
- “Configuring ISDN Terminals”
- “Configuring SIP Extensions”

Power Fail Transfer[319]

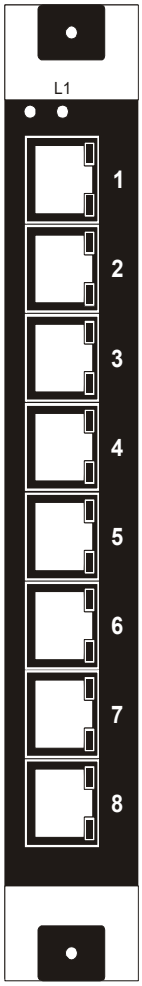
What's this?

When the power supply to the System fails, SLT 1 and SLT 2 are automatically connected to CO trunks CO1 and CO2 respectively of the same card. The System switches from normal functioning mode to the Power Failure Connections. Only the trunks handled by Power Failure Connections can be used during a power failure. Outgoing calls can be made from such extensions as well as incoming calls on the trunks can land on these extensions. When power resumes and normal system functioning is restored, calls on the Power Failure Connections remain in effect until they are terminated.

The following Cards of SARVAM UCS support Power Fail Transfer.

ETERNITY ME CO8+SLT24

Combination card, with 8 CO ports to connect 8 CO analog trunk lines and 24 SLT ports to connect 24 Single Line Telephones.

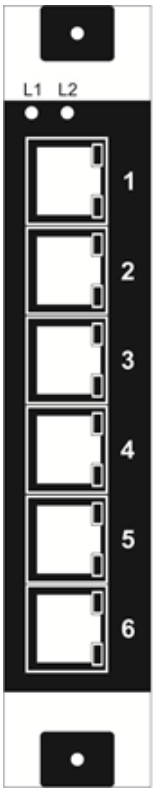


Connector	Color	Connection	H/w Port Offset
RJ45-1	Blue - (Blue & White)	SLT	01
	Orange - (Orange & White)	SLT	02
	Green - (Green & White)	SLT	03
	Brown - (Brown & White)	SLT	04
RJ45-2	Blue - (Blue & White)	SLT	05
	Orange - (Orange & White)	SLT	06
	Green - (Green & White)	SLT	07
	Brown - (Brown & White)	SLT	08
RJ45-3	Blue - (Blue & White)	SLT	09
	Orange - (Orange & White)	SLT	10
	Green - (Green & White)	SLT	11
	Brown - (Brown & White)	SLT	12
RJ45-4	Blue - (Blue & White)	SLT	13
	Orange - (Orange & White)	SLT	14
	Green - (Green & White)	SLT	15
	Brown - (Brown & White)	SLT	16
RJ45-5	Blue - (Blue & White)	SLT	17
	Orange - (Orange & White)	SLT	18
	Green - (Green & White)	SLT	19
	Brown - (Brown & White)	SLT	20
RJ45-6	Blue - (Blue & White)	SLT	21
	Orange - (Orange & White)	SLT	22
	Green - (Green & White)	SLT	23
	Brown - (Brown & White)	SLT	24
RJ45-7	Blue - (Blue & White)	CO	01
	Orange - (Orange & White)	CO	02
	Green - (Green & White)	CO	03
	Brown - (Brown & White)	CO	04
RJ45-8	Blue - (Blue & White)	CO	05
	Orange - (Orange & White)	CO	06
	Green - (Green & White)	CO	07
	Brown - (Brown & White)	CO	08

319. ETERNITY PENX does not support this feature.

ETERNITY GE CARD CO4+DKP2+SLT12

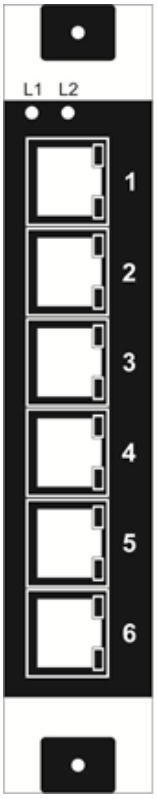
Combination Card, with 4 ports to connect 4 Two-wire Trunk lines, 2 ports to connect 2 Digital Key Phones and 12 ports to connect 12 Single Line Telephones.



Connector	Color	Connection	H/w Port Offset
RJ45-1	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	01 02 03 04
RJ45-2	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	05 06 07 08
RJ45-3	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	09 10 11 12
RJ45-4	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	CO CO CO CO	01 02 03 04
RJ45-5	Blue - (Blue & White) Orange - (Orange & White)	DKP DKP	01 02
RJ45-6	Unused		

ETERNITY GE CARD CO4+DKP2+SLT8

Combination Card, with 4 ports to connect 4 Two-wire Trunk lines, 2 ports to connect 2 Digital Key Phones and 8 ports to connect 8 Single Line Telephones.



Connector	Color	Connection	H/w Port Offset
RJ45-1	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	01 02 03 04
RJ45-2	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	05 06 07 08
RJ45-3	Unused		
RJ45-4	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	CO CO CO CO	01 02 03 04
RJ45-5	Blue - (Blue & White) Orange - (Orange & White)	DKP DKP	01 02
RJ45-6	Unused		

Feature Interactions

Force Call disconnection: If PFT ports are in conversation during power failure and power supply is restored, then any other extensions of the system cannot invoke Forced Call Disconnection on PFT SLT or PFT CO port.

ACB (Auto Call back): If PFT ports are in conversation during power failure and power supply is restored, then other extensions of the system will be able to set ACB on PFT SLT or CO port.

Raid: If PFT ports are in conversation during power failure and power supply is restored, then system shall not allow other extensions of system to invoke Raid feature on called PFT SLT or CO port.

Barge - In: If PFT ports are in conversation during power failure and power supply is restored, system shall not allow any other extensions of the system to access Barge in on busy PFT SLT port.

Interrupt Request: If PFT ports are in conversation during power failure and power supply is restored, system shall not allow any other extension to make an interrupt request on busy PFT SLT port.

Trunk Reservation: If PFT ports are in conversation during power failure and power supply is restored, system shall allow you to access Trunk reservation on busy PFT CO port.

QSIG

What's this?

Q-Signaling (QSIG) is an ISDN based protocol for signaling between two Systems. QSIG is a protocol based on internationally agreed Standards for ISDN. Using QSIG you can network two or more SARVAM UCS. This is known as 'Interoperability'. The SARVAM UCS supports QSIG on the T1E1PRI ports.

The basic call procedure in the QSIG is implemented as per the ECMA-143. The generic functional protocol for the Supplementary Services is implemented as per ECMA-165.

This provides details about following:

a. QSIG Supplementary Services
b. Configuring for - Basic Call and Supplementary Services
c. Configuring for - IC Call using DDI Routing Over QSIG

To use this feature you must purchase a License. Refer the topic "License Management" to know more.

a. QSIG Supplementary Services

The SARVAM UCS supports following supplementary services in QSIG:

- Advice of Charge (AOC)
- Identification:
  - The Calling Line Identification Presentation (CLIP)
  - Calling Line Identification Restriction (CLIR)
  - The Connected Line Identification Presentation (COLP)
  - Connected Line Identification Restriction (COLR)
- Name Identification:
  - The Calling Name Identification Presentation (CNIP)
  - The Connected Name Identification Presentation (CONP)
- Call Diversion:
  - Call Forward Unconditional (CFU)
  - Call Forward On Busy (CFB)
  - Call Forward on No Reply (CFNR)
- Call Completion:
  - Call Completion on Busy Subscriber (CCBS)
  - Call Completion on No Reply (CCNR)
- Do Not Disturb (DND)
- Do Not Disturb - Override (DNDO)
- Call Intrusion (CI)

- Call Transfer (CT)
- Recall (RE)
- Message Wait Indication (MWI)
- Call Offer (CO)

The implementation of these features in QSIG is as per specific ECMA standards as described below.

Advice on Charge:

- Advice on Charge (AOC) is implemented as per ECMA-211 and ECMA-212.
- Using this feature, the caller can know the cost of the call made to public network using networked trunk. The cost of the call will be determined by using Call Cost Calculation. Refer chapter “Call Cost Calculation (CCC)”

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- The SARVAM UCS supports 'AOC-E' End of the Call Charge Information. SARVAM UCS supports charging for calls made to public network. The Cost of the call is calculated by the end System from which the call is terminated on the public network.

CLIP/CLIR/COLP/COLR/CNIP/CONP

- The Identification features are implemented as per ECMA-163 and ECMA-164.
- Refer chapters “Calling Line Identification and Presentation (CLIP)” and “Calling Line Identification Restriction (CLIR)”.

Call Diversion

- Call Diversion is implemented as per ECMA-173 and ECMA-174. Refer chapters “Call Forward” and “Call Forward-Remote”.
- Remote activation/deactivation of Call Forward is not supported.

Call Completion-on Busy Subscriber and on No Reply (CCBS and CCNR)

- Call Completion (CC) features CCBS and CCNR are implemented as per standards ECMA-185 and ECMA-186.
- The CCBS is implemented as the feature Auto Call Back on Busy and the CCNR is implemented as the feature Auto Call Back on No Reply. Refer chapter “Auto Call Back (ACB)”.

Do Not Disturb (DND)

- DND is implemented as per standards ECMA-193 and ECMA-194. Refer chapters “Do Not Disturb (DND)”.

Do Not Disturb - Override (DNDO)

- DND is implemented as per standards ECMA-193 and ECMA-194. See the topic “Do Not Disturb (DND)”.

Call Intrusion (CI)

- Call Intrusion is implemented as per standards ECMA-202 and ECMA-203.
- The Call Intrusion (CI) is implemented as Raid feature. However the extension on which intrusion is made, will not get 'beeps'. On successful intrusion, the 'conference 3-party' type call will be established. See the topic “Raid” and “Conference-3 Party”.

Call Transfer

- Call Transfer is implemented as per standards ECMA-177 and ECMA-178. Refer chapter “Call Transfer”.
- The SARVAM UCS supports Call Transfer by Joining method.

'CLIP-Hold' flag on the Transferring Station:

- If 'CLIP-Hold' flag is enabled on the Transferring Station, the SARVAM UCS will send the held party's number as calling number while placing call on QSIG.
- If 'CLIP-Hold' flag is disabled on the Transferring Station, the SARVAM UCS will send the calling station's number as calling number. Refer chapter “Calling Line Identification and Presentation (CLIP)” and “Station Advanced Feature Template” for more details of CLIP-Hold.

- *It is recommended to enable CLIP Hold flag for the operator extension, so that when operator transfers the call to another System using QSIG, the terminating System (SARVAM UCS) can identify it as call from Public network and treat it as an incoming call.*
- *As a Terminating System, the SARVAM UCS will consider the call as internal or from public network depending upon the length of the digits received as the calling number.*
- *If Calling number is not received because of any reason, and CLIP- Hold flag is enabled, the SARVAM UCS will send the 'Trunk Name' configured for the trunk port as calling line identification.*

Recall (RE)

- Recall (RE) is implemented as per standards ECMA-213 and ECMA-214 and ECMA-143.
- The SARVAM UCS supports Recall-Busy and Recall-No Answer features like SARVAM UCS features, Call Transfer-On Busy and Call Transfer-While Ringing.
- Refer chapter “Call Transfer” for more details.

Message Wait Indication (MWI)

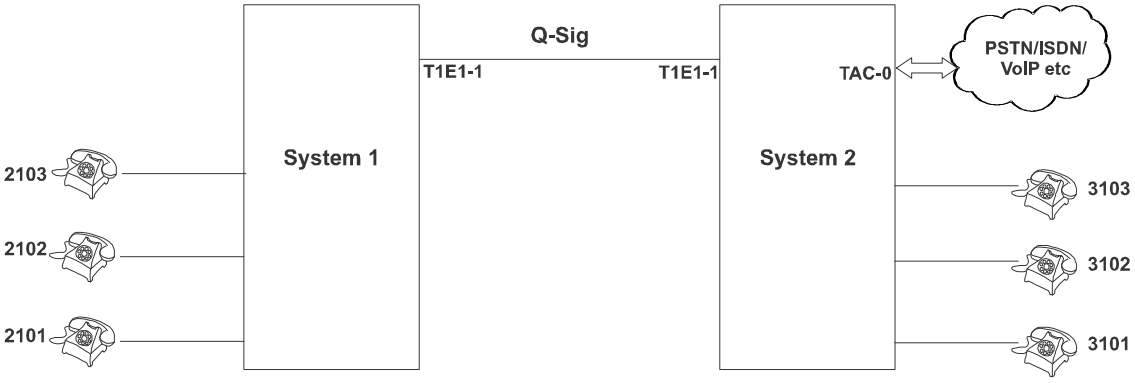
- Message Wait (MWI) is supported as per standards ECMA- 241 and ECMA-242.
- Refer chapter “Message Wait” for details about form of indications.

Call Offer

- Call Offer is implemented as per standards ECMA-191 and ECMA-192.
- The call offer feature is implemented as the 'Interrupt Request' feature.

- Refer chapter “Interrupt Request (IR)” for more details.

b. Configuring for - Basic Call and Supplementary Services



For simple application as shown in above figure, make settings as given below for System 1 and System 2.

- Connect T1E1PRI line between System 1 and System 2.
- Now open Web Jeeves and configure the settings in System 1 and System 2 as given below:

Settings for System 1:

Following basic settings are required in SARVAM UCS System 1.

- T1E1 Port Parameters:
  - Set Signal Type as "QSIG"
  - Set Orientation Type as "Network"
- T1E1 PRI/QSIG Signaling Parameters:
  - Set ISDN Switch Variant as "QSIG E1"
- Closed User Groups (CUG)

Index	Route Code	OGTBG	Strip Digit Count	Self Route	Dialed Digit Count	Apply Toll Control	Apply Call Cost
001	21	01	0	Enable	04		
002	31	01	0	Disable	04		
003	0	01	0	Disable	04		

- OG Trunk Bundle Groups

Group No.	Rotation Flag	LCR Type	OGTB Member 1	OGTB Member 2	OGTB Member 3	-------	OGTB Member 8
01	Disable	None	01	00	00	-------	00

- OG Trunk Bundle

Template Number	Trunk Port		Start Channel Number	Total Trunk Count	Rotation Type	Automatic Number Translation (ANT)	
	Port Type	Port No.				Apply	ANT Table No.
01	T1E1PRI	001	01	030	Cyclic	Disable	05

Following settings are required for CLI display.

- In T1E1PRI Port Parameters, under DDI Routing configure Outgoing (OG) Reference ID as "01" .

- DDI Routing-Outgoing Reference Table.

Table ID	OG Ref. ID	Start Channel No.	Channel Count	Pilot Number	DDI Routing Ref. ID
01	01	01	30	2100	01

- DDI Routing-DDI Routing Table

Table ID	Reference ID	Start DDI No.	Total DDI Numbers	DDI No. Digit Count	Port Type	Port Number	Start DDI Flexible No.
001	01	2100	0100	4	FLEXNUM	000	2100

Settings for System 2:

Following basic settings are required in System 2.

- T1E1 Port Parameters
  - Set Signal Type as "E1 QSIG"
  - Set Orientation Type as "Terminal"

- T1E1 PRI/QSIG Signaling Parameters:
  - Set ISDN PRI Variant as "QSIG E1"

- Closed User Groups (CUG)

Index	Route Code	OGTBG	Strip Digit Count	Self Route	Dialed Digit Count	Apply Toll Control	Apply Toll Control
001	21	02*	0	Enable	04		
002	31	02*	0	Disable	04		

**Assign OGTBG of T1E1PRI-1 used for QSIG connectivity with System 1.*

- OG Trunk Bundle Groups

Group Number	Rotation Flag	LCR Type	OGTB Member 1	OGTB Member 2	OGTB Member 3	-------	OGTB Member 8
01	Disable	None	01	00	00	-------	00
02	Disable	None	02	00	00	-------	00

- OG Trunk Bundle

Template Number	Trunk Port		Start Channel no.	Total Trunk Count	Rotation Type	Automatic Number Translation (ANT)	
	Port Type	Port No.				Apply	ANT Table No.
01	Mobile	001	01	008	Cyclic	Disable	05
02	T1E1PRI	001	01	030	Cyclic	Disable	05

Following settings are required for CLI display.

- In T1E1PRI Port Parameters, under DDI Routing configure Outgoing (OG) Reference ID as "01".
- DDI Routing-Outgoing Reference Table.

Table ID	OG Ref. ID	Start Channel No.	Channel Count	Pilot Number	DDI Routing Ref. ID
01	01	01	30	3100	01

- DDI Routing- DDI Routing Table

Table ID	Reference ID	Start DDI No.	Total DDI Numbers	DDI No. Digit Count	Port Type	Port Number	Start DDI Flexible No.
001	01	3100	0100	4	FLEXNUM	000	3100

Basic Call:

Given below are various examples of calls made/received between the systems.

Example 1:

To make call from station of System 1 to any station of System 2 or vice versa.

- From station 2101 of System 1, dial station number 3101 or from station 3101 of System 2, dial station number 2101.

Example 2:

To make outgoing call from station of System 1 using Trunk of System 2.

- From station 2101 of System 1, dial 0^{320} (TAC at System 2 for Trunk) and then dial 02652630555 (External Party number).

Example 3:

Incoming call on Trunk at System 2 using Auto Attendant[321].

- Enable Auto Attendant on Mobile Trunk at System 2.
- Now External Party dials the number of the Mobile port. Incoming call is answered by System 2. External Party gets Auto Attendant Music. Now External Party dials station number.
    - If External Party dials station number 3101 then call lands on 3101 of System 2.
    - If External Party dials station number 2101 then call lands on 2101 of System 1.

Supplementary Services:

Working of Supplementary Services of QSIG in SARVAM UCS is explained as below.

Identification:

Station 2101 of System 1, dial station number 3101.

- CLIP/CNIP:

  Name and Number of station 2101 is displayed as CLI on station 3101.

- COLP/CONP:

  When 3101 of System 2 answers the call then Name and Number of station 3101 is displayed on station 2101 using reverse DDI method.

- CLIR:

  To restrict CLI at station 3101, configure the following for station 2101.

  - Allow "CLI Restriction (CLIR)" in Class of Service for station 2101.
  - Enable CLIR by dialing ***103-1*** from station 2101.

  Here, Name and Number of station 2101 is not displayed on station 3101.

If "CLI Restriction (CLIR) Override" is enabled in Class of Service for station 3101, then Name and Number of station 2101 is displayed on station 3101.

- COLR:

  To restrict CLI when incoming call is answered by 3101, configure the following for station 3101.

  - Allow "CLI Restriction (CLIR)" in Class of Service for station 3101.
  - Enable CLIR by dialing 103-1 from station 3101.

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320. *Here 2101 of System 1 does not get dial tone after dialing 0 (TAC at System 2 for Trunk) because 0 is programmed in CUG table for System 1.*
321. *Refer "c. Configuring for-IC Call using DDI Routing Over QSIG" topic for how to route Incoming Call on T1E1PRI/SIP over QSIG using DDI Routing.*

Here, Name and Number of station 3101 is not displayed on station 2101 when station 3101 answers the incoming call.

If "CLI Restriction (CLIR) Override" is enabled in Class of Service for station 2101, then Name and Number of station 3101 is displayed on station 2101.

Call Diversion:

Station 2101 of System 1 wants to set Call Diversion (Call Forward) to station number 3101 of System 2.

- Call Forward-Unconditional:

  From station 2101 of System 1, dial ***131-3101-#****. Now all incoming calls on 2101 are forwarded to 3101 of System 2.

- Call Forward-on Busy:

  From station 2101 of System 1, dial ***132-3101-#****. Now all incoming calls on 2101 are forwarded to 3101 of System 2 when 2101 is busy.

- Call Forward- on No Reply:

  From station 2101 of System 1, dial ***133-3101-#****. Now all incoming calls on 2101 are forwarded to 3101 of System 2 when 2101 does not answer the incoming call for RBT- Transfer on No Reply Timer.

- *Call Follow Me does not work in QSIG.*
- *Call Forward-Remote does not work in QSIG.*
- *Call Forward-Dual Ring does not work in QSIG.*
- *Call Forward-External number on QSIG not supported, that is, if '0' is TAC code at System 2 and station 2101 of System 1 dials **131-0-02652630555 (External Number)-#*** then station 2101 gets error tone*.

Call Completion:

Station 2101 of System 1 wants to set CCBS/CCNR (ACB) on station number 3101 of System 2.

- CCBS/CCNR:

  From station 2101 of System 1, dial station number 3101 of System 2. Now dial Access code of ACB (Dial 2) to set CCBS/CCNR on station 3101 of System 2.

Call Intrusion:

Station 2101 of System 1 and station 3101 of System 2 are in speech. Now another station can intrude their conversation using the feature Call Intrusion.

Station 2102 of System 1 wants to intrude the conversation using Call Intrusion on station 2101 of System 1.

- Allow "Raid" feature in Class of Service for station 2102 of System 1.

- Set "Priority" level of station 2102 higher than "Priority" level of station 2101.

- Disable "Privacy from Raid" in Class of Service for station 2101 of System 1.

- Also disable "Privacy from Raid" in Class of Service for station 3101 of System 2.
- Now from station 2102 call 2101. After getting busy tone, dial Access code of Raid (Dial 5). Speech will be established between all three parties 2101, 2102 and 3101.

If "Privacy from Raid" is enabled in Class of Service for station 2101 of System 1 or 3101 of System 2, then Raid will not function.

Station 3102 of System 2 wants to intrude the conversation using Call Intrusion on station 2101 of System 1.

- Allow "Raid" feature in Class of Service for station 3102 of System 2.
- Disable "Privacy from Raid" in Class of Service for station 2101 of System 1.
- Also disable "Privacy from Raid" in Class of Service for station 3101 of System 2.
- Now from station 3102 of System 2 call station number 2101 of System 1. After getting busy tone, dial Access code of Raid (Dial 5). Speech will established between all three parties 2101, 3101 and 3102.

If "Privacy from Raid" is enabled in Class of Service for station 2101 of System 1 or 3101 of System 2, then Raid will not function.

Call Transfer:

Station 2101 and station 2102 of System 1 are in speech. Now station 2101 of System 1 wants to transfer the call to any station number or external number of System 2.

- From station 2101, dial Flash and then dial 3101 (Station number of System 2) or dial 0-02652630555 (External Number).
- Station 2101 goes ON hook or presses 'Transfer' key to transfer the call.
- User can transfer the call after speech is established with the second party or when it is 'Ringing' or 'Busy'.

Call Recall:

Station 2101 of System 1 transfers the call to station 3101 of System 2 when station 3101 of System 2 is 'Ringing' or 'Busy'. Station 3101 does not answer the call.

- Call is returned to station 2101 after expiry of 'Transfer on Busy Timer', in case of transfer- on busy and 'Transfer while Ringing Timer' in case of Transfer- while ringing.

Call Offer:

Station 2101 of System 1 and station 3101 of System 2 are in speech. Now station 2102 of System 1 wants to give Call Offer to station 3101 of System 2.

- Allow "Interrupt Request" feature in Class of Service for station 2102 of System 1.
- From 2102 of System 1, call station 3101 of System 2. After getting busy tone, dial Access code of Interrupt Request (Dial 3).
- Station 3101 of System 2 gets beeps.

If "Privacy from Interrupt Request and Barge-In" is enabled in class of Service for station 3101 of System 2, then Call Offer will not function.

Do Not Disturb (DND):

Station 2101 of System 1 wants to set Do Not Disturb on his/her station.

- Allow "Do Not Disturb" in Class of Service for station 2101 of System 1.
- From station 2101, activate Do Not Disturb by dialing 181.
- Now station 3101's call will not be placed on 2101 of System 1.
- Do Not Disturb-Override (DNDO):

  Station 2101 of System 1 has activated Do Not Disturb. Station 3101 of System 2 calls station 2101 of System 1.

  - Allow "Do Not Disturb- Override" in Class of Service for station 3101 of System 2.
  - Station 3101 of System 2 will be able to call station 2101 of System 1.

If "Privacy from DND Override" is enabled in Class of Service for station 2101, then station 3101's call will not be placed on station number 2101 of System 1.

Advice of Charge (AOC)

AOC is applicable for all external calls made by station of System 1 using trunk of System 2 (OG Gateway).

AOC-Outgoing Call:

Station 2101 of System 1 makes an external call by dialing 0-02652630555 (External Number).

- After completion of call, System 2 calculates the Cost of the call using AOC.
- Cost of call in AOC is calculated by System 2 as per parameters set in Jeeves for 'Call Cost Calculation'.
- If System 2 does not give any charge in AOC then no charge applies on station 2101 of System 1.
- In System 1 no parameters need to be configured.

AOC-Internal Call:

Station 2101 of System 1 calls station 3101 of System 2.

- All calls between two stations of both System are free of charge.

AOC-Call Transfer

Station 2101 and station 2102 of System 1 are in speech. Now station 2101 transfers the call to External Number 02652630555 using trunk of System 2.

- Cost of the call is calculated as per "Call Splitting" option and "When Call Splitting is OFF, charge calls to" option set in SMDR-Outgoing Calls at System 2.
- If "Call Splitting" is enabled in System 2, then charge before the call transfer is applied to station 2101 and charge after the call transfer is applied to station 2102.

- If "Call Splitting" is disabled in System 2, then charge is applied as per "When Call Splitting is OFF, charge calls to".

- If "When Call Splitting is OFF, charge calls to" is set to 'Originating' then all charges before and after the call transfer is applied on station 2101 who has transferred the call.

- If "When Call Splitting is OFF, charge calls to" is set to 'Terminating' then all charges before and after the call transfer is applied on station 2102.

Message Wait Indication (MWI):

Station 2101 of System 1 is an Operator and wants to set Message wait on station 3101 of System 2.

- In the "Station Advanced Feature Template" of System 2 and configure "MW Notification Type" as “Stuttered Dial Tone".

- From station 2101, dial Access Code "1076 - 3101-1", to set Message wait on station 3101 of System 2.

- Now Message wait is set on station 3101. When station user 3101 goes off hook, s/he will hear stuttered dial tone.

  Similarly, you can cancel message wait set on station 3101 by dialing access code "1076-3101-0".

c. Configuring for-IC Call using DDI Routing Over QSIG

If user wants DDI based routing over QSIG then following setup can be made (Not required for normal scenario explained in Basic Call and Supplementary Services).

The setup is explained below:

- The T1E1PRI trunk of System 2 has:
  - Pilot Number with MSN number as 0265304
  - DDI numbers from 2100-2199 and 3100-3199

- Station numbers from 2100 to 2199 are connected at System 1.

- Station numbers from 3100 to 3199 are connected at System 2.
- System 1 and System 2 are connected with each other using QSIG.
- If Calling Party number is received as 0265304-2100 to 0265304-2199 then the call is routed to the stations of System 1. If Calling Party number is received as 0265304-3100 to 0265304-3199 then the call is routed to the stations of System 2.

To configure various parameters in System 1 refer Configuration of "Basic Call" and "Supplementary Services".

In System 2 only the following needs to be done:

- Configuring the OG Trunk Bundle

Template No.	Trunk Port		Start Channel no.	Total Trunk Count	Rotation Type	Automatic Number Translation (ANT)	
	Port Type	Port No.				Apply	ANT Table No.
02	T1E1PRI	001	01	030	Cyclic	Enable	05

- Configuring the External Number List

Location	External No. List 05	External No. List 06
001	026530421	21

- Configuring the DDI based routing.
  - Program Incoming (IC) Reference ID for all Time Zones as "01" in T1E1 Port Parameters.
  - DDI Routing-Incoming Route Reference Table.

Table ID	IC Ref. ID	Start Channel No	Total Channel Count	DDI Routing Ref. ID	Route on First Destination	Ring Timer (Sec)	Route on TLG when No Reply	Route on TLG when Busy	Trunk Feature Templates
01	01	01	30	01	No	045	No	No	01
02	01	01	30	02	No	045	No	No	01

  - DDI Routing-DDI Routing Table

Table ID	Reference ID	Start DDI No.	Total DDI Numbers	DDI No. Digit Count	Port Type	Port Number	Start DDI Flexible No.
001	01	3100	0100	4	FLEXNUM	000	3100
002	02	2100	0100	4	ROUTGRP	009	

- Routing Group

Routing Group	Rotation Flag	Member 01	Member 02	Member 03	Member 04-32
09	Enable	OTBG-02	None	None	None

Basic Call:

Below examples explains the calls made/received using DDI Routing.

- External Party call the number 0265304-2101

  Call lands directly on station 2101 of System 1 through QSIG.

- External Party calls the number 0265304-3101

  Call lands on station 3101 of System 2.

Voice Mail for Extensions over QSIG

SARVAM UCS supports Voice Mail for extension users connected over QSIG. These are extensions of the remote System (may be SARVAM UCS or any other System) connected with SARVAM UCS over QSIG.

To provide Voice Mail to the remote system extension users you must configure the Voice Mail Settings.

For details, see “Extensions Over Q-SIG”.

Quick Dial

What's this?

Quick Dial provides DKP and Extended IP phone users the facility of 'One-touch' dialing of numbers stored in their Personal Directory and the Global Directory.

How it works

Quick Dial is based on "Abbreviated Dialing".

To be able to Quick Dial a number,

- the number must exist in the Personal or Global Directory assigned to the extension.
- Personal and Global Directory dialing must be allowed in the Class of Service of the extension.
- On the DKP and Extended IP Phones, DSS keys must be configured with the Short Codes or Abbreviated Numbers that are to be dialed out. These short codes are derived from the Index numbers of the Personal Directory and the Memory Location Index of the Global Directory.
- You can Quick Dial a number simply by pressing the DSS key.
- The system locates the number to be dialed out in the Personal/Global Directory on the basis of the Index Number/Memory Location Index configured on the DSS Key.

How to configure

See "Abbreviated Dialing" for instructions on configuring and assigning the Personal and Global Directories.

To assign the Short Codes or Abbreviated Numbers to be used for Quick Dial on DSS keys, for each DKP/ Extended IP Phone extension,

- List down the numbers from the Personal Directory and Global Directory to be used for Quick Dial.
- If the number is from the Personal Directory assigned to the extension, note the Index number at which it is stored in the Personal Directory: 001 to 025.
- If the number is from the Global Directory assigned to the extension, note the Memory Location Index at which it is stored in the Global Directory: 100 to 999.
- Now, configure the Quick Dial numbers on the DSS keys of the DKP/ Extended IP Phone.

  For detailed instructions on configuring DSS Keys on a Digital Key phone, see "DSS Keys Programming".

  For instructions on configuring DSS Keys on Matrix Extended IP Phone, see "Configuring SIP Extension Settings as per the Extended Phone Type" under "Configuring SIP Extensions".
- On the desired key, select **Quick Dial** as **Function Type**.

- As **Offset**, select the Index Number against which the number is stored in the Personal/Global Directory.

How to use

For EON and Extended IP Phone Users only

- Press the DSS Key assigned to the Quick Dial numbers.
- The number will be dialed out.
- Talk when the called party answers.

Raid

What's this?

Raid allows you to interrupt a telephone conversation between two extension users, turning the conversation into a three-way call.

You can use Raid to land in a conversation between two extension users, or between an extension user and an external caller, with a warning beep to the extension user. The extension user will hear a beep when you raid and you will enter in to three-way speech with both parties.

You may also Raid a conversation without any warning by disabling the beep.

How it works

- A, B and C are extension users.
- A and B are in speech.
- C calls A.
- C gets busy tone.
- C dials the feature access code for Raid.
- Beep is played on A. Three-way speech is established between A, B and C.
- If any of these three parties disconnects, two-way speech is established between the remaining parties.

Feature Interactions

- Raid works only if the dialed extension is busy in two-way speech. The two-way speech may be with another extension or with an external number on a trunk. However, it will not work if the conversation is being recorded.
- You cannot Raid on Trunks, that is, if two external numbers are in two-way speech. In this case, C cannot raid the conversation.
- Raid will not work if **Privacy against Raid** is enabled in the Class of Service of the extension being raided. In this case, if Extension A has Privacy against Raid in its Class of Service, C will not be able to Raid the conversation between A and B. To know more about this feature, see "Privacy".
- The extension using Raid must have higher Priority assigned to it than the extension being raided. In this case, C must have higher Priority than A to be able to invoke Raid.
- Raid will not work when the two-way conversation between the users is being taped.

Raid is a sensitive feature. You are advised to restrict access to this feature to selective extension users.

How to configure

To be able to use Raid, extension users must have this feature enabled in the “Class of Service (COS)” assigned to them for the time zones in their “Station Basic Feature Template”.

By default, beep is played as a warning to the extension being raided. If required, you may disable the beep played during Raid, by clearing the **Play Beep when Raid/Conference/Dial-In Conference begins** check box in the **System Parameters**. For instructions, see “System Parameters”.

How to use

For EON & Extended IP Phone Users

When dialed extension is busy,

- Press DSS Key assigned to Raid.
  OR
- Dial 5 on Busy Tone.

For SLT Users

When dialed extension is busy,

- Dial 5 on Busy Tone.

RCOC (Return Call to Original Caller)

This feature is not applicable if CDMA Mobile Card is installed in your system.

What is this?

Generally, extensions users of the System are given a trunk access to make outgoing calls from their phones. It is also common for a group of extensions to share the same trunks to make outgoing calls.

When an extension user of the System makes an outgoing call and the called party does not answer the call or is busy on another line, it is possible for the called party to return the call (made by the extension user) on the basis of the CLI number received.

However, when the called party returns the call, this incoming call is mostly likely to land on the Operator extension, as incoming calls are usually routed to the Operator.

Now, the Operator has no way of knowing which extension made the call so as to transfer the call to that extension.

Instead, the Operator must either ask the called party whom they wish to speak to and transfer the call or put the called party on hold and find out the extension that made the call. This is an unwieldy process for all concerned - the Operator, the called party and the extension user who originally made the call.

This can be overcome if the System is able to route the returned call to the original caller's extension.

SARVAM UCS makes this possible with the Return Call to Original Caller feature.

This feature is supported on Mobile ports, BRI, T1E1PRI (T1E1) and SIP Trunks.

RCOC is not supported when calls are made from analog trunks — CO and E&M — due to the signaling limitations of these trunks.

How it works

The Prerequisites

- RCOC is enabled on the desired Trunk/s - BRI, T1E1PRI, Mobile and SIP.
- RCOC is enabled in the Class of Service group assigned to the extension.

RCOC Table

RCOC table is maintained internally by the SARVAM UCS and it is non-programmable.

- SARVAM UCS can keep a record of 255[322] entries in the RCOC Table.
- Each entry is kept for the duration of the RCOC Record Delete Timer (programmable; default: 999 minutes). Whenever a record is stored in the RCOC database, the Record Delete Timer for that entry is activated. On the expiry of the Timer, the entry is deleted by the system.

322. ETERNITY LENX/MENX can store 512 entries in the RCOC table.

- If a same external number is dialed using 3 different SIP/Mobile Ports with RCOC enabled on all the Trunks and Extensions, then if the called party calls back, the call will land on the original callers using FIFO logic.

- Each record is deleted from the database either after the call is returned or on expiry of the Record Delete Timer.

- The RCOC database remains unaffected during power outages.

The Process

- When an extension having RCOC feature in its Class of Service makes an out going call, the system checks if RCOC is enabled on the trunk through which the outgoing call is routed.

- If RCOC is enabled on the trunk, the system stores the record of the outgoing call in the RCOC Table.

- The system sets RCOC for the outgoing call in the following conditions, according to the Destination Port[323]:

    - If the Destination Port is a Mobile Port, RCOC is set when:
        - called party is busy.
        - called party is out of coverage/mobile is switched off.
        - called party does not reply.
        - called party rejects the call.
        - caller (extension that made the call) goes ON-Hook before the called party answers the call.

    - If the Destination Port is a BRI or T1E1PRI Port or a SIP Trunk, RCOC is set when:
        - called party is busy.
        - called party does not answer the call.
        - caller (extension that made the call) goes ON-Hook before the called party answers the call.

    - When SARVAM UCS acts as Gateway,
        - If the Originating Port is either BRI-NT or T1E1-NT, the system checks the Class of Service allowed to the trunk port.
        - If the Originating Port is a trunk (T1E1-TE, BRI-TE, CO, MOBILE, SIP), RCOC will be set, if it is enabled on Destination Port.
        - RCOC shall be set only if the Calling Party's Number is available. If calling party number is missing, then RCOC shall not be set.

While returning call to the original caller, if you want SARVAM UCS to match the Trunk Port type and Trunk Port number of the incoming call with the Trunk Parameters of the entry stored in the RCOC table, then you must enable ***Apply RCOC only if the caller calls back on the same trunk from which the call was made*** *option in the System Parameters.*

- Whenever there is an incoming call on any trunk, the system checks the ***Apply RCOC only if the caller calls back on the same trunk from which the call was made*** *flag in the System Parameters.*

    - If enabled, the system matches the Trunk Port number and the Trunk Port type of the incoming call with the entry stored in the RCOC Table.

    - If disabled, the system matches the CLI of the incoming call with the entry stored in the RCOC Table.

---

323. The Trunk from which the outgoing call is made.

- If a matching record entry is found, the system routes the call to the original caller and clears the record entry from the RCOC Table.

- The return call rings on the original caller's extension for the period of the Ring Back Tone Timer (programmable; default 45 seconds). If the original caller does not answer the call within this Timer, the call is routed to the Trunk Landing Group programmed for that trunk.

  The Ring Back Tone Timer is common to all internal calls; calls made from one extension will ring on the destination extension till the end of this timer. Change in the Ring Back Tone Timer for RCOC returned calls on original caller's extension will also be applied on Ring Back Tone Timer for all internal calls. So, change this Timer taking this into consideration.

- If no match is found in the RCOC Table or the extension or the original caller is busy, the call will be routed according to the incoming call logic programmed (as programmed in the assigned Trunk Feature Template) in the system.

- *As RCOC is a "Class of Service (COS)" dependent feature, extensions that are not allowed this feature in their COS cannot have their calls returned; even if this feature is enabled on the Trunk they used to make the call.*

- *In case of Call Transfer, RCOC will be set for the extension on which the call is transfered.*

Feature Interaction: RCOC and DISA CLI Authentication

- When DISA CLI Authentication (Multiple Calls or One Call) is enabled on a trunk, whenever there is an incoming call on the trunk, the system will first check the DISA CLI Authentication Table.

- If a matching entry is found in the DISA CLI Authentication table, the system will give dial tone to the caller.

- The caller can now invoke RCOC feature by dialing ** (pressing Star key twice).

  OR

- The caller can make calls to a station or an external number or use a feature as required.

- If the caller invokes RCOC feature by dialing ** (pressing Star key twice), the system will check the RCOC Table.

- If a matching record entry is found, the system routes the call to the original caller and clears the record entry from the RCOC Table.

How to configure

For this feature to work, it must be enabled on the Trunk and in the Class of Service of the extensions. See "Enabling RCOC on Trunk using Jeeves", "Enabling RCOC on Trunk using SE Command" and "RCOC in Class of Service" for instructions.

If desired, the related Timers, that is, the RCOC Record Delete Timer and the Ring Back Tone Timer may also be changed. See "System Timers and Counts" for instructions.

If you want SARVAM UCS to match the Trunk Port type and Trunk Port number of the incoming call with the Trunk Parameters of the entry stored in the RCOC table, then you must enable **Apply RCOC only if the caller calls**

back on the same trunk from which the call was made flag in the System Parameters. See “System Parameters” for instructions.

RCOC on Trunk

Enabling RCOC on Trunk using Jeeves

- Login as System Engineer.
- Click the trunk parameters of the trunk type on which you want to enable this feature, namely:
  - SIP Parameters
  - BRI Parameters
  - T1E1 Port Parameters (T1E1PRI)
  - MOBILE Port Parameters
- The page containing the trunk parameters will open.
- Select the 'Return Call to Original Caller (RCOC)' check box on the page to enable this feature on the desired trunk port.
- Click 'Submit' at the bottom of the page to save changes.
- Log out of Jeeves or continue with other programming tasks.

Enabling RCOC on Trunk using SE Command

- Enter SE mode from a DKP/SLT.

  To enable RCOC on T1E1 Trunk
  - Dial ***6145-1-T1E1-Code*** to enable the feature on a single trunk.
  - Dial ***6145-2-T1E1-T1E1-Code*** to enable the feature on a range of trunks.
  - Dial ***6145-*-Code*** to enable the feature on all trunks.
    Where,
    T1E1 is the software port number of the trunk from 01 to 08.
    Code is
    0 for Disable
    1 for Enable
    Default: Disable

  To enable RCOC on BRI Trunk
  - Dial ***6220-1-BRI-Code*** to enable the feature on a single BRI trunk.
  - Dial ***6220-2-BRI-BRI-Code*** to enable the feature on a range of BRI trunks.
  - Dial ***6220-*-Code*** to enable the feature on all BRI trunks.
    Where,
    BRI is the software port number of the BRI trunk from 01 to 32.
    Code is
    0 for Disable
    1 for Enable
    Default: Disabled

To enable RCOC on Mobile Port

- Dial ***8030-1-Mobile -Code*** to enable the feature on a single Mobile port.
- Dial ***8030-2-Mobile-Mobile-Code*** to enable the feature on a range of Mobile ports.
- Dial ***8030-*-Code*** to enable the feature on all Mobile ports.
  Where,
  Mobile is the software port of the Mobile port from 01 to 32.
  Code is
  0 for Disable
  1 for Enable
  Default: Disabled

To change the RCOC Record Delete Timer

- Dial ***3521-Minutes***
  Where,
  Minutes are from 001 to 999
  Default: 999 minutes

To change the Ring Back Tone Timer

- Dial ***3503-Seconds***
  Where,
  Seconds is from 001 to 255.
  Default: 045 seconds.

- Exit SE mode.

RCOC in Class of Service

The feature 'RCOC' must be enabled in the "Class of Service (COS)" group assigned to the extensions for returned calls to land on them.

In the default Station Basic Feature Template 01 assigned to all extensions of the SARVAM UCS, the default Class of Service group 01 has the feature "RCOC" enabled. So, all extensions of SARVAM UCS are by default allowed this feature.

There is no need to program this feature if all extensions are to be allowed this feature.

However, if you want to deny this COS feature to certain extensions and allow this feature to all other extensions, follow these steps:

1. Define a CoS group with RCOC disabled.
2. Prepare a Station Basic Template with this CoS group applicable in all the time zones.
3. Assign this newly prepared Station Basic Feature Template to the extensions on which 'RCOC' is to be disabled.

Refer the topic "Class of Service (COS)" and "Station Basic Feature Template" for instructions.

Real Time Clock (RTC)

What's this?

Various features and facilities supported by SARVAM UCS, such as Alarms, Station Message Detail Records, System Activity Log, Time Zones, Daylight Savings, certain Voice Mail features need the correct time and date for their proper functioning.

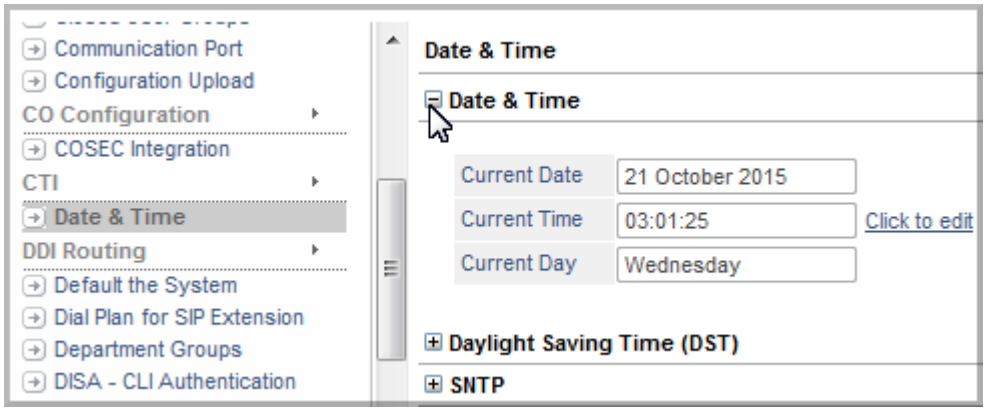
The SARVAM UCS has a built-in Real Time Clock (RTC) circuit that maintains date and time. When you select Region, the RTC is automatically set to the current date and time of the country/region where SARVAM UCS is installed.

Since the RTC circuit may drift over a period, it is recommended that you check and reset RTC values at least once every month to correct this drift. The RTC of SARVAM UCS takes care of leap years.

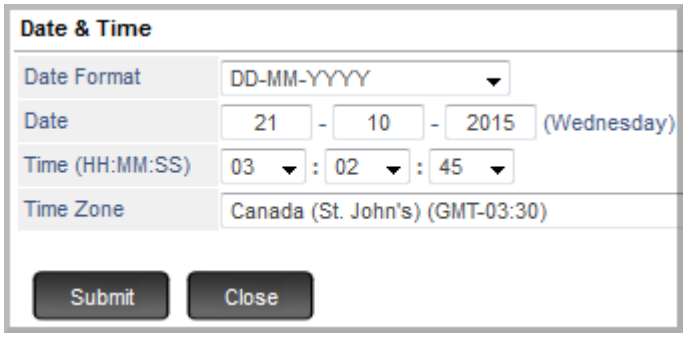
How to Configure

Configuring RTC using Jeeves

- Log in as System Engineer.
- Click **Configuration**, click **Date and Time**.
- Click **Date and Time** to expand.



- Click on **Click to edit** to set the date and time as per your requirement.



- Select the desired **Date Format**:
- Enter the current **Date**
- The current **Day** will appear as per the date you set.
- Set the current **Time** in HH:MM:SS (hours-minutes-seconds) format.
- Select **Time Zone** of the country/region where SARVAM UCS is installed.
- Click **Submit** to save RTC settings.
- You may log out of Jeeves

Configuring RTC using a Telephone

- Enter SE mode from a DKP/SLT.

To set the date format, dial:

- ***1000-Date Format***
  Where,
  Date format is
  1 for DD-MM-YYYY
  2 for MM-DD-YYYY
  Default: 1.

To set date, dial:

- ***1001-Date-Month-Year***
  Where,
  DD is 01 to 31 (leading zero mandatory for single digit date).
  MM is 01 to 12 (leading zero mandatory for single digit month).
  YYYY is year in four digits from 0000 to 9999.
  Default: Current date is set.

First set the date format and then set the date.

To set the current time, dial:

- ***1003-Hours-Minutes-Seconds***
  Where
  Hours is 00 to 23 (24-hour format in two digits)
  Minutes is 00 to 59 (in two digits)
  Seconds is 00 to 59 (in two digits)
  Default: Current Time is set.

To select a Time Zone, dial:

- ***1002-Time Zone Index Number***
  Where,
  Time Zone Index Number is from 001 to 124, refer the table below.
  Default: Index 050 - India.

- Exit SE mode.

Time Zones

Time Zone	Index
(GMT-12:00) US Minor Outlying Islands (Baker Island, Howland Island)	135
(GMT-11:00) American Samoa	126
(GMT-10:00) United States (Hawaii)	116
(GMT-09:30) French Polynesia	130
(GMT-09:00) United States (Juneau)	115
(GMT-08:00) United States (Las Vegas, Los Angeles, Phoenix, San Francisco, Seattle)	114
(GMT-08:00) Mexico (Tijuana)	072
(GMT-08:00) Canada (Vancouver)	031
(GMT-07:00) United States (Albuquerque, Boise, Cheyenne, Denver, Salt Lake City)	113
(GMT-07:00) Mexico (Chihuahua)	071
(GMT-07:00) Canada (Calgary)	030
(GMT-06:00) United States (Chicago, Dallas, Des Moines, Memphis, Minneapolis, New Orleans, Oklahoma, Omaha, St. Louis)	112
(GMT-06:00) Mexico (Mexico City)	070
(GMT-06:00) Costa Rica	035
(GMT-06:00) Canada (Winnipeg)	029
(GMT-05:00) United States (Atlanta, Augusta, Boston, Charlotte, Columbus, Detroit, Indiapolis, Miami, NY, Philadelphia, Washington)	111
(GMT-05:00) Peru	085
(GMT-05:00) Cuba	037
(GMT-05:00) Colombia	034
(GMT-05:00) Canada (Montreal, Ottawa, Toronto)	028
(GMT-05:00) Brazil (Acre)	022
(GMT-05:00) Bahamas	009
(GMT-04:30) Venezuela	118
(GMT-04:00) Paraguay	084
(GMT-04:00) Guyana	047
(GMT-04:00) Chile	032
(GMT-04:00) Canada (Halifax)	027
(GMT-04:00) Brazil (Manaus)	021
(GMT-04:00) Bolivia	015
(GMT-04:00) Antigua & Barbuda	003

Time Zone	Index
(GMT-03:30) Canada (St. John's)	026
(GMT-03:00) Brazil (Brasilia, Rio de Janeiro, Sao Paulo)	020
(GMT-03:00) Argentina	004
(GMT-02:00) Brazil (Fernando De Noronha)	019
(GMT-01:00) Cape Verde (Cabo Verde)	129
(GMT) Ireland	054
(GMT) Portugal	088
(GMT) United Kingdom	110
(GMT+01:00) Algeria	002
(GMT+01:00) Austria	008
(GMT+01:00) Belgium	013
(GMT+01:00) Bosnia & Herzegovina	016
(GMT+01:00) Cameroon	025
(GMT+01:00) Cote d'Ivoire	125
(GMT+01:00) Croatia	036
(GMT+01:00) Czech Republic	039
(GMT+01:00) Denmark	040
(GMT+01:00) France	044
(GMT+01:00) Germany	045
(GMT+01:00) Italy	056
(GMT+01:00) Namibia	076
(GMT+01:00) Netherlands	078
(GMT+01:00) Nigeria	080
(GMT+01:00) Norway	081
(GMT+01:00) Poland	087
(GMT+01:00) Slovakia	095
(GMT+01:00) Spain	097
(GMT+01:00) Sweden	100
(GMT+01:00) Switzerland	101
(GMT+02:00) Belarus	012
(GMT+02:00) Botswana	017
(GMT+02:00) Bulgaria	023
(GMT+02:00) Cyprus	038
(GMT+02:00) Egypt	041

Time Zone	Index
(GMT+02:00) Finland	043
(GMT+02:00) Greece	046
(GMT+02:00) Hungary	049
(GMT+02:00) Israel	055
(GMT+02:00) Jordan	058
(GMT+02:00) Lebanon	065
(GMT+02:00) Libya	066
(GMT+02:00) Mozambique	074
(GMT+02:00) Romania	090
(GMT+02:00) South Africa	096
(GMT+02:00) Syria	102
(GMT+02:00) Turkey	106
(GMT+02:00) Ukraine	108
(GMT+02:00) Yugoslavia	121
(GMT+02:00) Zambia	122
(GMT+02:00) Zimbabwe	123
(GMT+03:00) Yemen	120
(GMT+03:00) Bahrain	010
(GMT+03:00) Iraq	053
(GMT+03:00) Kenya	060
(GMT+03:00) Kuwait	063
(GMT+03:00) Qatar	089
(GMT+03:00) Russia (Moscow, St. Petersburg)	091
(GMT+03:00) Saudi Arabia	124
(GMT+03:00) Sudan	099
(GMT+03:00) Uganda	107
(GMT+03:30) Iran	052
(GMT+04:00) Mauritius	069
(GMT+04:00) Oman	082
(GMT+04:00) United Arab Emirates	109
(GMT+04:30) Afghanistan	001
(GMT+05:00) Maldives	068
(GMT+05:00) Pakistan	083
(GMT+05:00) Tajikistan	104

Time Zone	Index
(GMT+05:00) Uzbekistan	117
(GMT+05:30) Sri Lanka	098
(GMT+05:30) India	050
(GMT+05:45) Nepal	077
(GMT+06:00) Bangladesh	011
(GMT+06:00) Bhutan	014
(GMT+06:00) Kazakhstan	059
(GMT+06:00) Kyrgyzstan	064
(GMT+06:00) Russia (Novosibirsk)	092
(GMT+06:30) Myanmar	075
(GMT+07:00) Cambodia	024
(GMT+07:00) Indonesia	051
(GMT+07:00) Thailand	105
(GMT+07:00) Vietnam	119
(GMT+08:00) Australia (Perth)	005
(GMT+08:00) Brunei	018
(GMT+08:00) China	033
(GMT+08:00) Hong kong	048
(GMT+08:00) Malaysia	067
(GMT+08:00) Mongolia	073
(GMT+08:00) Philippines	086
(GMT+08:00) Singapore	094
(GMT+08:00) Taiwan	103
(GMT+08:45) Australia (Eucla)	127
(GMT+09:00) Japan	057
(GMT+09:00) Korea-North	061
(GMT+09:00) Korea-South	062
(GMT+09:30) Australia (Adelaide)	006
(GMT+10:00) Australia (Brisbane, Canberra, Melbourne, Sydney)	007
(GMT+10:00) Russia (Vladivostok)	093
(GMT+10:30) Australia (Lord Howe Island)	128
(GMT+11:00) Solomon Island)	134
(GMT+12:00) Fiji	042
(GMT+12:00) New Zealand (Auckland, Wellington)	079

Time Zone	Index
(GMT+12:45) New Zealand (Chatham Islands)	132
(GMT+13:00) Samoa	133
(GMT+14:00) Kiribati	131

Redundancy

What's this?

In today’s competitive era, the key motto of all the organizations is to provide best services to their clients and to achieve this, system’s reliability is the basic need of all the organizations. As time is a valuable asset for an organization, any kind of unplanned system downtime or system failure is not acceptable, as it effects the organization’s proficiency. Moreover, an unreliable system may cause delay in response time which may further lead to loss of business, clients and reputation of an organization. To meet this paramount need of the modern organizations, SARVAM UCS supports Redundancy.

In general, Redundancy is the process of restoring the functionality of a system, when any specific part or unit, which is critical to the functioning of the system fails. This is achieved by creating a replica of these specific units in the system. In other words, redundancy is the transferring of control from an active critical unit to a standby critical unit in a system.

SARVAM UCS supports redundancy for its two critical cards - Power Supply Card and CPU Card. To simplify, SARVAM UCS supports two types of redundancy:

- Power Redundancy
- Application and Hardware Redundancy

- *Redundancy is supported in ETERNITY LENX/MENX only.*
- *During Redundancy, all the mature calls will be retained in the system. However, calls other than the mature state will be dropped. Moreover, the call information and configurations made in the system will also be retained.*

Power Redundancy

SARVAM UCS supports two Power Supply Cards — Power Supply Card 1 and Power Supply Card 2.

Power Supply Card 1 acts as the Active[324] Card and the Power Supply Card 2 acts as a Standby to this Power Supply Card 1.

When the Power Supply Card 1 fails, the Power Supply Card 2 takes the control and becomes Active, thus providing persistent power supply for the proper functioning of the system.

After this Power Supply Card 1 is fixed, it now acts as a Standby to the Power Supply Card 2. In case, the Power Supply Card 2 fails, the Power Supply Card 1 takes the control and becomes Active again.

This procedure of transferring control from Power Supply Card 1 to Power Supply Card 2 and vice versa is termed as Power Redundancy.

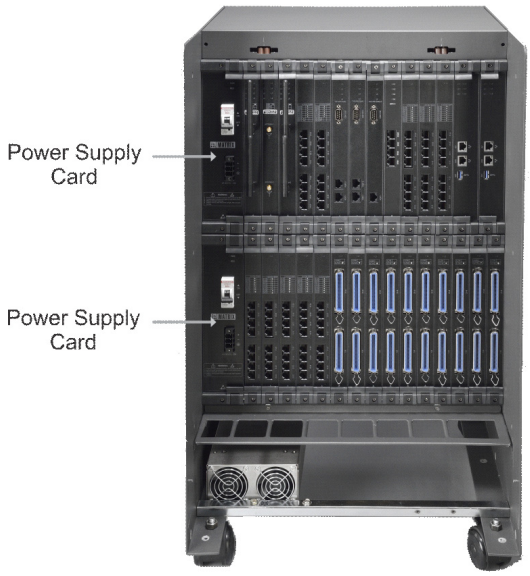
Power Redundancy is an automatic process and does not require user intervention. No prior configuration is required for power redundancy. However, for power redundancy, make sure you install two Power Supply Card, that is Power Supply Card 1 and Power Supply Card 2 properly in the system.

For Power Redundancy in ETERNITY LENX, install two PS48V DC Power Supply Cards.

324. The Power Supply Card which is detected first, when the system is powered on.

For Power Redundancy in ETERNITY MENX, install two PS48V DC Cards or two PS UNI Cards.

Illustrated below is the diagram of ETERNITY LENX and ETERNITY MENX for better understanding.



To know more about the power supply cards of ETERNITY LENX and ETERNITY MENX, refer to “The Power Supply Card” in Installing ETERNITY LENX and “The Power Supply Card” in Installing ETERNITYMENX respectively.

Hardware and Application Redundancy

SARVAM UCS supports two CPU[325] Cards — CPU Card 1 and CPU Card 2.

CPU Card 1 acts as the Active[326] Card and CPU Card 2 acts as a Standby Card.

When the CPU Card 1 fails, the CPU Card 2 takes the control and becomes Active, thus providing uninterrupted communication.

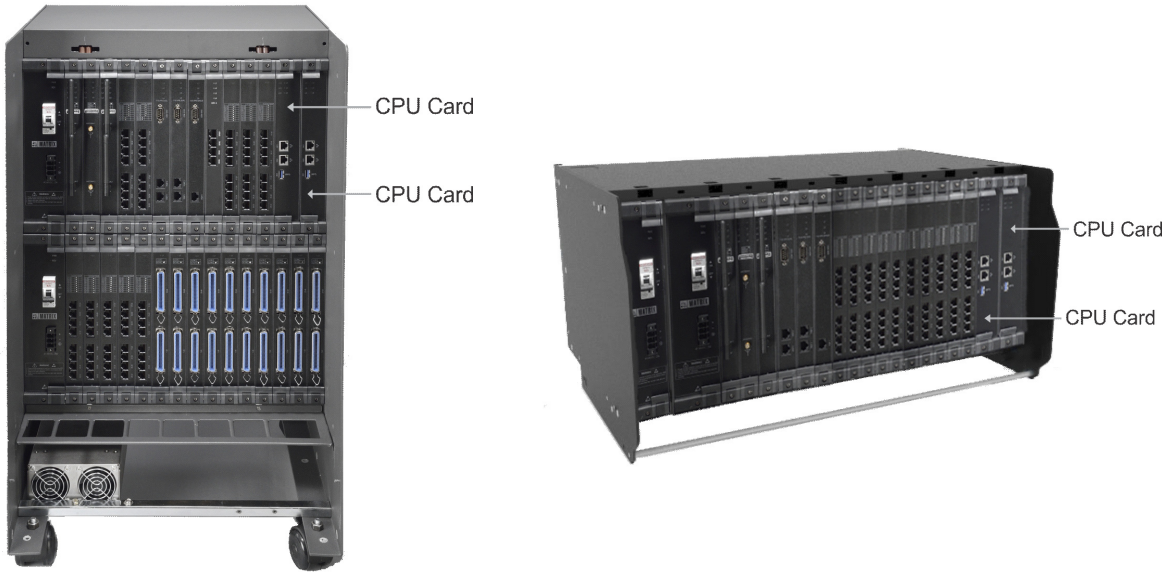
After this CPU Card 1 is fixed, it now acts as a Standby to the CPU Card 2. In case, the CPU Card 2 fails, the CPU Card 1 takes the control and becomes Active again.

This procedure of transferring control from CPU Card 1 to CPU Card 2 and vice versa is termed as Hardware and Application Redundancy.

For Hardware and Application Redundancy in ETERNITY LENX/MENX, install two CPU Cards.

325. *In case of ETERNITY LENX, make sure both the CPU Cards have the same IDT switches — 1K or 2K, else Redundancy will not be supported.*
326. *The CPU Card which is detected first, when the system is powered on.*

Illustrated below is the diagram of ETERNITY LENX and ETERNITY MENX for better understanding.



For Hardware and Application Redundancy, the system must have,

- Two CPU Cards - CPU Card 1 and CPU Card 2.
- Redundancy License, refer to “License Management”.
- Configured the Redundancy Parameters, refer to “Configuring Redundancy Parameters”.

Preferably, you must activate all the required licenses along with Redundancy License in one CPU Card, say CPU Card 1.

Let us understand Hardware and Application Redundancy in detail with the help of an example.

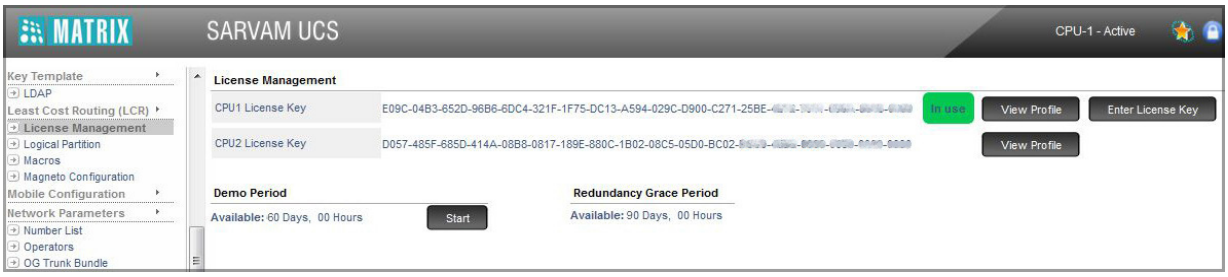
The below example is explained assuming that you have fulfilled the above listed pre-requisites for Hardware and Application Redundancy.

Assume that, you have activated all the required licenses along with the Redundancy License in CPU Card 1 and only default licenses are available in CPU Card 2. Lets say you have activated CTI and SMS Gateway licenses in CPU Card 1 and the same is not activated in CPU Card 2.

To have more clarity in understanding this example,

- Login as System Engineer.

- Under **Configuration**, click **License Management**.



Currently, the system is using the licenses as activated in the license key of CPU Card 1. This is because redundancy license is activated in CPU Card 1.

In case, where redundancy license is activated in both CPU Card 1 and CPU Card 2, the system will use the license key of the CPU Card which is installed in slot number 1. You can change the license key used by the system by selecting the license key of the desired CPU Card. To know more about this, refer to case 2 in Scenario 2, explained later in the topic.

The 'In Use' tag indicates the license key of the CPU Card which is currently used by the system. This implies the system will use the licenses as activated in this license key.

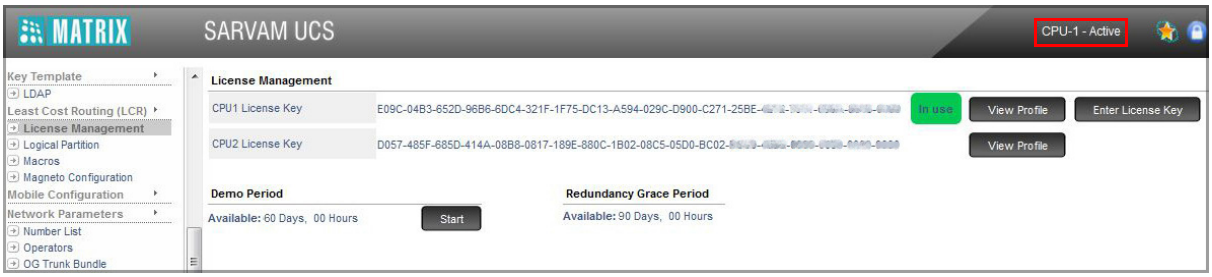
To view the list of licenses activated in CPU Card 1, click the **View Profile** button of CPU Card 1. Similarly, you can also view the default licenses present in the CPU Card 2. To do so, click the **View Profile** button of CPU Card 2.

You can also compare the licenses activated in both the CPU Cards, to do so, click the **View Profile** button of any of the desired CPU Card and click the **Compare All Licenses** check box.

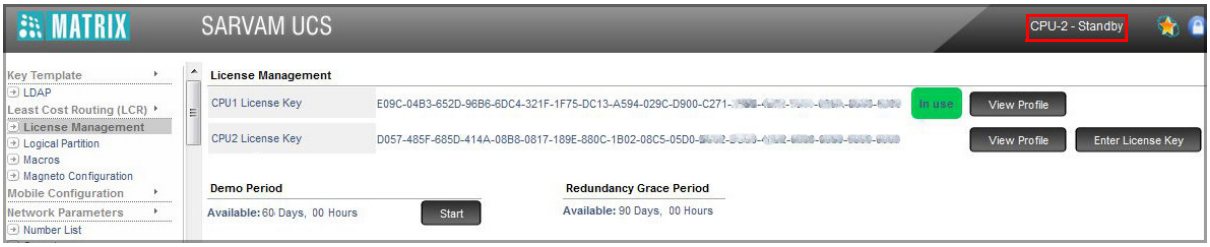
Service Profile	As per CPU1	As per CPU2
SARVAM UCS ENT	Yes	No
Expansion Slots	1-28	1-8
Vocoder Channels	64	4
VMS Channels	16	4
IP Subscribers	505	5
VARTA Essential Users	500	0
VARTA Professional Users	500	0
VARTA Collaboration Users	500	0
PLCC	Yes	No
Hospitality	Yes	No
Hospitality E911	Yes	No
PMS	Yes	No
PMS	Yes	No
Gateway	Yes	No
SMS Server	Yes	No
CTI	Yes	No
SMS Gateway	Yes	No
Redundancy	Yes	No

You can see, that Redundancy, CTI and SMS Gateway Licenses are activated in CPU Card 1 and the same is not activated in CPU Card 2.

Currently, CPU Card 1 is the Active Card.



The CPU Card 2 is the Standby Card.



To know about the IP address of the Standby Card, refer to “Redundant CPU IP Address and Port”.

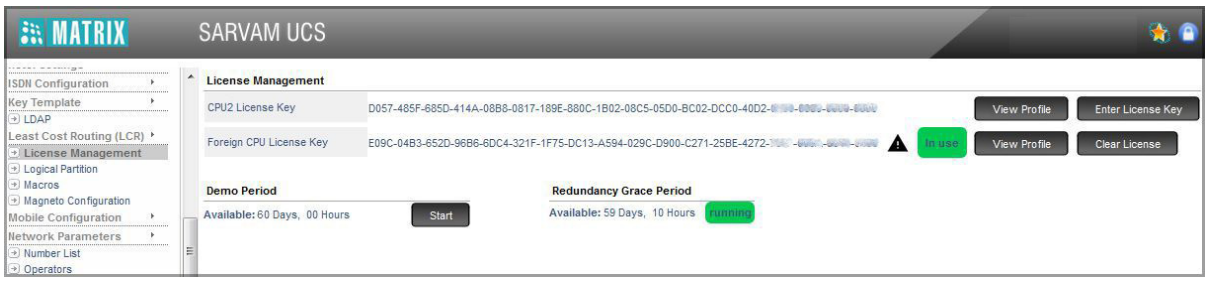
The scenario explained above is before Redundancy.

Now, when the CPU Card 1 fails, the CPU Card 2 takes over the control and becomes Active. The system will continue using the licenses as activated in the license key of CPU Card 1. Therefore, all the licenses activated in CPU Card 1 will be applicable to CPU Card 2. However, the same will not be reflected in the Service Profile of CPU Card 2.

You can use all the features and facilities as per the licenses activated in the license key of CPU Card 1, even though these are not activated in the license key of CPU Card 2, for example: CTI and SMS Gateway Licenses. However, you can use the features and facilities as activated in the licenses key of CPU Card 1 only for a limited period of time, known as redundancy grace period. This redundancy grace period is of 90 days and is decremented from CPU Card 2.

The redundancy grace period will start, when the CPU Card 1 is down and is not detected by the system within 10 minutes. So, in case, if you restart the system and the same is up within 10 minutes, the redundancy grace period will not be applicable.

As CPU Card 1 is no longer detected by the system, it is treated as a foreign card and the license key of the same is denoted as the Foreign CPU License Key.

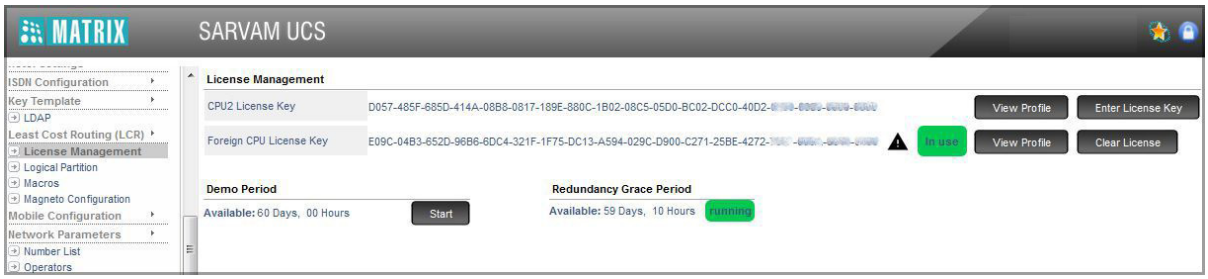


Now, let us take this example further to understand the different scenarios that may arise:

Scenario 1: The CPU Card 2 is active and the CPU Card 1 is detected as faulty and is sent to the technical support for assistance.

In this scenario, the CPU Card 2 will act as a standalone card, using the licenses as activated in the license key of CPU Card 1. These licenses of CPU Card 1 can be used only for the redundancy grace period, which is of 90 days.

The below screen displays the above explained scenario, where the CPU Card 2 is the standalone card in the system and is using the licenses of CPU Card 1(Foreign CPU License Key).



Let us consider CPU Card 1 is fixed within 10 days. Now, when this CPU Card 1 is inserted in the system, it acts as a Standby Card to the CPU Card 2. The CPU Card 2 will still act as an Active Card. The redundancy grace period will stop and you can still use the licenses as activated in the license key of CPU Card 1.

This will again create the ideal condition to support redundancy.

*For **Firmware Version later than V1R6.7** the Grace period will be incremented till it reaches 90 days for both the cards — CPU Card 1 and CPU Card 2.*

Now, when this CPU Card 2 fails, the CPU Card 1 becomes Active. However, in this case, the redundancy grace period will not be applicable as the system is using the license key of the CPU Card 1, which is already present in the system.

Redundancy grace period will be applicable only when the system uses the license key of the CPU Card, which is not present or detected by the system.

- *If you start the demo mode, the redundancy grace period will stop and the system will use the licenses allowed in demo mode. To view the licenses allowed in demo mode, click the **View Profile** button after starting the demo mode.*
- *You must activate redundancy license in the CPU Card in which you have activated the maximum number of licenses. This is important as during redundancy, the system will allow you to use the set of features and facilities as per the licenses activated in that CPU Card.*

Scenario 2: The CPU Card 2 is active and the CPU Card 1 is permanently detected as faulty.

In this scenario, the CPU Card 2 will act as an active standalone card, using the features and facilities as per the licenses activated in the Foreign CPU License Key (license key of CPU Card 1).

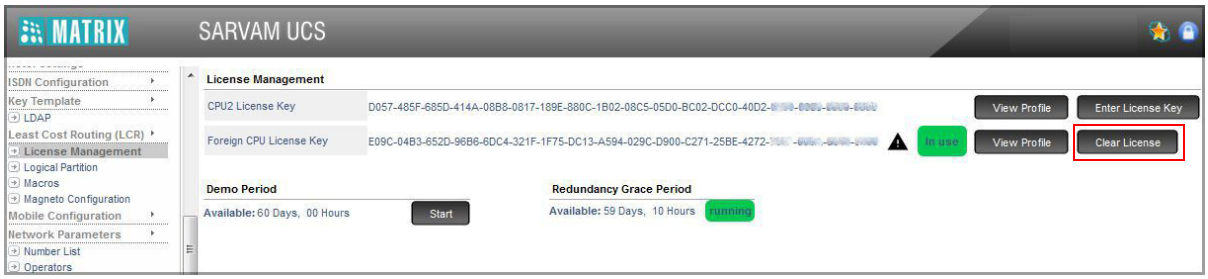
The system can use the features and facilities as per the licenses activated in Foreign CPU License Key only for a redundancy grace period, which is of 90 days. On the expiry of this redundancy grace period, you can use only the default licenses present in CPU Card 2.

Let us take this scenario further and understand different cases that may arise. The below cases are explained considering the redundancy grace period has not expired.

Case 1: You want to use CPU Card 2 as the standalone card without supporting redundancy temporarily.

Since the CPU Card 2 is using the features and facilities as per the licenses activated in the Foreign CPU License Key (license key of CPU Card 1), the redundancy grace period will be applicable.

If you want to use this CPU Card 2 as the standalone card, you must clear the Foreign CPU License Key (license key of CPU Card 1). To do so, click the **Clear License** button. As soon as you clear the Foreign CPU License Key, the redundancy grace period will stop and the system will now allow you to use the features and facilities as per the licenses activated in CPU Card 2.



Let us take this example further considering you have not cleared the Foreign CPU License Key.

As this Foreign CPU License Key is not cleared, the system will keep on using the licenses activated in this license key at the cost of the redundancy grace period. On the expiry of this redundancy grace period, you can use only the default licenses present in CPU Card 2.

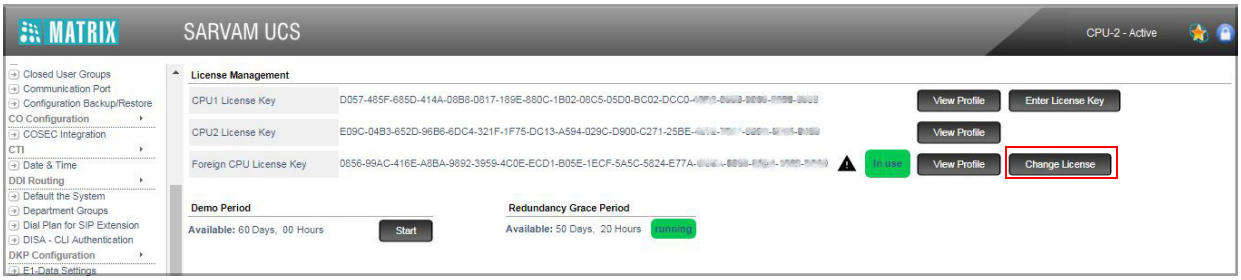
Case 2: You want to install a New CPU Card 1 in the system so as to support redundancy.

Currently, the CPU Card 2 (the card only present in the system) is using the licenses as per the Foreign CPU License Key (license key of CPU Card 1). Now, you install a new CPU Card, say New CPU Card 1 in the system which has all required licenses along with the redundancy license activated.

You can preserve the redundancy grace period by using the licenses activated in New CPU Card 1.

To do so, click the **Change License** button, a now pop up opens. Select **New CPU Card 1** as the option for **Select License Key.**

The system will now use the licenses as activated in New CPU Card 1.



*For **Firmware Version V1R6.7 onwards** the Grace period will be incremented till it reaches 90 days for CPU Card 2.*

When CPU Card 2 fails, this New CPU Card 1 will take over the control and will use the licenses as activated in the New CPU Card 1.

- *The **Change License** button will be visible when the redundancy license is activated in both the Active and Standby Cards. This option is also visible, if the redundancy license is activated in the Foreign CPU License Key or any one CPU Card, currently present in the system.*

- *The Standby Card, if present will restart once the redundancy grace period expires in the Active CPU Card.*

After the redundancy process is completed, SARVAM UCS intimates you about the same via a Redundancy Notification Call. To know how to receive the notification call, refer to “Redundancy Notification Call”.

To view the status of different parameters during the redundancy process, refer to “Redundancy Process”.

Redundancy Notification Call

What's this?

When the active card fails, the standby card takes over the control and becomes active. This process of restoring the functionality of the server by transferring the active operation to the standby card is known as redundancy. To know more, see “Redundancy”.

As redundancy is an important event, the intimation of the same can be provided to the concerned entities in an organization via an auto-generated notification call, known as Redundancy Notification Call.

This redundancy notification call is generated by the server and sent to the desired landing destination/s, after the redundancy process is completed.

You can either select a dedicated extension or a routing group as the landing destination for the redundancy notification call.

You can also define different landing destinations for each time zone, that is Working Hours (WH), Break Hours (BH) and Non-working Hours (NH) as per your requirement.

How it works

- After the redundancy process is completed, the extension configured to receive the redundancy notification call rings[327], displaying the message “Redundancy Notification Call”[328] on the LCD screen.
- When the extension user answers the redundancy notification call, s/he is played a piece of music or a voice message as uploaded in MoH management. To know more, refer to “Uploading Custom MoH”.
- The extension user can now acknowledge the redundancy notification call. The acknowledged redundancy notification call is logged in the “System Activity Log” as well as “Simple Network Management Protocol (SNMP)” with the details of the extension that acknowledged the call.
- If the extension configured to receive the redundancy notification call is busy, then the system will place the notification call only after the extension becomes idle.
- If the extension user does not answer or reject or answer but does not acknowledge the redundancy notification call till the expiry of the Ring Back Tone Timer, then the system makes one more attempt after an interval of 10 minutes and place the notification call on the landing destination again. If the redundancy notification call is not acknowledged, then the notification call is logged in the “System Activity Log” as well as “Simple Network Management Protocol (SNMP)”.

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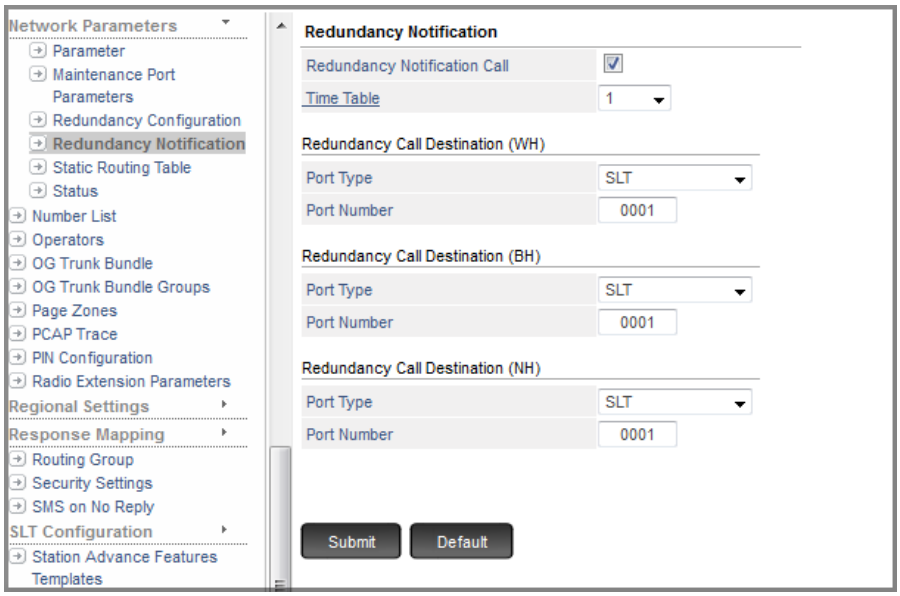
327. The extension rings for the duration of the Ring Back Tone Timer (configurable, default:45 minutes). See “System Timers and Counts”.

328. If you use a SLT, that does not supports FSK CLI as the landing destination, then the message “Redundancy Notification Call”, will not be displayed, on the screen.

How to configure

- Log in as System Engineer.
- Under **Configuration**, click **Network Parameters**.
- Click **Redundancy Notification**.



- **Redundancy Notification Call:** Enable this check box if you want the system to place the Redundancy Notification Call on extension/s after the redundancy process is completed. By default, this check box is enabled.
- **Time Table:** A Time Table is a schedule for the three Time Zones, namely: Working Hours, Break Hours, Non-Working hours for a week

  You can define and select the Time Table for the Redundancy Notification Call as per you requirement. There are 8 different Time Table templates to select from. By default, the Time Table 1 is selected.

  In Time Table 1, six days of the week - Monday to Saturday - have working hours from 9:00-18:00, break hours from 13:00-14:00 hours and non-working hours from 18:00 to 09:00. Sunday is a holiday, with all three Time Zones set to 00:00 hours.

  You may also customize the default Time Table 1 OR customize and assign a different Time Table as per your requirement. Refer to “Time Tables” for more details.

  You can select different landing destinations for each time zone, that is Working Hours (WH), Break Hours (BH) and Non-working Hours (NH) as per your requirement.

  For example, redundancy notification call can be routed to the Operator’s extension during working hours and the security personnel extension’s during the non-working hours.
- **Port Type:** Select the landing destination on which you want to place the Redundancy Notification Call. It may be a SLT, DKP, SIP Extension or Routing Group.

If you select routing group as the landing destination for the redundancy notification call, then you must configure a routing group. To know how to create a routing group, refer to "Routing Group".

By default, SLT is selected as the landing destination for all the three time zones.

- *This page displays only those Extension Port Type that are pre defined in the System Pre-requisites page. For instance, if the value selected for **SIP Extensions** under **Number of Ports Used** is zero in the **System Pre-requisites** page, then SIP Extension will not be displayed as one of the Port Type. To know more, refer to "Configuring System Pre-requisites".*

- *The Redundancy Notification Call will be displayed in the call logs, if you have configured the UC Clients (VARTA ADR100/ VARTA AMP100/ VARTA WIN200) or SPARSH VP330 or SPARSH VP210 as the landing destination. However, the notification call will not be displayed in the call logs, if you have configured, SPARSH VP310/ SPARSH VP510 or DKP as the landing destination.*

- *When the Redundancy Notification Call or Emergency Reporting Call is answered, MOH is played as a piece of music to the extension user. So, if a extension user uses a SLT, that does not has a display, as the landing destination, it becomes difficult for him/her to distinguish whether the incoming call is a Redundancy Notification Call or a Emergency Notification Call. So, to avoid such confusion, make sure you use a SLT that has a display or supports FSK CLI.*

- *Redundancy Notification Call is not supported on SPARSH VP248.*

- *Redundancy notification call will not be applicable, if you select Voice Mail Auto Attendant or OTBG as the members in the Routing Group.*

- **Port Number:** Enter the port number (software port or routing group number) on which the landing destination is configured.

  If you have selected *SLT, DKP or SIP Extension* as the Port Type, enter the software port number of the Extension.

  If you have selected *Routing Group* as the Port Type, enter the routing group number (01 to 96) in this field.

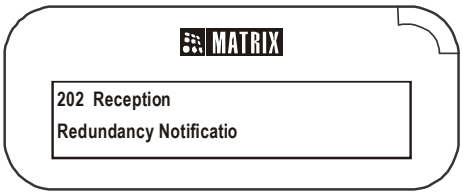
  Similarly, you can configure the landing destination for the Break Hours and the Non-Working hours as per your requirement.

- Click **Submit** at the bottom of the page to save changes.

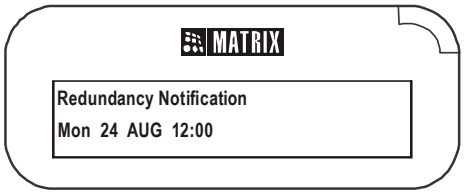
- *The Redundancy Notification call is generated 5 minutes after the redundancy process is completed.*
- *If you have set DND or Call Forward on a extension that is configured to receive the redundancy notification call, then in such case, the system will override these features and will place the redundancy notification call on the extension.*

Acknowledging Redundancy Notification Call from SPARSH VP310 Extended IP Phone

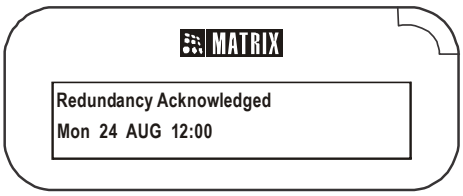
- You will receive a Redundancy Notification Call, after the redundancy process is completed.



- Lift the Handset/ press the Speaker Key/ press the Headset Key/ CA Key to answer the Redundancy Notification Call.
- You will hear a piece of music or a voice message.



- Press the digits (**0-9, * or #**) to acknowledge the Redundancy Notification Call.



- The Redundancy Notification Call is acknowledged.

Refer the respective User guide of the Extended IP Phones and UC Clients, to know how to acknowledge the Redundancy Notification Call.

Even though SARVAM UCS supports Redundancy Notification Call, it is advisable to enable the Email/ SMS notification for the system related activities and faults, so that, in case you miss the notification call, you still are notified about the redundancy process via an Email/ SMS. To know how to enable the Email/ SMS notifications, refer to "System Log Notification".

Reminder

What's this?

Reminders are a variation of the “Alarms” feature, requiring the Date and Time to be set for each Reminder call.

Reminder calls are useful for extension users who wish to be reminded of important tasks or appointments.

For Reminder calls, date and time are set in the following format:

Date is set, according to Date Format you selected in the “Real Time Clock (RTC)” parameters, as:

- Day-Month-Year (DD:MM:YYYY)

  Or

- Month-Date-Year (MM:DD:YYYY).

- Reminders can be set and canceled by:
  - the Operator from the Operator phone and from Jeeves.
  - extension users from their phones.

- Multiple Reminders can be set for an extension by the Operator and/or by the extension user.

- The mechanism for serving Reminders calls can be configured as 'Personalized' or 'Automated'.

- Reminders can be voice-guided, if the SARVAM UCS has a Voice Mail System module installed in it.

- SARVAM UCS can register as many as 999 Reminders[329] set by the Operator and extension users.

How it works

Personalized Reminder

When the Reminder call serving mechanism is configured as 'Personalized',

- The Operator Phone rings first[330], displaying the number of the extension to which the reminder call is to be served.

- When the Operator answers this call, a call is placed on the extension on which the reminder call is set.

- The extension phone rings for the duration of the Alarm Ring Timer.

- When the extension user answers the call, the Operator greets the extension user with the reminder message.

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329. ETERNITY PENX supports 48 Reminders.

330. The Operator phone rings for the duration of the Alarm Ring Timer. If the Operator does not answer the call, the SARVAM UCS will make two more Alarm Attempts at an Alarm Attempt Interval of 5 minutes to call the Operator.

- If the extension user does not answer the call till the *Alarm Ring Timer* has elapsed, the Operator phone will display a text message notifying 'No Reply' from the extension. The Reminder is now considered as served.

- If the extension is busy[331], the Operator phone will display a text message notifying that the extension number is 'Busy'.

- The Operator can now choose to

  - inform the extension user about the Reminder in person or send someone to do it.

    OR

  - try the busy extension again.

    OR

  - set “Auto Call Back (ACB)”.

Personal Reminders will work even if the extension user has set DND or Call Forward.

Automated Reminder

When the Alarm serving mechanism is configured as 'Automated',

- The extension phone rings at the set time till the end of the Alarm Ring Timer. If the extension phone is a DKP or the Matrix Extended IP Phone, Reminder message will appear on its display.

- When the extension user answers the call, the user may be played music-on-hold, or a pre-recorded voice message, or be connected to the Voice Mail, or routed to the Operator, depending upon the Alarm Notification Type you have configured for the extension.

- If the extension user does not answer the reminder call, the SARVAM UCS makes two more attempts (in all, 3 attempts) at an interval of 5 minutes between each attempt, to call the extension.

- If all Reminder call attempts go unanswered, the SARVAM UCS places the call on the Operator Phone. The Operator Phone rings till the end of the Alarm Ring Timer. The Operator Phone displays the number of the extension with the message 'No Reply'. The Reminder call is now considered as served.

- If the extension phone is busy, the SARVAM UCS will continue to make the Reminder call Attempts at the Alarm Interval programmed. When all Alarm Attempts go unanswered, SARVAM UCS will place a call on the Operator phone. The Operator Phone will display the number of the extension phone with the message 'Busy'.

331. An improperly placed receiver may also be the cause for the busy tone on the extension phone. In that case, the system will notify the Operator Phone with the 'OFF-Hook Alert'.

Snooze

The Snooze function can be added to Automated Reminders to ensure that the extension user answers the call. Snooze is a system-wide feature; when set, this function will be added to all Automated Reminder calls.

When Snooze is activated,

- The extension phone rings for the Number of Alarm Attempt configured, at the set Alarm Attempt Interval.
- The extension stops ringing when the user answers the call and dials **0** to acknowledge the Reminder call. This reminder call Acknowledgement Code **0** is non-configurable.

- *Reminder can be set for Operator phones also.*
- *Reminder settings will be retained in the system during power down and system upgrades.*
- *When multiple reminder requests have been set by an extension user, the extension user cannot selectively cancel a particular reminder request. Only the Operator can selectively cancel Reminders set for an extension user from the System Administrator pages of Jeeves.*
- *It is not possible to modify—change the date and time—of a reminder request. So, you may cancel the Reminder request and set a new one.*
- *Consider you have set a reminder with snooze enabled and Number of Alarm Attempts set as three (configurable). If this reminder call is not acknowledged by the extension user at the first attempt and due to some reason, the system restarts, then the pending two attempts will not be served. However, this reminder will be displayed under the pending reminder list.*

Reminder Status Report

The Operator can know the details of Reminders that have not been served on the *Reminder* page of Jeeves, from the System Administrator (SA) mode or by pressing the DSS key assigned to Wakeup Call Log. To view the log using DSS key, see “Alarm Status Report”.

The status of Reminders set by Operator as well as extension users appears on this page, with details of time (hours and minutes), and serving mechanism (personalized, automated).

The Operator can view the Reminder report whenever required and can also print this report.

How to configure

The configuration of Reminders is the same as *Alarms*.

To configure Reminders feature, do the following:

- Select the **Alarm Notification Type** for the extensions.
- Configure, as required, the Alarm Call related parameters: **Alarm Ring Timer**, **Number of Attempts**, **Alarm Attempt Interval**, **Configurable Alarm Type** and **Configurable Alarm Category**, and **Snooze**.

- Configure **Macros**, if the SLT extension has special function keys, and you want to a function key for the Reminder feature.

  For instructions, see the topic “How to configure” under “Alarms”.

How to use

Reminders can be set by the extension users by themselves. The extension users can also ask the Operator to set the Reminder for them.

If the Voice Mail System module is installed in the SARVAM UCS, it can offer voice-guided Reminders to extension users and the Operator.

Voice-guided reminders lead users through a menu, helping them set the alarm in a step-by-step manner.

Voice-Guided Reminders set/canceled by Operator

The Operator can set voice-guided Reminders for extension users

For EON & Extended IP Phones

- Press DSS Key assigned to 'Remote Voice-Guided Reminder' function.
  OR
- Dial 1072-035
- Follow Voice Mail System Prompts to set/cancel Reminder.

For SLT

- Pick up the handset.
- Dial 1072-035
- Follow Voice Mail System Prompts to set/cancel Reminder.
- Replace handset.

Voice-Guided Reminder set/canceled by Extension Users

For EON & Extended IP Phone Users

- Press the key assigned to Voice-guided 'Reminder' function.
  OR
- Dial 164
- Follow Voice Mail System prompts.

For SLT Users

- Pick up the handset.
- Dial 164
- Follow Voice Mail System Prompts.
- Replace handset.

If the SLT of the extension user has a special Reminder function key, the extension user can set the alarm using this key.

For SLT with 'Reminders' Key

- Press 'Reminders' key. (The label on the SLT key may differ from model to model)
- Follow the Voice Mail System prompts to set/cancel reminders.

- *SLTs with special function keys will work only if the corresponding Macros are programmed by the System Engineer.*

- *Without the Voice Mail System installed, the extension user having SLT with the special Reminder function key will not be able to set/cancel Reminders. This extension user can set/cancel Reminders only by dialing the feature access code for voice-guided Reminders.*

Non-Voice Guided Reminders set/canceled by Operator

For EON & Extended IP Phones

To set Reminder for the extension user,

- Press the DSS Key assigned the 'Remote Reminder' function.
  OR
- Dial 1072-033
- Enter the Extension Number.
- Enter Date and Time in the format:
  DD: MM: YYYY: HH: MM

  OR

  MM: DD: YYYY: HH: MM (users in USA)

- Select 'Personalized' or 'Automated'.
- Press 'Enter' key to set Reminder.
- You get a confirmation tone and message.

To cancel Reminder set for the extension user,

- Press key assigned for ‘Remote Reminder’ function.
  OR
- Dial 1072-033
- Enter Extension Number.
- Select 'Cancel All'.
- Press 'Enter' Key.

For SLT

To set Reminder for an extension,

- Lift handset.
- Dial 1072-033
- Dial Extension Number.
- Dial Date and Time in the format:
  DDMMYYYYHHMM
  OR

- MMDDYYYYHHMM (users in USA)
- Dial 1 for Personalized, Dial 2 for Automated.
- You get confirmation tone.
- Replace handset.

To cancel Reminder set for an extension,

- Lift handset.
- Dial 1072-033
- Dial Extension Number.
- Dial #
- You get confirmation tone.
- Replace handset.

To cancel reminder calls selectively, go to 'Reminder Status' page from the System Administrator of Jeeves.

If the 'Configurable Alarm Category' flag is enabled, only then the system will give the option of selecting 'Automated' or 'Personalized' as the serving mechanism. By default the serving mechanism is Automated.

Non-Voice Guided Reminders set/cancel by Extension Users

For EON & Extended IP Phone Users

To set Reminder:

- Press the 'Reminder' key.
  OR
- Dial 162
- Enter Date and Time in the format
  DD:MM:YYYY:HH:MM
  OR
  MM:DD:YYYY:HH:MM (users in USA)
- Press 'Enter' key.
- You get a confirmatory text message and confirmation tone.
- Go Idle.

To cancel Reminder:

- Press 'Reminder' Key again.
  OR
- Dial 162
- Select 'Cancel All'.
- Press Enter Key.

For SLT Users

To set Reminder,

- Lift handset.
- Dial 162
- Dial Date and Time in the format
  DDMMYYYYHHMM
  OR

MMDDYYYYHHMM (users in USA)

- You get confirmation tone.
- Replace handset.

To cancel Reminder,

- Pick up the handset.
- Dial 162
- Dial #
- You get confirmation tone.
- Replace handset.

Viewing and Printing Reminder Status

The Operator can view the status of Reminders that are yet to be served from the System Administrator pages of Jeeves or from an extension of the SARVAM UCS.

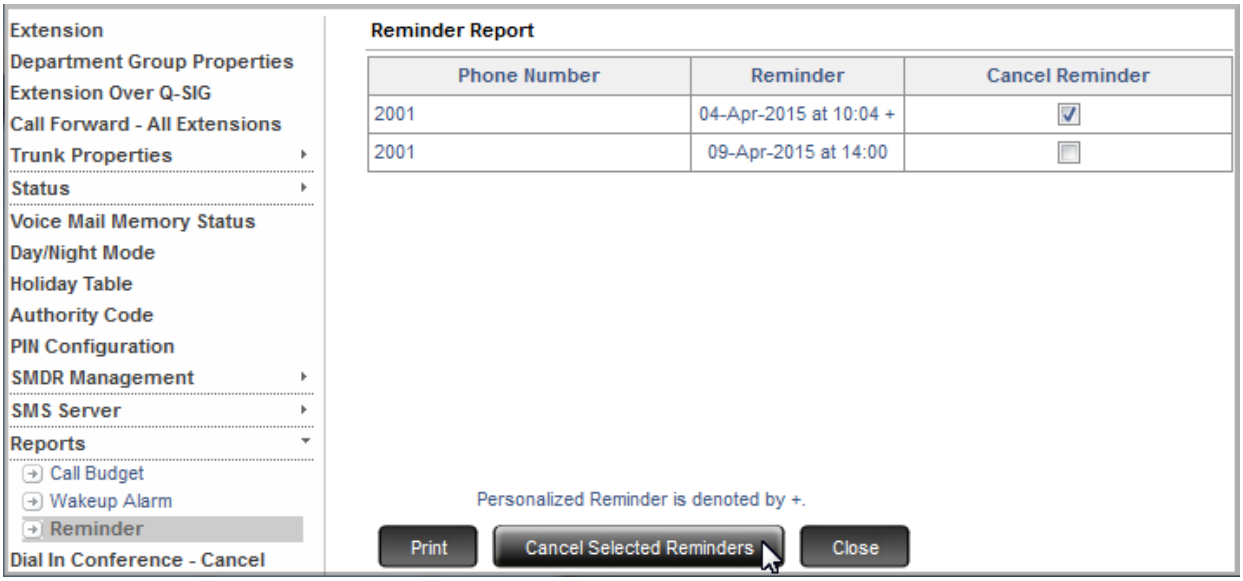
To view Reminder Status from the System Administrator pages of Jeeves,

- Open Jeeves.
- Log in as System Administrator.
- Click the **Reports** link.
- Under Reports, click the **Reminder** link.

The unserved Reminders appear on the page.

- To cancel any of the unserved Reminders,

- Select the **Cancel Reminder** check box of the extension number for which you want to cancel the reminder.



- Click the **Cancel Selected Reminders** button at the bottom of the page.

- To print this page, click the **Print** button.
- Click **Close** to exit the page.
- Click **Logout** to exit SA mode.

To print Reminder reports from an extension of SARVAM UCS,

- Pick up the handset.
- Dial ***1072-917.***
- Replace Handset.
- The report will be printed on the destination port assigned to Hotel/Motel reports.

```
Reminder Report           AS ON 05-05-2016(Thu) AT 23:50
--------------------------------------------------------------------------------
  Room#    Phone#  Reminder           P     Room#    Phone#  Reminder           P
--------------------------------------------------------------------------------
          3001    04-05-2016 12:25               3001    04-05-2016 12:14
          3001    05-05-2016 17:11
--------------------------------------------------------------------------------
 + indicates Personal Reminder                                    Page : 1
                     ---End of Report---
```

- *The '+' sign on the report indicates Personal Reminder.*

- *If the date format of SARVAM UCS is set as MM-DD-YYYY or the Region 'USA' is selected, then the reminder report will be printed according to this date format. To know more, see "Real Time Clock (RTC)".*

Remote Programming

What's this?

SARVAM UCS offers a Graphic User Interface, Jeeves, for configuration and maintenance. With the Jeeves and IP connectivity, SARVAM UCS can be configured from across the globe. However, in some cases, where IP connectivity from public network is not available, remote programming can be done using DISA.

Direct Inward System Access (DISA) facility of the system allows a remote user to login and use most of the functions of the system. Programming is one such function allowed to the remote user. The remote user can program the system using the same commands as used by the normal local station to program the system. The user can login into the SA programming or SE programming mode.

How to use

- Make a DISA call.
- Login as DISA user. Get the DISA login beeps. Enter the SE/SA mode. Get programming/Dial Tone.
- Make programming changes.
- Exit SE/SA mode.
- Terminate DISA call.
- In case none of the trunk lines have DISA facility and it is required to do remote programming, then follow following steps:
    - From the remote location, make a call to the system.
    - Ask the person answering the call at the system end to keep you on hold by dialing Flash. When the person at the system end dials 'Flash', you get hold on music.
    - Ask the person at the system end to dial ***1#91-SE Password***. On completion of the command, he gets confirmation tone. He can disconnect and carry on with the routine job.
    - The person at the remote end gets DISA beeps.
    - He dials #1 and gets the dial tone.
    - He can enter the SE/SA mode from the remote end.

SE Mode

Dial ***1#91-SE Password*** program the system. Dial '***00***' exit from the programming mode (SE Mode).

When you have logged into SE Mode from DISA, to program #, you need to press # four times.

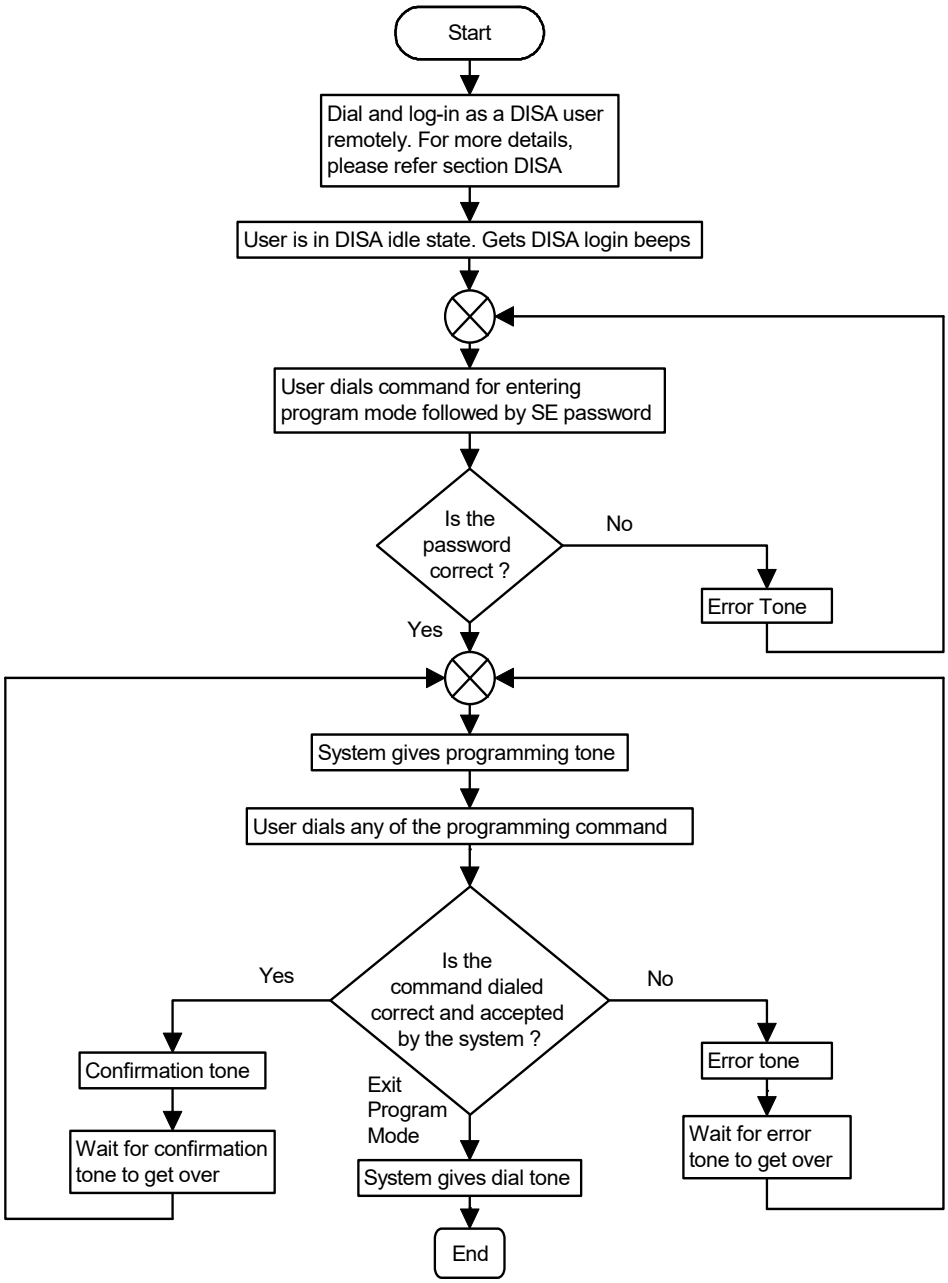
SA Mode

Dial ***1#92-SA Password*** program the system. Dial ***1#92*** exit from SA mode.

Once the user is out of SE/SA Mode the user gets DISA beeps.

How it works

Following flow chart depicts the process:



Room Monitor

What's this?

This feature enables the DKP and Extended IP Phone Extension users to listen to the conversations taking place in another location where a DKP/Extended IP Phone is present.

Room Monitor can be used to monitor activities on the Shop Floors / Manufacturing areas from another location.

- *Use this feature in accordance with the local privacy laws.*
- *Matrix Comsec is not responsible for any mis-/abuse of this feature by users.*

How it works

- A is a supervisor in a Manufacturing unit.
- A's room is on the second floor. The manufacturing area is on the ground floor.
- To keep track of the activities in the plant on the ground floor, there must be a DKP or an Extended IP Phone at the place where the activities are to be monitored, and A's extension must have higher "Priority" than the extension at the monitored location.
- If there is a DKP or an Extended IP Phone at the desired location, A can activate Room Monitor.

  A can activate Room Monitor only if the DKP/Extended IP Phone at the desired location is idle.
- When A activates Room Monitor, the microphone of the DKP/Extended IP Phone on the ground floor goes Off-hook. A can now hear all the sounds taking place on the ground floor, without anyone present there coming to know that they are being monitored.
- To end Room Monitor, A must disconnect.
- Room monitoring will be terminated on the DKP/Extended IP Phone on the ground floor, if someone lifts the handset of this phone or if there is a call on this phone from another extension.

You can activate Room Monitor from any extension port type, but the extension being monitored must be a DKP or an Extended IP Phone.

How to configure

To be able to use Room Monitor, extension users must have this feature enabled in the "Class of Service (COS)" in the "Station Basic Feature Template" assigned to their extensions.

How to use

For EON & Extended IP Phone Users

To enable Room Monitor on an extension,

- Press the DSS Key assigned to Room Monitor.
- Dial Extension Number to be monitored.
  OR
- Dial 1073-Extension Number to be monitored.

For SLT Users

- Dial 1073-Extension Number to be monitored.

Response Mapping

What's this?

For calls to be established between a SIP and ISDN networks, the two networks must interoperate. The mapping of ISDN causes of call failure or completion, to SIP codes and events, as well as mapping of SIP codes and events to ISDN causes are vital aspects to attain interoperation between the networks. Similarly, mapping of system disconnect/release events to SIP/ISDN networks is another important aspect.

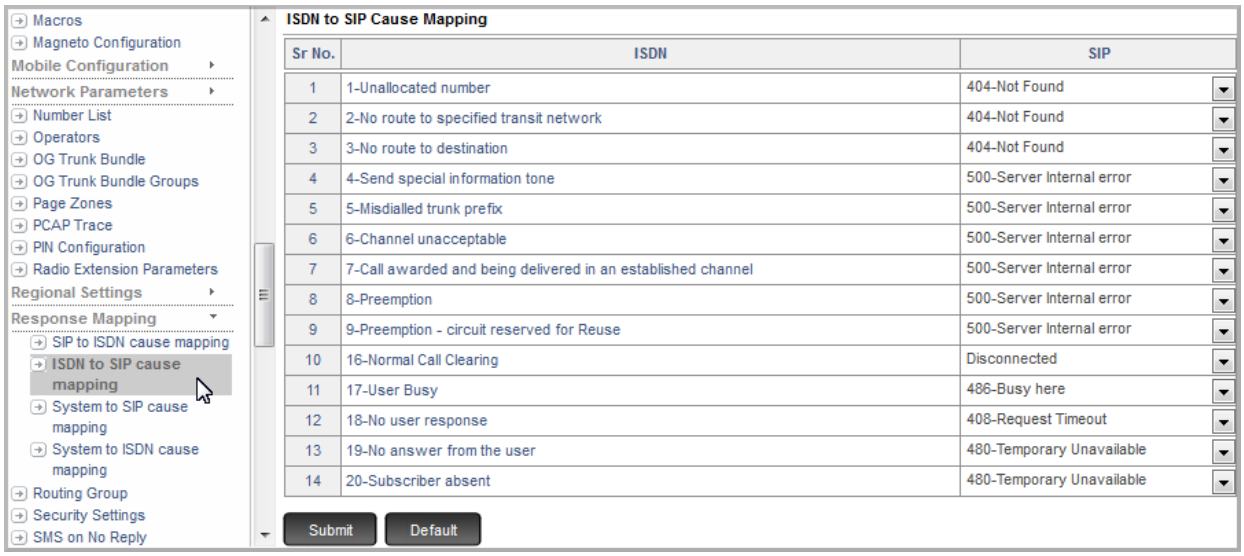
The SARVAM UCS supports programmable event cause mapping. You can configure the following as per your requirement:

- Disconnect/error cause for SIP to ISDN calling, that is, when the system receives a SIP disconnect/error cause from remote SIP peer then you can select the code and event the system must send on ISDN to the remote ISDN peer.
- Disconnect/error cause for ISDN to SIP calling, that is, when system receives the ISDN disconnect/release cause then you can select the code and event the system must send as the SIP response to the remote SIP Peer.
- Disconnects/release cause to be sent by the system to SIP network, that is, when the system disconnect/release the call with SIP then you can select the code and event the system must send as the SIP response to remote SIP Peer.
- Disconnects/release cause to be sent by the system to ISDN network, that is, when the system disconnect/release the call with ISDN then you can select the code and event the system must send as the ISDN cause to remote ISDN Peer.

Configuring Response Mapping using Jeeves

- Log in as System Engineer.
- Under **Configuration**, click **Response Mapping**.
- Click **SIP to ISDN cause mapping**.

- The **SIP** column contains the list of the SIP codes and events. For each SIP code and event you can select the cause that you want the system to send to the remote ISDN peer.
- Click **Submit**.
- Click the **ISDN to SIP cause mapping**.



- The **ISDN** column contains the list of ISDN causes of call failure or completion.
- For each ISDN cause you can select the SIP code and event you want the system to send as the SIP response to the remote SIP Peer.
- Click **Submit**.

Similarly, you can also map the system disconnect/release events to SIP/ISDN network. Click the respective link S**ystem to SIP cause mapping** or **System to ISDN cause mapping** to configure the mapping as per your requirement.

Routing Group

What's this?

Routing Group is a group of extensions used for landing incoming calls as a Trunk Landing Group, as Alarm Notification Group, as Floor Service Group and as Department Group.

How it works

SARVAM UCS supports the formation of 96 Routing Groups. In each group you can have upto 32 members.

The *member* of a Routing Group can be Single Line Telephones (SLT), Digital Key Phones (DKP), SIP Extensions, ISDN Terminals, Virtual Extensions, Voice Mail Auto Attendant and Outgoing Trunk Bundle Group.

These groups can be used:

- as Trunk Landing Groups to route incoming calls.
- as Alarm Notification Groups to server Alarm Notifications.
- as Floor Service Groups to provide Floor Service.
- as Department Groups to route incoming calls to a particular department.

This is how a Routing Group works,

- There is an incoming call on CO1.
- Routing Group 1 is assigned as the Trunk Landing Group for CO1. The Routing Group has DKP 2001, 2002 and 2003 as landing destinations.
- By default incoming calls will be placed on the members in rotation, that is first call on DKP 2001, second call on 2002 and so on.

  If you want incoming calls to be placed on DKP 2001 always, you must disable Rotation.
- By default, an incoming call will be placed on 2001. 2001 rings for the duration of the Ring Timer, if the call is unanswered the system re-directs the call to 2002 and so on, till the call is answered.

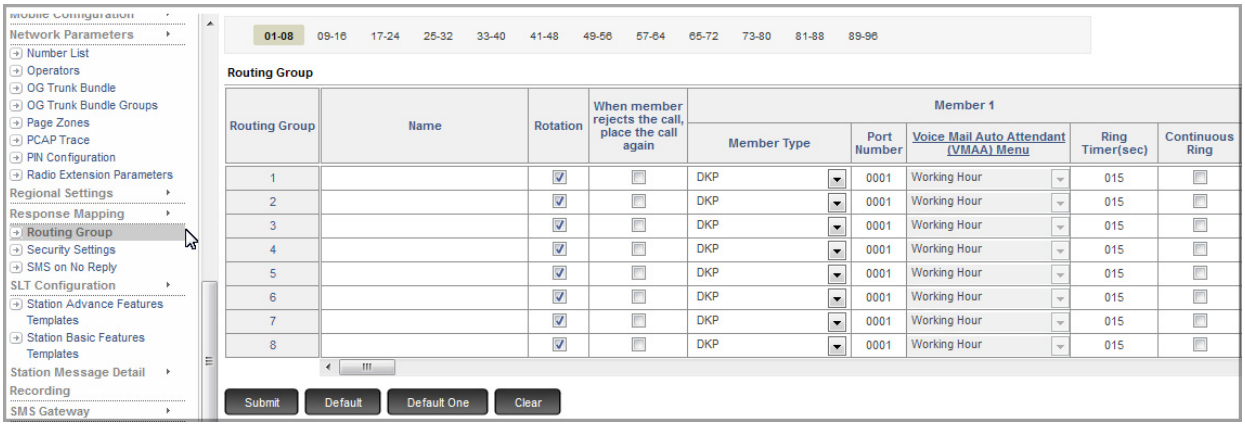
  If you want all the extensions to ring continuously till the call is answered by any member, you must enable C*ontinuous* Ring. 2001 will continue to ring even as the system hunts for other extensions in the routing group to land the call. If the call still remains unanswered, the system will return the call to 2001 once again.
- In this way the system places the incoming calls on the member extensions in a Routing Group till the call is answered.

If you have selected Voice Mail Auto Attendant or OTBG as members in a Routing Group, the parameters Ring Timer and Continuous Ring are not applicable.

How to configure

Configuring Routing Groups using Jeeves

- Login to Jeeves as System Engineer.
- Under **Configuration**, click **Routing Group**.



- Choose the Routing Group number (01-96) you want to use as routing group. In each routing group you can program maximum 32 'members'.
- Now program the following parameters for the selected routing group:
  - **Name:** You can assign a Name to the department group to facilitate identification. This name will appear in the Dial by Name directory along with the department group number. The Name can be a maximum of 18 characters.
  - **Rotation:** Select the Rotation check box to enable rotation of calls in the routing group which has multiple 'member' extensions. When enabled, each fresh call will land on the extension which is next to the one that received the last call. This ensures equal distribution of incoming calls to all the destinations within the routing group. The flag has no relevance if the routing group has only one member extension.
  - **When member rejects the call, place the call again:** If any SIP/DKP member extension rejects an incoming call and the system again checks the routing group sequence, you can allow or restrict placing the same call on this extension. Select the When member rejects the call, place the call again check box**,** if you want the system to place the call again on the extension.

    If this check box is cleared (disabled) and you have selected the Continuous check box, the extension that rejects the call stops ringing while the system hunts for other extensions in the routing group to land the call.
  - **Member Type:** Select the 'Member Type'. You can select SLT, DKP, SIP, Virtual Extensions, ISDN Terminal, OTBG or the Voice Mail Auto Attendant.

    Configure only as many extensions as you want in the routing group and set the remaining Member Types to 'None'.

    For example: if you want to program only one extension in the routing group, set the Member Type in the remaining columns (Member 02-Member 32) to 'None.'

- **Port Number:** Enter the software port number on which the SLT/DKP/SIP/Virtual Extension/ISDN Terminal is connected.

  If you have selected OTBG then enter the OTBG number here.

- **Voice Mail Auto Attendant (VMAA) Menu:** if you have selected the *Voice Mail Auto Attendant* as the Port Type, select the VMAA Menu to assign to the respective routing group.

  You may click the *Voice Mail Auto Attendant (VMAA) Menu* link to edit the parameters of desired VMAA Menu. For details, see “Voice Mail Auto-Attendant Menu”.

- **Ring Timer(s):** This timer defines the time for which the extension, on which the call lands, should ring. By default, the ring timer is set to 015 seconds and can be changed.

- **Continuous Ring:** Select the Continuous Ring check box, if you want an extension to ring continuously until the call is answered. The first extension will continue to ring even as the system hunts for other extensions in the routing group to land the call. If the call still remains unanswered, the system will return the call to the first extension once again. This parameter is not relevant, if there is only one member extension in a routing group.

- Click **Submit** to save.
- Repeat the same steps to create another Routing Group.
- To route incoming calls on a trunk, you must assign a Routing Group in the Trunk Landing Group in a Trunk Feature Template assigned to the trunk.
- To assign a Routing Group as a Trunk Landing Group, under “Configuring Trunks”, see “Trunk Feature Template”.
- To assign a Routing Group as a Department Group, see “Department Call”.
- To assign a Routing Group as a Alarm Notification Group, under “Alarms”, see “How to configure”.
- To assign a Routing Group as a Floor Service Group, see “Floor Service”.

Configuring Routing Groups using a Telephone

- Enter SE Mode from a DKP/SLT.

  To select the destination in the Routing Group, dial:
  - ***6502-1-Routing Group-Destination Index-Port Type-Port Number***
  - ***6502-2-Routing Group-Routing Group-Destination Index-Port Type-Port Number***
  - ***6502-*-Number Index-Port Type-Port Number***

  Where,
  Routing Group is from 01 to 96.
  Destination Index is from 01 to 32.

Port Type	Meaning	Port Number
00	None[a]	000
01	SLT	001-512

Port Type	Meaning	Port Number
02	DKP	001-128
16	OG Trunk Bundle Group[b]	001-025
28	ISDN Terminal	01-64
34	SIP Extension	001-999
36	Virtual Extension	01-64

a. Parameters, Ring Timer and Continuous Ring are not applicable if the SE selects None, OG Trunk Bundle Group as Member Type.

b. Same as per above note.

To program the time for which each station in the group should ring, dial:

- ***6503-1-Routing Group-Member Index-Ring Timer***
- ***6503-2-Routing Group-Routing Group-Member Index-Ring Timer***
- ***6503-*-Member Index-Ring Timer***

  Where,
  Routing Group is from 01 to 96.
  Member Index is from 01 to 32.
  Ring Timer is from 001 to 255.
  By default, Ring Timer is 015 seconds.

To program continuous or non-continuous ring for a station in the group, dial:

- ***6504-1-Routing Group-Member Index-Flag***
- ***6504-2-Routing Group-Routing Group-Member Index-Flag***
- ***6504-*-Member Index-Flag***

  Where,
  Routing Group is from 01 to 96.
  Member Index is from 01 to 32.

Flag	Meaning
0	No station rings for the time set (Disable continuous)
1	The station rings till the call matures (Enable continuous)

To program rotation, dial:

- ***6505-1-Routing Group-Rotation Method***
- ***6505-2-Routing Group-Routing Group-Rotation Method***
- ***6505-*-Rotation Method***

  Where,
  Routing Group is from 01 to 96.

Rotation Method	Meaning
0	Fresh call lands on the first station within the group (disable continuous)
1	Fresh call lands on following the rotation method (enable continuous)

To default a routing group, dial:

- ***6501-1-Routing Group***
- ***6501-2-Routing Group-Routing Group***
- ***6501-****

Where,
Routing Group is from 01 to 96.

On issuing the command all routing groups will be set to default values as follows:

Member Index	Port Type	Port Number	Ring Timer (Sec.)	Continuous Ring	Rotation
01	DKP	001	015	OFF	ON
02	SLT	001	015	OFF	ON
03	SLT	002	015	OFF	ON
04-32	None	000	015	OFF	ON

To clear the member of a Routing Group, dial:

- ***6510-1-Routing Group***
- ***6510-2-Routing Group-Routing Group***
- ***6510-****
  Where,
  Routing Group is from 01 to 96.

- Exit SE mode.

Selective Port Access

What's this?

SARVAM UCS supports different extension and trunk port types. In the Selective Port Access feature, each port type is assigned a Port Access Code. Extension users can access a particular port by dialing the Port Access Code assigned to the Port and its Port Number.

How it works

- Extension user A wants to access a particular Mobile port, *Mobile Port 1* to make a call. Extension A must dial the Selective Port Access Feature Code, followed by the Port Type Code for Mobile ports and then dial the Port Number.

- The following access codes are assigned to each Port Type:

Port Types	Port Access Code	Port Numbers[a]
SLT	01	001 to 240
DKP	02	001 to 096
CO	03	001 to 064
BRI	04	01 to 32
T1E1	05	01 to 08
E&M	06	001 to 032
Mobile	25	001 to 040
SIP Trunk	26	01 to 99
ISDN Terminal	28	01 to 64
SIP Extension	34	001 to 999
Magneto	29	001 to 016
Virtual Extension	36	01 to 64
Radio Ports	40	01 to 16

a.The number of ports mentioned here are for ETERNITY GENX. For details regarding the number of ports supported for ETERNITY LENX/MENX/PENX, see "Technical Specifications - SARVAM UCS".

Here, Extension A must dial ***69-25-01***, where ***69*** is the feature code for Selective Trunk Access, **25** is the port access code for the Mobile Port, and ***01*** is the number of the Mobile Port which A wants to access.

Similarly, if Extension A wants to call SIP Extension 10, A can dial ***69-34-010***.

How to configure

To be able to use Selective Port Access, extension users must have this feature enabled in their "Class of Service (COS)".

How to use

For EON & Extended IP Phone Users

To enable Selective Port Access on an extension,

- Press DSS key assigned to Selective Port Access code
- From the menu select the Port Type
- Enter the Port Number of the selected Port Type

  OR
- Dial 69 / 89 (for users in USA)
- Dial the Port Type - Port Number

For SLT Users

- Dial 69-Port Type-Port Number
- Dial 89-Port Type-Port Number (for users in USA)

Security Settings

The feature Security Settings enables you to restrict unauthorized access to SARVAM UCS using Remote login, Web, Third Party Auto Configuration as well as SIP Extension registration.

SARVAM UCS also supports TLS (Transport Layer Security) protocol. Based on the TLS version configured in the server, TLS negotiation takes place. This enables the SIP Extensions and Web server to connect securely with the system over TLS protocol.

For Remote Login, Web, Third Party Auto Configuration

Remote Login

For the Remote Login, the Jeeves provides you the facility to generate the keys. Once these keys are generated, you need to contact the Matrix Technical Support for Remote Login.

Web and Third Party Auto Configuration

When any user attempts to access Web Server or Third Party Auto Configuration using false credentials for 5 times consecutively within 10 minutes, SARVAM UCS blocks such IP Address for 10 minutes.

To allow access to Web Server and Third Party Auto Configuration, to specific trusted IP Address/es, you must configure them in the *Trusted IP Address/es* table. For instructions, see “How to configure” below.

For SIP Extensions

When any user attempts to register as a SIP Extension using false credentials— Authentication ID or Authentication Password and the authentication attempt fails for 10 times consecutively, SARVAM UCS blacklists the IP Address and port used for registration. See “Black List IP Address - SIP Extensions” for more details.

You are recommended to configure the trusted IP Addresses in the *Trusted IP Address/es* table to avoid blacklisting. For instructions, see “How to configure” below.

However, if any IP Address is already blacklisted, it will be stored in the **Black List IP Address - SIP Extensions** table. To allow access to such blacklisted IP Address, you must remove it from the **Black List IP Address - SIP Extensions** table manually.

How it works

For this feature to work,

- the *Trusted IP Address/es* table must be configured. You can configure a maximum of 25 addresses in this table.
- determine the facilities you want to allow to each IP Address — Remote Login, Web, Third Party Auto Configuration, SIP Extension registration.
- with this table configured,
  - access to Remote Login, Web Server, Third Party Auto Configuration will be allowed only to the configured Trusted IP Address/es.

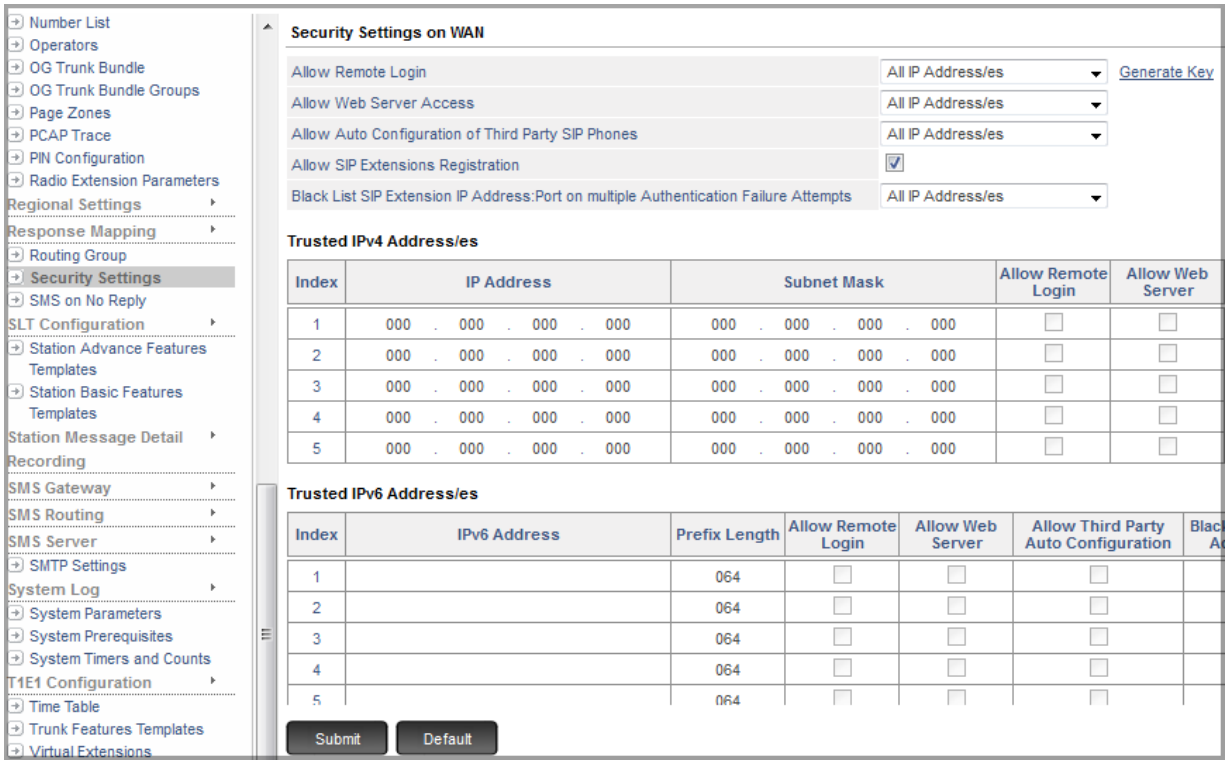
- Trusted IP Addresses configured for the registration of the SIP Extension, will not be blacklisted.

The successful attempt to access SARVAM UCS using Remote Login or Web will be logged in the System Activity Log.

When you change the jumper position on the CPU Card from BC to AB, the parameters Allow Remote Login, Allow Web Server Access will be set to default. The Jumper number depends on the model of SARVAM UCS. For details, see "System Security".

How to configure

- Log in as System Engineer.
- Under **Configuration**, click **Security Settings**.



Trusted IPv4 Address/es

- In **Allow Remote Login**, you can select Don't Allow, All IP Address/es or Only Trusted IP Address/es option. By default, **Don't Allow** is selected.

  If you want to allow access to generate the password for the Remote Login from all IP Addresses, select **All IP Address/es** option.

  If you want to allow access to generate the password for the Remote Login from specific IP Addresses only:
  - select **Only Trusted IP Address/es** option.
  - configure the IP Address/es and their respective Subnet Mask in the **Trusted IPv4 Address/es** table.
  - enable the **Allow Remote Login** check box in the **Trusted IPv4 Address/es** table.

If you select **All IP Address/es** or **Only Trusted IP Address/es** option, click **Generate Key**.
The Key Generation window opens.
In **Key Count**, select the number of keys you wish to generate. You can generate maximum 10 keys.

Click **Generate**.

The number of keys you selected are generated and saved in a file. Using these keys you can remotely login into the system.
For further assistance for Remote Login, along with this file contact Matrix Technical Support Team.

- In **Allow Web Server Access**, you can select Don't Allow, All IP Address/es or Only Trusted IP Address/es option. By default, **Don't Allow** is selected.

  If you want to allow access to the Web Server from all IP Addresses, select **All IP Address/es** option.

  If you want to allow access to the Web Server from specific IP Addresses only:
  - select **Only Trusted IP Address/es** option.
  - configure the IP Address/es and their respective Subnet Mask in the **Trusted IPv4 Address/es** table.
  - enable the **Allow Web Server** check box in the **Trusted IPv4 Address/es** table.

- In **Allow Third Party Auto Configuration**, you can select Don't Allow, All IP Address/es or Only Trusted IP Address/es option. By default, **Don't Allow** is selected.

  If you want to allow the third party configuration from all IP Addresses, select **All IP Address/es** option.

  If you want to allow Third Party Auto Configuration from specific IP Addresses only:
  - select **Only Trusted IP Address/es** option.
  - configure the IP Address/es and their respective Subnet Mask in the **Trusted IPv4 Address/es** table.
  - enable the **Allow Third Party Auto Configuration** check box in the **Trusted IPv4 Address/es** table.

- By default, the **Allow SIP Extensions Registration** check box is disabled. If you want to allow SIP extensions registration from the WAN Port, select the check box.

  - If you allow SIP Extension Registration, you can **Black List SIP Extension IP Address:Port on multiple Authentication Failure Attempts**.

    By default, **All IP Address/es** option is selected. If you do not want to include the Trusted IP Addresses, select **Except Trusted IP Address/es** option.

    If you select **Except Trusted IP Address/es,**

    - configure the Trusted IP Address/es and their respective Subnet Mask in the **Trusted IPv4 Address/es** table.

    - enable the **Black List SIP Extension IP Address:Port except** check box for those IP Addresses which you do not want to blacklist.

- Click **Submit** to save changes.

Trusted IPv6 Address/es

- In **Allow Remote Login**, you can select Don't Allow, All IP Address/es or Only Trusted IP Address/es option. By default, **Don't Allow** is selected.

If you want to allow access to generate the password for the Remote Login from all IP Addresses, select **All IP Address/es** option.
If you want to allow access to generate the password for the Remote Login from specific IP Addresses only:

- select **Only Trusted IP Address/es** option.
- configure the IPv6 Address/es and their respective Prefix Length in the **Trusted IPv6 Address/es** table. The Prefix Length is a decimal value that indicates how many of the high-order contiguous bits of the address comprise the prefix (the network portion of the address).
- enable the **Allow Remote Login** check box in the **Trusted IPv6 Address/es** table.

If you select **All IP Address/es** or **Only Trusted IP Address/es** option, click **Generate Key**.

The Key Generation window opens.

In **Key Count**, select the number of keys you wish to generate. You can generate maximum 10 keys.

Click **Generate**.

The number of keys you selected are generated and saved in a file. Using these keys you can remotely login into the system.

For further assistance for Remote Login, along with this file contact Matrix Technical Support Team.

- In **Allow Web Server Access**, you can select Don't Allow, All IP Address/es or Only Trusted IP Address/es. By default, **Don't Allow** is selected.

  If you want to allow access to the Web Server from all IP Addresses, select **All IP Address/es** option.

  If you want to allow access to the Web Server from specific IP Addresses only:
  - select **Only Trusted IP Address/es** option.
  - configure the IPv6 Address/es and their respective Prefix Length in the **Trusted IPv6 Address/es** table. The Prefix Length is a decimal value that indicates how many of the high-order contiguous bits of the address comprise the prefix (the network portion of the address).
  - enable the **Allow Web Server** check box in the **Trusted IPv6 Address/es** table.

- In **Allow Third Party Auto Configuration**, you can select Don't Allow, All IP Address/es or Only Trusted IP Address/es. By default, **Don't Allow** is selected.

  If you want to allow third party auto configuration from all IP Addresses, select **All IP Address/es** option.

  If you want to allow Third Party Auto Configuration from specific IP Addresses only:
  - select **Only Trusted IP Address/es** option.
  - configure the IPv6 Address/es and their respective Prefix Length in the **Trusted IPv6 Address/es** table. The Prefix Length is a decimal value that indicates how many of the high-order contiguous bits of the address comprise the prefix (the network portion of the address).

  - enable the **Allow Third Party Auto Configuration** check box in the **Trusted IPv6 Address/es** table.

- By default, the **Allow SIP Extensions Registration** check box is disabled. If you want to allow SIP extensions registration from the WAN Port, select the check box.

  - If you allow SIP Extension Registration, you can **Black List SIP Extension IP Address:Port on multiple Authentication Failure Attempts**.

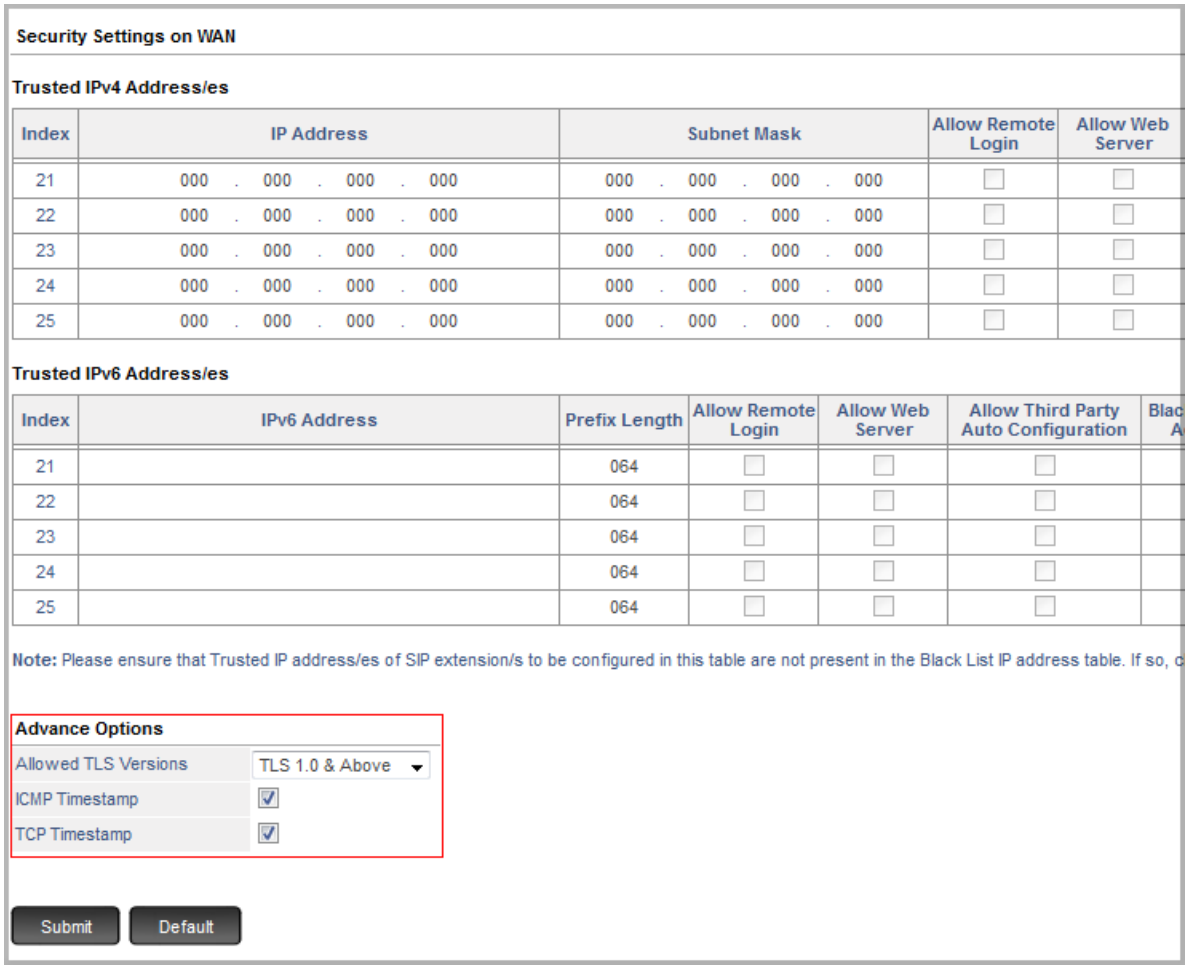
By default, **All IP Address/es** option is selected. If you do not want to include the Trusted IP Addresses, select **Except Trusted IP Address/es** option.

If you select **Except Trusted IP Address/es,**

- configure the IPv6 Address/es and their respective Prefix Length in the **Trusted IPv6 Address/es** table. The Prefix Length is a decimal value that indicates how many of the high-order contiguous bits of the address comprise the prefix (the network portion of the address).

- enable the **Black List SIP Extension IP Address:Port except** check box for those IP Addresses which you do not want to blacklist.

Advance Options



- In **Allowed TLS Versions,** select the TLS Version[332] you want the system to use to establish a secure connection with the clients. You may select — TLS 1.0 & Above, TLS 1.1 & Above or TLS 1.2 as per your requirement. Default: TLS 1.0 & Above.

  If the TLS version of the server and the client is not compatible, then secure connection will not be established.

332. SPARSH VP330/SPARSH VP210 supports TLS Version 1.0 only.

Changing the TLS Version may result in drop of all ongoing TLS connections.

- Select the **ICMP Timestamp** check box if you want to send Date and Time of SARVAM UCS in response to the ICMP request received from the remote device. By default, it is enabled.

- Select the **TCP Timestamp** check box if you want to send Date and Time of SARVAM UCS in response to the TCP request received from the remote device. By default, it is enabled.

Self Ring Test

What's this?

You can use Self Ring Test to check the functioning of your own extension phone. Self Ring Test allows you to call your own extension. You can check the ringing volume of your extension phone.

How to use

For EON & Extended IP Phone Users

- Go OFF-Hook.
- Press DSS Key assigned to Self Ring Test.
  OR
- Dial ***1057***.
- Go ON-Hook.
- Your phone rings.
- Go OFF-Hook to stop the ring.
- Go ON-Hook.

For SLT Users

- Lift the receiver.
- Dial ***1057***.
- Replace receiver.
- Your phone rings.
- Lift the receiver to stop the ring.
- Replace the receiver.

Shared Call Appearance

What is this?

Shared Call Appearance (SCA) allows Standard SIP Phones that are registered with SARVAM UCS at different locations with the same address/number, to get notification on call states of the call appearance(s) shared by them.

Whenever a call is made or received from a shared call appearance, SARVAM UCS sends each SIP Phone sharing the call appearance(s), a notification on the state of the call appearance. Through these notifications, each user sharing the same address/number can know the current state of the call appearance and act accordingly.

SARVAM UCS supports and displays the following call states for a shared call appearance on the SIP phones:

State	Meaning
Idle	When the call appearance is free.
Seized	When the call appearance is been seized from any User binding using the line-seize subscription.
Progressing	When the User has generated a call using the call appearance and the called destination is ringing.
Ringing	When the call is received on the User at the call appearance.
Active	When the call of the User at the call appearance is in matured state.
Held	When the call at the call appearance of the User has been put on public hold from the User binding.
Held-private	When the call at the call appearance of the User has been put on private hold from the User binding.

- *SARVAM UCS supports SCA as per the Broadsoft SCA feature Specifications.*
- *SARVAM UCS supports this feature only on Standard IP Phones.*
- *Standard IP Phones may differ in the type of indication (LED color and cadence, text message display) they provide for the Call States. Refer to the manufacturer's documentation for SCA indication supported on the phones.*
- *Calls put on 'Consultation Hold' from Location -1 of SIP Extension (binding -1), the SIP Extension registered at Location-2/3 (another binding) can not retrieve that call.*

How it works

SARVAM UCS supports up to 10 call appearances on SIP extensions. The number of call appearances that will be shared by the SIP Phones will depend on the number of call appearances you have configured for the SIP extension.

To provide SCA to the Standard IP Phones registered with the same ID, the Shared Call Appearance flag must be enabled in the SIP Extension Settings of SARVAM UCS.

On the Standard IP Phones, make sure you have configured as many call appearances as allowed on the SIP extension by SARVAM UCS and configure the corresponding number of CA keys.

Here is an example of how Shared Call Appearance works:

- A, B and C are Standard SIP phones registered with SARVAM UCS at three different locations with the same SIP ID, 602.
- Two Call Appearances are configured for A, B and C and Shared Call Appearance is enabled for SIP ID 602.
- Two keys are assigned for the two Call Appearances on A, B and C.
- There is an incoming call for SIP ID 602.
- SARVAM UCS presents the incoming call on a free call appearance, Call Appearance1, as ‘Ringing’.
- A, B and C get the same alert, ‘Ringing’ simultaneously on the same call appearance, Call Appearance 1.
- A answers the call first and gets connected to it on Call Appearance 1.
- A, B and C get indication of the current call state as ‘Active’ on the same call appearance. B and C will not be able to make or receive any new call from this busy call appearance. However, they can make or receive a new call from the other free call appearance, Call Appearance 2.
- When A makes an outgoing call using Call Appearance 2, SARVAM UCS presents the state of the same call appearance on A, B and C as ‘Seized’, then as ‘Progressing’ when the destination number starts ringing, and then as ‘Active’, when the call is answered.
- B and C will not be able to make or receive a call from Call Appearance 2.
- A can put an ‘Active’ call on public Hold or on private Hold. When A puts an ‘Active’ call on public hold, SARVAM UCS presents the state of this call as ‘Held’ to A, B and C. Now, B or C can retrieve the call by pressing the corresponding call appearance key.
- When A puts an ‘Active’ call on private Hold, SARVAM UCS presents the state of this call as ‘private-Held to A, B and C. Only A can retrieve the call. Thus, if a call is put on private hold (Held-private), it can be retrieved only from the IP Phone that put it on hold.

How to configure

You can provide this feature only to Standard IP Phones you have registered with SARVAM UCS. To provide this feature,

- In SARVAM UCS, you must enable the **Shared Call Appearance** check box on the SIP Extension Settings.For instructions, see “Configuring SIP Extensions”.
- On the Standard IP phones,
  - configure as many call appearances as allowed on the SIP extension.
  - For each shared call appearance, configure a corresponding call appearance key on the SIP Phone.

  For instructions refer to the manufacturer’s documentation (Installation Guide/User Guide) for the respective SIP Phones.

SIM Card Balance and Recharging

This feature is not applicable if CDMA Mobile Card is installed in your system.

What's this?

The SARVAM UCS supports Balance Inquiry and Recharging of the SIM Card installed in its Mobile Ports[333].

How to use

To be able to use this feature, first collect the following information from your Network Operator:

- **Balance Inquiry Number:** This is the number provided by the Network Operator to the subscribers to check Balance. Different Network Operators have different numbers. For example, the Balance Inquiry number of Vodafone is ****141#.***

- **Recharging Service Number:** This is the number provided by the Network Operators to their subscribers for Recharging Service. Different Network Operators have different numbers for Recharging Service. For example, the Recharging Service Number of Vodafone is ****140****.

This feature can be used from the SE Mode only on Jeeves.

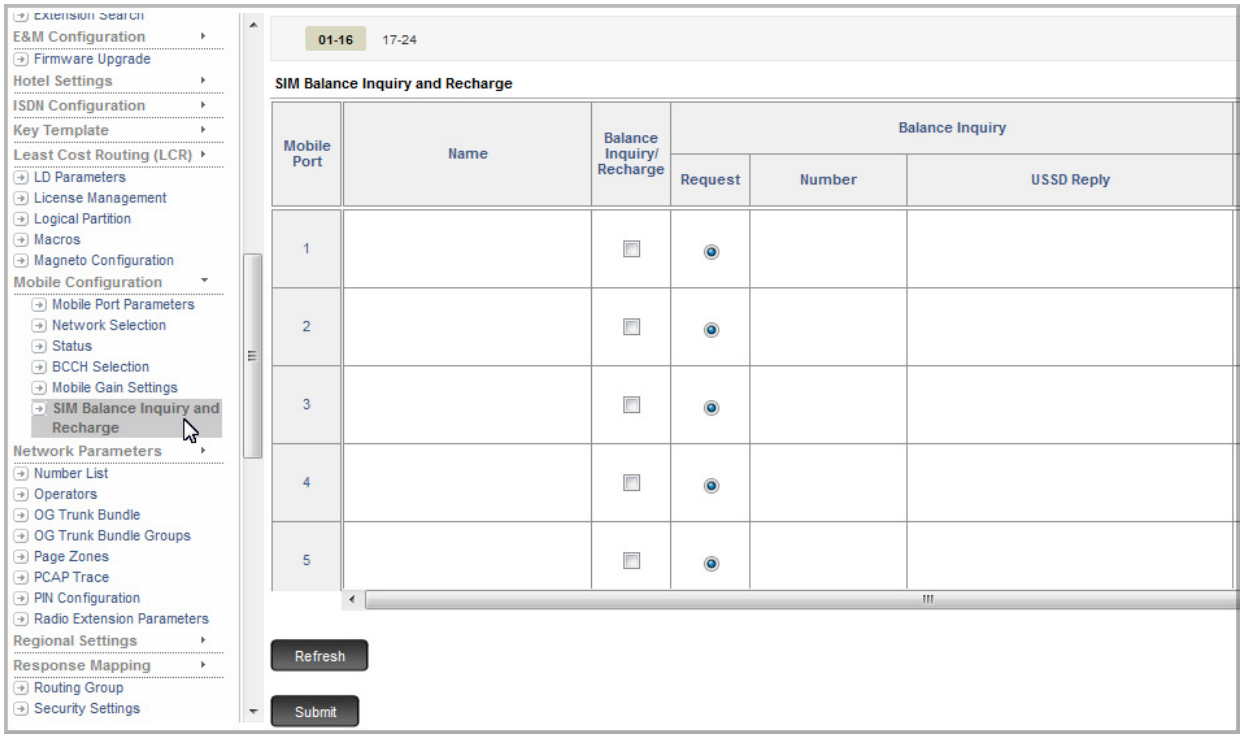
SIM Card Balance Inquiry and Recharging using Jeeves

- Login to Jeeves as System Engineer.

- Under **Configuration**, click **Mobile Configuration**.

333. SARVAM UCS supports Unstructured Supplementary Service Data (USSD), the standard for transmitting information over CSM signaling channels and a commonly used method to query the available balance and other similar information in pre-paid GSM services.

- Click **SIM Balance Recharge** to open the page.



- All the mobile ports configured in the system will appear on this page, by their Names you programmed when configuring the mobile trunk ports.

  If you have not programmed any name for a port, the Name field for that port will appear blank.

- Go to the Mobile Port on which you want to request SIM Balance/Recharge.

- Select the **Balance Inquiry/Recharge** check box to enable this feature.

Balance Inquiry

- To make Balance Inquiry,

  - Click the **Request** option button under **Balance Inquiry**, for all those Mobile Ports for which you want to request Balance Inquiry.

  - Enter the Balance Inquiry **Number** provided by the Network Operator whose SIM Card you have installed in the Mobile Port.

    A maximum of 16 digits are allowed. The valid digits for Balance Inquiry number are any digits from 0 to 9 and the characters * and #

  - Click **Submit**.

Recharging the SIM

- To recharge the SIM,
    - Click the **Request** option button under **Recharge**, for all those Mobile Ports for which you want to make a recharge request.
    - **Number:** Enter the Recharging Service Number provided by the Network Operator in this field.

      A maximum of 16 digits are allowed. The valid digits for Recharging Service number are any digits from 0 to 9 and the characters * and #
    - **PIN:** Enter the PIN number which is printed on the Recharge Voucher/Coupon. Your Recharge PIN number may consists of a maximum of 20 digits.

      The valid digits for PIN number are any digits from 0 to 9 and the characters * and #.

      Make sure you enter the digits and characters of the Recharge PIN number exactly as given on the Recharge Voucher/Coupon.
    - Click **Submit**.
    - Click **Refresh** at the bottom of the page.
    - In **USSD Reply** the response received from the GSM network (including possible error messages) will be displayed. When the USSD Reply is received from the network, it will appear with the Date and Time stamp of SARVAM UCS in this field.
- For each port that you send a Balance Inquiry/Recharge Request, you will get this USSD-Reply: "Please wait, processing the request. Refresh the page to see the current status."
- Click **Refresh** bottom of the page.
- The response received from the GSM network (including possible error messages) will be displayed under **USSD-Reply**. When the USSD Reply is received from the network, it will appear with the Date and Time stamp of SARVAM UCS in this field.
- You may log out of Jeeves.

- *The USSD code functionality is applicable when the Preferred Network Mode is not LTE only and if the Service provider supports USSD code.*
- *For each Mobile Port (SIM Card) at a time you can either request Balance Inquiry or Recharge the SIM Card.*

  However, you can send Balance Inquiry/Recharge request for all the Mobile Ports available in the system.
- *During Balance Inquiry/Recharge-Request, the status of the Mobile port will be 'busy'. It will become idle only after the USSD response is received from the GSM network.*
- *The SARVAM UCS will clear the USSD Reply after system restart. So each time you open the 'SIM Balance and Recharge' page after system restart, the USSD Reply box will be blank.*

SMS Gateway

This feature is not applicable if CDMA Mobile Card is installed in your system.

What's this?

SARVAM UCS offers the SMS Gateway interface to extend UC functionality.

The SMS Gateway feature of SARVAM UCS enables you to send/receive messages to/from individuals, selective groups or masses using the Mobile Port of SARVAM UCS.

SARVAM UCS allows you to register multiple SMPP Clients (Software Applications used for sending/receiving messages) with SARVAM UCS. SARVAM UCS functions as an SMPP Server. These Clients can send/receive messages using the Mobile port/s of SARVAM UCS.

The messages are sent using the Short Message Peer to Peer Protocol (SMPP Version 3.4). Using this encoding, it is possible to send up to 160 7-bit characters in one message, in the GSM network.

To use this feature you must purchase the SMS Gateway License. Refer to the topic "License Management" to know more.

How it works

For this feature to work,

- you must have the SMS Gateway license.
- you must have the SMPP Clients installed on a computer connected in the same LAN as SARVAM UCS.
- you must configure the required parameters in the SMPP Clients to register itself with SARVAM UCS
- you must define the Mobile port through which the messages are to be sent/received in the SMPP Clients.
- you must configure the SMPP Client parameters in SARVAM UCS.

This is how the SARVAM UCS SMS Gateway works,

- On successful registration of the SMPP Client with SARVAM UCS, a binding is established between the SMPP Client and SARVAM UCS.
- The SMPP Client can bind itself with SARVAM UCS as a Receiver, Transmitter or Transceiver.
    - As a Transmitter, the SMPP Client will only be able to send messages using the Mobile port of SARVAM UCS.
    - As a Receiver, the SMPP Client will only be able to receive messages from the Mobile port of SARVAM UCS.
    - As a Transceiver, the SMPP Client will be able to send and receive messages from the Mobile port of SARVAM UCS.

- The SMPP Client sends all the information required to send an SMS— the message content, the destination mobile number and the mobile port through which the message is to be sent to SARVAM UCS— in the Protocol Data Unit (PDU) format.
- The message will be sent to the destination number through the Mobile port, if it is idle or in speech. The message will be sent using the SMS Center Number configured for that port.

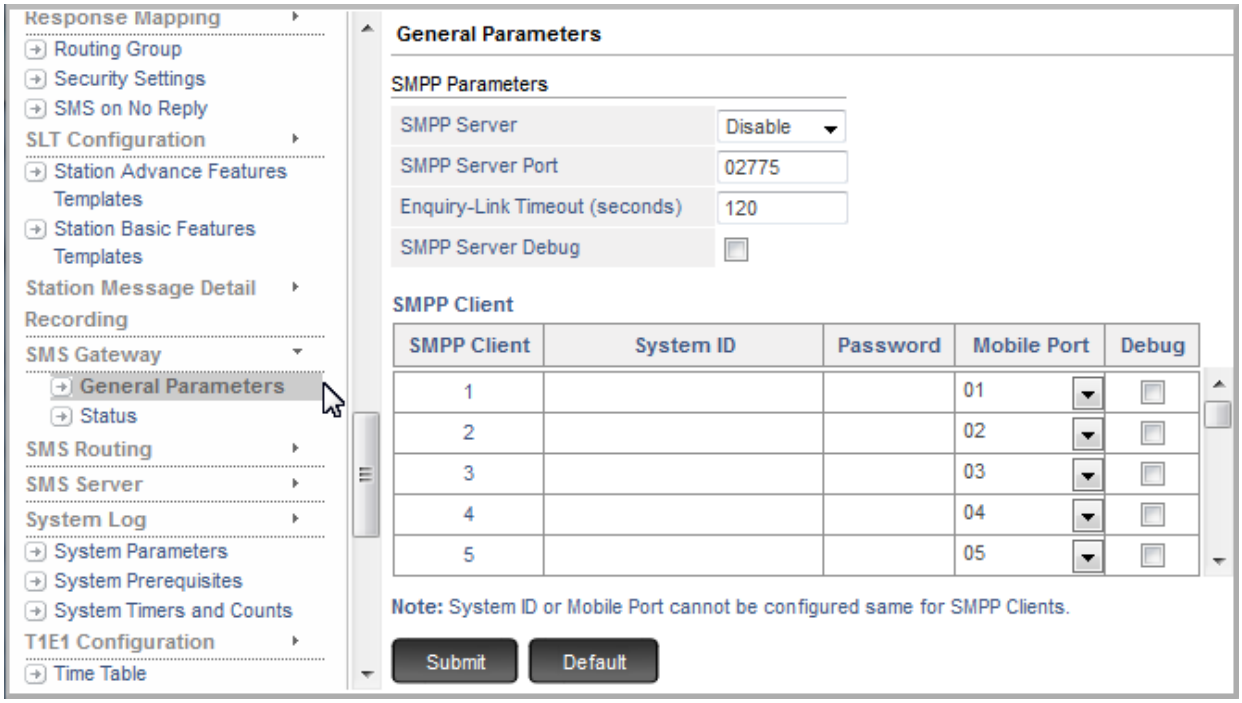
- *If the Mobile Port is any other state the SMS will not be sent.*
- *If the Mobile port is disabled/unregistered, the messages will not be sent.*

- SARVAM UCS will accept a new SMS sent by the SMPP Client for the same Mobile port only after the preceding SMS is successfully sent.
- All incoming SMS on the Mobile port will be sent to the SMPP Client.
- After the SMS is sent to the SMPP Client the same will be deleted from the SIM card of the Mobile port.

How to Configure

For the SMS Gateway you need to configure the following parameters,

- Log in as System Engineer.
- Under **Configuration**, click **SMS Gateway**.



- Click **General Parameters**. The SMPP Server - Parameters page opens.
  - **SMPP Server:** By default, the SMPP Server is disabled. To use the SMS Gateway feature, select Enable.

- **SMPP Server Port:** Enter the SMPP Server's listening port. The valid range is 1025 to 65535. Default: SMPP Server Port is 2775.

- **Enquiry-Link Timeout (seconds):** The SMPP Client sends the Enquiry-Link Requests to the SMPP Server at regular intervals to refresh its binding with the Server. The system re-loads this timer (default:120 seconds) every time it receives the request from the Client. If no response is received from the SMPP Client before the expiry of this timer, the Server considers the Client as disconnected. The valid range of the timer is 005 to 999. Default: 120 seconds.

- **SMPP Server Debug:** To monitor the events and processes of the SMPP Server, for trouble shooting and identifying faults and errors, select the SMPP Server Debug check box. The debug messages are sent to the remote Syslog Server. For detailed instructions on how to configure the Destination Port, Syslog Server IP Address and Port, see "System Debug".

- You can register multiple SMPP Clients with SARVAM UCS. Against each SMPP Client configure the following parameters:

The maximum number of SMPP Clients which you can register is equal to the maximum number of Mobile ports supported by SARVAM UCS, that is 64.

  - **SMPP Client System ID**: Enter the ID you want the Server to use to authenticate the Client. A maximum of 16 characters, including all ASCII characters are allowed.

  - **SMPP Client Password:** Enter the password you want the Server to use to authenticate the Client. To avoid unauthorized access, make sure the password is strong and is kept confidential. The Password can be of maximum of 9 characters, including all ASCII characters are allowed.

Make sure you:

- *Do not assign the same SMPP Client System ID and Password to multiple SMPP Clients.*

- *Configure the same SMPP Client System ID and Password in the SMPP Client. This ID is sent to the Server in the connection request made by the Client.*

  - **Mobile Port:** Select the Mobile Port using which the SMS is to be sent from/to the SMPP Client. The SARVAM UCS supports a maximum of 64 Mobile ports[334].

Make sure you do not assign the same Mobile Port to multiple SMPP Clients.

  - **Debug:** If you want to monitor the events and processes of the SMPP Client, for trouble shooting and identifying faults and errors, select the **Debug** check box. The debug messages are sent to the remote Syslog Server. For detailed instructions, for configuring the Destination Port, Syslog Server IP Address and Port, see "System Debug".

Viewing SMPP Server Status

You can view the status of SMPP Server on Jeeves only. To do this,

- Log in as System Engineer.

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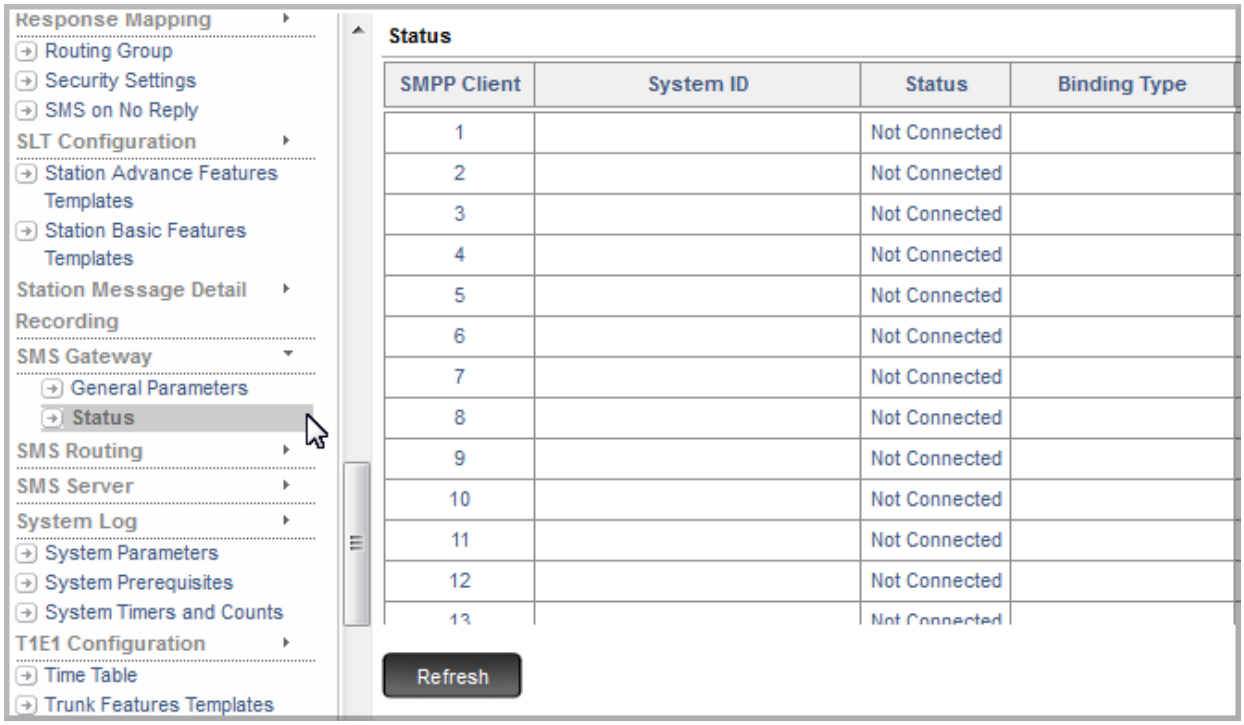
334. Depends on the model you have. Please refer the Appendix for an overview of the system resources and maximum expansion capacity.

- Under **Configuration**, click **SMS Gateway**.
- Click **Status**.



- The following parameters are displayed for each SMPP Client registered with SARVAM UCS:
  - **System ID:** This is the ID received in the SMPP Client's connection request.
  - **Status:** This field displays any one of the following:

Status	Description
Not Connected	When the SMPP Server does not receive any request from the SMPP Client.
Connected	When the SMPP Server receives a request from the SMPP Client and it accepts it. A connection is established between the Server and the Client.

  - **Binding Type:** This field displays the type of SMPP Client Binding, that is Transmitter, Receiver or Transceiver.
  - **Address:** This field displays the SMPP Client's IP Address and Port.

SMS on No Reply

This feature is not applicable if CDMA Mobile Card is installed in your system.

What's this?

Among the various UC features offered by SARVAM UCS, SMS on No Reply is the most useful business UC feature.

SMS on No Reply feature of SARVAM UCS is used to send an SMS to the mobile user, when an extension user calls the mobile user and the mobile user doesn't answer the call. The SMS that is sent to the mobile user contains the extension user's contact information and the message to be conveyed in brief, so that the mobile user can call back if necessary.

How it Works

For this feature to work,

- you must have installed a GSM Card.
- you must have the SMS Server license to route the SMS through the Mobile ports. For instructions, see *"License Management"*.
- you must enable the **Send SMS** check box in "Mobile Port Parameters".
- you must enable **SMS on No Reply** in the **Station Advanced Feature Template** assigned to the extension user. For instructions, see "How to Configure".
- you must configure the SMS on No Reply parameters. For instructions, see "How to Configure".

This is how the SMS on No Reply works,

- An extension user A calls an external mobile number 9898906336. This call is routed using the Mobile Port of the SARVAM UCS.

- The call routed through the Mobile Port will be considered as no reply only if 'No Reply' event is received from the service provider.

- *The SMS on No Reply will not be applicable for calls routed through the CO Port.*

- *If the call is routed through the SIP it will be considered as no reply when cause 408 is received from the network.*

- *If the call is routed through the BRI/ T1E1 it will be considered as no reply when cause 18 (No User Response) or cause 19 (No Answer from User) is received from the network.*

- After the No reply event is received from the service provider, the system matches the mobile users number with the numbers configured in the Number list for sending SMS.

- If a match is found, the system sends the SMS to the mobile user. The SMS contains the Name and/or Number of the extension user A as well as the message. This message can be customized as per extension user requirement.

- In this way the mobile user can callback the extension user, if required.

How to Configure

For the SMS on No Reply feature to work, you need to configure the following parameters:

- **SMS for OG Call - No Reply** in the Station Advanced Feature Template
- **Number list for sending SMS** for SMS on No Reply.
- **Send SMS** option for routing the SMS through the Mobile ports.
- **SMS Text** message to be sent as SMS to mobile users.

Configuring SMS on No Reply in the Station Advanced Feature Template

The default Station Advanced Feature Template 01 is assigned to all extension users of the SARVAM UCS. In this template SMS on No Reply is disabled.

Decide which extensions are to be allowed 'SMS on No Reply'. If you want to allow SMS on No Reply to all extensions, retain the default Station Advanced Feature Template 01 and configure the SMS on No Reply parameters. However, if you want to allow SMS on No Reply only to selected extensions, select another Station Advanced Feature Template and configure the SMS on No Reply parameters in that template.

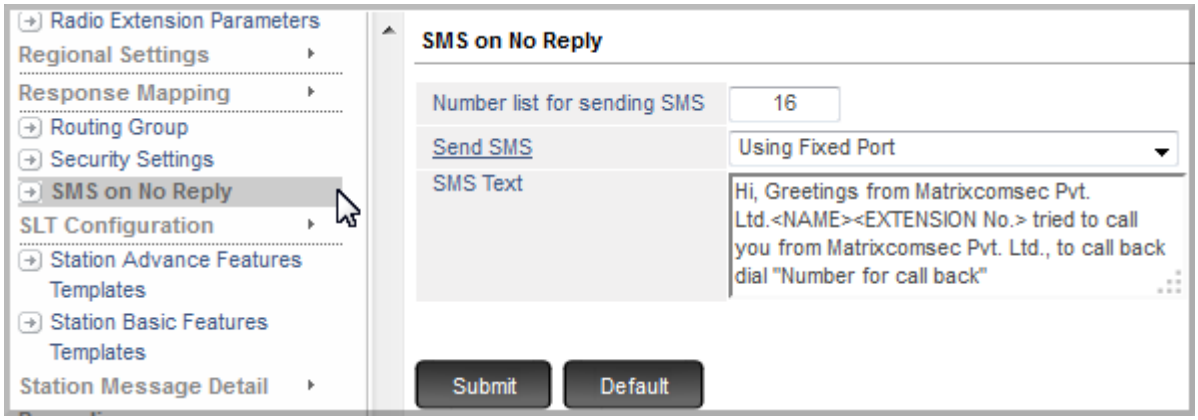
Now, to assign the template to selected extensions, follow these steps:

1. Prepare a Station Advanced Feature Template with SMS on No Reply.
2. Assign this newly prepared template to the desired extensions.

Refer the topics "Station Advanced Feature Template" for detailed programming instructions on how to customize a Station Advanced Feature Template, configure the SMS on No Reply and how to apply this template on extensions.

Configuring SMS on No Reply Parameters

- Log in as System Engineer.
- Under **Configuration**, click **SMS on No Reply**.



- Configure the **Number List for sending SMS.** In this list enter the number strings that you expect the callers to dial. By default, Number List 16 is assigned to SMS on No Reply.

Refer to the topic “Number Lists” to know more, and for configuration instructions.

- Select the desired option in **Send SMS** to route the SMS. You can select Using Fixed Port or Based on LCR Table. See “Fixed Port Routing (SMS Server)” and “Least Cost Routing” for instructions.

- You can customized the **SMS Text** as per your requirements. The default text message is **Hi, Greetings from Matrixcomsec Pvt. Ltd.<NAME><EXTENSION No.> tried to call you from Matrixcomsec Pvt. Ltd., to call back dial "Number for call back".**

  Here, in the SMS that is sent the **NAME** will be replaced with the name of the calling extension user and **EXTENSION No** with the extension number assigned to the calling user. Make sure that the **NAME** and **EXTENSION No** are written in the same format as displayed in the default message. The callback number can be mentioned as text, if required or it can be removed.

- Click **Submit**.

SMTP Settings

The SMTP Settings must be configured if you wish to send mails for VMS applications and/or register the SMS Server as SMTP Client.You can configure different account for each application. The systems allows you to configure a maximum of three accounts.

The Unified Messaging Functionality of SARVAM UCS includes using SMTP to send emails for the following functions:

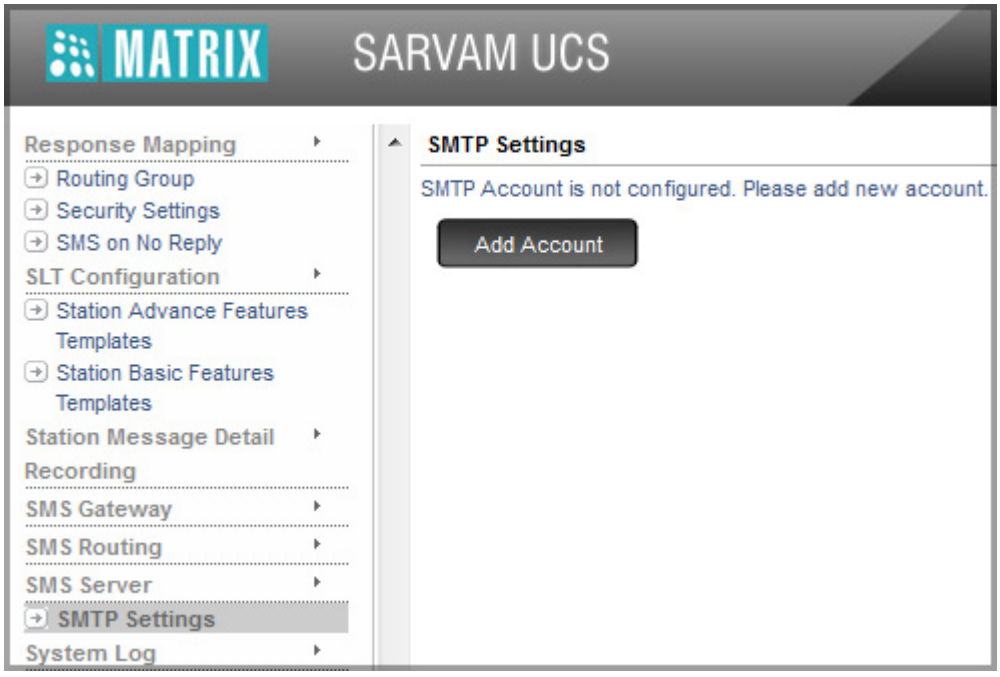
- send notification to the extension users about the arrival of new messages (with/without Voice Mail Attachment).
- send notification to the extension users about the memory usage status of their mailbox.
- memory usage notification to SE.
- send mails from SMTP Clients while using the SMS Server Application.

For email transmission, you must:

- configure the parameter *Message Wait Notification via Email*[335] in the VMS settings of the extension. For instructions, see “Email Based Notification”.
- configure the parameters *VMS E-mail Notification* and *Use SMTP Account* in VMS General Parameters. For instructions, see “Configuring VMS General Parameters”.
- configure the Mail Settings in SMS Server, see “SMTP Configuration”.
- configure the SMTP Accounts and its parameters.

To configure SMTP Settings using Jeeves,

- Log in as System Engineer.
- Under **Configuration**, click **SMTP Settings** to open the page.
- By default no account is configured. You can add a maximum of three accounts. Click **Add Account** and then configure the following parameters:



335. *You can also have the new voice message mailed as an attachment with the message wait notification.*

- Configure the **Account Name.** This will help to identify the application for which the account is being used.

- Configure the **SMTP Server Address** and **SMTP Server Port**. This is the Server's IP Address and Port number that will be used to send outgoing mails. Both IPv4 and IPv6 addresses are supported. If port is not programmed, the default port value 25, will be used. Valid Port range: 25;465;587;1025 to 65535. The Server Address can be a maximum of 40 characters.

- Configure the **Email ID** for the account registered with the Email Server, as provided by your network administrator. This Email ID will appear to the recipient as the originator of the email (that is in the FROM field). The Email ID you configure may consist of a maximum of 64 characters. Default: Blank.

- If your Email Server uses authentication, select the **Require Authentication** check box. Default: Disabled. If you have enabled authentication, you must also configure the *User ID* and the *Password.*

- If you have enabled authentication, configure the **User ID** and the Authentication **Password** as provided to you by your network administrator. The User ID may consist of a maximum of 40 characters and the Password can be a maximum of 24 characters. Default: Blank.

- To transport all data in a secure manner, select **Enable Secure Socket Layer (SSL)** check box. All the data to the Email Server will be transported over secure layer. Default: Disabled.

- Configure the **Display Name**. This name will be displayed to the mail recipient. You can configure a maximum of 24 characters. Only ASCII characters are allowed. Default: Blank.

- In **Connection Timeout Interval** configure the time duration for which you want the system to wait for a response from the SMTP server. You may change the Connection Timeout Interval timer, if required. The range of Connection Timeout Interval timer is 01 to 99 seconds. By default, it is set to 60 seconds.

- The **SMTP Application used by Application** displays the name of the application/s — SMS Server, VMS Notification — which is using the account. It will be blank if the account is not being used by any application.

- Click **Submit** to save SMTP settings.

Test SMTP Settings

- Click the **'Test Account'** button to check if the SMTP Parameters have been configured correctly.

  When you click this button, the alert message will appear: *"Testing SMTP can take up to 99 seconds. Would you like to continue?"* Click 'OK' button.

  The Test Result will be displayed in 'Test Status' field.

- **Test Status:** Any one of the results listed below may appear in this field:

Test Status Message	Description
"SMTP Server Connection Not Established"	When connection to SMTP server fails due to any reason.
"Login to SMTP Server Failed"	When connection to SMTP server is established but login to SMTP server fails due to any reason.

Test Status Message	Description
"Sending Test Mail Failed"	When connection to SMTP has been established successfully but there is no acknowledgment for the test mail sent.
"Test Mail Sent Successfully"	When acknowledgment for the test mail sent to SMTP server received successfully.
Already Test is running	When the test is already being processed for another SMTP account.
Timeout	When the test is in process and the Connection Timeout Interval expires.

Email Failure Errors

When message sending fails, the events will be logged into the System Fault Log with a specific code. For more information, see “System Fault Log”.

Simple Network Management Protocol (SNMP)

What's this?

Simple Network Management Protocol (SNMP) is an application-layer protocol used for exchanging management information between network devices. Using SNMP, you can manage and monitor network elements, audit network usage, detect network faults or inappropriate network access.

The SNMP architecture consists of:

- An **SNMP Agent** is a program that is bundled within the managed device. SNMP agent allows a managed device to collect the Management Information Base from the device and make it available to the SNMP Manager on request. It receives SNMP requests and generates SNMP responses or notifications (traps/informs). The SNMP Agents are SNMP Servers. Here, it is SARVAM UCS.

- **SNMP Manager**, usually the Network Management Station. The manager communicates with multiple SNMP Agents implemented in the network. It generates SNMP requests and receives SNMP responses and notifications (traps/informs). The SNMP Manager is an SNMP Client. Here, it is the SNMP Manager installed in the PC.

- **Managed device** or the network element is a part of the network that requires some form of monitoring and management. For example, switch, routers, servers.

- **Management Information Base** is the commonly shared database between the Agent and the Manager. This information is known as managed objects. These managed objects are defined in MIB (Management Information Base) module.Any sort of status or information that can be accessed by the Manager is defined in a MIB.

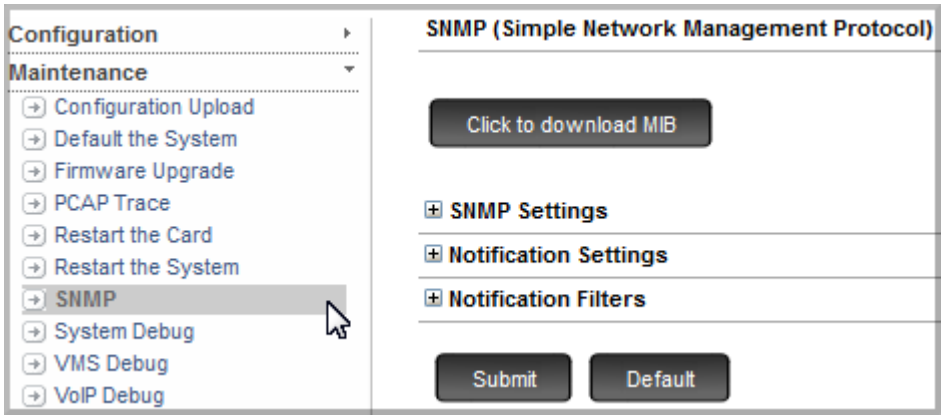
SNMP uses UDP (User Datagram Protocol) as the transport protocol for passing information between Managers and Agents. By default, the Agent listens on UDP port 161 for requests from Manager and the Manager listens on UDP port 162 for notification from Agent.

How to Configure

- Log into Jeeves.

- Click the **Maintenance** link.

- Click the **SNMP** link.



MIB Files

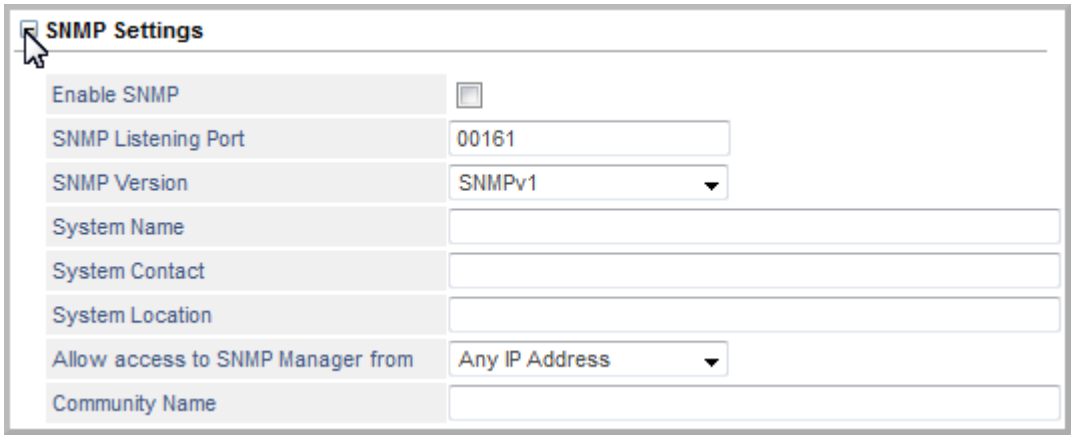
If the Manager needs to access any status or information about the client, the Manager must download and install the MIB files on the local disk.

To download these files,

- Click the **Click to download MIB** button.
- You will get a prompt with the option to open the file or save the file to a location. Save the file on the local disk.

SNMP Settings

- Click **SNMP Settings** to expand.



- Select the **Enable SNMP?** check box. Default: Disabled.
- Configure the **SNMP Listening Port**. Valid Range:161, 1025-65535. Default: 161.
- Select the **SNMP Version** as supported by your SNMP Manager. You can select from:
  - SNMPv1

- SNMPv2c
- SNMPv3

For enhanced security, you must select SNMPv3.

- Configure the **System Name**. When there are multiple devices connected in the same network, the name configured helps to identify the SARVAM UCS within the network. The System Name can be a maximum of 40 characters. Default: Blank.

- Configure the **System Contact**. It is the name and number of the person to be contacted, in case of notification. The System Contact can be of a maximum of 40 characters. Default: Blank.

- Configure the **System Location**. This is the physical location of SARVAM UCS. This information is helpful to the administrator. The System Location may consist of a maximum of 40 characters. Default: Blank.

- Configure **Allow access to SNMP Manager from** to allow/restrict accessibility to the SNMP Manager. You can select **Any IP Address** or **Specific IP Addresses**. Both IPV4 and IPv6 addresses are supported.

  If you want to restrict the accessibility of the SNMP Manager, select **Specific IP Addresses**, configure **IP Addresses 1** to **5**. Default: Blank.

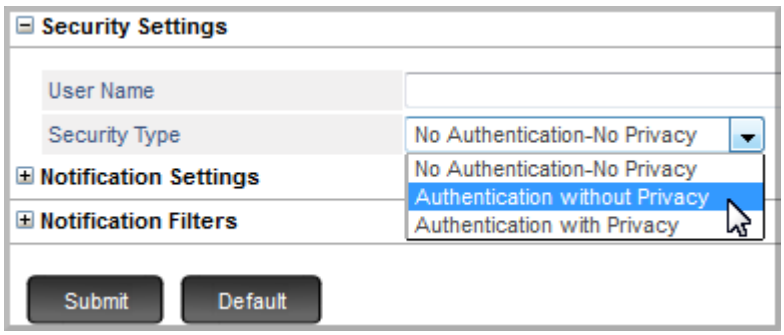
- If SNMP version is set as **SNMPv1** or **SNMPv2c,** configure **Community Name**.

  Community Name identifies the SNMP community in which the sender and recipient of the message are located. It enables communication between SARVAM UCS and the Manager. The Community Name can be a maximum of 40 characters. Default: Blank. To avoid unauthorized access, we recommend you to assign a strong Community Name.

- If SNMP version is set as **SNMPv3,** the **System's Engine ID** is displayed. This is a unique identification of the system. It is a hexadecimal field with length of 22 characters. The ID consists of:
  - Enterprise Number (800086df03 which is fixed)
  - MAC Address of the system (MAC address of Ethernet (LAN/WAN) Port)

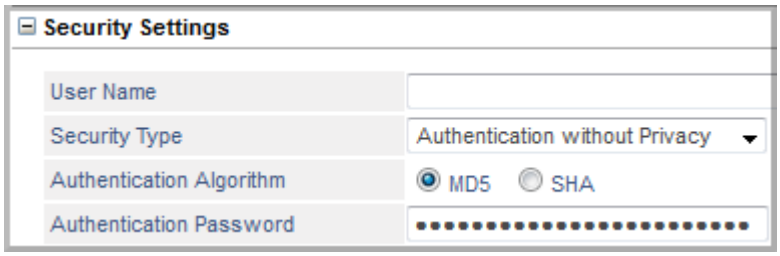
Security Settings

- If SNMP version is set as **SNMPv3**, click **Security Settings** to expand and configure the following.



- Enter the **User Name**. The User Name can be a maximum of 40 characters. User Name will be used for authentication and privacy in SNMPV3.

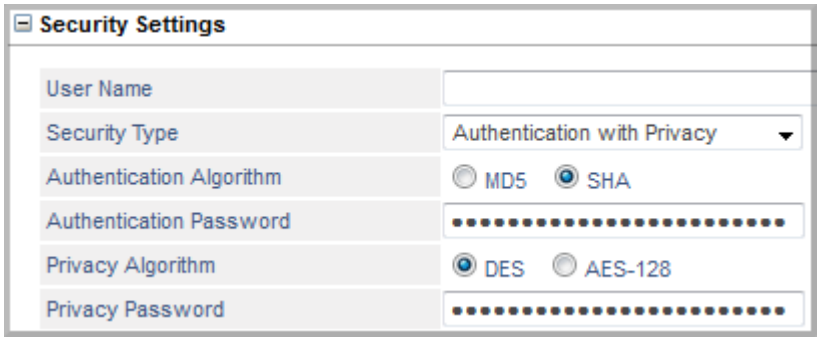
- Select the appropriate **Security Type** as per your requirement. Security Type defines the level of security.
    - When Authentication and Privacy are not required, select **No Authentication-No Privacy**
    - When only Authentication is required, select **Authentication without Privacy**. Incoming SNMP Messages will require authentication.



If you select this method, select the **Authentication Algorithm** as **MD5** or **SHA**. Default: MD5.

In the **Authentication Password**, enter a password of your choice as Authentication Password for the User Name you have assigned.To avoid unauthorized access, we recommend you to use a strong password and make sure it is kept confidential. The Authentication Password must be a minimum of 8 characters and may have upto 24 characters. Default: Blank.

- When both Authentication and Privacy are required, select **Authentication with Privacy**. Incoming SNMP Message will require authentication and these messages will be encrypted, which will be decrypted at the receivers end only.



If you select this method, select the **Authentication Type** as **MD5** or **SHA**. Default: MD5.

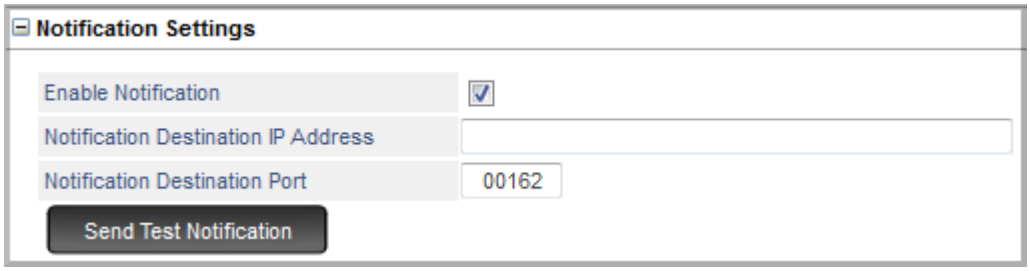
Enter **Authentication Password** for the User Name you have assigned. To avoid unauthorized access, we recommend you to use a strong password and make sure it is kept confidential. The Authentication Password must be a minimum of 8 characters and may have upto 24 characters. Default: Blank.

Select the **Privacy Algorithm** as **DES** or **AES**. Default: DES.

Enter the **Privacy Password** of your choice.We recommend you to use a strong password and make sure it is kept confidential. The Privacy Password must be a minimum of 8 characters and may have upto 24 characters. Default: Blank.

Notification Settings

- Click **Notification Settings** to expand.



If SNMP version is set as **SNMPv1**, configure the following parameters.

- Select **Enable Notification** check box, if you want SARVAM UCS to generate Trap message for an error.
    - You must configure the **Notification Destination IP Address** and **Notification Destination Port** of the Manager or of any other device where you want to receive the trap messages.

      SARVAM UCS will send the notification (error message) to the destination configured.

      Valid range of the Notification Destination Port is 162 or 0-65535. Default port is 162.
- Click **Submit** button to save the settings.

If SNMP version is set as **SNMPv2c or SNMPv3**, configure the following parameters.

- Select **Enable Notification** check box, if you want SARVAM UCS to generate Trap or Inform message for an error.
- Select the **Notification Type**. You may select **Trap** or **Inform**.

  If you want the system to send notification message without acknowledgment, select **Trap**.

  If you want the system to send notification message with acknowledgment, select **Inform**.
    - If you select **Inform** as the *Notification Type*, you must configure Retry Attempts and Retry Interval.

      If acknowledgment is not received from the Manager for the notification sent, the system will keep retransmitting the message for the number of attempts you have configured as the **Retry Attempts**. Default: 3.

      The system will retransmit the messages at regular time intervals you have configured as **Retry Interval**. Default: 10 seconds.
    - You must configure the **Notification Destination IP Address** and **Notification Destination Port** of the Manager or of any other device where you want to receive the trap messages.

      SARVAM UCS will send the notification (error message) to the destination configured.

      Valid range of the Notification Destination Port is 162 or 0-65535. Default port is 162.

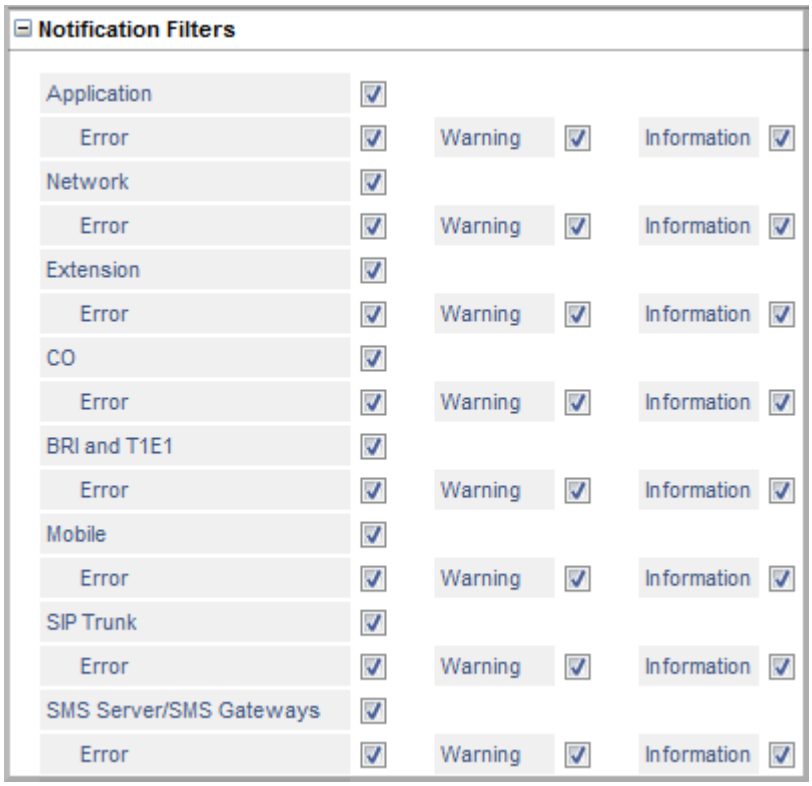
- Click **Submit** button to save the settings.
- Click the **Send Test Notification** button, to send a test notification message.

Notification Filters

By default, you get notifications of errors, information and warnings for events related to the Application, Network, Extension, Trunk Ports —CO, Mobile, SIP Trunks, T1E1and BRI, SMS Server/Gateway, Voice Mail System and VoIP. Refer to the table at the end of this topic for the event list. You can choose the type of notification you want by setting the notification filters.

To set the filters,

- Click **Notification Filters** to expand.



- By default, all the filters are enabled. To disable any filter, clear the respective check box.

Simple Network Time Protocol - SNTP

What's this?

SNTP is a protocol used to synchronize the clocks to some time reference. SARVAM UCS supports both, SNTP Client as well as SNTP Server.

SNTP Client

SARVAM UCS has its own Real Time Clock (RTC) to store date and time. When you select the Region, the RTC parameters are set automatically.

However, the RTC can drift over a long period. So, you may check and reset the RTC values at regular intervals to correct this drift. You can set the RTC manually (see "Real Time Clock (RTC)") or synchronize it with any SNTP Server in the Public Network. To synchronize time with the SNTP Server SARVAM UCS supports SNTP Client.

SNTP Server

SARVAM UCS supports IP Phones and these phones have SNTP Clients so that they can synchronize their clock with a SNTP Server.

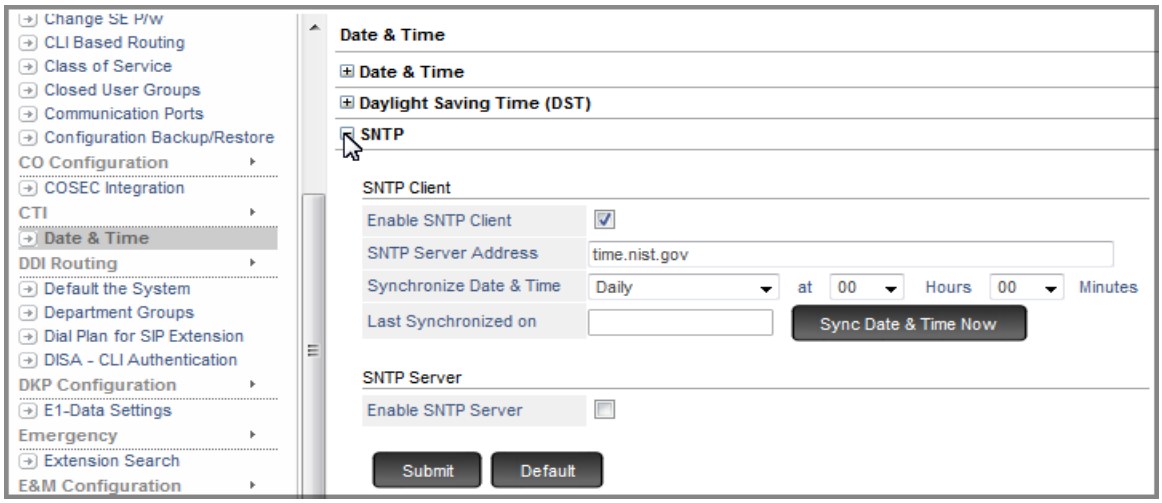
These SNTP Clients in the phones can synchronize their clock with the SNTP Server on the public network only if Internet is available in the network. Hence, to overcome this SARVAM UCS also supports an SNTP Server with which the SNTP Clients can synchronize their time without internet connectivity.

How to Configure

SNTP Client

To use SNTP for synchronizing with the Real Time Clock,

- Log in as System Engineer.
- Under **Configuration** click **Date and Time**.
- Click **SNTP** to expand.



- **Enable SNTP Client:** When this check box is enabled, SARVAM UCS behaves as SNTP Client. This parameter enables you to synchronize RTC of system with the SNTP server in the external network. Default: Enabled.

- **SNTP Server Address:** Enter the Time Server Address. SNTP Server address can be of maximum 64 characters. Both IPv4 and IPv6 addresses are supported. Default: time.nist.gov

- **Synchronize Date and Time:** You can Synchronize Date and Time with the SNTP Server on daily or weekly basis. If you select weekly, the system will sync the date and time with the SNTP Server on every Monday. Select the desired option as per your requirement. Default: Daily.

- **Last Synchronized On:** This field displays the time when the system last synchronized with the SNTP Server.

- **Sync Date and Time Now:** Click this button to synchronize date and time of SARVAM UCS with the SNTP server, whenever required.

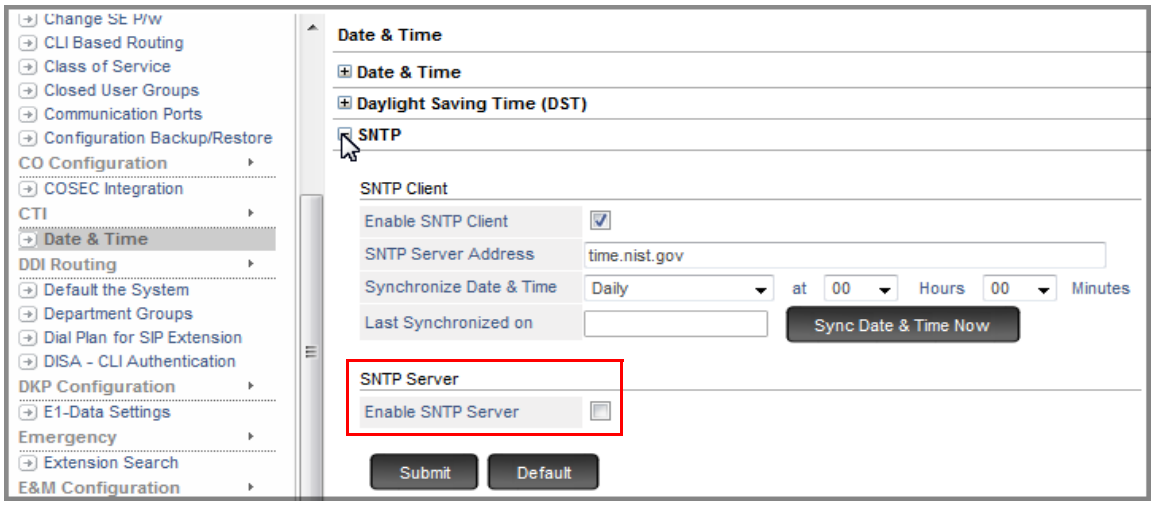
SNTP Server

To enable the SNTP Server,

- Log in as System Engineer.

- Under **Configuration** click **Date and Time**.

- Click **SNTP** to expand.



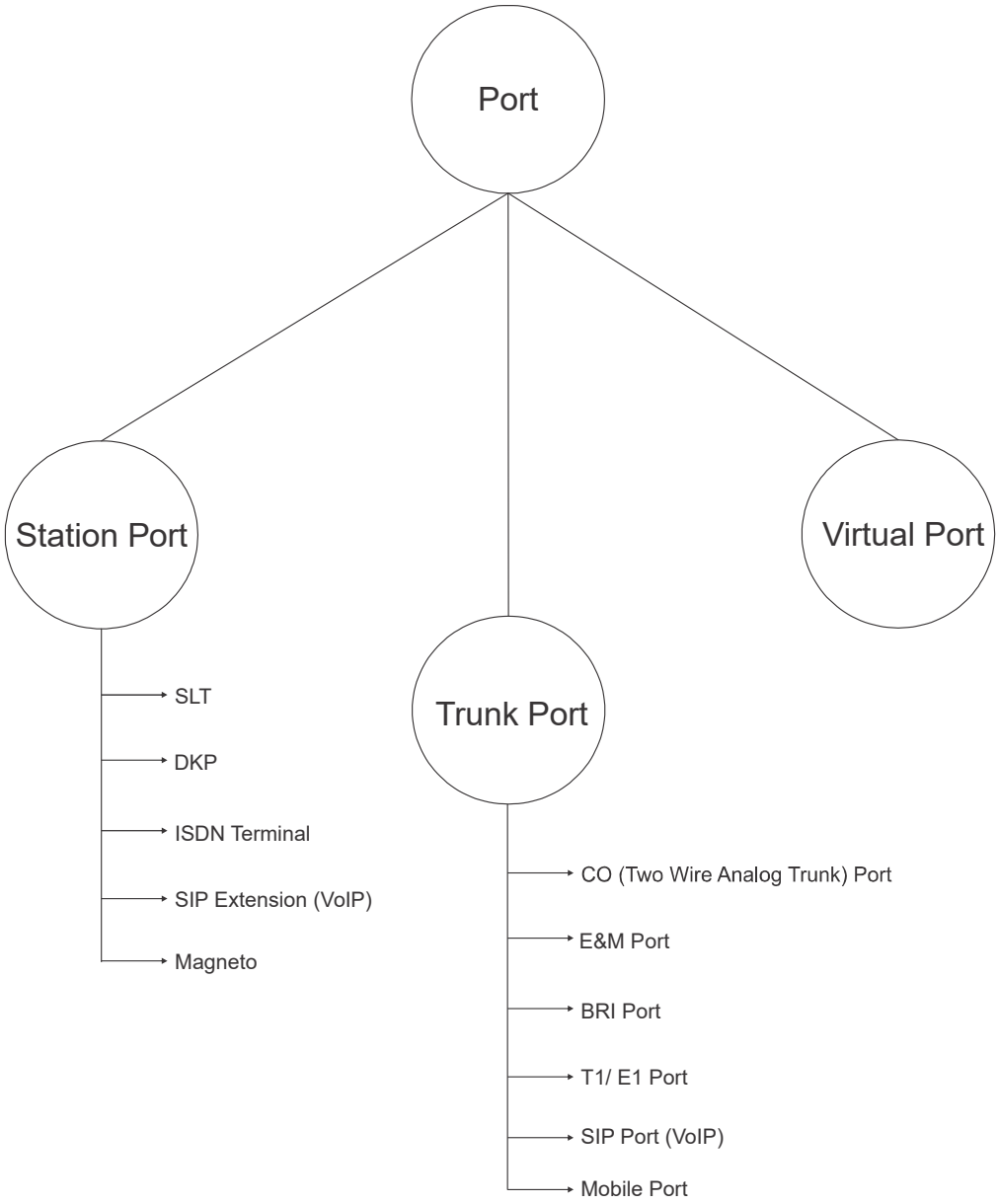
- **Enable SNTP Server:** When this check box is enabled, SARVAM UCS behaves as SNTP Server and the extensions (IP Phones) registered with it behave as the clients. Default: Disabled.

- Click **Submit** to save the changes.

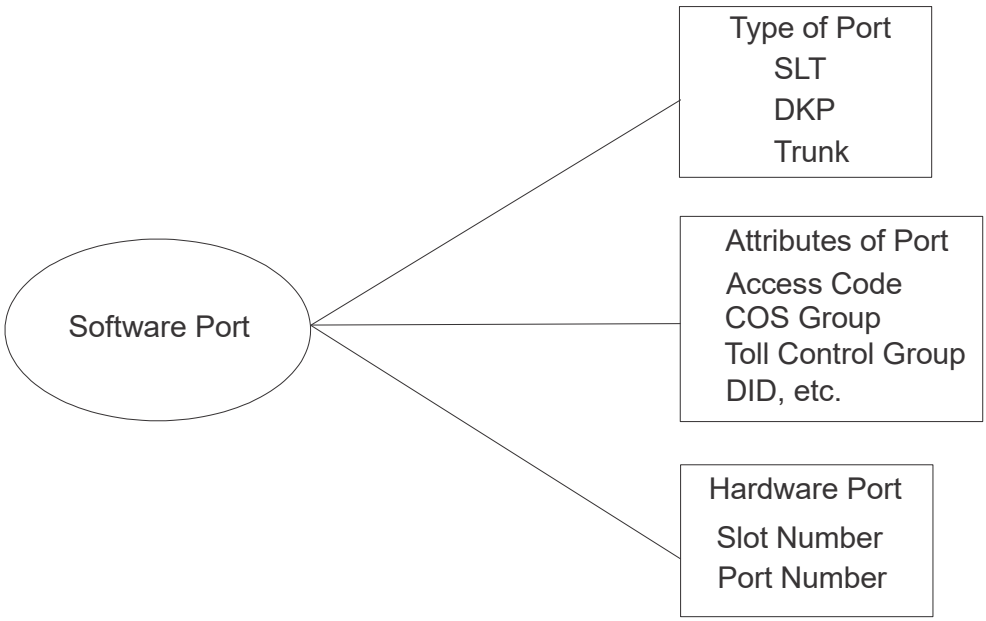
Software Port and Hardware ID

What's this?

The SARVAM UCS supports different types of ports as illustrated below:



The SARVAM UCS treats a port as an entity and processes it on the basis of port type and its programmed attributes.



Software Port

The SARVAM UCS takes a software port as fundamental entity. It processes the software port. Hardware ID and the access code are just two attributes of a software port and hence they are not used anywhere in processing and programming.

Software ports are always numbered from 1 to the maximum number of ports supported for each port type. The following table lists this range for each port type.

Port Type	Max. Ports Supported	Port No. Range[a]
SLT	240	001-240
DKP	96	001-096
CO Trunk	64	001-064
E&M	32	001-032
BRI	32	01-32
T1E1PRI	8	01-08
Mobile (GSM)	40	01-40
SIP Trunks	99	01-99
Magneto	16	001-016

a.The number of ports mentioned here are for ETERNITY GENX. For details regarding the number of ports supported for ETERNITY LENX/MENX/PENX, see “Technical Specifications - SARVAM UCS”.

Each type of software port has different attributes. The System Engineer (SE) programs these attributes using the corresponding template at the time of installing the SARVAM UCS.

Hardware ID is an attribute of a software port. Hence, all the programming is done for the software port and not for the hardware ID. Accordingly, the software port number is used for all programming tasks.

An access code is just a Flexible number assigned to the software port. Programming is for the software port and not for the access code (Flexible number). Accordingly, the software port number is used for all the programming.

The System Engineer allocates software port numbers to different users. This allocation is Flexible and any software port number can be assumed for any user. Hardware ID is not relevant at this stage. Hardware ID can be programmed for a software port any time. Further, it can be changed any time in case of hardware failure of a port.

Following example elaborates this point.

Name	Position	Port Type	S/w Port Used
Anil Sharma	Managing Director	DKP	001
Nikhil Rao	VP (Marketing)	DKP	002
Revathi Thyagarajan	VP (Finance)	DKP	003
Anand Chakraborty	Manager (MD)	SLT	001
Pankaj Shah	Accountant	SLT	003
Ravi Tandon	Sales Executive	SLT	002
Conference Room	--	DKP	005
Canteen	--	SLT	004
Leased Line	--	Trunk	003
STD Line	--	Trunk	001

Please take a note of following points:

- Software port numbers start from 001 for all different port types.
- Order is not important while allocating software port numbers.

Hardware ID

The hardware ID of a software port denotes where the port is physically located. To derive hardware ID of a software port, we need:

- Slot number of the card.
- Port number on that card (Port offset).

Slot Assignment for SARVAM UCS

Model	Port Name	Connector
ETERNITY LENX	01-27	Universal Slots
ETERNITY MENX	01-16	Universal Slots
ETERNITY GENX	01-12	Universal Slots
ETERNITY PENX	01-06	Universal Slots

Use following command to assign hardware ID to a SLT software port:

1101-SLT-Slot-Port offset on the card
Where,
SLT = Software port number of SLT from 001 to 512.
Slot=Slot number from, 01 to 16, in which the card is inserted.
Port offset = Port number on that card, from 01 to 99.
By default, it is auto assigned.
To clear the hardware ID assigned to a SLT software port use the following command:
1101-SLT-00-00

Use following command to assign hardware ID to a DKP software port:
1102-DKP-Slot-Port offset on the card
Where,
DKP = Software port number of DKP from 001 to 128.
Slot=Slot number from 01 to 16 in which the card is inserted.
Port offset = Port number on that card, from 01 to 99.

Use the following command to clear the hardware ID assigned to DKP software port:
1102-DKP-00-00

Use following command to assign hardware ID to DSS software port:
1103-DKP-DSS-Slot-Port offset on the card
Where,
DKP = Software port number of DKP from 001 to 128.
DSS is 0 or 1.
Slot=Slot number from 01 to 16 in which the card is inserted.
Port offset = Port number on that card, from 01 to 99.

Use the following command to clear the hardware ID assigned to the DSS software port:
1103-DKP-DSS-00-00
Use following command to assign hardware ID to a CO software port:
1104-CO-Slot-Port offset on the card
Where,
CO = Software port number of CO from 001 to 128.
Slot=Slot number from 01 to 16 in which the card is inserted.
Port offset = Port number on that card, from 01 to 99.
Use the following command to clear the hardware ID assigned to CO software port:
1104-CO-00-00

Use following command to assign hardware ID to an E&M software port:
1105-E&M-Slot-Port offset on the card
Where,
E&M = Software port number of E&M from 001 to 128.
Slot=Slot number from 01 to 16 in which the card is inserted.
Port offset = Port number on that card, from 01 to 99.

Use the following command to clear the hardware ID assigned to E&M software port:
1105-E&M-00-00

Use following command to assign hardware ID to a BRI software port:
1106-BRI-Slot-Port offset on the card

Where,
BRI is from 01 to 32.
Slot is from 01 to 16, in which the card is inserted.
Port Offset on the card is from 01 to 99. (port number 01 to 08 are practically used values).
By default, the Hardware Slot and Port Offset are auto assigned.

Use following command to de-assign the hardware slot and the hardware port assigned to the BRI software port.
1106-BRI-00-00

Use following command to assign hardware ID to a T1E1PRI software port:
1107-T1E1-Slot-Port offset on the card
Where,
T1E1PRI is from 01 to 08.
Slot = Slot number from 01 to 16, in which the card is inserted.
Port offset = Port number on that card, from 01 to 99 (port number 01 and 02 are practically used values).
By default, the Hardware Slot and Port Offset are auto assigned.

Use the following command to clear the hardware ID assigned to T1E1PRI software port:
1107-T1E1-00-00

Use following command to assign hardware ID to a Mobile port:
1108-Mobile Trunk Number-Slot-Port offset on the card
Where,
Mobile Trunk Number is from 001 to 064.
Slot is from 01 to 16.
Port Offset on the card is from 01 to 99. (port number 01 to 04 are practically used values).
By default, the Hardware Slot and Port Offset are auto assigned.

Use following command to de-assign the hardware slot and the hardware port assigned to the Mobile port.
1108-Mobile Trunk Number-00-00

Use following command to assign hardware ID to a Magneto port:
1110-Magneto-Slot-Port offset on the card
Where,
Magneto is from 001 to 128.
Slot is from 01 to 16
Port Offset on the card is from 1 to 8.

By default, the Hardware Slot and Port Offset is auto assigned.

Use following command to de-assign the hardware slot and the hardware port assigned to the Magneto port.
1110-Magneto port-00-00

Static Routing Table

What's this?

Static Routing Table is required when:

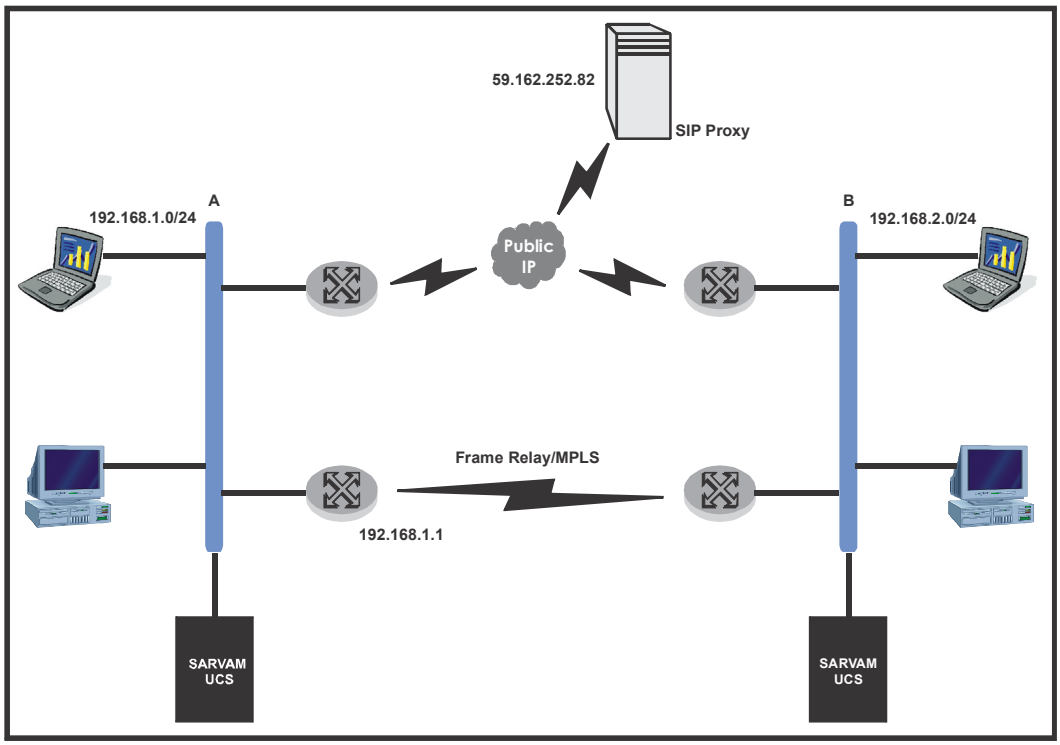
- you have more than one router (gateway) in your network and you want SARVAM UCS to send packets to multiple routers/gateways for different types of calls. These packets are routed through the WAN Port.
- you want to route specific packets through the LAN Port.

Static Routing Table helps route calls between point to point sites (connected through Multi Protocol Label Switching-MPLS, Frame Relay, etc.) and to public internet at the same time.

How it works

Routing the packets through the WAN Port

For example, two Local Area Networks, Network A and Network B, are connected through Frame Relay/ Multi Protocol Label Switching (MPLS) network to give access to local resources and also to make Peer-to-Peer calls.



At both sites SARVAM UCS is connected behind a router.

These sites are also connected to public IP network to:

- give internet access to local hosts.
- access DID service provided by ITSPs to make PSTN/ GSM calls over IP network.

Network A and Network B are in different subnets.

The Static Routing Table makes it possible to route different types of outgoing calls—Peer to Peer or Proxy—made to different subnets through different Gateways.

You can also access a PMS system installed in a different network using the Static Routing Table.

The Static Routing Table defines the appropriate Gateway Address (or Router's LAN Address) where the IP packets are to be sent.

In the Static Routing Table, you must configure:

- The address of the final Destination where the packets are to be sent.
- The Subnet Mask to be applied on the final destination address.
- The Gateway Address where the IP packets are to be sent.

When SARVAM UCS sends packets, if the final destination IP Address and SARVAM UCS are not in the same Subnet, the system will check the Static Routing Table.

If a perfect match is found, SARVAM UCS will start sending the IP packets to the corresponding Gateway Address configured in the table.

If no match is found, SARVAM UCS will send the IP Packets to the **Default Gateway Address** (Network Connection Type) you configured in the Network Parameters. For detailed instructions, see "Configuring Network Parameters".

Routing the packets through the LAN Port

In another scenario:

- SIP Trunk 1 is a proxy trunk
- Source Port IP Address on the SIP Trunk 1 is configured as LAN Port IP Address
- IP Address provided by the ITSP is 192.168.5.10 (Destination IP Address)
- LAN IP Address is 192.168.5.1

In the Static Routing Table, you must configure:

- The address of the final Destination where the packets are to be sent as 192.168.5.10.
- The Subnet Mask to be applied on the final destination address as 255.255.255.0.
- The Gateway Address where the IP packets are to be sent as 192.168.5.1.

When SARVAM UCS sends packets, it checks the final destination IP Address in the Static Routing Table.

If a perfect match is found, SARVAM UCS will start sending the IP packets to the corresponding Gateway Address configured in the table.In this case the packets will be sent through the LAN Port the Destination IP Address 192.168.5.10

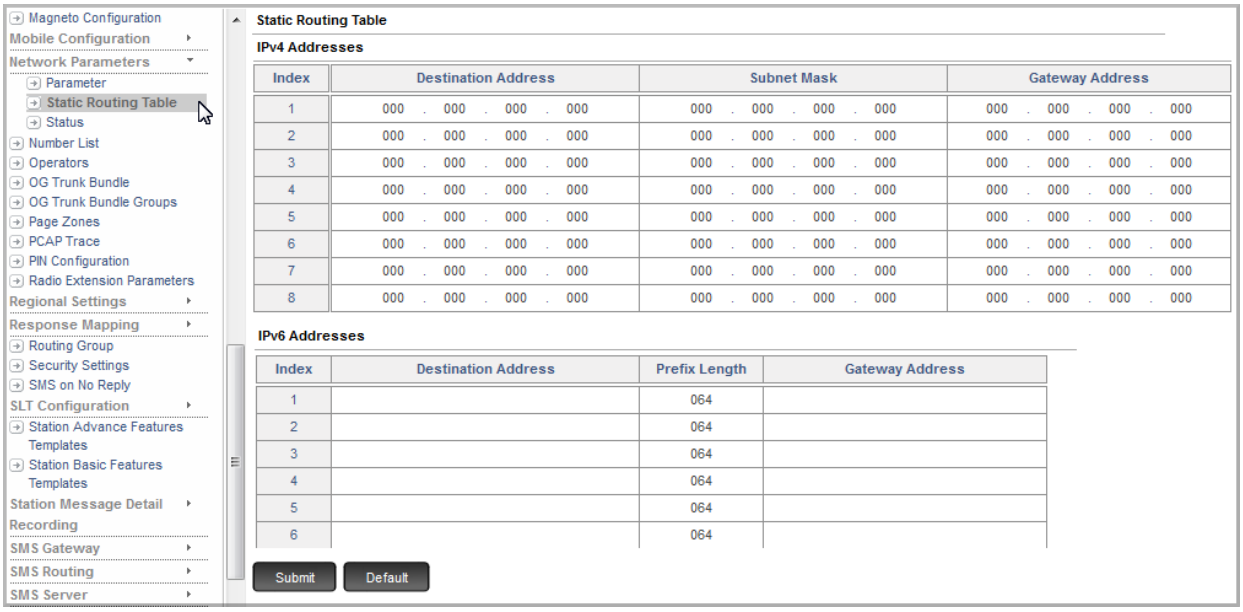
If no match is found, SARVAM UCS will send the IP Packets to the **Default Gateway Address** (Network Connection Type) you configured in the Network Parameters. For detailed instructions, see "Configuring Network Parameters".

How to configure

The Static Routing Table must be configured at each location where SARVAM UCS is installed. You may configure the Static Routing Table using Jeeves.

Configuring Static Routing Table using Jeeves

- Log in as System Engineer.
- Under **Configuration**, click **Network Parameters**.
- Click **Static Routing Table**.
- The Static Routing Table page opens.



The Static routing table allows you to configure upto 8 entries in each table, IPv4 Addresses table and IPv6 Addresses. Each entry is stored against an Index number.

IPv4 Addresses table

For each entry, you must configure the following fields:

- **Destination Address:** This is the address of the final destination where the call is to be made. This can be a device IP Address or Network Address.
- **Subnet Mask**: This is the mask to be applied on destination address.
- **Gateway Address**: This is the IP address of the node where the IP packets are to be sent. Generally, it is the IP address of the LAN interface of the Router.

  The Gateway Address must be in the same subnet as SARVAM UCS.

Click **Submit** to save your entries.

IPv6 Addresses table

For each entry, you must configure the following fields:

- **Destination Address:** This is the address of the final destination where the call is to be made. This can be a device IP Address or Network Address.

  Valid Range of the IPv6 Address is A to F, a to f, 0 to 9,:(colon). It can be a maximum of 39 characters. Default: Blank.

- **Prefix Length**: The Prefix Length is a decimal value that indicates how many of the high-order contiguous bits of the destination address comprise the prefix (the network portion of the address).

  The Prefix Length range is from 1 to 128 bits. Default: Blank.

- **Gateway Address**: This is the IP address of the node where the IP packets are to be sent. Generally, it is the IP address of the LAN interface of the Router.

  The Gateway Address must be in the same subnet as SARVAM UCS.

Click **Submit** to save your entries.

The CPU Card will restart after the Static Routing parameters have been configured.

To take the above example further, the Static Routing Table of SARVAM UCS at Location A should be configured as:

Index	Destination Address	Subnet Mask	Gateway Address
1	192.168.2.0	255.255.255.0	192.168.1.1
2			
:			
8			

If you configure the Subnet Mask as 255.255.255.255, then only the Destination Address will be accessible.

If you configure the Destination Address as 192.168.2.1, then only this specific address will be accessible.

- The Destination Address 192.168.2.0 specifies the network address of Location B.

- The Subnet Mask is the mask to be applied on the Destination address.

- The Gateway Address 192.168.1.1 specifies the LAN address of the Router A which connects location A and location B.

  The IP address of the LAN interface of the router which connects Location A to the public internet should be configured as Default Gateway in the Network Parameters of SARVAM UCS in location A.

  With the Static Routing Table configured thus, all calls made by SARVAM UCS to 192.168.2.0/ 24 will be routed through the router which connects Location A to Location B. Whereas, all calls made by SARVAM UCS to addresses other than 192.168.2.0/ 24 will be routed through the Default Gateway.

  Similarly, configure the Static Routing Table in SARVAM UCS at location B to enable calling from Location B to Location A.

- Click **Submit** to save entries.

Station Message Detail Recording (SMDR)

What's this?

The SARVAM UCS can record the details of Internal, Incoming and Outgoing calls made from/to all the extensions. This feature is called Station Message Detail Recording (SMDR).

You can store SMDR; obtain SMDR as a Report whenever you want or obtain it Online, immediately after the call has been made or received. You can also use SMDR to calculate the cost of the calls.

To be able to use SMDR, you must configure:

- **SMDR Storage:** These parameters are programmed to enable the storing of the IC, OG and Internal calls. To know more, see "Station Message Detail Recording-Storage".
- **SMDR Report:** These parameters are programmed to assign destination port for getting report of IC, OG and Internal calls and to get offline report. To know more, see "Station Message Detail Recording-Report".
- **SMDR Online:** These parameters enable you to obtain Online report of Incoming, Outgoing and Internal calls. With Online SMDR you can obtain details of each call immediately after the call has been made or received. You can also set the call record format you want for Incoming calls when SARVAM UCS is interfaced with a third party call accounting software (CAS). To know more see, "Station Message Detail Recording-Online".
- **SMDR Posting:** These parameters enable you to interface third-party call accounting software (CAS) with SARVAM UCS for call cost calculation. You can select the protocol supported by the call accounting software and further customize the handshaking parameters and call record formats. To know more see, "Station Message Detail Recording-Posting".

Refer to the Hospitality Manual for more details about PMS.

Station Message Detail Recording-Online

What's this?

The SARVAM UCS can generate report for the calls as and when the call is made and send the report to the computer.

SARVAM UCS also supports a Syslog Client for SMDR. The Syslog Client enables the system to send call records in syslog format to the remote 'Syslog Server'. You can view the call records on the remote server. SARVAM UCS supports SMDR only on TCP.

SMDR generated as and when calls are made or received is called SMDR **Online** report. In the SMDR **Online** report,

- Each **Internal call** is stored with following fields:
    - Extension who made the call.
    - Extension to which the call was made.
    - Date and time when the call was made.
    - Duration of the call in seconds.
- Each **Outgoing call** is stored with following fields:
    - Extension who made the call.
    - Trunk line port used for the call.
    - Number dialed.
    - Date and time when the call matured.
    - Duration of the call in seconds.
    - Call Units.
    - Call Maturity Type.
    - Call Type (Normal, DISA, ECF etc.).
- Each **Incoming call** is stored with the following fields:
    - The Trunk on which the call is received.
    - Extension number which answered the call.
    - Date and time when the call was received.
    - Calling Number.
    - Hold, speech, ring duration of the call in seconds.

How to configure

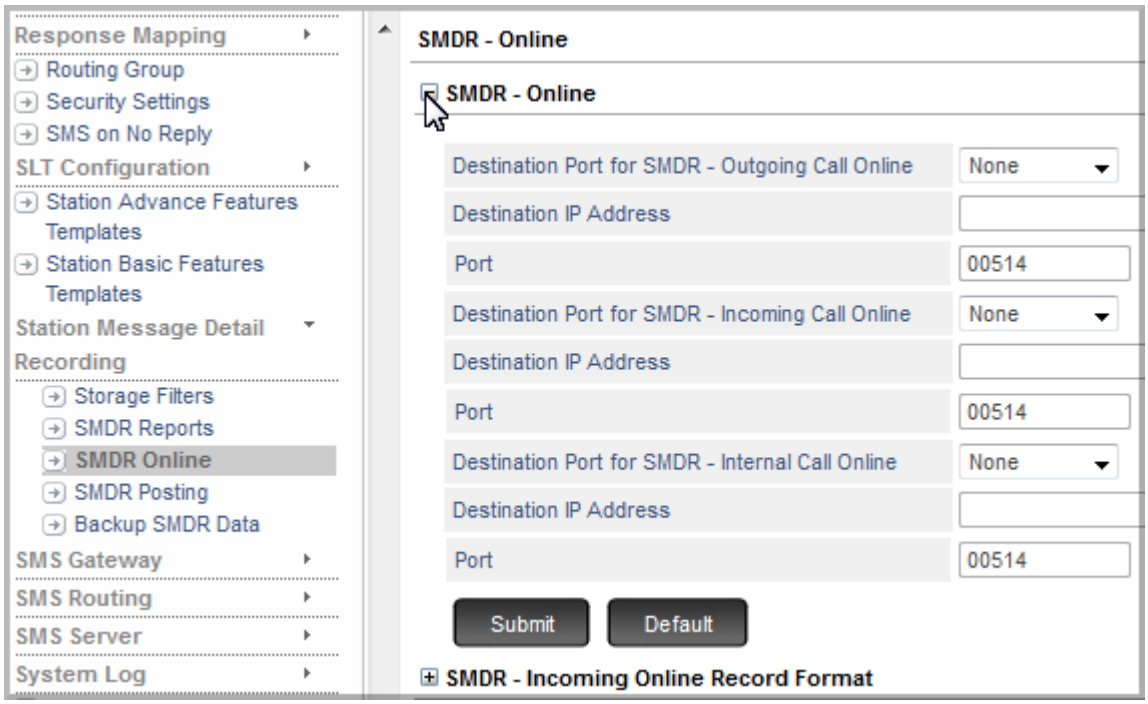
To get the **Online** report you must do the following:

- Enable SMDR Storage in the SMDR buffer. See "Station Message Detail Recording-Storage".
- Select the Destination port for Incoming, Outgoing and Internal calls. If you select Ethernet as the destination port, you must configure the Destination IP Address. The Online report is sent to this address as soon as the incoming call is completed.

- You may also change the default format for the SMDR Online report for incoming calls, like column position and field length for calling number, speech duration, type of call etc., as required. For this you need to configure the settings of **SMDR Incoming Online Record Format**.

Configuring SMDR-Online using Jeeves

- Login as System Engineer.
- Under **Configuration**, click **Station Message Detail Recording**.
- Click **SMDR Online** to open the page.
- Click **SMDR Online** to expand.



For **SMDR - Outgoing Call Online**,

- Select the **Destination Port**.

  If you select Ethernet, in **Destination IP Address**, enter the IP Address of the remote Syslog Server. Both IPv4 and IPv6 addresses are supported.

  In **Port**, enter the port of the remote Syslog Server. Valid port range is: 514; 1025 to 65535.

For **SMDR - Incoming Call Online**,

- Select the **Destination Port**.

  If you select Ethernet, in **Destination IP Address**, enter the IP Address of the remote Syslog Server. Both IPv4 and IPv6 addresses are supported.

  In **Port**, enter the port of the remote Syslog Server. Valid port range is: 514; 1025 to 65535.

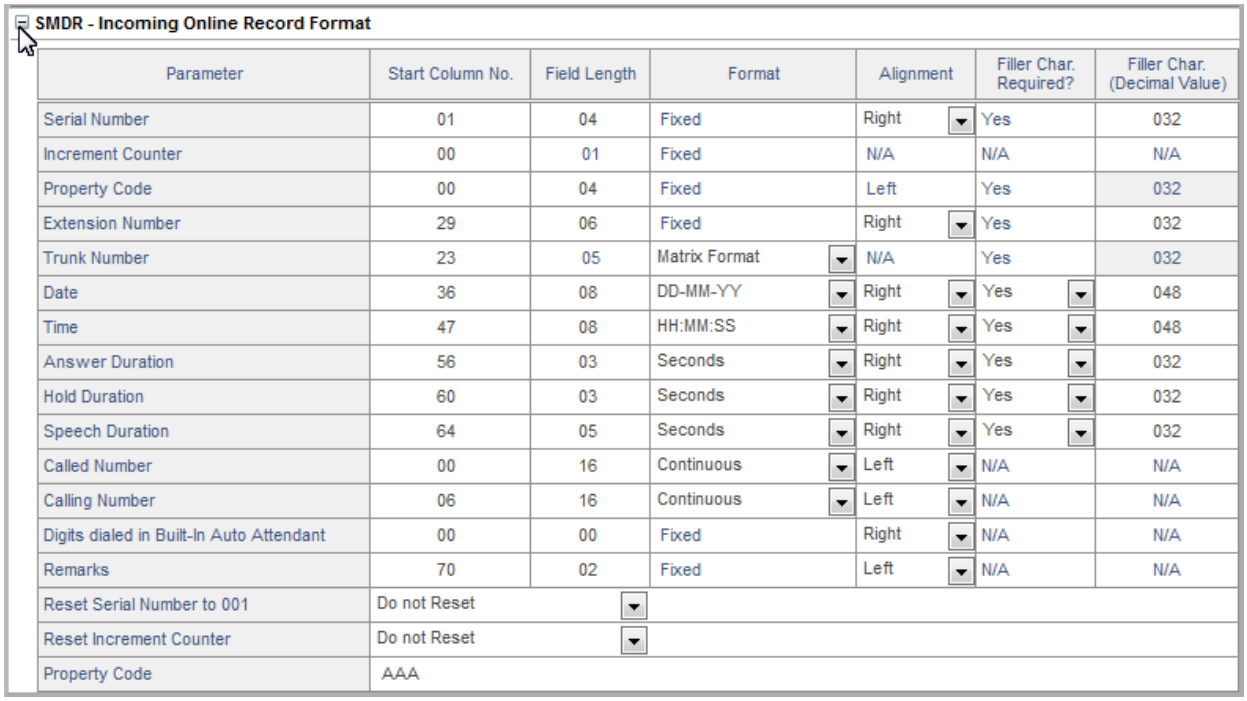
For **SMDR - Internal Call Online**,

- Select the **Destination Port**.

  If you select Ethernet, in **Destination IP Address**, enter the IP Address of the remote Syslog Server. Both IPv4 and IPv6 addresses are supported.

  In **Port**, enter the port of the remote Syslog Server. Valid port range is: 514; 1025 to 65535.

- To configure Call Record Format for Incoming Calls, click **SMDR Incoming Online Call Record Format** to expand.



SMDR - Incoming Online Record Format

Parameter	Start Column No.	Field Length	Format	Alignment	Filler Char. Required?	Filler Char. (Decimal Value)
Serial Number	01	04	Fixed	Right	Yes	032
Increment Counter	00	01	Fixed	N/A	N/A	N/A
Property Code	00	04	Fixed	Left	Yes	032
Extension Number	29	06	Fixed	Right	Yes	032
Trunk Number	23	05	Matrix Format	N/A	Yes	032
Date	36	08	DD-MM-YY	Right	Yes	048
Time	47	08	HH:MM:SS	Right	Yes	048
Answer Duration	56	03	Seconds	Right	Yes	032
Hold Duration	60	03	Seconds	Right	Yes	032
Speech Duration	64	05	Seconds	Right	Yes	032
Called Number	00	16	Continuous	Left	N/A	N/A
Calling Number	06	16	Continuous	Left	N/A	N/A
Digits dialed in Built-In Auto Attendant	00	00	Fixed	Right	N/A	N/A
Remarks	70	02	Fixed	Left	N/A	N/A
Reset Serial Number to 001	Do not Reset					
Reset Increment Counter	Do not Reset					
Property Code	AAA					

- Configure the following Call Record Format parameters as required.

  For each parameter explained briefly below, you can define the column position, field length (i.e. the number of digits), the alignment (whether left aligned or right), and the filler characters, wherever required.

  - **Serial Number:** This is the serial number generated for each call record. Serial numbers are generated from 001 to 999. When serial number '999' is reached, the numbers roll over to 001.

    When this field rolls over, it increments the increment counter.

  - **Increment Counter:** It increments when the serial number counter rolls over. The Increment counter starts from A, ending at Z, and then roll over back to A.

  - **Property Code:** This is the property code, if required. This may be an abbreviation of the property name.

  - **Extension Number:** This is the extension number that answered the call. You can define the column position and the field length for the extension number.

  - **Trunk Number:** This is the number of the trunk on which the call was received.

- *The Matrix Format occupies 5 character spaces.*
- *Check-Inn Format occupies 4 character spaces.*
- *The First Character in the Check-Inn Format is X (Fixed). The remaining three characters show the software port number.*
- **Date:** The date on which the call was received. The date fill flag is to be enabled.

- *Filler Character field is applicable for Date, Month and Year, i.e. whether the single digit date is to be printed as space-X or 0-X. For example, date = 1 is to be displayed as '1' or '01'.*
- *Where leading zeroes are not required, the date, month and year sub-fields are right aligned and the spaces are filled with character 'space'.*
- *The Date field is not linked to the global flag of Date Format. The global Flag of Date format is used, while using features or in configuration reports but not for SMDR Online. This is because the date format used by the CAS is not the same as used by the users of the system.*
- **Time:** The time when the call was received. The format of the time field and the time fill flag are to be programmed.

- *Filler Character field is applicable for Hours, Minutes and Seconds i.e. whether the single digit hour is to be printed as space-X or 0-X. For example, hour = 1 is to be displayed as '1' or '01'.*
- *In case leading zeroes are not required, Date, Month and Year sub-fields are right aligned and the spaces are filled with character 'space'.*
- **Answer Duration:** The time after which the call was answered. Program the duration unit and the duration fill flag.
- **Hold Duration:** The time for which the call was put on hold.
- **Speech Duration:** The time for which the call was in speech with the extension.

When Duration Unit = Minutes, the rounding off to the nearest whole number is done. For seconds <= 30, Minute is not incremented. For seconds > 30, minute is incremented.

- **Called Number:** This is applicable only for calls received on SIP trunks. The number dialed by the caller is referred to as Called Number.
- **Calling Number:** This is the number of the Caller.
- **Digits dialed in Built-In Auto Attendant:** This is the number dialed by the caller using Built-In Auto Attendant.
- **Remarks:** You may use this for indicating the Type of Call, for example, Built-In Auto Attendant.
- **Reset Serial Number to 001:** The Serial number counter can be reset to 001 after 24 hours (from 00:00 HH:MM) or every 6 hours. By default, 'No Compulsory Reset' is selected, which means the serial number counter will not be automatically reset.

- **Reset Increment Counter:** The Increment Counter can be reset to 001 after 24 hours (from 00:00 HH:MM) or every 6 hours. By default, 'No Compulsory Reset' is selected, which means the serial number counter will not be automatically reset.

- **Property Code:** This may be an abbreviation of the property name.

- Click **Submit** to save settings.

- You may log out of Jeeves.

Configuring SMDR-Online using a Telephone

- Enter SE mode from an SLT/DKP.

Internal Calls

To assign a destination port for Online SMDR-Internal call record, dial:

- ***2830-Code***

  Where,

Code	Meaning
0	None
1	COM Port
2	Ethernet Port
3	USB to COM Port

By default, the no port is assigned, that is Online printing is disabled.

To assign an IP Address, if you select Ethernet Port as the destination port, dial:

- ***2832-IP Address***

  By default, IP Address is 192.168.1.104

IPv6 address can be configured using Jeeves only.

To assign a Port, if you select Ethernet Port as the destination port, dial:

- ***2833- Port***

  Where,
  IP Port is from 514 and 1025-65535
  By default, IP Port is 514.

Outgoing Calls

To assign destination port for Online SMDR-OG Call Record, dial:

- ***2730-Code***

  Where

Code	Meaning
0	None
1	COM Port

Code	Meaning
2	Ethernet Port
3	USB to COM Port

By default, the port assigned is None. This means the On-line printing is disabled.

To assign an IP Address, if you select Ethernet Port as the destination port, dial:

- ***2732-IP Address***
  By default, IP Address is 192.168.1.104

IPv6 address can be configured using Jeeves only.

To assign the Port, dial:

- ***2733-IP Port***
  Where,
  Port is from 514 and 1025-65535
  By default, IP Port is 514.

To Start/Abort report generation, dial:

- ***1072-101-Flag***
  Where,

Flag	Meaning
0	Abort
1	Start

By default, Flag is 0.

The SARVAM UCS provides a facility to abort the report generation in midway (***1072-101-0***). Once the report generation is aborted, then it has to be explicitly started with command (***1072-101-1***). This command is issued from the SA mode.

Incoming Calls

To assign destination port for Online SMDR-IC Call Record, dial:

- ***2930-Code***
  Where,

Code	Meaning
0	None
1	COM Port
2	Ethernet Port
3	USB to COM Port

By default, the port assigned is None. This means the Online printing is disabled.

To assign an IP Address, if you select Ethernet Port as the destination port, dial

- ***2932-IP Address***
  By default, IP Address is 192.168.1.104.

IPv6 address can be configured using Jeeves only.

To assign the Port:

- ***2933-Port***
  Where,
  Port is from 514 and 1025-65535
  By default, IP Port is 514.

To start/abort report generation:

- ***1072-151-Flag***
  Where,

Flag	Meaning
0	Abort
1	Start

By default, Flag is 0.

The SARVAM UCS provides a facility to abort the report generation midway (***1072-151-0***). Once the report generation is aborted, then it has to be explicitly started with (***1072-151-1***). This command is issued from the SA mode.

SMDR record format for IC Call Printing-Online

You can customize the format of the incoming call records as per your requirement. This can be done with the commands given below:

Serial Number

To program column position for serial number, dial:

- ***8200-Column Position***
  Where,
  Column Position is from 00 to 78.
  By default, Column Position is 01.

To program field length for serial number, dial:

- ***8201-Field Length***
  Where,
  Field Length is from 00 to 78.
  By default, Field Length is 04.

To program alignment for serial number, dial:

- ***8202-Alignment***
  Where,

Alignment	Meaning
1	Left Alignment

Alignment	Meaning
2	Right Alignment

By default, Alignment is 2.

To program fill character for serial number, dial:

- ***8203-Fill Character***

  Where,

  Fill Character is 3 digit ASCII value.

  By default, Fill Character is ‘Space’.

To program the reset for serial number, dial:

- ***8204-Reset***

  Where,

Reset	Meaning
1	No Compulsory Reset
2	Reset to 001 every 24 hours (at 00:00 Hrs.)
3	Reset to 001 every 6 hours (at 00:00 Hrs.)

By default, Reset is ‘1’.

Serial Number starts from 1 and not 0.
When this field rolls over, it increments the increment counter.

Increment Counter

To program column position for increment counter, dial:

- ***8205-Column Position***

  Where,

  Column Position is from 00 to 78.

  By default, Column Position is 00 (This field is not available by default).

  By default, Field Length is 1, which is fixed.

To program the reset for increment counter, dial:

- ***8206-Reset***

  Where,

Reset	Meaning
1	No Compulsory Reset
2	Reset to 001 every 24 hours (at 00:00 Hrs.)
3	Reset to 001 every 6 hours (at 00:00 Hrs.)

By default, Reset is ‘1’.

Increment Counter starts from A to Z and then rolls over back to A.
Increment Counter increments when Serial Number Counter rolls over.

Property Code

To program column position for property code, dial:

- ***8207-Column Position***
  Where,
  Column Position is from 00 to 78.
  By default, Column Position is 00 (This field is not available by default).

To program field length for property code, dial:

- ***8208-Field Length***
  Where,
  Field Length is from 00 to 78.

To program property code string for property code, dial:

- ***8209-Property Code String***

- *This is a fixed code that appears in each call.*
- *The Enterprise can have its own code.*
- *This code is required by the Property Management System (PMS) when it is catering to more than one PMS interfaces.*
- *Refer separate Manual for Hotel/Motel Applications for more details about PMS and Hotel applications for this feature.*

Extension Number

To program column position for extension number, dial:

- ***8210-Column Position***
  Where,
  Column Position is from 00 to 78.
  By default, Column Position is 29.

To program field length for extension number, dial:

- ***8211-Field Length***
  Where,
  Field Length is from 00 to 78.
  By default, Field Length is 04.

To program alignment for extension number, dial:

- ***8212-Alignment***
  Where,

Alignment	Meaning
1	Left Alignment
2	Right Alignment

  By default, Alignment is 2.

To program fill character for extension number, dial:

- ***8213-Fill Character***
  Where,
  Fill Character is 3 digit ASCII value.
  By default, Fill Character is ‘Space’.

Trunk Number

To program column position for trunk number, dial:

- ***8214-Column Position***
  Where,
  Column Position is from 00 to 78.
  By default, Column Position is 23.

To program format type for trunk number, dial:

- ***8215-Format Type***
  Where,

Format Type	Meaning
1	Matrix Format
2	Check-In Format

By default, Format Type is '1'.

- *Matrix Format occupies 5 character space.*
- *Check In Format occupies 4 character space.*
- *First Character in Check In Format is **X** (Fixed). Remaining three characters show the software port number. However, this will not specify whether the call is made through CO 125 or E&M 125. Also the channel number will not be specified in case of call made through T1E1PRI port or BRI port.*

Date

To program column position for date field, dial:

- ***8216-Column Position***
  Where,
  Column Position is from 00 to 78.
  By default, Column Position is 34.

To program field length for date field, dial:

- ***8217-Field Length***
  Where,
  Field Length is from 00 to 78.
  By default, Field Length is 10.

To program alignment for date field, dial:

- ***8218-Alignment***
  Where,

Alignment	Meaning
1	Left Alignment
2	Right Alignment

By default, Alignment is 2.

To program fill character for date field, dial:

- ***8219-Fill Character***
  Where,

Fill Character is 3 digit ASCII value.
By default, Fill Character is ‘Zero’.

To program date format for date field, dial:

- ***8220-Date Format***

  Where,

Date Format	Meaning
01	DD-MM-YY
02	DD/MM/YY
03	DD.MM.YY
04	DD MM YY
05	DDMMYY
06	DD-MM-YYYY
07	DD/MM/YYYY
08	DD.MM.YYYY
09	DD MM YYYY
10	DDMMYYYY
11	MM-DD-YY
12	MM/DD/YY
13	MM.DD.YY
14	MM DD YY
15	MMDDYY
16	YY-MM-DD
17	YY/MM/DD
18	YY.MM.DD
19	YY MM DD
20	YYMMDD
21	YYYY-MM-DD
22	YYYY/MM/DD
23	YYYY.MM.DD
24	YYYY MM DD
25	YYYYMMDD
26	MM-DD
27	MM/DD
28	MM.DD
29	MM DD
30	MMDD

Date Format	Meaning
31	DD-MM
32	DD/MM
33	DD.MM
34	DD MM
35	DDMM

By default, Date Format is 1.

To program date fill flag for date field, dial:

- ***8257-Date Fill Flag***

  Where,

Date Fill Flag	Meaning
0	Disable
1	Enable

By default, Date Fill Flag is '1' (that is, single digit in Date, Month and year is printed with prefix '0').

- *Leading Zeros field is applicable for Date, Month and Year, that is, whether the single digit date is to be printed as space-X or 0-X. For example: date = 1 is to be displayed as '1' or '01'. In case when leading zeroes are not required, the date, month and year sub-fields are right aligned and the spaces are filled with character 'space'.*

- *This Date field is not linked to the global flag of Date Format. The global Flag of Date format is used while using features or in configuration reports but not in PMS. This is because the date format used by the PMS is not the same as used by the users of the system.*

Time

To program column position for time field, dial:

- ***8222-Column Position***

  Where,

  Column Position is from 00 to 78.

  By default, Column Position is 45.

To program field length for time field, dial:

- ***8223-Field Length***

  Where,

  Field Length is from 00 to 78.

  By default, Field Length is 08.

To program alignment for time field, dial:

- ***8224-Alignment***

  Where,

Alignment	Meaning
1	Left Alignment
2	Right Alignment

By default, Alignment is 2.

To program fill character for time field, dial:

- ***8225-Fill Character***

  Where,

  Fill Character is 3 digit ASCII value.

  By default, Fill Character is ‘Zero’.

To program time format for time field, dial:

- ***8226-Time Format***

  Where,

Time Format	Meaning
0	Disable
1	Enable

By default, Time Format is 1.

To program time fill flag for time field, dial:

- ***8258-Time Fill Flag***

  Where,

Time Fill Flag	Meaning
0	Disable
1	Enable

By default, Time Fill Flag is ‘1’.

Leading Zeros field is applicable for Hours, Minutes and Seconds, that is, whether the single digit hour is to be printed as space-X or 0-X. For example: hour = 1 is to be displayed as ‘1’ or ‘01’. In case when leading zeroes are not required, Date, Month and Year sub-fields are right aligned and the spaces are filled with character ‘space’.

Answer Duration

To program column position for answer duration field, dial:

- ***8227-Column Position***

  Where,

  Column Position is from 00 to 78.

  By default, Column Position is 54.

To program field length for answer duration field, dial:

- ***8228-Field Length***

  Where,

  Field Length is from 00 to 78.

  By default, Field Length is 03.

To program alignment for answer duration field, dial:

- ***8229-Alignment***

Where,

Alignment	Meaning
1	Left Alignment
2	Right Alignment

By default, Alignment is 2.

To program fill character for answer duration field, dial:

- ***8230-Fill Character***

  Where,

  Fill Character is 3 digit ASCII value.

  By default, Fill Character is ‘Space’.

To enable/disable the Filler character flag for Answer Duration, dial:

- ***8259-Filler Character Flag for Answer Duration***

  Where,

Filler Character Flag for Answer Duration	Meaning
0	Disable
1	Enable

Default = Enable, (that is, the Filler Character will used as programmed)

To program duration unit for answer duration field:

- ***8231-Duration Unit***

  Where,

Duration Unit	Meaning
1	HH:MM:SS
2	HHMMSS
3	Minutes
4	Seconds

By default, Duration Unit is ‘4’.

When Duration Unit = Minutes, rounding to nearest whole number is done. For seconds <= 30, Minute is not incremented and for seconds > 30, minute is incremented.

Hold Duration

To program column position for hold duration field, dial:

- ***8232-Column Position***

  Where,

  Column Position is from 00 to 78.

  By default, Column Position is 58.

To program field length for hold duration field, dial:

- ***8233-Field Length***

  Where,
  Field Length is from 00 to 78.
  By default, Field Length is 03.

To program alignment for hold duration field, dial:

- ***8234-Alignment***

  Where,

Alignment	Meaning
1	Left Alignment
2	Right Alignment

  By default, Alignment for hold duration field is 2.

To program fill character for hold duration field, dial:

- ***8235-Fill Character***

  Where,
  Fill Character is 3 digit ASCII value.
  By default, Fill Character is 'Space'.

To enable/disable the Filler character flag for Hold Duration, dial:

- ***8260-Filler Character Flag for Hold Duration***

  Where,

Filler Character Flag for Hold Duration	Meaning
0	Disable
1	Enable

  Default = Enable, (that is, Filler Character will used as programmed)

Speech Duration

To program column position for speech duration field, dial:

- ***8237-Column Position***

  Where,
  Column Position is from 00 to 78.
  By default, Column Position is 62.

To program field length for speech duration field, dial:

- ***8238-Field Length***

  Where,
  Field Length is from 00 to 78.
  By default, Field Length is 03.

To program alignment for speech duration field, dial:

- ***8239-Alignment***

Where,

Alignment	Meaning
1	Left Alignment
2	Right Alignment

By default, Alignment is 2.

To program fill character for speech duration field, dial:

- ***8240-Fill Character***
  Where,
  Fill Character is 3 digit ASCII value.
  By default, Fill Character is 'Space'.

To enable/disable the Filler character flag for Speech Duration, dial:

- ***8261-Filler Character Flag for Speech Duration***
  Where,

Filler Character for Speech Duration	Meaning
0	Disable
1	Enable

Default = Enable, (that is, Filler Character will used as programmed).

Called Number

To program column position for called number field, dial:

- ***8242-Column Position***
  Where,
  Column Position is from 00 to 78.
  By default, Column Position is 00. (This field is not available by default)

To program field length for called number field, dial:

- ***8243-Field Length***
  Where,
  Field Length is from 00 to 78.
  By default, Field Length is 16.

To program alignment for called number field, dial:

- ***8244-Alignment***
  Where,

Alignment	Meaning
1	Left Alignment
2	Right Alignment

By default, Alignment is 1.

To program number format for called number field, dial:

- ***8245-Number Format***

  Where,

Number Format	Meaning
1	Continuous
2	Separated

By default, Number Format is ‘1’.

When separated is selected, put ‘-‘ in the called party number of 10 digits. First after three digits and another after six digits.

Calling Number

To program column position for calling number field, dial:

- ***8246-Column Position***

  Where,

  Column Position is from 00 to 78.

  By default, Column Position is 06.

To program field length for calling number field, dial:

- ***8247-Field Length***

  Where,

  Field Length is from 00 to 78.

  By default, Field Length is 16.

To program alignment for calling number field, dial:

- ***8248-Alignment***

  Where,

Alignment	Meaning
1	Left Alignment
2	Right Alignment

By default, Alignment is 1.

To program number format for calling number field, dial:

- ***8249-Number Format***

  Where,

Number Format	Meaning
1	Continuous
2	Separated

By default, Number Format is 1.

When separated is selected, put '-' in the called party number of 10 digits. First after three digits and another after six digits.

Digits dialed in Built-in Auto Attendant

To program column position for the digits field, dial:

- ***8250-Column Position***
  Where,
  Column Position is from 00 to 78.
  By default, Column Position is 00, that is, this field is not available by default.

To program field length for the digits field, dial:

- ***8251-Field Length***
  Where,
  Field Length is from 00 to 78.
  By default, Field Length is 00.

To program alignment for the digits field, dial:

- ***8252-Alignment***
  Where,

Alignment	Meaning
1	Left Alignment
2	Right Alignment

Remarks

To program column position for remarks field, dial:

- ***8253-Column Position***
  Where,
  Column Position is from 00 to 78.
  By default, Column Position is 68.

To program field length for remarks field, dial:

- ***8254-Field Length***
  Where,
  Field Length is from 00 to 78.
  By default, Field Length is 02.

To program alignment for remarks field, dial:

- ***8255-Alignment***
  Where,

Alignment	Meaning
1	Left Alignment
2	Right Alignment

By default, Alignment is 1.

To assign default IC SMDR format, dial:

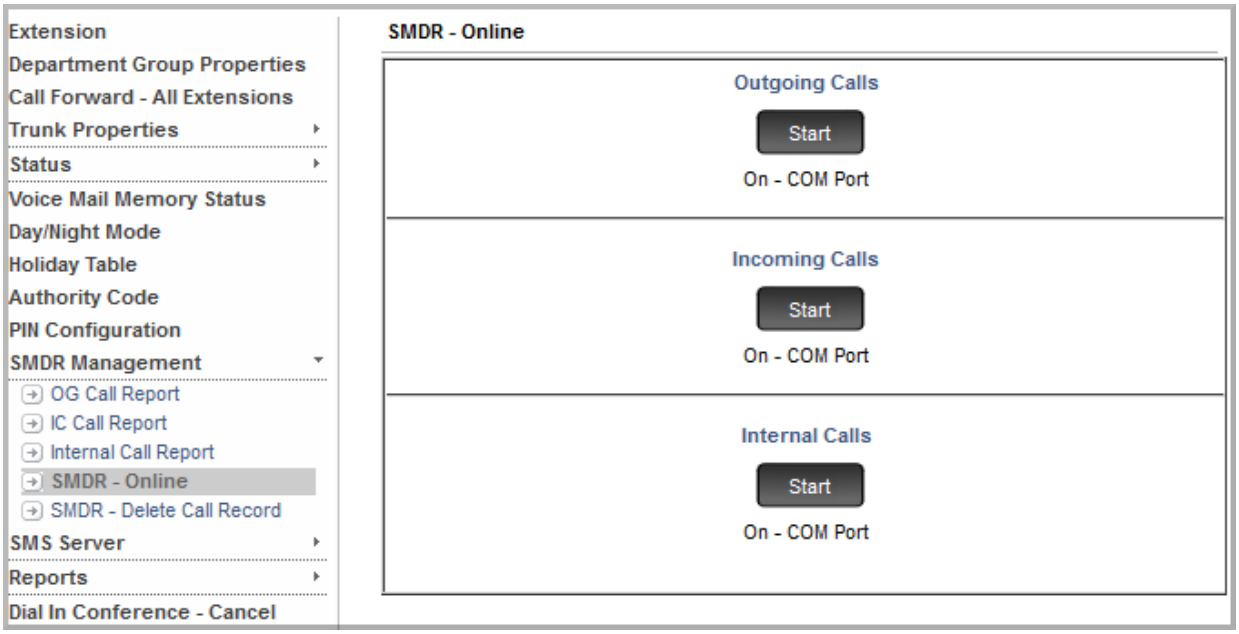
- ***8256***

How to use

You can start and stop SMDR Online report from the System Administrator mode using Jeeves or dialing SA Commands from an extension phone.

To start/stop Online report using Jeeves,

- Open Jeeves.
- Log in as System Administrator.
- Under **SMDR Management**, click **SMDR-Online** to open the page.



- To start SMDR Online for Outgoing Calls, Incoming Calls and Internal Calls, click the **Start** button.
- To stop SMDR Online for any of these call types, click **Abort** button.
- You may log out of Jeeves.

To generate SMDR Online report from an extension phone,

- Enter SA mode from a DKP/SLT/Extended IP Phone.

  To start/stop Online report for **Internal Calls**:

  - Dial **1072-136-1** to start
  - Dial **1072-136-0** to stop

  To start/stop Online report for **Outgoing Calls**:

  - Dial **1072-101-1** to start

- Dial **1072-101-0** to stop

To start/stop Online report for **Incoming Calls**:

- Dial ***1072-151-1*** to start
- Dial ***1072-151-0*** to stop

- Exit SA mode.

The Online report for calls looks like shown below:

IC Call Online Report

```
2   4002 000 V001 #1234          08-02-21 12:52:28    6    1    1.10  I
3   4002 000 V001 #1234          08-02-21 12:53:52    7    1    1.10  I
4   4002 000 V001 #1234          08-02-21 12:54:07   10    1    1.10  I
5   4002 000 V001 #1234          08-02-21 12:54:27    5    1    1.10  I
```

OG Call report

```
2 #1234@191.168.10 V001    4002 08-02-21   12:56:50   0   0     6 N
3 #1234@191.168.10 V001    4002 08-02-21   12:57:08   0   0     9 N
4 #1234@191.168.10 V001    4002 08-02-21   12:57:28   0   0    10 N
```

Internal Call report

```
IN      4002        4003      08-02-2021  12:59:58    10
IN      4002        4003      08-02-2021  13:00:10    11
IN      4002        4003      08-02-2021  13:00:26     4
IN      4002        4003      08-02-2021  13:00:34    10
```

Station Message Detail Recording-Posting

The Station Message Detail Record (SMDR)-Posting feature of SARVAM UCS is used for interfacing the system with a third party Call Accounting Software (CAS).

When SARVAM UCS is interfaced with a third party Call Accounting Software (CAS) to determine the cost of the calls made by the extension users, the system uses SMDR-Posting to send to CAS call record details, like number to which the call was made by the extension user, number of the extension from which the call was made, the date and time when the call was made, the duration of the call, metering pulses incurred for the call, etc. On receipt of this information, the CAS calculates the cost of the call for billing.

As different CAS interfaces support different protocols, the SARVAM UCS offers the flexibility to send call detail records using the protocol supported by CAS. SARVAM UCS supports as many as 16 different widely-used CAS protocols such as, Holidex, Hobic, Micros A, Micros B, Comm One, Call-Inn, Bell-HOBIC, XIOX, RSI and others.

Each posting protocol has its own handshaking protocol and call record format. You may configure any one of these depending upon the protocol supported by CAS you have interfaced with SARVAM UCS. It is also possible to customize the posting protocol to match the settings required by the CAS you have interfaced.

SARVAM UCS supports SMDR-Posting on Serial RS232 Communication Port as well as on TCP/IP Ethernet(LAN/WAN) Port. Thus, the CAS can be interfaced either on the COM port or the Ethernet(LAN/WAN) of the SARVAM UCS.

SMDR-Posting sends outgoing call records only.

SMDR-Posting Protocols

The SARVAM UCS supports different SMDR posting protocols from the system to CAS. The flow of messages between the SARVAM UCS and the protocols of CAS Interface (Matrix and Blind Send) are described below:

Matrix

- **Positive Response from the CAS**

SARVAM UCS to CAS	CAS to SARVAM UCS
<STX> -Call Record-<ETX> <BCC>	
	ACK

- **Negative Response from the CAS**

SARVAM UCS to CAS	CAS to SARVAM UCS
<STX> -Call Record-<ETX> <BCC> and wait for Response to Data Timeout (sec), default 3 sec.	
	NAK
Retransmit <STX> -Call Record-<ETX> <BCC> and wait for Data Transfer Retry Timer (sec) - on Negative Response, default 3 seconds.	
	NAK

SARVAM UCS to CAS	CAS to SARVAM UCS
Retransmit <STX> -Call Record-<ETX> <BCC> and wait for Data Transfer Retry Timer (sec) - on Negative Response, default 3 seconds.	
	NAK
Retransmit <STX> -Call Record-<ETX> <BCC> and wait for Data Transfer Retry Timer (sec) - on Negative Response, default 3 seconds.	
	NAK
Retransmit <STX> -Call Record-<ETX> <BCC> and wait for Data Transfer Retry Timer (sec) - on Negative Response, default 3 seconds.	
	NAK

The SARVAM UCS will make 5 attempts (default value of *Data Transfer Retry Count - on Negative Response*) to send the message after a regular interval of 3 seconds (default value of *Data Transfer Retry Timer - on Negative Response*). If the ACK is still not received from the CAS, the SARVAM UCS will proceed to the next message.

- **Busy Response from the CAS**

SARVAM UCS to CAS	CAS to SARVAM UCS
<STX> -Call Record-<ETX> <BCC> and wait for Response to Data Timeout (sec), default 3 sec.	
	NAK (CAS responds but cannot accept at this time)
Retransmit <STX> -Call Record-<ETX> <BCC> and wait for Data Transfer Retry Timer (sec) - on Negative Response, default 3 seconds.	
	NAK
Retransmit <STX> -Call Record-<ETX> <BCC> and wait for Data Transfer Retry Timer (sec) - on Negative Response, default 3 seconds.	
	NAK
Retransmit <STX> -Call Record-<ETX> <BCC> and wait for Data Transfer Retry Timer (sec) - on Negative Response, default 3 seconds.	
	NAK
Retransmit <STX> -Call Record-<ETX> <BCC> and wait for Data Transfer Retry Timer (sec) - on Negative Response, default 3 seconds.	
	NAK

The SARVAM UCS will make 5 attempts (default value of *Data Transfer Retry Count - on Negative Response*) to send the message after a regular interval of 3 seconds (default value of *Data Transfer Retry Timer - on Negative Response*). If the ACK is still not received from the CAS, the SARVAM UCS will proceed to the next message.

- **No Response from the CAS**

SARVAM UCS to CAS	CAS to SARVAM UCS
<STX> -Call Record-<ETX> <BCC> and wait for Response to Data Timeout (sec), default 3 sec.	
	(no response)
Retransmit <STX> -Call Record-<ETX> <BCC> and wait for Data Transfer Retry Timer (sec) - on No Response, default 3 seconds.	
Retransmit <STX> -Call Record-<ETX> <BCC> and wait for Data Transfer Retry Timer (sec) - on No Response, default 3 seconds.	
Retransmit <STX> -Call Record-<ETX> <BCC> and wait for Data Transfer Retry Timer (sec) - on No Response, default 3 seconds.	
Retransmit <STX> -Call Record-<ETX> <BCC> and wait for Data Transfer Retry Timer (sec) - on No Response, default 3 seconds.	

The SARVAM UCS will make 5 attempts (default value of *Data Transfer Retry Count - on No Response*) to send the message after a regular interval of 3 seconds (default value of *Data Transfer Retry Timer - on No Response*). If the ACK is still not received from the CAS, the SARVAM UCS send the new message to CAS.

Blind Send

If you select this protocol as the SMDR-OG Posting Protocol, SARVAM UCS sends the call details without waiting for any response from the CAS. Each record is sent with the End of Packet Character.

Customized

If you select this protocol as the SMDR-OG Posting Protocol, SARVAM UCS provides you the flexibility to set the values for the OG Handshaking Protocol and the OG Online Call Record Format as per your requirement.

Call Detail Record Formats

The default Call Detail Record formats for Blind Send and Matrix are given below.

Matrix

Parameter	Start Column Number	Field Length	Format	Alignment	Filler Char. Required?	Filler Char. Decimal Value	Remarks
Serial Number	01	04	Fixed	Right	Yes	032	Every 6 hours it is cleared to 001. (Starting from mid-night 00:00:00)
Increment Counter	00	01	Fixed	Left	NA	NA	Every 6 hours it is cleared to A. (Starting from mid-night 00:00:00)
Property Code	00	04	Fixed	Left	Yes	032	As per the Programmed String
Extension Number	06	05	Fixed	Right	Yes	032	
Authority Code	00	03	Fixed	Left	NA	NA	
Trunk Number	12	05	Matrix Format	Left	Yes	032	
Date	37	10	DD-MM-YYYY	Right	Yes	032	
Time	48	08	HH:MM:SS	Right	Yes	032	
Duration	057	005	Seconds	Right	Yes	032	
Units	063	004	Fixed	Right	Yes	032	
Amount	068	007	Currency with Decimal Point	Right	Yes	032	Format is DDD.CC
Currency	000	001	Fixed	Right	Space	032	Country Specific

Parameter	Start Column Number	Field Length	Format	Alignment	Filler Char. Required?	Filler Char. Decimal Value	Remarks
Call Type Indicator	000	001	Fixed	Right	NA	NA	As per the Call Type Indicator table programmed by the SE. The SE should program L = local, F=International and Space shall be used for long distance.
Location	000	005	Fixed	Right	NA	NA	
Called Number	18	19	Continuous	Left	Space	NA	
PIN	00	04	Fixed	Right	Yes	032	--
Account Code	00	04	Fixed	Right	Yes	032	
Remarks	76	02	Fixed	Left	Space	NA	
Reset Serial Number to 001	Do not Reset						
Staring Character - Increment Counter	A						
Reset Increment Counter	Do not Reset						
Prefix String Required	No						
Property Code	000						
Currency Symbol (Enter Decimal Value)	003 000 000 000 000 000 000 000						

AST

Parameter	Start Column Number	Field Length	Format	Alignment	Filler Char. Required?	Filler Char. Decimal Value	Remarks
Serial Number	01	04	Fixed	Right	Yes	032	Every 6 hours it is cleared to 001. (Starting from mid-night 00:00:00)
Increment Counter	00	01	Fixed	Left	NA	NA	Every 6 hours it is cleared to A. (Starting from mid-night 00:00:00)
Property Code	00	04	Fixed	Left	Yes	032	As per the Programmed String
Extension Number	06	05	Fixed	Right	Yes	032	
Authority Code	00	03	Fixed	Left	NA	NA	
Trunk Number	12	05	Matrix Format	Left	Yes	032	
Date	37	10	DD-MM-YYYY	Right	Yes	032	MM/DD
Time	48	08	HH:MM:SS	Right	Yes	032	HH:MM
Duration	057	005	Seconds	Right	Yes	032	Duration is in Minutes.
Units	063	004	Fixed	Right	Yes	032	
Amount	068	007	Currency with Decimal Point	Right	Yes	032	Format is DDD.CC
Currency	000	001	Fixed	Right	Space	032	$

Parameter	Start Column Number	Field Length	Format	Alignment	Filler Char. Required?	Filler Char. Decimal Value	Remarks
Call Type Indicator	000	001	Fixed	Right	NA	NA	As per the Call Type Indicator table programmed by the SE. The SE should program L = local, F=International and Space shall be used for long distance.
Location	000	005	Fixed	Right	NA	NA	
Called Number	18	19	Continuous	Left	NA	NA	Area code, Exchange code and Subscriber Number separated by dash. Space is sent in place of Area Code and first dash if area code is not present.
PIN	00	04	Fixed	Right	Yes	032	--
Account Code	00	04	Fixed	Right	Yes	032	--
Remarks	76	02	Fixed	Left	NA	NA	--
Reset Serial Number to 001	Do not Reset						
Staring Character - Increment Counter	A						
Reset Increment Counter	Do not Reset						
Prefix String Required	No						
Property Code	000						

Parameter	Start Column Number	Field Length	Format	Alignment	Filler Char. Required?	Filler Char. Decimal Value	Remarks
Currency Symbol (Enter Decimal Value)	000 000 000 000 000 000 000 000						

Blind Send

Parameter	Start Column Number	Field Length	Format	Alignment	Filler Char. Required?	Filler Char. Decimal Value	Remarks
Serial Number	01	04	Fixed	Right	Yes	032	
Increment Counter	00	01	Fixed	Left	NA	NA	
Property Code	00	04	Fixed	Left	Yes	032	As per the Programmed String
Extension Number	06	05	Fixed	Right	Yes	032	
Authority Code	00	03	Fixed	Left	NA	NA	
Trunk Number	12	05	Matrix Format	Left	Yes	032	
Date	37	10	DD-MM-YYYY	Right	Yes	032	
Time	48	08	HH:MM:SS	Right	Yes	032	
Duration	057	005	Seconds	Right	Yes	032	
Units	063	004	Fixed	Right	Yes	032	
Amount	068	007	Currency with Decimal Point	Right	Yes	032	Format is DDD.CC
Currency	000	001	Fixed	Right	Space	032	Country Specific

Parameter	Start Column Number	Field Length	Format	Alignment	Filler Char. Required?	Filler Char. Decimal Value	Remarks
Call Type Indicator	000	001	Fixed	Right	NA	NA	As per the Call Type Indicator table programmed by the SE. The SE should program L = local, F=International and Space shall be used for long distance.
Location	000	005	Fixed	Right	NA	NA	
Called Number	18	19	Continuous	Left	Space	NA	
PIN	00	04	Fixed	Right	Yes	032	--
Account Code	00	04	Fixed	Right	Yes	032	
Remarks	76	02	Fixed	Left	Space	NA	
Reset Serial Number to 001	Do not Reset						
Staring Character - Increment Counter	A						
Reset Increment Counter	Do not Reset						
Prefix String Required	No						
Property Code	000						
Currency Symbol (Enter Decimal Value)	013 010 000 000 000 000 000 000						

Customized SMDR-Posting Protocol

You can use the Customized SMDR Posting Protocol to match the settings required by the CAS you have interfaced. When you use Customized SMDR-Posting Protocol, you can customize the Call Detail Record format to match your requirement.

When the Call Detail Record format is customized, if there is a gap between two fields, these fields will be 'space' (ASCII-32).

It is also possible to customize the posting protocol to match the settings required by the CAS you have interfaced.

Setting up CAS Interface

You can setup the CAS Interface on Communication Port or the Ethernet (LAN/WAN) Port:

- To setup the CAS Interface on COM Port[336], the following functional components are required to make the interface work:

  - A PC with a spare serial/COM port (not supplied by Matrix).
  - The CAS Software (not supplied by Matrix).
  - The SARVAM UCS (supplied by Matrix).

  Now, connect the COM port of the PC with the COM port the SARVAM UCS using the communication cable supplied by Matrix[337].

- To setup the CAS Interface on the Ethernet (LAN/WAN) Port, the following functional components are required to make the interface work:

  - A PC with a spare LAN/WAN Port (not supplied by Matrix) Or any free LAN Port of the LAN Switch on which the CAS server application software is running.
  - The CAS Software (not supplied by Matrix).
  - The SARVAM UCS (supplied by Matrix).

  Now, connect the Ethernet (LAN/WAN) Port of the CPU Card of the SARVAM UCS with the Ethernet Port of the PC (on which CAS server application is running) or to one of the Ethernet port of the LAN Switch, if the CAS server is in the same LAN.

How to configure

Configuring the SMDR-Posting feature involves the following steps:

- Enabling storage of Outgoing (OG) SMDR. By default, OG SMDR storage is enabled. Refer “Station Message Detail Recording-Storage”.

- Selecting the appropriate SMDR-OG Posting Protocol to be used.

- Selecting the Destination Port for SMDR-Posting.

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336. This is applicable only for ETERNITY GENX.
337. This cable is supplied by Matrix as an optional item.

- If SMDR-Posting is through RS232 (that is, the CAS Interface is to be set up on the COM Port), program the attributes of the COM port. Refer the chapter "Communication Ports" to set attributes of the COM port.

- If SMDR-Posting is through TCP/IP (that is, the CAS Interface is to be set up on the Ethernet (LAN/ WAN) Port, program the destination IP address and Port.

- Refining the Handshake parameters, if required.

- Refining Call Detail Record format, if required.

- Starting SMDR-Posting process.

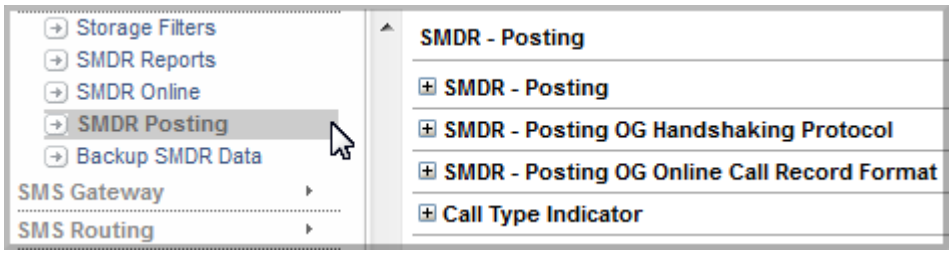
Configuring SMDR Posting using Jeeves,

- Log in as System Engineer.

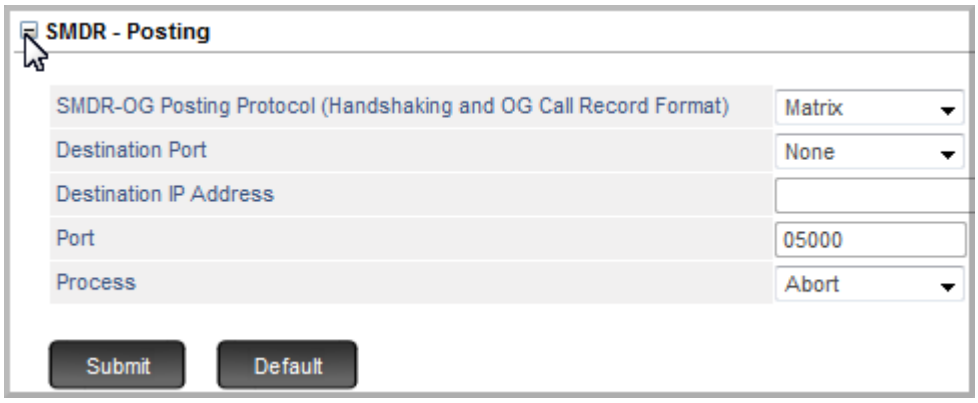
- Under **Configuration**, click **Station Message Detail Recording**.

Selecting SMDR-Posting Protocol for CAS

- Click the **SMDR Posting**. The SMDR-Posting parameters page opens.



- Click **SMDR-Posting** to expand.



- In the **SMDR-OG Posting Protocol** (Handshaking and OG Call Record Format) drop down list, select the appropriate protocol to be used. Default: Matrix.

- Select the **Destination Port** on which the SMDR Posting is set up. You can select **COM Port** or **Ethernet**. Default: None.

If you select **Ethernet**,

- In **Destination IP Address**, enter the IP Address of the PC on which the CAS server application software is running, that is, where SARVAM UCS should post SMDR. Both IPv4 and IPv6 addresses are supported.

- In **Port**, enter the port of the PC on which the CAS server application software is running, that is, where SARVAM UCS should post SMDR. Valid port range is: 5000; 514; 1025 to 65535.

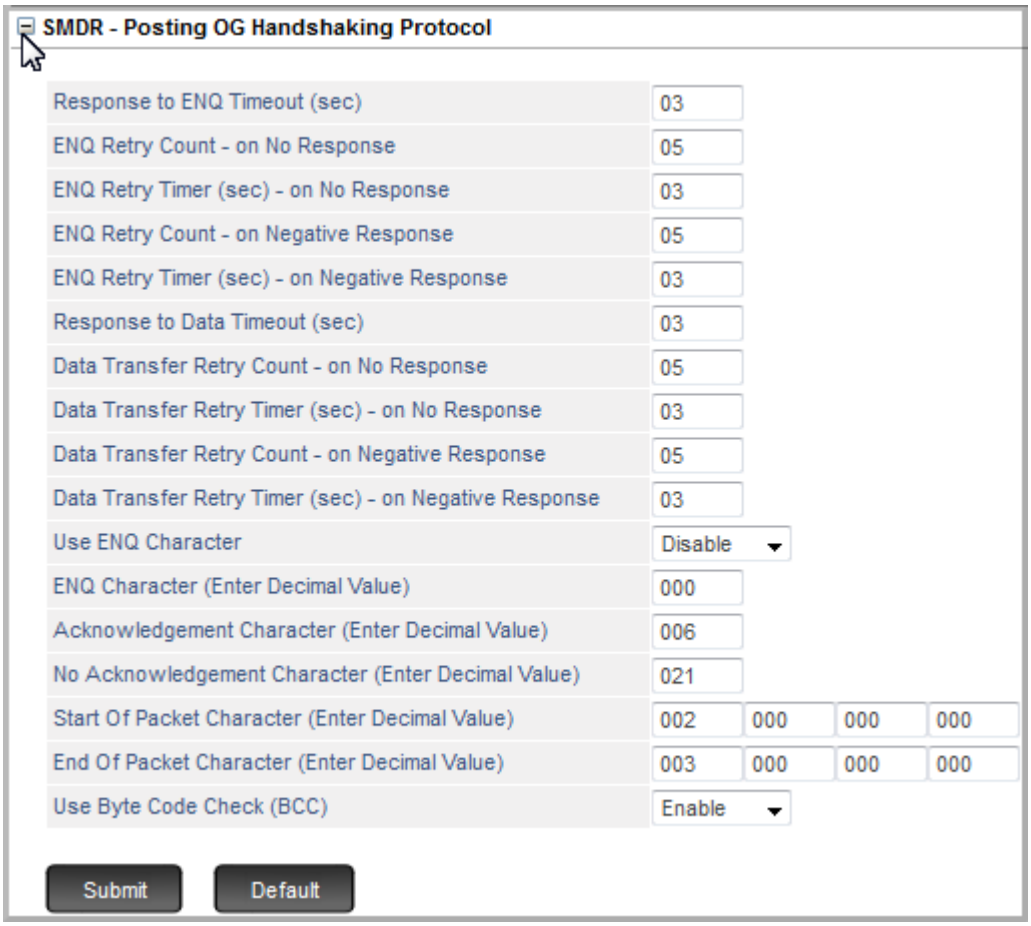
- To start SMDR posting, select **Process** as **Start**.

Refining Handshake Parameters

You may need to refine some of the Handshake parameters of the selected SMDR-Posting protocol, that is, change the factory default values of the protocol, to match the software requirements of the CAS being used in the organization. Refer the below table for default values of each protocol supported by SARVAM UCS.

To refine Handshake Parameters,

- Click **SMDR-Posting OG Handshaking Protocol**.



Configure the following parameters as required:

- **Response to ENQ Timeout (sec):** The time for which the sender waits for a response to ENQ from the receiver.

- **ENQ Retry Count - on No Response:** The number of times the sender should send ENQ before dropping the process, in case response is not received for the last message sent.

- **ENQ Retry Timer (sec) - on No Response:** The time after which the sender should sent the ENQ again, in case the response is not received for the last message sent.

- **ENQ Retry Count - on Negative Response:** The number of times the sender should send ENQ before dropping the process, in case of a negative response received for the last message sent.

- **ENQ Retry Timer (sec) - on Negative Response:** The time after which the sender should sent the ENQ again.

- **Response to Data Timeout (sec):** The time for which the sender waits for a response to data from the receiver.

- **Data Transfer Retry Count - on No Response:** The number of times the sender should send ENQ before dropping the process. This parameter is used when ACK is received against ENQ and there is some problem while sending the data.

- **Data Transfer Retry Timer (sec) - on No Response:** The time after which the sender should send the ENQ again before dropping the process. This parameter is used when ACK is received against ENQ and there is some problem in sending the data.

- **Data Transfer Retry Count - on Negative Response:** The number of times the sender should send ENQ before dropping the process. This parameter is used when ACK is received against ENQ and there is some problem in sending the data.

- **Data Transfer Retry Time (sec) - on Negative Response:** The time after which the sender should sent the ENQ again before dropping the process. This parameter is used when ACK is received against ENQ and there is some problem in sending the data.

- **Use ENQ Character:** This is enabled if the protocol uses ENQUIRE (ENQ) Signal.

- **ENQ Character:** This parameter signifies the ASCII character (Single Character) used to send ENQUIRE (ENQ) signal to the receiver.

- **Acknowledgement Character:** This parameter signifies the ASCII character (Single Character) used by the receiver to acknowledge the receipt of the Link Control Character/Message Data.

- **No Acknowledgement Character:** This parameter signifies the ASCII character (Single Character) used by the receiver to dis-acknowledge the receipt of the Link Control Character/Message Data.

- **Start of Packet Character:** A string of four ASCII characters used by the receiver to indicate Start of Packet. Each ASCII character is from 000 to 252. Start of Packet may be of one character only, in this case the string should be completed by programming remaining three characters with ASCII Null Character (000).

- **End of Packet Character:** A string of four ASCII characters used by the receiver to indicate End of Packet. Each ASCII character is from 000 to 252. End of Packet may be of one character only, in this case, the string should be completed by programming the remaining three characters should be programmed as ASCII Null (000).

- **Use Byte Code Check (BCC):** This flag is to be enabled when the protocol uses BCC Signal.

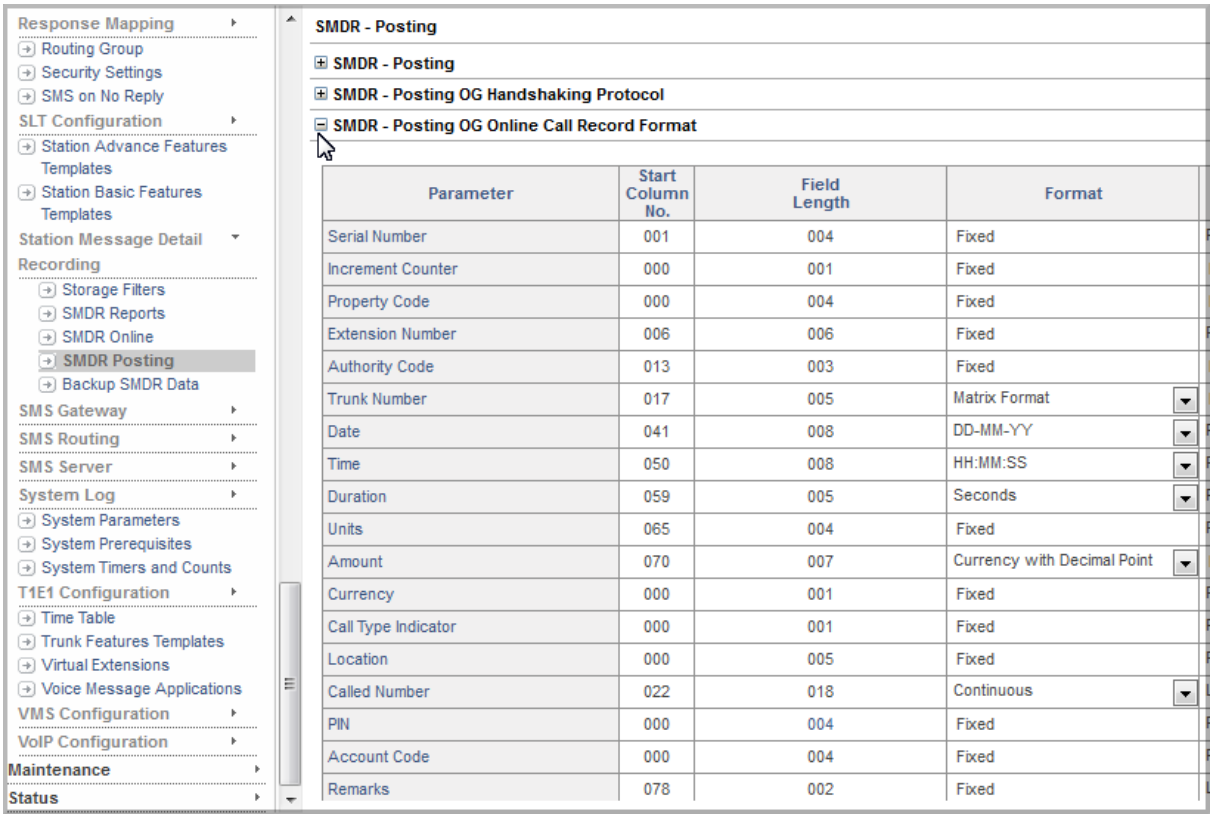
- Click **Submit** to save settings.

Refining Call Detail Record Format Parameters

The Call Detail Record (CDR) format for the selected SMDR-Posting protocol can also be refined to match the software requirements of the CAS being used by the organization.

This may be required if you have selected a 'customized' protocol. To refine Call Record Format,

- Click **SMDR-Posting OG Online Call Record Format** to expand.



Parameter	Start Column No.	Field Length	Format
Serial Number	001	004	Fixed
Increment Counter	000	001	Fixed
Property Code	000	004	Fixed
Extension Number	006	006	Fixed
Authority Code	013	003	Fixed
Trunk Number	017	005	Matrix Format
Date	041	008	DD-MM-YY
Time	050	008	HH:MM:SS
Duration	059	005	Seconds
Units	065	004	Fixed
Amount	070	007	Currency with Decimal Point
Currency	000	001	Fixed
Call Type Indicator	000	001	Fixed
Location	000	005	Fixed
Called Number	022	018	Continuous
PIN	000	004	Fixed
Account Code	000	004	Fixed
Remarks	078	002	Fixed

- **Serial Number:** This is the serial number generated for each call record. Serial numbers are generated from 000 to 999. When serial number '999' is reached, the numbers roll over to 000.

Serial Number starts from 1 and not 0.
When this field rolls over, it increments the increment counter.

- **Increment Counter:** It increments when the serial number counter rolls over. The Increment counter starts from A, ending at Z, and then roll over back to A.

- **Property Code:** This is the property code required by the CAS used in the organization. It is a string of alphanumeric characters and is to be terminated with #*. This field has a maximum of 128 alphanumeric characters.

- *You must program this string keeping in mind the field length used by the selected/customized posting protocol.*

- *The default value of the default Property Code String has been set as '000', as at least two known protocols use this field. You can set a different value here and the new value will appear in the CDR record, irrespective of the protocol type selected.*

- *If XIOX protocol has been selected as SMDR-OG Posting Protocol (Handshaking and OG Call Record Format), you should program Property Code as HTL.*

- **Extension Number:** This is the extension number from which the call was made. You can define the column position and the field length of the Extension number in the Call Detail Record.

- **Trunk Number:** This is the number of the trunk from which the call was made.

- *The Matrix Format occupies 5 character spaces.*

- *Check-Inn Format occupies 4 character spaces.*

- *The First Character in the Check-Inn Format is X (Fixed). The remaining three characters show the software port number. However, this does not specify whether the call is made through CO 125 or E&M 125. Also, the channel number is not specified in case of call made through T1E1PRI port or BRI port.*

- **Date:** The date on which the call was made. The date fill flag is to be enabled.

- *Filler Character field is applicable for Date, Month and Year, that is, whether the single digit date is to be printed as space-X or 0-X. For example, date = 1 is to be displayed as '1' or '01'.*

- *Where leading zeroes are not required, the date, month and year sub-fields are right aligned and the spaces are filled with character 'space'.*

- *The Date field is not linked to the global flag of Date Format. The global Flag of Date format is used, while using features or in configuration reports but not in CAS. This is because the date format used by the CAS is not the same as used by the users of the system.*

- **Time:** The time when the call was made. The format of the time field and the time fill flag are to be programmed.

- *Filler Character field is applicable for Hours, Minutes and Seconds, that is, whether the single digit hour is to be printed as space-X or 0-X. For example, hour = 1 is to be displayed as '1' or '01'.*

- *In case when leading zeroes are not required, Date, Month and Year sub-fields are right aligned and the spaces are filled with character 'space'.*

- **Duration:** The duration of each call. Program the duration unit and the duration fill flag.

When Duration Unit = Minutes, the rounding off to the nearest whole number is done. For seconds <= 30, Minute is not incremented. For seconds > 30, minute is incremented.

- **Units:** The duration of the call interpreted in terms of units. The number of units depends on the Pulse Rate. The number of units is derived from the Call Unit = Call duration in seconds/Pulse rate in seconds.

- **Amount:** This is the Amount of the call. Program the amount format and the fill flag.

- *Filler Character field is applicable for both the sub fields of Amount viz. Rupees/Paisa, that is, whether the single digit Rupee is to be printed as space-X or 0-X. For example, Rupee = 1 is to be displayed as*

'1' or '01'. Where leading zeroes are not required, the Rupee and Paisa are right aligned and the spaces are filled with character 'space'.

- *When Amount Format = Higher Currency, rounding to nearest whole number is done. For Lower Currency <= 50, Higher Currency is not incremented and for Lower currency > 50, Higher Currency is incremented.*

- **Currency:** This is the symbol of the currency in which the Amount is charged. A maximum of 8 ASCII Characters are allowed.

- *Generally, Currency Symbol field prefixes to Amount field. Hence, to comply with various CDR formats, it is recommended that the column position of Currency Symbol and Amount field should be programmed properly.*

- *You can change the Currency Symbol used in the OG-SMDR Format. However, this change will not be reflected in the Front Desk User Wizard.*

- **Call Type Indicator:** This indicates the type of call made, that is, whether local, international, information, etc.

  Your must program the Number String, the Text String and its Meaning as explained in following table:

Number Index	Number String	Text String	Meaning
01	0	LD	Long Distance
02	95	IC	Inter Circle
03	197	INFO	Information
04	0	INTL	International
:	:	:	:
36	2	L	Local

The Text String is a string of Alphanumeric characters. Number String is of a maximum 4-digits.

The Number Index is kept as '36' as one of the SMDR-OG Posting protocols, INN-FORM XL supports 24 different types of calls.

By default, all the entries in this table are blank.

You are advised to program the first 10 entries of this table as below if the selected posting protocol is Bell Hobic or XIOX.

Number Index	*Number String*	*Text String*	*Meaning*
01	*1*	*A*	
02	*2*	*A*	
03	*3*	*A*	
04	*4*	*A*	

Number Index	*Number String*	*Text String*	*Meaning*
05	*5*	*A*	
06	*6*	*A*	
07	*7*	*A*	
08	*8*	*A*	
09	*9*	*A*	
10	*0*	*A*	
:	*:*	*:*	
36	*Blank*	*Blank*	*Blank*

You are advised to program the first 11 entries of this table as below, if the selected posting protocol is Holidex or Hobic.

Number Index	*Number String*	*Text String*	*Meaning*
01	*1*	*L*	
02	*2*	*L*	
03	*3*	*L*	
04	*4*	*L*	
05	*5*	*L*	
06	*6*	*L*	
07	*7*	*L*	
08	*8*	*L*	
09	*9*	*L*	
10	*0*	*L*	
11	*0*	*F*	*International*
:	*:*	*:*	
36	*Blank*	*Blank*	*Blank*

- You are advised to use default (that is, Blank) table, if the selected protocol is Hilton, as Hilton uses blank entries in this field which is 12 bytes long.

- The Text String should preferably be same as Field Length. If not, the remaining spaces will be filled with character 'Space'. If the Field length is less than the Text string characters, then the number of text characters equal to the Field length will be printed.

- **Location:** This column indicates the location of the external number to which the call was made.

- *The system detects the location from the called location programmed in the Area and Country Code Tables.*

- *Called Location is programmed as one of the parameters of the Area Code Table and Country Code Table. Depending upon the prefix dialed, the Location string is picked up from either Country Code table or Area Code table.*

- *Called Location is not displayed for Local Calls.*

- *The Called Location parameter in the Country Code table and Area Code table is of 8 Characters.*

- *If the number of characters in the field Called Location is more than Field length then the remaining characters will not be printed (overlapped by next field).*

- *If the number of characters in the field Called Location is less than Field length then the remaining characters in the field Called Location will be filled by spaces.*

- **Called Number:** This is the external number to which the call was made.

- *One way to separate the called party number is by Area Code, Exchange code and Subscriber Number. This is difficult in an Open numbering system, in which the field size of area code, exchange code are not standard but vary from two digits to four digits (for example, the Area code for 'Mumbai' is of 2 digits, whereas that of 'Vadodara' is 3 digits).*

- *In the Closed numbering system, the Area Code, Exchange Code and the Subscriber number are of fixed length. In such case, including '-' in the called party number is not difficult. Hence, '-' is put in the called party number. The called party number is assumed to be of 10 digits. The first '-' is placed after four digits, counting from the right. The second '-'is placed after seven digits, counting from the right. If the dialed number is a local number of 7 digits then the second '-'is not placed. Also, the remaining three digits are not placed, but filled with character 'space'.*

- *In this case, even if the call is made to a geographical area where open numbering system is followed, '-' is placed in the same way.*

- **Account Code:** This is the Account Code (Refer Note4) using which the call was made.

- **Remarks:** This column indicates the details of the call; whether it was a DISA call, DOSA call, Auto Redial Call, type of call maturity.

  Fixed Characters are used to indicate the type of call, call details, etc. The notations for the Remarks field are:

D	DISA Call
A	Auto Redial Call
C	CPD
K	12KHz/16KHz
R	Reversal
D	Delay
I	Connect

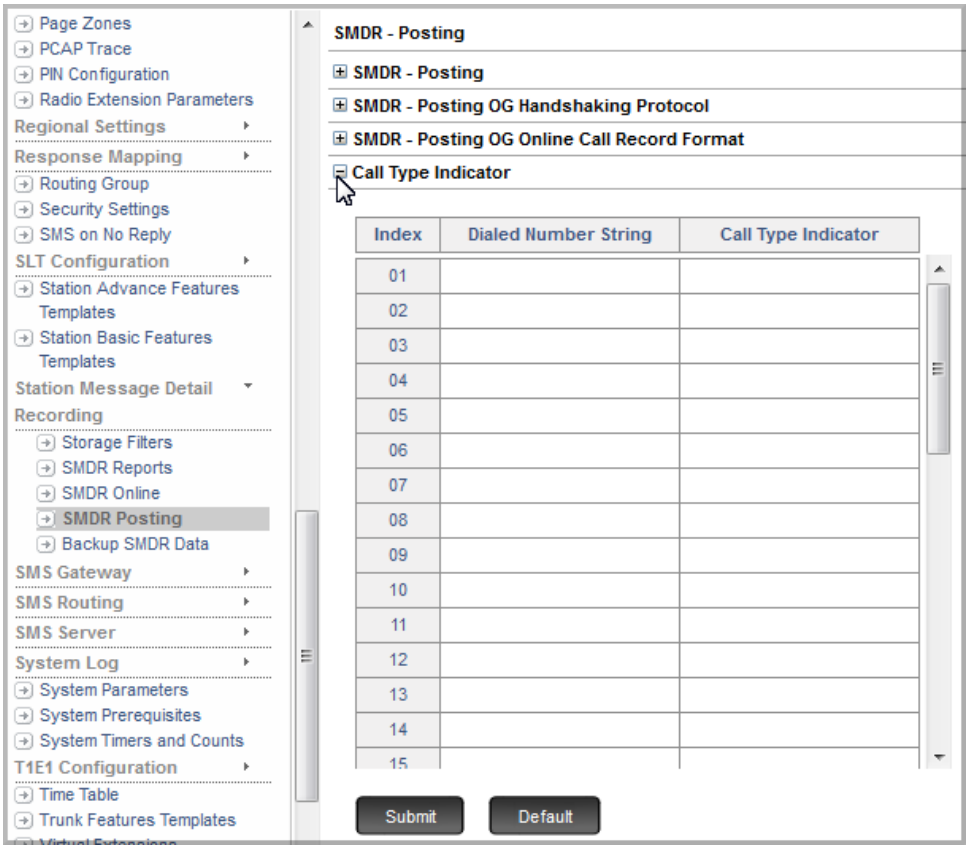
- **Reset Serial Number to 001:** The Serial number counter can be reset to 001 after 24 hours (from 00:00 HH:MM) or every 6 hours. By default, 'No Compulsory Reset' is selected, which means the serial number counter will not be automatically reset.

- **Starting Character - Increment Counter:** Specify the starting character of the increment counter as the serial number rolls over, in this field.
- **Reset Increment Counter:** The Increment Counter can be reset to 001 after 24 hours (from 00:00 HH:MM) or every 6 hours. By default, 'No Compulsory Reset' is selected, which means the serial number counter will not be automatically reset.
- **Prefix String Required:** This flag is to be programmed if the prefix string 0ac1 is to be sent when interfacing with OG-SMDR Posting Protocol.
- **Property Code:** Enter the property code required by the CAS.
- **Currency Symbol (Enter Decimal Value):** Enter currency symbol to be used.

Call Type Indicator

In the Call Record, you can also include the Type of Call: local, national, international, by configuring the Call Type Indicator Table.

- Click **Call Type Indicator** to expand.



- In the **Dialed Number String** column, enter the number strings for each Call Type. You can enter the prefix, e.g. 0 for long distance calls, 2 for local numbers, etc.
- For each **Dialed Number String**, define a **Call Type Indicator**, this is an abbreviation of the Call Type, e.g.: LD for long distance, INTL for International, etc.

Number String is of a maximum 4-digits. The Text String is a string of 4 alphanumeric characters. Your entries may look like these:

Number Index	Dialed Number String	Call Type Indicator (Text String)	Meaning
01	0	LD	Long Distance
02	95	IC	Inter Circle
03	197	INFO	Information
04	00	INTL	International
:	:	:	:
36	2	L	Local

You may enter as many Call Types as supported by the Posting Protocol you have selected.

- Click **Submit** to save entries.

Configuring SMDR Posting using a Telephone

Selecting SMDR -Posting Protocol for CAS

- Enter SE mode from a DKP/SLT.

  To enable/disable Storage of SMDR of Outgoing Calls, dial:
  - ***2701-Flag***
    Where,
    Flag is
    1 for Enable
    0 for Disable
    By default, storage of outgoing calls is enabled.

  To assign Destination Port for SMDR-OG Posting, dial:
  - **8330-Code**
    Where,

Code	Meaning
0	None
1	COM Port
2	Ethernet Port
3	USB to COM Port

    By default, the port assigned is Ethernet.

    For example, to select COM Port as destination port for SMDR Posting in SARVAM UCS, dial: ***8330-1***.

  To program CAS Server's IP Address (destination IP Address), dial:
  - **8331-Address-Address-Address-Address**
    Where,

Address is from 000 to 255 for the first 3 Octets and 001 to 254 for the fourth Octet.
By default, Destination Server IP Address is 192.168.1.103.

IPv6 address can be configured using Jeeves only.

To program CAS Server IP Port (Destination Port), dial:

- **8332-Destination IP Port**
  Where,
  Destination IP Port number is from 1025 to 65535.
  By default, destination IP Port is 5000.

To select appropriate SMDR-OG Posting Protocol:

- **8301-SMDR-OG Posting Protocol**
  Where,
  SMDR-OG Posting protocols is from 01 to 16.

Posting Protocol	Protocol Name
01	Blind Send
02	Matrix
03	Holidex
04	HOBIS A
05	HOBIS B
06	HOBIC
07	BELL HOBIC
08	MICROS A
09	MICROS B
10	Hilton
11	Xiox
12	Comm One
13	Call-Inn
14	RSI-CMS
15	Customized
16	AST

By default, Posting Protocol is 02, that is, Matrix.

When the above command is issued, the system will set the default values of the *OG Handshaking Protocol* and *OG Online Call Record Format* parameters as per the selected Posting Protocol.

To start/stop SMDR Posting Process, dial:

- **8333-Code**

Where,

Code	Meaning
0	Abort
1	Start

By default, SMDR Posting is Stop.

To restore the default OG SMDR-Posting Parameters, dial:

- **8300**

- Exit SE mode.

Handshaking Parameters

- Enter SE mode.

  To set ENQ no response timer, dial:

  - **8302-ENQ No Response Timer (Response to ENQ Timeout)**
    Where,
    ENQ No Response Timer is from 01-99 Seconds.

  To set ENQ no response retry count, dial:

  - **8303-ENQ Retry Count**
    Where,
    ENQ Retry Count is from 01-99.

  To set ENQ no response Retry Timer, dial:

  - **8304-ENQ No Response Retry Timer**
    Where,
    ENQ No Response Retry Timer is from 01-99 Seconds.

  To set ENQ Retry Count, dial:

  - **8305-ENQ Retry Count**
    Where,
    ENQ Retry Count is from 01-99.

  To set ENQ Retry Time, dial:

  - **8306-ENQ Retry Timer**
    Where,
    ENQ Retry Time is from 01-99 Seconds.

  To Response to Data Timeout, dial:

  - **8307-Response to Data Timeout**
    Where,
    Response to Data Timeout is from 01-99 Seconds.

  To set Data Transfer Retry Count, dial:

  - **8308-Data Transfer Retry Count**
    Where,
    Data Transfer Retry Count is from 01-99.

  To set Data Transfer Retry Time, dial:

- **8309-Data Transfer Retry Time**
  Where,
  Data Transfer Retry Time is from 01-99 Seconds.

To set Data Transfer Retry Count, dial:

- **8310-Data Transfer Retry Count**
  Where,
  Data Transfer Retry Count is from 01-99.

To set Data Transfer Retry Time, dial:

- **8311-Data Transfer Retry Time**
  Where,
  Data Transfer Retry Time is from 01-99 Seconds.

To enable/disable ENQUIRE Signal, dial:

- **8312-ENQUIRE Signal**
  Where

ENQUIRE Signal	Meaning
0	Disable
1	Enable

To set the ENQUIRE character, dial:

- ***8313-ENQUIRE***
  Where,
  ENQUIRE is an ASCII Character from 000 to 252.

To set the ACK Character, dial:

- **8314-Set ACK Character**
  Where,
  Set ACK Character is an ASCII Character from 000 to 252.

To program the NAK Character, dial:

- **8315-Set NAK Charact**er
  Where,
  Set NAK Character is an ASCII Character from 000 to 252.

To set the Start of Packet string, dial:

- **8316-Character 1-Character 2-Character 3-Character 4**
  Where,
  Start of Packet is a string of four ASCII Characters. Each ASCII character is from 000 to 252. If the Start of packet contains only one ASCII character then the string should be completed by programming remaining three characters with ASCII Null Character (000).

  If STX is to be programmed as 'Start of Packet', dial: ***8316-002-000-000-000***

To program the End of Packet string, dial:

- **8317-Character 1-Character 2-Character 3-Character 4**
  Where,
  End of Packet is a string of four ASCII Characters. Each ASCII character is from 000 to 252. If the End of packet contains only one ASCII character then the string should be completed by programming remaining three characters with ASCII Null Character (000).

If ETX is to be programmed as 'End of Packet', dial: ***8317-003-000-000-000***

To enable/disable BCC Flag, dial:

- **8318-BCC Flag**
  Where,

BCC Flag	Meaning
0	Disable
1	Enable

- Exit SE mode.

Call Detail Record format parameters

- Enter SE mode.

To program column position for serial number, dial:

- **8100-Column Position**
  Where,
  Column Position is from 000 to 128.
  By default, Column Position is 001.

To program field length for serial number, dial:

- **8101-Field Length**
  Where,
  Field Length is from 000 to 128.
  By default, Field Length is 004.

To program alignment for serial number, dial:

- **8102-Alignment**
  Where,

Alignment	Meaning
1	Left Alignment
2	Right Alignment

By default, Alignment is 2.

To program fill character for serial number, dial:

- **8103-Fill Character**
  Where,
  Fill Character is 3 digit ASCII value ranging from 032 to 254.
  By default, Fill Character is 'Zero'.

To program reset for serial number, dial:

- **8104-Reset**

Where,

Reset	Meaning
1	No Compulsory Reset
2	Reset to 001 every 24 hours (at 00:00 Hrs.)
3	Reset to 001 every 6 hours (at 00:00 Hrs.)

By default, Reset is '1'.

To program column position for increment counter, dial:

- ***8105-Column Position***
  Where,
  Column Position is from 000 to 128.
  By default, Column Position is 000.
  By default, Field Length is 1, which is fixed.

To program reset for increment counter, dial:

- **8106-Reset**
  Where,

Reset	Meaning
1	No Compulsory Reset
2	Reset to 001 every 24 hours (at 00:00 Hrs.)
3	Reset to 001 every 6 hours (at 00:00 Hrs.)

By default, Reset is '1'.

To program starting character for increment counter, dial:

- **8174-Starting Character**
  Where,
  Starting Character is from A to Z.
  By default, Starting Character is 'A'.

To program column position for property code, dial:

- **8107-Column Position**
  Where,
  Column Position is from 000 to 128.
  By default, Column Position is 000 (This field is not available by default)

To program field length for property code, dial:

- **8108-Field Length**
  Where,
  Field Length is from 000 to 128.
  By default, Field Length is 004.

To program property code string for property code, dial:

- **8109-Property Code String**
  Where,
  Property Code String is maximum of 16 characters.

By default, it is '000'.

To program column position for extension number, dial:

- **8110-Column Position**
  Where,
  Column Position is from 000 to 128.
  By default, Column Position is 006.

To program field length for extension number, dial:

- **8111-Field Length**
  Where,
  Field Length is from 000 to 128.
  By default, Field Length is 005.

To program alignment for extension number, dial:

- **8112-Alignment**
  Where,

Alignment	Meaning
1	Left Alignment
2	Right Alignment

By default, Alignment is 2.

To program fill character for extension number, dial:

- **8113-Fill Character**
  Where,
  Fill Character is 3 digit ASCII value and ranging from 032 to 254.
  By default, Fill Character is 'Space'.

To program column position for trunk number, dial:

- **8114-Column Position**
  Where,
  Column Position is from 000 to 128.
  By default, Column Position is 012.

To program format type for trunk number, dial:

- **8115-Format Type**
  Where,

Format Type	Meaning
1	Matrix Format
2	Check-Inn Format

By default, Format Type is '1'.

To program column position for date field, dial:

- **8116-Column Position**
  Where,
  Column Position is from 000 to 128.

By default, Column Position is 037.

To program field length for date field, dial:

- **8117-Field Length**
  Where,
  Field Length is from 000 to 128.
  By default, Field Length is 010.

To program alignment for date field, dial:

- **8118-Alignment**
  Where,

Alignment	Meaning
1	Left Alignment
2	Right Alignment

By default, Alignment is 2.

To program fill character for date field, dial:

- **8119-Fill Character**
  Where,
  Fill Character is 3 digit ASCII value from 032 to 254.
  By default, Fill Character is 'Space'.

To program date format for date field, dial:

- **8120-Date Format**
  Where,

Date Format	Meaning
01	DD-MM-YY
02	DD/MM/YY
03	DD.MM.YY
04	DD MM YY
05	DDMMYY
06	DD-MM-YYYY
07	DD/MM/YYYY
08	DD.MM.YYYY
09	DD MM YYYY
10	DDMMYYYY
11	MM-DD-YY
12	MM/DD/YY
13	MM.DD.YY
14	MM DD YY
15	MMDDYY

Date Format	Meaning
16	YY-MM-DD
17	YY/MM/DD
18	YY.MM.DD
19	YY MM DD
20	YYMMDD
21	YYYY-MM-DD
22	YYYY/MM/DD
23	YYYY.MM.DD
24	YYYY MM DD
25	YYYYMMDD
26	MM-DD
27	MM/DD
28	MM.DD
29	MM DD
30	MMDD
31	DD-MM
32	DD/MM
33	DD.MM
34	DD MM
35	DDMM

By default, the date format depends upon the Posting Protocol selected.

To program date fill flag for date field, dial:

- **8170-Date Fill Flag**

  Where,

Date Fill Flag	Meaning
0	Disable
1	Enable

By default, Date Fill Flag is '1'.

To program column position for time field, dial:

- **8122-Column Position**

  Where,

  Column Position is from 000 to 128.

  By default, Column Position is 048.

To program field length for time field, dial:

- **8123-Field Length**
  Where,
  Field Length is from 000 to 128.
  By default, Field Length is 008.

To program alignment for time field, dial:

- **8124-Alignment**
  Where,

Alignment	Meaning
1	Left Alignment
2	Right Alignment

  By default, Alignment is 2.

To program fill character for time field, dial:

- **8125-Fill Character**
  Where,
  Fill Character is 3 digit ASCII value ranging from 032 to 254.
  By default, Fill Character is 'Space'.

To program time format for time field, dial:

- **8126-Time Format**
  Where,

Time Format	Meaning
1	HH:MM:SS
2	HH:MM

  By default, Time format is 1.

To program time fill flag for time field, dial:

- **8171-Time Fill Flag**
  Where,

Time Fill Flag	Meaning
0	Disable
1	Enable

  By default, Time Fill Flag is '1'.

To program column position for duration field, dial:

- **8127-Column Position**
  Where,
  Column Position is from 000 to 128.
  By default, Column Position is 057.

To program field length for duration field, dial:

- **8128-Field Length**

Where,
Field Length is from 000 to 128.
By default, Field Length is 005.

To program alignment for duration field, dial:

- **8129-Alignment**

  Where,

Alignment	Meaning
1	Left Alignment
2	Right Alignment

  By default, Alignment is 2.

To program fill character for duration field, dial:

- **8130-Fill Character**

  Where,
  Fill Character is 3 digit ASCII value ranging from 032 to 254.
  By default, Fill Character is 'Space'.

To program duration unit for duration field, dial:

- **8131-Duration Unit**

  Where,

Duration Unit	Meaning
1	HH:MM:SS
2	HHMMSS
3	Minutes
4	Seconds

  By default, Duration Unit is '4'.

To program duration fill flag for duration field, dial:

- **8172-Duration Fill Flag**

  Where,

Duration Fill Flag	Meaning
0	Disable
1	Enable

  By default, Duration Fill Flag is '1'.

To program column position for units field, dial:

- **8132-Column Position**

  Where,
  Column Position is from 000 to 128.
  By default, Column Position is 063.

To program field length for units field, dial:

- **8133-Field Length**
  Where,
  Field Length is from 000 to 128.
  By default, Field Length is 004.

To program alignment for units field, dial:

- **8134-Alignment**
  Where,

Alignment	Meaning
1	Left Alignment
2	Right Alignment

  By default, Alignment is 2.

To program fill character for units field, dial:

- **8135-Fill Character**
  Where,
  Fill Character is 3 digit ASCII value ranging from 032 to 254.
  By default, Fill Character is 'Space'.

To program column position for amount field, dial:

- **8136-Column Position**
  Where,
  Column Position is from 000 to 128.
  By default, Column Position is 068.

To program field length for amount field, dial:

- **8137-Field Length**
  Where,
  Field Length is from 000 to 128.
  By default, Field Length is 007.

To program alignment for amounts field, dial:

- **8138-Alignment**
  Where,

Alignment	Meaning
1	Left Alignment
2	Right Alignment

  By default, Alignment is 2.

To program fill character for amounts field, dial:

- **8139-Fill Character**
  Where,
  Fill Character is 3 digit ASCII value ranging from 032 to 254.
  By default, Fill Character is 'Space'.

To program amount format for amount field, dial:

- **8140-Amount Format**

  Where,

Amount Format	Meaning
1	Higher Currency
2	Lower Currency
3	Spoken Currency with decimal point
4	Spoken Currency without decimal point

  By default, Amount Format is '3'.

To program the amount fill flag for amount field, dial:

- **8173-Amount Fill Flag**

  Where,

Amount Fill Flag	Meaning
0	Disable
1	Enable

  By default, Amount Fill Flag '1'.

To program column position for currency symbol field, dial:

- **8141-Column Position**

  Where,

  Column Position is from 000 to 128.

  By default, Column Position is 000.

To program field length for currency symbol field, dial:

- **8142-Field Length**

  Where,

  Field Length is from 000 to 128.

  By default, Field Length is 001.

To program alignment for currency symbol field, dial:

- **8143-Alignment**

  Where,

Alignment	Meaning
1	Left Alignment
2	Right Alignment

  By default, Alignment is '2'.

To program fill character for currency symbol field, dial:

- **8144-Fill Character**

  Where,

  Fill Character is 3 digit ASCII value ranging from 032 to 254.

By default, Fill Character is 'Space'.

To program symbol for currency symbol field, dial:

- **8145-Character 1-Character 2-Character 3-Character 4-Character 5-Character 6-Character 7-Character 8**
  Where,
  Character 1 to Character 8 shall be in 3 digit Decimal values.
  Decimal values 000 and 032 to 255 are allowed.

  If currency string/symbol to be used is fewer than 8 characters, terminate the command with ***#*.***

  Refer following table to know Decimal value of corresponding currency character:

Dec	ASCII Char.	Meaning	Dec	ASCII Char.	Meaning
0	NUL	Null	32		Space
1	SOH	Start of heading	33	!	!
2	STX	Start of text	34	"	"
3	ETX	Break/end of text	35	#	#
4	EOT	End of transmission	36	$	$
5	ENQ	Enquiry	37	%	%
6	ACK	Positive acknowledgment	38	&	&
7	BEL	Bell	39	'	'
8	BS	Backspace	40	(	(
9	HT	Horizontal tab	41	)	)
10	LF	Line feed	42	*	*
11	VT	Vertical tab	43	+	+
12	FF	Form feed	44	,	,
13	CR	Carriage return	45	-	-
14	SO	Shift out	46	.	.
15	SI	Shift in/XON (resume output)	47	/	/
16	DLE	Data link escape	48	0	Zero
17	DC1	XON - Device control character 1	49	1	One
18	DC2	Device control character 2	50	2	Two
19	DC3	XOFF - Device control character 3	51	3	Three
20	DC4	Device control character 4	52	4	Four
21	NAK	Negative Acknowledgment	53	5	Five
22	SYN	Synchronous idle	54	6	Six
23	ETB	End of transmission block	55	7	Seven
24	CAN	Cancel	56	8	Eight
25	EM	End of medium	57	9	Nine
26	SUB	substitute/end of file	58	:	:
27	ESC	Escape	59	;	;
28	FS	File separator	60	<	<
29	GS	Group separator	61	=	=
30	RS	Record separator	62	>	>
31	US	Unit separator	63	?	?

Dec	ASCII Char.	Meaning	Dec	ASCII Char.	Meaning
64	@	@	96	`	`
65	A	A	97	a	a
66	B	B	98	b	b
67	C	C	99	c	c
68	D	D	100	d	d
69	E	E	101	e	e
70	F	F	102	f	f
71	G	G	103	g	g
72	H	H	104	h	h
73	I	I	105	i	i
74	J	J	106	j	j
75	K	K	107	k	k
76	L	L	108	l	l
77	M	M	109	m	m
78	N	N	110	n	n
79	O	O	111	o	o
80	P	P	112	p	p
81	Q	Q	113	q	q
82	R	R	114	r	r
83	S	S	115	s	s
84	T	T	116	t	t
85	U	U	117	u	u
86	V	V	118	v	v
87	W	W	119	w	w
88	X	X	120	x	x
89	Y	Y	121	y	y
90	Z	Z	122	z	z
91	[	[	123	{	{
92	\	\	124	|	|
93	]	]	125	}	}
94	^	^	126	~	Tilde
95	_	_	127	DEL	Delete

To program column position for call type indicator field, dial:

- **8146-Column Position**
  Where,
  Column Position is from 000 to 128.
  By default, Column Position is 000.

To program field length for call type indicator field, dial:

- **8147-Field Length**
  Where,
  Field Length is from 000 to 128.
  By default, Field Length is 001.

To program alignment for call type indicator field, dial:

- **8148-Alignment**

  Where,

Alignment	Meaning
1	Left Alignment
2	Right Alignment

  By default, Alignment is '2'.

To program number string for call type indicator field, dial:

- **8149-Number Index-1-Number String**

  Where,

  Number Index is from 01 to 36.

  Number String is of four digits. Terminate with #* if the string is less than four digits.

To program text string for call type indicator field, dial:

- **8149-Number Index-2-Text String**

  Where,

  Number Index is from 01 to 36.

  Text String is a string of Alphanumeric characters. Terminate with #* if the string is less than four digits. Keep the Text String same as the Field Length.

  By default, all the entries in this table are blank.

To program column position for called location field, dial:

- **8150-Column Position**

  Where,

  Column Position is from 000 to 128.

  By default, Column Position is 000.

To program field length for called location field, dial:

- **8151-Field Length**

  Where,

  Field Length is from 000 to 128.

  By default, Field Length is 005.

To program alignment for called location field, dial:

- **8152-Alignment**

  Where,

Alignment	Meaning
1	Left Alignment
2	Right Alignment

  By default, Alignment is '2'.

To program column position for called number field, dial:

- **8154-Column Position**

  Where,

  Column Position is from 000 to 128.

By default, Column Position is 018.

To program field length for called number field, dial:

- **8155-Field Length**
  Where,
  Field Length is from 000 to 128.
  By default, Field Length is 019.

To program alignment for called number field, dial:

- **8156-Alignment**
  Where,

Alignment	Meaning
1	Left Alignment
2	Right Alignment

By default, Alignment is '1'.

To program number format for called number field, dial:

- **8157-Number Format**
  Where,

Number Format	Meaning
1	Continuous
2	Separated

By default, Number Format is '1'.

To program column position for account code field, dial:

- **8158-Column Position**
  Where,
  Column Position is from 000 to 128.
  By default, Column Position is 000.

To program field length for account code field, dial:

- **8159-Field Length**
  Where,
  Field Length is from 000 to 128.
  By default, Field Length is 004.

To program alignment for account code field, dial:

- **8160-Alignment**
  Where,

Alignment	Meaning
1	Left Alignment
2	Right Alignment

By default, Alignment is '2'.

To program fill character for account code field, dial:

- **8161-Fill Character**
  Where,
  Fill Character is 3 digit ASCII value ranging from 032 to 254.
  By default, Fill Character is 'Space'.

To program prefix string (ac01), dial:

- **8165-Code**
  Where,

Code	Meaning
0	No
1	Yes

  By default, Code is '0'.

To program column position for remarks field, dial:

- **8166-Column Position**
  Where,
  Column Position is from 000 to 128.
  By default, Column Position is 076.

To program field length for remarks field, dial:

- **8167-Field Length**
  Where,
  Field Length is from 000 to 128.
  By default, Field Length is 002.

To program alignment for remarks field, dial:

- **8168-Alignment**
  Where,

Alignment	Meaning
1	Left Alignment
2	Right Alignment

  By default, Alignment is '1'.

To assign default CDR format, dial:

- 8169

Configuring Area Code and Country Code

Country code and Area Code must be programmed so that a call can be placed to a particular location in a particular country. Country and Area Codes need to be programmed also for the purpose of calculating cost of calls. Refer chapter “Call Cost Calculation (CCC)”.

To program a country code, dial:

- **8321-Index-Country Code-#***
  Where,

Index is from 001 to 200.

Country Code is a number string of maximum of 4 digits, terminate with #* if less than four digits.

Index	Country Code
001	041
002	051
:	:
200	096

To program a location in country code, dial:

- **8322-Index-Country Name-#***

  Where,

  Index is from 001 to 200.

  Country Name is of 8 characters; terminate the command string with #*, if the name is less than 8 characters.

Index	Country Name
001	India
002	Kenya
:	:
200	UAE

Station Message Detail Recording-Report

What's this?

The SARVAM UCS can generate SMDR reports in two modes:

- Online: as and when a call is made or received (see “Station Message Detail Recording-Online”)
  Or
- Offline: whenever required, the records of calls stored in the buffer can be printed.

Generation of call record reports offline, is called SMDR - Report.

You can generate SMDR Report, either

- Manually: The report is generated whenever you want.
  Or
- As per Schedule: The report is generated on a preset Day, Date and Time.

SARVAM UCS allows you to set a variety of filters for printing SMDR Reports.

SARVAM UCS supports SMDR-Reports on Serial RS232 Communication Port as well as on TCP/IP Ethernet Port.

How to configure

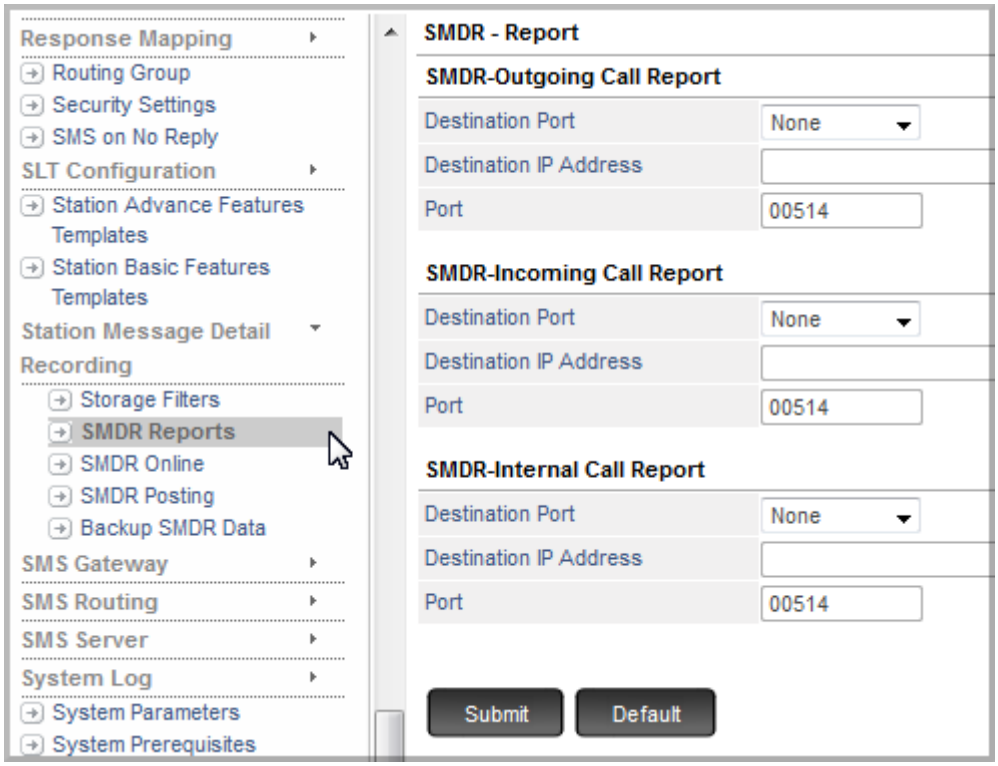
To be able to generate SMDR -Report, you must do the following:

- Enable SMDR Storage in the SMDR buffer. See “Station Message Detail Recording-Storage”.
- Assign the Destination port for Incoming, Outgoing and Internal calls.

Configuring SMDR-Online using Jeeves

- Login as System Engineer.
- Under **Configuration**, click **Station Message Detail Recording**.

- Click **SMDR Reports** to open the page.



- For **SMDR - Outgoing Call Report**,
  - Select the **Destination Port**. Default: None.
  - If you select Ethernet as the Destination Port,
    - In **Destination IP Address**, enter the IP Address of the remote Syslog Server. Both IPv4 and IPv6 addresses are supported.
    - In **Port**, enter the port of the remote Syslog Server. Valid port range is: 514; 1025 to 65535.
- For **SMDR - Incoming Call Report**,
  - Select the **Destination Port**. Default: None.
    - In **Destination IP Address**, enter the IP Address of the remote Syslog Server. Both IPv4 and IPv6 addresses are supported.
    - In **Port**, enter the port of the remote Syslog Server. Valid port range is: 514; 1025 to 65535.
- For **SMDR - Internal Call Report**,
  - Select the **Destination Port**. Default: None.
    - In **Destination IP Address**, enter the IP Address of the remote Syslog Server. Both IPv4 and IPv6 addresses are supported.
    - In **Port**, enter the port of the remote Syslog Server. Valid port range is: 514; 1025 to 65535.

- Click **Submit** to save settings.

Configuring SMDR-Online using a Telephone

Incoming calls report

Destination Port:

To assign a destination port for SMDR-IC Report, dial:

- ***2931-Code***

  Where,

Code	Meaning
0	None
1	COM Port
2	Ethernet Port
3	USB to COM Port

If you assigned Ethernet Port as destination port, to assign the IP Address to the Ethernet Port, dial:

- ***2934-IP Address***

  By default, IP Address is 192.168.1.104

IPv6 address can be configured using Jeeves only.

To assign the Port, dial:

- ***2935- Port***

  Where,

  Port is from 514 and 1025-65535

  By default, IP Port is 514.

Internal Calls Report

Destination Port

To assign destination port for Online SMDR-Internal Report, dial:

- ***2831-Code***

  Where,

Code	Meaning
0	None
1	COM Port
2	Ethernet Port
3	USB to COM Port

If you assigned Ethernet Port as destination port, to assign the IP Address to the Ethernet Port, dial:

- ***2834-IP Address***

  By default, IP Address is 192.168.1.104

IPv6 address can be configured using Jeeves only.

To assign the IP Port, dial:

- ***2835-IP Port***
  Where,
  IP Port is from 514 and 1025-65535
  By default, IP Port is 514.

OG Calls Report

Destination Port

To assign a destination port for SMDR-OG Report, dial:

- ***2731-Code***
  Where,

Code	Meaning
0	None
1	COM Port
2	Ethernet Port
3	USB to COM Port

If you assigned Ethernet Port as destination port, to assign the IP Address to the Ethernet Port, dial:

- ***2734-IP Address***
  By default, IP Address is 192.168.1.104

IPv6 address can be configured using Jeeves only.

To assign the IP Port, dial:

- ***2735-IP Port***
  Where,
  IP Port is from 514 and 1025-65535
  By default, IP Port is 514.

How to use

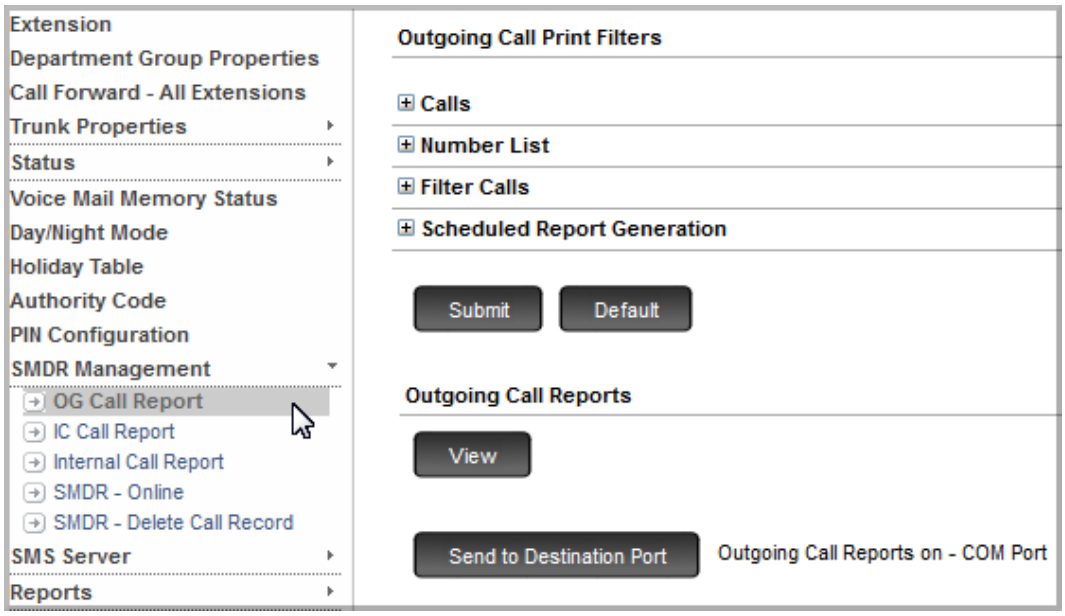
You can print SMDR Report whenever you want or schedule printing of the report from the System Administrator mode using Jeeves or dialing SA Commands from an extension phone.

Print SMDR-Report using Jeeves

- Open Jeeves.
- Log in as System Administrator.
- Click **SMDR Management** to expand.

Outgoing Calls

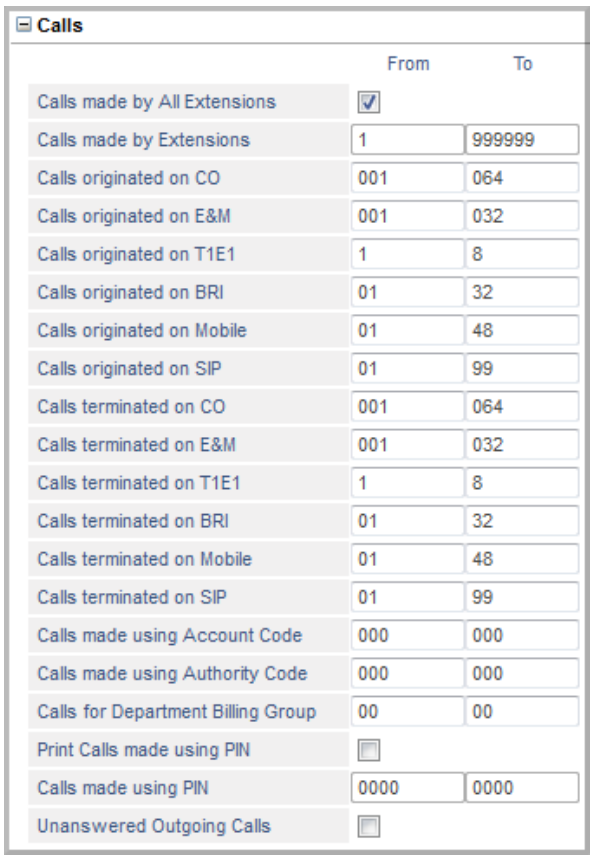
- To print Outgoing Calls with filters, click **OG Call Report**.



Setting Print Filters

To print calls originating and termination on specific trunks/extensions,

- Click **Calls** to expand.



- Set the following filters as desired. You can print records of outgoing calls made by all extensions or by specific extensions: **Calls made by All Extensions**, **Calls made by Extensions.** Default: Calls made by All Extensions

  To print records of specific extensions, clear the **Calls made by All Extensions** check box and enter the desired extension range in **From** and **To** in **Calls made by Extension**.

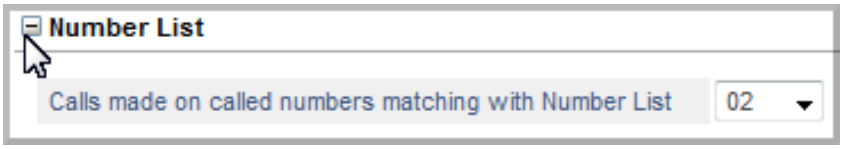
  You can also print records of outgoing calls Originating and outgoing calls Terminating on specific Extension numbers, and trunks: **CO**, **E&M**, **T1E1**, **BRI**, **Mobile** and **SIP**.

  You can also print outgoing calls made using **Account Code**, **Authority Codes**, **Calls for Department Billing Groups**, **PIN** and **Unanswered Outgoing Calls**.

*If you have extension numbers beginning with # or *, make sure the range you assign in From and To either have # or *. A mix of both will not work.*

To print outgoing calls made to certain numbers,

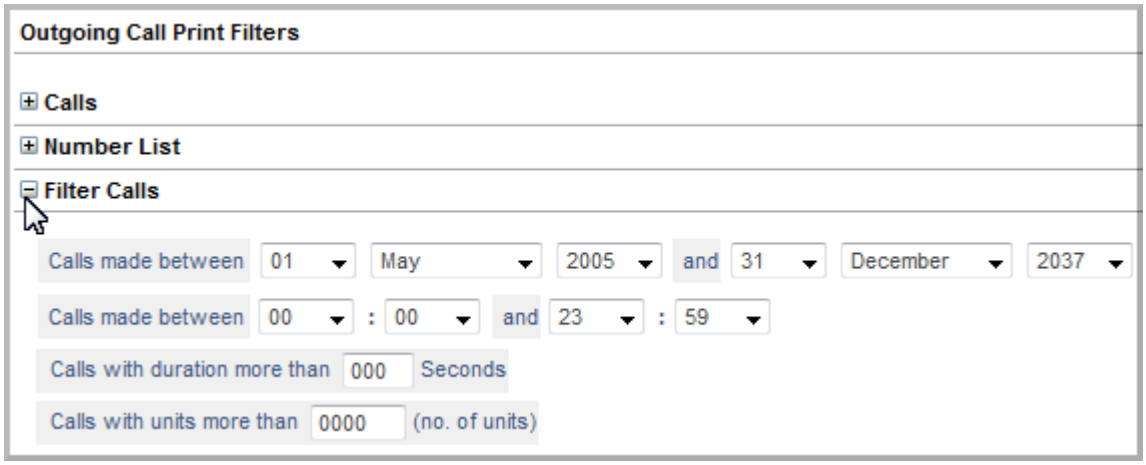
- Click **Number List** to expand.



  Enter the desired numbers in the Number List of your choice and select the same number list in **Calls made on called numbers matching with Number List**.

To filter calls according to specific dates, time and duration,

- Click **Filter Calls** to expand,



- To print outgoing calls made on a certain date or between a certain time period, set the filter **Calls made between**. To print calls made on a particular date, select the same Date, Month and Year in both fields.

- To print outgoing calls made at a particular time, set the Hours and Minutes in 24-hour format call in the filter **Calls made between 00: 00 and 23:59**.

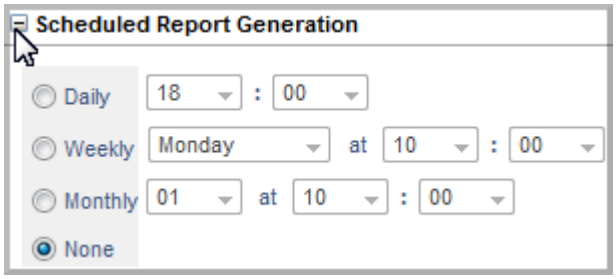
- If you want outgoing calls that exceed as certain duration to be printed, set the filter **Calls with duration more than (sec)** to the desired duration. All outgoing calls with duration greater than this value, will be printed.

- If you want outgoing calls exceeding certain metering units to be printed, set the filter **Calls with units more than (units)** to the desired value. All outgoing calls that have metering units greater than this value will be stored.

- Click **Submit** to save.

Scheduled Report Generation

- Click **Scheduled Report Generation** to expand.



- To generate SMDR Report of outgoing calls on a particular day, day of the week, or day of the month, set Scheduled Report Generation, as required.

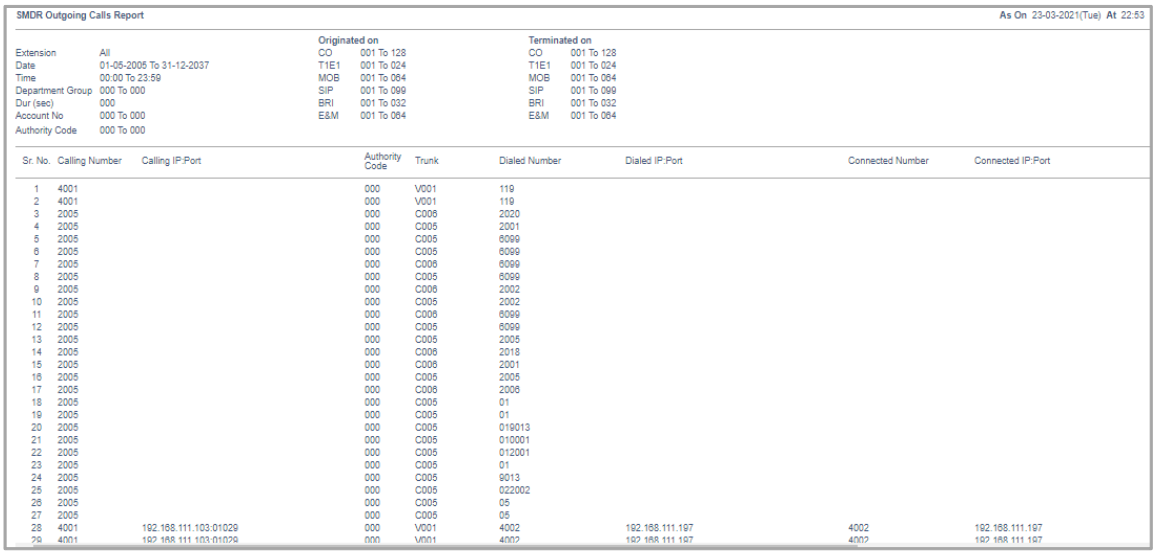
- Click **Submit** to save.

Manual Report Generation

- You can print the SMDR Report of outgoing calls any time you want. You can print the report on a local printer or the Destination Port of SARVAM UCS.

To view/print the report on the local printer,

- Click the **View** button.


SMDR Outgoing Calls Report

As On 23-03-2021(Tue) At 22:53

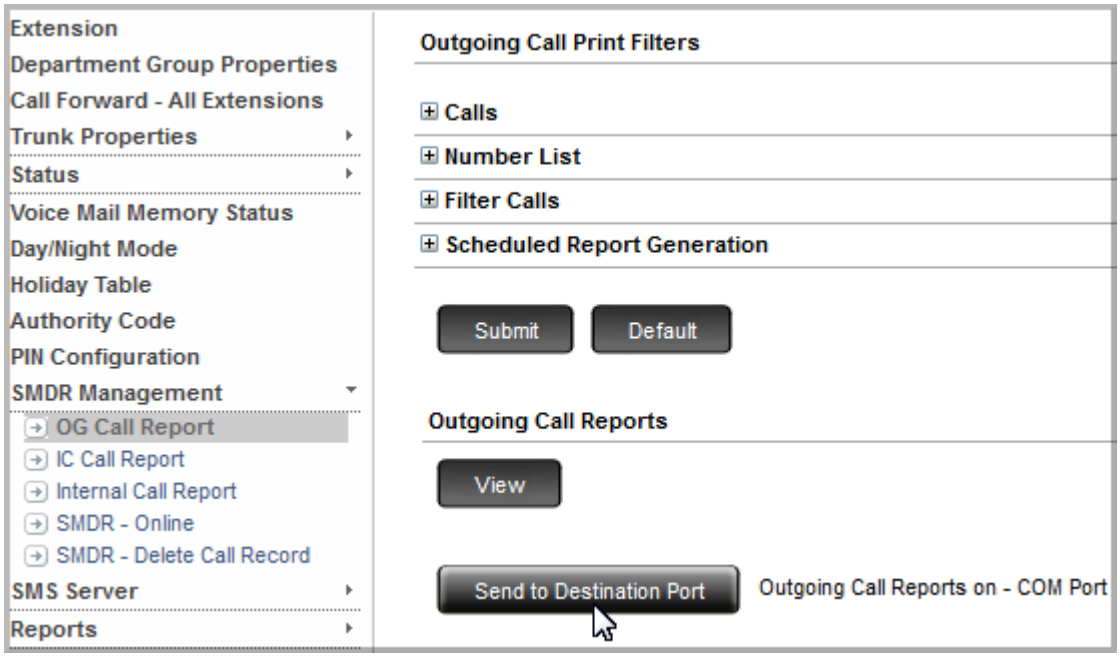
		Originated on		Terminated on	
Extension	All	CO	001 To 128	CO	001 To 128
Date	01-05-2005 To 31-12-2037	T1E1	001 To 024	T1E1	001 To 024
Time	00:00 To 23:59	MOB	001 To 064	MOB	001 To 064
Department Group	000 To 000	SIP	001 To 099	SIP	001 To 099
Dur (sec)	000	BRI	001 To 032	BRI	001 To 032
Account No	000 To 000	E&M	001 To 064	E&M	001 To 064
Authority Code	000 To 000				

Sr. No.	Calling Number	Calling IP:Port	Authority Code	Trunk	Dialed Number	Dialed IP:Port	Connected Number	Connected IP:Port
1	4001		000	V001	119			
2	4001		000	V001	119			
3	2005		000	C006	2020			
4	2005		000	C005	2001			
5	2005		000	C005	6099			
6	2005		000	C005	6099			
7	2005		000	C006	6099			
8	2005		000	C005	6099			
9	2005		000	C006	2002			
10	2005		000	C005	2002			
11	2005		000	C006	6099			
12	2005		000	C005	6099			
13	2005		000	C005	2005			
14	2005		000	C006	2018			
15	2005		000	C006	2001			
16	2005		000	C005	2005			
17	2005		000	C006	2006			
18	2005		000	C005	01			
19	2005		000	C005	01			
20	2005		000	C005	019013			
21	2005		000	C005	010001			
22	2005		000	C005	012001			
23	2005		000	C005	01			
24	2005		000	C005	9013			
25	2005		000	C005	022002			
26	2005		000	C005	05			
27	2005		000	C005	05			
28	4001	192.168.111.103:01029	000	V001	4002	192.168.111.197	4002	192.168.111.197
29	4001	192.168.111.103:01029	000	V001	4002	192.168.111.197	4002	192.168.111.197

- A detailed report is displayed on the screen.
- Click **Print** to print the report.

To print the report on the Destination Port,

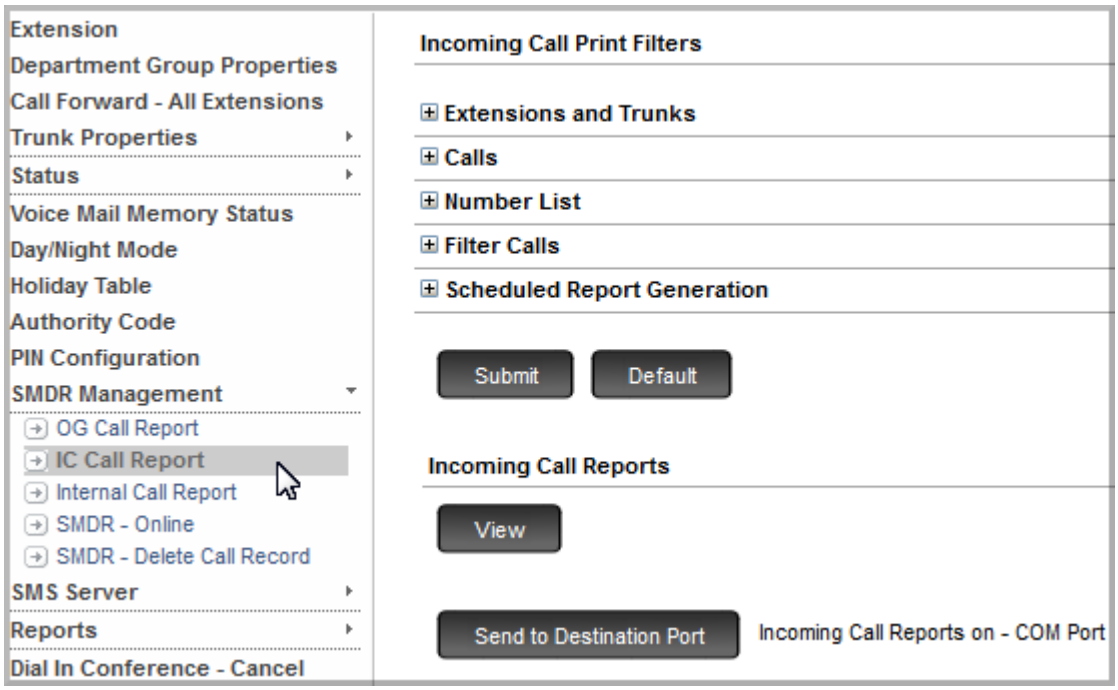
- Click the **Send to Destination Port** button.



- To stop printing, click **Abort** button.

Incoming Calls

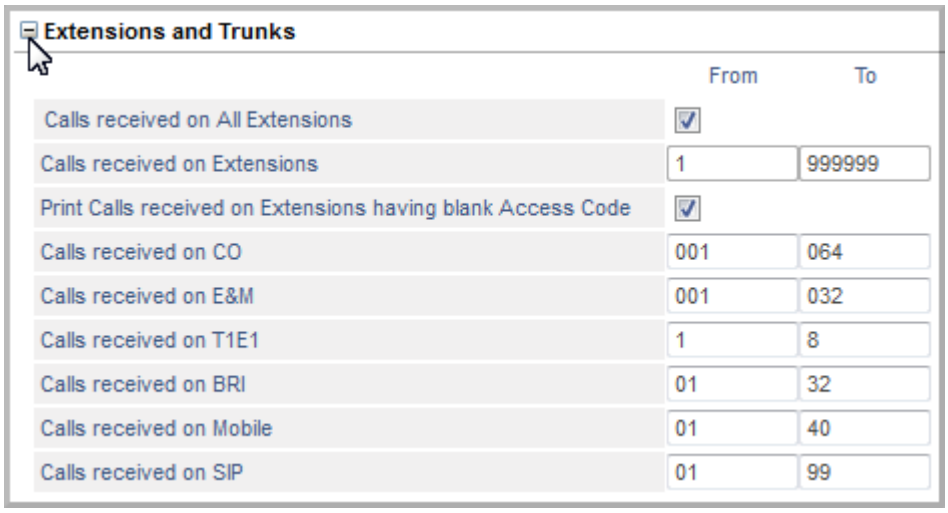
- To print Incoming Calls with filters, click **IC Call Report**.



Setting Print Filters

To print calls received on specific trunks/extensions,

- Click **Extension and Trunks** to expand.



- Set the filters as desired. You can print records of incoming calls received on a specific extension or a range of extensions and trunk ports: **CO**, **E&M**, **T1E1**, **BRI**, **Mobile** and **SIP**.

  To print records of specific extensions, clear the **Calls received on All Extensions** check box and enter the desired extension range in **From** and **To**.

- To print calls received only on extensions that have not been assigned Access Codes, select the **Print Calls received on Extensions having blank Access Code** check box. Make sure in **Calls received from Extensions**, you have assigned 0 in the **From** and **To** fields.

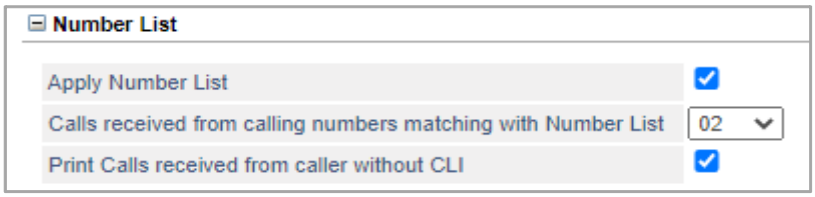
To print incoming calls according to Call Type,

- Click **Calls** to expand.

- You can also print incoming calls of different Call Types: **Normal** calls, calls received using **Built-In Auto Attendant**, calls that remained **Unanswered**, **Unanswered calls on Built-In Auto Attendant**, and calls made using **DISA**.

To print incoming calls received from certain numbers,

- Click **Number List** to expand.



- Select the **Apply Number List** check box and then you can configure **Calls received from calling numbers matching with Number List** and **Print Calls received from caller without CLI.**

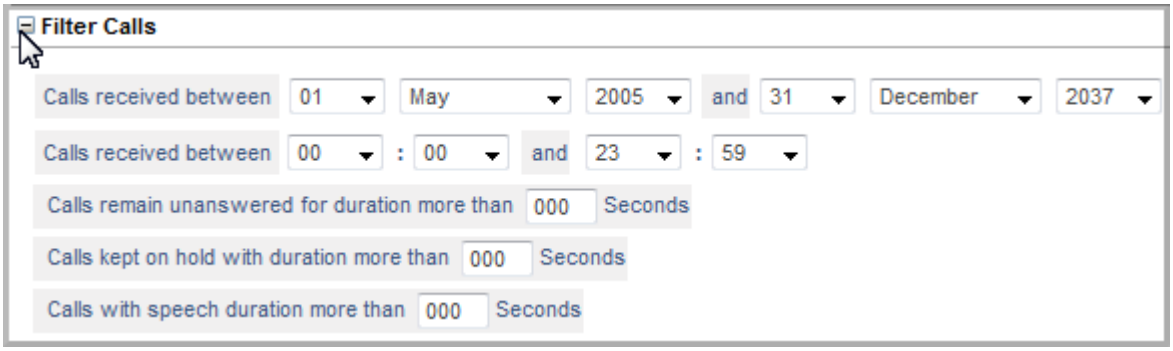
  To print outgoing calls received from certain numbers, enter the CLI of these numbers in the Number List of your choice and select the same number list in **Calls received from calling numbers matching with Number List**.

  To print calls received without CLI, select the **Print Calls received from caller without CLI** check box.

*If **Accept Anonymous Calls on SIP trunks** is enabled, to view the details in the SMDR Report, make sure you enable the **Print Calls received from caller without CLI** check box.*

To filter calls between specific dates, time or with a specific duration,

- Click **Filter Calls** to expand.



- To print incoming calls received on a certain date or between a certain time period, set the filter **Calls received between**. To print calls made on a particular date, select the same Date, Month and Year in both fields.

- To print incoming calls received at a particular time, set the Hours and Minutes in 24-hour format call in the filter **Calls received between 00: 00 and 23:59**.

- To print incoming calls that remained unanswered for more than a certain duration, set the filter **Calls remain unanswered for duration more than (sec)**.

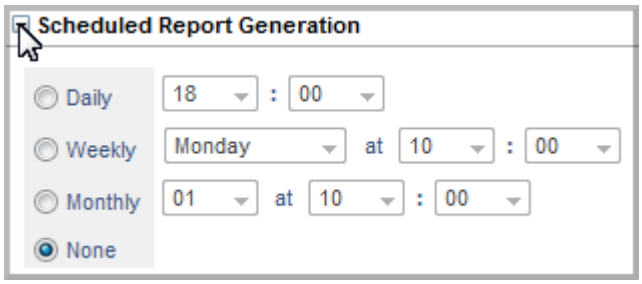
- To print calls that were kept on hold for more than a certain duration, set the filter **Calls kept on hold with duration more than (seconds)** to the desired value.

- To print calls with speech duration of a certain duration, set the filter **Calls with speech duration more than (seconds)**

- Click **Submit** to save.

Scheduled Report Generation

- Click **Scheduled Report Generation** to expand.



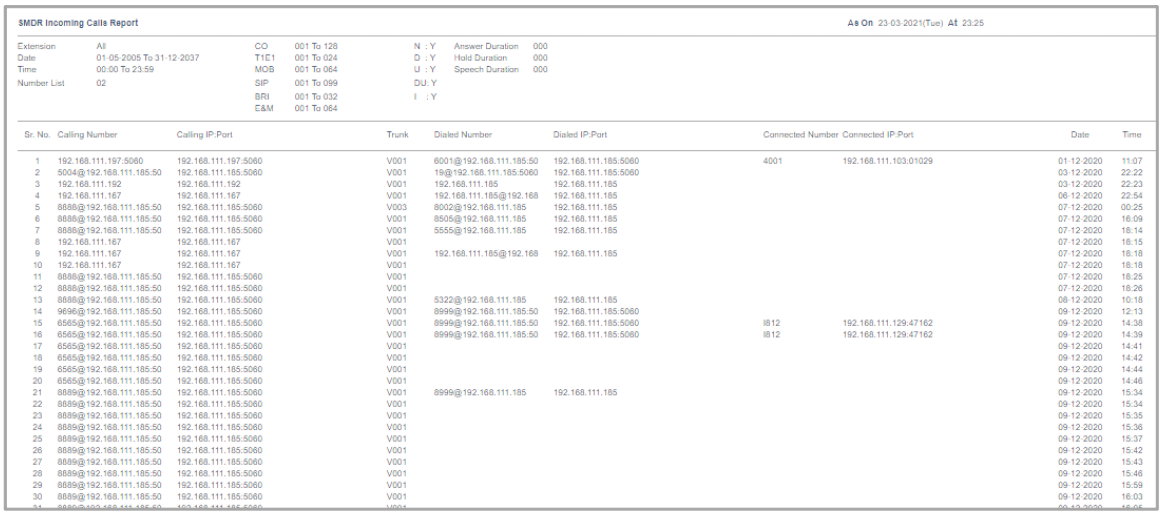
- To generate SMDR Report of incoming calls on a particular day, day of the week, or day of the month, set Scheduled Report Generation, as required.
- Click **Submit** to save.

Manual Report Generation

- You can print the SMDR Report of incoming calls any time you want. You can print the report on a local printer or the Destination Port of SARVAM UCS.

To view/print the report on the local printer,

- Click the **View** button.


SMDR Incoming Calls Report

As On 23-03-2021(Tue) At 23:25

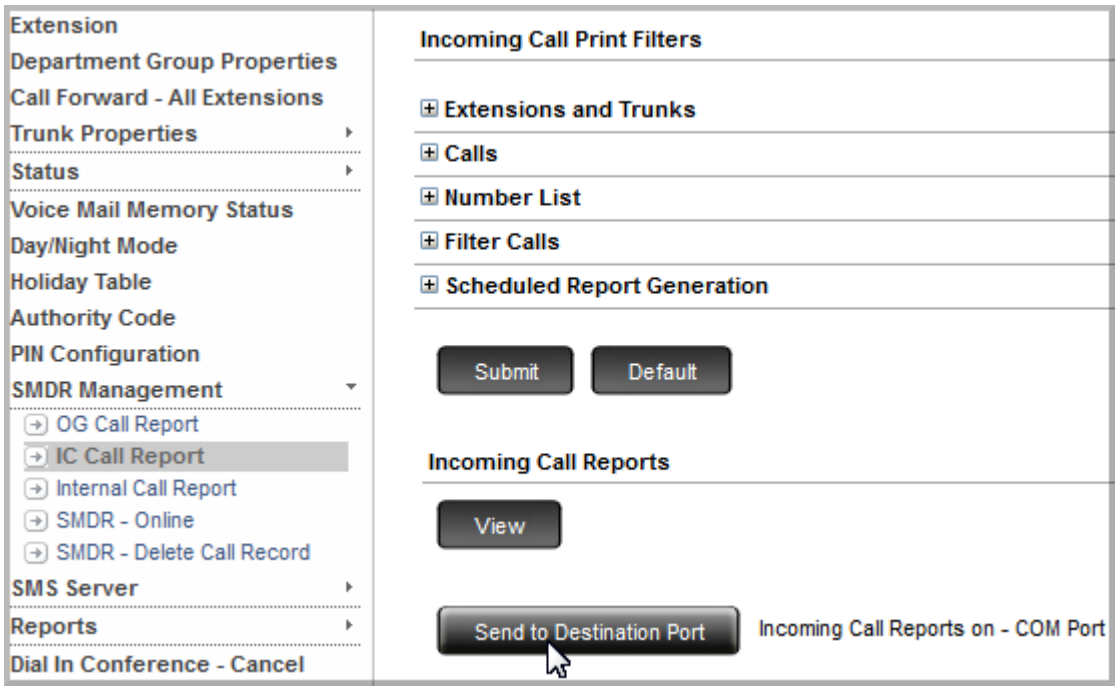
Extension	All	CO	001 To 128	N : Y	Answer Duration	000
Date	01-05-2005 To 31-12-2037	THE1	001 To 024	D : Y	Hold Duration	000
Time	00:00 To 23:59	MOB	001 To 064	U : Y	Speech Duration	000
Number List	02	SIP	001 To 099	DU: Y		
		BRI	001 To 032	I : Y		
		E&M	001 To 064			

Sr. No.	Calling Number	Calling IP:Port	Trunk	Dialed Number	Dialed IP:Port	Connected Number	Connected IP:Port	Date	Time
1	192.168.111.197:5060	192.168.111.197:5060	V001	6001@192.168.111.185:50	192.168.111.185:5060	4001	192.168.111.103:01029	01-12-2020	11:07
2	5004@192.168.111.185:50	192.168.111.185:5060	V001	19@192.168.111.185:5060	192.168.111.185:5060			03-12-2020	22:22
3	192.168.111.192	192.168.111.192	V001	192.168.111.185	192.168.111.185			03-12-2020	22:23
4	192.168.111.167	192.168.111.167	V001	192.168.111.185@192.168	192.168.111.185			06-12-2020	22:54
5	8888@192.168.111.185:50	192.168.111.185:5060	V003	8002@192.168.111.185	192.168.111.185			07-12-2020	00:25
6	8888@192.168.111.185:50	192.168.111.185:5060	V001	8505@192.168.111.185	192.168.111.185			07-12-2020	16:09
7	8888@192.168.111.185:50	192.168.111.185:5060	V001	5555@192.168.111.185	192.168.111.185			07-12-2020	18:14
8	192.168.111.167	192.168.111.167	V001					07-12-2020	18:15
9	192.168.111.167	192.168.111.167	V001	192.168.111.185@192.168	192.168.111.185			07-12-2020	18:18
10	192.168.111.167	192.168.111.167	V001					07-12-2020	18:18
11	8888@192.168.111.185:50	192.168.111.185:5060	V001					07-12-2020	18:25
12	8888@192.168.111.185:50	192.168.111.185:5060	V001					07-12-2020	18:26
13	8888@192.168.111.185:50	192.168.111.185:5060	V001	5322@192.168.111.185	192.168.111.185			08-12-2020	10:18
14	9696@192.168.111.185:50	192.168.111.185:5060	V001	8999@192.168.111.185:50	192.168.111.185:5060			09-12-2020	12:13
15	6565@192.168.111.185:50	192.168.111.185:5060	V001	8999@192.168.111.185:50	192.168.111.185:5060	I812	192.168.111.129:47162	09-12-2020	14:38
16	6565@192.168.111.185:50	192.168.111.185:5060	V001	8999@192.168.111.185:50	192.168.111.185:5060	I812	192.168.111.129:47162	09-12-2020	14:39
17	6565@192.168.111.185:50	192.168.111.185:5060	V001					09-12-2020	14:41
18	6565@192.168.111.185:50	192.168.111.185:5060	V001					09-12-2020	14:42
19	6565@192.168.111.185:50	192.168.111.185:5060	V001					09-12-2020	14:44
20	6565@192.168.111.185:50	192.168.111.185:5060	V001					09-12-2020	14:46
21	8889@192.168.111.185:50	192.168.111.185:5060	V001	8999@192.168.111.185	192.168.111.185			09-12-2020	15:34
22	8889@192.168.111.185:50	192.168.111.185:5060	V001					09-12-2020	15:34
23	8889@192.168.111.185:50	192.168.111.185:5060	V001					09-12-2020	15:35
24	8889@192.168.111.185:50	192.168.111.185:5060	V001					09-12-2020	15:36
25	8889@192.168.111.185:50	192.168.111.185:5060	V001					09-12-2020	15:37
26	8889@192.168.111.185:50	192.168.111.185:5060	V001					09-12-2020	15:42
27	8889@192.168.111.185:50	192.168.111.185:5060	V001					09-12-2020	15:43
28	8889@192.168.111.185:50	192.168.111.185:5060	V001					09-12-2020	15:46
29	8889@192.168.111.185:50	192.168.111.185:5060	V001					09-12-2020	15:59
30	8889@192.168.111.185:50	192.168.111.185:5060	V001					09-12-2020	16:03

- A detailed report is displayed on the screen.
- Click **Print** to print the report.

To print the report on the Destination Port,

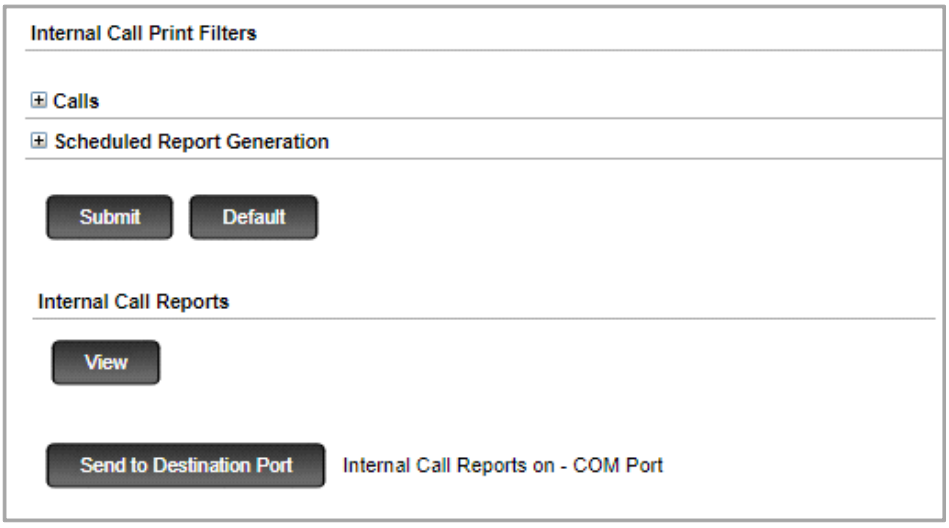
- Click the **Send to Destination Port** button.



- To stop printing, click **Abort** button.

Internal Calls

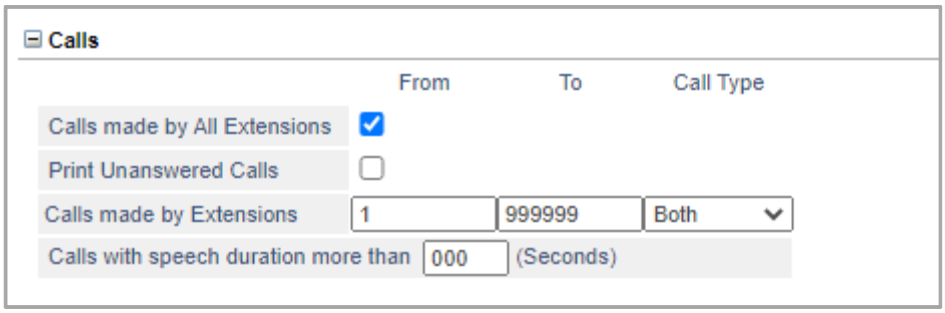
- To print Internal call Report with filters, click **Internal Call Report** link.



Setting Print Filters

To print calls made/received by particular extensions,

- Click **Calls** to expand.



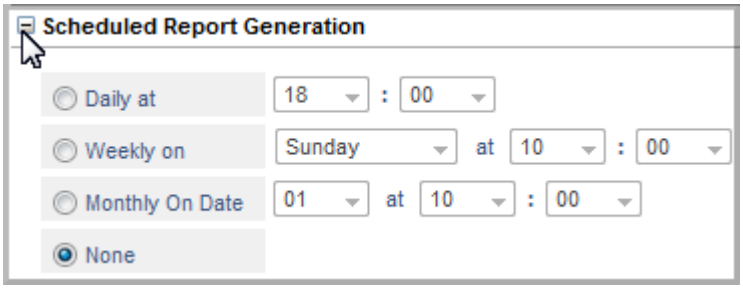
- By default, **Calls made by All Extensions** check box is selected. Hence calls made and calls received by all the extensions will be printed.
- Click **Submit**.
- To print calls made by a particular extension or a range of extensions, clear the **Calls made by All Extension** check box and set the filter **Calls made by Extensions**, by entering the extension numbers in the **From** and **To** fields.

  If you want to print calls made by a particular extension only, enter the same extension number in both **From** and **To** fields.
- Click **Submit**.

  You can also print calls made and calls received by these extensions by selecting the **Call Type**.
  - Select **Both** as Call Type, to print calls made and received by the extensions.
  - Select **Caller** as Call Type, to print only those calls that were made by the extension.
  - Select **Receiver** as Call Type, to print only those calls that were received by the extension.
  - Select **None**, if you do not want to use the Call Type filter.
- To print unanswered/missed calls, select the **Print Unanswered Calls** check box.
- To print calls with speech duration of a certain duration, set the filter **Calls with speech duration more than (Seconds).**
- Click **Submit** to save.

Scheduled Report Generation

- Click **Scheduled Report Generation** to expand.



- To generate SMDR Report of internal calls on a particular day, day of the week, or day of the month, set Scheduled Report Generation, as required.

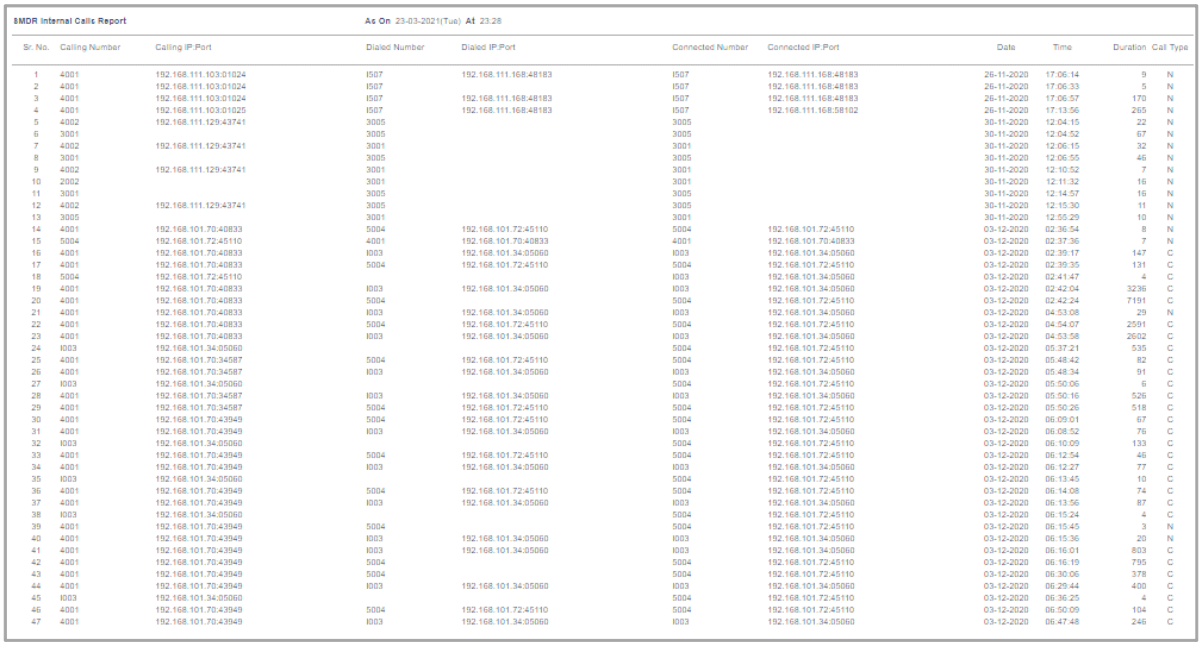
- Click **Submit** to save.

Manual Report Generation

- You can print the SMDR Report of internal calls any time you want. You can print the report on a local printer or the Destination Port of SARVAM UCS.

To view/print the report on the local printer,

- Click the **View** button.



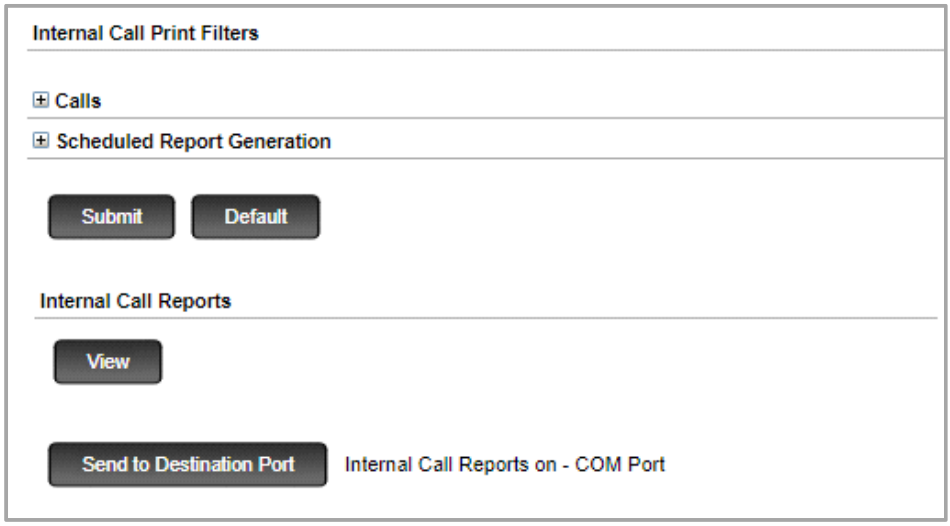
SMDR Internal Calls Report — As On 23-03-2021(Tue) At 23:28

Sr. No.	Calling Number	Calling IP:Port	Dialed Number	Dialed IP:Port	Connected Number	Connected IP:Port	Date	Time	Duration	Call Type
1	4001	192.168.111.103:01024	I507	192.168.111.168:48183	I507	192.168.111.168:48183	26-11-2020	17:06:14	9	N
2	4001	192.168.111.103:01024	I507		I507	192.168.111.168:48183	26-11-2020	17:06:33	5	N
3	4001	192.168.111.103:01024	I507	192.168.111.168:48183	I507	192.168.111.168:48183	26-11-2020	17:06:57	170	N
4	4001	192.168.111.103:01025	I507	192.168.111.168:48183	I507	192.168.111.168:58102	26-11-2020	17:13:56	265	N
5	4002	192.168.111.129:43741	3005		3005		30-11-2020	12:04:15	22	N
6	3001		3005		3005		30-11-2020	12:04:52	67	N
7	4002	192.168.111.129:43741	3001		3001		30-11-2020	12:06:15	32	N
8	3001		3005		3005		30-11-2020	12:06:55	46	N
9	4002	192.168.111.129:43741	3001		3001		30-11-2020	12:10:52	7	N
10	2002		3001		3001		30-11-2020	12:11:32	16	N
11	3001		3005		3005		30-11-2020	12:14:57	16	N
12	4002	192.168.111.129:43741	3005		3005		30-11-2020	12:15:30	11	N
13	3005		3001		3001		30-11-2020	12:55:29	10	N
14	4001	192.168.101.70:40833	5004	192.168.101.72:45110	5004	192.168.101.72:45110	03-12-2020	02:36:54	8	N
15	5004	192.168.101.72:45110	4001	192.168.101.70:40833	4001	192.168.101.70:40833	03-12-2020	02:37:36	7	N
16	4001	192.168.101.70:40833	I003	192.168.101.34:05060	I003	192.168.101.34:05060	03-12-2020	02:39:17	147	C
17	4001	192.168.101.70:40833	5004	192.168.101.72:45110	5004	192.168.101.72:45110	03-12-2020	02:39:35	131	C
18	5004	192.168.101.72:45110			I003	192.168.101.34:05060	03-12-2020	02:41:47	4	C
19	4001	192.168.101.70:40833	I003	192.168.101.34:05060	I003	192.168.101.34:05060	03-12-2020	02:42:04	3236	C
20	4001	192.168.101.70:40833	5004		5004	192.168.101.72:45110	03-12-2020	02:42:24	7191	C
21	4001	192.168.101.70:40833	I003	192.168.101.34:05060	I003	192.168.101.34:05060	03-12-2020	04:53:08	29	N
22	4001	192.168.101.70:40833	5004	192.168.101.72:45110	5004	192.168.101.72:45110	03-12-2020	04:54:07	2591	C
23	4001	192.168.101.70:40833	I003	192.168.101.34:05060	I003	192.168.101.34:05060	03-12-2020	04:53:58	2602	C
24	I003	192.168.101.34:05060			5004	192.168.101.72:45110	03-12-2020	05:37:21	535	C
25	4001	192.168.101.70:34587	5004	192.168.101.72:45110	5004	192.168.101.72:45110	03-12-2020	05:48:42	82	C
26	4001	192.168.101.70:34587	I003	192.168.101.34:05060	I003	192.168.101.34:05060	03-12-2020	05:48:34	91	C
27	I003	192.168.101.34:05060			5004	192.168.101.72:45110	03-12-2020	05:50:06	6	C
28	4001	192.168.101.70:34587	I003	192.168.101.34:05060	I003	192.168.101.34:05060	03-12-2020	05:50:16	526	C
29	4001	192.168.101.70:34587	5004	192.168.101.72:45110	5004	192.168.101.72:45110	03-12-2020	05:50:26	518	C
30	4001	192.168.101.70:43949	5004	192.168.101.72:45110	5004	192.168.101.72:45110	03-12-2020	06:09:01	67	C
31	4001	192.168.101.70:43949	I003	192.168.101.34:05060	I003	192.168.101.34:05060	03-12-2020	06:08:52	76	C
32	I003	192.168.101.34:05060			5004	192.168.101.72:45110	03-12-2020	06:10:09	133	C
33	4001	192.168.101.70:43949	5004	192.168.101.72:45110	5004	192.168.101.72:45110	03-12-2020	06:12:54	46	C
34	4001	192.168.101.70:43949	I003	192.168.101.34:05060	I003	192.168.101.34:05060	03-12-2020	06:12:27	77	C
35	I003	192.168.101.34:05060			5004	192.168.101.72:45110	03-12-2020	06:13:45	10	C
36	4001	192.168.101.70:43949	5004	192.168.101.72:45110	5004	192.168.101.72:45110	03-12-2020	06:14:08	74	C
37	4001	192.168.101.70:43949	I003	192.168.101.34:05060	I003	192.168.101.34:05060	03-12-2020	06:13:56	87	C
38	I003	192.168.101.34:05060			5004	192.168.101.72:45110	03-12-2020	06:15:24	4	C
39	4001	192.168.101.70:43949	5004		5004	192.168.101.72:45110	03-12-2020	06:15:45	3	N
40	4001	192.168.101.70:43949	I003	192.168.101.34:05060	I003	192.168.101.34:05060	03-12-2020	06:15:36	20	N
41	4001	192.168.101.70:43949	I003	192.168.101.34:05060	I003	192.168.101.34:05060	03-12-2020	06:16:01	803	C
42	4001	192.168.101.70:43949	5004		5004	192.168.101.72:45110	03-12-2020	06:16:19	795	C
43	4001	192.168.101.70:43949	5004		5004	192.168.101.72:45110	03-12-2020	06:30:06	378	C
44	4001	192.168.101.70:43949	I003	192.168.101.34:05060	I003	192.168.101.34:05060	03-12-2020	06:29:44	400	C
45	I003	192.168.101.34:05060			5004	192.168.101.72:45110	03-12-2020	06:36:25	4	C
46	4001	192.168.101.70:43949	5004	192.168.101.72:45110	5004	192.168.101.72:45110	03-12-2020	06:50:09	104	C
47	4001	192.168.101.70:43949	I003	192.168.101.34:05060	I003	192.168.101.34:05060	03-12-2020	06:47:48	246	C

- A detailed report is displayed on the screen.
- Click **Print** to print the report.

To print the report on the Destination Port,

- Click the **Send to Destination Port** button.



- To stop printing, click **Abort** button.

Print SMDR-Report using a Telephone

Print Filter Settings - Incoming Calls

The SA can program the filters as per user's need to generate a customized report. These commands enable the user to select the type of call reports generated viz. Normal calls, Built-In Auto Attendant calls, Unanswered calls, Long Speech Duration calls. It is also possible to generate call reports for a range of extensions (SLT, DKP), different type of trunks (CO, BRI, T1E1PRI etc.).

Filter	Commands	Values
To set filter to print all normal calls	***1072-152-Flag***	Flag: 0 = Disable 1 = Enable
To set filter to print all Built-In Auto Attendant calls	***1072-153-Flag***	Flag: 0 = Disable 1 = Enable
To set filter to print all Unanswered calls	***1072-154-Flag***	Flag: 0 = Disable 1 = Enable
To set filter to print all Built-In Auto Attendant Unanswered calls	***1072-155-Flag***	Flag: 0 = Disable 1 = Enable
To set filter to print all DISA calls	***1072-156-Flag***	Flag: 0 = Disable 1 = Enable
To set filter to print all calls with specific call duration	***1072-157-Seconds***	Seconds:000-255
To set filter to print all call with specific Unanswered duration	***1072-158-Seconds***	Seconds:000-255
To set filter to print all calls with specific hold duration	***1072-159-Seconds***	Seconds:000-255
To set filter to print all IC calls received by the extension	***1072-160-Flexible No.-Flexible No.***	Flexible Number: 000000 to 999999
To set filter to print all IC calls received on the CO	***1072-161-CO-CO***	CO: 000-128
To set filter to print all IC calls received on the BRI	***1072-162-BRI-BRI***	BRI: 00-32
To set filter to print all IC calls received on the T1E1PRI	***1072-163-T1E1PRI-T1E1PRI***	T1E1PRI: 01-08
To set filter to print all IC calls received on the E&M	***1072-164-E&M-E&M***	E&M: 000-128

Filter	Commands	Values
To set filter to print all IC calls received on Mobile	***1072-165-Mobile-Mobile***	Mobile: 00-64
To set filter to print all IC calls received on SIP	***1072-166-SIP-SIP***	SIP: 00-32
To set filter to print all IC calls on or from date	***1072-167-DD-MM-YYYY-DD-MM-YYYY***	Date: 01-31 Month: 01-12 Year: 0000-9999
To set filter to print all IC calls received At/from-to Time	***1072-168-HH-MM-HH-MM***	HH: 00-23 MM: 00-59
To set filter to print all IC calls from numbers matching in a Number list	***1072-169-Number List***	Number List: 00-16

Note: If you do not want to print calls for a particular type, set the start and end range as 0.

Default the Filters

To default the report generation filters, dial:

- ***1072-170***
  Calls made by all extensions, all CO, All BRI, All E&M and All T1E1PRI are printed.
  The Date range is 01-05-2005 to 31-12-2100.
  The Time range is 00:00 to 23:59.
  The Number List is 02.
  By default, All calls are printed as per the set filters.

Manual Report Generation

SA should issue an explicit start command, to initiate the report generation.

To start/abort report generation, dial:

- ***1072-171-Flag***

Where,

Flag	Meaning
0	Abort
1	Start

By default, the Flag is 0.

Once the Start command is issued, the report generation stops only after the complete report based on the filters set is generated. SARVAM UCS provides a facility to abort the report generation in midway (***1072-171-0***). Once the report generation is aborted, then it has to be explicitly started (***1072-171-1***) if the report is required again.

The Scheduled Report Generation

SA can issue the command to start report, automatically at a particular time (Daily) or at a particular time on a particular day of the week (Weekly) or at a particular time on a particular date of a month (Monthly).
Once these parameters are programmed and the report generation is enabled, there is no need of any initiation of the command.

Enable/disable Scheduled Report

To enable/disable the generation of daily/weekly/monthly reports, dial:

- ***1072-172-Flag***

  Where,

Flag	Meaning
0	Disable Schedule Report Generation
1	Enable Daily Scheduled Report
2	Enable Weekly Scheduled Report
3	Enable Monthly Scheduled Report

By default, the generation report is disabled.

Daily Reports

To program the time for daily scheduled reports, dial:

- ***1072-173-HH-MM***

  Where,
  HH and MM is in 24 hour format.
  By default, HH:MM is 18:00.

Weekly Reports

To program the day and time for weekly scheduled reports, dial:

- ***1072-174-Day-HH-MM***

  Where,
  HH and MM is in 24 hour format.
  Day is from 1 to 7 (1 is Sunday, 2 is Monday …… and 7 is Saturday).
  By default, HH:MM is 10:00 and Day is 2.

Monthly Reports

To program the date and time for monthly scheduled reports, dial:

- ***1072-175-Date-HH-MM***

  Where,
  HH and MM is in 24 hour format.
  Date is from 01 to 31.
  By default, HH:MM is 10:00 and Date is 01.

SARVAM UCS provides a facility to Abort the scheduled report generation midway (1072-171-0). This aborts the current report generation but does not affect any other scheduled report generation. The abortion of a report does not affect its consecutive schedule. That is if a daily report is aborted on Monday, it does affect the report generation schedule of Tuesday.

Print Filter Settings - Internal Calls

Filter	Command	Value
To set filter to print all internal calls from an extension.	***1072-137-Flexible No.-Flexible No.-Type***	Flexible Numbers: 000000-999999 Type: 0 - Do not print the calls made to or from the extension. 1 - Print calls made to this extension 2 - Print calls made from this extension 3 - Print all calls made to/from this extension By default, Type is 3 and seconds is 000.
To set filter to print all internal calls with duration greater than specified.	***1072-138-Seconds***	Seconds: 000-999

Manual Report Generation

SA should issue an explicit start command to initiate the report generation.

To start/abort Report generation, dial:

- 1072-141-Flag
  Where,

Flag	Meaning
0	Abort
1	Start

By default, flag is 0.

Once the start command is issued, the report generation stops only after the complete report based on the filters set is generated. SARVAM UCS provides a facility to abort the report generation in midway (***1072-141-0***). Once the report generation is aborted, then it has to be explicitly started (***1072-141-1***) if the report is required again.

Scheduled Report Generation

SA should issue the command to start the report, automatically at a particular time (Daily) or at a particular time on a particular day of the week (Weekly) or at a particular time on a particular date of a month (Monthly).

Once these parameters are programmed and the report generation is enabled, there is no need of any initiation command.

Enable/disable Scheduled Report

- To enable/disable the generation of daily/weekly/monthly scheduled reports, dial:
  1072-142-Flag

Where,

Flag	Meaning
0	Disable Schedule Report Generation
1	Enable Daily Scheduled Report
2	Enable Weekly Scheduled Report
3	Enable Monthly Scheduled Report

By default, the generation report is disabled.

Daily Reports

To program the time for daily scheduled reports, dial:

- ***1072-143-HH-MM***
  Where,
  HH and MM is in 24 hour format.
  By default, HH:MM is 18:00

Weekly Reports:

To program the day and time for weekly scheduled reports, dial:

- ***1072-144-Day-HH-MM***
  Where,
  HH and MM is in 24 hour format.
  Day is from 1 to 7 (1 is Sunday, 2 is Monday …… and 7 is Saturday).
  By default, HH:MM is 10:00 and Day is 2.

Monthly Reports:

To program the date and time for monthly scheduled reports, dial:

- ***1072-145-Date-HH-MM***
  Where,
  HH and MM is in 24 hour format.
  Date is from 01 to 31.
  By Default, HH:MM is 10:00 and Date is 01.

SARVAM UCS provides a facility to abort the scheduled report generation in midway (1072-141-0). This aborts the current report generation but does not affect any other scheduled report generation. The abortion of a report does not affect its consecutive schedule. That is if a daily report is aborted on Monday, it does affect the report generation schedule of Tuesday.

Print Filter Settings - Outgoing Calls

SA can configure various filters to generate a report as per the user's requirement. It is possible to program the following filters:

Filter	Commands	Values
To set filter to print all calls terminated on CO	***1072-103-CO-CO***	CO: 001-128
To set filter to print all calls terminated on BRI	***1072-104-BRI-BRI***	BRI: 01-32

Filter	Commands	Values
To set filter to print all calls terminated on T1E1PRI	***1072-105- T1E1PRI - T1E1PRI***	T1E1PRI : 01-08
To set filter to print all calls terminated on E&M	***1072-106-E&M-E&M***	E&M: 001-128
To set filter to print all calls terminated on Mobile	***1072-107-Mobile-Mobile***	Mobile: 00-64
To set filter to print outgoing calls terminated on SIP	***1072-108-SIP-SIP***	SIP: 00-32
To set filter to print outgoing calls for department billing group	***1072-109-Group No.-Group No.***	Dept Group: 00 to 99
To set filter to print all calls made on or from date	***1072-110-DD-MM-YYYY-DD-MM-YYYY***	Date: 01-31. Month: 01-12. Year: 0000-9999.
To set filter to print all calls made at or in between a time	***1072-111-HH-MM-HH-MM***	HH: 00-23 MM: 00-59
To set filter to print all calls of numbers matching in a Number list	***1072-112-External Number List***	Number List = 01-16.
To set filter to print all calls duration more than specified	***1072-113-Seconds***	Seconds = 000-999.
To set filter to print all calls with units more than specified	***1072-114-Units***	Unit = 0000-9999
To set filter to print all calls made using Account Code	***1072-115-Account Code-Account Code***	Account Code = 0000-9999
To set filter to print all calls originated on extensions	***1072-102-Flexible No.-Flexible No.***	Flexible Numbers: 000000 to 999999
To set filter to print calls originated on CO	***1072 - 183 - CO - CO***	CO = 000-128
To set filter to print calls originated on BRI	***1072 - 184- BRI- BRI***	BRI = 00-32
To set filter to print calls originated on T1E1PRI	***1072 - 185 - T1E1PRI - T1E1PRI***	T1E1PRI = 01-08
To set filter to print calls originated on E&M	***1072 - 186 - E&M- E&M***	E&M = 000-128
To set filter to print calls originated on Mobile	***1072 - 187 - Mobile- Mobile***	Mobile = 00-64
To set filter to print calls originated on SIP	***1072 - 188 - SIP- SIP***	SIP = 00-32
To change the CPU from Active to Standby mode	***1072-189***	

To default the report generation filters, dial:

- ***1072-120***
  By default, All calls are printed as per the set filters.
  Calls made by all extensions, all CO, ALL BRI, ALL E&M and All T1E1PRI are printed.
  The Date range = 01-05-2005 to 31-12-2100.
  The Time range = 00:00-23:59.
  The External Number list = 02.
  Units = 0000.
  Seconds = 000.

Delete calls made to/from an extension

The SARVAM UCS supports Deletion of SMDR Calls (From SA Mode) made from a particular extension or a range of extensions.

To delete the calls made by an extension or a range of extensions, dial:

- ***1072-131-Flexible Number-Flexible Number***

To delete the calls made on a Date or from a Date, dial:

- ***1072-132-DD-MM-YYYY-DD-MM-YYYY*** (The format of the date depends on the date format of the system)

Manual Report Generation

SA should issue an explicit start command to initiate the report generation.

To start/abort Report generation, dial:

- ***1072-121-Flag***
  Where,

Flag	Meaning
0	Abort
1	Start

By default, Flag is 0.
Once the start command is issued, the report generation stops only after the complete report based on the filters set is generated. SARVAM UCS provides a facility to abort the report generation in midway (***1072-121-0***). Once the report generation is aborted, then it has to be explicitly started (***1072-121-1***) if the report is required again.

Scheduled Report Generation

SA should issue the command to start the report, automatically at a particular time (Daily) or at a particular time on a particular day of the week (Weekly) or at a particular time on a particular date of a month (Monthly).
Once these parameters are programmed and the report generation is enabled, there is no need of any initiation command.

Enable/disable Scheduled Report

To enable/disable the generation of daily/weekly/monthly scheduled reports, dial:

- ***1072-122-Flag***

Where,

Flag	Meaning
0	Disable Schedule Report Generation
1	Enable Daily Scheduled Report
2	Enable Weekly Scheduled Report
3	Enable Monthly Scheduled Report

By default, the generation report is disabled.

Daily Reports

To program the time for daily scheduled reports, dial:

- ***1072-123-HH-MM***
  Where,
  HH and MM is in 24 hour format.
  By default, HH:MM is 18:00.

Weekly Reports

To program the day and time for weekly scheduled reports, dial:

- ***1072-124-Day-HH-MM***
  Where,
  HH and MM is in 24 hour format.
  Day is from 1 to 7 (1 is Sunday, 2 is Monday …… and 7 is Saturday).
  By default, HH:MM is 10:00 and Day is 2.

Monthly Reports

To program the date and time for monthly scheduled reports, dial:

- ***1072-125-Date-HH-MM***
  Where,
  HH and MM is in 24 hour format.
  Date is from 01 to 31.
  By Default, HH:MM is 10:00 and Date is 01.

*SARVAM UCS provides a facility to abort the scheduled report generation in midway (**1072-121-0**). This aborts the current report generation but does not affect any other scheduled report generation. The abortion of a report does not affect its consecutive schedule. That is if a daily report is aborted on Monday, it does affect the report generation schedule of Tuesday.*

The Offline report for calls looks like shown below:

```
       SMDR OUTGOING CALLS REPORT          AS ON 04-02-2021(Thu) AT 12:20
-------------------------------------------------------------------------------
                                        Originated on     Terminated on
 Extension  : ALL                      CO   :001 To 064  CO   :001 To 064
 Date       : 01-05-2005 To 31-12-2037 BRI  :001 To 032  BRI  :001 To 032
 Time       : 00:00 To 23:59           T1E1 :001 To 008  T1E1 :001 To 008
 Number List: 02                       E&M  :001 To 032  E&M  :001 To 032
 Department : 00 To 00                 MOB  :001 To 040  MOB  :001 To 040
 Dur - Sec  : 000                      SIP  :001 To 099  SIP  :001 To 099
 Units      : 0000
 Accout No  : 000 To 000
 Authrty Cod: 000 To 000
-------------------------------------------------------------------------------
 SrNo Caller AuC Trnk     ConnNo          Date     Time    Dur  Unit Amount  R
-------------------------------------------------------------------------------
   1   I007 000 V002                   12-01-21 11:34:37     3    0    0.00  I
   2   I006 000 V002                   12-01-21 11:34:46     4    0    0.00  I
   3   I006 000 V001                   12-01-21 11:40:51     2    1    1.10  I
   4   I007 000 V001                   12-01-21 11:40:53     8    1    1.10 TI
   5   I007 000 V002                   12-01-21 11:41:21     4    1    1.10 TI
   6   I007 000 V002                   12-01-21 11:41:46     3    1    1.10 TI
   7   I006 000 V002                   12-01-21 11:42:10    21    0    0.00  I
   8   I006 000 V002                   12-01-21 11:47:24   273    0    0.00  I
   9   I006 000 V002                   12-01-21 11:52:17    92    0    0.00  I
  10   I006 000 V002                   12-01-21 11:58:47   120    0    0.00  I
  11   5004 000 V001                   12-01-21 12:28:04    13    1    1.10  I
  12   5004 000 V001                   12-01-21 12:47:59    89    1    1.10  I
  13   5004 000 V001                   12-01-21 12:49:38    18    1    1.10  I
  14   5004 000 V002                   12-01-21 12:50:05    72    1    1.10  I

-------------------------------------------------------------------------------
 SARVAM UCS    V1R6                                              Page : 1
```

```
       SMDR INCOMING CALLS REPORT          AS ON 04-02-2021(Thu) AT 16:39
--------------------------------------------------------------------------------
 Extension  : ALL                  CO  :001 To 064   N : Y   Answer Dur:000
 Date       : 01-05-05 To 31-12-37 BRI :001 To 032   D : Y   Hold Dur  :000
 Time       : 00:00 To 23:59       T1E1:001 To 008   U : Y   Speech Dur:000
 Number List: 02                   E&M :001 To 032   DU: Y
                                   MOB :001 To 040   I : Y
                                   SIP :001 To 099
--------------------------------------------------------------------------------
 SrNo    Calling No    Trnk  ConnNo   Date       Time    Ans Hld Spch  R
                                                        Dur Dur  Dur
--------------------------------------------------------------------------------
    1                  P002    2001 04-02-21   12:09:57   6   0    86 N
    2                  P002    2001 04-02-21   12:24:18   5   0    26 N
    3                  P002    2001 04-02-21   12:26:25   5   0    20 N
    4                  P002    2001 04-02-21   12:31:10   2   0     5 N
    5                  P002    2001 04-02-21   14:40:57   6   0    14 N
    6                  P002         04-02-21   14:42:08   2   0     0 U
    7                  P002    2001 04-02-21   14:51:04   5   0    10 N
--------------------------------------------------------------------------------
 TOTAL CALLS   : 7
 TOTAL ANS DUR : 31       TOTAL HOLD DUR : 0        TOTAL SPCH DUR : 161
--------------------------------------------------------------------------------
Trunk    : C=CO, B=BRI, P=T1E1, E=E&M, M=MOB, V=SIP
Extension: S=SLT, D=DKP, G=MAG, I=SIP Extn, R=Virtual Extn, N=ISDN Extn
Additional Ports: X=MODEM,
R(CALL TYPE): N=Normal, U=UnAnswered, I=DISA, G=Gateway, Q=Qsig, T=Transfer
              D=Built-In Auto Attendant, DU=Built-In Auto Attendant UnAnswered
              C=Conference, F=Forward
--------------------------------------------------------------------------------
 SARVAM UCS    V1R6                                              Page : 1
```

SMDR Internal Calls Report

As On 08-03-2023(Wed)
At 09:49

Sr. No.	Calling Number	Calling IP:Port	Dialed Number	Dialed IP:Port	Connected Number	Connected IP:Port	DateTimeDuration	Call Type
1	3003		3004		3004		05-03-202313:19:35	8N
2	3004		3003		3003		05-03-202313:41:21	6N
3	3004		121		4004	191.168.10.158:59583	05-03-202316:15:13	45N
4	3003		121		4006	191.168.10.156:55592	05-03-202316:15:06	41N
5	3004		121		4004	191.168.10.158:59583	05-03-202316:16:44	35N
6	3003		121		4004	191.168.10.158:59583	05-03-202316:17:32	30N
7	4004	191.168.10.158:59583	4007	191.168.10.152:56600	4007	191.168.10.152:56600	05-03-202316:18:17	221N
8	3003		121		4006	191.168.10.156:55592	05-03-202316:18:23	334N
9	3003		4007	191.168.10.152:56600	4007	191.168.10.152:56600	05-03-202316:24:15	3N
10	3004				4006	191.168.10.156:55592	05-03-202316:17:22	1279N

CALL TYPE : N=Normal, C=Conference, F=Forward, U=Unanswered

Station Message Detail Recording-Storage

What's this?

SARVAM UCS stores SMDR of Incoming calls, Outgoing Calls and Internal Calls. The call records are stored in the SMDR buffer. For this, SMDR storage for these types of calls must be enable, and if required further filters can be set.

SARVAM UCS can store 6000 outgoing calls, 4999[338] incoming calls, and 999[339] internal calls in the SMDR buffer. Once the SMDR buffer is full, the next call is stored in place of the oldest call in the SMDR buffer, using the First In First Out (FIFO) logic.

The buffer can be cleared at any time from the System Administrator mode.

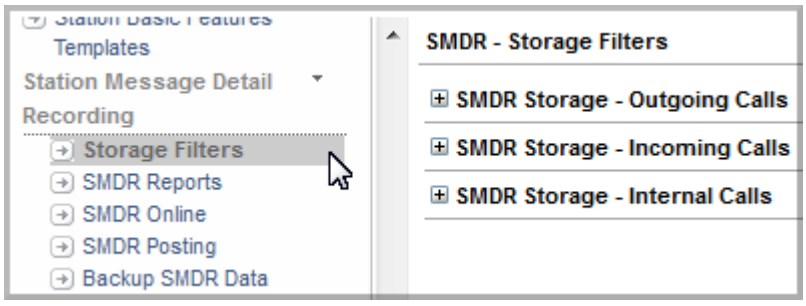
The SMDR buffer data is maintained even during power failures. However it is advisable to take frequent printouts of the calls to avoid accidental loss of the data.

How to configure

To enable storage of SMDR of Outgoing, Incoming and Internal Calls, you must enable this feature in the system and set the storage filters as per your requirement.

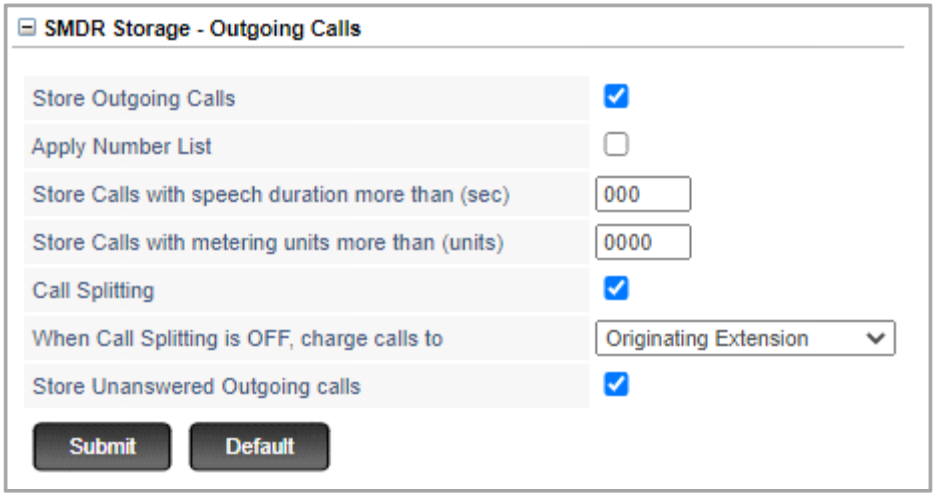
Configuring SMDR-Storage using Jeeves

- Login as System Engineer.
- Under **Configuration**, click **Station Message Detail Recording**.
- Click **Storage Filters** to open the page.



338. *ETERNITY PENX can store 5000 Incoming Calls and 1000 Internal Calls.*
339. *ETERNITY LENX/MENX can store 9999 Outgoing Calls, 9999 Incoming Calls and 2000 Internal Calls.*

- To configure storage filters for outgoing calls, click **SMDR Storage - Outgoing Calls** to expand,



  - Select the **Store Outgoing Calls** check box to enable storage of outgoing calls as per the filters you set. If outgoing call storage is disabled, no outgoing call will be stored.

  - Select the **Apply Number List** check box and then you can configure **Store Calls of Called Number matching with Number List**.

    You can limit the storage of calls to certain numbers. Select a number list and enter the desired called party numbers in the selected list. In **Store Calls of Called Number matching with Number List**, select the same list number.

  - If you want outgoing calls that exceed as certain duration to be stored, set the filter **Store Calls with speech duration more than (sec)** to the desired duration. All outgoing calls with duration greater than this value, will be stored.

  - If you want outgoing calls exceeding certain metering units to be stored, set the filter **Store Calls with metering units more than (units)** to the desired value. All outgoing calls that have metering units greater than this value will be stored.

  - Outgoing calls made by an extension user can be transferred to another extension. In such cases, you may enable **Call Splitting** if you want to charge the amount to each extension according to the duration of speech that each extension was involved in the call.

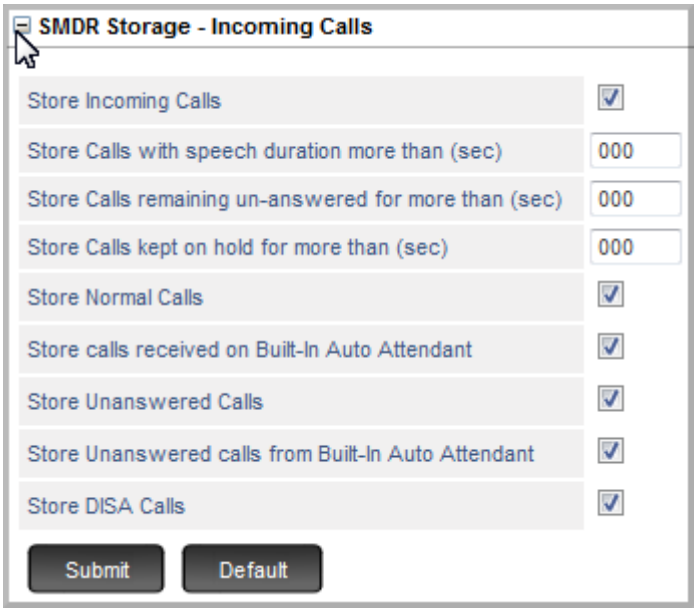
  - If Call Splitting is disabled, you have the option of charging the call amount either to the extension that originally made the call, i.e. the Originating Extension, or to the extension that was last in speech on the call, i.e. the Terminating Extension.

    In the **When Call Splitting is OFF, charge calls to** field, select the desired extension you want to charge the call to as **Originating Extension** or **Terminating Extension**.

  - Select the **Store Unanswered Outgoing calls** check box to enable storage of unanswered outgoing calls as per the filters you set. If this check box is disabled, unanswered outgoing calls will not be stored.

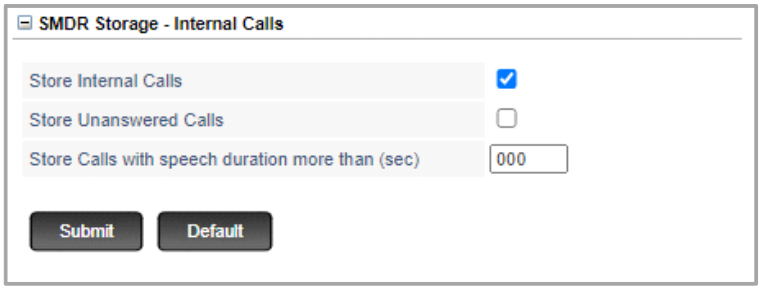
  - Click **Submit** to save your Outgoing call storage filter settings.

- To configure storage filters for incoming calls, click **SMDR Storage - Incoming Calls** to expand,



- Select the **Store Incoming Calls** check box to enable storage of incoming calls as per the filters you set. If incoming call storage is disabled, no incoming call will be stored.
- If you want incoming calls that exceed as certain duration to be stored, set the filter **Store Calls with speech duration more than (sec)** to the desired duration. All calls with duration greater than this value, will be stored.
- If you want incoming calls that remain unanswered for certain duration to be stored, set the filter **Store Calls remaining un-answered for more than (sec)** to the desired duration. All calls with duration greater than this value, will be stored.
- If you want incoming calls that were kept on hold for a certain duration to be stored, set the filter **Store Calls kept on hold for more than (sec)** to the desired duration.
- If you want all calls, except calls received using Auto Attendant to be stored, select the **Store Normal Calls** check box.
- If you want calls received on Auto Attendant to be stored, select the **Store calls received on Built-In Auto Attendant** check box.
- If you want all calls, that remained unanswered to be stored, select the **Store Unanswered Calls** check box.
- If you want all calls received using Auto Attendant that remained unanswered to be stored, select the **Store Unanswered calls from Built-In Auto Attendant** check box.
- If you want calls made using DISA, select the **Store DISA Calls** check box.
- Click **Submit** to save Incoming Call storage filters.

- To configure storage filters for internal calls, click **SMDR Storage - Internal Calls** to expand,



  - Select the **Store Internal Calls** check box to enable storage of internal calls as per the filters you set.
  - To store unanswered/missed internal call, select the **Store Unanswered Calls** check box.
  - To store internal calls that exceed as certain duration, set the filter **Store Calls with speech duration more than (sec)** to the desired duration. All internal calls with duration greater than this value, will be stored.
  - Click **Submit** to save.

Configuring SMDR-Storage using a Telephone

SMDR Storage-Incoming Calls

Enable SMDR-IC Storage

To set SMDR storage mode, dial:

- ***2901-Storage Flag***

  Where,

Storage Flag	Meaning
0	Don't store the incoming calls
1	Store the incoming calls

By default, all the calls are stored as per the filters set.

Filter Settings - Incoming Call Storage

These commands enable the user to select the type of calls to be stored, namely, All calls, Trunk wise calls, Un-answered calls, Built-In Auto Attendant calls, etc. Calls will be stored only if duration, exceeds the set value.

Filter	Commands	Values
Store Normal Calls	***2902-Flag***	Flag: 0 = Don't Store 1 = Store
Store Built-In Auto Attendant Calls	***2903-Flag***	Flag: 0 = Don’t Store 1 = Store

Store Unanswered Calls	***2904-Flag***	Flag: 0 = Don't Store 1 = Store
Store Unanswered Built-In Auto Attendant Calls	***2905-Flag***	Flag: 0 = Don't Store 1 = Store
Store DISA Calls	***2906-Flag***	Flag: 0 = Don't Store 1 = Store
Store Calls-Speech Duration More than	***2907-Seconds***	Seconds:000-999
Store Calls-Unanswered Duration for more than	***2908-Seconds***	Seconds:000-999
Store Calls-Hold Duration for more than	***2909-Seconds***	Seconds:000-999

Default IC Storage Filters

To default the incoming call storage filters, dial:

- ***2915***
  By default,
  All Normal, Built-In Auto Attendant, Unanswered, Unanswered Built-In Auto Attendant, DISA calls will be stored.
  The speech duration is set to 000.
  The unanswered duration is set to 000.
  The Hold Duration is set to 000.

SMDR Storage-Internal Calls

Enable SMDR-Internal Storage:

To set SMDR storage mode, dial:

- ***2801-Storage Flag***
  Where,

Storage Flag	Meaning
0	Don't store the internal calls
1	Store the internal calls

By default, all the calls are stored as per the filters set.

Filter Settings - Internal Call Storage

Filter	Command	Value
Store Calls-Speech duration	***2802-Seconds***	Seconds: 000-999

Default Internal Storage Filters

To default the internal call storage filters, dial:

- ***2815***

SMDR Storage-Outgoing Calls

Enable SMDR-Outgoing Storage

To set SMDR storage flag, dial:

- ***2701-Storage Flag***
  Where,

Storage Flag	Meaning
0	Don't store the outgoing calls
1	Store the outgoing calls

By default, the Storage Flag is 1.

Filter Settings - Outgoing Call Storage

These commands enable the user to select the type of calls to be stored, namely, destination number wise, duration wise or cost wise.

Destination Wise

It is possible to store outgoing calls selectively depending on the destination numbers. The SARVAM UCS supports this feature in association with Number Lists. An outgoing call will be stored only if the number matches with an entry in the Number List assigned.

To assign a Number list containing numbers for call storage, dial:

- ***2702-Number List***
  Where,
  Number List is 01-16.
  By default, Number List assigned is 02.

Duration wise

Sometimes it is required to filter out the calls of small durations. System will not store the calls with duration less than duration, programmed.

To set the filter of call duration, dial:

- ***2703-Seconds***
  Where,
  Seconds is from 000 to 999.
  By default, Seconds is 000.

Unit wise

Sometimes it is required to filter out the calls based on Call units. System will not store the calls with units less than the units programmed:

To set the filter for call Units, dial:

- ***2704-Units***
  Where,
  Units are from 0000 to 9999.
  By default, Units is 0000.

Default OG Storage Filters

To default the outgoing calls storage filters, dial:

- ***2715***
  By default, Number List is 02, Seconds is 000 and Units is 000.

Call Toggle

To set the Call Toggle flag, dial:

- ***2716-Toggle Flag***
  Where,

Toggle Flag	Meaning
0	Toggle OFF
1	Toggle ON

By default, Toggle is ON.

To set the originating flag, dial:

- ***2717-Originating Flag***
  Where,

Originating Flag	Meaning
0	OFF
1	ON

By default, the flag is originating extension.

When Call Toggle Flag is set OFF, then only above command (2717) can be effective. Refer chapter "Call Toggle" for more details.

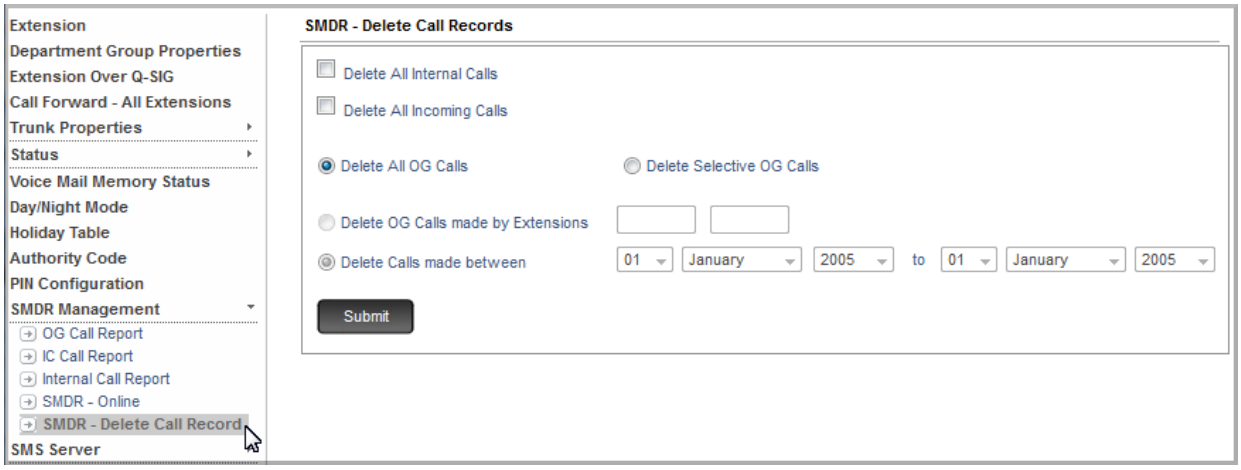
How to use

The SMDR stored in the buffer can be cleared at any time from the System Administrator mode, using Jeeves or by dialing SA commands from an extension phone.

To delete SMDR records using Jeeves,

- Log in as System Administrator.

- Under **SMDR Management**, click **SMDR - Delete Call Record** to open the page.



- To delete all records in the internal SMDR buffer, select **Delete All Internal Calls** check box.
- To delete all records in the Incoming SMDR buffer, select the **Delete All Incoming Calls** check box.
- If you want to delete all records in the Outgoing SMDR buffer, select the **Delete All OG Calls** radio button.
- You can also delete records of outgoing calls selectively, i.e. delete only records of outgoing calls made by a particular extension or a range of extensions, or calls made between a certain period.
    - To delete records of outgoing calls selectively, select the **Delete Selective OG Calls** radio button.
    - To delete calls made by a particular extension or a range of extensions, select the **Delete OG Calls made by Stations** button.
        - In the first edit box, enter the number of the first extension in the range. In the second box, enter the number of the last extension in the range. If you want to delete the records of a particular extension, enter the same extension number in both fields.
    - To delete the records of outgoing calls made on a particular date or during a certain period, select the **Delete calls made between** radio button. Select the start and end Date, Month and Year for this period. If you want to delete the records of a particular date, enter the same date as start and end.
- Click **Submit** to save.
- The SMDR buffer will be cleared according to the settings you enabled on this page.
- You may log out of Jeeves.

To delete SMDR records from an extension phone,

- Enter SA mode from a DKP/SLT/Extended IP Phone.

To delete all Incoming calls:

- Dial **1072-180-Reverse SA Password**

To delete all Internal calls:

- Dial **1072-150-Reverse SA Password**

To delete all Outgoing calls:

- Dial **1072-133-Reverse SA Password**

To delete calls made by a particular extension or range of extensions:

- Dial **1072-131-Extension Number-Extension Number**

To delete calls made by on a particular date or a between a certain time period:

- **1072-132-DD-MM-YYYY-DD-MM-YYYY** (The format of the date depends on the Date Format of the system)

- Exit SA mode.

System Activity Log

What's this?

The SARVAM UCS monitors all its activities and maintains records of these activities in the System Activity Log.

The System Activity Log has a buffer capacity of 500 records. The Activity Log stores records using the FIFO method.

This log can be printed on a local printer or downloaded on a computer in form of a report. The activity log can be printed or downloaded in two modes:

- **Online:** The activity report is printed/downloaded as and when an activity occurs.
- **Report (Offline):** The activity report is printed/downloaded whenever desired. In the Offline mode, the last 500 activities recorded by the system are printed/downloaded.

The System Administrator can print/download System Activity Log, online or offline using any serial device connected to the COM Port of SARVAM UCS.

SARVAM UCS also supports Syslog Client for System Activity Logs. The Syslog Client enables the system to send activity logs in syslog format to the remote 'Syslog Server'. You can view the logs on the remote server.

You may use Syslog for System Activity Log, if you have no spare COM Port on your SARVAM UCS.

How it works

- A destination port, serial or ethernet, must be assigned for activity logs. The system will send the activity log to this port.
- If the System Administrator extension is a DKP or an Extended IP Phone, a DSS Key can be assigned for System Activity Log.
- Each activity is stored in the Activity Log in this format:
  <DD-MM-YYYY> < HH:MM:SS> <Activity Text>
- Whenever an activity is recorded by the system, the DSS key, if assigned for this feature on the System Administrator's DKP/Extended IP Phone extension, is turned ON.
- The System Administrator can view the activity log by pressing the DSS key (if assigned). The DKP/ Extended IP Phone of the System Administrator will display the activity in this format:

  DD-MM HH:MM <Activity Index>

*The format of the Date will be DD-MM or MM-DD as per **Date Format** selected in the **Real Time Clock** settings of the system.*

Index of the Type of Activities recorded in the System Activity Log:

Event Index	Activity	Description
01	System VxxRyy.zz Started	xx = 2 digit Version number yy = 2 digit Revision number zz = Either 1 digit or 2 digits (not always 2 digits)
02	Default Configuration Loaded	
03	Card Present: Slot=xx, Type: CARD TYPE	xx = Slot number CARD TYPE = Card Type with version revision
04	SLT Normal: nnnnnn, Slot=xx, Port=yy	nnnnnn = max. 6 digit flexible number assigned to SLT xx = Slot number yy = Port number
05	DKP Normal: nnnnnn, Slot=xx, Port=yy	nnnnnn = max. 6 digit flexible number assigned to DKP xx = Slot number yy = Port number
06	BRI Layer-1 Up : Slot=xx, Port=yy	xx = Slot number yy = Port number
07	T1E1 Layer-1 Up : Slot=xx, Port=yy	xx = Slot number yy = Port number
08	SMDR-IC buffer deletion: nnnnnn	nnnnnn = max. 6 digit flexible number of the station
09	SMDR-OG buffer deletion: nnnnnn	nnnnnn = max. 6 digit flexible number of the station
10	SMDR-Internal buffer deletion: nnnnnn	nnnnnn = max. 6 digit flexible number of the station
11	SE Access From: nnnnnn	nnnnnn = max. 6 digit flexible number of the station
12	SA Access From: nnnnnn	nnnnnn = max. 6 digit flexible number of the station
13	Emergency Number Dialed: nnnnnn	nnnnnn = max. 6 digit flexible number of the station
14	Log In nnnnnn	nnnnnn = max. 6 digit flexible number of the station
15	Log Out nnnnnn	nnnnnn = max. 6 digit flexible number of the station
16	Reg Fail, Authentication PW Invalid, SIPTrk=xx	xx = SIP Trunk number
17	Reg Fail, Config Parameters Invalid, SIPTrk=xx	xx = SIP Trunk number
18	Stack Construct Authenticate Fail, Slot=xx, Port=yy	xx = Slot number yy = Port number
19	Stack Construct IP Addr Invalid, Slot=xx, Port=yy	xx = Slot number yy = Port number
20	Call Budget exhausted, xx - yyy, NAME	xx = Port Type yy = Port offset NAME = Name of the Trunk
21	Reserved	Reserved for future use
22	Reserved	Reserved for future use
23	Personal Mailbox is full for Extension nnnnnn	nnnnnn = max. 6 digit flexible number of the station

Event Index	Activity	Description
24	VMS USB Memory is Full	xx = Slot number yy = Port number
25	Restart due to change Network Para Slot=xx, Port=yy	xx = Slot number yy = Port number
26	Restart due to change SIP Para Slot=xx, Port=yy	xx = Slot number yy = Port number
27	Restart due to Master request Slot=xx, Port=yy	xx = Slot number yy = Port number
28	CPU Card- is in Active Mode.	
29	CPU Card- is in Standby Mode.	
30	Power Supply Card- O.K.	
31	CPU- Ethernet link up.	
32	CPU- Ethernet link down.	
33	Data Port link up, Port=xx	
34	Data Port link down, Port=xx	
35	Usb is full!!	
36	VMS USB Memory 80% Full	
37	VMS USB Memory Usage in Limit	
38	DSP Frame Sync error, Slot=xx	xx = Slot number
39	DSP Drop error, Slot=xx	xx = Slot number
40	DSP Time Out error, Slot=xx	xx = Slot number
41	DSP DMA Address error, Slot=xx	xx = Slot number
42	Message Notification Retry Count over for nnnnnn	nnnnnn = max. 6 digit flexible number of the station
43	Default Password restored for Authority Code:nnn	nnn = 3 digit Authority Code
44	Call between nnnnnn and COxxx dropped as prone to fake	nnnnnn = max. 6 digit flexible number of the station
45	Emergency Call of xxxxxx is Acknowledged by yyyyyy.	xxxxxx = max. 6 digit flexible number of the station from which the Emergency Call was made yyyyyy = max. 6 digit flexible number of the station which had acknowledged the Emergency call.
46	Reserved	Reserved for future use
47	System fan has failed. VoIP module will stop working	
48	System fan has failed. System will go in Thermal Sleep Mode	
49	Reserved	Reserved for future use

Event Index	Activity	Description
50	T1E1 Layer-2 Up : Slot=xx, Port=yy	xx = Slot number yy = Port number
51	BRI Layer-2 Up : Slot=xx, Port=yy	xx = Slot number yy = Port number
52	System Restart Using SMS	
53	Forward DST Applied at: DD-MMM HH:MM	DD = Date , MMM = Month, HH: Hour, MM: Minute at when Forward DST is applied
54	Backward DST Applied at: DD-MMM HH:MM	DD = Date , MMM = Month, HH: Hour, MM: Minute at when Backward DST is applied
55	"DST Config Change: Enabled/Disabled DST Config Change:"	When DST mode is set to either Manual or Scheduled from Disabled or vice versa then first message is displayed. If any change is made in DST -Manual/ Scheduled Parameters the second message is displayed
56	System VxxRyy.zz Started with System Command App	xx = 2 digit Version number yy = 2 digit Revision number zz = Either 1 digit or 2 digits (not always 2 digits)
57	Command will execute using System Function	At every power-on when the System is up using System Commands
58	LOGIN_NAME Login blocked for IP=IP_ADDRESS	"IP_ADDRESS" should be the IP Address from which the last failure attempt was made. LOGIN_NAME should be actual Login ID i.e. SE or SA or FDU.
59	IP= IP_ADDRESS blocked for Auto Prov. of User SIP_ID	"IP_ADDRESS" should be the IP Address from which the last failure attempt was made. SIP_ID should be SIP ID of user.
60	"IP= IP_ADDRESS:PORT blocked	IP_ADDRESS:PORT should be the IP Address and Port from which the last failure attempt was made.
61	Sendto Network Error for Connection ID XXX	XXX is the connection id for which the call is released by the VoIP.
62	Shadow+ file removed	
63	General Mailbox is Full	
64	Reserved for future use	
65	Reserved for future use	
66	XYZ Login IP Address	When anyone logs into the SE, SA or FDU mode. XYZ is the login mode - SE/SA or FDU.
67	System Restart by YY	When the system is restarted from the Web GUI or Extension number using commands. YY is the IP Address or the Extension Number.

Event Index	Activity	Description
68	CPU-X Restart by IP Address	When the CPU card is restarted from the Web GUI XX is CPU 1 or 2
69	Slot-XX Restart by YY	When any card is restarted from the Web GUI or Extension Number using commands. XX is slot number from 01 to 27 and YY is the IP Address or Extension Number.
70	Extended Firmware is not upgraded	Firmware is upgraded manually using the external USB and internal USB is the Extended Firmware is not upgraded
71	Internal USB not detected	While Firmware upgrade the internal USB is not detected, hence upgrade will not be possible.
72	Mobile SIM:XX	When a Mobile SIM is inserted or removed from a Mobile Card. XX is the Mobile Port Number.
73	LDAP Test completed successfully	
74	LDAP Sync done for Global Directory	
75	Complex SIP Password	
76	Redundancy grace period	
77	Validity expires in 0 days - CertificateFriendlyName	
78	Redundancy Call Acknowledged by xxxxxx	xxxxxx = max. 6 digit flexible number of the station that acknowledged the redundancy notification call.
79	Redundancy Call Unacknowledged by xxxxxx	xxxxxx = max. 6 digit flexible number of the station that did not acknowledged the redundancy notification call.
80	RAM_USAGE_HIGH	RAM Usage is High. System Performance may degrade
81	RAM_USAGE_BACK_TO_NORMAL	RAM Usage is now back to Normal
82	CPU_USAGE_HIGH	CPU Usage is High. System Performance may degrade
83	CPU_USAGE_BACK_TO_NORMAL	CPU Usage is now back to Normal

When installed in the Hotel mode, the SARVAM UCS captures Hotel-Motel Activity Log. To know more, see the SARVAM UCS Hospitality System Manual.

How to configure

The two functional parts of system activity log are: Storage and Report Generation in the Online or Report modes.

To be able to use this feature, you must enable storage of Activity Logs, and assign the Syslog Server address as Destination Port for the logs.

The destination port may be a COM Port or Ethernet Port.

If you want to use Syslog Server, you must assign the IP Address of the remote Syslog server as the Destination Port for the System Activity Log.

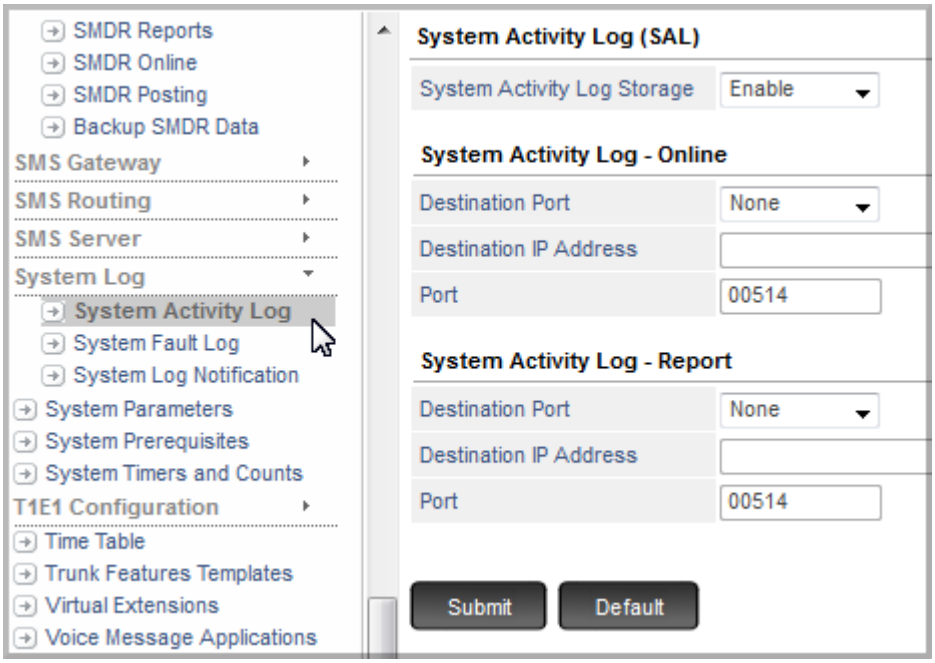
If the System Administrator phone is a DKP or an Extended IP Phone, you may assign a DSS key for System Activity Log.

For instructions on configuring a DSS key on a DKP, see “DSS Keys Programming”.

For instructions on configuring a DSS key on an Extended IP Phone, see “DSS Key Settings” in “Configuring Matrix SPARSH VP330”, “DSS Key Settings” in “Configuring Matrix SPARSH VP248”, “DSS Key Settings” in “Configuring Matrix SPARSH VP310” and “DSS Key Settings” in “Configuring Matrix SPARSH VP510”.

Configuring System Activity Log using Jeeves

- Open Jeeves
- Log in as System Engineer.
- Under **Configuration**, click **System Log** link.
- Click **System Activity Log**.



- By default, **System Activity Log Storage** is **Enabled**. Select Disable, if you do not want the system to keep a record of all the system activities.
- To generate **System Activity Log - Online**, that is, as and when the activity occurs, select **Destination Port** for Online SAL from the following options:
  - **COM Port:** Select a communication port if you want to use a serial device to capture the logs. Make sure the device is connected to the COM port.
  - **Ethernet:** Select Ethernet port if you want to use the remote syslog server for the logs.

- **USB to COM Port:** Select USB to COM port if you want to use the External USB as a COM port for capturing the logs. Make sure you connect the USB to COM Converter to the USB Port.

  - In **Destination IP Address**, enter the IP Address of the remote Syslog Server. Both IPv4 and IPv6 addresses are supported.

  - In **Port**, enter the port of the remote Syslog Server. Valid port range: 1025 to 65535;514.

  By default, no destination port is assigned for Online SAL.

- To generate **System Activity Log - Report**, that is, offline, whenever desired, select **Destination Port** for SAL Report from the options **COM Port**, **Ethernet Port** or **USB to COM Port**. Default: None.
  If you selected Ethernet as the Destination port,

  - In **Destination IP Address**, enter the IP Address of the remote Syslog Server. Both IPv4 and IPv6 addresses are supported.

  - In **Port**, enter the port of the remote Syslog Server. Valid port range: 1025 to 65535;514.

- Click **Submit** to save settings.

Configuring System Activity Log using a Telephone

- Enter SE mode from a DKP/SLT.

  To set enable/disable SAL storage, dial:
  - ***6401-Storage Flag***
    Where,
    Storage Flag is
    0 for Disable
    1 for Enable
    Default: Enable

  To select the Destination Port for SAL Online, dial:
  - ***6402-Port***
    Where,
    Port is
    0 for None
    1 for COM Port
    2 for Ethernet Port
    3 for USB to COM Port
    Default: None

  If you selected Ethernet Port as Destination port, to assign the IP Address for the Ethernet Port, dial:
  - ***6404-IP Address***

IPv6 address can be configured using Jeeves only.

  To assign Port for the IP Address, dial:
  - ***6405-Port***
    Where,
    Port is from 514 and 1025-65535
    Default: 514.

To select the Destination Port for SAL Report, dial:

- ***6403-Port***
  Where,
  Port is
  0 for None
  1 for COM Port
  2 for Ethernet Port
  3 for USB to COM Port
  Default: None

If you selected Ethernet Port as Destination port, to assign the IP Address for the Ethernet Port, dial:

- ***6406-IP Address***

IPv6 address can be configured using Jeeves only.

To assign Port for the IP Address, dial:

- ***6407-Port***
  Where,
  Port is from 514 and 1025-65535
  Default: 514.

To restore default values of the system activity log parameters, dial:

- ***6410***

- Exit SE Mode.

How to use

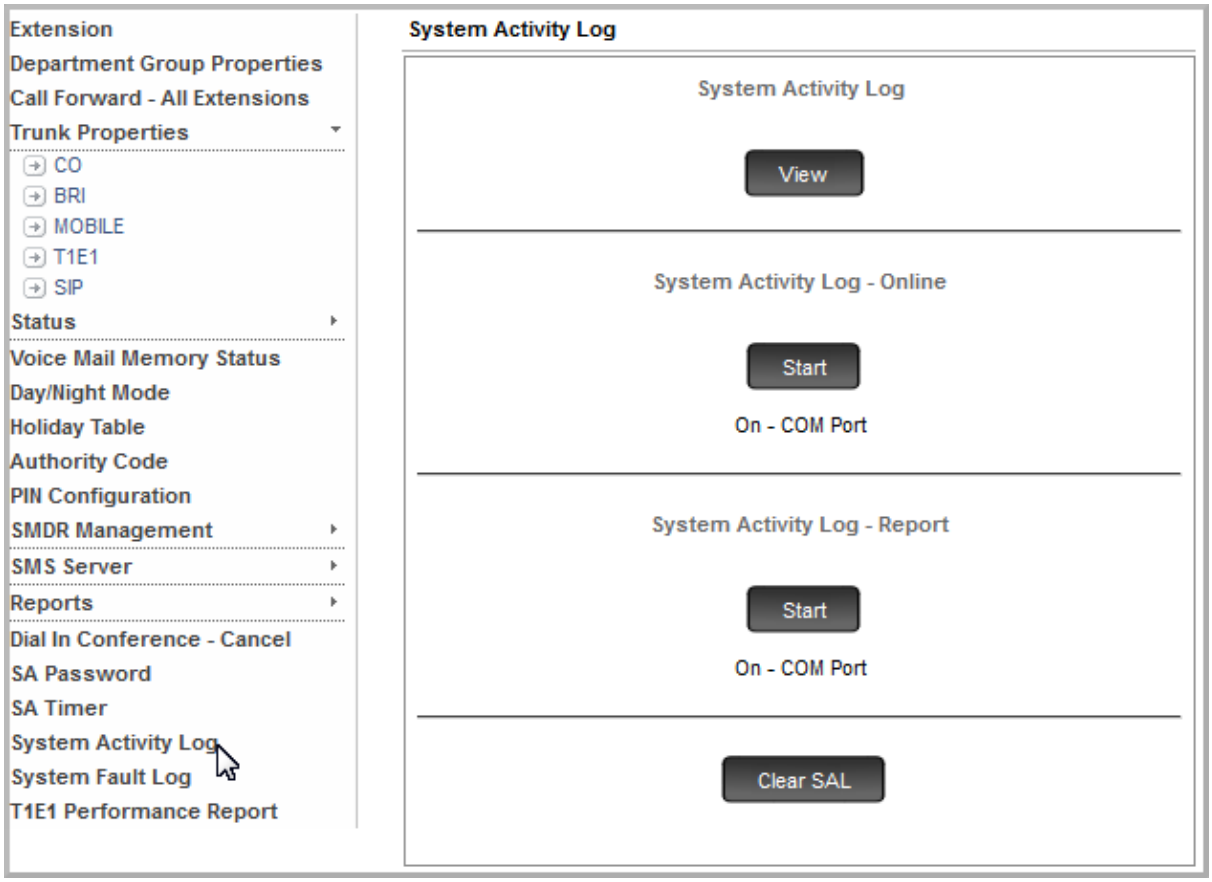
You can start and stop System Activity Log - Online and Report from the System Administrator mode using Jeeves or dialing SA Commands from an extension phone.

To start/stop report generation using Jeeves,

- Open Jeeves.

- Log in as System Administrator.

- Click the **System Activity Log** link.



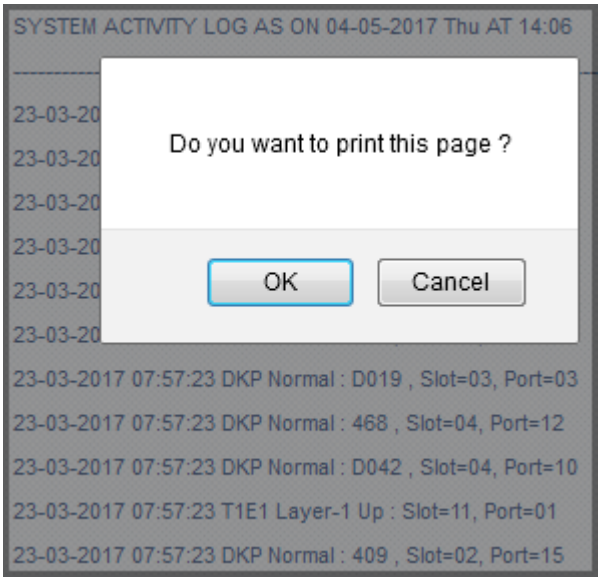
- To start **System Activity Log - Online**, click the **Start** button.
- To stop **System Activity Log - Online**, click the **Abort** button.
- To start **System Activity Log - Report**, click the **Start** button.
- To stop **System Activity Log - Report**, click the **Abort** button.
- To clear System Activity Logs from the buffer, click the **Clear SAL** button.

By default, the online and offline reports are printed on the destination port assigned by you.

- To view the **System Activity Log** on your computer screen, click the **View** button.

System Activity Log appears on your screen.

- To print this report on the local printer connected to your computer, click the **Print** button. An alert message will appear.



- Click the **OK** button. The System Activity Log - Report will be printed on the local printer connected to your computer.

Make sure you have selected 'NONE' as the destination port to print the report on the local printer connected to your computer.

- You may log out of the System Administrator mode.

To generate Activity logs from an extension phone,

- Enter SA mode from a DKP/SLT/Extended IP Phone.

To start/stop **Online** log generation:

- Dial **1072-024-1** to start
- Dial **1072-024-0** to stop

To start/stop offline **Report** generation:

- Dial **1072-023-1** to start
- Dial **1072-023-0** to stop

To clear Activity Logs,

- Dial **1072-022-Reverse SA Password**

- Exit SA mode.

You may print the logs captured on the Syslog Server after suitable modification of the format.

The **Online** System Activity Log report would look like this:

```
30-01-2017 17:11:11 Card Present: Slot=07, Type: DKP16 V3R0
```

The **Offline** System Activity Log report would look like this:

```
SYSTEM ACTIVITY LOG AS ON 27-02-2017 Mon AT 12:00
--------------------------------------------------------------------------------
30-01-2017 17:10:57 System V01R02.0 Started with System Command App
30-01-2017 17:11:11 Card Present: Slot=29, Type: VoIP NX CPU V1R4
30-01-2017 17:11:11 Card Present: Slot=07, Type: DKP16 V3R0
30-01-2017 17:11:11 Card Present: Slot=33, Type: SWITCH V0R0
30-01-2017 17:11:29 DKP Normal : 3001 , Slot=07, Port=01
30-01-2017 17:58:02 Restart due to change Network Para Slot=29, Port=01
30-01-2017 17:58:03 Stack Construct IP Addr Invalid Slot=29, Port=01
30-01-2017 17:58:13 Card Present: Slot=29, Type: VoIP NX CPU V1R4
31-01-2017 09:17:03 Restart due to change Network Para Slot=29, Port=01
31-01-2017 09:17:03 Stack Construct IP Addr Invalid Slot=29, Port=01
31-01-2017 09:17:14 Card Present: Slot=29, Type: VoIP NX CPU V1R4
31-01-2017 09:22:27 Restart due to change Network Para Slot=29, Port=01
--------------------------------------------------------------------------------
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```

System Activity Log Display

What's this?

The SARVAM UCS provides a facility to display the last activity monitored by the system on the System Administrator's extension phone. The system also provides you the facility to view all the activities through Jeeves. To know more, see "How to use" in "System Activity Log".

How to use

To be able to use this feature optimally, the System Administrator extension phone must be a DKP or an Extended IP Phone, and a DSS Key must be assigned on the phone to System Activity Log Display.

For instructions on configuring DSS Keys on a Digital Key phone, see "DSS Keys Programming".

For instructions on configuring DSS Keys on Matrix Extended IP Phone, see "DSS Key Settings" in "Configuring Matrix SPARSH VP330", "DSS Key Settings" in "Configuring Matrix SPARSH VP248", "DSS Key Settings" in "Configuring Matrix SPARSH VP310" and "DSS Key Settings" in "Configuring Matrix SPARSH VP510".

To view the System Activity Log from System Administrator Mode,

- Go Off-hook.
- Press the DSS key assigned to System Activity Log Display.
  OR
- Dial 1072-009
- The last recorded Activity log appears on your phone's display in this following format: ***Date-Time-Activity Index***
  The Date and Time are in <DD-MM-YYYY HH:MM:> format
  The Activity Index is a two digit number from 01 to 23.

  See System Activity Log Activity Index table in "System Activity Log".

The date and month format will be DD-MM or MM-DD as per date format set in the system. See "Real Time Clock (RTC)" for setting the date format.

System Fault Log

What's this?

The SARVAM UCS maintains a log of all system faults. The system Fault Log has a buffer capacity of 500 records. The Fault Log stores records using the FIFO method.

The System Fault log can be printed on a local printer or downloaded on a computer in form of a report. The report can be printed or downloaded by the System Administrator in two modes:

- **Online:** The fault report is printed/downloaded as and when a fault occurs.
- **Report (Offline):** The faulty report is printed/downloaded whenever desired. In the Report (Offline) mode, the last 100 faults recorded by the system are printed/downloaded.

The System Administrator can print/download System Fault Log, *Online* or *Report* using any serial device connected to the COM Port[340] of SARVAM UCS.

Matrix SARVAM UCS also supports Syslog Client for System Fault Logs. The Syslog Client enables the system to send fault logs in syslog format to the remote 'Syslog Server'. You can view the logs on the remote server.

You may use Syslog for System Fault Log, if you have no spare COM Port on your SARVAM UCS.

How it works

- A destination port for sending the report must be selected to which the system can send the log.
- If the System Administrator extension is a DKP or an Extended IP Phone, a DSS Key can be assigned for System Fault Log.
- Whenever a fault is detected, the LED of the Fault Log DSS key, if assigned, is turned ON.
- If more than one DKP/Extended IP extension is assigned Fault Log DSS Key, the LED of all keys will be turned ON.
- The System Administrator must acknowledge the Fault indication by pressing the Fault Log key or by dialing the Fault Log access code. The LED of the Fault Log key is turned OFF.

340. PENX does not support COM Port, hence Reports can be printed using USB to COM Port.

The different fault events that are logged are summarized in this table:

Event Index	Activity	Description
01	Card Absent: Slot=xx, Type: CARD TYPE	xx = Slot number CARD TYPE = Card Type with version revision
02	DKP Absent: nnnnnn, Slot=xx, Port=yy	nnnnnn = max. 6 digit flexible number assigned to DKP xx = Slot number yy = Port number
03	SLT Short : nnnnnn, Slot=xx, Port=yy	nnnnnn = max. 6 digit flexible number assigned to SLT xx = Slot number yy = Port number
04	SLT Standby-Open: nnnnnn, Slot=xx, Port=yy	nnnnnn = max. 6 digit flexible number assigned to SLT xx = Slot number yy = Port number
05	BRI Layer-1 Down : Slot=xx ,Port=yy	xx = Slot number yy = Port number
06	T1E1 Layer-1 Down : Slot=xx ,Port=yy	xx = Slot number yy = Port number
07	Reserved	
08	Com Port Link Break Down	
09	RTC Failure	Reserved for future use
10	VOPP Program Download Fail, Mod = x	x = VoIP module number
11	VOPP - SYNC Failed, Mod = x	x = VoIP module number
12	Registration Timer Fail, SIP Trunk=xx	xx = SIP Trunk number
13	CPU Card - gets Absent	
14	Power Supply Card- gets Absent	
15	HPI Queue Fail, Slot=xx	xx = Slot number
16	Slave DSP Alive Error, Slot=xx	xx = Slot number
17	SMTP Error Code[a] - xx	xx = Error Code Number
18	VOPP - Failed to Open Kernel File, Mod = x	x = VoIP module number
19	VOPP - Failed to Boot, Mod = x	x = VoIP module number

Event Index	Activity	Description
20	Power Supply Module Fail (LENX): STRING, Power Supply VER. Card x	STRING: One of the following will be displayed: 3.5 VDC Module 1 Fault 3.5 VDC Module 2 Fault 5 VDC Module Fault -27 VDC Module 1 Fault -27 VDC Module 2 Fault -87 VDC Module 1 Fault -87 VDC Module 2 Fault Over Temperature Alarm VER.: Version with Version Revision x = Card Number (1 or 2)
21	System fan has failed. VoIP module has stopped working	
22	T1E1 Layer-2 Down: Slot=xx ,Port=yy	xx = Slot number yy = Port number
23	BRI Layer-2 Down : Slot=xx ,Port=yy	xx = Slot number yy = Port number
24	Scheduled backup file of SMS Server is deleted	
25	SLT Thermal Shutdown: Slot=xx, Port=yy	xx = Slot number yy = Port number
26	System fan has failed. System is in Thermal Sleep Mode	
27	LAN and WAN Port IP Addresses are in the same subnet	
28	Power Supply Card Fail (MENX): Power Supply VER. Card x Fail/OFF	VER.: Version with Version Revision x = Card Number (1 or 2)
29	Unlicensed Slot, Card Inactive: Slot=xx, Type: CARD TYPE	xx = Slot number CARD TYPE = Card Type with version revision
30	CommMngr Health Failed: CPU-X	X = CPU 1 or 2 DSP of CPU card stops responding.
31	LDAP Test Failed	
32	LDAP Sync Failed	
33	Redundancy failed - License Not Activated	
34	Validity expired - FRIENDLY_NAME	FRIENDLY_NAME = Name of the Certificate that has expired.

a. For *SMTP Error Code details see "SMTP Errors" at the end of this topic.*

VOPP Fail: *If VoPP fail message is logged, the SARVAM UCS CPU Card will not be functional.*

Registration Timer Fail: *The system may fail to load either the Re-registration Timer or the Registration Retry Timer. In such a case the Proxy SIP trunk will remain un-registered and will not be functional.*

The system will decode the registration status message received from the VoIP module and, if it is found to be a problem caused by Registration Timer Failure, this will be logged in the System Fault Log.

This can happen to one or more SIP trunks, while the other SIP Trunks are functioning normally. You need to restart the system to resolve the problem.

How to configure

To be able to use this feature, you must enable storage of Fault Logs, and assign a Destination Port for the Fault Logs. The destination port may be a COM Port, Ethernet Port or USB to COM Port.

If the System Administrator phone is a DKP or an Extended IP Phone, you may assign a DSS key for System Fault Log.

For instructions on configuring DSS Keys on a Digital Key phone, see “DSS Keys Programming”.

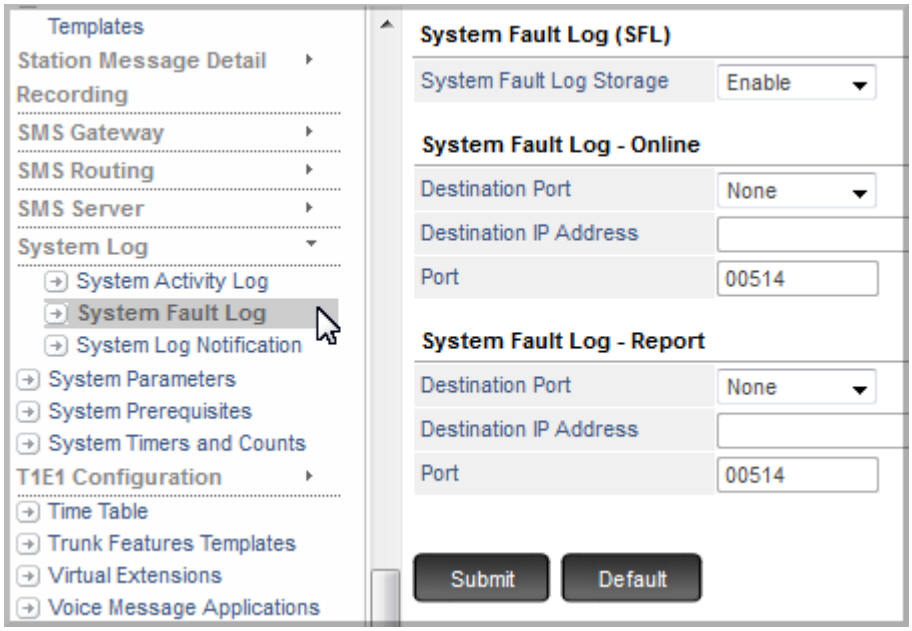
For instructions on configuring DSS Keys on Matrix Extended IP Phone, see “DSS Key Settings” in “Configuring Matrix SPARSH VP330”, “DSS Key Settings” in “Configuring Matrix SPARSH VP248”, “DSS Key Settings” in “Configuring Matrix SPARSH VP310” and “DSS Key Settings” in “Configuring Matrix SPARSH VP510”.

You may configure the System Fault Log settings using Jeeves and by dialing system commands from a telephone connected to the SARVAM UCS.

Configuring System Fault Log using Jeeves

- Open Jeeves
- Log in as System Engineer.
- Under **Configuration**, click **System Log** link.

- Click **System Fault Log**.



- By default, **System Fault Log Storage** is **Enabled**. Select Disable, if you do not want the system to keep a record of all the system faults.
- To generate **System Fault Log - Online**, that is, as and when the fault occurs, select the **Destination Port** for Online SFL from the following options:
  - **COM Port:** Select a communication port if you want to use a serial device to capture the logs. Make sure the device is connected to the COM port.
  - **Ethernet**: Select Ethernet port if you want to use the remote syslog server for the logs.
  - **USB to COM Port:** Select USB to COM port if you want to use the External USB as a COM port for capturing the logs. Make sure you connect the USB to COM Converter to the USB Port.
    - In **Destination IP Address**, enter the IP Address of the remote Syslog Server. Both IPv4 and IPv6 addresses are supported.
    - In **Port**, enter the port of the remote Syslog Server. Valid port range: 1025 to 65535;514.

  By default, no destination port is assigned for Online SFL.
- To generate **System Fault Log - Report**, that is, offline, whenever desired, select **Destination Port** for SFL Report from the options: **COM Port** or **Ethernet**. Default: None.
  - If you selected Ethernet as Destination Port,
    - In **Destination IP Address**, enter the IP Address of the remote Syslog Server. Both IPv4 and IPv6 addresses are supported.
    - In **Port**, enter the port of the remote Syslog Server. Valid port range: 1025 to 65535;514.
- Click **Submit** to save settings.

Configuring System Fault Log using a Telephone

- Enter SE mode from a DKP/SLT.

  To set enable/disable SFL storage, dial:
  - ***6451-Storage Flag***
    Where,
    Storage Flag is
    0 for Disable
    1 for Enable
    Default: Enable

  To select the Destination Port for SFL Online, dial:
  - ***6452-Port***
    Where,
    Port is
    0 for None
    1 for COM Port
    2 for Ethernet Port
    3 for USB to COM Port
    Default: None

  If you selected Ethernet Port as Destination port, to assign the IP Address for the Ethernet Port (Online), dial:
  - ***6454-IP Address***

IPv6 address can be configured using Jeeves only.

  To assign Port for the IP Address (Online), dial:
  - ***6455-Port***
    Where,
    Port is from 514 and 1025-65535
    Default: 514.

  To select the Destination Port for SFL Offline (Report), dial:
  - ***6453-Port***
    Where,
    Port is
    0 for None
    1 for COM Port
    2 for Ethernet Port
    3 for USB to COM Port
    Default: None

  If you selected Ethernet Port as Destination port, to assign the IP Address for the Ethernet Port (Report), dial:
  - ***6456-IP Address***

IPv6 address can be configured using Jeeves only.

To assign Port for the IP Address, dial:

- ***6457-Port***
  Where,
  Port is from 514 and 1025-65535
  Default: 514.

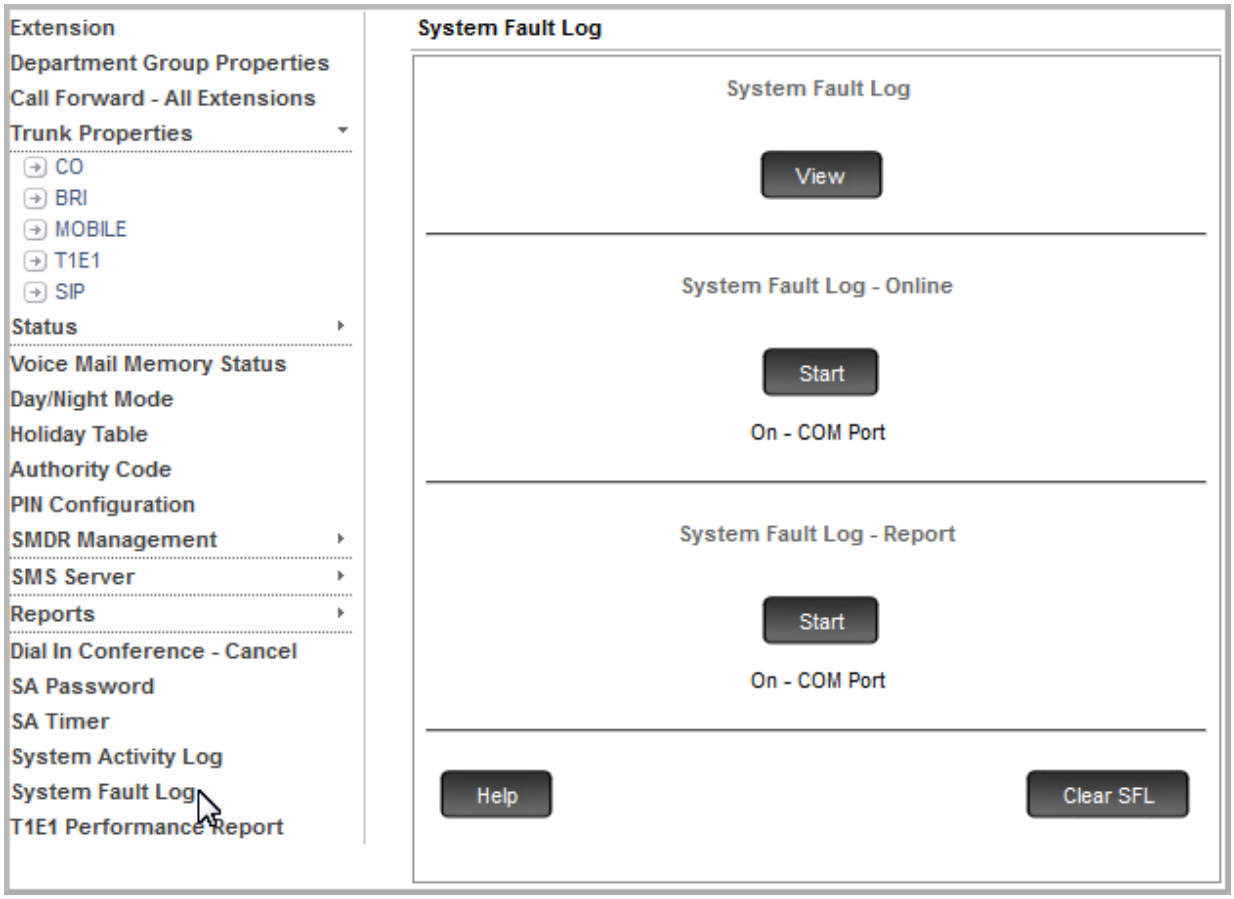
- Exit SE Mode.

How to use

You can start and stop System Fault Log - Online and Report from the System Administrator mode using Jeeves or dialing SA Commands from an extension phone.

To start/stop fault report generation using Jeeves,

- Open Jeeves.
- Log in as System Administrator.
- Click the **System Fault** Log link.



- To start **System Fault Log - Online**, click the **Start** button.
- To stop **System Fault Log - Online**, click the **Abort** button.

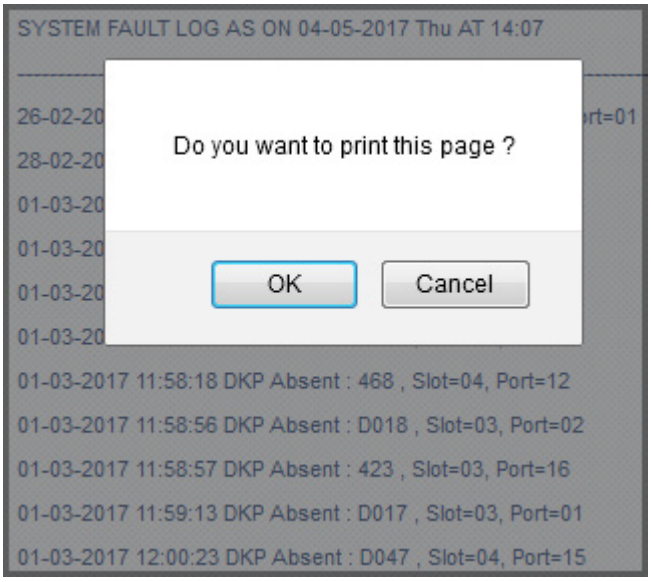
- To start **System Fault Log - Report**, click the **Start** button.
- To stop **System Fault Log - Report**, click the **Abort** button.
- To clear System Fault Logs from the buffer, click the **Clear SFL** button.

By default, the online and offline reports are printed on the destination port assigned by you.

- To view **System Fault Log** on your computer screen, click the **View** button.

System Fault Log appears on your screen.

- To print this report on the local printer connected to your computer, click the **Print** button. An alert message will appear.



- Click the **OK** button. The System Fault Log - Report will be printed on the local printer connected to your computer.

Make sure you have selected 'NONE' as the destination port to print the report on the local printer connected to your computer.

- You may log out of the System Administrator mode.

To generate Fault logs from an extension phone,

- Enter SA mode from a DKP/SLT/Extended IP Phone.

To start/stop **Online** fault log generation:

- Dial **1072-028-1** to start
- Dial **1072-028-0** to stop

To start/stop offline **Report** generation:

- Dial **1072-027-1** to start
- Dial **1072-027-0** to stop

- Exit SA mode.

You may print the logs captured on the Syslog Server after suitable modification of the format.

The format of the standard system fault report looks as below:

Field	Column Position
Blank	00
Date (As per Date format selected)	01

Field	Column Position
Time (HH:MM:SS)	12
Even Index No. (two characters)	21
Event Description	24

The **Online** report would look like this:

```
06-02-2017 15:59:54 DKP Absent : 3017 , Slot=06, Port=01
```

The **Offline** report would look like this:

```
SYSTEM FAULT LOG AS ON 27-02-2017 Mon AT 12:13
--------------------------------------------------------------------------------
30-01-2017 17:58:03 Card Absent: Slot=29, Type: VoIP NX CPU V1R4
31-01-2017 09:17:03 Card Absent: Slot=29, Type: VoIP NX CPU V1R4
31-01-2017 09:22:28 Card Absent: Slot=29, Type: VoIP NX CPU V1R4
31-01-2017 10:27:54 Card Absent: Slot=29, Type: VoIP NX CPU V1R4
31-01-2017 10:28:34 Card Absent: Slot=29, Type: VoIP NX CPU V1R4
06-02-2017 15:59:54 DKP Absent : 3017 , Slot=06, Port=01
07-02-2017 17:35:23 Card Absent: Slot=29, Type: VoIP NX CPU V1R4
09-02-2017 11:56:23 Card Absent: Slot=29, Type: VoIP NX CPU V1R4
09-02-2017 12:09:56 Card Absent: Slot=29, Type: VoIP NX CPU V1R4
09-02-2017 12:12:49 Card Absent: Slot=29, Type: VoIP NX CPU V1R4
09-02-2017 16:21:22 DKP Absent : 3019 , Slot=02, Port=01
11-02-2017 20:37:20 VOPP Program Download Fail, Mod=1
11-02-2017 20:38:52 Card Absent: Slot=01, Type: CO2 DKP2 SLT16 V3R0
--------------------------------------------------------------------------------
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```

SMTP Errors

The VMS may fail to send emails. These email failures are logged into the System Fault Log with a specific code. The table below describes the meaning of the codes.

Error Code	Error Description
SMTP Client Mail Failure Errors	
01	SMTP Client Failed to Receive Mail from IPCQ
02	File Attachment Failed
03	Invalid "To" Field
04	Invalid "From" Field
05	Invalid SMTP Server address or IP Specified
06	Invalid "HELO" domain
07	Failed to Connect to SMTP Server (Check SMTP Server Address or Port)
08	"HELO" or "EHLO" Command Failed
09	"AUTH CRAM-MD5" Command Failed
10	"CRAM-MD5" Login Failed
11	"AUTH LOGIN" Command Failed
12	Invalid Username for "LOGIN"
13	Invalid Password for "LOGIN"

Error Code	Error Description
14	"AUTH PLAIN" Login Failed
15	Can't Open Attachment File
16	"MAIL FROM" Command Failed
17	"RCPT TO" Command Failed
18	"DATA" Command Failed
19	SMTP Mail Failed (During final process to Prepare Mail)
20	Mail Failed (Wait For Server Response)
VMS Application Mail Failure Errors	
51	The calling process does not have write permission on the message queue, and does not have the CAP_IPC_OWNER capability
52	The message can't be sent due to the msg_qbytes limit for the queue and IPC_NOWAIT was specified in msgflg
53	The address pointed to by msgp isn't accessible
54	The message queue was removed
55	Sleeping on a full message queue condition, the process caught a signal
56	Invalid msqid value, or non-positive mtype value, or invalid msgsz value (less than 0 or greater than the system value MSGMAX)
57	The system does not have enough memory to make a copy of the message pointed to by msgp

System Fault Log Display

What's this?

The SARVAM UCS provides a facility to display the last fault monitored on the system on the System Administrator's extension phone. The system also provides you the facility to view all the faults through Jeeves. To know more, see "How to use" in "System Fault Log".

How it works

To be able to use this feature optimally, the System Administrator extension phone must be a DKP or an Extended IP Phone, and a DSS Key must be assigned on the phone to System Fault Log.

- When a fault occurs, the LED of the DSS Key assigned for the System Fault Log, glows.
- The System Administrator may press the DSS key or dial the System Fault Log feature access code to acknowledge it.
- On pressing the DSS Key or dialing of the acknowledgment command, the LED of the Fault Log key is turned OFF.

How to use

To view the System Fault Log from System Administrator Mode,

- Go Off-hook.
- Press the DSS key assigned to System Fault Log Display.
  OR
- Dial **1072-010**
- The Fault log appears on your phone's display in this format: ***Date-Time-Fault Index***
  The Date and Time are in <DD-MM-YYYY HH:MM:> format
  The Activity Index is a two digit number from 01 to 12.

  See System Fault Log Activity Index table in "System Fault Log".

System Log Notification

What's this?

Whenever a fault occurs in the system or an activity takes place, the details of the fault/ activity can be sent to the concerned person to notify him/ her of the same.

The notification can be sent,

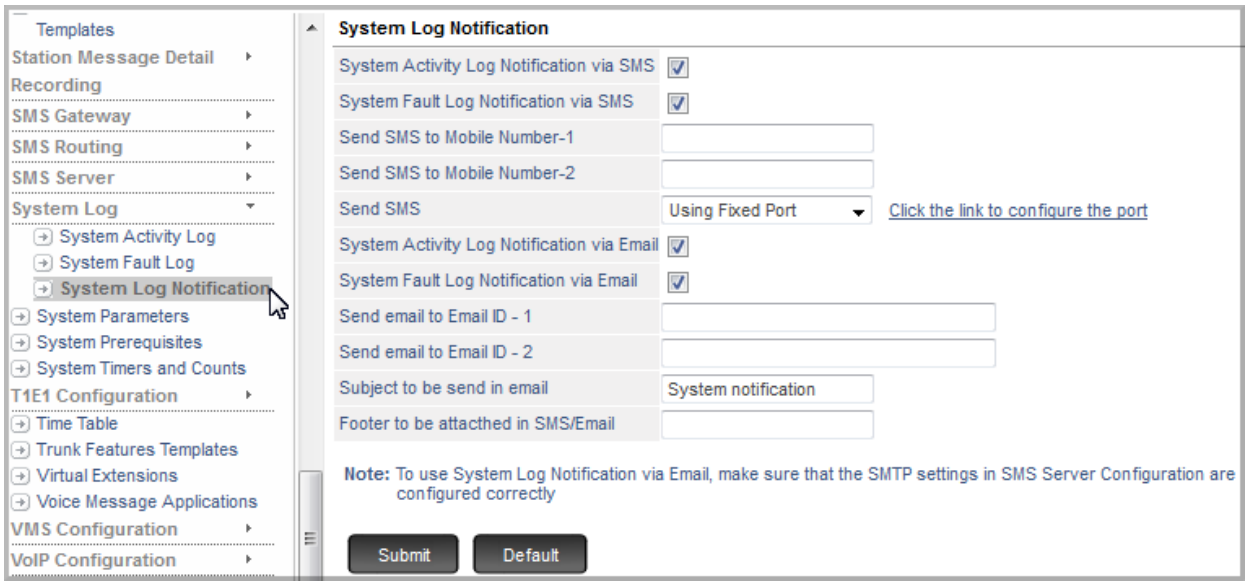
1. as a message to the specific mobile number.
   or/ and
2. as an email to the specific email id.

The SARVAM UCS maintains a log of all the system activities and faults. These logs can be printed on a local printer or downloaded on a computer in form of a report. To know more, see "System Activity Log" and "System Fault Log".

How to configure

To be able to use this feature, you must configure the following parameters.

- Open Jeeves.
- Log in as System Engineer.
- Under **Configuration**, click the **System Log** link.
- Click **System Log Notification**.



- By default, the **System Activity Log Notification via SMS** check box is enabled. Clear this check box, if you do not want the system to send notifications for the system activities via SMS.
- By default, the **System Fault Log Notification via SMS** check box is enabled. Clear this check box, if you do not want the system to send notifications for the system faults via SMS.

- In **Send SMS to Mobile Number-1** and **Send SMS to Mobile Number-2**, configure the mobile numbers to which you want the system to send notifications as SMS.
- You can **Send SMS** through a fixed Mobile Port or through different Mobile Ports according to specific numbers and time.
    - To send messages through fixed Mobile Ports, select **Using Fixed Port**. **Click the link to configure the port** link, the **SMS Routing-Fixed Port** table opens. Configure the **SMS Routing - Fixed Port Routing** table. For detailed information, see “Fixed Port Routing (SMS Server)”.
    - To send messages through certain preferred Mobile Port/s during a defined time interval, select **Based on LCR Table**. **Click the link to configure the port** link, the **SMS Routing-LCR** table opens. Configure the **SMS Routing-LCR** table. For detailed information, see “Least Cost Routing”.
- By default, the **System Activity Log Notification via Email** check box is enabled. Clear this check box, if you do not want the system to send notification for the system activities via Email.
- By default, the **System Fault Log Notification via Email** check box is enabled. Clear this check box, if you do not want the system to send notification for the system faults via Email.
- In **Send Email to Email ID-1** and **Send Email to Email ID-2**, configure the email ids to which you want the system to send notifications as an email.
- In **Subject to be Send in Email**, configure the text that is to be displayed as the subject to the receiver of the Email.
- In **Footer to be attached in SMS/Email**, configure the text you want to be displayed as the footer to the receiver of the SMS/Email.
- Click **Submit** to save settings.

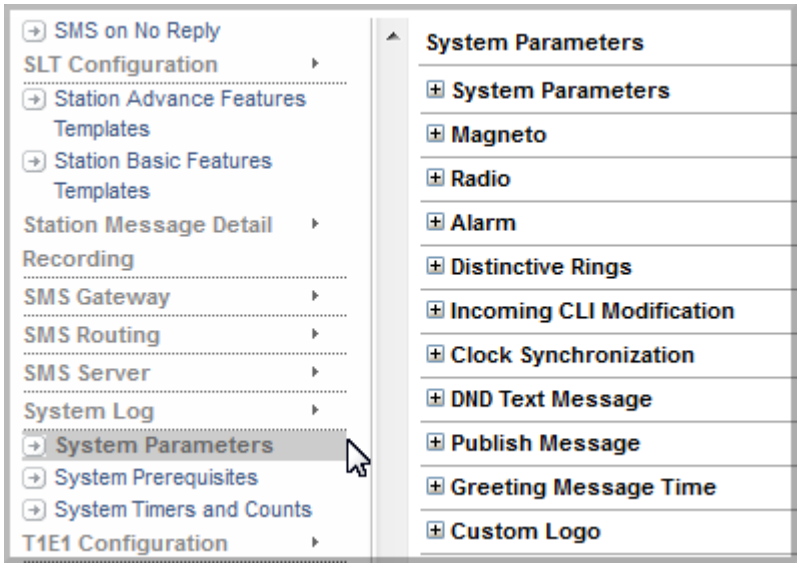
System Parameters

System Parameters are general parameters, related to features and facilities that are applied system-wide, such as customer name, Day-Night mode, storage of call logs, alarms, Built-In Auto Attendant call disconnect options, Presence, and DND messages. Each of these is described briefly along with the instructions for configuring them using the Jeeves and from a telephone.

How to configure

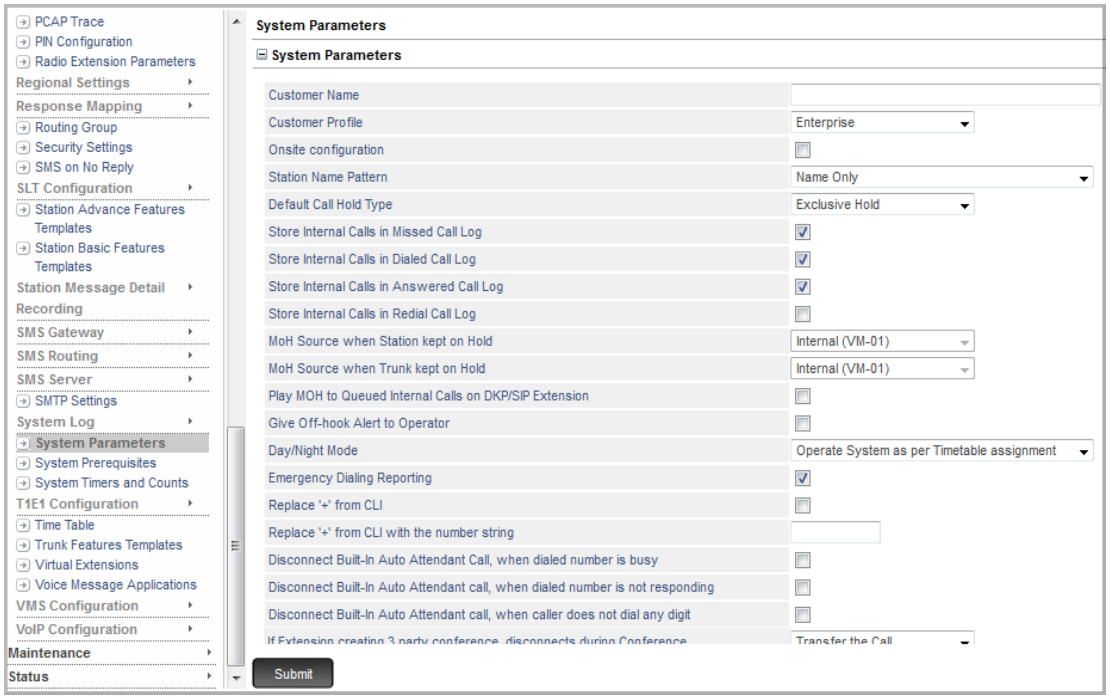
Configuring System Parameters using the Jeeves

- Log in to the Jeeves as System Engineer.
- Under **Configuration**, click **System Parameters**. The System Parameters page opens.



System Parameters

- Click **System Parameters** to expand. To view more parameters, use the vertical scroll bar on your right.



- Under *System Parameters*, do the following:

  - **Customer Name:** You can assign the name of the enterprise/organization that is using SARVAM UCS as the Customer Name. The Customer Name may contain up to 80 characters. You may enter the address of organization/enterprise along with the name.

    The Customer Name you assign will appear on the various System Reports generated and printed by the SARVAM UCS. Default: Blank.

You can assign Customer Name also on the "Configuring System Pre-requisites" page. If you have entered the Customer Name on this page, the same Name will appear on the System Parameters page.

  - SARVAM UCS two major applications: Enterprise application to meet requirements of businesses, and Hospitality application to meet the specific requirements of Hotels and Hospitals.

    You must select **Customer Profile** as **Enterprise** or **Hotel** according to the application you are using. When you select the Customer Profile, all the features and facilities specific to the application Enterprise/Hotel along with their default settings are loaded. By default, the Customer Profile of SARVAM UCS is defined as 'Enterprise'.

  - If you want the system to present for configuration only those trunks and extension ports that are detected by it at Power-On for configuration, select the **On-site Configuration** check box. Default: Disabled.

    When you enable On-site Configuration, the Jeeves, will show the pages for only those trunks and extension port types that are on board the system, that is, detected by the system at Power-On.

- **Station Name Pattern:** The Station Name Pattern is the format in which the names of extensions will be stored on the extension phones and displayed to other extensions. You can store names by First Names only, First names and Last Names. You can also add Titles indicating gender, designation, rank, social standing, like Mr. Mrs. Ms., Dr., Prof. Cmdr., Rev., to Names of extensions.

  SARVAM UCS supports the following Station Name patterns.

Option	Meaning
1	Title<space>First Name<space>Name
2	First Name only
3	Name only
4	First Name<space>Name
5	Title<space> First Name
6	Title<space>Name

  Station Name Pattern must be configured for the *Guest Name and Title* feature of the SARVAM UCS Hospitality module. To know more, refer the feature description in the *SARVAM UCS Hospitality System Manual*.

  By default, **Name Only** is selected as the Station Name Pattern when SARVAM UCS is operated in the Enterprise mode, and **Title<space>Name** is selected as the Station Name Pattern when SARVAM UCS is operated in the Hotel Mode.

- **Default Call Hold Type:** This parameter is related to the "Call Hold" feature of SARVAM UCS. You can select Global Hold or Exclusive Hold. Default: Exclusive Hold.

- To have SARVAM UCS store also internal calls in the Missed Call Log, select the **Store Internal Calls in Missed Call Log** check box. To know more about this feature, see "Call Logs". Default: Enabled.

- You may have SARVAM UCS store internal calls in the Dialed Call Log by selecting the **Store Internal Calls in Dialed Call Log** check box. To know more about this feature, see "Call Logs". Default: Enabled.

- To have the system store internal calls in the Answered Call Log, select the **Store Internal Calls in Answered Call Log** check box. To know more about this feature, see "Call Logs". Default: Enabled.

- To have the system store internal calls in the Redial Call Log, select the **Store Internal Calls in Redial Call Log** check box. To know more about this feature, see "Last Number Redial". Default: Disabled.

- In the **MoH Source when Station kept on Hold** box, keep the option **Internal (VM-01)**. To know more, read the feature description for "Music on Hold (MOH)". SARVAM UCS will play the Music-On-Hold recorded in the Voice Module Number 01 to the extension that is put on hold. Default: Internal (VM-01).

- In the **MoH Source when Trunk kept on Hold** box, keep the option **Internal (VM-01)**. To know more, read the feature description for "Music on Hold (MOH)". SARVAM UCS will play the Music-On-Hold recorded in the Voice Module Number 01 to the external callers who are put on hold. Default: Internal (VM-01).

- The DKP/SIP Extension users can set multiple call appearances on their phones. If you want the system to play MOH to all the queued internal calls when the user extension is busy, select the **Play**

MOH to Queued Internal Calls on DKP/Extended IP Phone check box. To know more, read the feature description for "Music on Hold (MOH)".

- If you select the **Give Off-hook Alert to Operator** check box, the system will detect extensions that are off-hook and ring on the Operator extension to alert the Operator about the state of the phone. This alert is useful for detecting whether the handset of extension phones are placed correctly**.** Read the feature description for "OFF-Hook Alert" to know more.

- You can set the Time Zone of the system as Working-Hours or Break Hours or Non-Working hours any time you want by setting the **Day/Night Mode**. You can set the system in the **Day Mode** or the **Break Hours Mode** or the **Night Mode**, or let the system **Operate as per the Time Table assignment**[341]. For more details see "Day Night Mode" and "Time Tables". Default: Operate System as per Time Table assignment.

- You can switch to Day Mode (Working Hrs) or Night Mode (Non-Working Hrs) on pressing the DSS key if you select the check box for **Toggle Day/Night mode through 'Set Day/Night Mode' key**. For more details, see "Day Night Mode". Default: disabled.

- If you want to remove the '+' prefix in the CLI of the calling party (presented by the Mobile Network) and replace it with another string, select the **Replace '+' from CLI** check box. Default: Disabled.

  If you want to program the number string with which the '+' prefix is to be replaced, in the **Replace '+' from CLI with the number string** field, enter the desired number string**.**

  If you keep the number string field blank, SARVAM UCS will remove '+' sign from the CLI of calling party and present the remaining digits on the CLI of the Called Party.

  For example:
  The number string +919327237228 is received as CLI.
  If '00' is configured as the replace string, the CLI number would become 00919327237228
  If no replacement string is configured (that is, left blank), the CLI number would be presented as 919327237228.

- To disconnect Built-In Auto Attendant Calls when the landing extensions are busy, select the **Disconnect Built-In Auto Attendant Call, when Dialed Number is Busy** check box. When you enable this flag, the call will not be routed to the Operator. Instead, it will be disconnected. Default: Disabled.

- To disconnect Built-In Auto Attendant Calls when there is no reply from the landing destination extensions, select the **Disconnect Built-In Auto Attendant Call, when Dialed Number is not Responding** check box.

  When this flag is enabled, the system disconnects the call if there is no reply from the landing destination extensions. The call will not be routed to the Operator. Default: Disabled.

- To disconnect Built-In Auto Attendant Calls if the caller fails to dial a digit, select the **Disconnect Built-In Auto Attendant Call, when Caller Does not Dial any Digit** check box.

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341. Certain features of the SARVAM UCS require extensions and trunks to behave differently according to the working hours, break hours and non-working hours, which are referred to as Time Zones. The Time Zones, are defined for the entire week in a Time Table. Time Table is assigned to trunks, extensions and other time-zone dependent features.

When this flag is enabled, the system will disconnect the call if the caller fails to dial a digit within the First Digit Wait Timer. The call will not be routed to the Operator. Default: Disabled.

- **If the Extension creating 3 party conference, disconnects during Conference**, you can select either to Transfer the call or Disconnect other parties.

  - If you select to **Transfer the call**, the 3-party conference is converted into a two-way speech between the other two parties.

  - If you select to **Disconnect other parties**, all the parties involved in the 3-party conference are disconnected.

  See "Conference-3 Party" for more details.

- To play a beep to participants of a conference to indicate inclusion of a new participant, select the **Play Beep when Conference/Dial-in Conference begins** check box selected. Default: Enabled.

  This flag is common for the features "Conference-Multiparty", "Conference Dial-In", "Emergency Conference". When this flag is enabled,

  - the system plays beeps to the other participants in a Dial-In Conference when a new participant joins in (dials into an on-going Dial-In Conference)

  - the system plays beeps to the other participants connected in a Multi-Party Conference and an Emergency Conference, when a new participant is included.

  If you disable this flag, the existing participants in a Dial-In or Multi-party conference will not hear any beep tone indicating the addition of a new participant.

- **Play Beep when Raid/Call Taping/Conversation Recording Starts** is a common flag for the features "Call Taping" and "Conversation Recording" and "Raid". When this flag is enabled,

  - the system plays a warning beep to the extension which is being raided by another extension, before establishing three-way speech.

  - the system plays beeps to the extensions/calling party before it starts taping the call in the common mailbox or recording the conversation in the extension's mailbox.

  When this flag is disabled, no warning beep will be played in Raid, no indication will be given to the opposite party when the call is being taped/conversation is being recorded. Default: Enabled.

- In **Play Feature Tone in place of Dial Tone when Call Forward is Set**, you can select whether you want the system to play **Feature Tone** instead of **Dial Tone** to the extensions when Call Forward is set on these extensions. When this flag is disabled, the system will play dial tone to the extension on which Call Forward is set, whenever the extension goes Off-hook. Default: Enabled.

- Select the **Ignore call forward set by member extension, when call is routed on Routing/Dept. Group** flag, if you want the system to place calls on member extensions in a Routing Group even if they have set Call Forward.

- Select **Call Proceeding Tone for Multistage Dialing**. Default: Network Tone.
  This parameter is used in Multistage Dialing where you need to configure Pause and Wait for Answer in the Dialed Number column for the number string dialed by the extension users.

When an extension user makes a call using a Calling Card, and the system dials out the number in stages, the extension user will get Ring Back Tone twice; first after the system has dialed the Calling Card Number, and again after the system has dialed out the destination number (called party number). Thus the extension user will get Ring Back Tone, twice. To avoid this, you may configure the 'Call Proceeding Tone' to be played by the system when using Multi-Stage Dialing.

You must configure the type of 'Call Proceeding Tone', according to your requirement; whether the extension user should be connected to the speech path when the Calling Card number is out dialed or when the called party number is out dialed. You can select any of the following Call Proceeding Tone options, as per your requirement:

- **Network Tone:** If this option is selected, the extension user will get Ring Back Tone after dialing the Calling Card number and again, after the system has dialed the called party number (when the system is dialing out the number with Pause and Wait for Answer configured in Dialed number string).

- **Pseudo Tone:** If this option is selected, the extension user will get Feature Tone when the user has completed dialing all the digits. At the end of the tone, the extension user gets connected to the called party (destination number).

- **Silent:** If this option is selected, the extension user will get Silence (no tone), after the extension user has completed dialing all digits. After dialing out the called party number in DTMF, the system will connect the caller to the called party number (destination number).

- You may select the **Companding Algorithm** according to the Regulatory Requirement of the country where SARVAM UCS is installed. Default: A-law

  The companding Algorithm —A law or μ law—is automatically selected when you select Region for SARVAM UCS. However, if necessary, you may change the default companding Algorithm that appears in this field.

If you change the Companding Algorithm, the terminals - EON48D and EON310.

- You can select the **Language of SE, SA and Front Desk User Web Interface** as per your requirement. The GUI of SARVAM UCS supports the languages English, Italian, Spanish, French, German, and Portuguese. When you select ‘Region’ for SARVAM UCS, one of these languages will be applied as appropriate for the region you selected. For instance, if you selected India, English will be applied. If you selected Spain, Spanish will be applied. If you selected a country where none of these languages are the local language, English will be applied.

  The language set by the system on Region selection will be applied on the pages of the GUI for every login session. You can change the default language set on Region selection, by configuring this parameter.

- To print each system report on a separate page, keep the **Form Feed in Report Printing** check box enabled. Default: Enabled.

- To be able to distinguish between incoming calls from the Public Network and those from the Private Network on the basis of the number digits received in the CLI, enter the desired number of digits in the **Minimum No. of digits received in CLI to consider the call is from Public N/w** box.

  This parameter is applicable to calls originating on the E1-PRI ports (Tie-Line) configured for the Q-Signaling. By default, CLI number with 8 or more digits will be considered as call from Public Network.

SARVAM UCS will check this parameter, whenever the incoming call is to be analyzed as call from PISN (Private Integrated Subscriber Network) or non-PISN (Public Network Number).

- If you want to enable the Operator Console to view the presence status of the extension they are calling, select the **Display Presence Status during Call on DKP/Extended IP Phone** check box. Default: Disabled.

- To use the Watch dog function supported by SARVAM UCS, select the **Enable Watch Dog** check box. SARVAM UCS supports the Watchdog function to detect and restart the system, whenever the system hangs. Default: Disabled.

  When Watch Dog function is disabled, you must manually restart the system when it hangs.

- Enable **Apply RCOC only if the caller calls back on the same trunk from which the call was made** check box, if you want the SARVAM UCS to match the Trunk Port Number and Trunk Port Type of the incoming call with the Trunk Parameters of the entry stored in RCOC table before routing the call to the original caller. Default: Disabled.

- Enable the **Stuttered Dial Tone When DND is set** check box, if you want SARVAM UCS to play a Stuttered Dial Tone on the extension when DND is set.

- If you have installed two CPU cards and want that each card must switch to the Stand-by mode after a fixed number of hours automatically, then select the desired option in **Set Active CPU Card to Stand-by after (Hrs.)**. This is applicable only for ETERNITY LENX/MENX.

- When there is an incoming call on the CO Trunk and if total 5 digits are dialed out, before expiry of Trunk Inter Digit Wait Timer, then the system will treat this call as an outgoing fraudulent call. To drop this call, select the **Detect Possible toll bypass attempt by Extn. during IC Call from CO Line & Drop the Call** check box.

- Select the **Play beeps when Assistant present in 3-Party Conference** check box, if you want the system to play beeps during the conference to indicate the presence of the Operator. For details, see “Conference-3 Party”.

- Select the **Play beeps when Assistant present in Multiparty Conference** check box, if you want the system to play beeps during the conference to indicate the presence of the Operator. For details, see “Conference-Multiparty” and “Conference Dial-In”.

- Select the **Play beeps when Assistant leaves the Conference** check box, if you want the system to play beeps to the conference participants when the Operator leaves the conference.

- By default, the **Display Guest Station in Directory** check box is enabled. This feature is required when you are using the sytem for the Hospitality Appplication.

  The system will display the contact details of the ***Guest - Station*** in the Contact List of all the other extensions.

  In the Hospitaity Application, there may be a requirement to keep the guest details confidential. In such cases you must keep this option disabled.That is the guest details will not appear in the contact list of the other extensions.

If a call is made to the guest number that is not visible in the contacts list from any extension, these details will appear in the call logs, even if this check box is disabled.

When disabled, the contact details of the Guest - Station will not appear in the contacts lists of the extensions assigned Station Type as Administration or Assistant.

- Select **Call Proceeding Tone for 1st caller of a SIP Extension**. Default: Ring Back Tone.

  You must configure the type of 'Call Proceeding Tone for 1st caller of a SIP Extension', according to your requirement. this tone will be played when the system is routing the call. You can select any of the following options:

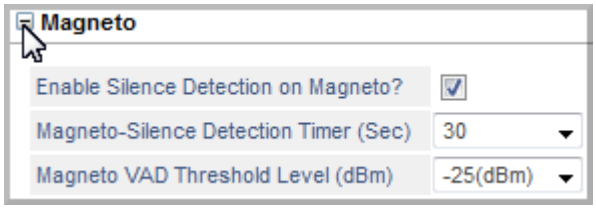
  - **Ring Back Tone:** If this option is selected, the extension user will get Ring Back Tone while the call is being routed.
  - **Pseudo Tone:** If this option is selected, the extension user will get Feature Tone while the call is being routed.
  - **Silent:** If this option is selected, the extension user will get Silence (no tone), while the call is being routed.

For routing the second call the system will always play Ring Back Tone to the second caller.

- **Include "Diversion" for call to SIP Extension:** Default: Disabled. If you have any SIP Phone/Entity registered with the system as Open SIP Standard and Call Forward is set on the same, then to include the called number in the forwarded call request, select this check box.

Magneto

- Click **Magneto** to expand.



- If you want calls from Magneto ports to be disconnected automatically, when silence is detected, select the **Enable Silence Detection on Magneto** check box. Default: Enabled.
- You may change the duration for the **Magneto Silence Detection Timer**, to the desired duration. The range of this timer is from 01 to 32 seconds. Default: 30 seconds.
- Set the **Magneto VAD Threshold Level** to the desired level. Default: -71(dBm).

Radio

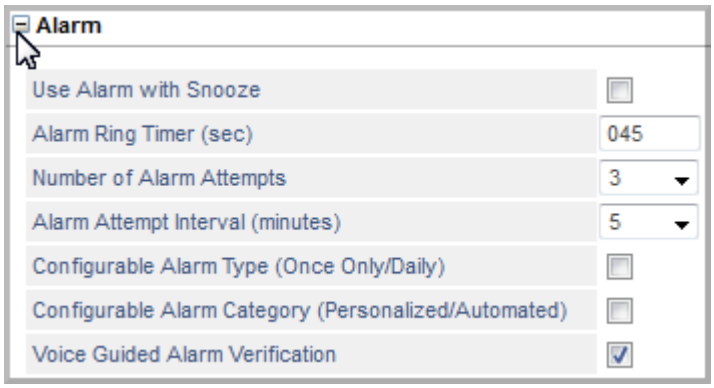
- Click **Radio** to expand.

- Configure the following parameters:
    - **DTMF Detection - Minimum Level (dB):** This parameter signifies the minimum level (dB) of the DTMF digit to be considered as valid. By default, Minimum levels set to -10.5dB.
    - **DTMF Detection - Minimum ON Time (msec):** This parameter signifies the minimum time period for which the DTMF signal should be present in order to be detected. The valid range of this time is 17 to 204 milliseconds. By default, Minimum ON Time is set to 17 milliseconds.
    - **DTMF Detection - Minimum OFF Time (msec):** This parameter signifies the minimum time period between successive DTMF digits. The valid range of this time is 17 to 204 milliseconds. By default, Minimum OFF Time is set to 17 milliseconds.
    - **DTMF Generation - DTMF ON Time (msec):** It is the width of DTMF digit to be dialed out by the Radio port. By default the ON Time is set to 102 milliseconds.
    - **DTMF Generation - Inter Digit Pause Time (msec):** When the Radio port dials out the DTMF digits, it waits for the Inter Digit Pause Timer, while dialing the DTMF digits. By default the timer is set to 102 milliseconds.

The DTMF Detection and Generation parameters are applicable only when there are Dial Pads connected to the Radio devices.

Alarm

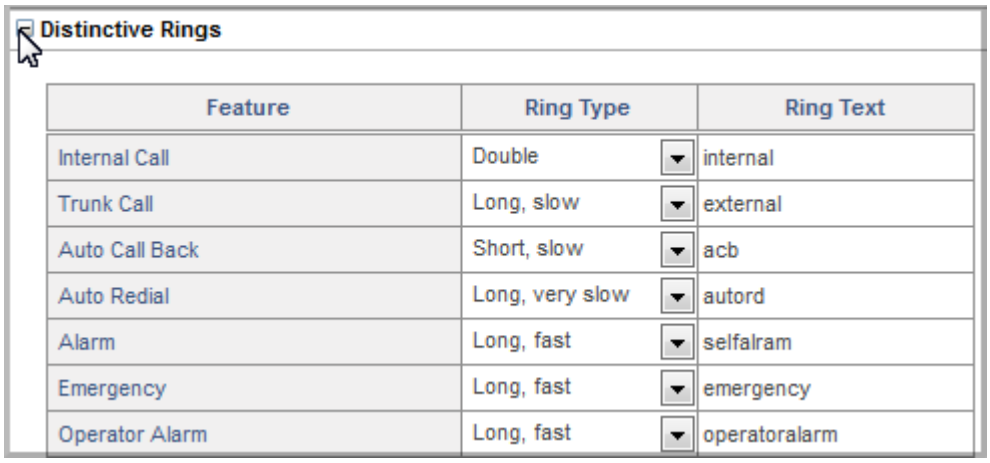
- Click **Alarm** to expand.



- Configure the following Alarm and Reminder parameters:
  - **Use Alarm with snooze**: Enable this flag if you want to use the Snooze function for the Alarm Call.
  - **Alarm Ring Timer (Sec.)**: You may change the time for which the Alarm Call will ring on the extension phone and the time for which the Operator phone will ring to notify an unanswered Alarm Call.
  - **Number of Alarm Attempts**: You may increase or decrease the number of attempts the system should make to serve an Alarm call.
  - **Alarm Attempt Interval**: You may increase or decrease the time gap between each attempt the system makes to serve an Alarm call.
  - **Configurable Alarm Type flag**: Disable this flag, if you do not what the system to provide the Operator and the extension users the option of setting 'Once Only' or 'Daily' Alarms. When this flag is disabled, the system will allow only 'Once Only' alarms to be set.
  - **Configurable Alarm Category**: Disable this flag, if you do not want the system to provide the Operator the option of setting 'Personalized' or 'Automated' Alarm calls. When this flag is disabled, the system will follow the 'Automated' Alarm call serving mechanism. The Operator will not be prompted to choose between 'Automated' and 'Personalized' Alarm calls when setting Alarm calls for an extension phone.
  - **Voice Guided Alarm Verification:** Enable this flag if you want to enable extension users to confirm the Time they have set for an alarm or the Date and Time they have set for a reminder. Default: Disabled.

Distinctive Rings

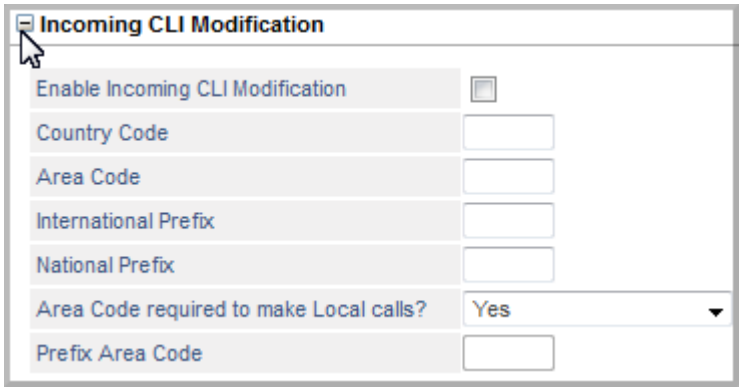
- Click **Distinctive Rings** to expand.



Distinctive Rings are ringing patterns used for distinguishing between different types of call events, like Internal Calls, Trunk Calls, Auto Call Back, Auto Redial, Alarm, Emergency call, Priority, etc. If you want to customize the Ringing pattern, for a call event, select the desired **Ring Type**. For more details see “Distinctive Rings”.

Incoming CLI Modification

- Click **Incoming CLI Modification** to expand.



Incoming CLI Modification is useful in countries where the Calling Line Identification (CLI) received by the System extension users must be suitably modified before it can be used to dial out the number. To know more, see "Incoming CLI Modification".

If you receive CLI in dialable format, there is no need to use this feature. In such case, keep the flag disabled. You do not need to program any of the CLI Modification parameters.

*For an incoming call on any trunk, the Incoming CLI Modification is applicable only when both — the **Allow Incoming CLI Modification** check box for the respective trunk and the **Enable Incoming CLI Modification** check box — are enabled. To know more, refer to "Configuring BRI Trunks", "Configuring CO Trunks", "Configuring E&M Lines", "Configuring Mobile Trunks", "Configuring SIP Trunks", "Configuring T1 Trunks" and "Configuring E1 Trunks".*

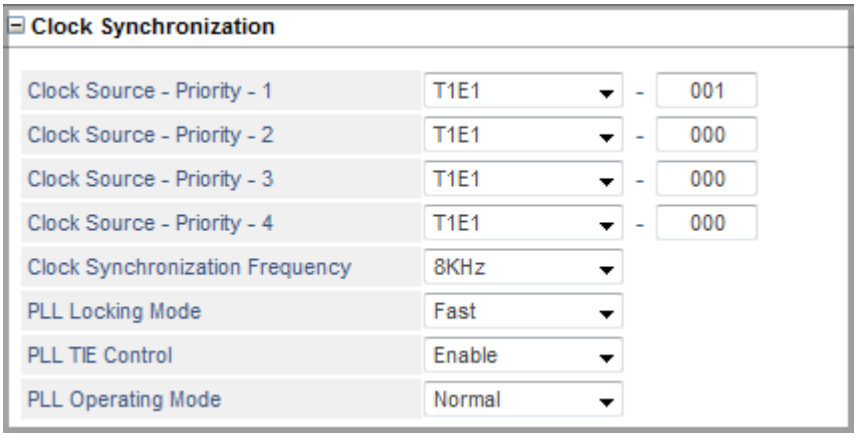
- To apply Incoming CLI Modification, select the **Enable Incoming CLI Modification** check box.
- Configure the following options for CLI modification:
  - **Country Code:** This is the Country Code of the country where SARVAM UCS is installed. The Country Code helps SARVAM UCS detect whether the Incoming CLI received is a national or an international number. Do not enter any prefix such as '+' or '00' for the Country Code. Default: '91' (India).
  - **Area Code:** This is the Area Code of the place where the SARVAM UCS is installed. The Area Code helps SARVAM UCS detect whether the Incoming CLI received is a local number. Do not enter any prefix for the Area Code. For example, if you want to enter Area Code for Mumbai, enter only '22'. Do not enter the prefix '0' to the area code. Default: '265' (Vadodara city).
  - **International Prefix***:* These are digits required as Prefix for dialing International Numbers. The prefix may be up to 5 digits, with numbers from 00000 to 99999. Default: '00'.
  - **National Prefix:** These are digits required as Prefix for dialing long distance, National (within the country) numbers. The prefix may be up to 5 digits, with numbers from 00000 to 99999. Default: '0'.
  - **Area Code required to make local calls?:** Depending on the dialing pattern of your local public telephone network, you may choose:
    - ***No** (Area Code not required),* if your public telephone network does not require the dialing of Area Code for local numbers.

- ***Yes*** *(Area Code is required),* if your public telephone network requires you to dial the Area Code for local numbers.

- ***Yes, with Prefix Digit****,* if your public telephone network requires you to dial Area Code with a particular Prefix for local numbers. If you select this option, you must also enter the prefix digits for the area code for local calls in the **Prefix Area Code** field.

Clock Synchronization

- Click **Clock Synchronization** to expand.



The SARVAM UCS supports four clock sources for Clock Synchronization for the E1-PRI and BRI ports. To know more, see “Clock Synchronization”.

- You must select the **Clock Source** in the order of Priority from 1 to 4. By default, the Clock Source priority is selected as follows:
  - Clock Source - Priority 1 - T1E1 - 001
  - Clock Source - Priority 2 - T1E1 - 002
  - Clock Source - Priority 3 - T1E1 - 003
  - Clock Source - Priority 4 - T1E1 - 004

- Also set the **Clock Synchronization Frequency** as: 8KHz Derived, 8KHz, 2.048MHz, 1.54 MHz Default: 8 KHz

- Set the **PLL Locking Mode** as **Slow** or **Fast**. Default: Fast.

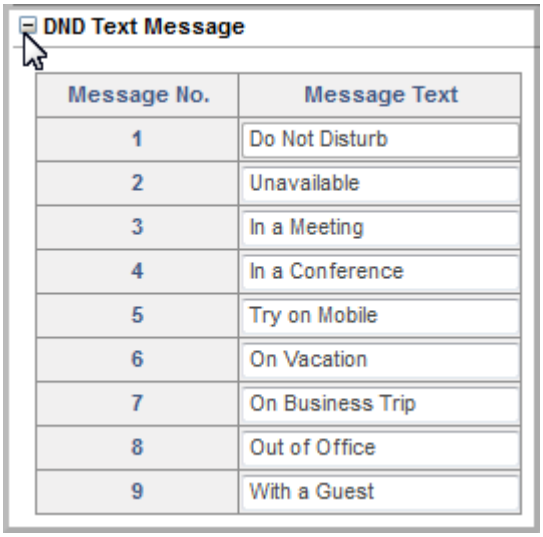
- Set the **PLL TIE Control** as **Enable** or **Disable**. Default: Enable.

- Select the **PLL Operating Mode** as **Normal**, **Hold Over** or **Free Run**. Default: Normal.

You can program PLL TIE Control and PLL Operating Mode using Jeeves only.

DND Text Message

- Click **DND Text Message** to expand.



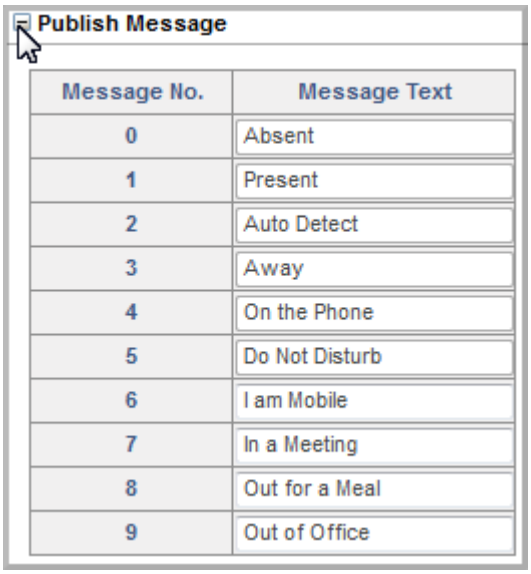
SARVAM UCS allows you to program DND different Text Messages, which extension users can select when setting DND on their extensions. The DND text message they select is displayed to the calling extensions. See “Do Not Disturb (DND)” to know more.

The default DND Text messages appear on your screen. You may customize messages 2 to 9 according to your requirement. Text Message can be of maximum 16 alphanumeric characters. All ASCII characters except < > and “ (double quote) are allowed.

Message No.	Message Text
1	Do Not Disturb
2	Unavailable
3	In Meeting
4	In Conference
5	Try on Mobile
6	On Vacation
7	On Business Trip
8	Out of Office
9	With a Guest

Publish Message

- Click **Publish Message** to expand.



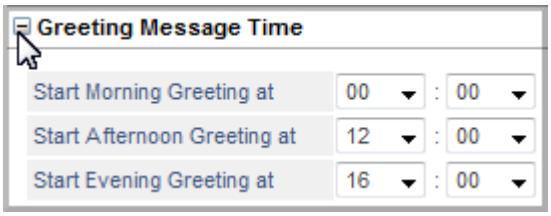
SARVAM UCS offers 10 different text Messages to Publish Message, as listed in the table below. You can customize message 6 to 9 to match your requirement. Text Message can be of maximum 16 alphanumeric characters. All ASCII characters except < > and “ (double quote) are allowed.

Message No.	Message Text
0	Absent
1	Present
2	Auto Detect
3	Away
4	On the Phone
5	Do Not Disturb
6	I am Mobile
7	In Meeting
8	Out for Meal
9	Out of Office

To know more about this feature, see “Presence”.

Greeting Message Time

- Click **Greeting Message Time** to expand.



- When **Auto Attendant** is enabled on trunks, the greeting messages are played to the callers according to the time of the day, morning, afternoon, evening.
    - If **Built-In Auto Attendant** is enabled on trunks, the system answers the call and plays the greeting message as per the voice modules.
    - If **Voice Mail Auto Attendant** is enabled on trunks, the Voice Mail System answers the call and plays the greeting message.

      To know more about this feature see “Auto Attendant”.

- You can set the desired Start Time for Morning, Afternoon and Evening greetings.

  In **Start Morning Greeting at** set the start time for the Morning Greeting Message. Similarly, in **Start Afternoon Greeting at** and **Start Evening Greeting at** set the start time for the Afternoon and Evening Greeting Message respectively.

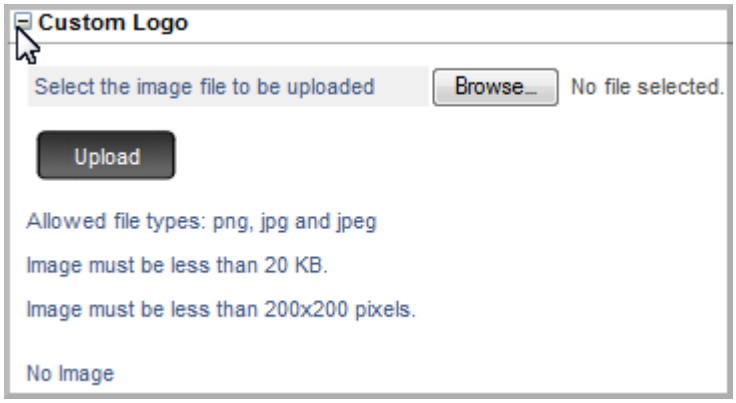
  The time must be in HH:MM format. The valid range for Hours (HH) is 00 to 23 and for Minutes (MM) is 00 to 59. By default the time in **Start Morning Greeting at** is set to 00:00, **Start Afternoon Greeting at** is set to 12:00 and **Start Evening Greeting at** is set to 16:00.

  The system plays the Morning Greeting Message between the Morning and Afternoon Greeting Start time, the Afternoon Greeting Message between the Afternoon and Evening Greeting Start time and the Evening Greeting Message between the Evening and Morning Greeting Start time.

As the system plays the Evening Greeting Message between the Evening and Morning Greeting Start time, to prevent the Evening Greeting Message from being played after midnight, you are recommended to set the Morning Greeting Start time to 00:00 hrs.

Custom Logo

- Click **Custom Logo** to expand.



For VARTA Mobile UC Clients — Android, iPhone — SARVAM UCS supports uploading of any customized logo. This logo shall appear on the Home screen of the VARTA ADR100 and VARTA AMP100.

Make sure the image you want to upload fulfils the following:
- The image must be a JPEG, JPG or PNG file
- Maximum Dimension of the image : 200 x 200px
- Maximum File Size of the image : 20KB

To upload the file,

- Click the **Choose File** button to **Select the image file to be uploaded** from the location on the local disk.
- Click the **Upload** button.

  A preview of the image you uploaded appears.
- If you wish to remove the image, click the **Remove** button.

If you upload another image directly without removing the previous image, the system will automatically overwrite the image.

- You may log out of Jeeves.

Configuring System Parameters using a Telephone

- Enter SE mode from a DKP/SLT.

Customer Name

- To enter Customer Name, dial:
  - ***5401-Customer Name-#****
- To clear the Customer Name, dial:
  - ***5401-#****

System Parameters

- To select Customer Profile, dial:
  - ***5315-Code***
    Where, Code is
    1 for Enterprise Mode
    2 for Hotel Mode
    Default: Enterprise Mode.

- To enable/disable On-site Configuration flag, dial:
  - ***2108-Flag***
    Where, flag is
    1 for Enable
    0 for Disable
    Default: Disable.

- To change Station Name Pattern, dial:
  - ***3615-Station Name Pattern Code***
    Where,
    Station Name Pattern Code is
    1 for Title <space> First Name <space> Name
    2 for First Name only
    3 for Name only
    4 for First Name <space> Name
    5 for Title<space>First Name
    6 for Title<space>Name
    Default: Name only

- To select Default Call Hold Type, dial:
  - ***5318-Hold Type***
    Where,
    Hold Type is
    1 for Exclusive Hold
    2 for Global Hold
    Default: Exclusive Hold.

- To enable/disable Log Internal Calls in Missed Calls, dial:
  - ***5361-Code***
    Where,
    Code is
    0 for Disable (Do not store internal calls in "Missed Calls" log)
    1 for Enable (Store internal calls in "Missed Calls" log)
    Default: Enable.

  To enable/disable Log Internal Calls in Answered Calls, dial:
  - ***5362-Code***
    Where,
    Code is
    0 for Disable (Do not store internal calls in "Answered" log)
    1 for Enable (Store internal calls in "Answered Calls" log)
    Default: Enabled.

  To enable/disable Log Internal Calls in Dialed Calls, dial:
  - ***5363-Code***

Where,
Code is
0 for Disable (Do not store internal calls in "Dialed Calls" log)
1 for Enable (Store internal calls in "Dialed Calls" log)
Default: Enabled.

- To select the type of music to be played when stations are kept on hold, dial:
  - ***3552-Code***
    Where,
    Code is
    1 for Voice Module 01
    Default: 1.

- To select the type of music to be played when trunks are kept on hold, dial:
  - ***3553-Code***
    Where,
    Code is
    1 for Voice Module 01
    Default: 1.

If all the extensions of the Routing Group you selected for Alarm Notification type are busy, the extension user will be played MoH (MoH can be Voice Module 01).

- To enable/disable OFF Hook Alert flag, dial:
  - ***5333-Code***
    Where, Code is
    1 for Enable
    2 for Disable
    Default: Enable

- To set the system in Day/Night mode, dial:
  - ***4801-Code***
    Where,
    Code is from 1 to 4.
    1 is for Day Mode.
    2 is for Night Mode.
    3 is for Operate system as per Time Table.
    4 is for Break Hours Mode.

- To enable/disable emergency reporting, dial:
  - ***5110-Code***
    Where,
    Code is 0 for Disable and 1 for Enable.
    Default: Enable

- To enable/disable Replace '+' from CLI, dial:
  - ***5334-Code***
    Where,
    Code is
    0 for Disable
    1 for Enable.
    Default: Disable

- To configure number string to be used as Replacement for '+' in CLI, dial:
  - ***5335-Replacement string***
    Where,
    Replacement string is any digit string of up to a maximum of 6 digits, from 0 to 9. Default: blank.

- To enable/disable the Disconnect when Caller Doesn't Dial a Digit' flag, dial:
  - ***5338-Code***
    Where, Code is
    0 for Disable.
    1 for Enable
    Default: Disable.

- To enable/disable the Disconnect Built-In Auto Attendant call, when Dialed Number Busy, dial:
  - ***5336-Code***
    Where, Code is
    0 for Disable.
    1 for Enable
    Default: Disable.

- To enable/disable Disconnect Built-In Auto Attendant Call, when Dialed Number does Not Reply, dial:
  - ***5337-Code***
    Where, Code is
    0 for Disable.
    1 for Enable
    Default: Disable

  To enable/disable Beep when Conference/Dial-In Conference, Raid starts, dial:
  - ***5331-Flag***
    Where,
    Flag is
    0 for Disable
    1 for Enable
    Default: Enable

- To enable/disable Beep when Call Taping and Conversation Recording starts, dial:
  - ***5332-Flag***
    Where,
    Flag is
    0 for Disable
    1 for Enable
    Default: Enable

- To disable/enable Feature Tone in place of Dial Tone when Call Forward is set, dial:
  - ***5312-Feature Tone Flag***
    Where,
    Feature Tone Flag is
    0 for Disable
    1 for Enable
    Default: Enable

- To select Call Proceeding Tone Type - Multistage Dialing, dial:
  - ***5311-Call Proceeding Tone Type***
    Where,

Call Proceeding Tone Type is
1 for Network Tone
2 for Pseudo Tone
3 for Silent
Default: Network Tone

- To select the Companding Algorithm, dial:
  - ***5322-Companding type***
    Where
    Companding type is
    1 is for A-Law
    2 is for μ-Law:
    Default: A-Law.

- To select a Language for SE, SA and Front Desk User Web Interface, dial:
  - ***5319-Language***
    Where,
    Language is
    1 for English
    2 for French
    3 for German
    4 for Spanish
    5 for Portuguese
    6 for Italian
    Default: English (country specific)

- To enable/disable Form Feed:
  - ***5321-Code***
    Where,
    Code is
    0 for Disable Form Feed.
    1 for Enable Form Feed.
    Default: Enable Form feed.

- To program minimum number of digits received in CLI to consider as call from the Public Network, dial:
  - ***5314-Minimum Caller ID Digits***
    Where,
    Minimum Caller ID Digits is from 01 to 16.
    Default:8

- To enable/disable 'Display Presence Status during Call on DKP', dial:
  - ***5320-Flag***
    Where,
    Flag is
    0 for Disable
    1 for Enable
    Default: Disable.

- To enable/disable Watch Dog function, dial:
  - ***5309-Flag***
    Where,
    Flag is
    1 for Enable

0 for Disable

- To enable/disable VMS Alarm Verification, dial:
  - ***2211-Code-#****
    Where,
    Where,
    Code is
    1 for Enabled
    0 for Disabled.

Magneto

To enable or disable the "Enable Silence Detection on Magneto?" flag, dial:

- ***5357-Flag***
  Where,
  Flag is
  1 for Enable
  0 for Disable
  Default: Enabled.

To set Magneto-Silence Detection Timer, dial:

- ***5356-Silence Detection Timer***
  Where,
  Silence detection timer value range is from 001 to 255 seconds.
  Default: 60 seconds.

To set the Magneto Threshold Level, dial:

- **5358 - Magneto VAD Threshold Level**
  Where,
  The value of the Level is from 0 to -96.
  Default: -25 dBm

Alarm

To configure Alarm Ring Timer, dial:

- ***2201-Seconds***
  Where,
  Seconds is from 001 to 255. Default: 45 seconds.

To configure Number of Alarm Attempts, dial:

- ***2202-Number of Alarm Attempts***
  Where,
  Number of Alarm Attempts is from 1 to 9. Default: 3.

To configure Alarm Attempt Interval, dial:

- ***2203-Alarm Attempt Interval***
  Where,
  Alarm Attempt Interval is from 1 to 9 minutes. Default: 5 minutes.

To configure Snooze function, dial:

- ***2204-Snooze***
  Where,
  Snooze is 0 or 1.
  Select '0' to disable Snooze and '1' to enable snooze.

To disable/enable Configurable Alarm Type, dial:

- ***2208-Flag***
  Where,
  Flag is 0 for Disable, 1 for Enable1. Default: Enabled.

To disable/enable Configurable Alarm Category, dial:

- ***2209-Flag***
  Where,
  Flag is 0 for Disable, 1 for Enable. Default: Enabled

To configure Alarm Notification Type, dial:

- ***5602-1-Template Number-12-Alarm Notification Type***
  Where,
  Template Number is Station Advanced Feature Template from 01 to 50.
  Default: 01.
  Alarm Notification Type is:
  1 for Music on Hold
  2 for Voice Message (Voice Modules)
  3 for Routing Group
  4 for Voice Mail

Incoming CLI Modification

- To enable/disable Incoming CLI Modification, dial:
  - ***5367-Flag***
    Where,
    Flag is
    0 for Disable
    1 for Enable
    Default: Disabled

- To program the Country Code, dial:
  - ***5368-Country Code***
    Where,
    Country Code is a number string up to 5 digits. The number string may consist of numbers from 00000 and 99999.
    Default: 91 (India)

    Program the Country Code without any prefix. For example, if you want to program USA as Country Code, enter ‘1’ only (without the prefix ‘00’ or ‘+’)

- To program the Area Code, dial:
  - ***5369-Area Code***
    Where,
    Area Code is a number string up to 5 digits. The number string may consist of numbers from 00000 and 99999.
    Default: 265 (Evacuator)

    Program the Area Code without any prefix. For example, if you want to program Mumbo as Area Code, enter ‘22’ only (without the prefix ‘0’)

- To program International Call Prefix, dial:
  - ***5370-International Call Prefix***
    Where,

International Call Prefix is a number string up to 5 digits. The number string may consist of numbers from 00000 and 99999.
Default: 00

- To program National Call Prefix, dial:
  - ***5371-National Call Prefix***
    Where,
    National Call Prefix is a number string up to 5 digits. The number string may consist of numbers from 00000 and 99999.
    Default: 0

- To enable/disable 'Area Code required to make local calls?', dial:
  - ***5372-Code***
    Where,
    Code is
    1 for No (Area Code is not required)
    2 for Yes (Area Code is required)
    3 for Yes (Area Code with Prefix required)
    Default: 1

- To program Prefix Digits to Area Code for Local Calls, dial:
  - ***5373-Prefix Digits***
    Where,
    Prefix Digits to Area Code is a number string up to 5 digits. The number string may consist of numbers from 00000 and 99999.
    Default: Blank

Clock Synchronization

- To select clock source, dial:
  - ***5341-Clock Source Index-Port Type-Port Offset***
    Where,
    Clock Source Index is 1 to 4.'
    Port Offset is 01 to 32.
    Clock Source Index is from 1 to 4.

Port Type	Meaning	Port Offset
05	T1E1	01-08
04	BRI	01-32
00	Null	000

0 is a valid port to follow internal clock.

Default:

Clock Source Index	Port Type-Port Offset
1	T1E1-1
2	T1E1-2

Clock Source Index	Port Type-Port Offset
3	T1E1-3
4	T1E1-4

- To select System Clock Synchronization, dial:
  - ***5342-System Clock Synchronization***
    Where,
    System Clock Synchronization from 1 to 4.

System Clock Synchronization	Meaning
1	8 KHz Derived
2	8 KHz
3	2.048 MHz
4	1.54 MHz

Default: 8 MHz

- To select PLL Locking Mode, dial:
  - ***5343-PLL Locking Mode***
    Where,
    PLL Locking Mode is
    1 for Fast
    2 for Slow
    Default: Fast.

Publish Message

You can customize Publish Messages using Jeeves only.

System Timers and Counts

What's this?

Several features of the SARVAM UCS are based on Timers and Counts. For example, how long and how many times an extension should ring when Message Wait is set, or how long the Busy Tone, the Ring Back Tone, or the Error Tone should be played to an extension. SARVAM UCS allows you to configure most of these Timers and Counts to suit your requirement. Listed below are the Timers and Counts related to the various features and facilities, along with a brief description and default value of each.

Auto Redial

Name	Description	Range	Default
Auto Redial - Ring Back Tone Wait Timer (sec.)	The time for which system waits to sense the Ring Back Tone from the PSTN/CO Network after dialing the requested number.	0 to 255	60 seconds
Auto Redial - Ring Timer (sec.)	The time for which the extension that has requested Auto Redial should ring.	0 to 255	45 seconds
Auto Redial - Normal Timer (sec.)	The time interval between auto redial attempts when Auto Redial 'Normal' is set.	0 to 255	45 seconds
Auto Redial - Normal Count	The number of auto redial attempts the system will make when Auto Redial 'Normal' is set.	0 to 255	5 tries
Auto Redial - Priority Timer (sec.)	The time interval between auto redial attempts when Auto Redial 'Priority' is set.	0 to 255	10 seconds
Auto Redial - Priority Count	The number of auto redial attempts the system will make when an extension having the feature Auto Redial Priority in its Class of Service uses Auto Redial 'Priority'.	0 to 255	20 attempts

Built-In Auto Attendant

Name	Description	Range	Default
Built-In Auto Attendant Inactivity Timer (sec.)	The time after which the system releases the trunk, if the caller has not dialed any digit, or when a Built-In Auto Attendant or Trunk Auto Answer call is not answered by the landing destination.	0 to 255	60 seconds
Built-In Auto Attendant Answer Wait Timer (sec.)	The time for which the system waits before answering a Built-In Auto Attendant call.	0 to 255	5 seconds
Built-In Auto Attendant Music Timer (sec.)	The time for which the system plays music after answering the Built-In Auto Attendant call.	0 to 255	5seconds
Built-In Auto Attendant Beeps Timer (sec.)	The time for which the system plays beeps to the caller to prompt the caller to dial the desired extension number when the call is answered by the Built-In Auto Attendant.	0 to 255	10 seconds

Name	Description	Range	Default
Built-In Auto Attendant Ring Timer (sec.)	The time for which the system rings on the landing destination extension in a Built-In Auto Attendant call.	0 to 255	30 seconds
Built-In Auto Attendant Busy Tone Timer (sec.)	In a Built-In Auto Attendant call, the time for which the system plays Busy Tone, if the dialed extension is busy.	0 to 255	15 seconds
Built-In Auto Attendant Error Tone Timer (sec.)	In a Built-In Auto Attendant call, the time for which the system plays Error Tone to the caller, if the caller has dialed an invalid code.	0 to 255	5 seconds

Call Progress Tones

Name	Description	Range	Default
Dial Tone Timer (sec.)	The time for which the system plays the Dial tone.	2 to 255	7 seconds
Ring Back Tone Timer (sec.)	The time for which the system plays the Ring Back Tone.	1 to 255	45 seconds
Busy Tone Timer (sec.)	The time for which the system plays the Busy Tone.	1 to 255	7 seconds
Error Tone Timer (sec.)	The time for which the system plays the Error Tone.	1 to 255	30 seconds
Feature Confirmation Tone Timer (sec.)	The time for which the system plays the Confirmation Tone when a feature is set or canceled.	1 to 255	7 seconds
Programming Error Tone Timer (sec.)	The time for which the system plays the Error Tone when you have entered an invalid command string while configuring a feature from a phone.	1 to 255	3 seconds
Programming Confirmation Tone Timer (sec.)	The time for which the system plays the Confirmation Tone when a system command is successfully executed when configuring the system from a phone.	1 to 255	3 seconds
Tone Demo Timer (sec.)	The time for which the system plays Call Progress Tone when you are demonstrating the tone.	1 to 255	30 seconds
Call Forward - No Reply Timer for Department Group (sec)	The time for which the system will wait for an extension (Department Group member) to answer an incoming call, before forwarding the call to the programmed destination.	1 to 255	30 seconds

Direct Inward System Access (DISA)

Name	Description	Range	Default
DISA Idle State Timer (sec.)	In a DISA PIN Authentication call, the time for which the system waits for the caller to go Off-hook after entering DISA. If the caller does not go Off-hook within this timer, the system releases the call.	0 to 255	20 seconds

Name	Description	Range	Default
DISA Inactivity Timer (min.)	In a DISA call, the time for which the system waits for the caller to dial digits. If the caller does not dial any digit within this timer, the system disconnects the call. This timer is applicable only for Analog Trunks.	0 to 255	2 minutes

Message Notification

Name	Description	Range	Default
Message Notification Retry Count	This count defines the number of times the system must make Message Wait Notification Calls to the destination number.	0 to 15	3
Message Notification Ring Timer (sec)	The time for which the extension on which the Message Wait Notification Call is made will ring.	0 to 255	45 seconds
Message Notification Interval (min)	This defines time interval between two retries. It is the time after which the system must make another attempt to place the notification call on the destination number.	1 to 255	5 minutes

Other Features

Name	Description	Range	Default
Auto Call Back Ring Timer (sec.)	The time for which the extension requesting the Auto Call Back and the destination extension will ring.	1 to 255	30 seconds
Interrupt Request Timer (sec.)	The time for which the extension on which the Interrupt Request is made will get the beeps.	1 to 255	45 seconds
Barge-In Timer (sec.)	The time after which the extension that has activated Barge-In gets connected to the extension which is barged in.	1 to 255	10 seconds
Trunk Reservation Timer (min.)	The time for which a trunk remains reserved for the extension that has reserved the trunk.	1 to 255	10 minutes
Transfer while Ringing Timer (sec.)	When an extension transfers a call to another extension after it starts ringing, this is the time for which the system will wait for the transfer target extension to answer the call. If the transfer target does not answer the call within this timer, the call is returned to the transferror.	1 to 255	30 seconds
Transfer on Busy Timer (sec.)	When a call is transferred to a Busy extension, this is the time for which beeps are played on the transfer target extension.	1 to 255	30 seconds
Trunk to Trunk Inactivity Timer (min.)	In a Trunk-to-Trunk call, this is the time for which the system waits after call maturity for any digit to be dialed. If no digit is dialed within this timer, the system drops the call.	1 to 255	2 minutes

Name	Description	Range	Default
Call Park Timer (min.)	The time after which the call comes back to the extension that has parked the call.	2 to 255	2 minutes
Call Park Release Timer (min.)	The time after which the parked call gets disconnected.	1 to 255	3 minutes
Live Call Screening (sec.)	The time for which the speaker of the DKP/ Extended IP Phone extension remains ON while the message from the caller is being recorded.	1 to 255	10 seconds
Message Wait Ring Count	It is the Number of times the extension should ring after the Message Wait is set on an extension. This count is applicable only when 'Ring' is selected as the Message Wait Notification type for the extension.	0 to 255	10 attempts
Message Wait Ring Timer (sec.)	The time for which the extension rings to indicate that Message Wait is set for the extension. This timer is applicable only when 'Ring' is selected as the Message Wait Notification type for the extension.	1 to 255	30 seconds
Message Wait Ring Interval Timer (min.)	The time after which the extension should ring again to indicate Message Wait is set. This timer is applicable only when 'Ring' is selected as the Message Wait Notification type for the extension.	1 to 255	30 minutes
Conflict Dialing Timer (sec.)	The time for which the system waits for the extension user to dial the next digit to resolve conflicting access codes dialed by the extension user.	1 to 255	2 seconds
Extension - Inter Digit Wait Timer (sec.)	The time for which the system waits for the extension user to dial the next digit. On the expiry of this timer, the system considers it as the end of number dialing.	2 to 255	7 seconds
SA Command - Inter Digit Wait Timer (sec.)	When the user dials SA Commands, the system waits for the SA Command - Inter Digit wait timer for the user to dial the next digit. On the expiry of this timer, the system considers it as end of number dialing and proceeds further with executing the command.	2 to 255	15 seconds
Trunk - First Digit Wait Timer (sec.)	the time for which the system waits for the extension user to dial the first digit, after grabbing the trunk.	1 to 255	25 seconds
Trunk - Inter Digit Wait Timer (sec.)	When an extension user has grabbed the trunk and is dialing a number, the system waits for the Trunk-Inter-Digit wait timer for the extension user to dial the next digit. On the expiry of this timer, the system considers it as end of number dialing and proceeds with the call.	1 to 255	3 seconds

Name	Description	Range	Default
Global Hold Retrieval Timer (sec.)	This is the time for which a call put on Global Hold remains connected in the system. If the call put on Global Hold is not retrieved within this timer, the call is returned to the DKP/Extended IP Phone which put it on hold.	1 to 999	60 seconds
Exclusive Hold Retrieval Timer (min.)	This is the time for which a call put on Exclusive Hold remains connected in the DKP/Extended IP Phone. If the call put on Exclusive Hold is not retrieved within this timer, the call is returned to the DKP/Extended IP Phone which put it on hold.	1 to 255	2 minutes
RCOC Record Delete Timer (min.)	This is the time for which the record of the outgoing call is stored in the RCOC Table. The timer is activated whenever a record is stored in the RCOC table. At the end of this timer, the system deletes this record from the table.	1 to 999	999 seconds
Release Conference if Idle for more than (min.)	This is the time for which the system will wait for participants of a Dial-In Conference to withdraw or release themselves from the conference, one-by-one. On the expiry of this timer, the system releases the Dial-In Conference and frees the resource occupied by this conference in the conferencing circuit.	1 to 255	2 minutes
Watchdog Refresh Wait Timer (sec.)	This is the time within which the system should signal to the Watchdog. If the system does not send signal to the Watchdog within this Timer, the watchdog device will restart SARVAM UCS.	1 to 255	60 seconds
Line Echo Cancellation Start Timer (sec.)	The time after which the SARVAM UCS sends the Line Echo Cancellation start request to the DKP port.	1 to 255	45 seconds
Retry Counts for Authority Code	When the user enters a wrong Authority Password, this count defines the number of times the system allows the user to re-enter the password.	0-9	3 attempts
Emergency Reporting Call - Ring Timer (min)	The time for which the Operators extension rings, to inform the operator the number of the extension that dialed out an emergency number.	1 to 255	10 minutes
Held Call Disconnection Timer (min)*	The time after which a held call will be disconnected, if not retrieved.	1 to 255	5 minutes
Conference - Assistant Present Beep Interval (sec)	The time after which you want the system to play beeps to indicate the presence of the Operator in the conference.	1 to 255	5 seconds
Radio Port Inactivity Timer (sec)	Set the duration of the Radio Port Inactivity Timer. When the Radio Port is in speech with another port and if no activity is detected on the Radio Port for the set duration, SARVAM UCS releases the Radio Port.	15 to 999	60 seconds

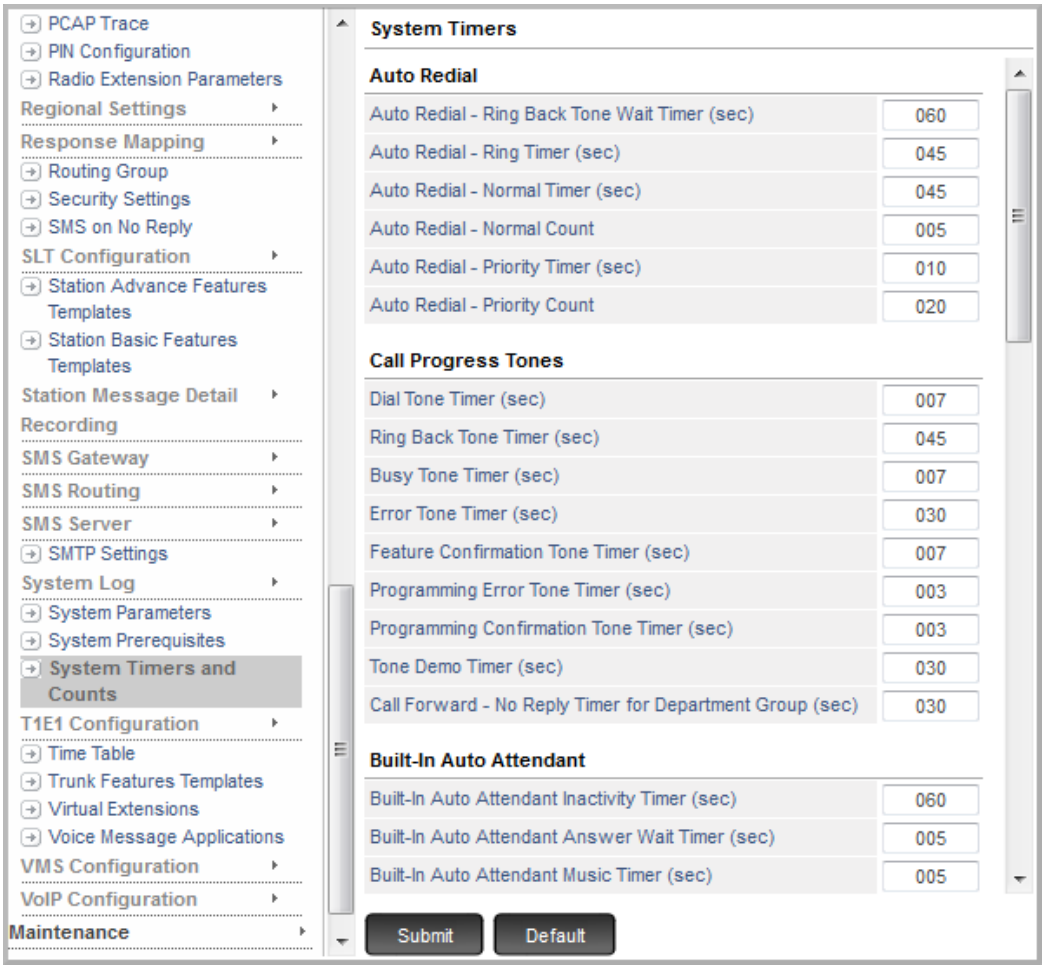
Name	Description	Range	Default
VARTA Client Inactivity Timer (days)	The time for which the server will send Push Notifications to the VARTA application. When the timer expires, the server will consider the application as unregistered and will stop sending Push Notifications to the application.	1 to 255	10 days

**Applicable for Matrix SPARSH VP330, SPARSH VP210 and Matrix VARTA Mobile UC Clients.*

How to configure

Configuring Timers and Counts using Jeeves

- Log into Jeeves as System Engineer.
- Under **Configuration**, click **System Timers and Counts**.



The Timers and Counts on this page are arranged by the name of the feature or function these are related to.

- Change the value of the Timer or Count by entering the desired duration or count in the respective fields.

- Click **Submit** to save settings.
- You may log out.

Configuring Timers and Counts using a Telephone

- Enter SE mode from a DKP/SLT.

Auto Redial

To set Auto Redial Normal - Timer, dial:

- ***1704-Seconds***
  Where,
  Seconds is from 000 to 255.

To set Auto Redial Normal - Count, dial:

- ***1705-Count***
  Where,
  Count is from 000 to 255.

To set Auto Redial - Priority, dial:

- ***1706-Seconds***
  Where,
  Seconds is from 000 to 255.

To set Auto Redial - Count, dial:

- ***1707-Count***
  Where,
  Count is from 000 to 255.

To change Auto Redial RBT Wait Timer, dial:

- ***1702-Seconds***
  Where,
  Seconds is from 000 to 255 seconds.

Do not set this timer to less than 2 seconds.

To change Redial Ring Timer, dial:

- ***1703-Seconds***
  Where,
  Seconds is from 000 to 255 seconds.

Do not set this timer to less than 2 seconds.

Call Progress Tones

To set Dial Tone Timer, dial:

- ***3502-Seconds***
  Where,
  Seconds is from 002 to 255.
  Default: 007 seconds

To set Ring Back Tone Timer, dial:

- ***3503-Seconds***
  Where,
  Seconds is from 001 to 255 seconds.
  Default: 045 seconds.

To set Busy Tone Timer, dial:

- ***3504-Seconds***
  Where,
  Seconds is from 001 to 255 seconds.
  Default: 007 seconds.

To set Error Tone Timer, dial:

- ***3505-Seconds***
  Where,
  Seconds is from 001 to 255 seconds.
  Default: 030 seconds.

To set Confirmation Tone Timer, dial:

- ***3506-Seconds*** to program the Confirmation Tone Timer.
  Where,
  Seconds is from 001 to 255 seconds.
  Default: 007 seconds.

To set Programming Confirmation Tone Timer, dial:

- ***3509-Seconds***
  Where,
  Seconds is from 001 to 255 seconds.
  Default: 003 seconds.

To set Programming Error Tone Timer, dial:

- ***3508-Seconds***
  Where,
  Seconds is from 001 to 255 seconds.
  Default: 003 seconds.

To set Tone Demo Timer[342], dial:

- ***3542-Seconds***.
  Where,
  Seconds is from 001 to 255.
  Default: 030 seconds.

Built-In Auto Attendant

To set Built-In Auto Attendant Inactivity Timer, dial:

- ***2411-Seconds***
  Where,
  Seconds is from 001 to 255 seconds.
  Default: 060 seconds.

To set Built-In Auto Attendant Answer Wait Timer, dial:

342. Time for which the system demonstrates the tone/ring to the user.

- ***2412-Seconds***
  Where,
  Seconds is from 001 to 255 seconds.
  Default: 005 seconds.

To set Built-In Auto Attendant Music Timer, dial:

- ***2413-Seconds***
  Where,
  Seconds is from 001 to 255 seconds.
  Default: 005 seconds.

To set Built-In Auto Attendant Beeps Timer, dial:

- ***2414-Seconds***
  Where,
  Seconds is from 001 to 255 seconds.
  Default: 010 seconds.

To set Built-In Auto Attendant Ring Timer, dial:

- ***2415-Seconds***
  Where,
  Seconds is from 001 to 255 seconds.
  Default: 030 seconds.

To set Built-In Auto Attendant Busy Tone Timer, dial:

- ***2416-Seconds***
  Where,
  Seconds is from 001 to 255 seconds.
  Default: 015 seconds.

To set Built-In Auto Attendant Error Tone Timer, dial:

- ***2417-Seconds***
  Where,
  Seconds is from 001 to 255 seconds.
  Default: 005 seconds.

DISA

To set DISA Idle State Timer, dial:

- ***2420-Seconds***
  Where,
  Seconds is from 001 to 255 seconds.
  Default: 020 seconds.

To set DISA Inactivity Timer, dial:

- ***2421-Minutes***
  Where,
  Minutes is from 001 to 255 minutes.
  Default: 002 minutes.

Other Features

To set Auto Call Back Ring Timer, dial:

- ***3801-Seconds***

Where,
Seconds is from 001 to 255 seconds.
Default: 030 seconds.

To set Interrupt Request Timer, dial:

- ***3802-Seconds***
  Where,
  Seconds is 001 to 255 seconds.
  Default: 045 seconds.

To set Barge-In Timer, dial:

- ***3803-Seconds***
  Where,
  Seconds is 001 to 255 seconds.
  Default: 010 seconds.

To set Trunk Reservation Timer, dial:

- ***3804-Minutes***
  Where,
  Minutes is 001 to 255 seconds.
  Default: 010 minutes.

To set Transfer While Ringing Timer, dial:

- ***3806-Seconds***
  Where,
  Seconds is from 001 to 255 seconds.
  Default: 030

To set Transfer on Busy Timer, dial:

- ***3807-Seconds***
  Where,
  Seconds is from 001 to 255 seconds.
  Default: 030

To set the Trunk-to-Trunk Inactivity Timer, dial:

- ***3808-Minutes***
  Where,
  Seconds is from 001 to 255 minutes.
  Default: 2 minutes.

To set Call Park Timer, dial:

- ***3809-Minutes***
  Where,
  Minutes is from 002 to 255 minutes.
  Default: 2 minutes.

To set Call Park Release Timer, dial:

- ***3810-Minutes***
  Where,
  Minutes is from 001 to 255 minutes.
  Default: 003 minutes.

To set Live Call Screening Timer, dial:

- ***3811-Seconds***
  Where,
  Seconds is from 001 to 255 seconds.
  Default: 010 seconds.

To set Message Wait Ring Count, dial:
- ***4403-Seconds***
  Where,
  Seconds is from 001 to 255 seconds.
  Default: 010.

To set Message Wait Ring Timer, dial:
- ***4404-Seconds***
  Where,
  Seconds is from 001 to 255 seconds.
  Default: 030 seconds.

To set Message Wait Ring Interval, dial:
- ***4405-Minutes***
  Where,
  Seconds is from 001 to 255 minutes.
  Default: 030 minutes.

To set Conflict Dialing Timer, dial:
- ***5351-Seconds***
  Where,
  Seconds is from 001 to 255 seconds.
  Default: 002 seconds.

To set Station - Inter Digit Wait Timer, dial:
- ***5352-Seconds***
  Where,
  Seconds is from 001 to 255 seconds.
  Default: 007 seconds.

To set Station - Inter Digit Wait Timer, dial:
- ***5352-Seconds***
  Where,
  Seconds is from 001 to 255 seconds.
  Default: 007 seconds.

To set Trunk - First Digit Wait Timer, dial:
- ***5353-Seconds***
  Where,
  Seconds is from 001 to 255 seconds.
  Default: 025 seconds.

To set Trunk - Inter Digit Wait Timer, dial:
- ***5354-Seconds***
  Where,
  Seconds is from 001 to 255 seconds.
  Default: 003 seconds.

To set Global Hold Retrieval Timer, dial:
- ***3805-Seconds***
  Where,

Seconds is from 001-999 Seconds.
Default: 060 seconds.

To set Exclusive Hold Retrieval Timer, dial:

- ***3812-Minutes***
  Where,
  Minutes is from 001- 255 Minutes.
  Default: 2 minutes.

To set the RCOC Record Delete Timer, dial:

- Dial ***3521-Minutes***
  Where,
  Minutes are from 001 to 999
  Default: 999 minutes.

To set Release Conference if idle for more than Timer, dial:

- ***5355-Minutes***
  Where,
  Seconds is from 001 to 255 minutes.
  Default: 002 minutes.

To set Watch Dog Refresh Wait Timer, dial:

- ***5316-Seconds***
  Where,
  Seconds is from 001 to 255 minutes.
  Default: 060 seconds.

- Exit SE mode.

System Security

What's this?

Access to the SARVAM UCS is secured at three levels by way of a password:

- at the System Engineer Level with the System Engineer (SE) password.
- at System Administrator Level with the System Administrator (SA) password.
- at the User Level with the User Password.

The System Engineer and the System Administrator passwords secure the system settings from access and alteration by unauthorized persons (anyone other than the System Engineer and the System Administrator), thus preventing possible misuse of the features and facilities.

System Engineer (SE) Password

For Accessing Jeeves

The SE password is a code used to prevent unauthorized access and alterations or misuse of the features and facilities. As this password is meant for restricting access to the SE mode, we strongly recommend you to:

- Keep the password secret.
- Select a complex password that cannot be easily guessed.
- Change the password regularly
- Not use the **"Remember Password**" property of your Web Browser.

The default SE Password is 1234. The password can be changed using Jeeves only and it must be as per the specifications given below:

- It must be a minimum of 6 characters and a maximum of 12 characters.
- It must include atleast one upper-case, one lower-case, one number and one special character.
- All ASCII characters (except Percentage **%**, Hash **#,** Equal to **=,** Plus **+**, And **&**, Backslash **\,** Less than **<,** Greater than **>,** Apostrophe **',** Double Quote **"** and **Space**) are allowed.

To provide additional security,

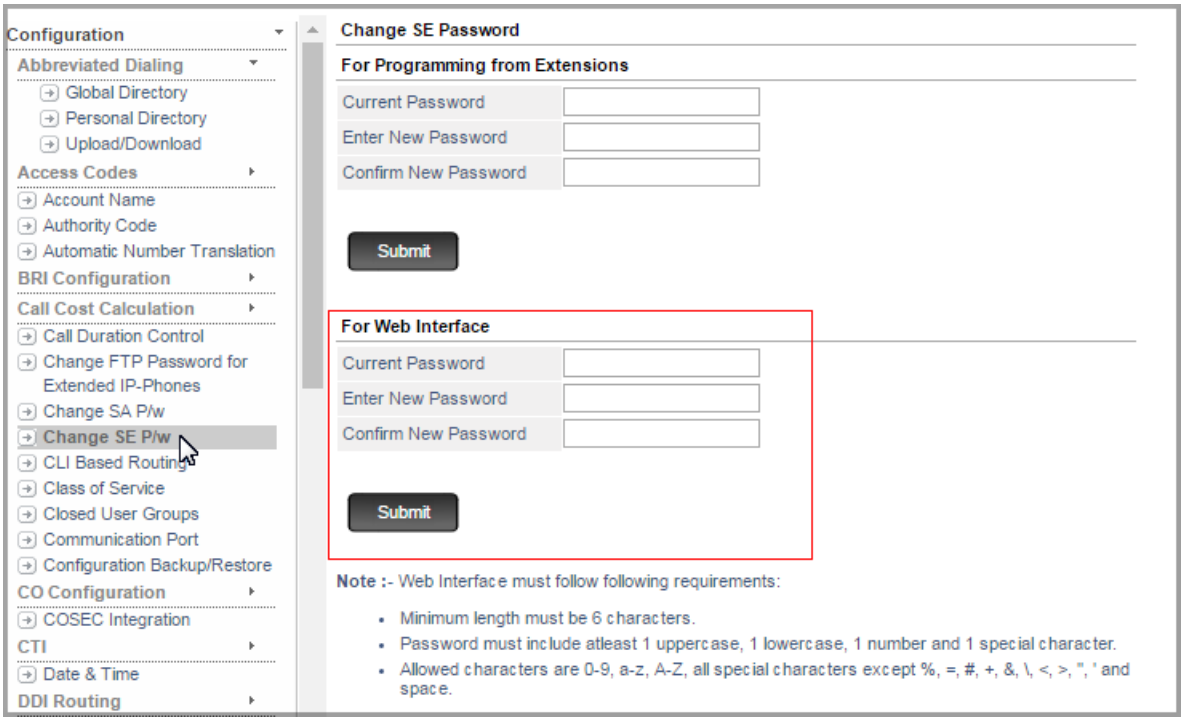
- the password will be valid for 90 days and you will not be able to login with the existing password. You will be prompted to change the password.
- if you enter a wrong password five times consecutively within 10 minutes, the system will block the source IP Address for 10 minutes. This activity will be logged in the "System Activity Log" as well as "Simple Network Management Protocol (SNMP)".

- *The SE password is stored in the CPU Card. If you forget the SE password, the only way to restore the default SE password is to change the Jumper settings of the CPU Card.*
- *You are advised to record and store the SE password at a safe place, where it can be accessed by you (the System Engineer) to avoid the inconvenience of restoring the default SE password.*

Changing the SE Password using Jeeves

- Log in to Jeeves as System Engineer.

- Under **Configuration**, click the **Change SE P/w**.



- Under **For Web Interface**,
  - Enter **Current Password.**
  - **Enter New Password**. The new password must be:
    - a minimum of 6 characters to a maximum of 12 characters.
    - all ASCII characters (except Percentage **%**, Hash **#,** Equal to **=,** Plus **+**, And **&**, Backslash **\,** Less than **<,** Greater than **>,** Apostrophe **',** Double Quote **"** and **Space**) are allowed.
    - include atleast one upper-case, one lower-case, one number and one special character.
  - In **Confirm New Password**, re-enter the new password to confirm.
  - Click **Submit** button to save your new password.

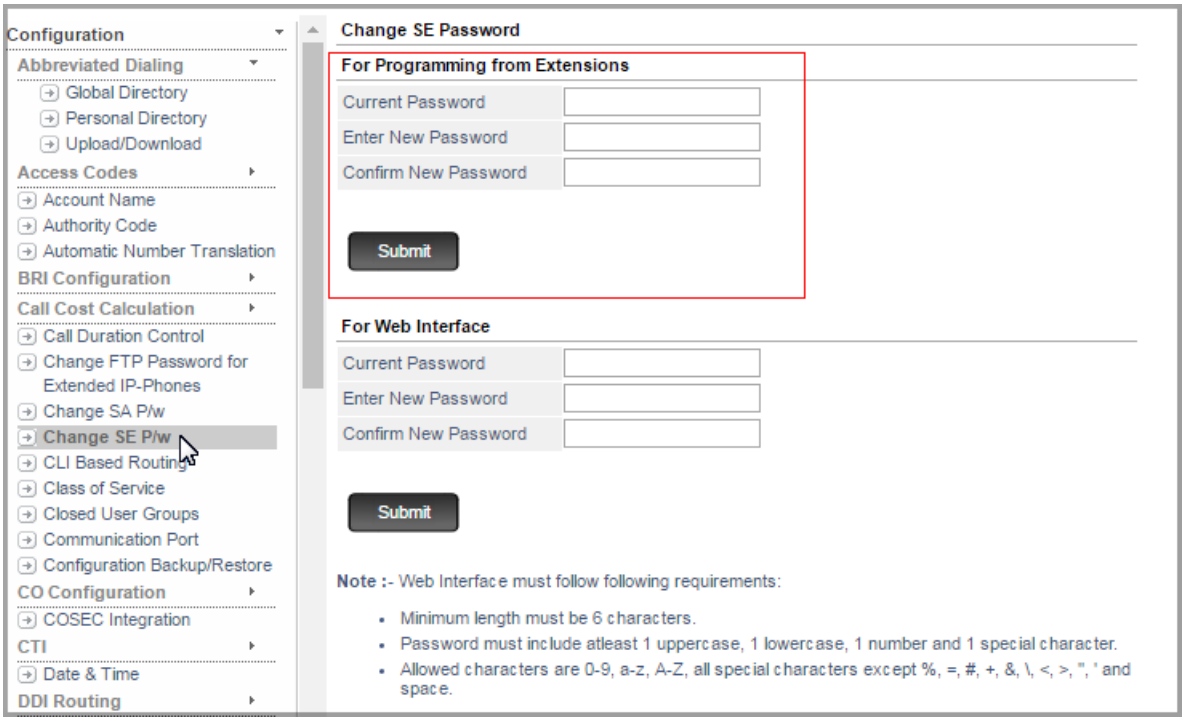
For programming from extensions

The SE password is a code used to restrict unauthorized access to the SE Mode. The password can be a minimum of 4 digits to a maximum of 12 digits. The valid digits are from 0 to 9. The default SE Password is 1234. To avoid unauthorized access, we recommend you to change the password. Make sure it is strong and is kept confidential. The SE Password can be changed using Jeeves as well using commands. However, the default SE password can not be changed from commands. To change the default SE password, refer to “1. System Engineer Mode”.

Changing the SE Password using Jeeves

- Log in to Jeeves as System Engineer.

- Under **Configuration**, click the **Change SE P/w**.



- Under **For Programming from Extensions**,
  - **Enter Current Password.**
  - **Enter New Password**. The new password can be a minimum of 4 digits to a maximum of 12 digits. The valid digits are from 0 to 9.
  - In **Confirm New Password**, re-enter the new password to confirm.
  - Click **Submit** button to save your new password.

Changing SE Password using a Telephone

- Enter SE mode from a DKP/SLT.
- Dial ***5306-New SE Password***
- Exit SE mode.

If the new password is less than 12 digits, you must dial 5306-New SE Password #.*
You cannot change the default SE password using a telephone.

Forgot the SE Password?

If you have already changed the default SE-Password for accessing Jeeves and are unable to recall or locate it, you must restore the default SE password.

Restoring Default SE Password

Restoring the Default SE Password requires you to change the Jumper/Switch Settings in the CPU Card. Follow the instructions given below for each model of SARVAM UCS.

ETERNITY LENX/MENX

- Remove the CPU Card from the slot.
- Locate the Switch **SW1** (PASSWORD_IP_DEFAULT) on the PCB of the CPU Card.
- Change the position of the Switch from ***1*** to ***ON*** (that is, from 'Normal' to 'Reset SE Password')
- Reinsert the CPU Card into the slot.
- Switch ON the system and wait for 15 seconds.
- Switch OFF the system and remove the CPU Card from the slot.
- Change the Switch position from ***ON*** to the original position ***1***.
- Insert the CPU Card back into the slot.
- Switch ON the system.

The following parameters are also set to default when Switch SW1 is turned ON:

- All the Network parameters except TCP NAT Keep Alive and UDP NAT Keep Alive.
- *Allow Remote Login* and *Allow Web Server access* parameters under Security Settings on WAN.
- *For Web Interface* parameter under Change SE Password.

ETERNITY GENX

- Remove the CPU Card from the slot.
- Locate Jumper ***J1*** on the PCB of the CPU Card.
- Change the position of the Jumper from ***BC*** to ***AB*** (that is, from 'Normal' to 'Reset SE Password')
- Reinsert the CPU Card into the slot.
- Switch ON the system and wait for 15 seconds.
- Switch OFF the system and remove the CPU Card from the slot.
- Change the Jumper position from ***AB*** to the original position ***BC.***
- Insert the CPU Card back into the slot.
- Switch ON the system.

The default SE password will be restored to 1234. You can now enter the programming mode by dialing ***1#91-1234*** (the default password). You can also change the SE password again using Jeeves or by dialing a command as described above.

If you do not restore the position of the Jumpers from AB back to BC, the new password will not be saved. Make sure you change the Jumper position to BC before you change the default password.

ETERNITY PENX

- Remove the cover.
- Locate Jumper ***J6*** on the PCB of the CPU Card.
- Change the position of the Jumper from ***BC*** to ***AB*** (that is, from 'Normal' to 'Reset SE Password')
- Switch ON the system and wait for 15 seconds.
- Switch OFF the system.
- Change the Jumper position from ***AB*** to the original position ***BC.***
- Switch ON the system.

The default SE password will be restored to 1234. You can now enter the programming mode by dialing ***1#91-1234*** (the default password). You can also change the SE password again using Jeeves or by dialing a command as described above.

If you do not restore the position of the Jumpers from AB back to BC, the new password will not be saved. Make sure you change the Jumper position to BC before you change the default password.

System Administrator (SA) Password

You can log into the SA mode through Jeeves or from extensions only after you have set the password from SE mode through Jeeves. For more information, see "2. System Administrator Mode".

For Accessing Jeeves

The SA password is a code for preventing unauthorized access to the SA mode. As this password is meant for restricting access to the SA mode, we strongly recommend you to:

- Keep the password secret.
- Select a complex password that cannot be easily guessed.
- Change the password regularly
- Not use the **"Remember Password**" property of your Web Browser.

The password can be changed using Jeeves only and it must be as per the specifications given below:

- It must be a minimum of 6 characters and a maximum of 12 characters.
- It must include atleast one upper-case, one lower-case, one number and one special character.
- All ASCII characters (except Percentage **%**, Hash **#,** Equal to **=,** Plus **+**, And **&**, Backslash **\,** Less than **<,** Greater than **>,** Apostrophe **',** Double Quote **"** and **Space**) are allowed.

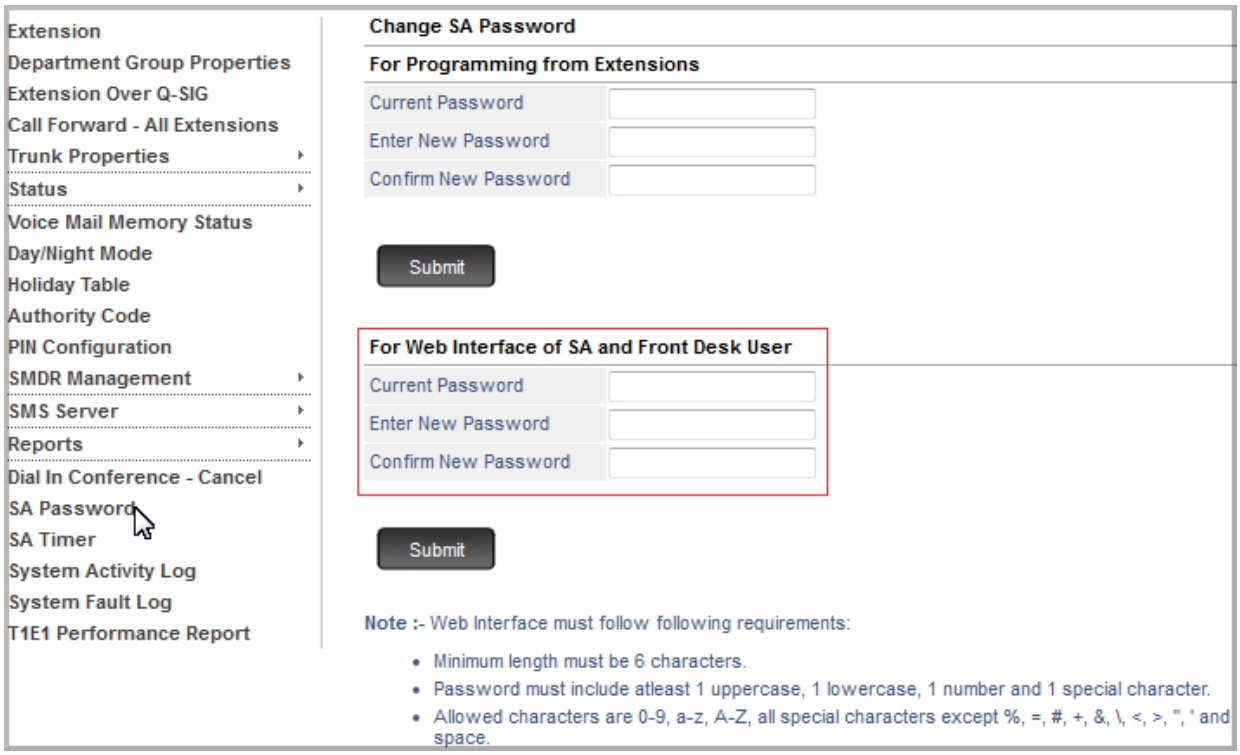
To provide additional security,

- the password will be valid for 90 days and you will not be able to login with the existing password. You will be prompted to change the password.
- if you enter a wrong password five times consecutively within 10 minutes, the system will block the source IP Address for 10 minutes. This activity will be logged in the "System Activity Log" Log as well as "Simple Network Management Protocol (SNMP)".

Changing SA Password using Jeeves

- Log in to Jeeves as System Administrator.

- Click **SA Password**.



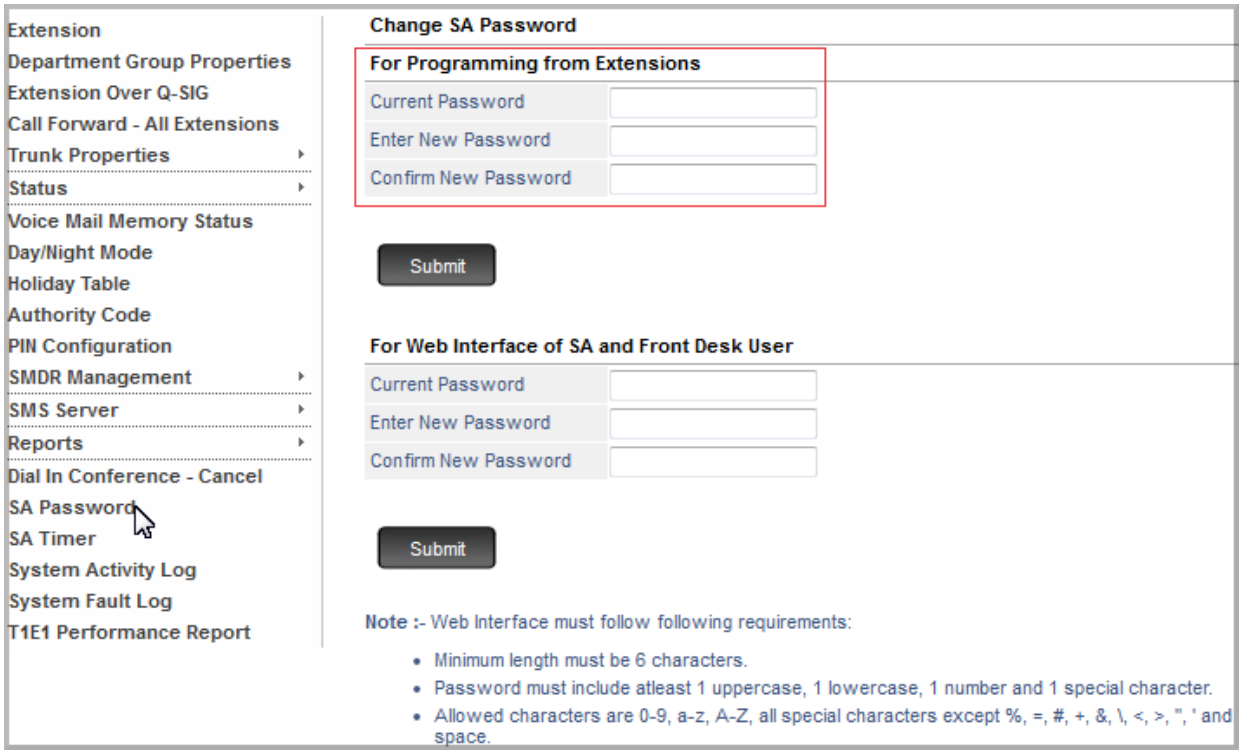
- Under **For Web Interface of SA and Front Desk User.**
  - Enter **Current Password.**
  - **Enter New Password**.The new password must be:
    - a minimum of 6 characters to a maximum of 12 characters.
    - all ASCII characters (except Percentage **%**, Hash **#,** Equal to **=,** Plus **+**, And **&**, Backslash **\,** Less than **<,** Greater than **>,** Apostrophe **',** Double Quote **"** and **Space**) are allowed.
    - include atleast one upper-case, one lower-case, one number and one special character.
  - In **Confirm New Password**, re-enter the new password to confirm.
- Click **Submit** button to save your new password.
- You may log out of Jeeves.

For programming from extensions

Changing SA Password using Jeeves

- Log in to Jeeves as System Administrator.

- Click **SA Password**.



- Under **For Programming from Extensions.**
- Enter **Current Password.**
- **Enter New Password**. The new password can be a minimum of 4 digits to a maximum of 12 digits. The valid digits are from 0 to 9.
- In **Confirm New Password**, re-enter the new password to confirm.
- Click **Submit** button to save your new SA password.
- You may log out of Jeeves.

Changing SA Password using a Telephone

- Enter SE mode from a DKP/SLT.
- Dial ***5310-New SA Password.***
- Exit SE mode.

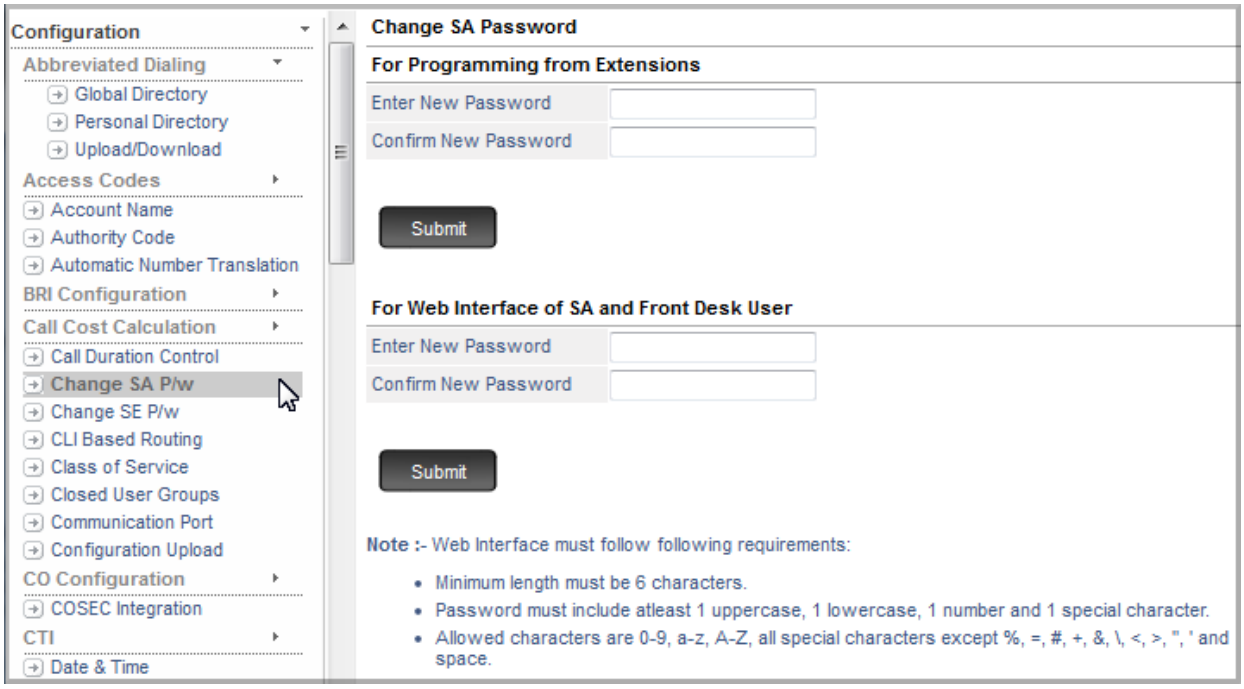
If the new password is less than 12 digits, you must dial 5310-New SA Password #.*

Forgot the SA Password?

In case the System Administrator has forgotten the password, a new password can be issued by the System Engineer only.

The System Engineer can assign a new SA password from Jeeves only.
To issue a new SA Password from Jeeves,

- Log in to Jeeves as System Engineer.
- Under **Configuration**, click **Change SA P/w**.



- Under **For Web Interface of SA and Front Desk User.**
- **Enter New Password**. The new password must be:
  - a minimum of 6 characters to a maximum of 12 characters.
  - all ASCII characters (except Percentage **%**, Hash **#,** Equal to **=,** Plus **+**, And **&**, Backslash **\,** Less than **<,** Greater than **>,** Apostrophe **',** Double Quote **"** and **Space**) are allowed.
  - include atleast one upper-case, one lower-case, one number and one special character.
- In **Confirm New Password**, re-enter the new password to confirm.
- Click **Submit** to save new SA password.

User Password

Extension Users can secure their respective stations/extensions from unauthorized use with a password unique to each extension. The User password too is a combination of any four digits, from 0 to 9. The default User Password is ***1111***, which each user can change from their respective extensions.

To avoid unauthorized access,

- the extension users must change the default password.
- make sure the new password is strong and is kept confidential.
- change the password regularly.

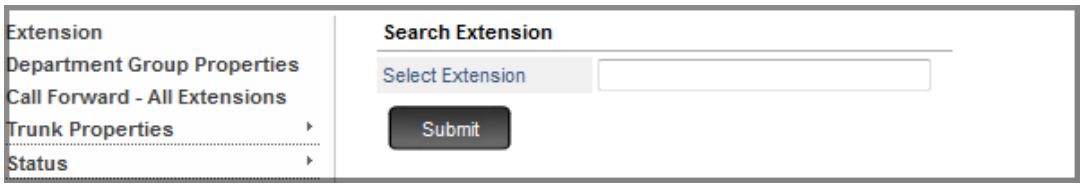
Refer the topic “User Password” to know more.

Forgot the User Password?

If an extension user forgets the User Password, a new password can be issued to the extension user by the System Administrator using Jeeves or by dialing a command.

To issue a new User password using Jeeves,

- Log in to Jeeves as System Administrator.
- Click **Extension**.



- In **Select Extension,** enter the Number or the Name of the extension you want to search.
- Click **Submit**.
- The page of the desired extension will appear.
- Click **Phone Properties** to expand.
- Enter the new User Password in the field **Change User Password to**. The User Password may be a combination of 4 digits. Valid digits: 0 to 9.
- Click **Submit** button to save your new SA password.
- You may log out of Jeeves.

To issue a new User password using a telephone,

- Enter SA mode from a DKP/Extended IP Phone/SLT.
- Dial ***1072-012-Extension Number-New User Password***
- Exit SA mode.

Additional Security to Extension Users

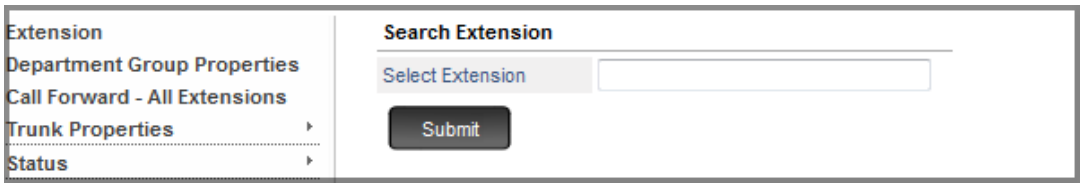
It is also possible to secure unattended extensions. This may be necessary when extension users forget to lock their extensions and are away from their desks. In such situations, the System Administrator can protect the extension from unauthorized access and use by:

a. locking the Keypad the extension phone; possible only on DKP/Extended IP Phone extensions.
b. setting the User Status for the extension as "Absent"; possible on both DKP/Extended IP Phones and SLT extensions. Read the feature description “User Absent/Present” to know more.

The System Administration can do this, lock the keypad of DKP/Extended IP extensions and set DKP/Extended IP Phones and SLT extension users as ‘Absent’ using Jeeves and by dialing commands from a telephone.

To secure an extension from Jeeves,

- Log in to Jeeves as System Administrator.
- Click **Extension**.



- **Select Extension:** Enter the Extension Number or the Extension Name of the extension you want to search.
- Click **Submit**.
- The page of the desired extension will appear.
- Click **Phone Properties** to expand.
- Change user status by selecting the option **Absent** for the parameter **Presence**. When you want to change user status again to present, select **Present.**
- Select the option **Lock** for the parameter **Keypad**. When you want to remove keypad lock, select **Unlock**.

When the keypad is locked, the features Call Log, Contact, Call Forward, Dynamic Lock, User Status, DND, Call Cost Display, Hotline, Alarm, Change User Password will not be accessible by the extension user.

- Click **Submit** button.
- Log out of Jeeves.

To secure an extension using a telephone,

- Enter SA mode from a DKP/Extended IP Phone/SLT.
- Dial ***1072-013-Extension Number-1*** to lock keypad of an extension.
- Dial ***1072-013-Extension Number-0*** to unlock keypad of an extension.
- Dial ***1072-014-Extension Number-0*** to change user status to Absent.
- Dial ***1072-014-Extension Number-1*** to change user status to Present.
- Exit SA mode.

- *Extension users can also set their status as 'Absent' or 'Present' from their respective extension phones. Refer "User Absent/Present").*

- DKP/Extended IP Phone *extension users can also lock the keypad of their phones from the Phone Menu. Refer "Digital Key Phone-Operation" and "Extended IP Phone/VARTA UC Client - Operation" for instructions on navigating the phone menu.*

- *It is also possible to lock/unlock the keypad and set the user extension status as 'Absent'/'Present' from a remote location using "Direct Inward System Access (DISA)".*

FTP Access for Extended IP Phones

The access to the FTP Server of the Extended IP Phones is secured by a password. The FTP password is a code for preventing unauthorized access. The default FTP Password is 1234. As this password is meant for restricting access, we strongly recommend you to:

- change the default password.
- make sure the new password is strong and is kept confidential.
- change the password regularly.

The password can be changed using Jeeves only. The password can be a minimum of 6 characters to a maximum of 12 characters. All ASCII characters (except Percentage **%**, Hash **#,** Equal to **=,** Plus **+**, And **&**, Backslash **\,** Less than **<,** Greater than **>,** Apostrophe **',** Double Quote **"** and **Space**) are allowed. It must include atleast one upper-case, one lower-case, one number and one special character.

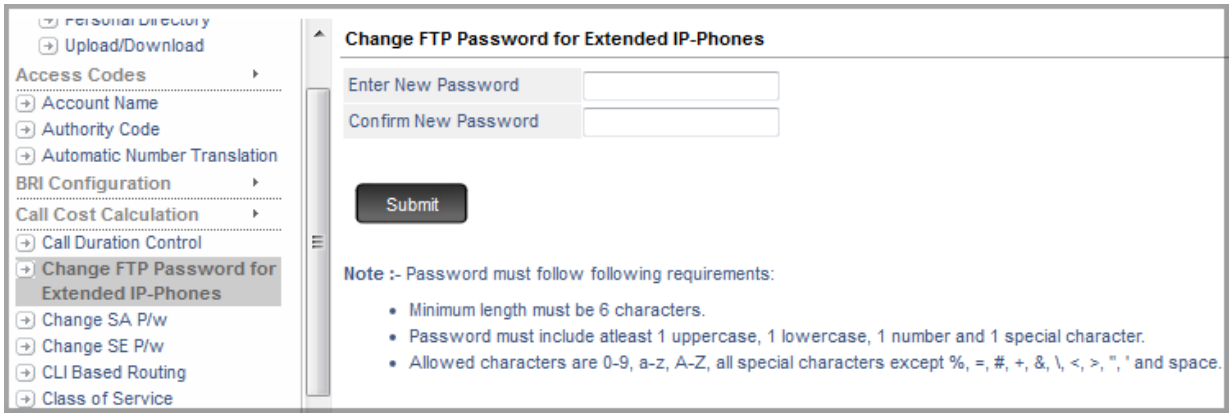
When you set the System to default, the FTP Password will not be set to default.

If you use jumpers to restore the default SE Password, the FTP Password will not be set to default.

Changing the FTP Password using Jeeves

- Log in to Jeeves as System Engineer.

- Under **Configuration**, click the **Change FTP Password for Extended IP-Phones**.



- **Enter New Password**.

- In **Confirm New Password**, re-enter the new password to confirm.

- Click **Submit** to save your new password.

Time Tables

What's this?

Certain features of the SARVAM UCS like Operator, Class of Service, Toll Control, Outgoing Trunk Access, among others, require stations and trunks to behave differently according to the time of the day, which is referred to as Time Zone.

For example, incoming calls are to be routed to the security personnel extension, instead of the Operator when the office is closed, or certain features in the Class of Service are to be allowed only during working hours, or access to outgoing long distance calls are to be denied during non-working hours, or the station must play a different greeting message to the callers during break hours and holidays.

Time Tables can be assigned to stations and trunks to define their behavior according to the time of the day, that is, Time Zone.

Time Zones

A day can be divided into three time zones: Working hours, Break hours and Non-working hours. The default Time Zones defined for each day are:

- Working hours: 09:00 to 18:00
- Break hours: 13:00 to 14:00
- Non-working hours: 18:00 to 09:00

Working, Break and Non-Working hours are set to 00:00 for Sunday.

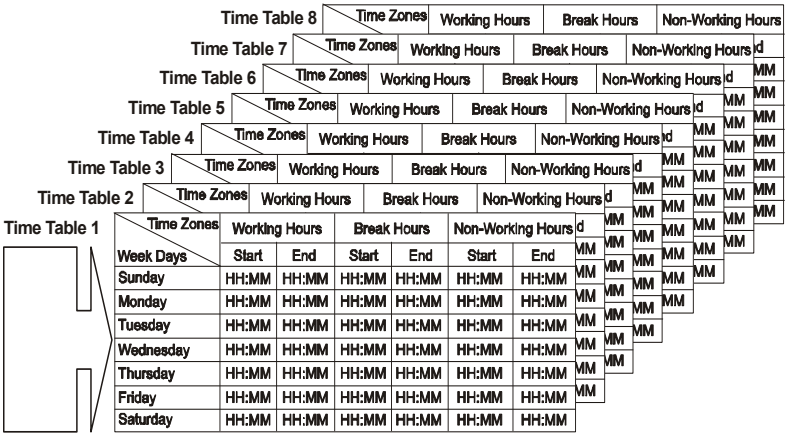
You can define a different Time Zone for your organization. Further, you can program each day of a week with different time zones. For example, you may define the Working hours from Monday to Friday as 09:30 to 18:30, and for Saturday, from 09:30 to 15:00. If you have a 24x7 business, you may set Working Hours also for Sunday.

Time Tables

A Time Table is a schedule of the three Time Zones, namely: Working Hours, Break Hours, Non-Working hours, for the entire week.

A Time Table is assigned to stations defining the Time Zones for the entire week, so that the system can execute the Time Zone-dependent features and facilities according to the Time Table.

There are 8 different Time Table templates to select from.



By default, the Time Table 1 is assigned to all stations and trunks in their Station Basic Feature Template and Trunk Feature Template respectively. In Time Table 1, six days of the week - Monday to Saturday -have working hours from 9:00-18:00, break hours from 13:00-14:00 hours and non-working hours from 18:00 to 09:00. Sunday is a holiday, with all three Time Zones set to 00:00 hours.

You may also customize the default Time Table 1 OR customize and assign a different Time Table to the stations and trunks.

SARVAM UCS offers the facility to switch the system manually into "Day/Night mode", at any point in time, by issuing a command. When you set the system in Day/Night Mode, the system overrides the Time Tables assigned to Trunks, Stations and Operator. According to the mode you selected, it applies Working Hours/Non-Working Hours to run all the Time-Zone dependent features of the system.

Refer the topic "Day Night Mode" to know more.

How to configure

A Trunk port can be assigned Time Table in the "Trunk Feature Template" assigned to it. A Station port can be assigned a time table in the "Station Basic Feature Template" assigned to it.

The default Time Table 1 is assigned to both stations and trunks of SARVAM UCS. Check if this time table matches the working hours of the organization, and the Time Zone requirements of the individual stations and trunks.

The following station parameters can be programmed differently for different Time Zones:

- Class of Service.
- Toll Control.
- OG Trunk Bundle Group.

The following trunk parameters can be programmed differently for different Time Zones:

- Auto Attendant
- DISA

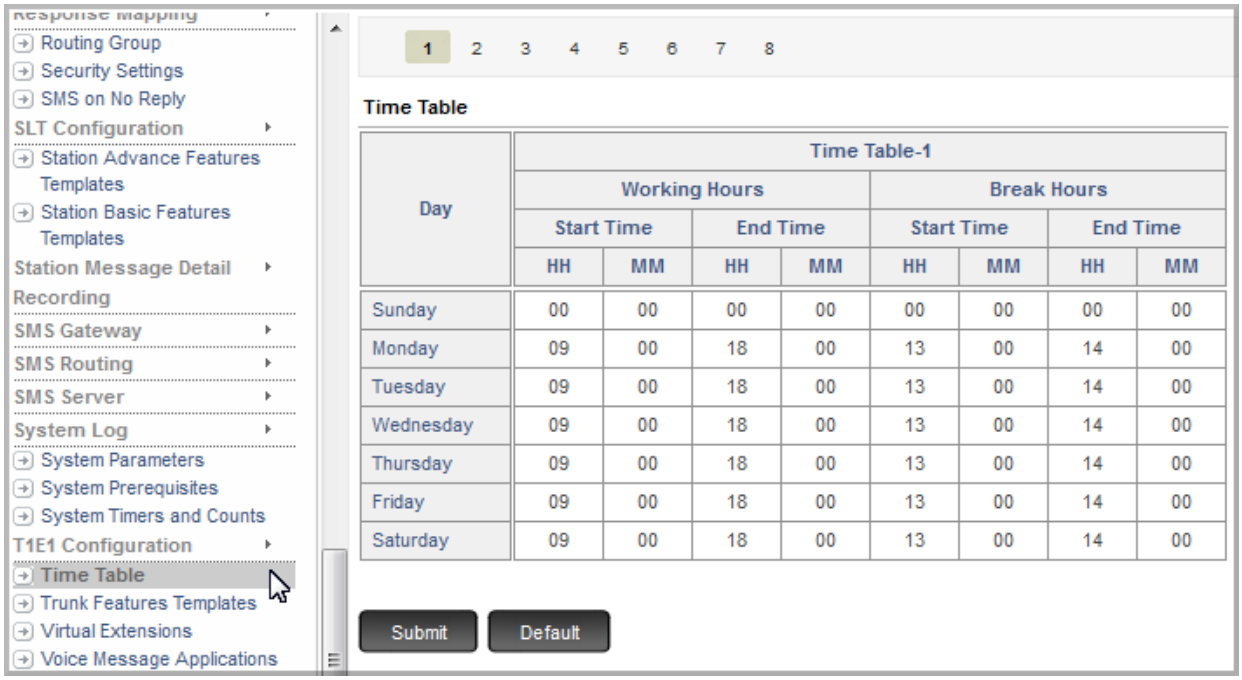
- Trunk Landing Group

You may retain the default Time Table 1 or customize it to suit your requirements. Or you may customize different Time Tables and assign them to different stations and trunks.

You can customize a Time Table using Jeeves or by dialing commands from a Telephone.

Programming Time Table using Jeeves

- Log in as System Engineer.
- Under **Configuration**, click **Time Table**.



- Select the desired Time Table number and define the Time Zones, that is, working hours, break hours and non-working hours.
- Click **Submit** at the bottom of the page to save your changes.
- Now, assign the Time Table you program to the desired Station/Trunk.
- To assign Time Table to Stations, go to Station Basic Feature Template. Refer the topic “Station Basic Feature Template” for instructions.
- To assign Time Table to Trunks, go to Trunk Feature Template. Refer the topic “Trunk Feature Template” for details a how to assign timetable to trunks.
- Log out of Jeeves or continue programming, as required.

Programming Time Table using a Telephone

- Enter SE mode from a DKP/SLT.

To program a timetable, dial:

- ***1052-1-Time Table-Day-Time Zone-Start Time-End Time*** to program a single time Table.
- ***1052-2-Time Table-Time Table-Day-Time Zone-Start Time-End Time*** to program the same time zones for a range of time tables.
- ***1052-*-Day-Time Zone-Start Time-End Time*** to program the same time zones for all time tables.
  Where,
  Time Table is from 1 to 8.

  Day is the Day of the week
  1 for Sunday
  2 for Monday
  3 for Tuesday
  4 for Wednesday
  5 for Thursday
  6 for Friday
  7 for Saturday

  Start Time is Time Zone start time in 24-hours, HH:MM format. Where Hours is 00 to 23 and Minutes is 00 to 59.

  End time is Time Zone end time in 24 hours, HH:MM format. Where Minutes is 00 to 59.

  By default, Time Zone 1 is 0900 to 1800 and Time Zone 2 is 1300 to 1400. The left over time automatically is treated as Non-Working Hours.

To default a time table, dial:

- ***1051-1-Time Table*** to default a single time table.
- ***1051-2-Time Table-Time Table*** to default a range of time tables.
- ***1051-**** to default all time tables.
  Where,
  Time Table is 1 to 8.

  To assign Time Table to Stations and Trunk ports using SE commands, refer the topics “Customizing Station Basic Feature Template using a Telephone” and “Customizing Trunk Feature Template using a Telephone”.

- Exit SE mode.

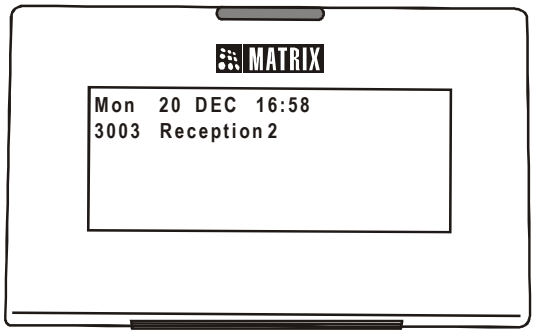
Time Zone Display

What's this?

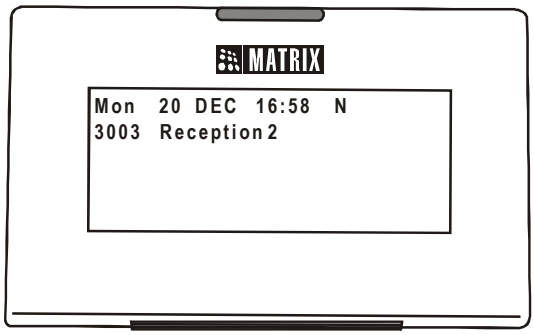
The current time zone—Working Hours, Break Hours and Non-working Hours—is displayed on the LCD of the DKP and the Extended IP Phone.

During Non-working hours the letter 'N' is displayed on the LCD of the DKP and the Extended IP Phone in the idle state, and during Break hours, the letter 'B' is displayed.

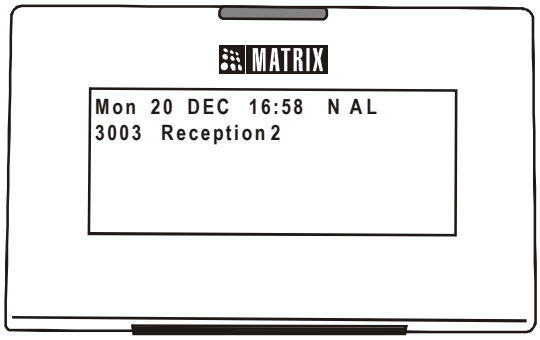
During working hours, in the idle state, the phone display will look like this:



During Non-working hours, in the idle state, the phone display will look like this:



During Non-working hours, in the idle state, if the extension user has set User Absent and activated Keypad Lock on the phone, the phone display will look like this:



Toll Control

What's this?

Toll Control (or Toll Restriction) is an expense control feature of SARVAM UCS. It enables you to program the system so that each extension has a designated calling permission referred to as 'Call Privilege'.

Each type Call Privilege allows the extension to call certain areas and restricts it from calling others. The extension can also be restricted from the dialing of specific telephone numbers.

The SARVAM UCS supports five types of Call Privileges, these are:

- **No Calls:** Dialing of all external numbers is restricted. Only internal (extension-to-extension) calls are allowed.

  If you have enabled Global Directory Part 1, Part 2 and 3 in the Class of Service of the extension, the numbers programmed in these will be allowed to be dialed out.

- **Local Calls:** Dialing of outgoing calls to Local area numbers, in addition to internal calls, is allowed. It is possible to restrict calls to certain local numbers. To apply this Call Privilege, you must configure the 'Local Numbers' list.

- **Regional Calls:** Dialing of outgoing calls to regional numbers is allowed, in addition to internal and local calls. It is possible to restrict calls to certain regions. To apply this Call Privilege type, you must configure the 'Regional Numbers' list.

- **National Calls:** Dialing of domestic, long-distance numbers within the country is allowed, in addition to internal and regional calls. You can also restrict calls to certain parts of the country. To apply this Call Privilege type, you must configure the 'National Numbers' list.

- **International Calls:** Dialing of international numbers is allowed, in addition to local area, long distance and internal numbers. You can also restrict calls to certain countries. To apply this Call Privilege type, you must configure the ‘International Numbers’ list.

- **All Calls:** Dialing of all types of numbers - local, regional, national, international- is allowed, without any restriction.

- **Limited Calls:** Dialing of only specific Telephone numbers (local, regional, national or international) is allowed. By applying this Call Privilege type, you can allow and restrict dialing of telephone numbers starting with a particular digit, or a particular area code, or certain telephone numbers only. To apply this Call Privilege type, you must program a list of the ‘Limited Numbers’ that are to be allowed and numbers that are to be restricted. You can configure three such ‘Limited Numbers’ lists.

Toll Control forms the basis of the features Dynamic Lock, Call Budget on Extension and Call Budget on Trunk.

Using “Dynamic Lock”, extension users can change the Toll Control (Call Privilege) of their extensions on their own. The Operator or System Administrator can also change the Toll Control of the extension using Dynamic Lock. To support this feature, SARVAM UCS offers fours levels of Toll Control, from 0 to 3.

Call Budget is a cost control feature that allows you to keep a control on the total cost of phone calls made by the extension users. You may also define the calling permission for extensions and trunks can be allotted a budget. For more details, see “Call Budget on Extension” and “Call Budget on Trunk”.

Toll Control Levels

For each Toll Control Level from 0 to 3, a 'Call Privilege' is defined. The system applies the Toll Control Level (that is, the Call Privilege) set by the extension users themselves or set by the System Administrator/Operator for the extensions using Dynamic Lock.

- **Toll Control - Level 0** is Time Zone based, wherein you must define the Call Privilege Type for each Time Zone, that is, Working Hours, Break Hours and Non-Working Hours. For instance, you may define 'All Calls' as Call Privilege for Working Hours, 'Local Calls' as Call Privilege for Break Hours and 'No Calls' as Call Privilege for 'Non-Working' Hours.

  By default, Call Privilege 'No Calls' is selected for all three Time Zones.

- **Toll Control - Level 1** is not based on Time Zones. By default, the Call Privilege Type for this level is 'No Calls'.

- **Toll Control - Level 2** is not based on Time Zones. By default, the Call Privilege type set for this level is 'No Calls'.

- **Toll Control - Level 3** is not based on Time Zones. By default, Call Privilege 'No Calls' is selected for this level.

- **Toll Control - Call Budget Consumed** is applied only if the “Call Budget on Extension” feature is enabled on the extension.

- If the feature Call Budget on Trunks is enabled and the budget is exhausted, no calls will be routed through the trunk. For details, see “Call Budget on Trunk”.

SARVAM UCS offers you the flexibility to redefine the Call Privilege for each of the above Toll Control Levels according to user requirements.

How it works

- When a call is made, the SARVAM UCS checks the Toll Control Level assigned to the extension making the call.

- The system checks the 'Call Privilege' programmed in the Toll Control Level of the extension.

- For each call privilege type detected, the system will check the following to determine if call is to be allowed or denied, as summarized in the table below:

Type Call Privilege detected	Logic will check
Local calls	Local Number List programmed.
Regional calls	Regional Number List programmed
National calls	National Number List programmed
Internationals calls	International Number List programmed

Type Call Privilege detected	Logic will check
Limited Calls	Limited Number list (1,2 or 3) assigned to the extension

- The Local, Regional, National, International and Limited Calls Number Lists consist of Allowed Numbers and Denied Numbers.
- The system compares the each digit of the dialed number string with the number strings programmed in the Allowed and Denied Number Lists of the Local/Regional/National/International/Limited Number Lists, using the following logic:

Allowed Number List	Denied Number List	Result
match found	match found	Call allowed
match found	no match found	Call allowed
no match found	no match found	Call allowed
no match found	match found	Call denied

- The call is allowed to be made, if the dialed number:
  - matches with Allowed Number list and the Denied Number list.
  - matches with Allowed Number list, but not with the Denied Number list.
  - matches with neither the Allowed List nor the Denied List.
- The call is restricted, if the dialed number matches with the Denied Number list, but not with the Allowed Number list.

How to configure

Decide the type of Call Privilege you wish to assign to each extension port type: SLT, DKP, SIP, ISDN Terminals.

For Toll Control to work, you must first program the lists of Local Numbers, Regional Numbers, National Numbers, International and Limited Numbers, according to the type of Call Privilege you want to assign to the extensions. To do this,

Make a two-column tables each for Local, Regional, National, International and Limited Call Numbers on paper or using a computer.

In one column of each list, write down the numbers you want to permit as Allowed Numbers. In the other column write down the numbers you want to restrict as Denied Numbers. Your Table may look like these:

List of Local Numbers for Call Privilege - Local Calls

Sr. No.	Allowed List	Denied List
1		
2		
:		
:		

Sr. No.	Allowed List	Denied List
999		

List of Regional Numbers for Call Privilege - Regional Calls

Sr. No.	Allowed List	Denied List
1		
2		
:		
:		
999		

List of Regional Numbers for Call Privilege - International Calls

Sr. No.	Allowed List	Denied List
1		
2		
:		
:		
999		

List of Limited Numbers for Call Privilege - Limited Calls 1

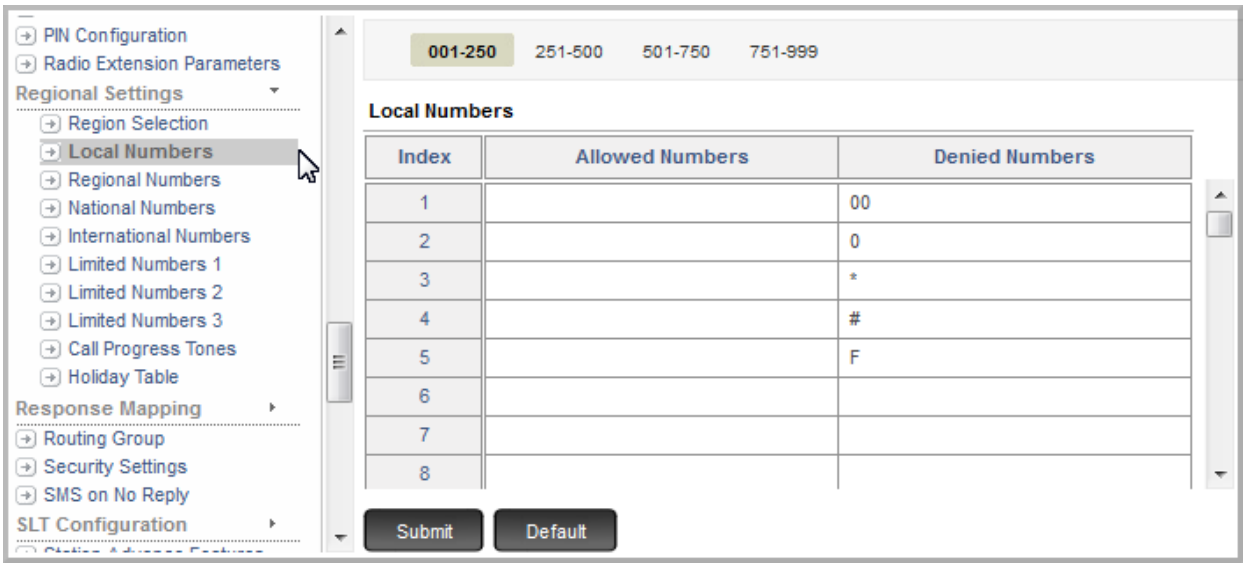
Sr. No.	Allowed List	Denied List
1		
2		
:		
:		
999		

Configuring Toll Control using Jeeves

- Login as System Engineer.
- Under **Configuration**, click **Regional Settings**.

Local Numbers

- Click **Local Numbers** to open the page.

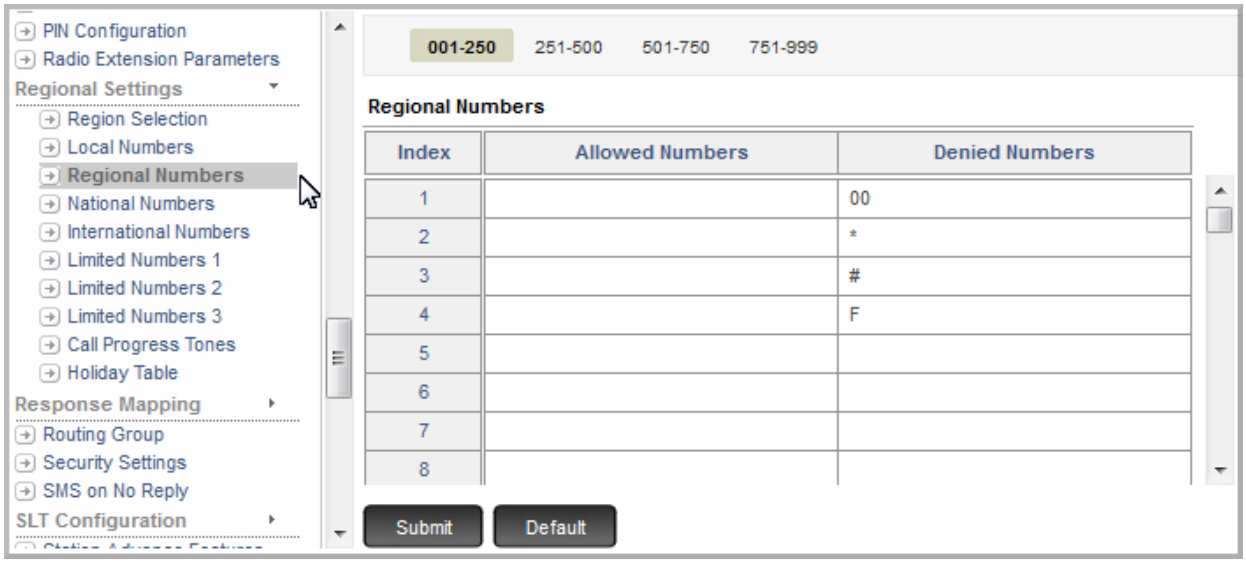


- Enter the local area numbers that are permitted to be dialed in the **Allowed Numbers** list and the numbers that are to be restricted in the **Denied Numbers** list. You may enter as many as 999 numbers in each list.

- Click **Submit** at the bottom of the page to save the entries.

Regional Numbers

- Click **Regional Numbers** to open the page.



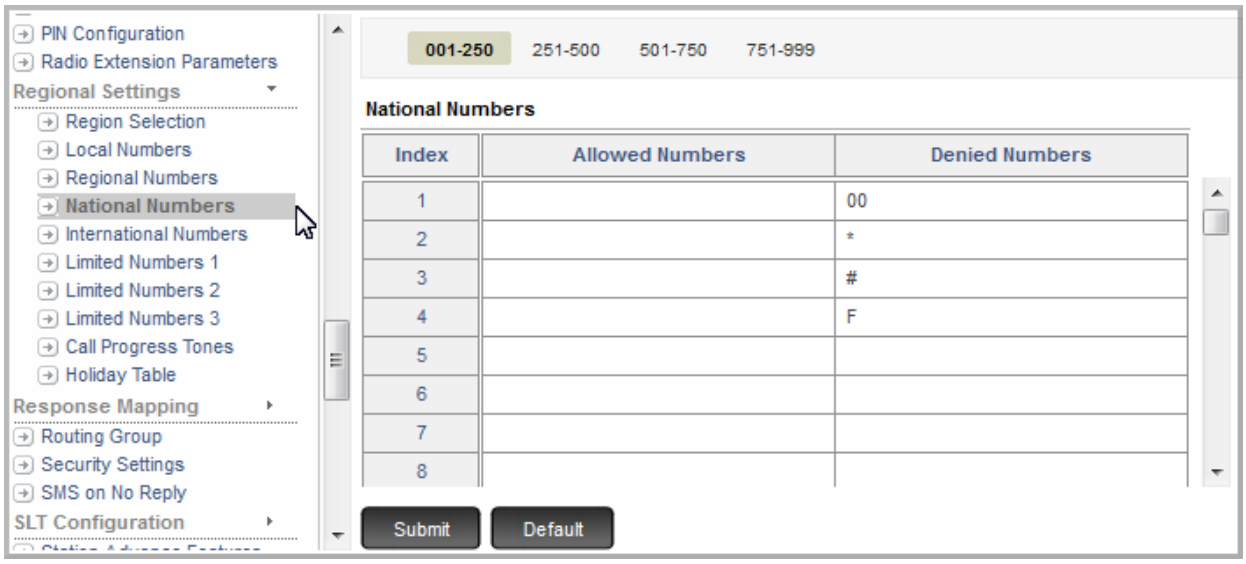
- Enter the regional area numbers that are permitted to be dialed in the **Allowed Numbers** list and the numbers that are to be restricted in the **Denied Numbers** list.

- Repeat the entries you made in the **Local Numbers** list also in the **Regional Numbers** list.

- Click **Submit** at the bottom of the page to save the entries.

National Numbers

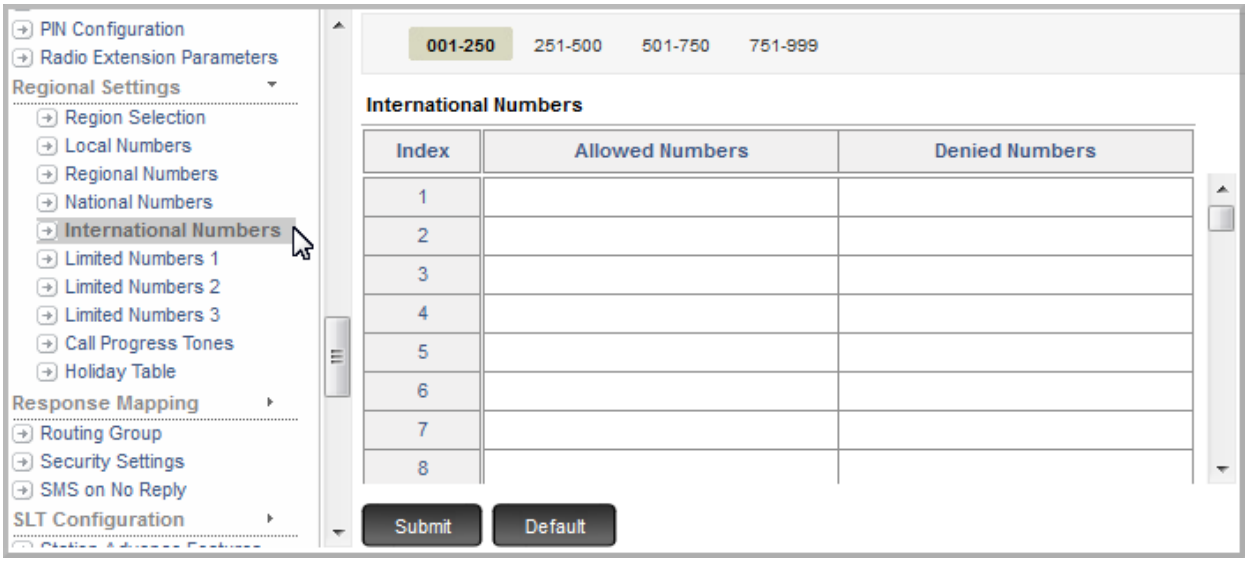
- Click **National Numbers** to open the page.



- Enter the long distance numbers within the country that are to be permitted in the **Allowed Numbers** list and the numbers that are to be restricted in the **Denied Numbers** list.
- Repeat the entries you made in the **Local Numbers** and **Regional Numbers** lists in this list.
- Click **Submit** at the bottom of the page to save the entries.

International Numbers

- Click **International Numbers** to open the page.

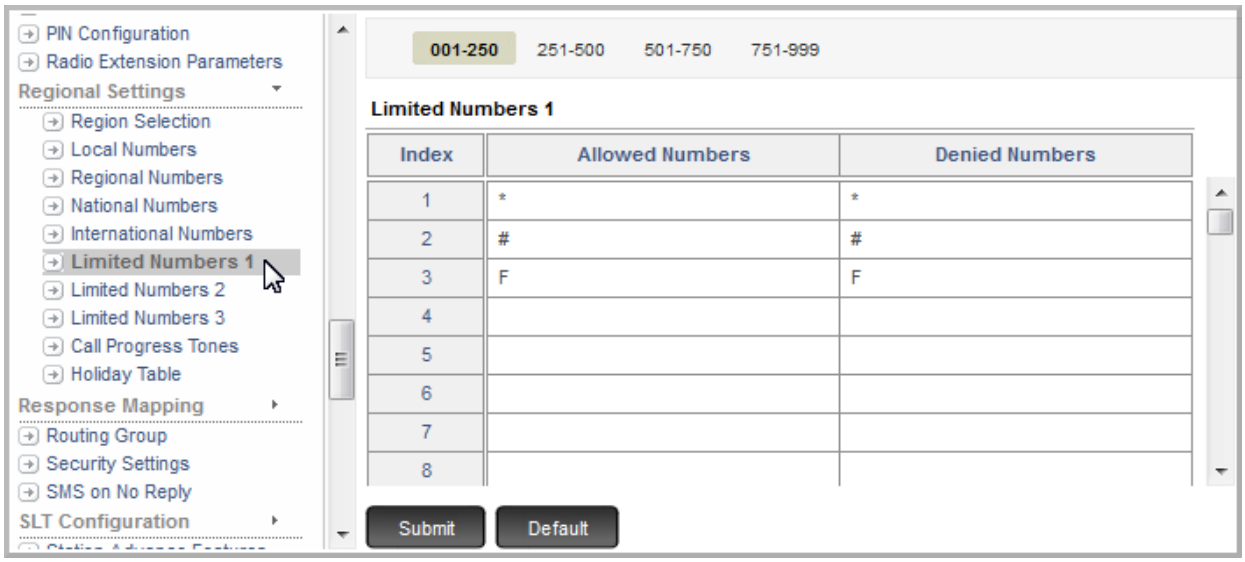


- Enter the overseas numbers that are to be permitted in the in the **Allowed Numbers** list and the numbers that are to be restricted in the **Denied Numbers** list.
- Repeat the entries you made in the **Local Numbers**, **Regional Numbers** and **National Numbers** lists in this list.

- Click **Submit** at the bottom of the page to save the entries.

Limited Numbers

- Click **Limited Numbers 1** to open the page.



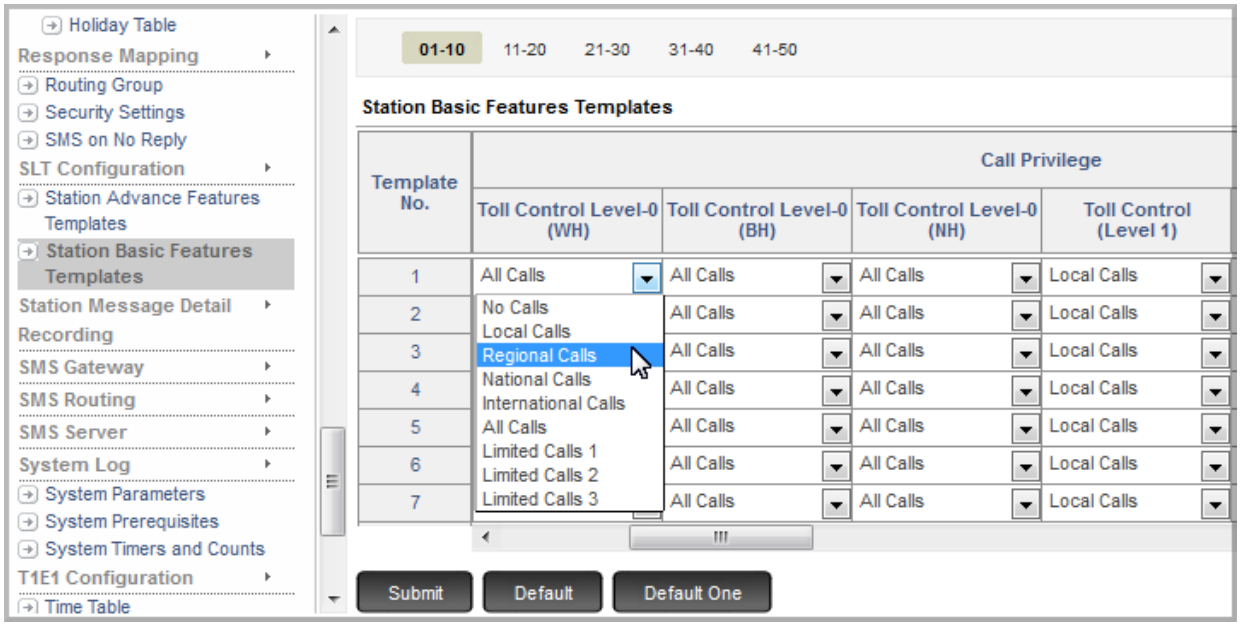
- Enter the specific numbers or digits that are to be allowed to be dialed in the **Allowed Numbers** list.
- Enter the specific numbers or digits that are to be restricted from being dialed in the **Denied Numbers** this list.
- Click **Submit** to save your entries.

It is not mandatory to assign the same Limited-Calls Allowed-List and Denied-List for all Time Zones of Toll Control Level 0 or to other Toll Control Levels. You can prepare different Allowed and Denied Lists for each Toll Control Level.

Toll Control Levels

- Click **Station Basic Feature Template** to open the page.
- By default all stations of SARVAM UCS are assigned Template 01. You may customize this template or select another template.

- Go to Toll Control Level 0.



- Select the desired Call Privilege Type for each Time Zone - Working Hours, Break Hours, Non-Working hours.
- Similarly, select the Call Privilege type for other Toll Control Levels 1, 2 and 3.
- For the type of call privilege you select, the respective number list - Local, Regional, National, International or Limited- you configured will be automatically assigned.
- Click **Submit** at the bottom of the page to save the entries.
- Log out of Jeeves or continue programming.

Programming Toll Control using a Telephone

- Enter SE mode from a DKP/SLT.

Local Numbers

To program Local Numbers Allowed List, dial:

- ***4303-Index-Number String-#****

  Where,

  Index is the location at which the number should be stored in the list, from 001 to 999.

  Number String can be maximum 16 digits and should be terminated with #* if it has fewer than 16 digits.

  Refer the following table for codes for dialing special digits: 0-9, #, *, A, B, C, D, Flash (F), Pause (P), +, Dot (.).

Special Digit	Code
Flash (F)	#2
Pause (P)	#3

Special Digit	Code
A	#4
B	#5
C	#6
D	#7
+	#8
Dot (.)	#9
#	##
*	**

To program Local Numbers Denied List, dial:

- ***4304-Index-Number String-#****

To default the Local Numbers List, dial:

- ***4311-Reverse SE Password***

Regional Numbers

To program Regional Numbers Allowed List, dial:

- ***4305-Index-Number String-#****
  Where,
  Index is the location at which the number should be stored in the list, from 001 to 999.

  Number String can be maximum 16 digits and should be terminated with #* if it has fewer than 16 digits.

  For keying in special digits, refer the above table.

To program Regional Numbers Denied List, dial:

- ***4306-Index-Number String-#****

To default the Regional Numbers List, dial:

- ***4312-Reverse SE Password***

National Numbers

To program National Numbers Allowed List, dial:

- ***4307-Index-Number String-#****
  Where,
  Index is the location at which the number should be stored in the list, from 001 to 999.

  Number String can be maximum 16 digits and should be terminated with #* if it has fewer than 16 digits.

  For keying in special digits, refer the above table.

To program National Numbers Denied List, dial:

- ***4308-Index-Number String-#****

To default the National Numbers List, dial:

- ***4313-Reverse SE Password***

To set the number list to default, dial:

- **4301-1-List Number**
  Where,
  List Number is the number of the List you customized as Allowed/Denied List, from 01 to 16.

To program a number in a list, dial:

- ***4302-List Number-Location Index-Number-#****
  Where,
  List is the number of the List you customized as Allowed/Denied List, from 01 to 16.
  Location Index is the index number at which the number should be stored in the list, from 001 to 999.

  Number is the number string you want to store at this location index on the list. The number can be maximum 16 digits and should be terminated with #* if it has fewer than 16 digits. Refer the following table for codes for dialing special digits: 0-9, #, *, A, B, C, D, Flash (F), Pause (P), +, Dot (.).

Special Digit	Code
Flash (F)	#2
Pause (P)	#3
A	#4
B	#5
C	#6
D	#7
+	#8
Dot (.)	#9
#	##
*	**

To clear a number programmed in a List, dial:

- ***4302-List Number-Location Index-#****
  Where,
  List Number is the number of the list from which the number is to be cleared, from 01 to 16.
  Location Index is the number on this list at which the number to be cleared is stored, from 001 to 999.

Toll Control Levels

To program Toll Control - Level 0 (WH), dial:

- ***5502-1-Template-07-Call Privilege Type***
  Where,
  Template is number of the Station Basic Feature Template, from 01 to 50
  07 is the feature number for Toll Control Level 0 (WH) on the template.
  Call Privilege Type is
  1 for No Calls
  2 for Local Calls
  3 for Regional Calls
  4 for National Calls
  5 for International Calls
  6 for All Calls
  7 for Limited Calls 1

8 for Limited Calls 2
9 for Limited Calls 3

To program Toll Control - Level 0 (WH) Allowed List, dial:

- ***5502-1-Template-08-Number List***

  Where,
  Template is number of the Station Basic Feature Template, from 01 to 50
  08 is the feature number on the template for Toll Control Level 0 (WH) Allowed List for Limited Calls.
  Number List is the number of the list prepared as Allowed List for Limited Calls, from 01 to 16.

To program Toll Control - Level 0 (WH) Denied List, dial:

- ***5502-1-Template-09-Number List***

  Where,
  Template is number of the Station Basic Feature Template, from 01 to 50
  09 is the feature number on the template for Toll Control Level 0 (WH) Denied List for Limited Calls.
  Number List is the number of the list prepared as Denied List for Limited Calls, from 01 to 16.

To program Toll Control - Level 0 (BH), dial:

- ***5502-1-Template-10-Call Privilege Type***

  Where,
  Template is number of the Station Basic Feature Template, from 01 to 50
  10 is the feature number for Toll Control Level 0 (BH) on the template.
  Call Privilege Type is
  1 for No Calls
  2 for Local Calls
  3 for Regional Calls
  4 for National Calls
  5 for International Calls
  6 for All Calls
  7 for Limited Calls 1
  8 for Limited Calls 2
  9 for Limited Calls 3

To program Toll Control - Level 0 (BH) Allowed List, dial:

- ***5502-1-Template-11-Number List***

  Where,
  Template is number of the Station Basic Feature Template, from 01 to 50
  11 is the feature number on the template for Toll Control Level 0 (BH) Allowed List for Limited Calls.
  Number List is the number of the list prepared as Allowed List for Limited Calls, from 01 to 16.

To program Toll Control - Level 0 (BH) Denied List, dial:

- ***5502-1-Template-12-Number List***

  Where,
  Template is number of the Station Basic Feature Template, from 01 to 50
  09 is the feature number on the template for Toll Control Level 0 (BH) Denied List for Limited Calls.
  Number List is the number of the list prepared as Denied List for Limited Calls, from 01 to 16.

To program Toll Control - Level 0 (NH), dial:

- ***5502-1-Template-13-Call Privilege Type***

  Where,
  Template is number of the Station Basic Feature Template, from 01 to 50
  10 is the feature number for Toll Control Level 0 (NH) on the template.
  Call Privilege Type is:

1 for No Calls
2 for Local Calls
3 for Regional Calls
4 for National Calls
5 for International Calls
6 for All Calls
7 for Limited Calls 1
8 for Limited Calls 2
9 for Limited Calls 3

To program Toll Control - Level 0 (NH) Allowed List, dial:

- ***5502-1-Template-14-Number List***
  Where,
  Template is number of the Station Basic Feature Template, from 01 to 50
  11 is the feature number on the template for Toll Control Level 0 (NH) Allowed List for Limited Calls.
  Number List is the number of the list prepared as Allowed List for Limited Calls, from 01 to 16.

To program Toll Control - Level 0 (NH) Denied List, dial:

- ***5502-1-Template-15-Number List***
  Where,
  Template is number of the Station Basic Feature Template, from 01 to 50
  09 is the feature number on the template for Toll Control Level 0 (NH) Denied List for Limited Calls.
  Number List is the number of the list prepared as Denied List for Limited Calls, from 01 to 16.

To program Toll Control - Level 1, dial:

- ***5502-1-Template-16-Call Privilege Type***
  Where,
  Template is number of the Station Basic Feature Template, from 01 to 50
  16 is the feature number for Toll Control Level 1 on the template.
  Call Privilege Type is:
  1 for No Calls
  2 for Local Calls
  3 for Regional Calls
  4 for National Calls
  5 for International Calls
  6 for All Calls
  7 for Limited Calls 1
  8 for Limited Calls 2
  9 for Limited Calls 3

To program Toll Control - Level 2, dial:

- ***5502-1-Template-17-Call Privilege Type***
  Where,
  Template is number of the Station Basic Feature Template, from 01 to 50
  17 is the feature number for Toll Control Level 2 on the template.
  Call Privilege Type is

  1 for No Calls
  2 for Local Calls
  3 for Regional Calls
  4 for National Calls
  5 for International Calls
  6 for All Calls

7 for Limited Calls 1
8 for Limited Calls 2
9 for Limited Calls 3

To program Toll Control - Level 3, dial:

- ***5502-1-Template-18-Call Privilege Type***
  Where,
  Template is number of the Station Basic Feature Template, from 01 to 50
  18 is the feature number for Toll Control Level 3 on the template.
  Call Privilege Type is

  1 for No Calls
  2 for Local Calls
  3 for Regional Calls
  4 for National Calls
  5 for International Calls
  6 for All Calls
  7 for Limited Calls 1
  8 for Limited Calls 2
  9 for Limited Calls 3

To program Toll Control - Call Budget Consumed, dial:

- ***5502-1-Template-19-Call Privilege Type***
  Where,
  Template is number of the Station Basic Feature Template, from 01 to 50
  19 is the number for this feature on the template.
  Call Privilege Type is

  1 for No Calls
  2 for Local Calls
  3 for Regional Calls
  4 for National Calls
  5 for International Calls
  6 for All Calls
  7 for Limited Calls 1
  8 for Limited Calls 2
  9 for Limited Calls 3

- Exit SE mode.

Also refer the topics “Configuring Extensions”, “Station Basic Feature Template”, “Number Lists”.

Trunk Auto Answer

What's this?

Trunk Auto Answer enables calls landing on a trunk to be answered automatically by greeting the caller with a voice message before the call is actually handled.

Trunk Auto Answer is useful when you want callers to remain connected until one of the landing destinations selected for incoming trunk calls becomes free to attend to the caller.

Trunk Auto Answer is useful in call centres, railway enquiry, banks, where callers need to be notified that they would be attended shortly, so that they do not disconnect the call.

SARVAM UCS offers three types of Trunk Auto Answer**:**

- **For all Calls:** the system answers all incoming calls landing on the trunk line.
- **When Busy:** the system answers incoming calls on the trunk, only if the landing destinations are busy.
- **Delayed:** the system first routes the calls to the Trunk Landing Group. If not answered by any extension, the call is answered by the system.

How it works

SARVAM UCS handles incoming calls on the trunk according to the type of Trunk Auto Answer selected for the trunk: **For all Calls** or **When Busy** or **Delayed.**

When **Trunk Auto Answer–For all Calls** is enabled on a Trunk, for each incoming call on the trunk,

- The System answers the call with a Greeting message, known as the *Trunk Auto Answer Greeting*, and rings the landing destination selected for the time of the day.

  The system starts the *Built-In Auto Attendant Inactivity Timer* (default: 60 seconds). The Trunk Auto Greeting message is played once. You may assign a Trunk Auto Answer Greeting of your preference.

- If the landing destination does not answer before the Trunk Auto Answer Greeting message ends, the system plays *Trunk Auto Answer Ring Back Tone* message to the caller.

  The Ring Back Tone message is played repeatedly for the duration of the Built-In Auto Attendant Inactivity Timer.

  However, if no Trunk Auto Answer Ring Back Tone message is assigned, the system will plays Ring Back Tone to the caller for the duration of this timer.

- If any of the landing destinations answers the call before the expiry of the Built-In Auto Attendant Inactivity Timer, the system stops the Built-In Auto Attendant Inactivity Timer and the Ring Back Tone message, and connects the caller to the extension that answered the call.

- If none of the landing extensions answers the call before the expiry of the Built-In Auto Attendant Inactivity Timer, the system plays the *Trunk Auto Answer Busy Bye* message and releases the trunk port.

If no Trunk Auto Answer Busy Bye message is assigned, the system plays the Busy Tone for the duration of the Busy Tone Timer and releases the trunk port.

When **Trunk Auto Answer–When Busy** is enabled on a Trunk, for each incoming call on the trunk,

- The System answers the with the Trunk Auto Answer Greeting message and loads the Built-In Auto Attendant Inactivity Timer.

  The Greeting message is played once.

- The System waits for any of the landing destinations selected for the time of the day to be free.

- If no landing destination is free at the end of the Trunk Auto Answer Greeting message, the system plays Ring Back Tone or *Trunk Auto Answer Ring Back Tone* message, if assigned, to the caller for the duration of the Built-In Auto Attendant Inactivity Timer.

- If any of the landing destinations is free before the expiry of the Built-In Auto Attendant Inactivity Timer, the system places the call on that destination.

- If none of the landing destinations is free at the end of the Built-In Auto Attendant Inactivity Timer, the system plays the Trunk Auto Answer Busy Bye message, if assigned, and releases the trunk port.

  If the Busy Bye message is not assigned, the system will play the Busy Tone to the caller for the duration of the Busy Tone Timer.

When **Trunk Auto Answer–Delayed** is enabled on a Trunk, for each incoming call on the trunk,

- The system first routes the incoming calls to the Trunk Landing Group. It waits for the duration of the *Delayed Trunk Auto Answer Timer* (programmable; default:10 seconds) for any of the extensions in the Trunk Landing Group to answer the call.

- If the call is not answered by any of the extensions, the System answers the call with a Greeting message, known as the *Trunk Auto Answer Greeting*, and rings the landing destination selected for the time of the day.

  The system starts the *Built-In Auto Attendant Inactivity Timer* (default: 60 seconds). The Trunk Auto Greeting message is played once. You may assign a Trunk Auto Answer Greeting of your preference.

- If the landing destination does not answer before the Trunk Auto Answer Greeting message ends, the system plays *Trunk Auto Answer Ring Back Tone* message to the caller.

  The Ring Back Tone message is played repeatedly for the duration of the Built-In Auto Attendant Inactivity Timer.

  However, if no Trunk Auto Answer Ring Back Tone message is assigned, the system will plays Ring Back Tone to the caller for the duration of this timer.

- If any of the landing destinations answers the call before the expiry of the Built-In Auto Attendant Inactivity Timer, the system stops the Built-In Auto Attendant Inactivity Timer and the Ring Back Tone message, and connects the caller to the extension that answered the call.

- If none of the landing extensions answers the call before the expiry of the Built-In Auto Attendant Inactivity Timer, the system plays the *Trunk Auto Answer Busy Bye* message and releases the trunk port.

If no Trunk Auto Answer Busy Bye message is assigned, the system plays the Busy Tone for the duration of the Busy Tone Timer and releases the trunk port.

How to configure

For this feature to work, you must do the following:

- Enable Trunk Auto Answer on the Trunk Feature Template of the desired trunks. To know more and for instructions, see “Trunk Feature Template”.

- Select the **Trunk Auto Answer Greeting message**, the **Trunk Auto Answer Ring Back Tone Message**, and the **Trunk Auto Answer Busy Bye Message** for the Working Hours, Break Hours and Non-Working Hours.

  You may select different Greeting, Ring Back Tone and Busy Bye Message for each time zone. You can also select Music-On-Hold instead of Ring Back Tone Message for the time zones.

  Configure the **Delayed Trunk Auto Answer** timer, if required.

- Configure the Trunk Auto Answer related **Timers**, if required. The following Timers are of relevance to the Trunk Auto Answer Feature:
  - The Built-In Auto Attendant Inactivity Timer (default: 60 seconds)
  - The Ring Back Tone Timer (default: 45 seconds)
  - The Busy Tone Timer (default: 7 seconds)

  - You may change the duration of these timers from the “System Timers and Counts” page.

The Ring Back Timer and the Busy Tone Timer are also applicable for the Ring Back Tone and the Busy Tone played for internal calls.

- Record and assign Voice Modules for the following Voice Messages related to this feature:
  - Trunk Auto Answer Greeting Message
  - Trunk Auto Answer Ring Back Tone Message
  - Trunk Auto Answer Busy Bye Message

  For each of these messages, you can record four different messages.

  See the topic “Voice Message Applications” for instructions on recording and assigning voice modules to greeting messages.

Trunk Call Waiting

What's this?

The Trunk Call Waiting feature gives indication to the user of a busy extension about the waiting call on a Trunk.

This is a "Class of Service (COS)" dependent feature. Only those extensions which have this feature enabled in the COS allowed to them, will be given indication of the incoming call waiting on the trunk.

How it works

This feature is not supported on SIP Extensions.

- A and B are extensions.
- Trunk Call Waiting feature is enabled in the Class of Service of B but not on A.
- There is an incoming call on a trunk for B.
- B is busy on a call with A.
- SARVAM UCS plays Beeps to B to indicate the call waiting.
- To answer the waiting call, B may dial Flash or press the Transfer key.
- B will be connected to the caller.
- A will be put on hold.
- When there is an incoming call on a trunk for A, but A is busy on another call, A will not be provided any indication of the waiting call.

How to configure

For Trunk Call Waiting to work on an extension, it must be enabled in the Class of Service allowed to that extension. This can be done using Jeeves as well as a Telephone.

In the default factory settings, Station Basic Feature Template Number 01 is assigned to all the extensions of SARVAM UCS. Template 01 is assigned CoS group 01. Trunk Call Waiting is disabled in the CoS. So, none of the extensions of SARVAM UCS are provided call waiting indication for incoming trunk calls.

If you want to allow Trunk Call Waiting uniformly to all extensions of SARVAM UCS, simply enable this feature in the default CoS group (01) in the default Template (01).

However, if you want to allow this feature to only selected extensions,

1. Define a CoS group with Trunk Call Waiting enabled.
2. Prepare a Station Basic Feature Template with this CoS group applicable in all the "Time Zones".
3. Assign this new Template to the extensions to which Trunk Call Waiting is to be allowed.

Refer the topics "Class of Service (COS)" and "Station Basic Feature Template" for programming instructions.

Trunk Landing Group (TLG)

What's this?

A Trunk Landing Group is a group of extensions on which incoming calls on a particular trunk are landed.

Trunk Landing Groups are formed for efficient call management. Generally, incoming calls on a trunk are landed on the Operator extensions. However, when several trunks are interfaced with the system, it becomes difficult for the operator to answer all calls efficiently. Trunk Landing Groups relieve the Operator to a great extent, as the incoming calls get distributed among several extensions.

How it works

- A Trunk Landing Group (TLG) is a "Routing Group".
- You can configure as many as 95 TLGs. Each group is numbered from 01 to 96.
- A maximum of 32 stations — SLT, DKP, SIP, ISDN, OGTB or Virtual Extensions — can included in each Trunk Landing Group.

  To use the Gateway Application of SARVAM UCS, select OGTB as the station.
- For each group that you create, you can do the following:
  - set the Sequence in which the stations in the group should ring, by selecting the member stations in a sequence from 1 to 32.
  - set the Time for which each station in the group should ring, by setting the *Ring Timer* (default: 15 seconds).
  - set each station to ring continuously till the call matures by enabling *Continuous Ring* (default: disabled).

    When Continuous Ring is enabled, once a station receives a ring, it rings continuously till the call matures. The station continues to ring even as other stations in the group are hunted.

    If the call is not answered even after the last station in the group has been hunted, the system will loop back and start hunting from the first station, all over again.
  - have a number of stations in the group ring simultaneously by enabling *Continuous Ring* on these stations and setting the *Ring Timer* for these stations to *'00'* seconds.
  - set equal distribution of incoming calls on all stations in the group, by enabling *Rotation* for the entire group (default: disabled).

    When Rotation is enabled on a TLG, for each new call on a trunk, the system will land the call on the extension next to the one that received the last call.

    When Rotation is disabled in a TLG, for each new call on a trunk, the system will land the call on the first free station of the TLG.

- To each Trunk, you must assign a TLG for the Time Zones, working hours, break hours and non-working hours. You may assign the same TLG for all three Time Zones, or a different TLG for each Time Zone.

How to configure

- According to the number of trunks interfaced with your SARVAM UCS and the number of extensions you have, identify the trunks to which you want to assign TLG for each Time Zone. This will help you decide the number of TLGs to be formed, the type and number of extensions in each group, and their sequence.
- Decide the Trunk number to which each TLG is to be assigned.
- Configure each TLG as a Routing Group. See “Routing Group” for instructions.
- Assign the TLGs you formed for each trunk for the three Time Zones in its Trunk Feature Template. for instructions, see “Trunk Feature Template”.

Example:

Two CO Lines (configured on software ports 001 and 002) are interfaced with SARVAM UCS.

- Incoming calls on CO 1 during working hours should land on SLT stations 2001, 2003, 2005 (configured on software ports 008, 010, 012 respectively).

  Incoming calls on CO 1during break hours and non-working hours should land on SLT stations 2002, 2004 (configured on software ports 009, 011 respectively).
- Incoming calls on CO 2 should land always on DKP stations 3001, 3002, 3003, 3009, 3010 (configured on software ports 013, 014, 015, 016, 017).
- Incoming call on CO 1 should ring for 10 seconds on each station in the TLG.
- Incoming call on CO 2 should ring for 20 seconds on each station in the TLG.
- The stations of the TLGs of CO 1 and CO2 should ring for the set time only.
- “Rotation Method” to be followed in the TLG of CO1 and CO2.

In this example, for CO 1, you would need to form 2 TLGs; one TLG for working hours and one for break hours and non-working hours. So, form two Routing Groups. For example, Routing Group 10 for working hours and Routing Group 11 for break hours and non-working hours.

In Routing Group 10, select the member SLTs in this sequence: 2001, 2003 and 2005. Set the Ring Timer for each member SLT to 10 seconds. Disable Continuous Ring for the SLT, as each station in the group must ring for the set time.

In Routing Group 11, select the member SLT in this sequence: 2002 and 2004, and set the Ring Timer for each SLT to 10 seconds. Disable Continuous Ring for the SLT, as each station in the group is required to ring for the set time only.

Enable Rotation for Routing Group 10 and Routing Group 11.

For CO 2 you would need to form a common TLG for working hours, break hours and non-working hours. So, configure as single routing group, for example, Routing Group 13 for CO 2. In Routing Group 13, select the

member stations in this sequence: 3001, 3002, 3003, 3009, 3010. Set the Ring Timer for each DKP member station to 20 seconds. Disable Continuous ring for each DKP in the group. Enable Rotation for Routing Group 13.

Select a Trunk Feature Template number for CO 1 and CO 2. For example, feature template 04 for CO 1 and template 05 for CO 2.

In the Feature Template of CO 1, assign TLG. For working hours, assign Routing Group 10, for break and non-working hours, assign Routing Group 11.

In the feature Template of CO 2, assign Routing group 13 as TLG for all three time zones.

Trunk Reservation

What's this?

This feature enables any extension user to reserve a trunk for exclusive use, for a specific time period.

Trunk Reservation can be requested from an SLT extension, a DKP extension and from an Extended IP Phone extension.

This feature is not supported on SIP Trunks.

How it works

Let us understand this feature with the help of an example:

In an organization there are four CO trunk lines, CO 1, 2, 3 and 4, but all of these have full traffic throughout the day.

Extension user A is a sales executive. To complete the sales target, A needs to make long-distance calls to customers. Since there is full traffic on all the four trunks throughout the day, and these trunks are constantly busy, A would need a dedicated trunk line to save time and complete the target.

So, A can reserve one of the four trunk lines for the desired duration. To do this,

- Extension A grabs a trunk by dialing the Trunk Access Code.
- The Trunk is busy.
- Extension A dials the feature access code for Trunk Reservation for the busy trunk.
- As soon as the trunk is free, A's extension rings.
- A answers the call and gets connected to the trunk, and gets dial tone.
- A can now make as many calls as required.
- The trunk remains reserved for the duration of the Trunk Reservation Timer. This timer is configurable, and by default is set to 10 minutes. A can have this Timer configured to the desired duration.
- All other extension users who try to access this trunk get error tone, even if this trunk is free.
- If A is finished with the calls before the expiry of the Trunk Reservation Timer, A has two options:
  a) release the trunk manually, by cancelling Trunk Reservation.
  or
  b) wait for the expiry of Trunk Reservation Timer.
- Only when the trunk is released (by A or at the end of the Timer) will other users be able to access it.

How to configure

To be able to use Trunk Reservation, extension users must have this feature enabled in their Class of Service for the time zone. For configuration instructions, see “Class of Service (COS)”, and “Station Basic Feature Template”.

You may increase or decrease the duration of the Trunk Reservation Timer. See “System Timers and Counts” for instructions.

How to use

For EON & Extended IP Phone Users

To set Trunk Reservation, when Trunk you access is busy,

- Press DSS Key assigned to Trunk Reservation

  OR

- Dial 6 on Busy Tone

To release a reserved Trunk, wait for the Timer to expire, or cancel Trunk Reservation, manually.

To cancel Trunk Reservation Manually,

- Press Call Back Key / DSS Key assigned to Auto Call Back.

  OR

- Dial 102

For SLT Users

To set Trunk Reservation, when Trunk you access is busy,

- Lift handset.
- Dial 6 on Busy Tone.
- Replace handset.

To cancel Trunk Reservation Manually,

- Lift handset.
- Dial 102
- Replace handset.

User Absent/Present

What's this?

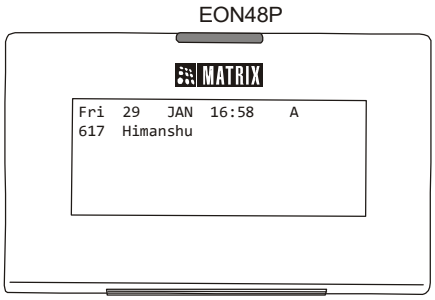
Extension users may sometimes want to leave their desks, and expecting to return soon, they may not have forwarded their calls or set Do Not Disturb on their extensions. In such cases, incoming calls will continue to land on the extension and go unanswered. The callers have no way of knowing that the extension user is not present at the extension and may try the extension number repeatedly.

With the User Absent/Present feature of SARVAM UCS, extension users, including the Operator, can set 'User Absent' when they leave their desks. By doing so, they can block all incoming external as well as internal calls from landing on their extension. When they return to their desks, they can set 'User Present' and receive incoming calls again.

There are more options for indicating availability to other extensions. Refer the topic "Presence" to know more.

How it works

When an extension user of EON/Extended IP Phone user sets 'User Absent', the letter 'A' appears on the phone's display:



The letter 'A' disappears when the extension user sets 'User Present'.

When an extension user of DKP/Extended IP Phone calls the extension which has set 'User Absent', the text message 'User Absent' will appear on the caller's phone display.

When an SLT extension user calls the extension which has set 'User Absent', callers who dial this extension will get an error tone.

External callers who call the extension, on which 'User Absent' is set, will get an error tone only.

- *Outgoing calls can be made from the extension which has set 'User Absent'. Only incoming calls are restricted.*
- *User Absent/Present can be set on an extension from the SA mode.*
- *If more than one extension is configured as "Operator' (routing group), incoming calls will be blocked only on the Operator extension which has set User Absent.*

- *User Password is required for this feature. The default User Password, 1111, will not work. Change the User Password first.*

How to use

For Extension Users

To set User Absent on your extension:

- Dial ***104-User Password-0***

To set User Present on your extension:

- Dial ***104-User Password-1***

For System Administrator

To set User Absent on an extension:

- Dial ***1072-014-Extension Number-0***

To set User Present on an extension:

- Dial ***1072-014-Extension Number-1***

The System Administrator can also set an extension as Absent/Present using Jeeves. For instructions, read "Additional Security to Extension Users" under the topic *System Security.*

User Password

What's this?

The User Password is a 4-digit code for extension users to protect their extension phones from unauthorized use. The default User Password is ***1111***. It must be changed by the extension users from their phones to any desired value, not exceeding 4 digits.

To avoid unauthorized access,

- the extension users must change the default password.
- make sure the new password is strong and is kept confidential.
- change the password regularly.

In case the extension user forgets the password, it can be cleared and restored to the default value *1111* by the System Engineer (SE) or the System Administrator (SA). Refer the topic "System Security" for instructions.

The User Password is also required to access and use certain features of SARVAM UCS, which are listed below.

- Call Follow Me
- Dynamic Lock
- Direct Inward System Access (DISA)
- Walk-In Class of Service
- Presence
- Hot Desk
- Phone Settings of the DKP
- Mailbox of Voice Mail

The extension user must change the default password for all the above listed features except: Phone Settings, Mailbox of Voice Mail. Both these features allow the extension user to use the default User Password, whereas in the case of others, the system will not allow feature access without changing the User Password.

In the case of Hot Desking, the default password will work only for one extension involved.

The User Password for an extension can be changed only from that extension phone only.

Since the Mailbox can be accessed using the default User Password, extension users who are assigned a mailbox are recommended to change their User Password to prevent unauthorized access to their mailbox.

How to use

For EON and Extended IP Phone Users

- Press 'Enter' key to enter DKP menu.
- Scroll to Change User Password.
- Press 'Enter' key.
- You get the prompt: Enter Old User Password.
- Dial you current user password.
  If the default password *1111* has not be changed, enter default password.
- You get the prompt: Enter New User Password.
- Dial your new user password, not exceeding 4-digits.
  You get confirmation tone and the confirmatory message 'User Password Changed' on your phone display.

For SLT Users

- Lift the handset.
- Dial 114
- Dial Current User Password (default password *1111* if not changed already).
- Dial New User Password.
- You get confirmation tone.
- Replace handset.

Video Call

What's this?

Video calling has become an increasingly important tool in today's business world. It offers people the power of face-to-face communication, at reasonable cost without incurring the expense of traveling. SARVAM UCS allows you to make and receive video calls by connecting video capable user terminals.

Among the various UC features offered by SARVAM UCS, Video calling is the most important.

SARVAM UCS supports video calling over:

- T1E1 PRI Trunks and ISDN Terminals
- SIP Trunks and SIP Extensions

How it Works

Video Calling over T1E1 PRI Trunks and ISDN Terminals

For video calling over T1E1 PRI Trunks, you require the following:

- Connect the T1E1 PRI Trunks to SARVAM UCS.
- Make sure you have a BRI Card installed in the system and set the Orientation of the BRI Port as Network.
- Connect ISDN Terminals with video capability to the BRI ports.

The minimum bandwidth required for a video stream to flow is 128 kbps. Hence, 2 BRI channels will be consumed. When this video stream flows through the PRI line it will consume 2 PRI channels too. However, for higher resolution you can connect multiple BRI ports, if supported by the ISDN Terminal.

To conduct a video call,

- From the ISDN Terminal dial the number of the desired party (other ISDN Terminal). This is a one to one call.

- You can also make a video call from one ISDN Terminal of SARVAM UCS to another ISDN Terminal of SARVAM UCS. In this case 4 BRI channels will be consumed.

How to configure

To conduct a video call you must configure the following:

- Assign Access Codes to the ISDN Terminals.
- Assign the BRI ports to the ISDN Terminals.
- For outgoing calls, assign a Station Basic Feature Template to the ISDN Terminal. Make sure the Outgoing Trunk Bundle Group in the Station Basic Feature Template has the desired PRI lines as members to route the outgoing calls.
- For incoming calls, configure the Outgoing Reference ID, the Incoming Reference ID and the DDI Routing Table. Configure the same so that the incoming calls on the T1E1 ports are routed directly to the ISDN Terminals.

For detailed information, see “Configuring BRI Trunks”, “Configuring ISDN Terminals”, “Configuring E1 Trunks”, “Configuring T1 Trunks” and “Direct Dialing-In (DDI)”.

To reduce call processing time, it is recommended that you enable Number based LCR on the OG Trunk Bundle Group assigned in the Station Basic Feature Template. You must also configure the Number Based LCR Table.

Video Calling over SIP Trunks and SIP Extensions

SARVAM UCS supports video calling for SIP to SIP calls only. SARVAM UCS only acts as a relay agent to place video calls between two video capable user terminals with no additional supplementary features.

Video capable user terminals which are communicating through SARVAM UCS must confirm to the same type of video codecs[343] as per industry standards. If video codec negotiation is not successful between the end-to-end video terminals, either the call will be converted into an audio call or the call will be dropped.

For enhanced video quality during the call, some of the advanced video attributes including profile/level (H.263+, MPEG4, H.264), bandwidth, standard annexes, frame rate, image size must be confirmed between the communicating video terminals. SARVAM UCS will not modify or alter any of these attributes which can effect the characteristics of the video stream during communication.

For a video call, two VOCODER channels[344] of SARVAM UCS are consumed. During the entire video call, those VOCODER channels are reserved and used till either of the parties disconnects the call.

Since SARVAM UCS support a maximum of 128 VOCODER channels. A maximum of 55[345] simultaneous video calls are supported.

You can convert a normal SIP to SIP audio call into a video call only if both the SIP extensions support video calling.

- *Video calling is supported in VARTA UC Clients.*
- *None of the proprietary Extended SIP Phones of Matrix support video calling.*
- *SRTP is not supported for video calling.*

Feature Interactions

When the below mentioned features are accessed during an ongoing Video call, the call will be established into an Audio call:

- Features where-in the system establishes a conference such as 3-Party Conference, Multiparty Conference, Conversation Recording, Call Taping, Raid etc.
- When users access Barge-In or Forced Answer.
- When users unhold, unpark or Blind Transfer the call.
- When system answers the call on a trunk, such as DID, DISA, Trunk Auto Answer, Callback on Trunk etc.

When the below mentioned features are accessed the call will always be an Audio call:

- When users make a Paging or Meet Me Paging call, access Voicemail , initiate an Emergency Conference or make an Intercom request.
- Features where-in the system initiates the call such as ACB Return call, Auto Redial Return call, Alarm call, Handover call etc.

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343. Current industry standard video codecs include H.261, H.263, H.263p, H.264, MPEG4 etc. For more details, refer the documentation of the corresponding video terminals you are using for video calling.

344. The Channels will be reserved only for Transcoding Mode.

345. For PENX, maximum 16 Video Calls are supported in SARVAM SMB.

- *To dial Flash, make sure your IP Phone supports Flash dialing.*
- *You can use #2, if your IP Phone does not support Flash dialing.*

Virtual Extension

What's this?

The Virtual Extension feature of SARVAM UCS enables multiple users to share one telephone instrument as their extension, yet be considered as individual extensions by the system, with distinct extension properties and class of service.

Such shared extensions are called Virtual Extensions, as their users do not have individual phones for their use.

Virtual Extensions are useful in laboratories, common rooms, dormitories, shop floors, and wherever it is not feasible to provide dedicated telephone instruments to individual extension users. Virtual extensions allow you make optimum use of the existing phones without investing in new ones.

How it works

The shared telephone instrument is called the Master Extension. A Master Extension can be an SLT, a DKP or the Matrix Extended IP Phone.

- Virtual Extensions are assigned to the Master Extension. A Master Extension can have multiple Virtual Extensions, but a Virtual Extension can have only one Master Extension.
- The Virtual Extension functions as any other extension of SARVAM UCS. It can be assigned all features and facilities, like Class of Service, Toll Control, Call Forward, just like any other physical extension of SARVAM UCS. You can assign Station Basic Feature Template and Advanced Feature Template to the Virtual Extension.
- Incoming calls to a Virtual Extension will ring on the Master Extension.
- All incoming, outgoing, internal and external calls of the Virtual Extensions are recorded in the Station Message Detail Records.
- To make outgoing calls, the Virtual Extension user must use the feature "Walk-In Class of Service".
- The Virtual Extension user is logged out of the Master Extension according to the Walk Out mode assigned to it: *Walk out single call* or *Walk out multiple calls.*

How to configure

For this feature to work, you must do the following:

- Make a list of the number of Virtual Extensions required by you along with their names and numbers.
- Decide the Master Extension (landing destination) that is, the Port Type and Port Number. The Port Type may be SLT, DKP, ISDN Terminal, or SIP Extension. Port Number is the number of the software port to which the landing destination extension is connected.

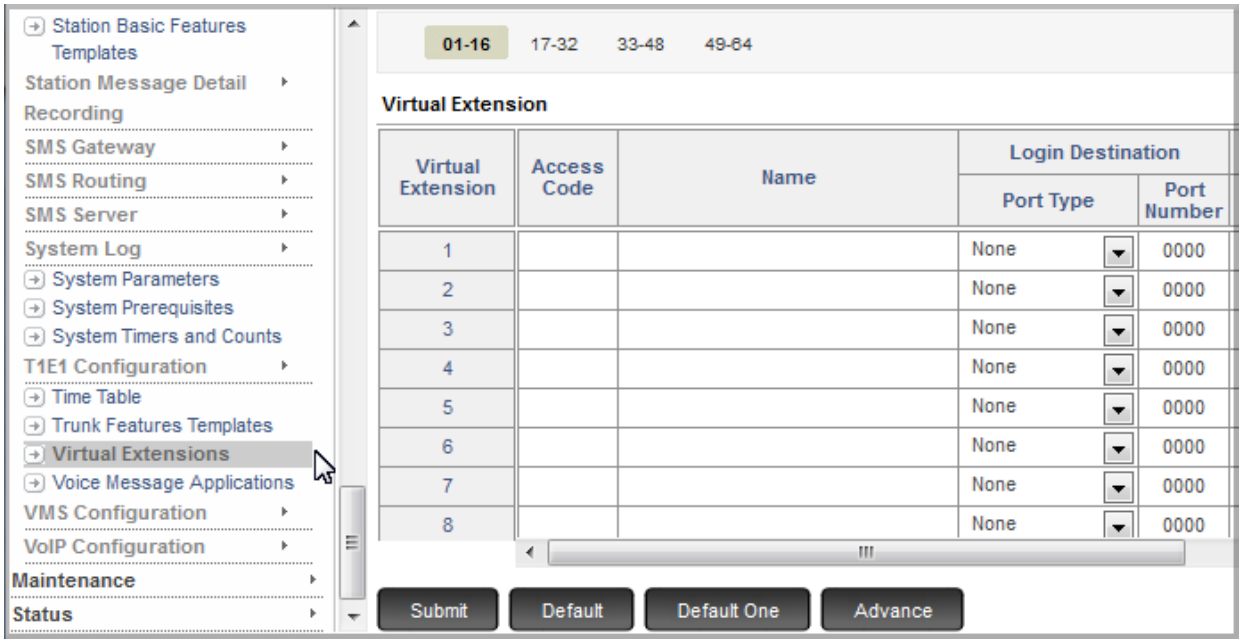
- If required, you can customize the Station Basic Feature Template and Station Advanced Feature Template you want to assign to the Virtual Extensions. For making outgoing calls users of Virtual Extensions must have "Walk-In Class of Service" enabled in their COS.

- Decide the Priority you want to assign to the Virtual Extensions. When an outgoing call is made from any Virtual Extension, the ring type played to the called party will be as per the set priority.

Configuring Virtual Extensions using Jeeves

- Log in as System Engineer.

- Under **Configuration**, click **Virtual Extensions**.



Configure the following parameters for each Virtual Extension:

- **Access Code:** Assign Station Access Codes to the Virtual Extensions. Station Access Codes are commonly referred to as Extension Numbers. These may be a combination of 1, 2, 3, 4, 5 and 6 digits, which are dialed to call the Virtual Extension to which they are assigned.

  To assign Station Access Codes according to your preference and requirement to a range of Virtual Extensions, see "Assigning Access Codes to a Range of Extensions".

If you decide to customize the Station Access Codes, make sure that the numbers do not clash with any other Access Code in the 'Dial' phase. Refer the topics "Access Codes" and "Conflict Dialing" to know more.

- **Name:** Assign a 'Name' to the Virtual Extension. The name may be of the person who will use the extension. This name will be displayed on the LCD of the remote user's phone, if it is equipped with Caller ID.

  You can program a name of a maximum of 18 alphanumeric characters.

- **Login Destination:** Configure the **Port Type** and **Port Number** you want the Virtual Extension user to log into to make outgoing calls.

- **Landing Destination:** Configure the **Port Type** and **Port Number on which** you want the Virtual Extension user to receive incoming calls.

- **Priority:** Select a Priority Level for the Virtual Extension.

  Each extension of the SARVAM UCS is assigned a Priority Level starting from 1, 2, 3... to 9, with '1' being lowest Priority and '9' being highest Priority. Whenever an extension phone with higher priority calls an extension with lower priority, a triple ring is placed on the called station. To know more, read the feature description “Priority”.

  By default, the Priority of all Virtual Extensions is set to '1-None'. So, decide what Priority Level you will assign to each of the Virtual Extensions and set the desired level for each extension.

Advanced Configuration Parameters

- Click the **Advanced** button at the bottom of the page and configure the following parameters:

  - **Mobile Number:** Enter the Mobile Number of the extension user you wish to store. The Number can be a maximum of 16 digits.

  - **Email ID:** Enter the Email ID of the extension user you wish to store. The Email ID can be a maximum of 64 characters.

  - **Group:** You can assign the extension user to a **Group**. The system clubs together extension users assigned the same Group. The Group can be a maximum of 16 characters. Default: Blank.

- If you have completed configuring the parameters, click **Submit** at the bottom of the page to save your settings.

  It is possible to default all the parameters by clicking the **Default** button. You can also restore default values of the parameters of a single Virtual Extension by clicking the **Default One** button and specifying the Virtual Extension Number you want to set to default.

Configuring Virtual Extension Parameters using a Telephone

To assign Access Code to a Virtual Extension, dial:

- ***3109-1-Virtual Extension-Access Code-#**** to assign the Access Codes to a single Virtual Extension port.
- **3109-2-Virtual Extension-Virtual Extension-Access Code** to assign the Access Codes for a range of Virtual Extension ports.
- **3109-1-Virtual Extension-Access Code-#*** to assign the Access Codes to all the Virtual Extension ports.
  Where,
  Virtual Extension is from 01 to 64.
  Access Code is a number string of any combination of 1, 2, 3 or 4 digits. Terminate the command with #* if the number string has fewer than 4 digits.

To clear the access codes for all the Virtual Extension, dial:

- **3109-***

To assign a Name to an Virtual Extension, dial:

- ***5414-1-Virtual Extension-Name-#**** to assign a Name to a single Virtual Extension port.
- ***5414-2-Virtual Extension-Virtual Extension-Name-#**** to assign the same Name to a range of Virtual Extension ports.
- ***5414-*-Name-#**** to assign the same Name to all Virtual Extension ports.
  Where,
  Virtual Extension is the Software Port number of the Virtual Extension port from 001 to 512.
  Name is a string of a maximum 18 alphanumeric characters. Terminate the commands with #* if the number string has fewer than 18 characters.

To clear the name of the Virtual Extension, dial:

- ***5414-1-Virtual Extension-#**** to clear the Name of a single Virtual Extension port.
- ***5414-2-Virtual Extension-Virtual Extension-#**** to clear the Names of a range of Virtual Extension ports.
- ***5414-*-#**** to clear the Names of all Virtual Extension ports.

To assign a Station Basic Feature Template to a Virtual Extension, dial:

- ***5613-1-Virtual Extension-Template Number*** to assign a Template to a single Virtual Extension.
- ***5613-2-Virtual Extension-Virtual Extension-Template Number*** to assign the same Template to a range of Virtual Extension.
- ***5613-*-Template Number*** to assign the same Template to all Virtual Extension.
  Where,
  Virtual Extension is from 01 to 64.
  Template is the Station Basic Feature Template from 01 to 50. Default: 01.

To assign a Station Advanced Feature Template to a Virtual Extension, dial:

- ***5611-1-Virtual Extension-Station Advanced Feature Template*** to assign a Template to a single Virtual Extension.
- ***5611-2-Virtual Extension-Virtual Extension-Station Advanced Feature Template*** to assign the same Template to a range of Virtual Extensions.
- ***5611-*-Station Advanced Feature Template*** to assign the same Template to all Virtual Extensions.
  Where,
  Virtual Extension is from 01 to 64.
  Station Advanced Feature Template is the Station Advanced Feature Template from 01 to 50. Default: 01.

To define the Priority for a Virtual Extension, dial:

- ***3919-1-Virtual Extension-Priority*** to define Priority for a single Virtual Extension.
- ***3919-2-Virtual Extension-Virtual Extension-Priority*** to define the same Priority for a range of Virtual Extensions.
- ***3919-*-Priority*** to define the same Priority for all Virtual Extensions.
  Where,
  Virtual Extension is from 01 to 64.
  Priority is from 1 to 9. Default: 1-None.

To program Landing Destination for Virtual Extension, dial:

- **3001-1-Virtual Extension-Port Type-Port Number** to program landing destination for a single Virtual Extension.
- **3001-2-Virtual Extension- Virtual Extension -Port Access Code - Port Number** to program landing destination for a range of Virtual Extensions.
- **3001-*-Virtual Extension-Port Access Code-Port Number** to program landing destination for all the Virtual Extensions.
  Where,

Virtual Extension is from 01 to 64.
Port Type and Port Number is:

Port Type	Port Access Code	Port Number
SLT	01	001-512
DKP	02	001-128
ISDN Terminal	28	01-64
SIP Extension	34	001-500

Voice Help

What's this?

The Voice Help feature of SARVAM UCS allows you to record and play short (16 second duration) voice messages to provide quick help to extension users. Voice Help messages may contain important instructions, or frequently accessed feature codes, or important phone numbers, etc.

For example, the Access Codes of the frequently used features can be recorded in the Voice Help message. Extension users may simply dial the Voice Help feature code and listen to the voice message.

How to configure

To be able to use Voice Help, you must first record a voice module with the contents you wish to provide as help. Record the voice module considering the maximum duration of the voice module, so that the message is not truncated.

The voice module must be assigned to the Voice Help application. Refer topic "Voice Message Applications" for instructions on recording the voice module and assigning it to Voice Help.

How to use

For EON and Extended IP Phone Users

- Press DSS Key assigned to Voice Help.
  OR
- Dial ***1090***
  You will hear the Voice Help message.
- Go idle after the message ends.

For SLT Users

- Lift the handset.
- Dial ***1090***.
- You will hear the Voice Help message.
- Replace handset after the message ends.

Voice Message Applications

What's this?

SARVAM UCS allows you to record different voice messages which can be played to callers/extension users according to the situation. For example, if you have activated "Auto Attendant" on some of the trunk lines, you can use an appropriately recorded Voice Message to guide callers to reach the desired party/destination extension. Similarly, an extension user who wants to set an alarm can record a personal voice message on his/her own, which will be played to him/her when the alarm request is served.

If RTP mode is set as RTP Relay or Direct RTP and when calls are made to/from SIP, voice messages will not be played for the following tones — Dial Tone, Ringback Tone, Busy Tone, Error Tone and Confirmation Tone. See RTP Mode under "Configuring VoIP Parameters" for more details.

How it works

The voice messages are recorded in 'Voice Modules' and the voice modules are assigned to the features/ applications with which they are to be used.

The SARVAM UCS supports 16 Voice Modules of a maximum duration of 16 seconds each as well as a customized option wherein you can customize the module size as per your requirements. By default, the option is 16 Voice Modules of 16 seconds. You can record up to 16 short messages of a maximum 16 seconds each or less or you can record a single message of maximum 240 sec. The total size of the Voice module is 256 sec.

A voice message can be of two types:

- **Once-Only:** the message is played only once from its start to its end.
- **Continuous:** the message is played repeatedly from the start to the end.

The Voice Messages you. recorded can also be Uploaded and Downloaded through Jeeves. For instructions, see "How to configure".

When the recorded voice modules are assigned to the features/applications, they are played to the callers/ extension users whenever the feature/application is activated.

As many as nine[346] different voice messages can be played simultaneously to a caller/extension user.

Voice messages can be used for different applications or situations as described in the following.

Built-In Auto Attendant Greeting Messages

On Trunk lines that have Built-In Auto Attendant enabled, you can use a recorded Voice Message to guide callers to reach the desired party/destination extension. It is possible to play Built-In Auto Attendant Welcome Greeting Messages according to the time zone, that is, working hours, break hours, and non-working hours.

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346. In ETERNITY LENX/MENX, 11 different voice messages can be played simultaneously to the caller/extension user. In ETERNITY PENX 05 different voice messages can be played simultaneously to the caller/extension user.

For example:

- **Built-In Auto Attendant-Time-based Greeting message in the morning:** Good Morning.
- **Built-In Auto Attendant-Time-based Greeting message in the afternoon:** Good Afternoon.
- **Built-In Auto Attendant-Time-based Greeting message in the evening:** Good Evening.
- **Built-In Auto Attendant - Welcome Greeting message for working hours:** “Welcome to Cotton Software”.
- **Built-In Auto Attendant - Welcome Greeting message for lunch hours:** “Welcome to Cotton Software. This is lunchtime. Please call after 2.00 pm”.
- **Built-In Auto Attendant - Welcome Greeting Message for non-working hours:** “Welcome to Cotton Software. We are closed for the day. Please call later”.
- **Built-In Auto Attendant - Dial Message:** “Please dial the desired station number”.

Built-In Auto Attendant Guidance Messages

You can use voice messages to guide Auto Attendant callers at various stages of the call. For instance, prompting them to dial a number, or giving an alert when they dial a wrong or invalid number, or informing them when the desired destination extension is busy or when there is no response from the dialed extension, etc.

For example,

- **Built-In Auto Attendant - Dial Message:** "Please dial the desired Extension Number".
- **Built-In Auto Attendant - Wrong Dial Message:** "Sorry you have dialed an invalid number”.
- **Built-In Auto Attendant - Ring Back Tone (RBT) Message:** "The number you dialed is ringing.”
- **Built-In Auto Attendant - Busy Message:** "The extension you have dialed is busy".
- **Built-In Auto Attendant - No-Reply Message:** "The Extension you have dialed is not responding".
- **Built-In Auto Attendant - No-Dial Message:** "Sorry you have not dialed any Extension Number".
- **Built-In Auto Attendant - Conference Number**[347] **Message:** "Please dial the Conference Number".
- **Built-In Auto Attendant - Conference Password Message:** "Please dial the Conference Password".
- **Built-In Auto Attendant - Call Transfer Message:** "Transferring the call to the Operator". This message is played to the caller, when he does not dial any number and the call is transferred to the Operator.

Voice Message for Features

- **Alarms:** A pre-recorded voice message is played to the extension on which Alarm is set when the wake-up call is served. This feature is very useful in hotels where wake-up alarms are to be set for guests at the

347. Refer the description of the feature “Conference Dial-In”

oddest hours. With the Voice Message for alarms, guests can be greeted when they pick up the handset to answer their phone.
For example, "Good Morning. This is a wake up call. You may please call room service for any assistance. Thank you. Have a nice day".

You are recommended to record the message "Please press 0 to acknowledge the Alarm and Reminder" as a voice module for the Alarm/Reminder message, so that extension users can acknowledge Snooze calls. Refer the topics "Alarms" and "Reminder" to know more.

- **Security/Emergency Message:** A voice message is played to the external number and to the operator extension which answers the Emergency call.

- **Voice Help:** You can record and play voice message to provide quick help to extension users. The Voice Help message may contain important instructions, or frequently accessed feature codes, or important phone numbers, etc.

  For example, the Access Codes of the frequently used features can be recorded in the Voice Help message. Extension users may simply dial the Voice Help feature code and listen to the voice message.

  Help message must be recorded considering the maximum duration of the Voice Module (16 seconds), so that voice messages are not truncated.

- **Music-on-Hold:** Callers who are put on hold are usually played music from an internal/external source as they wait. You can play a voice message instead of music to the callers. The message may contain any promotional information about your company or services provided by your organization, etc.

  For example, "Welcome to Progressive Bearings. We are glad to announce that we are now an ISO 9001 company."

- **Message Waiting:** Whenever there is a new message in the mailbox of the extension user and if the VMS informs the SARVAM UCS about the new message, the SARVAM UCS changes the dial tone of the extension to a stuttered dial tone. The SARVAM UCS also offers the facility to playback a message instead of the stuttered dial tone to indicate the waiting message. An appropriate voice message can be played back to the extension user when he lifts the handset.

  For example, "You have a new message in your Mailbox. Please access your mailbox".

- **DND Notification:** A voice message can be played to extension users who try to call an extension that has set DND.

Voice Message for Dial Tone

Normally, when a caller goes OFF-Hook to dial an extension, he gets the dial tone for a fixed time interval (Dial Tone Timer). If the caller does not respond within this time, he gets an error tone. The caller however, may not know about the Timer. So, a message may be played to the caller.

For example, "Please dial the number immediately after the beep".

If you have enabled the Hotline feature, the Hotline Timer will override the Dial Tone timer, because of which the voice message may get truncated.

Voice Message for Ring Back Tone

When a caller dials an extension, if the station is free, the caller gets a Ring Back Tone. Instead of the ring Back Tone, the caller can be played a message like: "The station you have dialed is free. Please wait till the station responds".

Voice message for Busy Tone

When a caller dials an extension, if the extension is busy, the caller gets a busy tone. The SARVAM UCS can be made to play back message instead of the busy tone, like: "The station you have dialed is currently busy. Please dial after some time".

Voice message for Error Tone

When a caller performs a wrong operation or uses a feature without access, it is possible to play a voice message to the caller instead of the Error Tone, such as: "Please check the number you have dialed".

Voice message for Confirmation Tone

The SARVAM UCS confirms the successful usage of features with a confirmation tone which could be replaced with a voice message like: "The requested operation is performed successfully".

Voice Message for Toll Control/CoS Violation

When an extension dials out a number, the system checks the Toll Control assigned to that extension. If the dialed number does not match with the number strings programmed for the Call Privilege type set for the Toll Control assigned to the extension, the following message can be played: "Please check the number you have dialed. This facility is not available on your Telephone".

Similarly, a voice message can be played to the extension user who attempts to invoke a feature that is not allowed to his/her extension phone in its Class of Service (COS).

Voice Message for Trunk Auto Answer

Trunk Auto Answer enables calls landing on a trunk to be answered automatically by greeting the caller with a voice message before the call is actually handled. For example, the caller can be played the message: "Please Wait! Your call will be attended shortly".

SARVAM UCS supports 4 Trunk Auto Answer messages.

Direct Dialing-In Guidance Messages

You can use voice messages to inform callers that the dialed extension is busy or there is no response from the dialed extension or to inform the caller that the call is being transferred to a Trunk Landing Group.

Voice Messages for Station User Greetings

SARVAM UCS supports two greeting messages. These messages are played to the extension users when they lift the handset. These Messages may be festive greetings like 'Happy New Year", or an announcement like "Meeting in the Conference Room at 4 O'clock", or any other voice message to be played to the extension users when they lift the handset. Once the greeting message is in effect, on lifting the handset Station User Greeting1 and Greeting2 are played followed by dial tone.

Voice Guidance-Time Based Greetings

SARVAM UCS supports 3 time-based greetings: Morning, Afternoon, Evening greetings. Messages like "Good Morning", "Good Afternoon", "Good Evening" can be recorded.

These time based greetings are played to callers before the Auto Attendant -Greeting Message is played on a Built-In Auto Attendant enabled trunk. You can set the desired Start Time for Morning, Afternoon and Evening greetings, see “Greeting Message Time” under *System Parameters*.

OG Call - Called Party Busy

SARVAM UCS allows extension users to make outgoing calls that can be routed through different networks. When outgoing calls are made using the Mobile, SIP or ISDN networks, if the called party is busy, the network provides a busy signal along with a busy message or tone.

In certain cases, when SARVAM UCS receives only busy signal without any message or tone from the network, you can use a voice message (Outgoing Call-Busy message), that can be played to the callers.

Radio Extension-Dial Messages

You can use voice messages to guide Radio extension users callers at various stages of the call. For instance, prompting them to dial a number, or giving an alert when they dial a wrong or invalid number, or informing them when the desired destination extension is busy or when there is no response from the dialed extension, etc.

PIN Dialing

You can record and play a voice message instead of beeps to prompt the user to dial PIN while using PIN Dialing feature. See “PIN Dialing” for more details.

How to configure

To be able to play Voice Messages, you must first record them in voice modules. Once you have recorded the voice messages, you must assign the voice module to the appropriate Voice Message Application.

Pre-recorded voice messages are in-built in the SARVAM UCS. You may either use them when recording the voice modules or you may record messages of your choice.

Recording Voice Messages

You can record Voice Messages on voice modules from a DKP/SLT connected to the SARVAM UCS.

When you record voice messages, make sure that the audio files are recorded in *.wav* file format, have the attributes listed below:

- Chunk ID : "RIFF"
- ChunkSize : Filesize-8 (check file size using right click property)
- SubChunk1ID : "frnt"
- Channel :1 (mono)
- Samples per seconds per channels : 8000
- Wav Byte Rate : 8000
- Bit Per Sample: 8

Recording Voice Messages from a Phone

To record a Voice Message from a phone,

- Lift the handset.
- Enter SE mode from the DKP/SLT.

Record the Voice Message, dial:

- ***2502-Voice Module Number***
  Where,
  Voice Module is from 01 to 16.

- If using a DKP, press Enter key.
- You will get a beep.
- At the end of the beep, start recording the message, by speaking into your telephone instrument or if using an external music source (PC or Music System), play the voice message.
- Limit your message to 16 seconds.
- Go ON-Hook on completion of the message.

To check/verify the Voice Message you recorded, dial:

- ***2503-Voice Module***
  Where,
  Voice Module is from 01 to 16.
  In this case, dial the number of the Voice Module you just recorded.

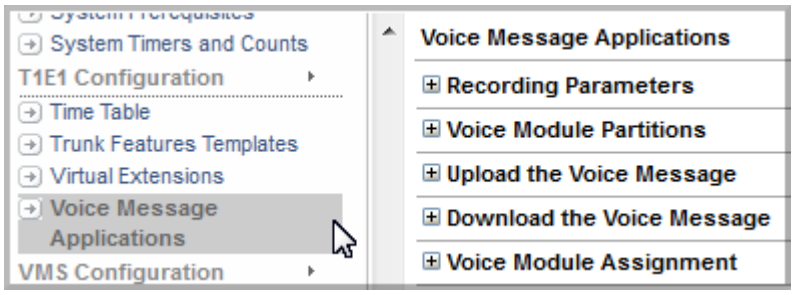
  If the audibility of the recorded message is not satisfactory, you may repeat this procedure again.

Voice Module 01 is reserved for Music-on-Hold by default. You are advised not to assign this module to any other Voice Message Application.

Configuring Voice Messages Application parameters using Jeeves

- Log in to Jeeves as System Engineer.

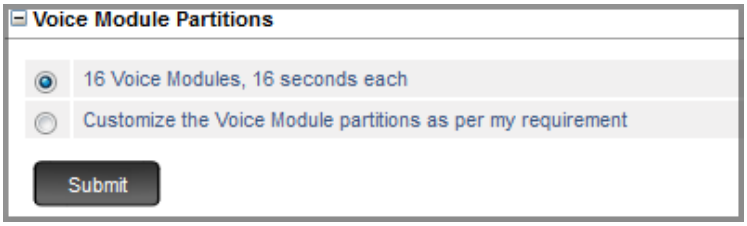
- Under **Configuration**, click **Voice Message Applications** to open the page.



- Click **Recording Parameters** to expand.

  - **Recording Source:** This parameter displays the default source of recording as the Telephone Instrument.

- Click **Voice Module Partitions** to expand.



  - By default, SARVAM UCS supports 16 Voice Modules of 16 seconds each. If you want to change the Voice Module size, select the **Customize the Voice Module partitions as per my requirement** option.

  - If you select **Customize the Voice Module partitions as per my requirement**, for each **Module Number** you can customize the **Module Size**.

  - *Total Voice Module Size available for the Voice Message Applications is 256 seconds—16 Modules of 16 seconds each or you can customize the Module Size as per your requirement — 16, 32, 48, 64, 80, 96, 112, 128, 144,160, 176, 192, 208, 224, 240.*

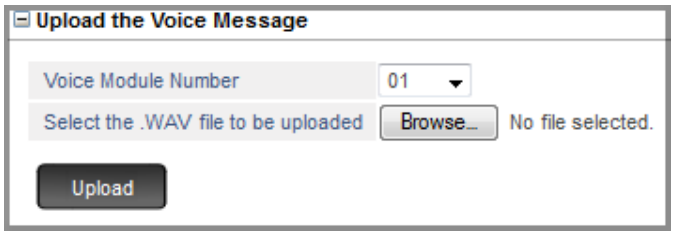
  - *Module Number 1 is reserved for the MOH. A default MOH file is already uploaded in the system. This default MOH will be over-written, when a new MOH file is uploaded. You cannot delete this default MOH file.*

  - *Only the first or the second Module Number can be assigned the Module Size as 240 seconds.*

  - *If you want to upload a customized MOH of 240 seconds, then make sure you also assign the Voice Module size as 240 seconds to that Module Number. In case, you upload a MOH of 240 seconds in a Module Number having the module size defined as 16 seconds, the System will play this uploaded MOH for only 16 seconds, the rest message will be truncated.*

- *When the modules sizes are changed, the system will automatically disable/enable the remaining module sizes.*
- *The messages from the disabled/resized Module Number will be deleted. In future, if you wish to use these Modules Numbers, you need to upload the messages in these modules again.*
- *The Voice Module Partitions will not be set to default values even when the system is set to default.*

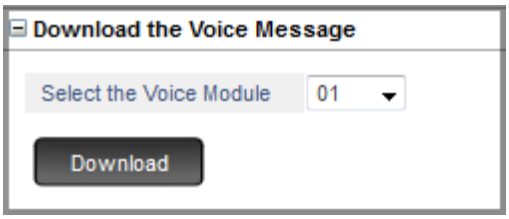
- Click **Upload the Voice Message** to expand.



- To upload recorded voice messages,
  - Select the desired **Voice Module Number**.
  - Click **Browse** to **Select the .WAV file to be uploaded**.
- Click **Upload**.

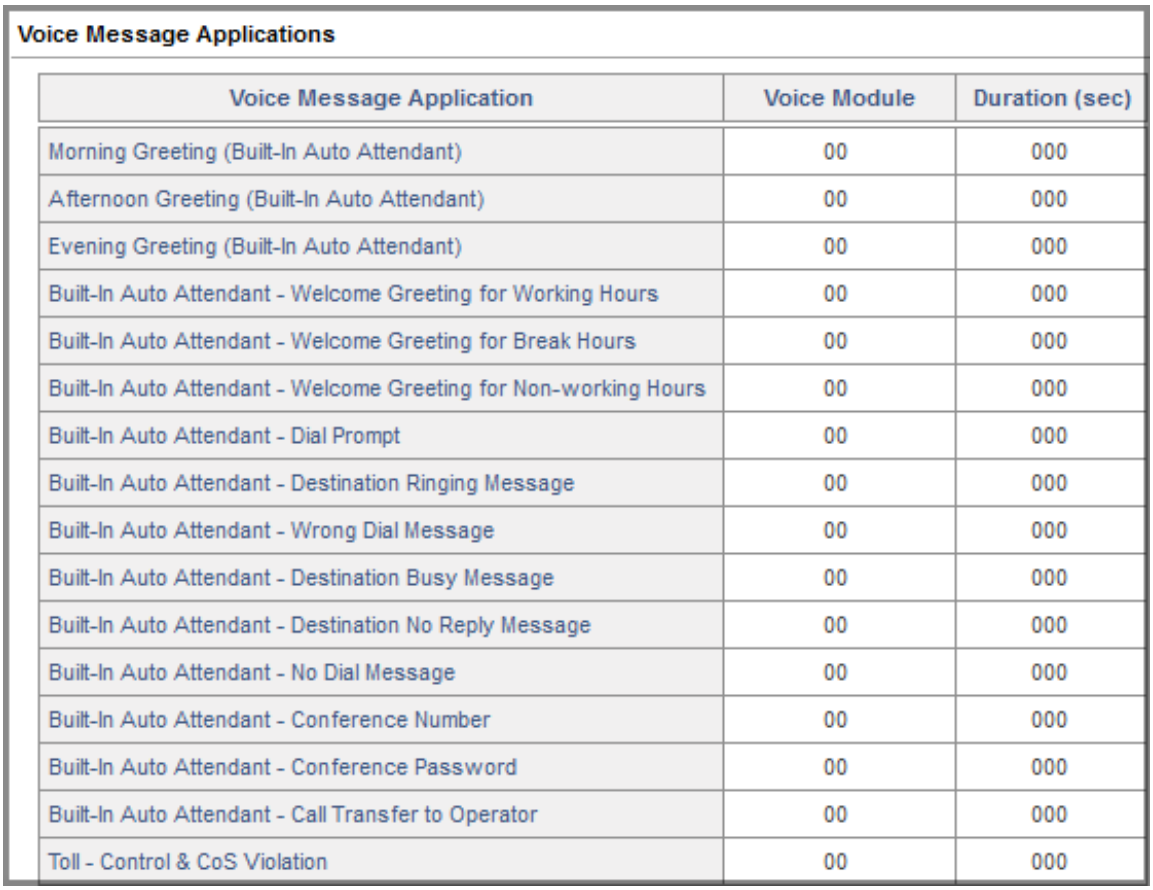
After you have uploaded a file of 240 sec duration in module 1 in SARVAM UCS SME/SMB, it is recommended to wait for 20 min atleast before uploading the next file. In the case of SARVAM UCS ENT, the wait period is of 4 min for the same.

- Click **Download the Voice Message** to expand.



- To download the voice messages,
  - Select the desired **Voice Module Number**.
- Click **Download**.

- Click **Voice Module Assignment** to expand.



Voice Message Applications

Voice Message Application	Voice Module	Duration (sec)
Morning Greeting (Built-In Auto Attendant)	00	000
Afternoon Greeting (Built-In Auto Attendant)	00	000
Evening Greeting (Built-In Auto Attendant)	00	000
Built-In Auto Attendant - Welcome Greeting for Working Hours	00	000
Built-In Auto Attendant - Welcome Greeting for Break Hours	00	000
Built-In Auto Attendant - Welcome Greeting for Non-working Hours	00	000
Built-In Auto Attendant - Dial Prompt	00	000
Built-In Auto Attendant - Destination Ringing Message	00	000
Built-In Auto Attendant - Wrong Dial Message	00	000
Built-In Auto Attendant - Destination Busy Message	00	000
Built-In Auto Attendant - Destination No Reply Message	00	000
Built-In Auto Attendant - No Dial Message	00	000
Built-In Auto Attendant - Conference Number	00	000
Built-In Auto Attendant - Conference Password	00	000
Built-In Auto Attendant - Call Transfer to Operator	00	000
Toll - Control & CoS Violation	00	000

- **Voice Message Application-Voice Module-Duration (sec):** Go to the desired Voice Message Application to assign the voice module number you recorded for this application. For example, you have recorded Voice Module 08 as DND Notification Message. Enter this number against "DND Notification".

  - By default, the following Voice Messages Applications have been assigned to Voice Modules 01 to 13 (see table below). If the default Message recorded in the module suits your purpose, simply assign the Voice Module number to the relevant Voice Application. The system will automatically set the duration of the Voice Message Application.

    If you want to change the duration of a recorded Voice Module, you can do so, by changing the duration manually for the particular recorded message.

  - For example, if you want to use Morning, Afternoon and Evening Greetings. You may simply assign the default Voice Modules 02, 03, and 04.

  - Refer the following table for Voice Message Applications assigned by default to the Voice Modules:

Default Voice Message Applications assigned to Voice Modules in the Enterprise Mode

Voice Module Number	Voice Message Application	Voice Message
01	Music-On-Hold	

Voice Module Number	Voice Message Application	Voice Message
02	Built-In Auto Attendant -Morning Greeting	Good Morning!
03	Built-In Auto Attendant -Afternoon Greeting	Good Afternoon!
04	Built-In Auto Attendant -Evening Greeting	Good Evening!
05	Built-In Auto Attendant - Welcome Greeting for Day Time (Working Hours)	Welcome!
06	Built-In Auto Attendant - Welcome Greeting for Night time (Non-working and Break hours)	Welcome! I am sorry, we are closed.
07	Built-In Auto Attendant - Dial prompt / Radio Extension-Dial Message	Please dial the desired number.
08	Built-In Auto Attendant - No Dial message	Sorry! You have not dialed any number.
09	Built-In Auto Attendant - Wrong Dial message	Sorry! The number is not valid.
10	Built-In Auto Attendant - Destination Busy message	The person you dialed is busy.
11	Built-In Auto Attendant - Destination Ringing message (Ring Back Tone)	The number you have dialed is ringing.
12	Built-In Auto Attendant - Destination No Reply message	The person you dialed is not responding.
13	Built-In Auto Attendant - Call Transfer to Operator message	Please hold, transferring your call to the Operator.

Default Voice Message Applications assigned to Voice Modules in the Hotel Mode

Voice Module Number	Voice Message Application	Voice Message
01	Music-On-Hold	
02	Built-In Auto Attendant -Morning Greeting	Good Morning!
03	Built-In Auto Attendant -Afternoon Greeting	Good Afternoon!
04	Built-In Auto Attendant -Evening Greeting	Good Evening!
05	Built-In Auto Attendant - Welcome Greeting for Day Time (Working Hours)	Welcome!

Voice Module Number	Voice Message Application	Voice Message
06	Built-In Auto Attendant - Welcome Greeting for Night time (Non-working and Break hours)	Welcome! I am sorry, we are closed.
07	Built-In Auto Attendant - Dial prompt	Please dial desired number.
08	Built-In Auto Attendant - No Dial message	Sorry! You have not dialed any number.
09	Built-In Auto Attendant - Wrong Dial message	Sorry! The number is not valid.
10	Built-In Auto Attendant - Destination Busy message	The person you dialed is busy.
11	Built-In Auto Attendant - Destination Ringing message (Ring Back Tone)	The number you have dialed is ringing.
12	Built-In Auto Attendant - Destination No Reply message	The person you dialed is not responding.
13	Built-In Auto Attendant - Call Transfer to Operator message	Please hold, transferring your call to the Operator.
14	Alarm	This is your wake up call.
15	Built-In Auto Attendant - Dial prompt	Please dial Room Number.
16	Blank	

- *It is possible to assign the same Voice Module to more than one Voice Message Application.*
- *If you have already recorded Voice Module 08, SARVAM UCS will automatically detect and display the duration of the Voice Module you recorded. So you need not define the duration of the Voice Module.*
- *You may define the duration of the Voice Module, only if you want the recorded voice message to be played for a specific duration. For example, the message you recorded in the voice module is 15 seconds long, but you want to play only the message contents of the first 8 seconds, you can define the duration of the message as 8 seconds.*
- *Voice Module 01 is reserved for Music-on-Hold by default. You are advised not to assign this module to any other Voice Message Application.*

- Click **Submit** to save your settings.
- You may log out of Jeeves.

Assigning Voice Modules to Voice Message Applications using a Telephone

- Enter SE mode.

To assign a voice message application to a voice module:

- Dial ***2505-Voice Message Application Number-Voice Module***
  Where,

Voice Message Application Number is from 01 to 47. See table below.
Voice Module is the voice module number from 1 to 16.
By default, Voice Module assigned is 00.

Voice Message Application Number	Meaning	Once/Continuous (Not Programmable)
01	Built-In Auto Attendant -Morning Greeting	Once
02	Built-In Auto Attendant - Afternoon Greeting	Once
03	Built-In Auto Attendant - Evening Greeting	Once
04	Built-In Auto Attendant - Welcome Greeting for Working Hours	Once
05	Built-In Auto Attendant - Welcome Greeting for Break Hours	Once
06	Built-In Auto Attendant - Welcome Greeting for Non-working Hours	Once
07	Built-In Auto Attendant - Dial Prompt	Once
08	Built-In Auto Attendant - Destination Ringing Message	Continuous
09	Built-In Auto Attendant - Wrong Dial Message	Once
10	Built-In Auto Attendant - Destination Busy Message	Once
11	Built-In Auto Attendant - Destination No Reply Message	Once
12	Built-In Auto Attendant - No Dial Message	Once
13	Built-In Auto Attendant - Conference Number	Once
14	Built-In Auto Attendant - Conference Password	Once
15	Built-In Auto Attendant - Call Transfer to Operator	Once
16	Toll - Control & COS Violation	Once
17	Trunk Auto Answer Greeting 1	Once
18	Trunk Auto Answer Greeting 2	Once
19	Trunk Auto Answer Greeting 3	Once
20	Trunk Auto Answer Greeting 4	Once
21	Trunk Auto Answer RBT Message 1	Continuous
22	Trunk Auto Answer RBT Message 2	Continuous
23	Trunk Auto Answer RBT Message 3	Continuous
24	Trunk Auto Answer RBT Message 4	Continuous
25	Trunk Auto Answer Bye Message 1	Once
26	Trunk Auto Answer Bye Message 2	Once
27	Trunk Auto Answer Bye Message 3	Once

Voice Message Application Number	Meaning	Once/Continuous (Not Programmable)
28	Trunk Auto Answer Bye Message 4	Once
29	DDI - Destination No Reply Message	Once
30	DDI - Destination Busy Message	Once
31	DDI - Call Transfer to TLG Message	Once
32	Alarm	Continuous
33	Station User Greeting 1	Once
34	Station User Greeting 2	Once
35	Message Wait Notification	Continuous
36	Message Wait - No message	Once
37	DND Notification	Once
38	Security/Emergency Message	Continuous
39	Dial Tone	Once
40	Ring Back Tone	Continuous
41	Busy Tone	Once
42	Error Tone	Once
43	Confirmation Tone	Once
44	Voice Help	Continuous
45	OG Call - Called Party is Busy	Once
46	Radio Extension-Dial Message	Once
47	PIN Dialing	Once

To use the default Voice Modules, refer the table "Default Voice Message Applications assigned to Voice Modules in the Enterprise Mode", and assign the desired Voice Module to a Voice Message Application.

To define voice message duration for voice modules[348]:

- Dial ***2504-Voice Module-Duration***
  Where,
  Voice Module is 01 to 16.
  Duration is from 00 to 96 seconds.
  By default, Duration of Voice Modules is 16 seconds.

To clear the voice module assigned to a voice message application:

- Dial ***2505-Voice Message Application Number-00***
  Where,
  VM Application Number is from 01 to 47.

- Exit SE mode.

348. *Voice module duration up to 96 seconds can be assigned to a Voice Module through telephone.To configure more than 96 seconds, refer to "Configuring Voice Messages Application parameters using Jeeves".*

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Assigning Station User Greeting Message to Extensions

This is done from SA mode only, using a DSS Key (if the phone is EON and a key is programmed for this function) or by dialing SA command strings.

To set/cancel a User Greeting message on an extension phone, from SA mode:

- Dial ***1072-008-Extension Number-Greeting Number-Code***
  Where,
  Greeting Number is
  0 for Greeting Message 1
  1 for Greeting Message 2

  Code is
  0 for Cancel
  1 for Set

VoLTE Configuration

What's this?

Voice over Long Term Evolution (VoLTE) is a standard that is used for high-definition voice calling service. VoLTE offers faster connection speed and advance technology for VoIP calls with better audio clarity and added security.

SARVAM UCS gives you the facility to access 4G LTE network depending upon the GSM module mounted on the Mobile card. This is made possible through an MBN file which is required for voice calling over LTE. Every mobile service provider has a unique MBN file that is already preloaded in the 4G module. The system also provides additional facility to manually upload MBN file of newly emerging mobile service provider in the market that offer VoLTE services.

The 4G module of SARVAM UCS offers backward compatibility, thereby enabling the devices to automatically get registered with 3G/2G network when not within the coverage area of 4G LTE network. To know more, refer *"The Mobile Card for ETERNITY LENX"*, *"The Mobile Card for ETERNITY MENX"*, *"The Mobile Card for ETERNITY GENX"* and *"The Mobile Card for ETERNITY PENX"*

How it works

Pre-requisites

To access 4G LTE network ensure you have performed the following:

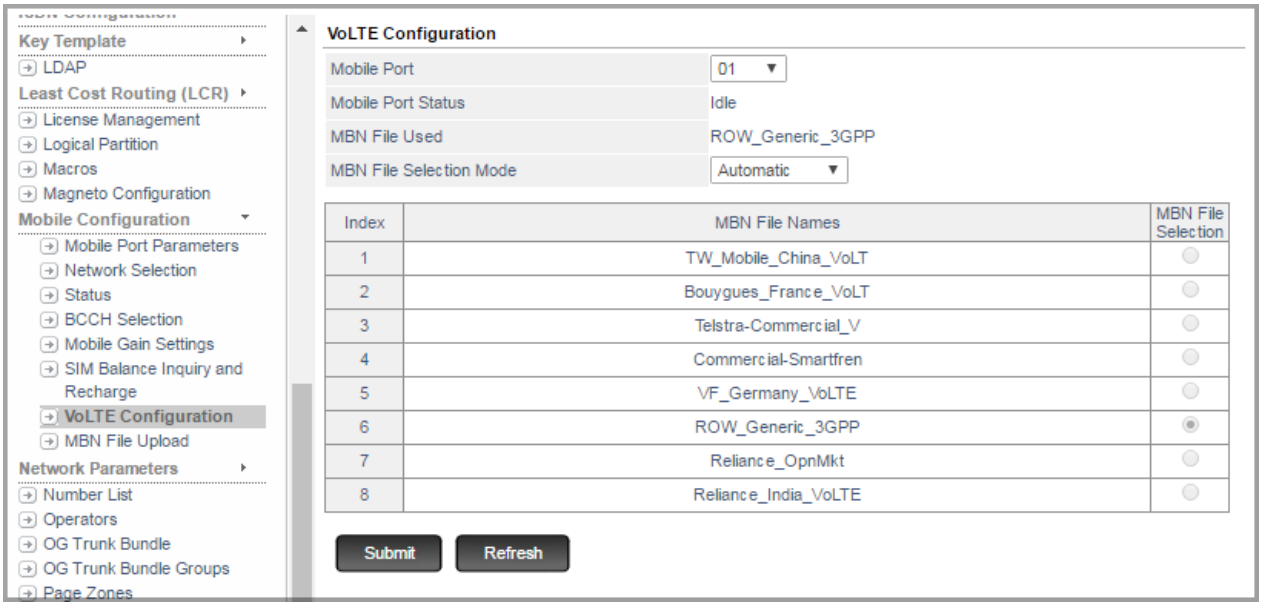
- Make sure the Mobile card with 4G module is installed in the system.
- Insert the Nano SIM cards of the desired service providers.
- Configure the VoLTE parameters. For details refer to *"Configuring VoLTE parameters"*.
- Uploading MBN files, For details refer to *"MBN File Upload"*
- Configure the Mobile Trunks. For details refer to *"Configuring Mobile Trunks"*.

How to configure

Configuring VoLTE parameters

- Log into Jeeves as System Engineer.

- Under **Mobile Configuration**, click **VoLTE Configuration** to open the page.



The VoLTE parameters will be visible only when 4G SIM is inserted.

Configure the following port parameters:

- **Mobile Port:** This is number of the Mobile port for which VoLTE Configuration is displayed. You can choose a different Mobile Port number from the drop down list. The page will display the VoLTE Configuration related parameters for the selected mobile port.

- **Mobile Port Status:** This is the status of the connection - showing Initialization with the Network, Registering with the Network, Idle or Busy state of the network. It also shows errors and alerts when SIM is absent, the wrong SIM PIN has been entered, SIM PUK is required.

- **MBN File Used:** The module stores varies MBN files. This displays the name of currently used MBN file for the particular Mobile Port.

- **MBN File Selection Mode:** Select the desired option - Automatic or Manual for selection of MBN file. By default, it is Automatic. The MBN files of different service providers are pre-loaded inside the module and the same are displayed in the table**.**

    - When you enable **Automatic** mode for selecting the MBN file, the system automatically selects the MBN file of the mobile service provider depending upon the SIM card inserted into the slot.

      The table is un-editable and displays the selected MBN file.

    - When you enable **Manual** mode for selecting the MBN file, then make sure you select the MBN file from the list of available MBN files.

      The table displays all the MBN files supported by the module. Click on the corresponding radio button to select the desired MBN file.

You can also upload MBN file of a newly emerged VoLTE supporting service provider in the market. To do so, refer *"MBN File Upload"*. After the file is uploaded it appears in the list.

- Click **Submit** to save your settings.

MBN File Selection Mode and Manual Selection table will be disabled when the 4G SIM is registered with 2G/3G network.

MBN File Upload

To upload the new MBN file,

- **Mobile Slot:** Select the desired slot number in which you have inserted the 4G card. The new MBN file will be uploaded in this card.
- **Select MBN File:** Click **Browse** button to reach the location where MBN file is stored.

Make sure the file to be uploaded is in .mbn format.

- Click **Upload**.

The maximum number of MBN files that can be manually uploaded is 20.
While uploading the MBN, make sure:
- *you do not remove or insert any SIM.*
- *the Mobile Port is in Ideal state or no SIM is inserted.*
- *you check all the ports for the updated MBN files.*

On successful upload a message is displayed in Jeeves.

To view uploaded file in the list of MBN files,

- Under **Mobile Configuration**, click **VoLTE Configuration**.

Walk-In Class of Service

What's this?

Every extension of SARVAM UCS is assigned a Class of Service and Toll Control defining its access to features and its calling permission.

With Walk-In Class of Service, extension users of SARVAM UCS can make calls or access features from any other extension of the system as per the Class of Service, Toll Control and other features / facilities assigned to their own extension.

This feature is useful to extension users who frequently move away from their desk, as it allows them the same level of feature access and calling permission as their own extension, from another extension.

Extension users can 'Walk-In' from any extension port: DKP, SLT, ISDN Terminal, E&M (with Station as Orientation Type), and SIP.

SARVAM UCS offers two types of Walk-In:

- **One call:** The extension user is automatically logged out from the extension into which the user has walked-in, after one call.

- **Multiple Calls:** The extension user can make as many calls as desired, and remains 'walked-in' until the user dials the feature code to 'Walk-Out', or until another extension user walks into the same extension.

To allow 'One call' or 'Multiple Calls' to an extension, you need to set the 'Walk-Out Mode' in the "Station Advanced Feature Template" of the extensions.

Walk-In Class of Service is a password-protected facility and the default User Password ***1111*** will not be accepted. To be able to walk into another extension, extension users must first change their User Password to another value.

How it works

With the help of this illustration, let us understand how Walk-in Class of Service works.

In this illustration, Extension user A has a DKP with the number 3001, with long distance calling facility (toll control: All calls). Extension user B has an SLT with the number 2001, without long distance calling (toll control: local calls).

Here,

- 3001 is the **Source Extension**, whose CoS, Toll Control and other features / facilities (assigned in the Station Basic Feature template) are used from another extension (2001) by performing Walk-In.

- 2001 is the **Destination Extension** on which Walk-In is performed.

Walking into another extension to make calls

- Now, extension user A is at B's desk and needs to make a long distance call. B's extension does not have long distance calling.

- Extension user A can 'Walk-In' into B's extension (2001) by dialing
  - the feature code for 'Walk-In Class of Service'.
  - A’s extension number, 3001.
  - A’s User Password (the default password 1111 will not be accepted, it must be changed first).

- On successful Walk-In, SARVAM UCS applies the Class of Service, Toll Control and other features / facilities as per the Station Basic Feature Template of the **Source Extension** 3001 on the **Destination Extension** 2001.

- Extension user A can make external and internal call from extension B.

- If Extension 2001 has 'One Call' selected as the 'Walk-Out Mode' for the extension, A will be 'Walked-Out' when the current call ends or if A goes ON-Hook at any time after walking into extension 2001.

- If Extension 2001 has 'Multiple Calls' selected as the 'Walk-Out Mode' for the extension, extension user A must manually walk out by dialing the feature code for 'Walk-Out'.

- If Extension user A does not 'Walk-Out', the system will perform a walk out for A only when another extension user walks into extension 2001.

At a time, only one extension user can walk in to another extension.

Calls made after walk-in will be charged to the Source Extension. Here, calls made by Extension user A from extension 2001 using Walk-In will be calculated and charged to extension 3001 only.

Call record details of calls made after walk-in will be recorded for the Source Extension. Here, call record details of calls made by Extension user A from extension 2001, using Walk-In, will be recorded in the Station Message Detail Record of extension 3001 only.

Walking into another extension to access a feature

- Extension user A with number 3001 has Call Forward in the Class of Service.

- Extension user B with number 2001 does not have Call Forward in the Class of Service.

- Extension user A is currently at B’s desk. A needs to forward calls of own extension to an external number. To do this,
  - A walks into B’s extension by dialing
    - the feature code for 'Walk-In Class of Service'.
    - A’s extension number (3001)
    - A’s User Password[349]

- On successful Walk-In, the system applies the Class of Service, Toll Control and other features / facilities as per the Station Basic Feature Template of the Source Extension 3001 on the Destination Extension 2001.

- From extension 2001, extension user A sets Call Forward for 3001, to an external number.

349. The default password 1111 will not be accepted, it must be changed first.

- Call Forward can be canceled from the Source Extension or from the Destination Extension. To cancel Call Forward, extension user A can go back to 3001 and cancel Call Forward from 3001, or can cancel Call Forward from 2001, if user A is still walked-in on 2001.

CAUTION! *The Destination Extension user can access their Class of Service or Toll Control only after the Source Extension user has walked out from their extension. For example, user B cannot set or cancel Call forward on extension 2001, until user A has walked out from 2001.*

Incoming calls on the Destination Extension (2001) will work according to the setting of the Destination Extension only, whereas outgoing calls on the Destination Extension will work according the settings of the Source Extension (3001).

The following set of features of the Destination Extension will remain unaffected:

- *Key map*
- *Language*
- *Priority*
- *Call Pick-Up group*

CAUTION! *There is a risk of fraudulent calls being made from your extension, if a third party comes to know the User Password of your extension. The cost of such fraudulent calls will have to be borne by the owner of SARVAM UCS.*

So, protect your system from unauthorized access and misuse by putting strong authentication mechanisms in place.

- *Keep Passwords strictly confidential.*
- *Change Passwords regularly.*
- *Choose Passwords that are complex and difficult to guess.*

How to configure

This feature is available to all extensions of SARVAM UCS. All you need to do is, select of the 'Walk-Out Mode' for the extensions in their “Station Advanced Feature Template”.

By default, ‘One call’ is selected as the ‘Walk -Out’ mode in the default Station Advanced Feature Template 01 assigned to all extensions.

If you want to allow different walk-out modes to different extensions, use different Templates, but make sure other features on these templates are also configured according to requirement.

For detailed instructions refer the topics “Customizing Station Advanced Feature Template using Jeeves” and “Customizing Station Advanced Feature Template using a Telephone”.

How to use

For EON and Extended IP Phone Users

To perform a Walk-In, on the Destination Extension,

- Press DSS Key assigned to Walk-In Class of Service (if programmed).
  OR

- Dial 111
- Select ‘Walk-in’ and press Enter key.
- You get the prompt 'Walk in from which Station?'
- Dial your extension number.
- You get the prompt 'Enter your User Password'.
- Dial your User Password (default password will not be accepted).
- You get the confirmation message 'Walked In Successfully' on the phone’s display and a confirmation tone.
- You can now make your call(s) or access a feature.

To perform a Walk-Out, on the Destination Extension or on the Source Extension,

- Press DSS Key assigned to Walk-In Class of Service.
  OR
- Dial 111.
- Select 'Walk-Out’ on your phone's display.
- You get the confirmation message 'Walked Out' on your phone's display and confirmation tone.

You need to perform a Walk-Out only if the Source Extension as Multiple Calls set as Walk-Out mode. If the Source Extension has 'One Call’ set as the Walk-Out mode, you will be walked out of the Destination Extension when you go ON-Hook after making a call or accessing a feature.

If the extension you are walking in has 'One Call’ as the Walk-Out mode, and you go ON-Hook before you make the call, you will be 'Walked Out'. You must Walk-In again.

For SLT Users

To perform a Walk-In:

- Lift the handset of the Destination Extension.
- Dial 111-1
- Dial your extension number.
- Dial your User Password.
- You get a confirmation tone.
- You can make your calls or access features now, during the confirmation tone or after you get the dial tone.

To perform a Walk-Out:

- Lift the handset of the Destination Extension or the Source Extension.
- Dial 111-0
- You get a confirmation tone.
- Replace the handset.

CHAPTER 16

Voice Mail Features

If you have CDMA Mobile Card installed in your system, it is recommended to avoid using Voice Mail features as the DTMF Detection might not work efficiently.

Accessing your Mailbox

You will be able to access your mailbox only if there is a free VMS channel. For more details, see “Configuring VMS General Parameters”.

To access your Mailbox from your own extension:

- Dial 3931.
- System prompts you with:

  “You have no new messages”, if there are no new messages in your mailbox.

  “You have n new messages”, if there are new messages in your mailbox (n = No. Of messages).
- Enter your Mailbox password.
- You will enter the “Mailbox Access” menu.

To access your Mailbox from another extension:

- Dial 3941/3942/ 3943.
- System prompts you with, “Welcome! Please dial the extension number or To dial by name press ‘6’,To leave a message press ‘7’, To access your Personal Mailbox press ‘8’, For further assistance press ‘9’, To disconnect the call press ‘#’.
- Dial 8. System prompts you to dial extension number.
- Dial your extension number to access your mailbox.
- System prompts you with:

  “You have no new messages”, if there are no new messages in your mailbox.

  “You have n new messages”, if there are new messages in your mailbox (n = No. of messages).

- Enter your Mailbox password.
- You will enter the “Mailbox Access” menu.

To access your Mailbox through an external number using DISA Login:

- Log into the SARVAM UCS using DISA.
- If CLI Based Authentication is enabled, the system prompts you to dial desired extension number.
- If PIN Authentication is enabled, the system prompts you to dial your Extension Number and Password.
- Dial 3931 to access your Mailbox.
- System prompts you with:

  “You have no new messages”, if there are no new messages in your mailbox.

  “You have n new messages”, if there are new messages in your mailbox (n = No. of messages).
- Enter your Mailbox password.
- You will enter the “Mailbox Access” menu.

To access your Mailbox using VMS Auto Attendant:

- Call the trunk on which Voice Mail Auto Attendant is enabled.
- The incoming call on the trunk is answered by the VMS Auto Attendant.
- The VMS greets the caller with the Greeting message followed by the Welcome Message: “Welcome! Please dial the extension number or To dial by name press ‘6’,To leave a message press ‘7’, To access your Personal Mailbox press ‘8’, For further assistance press ‘9’, To disconnect the call press ‘#’.
- Dial 8. System prompts you to dial extension number.
- Dial your extension number to access your mailbox.
- System prompts you with:

  “You have no new messages”, if there are no new messages in your mailbox.

  “You have n new messages”, if there are new messages in your mailbox (n = No. of messages).
- Enter your Mailbox password.
- You will enter the “Mailbox Access” menu.

Alarms and Reminders

What’s this?

The Voice Mail System of SARVAM UCS offers voice-guided Alarms and Reminders, which can be set by the Operator as well as extension users.

How it works

Voice-guided Alarm and Reminder requests are served as per the date and time set by extension users. The different ways in which Alarm or Reminder requests will be served are described in the following:

- **Alarm with Snooze Off (Once Only)**
    - The VMS plays system greeting for the current time zone and the Extension Name followed by the message "This is your Wake up Call. Music of 5 seconds."
- **Alarm with Snooze On (Once Only)**
    - The VMS plays system greeting as per time zone and the Extension Name followed by the message "This is your Wake up Call. For Acknowledge, Please Press 0. Music of 5 seconds."
    - When user press '0', VMS prompts: "Your Alarm is Acknowledged."
- **Daily Alarm with Snooze Off**
    - The VMS plays system greeting as per the time zone and the Extension Name followed by the message "This is your Daily Wake up Call. Music of 5 seconds."
- **Daily Alarm with Snooze On**
    - VMS plays system greeting as per time zone and the Extension Name followed by the message "This is your Daily Wake up Call. For Acknowledge, Please Press 0. Music of 5 seconds."
    - When user press '0', VMS prompts: "Your Alarm is Acknowledged."
- **Reminder with Snooze Off**
    - VMS plays system greeting as per time zone and the Extension Name followed by the message "This is your Reminder call. Music of 5 seconds."
- **Reminder with Snooze On**
    - VMS plays system greeting as per time zone and the Extension Name followed by the message "This is your Reminder Call. For Acknowledge, Please Press 0. Music of 5 seconds."
    - When user press '0', VMS prompts: "Your Reminder is Acknowledged."

How to configure

The VMS allows you to enable/disable the **Alarm Verification** for alarms and reminders, allowing extension users who want to use alarms and reminders to confirm

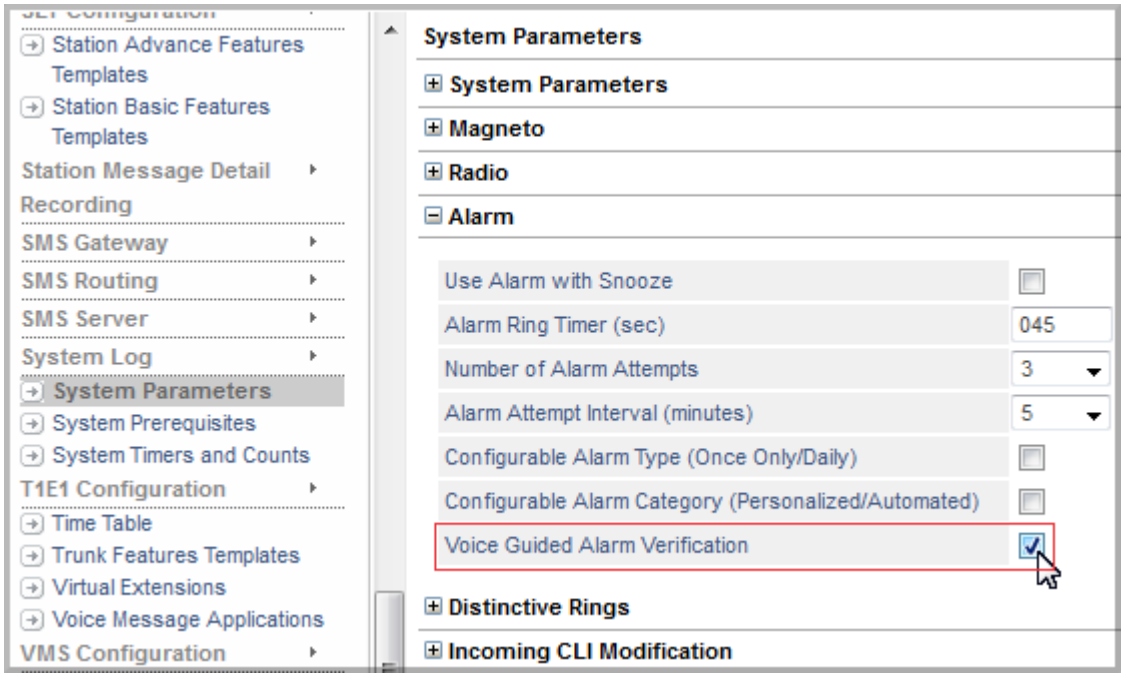
- Time set for an alarm
- Date and time set as a reminder.

When Alarm Verification is disabled, the VMS will not confirm the alarm and reminder set by the extension user.

If you want the VMS prompts to be played when the extension users answer an Alarm Call/Reminder Call and acknowledge it, make sure you have selected *Voice Mail* as *Alarm Notification Type*.

To configure Alarm Verification,

- Log in as System Engineer.
- Under **Configuration**, click **System Parameters**.
- On the System Parameters page, click **Alarm** to expand.



- Select the **Voice Guided Alarm Verification** check box to enable extension users to confirm the Time they have set for an alarm or the Date and Time they have set for a reminder. Default: Enabled.
- Click **Submit** to save changes.

To configure Alarm Notification Type,

- Under **Configuration**, click **Station Advanced Feature Template** to open the page.
- Scroll to **Alarm Notiifcation Type** and select **Voice Mail**.
- Click **Submit** to save changes.

- Assign this **Station Advanced Feature Template** to the desired extensions.
- Log out of Jeeves.

How to use

Alarm set by Extension Users

- Pick up the handset of your telephone and dial **163** → VMS prompts: "Enter the time, HH MM in twenty four hour format[350]. To cancel all alarms, press # (pound/hash)."
    - If no time is entered, VMS prompts: "You have not entered any input"
    - If invalid time is entered, VMS prompts: "You have entered invalid input."
- To set alarm, dial valid time → VMS prompts: "To set once, Press 1, To set Daily Press 2."

Once Only

- Dial 1 → VMS responds with: "You have set Wake up Alarm at …." (WakeupVeri.wav) followed by the prompt: To Confirm[351], Press 1, To Re-enter, Press 2."
- Dial 1 to confirm the time set for alarm → VMS responds with: "Your Wake up Alarm is set." followed by the prompt: "Thanks for using this Service."
    - If alarm is not set, the VMS responds with: "Sorry! Your Wake Up Alarm cannot be set. Please call Operator for further assistance." VMS further responds with: "Thanks for using this Service."

Daily Alarm

- Dial 2 → VMS responds with: "You have set Daily Wake up Alarm at …." followed by the prompt: "To Confirm, Press 1, To Re-enter, Press 2."
- Dial 1 to confirm the time set for alarm → VMS responds with: "Your Daily Wake up Alarm is set." followed by the prompt: "Thanks for using this Service."
    - If no alarm is set, the VMS responds with: "Sorry! Your Wake Up Alarm cannot be set. Please call Operator for further assistance." The VMS further responds with: "Thanks for using this Service."
- Dial # (pound/hash) to cancel all alarms → VMS responds with: "Your all Wake up Alarm are canceled." followed by the prompt: "Thanks for using this Service."
    - If no alarms are set, the VMS responds with: "Sorry! There is no Alarm to cancel." followed by the prompt: "Thanks for using this Service."

Alarm set by Operator

- Enter System Administrator Mode, dialing 1072.
- Dial 034 → the VMS prompts: "Enter the Extension number for which you have to set or cancel Wake Up Alarm."

350. The Date and time format depends on the Region/Country selected for the system.
351. This option will not be played if Alarm Verification is disabled in the System Parameters.

- Dial 1 to select the extension user for which the Alarm is to be set. VMS responds with: "Enter the time, HH MM in twenty-four hour format. To cancel all alarms, press '#' (pound/hash)'."
    - If no time is entered, the VMS prompts: "You have not entered any input"
    - If invalid time is entered, the VMS prompts: "You have entered invalid input."
- To set alarm, dial valid time → VMS prompts: "To set once, Press '1', To set Daily Press '2'."

Once Only

- Dial 1 → the VMS responds with: "To set it as Personal, Press 1. To set it as Automated, Press 2."
    - Dial 1 → VMS responds with: "You have set Personal Wake up alarm at…." followed by the prompt: "To Confirm, Press 1, To Re-enter, Press 2."
    - Dial 1 → VMS responds with: "Your Personal Wake up Alarm is set." followed by the prompt: "Thanks for using this Service."
    - If alarm is not set, the VMS responds with: "Sorry! Your Wakeup Alarm cannot be set. Please call Operator for further assistance." VMS further responds with: "Thanks for using this Service."

OR

- Dial 2 → the VMS responds with: "You have set Automated Wake up alarm at…." followed by the prompt: "To Confirm, Press 1, To Re-enter, Press 2."
- Dial 1 → the VMS responds with: "Your Automated Wake up Alarm is set." followed by the prompt: "Thanks for using this Service."
    - If alarm is not set, VMS responds with: "Sorry! Your Wakeup Alarm cannot be set. Please call Operator for further assistance." VMS further responds with: "Thanks for using this Service."

Daily Alarm

- Dial 2 → VMS responds with: "To set it as Personal, Press 1. To set it as Automated, Press 2."
    - Dial 1 → VMS responds with: "You have set Daily Personal Wake up alarm at…." followed by the prompt: "To Confirm, Press 1, To Re-enter, Press 2."
    - Dial 1 → VMS responds with: "Your Daily Personal Wake up Alarm is set." followed by the prompt: "Thanks for using this Service."
        - If the alarm is not set, the VMS responds with: "Sorry! Your Wakeup Alarm cannot be set. Please call Operator for further assistance." VMS further responds with: "Thanks for using this Service."

OR

- Dial 2 → the VMS responds with: "You have set Daily Automated Wake up alarm at…." followed by the prompt: "To Confirm, Press 1, To Re-enter, Press 2."
- Dial 1 → VMS responds with: "Your Daily Automated Wake up Alarm is set." followed by the prompt: "Thanks for using this Service."

- If alarm is not set, VMS responds with: "Sorry! Your Wakeup Alarm cannot be set. Please call Operator for further assistance." VMS further responds with: "Thanks for using this Service."

Reminders set by Extension Users

- Pick up handset of your telephone and dial **164** → VMS prompts: "Enter the Date in DD MM YYYY format[352]. To Cancel all Reminders, Press '#' (pound/hash)'. For example, To enter Date 17th March 2008, Dial One Seven Zero Three Two Zero Zero Eight."
  - If no date is entered then VMS prompts: "You have not entered any input"
  - If invalid date is entered then VMS prompts: "You have entered invalid input."
- Dial valid Date → the VMS prompts: "Enter the time, HH MM in twenty four hour format."
  - If no time is entered, the VMS prompts: "You have not entered any input"
  - If invalid time is entered, the VMS prompts: "You have entered invalid input."
- Dial valid time → VMS prompts: "You have set Reminder for..." followed by the prompt: "To Confirm, Press 1, To Re-enter, Press 2."
- Dial 1 to confirm the date and time set for Reminder → the VMS responds with: "Your Reminder is set." followed by the prompt: "Thanks for using this Service."
  - If Reminder is not set, the VMS responds with: "Sorry! Your Reminder cannot be set. Please call Operator for further assistance." VMS further responds with: "Thanks for using this Service."
- Dial # (pound/hash) to cancel Reminder → the VMS responds with: "Your Reminder is canceled." followed by the prompt: "Thanks for using this Service."
  - If no reminder is set, the VMS responds with: "Sorry! There is no Reminder to cancel." followed by the prompt: "Thanks for using this Service."

Reminders set by Operator

- Enter System Administrator Mode, dialing 1072.
- Dial 1072 to enter SA mode followed by 035 → VMS prompts: "Enter the Extension number for which you have to set or cancel Reminder."
- Dial 1 to select the station user for which the Reminder is to be set. VMS responds with: "Enter the Date in DD MM YYYY format. To Cancel all Reminders, Press '# (pound/hash)'. For example, To enter Date 17th March 2008, Dial One Seven Zero Three Two Zero Zero Eight."
  - If no date is entered, the VMS prompts: "You have not entered any input"
  - If invalid date is entered then VMS prompts: "You have entered invalid input."
- Dial valid date → VMS prompts: "Enter the time, HH MM in twenty-four hour format."

352. The date format in the prompt will be MM DD YYYY, if you selected USA as the Region/Country for your system.

- If no time is entered then VMS prompts: "You have not entered any input"
- If invalid time is entered then VMS prompts: "You have entered invalid input."

- Dial valid time → VMS prompts: "To set it as Personal, Press 1. To set it as Automated, Press 2."
  - Dial 1 → VMS responds with: "You have set Personal Reminder for…." followed by the prompt: "To Confirm[353], Press 1, To Re-enter, Press 2."
  - Dial 1 → VMS responds with: "Your Personal Reminder is set." followed by the prompt: Thanks for using this Service."
  - If Reminder is not set, the VMS responds with: "Sorry! Your Reminder cannot be set. Please call Operator for further assistance." VMS further responds with: "Thanks for using this Service."

  OR
  - Dial 2 → VMS responds with: "You have set Automated Reminder for…." followed by the prompt: "To Confirm, Press 1, To Re-enter, Press 2."
  - Dial 1 → VMS responds with: "Your Automated Reminder is set." followed by the prompt: "Thanks for using this Service."
  - If Reminder is not set, VMS responds with: "Sorry! Your Reminder cannot be set. Please call Operator for further assistance." VMS further responds with: "Thanks for using this Service."
- Dial # to cancel Reminder → VMS responds with: "Your Reminder is canceled.” followed by the prompt: "Thanks for using this Service."
  - If no reminder is set, VMS responds with: "Sorry! There is no Reminder to cancel." followed by the prompt: "Thanks for using this Service."

353. This option will not be played, if Alarm Verification is disabled in the System parameters.

VMS DISA Login

What's this?

This feature allows the remote users to log into the DISA mode using the trunks on which the Voice Mail Auto Attendant is enabled. The remote user can access and use the system's features and facilities using the DISA enabled trunks.

The VMS supports DISA-PIN Authentication-Multiple Calls only. For detailed information on the types of DISA Variants, see "Direct Inward System Access (DISA)".

Using VMS DISA login, remote users can:

- call any extension.
- make external calls.
- use features and facilities of the system.
- configure features and facilities of the system and administer the system.

All these can be done as if being done from a local extension of the SARVAM UCS.

How it works

For this feature to work,

- a VMS module must be installed in the system.
- you must enable **DISA-PIN Authentication-Multiple Calls** on the desired trunk: CO, Mobile, SIP, T1E1PRI, BRI.
- you must select **Voice Mail Auto Attendant** as the **Auto Attendant** option on the desired trunks: CO, Mobile, SIP, T1E1PRI, BRI.
- you must enable DISA in the "Class of Service (COS)" of the extension the caller is allowed to log into (using PIN Authentication).
- you must change the default **User Password** (1111) of the extension the caller is allowed to log into.

This is how VMS DISA Login works:

- A call lands on a DISA enabled Trunk.
- The incoming call on the trunk is answered by the VMS Auto Attendant. By default, the VMS greets the caller with the Greeting message followed by the Welcome Message: "Welcome! Please dial the extension number or To dial by name press '6',To leave a message press '7', To access your Personal Mailbox press '8', For further assistance press '9', To disconnect the call press '#'.
- The caller must dial the DISA Login Code, 1079. The VMS checks if DISA-PIN Authentication-Multiple Calls is enabled on the trunk for the current time zone, that is, working hours, break-hours and non-working hours.

- The VMS finds DISA is enabled on the trunk and prompts the caller: “Please enter the Extension Number.” and starts the First Digit Wait Timer (programmable; default: 25 seconds).
- The caller must dial the DISA Extension Number before the expiry of this timer.
- The VMS checks the CoS of the dialed Extension Number.
- DISA is enabled in the CoS of the dialed Extension Number, the VMS prompts: “Please enter your password.” and waits to receive digits till the expiry of the First Digit Wait Timer.
- The Password must be dialed by the caller before the expiry of this timer.
- The VMS checks if the Password is valid.
- The Password is valid and the DISA Login is successful. The caller is logged into the dialed Extension Number.
- The VMS hands over the DISA call to the SARVAM UCS.
- When the caller goes Off-hook by dialing the Off-hook code #1, the system plays the internal dial tone and waits for the caller to dial digits.
- If the caller dials an external number using a CO trunk, the system starts the *DISA Inactivity Timer* (configurable; default: 2 minutes)[354].
- The system waits for the caller to dial digits within the DISA Inactivity Timer.
- The system reloads this timer each time it receives digits from the caller. If the caller fails to dial any digit within this timer, the system plays beeps for the duration of the *DISA Warning Beeps Timer* (fixed; 15 seconds). If no digit is received at the end of the Warning Beeps, the system terminates the DISA session. If digits are received before the end of the Warning Beeps, the system reloads the DISA Inactivity Timer.
- The caller can make as many trunk calls and internal calls as the caller wants.
- The caller can terminate the DISA login session either by disconnecting from the remote end or by dialing the Termination Code *#9*.

The VMS plays the default Greeting Message, Welcome Greeting and DISA prompts to the callers. You can customize them as per your requirement, if required.

How to configure

- For instructions to enable DISA-PIN Authentication-Multiple Calls and Voice Mail Auto Attendant on the desired trunks, see the topic “Trunk Feature Template” in *Configuring Trunks*.
- For instructions to enable DISA in the CoS of the extensions which you want to allow callers to access using DISA, see “Class of Service (COS)”.
- To change the default User Password (1111) of the extensions which you want to allow callers to access using DISA, see “User Password” and “System Security”.

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354. DISA Inactivity Timer is not applicable for T1E1PRI lines, BRI lines, SIP and Mobile trunks.

- To set the DISA Timers as per your requirement, see “System Timers and Counts”.
- To customize the DISA prompts as per your requirement, see “Recording Voice Messages”.
- To upload the customized Voice Prompts, see “Prompts Management”.

Join Conference Dial-In using VMS

What's this?

This feature allows users to Join a Dial-in Conference using the trunks on which the Voice Mail Auto Attendant is enabled. After Joining the Conference users can also Temporary Leave, Rejoin or Permanently Leave the Conference.

How it works

For this feature to work,

- a VMS module must be installed in the system.
- you must select **Voice Mail Auto Attendant** as the **Auto Attendant** option on the desired trunks: CO, Mobile, SIP, T1E1PRI, BRI.
- you must inform the caller the Dial-In Conference Number and Password along with the time of Conference.

This is how Join Conference Dial-In using VMS works:

- A call lands on a VMS enabled Trunk.
- The incoming call on the trunk is answered by the VMS Auto Attendant. By default, the VMS greets the caller with the Greeting message followed by the Welcome Message: "Welcome! Please dial the extension number or To dial by name press '6',To leave a message press '7', To access your Personal Mailbox press '8', For further assistance press '9', To disconnect the call press '#'.
- While this prompt is being played, the caller must dial the Dial-In Conference Code, *19. The VMS prompts the caller: "Please enter your number and password. Caller must first dial 2 (the Join Dial-In Conference code) and then dial the Dial-In Conference Number and Password.
- System checks if the Conference Number and password is valid or not.
- The Number and Password is valid the caller will be able to join the conference.
- The Caller can temporarily leave from the Dial-In conference, to do so, while in speech, dial Flash-191.
- The Caller can rejoin the Dial-In conference, to do so, go Off-hook (by dialing the Off-hook code #1) and then dial 191.
- The Caller can permanently leave the Dial-In conference, to do so, go Off-hook (by dialing the Off-hook code #1).

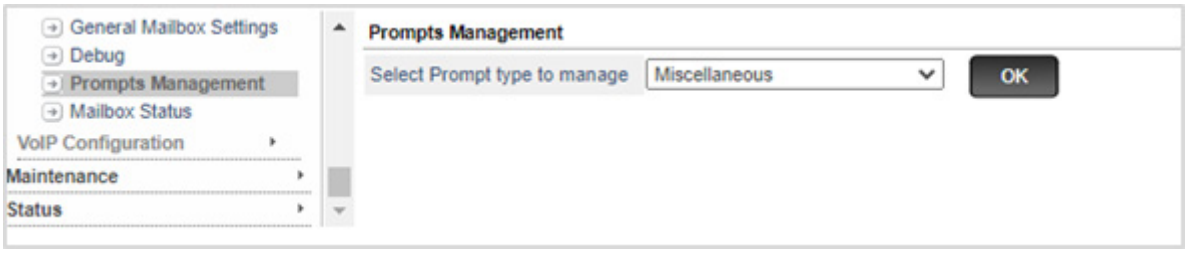
How to configure

- For instructions to enable Voice Mail Auto Attendant on the desired trunks, see the topic "Trunk Feature Template" in *Configuring Trunks*.
- To customize the Conference prompts as per your requirement, see "Recording Voice Messages".
- To upload the customized Voice Prompts, see "Prompts Management". If you are upgrading the firmware with Firmware later that V1R6.7, you need to manually upload the prompt. Refer to "Dial-In Prompt Upload".

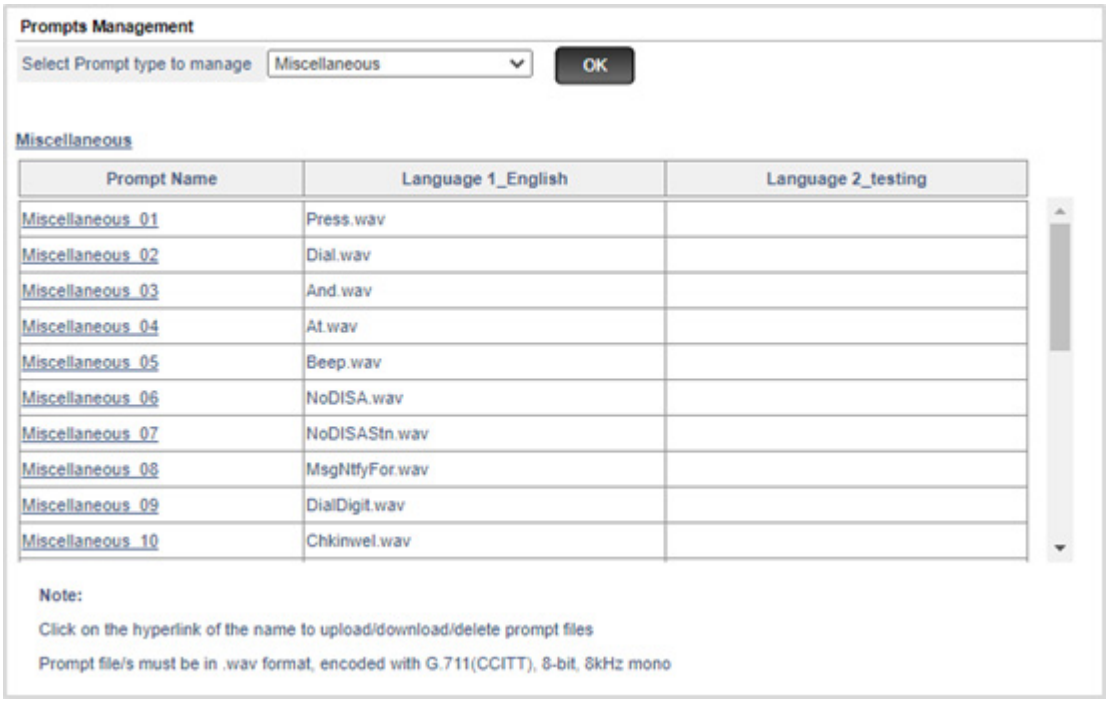
Dial-In Prompt Upload

After upgrading the system, to upload the VMS Dial-In Prompt, follow the steps mentioned below:

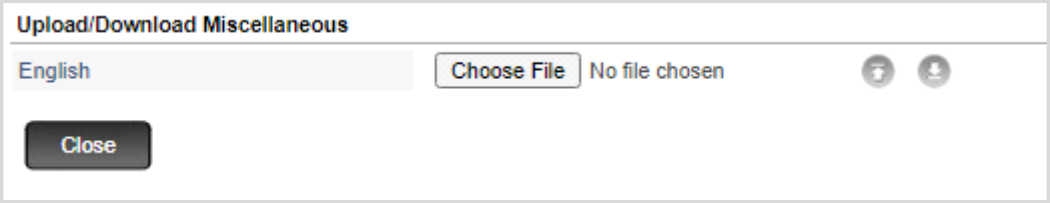
- Under **Configuration**, click **VMS Configuration**.
- Click **Prompts Management.**



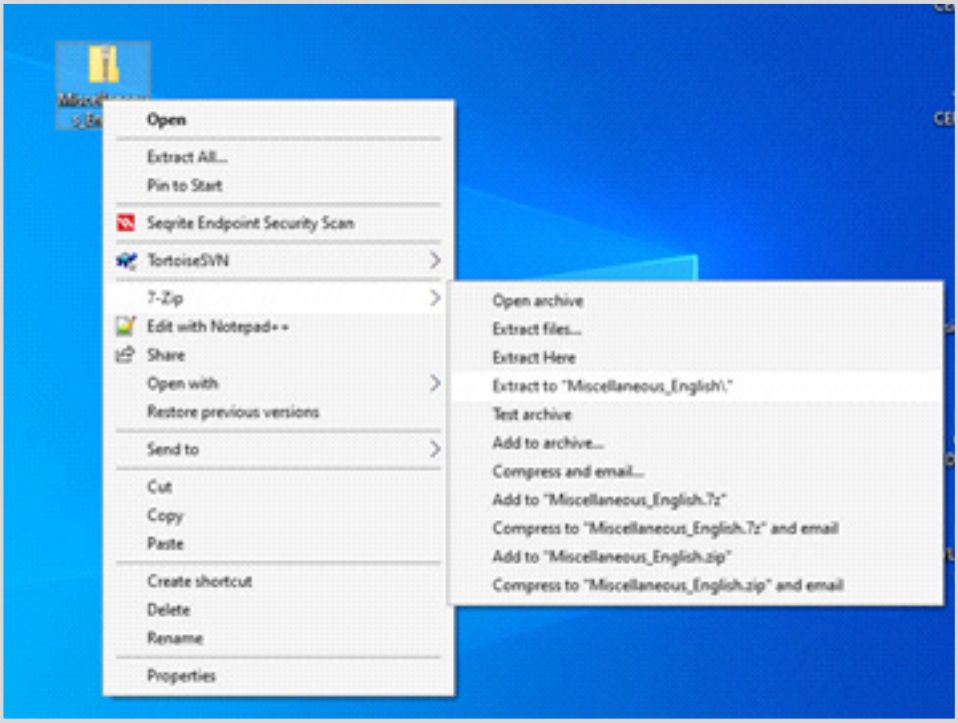
- In **Select Prompt type to manage** select **Miscellaneous**.



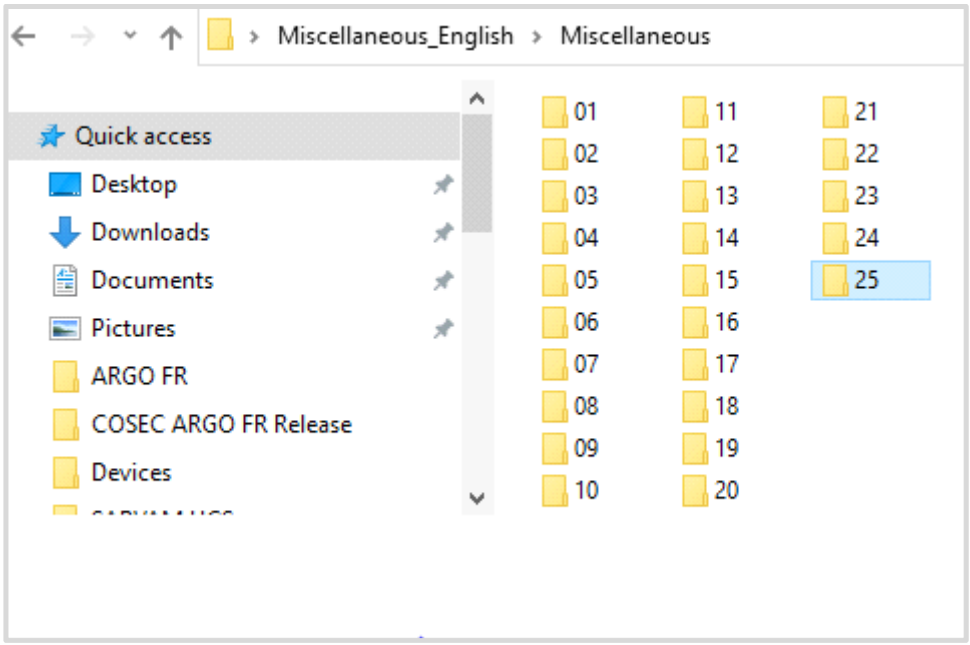
- Click **OK**.
- Click on **Miscellaneous**
- To download all the prompts of, click .



- The Miscellaneous_English zip folder is downloaded. Extract the folder files.



- Now click the extracted folder and create a new folder 25 in it.



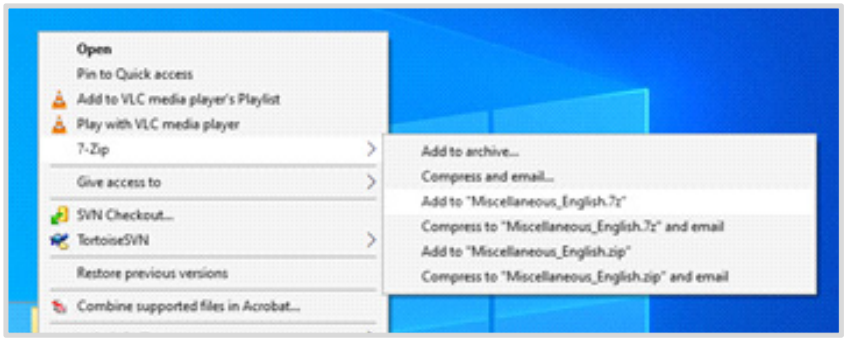
- Click to open Folder number 01, it has two files. Make sure you delete one file which you do not require. This folder must not have two files.

- Copy the new prompt in folder 25 from the USB package (path:Voicemail\database\prompts\language_1\Miscellaneous\25)

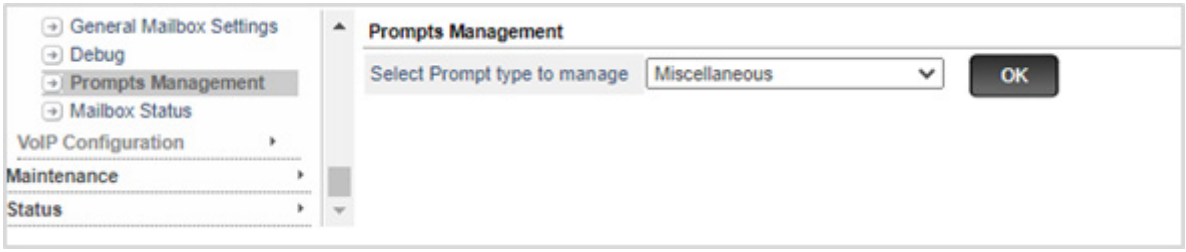
Contact the Dealer/Distributor or Technical Support for the USB package.

- Go back to the extracted Miscellaneous folder and open the folder 25. Paste the prompt here.

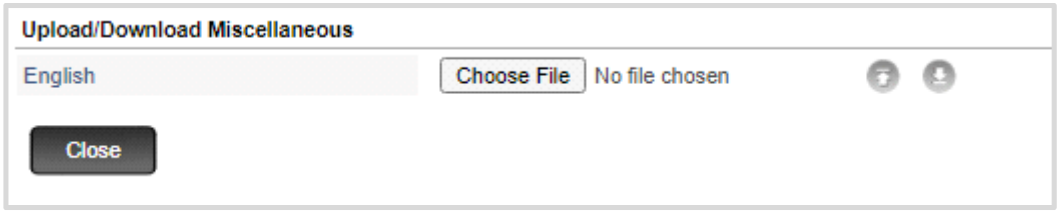
- Zip the Miscellaneous folder.



- Now, click **Prompts Management** again.



- In **Select Prompt type to manage** select **Miscellaneous**.
- Click **OK**.
- Click on **Miscellaneous**.



- Click the **Choose File** button to reach the location on the local disk where the Miscellaneous zip folder is stored in your PC.
- Click to upload.

Sending Messages

What's this?

The VMS enables extension users to send messages to other extensions that have a mailbox. An extension user can send a message to as many as 10 destinations at a time. The extension user can send the message either to a specific mailbox or to a Distribution List.

VMS also gives facility to the Sender of the message to request a read receipt of the message sent. When the Recipient has read the message, the VMS generates a file containing the first 5 seconds of the message that was sent and delivers it to the Sender's mailbox in the form of a new message with the Date and Time stamp and the prompt: "This message was read by <Extension Name> <5 seconds of message sent>". If the Sender does not request 'read receipt', no such message is delivered to the Sender.

How to use

- Call the VMS by dialing **3931**,
- The VMS takes you to your mailbox.
- VMS responds with: "You have <n> new messages" followed by the prompt: "Enter your mailbox password".
- Enter your mailbox password → VMS prompts: "You have n new/no new messages."
- VMS prompts: "To listen to new messages press '1', to listen to old messages press '2', to send a message press '3', to change your mailbox settings press '4'."
- Dial 3 → VMS prompts: "Enter the destination/s and dial hash to end."
- Dial valid extension numbers/distribution list number → VMS prompts: "Record your message after the beep and press any digit to end".
- Speak to record the message and press any digit to end.

The system provides an option to the caller to verify the message before storing the same. Enable the Message Verification check box and configure the digits for the Message Leave Options. The message will be played as per the digits selected for the options. For detailed instructions, see "Message Send/Forward Settings"

- VMS prompts "To request read receipt press '1', to ignore read receipt press '2'."
- If Message Verification flag is disabled then VMS will not playback the recorded message.
- Dial 1 to request read receipt of your message, else dial 2.
- VMS responds: "Message sent as normal."

Extension users must be careful in dialing destination numbers. If invalid destination is entered then the VMS will clear all the entries and will ask the mailbox owner to re-enter all the destinations again.

Once a valid destination number is entered and no more extensions are selected, the VMS understands it to be the end of list and sends the message.

Redirecting Messages

What's this?

The VMS offers extension users to re-direct the messages in their mailbox to another mailbox. The feature can be used by employees who are out of office or unable to access their mailbox. Using Redirect Messages, they can ensure that important messages are attended to by their colleagues in their absence.

To be able to use this feature, make sure a digit has been assigned to Mailbox Management. For instructions, see "Mailbox Access".

How to use

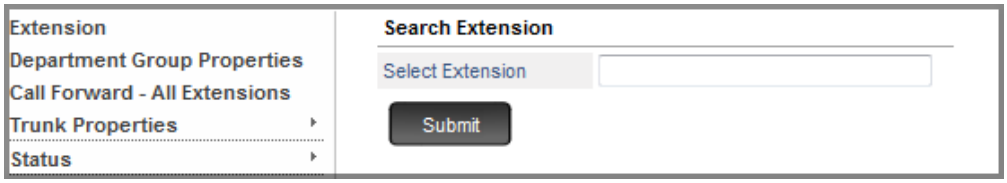
- Call the VMS by dialing **3931**,
- The VMS takes you to your mailbox.
- VMS responds with: "You have <n> new messages" followed by the prompt: "Enter your mailbox password".
- Enter your mailbox password → VMS prompts: "You have n new/no new messages."
- VMS prompts: "To listen to new messages press '1', to listen to old messages press '2', to send a message press '3', to change your mailbox settings press '4'."
- Dial 4 → The VMS responds with: "For Mailbox Name, press '1', For message redirection, press '2', To delete all old messages of your mailbox, press '3', To delete all messages of your mailbox, press '4', For Mailbox Greetings, press '5', To go to previous menu press '#'."
- Dial 2 → VMS prompts: "To set message redirection press '1', to cancel message redirection press '2', to go to previous menu press '#."
- Dial 1 to set message redirection → VMS prompts: "Enter the destination extension."
    - Dial valid destination extension → VMS responds: "Message redirection set."
- Dial 2 to cancel message redirection → VMS responds: "Message redirection cancelled."

Message Redirection using SA Jeeves

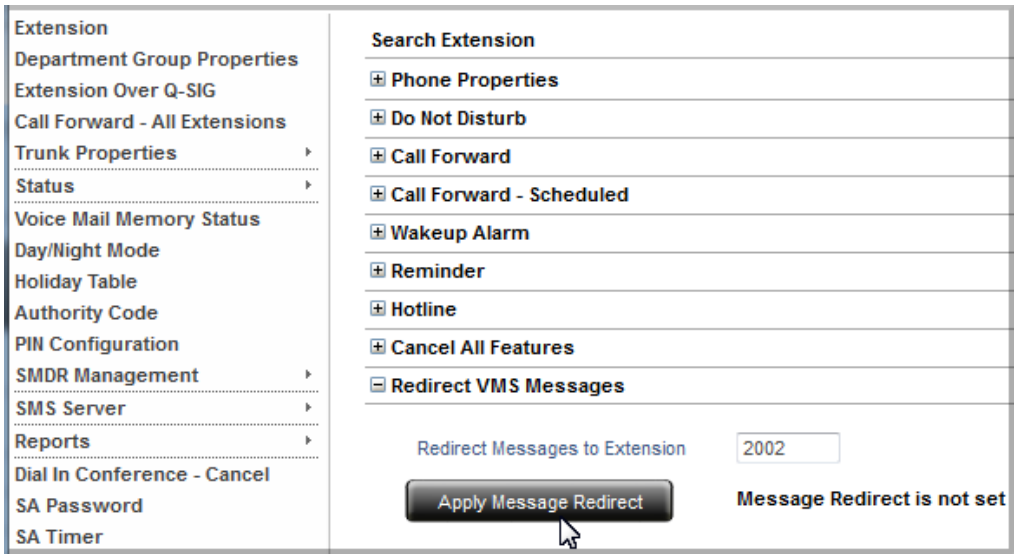
For Extension Users

- Log in to Jeeves as System Administrator.

- Click **Extension**.



- In **Select Extension,** enter the Number or the Name of the extension on which you want to set this feature.
- Click **Submit**.
- The searched extension users details appear on your screen.
- Click **Redirect VMS Messages** to expand.

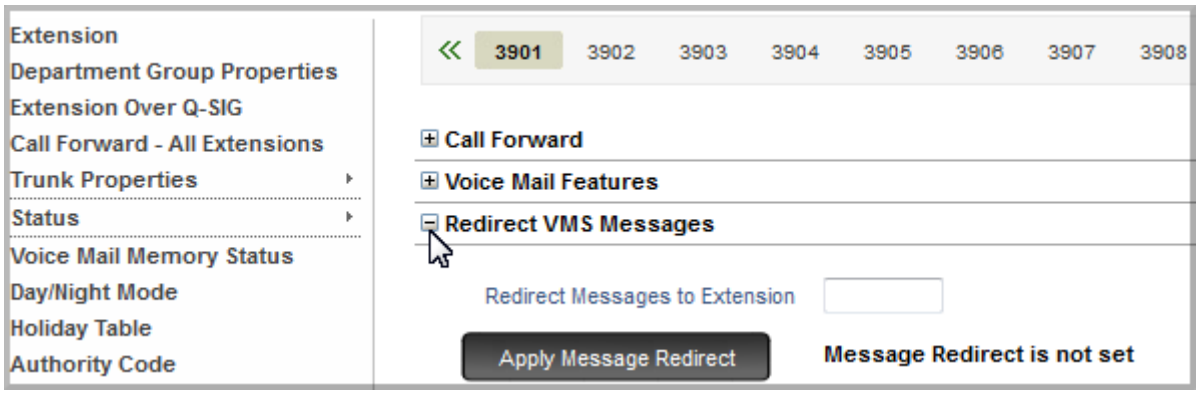


- In **Redirect Messages to Extension**, enter the extension number on which you want the messages to be redirected.
- Click the **Apply Message Redirection** button.

For Department Group

- Log in to Jeeves as System Administrator.

- Click **Department Group Properties**.



- Click the desired Department Group Number tab for which you want to set message redirection.
- Click **Redirect VMS Messages** to expand.
- In **Redirect Messages to Extension**, enter the extension number on which you want the messages to be redirected.
- Click the **Apply Message Redirection** button.

Message Redirection using SA Commands

- Enter SA mode from a DKP/SLT.

  To redirect Messages, dial:
  - ***1072-314-Source Station-Code-Destination Station***
    Where,
    Source Station is the Access Code of any SLT, DKP, SIP Extension, ISDN, Department Group.
    Destination Station is the Access Code of any SLT, DKP, SIP Extension, ISDN, Department Group or General Mailbox.

    Code is
    0 to Cancel Message Redirection
    1 is to Set Message Redirection

- Exit SA mode.

Recording Personal Greetings

What's this?

The VMS offers extension users the facility to customize the greeting messages for their mailbox. Extension users can record a different message for each time zone, namely working hours, break hours and non-working hours.

To be able to use this feature, make sure a digit has been assigned to Mailbox Management. For instructions, see "Mailbox Access".

How to use

- Call the VMS by dialing **3931**,
- The VMS takes you to your mailbox.
- VMS responds with: "You have <n> new messages" followed by the prompt: "Enter your mailbox password".
- Enter your mailbox password and the VMS prompts: "You have n new/no new messages."
- VMS prompts: "To listen to new messages press '1', to listen to old messages press '2', to send a message press '3', to change your mailbox settings press '4'."
- Dial 4 → The VMS responds with: "For Mailbox Name, press '1', For message redirection, press '2', To delete all old messages of your mailbox, press '3', To delete all messages of your mailbox, press '4', For Mailbox Greetings, press '5', To go to previous menu press '#'."
- Dial 5 → The VMS prompts: For Personal Greetings, press '1', For Conditional Greetings, press '2', To go to previous menu press '#'.
- Dial 1 → The VMS prompts: "For working hours greeting, press '1', for break hours greetings, press '2', for non-working hours greeting, press '3, to go to the previous menu, press '#''.
- Dial the digit of the desired time zone. The VMS prompts: "To record, press '1', to play, press '2', to erase, press '3', to go to the previous menu, press '#' ".
- To record the message, press 1. You can record a greeting message of 120 seconds. If the message duration is less than 60 seconds, press # after you have completed the message.
- The VMS prompts: "To record, press '1', to play, press '2', to erase, press '3', to go to the previous menu, press '#' ".
- To listen to the greeting message you recorded, press '2'.
- On completion of play back of the greeting message, the VMS prompts: "To record, press '1', to play, press '2', to erase, press '3', to go to the previous menu, press '#' ".

Recording Conditional Greetings

What's this?

The VMS offers extension users the facility to customize the greetings messages played to callers for certain conditions—busy, no reply or unconditional/unregistered Call Forward. Extension users can record a different message for each condition.

To be able to use this feature, make sure a digit has been assigned to Mailbox Management. For instructions, see "Mailbox Access".

How to use

- Call the VMS by dialing **3931**,
- The VMS takes you to your mailbox.
- VMS responds with: "You have <n> new messages" followed by the prompt: "Enter your mailbox password".
- Enter your mailbox password and the VMS prompts: "You have n new/no new messages."
- VMS prompts: "To listen to new messages press '1', to listen to old messages press '2', to send a message press '3', to change your mailbox settings press '4'."
- Dial 4 → The VMS responds with: "For Mailbox Name, press '1', For message redirection, press '2', To delete all old messages of your mailbox, press '3', To delete all messages of your mailbox, press '4', For Mailbox Greetings, press '5', To go to previous menu press '#'."
- Dial 5 → The VMS prompts: For Personal Greetings, press '1', For Conditional Greetings, press '2', To go to previous menu press '#'.
- Dial 2 → The VMS prompts: "For Busy press '1', for No Reply press '2', for Unconditional press '3', To go to previous menu press '#'.
- Dial the desired digit. The VMS prompts: "To record press '1', to play press '2', to erase press '3', to go to the previous menu press '#' ".
- To record the message, press 1. You can record a greeting message of 120 seconds. If the message duration is less than 60 seconds, press # after you have completed the message.
- The VMS prompts: "To record press '1', to play press '2', to erase press '3', to go to the previous menu press '#' ".
- To listen to the greeting message you recorded, press '2'.
- On completion of play back of the greeting message, the VMS prompts: "To record press '1', to play press '2', to erase press '3', to go to the previous menu press '#' ".

Message Verification

What's this?

Message Verification enables

- extension users to check the message they have recorded before sending it to someone.
- extension user to check the message they have recorded as a reply to any message.
- external callers to check the message they have recorded in the mailbox of an extension user.

Thus Message Verification is used in the VMS features — Messages left by External Callers, "Sending Messages", and "Broadcast Message".

How it works

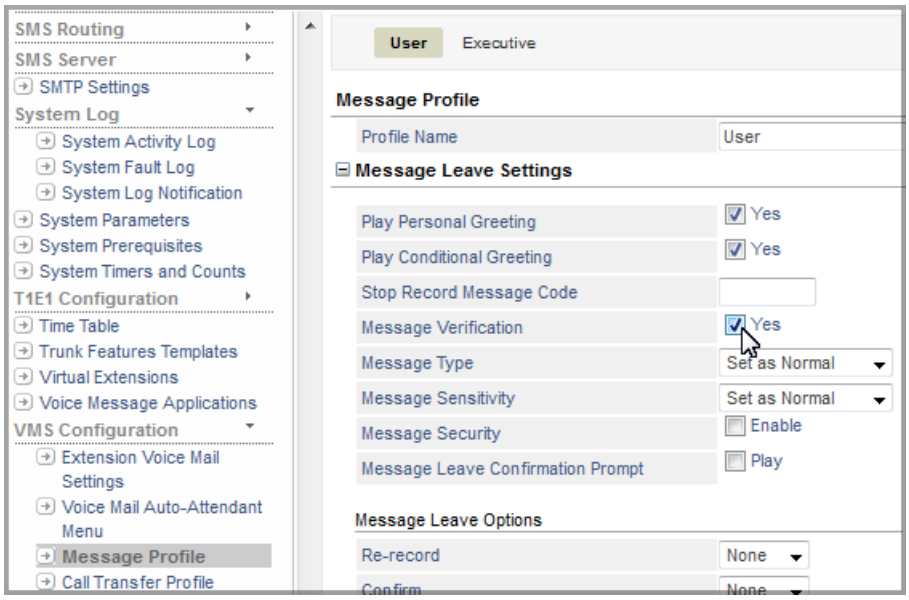
- For Message Verification to work, it must be enabled in the VMS.
- With Message Verification enabled, each time a caller or an extension user records a message, the VMS offers to the caller/extension user the option to verify the recorded message and re-record the message, if they want.
- When the caller/extension user uses the option to verify and re-record the message, the VMS sends the message to the mailbox of the receiver.

How to configure

By default, Message Verification is disabled in the VMS. So, callers and all extension users with voice mail facility cannot verify the message they have recorded. If you want to enable this feature,

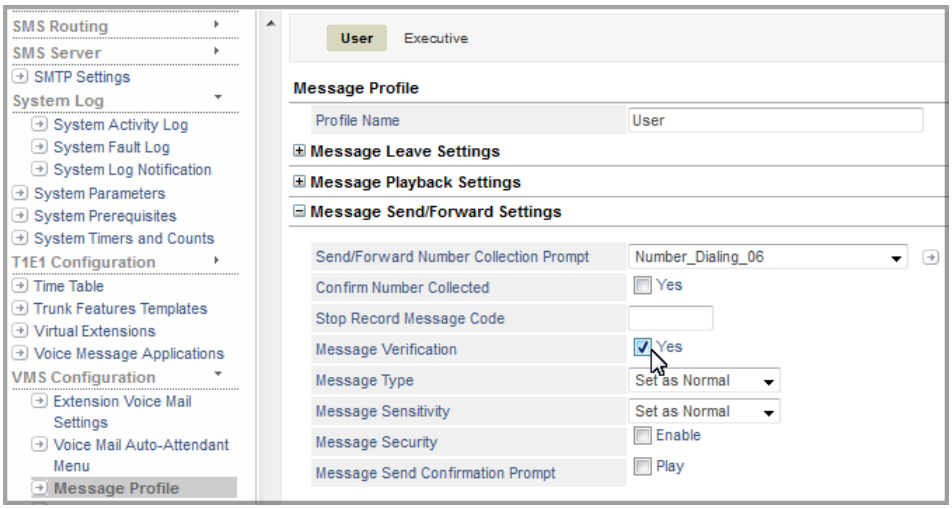
- Open Jeeves.
- Log in as System Engineer.
- Click **Configuration**, click **VMS Configuration**.
- Click **Message Profile**.
- To allow external callers to check the message they have recorded in the mailbox of an extension user or extension user to check the message they have recorded as a reply to any message.

- Click **Message Leave Settings** to expand.



- Select the **Message Verification** check box, to enable.
- Click **Submit** to save changes.

- To allow extension users to check the message they have recorded before sending it to someone,
  - Click **Message Send/Forward Settings** to expand.



- Select the **Message Verification** check box, to enable.
- Click **Submit** to save changes.

Message Notification

What's this?

The VMS sets Message Wait on the extension, whenever a new message arrives in its personal Mailbox of the extension. The VMS indicates the new message to the extension as per the Type of *Message Wait Indication* set for the extension. This may be in the form of a Stuttered Dial Tone, a Voice Message, Ring or LED Lamp. See "Message Wait" to know more.

The Message Notification feature of the VMS is an extension of the Message Wait feature. When **Message Wait Indication** for an extension is set as **Ring**, the VMS makes a Message Notification call to the extension.

How it works

- Extension A has Message Wait Indication type set as Ring.
- Whenever there is a new message in A's mailbox, the system will play the *Message Wait Ring* (Short, Fast) on extension A. See "Distinctive Rings".
- Extension A will ring for the duration of the Message Wait Ring Timer (configurable; default: 30 seconds).
- When Extension A answers the call within this timer, Extension A gets connected to the VMS.
- The VMS answers the call and allows access to Extension A's mailbox.
- If Extension A does not answer the Message Notification call within the Message Wait Timer, the system will ring on the extension again for as many times as the Message Wait Ring Count (configurable; default: 10 times), and at the interval set as the Message Wait Ring Timer Interval (configurable; default: 30 minutes).

How to configure

To provide Message Notification Call to extensions, you must configure **Ring** as **Message Wait Indication** for the extension. For detailed instructions, refer "Message Wait Settings" in "Extension Voice Mail Settings".

You may also configure the related Message Wait Timer, the Message Wait Ring Count and the Message Wait Ring Interval, if required. See "System Timers and Counts" for instructions.

How to use

- Go Off-hook, when your extension rings to indicate Message Wait (short, fast ring),
- The VMS greets with the message: "This is your message notification call".
- The VMS plays the message: "You have <n> new messages" followed by the prompt: "Enter your mailbox password".
- Dial your mailbox password, and follow VMS prompts.

Mailbox Settings

What's this?

The VMS allows extension users to change the settings of the following facilities of their mailbox:

- Record the Extension Name for their mailbox.
- Redirect Messages from their mailbox.
- Delete all old messages.
- Delete all messages.
- Record Personal and Conditional Greetings for their mailbox.
- Configure the Personal (Mobile/Alternate) Number and the Assistance Number.

To be able to use this feature, make sure a digit has been assigned to Mailbox Management. For instructions, see "Mailbox Access".

How to use

- Call the VMS by dialing **3931**,
- The VMS takes you to your mailbox.
- VMS responds with: "You have <n> new messages" followed by the prompt: "Enter your mailbox password".
- VMS prompts: "To listen to new messages press '1', to listen to old messages press '2', to send a message press '3', to change your mailbox settings press '4'."
- Dial 4 to go to Mailbox Settings.
- The VMS responds with: "For Mailbox Name press '1', For message redirection press '2', To delete all old messages of your mailbox press '3', To delete all messages of your mailbox press '4', For Mailbox Greetings press '5', To go to previous menu press '#'."

By default no digits are assigned for the options - Assistance Number and Personal Number, hence these options will not be played to the caller. For details, see "Mailbox Management".

To assign Name to your Mailbox:

- Dial 1 → VMS prompts: "To record press '1'. To play press '2'. To erase press '3'. To go to previous menu, press '#'."
- Dial 1 to record name for mailbox → VMS prompts: "Record your name after the beep and press any digit to end."

Names for extensions can also be recorded from the System Administrator mode. See "Record Greetings/ Name".

To delete old Messages in your Mailbox:

- Dial 3 → VMS prompts: “You are about to delete all old messages of your mailbox. To proceed press 1, to cancel press press any digit.”
- Dial 1 to delete all old messages in your mailbox → VMS responds with: “Your old messages have been deleted”.

To delete all Messages in your Mailbox:

- Dial 4 → VMS prompts: “You are about to delete all messages of your mailbox. To proceed press 1, to cancel press press any digit.”
- Dial 1 to delete all messages in your mailbox → VMS responds with: “Your messages have been deleted”.

To programme Assistance Number for your Mailbox:

- Dial 6 → VMS prompts: “To enter number press ‘1’, to play number press ‘2’, to clear number press ‘3’, to go to previous menu press ‘#’.”

Make sure the Assistance Number is an extension number.

To programme Personal (Mobile/Alternative) Number for your Mailbox:

- Dial 7 → VMS prompts: “To enter number press ‘1’, to play number press ‘2’, to clear number press ‘3’, to go to previous menu press ‘#’.”

Any external number can be configured as the Personal (Mobile/Alternative) Number.

To redirect Messages in your Mailbox, see “Redirecting Messages”.

To record personal greetings for your mailbox, see “Recording Personal Greetings”.

To record conditional greetings for your mailbox, see “Recording Conditional Greetings”.

Listening to Messages

What’s this?

The callers/extension user leave messages in the mailbox of extension users, when they are inaccessible or the user has forwarded his calls to the mailbox. Messages may also be received by the extensions users as notifications for certain events. User should access their mailboxes to listen to the messages.

VMS offers two options:

- To listen to old messages.
- To listen to new messages.

Once the message is heard by the mailbox owner, VMS treats it as an old message and places it in the old message list. The VMS also offers you the option of saving the message you have heard, as a new message.

How to use

- Call the VMS by dialing **3931**,
- The VMS takes you to your mailbox.
- VMS responds with: "You have <n> new messages" followed by the prompt: "Enter your mailbox password".
- Enter your mailbox password.
- The VMS checks the utilized mailbox memory,
    - if 80% of the mailbox memory has been consumed, the VMS prompts the caller: “Your Mailbox is 80% Full. Please Delete old messages of your mailbox.”
    - if 100% of the mailbox memory has been consumed, the VMS prompts the caller: ”Your Mailbox is Full. Please Delete old messages of your mailbox.”
- VMS prompts: "To listen to new messages press '1', to listen to old messages press '2', to send a message press '3', to change your mailbox settings press '4’."
- Dial 1 to listen to new messages Or dial 2 to listen to old messages.
- On the completion of a message, the VMS plays the message: ‘To replay the message press '1', for Date and time stamp press '2', to reply to message press ‘3’, to delete the message press '4', to play the next message press '5', to forward the message press '6', to save the message as new press '7', to go to previous menu press '#'.

Leaving a Message

What's this?

The VMS allows,

- External callers to leave messages in the extension user's mailbox.
- Extension users to leave message in the mailbox of other extensions.

To leave a message, the called extension must have a mailbox. The length of message recorded by the callers/ extension users must not exceed the message length set for the called extension's mailbox. If the message recorded by the callers/extension users exceeds the message length set for the called extension's mailbox, the VMS will stop recording the message after the time set and save the partially recorded message.

How to use

- The incoming call on the trunk is answered by the VMS Auto Attendant.The VMS greets the caller with the Greeting message followed by the Welcome Message: "Welcome! Please dial the extension number or To dial by name press '6',To leave a message press '7', To access your Personal Mailbox press '8', For further assistance press '9', To disconnect the call press '#'.

The System Greeting may vary depending on the timezone — Working Hours, Break Hours, Non-working Hours. In the above prompt the greeting is as per working hours.

- The caller dials 7 to leave message → VMS prompts: "Enter the Extension number for which you wish to leave message."

- The caller dials the desired extension number → VMS prompts: "Record your message after the beep and press any digit to end."

- VMS prompts: " Thank you for your call."

The system provides an option to the caller to verify the message before storing the same. Enable the Message Verification check box and configure the digits for the Message Leave Options. The message will be played as per the digits selected for the options. For detailed instructions, see "Message Send/Forward Settings"

It is mandatory for the caller/extension user to terminate the recording by dialing # (or the digit configured). If recording of the message is terminated simply by going on-hook, the VMS will not terminate the recording and the call will be disconnected only after time-out, that is, the Maximum Message Length configured for the extension.

If extension users wish to leave a message for another extension, refer "Sending Messages".

Accessing the General Mailbox

What's this?

A General mailbox is a shared mailbox between extension users. The General Mailbox is used for recording messages when the mailbox of an extension is full.

How it works

The VMS offers the following options to extensions when their mailbox is full:

- Not offer the caller to record a message.
- Overwrite the existing messages in the mail box with the new message.
- Deliver the new message to the General Mailbox.

If you configure Delivery of new messages to General Mailbox on an extension, whenever the mailbox of the extension is full, the VMS will offer the caller to record a message. This message will be recorded in the General Mailbox.

The extension user, whose mail box is full, can listen to the new message by accessing the General Mailbox.

The extension user can access the General Mailbox, if this feature is enabled in the Class of Service of the extension and the Password assigned to the General Mailbox (if configured) is known to the extension user. For details see "General Mailbox Settings".

How to configure

To offer extension users the facility of the General Mailbox when their mailbox is full, you must do the following:

- Configure the option **New Message Delivery Option in Mailbox Full Condition** in the **Extension Voice Mail Settings** of the extension.

  For the option **New Message Delivery Option in Mailbox Full Condition**, select **Deliver to General Mailbox**.

  For instructions on configuring the parameter, see "Extension Voice Mail Settings".

- Enable the feature General Mailbox in the "Class of Service (COS)" of the extension.

- If required, assign a Password for accessing the General Mailbox, see "General Mailbox Settings".

How to use

For EON & Extended IP Phone Users

- Press DSS Key assigned to General Mailbox.
  OR
- Dial 1176.
- Follow VMS prompts.

For SLT Users

- Lift handset.
- Dial 1176.
- Follow VMS prompts.

Forwarding Messages

What’s this?

The VMS enables extension users to forward messages of their mailbox to other mailboxes.

How it works

The Forwarding Messages feature of the VMS offers to extension users the following options:

- forward messages after adding a comment.
- forward messages without adding comment.
- forwarding messages with Message Read Receipt request.

Before forwarding a message, the VMS asks the Sender, if the Sender needs a confirmation that the message has been read by the Recipient.

If the Sender requests for 'Message Read Receipt', the VMS stores this request. When the Recipient reads the message, the VMS generates a file containing the first 5 seconds of the message that was sent by the Sender and delivers it to the Sender's mailbox in the form of a new message with a prompt: "This message was read by <Extension Name><5 seconds of the message sent>" with the Date and Time at which the message was read.

How to use

- Call the VMS by dialing **3931**.
- The VMS takes you to your mailbox.
- VMS responds with: "You have <n> new messages" followed by "Enter your mailbox password".
- Enter your mailbox password → VMS responds with: "You have no/n new messages".
- The VMS prompts: "To listen to new messages press 1, to listen to old message press 2, to send a message press 3, to change your mailbox settings press 4”.
- Dial 1 or Dial 2 and listen to the messages → VMS prompts: "To replay the message press 1' for Date and Time stamp press 2, to reply to message press 3, to delete the message press 4, to play the next message press 5, to forward the message press 6, to save the message as new press 7, to go to previous menu press #."
- Dial 6 → VMS prompts: "To forward the message with comment before the message press 1, to forward the message with comment after the message press 2, to forward the message without comment press 3, to go to previous menu press #."
- To forward message with comment before the message dial '1' or To forward message with comment after the message dial ‘2’. The VMS prompts: "Enter the Destinations".
    - Dial Destinations/Distribution list number to forward the message.
    - The VMS prompts: "Record your message after the beep and press any digit to end."

- Speak to record your comment and press # to end recording.
- The VMS prompts: "To request read receipt press 1, to ignore read receipt press 2."

- To forward message without comment dial '3'. The VMS prompts: "Enter the Destinations".
  - Dial Destination number/Distribution list number to forward the message.

The system provides an option to the caller to verify the message before storing the same. Enable the Message Verification check box and configure the digits for the Message Leave Options. The message will be played as per the digits selected for the options. For detailed instructions, see "Message Send/Forward Settings"

Extension users must be careful in dialing destination numbers. If invalid destination is entered then the VMS will clear all the entries and will ask the mailbox owner to re-enter all the destinations again.

A message can be forwarded to maximum 10 destinations. A destination can be an extension or a distribution list.

Email Based Notification

What's this?

VMS supports E-mail Based Notification, the Unified Messaging feature. This is used to inform the extension users, the arrival of new messages in their mailbox and the memory usage status of their mailbox.

Extension users can also receive new messages as attachments to the email.

Extension users will receive the notification for the mailbox memory usage for the following:

- when 80% of their mailbox memory has been consumed.
- when 100% of their mailbox memory has been consumed.

For Email Notification to function, you must configure the "SMTP Settings" .

How to configure

To be able to use this feature, you must configure the parameters for **Message Wait Notification via Email** under Message Wait Settings in Extension Voice Mail Settings.

- You can select the desired option as Notification: **Without attachment** , **With attachment** or **With attachment and mark voicemail as read**.

- Specify the **E-mail Address** of the extension user to which the notifications are to be sent.

  For the General Mailbox you only need to specify the **E-mail Address** to which the notifications are to be sent.

For details see "Message Wait Notification via E-Mail" in "Extension Voice Mail Settings".

The Notification sent for arrival of new messages in the mailbox and the memory usage status users mailbox will be sent to the *Email ID* configured under *Message Wait Notification via Email*.

The messages sent can be customized as per your requirement. For detailed instructions, refer "VMS E-Mail Notification".

Message Wait Notification via Call

What's this?

The VMS supports Notification via Call to inform the extension users about the arrival of new messages in their mailbox.

Extension users can receive new message notification calls on a phone number of their choice. This number may be another extension number or an external number. You can set the Type of notification calls as

- **Immediate:** Users will receive notifications as soon as a new message arrives in their mail box.
  Or
- **Scheduled:** Extension users will receive notification at specified time intervals.

You can set the preferred time slots in a day during which notification calls should be made to extension users. In addition to the time slot preference, you can also choose to receive notification calls on a Holiday.

Message Notification via Call for Department Group will not work if the destination number is an external number.

How it works

For this feature to work, you must do the following configuration for the extension:

- Select the type of Notification call.
- Define the preferred time slots by configuring Time Zones. You can configure four different Time Zones, defining the Start Time and End Time for each Time Zone.
- Configure the phone number to which the notification call is to be made. If the number is an external number, configure the Trunk Access Code to be used for making the calls.

When **Immediate** is selected as the Type of notification,

- A new message arrives in the mailbox of the extension user.
- The system checks the preferred start and end time of the time zones configured for the extension. If the message has arrived within the preferred time slot (Start and End Time) it immediately makes the notification call on the number configured for the user.

  If the number is an external number, the system dials out the number using the Trunk Access Code (TAC) assigned for making notification calls.
- When the call is answered, the extension user gets connected to the VMS and can listen to the message.
- If the notification call is not answered, by default, the system makes three attempts (Message Notification Retry Count; programmable) at an interval of 5 minutes (Message Notification Interval; programmable) between each attempt.
- If the notification call remains unanswered after the third attempt, the system will not make any more attempts to place this notification call. The next notification call will be made only when another new message arrives in the mailbox of the user between the start and end time of the configured time zone.

When **Schedule** is selected as the Type of notification,

- A new message arrives in the mailbox of the extension user.
- The system checks the start time of the time zone(s) configured and the notification call will be made on the number at the subsequent start time.

  If the number is an external number the system dials out the number through the Trunk Access Code (TAC) assigned for making notification calls.

- When the call is answered the extension user gets connected to the VMS and can listen to the message.

- If the notification call is not answered, the system makes three attempts (Message Notification Retry Count; programmable) at an interval of 5 minutes (Message Notification Interval; programmable) for each time zone. The system will continue to make attempts to place the notification call till the call is answered.

Thus, when Notification type is Immediate, notification call is made for each message that is received within the start and end time configured in the time zone.

When Notification type is Scheduled, notification call is made for all messages received before the start time configured in the time zone. Where multiple time zones are configured, notification call will be made at the start time of the next time zone.

How to configure

For Message Wait Notification via Call, you need to configure:

- the parameters for **Message Wait Notification via Call** under Message Wait Settings in “Extension Voice Mail Settings”.

- select the desired profile in Schedule Profile. For instructions to configure the profile parameters, see “Configuring Notification via Call - Profile”.

- Make Message Notification call using TAC for calls to be made to external numbers. See “Configuring VMS General Parameters”.

- if required, the Message Notification Retry Count, Message Notification Interval and Message Notification Ring. See “System Timers and Counts”.

Dial By Name

What's this?

The VMS Auto Attendant allows external callers and extension users to reach the desired person in an organization by dialing the name of that person. This feature is useful when caller/extension user cannot recall the extension number of the person they want to speak to.

How to configure

For this feature to work, each extension user's name must be abbreviated and configured on the extension. It is recommended that the extension users' names be abbreviated to the first three letters of the name. As far as possible, abbreviate names such that no two names are the same.

You must configure the **Abbreviated Name** in the *Extension Voice Mail Settings*. For detailed instructions, refer "Extension Voice Mail Settings".

Make sure the names for the desired extension users are recorded. For details, see "Mailbox Settings".

How to use

- The incoming call on the trunk is answered by the VMS Auto Attendant. The VMS greets the caller with the Greeting message followed by the Welcome Message: "Welcome! Please dial the extension number or To dial by name press '6',To leave a message press '7', To access your Personal Mailbox press '8', For further assistance press '9', To disconnect the call press '#'.

The System Greeting may vary depending on the timezone — Working Hours, Break Hours, Non-working Hours. In the above prompt the greeting is as per working hours.

- The caller dial 7 → VMS prompts: "Please enter first three letters of the name."
- If multiple matches are found the VMS prompts: "More than one match found. Matching Names will be played one by one. To Select the name press '1', to Skip the name press '2', To Repeat the last name press '3'."
- Dial 1 → VMS prompts: "To confirm press '1', to Re-enter press '2'."
- The caller dials 1 → VMS transfers the call as per the transfer type assigned to the selected station. Talk.
    - If the caller dials invalid digits, the VMS prompts: "No match found." followed by the prompt: "Please enter first three letters of the name."

Extension users can use "Dial By Name" to reach another extension.

Dial by Extension Number

What's this?

The VMS Auto Attendant allows external callers and extension users to reach directly the desired person in an organization by dialing the extension number of that person.

How to use

- The incoming call on the trunk is answered by the VMS Auto Attendant. The VMS greets the caller with the Greeting message followed by the Welcome Message: “Welcome! Please dial the extension number or To dial by name press ‘6’,To leave a message press ‘7’, To access your Personal Mailbox press ‘8’, For further assistance press ‘9’, To disconnect the call press ‘#’.

The System Greeting may vary depending on the timezone — Working Hours, Break Hours, Non-working Hours. In the above prompt the greeting is as per working hours.

- The caller dials valid extension number.
    - VMS prompts: "Please enter first three letters of the name.”

Make sure you have:
- *enabled the Confirm Name check box.Refer to “Auto-Attendant Settings”.*
- *configured the **Abbreviated Name** for the respective extension. To know more, refer to “Extension Voice Mail Settings”.*

    - The caller enters the first three letters of the name.
        - If the caller dials valid digits and if the **Abbreviated Name** is not configured → VMS transfers the call as per the **Call Transfer Profile** selected for the extension. Talk.
        - If the caller dials invalid digits → VMS prompts: "No match found." followed by the prompt: "Please enter first three letters of the name."

- If the extension number dialed by the caller is invalid, the VMS prompts: "The number is not valid." followed by the prompt: "Please dial the extension number or To dial by name press ‘6’,To leave a message press ‘7’, To access your Personal Mailbox press ‘8’, For further assistance press ‘9’, To disconnect the call press ‘#’.
- Dial valid extension number and talk.

Call Transfer Types

What's this?

The VMS Auto Attendant answers calls of external callers and extension users (referred to here as 'callers') and transfers the call to the extensions according to the Call Transfer type set for the extension.

You must configure the Call Transfer Type in the Call Transfer Profile assigned to the extension. Different Call Transfer Profiles can be assigned for Working, Break and Non-working hours.

The VMS Auto Attendant offers the following types of Call Transfers:

- **Blind**: When the caller dials the extension number, the VMS Auto Attendant transfers the call on the extension without checking whether it is busy or free.
- **Wait for Ring**: When the caller dials the extension number, the VMS Auto Attendant waits for the extension to start ringing and then transfers the call.
- **Wait for Answer**: When the caller dials the extension number, the VMS Auto Attendant transfers the call when the extension answers (goes OFF-Hook).
- **Screened**: The VMS Auto Attendant prompts the caller to record his/her name. It puts the caller on hold and places the call on the desired extension. If the extension is free and answers the call, the VMS announces the caller's name to the extension user and prompts the extension user to choose whether or not to speak to the caller. If the extension user chooses to talk, the VMS transfers the call.
- **None**: When the caller dials the extension number, the VMS Auto Attendant transfers the call to the desired extension users mailbox directly.

How to configure

Call Transfer Type must be configured in the Call Transfer Profile assigned to each extension. For instructions, refer "Call Transfer Settings" in "Extension Voice Mail Settings" and "Call Transfer Profile".

For calls received on trunks the VMS Auto Attendant transfer the call as per the configuration in "Auto-Attendant Settings" in "Voice Mail Auto-Attendant Menu".

Broadcast Message

What's this?

Broadcasting Message allows you to send the same message to all extension users having voice mail, at the same time. You can use Broadcast Message to make general announcements like hosting of an event, an unplanned day off, and other such activities or events.

How to use

To Broadcast Message,

- Enter SA Mode.
- Dial 1072-301
- VMS prompts: "Record your message after the beep and press any digit to end". (Recmsg.wav)
- Speak to record your message after the beep, and press # (hash/pound) to end the message.

The system provides an option to the caller to verify the message before storing the same. Enable the Message Verification check box and configure the digits for the Message Leave Options. The message will be played as per the digits selected for the options. For detailed instructions, see "Message Send/Forward Settings"

The length of the message you want to broadcast must be equal to or less than the minimum of message length programmed for the mailboxes, or your message will be truncated. For instance, if the Maximum message length for a mailbox is configured as 15 seconds, maximum length of the message to be broadcast must be less than or equal to 15 seconds. If the broadcast message exceeds this limit, the system will play the first 15 seconds and truncate the remaining part of the message.

CHAPTER 17 System Maintenance

Certificate Management

What's this?

SARVAM UCS supports certification for TLS and Web Server. In case, the Clients (Standard SIP phones) require this certificate, you can download the same and install it in them.

SARVAM UCS supports the following types of Certificates.

1. Root CA Certificate
2. Self-Signed System Certificate
3. CA-Signed System Certificate

Certificate Authority (CA) is a trusted (third-party) organization which creates and sells TLS Certificates to websites or organizations. CAs issue a TLS Certificate to the organizations/websites after verifying their credentials.

Generally, one TLS Certificate is issued for a particular server/website domain and it is valid for a limited period of time.

SARVAM UCS supports upto 5 (1 Default Root CA + 1 default Self-Signed +3 newly generated Self-Signed or CA-Signed) Certificates in the system.

The Default Root Certificate and Default System Certificate cannot be deleted from the system.

Now, let us understand the purpose of each type of certificate.

Root CA Certificate

The Root CA Certificate is an in-built Certificate Authority which signs the System Certificates created by our own server or by other servers. The Root CA Certificate is also used by the clients to validate the Server Certificate that is received during TLS negotiation.

By default, a Root CA Certificate is already provided in the system. You may also regenerate a Root CA Certificate to bind the Certificate with the details of your organization. Refer “Regenerate Root CA” to know more.

Self-Signed System Certificate

When a System Certificate is signed using a Root CA Certificate, it generates a Self-Signed System Certificate. This certificate is generated by the clients themselves or by the Servers and then given to their clients. The Self-Signed System Certificate is faster to create since it is self-issued but it is not as robust as CA-Signed Certificate.

This certificate must be installed in the trusted list of clients that connect over TLS with the Server. Since the certificate is self-signed, it is not likely to be in the clients' trust file, hence, they need to add it. To know more, refer "Generate Self-Signed System Certificate".

If a remote client has a policy of accepting certificates only from trusted CAs, then it is likely that the Self-Signed Certificate sent by the server during TLS negotiation might get rejected. In such cases, you need to install a CA-Signed System Certificate in the system.

CA-Signed System Certificate

CA-Signed System Certificates are the TLS Certificates which are created by trusted (third-party) Certificate Authorities, signed and sold to any applicant. These certificates contains the identity of the owner. It is the responsibility of the CA to verify the owner's (applicant's) credentials.

Since the CA-Signed System Certificate is issued by a trusted CA, it ensures complete protection from security threats.

If you wish to install a CA Signed Certificate in your system, you must do the following.

1. Generate and enroll the Certificate Signing Request (CSR). For more details, refer "Generate Certificate Signing Request".
2. Get the Certificate Signing Request (CSR) verified and signed by the Certified Authority (CA).
3. Upload the CA-signed system certificate in the server. For more details, refer "Upload Certificates".

Enrolling the Certificate Signing Request with CA

Enrollment is a process of obtaining a certificate from any trusted third party (CA). After you have generated and downloaded the Certificate Signing Request (CSR), you must contact any authorized third party that issues TLS Certificates to companies or web owners, such as GoDaddy, DigiCert, Thawte, VeriSign, etc. and enroll the Certificate Signing Request (CSR) with them. These third parties Certificate Authorities (CA) have their charges to sign and validate the Certificate Signing Request (CSR) for a fixed time interval.

Verification and Signing of the Certificate Signing Request by CA

On receiving the Certificate Signing Request (CSR), the CA verifies the Server's / User's credentials. After successful verification, the CA signs and sends the signed certificate to the server. These signed certificates are called as CA-Signed System Certificate.

Upload of CA-Signed System Certificate

After the CA-signed system certificate is received, upload it in your server along with the private key. This certificate will be sent by the server to the clients, if assigned for the service, during TLS negotiation.

How it works

To be able to use this feature,

- Select the **Enable SIP over TLS** check box in “Configuring VoIP Parameters”.
- Select **Transport Mode**[355] as TLS in Location-1 of SIP Extension Settings of the respective Extended IP phone.
- Make sure the clients communicating with the server over TLS protocol are compatible with the TLS version configured in the system. To know more, refer **Allowed TLS Versions** in “Security Settings”.
- Make sure the Date and Time of the server is synchronized with the SNTP Server.
- Make sure that the validity of the Certificate has not expired to ensure secure connection between the clients and the server.

Now, let us understand how it works,

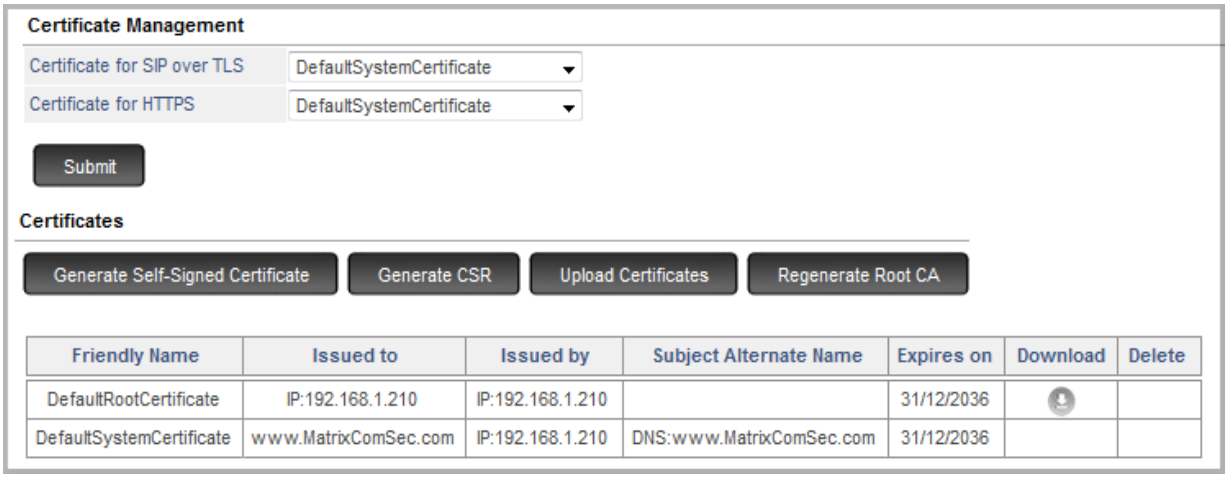
- The system has a default Root CA certificate installed which is generated using your own company related details.
- The Root CA certificate is used to create the System Certificate (self-signed), which is used by the clients for secure communication with the server.
- If you want to use System Certificate signed by some third-party Certificate Authority, then you need to generate a CSR (Certificate signing request). For example, for Standard SIP clients and third-party servers.
- When you generate and download the CSR, two files are saved — the CSR file and the Private key. You may secure the Private key using a Pass-phrase.
- Send the CSR file only to the authorized Certificate Authority (CA), get it signed and upload the CA-Signed System Certificate along with the private key in your system. The same pass phrase, if configured while generating CSR, needs to be entered while uploading the CA-signed certificate in your system.
- After the Certificate is successfully uploaded, it is displayed in the list of certificates.
- The System Certificates (Self-signed or CA-signed) thus ensure secure communication for SIP over TLS and HTTPS between the server and clients.

How to configure

- Log in as System Engineer.
- Click **Maintenance**.

355. Not applicable for SPARSH VP248 and Standard SIP Phones.

- Under Maintenance, click **Certificate Management**.



Certificate Management

- **Certificate for SIP over TLS:** Select the Certificate that will be used to establish secure SIP connection between server and IP devices. Default: Default System Certificate.
- **Certificate for HTTPS:** Select the Certificate that will be used to establish secure HTTP connection between server and clients. Default: Default System Certificate.

*If you change the Certificate in **Certificate for SIP over TLS** or **Certificate for HTTPS**, then it may result in drop of all ongoing TLS connections and VoIP calls.*

- Click **Submit** to save the changes you made.

Certificates

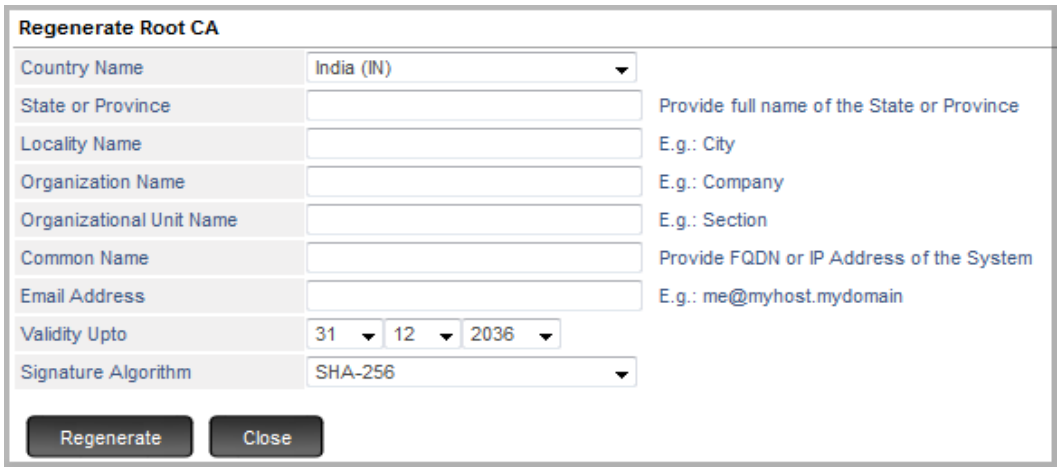
Regenerate Root CA

You may use this facility if you want to generate a Root Certificate depending upon the installation scenario.

On regeneration of a Root CA Certificate, the default Root CA Certificate will get replaced. Additionally, all the Self-Signed System Certificates are regenerated and replace the existing ones.

To regenerate a Root CA Certificate,

- Click the **Regenerate Root CA** button.



A new window opens which displays the parameters to be configured.

- **Country Name:** - Select the name of the country from the list. Default: India (IN).
- **State or Province:** - Enter the full name of the state or province. For example, Gujarat.
- **Locality Name:** Enter the name of the city. For example, Vadodara.
- **Organization Name:** Enter the name of your organization where SARVAM UCS is installed. For example, Matrix.
- **Organizational Unit Name:** Enter the name of the unit or section or domain of your organization, where SARVAM UCS is installed. For example, Factory.
- **Common Name:** Enter the FQDN or IP Address of the system. This Common Name serves as the distinguishing factor. The Common Name can be a maximum of 64 characters. For example, www.sarvam.com.
- **Email Address**: Enter the Email address of your host. For example, xyz@matrix.com. The Email Address can be a maximum of 40 characters.
- **Validity Upto:** Select any date from the present date of the system to 31st December 2036, the duration for which the certificate should be valid. Default: 31 December 2036.
- **Signature Algorithm:** Select the algorithm using which the certificate has to be signed. You may select — SHA-1, SHA-256,SHA-512. Default: SHA-256
- Click **Regenerate**.

  A message appears to confirm that the Certificates have been generated successfully.

  Once the Root CA has been successfully generated, the system automatically regenerates the Self-Signed Certificates using the newly generated Root CA Certificate.
- Click **OK** to close the window. You will be redirected to the Certificate Management page.

Generate Self-Signed System Certificate

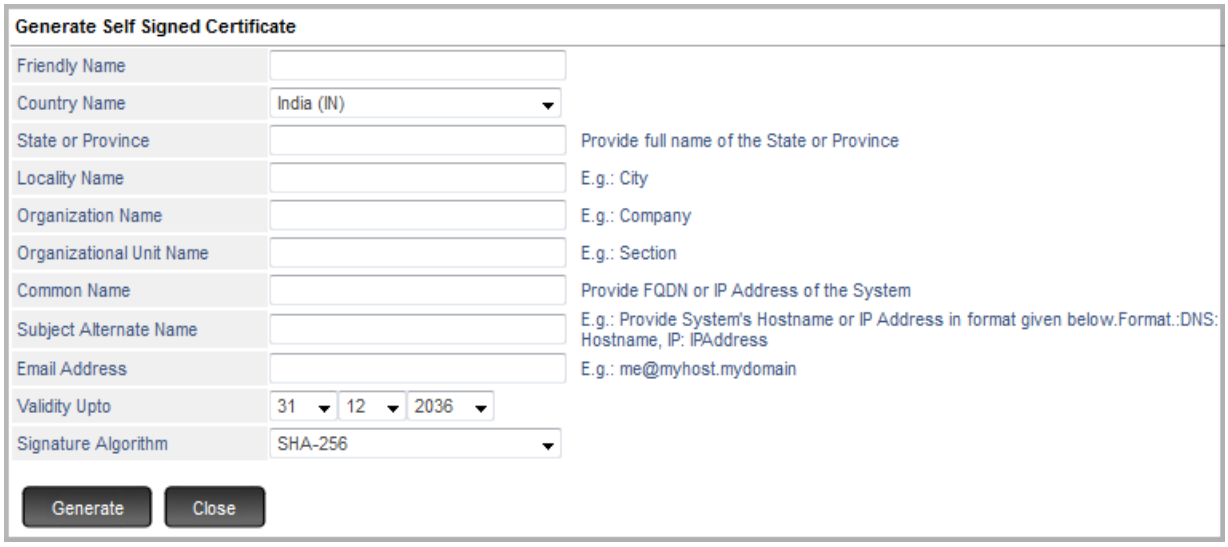
Depending on your installation scenario, you can generate a Self-Signed System Certificate using the details of your own organization.

The Self-Signed System Certificates cannot be downloaded.

You can generate multiple Self-Signed System Certificates using the same Root CA Certificate.

To generate a Self-Signed System Certificate,

- Click the **Generate Self-Signed Certificate** button.



A new window opens which displays the default parameters.

- **Friendly Name:** Enter the name you want to assign to the certificate. Make sure the name you enter is unique and does not match with the name of any other certificate configured in the system. The name can be a maximum of 32 characters. For example, MatrixComsec
- **Country Name:** - Select the name of the country from the list. Default: India (IN).
- **State or Province:** - Enter the full name of the state or province. For example, Gujarat.
- **Locality Name:** Enter the name of the city. For example, Vadodara
- **Organization Name:** Enter the name of your organization where SARVAM UCS is installed. For example, Matrix.
- **Organizational Unit Name:** Enter the name of the unit or section or domain of your organization, where SARVAM UCS is installed. For example, Factory.
- **Common Name:** Enter the FQDN or IP Address of the system. This Common Name serves as the distinguishing factor. For example, www.sarvam.com.
- **Subject Alternate Name:** Enter the names of the multiple domains separated by comma (if the same certificate is to be issued for multiple domains of an organization). The Subject Alternate Name can be a maximum of 255 characters. For example, DNS:matrixcomsec.com, IP: 192.168.101.123

Make sure you enter “DNS:” before the domain name and “IP:” before the IP address in the Subject Alternate Name.

- **Email Address**: Enter the Email address of your host. For example, xyz@matrix.com. The Email Address can be a maximum of 40 characters.
- **Validity Upto:** Select any date from the present date of the system to 31st December 2036, the duration for which the certificate should be valid. Default: 31 December 2036.
- **Signature Algorithm:** Select the algorithm using which the certificate has to be signed. You may select — SH-1, SHA-256,SHA-512. Default: SHA-256
- Click **Generate** to generate the Self-Signed Certificate.

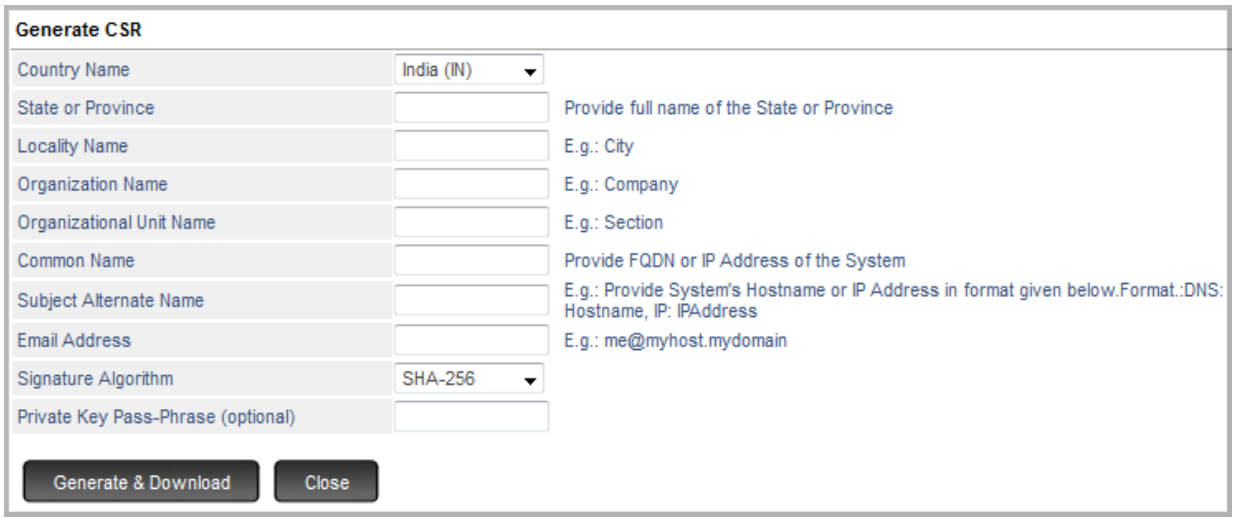
  A message appears to confirm that the Certificates have been generated successfully.
- Click **OK** to close the window**.** You will be redirected to the Certificate Management page.

  The newly generated Self-Signed System Certificate will appear in the **Certificates** list.

Generate Certificate Signing Request

To generate a Certificate Signing Request (CSR),

- Click the **Generate CSR** button.



A new window opens which displays the parameters to be configured for generating CSR.

- **Country Name:** - Select the name of the country from the list. Default: India (IN).
- **State or Province:** - Enter the full name of the state or province. For example, Gujarat.
- **Locality Name:** Enter the name of the city. For example, Vadodara
- **Organization Name:** Enter the name of the organization where SARVAM UCS is installed. For example, Matrix.

- **Organizational Unit Name:** Enter the name of the unit or section or domain of your organization, where SARVAM UCS is installed. For example, Factory.

- **Common Name:** Enter the FQDN or IP Address of the system. This Common Name serves as the distinguishing factor. For example, www.sarvam.com. The Common Name should not exceed 64 characters.

- **Subject Alternate Name:** Enter the names of the multiple domains separated by comma (if the same certificate is to be issued for multiple domains of an organization). For example, DNS:matrixcomsec.com, IP: 192.168.101.123

Make sure you enter "DNS:" before the domain name and "IP:" before the IP address in the Subject Alternate Name.

- **Email Address**: Enter the Email address of your host. For example, xyz@matrix.com. The Email Address should not exceed 40 characters.

- **Signature Algorithm:** Select the algorithm using which the certificate has to be signed. You may select — SH-1, SHA-256,SHA-512. Default: SHA-256

- **Private Key Pass-Phrase (Optional):** Enter the Private Key Pass-Phrase for encrypting the private key. The password must
  - be of minimum 6 characters and can be a maximum of 24 characters.
  - include atleast one upper-case, one lower-case, one number and one special character.
  - all ASCII characters (except Percentage %, Hash #, Equal to =, Plus +, And &, Backslash \, Less than <, Greater than >, Apostrophe ', Double Quote " and Space) are allowed.

    Default: Blank.

- Click **Generate & Download** to generate and download the Certificate Signing Request and the Private key.
  OR
  Click **Close** to close the window.

- *The downloaded file will be a zip file. Extract the files and send the CSR file only to any trusted Certificate Authority.*

- *The zip folder download depends upon the browser you are using. Check the **Download Settings** of your browser and set the Download path accordingly.*
  OR
- *If your browser does not ask you for the location you want to save your file, it saves it in the default location according to the download path specified for that browser.*

- *If you are using Mozilla Firefox (version 3.5 recommended), before you save the configuration files, set the **Downloads** option of your browser as **Always ask me where to save files**.*

- *The downloaded Private Key should not be sent to the trusted (third-party) Certificate Authority to prevent it from being mishandled by some malicious user.*

- *Make sure that you do not lose the Private Key or forget the Pass-Phrase; otherwise, you will have to generate a new CSR and set the Pass-Phrase again.*

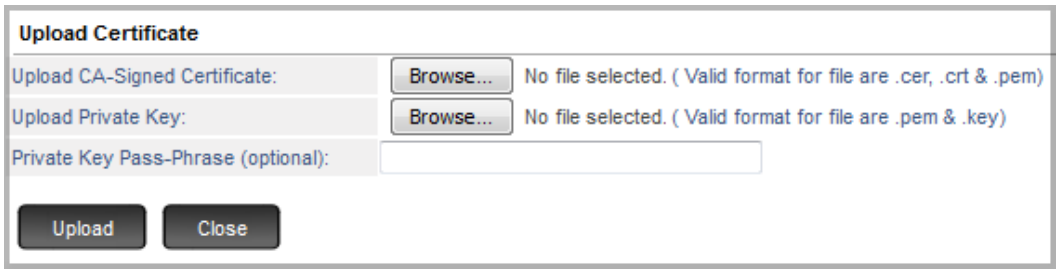
Upload Certificates

The size of the file to be uploaded should not exceed 5MB.

After you download the CSR, you must get it signed from a Trusted Certificate Authority (CA) and then upload it in your system.

To upload the CA-Signed Certificate,

- Click the **Upload Certificates** button.



  A new window opens which provides the option for uploading the CA-signed certificate and Private key.

- **Upload CA-signed Certificate:** Click the **Browse** button to reach the location on the local disk where the CA-Signed Certificate is stored in the PC. The valid formats for certificate are .cer, .crt and .pem.

- **Upload Private Key:** Click the **Browse** button to reach the location on the local disk where the Private key is stored in the PC. The valid formats for key are .pem and .key.

- **Private Key Pass-Phrase (optional):** Enter the Private Key Pass-Phrase for decrypting the private key. Make sure that you enter the same Private Key Pass-Phrase that was configured during generation of the CSR.

  If you have not configured any Pass-Phrase during generation of CSR, that is, Private Key is not Pass-Phrase protected, then you may leave this field blank.

- Click the **Upload** button to upload the CA-Signed Certificate and the Private key.

  A message appears to confirm that the Certificate has been uploaded successfully.

- Click **OK** to close the window. You will be redirected to the Certificate Management page.

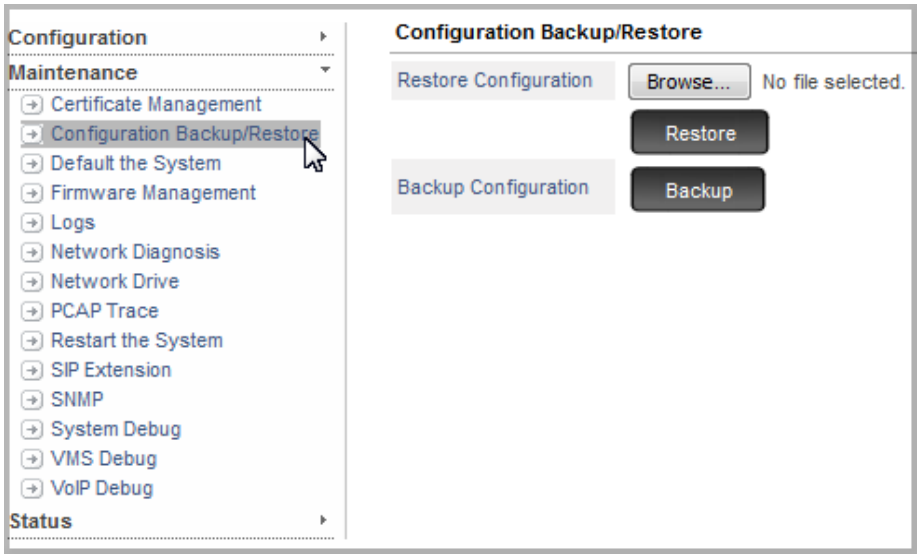
  The CA-Signed System Certificate you uploaded will appear in the **Certificates** list.

  To delete a System Certificate, click .

  To download the Root Certificate or CA-Signed System Certificate, click .

Configuration Backup/Restore

SARVAM UCS provides you the facility to Backup the configuration files from the system to the hard drive and Restore the configuration files from the hard drive to the system at the click of a button.



Restore Configuration

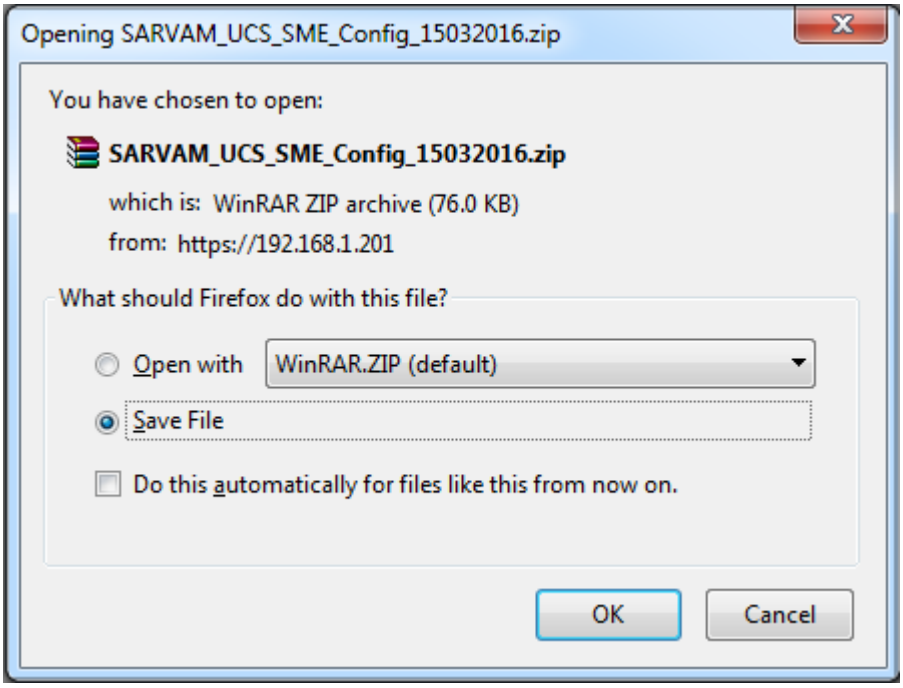
- To restore configuration files from any hard drive to the system, **Restore Configuration** option is provided.
- Click on the **Browse** button to reach the location on the local disk where the configuration files are stored in PC. Make sure that the file is a zip file.
- After selecting the required configuration zip file from PC, click on the **Restore** button.

  The system initiates the restoring of configuration zip file in flash storage. After successfully restoring and validating the file, the system restarts with the new configuration.

If you select a file other than zip file, an error message is displayed when you click on the Restore button.

Backup Configuration

- To save the existing configuration files as backup, click the **Backup Configuration** button. The **SARVAM_UCS_SME_Config_ddmmyyyy.zip** window will open; where ddmmyyyy signifies the current date.

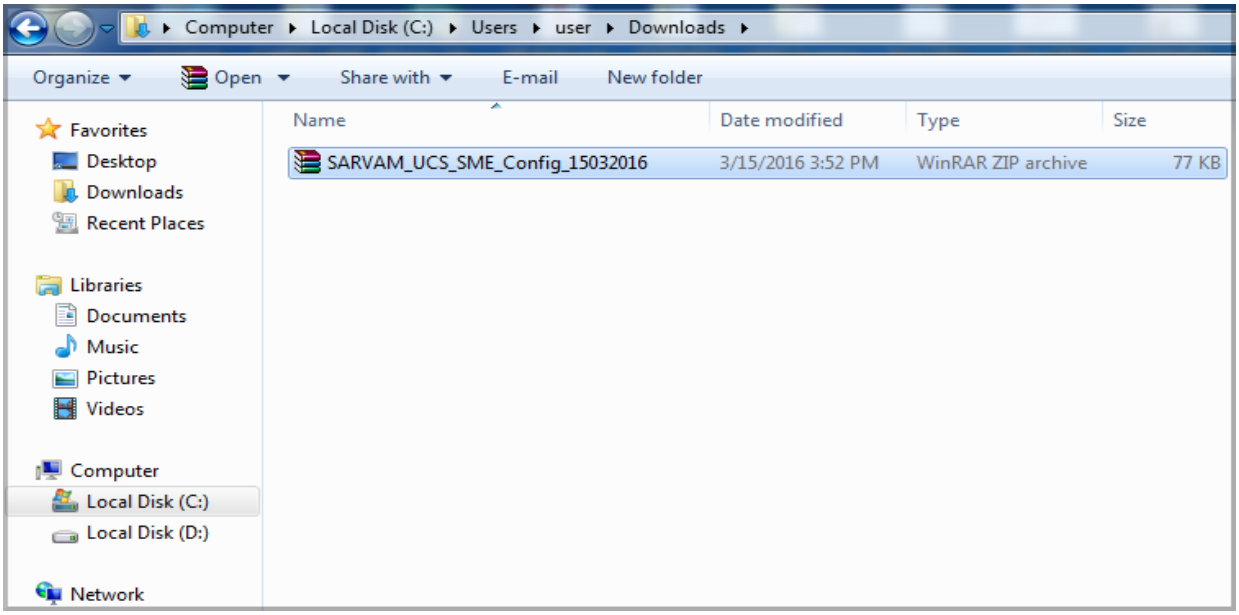


- You can either open the zip file or save the file to a location.

*The above window display depends upon the browser you are using. Check the **Download Settings** of your browser and set the Download path accordingly.*
OR
If your browser does not ask you for the location you want to save your file, it saves it in the default location according to the download path specified for that browser.

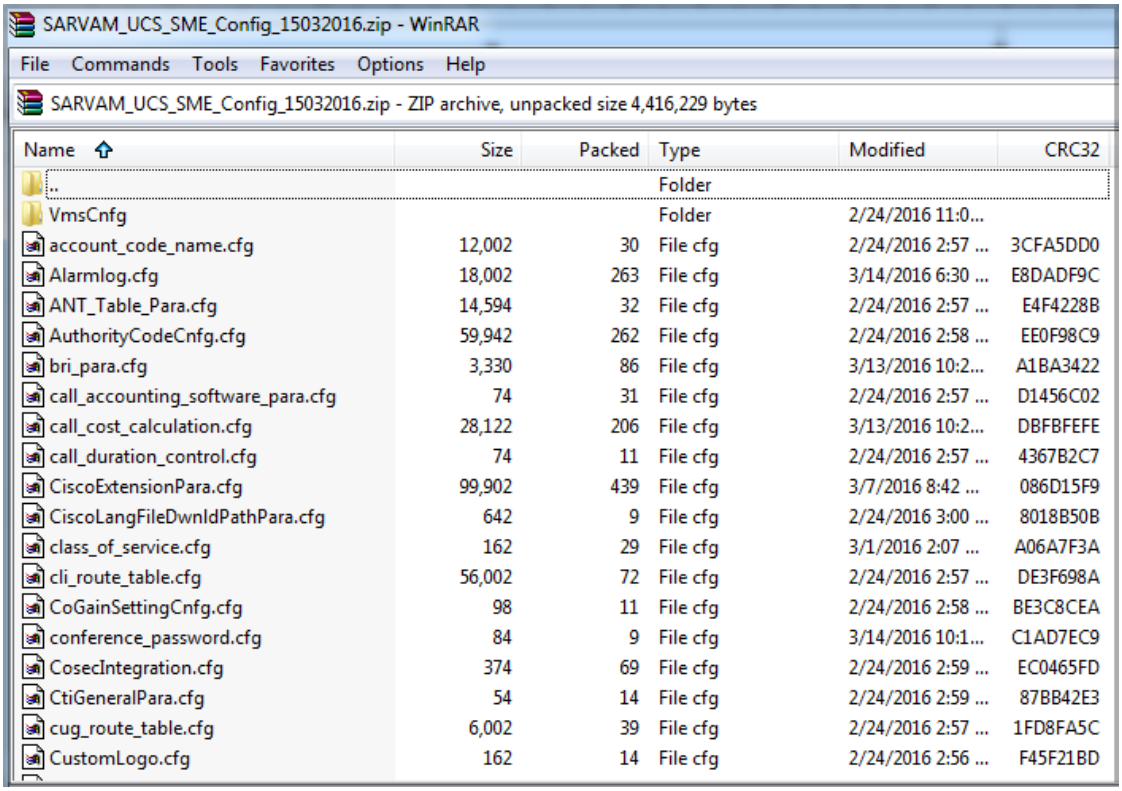
*If you are using Mozilla Firefox (version 3.5 recommended), before you save the configuration files, set the **Downloads** option of your browser as **Always ask me where to save files**.*

- Save the file on the local disk.



Save the back up configuration files by tagging the file name with the Version-Revision of the Firmware and tag the name of the backup folder on your computer with the date. This will help you at the time of restoring the back up configuration files.

- Open the SARVAM_UCS_SME_Config_15032016 folder to view the configuration files.



- Keep this folder as a backup. In case there is a problem with the system configuration files these backup files can be restored back in the system.

Default Settings

What's this?

SARVAM UCS is supplied with preset values for system and feature settings, as which may be altered and customized by users to match their requirements and preferences. The factory-set values for system and feature settings that are automatically assigned by the system are referred to as Default Settings or standard settings.

Every configurable parameter in the system has factory-set default values, which may be changed or customized to match user requirements and preferences.

If you have purchased / upgrading the system with Firmware Version later than V1R6.7, due to security concerns the default values of certain parameters have been changed. For new systems purchased these Default Settings will the applicable automatically. For details refer to “Modified default parameter values for Firmwares later than V1R6.7”. If you are upgrading the system, refer to “After updating Firmware later than V1R6.7”.

How it works

The default settings are to be loaded or restored in the following situations:

1. ***Installing the SARVAM UCS in a country other than India.***

   SARVAM UCS provides default settings to match country/region-specific requirements of users worldwide.

   The default settings are set to match user requirements of India.

   So, you must select the appropriate “Region” for the country/region in which the system is installed.

   The system will load the default settings for the country/geographical region where the system is installed.

   The system is designed to work efficiently with the default settings. So, if the country/region-specific default settings match their requirements, you may not even need to alter or customize the values of various parameters.

   They may work with default settings for the most part, customizing only some of the parameters to match their specific requirements.

   The country-specific default settings of various parameters that will be loaded on changing the 'Region' are presented in the table below. For default values of Trunk Access Codes, Configuring Emergency Number Dialing, Distinctive Rings, for various countries refer the respective topics.

Country Code	Country Name	Default Time Zone	Default DST Mode	Default DST Schedule Type	CPTG[a]	Default DKP Language	Distinctive Ring[b]	Opr	TAC	Abbr. Dialing
001	Afghanistan	GMT+04:30				English				
002	Algeria	GMT+01:00				English				
003	Antigua and Barbuda	GMT-04:00				English				
004	Argentina	GMT-03:00			04	Spanish				
005	Australia (Perth)	GMT+08:00	Scheduled	2	05	English		9	0	

Country Code	Country Name	Default Time Zone	Default DST Mode	Default DST Schedule Type	CPTG[a]	Default DKP Language	Distinctive Ring[b]	Opr	TAC	Abbr. Dialing
006	Australia (Note2) (Adelaide)	GMT+09:30	Scheduled	2	05	English		9	0	
007	Australia (Brisbane, Canberra, Melbourne, Sydney)	GMT+10:00			05	English		9	0	
008	Austria	GMT+01:00	Scheduled	1		German		9	0	
009	Bahamas	GMT-05:00				English		9	0	
010	Bahrain	GMT+03:00	Scheduled	3		English		9	0	
011	Bangladesh	GMT+06:00				English				
012	Belarus	GMT+02:00				English				
013	Belgium	GMT+01:00	Scheduled	2		French		0	0	
014	Bhutan	GMT+06:00				English				
015	Bolivia	GMT-04:00				Spanish				
016	Bosnia and Herzegovina	GMT+01:00				English				
017	Botswana	GMT+02:00				English				
018	Brunei	GMT+08:00				English				
019	Brazil (Fernando De Noronha)	GMT-02:00			06	Portuguese		0	9	
020	Brazil (Brasilia, Rio de Janeiro, Sao Paulo)	GMT-03:00	Scheduled	4	06	Portuguese		0	9	
021	Brazil (Manaus)	GMT-04:00			06	Portuguese		0	9	
022	Brazil (Acre)	GMT-05:00			06	Portuguese		0	9	
023	Bulgaria	GMT+02:00				English				
024	Cambodia	GMT+07:00				English				
025	Cameroon	GMT+01:00				English				
026	Canada (St. John's)	GMT-03:30	Scheduled	5	03	English	T3	0	9	6
027	Canada (Halifax)	GMT-04:00	Scheduled	5	03	English	T3	0	9	6
028	Canada (Montreal, Ottawa, Toronto)	GMT-05:00	Scheduled	5	03	English	T3	0	9	6
029	Canada (Winnipeg)	GMT-06:00	Scheduled	5	03	English	T3	0	9	6
030	Canada (Calgary)	GMT-07:00	Scheduled	5	03	English	T3	0	9	6
031	Canada (Vancouver)	GMT-08:00	Scheduled	5	03	English	T3	0	9	6
032	Chile	GMT-04:00	Scheduled	6		Spanish		0	9	
033	China	GMT+08:00			08	English		0	9	
034	Colombia	GMT-05:00				Spanish				
035	Costa Rica	GMT-06:00				Spanish				
036	Croatia	GMT+01:00				English				
037	Cuba	GMT-05:00	Scheduled	18		Spanish		0	9	
038	Cyprus	GMT+02:00				English				
039	Czech Republic	GMT+01:00				English				
040	Denmark	GMT+01:00	Scheduled	7		English		0	9	
041	Egypt	GMT+02:00	Scheduled	11	09	English		9	0	

Country Code	Country Name	Default Time Zone	Default DST Mode	Default DST Schedule Type	CPTG[a]	Default DKP Language	Distinctive Ring[b]	Opr	TAC	Abbr. Dialing
042	Fiji	GMT+12:00				English				
043	Finland	GMT+02:00	Scheduled	8		English		9	0	
044	France	GMT+01:00	Scheduled	2	10	French		9	0	
045	Germany	GMT+01:00	Scheduled	2	11	German		9	0	
046	Greece	GMT+02:00	Scheduled	2	12	English		9	0	
047	Guyana	GMT-04:00				English				
048	Hong Kong	GMT+08:00				English		9	0	
049	Hungary	GMT+02:00	Scheduled	2		English		9	0	
050	India	GMT+05:30			01	English	T1	9	0	8
051	Indonesia	GMT+07:00			14	English		0	0	
052	Iran	GMT+03:30			15	English		9	0	
053	Iraq	GMT+03:00	Scheduled	9	16	English		9	0	
054	Ireland	GMT	Scheduled	7		English		0	9	
055	Israel	GMT+02:00			17	English				
056	Italy	GMT+01:00	Scheduled	2	18	Italian		9	0	
057	Japan	GMT+09:00			19	English				
058	Jordan	GMT+02:00				English	T1	0	9	
059	Kazakhstan	GMT+06:00				English				
060	Kenya	GMT+03:00			20	English				
061	Korea - North	GMT+09:00			21	English				
062	Korea - South	GMT+09:00				English				
063	Kuwait	GMT+03:00				English				
064	Kyrgyzstan	GMT+06:00	Scheduled	10		English		9	0	
065	Lebanon	GMT+02:00	Scheduled	12		English		9	0	
066	Libya	GMT+02:00				English				
067	Malaysia (Note1)	GMT+08:00			22	English		0	9	6
068	Maldives	GMT+05:00				English				
069	Mauritius	GMT+04:00				English				
070	Mexico (Mexico City)	GMT-06:00	Scheduled	3	03	Spanish	T3	0	9	6
071	Mexico (Chihuahua)	GMT-07:00	Scheduled	3	03	Spanish	T3	0	9	6
072	Mexico (Tijuana)	GMT-08:00	Scheduled	3	03	Spanish	T3	0	9	6
073	Mongolia	GMT+08:00				English				
074	Mozambique	GMT+02:00				Portuguese				
075	Myanmar	GMT+06:30				English				
076	Namibia	GMT+01:00	Scheduled	13	03	English	T3	9	0	6
077	Nepal	GMT+05:45				English				
078	Netherlands	GMT+01:00				English				
079	New Zealand	GMT+12:00	Scheduled	14	24	English		0	1	
080	Nigeria	GMT+01:00				English				
081	Norway	GMT+01:00	Scheduled	15		English		9	0	
082	Oman	GMT+04:00				English				
083	Pakistan	GMT+05:00				English				
084	Paraguay	GMT-04:00	Scheduled	16		Spanish		9	0	
085	Peru	GMT-05:00				Spanish				
086	Philippines	GMT+08:00			25	English				
087	Poland	GMT+01:00	Scheduled	1	26	English		9	0	
088	Portugal	GMT	Scheduled	7	27	Portuguese		9	0	
089	Qatar	GMT+03:00				English				
090	Romania	GMT+02:00				English				

Country Code	Country Name	Default Time Zone	Default DST Mode	Default DST Schedule Type	CPTG[a]	Default DKP Language	Distinctive Ring[b]	Opr	TAC	Abbr. Dialing
091	Russia (Moscow, St. Petersburg)	GMT+03:00	Scheduled	1	28	English		9	0	
092	Russia (Novosibirsk)	GMT+06:00	Scheduled	1	28	English		9	0	
093	Russia (Vladivostok)	GMT+10:00	Scheduled	1	28	English		9	0	
094	Singapore	GMT+08:00			30	English		9	0	
095	Slovakia	GMT+01:00				English				
096	South Africa	GMT+02:00			31	English				
097	Spain	GMT+01:00	Scheduled	1	32	Spanish		9	0	
098	Sri Lanka	GMT+05:30				English				
099	Sudan	GMT+03:00				English				
100	Sweden	GMT+01:00	Scheduled	2		English		9	0	
101	Switzerland	GMT+01:00	Scheduled	2		German		9	0	
102	Syria	GMT+02:00	Scheduled	17		English		9	0	
103	Taiwan	GMT+08:00				English		0	0	
104	Tajikistan	GMT+05:00				English				
105	Thailand	GMT+07:00			33	English				
106	Turkey	GMT+02:00			34	English		9	0	
107	Uganda	GMT+03:00				English				
108	Ukraine	GMT+02:00				English				
109	United Arab Emirates	GMT+04:00			35	English		9	0	
110	United Kingdom	GMT	Scheduled	7	02	English	T2	0	9	8
111	United States (Atlanta, Augusta, Boston, Charlotte, Columbus, Detroit, Indiapolis, Miami, NY, Philadelphia, Washington)	GMT-05:00	Scheduled	3	03	English	T3	0	9	6
112	United States (Chicago, Dallas, Des Moines, Memphis, Minneapolis, New Orleans, Oklahoma, Omaha, St. Louis)	GMT-06:00	Scheduled	3	03	English	T3	0	9	6
113	United States (Albuquerque, Boise, Cheyenne, Denver, Salt Lake City)	GMT-07:00	Scheduled	3	03	English	T3	0	9	6
114	United States (Las Vegas, Los Angeles, Phoenix, San Francisco, Seattle)	GMT-08:00	Scheduled	3	03	English	T3	0	9	6

Country Code	Country Name	Default Time Zone	Default DST Mode	Default DST Schedule Type	CPTG[a]	Default DKP Language	Distinctive Ring[b]	Opr	TAC	Abbr. Dialing
115	United States (Juneau)	GMT-09:00	Scheduled	3	03	English	T3	0	9	6
116	United States (Hawaii)	GMT-10:00			03	English	T3	0	9	6
117	Uzbekistan	GMT+05:00				English				
118	Venezuela	GMT-04:30				Spanish				
119	Vietnam	GMT+07:00				English				
120	Yemen	GMT+03:00				English				
121	Yugoslavia	GMT+02:00				English				
122	Zambia	GMT+02:00				English				
123	Zimbabwe	GMT+02:00				English				
124	Saudi Arabia	GMT +3.00				English	T1	9	0	
125	Cote d'Ivoire	GMT+01:00	Scheduled	2	10	French		9	0	
126	American Samoa	GMT-11:00			03	English	T3	0	1	8
127	Australia (Eucla)	GMT+08:45			05	English	T1	0	9	6
128	Australia (Lord Howe Island)	GMT+10:30			05	English	T1	0	9	6
129	Cape Verde (Cabo Verde)	GMT-01:00			03	English	T3	9	0	8
130	French Polynesia	GMT-09:30			10	French	T1	9	0	8
131	Kiribati	GMT+14:00			05	English	T1	9	0	8
132	New Zealand (Chatham Islands)	GMT+12:45			24	English	T1	0	1	8
133	Samoa	GMT+13:00			05	English	T1	9	0	8
134	Solomon Island	GMT+11:00			02	English	T2	9	0	8
135	US Minor Outlying Islands(Baker Island, Howland Island)	GMT-12:00			03	English	T3	0	9	6

a. See “Call Progress Tones” for more information on CPTG.
b. See “Distinctive Rings” for more information on Ring Type and Cadence.

2. ***Installing the SARVAM UCS in a Hospitality Application.***

The two main applications of the SARVAM UCS are:

- Enterprise application to meet the communication requirements of businesses.
- Hospitality application to meet the specific requirements of Hotels and Hospitals.

In addition to the common set of System features, there is a distinct set of in-built features for each of these applications. When SARVAM UCS is to be installed in any of the two application scenarios, the 'Customer Profile' - whether the user is an Enterprise or a Hotel - is to be defined at the time of installation.

When the Customer Profile is defined, all features specific to the application Enterprise/Hotel, along with their default settings are loaded. By default the Customer Profile of SARVAM UCS is defined as 'Enterprise'.

Refer the *SARVAM UCS Hospitality System Manual* to know more.

3. ***Malfunctioning of the System.***

   When there is a system malfunction, possibly caused by a programming error that you are unable to diagnose, you may restore default settings.

Restoring Default Settings

Default settings can be loaded or restored from the programming mode (software default).

To be able to do this, you must have the Programming Password, also referred to as the "SE password".

Whenever you restore the default settings in the system, all the programmable parameters except Network Port Parameters[356] *and the "Region" will be set back to their default values.*

All parameters will be assigned default values except the following, when the system is set to default:

- SE Password
- SA Password
- Date Format
- CPTG Type
- Companding Algorithm - A-Law / u-Law
- Voice Module
- Clock Synchronization Parameters
  - Source Port
  - Clock Type
  - PLL TIE Control
  - PLL Operating Mode
  - Locking Mode
- System Activity Log
- System Fault Log
- SMDR
- SIM PIN of Mobile
- Network Parameters
- Wizard Parameters

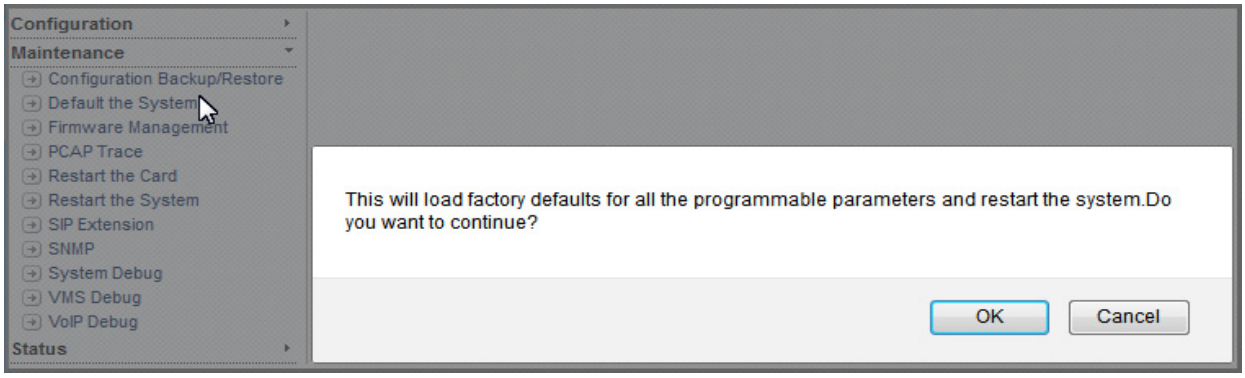
Restoring Default Settings using Jeeves

- Log in as System Engineer.

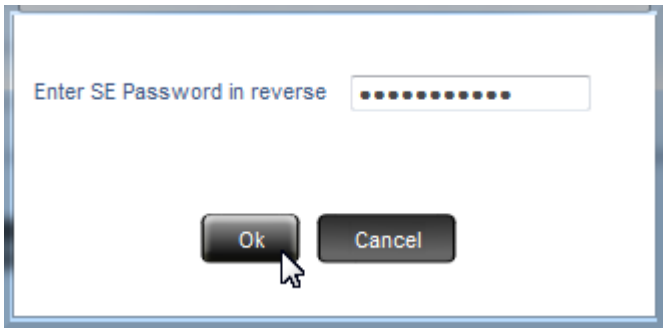
- Under **Maintenance**, click **Default the System**.

356. The IP Address, Subnet Mask, Primary DNS, Secondary DNS, Host Name, Domain Name, DHCP Server Address.

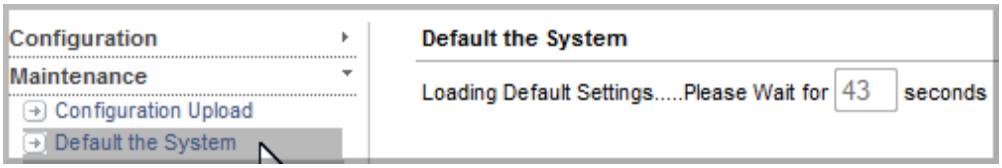
- You will be prompted whether you want to default the system.



- Click **OK**.
- Enter Reverse SE Password on the prompt.



- The SE password you enter must be the current password. For Example: if it is 1234, enter 4321 and click OK.
- The system will restart and load the default settings. It takes 50 seconds to load default settings. The system count down in the seconds will appear on your screen as “Loading Default Settings...Please Wait for ____ seconds.



- You are returned to the Login page of Jeeves.
- You may now enter programming mode again.

Without the SE Password, you cannot restore default values via the software. If you forget the SE Password, you must resort to hardware default of the SE-Programming Password first. Refer the topic “System Security” for instructions on restoring the default SE Password.

Modified default parameter values for Firmwares later than V1R6.7

Page Name	Feature/Parameter	Old Default Value	New Default Value	Impact on behavior after update
Station Basic Feature Template [1-50]	Call Privilege ->Toll Control Level-0 (WH)	All Calls	No Calls	Extension Users will not be able to make external calls.
Station Basic Feature Template [1-50]	Call Privilege ->Toll Control Level-0 (BH)	All Calls	No Calls	Extension Users will not be able to make external calls.
Station Basic Feature Template [1-50]	Call Privilege ->Toll Control Level-0 (NH)	All Calls	No Calls	Extension Users will not be able to make external calls.
Station Basic Feature Template [1-50]	Call Privilege ->Toll Control Level-1	Local Calls	No Calls	Extension Users will not be able to make external calls.
Station Basic Feature Template [1-50]	Call Privilege ->Toll Control Level-2	National Calls	No Calls	Extension Users will not be able to make external calls.
Station Basic Feature Template [1-50]	Call Privilege ->Toll Control Level-3	No Calls	No Calls	Extension Users will not be able to make external calls.
Class of Service [1-20]	Closed User Group (CUG)	Enable	Disable	Extension Users will not be able to access this feature by default.
Class of Service [1-20]	Global Directory Part-1	Enable	Disable	Extension Users will not be able to access this feature by default.
Class of Service [1-20]	Trunk-Trunk Transfer	Enable	Disable	Extension Users will not be able to access this feature by default.
Logical Partition	VoIP-to-VoIP	Enable	Disable VoIP has been split into SIP Trunk and SIP Extension	Users will not be able to make SIP Trunk calls from SIP Extension. **Logical Partition Table will be set to default for all regions respectivelys**
OG Trunk Bundle Groups	OG Trunk Bundle Members	1,2,3,4	0	Extension Users will not be able to make eternal calls from their extensions.
OG Trunk Bundle	Trunk Port -Type	CO, Mobile, BRI, T1E1 and SIP Trunk	None	Extension Users will not be able to make eternal calls from their extensions.
SIP Trunk Parameter	Accept anonymous calls?	Enable	Disable	Anonymous Calls will not be accepted.
Mobile Trunk Parameter	Accept Anonymous Call	Enable	Disable	Anonymous Calls will not be accepted.
Call Duration Control for Table-1	Apply CDC for incoming calls received from trunk	Disable	Enable	CDC will be applied on All trunks and Extensions for external calls.

Call Duration Control for Table-1	Apply CDC for outgoing calls made from trunk	Disable	Enable	CDC will be applied on All trunks and Extensions for external calls.
Call Duration Control for Table-1	CDC Timer (sec)	160	300	CDC will be applied on All trunks and Extensions for external calls.
Call Duration Control for Table-1	Disconnect Call after CDC Timer	Disable	Enable	CDC will be applied on All trunks and Extensions for external calls.
Call budget	For All Trunk	None	Minutes (300 minutes)	Budget will be applied on all the Trunks.

Firmware Management

What's this?

SARVAM UCS provides you a system to manage the upgradation of the system software with a click of a button.

Due to security concerns the Default Settings have been changed for systems purchased/upgraded with Firmwares later than V1R6.7. For new systems purchased these Default Settings will the applicable automatically, refer to “Modified default parameter values for Firmwares later than V1R6.7”. If you are upgrading the system, refer to “After updating Firmware later than V1R6.7” and “Modified default parameter values for Firmwares later than V1R6.7”.

For upgradation, make sure the internal USB is connected.

The Firmware upgrade package contains — System Firmware as well as the Extended Firmware.

You can upgrade the firmware from the external USB also. In this case, while upgradation if the internal USB is not detected, only the system firmware will get upgraded. The extended firmware will not get upgraded.

The new ETERNITY LENX/MENX/GENX systems (sent from production) have benchmarks as mentioned below and hence cannot be downgraded to version lower than the ones mentioned below:

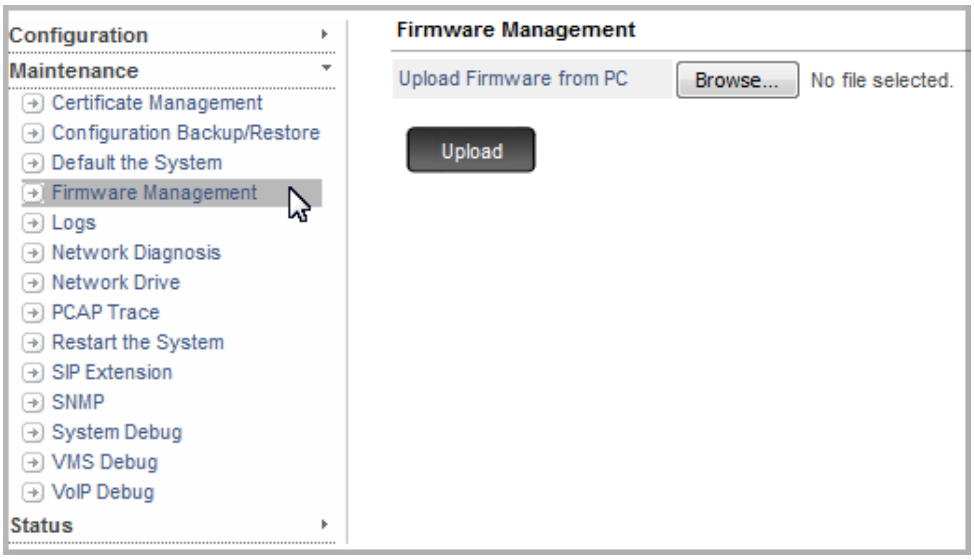
- *Systems with the firmware V1R3 and later cannot be downgraded below V1R3.*
- *Systems with the firmware V1R5.1 and later cannot be downgraded below V1R5.1.*
- *Systems with the firmware V1R6.1 and later cannot be downgraded below V1R6.1.*
- *Systems with the firmware V1R6.3 and later cannot be downgraded below V1R6.3.*
- *If you have system with firmware V1R3 and later or firmware V1R5.1 and later and you upgrade the same with firmware V1R6.1, the system cannot be downgraded again.*

In case you downgrade any of the above with the firmware below the benchmark, the system will reboot and stop functioning. Consult Matrix Technical Support Team for further assistance.

How to Configure

- Log into Jeeves.
- Click the **Maintenance** link.

- Click the **Firmware Management** link.



- In **Upgrade Firmware from PC**, click the **Browse** button to reach the location on the local disk where the firmware files are stored in PC. Make sure that the file is a zip file with extension .zip.
  The table below describes a few possible cases and the corresponding action taken by SARVAM UCS.

Version-Revision of your SARVAM UCS	Manual Selection for Firmware Upgrade	Action Taken by SARVAM UCS
V1R5.1.0	V1R6.1.0	SARVAM UCS will upgrade the firmware. Once upgraded with this version, the system cannot be downgraded to any other version.
	V1R4.1.0	SARVAM UCS will downgrade the firmware.
	V1R1.1.0	SARVAM UCS will downgrade the firmware.
	V1R5.1.0	SARVAM UCS will neither upgrade nor downgrade and an error message will be displayed for the same.
	V1R7.1.0 and benchmark version is V1R6.1.0	SARVAM UCS will not upgrade the Firmware and display error message.

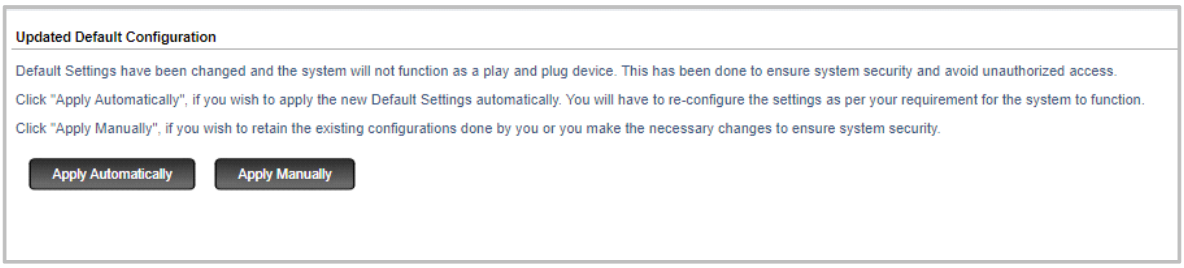
  The lowest benchmark version for SARVAM UCS SME is V1R1.1.0 and for SARVAM UCS ENT is V1R2.1.0.

- After selecting the required firmware zip file from PC, click on the **Upgrade** button.
- After successfully uploading the file, you can select any of the following options:
  - **Restart:** Click this button, if you want to finally update the firmware.Once you click this button, the system restarts and the new firmware is uploaded.
  - **Discard:** Click this option if you do not want to upload the selected firmware.

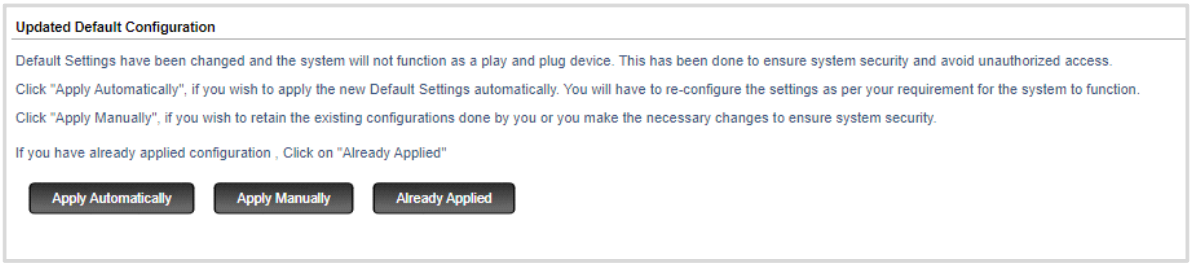
- *If you select a file other than .zip file, an error message is displayed when you click on the Upload button.*
- *Maximum size of the file that can be uploaded is 100 MB.*

After updating Firmware later than V1R6.7

- If you select, Restart, the system restarts and the new firmware is applied
- Re-login into the system. The following screen appears.



- To ensure system security and avoid unauthorized access, the System Default Settings have been changed and the system will not function as a play and plug device.
  - Click **Apply Automatically**, if you wish to apply the new Default Settings automatically. You will have to re-configure the settings as per your requirement for the system to function. Before you click this options make sure you have checked the new default parameters. For details, refer to “Modified default parameter values for Firmwares later than V1R6.7”.
  - Click **Apply Manually**, if you wish to retain the existing configurations done by you and you will have to modify the settings yourself to ensure system security.
- If you click **Apply Automatically**, the system default settings will be changed and system access will be provided.
- If you click **Apply Manually**, system access will be provided. When you re-login the following screen will appear.



  - To avoid re-appearance of the above screen at every re-login, click **Already Applied**. System access will be provided. If you select this option, make sure you have made appropriate changes in the settings to ensure maximum system security and to avoid unauthorized access. Also refer to “System Security” to know more about the login levels and how to set secure passwords.

Matrix will not be responsible for losses/issues arising due to inappropriate configuration.

  - If you click **Apply Manually** again, system access will be provided but the above screen will re-appear at every login.

- If you click **Apply Automatically**, the system default settings will be changed and system access will be provided. Before you click this options make sure you have checked the new default parameters. For details, refer to “Modified default parameter values for Firmwares later than V1R6.7”.

Refer to “Modified default parameter values for Firmwares later than V1R6.7”, to know the changes in default configurations

Logs

What's this?

The SARVAM UCS monitors the boot-up process activities and maintains records of these in the Logs. This is useful source of information for troubleshooting.

You can download these files and use the same for analysis.

Viewing Logs in Jeeves

- Login as System Engineer.
- Under **Maintenance**, click **Logs**.

- Click on the desired file to downloaded.

Auto Update

SARVAM UCS provides you the facility to selectively update the firmware of the cards installed in the system. The system displays the respective card firmware uploaded using "Firmware Management".

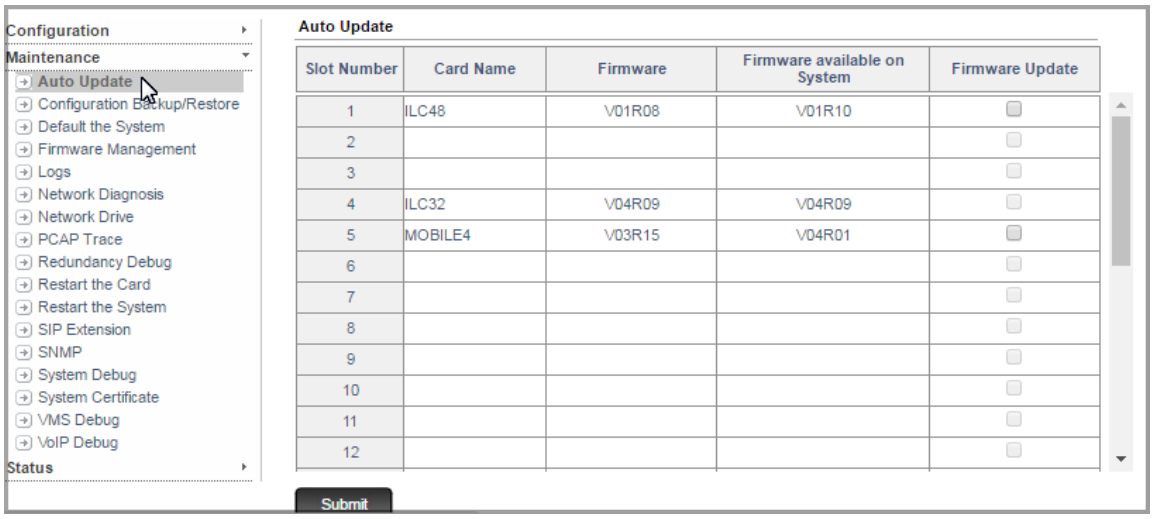
This feature is supported in ETERNITY LENX and ETERNITY MENX only.

To upgrade Data Card Firmware, refer "DATA Card Firmware Upgrade".

If the CO Card is being updated, you will not be able to conduct the AC Impedance Test.

To auto update the firmware of cards,

- Log in as System Engineer.
- Under **Maintenance**, click **Auto Update**.



The following details are displayed:

- **Slot No.:** These are the number of universal slots in the system. These will differ according the variant of SARVAM UCS you have.
- **Card Name:** The type of card installed in the slots. For example, SLT16, SLT8, Switch.
- **Firmware:** The current firmware version and revision of the card, for example V05R10.
- **Firmware Available on the System:** The firmware version and revision of the card available on the SARVAM UCS.
- To update the current firmware of any of the cards that appear on this page, with the firmware available on the system, select the **Firmware Update** check box.

.

For example, in slot 8 the T1E1 Dual Card is installed. The **Firmware** field displays the current version and revision of the card, V05R01 and the **Firmware available on the System** field displays V05R02. If you

select the **Firmware Update** check box of this card, SARVAM UCS will update the firmware of the T1E1 Dual Card to V05R02.

- Click **Submit**. The system starts the updating process.

  The status of the updating process is displayed in the **Firmware** column. After the updating process is completed, the **Firmware Update** check box is unavailable.

*If you have selected the **Firmware Update** check box for multiple cards, SARVAM UCS will update only two cards simultaneously.*

Uploading Custom MoH

If the RTP Mode is set as RTP Relay or Direct RTP, the default MoH is played to the users from the CPU Card.

No special programming is required for using the default MoH. However, if you want to use the customized MoH, you must first:

- record the message
- upload the new recorded message file in the CPU Card.

Recording Voice Messages

When you record MoH of your choice, consider these important points:

- The MoH file is in WAV format, so the customized MoH must also be in the same format and can be of maximum 240 seconds.

- The MoH file has a unique name MoH.wav, make sure the audio file of the custom MoH you have recorded must have the same unique file name as the existing default audio file.

- You can record custom MoH from any external source, and upload the audio file. Refer “Uploading Custom MoH”.

- When you record the MoH from any external source and upload it, make sure that the audio file is recorded in.*wav* file format, with the attributes listed below:
  - Bit Rate: 128 kbps
  - Audio Sample Size: 16 bit
  - Channels: 1 (mono)
  - Audio Sample Rate: 8 KHz
  - Audio Format: PCM

Uploading Custom MoH

You can record a piece of music or a message of maximum 240 seconds duration and assign it to Voice Module 01, which is reserved for Music-on-Hold.

Refer the topic “Voice Message Applications” for instructions on recording and assigning voice modules.

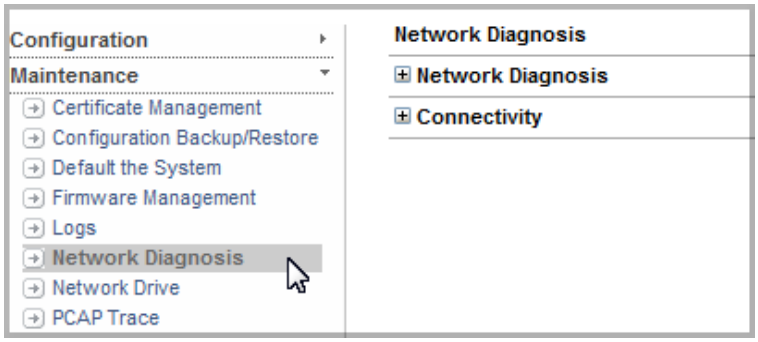
Network Diagnosis

What's this?

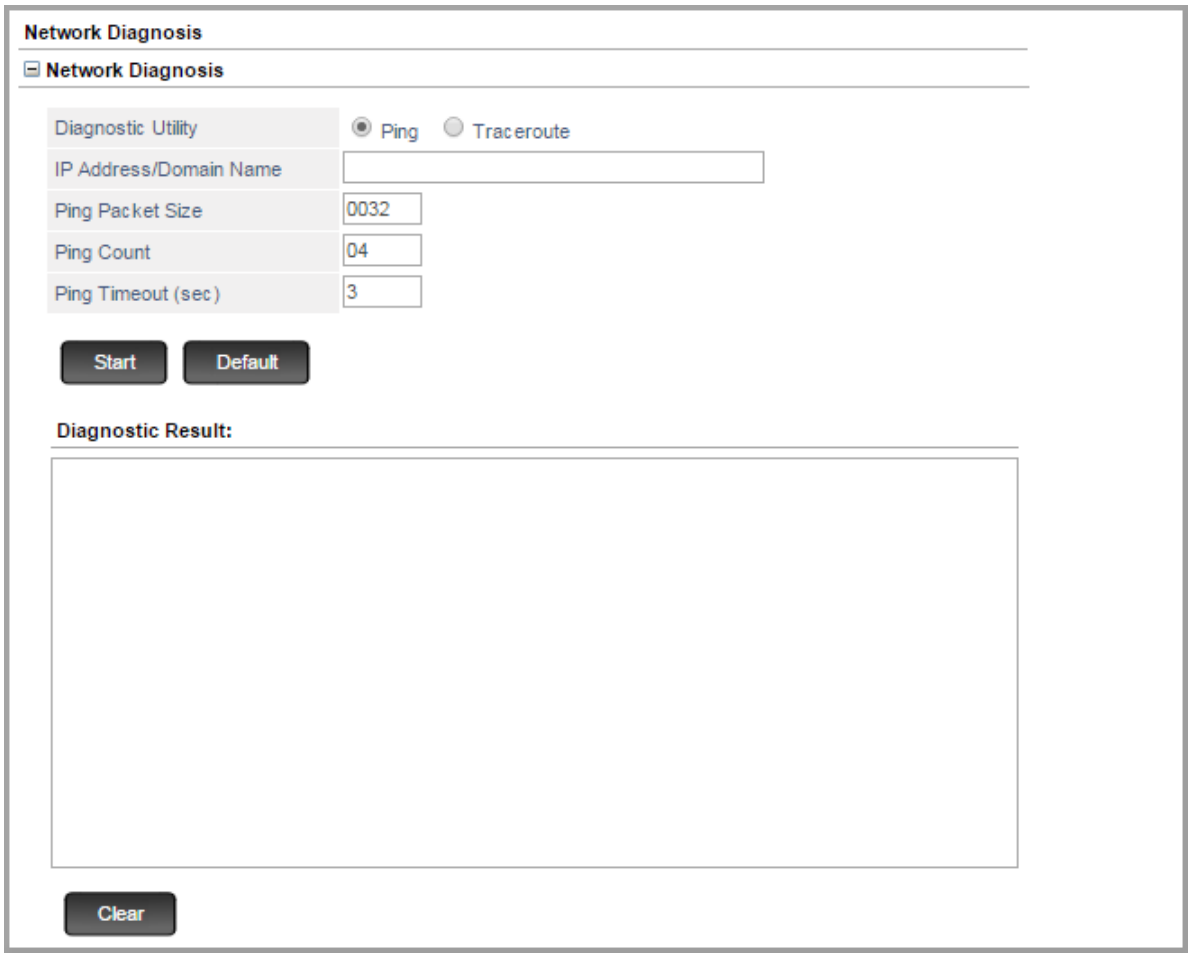
SARVAM UCS provides you an option to check the Internet/WAN connectivity using Ping and Traceroute as the diagnostic tools.

How to Configure

- Log into Jeeves.
- Click the **Maintenance** link.
- Click the **Network Diagnosis** link.



- Click the **Network Diagnosis** link to expand.



- In **Diagnostic Utility**, select the diagnostic tool — Ping or Traceroute — to check the Internet/WAN connectivity.

- In **IP Address/Domain Name**, enter the IPV4 or IPV6 Address or the Domain Name of the system whose connectivity you wish to test. Default: Blank

  If you have selected *Ping* as the *Diagnostic Utility* option, configure the following parameters:

  - In **Ping Packet Size**, enter the number of bytes you want the system to send for Ping test. Valid Range: 4 to 1024. Default: 32 bytes.
  - In **Ping Count**, enter the number of times you want system to send the request message for Ping test. Valid Range: 1 to 50. Default: 4 times.

  - In **Ping Timeout (sec)**, enter the time for which you want the system to wait to get the response for each request message sent. Valid Range: 1 to 9. Default: 3 sec.

  If you have selected *Traceroute* as the *Diagnostic Utility* option, configure the following parameters:

  - In **Traceroute Max TTL**, enter the maximum number of hops (Time-To-Live value) you want the system to take in the path to find the IP Address configured. Valid Range: 1 to 255. Default: 30.

  - In **Traceroute Protocol**, select the protocol — ICMP or UDP — which you want the system to use for traceroute functionality.

- To start the Network Diagnosis, click **Start** button.

  The Diagnostic result will appear on the screen.

- To clear the Diagnostic result, click **Clear** button.

To know about the Connectivity between the third party server and SARVAM UCS system, refer to “APNS Connectivity” and “FCM Connectivity”.

Network Drive Settings

The Network Drive Settings allows you to test the connectivity between the system and the Network Drive where you wish to save the mailbox backup files. The authenticity to access the shared folder is also verified by the Network drive for security purpose.

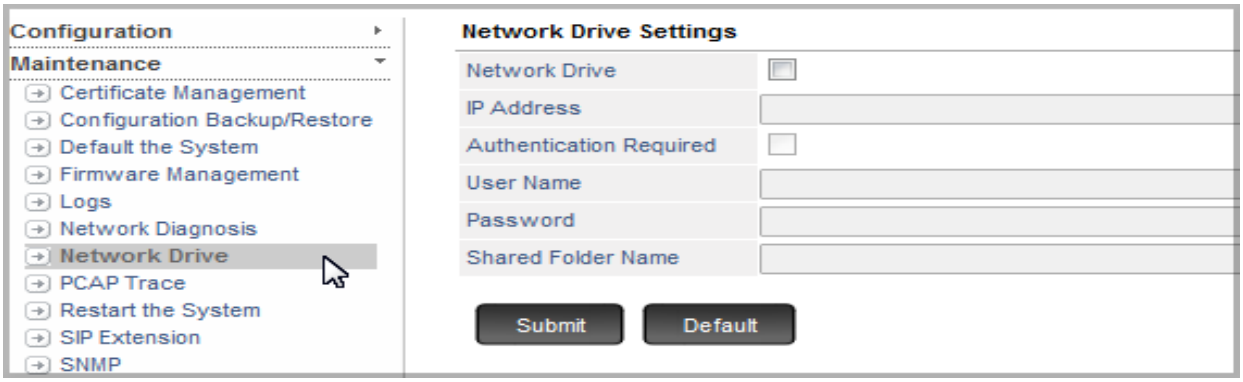
Backup on Network Drive provides you the facility to retrieve valuable information in case internal USB is full.

For information regarding the Voicemail Backup, see “Voicemail Backup”.

Make sure the System and the Network Drive are in the same Local Network.

How to Configure

- Log in as System Engineer.
- Under **Maintenance**, click **Network Drive**.



- Select the **Network Drive** check box to enable the Network Drive.

  Clear the check box to disable the Network Drive.

  By default, it is disabled.

  - In **IP Address**, enter the IP Address of the Network Drive where you wish to store the mailbox backup files.
  - Select the **Authentication Required** check box to enable the authentication process. Enable this check box if the Network Drive folder requires authentication for accessibility.
    - In **User Name**, enter the authentication ID set for the Shared Folder of the Network Drive.
    - In **Password**, enter the authentication password set for the Shared Folder of the Network Drive.
  - In **Shared Folder Name**, enter the name of the Shared Folder where you wish to store the mailbox backup files.
- Click **Submit**.

You may now test the connection between the system and the Network Drive. To do so,

- Click **Test**.

  A message will be displayed showing whether the connection is successful or not. In case of an error, the system will display the error message.

PCAP Trace

What's this?

PCAP or packet capture consists of intercepting and logging the traffic passing over a digital network or a part of a network. PCAP intercepts each packet in the data streams that flow across the network, and can decode and analyze its contents.

PCAP can be used, among others, to monitor the network, detect and analyze network problems, debug client/server communications, debug network protocol implementations.

SARVAM UCS supports PCAP Trace for the LAN Port, WAN Port, Internal Interface as well as RTP packets of the system and the Matrix Extended IP Phones. PCAP Trace is supported for both IPv4 and IPv6 addresses.

The PCAP Trace of the Extended IP Phones can be accessed using the FTP of the phone. The access to the FTP is secured by a password and it can be changed. For detailed instructions, see “FTP Access for Extended IP Phones”.

Packets traveling over a network are captured and saved in the system. You can save these trace files (packets captured by the system) on a PC and open these trace files using a graphical packet capture and protocol analysis tool such as Wireshark or Ethereal.

SARVAM UCS also supports Filters and 'Promiscuous' mode for capturing packets, which you can use to specify the types of data packets to be captured.

How to use

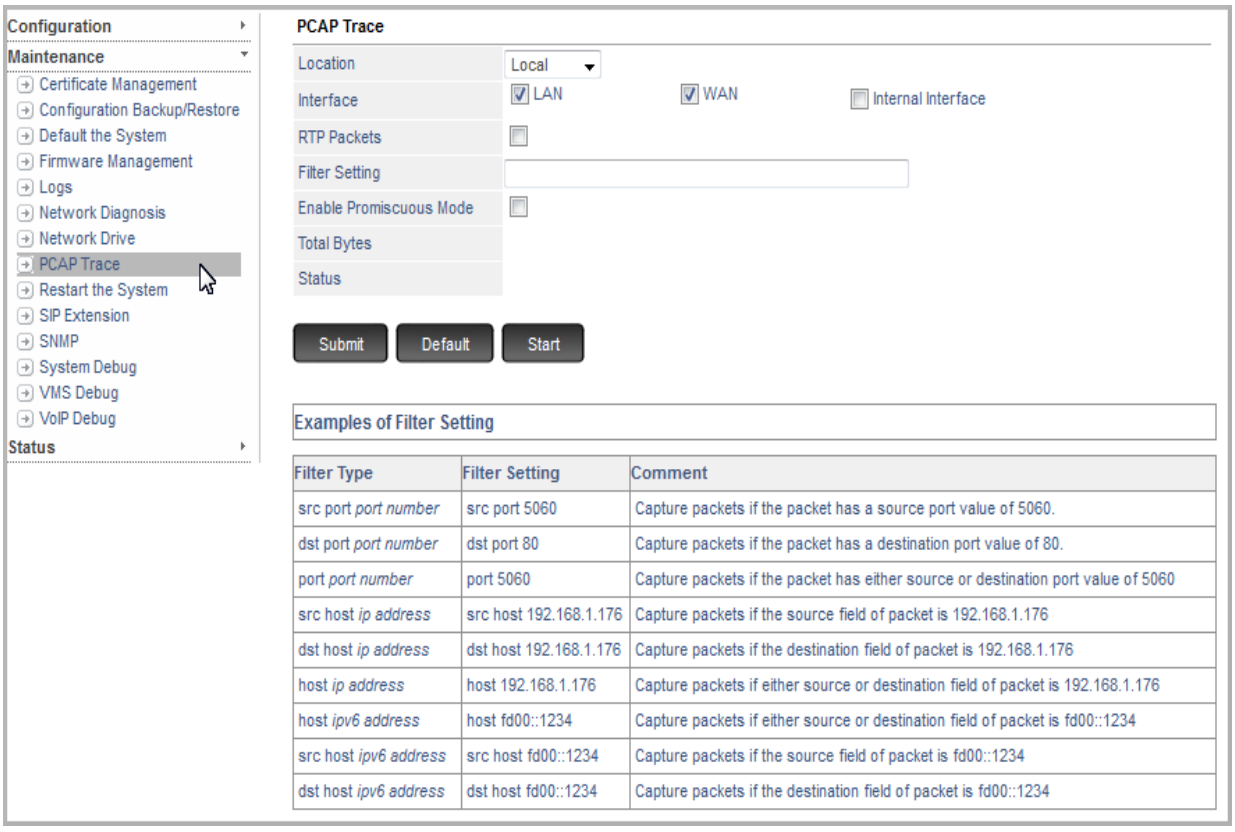
Using PCAP Trace of SARVAM UCS

When the PCAP Trace data are stored locally a maximum of 10 MB of packets can be captured and stored in the SARVAM UCS.

To use PCAP Trace of SARVAM UCS,

- Log in to Jeeves as System Engineer.

- Under **Maintenance**, click **PCAP Trace** link to open the page.



- **Location:** Select the location — Local or Remote — on which you want to store the PCAP data.

  If you select **Local** as the Location:

  - **Interface:** Select the interfaces for which you want to capture the packets — LAN, WAN, Internal. Select the check box of the desired interfaces.

  - **RTP Packets:** If you want to capture RTP packets, select the **RTP** check box.

IPv4 RTP Packets only will be capture.

  - **Filter Settings**: Decide the type of packets to be captured and set the Filter accordingly. The Filter Settings parameter should be maximum 60 characters in length; all ASCII characters are allowed. By default, this field is blank. So all packets will be captured.

    Refer the following examples to know how to set the Filters.

    Examples of Filter settings:

    - To capture only SNMP traces:
      Filter Settings = port 161
      Where, 161 is the SNMP Port number for which the traces are to be captured.

    - To capture packets which are transmitted from the system, *from* IP address 192.168.1.191:
      Filter Settings = src 192.168.1.191

    - To capture packets which are received for the system, *to* IP address 92.168.1.191:

Filter Settings = dst 192.168.1.191

- To capture only packets which are transmitted from the system and received to the system, IP address 192.168.1.191:
  Filter Settings = src 192.168.1.191 or dst 192.168.1.191

- To capture packets which are transmitted from the system for particular port number only, *from* IP address 192.168.1.191 and port number 161
  Filter Settings = src 192.168.1.191 and port 161

If you do not enter a valid filter, you will get the message: *'Invalid filter! Please enter valid filter'*.

It is not mandatory to set Filters. When the Filter Settings field is left blank, the system will capture all packets.

- You can set the **Enable Promiscuous Mode?** to **Yes**, if you want.

  When you enable Promiscuous mode, the SARVAM UCS will capture all network traffic. However, this will work only in a non-switched environment.

  When Promiscuous Mode flag is disabled, the system will capture only traffic that is directly related to it. Only traffic to, from or routed through the SARVAM UCS will be picked up by the PCAP Trace.

'Filter Settings' and 'Promiscuous Mode' (enabled) will not be cleared during power down.

- Click the **Start** button to begin the capturing of the packets.

- Click the **Stop** button to stop packet capture.

OR

Wait for the system to stop packet capturing. The system stops packet capturing once the maximum allotted memory of 10 MB (RAM) is utilized.

- **Total Bytes:** Bytes captured as per the filter setting will be displayed as **Total Bytes**.

Capturing of packets will not stop if you open any other page of Jeeves. So, you may continue using Jeeves for any other purpose while PCAP Trace is being used.

- When the packet capturing is stopped (by you or the system), click the **Save Trace File** button to save the files on the PC.

The current packets captured will not be deleted after you have saved the trace file. The current packets will be deleted when you start the PCAP capture again.

- Now, you can open the downloaded trace file using Wireshark or Ethereal or any other similar software which supports opening of trace files.

If you select **Remote** as the Location:

Configure the following:

- **Listening Interface:** Select WAN or LAN as per your requirement.

- **Listening Port:** Enter the port number using which you want SARVAM UCS to send the packets to the remote device. Make sure this port number is also configured in the remote device.

- **Remote Device IP Address:** This is IP address of the Remote Device on which SARVAM UCS will send the PCAP data.

  The PCAP packets will be captured on the remote device as per the filters set in Wireshark or Ethereal or any other similar software at the remote end.

- Click the **Start** button to begin the capturing of the packets.

- Click the **Stop** button to stop packet capture.

If you are not changing any configuration related to the remote location, we recommend you not to press the Start-Stop-Start buttons frequently within a 1 minute, as it may lead to improper functioning.

Using PCAP Trace for Matrix SPARSH VP248 Extended IP Phone

PCAP Trace supported by the Matrix Extended IP Phone can capture up to 1 MB of packets and store them in the embedded FTP server of the phone. The access to the FTP is secured by a password, for details see “FTP Access for Extended IP Phones”.

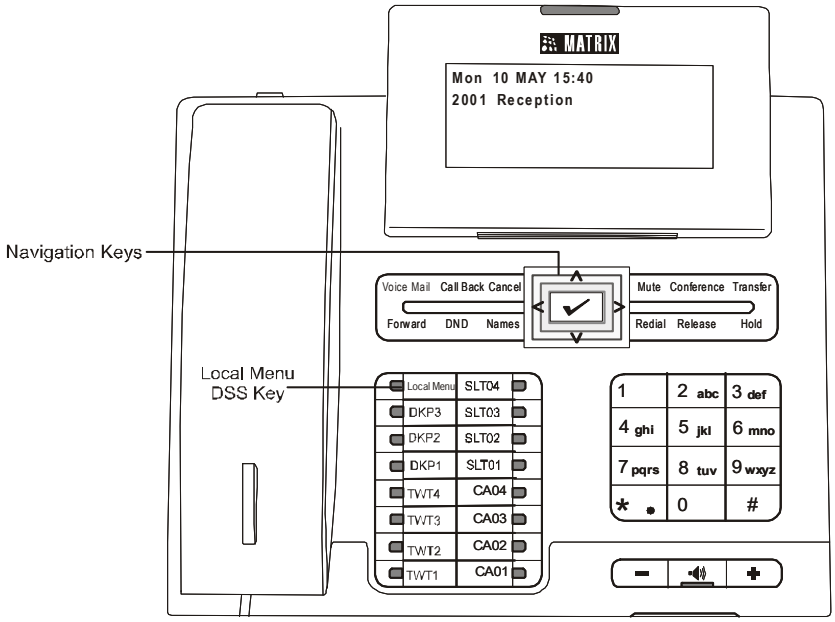
You are not required to set any filters; the phone captures all packets that it sends and receives.

To be able to use PCAP Trace for a Matrix Extended IP Phone, there must be a DSS Key for accessing the Local Menu on the phone. If you have not already configured the DSS Key for Local Menu, you may do so now. For instructions on configuring DSS keys of the Extended IP Phone, see “Configuring Matrix SPARSH VP248”.

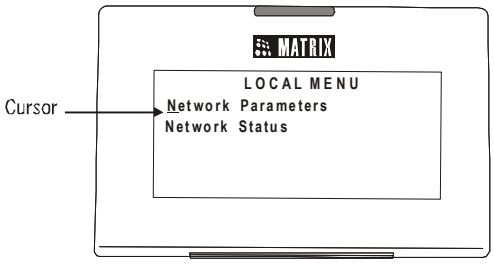
To use PCAP Trace from an Extended IP Phone extension,

- Press the DSS Key assigned to ‘Local Menu’.

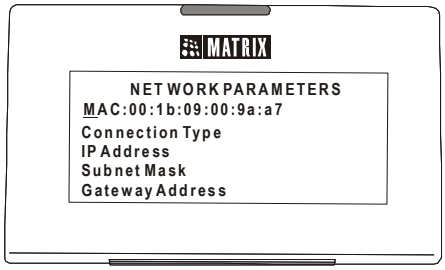
You can access the Local Menu only when your phone is in idle state.



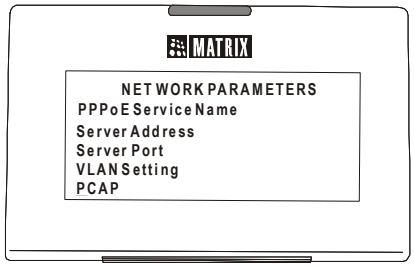
- The Local Menu appears on your phone display.



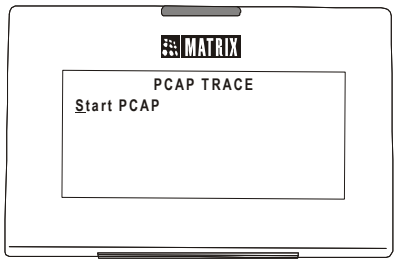
- The cursor appears under ‘Network Parameters’.
- Press Enter key to select ‘Network Parameters’.
- The Network Parameters submenu appears.



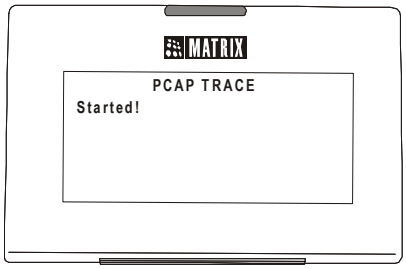
- Scroll with the DOWN navigation key to PCAP.



- Press Enter key.
- ‘Start PCAP’ appears on your phone display.

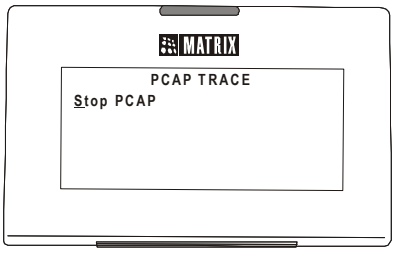


- Press Enter key. You get a confirmation message ‘Started!’

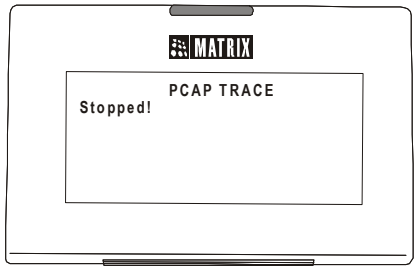


To stop PCAP,

- Enter the Local Menu of the phone again by pressing the DSS key.
- Enter the Network Parameters submenu.
- Select PCAP and press Enter.
- The message Stop PCAP appears on your phone display.



- Press Enter key. You get the confirmation message: ‘Stopped!’



- Go idle.

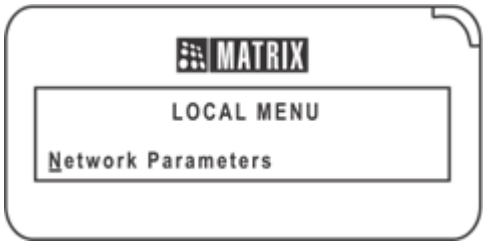
Using PCAP Trace for Matrix SPARSH VP310 Matrix Extended IP Phone

PCAP Trace supported by the Matrix Extended IP Phone can capture up to 1 MB of packets and store them in the embedded FTP server of the phone.

You are not required to set any filters; the phone captures all packets that it sends and receives.

To use PCAP Trace,

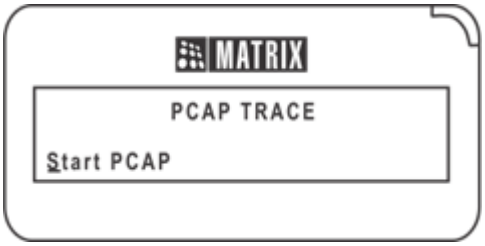
- When the phone is in idle state, press the DOWN key ⋁ to access the Network Settings.
- The cursor appears under ‘Network Parameters’.



- Press Enter key to select ‘Network Parameters’.
- The Network Parameters submenu appears.
- Scroll with the DOWN navigation key to PCAP.

- Press Enter key.

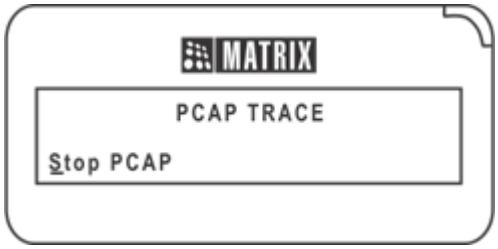
- ‘Start PCAP’ appears on your phone display.



- Press Enter key. You get a confirmation message ‘Started!’

To stop PCAP,

- Press the DOWN key ⋁ to access the Network Settings.
- The cursor appears under ‘Network Parameters’.
- Press Enter key to select ‘Network Parameters’.
- The Network Parameters submenu appears.
- Scroll with the DOWN navigation key to PCAP and press Enter key.
- The message Stop PCAP appears on your phone display.



- Press Enter key. You get the confirmation message: ‘Stopped!’

- Go idle.

You can download the Trace file from the embedded FTP server of the Extended IP Phone. To access the FTP server using Windows FTP, do the following:

- Go to **My Computer**.
- Type the current IP Address of the Extended IP Phone in the Address bar. For example: **ftp:// 192.168.201.134**
- Click **Go** or Press Enter key on your keyboard**.**
- The **Log on as** window of the FTP server opens.
- In **User Name**, type **se** (lower case).
- In **Password**, enter the FTP Password for the phone.
- Click **Log on**.
- On successful login, the FTP window will open. You will see the different Configuration folders in this window.
- Click the folder **ramdisk**.
- In the **ramdisk** folder, right click the file **trace.pcap** and copy it on to your local disk.
- Open the **trace.pcap file** using Wireshark or Ethereal or any other similar software which supports opening of trace files.

*You may also use FireFTP, if you are using Mozilla Fire Fox. Make sure your browser has the **FireFTP** Add-on installed.*

To download the Trace file for SPARSH VP330, SPARSH VP210 and SPARSH VP510, refer to their respective User Guides.

Redundancy Debug

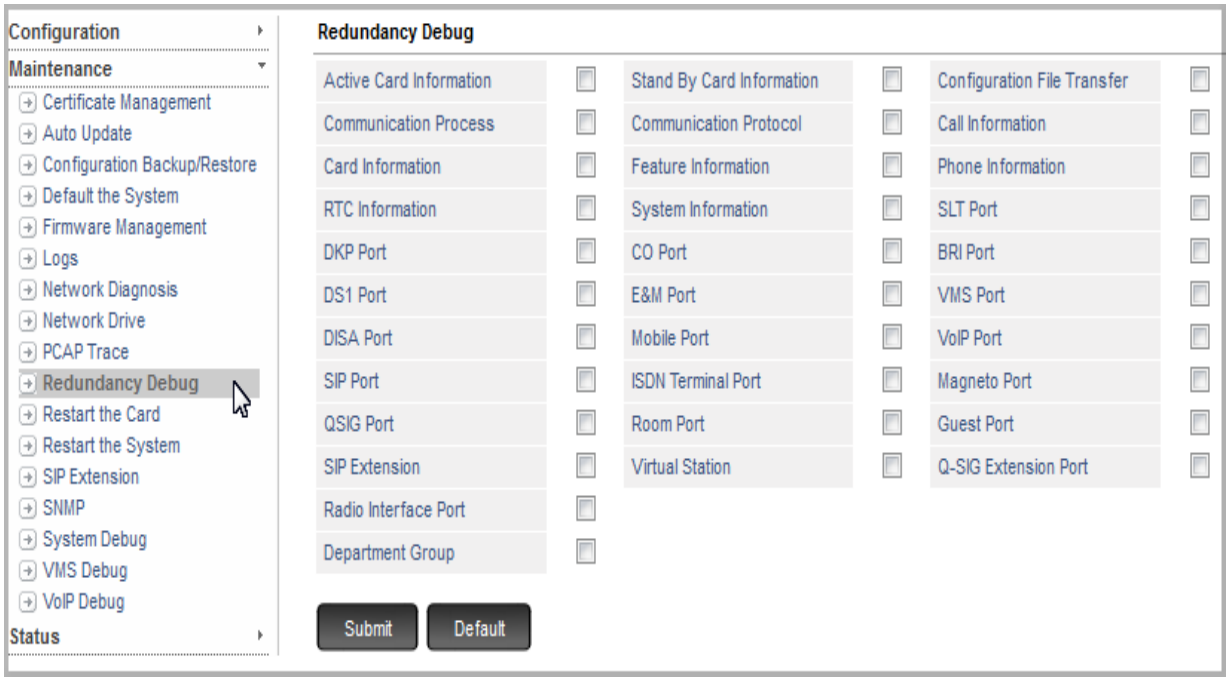
Redundancy Debug is applicable only for ETERNITY LENX/MENX system.

You can monitor the state of — Active and Stand by Card information, Feature Information, Call Information, the states of the various software ports, Department Group etc. — being transfered from the Active Card to the Stand by Card using Redundancy Debug.

SARVAM UCS supports Syslog Client for debugging. Debug messages are sent to the remote Syslog Server.

Configuring Redundancy Debug using Jeeves

- Log in as System Engineer.
- Under **Maintenance**, click **Redundancy Debug**.



- Select the respective check box of the events/processes you want to debug:
  - **Active Card Information**
  - **Stand By Card Information**
  - **Configuration File Transfer**
  - **Communication Process**
  - **Communication Protocol**
  - **Call Information**
  - **Card Information**
  - **Feature Information**
  - **Phone Information**
  - **RTC Information**
  - **System Information**
  - **SLT Port**

- **DKP Port**
- **CO Port**
- **BRI Port**
- **DS1 Port**
- **E&M Port**
- **VMS Port**
- **DISA Port**
- **Mobile Port**
- **VoIP Port**
- **SIP Port**
- **ISDN Terminal Port**
- **Magneto Port**
- **QSIG Port**
- **Room Port**
- **Guest Port**
- **SIP Extension**
- **Virtual Station**
- **Q-SIG Extension Port**
- **Radio Interface Port**
- **Department Group**

- Click **Submit**.

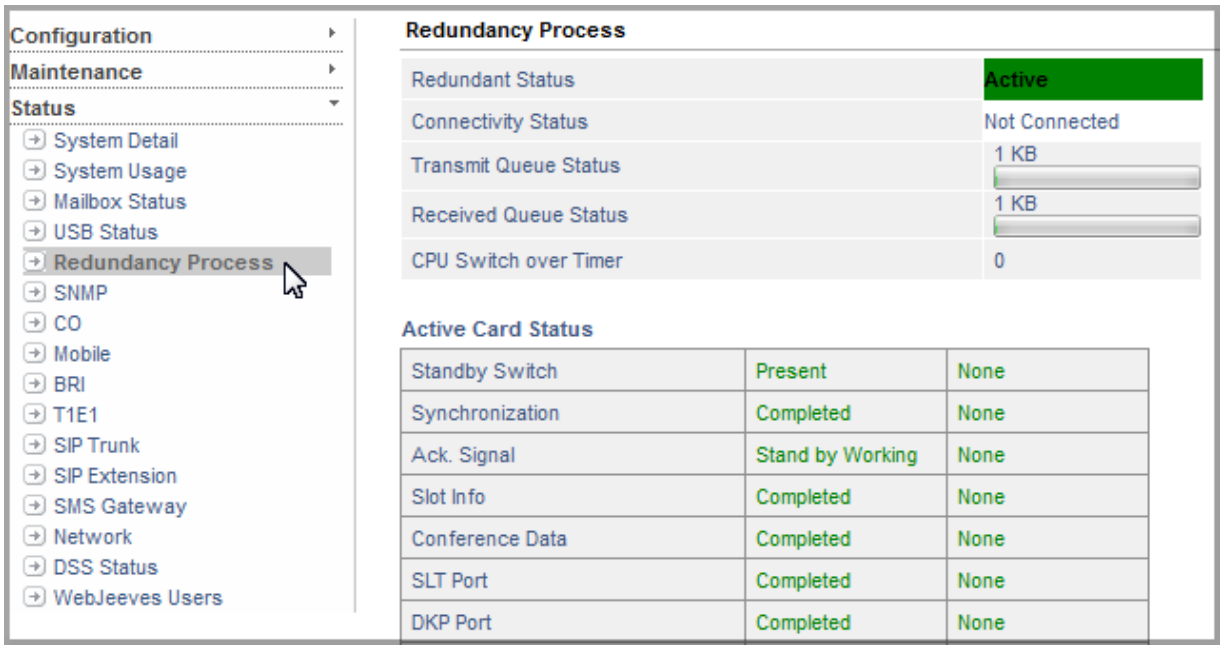
Redundancy Process

Redundancy Process is applicable only for ETERNITY LENX/MENX.

Viewing Redundancy Process Status

You can view the status of the Redundancy Process in Jeeves. To do so,

- Under **Status**, click **Redundancy Process**.



The following details are displayed:

- **Redundant Status** displays the status of the CPU Cards — Active or Standby.

  If the CPU Card is **Active**, the following details of the same will be displayed:

  - The **Connection Status** will display the connectivity status of the CPU Card — Connected or Not Connected.
  - **Transmit Queue Status** displays the status of the data that is pending to be transmitted. The Redundancy is executed properly if there is no pending data left to be transmitted.
  - **Received Queue Status** displays the status of the received data that is still remaining to be read and executed. The Redundancy is executed properly if all the data is received.
  - **CPU Switch over Timer** displays the time left (in seconds) for the CPU Card to completely switch from the Stand by mode to the Active mode and vice-versa. The time left will be calculated as per the time you configured for the **Set Active CPU Card to Stand-by after (Hrs.)** in the “System Parameters”.
  - **Active Card Status** displays the status of the data (Configuration and Call Information) being transfered from the Active Card to the Standby Card.

Make sure that both the active and standby CPU Cards have the same firmware installed. Only then all the details on this page will be displayed.

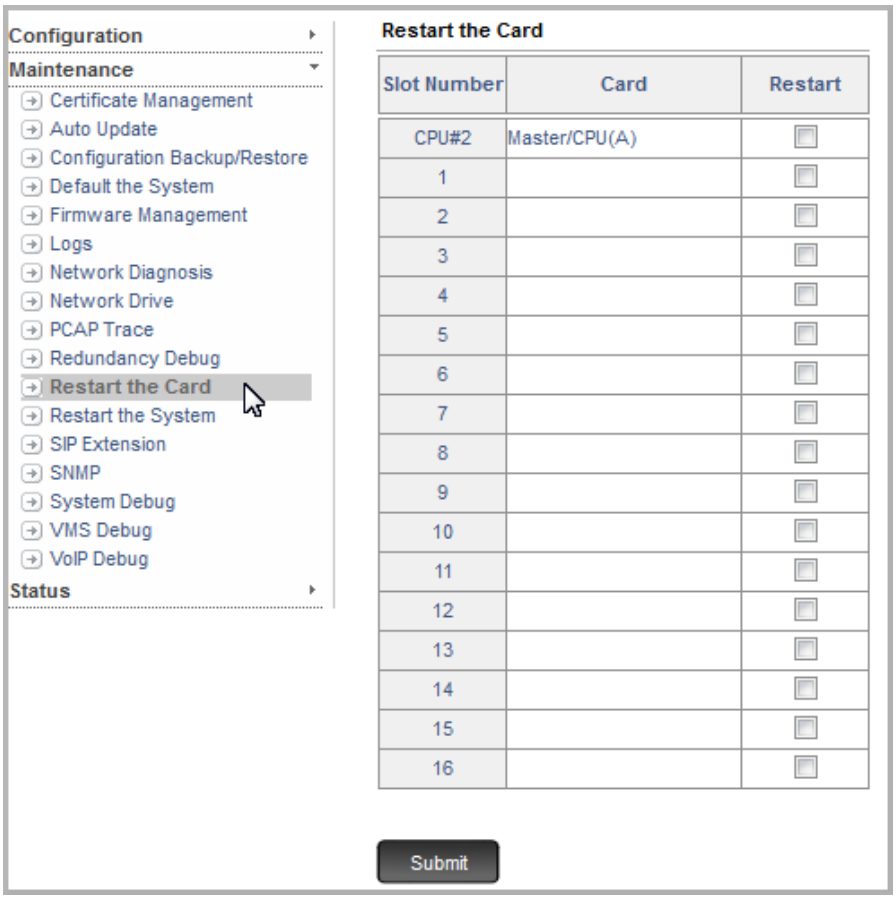
Restart the Card

Applicable only for ETERNITY LENX and MENX.

You can restart any of the cards installed in the system.

To do this,

- Log in as System Engineer.
- Under **Maintenance**, click **Restart the Card** to open the page.
- The page displays the Slot Numbers and the card installed in the slot.

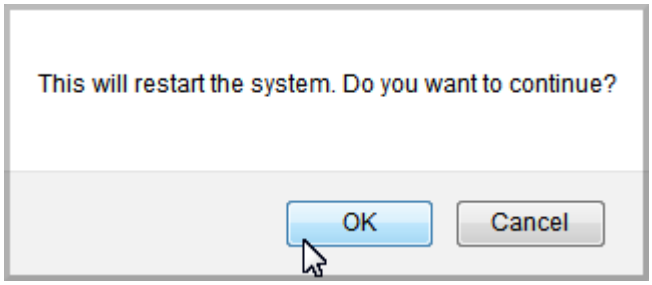


- To restart a card, select the **Restart** check box of the respective card.
- Click **Submit**.
- The system will restart the card.

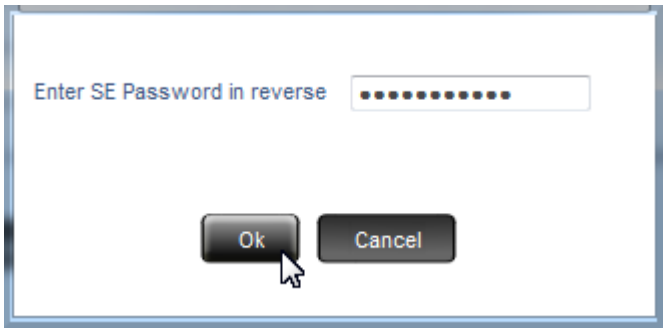
Restart the System

To restart the System,

- Log in as System Engineer.
- Under **Maintenance**, click **Restart the System** to open the page.



- You will get the message, “This will restart the system. Do you want to continue?”.
- Click **OK**.
- Enter Reverse SE Password on the prompt.



- The SE password you enter must be the current password. For Example: if it is Matrix@1234, enter 4321@xirtaM and click OK.
- The system will restart.

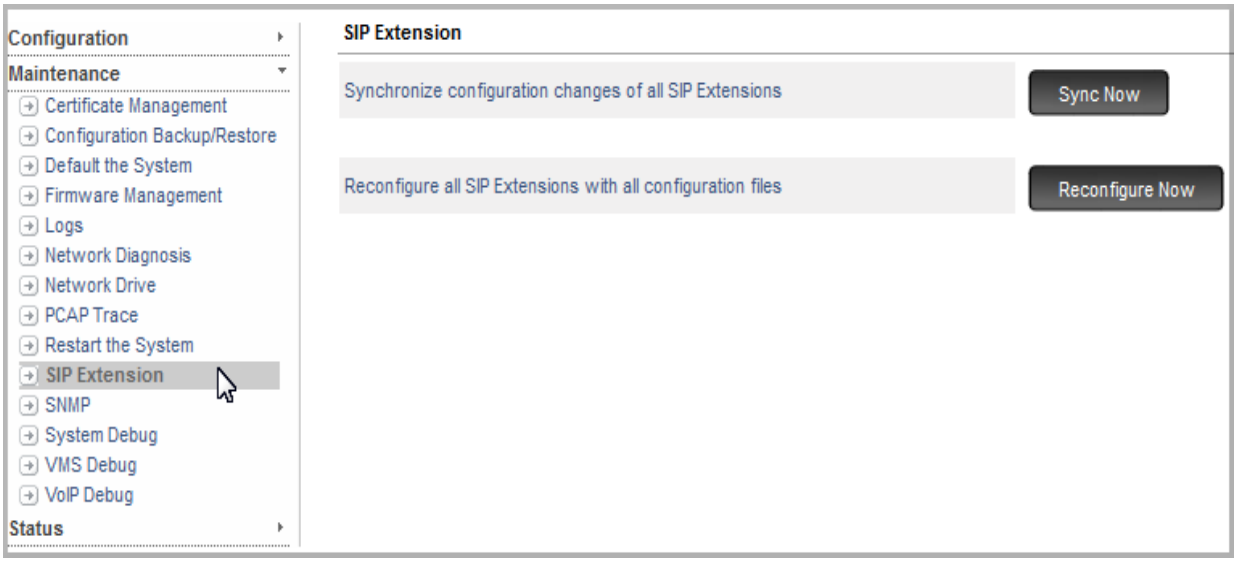
SIP Extension

What's this?

SARVAM UCS provides you an option to synchronize configuration changes of all SIP Extensions and reconfigure all SIP Extensions with all configuration files with a click of a button.

How to Configure

- Log into Jeeves.
- Click the **Maintenance** link.
- Click the **SIP Extension** link.



- In **Synchronize configuration changes of all SIP Extensions**, click the **Sync Now** button to synchronize the configuration of Extended Clients, VP110 and Third Party IP-Phones.

  Only those configuration files in which there are changes will be synchronized.

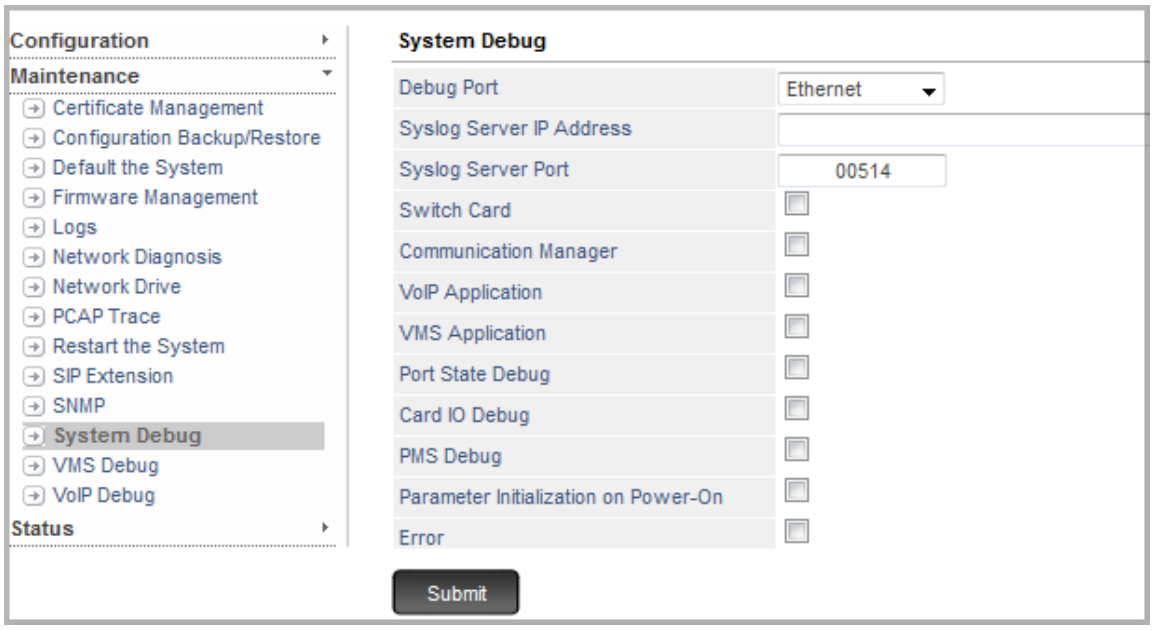
- In **Reconfigure all SIP Extensions with all configuration files**, click the **Reconfigure Now** button to reconfigure all the configuration files of Extended Clients, VP110 and Third Party IP-Phones.

System Debug

You can monitor the state of software ports and IO operations for trouble shooting and identifying faults and errors using System Debug. SARVAM UCS supports Syslog Client for debugging. Debug messages are sent to the remote Syslog Server.

Configuring System Debug using Jeeves

- Log in as System Engineer.
- Under **Maintenance**, click **System Debug**.



Debug Settings

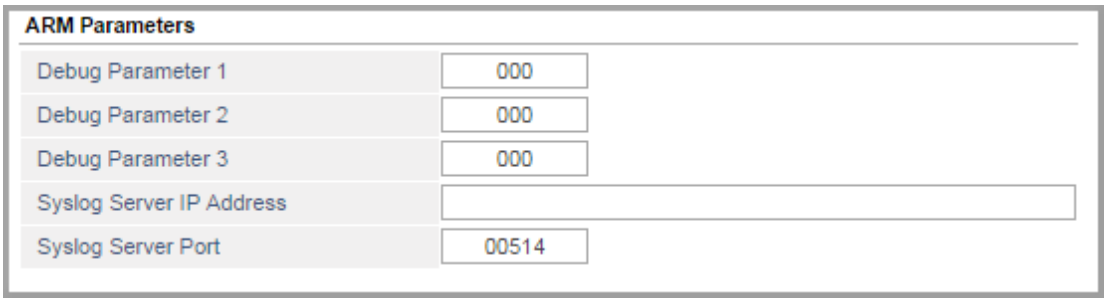
- In **Debug port**, select the destination port for sending debug messages as either **COM Port**, **Ethernet Port** or **USB to COM Port**.

  If you are using the Syslog Server for debugging, select Ethernet, and configure **Syslog Server IP Address** and **Port**. Both IPV4 and IPv6 addresses are supported. Valid port range is: 1025 to 65535;514.

- Now, select the respective check box of the events/processes you want to debug:
  - **Switch Card**
  - **Communication Manager**
  - **VoIP Application**
  - **VMS Application**
  - **Port State Debug**
  - **Card IO Debug**
  - **PMS Debug**
  - **Parameter Initialization on Power-On**
  - **Error**
  - **File Open/Read/Write**

- **ACB Debug**
- **Maturity/AOC Debug**
- **Auto-upgradation/RCOC Debug**
- **WebJeeves**
- **T1E1 Alarm Status**
- **SMS Server Main Application**

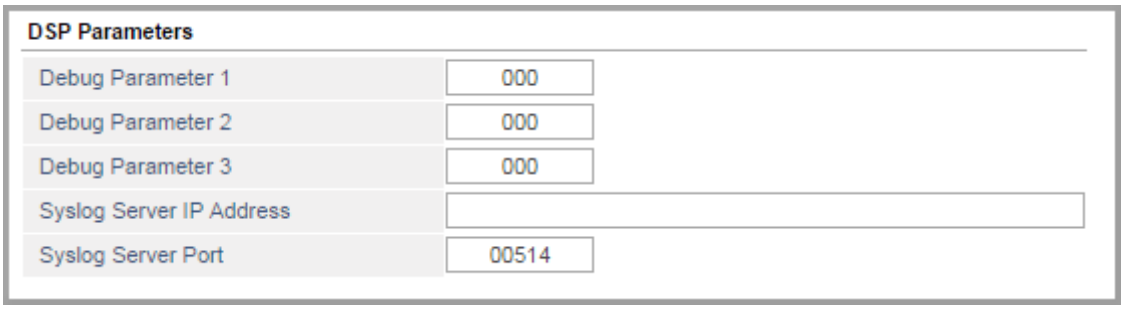
ARM Debug



- In **Syslog Server IP Address,** enter the server address where the debug log is to be sent and in **Port** enter the server port number where the debug log is to be sent. Both IPV4 and IPv6 addresses are supported. Default port number is 514. The valid range of this port is 1025 to 65535; 514.

- Enter the Debug levels in the **Debug Parameter 1, 2 and 3** fields.

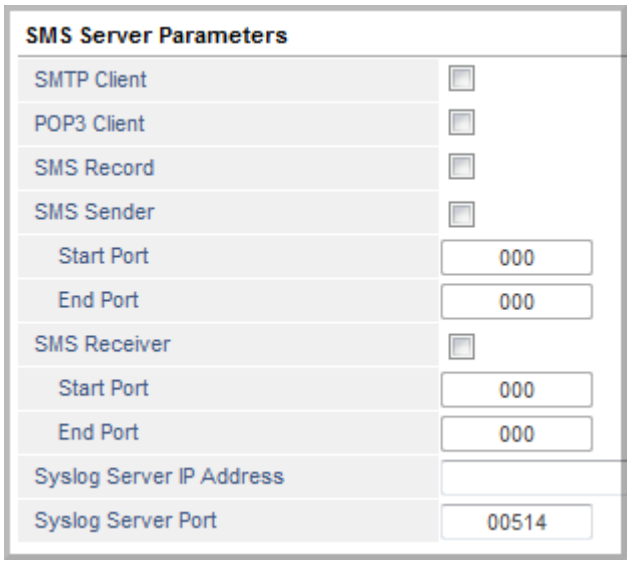
DSP Debug



- In **Syslog Server IP Address,** enter the server address where the debug log is to be sent and in **Port** enter the server port number where the debug log is to be sent. Both IPV4 and IPv6 addresses are supported. Default port number is 514. The valid range of this port is 1025 to 65535; 514.

- Enter the Debug levels in the **Debug Parameter 1, 2 and 3** fields.

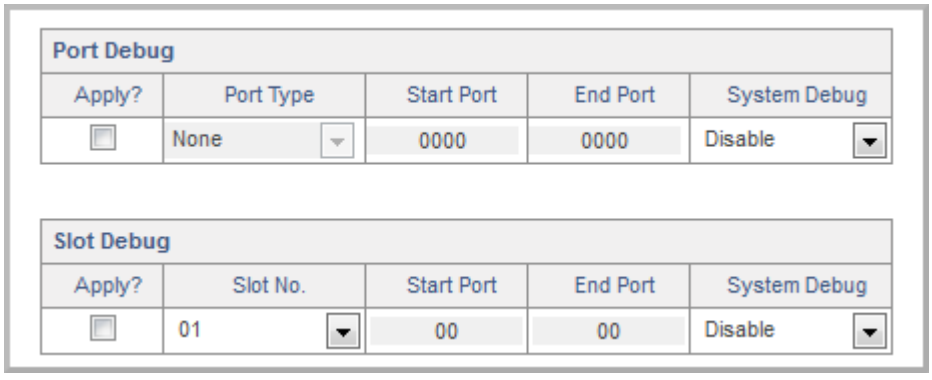
SMS Server Debug



- You can enable the Debug for the SMS Server Parameters. Select the check box of the respective parameter you want to debug:
  - **SMTP Client**
  - **POP3 Client**
  - **SMS Record**
  - **SMS Sender**
  - **SMS Receiver**

  - For the SMS Server debug, configure the **Syslog Server IP Address** and **Port**. Both IPV4 and IPv6 addresses are supported. Valid port range is: 1025 to 65535;514.

Port Debug



- You can also debug a particular Port, by configuring the **Port Debug** settings.
- You can also debug a card installed in a particular slot, by configuring the **Slot Debug** settings.
- Click **Submit**.

Configuring System Debug using Telephone

- Enter SE mode from an extension phone.

To program the IP Address of the Syslog Server (the computer on which the Syslog Server is running), dial:

- ***2178-Syslog Server IP Address***
  Default: 192.168.1.104.

IPv6 address can be configured using Jeeves only.

To program the Port of the Syslog Server, dial:

- ***2179-Syslog Server's Listening Port***
  Listening port can be from 1025 - 65535;514.
  Default: 514

To start/stop debug for required process, dial:

- ***2104-Value-Code***
  Where,
  Code is 0 to Enable and 1 to Disable.
  Value is a 2-digits string for each process type, as in this table:

Process Type	Value
Switch Card	01
Communication Manager Card	02
VoIP Application	03
VMS Application	04
Power Supply Card	05
Port State Debug	06
Card IO Debug	07
PMS Debug	08
Parameter Initialization on Power-On	09
Error	10
File Open/Read/Write	11
ACB Debug	12
Maturity/AOC Debug	13
Auto-upgradation/RCOC Debug	14
WebJeeves	15
T1E1 Alarm Status	16

- This command will be saved in the configuration and not changed even in the case of power failure.

To start/stop state debug, dial:

- ***2105-Port Type-Start Port Number-End Port Number-Flag***

  Where,

Port Type	Meaning
01	SLT
02	DKP
03	CO
04	BRI
05	T1E1
06	E&M
25	MOBILE
26	SIP TRUNK
29	MAGNETO

  Flag is 0 for Enable, 1 for Disable

  This command will not be saved in the configuration. If the system restarts, you need to dial this command again.

To enable SARVAM UCS HOST Debug, dial:

- ***2181-1-Code***

  Where,

  Code is from 000 to 255.

Code	Meaning
001	Enables HOST Debug on Ethernet
002	Enables SARVAM UCS Master Cmd/Event Debug
004	Enable ISDN Relate Host to DSP Cmd/Event Debug
007	Enable all three (above given) debugs
000	Enables DSP Serial Real Time Debug

  Default is 000.

To program IP Address for ARM Debug, dial:

- **2182-IP Address**

  Default: Blank.

To program ARM Syslog Port, dial:

- **2183- Port**

  Where,

  ARM Syslog Port range is from 514, 1025 to 65535.

  Default: 514.

To enable SARVAM UCS DSP Para 1 Debug, dial:

- ***2184-1-Code***
  Where,
  Code is from 000 to 255.

Code	Meaning
000	Enables DSP Serial Real Time Debug
001	Enables DSP Serial Slow Debug
002	Enables DSP Ethernet Debug

Default is 000.

It is not possible to enable all three debugs at a time, only one of the three can be enabled at a time.

To enable SARVAM UCS DSP Para 2 Debug, dial:

- ***2184-2-Code***
  Where,
  Code is from 000 to 255.

Code	Meaning
001	Enable SLT Debug
002	Enable CO Debug
004	Enable DKP Debug
08	Enable ENM Debug
016	Enable ISDN Debug
032	Enable GSM Debug
064	Enable VOIP Debug
128	Enable VMS Debug
255	Enable debug for all the ports listed above
000	Disable debug for all the ports listed above

Default is 000.

To program IP Address for DSP Debug, dial:

- **2185-IP Address**
  Default: Blank.

To program IP Port for DSP Debug, dial:

- **2186- Port**
  Where,
  IP Port range for DSP Debug is from 514, 1025 to 65535.
  Default: 514.

To enable SARVAM UCS PCM Capture Debug, dial:

- **2172-Slot Number-Hardware Port Offset[357]- Code**
  Where,
  Code is
  Slot Number is 01 to 35.
  0 for Rx and Tx PCM Capture Disable
  1 for Rx PCM Capture Enable
  2 for Tx PCM Capture Enable
  3 for Rx and Tx PCM Capture Enable

To initiate the debug of IO operations, dial:

- **2199-Slot Number-1-Port Number[358]-Code**
- **2199-Slot Number-2-Port Number-Port Number-Code**
- **2199-Slot Number-*-Code**
  Where,
  Slot Number is 01 to 12 (GENX)
  Slot Number is 01 to 27 (LENX/MENX)
  Port Number is the Port offset on the card in 2 digit, that is 01 to 32 (GENX)
  Port Number is the Port offset on the card in 2 digit, that is 01 to 64 (LENX/MENX)
  Code is 1 for Enable, 0 for Disable

  If the Slot No. and the Port Number are programmed as 99 the debug of all slots and ports is generated.

- Exit SE mode.

You are advised to backup System Configuration before you upgrade Firmware.

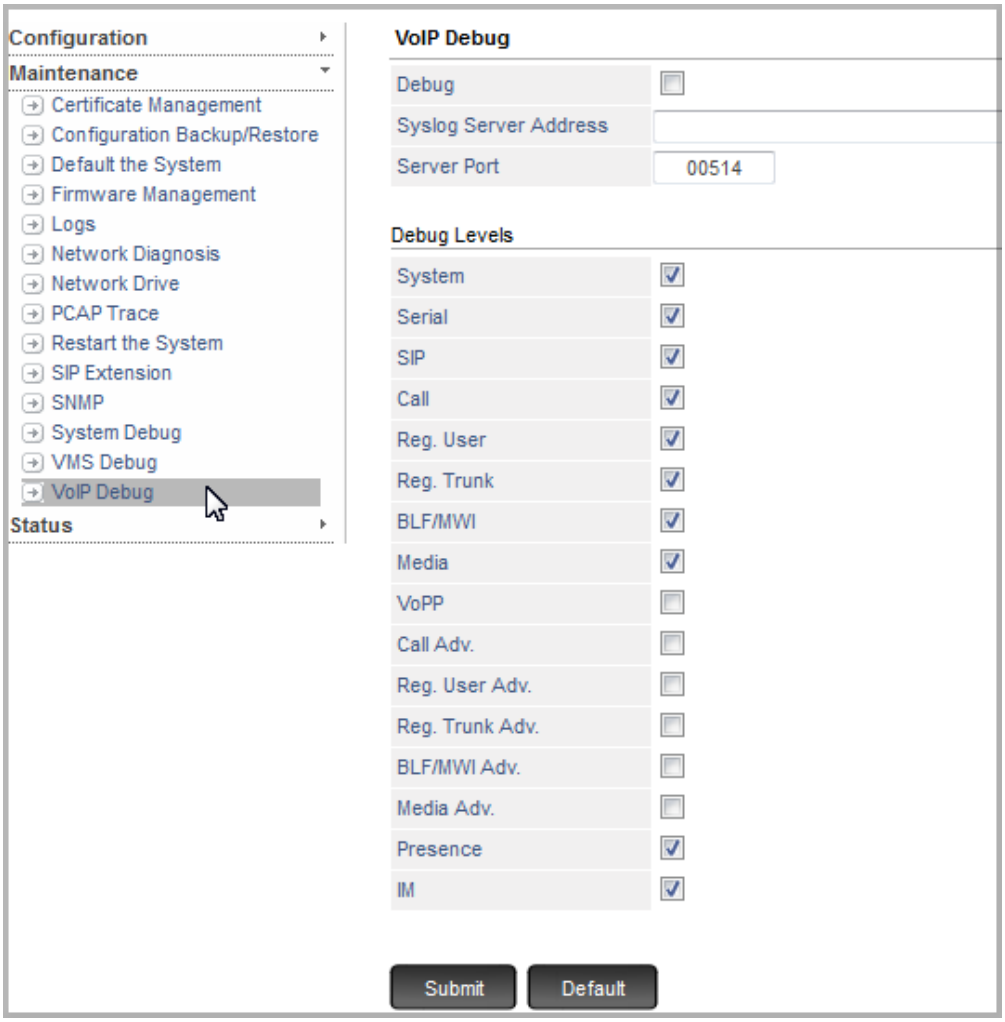
357. *For Slot Assignment in ETERNITY LENX, refer "Software Port and Hardware ID".*
358. *For Slot Assignment in ETERNITY LENX, refer "Software Port and Hardware ID".*

VoIP Debug

Matrix SARVAM UCS supports a feature by which SE can view the VoIP debugs on the server over IP network. This is done by 'Syslog Client' in the SARVAM UCS which supports multiple debug levels.

Configuring VoIP Debug using Jeeves

- Log in as System Engineer.
- Under **Maintenance**, click **VoIP Debug**.



- Configure the following:
  - Select the **Debug** check box to enable.
  - Configure **Syslog Server IP Address** and **Port**.
    Both IPv4 and IPv6 addresses are supported. By default, the Port is 514. Valid range of the port is from 1025 to 65535 and 514.
  - Now, select the respective Debug Level check box to enable:
    - System

- Serial
- SIP
- Call
- Reg.User (Registered User)
- Reg. Trunk (Registered Trunk)
- BLF/MWI
- Media
- VoPP
- Call Adv. (Advance)
- Reg. User Adv. (Registered User Advance)
- Reg. Trunk Adv. (Registered Trunk Advance)
- BLF/MWI Adv. (BLF/MWI Advance)
- Media Adv. (Media Advance)
- Presence
- IM
- RTCP

As per the level selected, debug log will be generated. For example: if debug log of Call is required, enable 'CALL' level and disable all other debug levels.

- Click **Submit**.

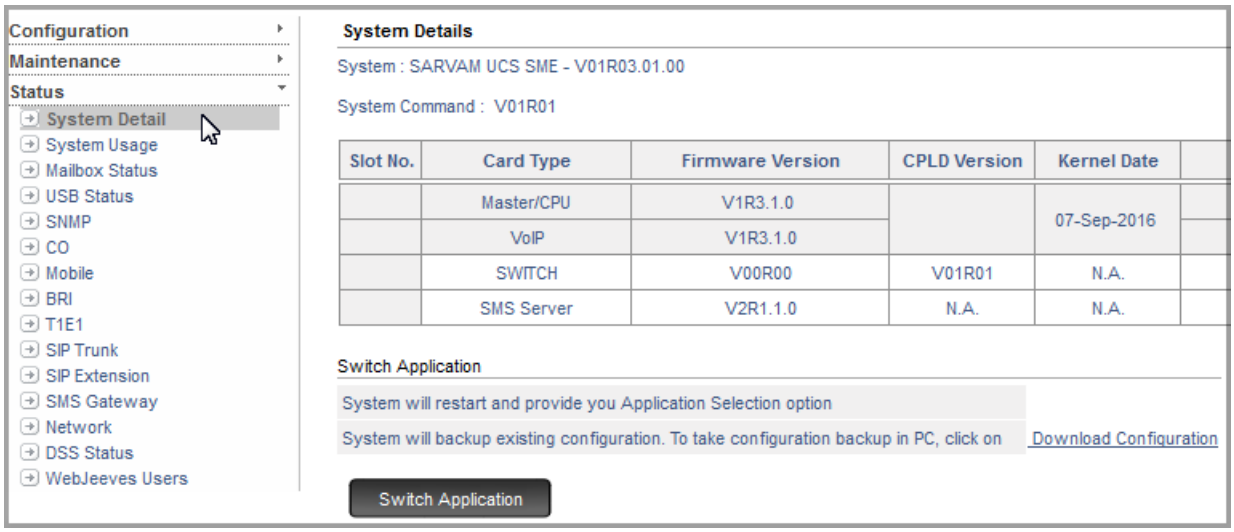
We recommend you to consult the Matrix support team before you enable Advance debugs. These debugs lead to debug data streaming in huge quantities which may affect the system performance.

System Details

The System Details displays the details of the system along with the slot details of the cards which are installed in the system with the Slot Number they are installed in. It also displays their respective Firmware Version, CPLD Version and Kernel Date.

To view the System Details,

- Log in as System Engineer.
- Under **Status**, click **System Details**.



If you have installed ETERNITY LENX, this page will also display the details of the Power Supply card. See "The Power Supply Card" for details.

Switch Application

If you want to run some other application on the ETERNITY GENX platform, click on the **Switch Application** button.

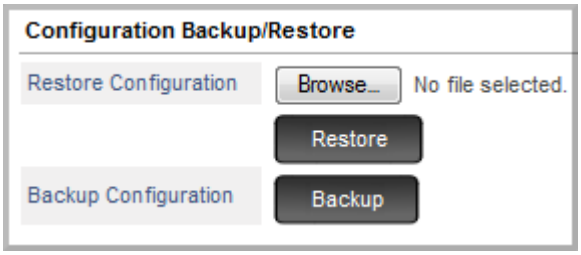
By clicking on the **Switch Application**, you will be diverted to the ETERNITY GENX Login page.

You must login with the same password you kept for the Application.

Select the Application Selection link and then the desired application you wish to run on the ETERNITY GENX Platform. Click on Next button. See "Application Selection".

The system will backup the existing configuration you did on this application. If you wish to take the backup of configuration in your PC, click on the **Download Configuration** link.

- **Configuration Backup/Restore** page opens.



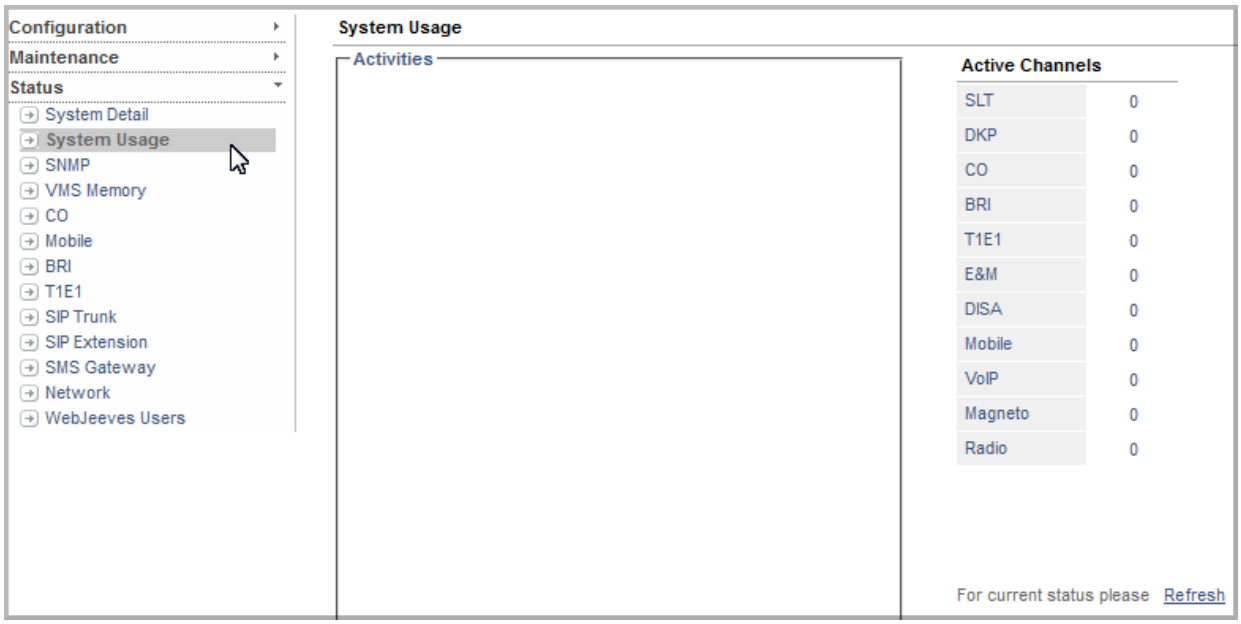
You can restore or backup configuration. For instructions, see “Configuration Backup/Restore”.

System Usage

To view the active channels and their activities,

- Log in as System Engineer.
- Under **Status**, click **System Usage**.

  The page displays the status of the calls along with the ports and trunks involved.



System Performance

What's this?

SARVAM UCS provides the facility to view the performance of the system online. You can check — the CPU usage, Memory (RAM) usage, speed and status of the Transmitted and Received data from the LAN port as well as the WAN port, Disk storage and Disk Activity. You can also view the system uptime, that is, the time for which the system has been functioning after the last restart.

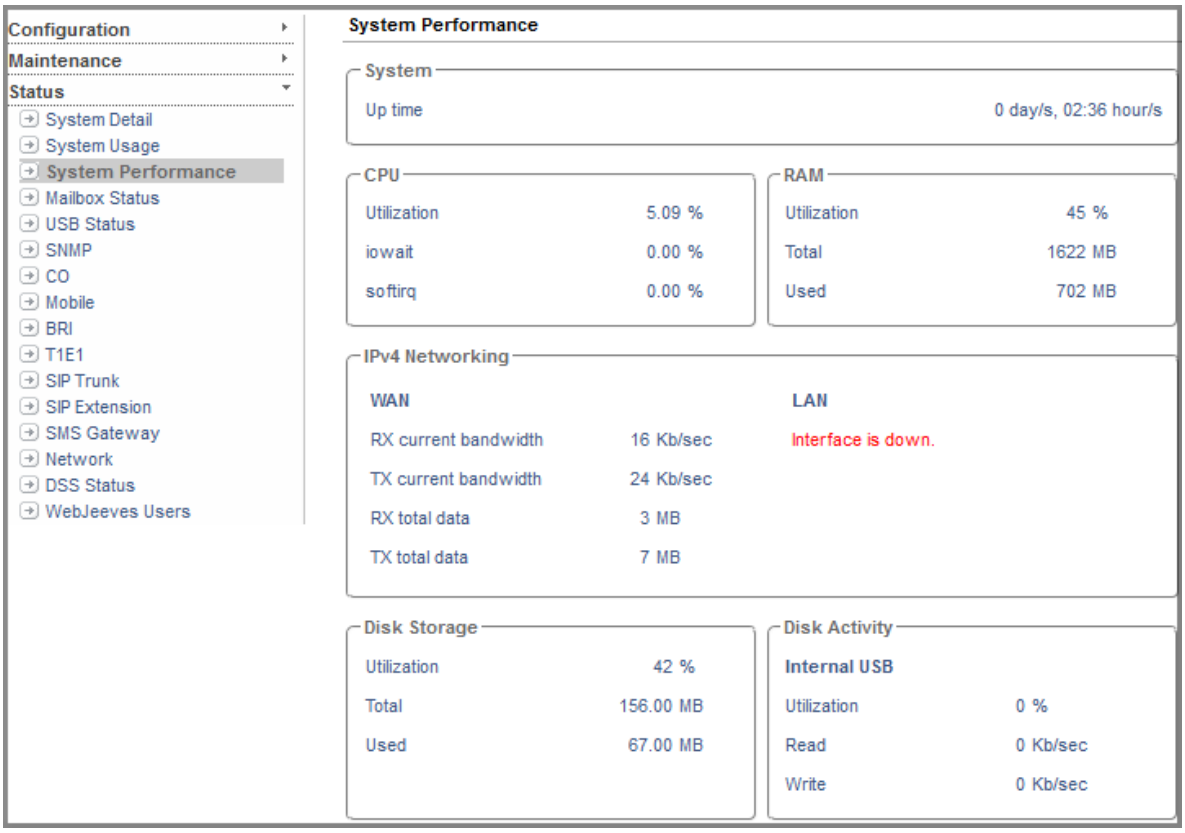
When the CPU and/or Memory (RAM) usage exceeds 80%, it is considered as high usage and the system sends notifications through Email, if configured. These events are also logged in the System Activity Log.

Similarly, when the CPU and/or Memory (RAM) usage is back to normal the system sends notifications through Email, if configured. These events are also logged in the System Activity Log.

For details, refer “System Log Notification” and “System Activity Log”.

To view the System Performance,

- Login as System Engineer.
- Under **Status**, click the **System Performance** link.



USB Status

What's this?

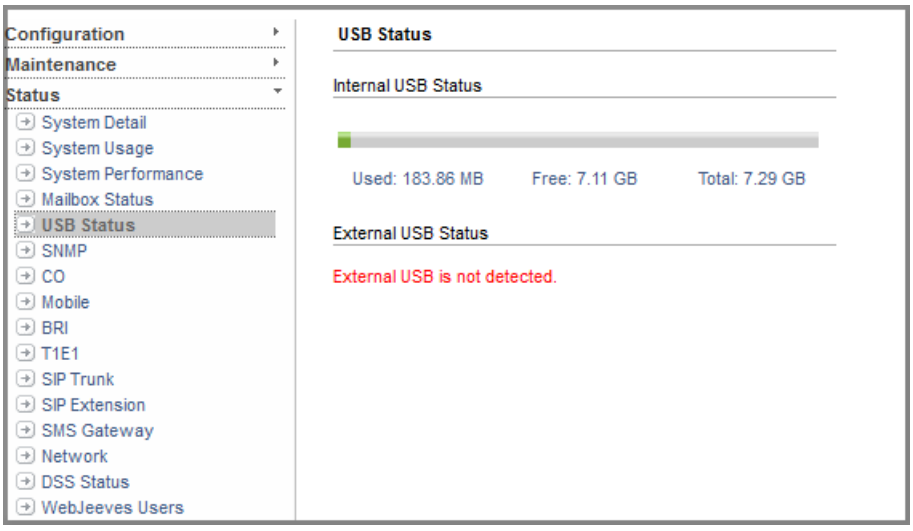
SARVAM UCS supports one internal USB (factory fitted) and one external USB.

The USB Status page displays the status of the internal as well external USB connected with the system. It also displays the memory status (used, free, total memory space) of the connected USBs.

If the external USB is not connected with the system, the status is displayed as *External USB is not detected.*

To view the USB Status,

- Login as System Engineer.
- Under **Status**, click the **USB Status** link.



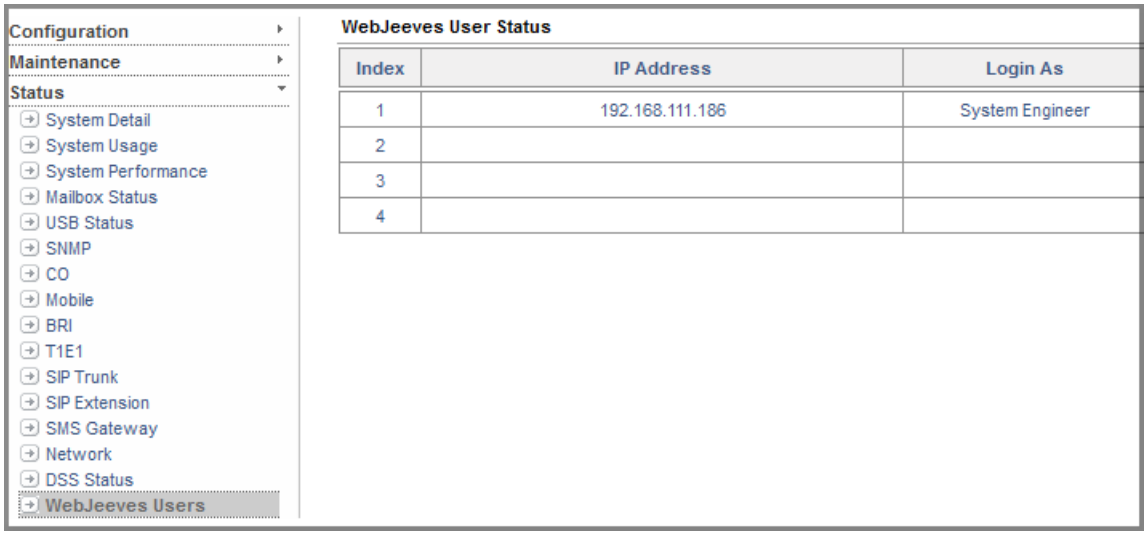
WebJeeves Users

What's this?

The WebJeeves Users Page displays the IP Address and Login Mode of the users accessing the Jeeves. It also displays the type of protocol used for accessing Jeeves.

To view the status of WebJeeves User,

- Login as System Engineer.
- Under **Status**, click **WebJeeves Users**.



Appendix

Technical Specifications - ETERNITY LENX

System Resource	Specifications
Processor Type	32 bit RISC processor
Processor speed	900 MHz dual core
Universal Slots	27
Number of USB Ports	1 Internal USB 2.0 1 External USB 3.0 (for future use)
USB Storage Capacity	Internal USB = 64 GB
Ethernet Ports	1 Gbps Eth for WAN 1 Gbps Eth for LAN
Vocoder Modules for VoIP[a]	64 Channels supported by each module
VMS Module for VMS[b]	64 Channels
Mounting Options	1. 19” Rack Mountable 2. Table Top 3. Wall Mountable
Dimensions (LxBxH)	715 X 474.25 X 449 mm (W/o caster wheel, Side clamp & Wall mounting clamp) 749 X 474.25 X 449 mm (With caster wheel & w/o Side clamp & Wall mounting clamp) 749 X 495.5 X 482.50 mm (With caster wheel, Side clamp & Wall mounting clamp)
Shipping Weight (Kg)	52 kg approx. (with packing)
Shipping Material Type	Corrugated Box
Power Supply Options	DC PS
Power Supply Input Range	DC PS :48 VDC (+20% to -15%)
Power Consumption[c]	For DC PS :688.19 W
Operating Temperature	0^{o} C to 45^{o} C
Operating Humidity	5 - 95 RH, Non-Condensing
Storage Temperature	- 20 Degree to +70 Degree
Storage Humidity	0 - 95% RH, Non-Condensing

a. Max number of VoIP modules that can be placed on the board are 4.
b. Max number of VMS module that can be placed on the board is 1.
c. Considering 30% SLT off-hook when 1296 (if the CPU Card has 1K IDT Switch then 1232) SLTs are connected.

Technical Specifications - ETERNITY MENX

System Resource	Specifications
Processor Type	32 bit RISC processor
Processor speed	900 MHz dual core
Universal Slots	16
Number of USB Ports	1 Internal USB 2.0 1 External USB 3.0 (for future use)
USB Storage Capacity	Internal USB = 64 GB
Ethernet Ports	1 Gbps Eth for WAN 1 Gbps Eth for LAN
Vocoder Modules for VoIP[a]	64 Channels supported by each module
VMS Module for VMS[b]	64 Channels
Mounting Options	1. 19” Rack Mountable 2. Table Top 3. Wall Mountable
Dimensions (WxHxD)	612 x 405.5 x 325.55 mm
Shipping Weight (Kg)	18 Kg approx. (with packing)
Shipping Material Type	Corrugated Box
Power Supply Options	AC PS DC PS
Power Supply Input Range	AC PS : 100 - 240 VAC DC PS : 48 VDC (+20% & -15%)
Power Consumption[c]	For AC PS : 402.17 W For DC PS : 325.90 W
Operating Temperature	0^{o} C to 45^{o} C
Operating Humidity	5 - 95 RH, Non-Condensing
Storage Temperature	- 20 Degree to +70 Degree
Storage Humidity	0 - 95% RH, Non-Condensing

a. Max number of VoIP modules that can be placed on the board are 4.
b. Max number of VMS module that can be placed on the board is 1.
c. Considering 30% SLT off-hook when 512 SLTs are connected.

Technical Specifications - ETERNITY GENX

System Resource	Specifications
Processor Type	32 bit RISC processor
Processor speed	900 MHz dual core
Universal Slots	12
Number of USB Ports	1 Internal USB 2.0 1 External USB 3.0 (for future use)
USB Storage Capacity	Internal USB = 64 GB
Ethernet Ports	1 Gbps Eth for WAN 1 Gbps Eth for LAN
Communication Ports	1
Vocoder Modules for VoIP[a]	64 Channels supported by each module
VMS Module for VMS[b]	64 Channels
Mounting Options	1. 19” Rack Mountable 2. Table Top 3. Wall Mountable
Dimensions (WxHxD)	436.5 x 226 x 173.5 mm
Unit Weight (Kg)	5.00 Kg
Shipping Weight (Kg)	8.20 Kg
Shipping Material Type	Corrugated Box
Power Supply Options	AC PS DC PS
Power Supply Input Range	AC PS : 100 - 240 VAC DC PS : 48 VDC (+20% & -15%)
Power Consumption[c]	For AC PS : 227 W For DC PS :183 W
Operating Temperature	0^{o} C to 45^{o} C
Operating Humidity	5 - 95 RH, Non-Condensing
Storage Temperature	- 20 Degree to +70 Degree
Storage Humidity	0 - 95% RH, Non-Condensing

a. Max number of VoIP modules that can be placed on the board are 2.
b. Max number of VMS module that can be placed on the board is 1.
c. Considering 30% SLT off-hook when 240 SLTs are connected.

Technical Specifications - ETERNITY PENX

System Resource	Specifications
Processor Type	32 bit RISC processor
Processor speed	1 GHz single core
Universal Slots	6
Number of USB Ports	1 Internal USB 2.0 1 External USB 2.0
USB Storage Capacity	Internal USB = 64 GB
Ethernet Ports	1 Gbps Eth for WAN 1 Gbps Eth for LAN
Communication Ports	1
Vocoder Modules for VoIP	1 Module (64 Channels supported)
VMS Module for VMS	1 Module (16 Channels)
Mounting Options	1. 19” Rack Mountable 2. Table Top 3. Wall Mountable
Dimensions (WxHxD)	482 X 331 X 55 in mm – with Clamps 448 X 331 X 55 in mm – without clamps
Unit Weight (Kg)	2.40 Kg Approximately
Shipping Weight (Kg)	3.80 Kg
Shipping Material Type	Breakable
Power Supply Options	AC PS
Power Supply Input Range	AC PS : 100 - 240 VAC
Power Consumption[a]	For AC PS : 40W
Operating Temperature	Minimum Operating Temperature: 0°C Maximum Operating Temperature: 45°C
Operating Humidity	5 - 95 RH, Non-Condensing
Storage Temperature	Minimum Storage Temperature: -20°C Maximum Storage Temperature: +70°C
Storage Humidity	0 - 95% RH, Non-Condensing

a. Considering 30% SLT off-hook when 48 SLTs are connected.

Technical Specifications - SARVAM UCS

ETERNITY LENX

System Resource	Specifications
SIP Extensions	2000 Max
SIP Trunks	99 Max
DKP Ports	128 Max
SLT Ports	1296 Max[a]
T1E1 Ports	24 Max
BRI Ports	32 Max
GSM Ports	64 Max
CO Ports	128 Max
E&M Ports	64 Max
Radio Ports	16 Max
Magneto Ports	16 Max
Vocoder Channels[b]	248 Max
Voice Mail Channels[c]	64 Max
IP to IP audio Calls with DRTP/RTP Relay	550 Max
IP to IP audio Calls with Vocoder channels	64 Max[d]
IP to non-IP audio Calls	128 Max[e]
IP to IP Video Calls	55 Max
Maximum 3-Party Conference	15
Maximum Parties in a Single Multi-Party Conference	21
Maximum Multi-Party Conference	2 x 21 Parties + 1 x 3 Parties
Maximum Call Taping	15
Number of Voice Modules[f]	16
Number of voice modules that can be simultaneously played	11

a. If you have CPU Card with 1K IDT Switch (PCB Version V1R3), the maximum number of SLT ports supported are 1232.
b. Four optional Vocoder Modules (NX DBM VOCODER64) can be installed.
c. Optional VMS Module (NX DBM VMS64) required.
d. When two NX DBM VOCODER64 Modules are installed.
e. When two NX DBM VOCODER64 Modules are installed.
f. 16 seconds each.

Supported Cards

Cards	Remarks
ETERNITY LE Card SLT48	
ETERNITY ME Card SLT32	
ETERNITY ME Card SLT16	
ETERNITY ME Card SLT8	
ETERNITY ME Card CO8+SLT24	
ETERNITY ME CARD DKP16	
ETERNITY LE Card ILC48	
ETERNITY ME Card ILC32	
ETERNITY ME DKP32	
ETERNITY ME DKP16	
ETERNITY ME DKP8	
ETERNITY ME Card CO16	
ETERNITY ME Card CO8	
ETERNITY ME Card BRI8	
ETERNITY ME Card BRI4	
ETERNITY ME Card T1E1PRI Dual	
ETERNITY ME Card T1E1PRI Single	
ETERNITY ME Card E1FO Dual	
ETERNITY ME Card E1FO Single	
ETERNITY ME Card E&M8	
ETERNITY ME Card E&M4	
ETERNITY ME Card GSM8	2G/3G/4G
ETERNITY ME Card CDMA2	
ETERNITY ME Card Magneto8	
ETERNITY ME Card Radio8	
ETERNITY ME Card Radio4	

ETERNITY MENX

System Resource	Specifications
SIP Extensions	2000 Max
SIP Trunks	99 Max
DKP Ports	128 Max

System Resource	Specifications
SLT Ports	512 Max
T1E1 Ports	8 Max
BRI Ports	32 Max
GSM Ports	64 Max
CO Ports	128 Max
E&M Ports	64 Max
Radio Ports	16 Max
Magneto Ports	16 Max
Vocoder Channels[a]	248 Max
Voice Mail Channels[b]	64 Max
IP to IP audio Calls with DRTP/RTP Relay	550 Max
IP to IP audio Calls with Vocoder channels	64 Max[c]
IP to non-IP audio Calls	128 Max[d]
IP to IP Video Calls	55 Max
Maximum 3-Party Conference	15
Maximum Parties in a Single Multi-Party Conference	21
Maximum Multi-Party Conference	2 x 21 Parties + 1 x 3 Parties
Maximum Call Taping	15
Number of Voice Modules[e]	16
Number of voice modules that can be simultaneously played	11

a. Four optional Vocoder Modules (NX DBM VOCODER64) can be installed.
b. Optional VMS Module (NX DBM VMS64) required.
c. When two NX DBM VOCODER64 Modules are installed.
d. When two NX DBM VOCODER64 Modules are installed.
e. 16 seconds each.

Supported Cards

Cards	Remarks
ETERNITY ME Card SLT32	
ETERNITY ME Card SLT16	
ETERNITY ME Card SLT8	
ETERNITY ME Card CO8+SLT24	
ETERNITY ME CARD DKP16	
ETERNITY ME Card ILC32	
ETERNITY ME DKP32	

Cards	Remarks
ETERNITY ME DKP16	
ETERNITY ME DKP8	
ETERNITY ME Card CO16	
ETERNITY ME Card CO8	
ETERNITY ME Card BRI8	
ETERNITY ME Card BRI4	
ETERNITY ME Card T1E1PRI Dual	
ETERNITY ME Card T1E1PRI Single	
ETERNITY ME Card E1FO Dual	
ETERNITY ME Card E1FO Single	
ETERNITY ME Card E&M8	
ETERNITY ME Card E&M4	
ETERNITY ME Card GSM8	2G/3G/4G
ETERNITY ME CARD CDMA2	
ETERNITY ME Card Magneto8	
ETERNITY ME Card Radio8	
ETERNITY ME Card Radio4	

ETERNITY GENX

System Resource	Specifications
SIP Extensions	999 Max
SIP Trunks	99 Max
DKP Ports	96 Max
SLT Ports[a]	240 Max
T1E1 Ports	8 Max
BRI Ports	32 Max
GSM Ports	40 Max
CO Ports	64 Max
E&M Ports	32 Max
Radio Ports	16 Max
Magneto Ports	16 Max
Vocoder Channels[b]	128 Max
Voice Mail Channels[c]	64 Max

System Resource	Specifications
IP to IP audio Calls with DRTP/RTP Relay	550 Max
IP to IP audio Calls with Vocoder channels	64 Max[d]
IP to non-IP audio Calls	128 Max[e]
IP to IP Video Calls	55 Max
Maximum 3-Party Conference	20
Maximum Parties in a Single Multi-Party Conference	21
Maximum Multi-Party Conference	2 x 21 Parties + 1 x 20 Parties
Maximum Call Taping	20
Number of Voice Modules[f]	15
Number of voice modules that can be simultaneously played	9

a. The maximum number of simultaneous off-hook SLT ports supported are 120.
b. Two optional Vocoder Modules (NX DBM VOCODER64) can be installed.
c. Optional VMS Module (NX DBM VMS64) required.
d. When two NX DBM VOCODER64 Modules are installed.
e. When two NX DBM VOCODER64 Modules are installed.
f. 16 seconds each.

Supported Cards

Cards	Remarks
ETERNITY GE CARD SLT8	
ETERNITY GE CARD SLT16	
ETERNITY GE CARD SLT20	
ETERNITY GE CARD ILC20	
ETERNITY GE CARD DKP8	
ETERNITY GE CARD DKP16	
ETERNITY GE CARD CO8	
ETERNITY GE CARD CO16	
ETERNITY GE CARD CO4+DKP2+SLT12	With inbuilt PFT
ETERNITY GE CARD CO4+DKP2+SLT8	With inbuilt PFT
ETERNITY GE CARD CO2+DKP2+SLT16	
ETERNITY GE CARD CO4+SLT16	
ETERNITY GE CARD DKP4+SLT16	
ETERNITY GE CARD GSM4	2G/3G/4G
ETERNITY GE CARD CDMA2	
ETERNITY GE CARD T1E1PRI SINGLE	

Cards	Remarks
ETERNITY GE CARD E1FOPRI SINGLE	
ETERNITY GE CARD RIC4	
ETERNITY GE CARD MAGNETO4	
ETERNITY GE CARD E&M4	
ETERNITY GE CARD BRI4	

ETERNITY PENX

System Resource	Specifications
Universal Slots	6
Pluggable Modules	1 On-Board Octasic Module – 64 Channel 1 On-Board VMS Module – 64 Channel
Internal USB Port	1
External USB Port	1
LAN (RJ-45)	1 (Gigabit Port)
WAN(RJ-45)	1 (Gigabit Port)
LED for System Health	2
LED for Power	1
SIP Extensions	100 Max
SIP Trunks	50 Max
DKP Ports	16 Max
SLT Ports	48 Max
T1E1 Ports	2 Max
GSM Ports	8 Max
CO Ports	16 Max
Voice Mail	1
VoIP Module	1
VoIP Channels	64 Max
No.of Voice Modules	16 Max
Simultaneous voice module playback	5
Total Number of Audio Conference Participants (System Wide)	48
Max number of simultaneous 3-party conference (System Wide)	16
IP to IP audio Calls with DRTP/RTP Relay	64 Max
IP to IP audio Calls with Vocoder channels	32 Max

System Resource	Specifications
IP to non-IP audio Calls	64 Max (No Video calls will not be supported simultaneously)
IP to IP Video Calls	16 Max (No Audio calls will not be supported simultaneously)
TDM-TDM Audio Calls	Non-Blocking
Maximum Parties in a Single Multi-Party Conference	48
Maximum Multi-Party Conference	16 (3-Party Conference)
Maximum Call Taping	16

Supported Cards

Cards	Remarks
ETERNITY PE CARD SLT8	
ETERNITY PE CARD SLT4	
ETERNITY PE CARD DKP8	
ETERNITY PE CARD CO8	
ETERNITY PE CARD DKP2+SLT6	
ETERNITY PE CARD CO4+SLT4	
ETERNITY PE CARD CO2+DKP2+SLT4	
ETERNITY PE CARD DKP2+SLT6	
ETERNITY PE CARD T1E1PRI SINGLE	
ETERNITY PE CARD GSM4	2G/3G/4G

Supported Terminals

UC Clients

VARTA ADR100	Unified Communication Client for Android
VARTA AMP100	Unified Communication Client for iOS
VARTA WIN200	Unified Communication Client for Windows

Desk Phones

EON48	The Digital Key Phone
EON310	Executive Digital Key Phone
EON510	Premium Digital Key Phone
SPARSH VP248	The High-Definition Edge to your IP Communication

SPARSH VP310E	The Executive IP Phone
SPARSH VP330E	Intuitive Touch-Screen IP Phone
SPARSH VP510E	Premium IP Phone
SPARSH VP110	The Business IP Phone
SPARSH VP710	The Smart Video IP Desk phone

Analog SLT ports supported for Short Loop with Loop Current programmed

Loop Current Programmed	25mA	30mA	35mA	40mA
Number of ports supported in talk mode (OFF-Hook short loop) according to loop current programmed	200	175	150	128

SLT (Analog Station)

Signaling	Loop Start
Dialing	DTMF and Pulse (10/20PPS)
Off Hook AC Impedance	600/900/Complex
Off Hook Current	39mA max
Loop Limit	1800Ω max (excluding Telephone)
On-Hook Voltage (Tip/Ring)	-48V nominal
DTMF Detection	ITU-T Q.24
Return Loss	>18dB
Longitudinal Balance	>50dB
Transmission Level Adjust	Tx Gain: -3dB to +6dB, Rx Gain: -3dB to 6dB
Ringing	Trapezoidal 60VRMS/25Hz and Sinusoidal 52VRMS/25Hz
REN	3
CLI Reception	DTMF, FSK ITU-T V.23 and FSK Bellcore 202
Protection	Over Voltage Secondary Protection
Physical Connector	RJ45

DKP (Digital Station)

Signaling	Proprietary Digital (2B+D)
Interface	Single Pair for Speech, Signaling and Power
Loop Limit	100Ω
Speech Level	Adjustable Tx and Rx Gain for Handset and Hands-free
Protection	Over Voltage Secondary Protection

Physical Connector	RJ45

EON48

	EON48S	**EON48P**
LCD	2 lines x 24 characters	6 lines x 24 characters
Dimensions	241x203x77.50mm (L x B x H)	241x203x77.50mm (LxBxH)
Weight	1.174 kg	1.302 kg
Environmental		
Operating Temperature	0° C to 45°C	0° C to 45°C
Operating Humidity	5 to 95% RH, Non-Condensing	5 to 95% RH, Non-Condensing
Storage Temperature	-20°C to +70°C	-20°C to +70°C
Storage Humidity	5 to 95% RH, Non-Condensing	5 to 95% RH, Non-Condensing

EON310

LCD	2*24 Character display
Dimensions	207x335x78 mm (LxBxH)
Weight	820gms
Environmental	
Operating Temperature	0° C to 45°C
Operating Humidity	5 to 95% RH, Non-Condensing
Storage Temperature	-20°C to +70°C
Storage Humidity	5 to 95% RH, Non-Condensing

EON510

LCD	240*64 Pixel Graphical LCD display
Dimensions	247x183x43 mm (LxBxH)
Weight	805gms
Environmental	
Operating Temperature	0° C to 45°C
Operating Humidity	5 to 95% RH, Non-Condensing
Storage Temperature	-20°C to +70°C
Storage Humidity	5 to 95% RH, Non-Condensing

SPARSH VP248

VoIP	
VoIP Protocols	SIP v2, SDP, RTP, RFC 2833
Network Protocol	IPv4, TCP, UDP, DHCP, STUN, PPPoE
NAT	STUN and NAT Keep Alive
Voice CODECS	G.722 Wideband, G.711 A/μ-Law, G.723-5.3, G.723-6.3, G.726-16, G.726-24, G.726-32, G.726-40, G.729AB
Call Progress Tones	Dial Tone, Ring Back Tone, Busy Tone, Error Tone, Waiting Tone
Voice	Dynamic Jitter Buffer (Adaptive), Comfort Noise Generation and Voice Activity Detection
Quality of Service	Layer 3 DIFFServ and TOS
Data Network	LAN Port (RJ45), 10/100 Base T (PoE Optional) PC Port (RJ45), 10/100 Base T
LCD Display	2 Lines and 6 Lines Display
Security	Password Protected Configuration
Power Supply	
Input	5VDC @2A through External Adapter (90 - 265 VAC, 47 - 63Hz, Optional) and Power-over-Ethernet (PoE)
Power Consumption	5W (Typical)
Mechanical	
Dimensions (WxHxD)	20.7 x 23.2 x 4.5 cm
Material	ABS Plastic
Installation Mounting	Wall Mount and Table - Top
Environmental	
Operating Temperature	0°C to 45°C
Operating Humidity	5 to 95% RH, Non-Condensing
Storage Temperature	-20°C to +70°C
Storage Humidity	0 to 95% RH, Non-Condensing
Unit Weight	1.18 Kgs (2.6 lbs) Approx.

SPARSH VP310

VoIP	
VoIP Protocols	SIP v2, SDP, RTP, RFC 2833
Network Protocol	IPv4, TCP, UDP, DHCP, PPPoE
Voice CODECS	G.722 Wideband, G.711 A/μ-Law, G.723, G.729, iLBC - 20/30 msec

Call Progress Tones	Dial Tone, Ring Back Tone, Busy Tone, Error Tone, Waiting Tone
Voice	Dynamic Jitter Buffer (Adaptive)
Quality of Service	Layer 2 CoS, Layer 3 DIFFServ and TOS
Data Network	LAN Port (RJ45), 10/100 Base T (PoE Optional) PC Port (RJ45), 10/100 Base T
LCD Display	2 x 24 Character Display
Security	TLS, SRTP
Power Supply	
Input	5VDC @2A through External Adapter (100-240 VAC, 50 - 60 Hz, Optional) and Power-over-Ethernet (PoE)
Power Consumption	4W (Typical)
Mechanical	
Dimensions (WxHxD)	20.7 x 23.2 x 4.5 cm
Material	ABS Plastic
Installation Mounting	Wall Mount and Table - Top
Environmental	
Operating Temperature	0°C to 45°C
Operating Humidity	5 to 95% RH, Non-Condensing
Storage Temperature	-20°C to +70°C
Storage Humidity	5 - 95% RH, Non-Condensing
Weight (Without Foot Stand)	830 gms Approx.

SPARSH VP330

LCD Display	4.3" Colour TFT Touch Screen Display
VoIP	
VoIP Protocols	SIP v2, SDP, RTP, RFC 2833
Network Protocol	IPv4, TCP, UDP, DHCP, PPPoE
Voice CODECS	G.722 Wideband, G.711 A/μ-Law, G.723, G.729
Call Progress Tones	Dial Tone, Ring Back Tone, Busy Tone, Error Tone, Waiting Tone
Voice	Dynamic Jitter Buffer (Adaptive)
Quality of Service	Layer 2 CoS, Layer 3 DIFFServ and TOS
Data Network	LAN Port (RJ45), 10/100 Base T (PoE Optional) PC Port (RJ45), 10/100 Base T
LCD Display	2 x 24 Character Display
Security	TLS, SRTP

Power Supply	
Input	5VDC @2A through External Adapter (100-240 VAC, 50 - 60 Hz, Optional) and Power-over-Ethernet (PoE)
Power Consumption	5.50 W (Typical)
Mechanical	
Dimensions (WxHxD)	20.7 x 23.2 x 4.5 cm
Material	ABS Plastic
Installation Mounting	Wall Mount and Table - Top
Environmental	
Operating Temperature	0° C to 45°C
Operating Humidity	5 to 95% RH, Non-Condensing
Storage Temperature	-20°C to +70°C
Storage Humidity	5 to 95% RH, Non-Condensing
Weight (Without Foot Stand)	840 gms Approx.

SPARSH VP510

VoIP	
VoIP Protocols	SIP v2, SDP, RTP, RFC 2833
Network Protocol	IPv4/IPv6, TCP, UDP, DHCP, PPPoE
Voice CODECS	G.722 Wideband, G.711 A/μ-Law, G.723, G.729
Voice	Dynamic Jitter Buffer (Adaptive)
Quality of Service	Layer 2 CoS, Layer 3 DIFFServ and TOS
Data Network	LAN Port (RJ45), 10/100 Base T (PoE Optional) PC Port (RJ45), 10/100 Base T
Security	TLS, SRTP
LCD	240*64 Pixel Graphic LCD Display
Power Supply	
Input	5VDC @2A through External Adapter (100-240 VAC, 50 - 60 Hz, Optional) and Power-over-Ethernet (PoE)
Power Consumption	2.25W (stand alone)
Mechanical	
Weight	805 gm
Dimension in mm [L*B*H]	247*183*43
Material	ABS Plastic
Installation Mounting	Wall Mount and Table - Top
Environmental Conditions	
Operating Temperature Range	0°C to 45°C

Storage Temperature	-20°C to +70°C
Operating and Storage Humidity	5 to 95%, RH, Non-Condensing

SPARSH VP248 (Standard SIP Phone)

VoIP	
VoIP Protocols	SIP v2, SDP, RTP, RFC 2833
Network Protocol	IPv4, TCP, UDP, DHCP, STUN, PPPoE
SIP	9 Multiple SIP Accounts Out Bound Proxy Support Main and Secondary DNS Server Support
NAT	STUN and NAT Keep Alive
Voice CODECS	G.722 Wideband, G.711 A/µ-Law, G.723-5.3, G.723-6.3, G.726-16, G.726-24, G.726-32, G.726-40, G.729AB
Call Progress Tones	Dial Tone, Ring Back Tone, Busy Tone, Error Tone, Waiting Tone
Voice	Dynamic Jitter Buffer (Adaptive), Comfort Noise Generation and Voice Activity Detection
Quality of Service	Layer 3 DIFFServ and TOS
Data Network	LAN Port (RJ45), 10/100 Base T (PoE Optional) PC Port (RJ45), 10/100 Base T
Security	Password Protected Administration
Power Supply	
Input	5VDC @2A through External Adapter (90-265VAC, 47-63Hz, Optional) Power-over-Ethernet (PoE)
Power Consumption	5W (Typical)
Mechanical	
Dimensions (WxHxD)	24.0 x 20.0 x 9.9 cm (9.4”x7.9”x3.9”)
Material	ABS Plastic
Installation Mounting	Wall Mount and Table-Top
Environmental	
Operating Temperature	0°C to 45°C

Physical Features of SPARSH VP110

This section lists the available physical features of SPARSH VP110 IP Phones.

- 132 x 64 Graphic LCD
- Single VoIP account
- 29 keys including 4 soft keys
- 1 x RJ9 (4P4C) Handset port

- 1 x RJ9 (4P4C) Headset port
- 2 x RJ45 10/100Mbps Ethernet ports
- 1 LED: 1 x Power
- Power Adapter (Optional): AC 100~240V input and DC 5V/600mA output
- Power over Ethernet (IEEE 802.3af)

Technical Specifications

Audio Features

- Full-duplex Hands-free Speakerphone with AEC
- Codecs: G.711(A/μ), G.722, G.723, G.729, G.726, iLBC
- DTMF: In-band, Out-of-band (RFC 2833) and SIP INFO
- VAD, CNG, AEC, PLC, AJB, AGC

Phone Book

- Local Phone book up to 1000 entries Black List
- XML Remote Phone book
- Intelligent Search Method
- Phone book Search/Import/Export
- Call History: Dialed/Received/Missed/Forwarded

Phone Features

- Single VoIP Account
- Call Hold, Mute, DND
- One-touch Speed Dial, Hotline
- Redial, Call Return, Auto Answer
- Call Forward, Call Waiting, Call Transfer
- Local 3-way Conference
- Direct IP Call without SIP Proxy
- Ringtone Selection/Import/Delete
- Keypad Lock, Emergency Call
- Set Date & Time Manually or Automatically
- Dial Plan, XML Browser, Action URL/URI
- Instant Messaging (Web UI and Phone)

Call Management

- Anonymous Call (CLIR)
- Anonymous Call Rejection
- Message Waiting Indicator (MWI)

- Voicemail, Call Pickup
- Intercom, Music on Hold
- Call Completion, Hot-desking
- Dial out Number from Web UI

Display

- 132x64-pixel Graphical LCD
- LED for Indicating Incoming calls, Voice/Text Messages, Mute, Call Hold/Held, On call
- Intuitive User Interface with Icons and Soft keys
- Multiple Language Options
- Caller ID with Name, Number

Networking and Security

- SIP v1 (RFC2543), v2 (RFC3261)
- IPv6
- NAT Transverse: STUN Mode
- Proxy Mode and Peer-to-Peer SIP Link Mode
- IP Assignment: Static/DHCP/PPPoE
- HTTPS Web Server
- Time and Date Synchronization using SNTP
- UDP/TCP/DNS-SRV (RFC 3263)
- QoS: 802.1p/Q Tagging (VLAN), Layer 3 ToS, DSCP
- SRTP for Voice
- Transport Layer Security (TLS)
- HTTPS Certificate Manager
- AES Encryption for Configuration File
- Digest Authentication
- IEEE802.1X
- SNMP v1/v2

Management

- Configuration: Browser/Phone/Auto-Provision
- Auto Provision via FTP/TFTP/HTTP/HTTPS for Mass Deploy
- Server Redundancy
- Factory Reset
- Soft Reboot
- Packet Tracing Export
- System Log

Physical Features

- 2 x 10/100 Mbps LAN & PC Ports
- 29 keys including 4 Soft Keys
- 1 x RJ9 Handset Port
- 1 x RJ9 Headset Port
- Dimension (W x D x H): 185 x 188 x 143 mm

Power Supply

- Power Adapter (Optional): 5VDC/600mA
- Power over Ethernet (IEEE 802.3af)
- Power Consumption: 5W (Typical)
- Connector: DC Power Jack

Mechanical

- Packaging: 10 Qty/CTN
- Net Weight: 9.8 Kg
- Gross Weight: 10.8 Kg
- Gift Box: 215 x 200 x 121 mm
- Installation: Wall Mount, Table-top
- Color: Gray

Environmental

- Operating Temperature: -10° C to 50°C (14° F to 122° F)
- Operating Humidity: 10 - 95% (Non-Condensing)

Certifications

CE, FCC-15 (Class-B), RCM, RoHS

Physical Features of SPARSH VP710

This section lists the available physical features of SPARSH VP710 IP phone.

- 7” 1024 x 600 pixel color touch screen with backlight
- Operating System: Android™ 5.1.1
- 16 VoIP accounts
- HD Voice: HD Codec, HD Handset, HD Speaker
- 20 dedicated hard keys, 3 dedicated soft Android keys for BACK, HOME and RECENT
- 1*RJ9 (4P4C) handset port
- 1*RJ9 (4P4C) headset port
- 2*RJ45 10/100/1000Mbps Ethernet ports
- 4 LEDs: 1*power, 1*mute, 1*headset, 1*speakerphone
- Power adapter: AC 100~240V input and DC 5V/2A output

- 1*USB2.0 port (on the top of the phone), support, USB camera.
- 1*USB2.0 port (on the rear of the phone), USB flash drive or USB headset
- Built-in Wi-Fi, support 802.11b/g/n
- Built-in Bluetooth 4.0, support Bluetooth headset
- Power over Ethernet (IEEE 802.3af)
- Wall Mountable

Key Features of the IP Phone

In addition to physical features introduced above, IP phone also supports the following key features when running the latest firmware:

Phone Features

- **Call Options:** emergency call, call waiting, call hold, call mute, call forward, call transfer, call pickup, five-way audio-only conference, five-way audio-only and video mixed conference (up to three-way video conference.
- **Basic Features:** DND, auto redial, live dialpad, dial plan, hotline, caller identity, auto answer.
- **Advanced Features:** BLF, server redundancy, distinctive ring tones, remote phone book, LDAP.

Codecs and Voice Features

- Wideband codec: G.722, Opus
- Narrowband codec: G.711, G.726, G.729, iLBC, G.723
- VAD, CNG, AEC, PLC, AJB, AGC
- Full-duplex speakerphone with AEC

Video Features

- Video codec: H264HP, H264, VP8
- Image codec: JPEG, PNG, BMP
- Adaptive bandwidth adjustment

Network Features

- SIP v1 (RFC 2543), v2 (RFC 3261)
- NAT Traversal: STUN mode
- DTMF: INBAND, RFC 2833, SIP INFO
- Proxy mode and peer-to-peer SIP link mode
- IP assignment: Static/DHCP/PPPoE
- VLAN assignment: LLDP/Static/DHCP/CDP
- Bridge mode for PC port
- HTTP/HTTPS server
- DNS client
- NAT/DHCP server
- IPv6 support
- Wi-Fi

Management

- FTP/TFTP/HTTP/PnP auto-provision
- Configuration: browser/phone/auto-provision
- Direct IP call without SIP proxy

- Dial number via SIP server
- Dial URL via SIP server
- TR-069

Security

- HTTPS (server/client)
- SRTP (RFC 3711)
- Transport Layer Security (TLS)
- VLAN (802.1q), QoS
- Digest authentication using MD5/MD5-sess
- Secure configuration file via AES encryption
- Phone lock for personal privacy protection
- Admin/User configuration mode
- 802.1X authentication

SPARSH VP210 (Extended)

LCD Display	128 x 64 Graphical LCD
VoIP	
VoIP Protocols	SIP v2, SDP, RTP, RFC 2833
Network Protocol	IPv4/IPv6, TCP, UDP, DHCP, PPPoE
Voice CODECS	G.722 Wideband, G.711 A/μ-Law, G.723, G.729
Call Progress Tones	Dial Tone, Ring Back Tone, Busy Tone, Error Tone, Waiting Tone
Voice	Dynamic Jitter Buffer (Adaptive)
Quality of Service	Layer 2 CoS, Layer 3 DIFFServ and TOS
Data Network	LAN Port (RJ45), 10/100/1000 Base T (PoE Optional) PC Port (RJ45), 10/100/1000 Base T
Security	TLS, SRTP
Power Supply	
Input	5VDC @2A through External Adapter (100-240 VAC, 50 - 60 Hz, Optional) and Power-over-Ethernet (PoE)
Power Consumption	1.0 W (Typical)
Mechanical	
Dimensions (WxHxD)	163 x 210 x 101 (mm) without stand and with receiver placed on the phone
Material	ABS Plastic
Installation Mounting	Table - Top
Environmental	
Operating Temperature	0° C to 45°C
Operating Humidity	5 to 95% RH, Non-Condensing
Storage Temperature	-20°C to +70°C

Storage Humidity	5 to 95% RH, Non-Condensing
Weight (Without Foot Stand)	650 gms Approx.

SPARSH VP210 (Standard)

LCD Display	128 x 64 Graphical LCD
VoIP	
VoIP Protocols	SIP v2, SDP, RTP (RFC 2833), SRTP
Network Protocol	IPv4/IPv6, TCP, UDP, DHCP, SNTP, NAT, STUN, HTTP, TLS
Voice CODECS	G.722, G.711 A/μ-Law, G.723, G.729
Call Progress Tones	Dial Tone, Ring Back Tone, Busy Tone, Error Tone, Waiting Tone
Quality of Service	Layer 2 CoS, Layer 3 DIFFServ and TOS
Data Network	LAN Port (RJ45), 10/100/1000 Base T (PoE Optional) PC Port (RJ45), 10/100/1000 Base T
Security	Password Protected Administration
Power Supply	
Input	5VDC(+/-0.25V)@2A through External Adapter (100-240 VAC, 50 - 60 Hz, Optional) and Power-over-Ethernet (PoE)
Power Consumption	1.0 W (Typical)
Mechanical	
Dimensions (WxHxD)	163 x 210 x 101 (mm) without stand and with receiver placed on the phone
Material	ABS Plastic
Installation Mounting	Table - Top
Environmental	
Operating Temperature	0° C to 45°C
Operating Humidity	5 to 95% RH, Non-Condensing
Storage Temperature	-20°C to +70°C
Storage Humidity	5 to 95% RH, Non-Condensing
Weight (Without Foot Stand)	650 gms Approx.

DSS64

Mechanical	
Weight (Product + Stand)	639gm
Product	487gm
Stand	152gm

Dimension in mm [L*B*H]	185 x 200 x 46.50
Environmental Conditions	
Operating Temperature Range	0°C to 45°C
Operating and Storage Humidity	5 to 95%, RH, Non-Condensing
Storage Temperature	-20°C to +70°C
Storage Humidity	0 to 95%, RH, Non-Condensing

DSS532

Terminals Supported	EON510, SPARSH VP510
Source of Power	Powered from the Host Phone
Programmable Keys	32
Stackable	upto 4 modules
LED Indicator	Dual Colour, Red/Blue
Signaling	Proprietary Digital (2B+D)
Interface	Single Pair for Speech, Signaling and Power
Physical Connector	RJ11
Physical Port	IN Port: Connects with the AUX port of the phone or OUT Port of the preceding DSS532 attached.
	OUT Port: Connects with IN Port of the succeeding DSS532 attached.
Installation Option	Table Mount
Mechanical	
Weight (Product + Stand)	235g
Product Weight	218g
Stand Weight	17g
Dimension [L*B*H]	178.7mm * 100mm * 40.5mm
Environmental Conditions	
Operating Temperature Range	0°C to 45°C
Operating Humidity	5 to 95%, RH, Non-Condensing
Storage Temperature	-20°C to +70°C
Storage Humidity	0 to 95%, RH, Non-Condensing

CO (Central Office) / Two-Wire Trunk (TWT)

Signaling	Loop Start
Loop Limit	1200Ω

Off Hook AC Impedance	600/900/Complex
Pulse Dialing	10/20 PPS
DTMF Dialing and Reception	ITU-T Q.23 and Q.24
Return Loss	>18dB
Longitudinal Balance	>50dB
Transmission Level Adjust	Tx Gain: -15dB to +10dB, Rx Gain: -15dB to 10dB
CLI Reception	DTMF, FSK ITU-T V.23 and FSK Bellcore 202
Call Maturity	Delay and Polarity Reversal
Protection	Over Voltage and Over Current Secondary Protection
Physical Connector	RJ45

ISDN BRI

Channels	2B+D
Personality	Network (NT) and Terminal (TE)
Switch Variant	AT&T 4ESS, DMS-100, ETSI NET3, ITU-T Q.921, ITU-T Q.931, NTT INS64, US NI1 (National ISDN1), France VNx
Protection	Solid state (Over Voltage and Over Current) Built-in Secondary Protection
Physical Connector	RJ45 (120Ω)

ISDN PRI

Channels	23B+D and 30B+D
Personality	Network (NT) and Terminal (TE)
Line Coding	AMI/B8ZS for T1 and HDB3 for E1
Framing	ESF for T1 and CEPT1 (with/without CRC) for E1
Switch Variant	AT&T 4ESS, AT&T 5ESS, DMS-100, ETSI NET5, ITU-T Q.921, ITU-T Q.931, NTT INS64, US NI2 (National ISDN 2), QSIC ECMA, France VN
Protection	Solid State (Over Voltage and Over Current) Built-in Secondary Protection
Physical Connector	RJ45 (Impedance Selectable) and fiber optic
Supplementary Services	QSIG ECMA

E1 CAS

Bit Rate	2048 kbps +/-50 ppm

Line Coding	HDB3
Framing	CEPT1 (with/without CRC) with CAS MF
Line Signaling	ITU-T Q.400 - Q. 490
Register Signaling	MFC-R2
Alarms	I.431. G.732, ETSI 300-233
Protection	Solid State (Over Voltage and Over Current) Built-in Secondary Protection
Physical Connector	RJ45 (Impedance Selectable) and fiber optic

T1 RBS

Bit Rate	1544 kbps +/- 50 ppm
Line Coding	AMI and B8ZS
Line Signaling	FXS Loop Start, FXO Loop Start, FXS Ground Start, FXO Ground Start, E&M (Immediate, Wink Start, Wink Start FGD)
Framing	D4, ESF
Digit Dialing	DTMF
Alarms	ANSI T1.231
Performance	ANSI T1.403, ANSI T1.231, AT&T TR54016
Protection	Solid state (Over Voltage and Over Current) Built-in Secondary Protection
Physical Connector	RJ45 (Impedance Selectable)

GSM Trunks

Frequency Band

Module	GSM 850	EGSM 900	DCS 1800	PCS 1900	UMTS 850	UMTS 900	UMTS 1900	UMTS 2100	Rx-diversity	GNSS
Quectel UC20-G	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓
Quectel M95	✓	✓	✓	✓	×	×	×	×	×	×

Compliant	ETSI GSM Phase2/2+
SIM Card	One SIM per GSM Port
SIM Interface	1.8V, 3V
RF Transmission Power	Class 4 (2W) at GSM850MHz and EGSM900MHz band Class 1 (1W) at DCS1800MHz and PCS1900MHz band
RF Sensitivity	Better than -102dBm
Protocol	AT Command Interface

External Antenna	One Antenna per 4 GSM Ports,1.8/3.0*dBi, 50Ω, SMA (Male) Connector, Omni-Directional with cable of 3 meters length

** Depends on GSM Frequency Band.*

3G

Frequency Band

Module	GSM 850	EGSM 900	DCS 1800	PCS 1900	UMTS 850	UMTS 900	UMTS 1900	UMTS 2100	Rx-diversity	GNSS
Quectel UC20-G	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓
Quectel M95	✓	✓	✓	✓	×	×	×	×	×	×

Compliant	ETSI GSM Phase2/2+
SIM Card	One SIM per GSM Port
SIM Interface	1.8V, 3V
Transmission Power	Output Power Class 4 (2W) at GSM850/GSM900 Class 1 (1W) at DCS1800/PCS1900 Class 3 (0.25W) at UMTS 850/900/1900/2100
RF Sensitivity	< -106dBm at GSM850, EGSM900, DCS1800, PCS1900 < -108dBm at WCDMA2100, WCDMA1900 < -106dBm at WCDMA850
Protocol	AT Command Interface
External Antenna	One Antenna per 4 GSM Ports,1.8/3.0*dBi, 50Ω, SMA (Male) Connector, Omni-Directional with cable of 3 meters length

** Depends on GSM Frequency Band.*

Module Supported : Quectel UC20-G		
Standards and directive	**Applied / Complied Harmonized Standards**	
	RE Directive 2014/53 EU, Article 3(1)(a) ● Safety	EN 60950-1:2006+A11:2009+A1:2010+A12:2011+A2:2013
	RE Directive 2014/53 EU, Article 3(1)(a) ● Health	EN 62311:2008
	RE Directive 2014/35 EU, Article 3(1)(b) ● EMC	ETSI EN 301 489-1 V2.1.1, ETSI EN 301-408-52 V1.1.0 ETSI EN 301 489-19 V2.1.0
	RE Directive 2014/53 EU, Article 3(2) ● Radio	EN 301 908-1 V11.1.1, EN 301 908-2 V11.1.1 EN 301 511 V12.5.1*, Draft EN 303 412 V1.1.0*
	* Note: This is non-harmonized radio standard accepted by the RED (Radio Equipment Directive)	
FCC Identifier	XMR201510UC20	
Moducations Supported	GSM: GMSK, 8PSK, WCDMA: BPSK, QPSK, 16QAM GPS: BPSK GLONASS: OFDM	

Module Supported: Quectel M95	
Standards and directive	2014/53/EU Radio Equipment Directive ETSI EN 301 489-1 V1.9.2 (2011-09), ETSI EN 301 489-7 V1.3.1 (2005-11) ETSI EN 301 511 V9.0.2 (2003-03), 3GPP TS 51.010-1 V9.1.0 (2010-03) EN 62311:2008 EN 60950-1:2006+A11:2009+A1:2010+A12:2011+A2:2013
FCC Identifier	XMR201512M95
Modulations Supported	GMSK(EGSM), GMSK(DCS)

Antenna for 2G and 3G

Antenna Type : Fixed Omni Directional Swivel Antenna

Antenna Gain : Dipole = 1.8/2.5 dBi

Antenna Connector : SMA (Male), 50Ω

EG25-G

Module Supported: EG25-G

Technology LTE/VoLTE

Frequency Bands	**LTE - FDD:** B1/B2/B3/B4/B5/B7/B8/B12/B13/B18/B19/B20/ B25/B26/B28 **LTE - TDD**: B38/B39/B40/B41 **WCDMA:** B1/B2/B4/B5/B6/B8/B19 **GSM:** B2/B3/B5/B8
Modulation Supported	QPSK, 16 QAM and 64 QAM

EC25

Module Supported: Quectel EC25-A	
Technology	**LTE / VoLTE** LTE Version : 3GPP E-UTRA Release 10
Frequency Bands	**FDD LTE:** B2/B4/B12 **Uplink Frequency band:** 1850 MHz – 1910 MHz 1710 MHz – 1755 MHz 699 MHz – 716 MHz **Downlink Frequency band:** 1930 MHz – 1990 MHz 2110 MHz – 2155 MHz 729 MHz – 746 MHz **WCDMA: B2/B4/B5** 1,900 MHz 2,100 / 1,700 MHz 850 MHz (for U.S.)
FCC Identifier	XMR201605EC25A
Modulation Supported	QPSK, 16QAM and 64QAM

Module Supported: Quectel EC25-AUT		
Standards and Directives	**SAFETY (RCM)**	EN60950-1:2006/A11:2009/A1:2010/A12:2011/A2:2013
		AS/NZS 60950.1 2011
	EMC	AS/NZS CISPR 32-2015
		ETSI EN 301908-13 (ETSI TS 136 521-1 V13.4.0)
		ETSI EN 301 908-1
	FCC	FCC CFR47 Part 2(2017)/FCC CFR 47 Part 22H (2017)
		AS/CA S042.1:2015, AS/CA S042.4:2015
	HEALTH	AS&NZS 2772.2-2016/ARPANSA standard RPS3-2002
Technology	**LTE / VoLTE** LTE Version : 3GPP E-UTRA Release 10	
Frequency Bands	**FDD LTE: B1/B3/B5/B7/B28** **Uplink Frequency band:** 1920 MHz – 1980 MHz 1710 MHz – 1785 MHz 824 MHz – 849 MHz 2500 MHz – 2570 MHz 703 MHz – 748 MHz **Downlink Frequency band:** 2110 MHz – 2170 MHz 1805 MHz – 1880 MHz 869 MHz – 894MHz 2620 MHz – 2690 MHz 758 MHz – 803 MHz **WCDMA: B1/B5** 2,100 MHz 850 MHz (for U.S.)	
FCC Identifier	NA	
Modulation Supported	(WCDMA)QPSK , (LTE)QPSK 16QAM	

Module Supported: Quectel EC25-E , EC25-E Minipcie		
Standards and Directives	SAFETY	EN 60950-1:2006+A11:2009+A1:2010+A12:2011+A2:2013
	EMC	DRAFT EN301 489-1 V2.2.0
		DRAFT EN301 489-19 V2.1.0
		DRAFT EN301 489-52 V1.1.0
		EN 55032:2015
		EN 55024:2010+A1:2015
	RADIO SPECTRUM	EN301 511 V1235.1
		EN301 908-1 V11.1.1
		EN301 908-1 V11.1.1
		EN301 908-13 V11.1.1
		DRAFT EN303 413 V1.1.0
	HEALTH	EN 62311:2008
Technology	**LTE / VoLTE** LTE Version : 3GPP E-UTRA Release 10	
Frequency Bands	**FDD LTE:** B1/B3/B5/B7/B8/B20 **Uplink Frequency band:** 1920 MHz – 1980 MHz 1710 MHz – 1785 MHz 824 MHz – 849 MHz 2500 MHz – 2570 MHz 880 MHz – 915 MHz 832 MHz – 862 MHz **Downlink Frequency band:** 2110 MHz – 2170 MHz 1805 MHz – 1880 MHz 869 MHz – 894 MHz 2620 MHz – 2690 MHz 925 MHz – 960 MHz 791 MHz – 821 MHz **TDD LTE: B38/B40/B41** 2570 MHz - 2620 MHz 2300 MHz - 2400 MHz 2496 MHz - 2690 MHz **WCDMA: B1/B5/B8** 2,100 MHz 850 MHz (for U.S.) 900 MHz **GSM:** 900/1800 MHz	
FCC Identifier	NA	
Modulation Supported	GMSK , 8PSK , QPSK , 16QAM, 64QAM(DL)	

Module Supported: Quectel EC25-V	
Technology	**LTE / VoLTE** LTE Version : 3GPP E-UTRA Release 10
Frequency Bands	**FDD LTE**: B4/B13 **Uplink Frequency band:** 1710 MHz – 1755 MHz 777 MHz – 787 MHz **Downlink Frequency band:** 2110 MHz – 2155 MHz 746 MHz – 756 MHz
FCC Identifier	XMR201607EC25V
Modulation Supported	QPSK, 16QAM and 64QAM

Antenna for 4G LTE

Antenna Type	:	Monopole Omni Directional Swivel Antenna
Antenna Gain	:	1/3 dBi
Antenna Connector	:	SMA (Male), 50Ω

CDMA

CDMA Frequency Band	800MHz CDMA cellular; 1900MHz PCS
Compliant	CDMA 1x; IS-95A
RUIM Card	One RUIM per CDMA Port
RUIM Interface	1.8V, 3V
Max. Output Power	+25dBm
RF Sensitivity	Typical -107dBm for 800MHz cellular and 1900MHz PCS

VoIP

VoIP Protocols	SIP v2, SIP over TCP, Symmetric RTP, RTCP, 100rel/PRACK
Network Protocol	IPv4/IPv6, TCP, UDP, SNTP, STUN, ARP, ICMP, PPPoE, DHCP, DNS, SMTP
SIP	Maximum 99 SIP Accounts per system, Outbound Proxy Support, Display Name, User Name, Password, URL, Proxy URL, Register URL, Register Interval
NAT/Firewall Support	STUN and NAT Keep Alive

Voice Codecs	G.711 (A-Law, μ-Law), G.723, G.729AB, GSM-FR, iLBC
Line Echo Cancellation	G.168 with 64 /128ms Tail Length
Voice	Dynamic Jitter Buffer (Adaptive), Comfort Noise Generation and Voice Activity Detection
Call Progress Tones	Dial Tone, Ring Back Tone, Busy Tone, Error Tone
Fax	T.38 Relay and Pass Through
Quality of Service	SIP QoS and RTP QoS
Security	MD5 Authentication for SIP, Password Protected Configuration by Admin and User

E&M Trunks

Signaling	Type IV and Type V
Speech Interface	2W/4W
Trunk Seizure Type	Immediate, Immediate+Wink, Seizure Pulse, Seizure Pulse+Wink, Express, Compander Control Signal (CCS)
Address Signaling	Pulse Dial - Pulse 10PPS, Pulse 20PPS Tone Dial - DTMF
Return Loss	20dB
Transhybrid Loss	20dB against configurable balance of 600 ohm or AT&T complex impedance
Transmit Gain	+/- 1.0 dB
Receive Gain	+/- 1.0 dB
Physical Connector	RJ45

Packing List

Verify contents of the package shipped to you with the contents listed below. If any of the items is missing or damaged, contact your Dealer/Reseller.

You can view the documentation of the following products by scanning the QR code printed on the Product Label/ Packaging Label of the respective product.

ETERNITY LENX

Sr. No	Item Name	Qty
1	ETERNITY LENX[a] with Caster Wheels	1
2	Power Cable with a 3-way Socket for 48VDC Power Supply Card	1

a. Factory fitted with the Power Supply Card and CPU Card.

ETERNITY LENX Cards

ETERNITY MENX-LENX CPU Card

Sr. No.	Item	Quantity
1	ETERNITY MENX-LENX CARD CPU	1

ETERNITY LE Power Supply Card

Sr. No.	Item	Quantity
1	ETERNITY LE CARD PS48VDC	1
2	Cable	1

ETERNITY LE ILC48 Card

Sr. No.	Item	Quantity
1	ETERNITY LE CARD ILC48	1
2	Cable with 50 Pin Centronics Connector (Male)	2

ETERNITY LE SLT48 Card

Sr. No.	Item	Quantity
1	ETERNITY LE CARD SLT48	1
2	Cable with 50 Pin Centronics Connector (Male)	2

ETERNITY LE Emerson PS Tray

Sr. No.	Item	Quantity
1	ETERNITY LE Emerson PC Tray	1
2	M4/10 CSK PH Screws (to affix the PS onto the Tray)	3
3	M3/8 CSK PH Screws (to fix the Tray onto the LE rack))	2
4	Tray Mounting Template	1

ETERNITY MENX

Sr. No	Item Name	Qty
1	ETERNITY MENX16SAC IN [a]	1
2	3-pin Power Cord, MC-4 Black	1
3	Screw 50/16 for Wall Mounting for Wall Mounting	4
4	Mounting Template	1

a. Factory fitted with the Power Supply Card and CPU Card.

ETERNITY MENX Cards

ETERNITY MENX-LENX CPU Card

Sr. No.	Item	Quantity
1	ETERNITY MENX-LENX CARD CPU	1

ETERNITY ME Card PS48VDC 500W (Power Supply)

Sr. No.	Item	Quantity
1.	ETERNITY ME CARD PS48VDC-500 Watts	1
2.	3-Way Battery Cable	1

ETERNITY ME Card PSUNI 500W

Sr. No.	Item	Quantity
1.	ETERNITY ME Card PSUNI 500W	1
2.	3-pin Power Cord, MC-4 Black	1

ETERNITY ME Card SLT32

Sr. No.	Item	Quantity
1.	ETERNITY ME Card SLT32	1
2.	Cable with RJ45 connector on one end	8

ETERNITY ME Card SLT16

Sr. No.	Item	Quantity
1.	ETERNITY ME Card SLT16	1
2.	Cable with RJ45 connector on one end	4

ETERNITY ME Card SLT8

Sr. No.	Item	Quantity
1.	ETERNITY ME Card SLT8	1
2.	Cable with RJ45 connector on one end	4

ETERNITY ME Card DKP32

Sr. No.	Item	Quantity
1.	ETERNITY ME Card DKP32	1
2.	Cable with RJ45 connector on one end	8

ETERNITY ME Card DKP16

Sr. No	Item Name	Qty
1	ETERNITY ME CARD DKP16	1
2	Cable with RJ45 connector on one end	4

ETERNITY ME Card DKP8

Sr. No	Item Name	Qty
1	ETERNITY ME CARD DKP8	1
2	Cable with RJ45 connector on one end	4

ETERNITY ME Card CO32

Sr. No	Item Name	Qty
1	ETERNITY ME CARD CO32	1
2	Cable with RJ45 connector on one end	8

ETERNITY ME Card CO16

Sr. No	Item Name	Qty
1	ETERNITY ME CARD CO16	1
2	Cable with RJ45 connector on one end	4

ETERNITY ME Card CO8

Sr. No	Item Name	Qty
1	ETERNITY ME CARD CO8	1
2	Cable with RJ45 connector on one end	4

ETERNITY ME Card CO8+SLT24

Sr. No	Item Name	Qty
1	ETERNITY ME CARD CO8+SLT24	1
2	Cable with RJ45 connector on one end	8

ETERNITY ME Card BRI8

Sr. No	Item Name	Qty
1	ETERNITY ME CARD BRI8	1
2	Cable with RJ45 connectors on both ends	8
3	Mini jumper	16

ETERNITY ME Card BRI4

Sr. No	Item Name	Qty
1	ETERNITY ME CARD BRI4	1
2	Cable with RJ45 connectors on both ends	4
3	Mini jumper	8

ETERNITY ME Card T1E1PRI Dual

Sr. No.	Item	Quantity
1.	ETERNITY ME Card T1E1PRI Dual	1
2.	Cable with RJ45 connectors on both ends	2

ETERNITY ME Card T1E1PRI Single

Sr. No.	Item	Quantity
1.	ETERNITY ME Card T1E1PRI Single	1
2.	Cable with RJ45 connectors on both ends	1

ETERNITY ME Card GSM8

Sr. No.	Item	Quantity
1.	ETERNITY ME Card GSM8	1
2.	GSM Antenna External SMA	2

ETERNITY ME Card GSM4

Sr. No.	Item	Quantity
1.	ETERNITY ME Card GSM4	1
2.	GSM Antenna External SMA	1

ETERNITY ME Card E&M8

Sr. No.	Item	Quantity
1.	ETERNITY ME Card E&M8	1
2.	Cable with RJ45 connector on one end	8

ETERNITY ME Card E&M4

Sr. No.	Item	Quantity
1.	ETERNITY ME Card E&M4	1
2.	Cable with RJ45 connector on one end	4

ETERNITY ME Card ILC32

Sr. No.	Item	Quantity
1.	ETERNITY ME Card SLT32	1
2.	Cable with RJ45 connector on one end	8

ETERNITY ME Card E1FO Dual

Sr. No.	Item	Quantity
1.	ETERNITY ME Card E1FO Dual	1
2.	Cable with RJ45 connectors on both ends	2

ETERNITY ME Card E1FO Single

Sr. No.	Item	Quantity
1.	ETERNITY ME Card E1FO Single	1
2.	Cable with RJ45 connectors on both ends	1

ETERNITY GENX

Sr.No.	Item Name	Qty
1	ETERNITY GENX12S[a]	1
2	3-pin Power Cord, MC-4 Black[b]	1
3	3-pin DC Input Cable[c]	1
4	Screws M 6/30 for Wall Mounting	2
5	Plastic Plugs for Wall Mounting	2
6	Wall Mounting Template	1
7	Side Clamps	2

a. ETERNITY GENX12S AC with factory fitted AC Power Supply and CPU Cards.
ETERNITY GENX12S DC with factory fitted DC Power Supply and CPU Cards.
b. Supplied with ETERNITY GENX12S AC.
c. Supplied with ETERNITY GENX12S DC.

ETERNITY GENX Cards

ETERNITY GE CARD PS48VDC 250W

Sr. No.	Item	Quantity
1.	ETERNITY GE CARD PS48VDC	1
2.	3-pin DC Input Cable	1

ETERNITY GE CARD PSUNI 250W

Sr. No.	Item	Quantity
1.	ETERNITY GE CARD PSUNI	1
2.	3-pin Power Cord, MC-4 Black	1

ETERNITY GENX CARD CPU

Sr. No.	Item	Quantity
1.	ETERNITY GENX CARD CPU	1

ETERNITY GE CARD SLT20

Sr. No.	Item	Quantity
1.	ETERNITY GE CARD SLT20	1
2.	Cable with RJ45 connector on one end	6

ETERNITY GE CARD SLT16

Sr. No.	Item	Quantity
1.	ETERNITY GE CARD SLT16	1
2.	Cable with RJ45 connector on one end	4

ETERNITY GE CARD SLT8

Sr. No.	Item	Quantity
1.	ETERNITY GE CARD SLT8	1
2.	Cable with RJ45 connector on one end	2

ETERNITY GE CARD DKP16

Sr. No.	Item	Quantity
1.	ETERNITY GE CARD DKP16	1
2.	Cable with RJ45 connector on one end	4

ETERNITY GE CARD DKP8

Sr. No.	Item	Quantity
1.	ETERNITY GE CARD DKP8	1
2.	Cable with RJ45 connector on one end	3

ETERNITY GE CARD DKP4+SLT16

Sr. No.	Item	Quantity
1.	ETERNITY GE CARD DKP4+SLT16	1
2.	Cable with RJ45 connector on one end	6

ETERNITY GE CARD DKP4+SLT12

Sr. No.	Item	Quantity
1.	ETERNITY GE CARD DKP4+SLT12	1
2.	Cable with RJ45 connector on one end	6

ETERNITY GE CARD CO2+DKP2+SLT16

Sr. No.	Item	Quantity
1.	ETERNITY GE CARD CO2+DKP2+SLT16	1
2.	Cable with RJ45 connector on one end	6

ETERNITY GE CARD CO2+DKP2+SLT8

Sr. No.	Item	Quantity
1.	ETERNITY GE CARD CO2+DKP2+SLT8	1
2.	Cable with RJ45 connector on one end	4

ETERNITY GE CARD CO4+DKP2+SLT12

Sr. No.	Item	Quantity
1.	ETERNITY GE CARD CO4+DKP2+SLT12	1
2.	Cable with RJ45 connector on one end	6

ETERNITY GE CARD CO4+SLT16

Sr. No.	Item	Quantity
1.	ETERNITY GE CARD CO4+SLT16	1
2.	Cable with RJ45 connector on one end	6

ETERNITY GE CARD CO16

Sr. No.	Item	Quantity
1.	ETERNITY GE CARD CO16	1
2.	Cable with RJ45 connector on one end	4

ETERNITY GE CARD CO8

Sr. No.	Item	Quantity
1.	ETERNITY GE CARD CO8	1
2.	Cable with RJ45 connector on one end	2

ETERNITY GE CARD BRI4

Sr. No.	Item	Quantity
1.	ETERNITY GE CARD BRI4	1
2.	Cable with RJ45 connectors on both ends	4

ETERNITY GE CARD T1E1PRI Single

Sr. No.	Item	Quantity
1.	ETERNITY GE CARD T1E1PRI Single	1
2.	Cable with RJ45 connectors on both ends	1

ETERNITY GE CARD E1FOPRI Single

Sr. No.	Item	Quantity
1.	ETERNITY GE CARD E1FOPRI Single	1
2.	Cable with RJ45 connectors on both ends	1

ETERNITY GE CARD GSM4

Sr. No.	Item	Quantity
1.	ETERNITY GE CARD GSM4	1
2.	GSM Antenna External SMA	1

ETERNITY GE CARD E&M4

Sr. No.	Item	Quantity
1.	ETERNITY GE CARD E&M4	1
2.	Cable with RJ45 connector on one end	4

ETERNITY GE CARD ILC20

Sr. No.	Item	Quantity
1.	ETERNITY GE CARD ILC20	1
2.	Cable with RJ45 connector on one end	6

ETERNITY PENX

Sr.No.	Item Name	Qty
1	ETERNITY PENX 6S	1
2	3-pin Power Cord	1
3	Screws for Wall Mounting	2
4	Plastic Plugs for Wall Mounting	2
5	Mounting Template	1

The following materials can be purchased separately if required.

Sr. No.	Item	Quantity
1.	Side Clamps for Rack Mounting	2
2.	Screw for Side Clamp	4

ETERNITY PE Card SLT8

Sr. No.	Item	Quantity
1.	ETERNITY PE Card SLT8	1
2.	Cable with RJ45 connector on one end	3

ETERNITY PE Card SLT4

Sr. No.	Item	Quantity
1.	ETERNITY PE Card SLT4	1
2.	Cable with RJ45 connector on one end	1

ETERNITY PE Card DKP8

Sr. No.	Item	Quantity
1.	ETERNITY PE Card DKP8	1
2.	Cable with RJ45 connector on one end	3

ETERNITY PE Card DKP2 + SLT6

Sr. No.	Item	Quantity
1.	ETERNITY PE Card DKP2+SLT6	1
2.	Cable with RJ45 connector on one end	3

ETERNITY PE Card CO8

Sr. No.	Item	Quantity
1.	ETERNITY PE Card CO8	1
2.	Cable with RJ45 connector on one end	3

ETERNITY PE Card CO4 + SLT4

Sr. No.	Item	Quantity
1.	ETERNITY PE Card CO4+SLT4	1
2.	Cable with RJ45 connector on one end	3

ETERNITY PE Card CO2 + SLT6

Sr. No.	Item	Quantity
1.	ETERNITY PE Card CO2+SLT6	1
2.	Cable with RJ45 connector on one end	3

ETERNITY PE Card CO2+DKP2+SLT4

Sr. No.	Item	Quantity
1.	ETERNITY PE Card CO2+DKP2+SLT4	1
2	Cable with RJ45 connector on one end	3

ETERNITY PE Card T1E1PRI Single

Sr. No.	Item	Quantity
1.	ETERNITY PE Card T1E1PRI Single	1
2.	Cable with RJ45 connectors on both ends	1

ETERNITY PE Card GSM4

Sr. No.	Item	Quantity
1.	ETERNITY PE Card GSM4	1
2.	GSM Antenna External SMA	1

ETERNITY PE Card VoIP64

Sr. No.	Item	Quantity
1.	ETERNITY PE Card VoIP64	1
2.	Cable with RJ45 connectors on both ends	1

ETERNITY PE Card VMS16

Sr. No.	Item	Quantity
1	ETERNITY PE Card VMS16	1
2.	4 GB USB Flash Drive[a]	1
3.	User Card	1

a. Factory fitted.

EON48

Sr. No.	Item	Quantity
1.	Phone, Handset and Spring Cord	1
2.	Line Cord	1
3.	Foot Stand	1
4.	Wall Mounting Template	1
5.	Self tapping screws and screw grips for wall mounting	2

EON310

Sr. No.	Item	Quantity
1.	Phone, Handset and Spring Cord	1
2.	Line Cord	1
3.	Foot Stand	1
4.	Wall Mounting Template	1
5.	Self tapping screws and screw grips for wall mounting	2

EON510

Sr. No.	Item	Quantity
1.	Phone, Handset and Spring Cord	1
2.	Line Cord	1
3.	Foot Stand	1
4.	Wall Mounting Template	1
5.	Self tapping screws and screw grips for wall mounting	2

SPARSH VP248 (Extended/Standard)

Sr. No.	Item	Quantity
1.	Phone, Handset and Spring Cord	1
2.	Ethernet Cable	1
3.	Foot Stand	1
4.	Wall Mounting Template	1

Sr. No.	Item	Quantity
5.	Self tapping screws and screw grips for wall mounting	2

SPARSH VP310 (Extended)

Sr. No.	Item	Quantity
1.	Phone, Handset and Spring Cord	1
2.	Ethernet Cable	1
3.	Foot Stand	1
4.	Wall Mounting Template	1
5.	Self tapping screws and screw grips for wall mounting	2

SPARSH VP330 (Extended)

Sr. No.	Item	Quantity
1.	Phone, Handset and Spring Cord	1
2.	Ethernet Cable	1
3.	Foot Stand	1
4.	Wall Mounting Template	1
5.	Self tapping screws and screw grips for wall mounting	2

SPARSH VP510(Extended)

Sr. No.	Item	Quantity
1.	Phone, Handset and Spring Cord	1
2.	Ethernet Cable	1
3.	Foot Stand	1
4.	Wall Mounting Template	1
5.	Self tapping screws and screw grips for wall mounting	2

SPARSH VP110

Sr. No.	Item	Quantity
1.	Matrix SPARSH VP110 IP Phone Unit	1

2.	Phone Stand	1
3.	Handset & Handset Cord	1
4.	Ethernet Cable	1
5.	Power Adapter	1

SPARSH VP710

Sr. No.	Item	Quantity
1.	Matrix SPARSH VP710 IP Phone Unit	1
2.	Phone Stand	1
3.	Handset & Handset Cord	1
4.	Ethernet Cable	1
5.	Camera	1
6.	Power Adapter (Optional)	1
7.	Wall Mount Bracket (Optional)	1

SPARSH VP210 (Extended/Standard)

Sr. No.	Item	Quantity
1.	Phone, Handset and Spring Cord	1
2.	Ethernet Cable	1
3.	Foot Stand	1

DSS64

Sr. No.	Item	Quantity
1.	DSS64 Console	1
2.	RJ45 Cable	1

DSS532

Sr. No.	Item	Quantity
1.	DSS532 Console Unit	1
2.	DSS Extender	1
3.	Foot Stand	1
4.	RJ11 Cable	1

Sr. No.	Item	Quantity
5	Clamps (2 DSS-Phone Clamps and 2 DSS-DSS Clamps)	4
6.	DSS532 Quick Installation Guide (printed copy)	1

EONSOFT

Sr. No.	Item	Quantity
1.	EONSOFT Dongle	1
2.	Handset with Spring Cord	1
3.	Communication Cable	1
4.	Screw m7/30	2
5.	Screw Grip	2
6.	Mounting Template	1

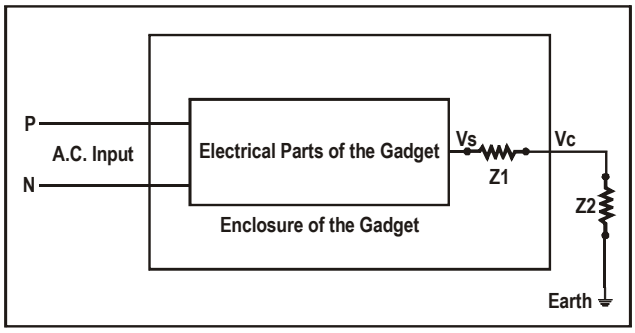
PPM4

Sr. No.	Item	Quantity
1.	Matrix PPM4	1
2.	Wall Mounting Screws	2
3.	Wall Mounting Screw Grips	2
4.	Mounting Template	1

How to Make the Telecom Earth

The Earth (Ground) is the most important safety procedure to prevent electrical shocks and fires. It protects from lightning strikes, electrical transients, static discharges, electromagnetic interference and electrical hazards.

A proper earth must be in place to protect people and the system. The following explanation shows how a perfect electrical earth can save lives.



In the above diagram, Vc = VS *Z2 / Z1 + Z2
Where,

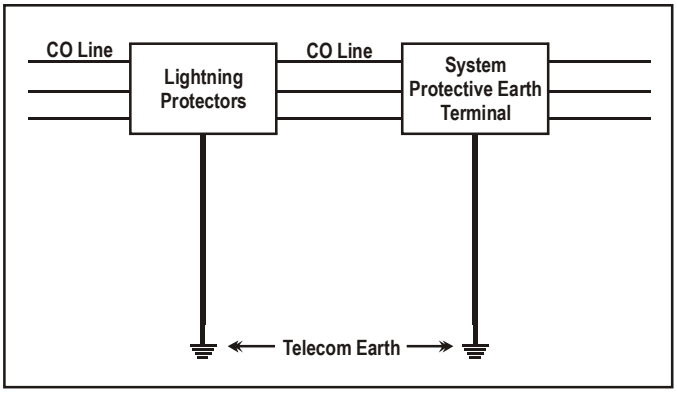
- Z1 is the stray impedance between the electrical parts of the Gadget and the Chassis.
- Z2 is the stray impedance between the Chassis and the Earth.
- If Z2 = 0 then VC = 0

This formula implies that if the impedance between the Chassis and the Earth is reduced to 0 then the Voltage on the Chassis, that is, VC, would be Zero and hence any person touching the enclosure will not get an electric shock. Hence Z2 should be made Zero.

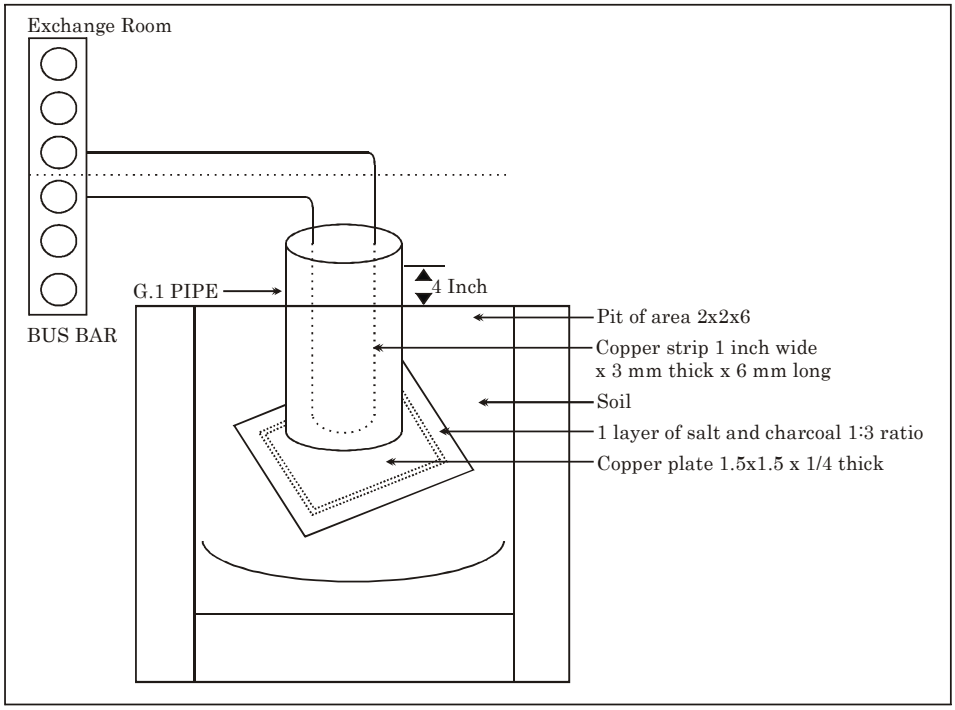
This can be done by providing a perfect earth to the electrical equipment.

It is recommended that you provide a dedicated earth for the *ETERNITY GENX*/any other telecom equipment. This dedicated earth is called the Telecom Earth (Ground).

Providing a separate Telecom Earth to the telecom equipment eliminates the possibility of any back-voltage on the earth.



How to make a perfect Earth



- Dig a pit of area 2 feet x 2 feet x 6 feet (L x B x D).
- Get a copper plate of size 1.5 feet x 1.5 feet x 0.25 feet.
- Connect a copper strip of size 1-inch wide, 3 mm thick and 6 feet length at the center of the copper plate by welding or nuts and bolts.
- Insert a G.I. pipe onto the copper strip till it reaches the copper plate.
- Place this set up into the pit. Make sure that at least 4 inches of the G.I. pipe is above the ground level.
- Fill the bottom of the pit with a 1-inch layer of charcoal and salt in the proportion of 3:1 (3 parts charcoal, 1 part salt) and then cover with the soil.
- Connect a bare 14 SWG copper wire (double) on the top of the copper strip and run it to the exchange room and connect it on the bus bar.
- The Bus bar is a copper strip, 4 inches long with 6 screws and nuts mounted on it. It has to be fixed on the wall in the exchange room.
- The earth wire of the Primary Protection Modules (PPM) should be connected to this Bus bar.
- Water the earth at regular intervals.

Connecting ETERNITY GENX to the Earth

Follow the instructions given below to connect the ETERNITY GENX to the Earth. The instructions are given with reference to the above diagram:

1. Loosen the screw.
2. Insert an earthing wire into the lug.
3. Crimp the earth wire with an appropriate tool.
4. Tighten the screw.
5. Connect the earthing wire to the earth.

*Make sure you use an earthing wire that has a conductor with a **cross-sectional area greater than 1.0 mm2 or AWG less than or equal to 16 AWG** with Green and Yellow insulation.*

- *Make sure you comply with all applicable laws, regulations and guidelines.*
- *Proper earthing is very important to protect the ETERNITY GENX from external noise and to reduce the risk of electrocution in the event of a lightning strike.*
- *The AC cable's earthing pin may not be enough to protect the ETERNITY GENX from external noise and lightning strikes. A permanent connection must be made between earth and the earth terminal of the main unit.*

VMS Prompts

Given below are the prompts which are already recorded in English language and are provided to you by default. Along with the prompts, the Prompt Names and File Names are also listed below. You must use these prompts as a reference while recording and uploading the prompts for other languages. For further information, refer “Prompts Management”.

Greetings		
Prompt Name	**File Name**	**Prompts**
Greeting 01	Morning.wav	Good Morning!
Greeting 02	Afternoon.wav	Good Afternoon!
Greeting 03	Evening.wav	Good Evening!

Auto Attendant Prompts		
Prompt Name	**File Name**	**Prompts**
Auto Attendant 01	Working.wav	Welcome! Please dial the extension number or To dial by name press 6. To leave a message press 7. To access your Personal Mailbox press 8. For further assistance press 9. To disconnect the call press # (HASH).
Auto Attendant 02	Break.wav	Thank you for calling! We are closed at this time. To leave a message press 7. To access your Personal Mailbox press 8. For further assistance press 9. To disconnect the call press # (HASH).
Auto Attendant 03	Nonworking.wav	Thank you for calling! We are closed at this time. To leave a message press 7. To access your Personal Mailbox press 8. For further assistance press 9. To disconnect the call press # (HASH).
Auto Attendant 04	WH_Hotel.wav	Welcome! Please dial the room number or to dial by name press 6. To leave a message press 7. To access your Personal Mailbox press 8. For further assistance press 9. To disconnect **the call** press # (HASH).
Auto Attendant 05	BH_Hotel.wav	Welcome! Please dial the room number or to dial by name press 6. To leave a message press 7. To access your Personal Mailbox press 8. For further assistance press 9. To disconnect **the call** press # (HASH).

Auto Attendant 06	NWH_Hotel.wav	Welcome! Please dial the room number or to dial by name press 6. To leave a message press 7. To access your Personal Mailbox press 8. For further assistance press 9. To disconnect **the call** press #. (HASH)
Auto Attendant 07	WH_USA.wav	Welcome! Please dial the extension number or to dial by name press 6. To leave a message press 7. To access your Personal Mailbox press 8. To connect to the operator press 0. To disconnect the call press £. (POUND)
Auto Attendant 08	BH_USA.wav	Thank you for calling! We are closed at this time. To leave a message press 7. To access your Personal Mailbox press 8. For further assistance press 9. To disconnect the call press £. (POUND)
Auto Attendant 09	NWH_USA.wav	Thank you for calling! We are closed at this time. To leave a message press 7. To access your Personal Mailbox press 8. For further assistance press 9. To disconnect the call press £. (POUND)
Auto Attendant 10	WH_Hotel_USA.wav	Welcome! Please dial the room number or to dial by name press 6. To leave a message press 7. To access your Personal Mailbox press 8. For further assistance press 0. To disconnect the call press £. (POUND)
Auto Attendant 11	BH_Hotel_USA.wav	Welcome! Please dial the room number or to dial by name press 6. To leave a message press 7. To access your Personal Mailbox press 8. For further assistance press 0. To disconnect the call press £. (POUND)
Auto Attendant 12	NWH_Hotel_USA.wav	Welcome! Please dial the room number or to dial by name press 6. To leave a message press 7. To access your Personal Mailbox press 8. For further assistance press 0. To disconnect the call press £. (POUND)
Auto Attendant 13	Holiday.wav	Thank you for calling! We are closed due to holiday. To leave a message press 7. For further assistance press 9. To disconnect the Call press #. (HASH)

Auto Attendant 14	Holiday_Hotel.wav	Welcome! Please dial the room number or to dial by name press 6. To leave a message press 7. To access your Personal Mailbox press 8. For further assistance press 9. To disconnect the Call press #. (HASH)
Auto Attendant 15	Holiday_USA.wav	Thank you for calling! We are closed due to holiday. To leave a message press 7. To access your Personal Mailbox press 8. For further assistance press 0. To disconnect the Call press £. (POUND)
Auto Attendant 16	Holiday_Hotel_USA.wav	Welcome! Please dial the room number or to dial by name press 6. To leave a message press 7. To access your Personal Mailbox press 8. For further assistance press 0. To disconnect the Call press £. (POUND)

Number Dialing		
Prompt Name	**File Name**	**Prompts**
Number Dialing 01	Entextn.wav	Please enter the extension number.
Number Dialing 02	DialName.wav	Please enter the first three letters of the name.
Number Dialing 03	MBAccess.wav	Please enter your extension number.
Number Dialing 04	Leavemsg.wav	Please enter the extension number for which you wish to leave message.
Number Dialing 05	LeavemsgH.wav	Please enter the room number or the extension number for which you wish to leave message.
Number Dialing 06	MsgdestI.wav	Please enter the destination and press hash [#] to end.
Number Dialing 07	MsgdestU.wav	Please enter the destination and press Pound [£] to end.

No Digit Dialed		
Prompt Name	**File Name**	**Prompts**
No Digit Dialed 01	NoDigitDialed.wav	Sorry, You have not dialed any digit.
No Digit Dialed 02	NoOptionSelected.wav	Sorry, You have not selected any option.
No Digit Dialed 03	NoInput.wav	Sorry, You have not entered any input.
No Digit Dialed 04	NoDestination.wav	Sorry, this is an invalid input.

Invalid Digit Dialed		
Prompt Name	**File Name**	**Prompts**
Invalid Digit Dialed 01	InvalidInput.wav	Sorry, this is an invalid input.
Invalid Digit Dialed 02	InvalidDigit.wav	Sorry, this is an Invalid digit.

Invalid Digit Dialed 03	Invalidno.wav	Sorry, this is an Invalid Number.
Invalid Digit Dialed 04	NoMatchFound.wav	No match found.
Invalid Digit Dialed 05	InvalidDest.wav	Invalid recipient. Please enter all recipients again.

Expiry of Count		
Prompt Name	**File Name**	**Prompts**
Expiry Of Count 01	RetryCountOver.wav	Sorry! Maximum attempts exceeded.

Call Transfer Type		
Prompt Name	**File Name**	**Prompts**
Call Transfer Type 01	PlsHold.wav	Please hold.
Call Transfer Type 02	XfrCall.wav	Please hold, transferring the call.
Call Transfer Type 03	Xfrcallto.wav	Please hold, transferring the call to
Call Transfer Type 04	XfrcalltoOpr.wav	Please hold, transferring the call to Operator.
Call Transfer Type 05	XferMailbox.wav	Please hold, transferring the call to mailbox.
Call Transfer Type 06	XferMailboxof.wav	Please hold, transferring the call to mailbox of

Call Transfer Unsuccessful		
Prompt Name	**File Name**	**Prompts**
Call Transfer Unsuccessful 01	Noreply.wav	Sorry! The person is unavailable to take your call right now.
Call Transfer Unsuccessful 02	Extbusy.wav	Sorry! The person you are trying to call is busy.
Call Transfer Unsuccessful 03	Noattnd.wav	Sorry! The call could not be attended.

MoH		
Prompt Name	**File Name**	**Prompts**
MoH 1	Holdmusic.wav	<Music>

Disconnect		
Prompt Name	**File Name**	**Prompts**
Disconnect 01	Thankyou.wav	Thank you for your call.

Language		
Prompt Name	**File Name**	**Prompts**
Language 01	english.wav	For English,

Dial by Name		
Prompt Name	**File Name**	**Prompts**
Dial by Name 01	PreName.wav	The following matches are found
Dial by Name 02	selectname.wav	To select the name,
Dial by Name 03	nextname.wav	To skip to the next name,
Dial by Name 04	repeatlastname.wav	To repeat the previous name,
Dial by Name 05	repeatnames.wav	To repeat all the names,
Dial by Name 06	enternameagain.wav	To re-enter the name,
Dial by Name 07	previousmenu.wav	To return to the previous menu,
Dial by Name 08	disconnect.wav	To disconnect,
Dial by Name 09	Selected.wav	Name is selected,
Dial by Name 10	NameSelected.wav	Selected name is,
Dial by Name 11	ConfirmName.wav	You have selected,
Dial by Name 12	confirm.wav	To confirm,
Dial by Name 13	reenter.wav	To re-enter,

Call Transfer		
Prompt Name	**File Name**	**Prompts**
Call Transfer 01	RecName.wav	Please Record your name after the beep and press any digit to end
Call Transfer 02	Callfrom.wav	Hello! There is a call from,
Call Transfer 03	acceptcall.wav	To accept the call,
Call Transfer 04	reject.wav	To reject the call,
Call Transfer 05	rejectbusy.wav	To reject the call as busy,
Call Transfer 06	rejectnoreply.wav	To reject the call as no reply,
Call Transfer 07	Leavemsg1.wav	To leave a message,
Call Transfer 08	XfertoOperator.wav	To transfer the call to operator,
Call Transfer 09	XfertoAssitance.wav	To transfer the call to assistant,
Call Transfer 10	XfertoAlternate.wav	To transfer the call to alternate number,
Call Transfer 11	DialExtn.wav	To dial an extension,
Call Transfer 12	XferMainMenu.wav	To return to Main Menu,
Call Transfer 13	XferPreviousMenu.wav	To return to Previous Menu,
Call Transfer 14	StayonHold.wav	To stay on hold,
Call Transfer 15	disconnect.wav	To disconnect,
Call Transfer 16	nonumber.wav	Number not programmed.

Message Record		
Prompt Name	**File Name**	**Prompts**
Message Record 01	Recmsg.wav	Please Record your Message after the beep and press any digit to end.
Message Record 02	RecMsgStopCode.wav	Please Record your message after the beep and press

Message Record 03	RecMsgEnd.wav	To end,
Message Record 04	MsgLeaveConfirm.wav	Message sent as,
Message Record 05	Rerecord.wav	To re-record the message,
Message Record 06	Confirm.wav	To confirm the message as normal,
Message Record 07	urgent.wav	To confirm the message as urgent,
Message Record 08	private.wav	To confirm the message as private,
Message Record 09	listenmsg.wav	To play the recorded message,
Message Record 10	appendmsg.wav	To append to the recorded message,
Message Record 11	disconnect.wav	To disconnect,
Message Record 12	normalset.wav	Normal
Message Record 13	urgentset.wav	Urgent
Message Record 14	privateset.wav	Private
Message Record 15	securedset.wav	Secured
Message Record 16	NoMailbox.wav	Mailbox not assigned.
Message Record 17	Mailboxfull.wav	Sorry! your message cannot be delivered as Mailbox is full.
Message Record 18	Memoryfull.wav	Sorry! your message cannot be delivered as System Storage is full.
Message Record 19	appendfail.wav	Sorry! message cannot be appended as System Storage is full.
Message Record 20	urgentandprivate.wav	To confirm the message as urgent and private,

Message Send Forward		
Prompt Name	**File Name**	**Prompts**
Message Send Forward 01	Recmsg.wav	Please Record your Message after the beep and press any digit to end
Message Send Forward 02	RecMsgStopCode.wav	Please Record your message after the beep and press
Message Send Forward 03	RecMsgEnd.wav	to end
Message Send Forward 04	MsgSendConfirm.wav	Message sent as
Message Send Forward 05	reenter.wav	To re-enter
Message Send Forward 06	Confirm.wav	To confirm
Message Send Forward 07	commentatstart.wav	To forward the message with comment before the message
Message Send Forward 08	commentatend.wav	To forward the message with comment after the message
Message Send Forward 09	nocomment.wav	To forward the message without comment
Message Send Forward 10	Rerecord.wav	To re-record the message
Message Send Forward 11	Confirm.wav	To confirm the message as normal
Message Send Forward 12	urgent.wav	To confirm the message as urgent
Message Send Forward 13	private.wav	To confirm the message as private
Message Send Forward 14	listenmsg.wav	To play the recorded message
Message Send Forward 15	appendmsg.wav	To append to the recorded message
Message Send Forward 16	Requestreceipt.wav	To request read receipt
Message Send Forward 17	ignorereceipt.wav	To ignore read receipt

Message Send Forward 18	previousmenu.wav	To return to previous menu
Message Send Forward 19	normalset.wav	Normal
Message Send Forward 20	urgentset.wav	Urgent
Message Send Forward 21	privateset.wav	Private
Message Send Forward 22	securedset.wav	Secured
Message Send Forward 23	NumberCollected.wav	Recipient number entered is
Message Send Forward 24	FwdMsgFail.wav	Sorry! Message cannot be forwarded as message length has exceeded
Message Send Forward 25	Pending.wav	Sorry! the message cannot be sent
Message Send Forward 26	NoDigitDialed.wav	You have not dialed any digit
Message Send Forward 27	nextnumber.wav	Enter the next recipient number
Message Send Forward 28	NoMailbox.wav	Sorry! Mailbox is not assigned
Message Send Forward 29	appendfail.wav	Sorry, message cannot be appended as System Storage is full
Message Send Forward 30	Mailboxfull.wav	Sorry, your message cannot be delivered as Mailbox is full
Message Send Forward 31	enterDDMMYY24.wav	Enter Delivery Date in DD MM YYYY format and time in Twenty four hour format
Message Send Forward 32	enterDDMMYY12.wav	Enter Delivery Date in DD MM YYYY format and time in Twelve hour format. For AM, press 1, for PM, press 2.
Message Send Forward 33	enterMMDDYY24.wav	Enter Delivery Date in MM DD YYYY format and time in Twenty four hour format
Message Send Forward 34	enterMMDDYY12.wav	Enter Delivery Date in MM DD YYYY format and time in Twelve hour format. For AM, press 1, for PM, press 2.
Message Send Forward 35	checkfuturedelivery.wav	To check the last entered Future Delivery
Message Send Forward 36	cancelfuturedelivery.wav	To cancel the last entered Future Delivery
Message Send Forward 37	futuredelconf.wav	To confirm press 1. To cancel press any digit
Message Send Forward 38	DiscardMsg.wav	To discard the message and return to previous menu

Mailbox Access		
Prompt Name	**File Name**	**Prompts**
Mailbox Access 01	Enterpwd.wav	Please Enter your mailbox password.
Mailbox Access 02	Youhave.wav	You have,
Mailbox Access 03	Newmsg.wav	New message.
Mailbox Access 04	Newmsgs.wav	New messages.
Mailbox Access 05	Nonewmsg.wav	You have no new messages.
Mailbox Access 06	Urgentmsg.wav	Urgent message
Mailbox Access 07	Urgentmsgs.wav	Urgent messages
Mailbox Access 08	Nooldmsg.wav	You have no old messages.
Mailbox Access 09	PerMB.wav	In your personal mailbox
Mailbox Access 10	DeptMB.wav	In your department group mailbox

Mailbox Access 11	MessageCount.wav	Message
Mailbox Access 12	Msgrecon.wav	This message was recorded on,
Mailbox Access 13	ConvBetn.wav	This conversation was between,
Mailbox Access 14	Callnum.wav	And calling number was,
Mailbox Access 15	MsgreadBy.wav	This message was read by,
Mailbox Access 16	DeptOrStn.wav	Press '1' to access personal mailbox, Press '2' to access department group mailbox
Mailbox Access 17	RecdCncl.wav	Sorry! Your message cannot be delivered.
Mailbox Access 18	MB80Full.wav	Your Mailbox is 80% Full. Please Delete old messages of your mailbox.
Mailbox Access 19	MBFull.wav	Your Mailbox is Full. Please Delete old messages of your Mailbox.
Mailbox Access 20	Invalpwd.wav	Sorry! This is an invalid password
Mailbox Access 21	MBblocked.wav	Sorry! Mailbox is currently in use. Please try again later.
Mailbox Access 22	NoMailbox.wav	Sorry! Mailbox is not assigned
Mailbox Access 23	SystemMemFull.wav	System Storage is Full. Please Delete some stored messages.
Mailbox Access 24	privatemsg.wav	Private Message
Mailbox Access 25	urgentprivatemsg.wav	Urgent and Private message
Mailbox Access 26	Disconnect.wav	Thank you for your call.
Mailbox Access 27	Messagefor.wav	Message for extension,
Mailbox Access 28	MsgDelete.wav	Message deleted.
Mailbox Access 29	MsgSaveCnf.wav	Your message has been saved as new.
Mailbox Access 30	recorddel.wav	Recording erased.
Mailbox Access 31	Numdel.wav	Number erased.

Mailbox Access Menu		
Prompt Name	**File Name**	**Prompts**
Mailbox Access Menu 01	listennewmsg.wav	To listen to a new message,
Mailbox Access Menu 02	listenoldmsg.wav	To listen to an old message,
Mailbox Access Menu 03	sendmsg.wav	To send a message,
Mailbox Access Menu 04	Mbmgnt.wav	For mailbox management,
Mailbox Access Menu 05	replaymsg.wav	To re-play the message,
Mailbox Access Menu 06	playmsgdetails.wav	For Date and Time of the message,
Mailbox Access Menu 07	replymsg.wav	To reply the message,
Mailbox Access Menu 08	deletemsg.wav	To delete the message,
Mailbox Access Menu 09	listennextmsg.wav	To Play the next message,
Mailbox Access Menu 10	forwardmsg.wav	To forward the message,
Mailbox Access Menu 11	msgasnew.wav	To mark the message as unread,
Mailbox Access Menu 12	Mbname.wav	For mailbox name,
Mailbox Access Menu 13	MsgRedirection.wav	For message redirection,
Mailbox Access Menu 14	DeleteAllOldmsgs.wav	To delete all old messages of your mailbox,
Mailbox Access Menu 15	RecMBgrt.wav	For mailbox greetings,
Mailbox Access Menu 16	Assistance.wav	For assistant number,

Mailbox Access Menu 17	Personal.wav	For personal number,
Mailbox Access Menu 18	SetMsgRedirection.wav	To set message redirection,
Mailbox Access Menu 19	CancelMsgRedirection.wav	To cancel message redirection,
Mailbox Access Menu 20	MsgRedirectNo.wav	Enter the recipient number,
Mailbox Access Menu 21	PersonalGrt.wav	For personal greetings,
Mailbox Access Menu 22	WHGrt.wav	For working hours greeting,
Mailbox Access Menu 23	BHGrt.wav	For break hours greeting,
Mailbox Access Menu 24	NWHGrt.wav	For non-working hours greeting,
Mailbox Access Menu 25	ConditionalGrt.wav	For conditional greetings,
Mailbox Access Menu 26	BusyGrt.wav	For busy,
Mailbox Access Menu 27	NoReplyGrt.wav	For no-reply,
Mailbox Access Menu 28	UnconditionalGrt.wav	For unconditional,
Mailbox Access Menu 29	Record.wav	To record,
Mailbox Access Menu 30	Play.wav	To play,
Mailbox Access Menu 31	Erase.wav	To erase,
Mailbox Access Menu 32	EnterNum.wav	To enter number,
Mailbox Access Menu 33	PlayNum.wav	To play number,
Mailbox Access Menu 34	ClearNum.wav	To clear number,
Mailbox Access Menu 35	EnterAssistanceNum.wav	Enter the assistant extension number.
Mailbox Access Menu 36	EnterPersonalNum.wav	Enter the personal number and Press Hash [#] to end
Mailbox Access Menu 37	previousmenu.wav	To return to Previous Menu
Mailbox Access Menu 38	Recname.wav	Please Record your name after the beep and press any digit to end.
Mailbox Access Menu 39	Recgreeting.wav	Please Record the greeting after the beep and press any digit to end.
Mailbox Access Menu 40	DelAllOldCnf.wav	You are about to delete all read messages of your mailbox. To proceed, Press 1, to Cancel, Press any digit.
Mailbox Access Menu 41	DelAllOldDone.wav	All old messages are deleted.
Mailbox Access Menu 42	MsgRedirectSet.wav	Message redirection set.
Mailbox Access Menu 43	MsgRedirectCancel.wav	Message redirection canceled.
Mailbox Access Menu 44	Numbersaved.wav	Number saved.
Mailbox Access Menu 45	Numberdelete.wav	Number deleted.
Mailbox Access Menu 46	Invalidno.wav	Sorry, this is an invalid number.
Mailbox Access Menu 47	NoDigitDialed.wav	Sorry, you have not dialed any digit.
Mailbox Access Menu 48	DelAllMsgsCnf.wav	You are about to delete all messages of your mailbox. To proceed press 1. To cancel, press any digit.
Mailbox Access Menu 49	DelAllMsgsDone.wav	All messages are deleted.
Mailbox Access Menu 50	DelAllMsgs.wav	To delete all messages of your mailbox,
Mailbox Access Menu 51	Recordingdelete.wav	Recorded file deleted.

Number and Month		
Prompt Name	**File Name**	**Prompts**

Number and Month 01	Num0.wav	Zero
Number and Month 02	Num1.wav	One
Number and Month 03	Num2.wav	Two
Number and Month 04	Num3.wav	Three
Number and Month 05	Num4.wav	Four
Number and Month 06	Num5.wav	Five
Number and Month 07	Num6.wav	Six
Number and Month 08	Num7.wav	Seven
Number and Month 09	Num8.wav	Eight
Number and Month 10	Num9.wav	Nine
Number and Month 11	Num10.wav	Ten
Number and Month 12	Num11.wav	Eleven
Number and Month 13	Num12.wav	Twelve
Number and Month 14	Num13.wav	Thirteen
Number and Month 15	Num14.wav	Fourteen
Number and Month 16	Num15.wav	Fifteen
Number and Month 17	Num16.wav	Sixteen
Number and Month 18	Num17.wav	Seventeen
Number and Month 19	Num18.wav	Eighteen
Number and Month 20	Num19.wav	Nineteen
Number and Month 21	Num20.wav	Twenty
Number and Month 22	Num30.wav	Thirty
Number and Month 23	Num40.wav	Forty
Number and Month 24	Num50.wav	Fifty
Number and Month 25	Num60.wav	Sixty
Number and Month 26	Num70.wav	Seventy
Number and Month 27	Num80.wav	Eighty
Number and Month 28	Num90.wav	Ninety
Number and Month 29	Num100.wav	Hundred
Number and Month 30	Num1000.wav	Thousand
Number and Month 31	Month1.wav	January
Number and Month 32	Month2.wav	February
Number and Month 33	Month3.wav	March
Number and Month 34	Month4.wav	April
Number and Month 35	Month5.wav	May
Number and Month 36	Month6.wav	June

Number and Month 37	Month7.wav	July
Number and Month 38	Month8.wav	August
Number and Month 39	Month9.wav	September
Number and Month 40	Month10.wav	October
Number and Month 41	Month11.wav	November
Number and Month 42	Month12.wav	December
Number and Month 43	Hash.wav	Hash
Number and Month 44	Pound.wav	Pound
Number and Month 45	Star.wav	Star
Number and Month 46	asterisk.wav	Asterisk

Alarm		
Prompt Name	**File Name**	**Prompts**
Alarm 01	EntertimeI.wav	Enter the time, HH:MM in Twenty four hour format. To cancel all alarms, press Pound [£]
Alarm 02	EntertimeU.wav	Enter the time, HH:MM in Twelve hour format. For AM, press 1. For PM, press 2 To cancel all alarms, press Pound [£]
Alarm 03	WakeupCancel.wav	Your all wake up alarms are canceled.
Alarm 04	SetOnceDaily.wav	To set once press '1'. To set daily press '2'
Alarm 05	WakeupVeri.wav	You have set wake up alarm for
Alarm 06	DailyWakeupVeri.wav	You have set daily wake up alarm for
Alarm 07	WakeupSet.wav	Your wake up alarm is set.
Alarm 08	DailyWakeupSet.wav	Your daily wake up alarm is set.
Alarm 09	Am.wav	A.M.
Alarm 10	Pm.wav	P.M.
Alarm 11	AlarmConf.wav	To confirm press 1, To re-enter press 2
Alarm 12	AlarmDateI.wav	Enter the Date in DD MM YYYY format. To cancel all reminders, press Pound [£]
Alarm 13	AlarmDateU.wav	Enter the Date in MM DD YYYY format. To cancel all reminders, press Pound [£]
Alarm 14	ReminderCancel.wav	Your all reminders are canceled.
Alarm 15	ReminderVeri.wav	You have set reminder for
Alarm 16	ReminderSet.wav	Your reminder is set
Alarm 17	Alarmnoset.wav	Sorry! Your wake up alarm cannot be set. Please call operator for further assistance.
Alarm 18	Remindernoset.wav	Sorry! Your reminder cannot be set. Please call operator for further assistance.
Alarm 19	RemoteExt.wav	Please Enter the extension number for which you wish to set or cancel wake up alarm.
Alarm 20	RemoteExtH.wav	Please Enter the room number for which you wish to set or cancel wake up alarm.
Alarm 21	PerWakeupVeri.wav	You have set personal wake up alarm for,
Alarm 22	AutoWakeupVeri.wav	You have set automated wake up alarm for,
Alarm 23	PerWakeupSet.wav	Your personal wake up alarm is set.
Alarm 24	AutoWakeupSet.wav	Your automated wake up alarm is set.

Alarm 25	DailyPerWakeupSet.wav	Your daily personal wake up alarm is set.
Alarm 26	DailyAutoWakeupVeri.wav	You have set daily automated wake up alarm for,
Alarm 27	DailyPerWakeupVeri.wav	You have set daily personal wake up alarm for,
Alarm 28	DailyAutoWakeupSet.wav	Your daily automated wake up alarm is set.
Alarm 29	PerReminderVeri.wav	You have set personal reminder for,
Alarm 30	AutoReminderVeri.wav	You have set automated reminder for,
Alarm 31	PerReminderSet.wav	Your personal reminder is set.
Alarm 32	AutoReminderSet.wav	Your automated reminder is set.
Alarm 33	Alarmmode.wav	To set it as Personal, Press 1. To set it as Automated, Press 2.
Alarm 34	Alarmnocancel.wav	Sorry! There is no alarm set to cancel.
Alarm 35	Remindernocancel.wav	Sorry! There is no reminder set to cancel.
Alarm 36	WakeUpgreeting.wav	This is your wake up call.
Alarm 37	DailyWakeUpgreeting.wav	This is your daily wake up call.
Alarm 38	Remindergreeting.wav	This is your reminder call.
Alarm 39	SWakeUpgreeting.wav	This is your wake up call. To acknowledge press '0'.
Alarm 40	SDailyWakeUpgreeting.wav	This is your daily wake up call. To acknowledge press '0'.
Alarm 41	SRemindergreeting.wav	This is your reminder call. To acknowledge press '0'.
Alarm 42	Acknowledge.wav	Your alarm is acknowledged.
Alarm 43	Entertime2I.wav	Enter the time, HH MM in Twenty four hour format
Alarm 44	Entertime2U.wav	Enter the time, HH MM in Twelve hour format. For AM, press 1. For PM, press 2.
Alarm 45	RemoteExtRem.wav	Please Enter the extension number for which you wish to set or cancel reminder.
Alarm 46	RemoteExtHRem.wav	Please Enter the room number for which you wish to set or cancel reminder.
Alarm 47	RemAcknowledge.wav	Your reminder is acknowledged.
Alarm 48	Thankservice.wav	Thank You for using this service.
Alarm 49	Morning.wav	Good Morning!
Alarm 50	Afternoon.wav	Good Afternoon!
Alarm 51	Evening.wav	Good Evening!
Alarm 52	NoInput.wav	Sorry, You have not entered any input.
Alarm 53	InvalidInput.wav	Sorry, this is an invalid input.

Miscellaneous		
Prompt Name	**File Name**	**Prompts**
Miscellaneous 01	Press.wav	Press,
Miscellaneous 02	Dial.wav	Dial,
Miscellaneous 03	And.wav	and,
Miscellaneous 04	At.wav	At,
Miscellaneous 05	Beep.wav	<Beep>
Miscellaneous 06	NoDISA.wav	Sorry! DISA feature is not allowed.
Miscellaneous 07	NoDISAStn.wav	Sorry! DISA feature is not allowed for dialed extension.
Miscellaneous 08	MsgNtfyFor.wav	Message notification for extension number,
Miscellaneous 09	DialDigit.wav	Press any digit to proceed.
Miscellaneous 10	Chkinwel.wav	Welcome! It is our pleasure to serve you. We will do our best to make your stay comfortable.
Miscellaneous 11	Checkout.wav	Sorry! The Guest has checked out.

Miscellaneous 12	to.wav	to,
Miscellaneous 13	Holdmusic.wav	<Music>
Miscellaneous 14	Entextn.wav	Please enter the extension number.
Miscellaneous 15	EntPwd.wav	Please enter your password.
Miscellaneous 16	Invalpwd.wav	Sorry, this is an invalid password.
Miscellaneous 17	enterfoldernum.wav	Enter folder number.
Miscellaneous 18	enterfilenum.wav	Enter file number.
Miscellaneous 19	RecPrompt.wav	Please Record your prompt after the beep and press any digit to end.
Miscellaneous 20	InvalidInput.wav	Sorry, this is an invalid input
Miscellaneous 21	NoInput.wav	Sorry, you have not entered any digit
Miscellaneous 22	Record.wav	To record
Miscellaneous 23	Play.wav	To play
Miscellaneous 24	Erase.wav	To erase
Miscellaneous 25	EntDialInnum.wav	Please enter your number and password

SARVAM UCS Features Supported in Terminals

Sr. No.	Features	Supported In						
		VP330	VP210	VP248/310 /510	Extended VP710	VARTA ADR100	VARTA AMP100	VARTA WIN200
1	Abbreviated Dialing	Yes	Yes	Yes	Yes	Yes	Yes	Yes
2	Access Codes	Yes	Yes	Yes	Yes	Yes	Yes	Yes
3	Account Codes	Yes	Yes	Yes	Yes	Yes	Yes	Yes
4	Alarms	Yes	Yes	Yes	No	No	No	Yes
5	Authority Codes	No	No	Yes	No	No	No	No
6	Auto Answer	Yes	Yes	Yes	Yes	No	No	No
7	Auto Call Back	Yes	Yes	Yes	Yes	Yes	Yes	Yes
8	Auto Redial	Yes	Yes	Yes	Yes	Yes	Yes	Yes
9	Background Music	No	No	No	No	No	No	No
10	Barge-In	Yes	Yes	Yes	Yes	Yes	Yes	Yes
11	BLF for Trunks	Yes	Yes	Yes	Yes	Yes	Yes	Yes
12	Call Chaining	Yes	Yes	Yes	No	No	No	Yes
13	Call Cost Display	No	No	Yes	No	No	No	No
14	Call Duration Display	Yes	Yes	Yes	Yes	Yes	Yes	Yes
15	Call Forward	Yes	Yes	Yes	Yes	Yes	Yes	Yes
16	Call Forward - Remote	No	No	Yes	No	No	No	No
17	Call Forward - Scheduled	Yes	Yes	Yes	Yes	Yes	Yes	Yes
18	Call Forward - Not Registered	Yes	Yes	Yes	Yes	Yes	Yes	Yes
19	Call Hold	Yes	Yes	Yes	Yes	Yes	Yes	Yes
20	Call Park	Yes	Yes	Yes	No	No	No	Yes
21	Call Logs	Yes	Yes	Yes	Yes	Yes	Yes	Yes
22	Call Pickup	Yes	Yes	Yes	Yes	Yes	Yes	Yes
23	Call Toggle	Yes	Yes	Yes	Yes	Yes	Yes	Yes
24	Call Transfer	Yes	Yes	Yes	Yes	Yes	Yes	Yes
25	CLIP	Yes	Yes	Yes	Yes	Yes	Yes	Yes
26	CLIR	Yes	Yes	Yes	No	No	No	Yes
27	Cancel All Station Features	No	No	Yes	No	No	No	No
28	CUG	Yes	Yes	Yes	Yes	Yes	Yes	Yes
29	CUG with Exchange ID	Yes	Yes	Yes	Yes	Yes	Yes	Yes
30	Conference 3-Party	Yes	Yes	Yes	Yes	Yes	Yes	Yes
31	Conference Multiparty	Yes	Yes	Yes	Yes	Yes	Yes	Yes
32	Conference Dial In	Yes	Yes	Yes	Yes	Yes	Yes	Yes
33	Conversation Recording	Yes	Yes	Yes	Yes	Yes	Yes	Yes
34	COSEC Integration	Yes	Yes	Yes	Yes	Yes	Yes	Yes
35	Daylight Saving Time (DST)	Yes	Yes	Yes	NA	NA	NA	NA
36	Department Call	Yes	Yes	Yes	Yes	Yes	Yes	Yes
37	Digital Key Phone-Operation	Yes	Yes	Yes	NA	NA	NA	NA
38	Distinctive Rings	Yes	Yes	Yes	No	No	No	Yes
39	Do Not Disturb (DND)	Yes	Yes	Yes	Yes	Yes	Yes	Yes
40	DSS Call Pick-Up	Yes	Yes	Yes	Yes	Yes	Yes	Yes
41	Dynamic Lock	Yes	Yes	Yes	No	No	No	Yes
42	Emergency Conference	Yes	Yes	Yes	Yes	Yes	Yes	Yes
43	Emergency Dialing	Yes	Yes	Yes	Yes	Yes	Yes	Yes

Sr. No.	Features	Supported In						
		VP330	VP210	VP248/310 /510	Extended VP710	VARTA ADR100	VARTA AMP100	VARTA WIN200
44	Extended IP Phone/Mobile Softphone Client - Operation	Yes	Yes	Yes	Yes	Yes	Yes	Yes
45	Flashing on Trunks (Continued Dialing)	Yes	Yes	Yes	No	No	No	Yes
46	Flexible Numbers	Yes	Yes	Yes	Yes	Yes	Yes	Yes
47	Follow Me	Yes	Yes	Yes	No	No	No	No
48	Forced Answer	Yes	Yes	Yes	Yes	Yes	Yes	Yes
49	Forced Call Disconnection	Yes	Yes	Yes	No	No	No	Yes
50	Handover (Manual) and	Yes	Yes	No	Yes	Yes	Yes	Yes
51	Handover (Automatic)	Yes	Yes	No	No	Yes	Yes	Yes
52	Hot Desking	No	No	No	No	No	No	No
53	Hotline	Yes	Yes	Yes	No	No	No	No
54	Intercom	Yes	Yes	Yes	Yes	Yes	Yes	Yes
55	Interrupt Request (IR)	Yes	Yes	Yes	Yes	Yes	Yes	Yes
56	Last Caller Recall	Yes	Yes	Yes	Yes	Yes	Yes	Yes
57	Last Number Redial	Yes	Yes	Yes	Yes	Yes	Yes	Yes
58	Live Call Screening	No	No	No	No	No	No	No
59	Live Call Supervision	Yes	Yes	Yes	Yes	Yes	Yes	Yes
60	Macros	No	No	Yes	No	No	No	No
61	Meet Me Paging	Yes	Yes	Yes	No	No	No	No
62	Message Wait	Yes	Yes	Yes	No	No	No	Yes
63	Mute	Yes	Yes	Yes	Yes	Yes	Yes	Yes
64	OFF-Hook Alert	No	No	No	No	No	No	No
65	One Touch Transfer	Yes	Yes	Yes	Yes	Yes	Yes	Yes
66	Paging	Yes	Yes	Yes	Yes	Yes	Yes	Yes
67	PIN Dialing	Yes	Yes	Yes	No	No	No	No
68	Presence	Yes	Yes	Yes	Yes	Yes	Yes	Yes
69	Quick Dial	Yes	Yes	Yes	No	No	No	No
70	Raid	No	No	Yes	No	No	No	No
71	Reminder	Yes	Yes	Yes	No	No	No	Yes
72	Room Monitor	Yes	Yes	Yes	Yes	Yes	Yes	Yes
73	Selective Port Access	Yes	Yes	Yes	No	No	No	No
74	Self Ring Test	No	No	Yes	No	No	No	No
75	System Activity Log Display	No	No	Yes	No	No	No	No
76	System Fault Log Display	No	No	Yes	No	No	No	No
77	Time Zone Display	No	No	Yes	No	No	No	No
78	Trunk Call Waiting	No	No	No	No	No	No	No
79	User Absent/Present	Yes	Yes	Yes	Yes	Yes	Yes	Yes
80	User Password	Yes	Yes	Yes	Yes	Yes	Yes	Yes
81	Video Call	No	No	No	Yes	Yes	Yes	Yes
82	Virtual Extension	Yes	Yes	Yes	Yes	Yes	Yes	Yes
83	Walk-In Class of Service	Yes	Yes	Yes	No	No	No	No
84	Voice Mail Features	Yes	Yes	Yes	Yes	Yes	Yes	Yes

Sr. No.	Features	Supported In						
		VP330	VP210	VP248/310 /510	Extended VP710	VARTA ADR100	VARTA AMP100	VARTA WIN200
	Others*							
85	IM	No	No	No	Yes	Yes	Yes	Yes
86	SMS	No	No	No	Yes	Yes	Yes	Yes
87	BLF for Extensions	Yes	No	No	Yes	Yes	Yes	Yes
88	Soft Keys	Yes	No	No	Yes	Yes	Yes	Yes
89	Contact Grouping	No	No	No	No	No	No	Yes
90	Favorites	Yes	No	No	Yes	Yes	Yes	Yes
* Refer to the respective User Guide for details								

Features at a Glance

Abbreviated Dialing

Personal/Global Abbreviated Dialing	***8-Location Code***
Program Personal memory	***1071-Location Code-Number-#*-TAC***

Account Code

Account Code by Number	***1058-Account Code***
Account Code by Name	***1059-Account Name***

Alarms

Once Only Alarm	***161-Hours-Minutes-1***
Daily Alarms	***161-Hours-Minutes-2***
Cancel Once Only/Daily Alarm	***161-#***
Set/Cancel Voice Guided Alarm	***163-Follow VMS Prompts***

Auto Call Back

Auto Call Back-On Busy	***2***
Auto Call Back-On No Reply	***2***
Cancel Auto Call Back	***102***

Auto Redial

Auto Redial	***17***
Cancel Auto Redial	***1070***

Barge-In

Barge-In	***4***

Call Cost Display

Last Ten dialed numbers Cost display	***1075***

Call Chaining

Call Chaining	***1050***

Call Forward

Call Forward-All Calls to Another Station	***131-Station/Department Group/VMS***
Call Forward-All Calls to External Station	***131-Trunk Access Code-Dest. Number-#****
Call Forward-If Busy	***132-Station/Department Group/VMS***
Call Forward-If Busy-All Calls to External Number	***132-Trunk Access Code-Dest. Number-#****
Call Forward-If No Reply	***133-Station/Department Group/VMS***
Call Forward-If No Reply-All Calls to External Number	***133-Trunk Access Code-Dest. Number-#****
Call Forward-If Busy or No Reply	***134-Station/Department Group/VMS***
Call Forward-If Busy or No Reply-All to External Number	***134-Trunk Access Code-Dest. Number-#****
Call Forward-Dual Ring	***1361***
Disable Call Forward-Dual Ring	***1360***
Cancel Call Forward	***130***

Call Forward - Scheduled

For Working Hours	
Call Forward-Scheduled - Unconditional	***1175-1-1-Destination Number***
Call Forward - Scheduled - Busy	***1175-1-2-Destination Number***
Call Forward - Scheduled - No Reply	***1175-1-3-Destination Number***
Call Forward - Scheduled - Busy/No Reply	***1175-1-4-Destination Number***
Call Forward - Scheduled - Set Dual Ring	***1175-1-5-1***
Call Forward - Scheduled - Cancel Dual Ring	***1175-1-5-0***
Call Forward - Scheduled - Cancel for Working Hours	***1175-1-0***
For Break Hours	
Call Forward-Scheduled - Unconditional	***1175-2-1-Destination Number***
Call Forward - Scheduled - Busy	***1175-2-2-Destination Number***
Call Forward - Scheduled - No Reply	***1175-2-3-Destination Number***
Call Forward - Scheduled - Busy/No Reply	***1175-2-4-Destination Number***
Call Forward - Scheduled - Set Dual Ring	***1175-2-5-1***
Call Forward - Scheduled - Cancel Dual Ring	***1175-2-5-0***
Call Forward - Scheduled - Cancel for Break Hours	***1175-2-0***
For Non-Working Hours	
Call Forward-Scheduled - Unconditional	***1175-3-1-Destination Number***
Call Forward - Scheduled - Busy	***1175-3-2-Destination Number***
Call Forward - Scheduled - No Reply	***1175-3-3-Destination Number***

Call Forward - Scheduled - Busy/No Reply	***1175-3-4-Destination Number***
Call Forward - Scheduled - Set Dual Ring	***1175-3-5-1***
Call Forward - Scheduled - Cancel Dual Ring	***1175-3-5-0***
Call Forward - Scheduled - Cancel for Non-Working Hours	***1175-3-0***
Cancel Call Forward - Scheduled for all Time Zones	***1175-0***

Call Forward - Department Group

Call Forward-Department Group - Unconditional	***1179-Department Group Number (Access Code)-1-Destination Number***
Call Forward-Department Group - Busy	***1179-Department Group Number (Access Code)-2-Destination Number***
Call Forward-Department Group - No Reply	***1179-Department Group Number (Access Code)-3-Destination Number***
Call Forward-Department Group - Busy/No Reply	***1179-Department Group Number (Access Code)-4-Destination Number***
Call Forward-Department Group - Cancel	***1179-Department Group Number (Access Code)-0***

Call Hold

Put the caller on Hold	***Flash***
Retrieve the caller	***Flash***

Call Park

To Park a Call	***115-Orbit Number***
To Automatically Park a Call in General Orbit	***1150***
To Retrieve the Parked Call	***116-Orbit Number***

Call Pick Up

Call Pick Up-General	***4***
Call Pick Up-Selective	***12-Station***

Call Toggle

Call Toggle (Toggle)	***Flash-1 (for SLT), Transfer-1 (for DKP/Extended IP phone)***

Call Transfer

Call Transfer to Station	***Speech with Station-Flash-Station (Transfer Target)-OnHook***

Call Transfer to Station (External Number)	***Speech with External Number-Flash-Station (Transfer Target)-OnHook***
Call Transfer to Trunk (External)	***Speech with External Number -Flash-TAC-External Number (Transfer Target)-OnHook***
Call Transfer to Trunk	***Speech with Station -Flash-TAC-External Number-Go OnHook***
Blind Transfer to Mail Box (VMS)	***Flash-1078-Station***

Calling Line Identification Restriction

Enable/disable CLIR	***103***

Cancel All Station Features

Cancel all station features	***1051***

Conference 3-Party

Conference 3-Party	***Flash-*3***

Conference Dial-In

Schedule a Conference	****19-1-Conference Number-Conference Password***
Initiate a Conference	****19-2-Conference Number-Conference Password***
Cancel a Conference	****19-0***

Conference Multiparty

To Temporarily Leave / Rejoin a Conference	***191***
Terminate Conference	***190***

Conversation Recording

Conversation Recording	***Flash-1095***

Department Call

Department Call	***Department Number (3901-3916)***

Direct Inward System Access

Enter DISA Mode	***1079-Station Number-User Password***

Do Not Disturb

Set Do Not Disturb with Text Message	***18-DND Message Number***
Cancel Do Not Disturb	***18-0***
DND Override	***4***

Dynamic Lock

Set Dynamic Timer	***142-User Password-Minutes***
Change Toll Control Level	***141-User Password-Level***

Emergency Conference

Make Emergency Conference	***1177-Department Group Number***
Cancel Emergency Conference	***While in speech with any one party, Go Off-Hook and dial 190***

Emergency Dialing

Dial Emergency Number	***Go Off-Hook and dial Emergency Number*** ***or*** ***Dial TAC-Emergency Number***

Flashing on Trunks

Flashing on Trunks	***Flash-****

Floor Service

To access Floor Service	***38***

Follow Me

Set Call Follow Me	***135-Station-User Password***
Cancel Call Follow Me	***130***

Forced Answer

Forced Answer	***5***

Forced Call Disconnection

Forced Call Disconnection	***#****

Hot Desking

Set Hot Desk	***1091-Your Station Number-User Password***

Hotline

Set Hotline	***151-Station***
Set Hotline Timer	***154-Seconds***
Hot Outward Dialing	***152-Trunk Access Code***
HOD with Number	***153-Trunk Access Code-Number-#****
Cancel Hotline/HOD	***150***

Interrupt Request

Interrupt Request	***3***

Last Caller Recall

Last Caller Recall	***1092***

Last Number Redial

Last Number Redial	***7***

Live Call Screening

To Activate LCS	***1094-1***
To Deactivate LCS	***1094-0***

Live Call Supervision

Set Live Call Supervision	***1098-Destination Station***

Maid-In

Maid Status from a Room	***1054-Code***

Meet Me Paging

Meet Me Paging-Caller	***1093-Page Zone Number***
Meet Me Paging-Called Party	***1093-Station Number of the Caller***

Message Wait

Message Wait Set/Cancel	***1076-Station-Code***

Message Wait Retrieval	***1077***

Mini Bar

To check the utilities of the room	***1056-Item Number-Quantity***

Mute

Mute	***1052***

OG Trunk Bundle Group

To grab OG Trunk Bundle Group	***0/5/6-OGTBG Index***

Operator

Call to Operator	***9***

Paging

Paging	***1074-Page Zone Number***

Presence

Publishing Presence by extension user	***104-Password-Index No.***
To view Presence Status	***1097-Extension Number***

Priority Calls in E&M MFCR2 Signaling

Manual Priority Intrusion	****37***
Forced Released Order	****38***

Raid

Raid	***5***

RCOC

RCOC in DISA Mode	***** on dial tone***

Reminder

Set Reminder	***162-DD-MM-YYYY-HH-MM***
Cancel Reminder	***162-#***
Set/Cancel Voice Guided Reminder	***164-Follow VMS Prompts***

Room Monitor

Room Monitor	***1073-Station***

Selective Port Access

Selective Port Access	***69-Port Type-Port Offset***

Self Ring Test

Self Ring Test	***1057***

System Administrator Mode

Enter System Administrator Mode	***1#92-SA Password***
Exit SA Mode	***1#92***
SA Command	***1072***

System Engineer Mode

Enter System Engineer Mode	***1#91-SE Password***
Exit SE Mode	***00***

Trunk Reservation

Reserve a Trunk	***6***
Cancel a Reserved Trunk	***102***

User Absent/Present

User Absent	***104-User Password-0***
User Present	***104-User Password-1***

User Password

Change User Password	***114-Old User Password-New User Password***

Voice Help

Voice Help	***1090***

Walk-In Class Of Service

To Walk-In	***111-1-Your Extension Number-Your User Password***
To Walk-Out	***111-0***

SA Commands

Feature Name	Feature Number	Access Code	Call Phases					
			Dial	Routing	Blocked	Placed	Matured 2-Way	Matured 3-way
Check-In	001	1072-901	Y				Y	
Check-Out	002	1072-902	Y				Y	
Guest Name	003	1072-903	Y				Y	
Guest Group	004	1072-904	Y				Y	
Guest-In/Out	005	1072-905	Y				Y	
Guest Title	006	1072-906	Y				Y	
Change Check-In Profile of Room	007	1072-907	Y				Y	
Change Occupancy Status of Room / Extension	008	1072-908	Y				Y	
Change Clean status of Room/ Extension	009	1072-909	Y				Y	
Room Shift	010	1072-910	Y				Y	
Reprint Check Out Report	011	1072-911	Y				Y	
Print Room Status Report	012	1072-912	Y				Y	
Print Alarm Status Report	013	1072-913	Y				Y	
Delete Checked Out calls	014	1072-914	Y				Y	
Set DND-Remote	015	1072-001	Y				Y	
Set Dynamic Lock settings – Remote	016	1072-002	Y				Y	
Set Wakeup Alarms –Remote	017	1072-003	Y				Y	
Set Call Budget for Remote Station	018	1072-004	Y				Y	
Assign/De-assign Mailbox to a Station/Department Group - Remote	019	1072-005	Y				Y	
Set Call Forward – Remote	020	1072-006	Y				Y	
Set Call forward for all stations-Remote	021	1072-007	Y				Y	
Assign Station User Greeting Message	022	1072-008	Y				Y	
Display & Acknowledge System Activity	023	1072-009	Y				Y	
Display & Acknowledge System Fault	024	1072-010	Y				Y	
Station Budget Display	025	1072-011	Y				Y	

Feature Name	Feature Number	Access Code	Call Phases					
			Dial	Routing	Blocked	Placed	Matured 2-Way	Matured 3-way
Change User password of a Station	026	1072-012	Y				Y	
Lock/Unlock DKP’s Keypad	027	1072-013	Y				Y	
User Absent / Present	028	1072-014	Y				Y	
Change SA password	029	1072-015	Y				Y	
Change SA mode timer	030	1072-016	Y				Y	
Display Registered Mobile Operator ID	031	1072-017	Y					
Set Day/Night mode	032	1072-018	Y					
Clear System Activity Log	036	1072-022	Y				Y	
Start/Abort SAL in Offline mode	037	1072-023	Y				Y	
Start/Abort SAL in Online mode	038	1072-024	Y				Y	
Clear Mailbox for a range of extensions	039	-	Y				Y	
Cancel Dial in Conference	040	1072-026	Y				Y	
Start/Abort SFL in Offline mode	041	1072-027	Y				Y	
Start/Abort SFL in Online mode	042	1072-028	Y				Y	
Display Port Parameters	043	1072-029	Y				Y	
Start/Abort Online OG Report	044	1072-101	Y				Y	
OG Print Filter: Print calls originated from station/s	045	1072-102	Y				Y	
OG Print Filter: To Print calls terminated from CO	046	1072-103	Y				Y	
OG Print Filter: To Print calls terminated from BRI	047	1072-104	Y				Y	
OG Print Filter: To Print calls terminated from T1E1	048	1072-105	Y				Y	
OG Print Filter: To Print calls terminated from E&M	049	1072-106	Y				Y	
OG Print Filter: To Print calls terminated from Mobile	050	1072-107	Y				Y	
OG Print Filter: To Print calls terminated from SIP	051	1072-108	Y				Y	
OG Print Filter: To Print calls Department Bill Group wise	052	1072-109	Y				Y	
OG Print Filter: To print calls made on dates	053	1072-110	Y				Y	

Feature Name	Feature Number	Access Code	Call Phases					
			Dial	Routing	Blocked	Placed	Matured 2-Way	Matured 3-way
OG Print Filter: Print calls made between time	054	1072-111	Y				Y	
OG Print Filter: To Print calls made to numbers matching with the numbers programmed in the Number List	055	1072-112	Y				Y	
OG Print Filter: To Print calls of Duration more than this time	056	1072-113	Y				Y	
OG Print Filter: To Print calls of Units more than the units programmed	057	1072-114	Y				Y	
OG Print Filter: To Print calls made to account code	058	1072-115	Y				Y	
Assign default OG Print filters	059	1072-120	Y				Y	
Start/Abort offline report	060	1072-121	Y				Y	
Enable/ Disable OG Schedule Reports	061	1072-122	Y				Y	
Program Time for Daily OG Scheduled Reports	062	1072-123	Y				Y	
Program Day and Time for OG Weekly Scheduled Reports	063	1072-124	Y				Y	
Program Date and Time for OG Monthly Scheduled Reports	064	1072-125	Y				Y	
Delete calls made by stations	065	1072-131	Y				Y	
Delete calls made on/from date	066	1072-132	Y				Y	
To clear SMDR OG buffer	067	1072-133	Y				Y	
Start/Abort Internal calls Report	068	1072-136	Y				Y	
Set filter to print Internal calls Report Station wise	069	1072-137	Y				Y	
Set filter to print internal calls with duration greater than that given here	070	1072-138	Y				Y	
Start/Abort Offline Internal Call Report	071	1072-141	Y				Y	
Enable/ Disable Internal Scheduled Reports	072	1072-142	Y				Y	
Program Time for Internal Daily Scheduled Reports	073	1072-143	Y				Y	
Program Day and Time for Internal Weekly Scheduled Reports	074	1072-144	Y				Y	

Feature Name	Feature Number	Access Code	Call Phases					
			Dial	Routing	Blocked	Placed	Matured 2-Way	Matured 3-way
Program Date and Time for Internal Monthly Scheduled Reports	075	1072-145	Y				Y	
To Clear SMDR Internal Buffer	076	1072-150	Y				Y	
Start/Abort Online – IC Report	077	1072-151	Y				Y	
Set filter to print all Normal calls	078	1072-152	Y				Y	
Set filter to print all Built-In Auto Attendant calls	079	1072-153	Y				Y	
Set filter to print all Unanswered calls	080	1072-154	Y				Y	
Set filter to print all Built-In Auto Attendant Unanswered calls	081	1072-155	Y				Y	
Set filter to print all DISA calls	082	1072-156	Y				Y	
Set filter to print all calls with speech duration More than timer	083	1072-157	Y				Y	
Set filter to print all calls unanswered for duration More than timer	084	1072-158	Y				Y	
Set filter to print all calls kept on hold for duration more than timer	085	1072-159	Y				Y	
Set filter to print all IC calls received by the station	086	1072-160	Y				Y	
Set filter to print all IC calls recd. On the CO	087	1072-161	Y				Y	
Set filter to print all IC calls recd. On the BRI	088	1072-162	Y				Y	
Set filter to print all IC calls recd. On the T1E1	089	1072-163	Y				Y	
Set filter to print all IC calls recd. On the E&M	090	1072-164	Y				Y	
Set filter to print all IC calls from Mobile	091	1072-165	Y				Y	
Set filter to print calls received from SIP	092	1072-166	Y				Y	
Set filter to print all IC calls recd. On/from date	093	1072-167	Y				Y	
Set filter to print all IC calls recd. At/from-to Time	094	1072-168	Y				Y	
Set filter to print all IC calls recd. From nos. matching the External Number List	095	1072-169	Y				Y	

Feature Name	Feature Number	Access Code	Call Phases					
			Dial	Routing	Blocked	Placed	Matured 2-Way	Matured 3-way
Default IC Print filters	096	1072-170	Y				Y	
Abort/Start IC Offline Report	097	1072-171	Y				Y	
Enable/ Disable IC Scheduled Report	098	1072-172	Y				Y	
Program Time for IC Daily Scheduled Reports	099	1072-173	Y				Y	
Program Day and Time for IC Weekly Scheduled Reports	100	1072-174	Y				Y	
Program Date and Time for IC Monthly Scheduled Reports	101	1072-175	Y				Y	
Clear SMDR-IC buffer	102	1072-180	Y				Y	
Start/Abort Printing of Online T1E1 Performance Report	103	1072-030	Y				Y	
Start/Abort Offline T1E1 Performance Report	104	1072-031	Y				Y	
Signal Strength of Mobile Port	105	1072-032	Y				Y	
Enable/Disable Call Cost Display for a Station	106	1072-181	Y				Y	
Start/Abort Hotel/Motel Activity log in Offline mode	107	1072-176	Y				Y	
Start/Abort Hotel/Motel Activity log in Online mode	108	1072-177	Y				Y	
Display and Acknowledge Hotel/ Motel Activity	109	1072-178	Y				Y	
Change Guest VIP Status of Station	110	1072-915	Y				Y	
Change Phone Ringing Pattern of Room	111	1072-916	Y				Y	
Print Reminder Status Report	112	1072-917	Y				Y	
Remote Reminder	113	1072-033	Y				Y	
Remote Voice Guided Alarm	114	1072-034	Y				Y	
Remote Voice Guided Reminder	115	1072-035	Y				Y	
Redirect Messages of a Station	116	1072-314	Y				Y	
To record Holiday message	117		Y				Y	
Enable/Disable Scheduled Alarm Report	118	1072-036	Y				Y	
Program Time for Scheduled Alarm Report	119	1072-037	Y				Y	

Feature Name	Feature Number	Access Code	Call Phases					
			Dial	Routing	Blocked	Placed	Matured 2-Way	Matured 3-way
Enable/Disable Scheduled Reminder Report	120	1072-038	Y				Y	
Program Time for Scheduled Reminder Report	121	1072-039	Y				Y	
Request Database Synchronization to PMS	122	1072-040	Y				Y	
Enable/Disable Scheduled Room Status Report	123	1072-041	Y				Y	
Program Time for Schedule Room Status Report	124	1072-042	Y				Y	
Enable/Disable Scheduled Change of Room Clean Status	125	1072-043	Y				Y	
Program Time for Schedule Change of Room Clean Status	126	1072-044	Y				Y	
To playback Holiday message	127		Y				Y	
Enable/Disable Internal Call Block For Guest Phones	128	1072-045	Y				Y	
Software Version Revision Display of CPU Card	129	1072-191	Y				Y	
User Definable Fields	130	1072-920	Y				Y	
OG Print Filter: To print calls originated from CO	131	1072-183	Y				Y	
OG Print Filter: To print calls originated from BRI	132	1072-184	Y				Y	
OG Print Filter: To print calls originated from T1E1	133	1072-185	Y				Y	
OG Print Filter: To print calls originated from E&M	134	1072-186	Y				Y	
OG Print Filter: To print calls originated from Mobile	135	1072-187	Y				Y	
OG Print Filter: To print calls originated from SIP	136	1072-188	Y				Y	
OG Print Filter: To print calls originated from Magneto	137	1072-189	Y				Y	
Set Budget Type for CO Port	138	1072-192	Y				Y	
Program Budget Amount for CO	139	1072-193	Y				Y	
Program Free Minutes on CO	140	1072-194	Y				Y	
Set Budget Type for T1E1 Port	141	1072-195	Y				Y	
Program Budget Amount for T1E1	142	1072-196	Y				Y	

Feature Name	Feature Number	Access Code	Call Phases					
			Dial	Routing	Blocked	Placed	Matured 2-Way	Matured 3-way
Program Free Minutes on T1E1	143	1072-197	Y				Y	
Set Budget Type for BRI Port	144	1072-198	Y				Y	
Program Budget Amount for BRI	145	1072-199	Y				Y	
Program Free Minutes on BRI	146	1072-200	Y				Y	
Set Budget Type for SIP	147	1072-201	Y				Y	
Program Budget Amount for SIP	148	1072-202	Y				Y	
Program Free Minutes on SIP	149	1072-203	Y				Y	
Set Budget Type for MOBILE	150	1072-204	Y				Y	
Program Budget Amount for MOBILE	151	1072-205	Y				Y	
Program Free Minutes on MOBILE	152	1072-206	Y				Y	
Call Budget Reset Mode for CO	153	1072-207	Y				Y	
Scheduled Date to Rest Call Budget Statistics on CO	154	1072-208	Y				Y	
Reset consumed Budget Amount / minutes on CO manually	155	1072-209	Y				Y	
Call Budget Reset Mode for T1E1	156	1072-210	Y				Y	
Scheduled Date to Rest Call Budget Statistics on T1E1	157	1072-211	Y				Y	
Reset consumed Budget Amount / minutes on T1E1 manually	158	1072-212	Y				Y	
Call Budget Reset Mode for BRI	159	1072-213	Y				Y	
Scheduled Date to Rest Call Budget Statistics on BRI	160	1072-214	Y				Y	
Reset consumed Budget Amount / minutes on BRI manually	161	1072-215	Y				Y	
Call Budget Reset Mode for SIP	162	1072-216	Y				Y	
Scheduled Date to Rest Call Budget Statistics on SIP	163	1072-217	Y				Y	
Reset consumed Budget Amount / minutes on SIP manually	164	1072-218	Y				Y	
Call Budget Reset Mode for MOBILE	165	1072-219	Y				Y	
Scheduled Date to Rest Call Budget Statistics on MOBILE	166	1072-220	Y				Y	
Reset consumed Budget Amount / minutes on MOBILE manually	167	1072-221	Y				Y	

Feature Name	Feature Number	Access Code	Call Phases					
			Dial	Routing	Blocked	Placed	Matured 2-Way	Matured 3-way
Swap Ports for CO8/SLT8 - MAG8 Card	168	1072-046	Y				Y	
Reset ASR and ACD for Mobile port	169	1072-222	Y				Y	
Set/Cancel Scheduled Call Forward	170	1072-223	Y				Y	
To Broadcast Message	175	1072-301	Y				Y	
To Play/Record VMS Prompt	176	1072-302	Y				Y	
To Play/Record VMS Mailbox Greeting	177	1072-303	Y				Y	
To Play/Record VMS Mailbox Name	178	1072-304	Y				Y	

System Commands

Abbreviated Dialing

Program telephone number in a personal directory	***1902-Personal Directory-Location Code-Number***
Clear telephone number from a location in a PM Group	***1902-Personal Directory-Location Code-#****
Program a name in PM Group	***1903-Personal Directory-Location Code-Name***
Clear a name from location in a PM Group	***1903-Personal Directory-Location Code-#****
To program a TAC Index to PM Group	***1904-Personal Directory-Location Code-TAC Index***
To assign PM Group to a SLT	***1905-1-SLT-Personal Directory***
To clear PM Group assigned to a SLT	***1905-1-SLT-00***
To assign PM Group to a DKP	***1906-1-DKP-Personal Directory***
To clear PM Group assigned to a DKP	***1906-1-DKP-00***
To assign PM Group to an ISDN Terminal	***1907-1-ISDN-Personal Directory***
To clear PM Group assigned to an ISDN Terminal	***1907-1-ISDN-00***
To clear a PM Group	***1901-1-PM Group***
To program a telephone number in global directory	***1801-Location Code-Number***
Clear telephone number from a location in global directory	***1801-Location Code-#****
Program a name in global directory	***1802-Location Code-Name***
Clear a name from a location in the global directory	***1802-Location Code-#****
Program the OGTBG for global directory	***1803-1-Location Code-OGTBG***
Clear a location code in the global directory	***1800-1-Location Code***

Access Codes

Program the access code for features	***3111-1-Feature Number-Access Code-#****
Assign default access code	***3161-1-Feature Number***

Account Codes

Program account name for the account code	***4851-1-Account Code-Account Name***
To enable Forced Account Code flag in Station Advanced Feature Template	***5602-1-Template Number-Feature Number-Flag Code***
To enable Forced Account Code flag on a Trunk	***5802-1-Template Number-Feature Number-Flag Code***

AC Impedance Test

To run the AC Impedance Test using the Accurate Mode	**3361-1-CO Port Number**
To view the report	**3362**

Alarms

Program alarm/reminder ring timer	***2201-Seconds***
Program number of alarm attempts	***2202-Number of Alarm Attempts***
Program alarm attempt interval	***2203-Alarm Attempt Interval***
Enable/disable snooze alarm/reminder	***2204-Code***
To disable/enable Configurable Alarm Type	***2208-Flag***
To disable/enable Configurable Alarm Category	***2209-Flag***
To configure Alarm Notification Type	***5602-1-Template Number-12-Alarm Notification Type***
To program Macros	***1810-Macro Index-Number String***
To clear a Macro	***1810-Macro Index-#****
To program Access Code for Macro	***3115-1-Macro Index-Access Code***
To clear the Access Code for the Macro	***3115-1-Macro Index***
To assign Destination Port for Hotel Reports	***3701-Destination Port Code***

Alternate Number Dialing

To assign an alternate group number to a location code	***1804-1-Memory Location Code-Alternate Number Group***
To clear an alternated group number to a location code	***1804-1-Memory Location Code-000***

Auto Answer

To set auto answer on DKP	***1214-1-DKP-Auto Call Answer Mode***
To set Auto Answer Timer	***1215-1-DKP-Auto Call Answer Timer***
To enable/disable Headset Connectivity flag	***1213-1-DKP-Headset Connectivity Flag***

Auto Call Back

To program Auto Call Back when Busy/No Reply in a CoS group	***1302-1-COS Group-Feature Number-Code***
To assign the CoS group with Auto Call Back Busy/No Reply to a Station Basic Feature Template	***5502-1-Template Number-Feature Number-Code***
To apply the Station Basic Feature Template with Auto Call Back when Busy and No Reply, to an SLT	***5503-1-SLT-Template Number***

To apply the Station Basic Feature Template with Auto Call Back when Busy and No Reply, to a DKP	***5504-1-DKP-Template Number***
To program the Auto Call Back Ring Timer	***3801-Seconds***

Auto Redial

Program time duration between two trials for low priority	***1704-Seconds***
Program the number of trials for low priority	***1705-Count***
Program time duration between two trials for high priority	***1706-Seconds***
Program the number of trials for high priority	***1707-Count***
Program auto redial RBT wait timer	***1702-Seconds***
Program auto redial ring timer	***1703-Seconds***

Automatic Number Translation

To enable Automatic Number Translation Flag on OGTB	***6702-1-OG Trunk Bundle Number-Feature Number-Code***
To assign an Automatic Number Translation Table	***6702-1-OG Trunk Bundle Number-Feature Number-Code***
To default the ANT Table	***4750-1-ANT Table No.***

Barge-In

To program Barge-In Timer	***3803-Seconds***

Call Back on Trunk Ports

To enable / disable Call Back on CO Port	***3310-1-CO-Code***
To program Call Back Timer on CO Port	***3311-1-CO-Call Back Timer***
To select Call Back Mode for CO Port	***3312 -1-CO-Call Back Mode***
To program Call Back On for CO Port	***3313-1-CO-Call Back on***
To assign Call Back - Incoming Number List to a CO Port	***3314-1 -CO-Incoming Number List***
To assign Call Back - Outgoing Number List to a CO Port	***3315-1 -CO-Outgoing Number List***
To select Call Back From port for a CO Trunk port	***3316-1-CO-Call Back From***
To assign Call Back - OGTB Group to a CO port	***3317-1-CO-OGTB Group***
To enable/disable Call Back on Mobile Port	***8010-1-Mobile-Code***
To change Call Back Timer	***8011-1-Mobile-Call Back Timer***
To select the Call Back Mode	***8012-1-Mobile-Call Back Mode***

To program Call Back on for the Mobile Port	***8013-1-Mobile-Call Back on***
To assign a Call Back - Incoming Number List	***8034-1-Mobile-Incoming Number List***
To assign a Call Back - Outgoing Number List to a Mobile Port	***8035-1-Mobile-Outgoing Number List***
To select Call Back From for a Mobile Port	***8036-1-Mobile-Call Back From***
To assign a Call Back - OGTB Group for a Mobile Port	***8037-1-Mobile-OGTB Group***
To enable/disable Call Back on BRI port	***6242-1-BRI- Call Back Flag***
To program Call Back Timer for BRI port	***6243-1-BRI-Call Back Timer***
To program Call Back Mode on BRI port	***6244-1-BRI-Call Back Mode***
To program Call Back On method for BRI port	***6245-1-BRI-Call Back on selection***
To assign Call Back - Incoming Number List to a BRI port	***6246-1-BRI-Incoming Number List***
To assign a Call Back - Outgoing Number List to a BRI port	***6247-1-BRI-Outgoing Number List***
To define Call Back From for a BRI port	***6248-1-BRI-Call Back From***
To assign a Call Back - OGTB Group for a BRI port	***6249-1-BRI-OGTB Group***
To enable/disable Call Back on T1E1 port	***6176-1-T1E1- Call Back Flag***
To program Call Back Timer for T1E1 port	***6177-1-T1E1-Call Back Timer***
To program Call Back Mode on T1E1 port	***6178-1-T1E1-Call Back Mode***
To program Call Back On method for T1E1 port	***6179-1-T1E1-Call Back on selection***
To assign Call Back - Incoming Number List to a T1E1 port	***6180-1-T1E1-Incoming Number List***
To assign a Call Back - Outgoing Number List to a T1E1 port	***6147-1-T1E1-Outgoing Number List***
To define Call Back From for a T1E1 port	***6148-1-T1E1-Call Back From***
To assign a Call Back - OGTB Group for a T1E1 port	***6149-1-T1E1-OGTB Group***

Call Budget

To program default Call Budget Amount	***3710-Preset Call Budget Amount***

Call Budget on Trunk

To program Call Budget Type on CO	***3301-1-CO-Budget Type***
To program Call Budget Amount on CO	***3302-1-CO-Budget Amount***
To program Free minutes on CO	***3303-1-CO-Minutes***
To program Call Budget Reset Mode for CO	***3304-1-CO-Call Budget Reset Mode***
To program the Date for Scheduled Reset mode	***3305-1-CO-Date***
To program Call Budget - Number of Calls on CO	***3309-1-CO-Number of Calls***

To program Call Budget Type on Mobile trunk port	***8019-1-Mobile-Budget Type***
To program Call Budget Amount on Mobile trunk port	***8020-1-Mobile-Budget Amount***
To program Call Budget Minutes on Mobile trunk port	***8021-1-Mobile-Minutes***
To program Call Budget Number of Calls on Mobile trunk port	***8033-1-Mobile-Number of Calls***
To program Call Budget Reset Mode for Mobile trunk port	***8022-1-Mobile-Call Budget Reset Mode***
To program the Date for Scheduled Reset mode for Mobile trunk port	***8023-1-Mobile-Date***
To program Call Budget Type on BRI	***6214-1-BRI-Budget Type***
To program Call Budget Amount on BRI	***6215-1-BRI-Budget Amount***
To program Call Budget Minutes on BRI	***6216-1-BRI-Minutes***
To program Call Budget Number of Calls on BRI	***6205-1-BRI-Number of Calls***
To program Call Budget Reset Mode for BRI	***6217-1-BRI-Reset Mode***
To program the Date for Scheduled Reset mode for BRI	***6218-1-BRI-Date***
To program Call Budget Type on T1E1 Port	***6122-1-T1E1-Budget Type***
To program Call Budget Amount on T1E1 Port	***6123-1-T1E1-Budget Amount***
To program Call Budget Minutes on T1E1 Port	***6124-1-T1E1-Minutes***
To program Call Budget Number of Calls on T1E1 Port	***6125-1-T1E1-Number of Calls***
To program Call Budget Reset Mode for T1E1	***6138-1-T1E1-Call Budget Reset Mode***
To program the Date for Scheduled Reset mode for T1E1	***6139-1-T1E1-Date***
To manually reset consumed Call Budget of CO trunks	***3306-1-CO***
To manually reset consumed Call Budget of Mobile trunks	***8024-1-Mobile***
To manually reset consumed Call Budget of BRI trunks	***6219-1-BRI***
To manually reset consumed Call Budget of T1E1 trunks	***6140-1-T1E1***

Call Cost Calculation (CCC)

Program the unit charge for first unit when 16 KHz metering is used	***2600-Unit Charge for First Unit***
Program the unit charge for additional unit when 16 KHz metering is used	***2601-Unit Charge for Additional Unit***
To select service charge type	***2602-Service Charge Type***
To program service charge	***2603-Service Charge***
To set the service charge percentage when is selected based on the percentage	***2604-Percentage***

Program duration of first unit for a pulse rate type on normal days	***2607-Pulse Rate Type-Time Zone-Duration of First Unit***
Program duration of additional unit for a pulse rate type on normal days	***2608-Pulse Rate Type-Time Zone-Duration of Additional Unit***
Load default normal pulse rate type	***2606***
Program cost of first unit for a pulse rate type on normal days	***2609-Pulse Rate Type-Time Zone-Cost of First Unit***
Program cost of additional unit for a pulse rate type on normal days	***2610-Pulse Rate Type-Time Zone-Cost of Additional Unit***
Program duration of first unit for a pulse rate type on holidays	***2612-Pulse Rate Type-Time Zone-Duration of First Unit***
Program duration of additional unit for a pulse rate type on holidays	***2613-Pulse Rate Type-Time Zone-Duration of Additional Unit***
Program cost of first unit for a pulse rate type on holidays	***2614-Pulse Rate Type-Time Zone-Cost of First Unit***
Program cost of additional unit for a pulse rate type on holidays	***2615-Pulse Rate Type-Time Zone-Cost of Additional Unit***
Load default Discounted pulse rate type	***2611***
To program an area code	***2620-Area Code Index-Area Code-#****
To clear an area code for an Index	***2620-Area Code Index-#****
To program Pulse Rate Type for Pulse Rate Option of area code index	***2621-Area Code Index-Pulse Rate Option-Pulse Rate Type***
To program pulse rate for an area code	***2621-Area Code Index-Pulse Rate Type***
To delete the complete area code table	***2622-Reverse SE Password***
To program ignore digit count when SP_SP LCR is used	***2623-Area Code Index-Ignore Digit Count***
Program weekly off day	***2630-Day-Code***
Program a Special date	***2631-Special Date Index-Date-Month***
To clear a Special date index	***2632-Special Date Index***
To program area code name	***2633-Area Code Index-Name***
To clear an area code name	***2633-Area Code Index-#****

Call Duration Control (CDC)

To enable CDC for outgoing call	***4202-1-CDC Table-Code***
To enable CDC for incoming call	***4203-1-CDC Table-Code***
To enable CDC for internal call	***4204-1-CDC Table-Code***
To assign an external number to a table (allowed)	***4205-1-CDC Table-Number List***
To assign an external number to a table (denied)	***4206-1-CDC Table-Number List***
To assign CDC time to CDC table	***4207-1-CDC Table-CDC Timer***

To assign disconnection flag to a CDC table	***4208-1-CDC Table-Disconnection Flag***
To default a CDC Table	***4201-1-CDC Table***

Call Hold

To select Default Call Hold Type	***5318-Hold Type***
To change Global Hold Retrieval Timer	***3805-Seconds***
To change Exclusive Hold Retrieval Timer	***3812-Minutes***

Call Logs

To enable/disable Log Internal Calls in Missed Calls	***5361-Code***
To enable/disable Log Internal Calls in Answered Calls	***5362-Code***
To enable/disable Log Internal Calls in Dialed Calls	***5363-Code***

Call Park

Set call park timer	***3809-Minutes***
Set call park release timer	***3810-Minutes***

Call Pick Up

To assign call pickup group for SLT	***3901-1-SLT-Call Pickup Group***
To de-assign call pickup group for SLT	***3901-1-SLT-00***
To assign call pickup group for DKP	***3902-1-DKP-Call Pickup Group***
To de-assign call pickup group for DKP	***3902-1-DKP-00***
To assign call pickup group for ISDN Terminal	***3903-1-ISDN Terminal-Call Pickup Group***
To de-assign call pickup group for ISDN Terminal	***3903-1-ISDN Terminal-00***

Call Progress Tones

To select the region	***3501-Region Code***
To change CPT-related Timers	***3502-Seconds***
To program the ring back tone timer	***3503-Seconds***
To program busy tone timer	***3504-Seconds***
To program error tone timer	***3505-Seconds***
To program confirmation tone timer	***3506-Seconds***
To program the programming confirmation tone timer	***3509-Seconds***
To program the programming error tone timer	***3508-Seconds***
To select a dial tone	***5307-Flag***

To demonstrate call progress tones	***3541-Code***
To set demonstration timer	***3542-Seconds***

Call Taping

To enable/disable beeps during conversation recording	***5332-Code***
To program a number in a List	***4302-List Number-Location Index-Number-#****
To clear a number from a Location Index	***4302-List Number-Location Index-#****
To program Number List - Incoming in Station Advanced Feature Template	***5602-1-Template Number-Feature Number-Code***
To enable Call Taping Internal Flag in a Station Advanced Feature Template	***5602-1-Template Number-Feature Number-Code***
To enable Tape calls coming without CLI Flag in a Station Advanced Feature Template	***5602-1-Template Number-Feature Number-Code***

Call Transfer

To program transfer while ringing timer	***3806-Seconds***
To program transfer-on busy timer	***3807-Seconds***
To program trunk-to-trunk inactivity timer	***3808-Minutes***

Call Line Identification and Presentation

To program parameter in a SLT hardware Template	***5702-1-Template Number-Parameter Number-Code***
To program the feature in a Station Advanced Feature Template	***5602-1-Template Number-Feature Number-Code***
To program replacement ‘+’ string in CLI	***5334-Code***
To program replacement ‘+’ string in CLI	***5335-Replacement String-#****
To removed replacement ‘+’ string in CLI	***5335-#****

Class of Service (COS)

To enable a feature in a CoS group	***1302-1-CoS Group-Feature Number-Code***
To default a CoS group	***1301-1-CoS Group***
To assign CoS group to a template	***5502-1-Template Number-Feature Number-Code***

CLI Based Routing

To program the incoming number in a CLI table	***4101-Index-Telephone Number-#****
To clear a number from the CLI table	***4101-Index-#****
To program name of calling party	***4102-Index-Name-#****

To clear a name of calling party	***4102-Index-#****
To assign landing destination for the incoming number	***4103-Index-Port Type-Port Number***
To clear a location index in the CLI table	***4104-1-Index***

Clock Synchronization

To program the clock sources	***5341-Clock Source Index-Port Type-Port Offset***
To select 'System Clock Synchronization'	***5342-System Clock Synchronization***
To select the 'PLL Locking Mode'	***5343-PLL Locking Mode***

Closed User Group (CUG)

To program route code	***4502-1-Route Index-Route Code-#****
To clear a particular route code	***4502-1-Route Index-#****
To assign OGTBG to a router code	***4503-1-Route Index-OGTBG***
To program strip digit count for a route	***4504-1-Route Index-Strip Digit Count***
To program self route flag for a route	***4505-1-Route Index-Code***
To program maximum dialed digits to select router for a route code	***4506-1-Route Index-Dialed Digit Count***
To clear an entry in a routing table	***4501-1-Route Index***

Communication Ports

To set data transfer rate of a COM port	***3201-Port-Speed***
To set data bit of a COM port	***3202-Port-Data Bits***
To set parity of a COM port	***3203-Port-Parity***
To set stop bit of a COM port	***3204-Port-Stop Bits***
To set default parameters of a COM port	***3210-Port***

Configuring using a Telephone

Changing Login Session Time Out of Jeeves	***2118-Time***
Logging Out users from Jeeves	***2188***

Configuring WAN Port

To select Connection Type	***2116-Connection Type***
To program PPPoE User ID	***2117 - PPPoE User ID***
To program PPPoE Password	***2123- PPPoE Password***
To program PPPoE Service Name	***2124-PPPoE Service Name***
To assign IP Address to WAN Port	***2110-IP Address***

To assign Subnet Mask to the WAN Port	***2111-Subnet Mask***
To program the Gateway IP Address	***2112-Gateway IP Address***
To select DNS Assignment Type	***2115-DNS Address Assignment Type***
To program Primary DNS Server Address	***2113-Primary DNS Server Address***
To program Secondary DNS Server Address	***2114-Secondary DNS Server Address***
To program Web Server Port (Listening port of Jeeves)	***2121-Port***
To enable/disable Dynamic DNS	***2125-Code***
To program Dynamic DNS User ID	***2126-DDNS User ID***
To program Dynamic DNS User Password	***2127-DDNS User Password***
To program Dynamic DNS Host Name	***2128 - DDNS Host Name***
To program Dynamic DNS - Retry Trial Count	***2129 - DDNS Retry Count***
To Update Dynamic DNS IP Address binding ('Update IP Address Now?' flag)	***2130***
To program Router's Public IP Address	***2132-IP Address***
To program STUN Server Address	***2133-STUN Server Address***
To program STUN Server Port	***2134- Port***
To set STUN Query Interval	***2135 - Interval***
To view IP Address of the Master Ethernet Port	***2150***
To view Subnet Mask of the Master Ethernet Port	***2151***
To view Gateway Address of the Master Ethernet Port	***2152***
To view current STUN query status	***2159***
To view Router's Public IP Address fetched by STUN	***2160***
To view Dynamic DNS Status	***2161***
To view Ethernet link Status	***2162***

Configuring Extensions

To change the default value of a SLT Hardware Template	***5702-1-Template Number-Parameter Number-Code***
To default SLT Hardware Template	***5701-1-Template Number***
To apply the Customized SLT Hardware Template to SLT port	***5703-1-SLT-Template Number***
To program a feature in a Station Basic Feature Template	***5502-1-Template Number-Feature Number-Code***
To default a Station Basic Feature Template	***5501-1-Template Number***
To assign a Station Basic Feature Template to SLT	***5503-1-SLT-Template Number***
To assign a Station Basic Feature Template to a DKP	***5504-1-DKP-Template Number***

To assign a Station Basic Feature Template to an ISDN Terminal port	***5507-1-ISDN-Template Number***
To assign a Template to an E&M (Station) port	***5505-1-E&M-Template Number***
To assign a Station Basic Feature Template to a T1E1PRI port	***5506-1-T1E1PRI-Template Number***
To assign a Station Basic Feature Template to a BRI port	***5509-1-BRI-Template Number***
To assign a Station Basic Feature Template to a Magneto Port	***5511-1-Magneto Port-Template Number***
To assign a Station Basic Feature Template to a Virtual Extension.	***5513-1-Virtual Extension-Template Number***
To program a feature in a Station Advanced Feature Template	***5602-1-Template Number-Feature Number-Code***
To default a Station Advanced Feature Template	***5601-1-Template Number***
To assign a Station Advanced Feature Template to SLT	***5603-1-SLT-Template Number***
To assign a Station Advanced Feature Template to a DKP	***5604-1-DKP-Template Number***
To assign a Station Advanced Feature Template to an ISDN Terminal port	***5607-1-ISDN-Template Number***
To assign a Station Advanced Feature Template to an E&M (Station) port	***5605-1-E&M-Template Number***
To assign a Station Advanced Feature Template to a T1E1PRI port	***5606-1-T1E1PRI-Template Number***
To assign a Station Advanced Feature Template to a BRI port	***5609-1-BRI-Template Number***
To assign a Station Advanced Feature Template to a Virtual Extension.	***5613-1-Virtual Extension-Station Advance Feature Template***
To assign a Station Advanced Feature Template to a Magneto Port.	***5611-1-Magneto-Station Advanced Feature Template Number***

Configuring SLT Extensions

To assign Hardware Slot-Port to an SLT	***1101-SLT-Port Offset on the Card***
To de-assign the hardware slot and the hardware port of an SLT port	***1101-SLT-00-00***
To assign Access Code to an SLT Port	***3101-1-SLT-Access Code-#****
To clear the Access Code to assigned to the SLT Port	***3101-1-SLT-#****
To assign default Access Codes assigned to SLT	***3151-1-SLT***
To assign a Name to an SLT	***5402-1-SLT-Name***
To clear the name of the SLT	***5402-1-SLT-#****
To assign an SLT Hardware Template to an SLT port	***5703-1-SLT-Template Number***

To assign a Station Basic Feature Template to an SLT Port	***5503-1-SLT-Template Number***
To assign a Station Advanced Feature Template to an SLT Port	***5603-1-SLT-Template Number***
To assign a Call Pick-Up Group to an SLT Port	***3901-1-SLT-Call Pickup Group***
To remove an SLT from a Call Pick Up group	***3901-1-SLT-00***
To assign a Personal Directory to an SLT Port	***1905-1-SLT-Personal Directory***
To clear the Personal Directory assigned to the SLT	***1905-1-SLT-00***
To define the Priority for an SLT Port	***3911-1-SLT-Priority***

Configuring DKP Extensions

To assign Hardware Slot-Port to a DKP	***1102-DKP-Port Offset on the Card***
To de-assign the hardware slot and the hardware port of a DKP port	***1102-DKP-00-00***
To assign Access Code to a DKP Port	***3102-1-DKP-Access Code-#****
To clear the Access Code to assigned to the DKP Port	***3102-1-DKP-#****
To assign default Access Codes assigned to DKP	***3152-1-DKP***
To assign a Name to a DKP	***5403-1-DKP-Name***
To clear the name of the DKP	***5403-1-DKP-#****
To assign a Station Basic Feature Template to a DKP Port	***5504-1-DKP-Template Number***
To assign a Station Advanced Feature Template to a DKP Port	***5604-1-DKP-Template Number***
To define the Call Capacity of a DKP Port	***1201-1-DKP-Call Capacity***
To assign a Key Map for a DKP Port	***1221-1-DKP-DKP Key Template Number***
To assign the DKP Port to a Call Pick-Up Group	***3902-1-DKP-Call Pickup Group***
To remove a DKP from a Call Pick Up group	***3902-1-DKP-00***
To assign a Personal Directory to a DKP Port	***1906-1-DKP-Personal Directory***
To clear the Personal Directory assigned to the DKP	***1906-1-DKP-00***
To define the Priority for a DKP Port	***3912-1-DKP-Priority***
To select CO CLIP Pattern for a DKP Port	***1243-1-DKP-CO CLIP Pattern***
To select Language for a DKP port	***1224-1-DKP-Language***
To select Ringer Mode for a DKP Port	***1204-1-DKP-Ringer Mode***
To select Ring Delay Timer for a DKP Port	***1205-1-DKP-Ring Delay Timer***
To set the Acknowledgement Mode	***1206-1-DKP-Ringer Auto Acknowledge Mode***
To set the Ringer Auto Acknowledge Timer	***1207-1-DKP-Ringer Auto Acknowledge Timer***
To select Destination for 'Play Ring ON' for a DKP Port	***1220-1-DKP-Ring Destination***

To select Ring Tune for a DKP Port	***1202-1-DKP-Ring Tune***
To set Ringer Volume for a DKP Port	***1203-1-DKP-Ringer Volume***
To set Handset Transmit (Tx) Volume Level for a DKP	***1208-1-DKP-Handset MIC Volume Level***
To set Handset Receive (Rx) Volume Level for a DKP	***1209-1-DKP-Handset Speaker Volume Level***
To set Right Handset Transmit (Tx) Volume Level for a DKP	***1225-1-DKP-Handset MIC Volume Level***
To set Right Handset Receive (Rx) Volume Level for a DKP	***1226-1-DKP-Handset Speaker Volume Level***
To set Headset Transmit (Tx) Transmit Volume Level for a DKP	***1222-1-DKP-Headset MIC Volume Level***
To set Headset Receive (Rx) Volume Level for a DKP	***1223-1-DKP-Headset Speaker Volume Level***
To set Hands-free Transmit (Tx) Volume Level for a DKP	***1210-1-DKP-Speaker Phone MIC Volume Level***
To set Hands-free Receive (Rx) Volume Level for a DKP	***1211-1-DKP-Speaker Phone Speaker Volume Level***
To set Key Click Volume Level for a DKP	***1212-1-DKP-Key Click Volume***
To enable/disable DTMF Generation Flag for a DKP	***1241-1-DKP-DTMF Generation***
To set DTMF Transmit Level for a DKP	***1218-1-DKP-DTMF Transmit Level***
To enable/disable Headset Connectivity for a DKP	***1213-1-DKP-Headset Connectivity Flag***
To enable/disable Auto Answer for a DKP	***1214-1-DKP-Auto Call Answer Mode***
To set Auto Answer Timer (sec) for a DKP	***1215-1-DKP-Auto Call Answer Timer***
To set LCD Back Light Level of a DKP	***1216-1-DKP-LCD Backlight Level***
To change LCD Backlight OFF Timer of a DKP	***1219-1-DKP-LCD Backlight OFF Timer***
To change LCD Contrast Level of a DKP	***1217-1-DKP-LCD Contrast Level***
To assign Hardware Slot-Port to DSS connected to a DKP	***1103-DKP-DSS-Slot-Port Offset on the Card***
To clear the hardware Slot-Port assigned to the DSS software port	***1103-DKP-DSS-00-00***

Configuring ISDN Terminal

To assign BRI Software Port to an ISDN Terminal	***7301-1-ISDN Terminal-BRI***
To de-assign ISDN Terminal from a BRI Software Port	***7301-1-ISDN Terminal-00***
To assign Access Code to an ISDN Terminal	***3103-1-ISDN Terminal-Access Code-#****
To clear the Access codes assigned to the ISDN Terminal	***3103-1-ISDN Terminal-#****
To assign default Station Access Codes to ISDN Terminal	***3153-1-ISDN Terminal***
To assign a Name to an ISDN Terminal	***5409-1-ISDN Terminal-Name-#****
To clear the Name of the ISDN Terminal	***5409-1-ISDN Terminal-#****

To assign a Station Basic Feature Template to an ISDN Terminal	***5507-1-ISDN Terminal-Template Number***
To assign a Station Advanced Feature Template to an ISDN Terminal	***5607-1-ISDN Terminal-Template Number***
To assign a Personal Directory to an ISDN Terminal	***1907-1-ISDN Terminal-Personal Directory***
To clear the Personal Directory assigned to an ISDN Terminal	***1907-1-ISDN Terminal-00***
To define the Priority for an ISDN Terminal	***3913-1-ISDN Terminal-Priority***
To assign an ISDN Terminal to a Call Pick-Up Group	***3903-1-ISDN Terminal-Call Pick-Up Group***
To remove an ISDN Terminal from a Call Pick-Up Group	***3903-1-ISDN Terminal-00***

Configuring Region

To select Region	***5301-Region Code***

Configuring Operator

To assign a Time Table to an Operator	***1602-1-Operator-Time Table***
To define an Operator for Working Hours	***1611-1-Operator-Routing Group***
To define an Operator for Break Hours	***1612-1-Operator-Routing Group***
To define an Operator for Non-working Hours	***1613-1-Operator-Routing Group***
To default an Operator	***1601-1-Operator***

Configuring Trunks

To change the default value of a CO Hardware Parameter in a Template	***5902-1-CO Hardware Template Number-Feature Number-Code***
To default CO Hardware Template	***5901-1-CO Hardware Template***
To assign a CO Hardware Template to a CO port	***5903-1-CO-CO Hardware Template***
To change the default value of a Trunk Feature Parameter in a Template	***5802-1-Trunk Feature Template Number-Feature Number-Code***
To default Trunk Feature Templates	***5801-1-Trunk Feature Template Number***
To assign a Trunk Feature Template to a CO Trunk	***5803-1-CO-Trunk Feature Template Number***
To assign a Trunk Feature Template to a BRI Trunk	***5804-1-BRI-Trunk Feature Template Number***
To assign a Trunk Feature Template to an E&M Trunk	***5805-1-E&M-Trunk Feature Template Number***
To assign a Trunk Feature Template to a T1E1 Trunk	***5806-1-T1E1-Trunk Feature Template Number***
To assign a Trunk Feature Template to a Mobile Trunk	***5807-1-Mobile-Trunk Feature Template Number***
To change the default values of a Parameter in an E&M Feature Template	***6002-1-Template Number-Parameter Number-Code***
To restore default values to the Parameters of an E&M Feature Template	***6001-1-Template Number***

To assign an E&M Feature Template to an E&M port	***6003-1-E&M-Template Number***
To assign E&M Feature Template to a T1E1 Port	***6004-1-T1E1-Template Number***

Configuring CO Trunks

To assign Hardware Slot and Port to the CO Port	***1104-CO-Slot-Port offset on the Card***
To clear the Hardware ID assigned to a CO Port	***1104-CO-00-00***
To program a Name for a CO Port	***5404-1-CO-Name***
To clear the Name of a CO trunk	***5404-1-CO-#****
To enable/disable the CO Trunk	***3307-1-CO-Flag***
To assign a Trunk Feature Template to a CO Port	***5803-1-CO-Trunk Feature Template Number***
To assign a CO Hardware Template to a CO Port	***5903-1-CO-Hardware Template Number***
To assign a Cost Factor to a CO Port	***3308-1-CO-Cost Factor***
To enable / disable Call Back on CO Port	***3310-1-CO-Code***
To program Call Back Timer on CO Port	***3311-1-CO-Call Back Timer***
To select Call Back Mode for CO Port	***3312 -1-CO-Call Back Mode***
To program Call Back On for CO Port	***3313-1-CO-Call Back on***
To assign Call Back - Incoming Number List to a CO Port	***3314-1 -CO-Incoming Number List***
To assign Call Back - Outgoing Number List to a CO Port	***3315-1 -CO-Outgoing Number List***
To select Call Back From port for a CO Trunk port	***3316-1-CO-Call Back From***
To assign Call Back - OGTB Group to a CO port	***3317-1-CO-OGTB Group***

Configuring Mobile Trunks

To assign Hardware Slot and Port	***1108-Mobile-Slot-Port Offset on the Card***
To de-assign Hardware Slot and Port	***1108-Mobile-00-00***
To enable/disable the Mobile Port	***8000-1-Mobile-Flag***
To assign a Name to the Mobile Port	***5408-1-Mobile-Name-#****
To clear the Name of the Mobile Port	***5408-1-Mobile-#****
To select the frequency Band for the Mobile Port	***8009-1-Mobile Port Number-Mobile Frequency Band Code***
To select the Preferred Network Mode for the Mobile Port	***8038-1-Mobile Port Number-Preferred Network Mode***
To assign SIM PIN to the Mobile Port	***8006-1-Mobile Port Number-SIM PIN-#****
To select Incoming Call mode on the Mobile port	***8005-1-Mobile-Mode***
To assign a Trunk Feature Template to the Mobile Port	***5807-1-Mobile Port Number-Template Number***

To enable/disable CLIR on the Mobile Port	***8031-1-Mobile Port Number-CLIR***
To enable RCOC on Mobile Port	***8030-1-Mobile-Code***
To assign Cost Factor to a Mobile Port	***8001-1-Mobile-Cost Factor***
To set Network Registration Retry Count	***8004-1-Mobile-N/w Registration Retry Count***
To enable/disable Call Back on Mobile Port	***8010-1-Mobile-Code***
To change Call Back Timer	***8011-1-Mobile-Call Back Timer***
To select the Call Back Mode	***8012-1-Mobile-Call Back Mode***
To program Call Back on for the Mobile Port	***8013-1-Mobile-Call Back on***
To assign a Call Back - Incoming Number List	***8034-1-Mobile-Incoming Number List***
To assign a Call Back - Outgoing Number List to a Mobile Port	***8035-1-Mobile-Outgoing Number List***
To select Call Back From for a Mobile Port	***8036-1-Mobile-Call Back From***
To assign a Call Back - OGTB Group for a Mobile Port	***8037-1-Mobile-OGTB Group***
To assign a Trusted Caller List Number to the Mobile Port	***8013-1-Mobile-Number List***
To enable/disable Accept Anonymous Calls on Mobile Port	***8029-1-Mobile-Code***
To program the Pause Timer on Mobile Port	***8014-1-Mobile-Pause Timer***
To program DTMF ON Time on Mobile Port	***8015-1-Mobile-DTMF ON Time***
To set DTMF Detection Mode on Mobile Port	***8051-1-Mobile-Mode***
To program DTMF Detection Duration on Mobile Port	***8052-1-Mobile-Duration***
To assign the Mobile Port to a 'Category' for Logical Partition	***8018-1-Mobile-Category***
To enable/disable Gateway Application on the Mobile Port	***8016-1-Mobile-Gateway Application Flag***
To program the DTMF String for the Gateway Application on the Mobile Port	***8017-1-Mobile-DTMF String***
To enable/disable Debug on a Mobile Port	***8028-1-Mobile Port Number-Debug Code***
To program SIP Rx Gain on the Mobile Port	***8041-1-Mobile-SIP Rx Gain***
To program SIP Tx Gain on the Mobile Port	***8042-1-Mobile-SIP Tx Gain***
To select Network Selection mode	***8007-1-Mobile Port Number-Code***
To program the network operator codes in order of priority for Mobile port	***8008-1-Mobile-Priority-Network Operator Code-#****
To reset ASR and ACD calculation for Mobile Port	***8032-1-Mobile-1***

Configuring E&M Lines

To assign Hardware ID to an E&M Software Port	***1105-E&M-Port offset on the Card***
To clear the Hardware ID assigned to an E&M Software Port	***1105-E&M-00-00***

To enable/disable an E&M Port	***3321-1-E&M-Code***
To program a Name for an E&M Port	***5406-1-E&M-Name***
To clear a name assigned to an E&M port	***5406-1-E&M-#****
To assign an E&M Feature Template to an E&M Port	***6003-1-E&M-Template Number***
To assign a Trunk Feature Template to an E&M Port	***5805-1-E&M Template Number***
To assign a Station Basic Feature Template to an E&M Port	***5505-1-E&M-Template Number***
To assign a Station Advanced Feature Template to an E&M Port	***5605-1-E&M-Template Number***
To set a Priority Level for an E&M Port	***3915-1-E&M-Priority***
To assign a Cost Factor to an E&M port	***3322-1-E&M-Cost Factor***

Configuring Magneto Interface

To assign Slot-Port Assignment to a Magneto Port	***1110-Magneto Trunk-Slot-Port offset on the Card***
To enable/disable Magneto Port	***6801-1-Magneto-Code***
To program Access Code for Magneto Port	***3107-1-Magneto-Access Code-#****
To clear the Access Code assigned to the Magneto port	***3107-1-Magneto-#****
To assign default Access Codes assigned to Magneto Port	***3156-1-Magneto***
To assign a Name to the Magneto Port	***5411-1-Magneto-Name-#****
To clear the Name of the Magneto Port	***5411-1-Magneto-#****
To assign Priority level of a Magneto Port	***3919-1-Magneto-Priority***
To assign a Station Basic Feature Template to Magneto Port	***5511-1-Magneto-Station Basic Feature Template Number***
To assign a Station Advanced Feature Template for Magneto Port	***5611-1-Magneto-Station Advanced Feature Template Number***
To assign MRE Key to a DKP	***1261-1-DKP Key Template-EON Terminal Type-Key Number-Function Type-Function Number-Channel***
To enable or disable the "Enable Silence Detection on Magneto?" flag	***5357-Flag***
To set Magneto-Silence Detection Timer	***5356-Silence Detection Timer***
To set the Magneto Threshold Level	**5358 - Magneto VAD Threshold Level**

Configuring Virtual Extensions

To assign Access Code to a Virtual Extension	***3109-1-Virtual Extension-Access Code-#****
To clear the access codes for all the Virtual Extension	**3109-***
To assign a Name to a Virtual Extension	***5414-1-Virtual Extension-#****
To clear the name of the Virtual Extension	***2152-1-Virtual Extension-#****

To assign a Station Basic Feature Template to a Virtual Extension	***5613-1-Virtual Extension-Template Number***
To assign a Station Advanced Feature Template to a Virtual Extension	***5613-1-Virtual Extension-Station Advanced Feature Template***
To define the Priority for a Virtual Extension	***3919-1-Virtual Extension-Priority***
To program Landing Destination for Virtual Extension	**3001-1-Virtual Extension-Port Type-Port Number**

Configuring LCR

To program Time Zone at a Time Zone index	***3402-Time Zone Index-Start Time-End Time***
To program the Cost Factor (Service Provider preference) for the Time Zone	***3403-Time Zone Index-CF1-CF2-CF3-CF4***
To default the Time Zone based LCR table	***3401***
To program Cost Factor (Service Provider preference) for the each Number	***3412-Number Index-CF1-CF2-CF3-CF4***
To default the Number-based LCR table	***3410***
To define Time Zone for Time+Number-based LCR	***3421-Time Zone Index-Start Time-End Time***
To program Cost Factor (Service Provider preference) for the each Number and Time Zone	***3423-Number Index-Time Zone Index-CF1-CF2-CF3-CF4***
To default the Time and Number-based LCR table	***3420***
To program Area Code in the Area Code Table	***2620-Area Code Index-Area Code-#****
To clear an Area Code in the Area Code Table	***2620-Area Code Index-#****
To program Ignore Digit Count for an Area Code	***2623-Area Code Index-Ignore Digit Count***
To program Cost Factor (Service Provider preference) for the each Number	***3442-Number Index-CF1-CF2-CF3-CF4***
To default the Number-based LCR table	***3440***
To select LCR type in OG Trunk Bundle Group	***1404-1-OGTBG-LCR Type***

Configuring Emergency Number Dialing

To program Emergency Numbers in the table	***3116-Index-Emergency Number-#****
To program OG Trunk Bundle Group (OTBG) for an emergency number	***3117-Index-OG Trunk Bundle Group***

Conflict Dialing

To program conflict dialing timer	***5351-Seconds***

Conversation Recording

To enable/disable conversation recording beeps	***5332-Code***

Customer Name

To program the customer name	***5401-Customer Name-#****
To clear the customer name	***5401-#****

Day Night Mode

To set the system in Day/Night mode	***4801-Code***

Daylight Saving Time (DST)

To program the DST Mode	***1010-DST Mode***
To program the start time of DST when manual selected	***1011-Date-Month-Current Time-Advance Time***
To program the end time of DST when manual selected	***1012-Date-Month-Current Time-Delay Time***
To program the DST Type	***1013-DST Type***

DDI Routing Table

To program the feature in a DDI Routing Table	***6322-1-DDI Routing Table ID-Parameter Number-Value***
To default a DDI table	***6321-1-DDI Routing Table ID***

Department Call

To program the destination in the routing group	***6502-1-Routing Group-Destination Index-Port Type-Port Number***
To program the time for which each station in the group should ring	***6503-1-Routing Group-Destination Index-Ring Timer***
To program continuous or non-continuous ring for a station in the group	***6504-1-Routing Group-Destination Index-Flag***
To program rotation method of a department group	***6505-1-Routing Group-Routing Flag***
To assign routing group to the department group	***2001-1-Department Group Index-Routing Group***
To clear the routing group assigned to the department group	***2001-1-Department Group Index-00***
To program the access code for a department number	***3113-1-Department Group Index-Access Code***
To default the access code for a department group	***3163-1-Department Group Index***
To clear the access code for a department number	***3113-1-Department Group Index-#****

Digest Authentication

To program User Password in the Digest Authentication Table	***4119-Index-User Password***

Digital Key Phone-Soft Keys Programming

To assign a feature to a key in a Template	***1261-1-DKP Key Template-EON Terminal Type-Key Number-Function Type-Function Number***
To assign a DKP Key template to a DKP	***1221-1-DKP-DKP Key Template***
To assign a Personalized Key map to a DKP	***1221-1-DKP-0***
To assign a function to a DKP key	***1252-1-DKP-Key Number-Function Type-Function Number***

Built-In Auto Attendant

To set Built-In Auto Attendant inactivity timer	***2411-Seconds***
To set Built-In Auto Attendant answer wait timer	***2412-Seconds***
To set music timer	***2413-Seconds***
To set beeps timer	***2414-Seconds***
To set ring timer	***2415-Seconds***
To set busy tone timer	***2416-Seconds***
To set error tone timer	***2417-Seconds***
To enable/disable the Disconnect when Caller Doesn't Dial a Digit' flag	***5338-Code***
To enable/disable the Disconnect Built-In Auto Attendant call when Dialed Number Busy:	***5336-Code***
To enable/disable Disconnect Built-In Auto Attendant Call, when Dialed Number does Not Reply	***5337-Code***

Direct Inward System Access (DISA)

To program DISA idle state timer	***2420-Seconds***
To program DISA inactivity timer	***2421-Minutes***
To program Calling Number on DISA Application	***4111- Index-Calling Number-#****
To program Port Type and Port Number for DISA Application	***4112- Index-Port Type-Port Number***

Distinctive Rings

To demonstrate the ring	***4003-Ring Pattern***
To change demonstration timer	***3542-Seconds***
To make default all ring type situation	***4001***
To assign ring type to a situation	***4002-Event-Ring Pattern***

Do Not Disturb (DND)

To default all DND Text Messages	***1501***

Emergency Detection and Reporting

To enable/disable emergency reporting	***5110-Code***

Emergency Dialing

To program emergency numbers	***3116-Index-Emergency Number-#****
To program an OGTBG for emergency number	***3117-Index-OG Trunk Bundle Group***

Flexible Numbers

To program the access code for a SLT	***3101-1-SLT-Access Code-#****
To clear the access code for a SLT	***3101-1-SLT-#****
To assign the default flexible number of a SLT	***3151-1-SLT***
To program the access code for a DKP	***3102-1-DKP-Access Code-#****
To clear the access code for a DKP	***3102-1-DKP-#****
To assign the default flexible number of a DKP	***3152-1-DKP***

Floor Service

To program a routing group with member extensions	***6502-1-Routing Group-Destination Index-Port Type-Port Number***
To program the Ring Timer for the routing group	***6503-1-Routing Group-Destination Index-Ring Timer***
To program the Continuous Ring Flag for the routing group	***6504-1-Routing Group-Destination Index-Flag***
To program the routing group in a Station Advanced Feature Template	***5602-1-Template Number-11-Routing Group***

IC Reference Table

To default the IC Reference Table	***6301-1-IC Reference Table Index***
To program IC Reference Table Index	***6302-1-IC Reference Table Index-feature Number-Value***

Interrupt Request

To set interrupt request timer	***3802-Seconds***

ISDN BRI

To assign a name to the BRI port	***5405-1-BRI-Name***
To enable/disable BRI port	***6201-1-BRI-Port Status***
To assigning a service provider to a BRI port	***6202-1-BRI-SP***
To program the BRI ISDN switch variant of the BRI port	***6203-1-BRI-BRI ISDN Switch Variant***
To program orientation type for a BRI port	***6204-1-BRI-Orientation Type***
To program the type of interface companding for the BRI	***6205-1-BRI-Companding***
To program Power Feed	**6227-1- BRI Port- Feed Power to Port**
To program the idle code for the BRI	***6207-1-BRI-Idle Code***
To set overlap receiving timer for BRI	***6208-1-BRI-Timer***
To program Pause Timer	***6209-1-BRI- Pause Timer***
To program DTMF ON Time	***6210-1-BRI-DTMF ON Time***
To program DTMF Inter digit Pause Timer	***6211-1-BRI- DTMF Inter digit Pause Time***
To get appropriate debug information	***6291-1-BRI-Level-Code***
To program a caller TON for the BRI	***6221-1-BRI-Caller TON***
To program a caller NPI for the BRI	***6222-1-BRI-Caller NPI***
To program a called party TON for the BRI	***6223-1-BRI-Called Party TON***
To program a called party NPI for the BRI	***6224-1-BRI-Called Party NPI***
To enable/disable Call Back on BRI port	***6242-1-BRI- Call Back Flag***
To program Call Back Timer for BRI port	***6243-1-BRI-Call Back Timer***
To program Call Back Mode on BRI port	***6244-1-BRI-Call Back Mode***
To program Call Back On method for BRI port	***6245-1-BRI-Call Back on selection***
To assign Call Back - Incoming Number List to a BRI port	***6246-1-BRI-Incoming Number List***
To assign a Call Back - Outgoing Number List to a BRI port	***6247-1-BRI-Outgoing Number List***
To define Call Back From for a BRI port	***6248-1-BRI-Call Back From***
To assign a Call Back - OGTB Group for a BRI port	***6249-1-BRI-OGTB Group***
To program an OG reference ID-WH for the BRI	***6231-1-BRI-OG Reference ID***
To program an OG reference ID-BH for the BRI	***6241-1-BRI-OG Reference ID***
To program an OG reference ID-NH for the BRI	***6242-1-BRI-OG Reference ID***
To program an IC reference ID-WH on the BRI	***6232-1-BRI-IC Reference ID***
To program an IC reference ID-BH on the BRI	***6233-1-BRI-IC Reference ID***
To program an IC reference ID-NH on the BRI	***6234-1-BRI-IC Reference ID***

To program number of channels reserved for data transmission	***6235-1-BRI-Channel Count***
To reserve the number of channels for OG calls on BRI	***6236-1-BRI-Channel Count***
To reserve the number of channels for IC calls on a BRI	***6237-1-BRI-Channel Count***
To select TEI Negotiation on a BRI port	***6238-1-BRI-TEI Negotiation Mode***
To program TEI Negotiation value when programmed as Fixed	***6239-1-BRI-TEI Value***
To program access code for ISDN terminal	***3103-1-ISDN Terminal-Access Code-#****
To default access code for ISDN terminal	***3153-1-ISDN Terminal***
To assign a BRI software port to ISDN terminal	***7301-1-ISDN Terminal-BRI***
To de-assign a BRI software port to ISDN terminal	***7301-1-ISDN Terminal-00***
To assign personal directory to an ISDN terminal	***1907-1-ISDN Terminal-Personal Directory***
To program Gateway Application-Answer Signaling Flag	***6206-1-BRI-Gateway Application-Answer Signaling Flag***
To program "Gateway Application-Answer Signaling DTMF String".	***6212-1-BRI-Gateway Application-Answer Signaling DTMF String***
To program Layer 1 Mode for the BRI Port	***6225-1-BRI-Layer 1 Mode***

Key Board Macro

To create a macro	***1810-Macro Index-Number String***
To clear a macro	***1810-Macro Index-#****
To program access codes for the macro	***3115-1-Macro Index-Access Code***
To clear access code	***3115-1-Macro Index***
To program to default access code for the macro	***3165-1-Macro Index***

Logical Partition

To define call permission across and between Categories	***5317-Category-Category-Flag***

Live Call Screening

To program LCS timer	***3811-Seconds***

Message Wait

To program Message Wait Ring Count	***4403-Message Wait Ring Count***
To program the message wait ring timer	***4404-Message Wait Ring Timer***
To program the message wait ring interval timer	***4405-Message Wait Ring Interval Timer***

Number List

To program a number in a Number List	***4302-List Number-Location Index-Number-#****
To clear the number programmed in a Number List	***4302-List Number-Location Index-#****
To default a Number List	***4301-1-List Number***

OFF-Hook Alert

To enable/disable OFF-Hook Alert to Operator	***5333-Code***

OG Reference Table

To program an OG reference table	***6312-1-OG Reference Table Index-Parameter Number-Value***
To default an OG reference table	***6311-1-OG Reference Table Index***

OG Trunk Bundle

To program the feature in OG trunk bundle	***6702-1-OG Trunk Bundle Number-Port Type-Port Number-Code***
To set default values for OG trunk bundle	***6701-1-OG Trunk Bundle Number***

OG Trunk Bundle Group

To make default OGTBG	***1401-1-OGTBG Number***
To set OG trunk bundle	***1402-1-OGTBG No-Destination Index-OG Trunk Bundle***
To set rotation flag	***1403-1-OGTBG Number-Flag***
To program the desirable access code for a trunk access index	***3112-1-OGTBG Index-Access Code-#****
To clear the access codes for an OGTBG index	***3112-1-OGTBG Index-#****
To assign default access code for a OGTBG index	***3162-1-OGTBG Index***

Paging

To program a DKP in a Page Zone	***2302-1-Page Zone-Member-DKP***
To include/exclude a member DKP in/from a Page Zone	***2303-1-Page Zone-Member-Flag***

Presence

To enable/disable 'Display Presence Status during Call on DKP'	***5320-Flag***

Peer-to-Peer Calling

To program the number string in peer-to-peer table	***7801-1-Index-Number String-#****
To clear the number string for an index	***7801-1-Index-#****
To program the destination address in peer-to-peer table	***7802-1-Index-Destination Address-#****
To clear the destination address for an Index	***7802-1-Index-#****
To program the name in peer-to-peer table	***7803-1-Index-Name-#****
To clear the name for an index	***7803-1-Index-#****

Priority

To assign priority to SLT	***3911-1-SLT-Priority***
To assign priority to DKP	***3912-1-DKP-Priority***
To assign priority to ISDN Terminal	***3913-1-ISDN Terminal -Priority***
To assign priority to T1E1	***3914-1-T1E1-Priority***
To assign priority to E&M	***3915-1-E&M-Priority***
To assign priority to BRI	***3916-1-BRI-Priority***
To assign priority to Magneto	***3919-1-Magneto-Priority***

RCOC

To enable RCOC on T1E1 Trunk	***6145-1-T1E1-Code***
To enable RCOC on BRI Trunk	***6220-1-BRI-Code***
To enable RCOC on Mobile Port	***8030-1-Mobile-Code***
To change the RCOC Record Delete Timer	***3521- Minutes***
To change the Ring Back Tone Timer	***3503-Seconds***

Real Time Clock

To program the date format	***1000-Date Format***
To set the date	***1001-Date-Month-Year***
To select the Country for the time zone	***1002-Time Zone***
To set time	***1003-Hours-Minutes-Seconds***

Reminder

Program alarm/reminder ring timer	***2201-Seconds***
Program number of alarm attempts	***2202-Number of Alarm Attempts***
Program alarm attempt interval	***2203-Alarm Attempt Interval***

Assign destination port to the Hotel/Motel report	***3701-Flag***
Enable/disable snooze alarm/reminder	***2204-Code***

Routing Group

To program the destination in the routing group	***6502-1-Routing Group-Destination Index-Port Type-Port Number***
To program the time for which each station in the group should ring	***6503-1-Routing Group-Member Index-Ring Timer***
To program continuous or non-continuous ring for a station in the group	***6504-1-Routing Group-Member Index-Flag***
To program rotation method of a department group	***6505-1-Routing Group-Rotation Method***
To default a routing group	***6501-1-Routing Group***
To clear the destination in a routing group	***6510-1-Routing Group***

Software Port and Hardware ID

To assign hardware ID to a SLT software port	***1101-SLT-Slot-Port offset on the Card***
To clear the hardware ID assigned to a SLT software port	***1101-SLT-00-00***
To assign hardware ID to a DKP software port	***1102-DKP-Slot-Port offset on the Card***
To clear the hardware ID assigned to a DKP software port	***1102-DKP-00-00***
To assign hardware ID to DSS software port	***1103-DKP-DSS-Slot-Port offset on the Card***
To clear the hardware ID assigned to DSS software port	***1103-DKP-DSS-00-00***
To assign hardware ID to a CO software port	***1104-CO-Slot-Port offset on the Card***
To clear the hardware ID assigned to a CO software port	***1104-CO-00-00***
To assign hardware ID to an E&M software port	***1105-E&M-Slot-Port offset on the Card***
To clear the hardware ID assigned to an E&M software port	***1105-E&M-00-00***
To assign hardware ID to a BRI software port	***1106-BRI-Slot-Port offset on the Card***
To clear the hardware ID assigned to a BRI software port	***1106-BRI-00-00***
To assign hardware ID to a T1E1 software port	***1107-T1E1-Slot-Port offset on the Card***
To clear the hardware ID assigned to a T1E1 software port	***1107-T1E1-00-00***
To assign hardware ID to a mobile software port	***1108-Mobile-Slot-Port offset on the Card***
To clear the hardware ID assigned to mobile software port	***1108-Mobile-00-00***
To assign hardware ID to a Magneto software port	***1110-Magneto-Slot-Port offset of the card***

To de-assign hardware ID to a Magneto software port	***1110-Magneto-00-00***

Static Routing Table

To program Destination Address in Static Routing Table	***7811-1-Index-Destination Address***
To program Subnet Mask in the Static Routing Table	***7812-1-Index-Subnet Mask***
To program Gateway Address in the Static Routing Table	***7813-1-Index-Gateway Address***
To clear specific entries of the index of Static Routing Table	***7814-1-Index***

Station Message Detail Recording-Online

To assign destination port for Online SMDR-IC Call Record	***2930-Code***
To assign the IP Address to the Ethernet Port	***2932-IP Address***
To assign the IP Port	***2933-IP Port***
To assign a destination port for Online SMDR-Internal call record	***2830-Code***
To assign the IP Address to the Ethernet Port	***2832-IP Address***
To assign the IP Port	***2833-IP Port***
To assign destination port for Online SMDR-OG Call Record	***2730-Code***
To assign the IP Address to the Ethernet Port	***2732-IP Address***
To assign the IP Port	***2733-IP Port***
To program column position for serial number	***8200-Column Position***
To program field length for serial number	***8201-Field Length***
To program alignment for serial number	***8202-Alignment***
To program fill character for serial number	***8203-Fill Character***
To program reset for serial number	***8204-Reset***
To program column position for increment counter	***8205-Column Position***
To program reset for increment counter	***8206-Reset***
To program column position for property code	***8207-Column Position***
To program field length for property code	***8208-Field Length***
To program property code string for property code	***8209-Property Code String***
To program column position for station number	***8210-Column Position***
To program field length for station number	***8211-Field Length***
To program alignment for station number	***8212-Alignment***
To program fill character for station number	***8213-Fill Character***

To program column position for trunk number	***8214-Column Position***
To program format type for trunk number	***8215-Format Type***
To program column position for date field	***8216-Column Position***
To program field length for date field	***8217-Field Length***
To program alignment for date field	***8218-Alignment***
To program fill character for date field	***8219-Fill Character***
To program date format for date field	***8220-Date Format***
To program date fill flag for date field	***8257-Date Fill Flag***
To program column position for time field	***8222-Column Position***
To program field length for time field	***8223-Field Length***
To program alignment for time field	***8224-Alignment***
To program fill character for time field	***8225-Fill Character***
To program time format for time field	***8226-Time Format***
To program time fill flag for time field	***8258-Time Fill Flag***
To program column position for answer duration field	***8227-Column Position***
To program field length for answer duration field	***8228-Field Length***
To program alignment for answer duration field	***8229-Alignment***
To program fill character for answer duration field	***8230-Fill Character***
To enable/disable the Filler char. flag for Answer Duration	***8259-Filler Character Flag for Answer Duration***
To program duration unit for answer duration field	***8231-Duration Unit***
To program column position for hold duration field	***8232-Column Position***
To program field length for hold duration field	***8233-Field Length***
To program alignment for hold duration field	***8234-Alignment***
To program fill character for hold duration field	***8235-Fill Character***
To enable/disable the Filler character flag for Hold Duration	***8260-Filler Character Flag for Hold Duration***
To program column position for speech duration field	***8237-Column Position***
To program field length for speech duration field	***8238-Field Length***
To program alignment for speech duration field	***8239-Alignment***
To program fill character for speech duration field	***8240-Fill Character***
To enable/disable the Filler char. flag for Speech Duration	***8261-Filler Character Flag for Speech Duration***
To program column position for called number field	***8242-Column Position***
To program field length for called number field	***8243-Field Length***
To program alignment for called number field	***8244-Alignment***
To program number format for called number field	***8245-Number Format***

To program column position for calling number field	***8246-Column Position***
To program field length for calling number field	***8247-Field Length***
To program alignment for calling number field	***8248-Alignment***
To program number format for calling number field	***8249-Number Format***
To program column position for Digits dialed in Built-in Auto Attendant	***8250-Column Position***
To program field length for Digits dialed in Built-in Auto Attendant	***8251-Field Length***
To program alignment for Digits dialed in Built-in Auto Attendant	***8252-Alignment***
To program column position for remarks field	***8253-Column Position***
To program field length for remarks field	***8254-Field Length***
To program alignment for remarks field	***8255-Alignment***
To assign default IC SMDR format	***8256***

Station Message Detail Recording-Posting

To enable/disable storage of SMDR OG Calls	***2701-Code***
To assign Destination Port for SMDR-OG Posting	***8330-Code***
To program SMDR OG Posting Protocol	***8301-SMDR OG Posting Protocol***
To set ENQ no response timer	***8302-ENQ No Response Timer***
To set ENQ no response retry count	***8303-ENQ Retry Count***
To set ENQ no response Retry Timer	***8304-ENQ No Response Retry Timer***
To set ENQ Retry Count	***8305-ENQ Retry Count***
To set ENQ Retry Time	***8306-ENQ Retry Time***
To set SMDR-OG posting Data response to Data Timeout	***8307-Response to Data Timeout***
To set Data Transfer Retry Count (No Response)	***8308-Data Transfer Retry Count***
To set Data Transfer Retry Time (No Response)	***8309-Data Transfer Retry Time***
To set Data Transfer Retry Count (Negative Response)	***8310-Data Transfer Retry Count***
To set Data Transfer Retry Time (Negative Response)	***8311-Data Transfer Retry Time***
To enable/disable ENQUIRE Signal	***8312-ENQUIRE Signal***
To set the ENQUIRE character	***8313-ENQUIRE***
To set the ACK Character	***8314-Set ACK Character***
To program the NAK Character	***8315-Set NAK Character***
To set the Start of Packet string	***8316-Start of Packet***
To program the End of Packet string	***8317-End of Packet***

To enable/disable BCC Flag	***8318-BCC Flag***
To set SMDR-OG Posting Parameters to Default Value	***8300***
To program column position for serial number	***8100-Column Position***
To program field length for serial number	***8101-Field Length***
To program alignment for serial number	***8102-Alignment***
To program fill character for serial number	***8103-Fill Character***
To program reset for serial number	***8104-Reset***
To program column position for increment counter	***8105-Column Position***
To program reset for increment counter	***8106-Reset***
To program starting character for increment counter	***8174-Starting Character***
To program column position for property code	***8107-Column Position***
To program field length for property code	***8108-Field Length***
To program property code string for property code	***8109-Property Code String***
To program column position for station number	***8110-Column Position***
To program field length for station number	***8111-Field Length***
To program alignment for station number	***8112-Alignment***
To program fill character for station number	***8113-Fill Character***
To program column position for trunk number	***8114-Column Position***
To program format type for trunk number	***8115-Format Type***
To program column position for date field	***8116-Column Position***
To program field length for date field	***8117-Field Length***
To program alignment for date field	***8118-Alignment***
To program fill character for date field	***8119-Fill Character***
To program date format for date field	***8120-Date Format***
To program date fill flag for date field	***8170-Date Fill Flag***
To program column position for time field	***8122-Column Position***
To program field length for time field	***8123-Field Length***
To program alignment for time field	***8124-Alignment***
To program fill character for time field	***8125-Fill Character***
To program time format for time field	***8126-Time Format***
To program time fill flag for time field	***8171-Time Fill Flag***
To program column position for duration field	***8127-Column Position***
To program field length for duration field	***8128-Field Length***
To program alignment for duration field	***8129-Alignment***
To program fill character for duration field	***8130-Fill Character***

To program duration unit for duration field	***8131-Duration Unit***
To program duration fill flag for duration field	***8172-Duration Fill Flag***
To program column position for units field	***8132-Column Position***
To program field length for units field	***8133-Field Length***
To program alignment for units field	***8134-Alignment***
To program fill character for units field	***8135-Fill Character***
To program column position for amount field	***8136-Column Position***
To program field length for amount field	***8137-Field Length***
To program alignment for amounts field	***8138-Alignment***
To program fill character for amounts field	***8139-Fill Character***
To program amount format for amount field	***8140-Amount Format***
To program amount fill flag for amount field	***8173-Amount Fill Flag***
To program column position for currency symbol field	***8141-Column Position***
To program field length for currency symbol field	***8142-Field Length***
To program alignment for currency symbol field	***8143-Alignment***
To program fill character for currency symbol field	***8144-Fill Character***
To program symbol for currency symbol field	***8145-Character1.....Character8***
To program column position for call type indicator field	***8146-Column Position***
To program field length for call type indicator field	***8147-Field Length***
To program alignment for call type indicator field	***8148-Alignment***
To program number string for call type indicator field	***8149-Number Index-1-Number String***
To program text string for call type indicator field	***8149-Number Index-2-Text String***
To program column position for called location field	***8150-Column Position***
To program field length for called location field	***8151-Field Length***
To program alignment for called location field	***8152-Alignment***
To program column position for called number field	***8154-Column Position***
To program field length for called number field	***8155-Field Length***
To program alignment for called number field	***8156-Alignment***
To program number format for called number field	***8157-Number Format***
To program column position for account code field	***8158-Column Position***
To program field length for account code field	***8159-Field Length***
To program alignment for account code field	***8160-Alignment***
To program fill character for account code field	***8161-Fill Character***
To program prefix string (ac01)	***8165-Code***
To program column position for remarks field	***8166-Column Position***

To program field length for remarks field	***8167-Field Length***
To program alignment for remarks field	***8168-Alignment***
To assign default CDR format	***8169***
To program a country code	***8321-Index-Country Code-#****
To program a location in country code	***8322-Index-Country Code-#****
To start/stop SMDR Posting Process	***8333-Code***
To program the destination IP Address for Posting SMDR-OG Call Record	***8331-Address-Address-Address-Address***
To program the destination IP Port for Posting SMDR-OG call Record	***8332-Destination IP Port***
To program Listening Port of ETERNITY GENX	***8334-Listening Port***

Station Message Detail Recording-Report

To assign a destination port for SMDR-IC Report	***2931-Code***
To assign the IP Address to the Ethernet Port	***2934-IP Address***
To assign the IP Port	***2935-IP Port***
To assign destination port for Online SMDR-Internal Report	***2831-Code***
To assign the IP Address to the Ethernet Port	***2834-IP Address***
To assign the IP Port	***2835-IP Port***
To assign a destination port for SMDR-OG Report	***2731-Code***
To assign the IP Address to the Ethernet Port	***2734-IP Address***
To assign the IP Port	***2735-IP Port***

Station Message Detail Recording-Storage

To set SMDR storage mode (IC)	***2901-Storage Flag***
Store Normal Calls	***2902-Flag***
Store Built-in Auto Attendant Calls	***2903-Flag***
Store Unanswered Calls	***2904-Flag***
Store Unanswered Built-in Auto Attendant Calls	***2905-Flag***
Store DISA Calls	***2906-Flag***
Store Calls-Speech Duration More than	***2907-Seconds***
Store Calls-Unanswered Duration for more than	***2908-Seconds***
Store Calls-Hold Duration for more than	***2909-Seconds***
To default the incoming call storage filters	***2915***
To set SMDR storage mode (Internal)	***2801-Storage Flag***
Store Calls-Speech duration	***2802-Seconds***

To default the internal call storage filters	***2815***
To set SMDR storage flag (OG)	***2701-Storage Flag***
To assign a Number List containing numbers for call storage	***2702-Number List***
To set the filter of call duration	***2703-Seconds***
To set the filter for call Units	***2704-Unit***
To default the outgoing call storage filters	***2715***
To set the Call Toggle flag	***2716-Toggle Flag***
To set the originating flag	***2717-Originating Flag***

System Activity Log

To program the storage flag to enable/disable SAL	***6401-Storage Flag***
To assign a port for online printing of SAL	***6402-Port***
To assign the IP Address to the Ethernet Port	***6404-IP Address***
To assign the IP Port	***6405-IP Port***
To assign a port for offline printing of SAL	***6403-Port***
To assign the IP Address to the Ethernet Port	***6406-IP Address***
To assign the IP Port	***6407-IP Port***
To default system activity log parameters for SAL	***6410***

System Debug

To start/stop debug for required process	***2104-Value***
To start/stop state debug	***2105-Port Type-Port Number Start-Port Number End-Flag***
To enable ETERNITY GENX Host debug	***2181-1-Code***
To set IP Address for ARM Debug	***2182-IP Address***
To set Port for ARM Debug	***2183-Port***
To enable ETERNITY GENX DSP Para 1 Debug	***2184-1-Code***
To enable ETERNITY GENX DSP Para 2 Debug	***2184-2-Code***
To set IP Address for DSP Debug	***2185-IP Address***
To set IP Port for DSP Debug	***2186-Port***
To enable or disable PCM capture - Debug	***2172-Slot Number-Hardware Port Offset- Code***
To program the IP Address of the Syslog Server	***2178-Syslog Server IP Address***
To program Port number on which ETERNITY GENX shall send debug to Syslog Server	***2179-Syslog Server's Listening Port***
To initiate the debug of IO operations	***2199-Slot Number-1-Port Number-Code***

System Fault Log

To enable/disable storage of faults	***6451-Flag***
To assign the port to system fault log-online printing	***6452-Port***
To assign the IP Address to the Ethernet Port	***6454-IP Address***
To assign the IP Port	***6455-IP Port***
To assign a port to system fault log-offline printing	***6453-Port***
To assign the IP Address to the Ethernet Port	***6456-IP Address***
To assign the IP Port	***6457-IP Port***

System Parameters

To assign Station Type to SLT	***3921-1-SLT-Station Type***
To assign Station Type to DKP	***3922-1-DKP-Station Type***
To assign Station Type to ISDN Terminal	***3923-1-ISDN Terminal-Station Type***
To restart the Card installed in Universal Slot of the ETERNITY GENX	***2187-Slot No***
To restart the system	***5305-Reverse SE Password***
To load default values of all system timers	***5303***
To know the software version/revision of the system	***5304***
To enable/disable form feed	***5321-Code***
To select a Language for SE, SA and Front Desk User Web Interface	***5319-Language***
To set the A-law/M-law	***5322-Type***
To monitor a port	***7902-Slot-LED Number-Port***
To enable/disable the watch dog	***5309-Flag***
To enable/disable feature tone	***5312-Feature Tone Flag***
To program the end of dialing digit	***5313-End of Dialing Digit***
To program Caller ID digits to be considered as call from Public Network	***5314-Minimum Caller ID Digits***
To program Listening Port of Web Server	***2121-Port***
To select Hotel/Enterprise Mode	***5315-Code***
To program default Call Budget Amount	***3710-Preset Call Budget Amount***
To display 'MAC Address' on DKP	***2122***
To program Call Proceeding Tone Type for multi-stage dialing	***5311-Call Proceeding Tone Type***

System Security

To change SE password	***5306-New SE Password***

To assign a new password to the SA	***5310-New SA Password***

T1 Maintenance

To enable/disable T1 FDL on a T1E1 port	***6164-1-T1E1-T1 FDL***
To program the T1 FDL protocol for a T1E1 port	***6165-1-T1E1-T1 FDL Protocol***

T1 RBS Parameters

To program the T1 line signaling variants for the T1E1 port	***6181-1-T1E1-T1 Line Signaling Variants***
To program T1 wink timer for T1E1	***6182-1-T1E1-Wink Timer***
To program the T1 wink wait timer for T1E1	***6183-1-T1E1-Wink Wait Timer***
To program the T1 wait wink timer for T1E1	***6184-1-T1E1-Wait Wink Timer***
To program the T1 delay duration for T1E1	***6185-1-T1E1-Delay Duration***
To program the T1 Start Delay timer for T1E1	***6186-1-T1E1-Start Delay timer***
To program T1 register signaling variant for the T1E1 port	***6161-1-T1E1-T1 Register Signaling Variant***
To select the Inbound ANI/DNIS Format	**6166-1-T1E1-Code**
To assign the Inbound Delimiter (?) character	**6167-1-T1E1-Character**
To select the Outbound ANI/DNIS Format	**6168-1-T1E1-Code**
To assign the Outbound Delimiter (?) character	**6169-1-T1E1-Character**
To program digital pulse dial ratio for the T1E1 port	***6163-1-T1E1-Code***

T1E1 Trunks

To enable/disable the port	***6101-1-T1E1-Port Status***
To assign a name to the T1E1 port	***5407-1-T1E1-Name***
To select the line Carrier	***6108-1-T1E1-Carrier***
To program the line coding mechanism for the E1 port	***6103-1-T1E1-Line Coding***
To program the line coding mechanism for the T1 port	***6195-1-T1E1-Line Coding***
To program the framing mode for the E1 port	***6104-1-T1E1-Framing***
To program the framing mode for the T1 port	***6196-1-T1E1-Framing***
To program Signaling type/line type for E1	***6105-1-T1E1-Line Type***
To program Signaling type/line type for T1	***6151-1-T1E1-Line Type***
To program interface companding of a T1E1	***6108-1-T1E1-Interface Companding***
To program auto receive equalization mode	***6110-1-T1E1-Mode***
To program the receive equalization parameters of a T1E1	***6111-1-T1E1-Receive Equalization Parameters***

To program glare option for the T1E1 port	***6112-1-T1E1-Glare Option***
To program the Category for the T1E1 port	***6121-1-T1E1-Category***
To program the idle code of a T1E1	***6113-1-T1E1-Idle Code***
To program overlap receiving timer	***6114-1-T1E1-Timer***
To program Pause Timer	***6109-1-T1E1-Pause Timer***
To program DTMF ON Time	***6117-1-T1E1-DTMF ON Time***
To program DTMF Inter Digit Pause Timer	***6118-1-T1E1-DTMF Inter Digit Pause Timer***
To program orientation type for the T1E1 port	***6106-1-T1E1-Orientation Type***
To program source TON for a T1E1	***6126-1-T1E1-Source TON***
To program source NPI for T1E1	***6127-1-T1E1-Source NPI***
To program OG destination party TON or a T1E1	***6128-1-T1E1-Destination TON***
To program OG destination party NPI for T1E1	***6129-1-T1E1-Destination NPI***
To program to select whether the inband tones should be feed on T1E1-NT before sending DISCONNECT message?	***6130-1-T1E1-Flag***
To program the dial tone flag for T1E1	***6115-1-T1E1-Flag***
To program the routing tone flag for T1E1	***6116-1-T1E1-Flag***
To reserved channels for data transmission on T1E1	***6135-1-T1E1-Channel Count (Data)***
To program number of channels reserved for OG	***6136-1-T1E1-Channel Count (OG)***
To program number of channels reserved for IC	***6137-1-T1E1-Channel Count (IC)***
To assign OG reference ID to T1E1	***6131-1-T1E1-OG Reference ID***
To assign IC reference ID for working hour	***6132-1-T1E1-IC Reference ID***
To assign IC reference ID for break hour	***6133-1-T1E1-IC Reference ID***
To assign IC reference ID for non-working hour	***6134-1-T1E1-IC Reference ID***
To assign a Cost Factor to the T1E1 Trunk	***6102-1-T1E1-Cost Factor***
To active/deactivate near end loopback for T1E1	***6141-1-T1E1-Loopback***
To start/stop far end loopback test for T1E1	***6142-1-T1E1-Code***
To assign the port to performance report for online printing	***6143-Port***
To assign the port to performance report for offline printing	***6144-Port***
To program E1 line signaling variant for the E1	***6152-1-T1E1-E1 Line Signaling Variant***
To program E1 register signaling variant for the E1	***6153-1-T1E1-E1 Register Signaling Variant***
To enable/disable E1 auto alarm for the T1E1 E1 signaling	***6154-1-T1E1-Flag***
To program the line build out parameters of a T1E1	***6162-1-T1E1-Code***
To enable/disable customer pulse width flag for T1E1 T1 signaling	***6171-1-T1E1-Flag***

To program customer pulse width word1 for T1E1 T1 signaling	***6172-1-T1E1-Customer Pulse Width Word 1***
To program customer pulse width word2 for T1E1 T1 signaling	***6173-1-T1E1-Customer Pulse Width Word 2***
To program customer pulse width word3 for T1E1 T1 signaling	***6174-1-T1E1-Customer Pulse Width Word 3***
To program customer pulse width word4 for T1E1 T1 signaling	***6175-1-T1E1-Customer Pulse Width Word 4***
To enable/disable customer pulse width flag for T1E1 E1 signaling	***6155-1-T1E1-Flag***
To program customer pulse width word1 for T1E1 E1 signaling	***6156-1-T1E1-Customer Pulse Width Word 1***
To program customer pulse width word2 for T1E1 E1 signaling	***6157-1-T1E1-Customer Pulse Width Word 2***
To program customer pulse width word3 for T1E1 E1 signaling	***6158-1-T1E1-Customer Pulse Width Word 3***
To program customer pulse width word4 for T1E1 E1 signaling	***6159-1-T1E1-Customer Pulse Width Word 4***
To program the T1 DTMF digit timer for T1E1	***6187-1-T1E1-DTMF Digit Timer***
To program the T1 DTMF inter digit timer for T1E1	***6188-1-T1E1-DTMF Inter Digit Timer***
To program the ISDN PRI switch variant	***6107-1-T1E1-ISDN PRI Switch Variant***
To program Send Called Party Number Using for T1E1 port	***6146-1-T1E1-Send Called Party Number Using***
To program forward tone maximum ON timer	***7101-1-T1E1-Forward Tone Maximum ON Timer***
To program forward tone maximum OFF timer	***7102-1-T1E1-Forward Tone Maximum OFF Timer***
To program the maximum compelled cycle time	***7103-1-T1E1-Maximum Compelled Cycle Time***
To program pulse duration for pulsed signals	***7104-1-T1E1-Pulse Duration for Pulsed Signals***
To program the pulsed signal maximum wait timer	***7105-1-T1E1-Pulsed Signal Maximum Wait Timer***
To program first forward tone wait timer	***7106-1-T1E1-First Forward Tone Wait Timer***
To program minimum MF signal persist timer	***7107-1-T1E1-Minimum MF Signal Persist Timer***
To program to set DNIS END type (outbound) for T1E1	***7108-1-T1E1-End of DNIS***
To program DNIS end type (inbound) for T1E1	***7109-1-T1E1-DNIS End Type***
To program the ANI send position	***7110-1-T1E1-ANI Send Position***
To program the Is ANI available (outbound)	***7111-1-T1E1-Is ANI Available***
To program the positive response to Is ANI available (outbound)	***7112-1-T1E1-Positive Response to Is ANI Available***
To program the negative response to Is ANI available (outbound)	***7113-1-T1E1-Negative Response to Is ANI Available***
To program the end of ANI with presentation allowed (outbound)	***7114-1-T1E1-ANI End Tone with Presentation Allowed***

To program the end of ANI with presentation restrict (outbound)	***7115-1-T1E1-ANI End Tone with Presentation Restrict***
To program the DNIS digit length	***7116-1-DNIS Digit Length***
To program the DNIS request position	***7117-1-T1E1-ANI Request Position***
To program ANI length	***7118-1-T1E1-ANI Length***
To program the Ask ANI available	***7119-1-T1E1-Ask ANI***
To program positive response to ask ANI	***7120-1-T1E1-Positive Response to Ask ANI***
To program negative response to ask ANI	***7121-1-T1E1-Negative Response to Ask ANI***
To program end tone presentation allowed (inbound)	***7122-1-T1E1-ANI End Tone Presentation Allowed***
To program end tone presentation restrict (inbound)	***7123-1-T1E1-ANI End Tone Presentation Restrict***
To program Ask Calling Party Sub Category	***7160-1-T1E1-Ask Calling Party Sub Category***
To program the ordinary subscriber	***7124-1-T1E1-Ordinary Subscriber***
To program the priority subscriber	***7125-1-T1E1-Priority Subscriber***
To program the maintenance equipment	***7126-1-T1E1-Maintenance Equipment***
To program the operator	***7127-1-T1E1-Operator***
To program the pay phone	***7128-1-T1E1-Pay Phone***
To program the data transmission	***7129-1-T1E1-Data Transmission***
To program the interception operator	***7130-1-T1E1-Interception Operator***
To program the send next digit	***7131-1-T1E1-Send Next Digit***
To program the send last but one digit	***7132-1-T1E1-Send Last but One Digit***
To program the send last but two digit	***7134-1-T1E1-Send Last but Two Digit***
To program the send last but three digit	***7135-1-T1E1-Send Last but three Digit***
To program Send Caller Party Category and ANI Digit	***7133-1-T1E1-Send Caller Party Category and ANI Digit***
To program the address complete, change over to reception of group B signals	***7136-1-T1E1-Address Complete, Change Over to Reception of Group B***
To program the send calling party category and change to group C	***7137-1-T1E1-Send Calling Party Category and Change to Group C***
To program congestion in the national network	***7138-1-T1E1-Congestion in National Network***
To program the send caller party's category	***7139-1-Send Caller Party Category***
To program the address-complete, charge, setup speech of condition group B signals	***7140-1-T1E1-Address Complete, Charge, Setup Speech Condition***
To program repeat DNIS digits from beginning of group B signals	***7141-1-T1E1-Repeat DNIS Digits from Beginning***
To program the send next ANI digit	***7142-1-T1E1-Send Next ANI Digit***
To program the send special information tone	***7143-1-T1E1-Send Special Information Tone***
To program the send special information tone and setup speech condition	***7144-1-T1E1-Send Special Information Tone and Setup Speech Condition***

To program the subscriber line busy	***7145-1-T1E1-Subscriber Line Busy***
To program the subscriber line free, charge	***7146-1-T1E1-Subscriber Line Free, Charge***
To program the subscriber line free, no charge	***7147-1-T1E1-Subscriber Line Free, No Charge***
To program congestion	***7148-1-T1E1-congestion***
To program the unallocated number	***7149-1-T1E1-Unallocated Number***
To program the subscriber's line out of order	***7150-1-T1E1-Subscriber's Line Out of Order***
To program the call rejected, no indication	***7151-1-T1E1-Reject Call***
To program the alternative answer tone	***7152-1-T1E1-Alternative Answer Tone***
To program the changed number (announcement on line)	***7153-1-T1E1-Changed Number (Announcement on Line)***
To program the send next ANI digit (group C)	***7154-1-T1E1-Send Next Digit***
To program the request transition to group A and restart from first DNIS	***7155-1-T1E1-Request Transition to Group A and Restart from First DNIS***
To program the address completed, change to reception of Group B signal	***7156-1-T1E1-Address Completed, Change to Reception of Group B Signal***
To program the tone for congestion	***7157-1-T1E1-Congestion***
To program the tone for request transition back to group A and send next DNIS signal	***7158-1-T1E1-Request Transition Back to Group A, and Send Next DNIS***
To program the tone for request transition back to group A and restart the last DNIS signal	***7159-1-T1E1-Request Transition Back to Group A, and Restart the Last DNIS***
To program the CD bits of the T1E1 port	***7161-1-T1E1-CD Bits***
To program the invert/don't invert bit A for the T1E1	***7162-1-T1E1-Invert Bit A***
To program the invert/don't invert bit B for the T1E1	***7163-1-T1E1-Invert Bit B***
To program the invert/don't invert bit C for the T1E1	***7164-1-T1E1-Invert Bit C***
To program the invert/don't invert bit D for the T1E1	***7165-1-T1E1-Invert Bit D***
To program the E1 metering bit for the T1E1 port	***7166-1-T1E1-E1 Metering Bit***
To program the metering pulse minimum timer for the T1E1	***7167-1-T1E1-E1 Metering Pulse Minimum Timer***
To program the clear back signal for the T1E1	***7168-1-T1E1-Clear Back Signal***
To program the release timer for the T1E1	***7169-1-T1E1-Release Timer***
To program line seizure acknowledge wait timer	***7170-1-T1E1-Line Seizure Acknowledge Wait Timer***
To program release guard timer	***7171-1-T1E1-Release Guard Timer***
To program signaling type/line type of a T1E1	***6105-1-T1E1-Line Type***
To assign E&M Feature Template to T1E1	***6004-1-T1E1-Template Number***
To select B Bit value	***7191-1-T1E1-Code***
To program B Bit value	***7192-1-T1E1-B Bit Value***
To program CD Bit value	***7193-1-T1E1-CD Bit Value***

To program to invert/don't invert Bit A for the T1E1 port	***7162-1-T1E1-Invert Bit A***
To program to invert/don't invert Bit B for the T1E1 port	***7163-1-T1E1-Invert Bit B***
To program to invert/don't invert Bit C for the T1E1 port	***7164-1-T1E1-Invert Bit C***
To program to invert/don't invert Bit D for the T1E1 port	***7165-1-T1E1-Invert Bit D***
To program the line build out parameters of a T1E1	***6162-1-T1E1-Code***
To program Gateway Application-Answer Signaling Flag	***6119-1-T1E1-Gateway Application-Answer Signaling Flag***
To program "Gateway Application-Answer Signaling DTMF String".	***6120-1-T1E1-Gateway Application-Answer Signaling DTMF String***
To enable/disable Call Back on T1E1 port	***6176-1-T1E1- Call Back Flag***
To program Call Back Timer for T1E1 port	***6177-1-T1E1-Call Back Timer***
To program Call Back Mode on T1E1 port	***6178-1-T1E1-Call Back Mode***
To program Call Back On method for T1E1 port	***6179-1-T1E1-Call Back on selection***
To assign Call Back - Incoming Number List to a T1E1 port	***6180-1-T1E1-Incoming Number List***
To assign a Call Back - Outgoing Number List to a T1E1 port	***6147-1-T1E1-Outgoing Number List***
To define Call Back From for a T1E1 port	***6148-1-T1E1-Call Back From***
To assign a Call Back - OGTB Group for a T1E1 port	***6149-1-T1E1-OGTB Group***
To start/stop debug the parameters for the T1E1 port	***6191-1-T1E1-Level-Code***
To enabled Global Level-1 debug for T1E1 Port	***6191-1-T1E1-1-Code***
To enabled Global Level-2 debug for T1E1 Port	***6191-1-T1E1-2-Code***
To enabled T1E1 Port Level Debug	***6192-1-T1E1-1-Code***
To enable T1E1 Port-Port Level Physical Layer Debug	***6192-1-T1E1-2-Code***

Time Tables

To program a timetable	***1052-1-Time Table-Day-Time Zone-Start Time-End Time***
To default a time table	***1051-1-Time Table***

Toll Control

To program Local Numbers Allowed List	***4303-Index-Number String-#****
To program Local Numbers Denied List	***4304-Index-Number String-#****
To default the Local Numbers List	***4311-Reverse SE Password***
To program Regional Numbers Allowed List	***4305-Index-Number String-#****

To program Regional Numbers Denied List	***4306-Index-Number String-#****
To default the Regional Number List	***4312-Reverse SE Password***
To program National Numbers Allowed List	***4307-Index-Number String-#****
To program National Numbers Denied List	***4308-Index-Number String-#****
To default the National Number List	***4313-Reverse SE Password***
To default selected number list	***4301-1-List Number***
To program Limited Calls - Allowed/Denied List	***4302-List Number-Location Index-Number-#****
To clear a number programmed in a List	***4302-List Number-Location Index-#****
To program Toll Control - Level 0 (WH)	***5502-1-Template-07-Call Privilege Type***
To program Toll Control - Level 0 (WH) Allowed List	***5502-1-Template-08-Number List***
To program Toll Control - Level 0 (WH) Denied List	***5502-1-Template-09-Number List***
To program Toll Control - Level 0 (BH)	***5502-1-Template-10-Call Privilege Type***
To program Toll Control - Level 0 (BH) Allowed List	***5502-1-Template-11-Number List***
To program Toll Control - Level 0 (BH) Denied List	***5502-1-Template-12-Number List***
To program Toll Control - Level 0 (NH)	***5502-1-Template-13-Call Privilege Type***
To program Toll Control - Level 0 (NH) Allowed List	***5502-1-Template-14-Number List***
To program Toll Control - Level 0 (NH) Denied List	***5502-1-Template-15-Number List***
To program Toll Control - Level 1	***5502-1-Template-16-Call Privilege Type***
To program Toll Control - Level 2	***5502-1-Template-17-Call Privilege Type***
To program Toll Control - Level 3	***5502-1-Template-18-Call Privilege Type***
To program Toll Control - Call Budget Consumed	***5502-1-Template-19-Call Privilege Type***

Trunk Reservation

To program trunk reservation timer	***3804-Minutes***

Using Full Programming Access

Changing Region using a Telephone	***5301-Region Code***

Voice Message Applications

To select a source to record a VM	***2501-Code***
To record a voice message	***2502-Voice Module Number***
To verify a message	***2503-Voice Module***
To define voice message duration for voice modules	***2504-Voice Module-Duration***
To assign a voice message application to a voice module	***2505-Voice Message Application Number-Voice Module***

To de-assign the voice module from a voice message application	***2505-Voice Message Application Number-00***

Basic Network

To select the Connection Type	***2116-Network Connection Type***
To assign IP Address to WAN Port	***2110-IP Address***
To assign Network Mask to the WAN Port	***2111-Network Mask***
To assign Gateway Address	***2112-Gateway Address***
To program Web Server Port	***2121-HTTP Port***
To view Address of Ethernet port	***2150***
To view Subnet of Ethernet port	***2151***
To view Gateway of Ethernet port	***2152***
To view Ethernet Link Status	***2162***
To exit the SE Programming Mode	***00***

System Commands for PENX

This section lists the Basic SE Commands required for configuring the important network parameters using a telephone. However, it is highly recommended to configure the system using Jeeves.

To assign IP Address to WAN Port	***2110-IP Address*** where, IP Address is 15 digits maximum. 000 to 255 for the first 3 Octets and 001 to 254 for the 4th Octet. Use zeros as fillers and dial the digits in a continuous sequence. Do not dial '.' in the IP Address. **For example**: To assign the IP Address to 192.168.50.10 dial 2110-192168050010.
To assign Subnet Mask to WAN Port	***2111-Subnet Mask*** **For example:** To assign the Subnet Mask to 255.255.255.0 dial 2111-255255255000.
To assign Gateway IP Address	***2112-Gateway IP Address*** where, Gateway IP Address may be a maximum of 15 digits max. Follow the same instructions as assigning IP Address to the Ethernet Port.
To select Connection Type	***2116-Connection Type*** where, Connection Type is 1 for Static 2 for DHCP 3 for PPPoE Default: Static
To configure Web Server Port	***2121-Port number*** Port range is: 80, 1025 to 65535.
To view IP Address of WAN Port	***2150***
To view Subnet Mask of WAN Port	***2151***
To view Gateway Address of WAN Port	***2152***
To view WAN link Status	***2162***
To set RTC - Date Format	***1000-Date Format*** where, 1 for DD-MM-YYYY 2 for MM-DD-YYYY Default: 1

To set RTC - Date-Month-Year Format	***1001-DD-MM-YYYY or 1001-MM-DD-YYYY (as per the configured Date Format)*** where, DD is 01 to 31 (leading zero mandatory for single digit date). MM is 01 to 12 (leading zero mandatory for single digit month). YYYY is year in four digits from 0000 to 9999. Default: Current Date is set.
To set the Current Time	***1003-HH-MM-SS*** where, Hours is 00 to 23 (24-hour format in two digits) Minutes is 00 to 59 (in two digits) Seconds is 00 to 59 (in two digits) Default: Current Time is set.
To select a Time Zone	***1002-Time Zone Index Number*** where, Time Zone Index Number is from 001 to 124, refer the table below. Default: Index 050 - India.

Time Zones

Time Zone	Index
(GMT-12:00) US Minor Outlying Islands (Baker Island, Howland Island)	135
(GMT-11:00) American Samoa	126
(GMT-10:00) United States (Hawaii)	116
(GMT-09:30) French Polynesia	130
(GMT-09:00) United States (Juneau)	115
(GMT-08:00) United States (Las Vegas, Los Angeles, Phoenix, San Francisco, Seattle)	114
(GMT-08:00) Mexico (Tijuana)	072
(GMT-08:00) Canada (Vancouver)	031
(GMT-07:00) United States (Albuquerque, Boise, Cheyenne, Denver, Salt Lake City)	113
(GMT-07:00) Mexico (Chihuahua)	071
(GMT-07:00) Canada (Calgary)	030
(GMT-06:00) United States (Chicago, Dallas, Des Moines, Memphis, Minneapolis, New Orleans, Oklahoma, Omaha, St. Louis)	112
(GMT-06:00) Mexico (Mexico City)	070
(GMT-06:00) Costa Rica	035
(GMT-06:00) Canada (Winnipeg)	029
(GMT-05:00) United States (Atlanta, Augusta, Boston, Charlotte, Columbus, Detroit, Indiapolis, Miami, NY, Philadelphia, Washington)	111

Time Zone	Index
(GMT-05:00) Peru	085
(GMT-05:00) Cuba	037
(GMT-05:00) Colombia	034
(GMT-05:00) Canada (Montreal, Ottawa, Toronto)	028
(GMT-05:00) Brazil (Acre)	022
(GMT-05:00) Bahamas	009
(GMT-04:30) Venezuela	118
(GMT-04:00) Paraguay	084
(GMT-04:00) Guyana	047
(GMT-04:00) Chile	032
(GMT-04:00) Canada (Halifax)	027
(GMT-04:00) Brazil (Manaus)	021
(GMT-04:00) Bolivia	015
(GMT-04:00) Antigua & Barbuda	003
(GMT-03:30) Canada (St. John's)	026
(GMT-03:00) Brazil (Brasilia, Rio de Janeiro, Sao Paulo)	020
(GMT-03:00) Argentina	004
(GMT-02:00) Brazil (Fernando De Noronha)	019
(GMT-01:00) Cape Verde (Cabo Verde)	129
(GMT) Ireland	054
(GMT) Portugal	088
(GMT) United Kingdom	110
(GMT+01:00) Algeria	002
(GMT+01:00) Austria	008
(GMT+01:00) Belgium	013
(GMT+01:00) Bosnia & Herzegovina	016
(GMT+01:00) Cameroon	025
(GMT+01:00) Cote d'Ivoire	125
(GMT+01:00) Croatia	036
(GMT+01:00) Czech Republic	039
(GMT+01:00) Denmark	040
(GMT+01:00) France	044
(GMT+01:00) Germany	045
(GMT+01:00) Italy	056

Time Zone	Index
(GMT+01:00) Namibia	076
(GMT+01:00) Netherlands	078
(GMT+01:00) Nigeria	080
(GMT+01:00) Norway	081
(GMT+01:00) Poland	087
(GMT+01:00) Slovakia	095
(GMT+01:00) Spain	097
(GMT+01:00) Sweden	100
(GMT+01:00) Switzerland	101
(GMT+02:00) Belarus	012
(GMT+02:00) Botswana	017
(GMT+02:00) Bulgaria	023
(GMT+02:00) Cyprus	038
(GMT+02:00) Egypt	041
(GMT+02:00) Finland	043
(GMT+02:00) Greece	046
(GMT+02:00) Hungary	049
(GMT+02:00) Israel	055
(GMT+02:00) Jordan	058
(GMT+02:00) Lebanon	065
(GMT+02:00) Libya	066
(GMT+02:00) Mozambique	074
(GMT+02:00) Romania	090
(GMT+02:00) South Africa	096
(GMT+02:00) Syria	102
(GMT+02:00) Turkey	106
(GMT+02:00) Ukraine	108
(GMT+02:00) Yugoslavia	121
(GMT+02:00) Zambia	122
(GMT+02:00) Zimbabwe	123
(GMT+03:00) Yemen	120
(GMT+03:00) Bahrain	010
(GMT+03:00) Iraq	053
(GMT+03:00) Kenya	060

Time Zone	Index
(GMT+03:00) Kuwait	063
(GMT+03:00) Qatar	089
(GMT+03:00) Russia (Moscow, St. Petersburg)	091
(GMT+03:00) Saudi Arabia	124
(GMT+03:00) Sudan	099
(GMT+03:00) Uganda	107
(GMT+03:30) Iran	052
(GMT+04:00) Mauritius	069
(GMT+04:00) Oman	082
(GMT+04:00) United Arab Emirates	109
(GMT+04:30) Afghanistan	001
(GMT+05:00) Maldives	068
(GMT+05:00) Pakistan	083
(GMT+05:00) Tajikistan	104
(GMT+05:00) Uzbekistan	117
(GMT+05:30) Sri Lanka	098
(GMT+05:30) India	050
(GMT+05:45) Nepal	077
(GMT+06:00) Bangladesh	011
(GMT+06:00) Bhutan	014
(GMT+06:00) Kazakhstan	059
(GMT+06:00) Kyrgyzstan	064
(GMT+06:00) Russia (Novosibirsk)	092
(GMT+06:30) Myanmar	075
(GMT+07:00) Cambodia	024
(GMT+07:00) Indonesia	051
(GMT+07:00) Thailand	105
(GMT+07:00) Vietnam	119
(GMT+08:00) Australia (Perth)	005
(GMT+08:00) Brunei	018
(GMT+08:00) China	033
(GMT+08:00) Hong kong	048
(GMT+08:00) Malaysia	067
(GMT+08:00) Mongolia	073

Time Zone	Index
(GMT+08:00) Philippines	086
(GMT+08:00) Singapore	094
(GMT+08:00) Taiwan	103
(GMT+08:45) Australia (Eucla)	127
(GMT+09:00) Japan	057
(GMT+09:00) Korea-North	061
(GMT+09:00) Korea-South	062
(GMT+09:30) Australia (Adelaide)	006
(GMT+10:00) Australia (Brisbane, Canberra, Melbourne, Sydney)	007
(GMT+10:00) Russia (Vladivostok)	093
(GMT+10:30) Australia (Lord Howe Island)	128
(GMT+11:00) Solomon Island)	134
(GMT+12:00) Fiji	042
(GMT+12:00) New Zealand (Auckland, Wellington)	079
(GMT+12:45) New Zealand (Chatham Islands)	132
(GMT+13:00) Samoa	133
(GMT+14:00) Kiribati	131

SARVAM UCS Features tested on IP Phones of different Brands

Features and Supportive Phones		
S.NO	**Feature**	**Phones Supported**
1	Intercom	1. GrandStream GXP2020
		2. GrandStream GXV3140
		3. GrandStream GXP2120
		4. Yealink T28P
		5. Yealink T26P
		6. Yealink T22P
		7. Yealink T20P
		8. Snom 300
		9. Cisco SPA504G
2	Distinctive Ring	1. Snom 300
		2. Yealink T28P
		3. Yealink T26P
		4. Yealink T22P
		5. Yealink T20P
		6. Polycom VVX1500D
3	Last Caller Recall	1. Polycom VVX1500D
		2. Cisco SPA504G
4	Paging	1. Grandstream GXP2020
		2. Grandstream GXV3140
		3. GrandStream GXP2120
		4. Yealink T28P
		5. Yealink T26P
		6. Yealink T22P
		7. Yealink T20P
		8. Snom 300
		9. Cisco SPA504G
5	Conversation Recording	1. Yealink T28P/T26P/T22P/T20P
		2. Snom 300
6	Call Park and Retrieve	1. Cisco SPA504G
		2. Polycom WX1500D
		3. Snom 300
7	Group Call Pickup and Selective Call Pickup	1. Cisco SPA504G
		2. Polycom WX1500D
8	SCA and Line Seize	1. Cisco 504G
		2. Polycom WX1500D
		3. Snom 300
		4. Yealink T28P/T26P/TT22P/T20P
		5. Grandstream GXP2120
9	Resume Call Transfer	1. Yealink T28P and T26P
10	Semi-Attend Transfer	1. Grandstream GXP2020
		2. Grandstream GXP2120
		3. Yealink T28P/T26P/TT22P/T20P
		4. Polycom WX1500D
11	Busy Lamp Field	1. Snom 300
		2. Cisco SPA504G
		3. Yealink T28P/T26P/TT22P/T20P
		4. Grandstream GXP2020
		5. Grandstream GXP2120
		6. Polycom WX1500D
12	Support of Call Hold	1. Snom 300 (only when a Key is configured as Extension)

Features on ports of SARVAM UCS and Supportive Phones		
S.NO	**Feature**	**Supportable Phone**
1	SARAVAM UCS allows these extensions to be monitored using Busy Lamp Field for Station	1. SLT 2. SIP Extension
2	SARAVAM UCS allows these trunks to be monitored using Busy Lamp Field for Trunk	1. CO 2. Mobile 3. SIP
3	Support of Call Hold Indication in BLF	1. SLT (Consultation Hold) 2. CO (Consultation Hold, Exclusive Hold, Global Hold) 3. Mobile (Consultation Hold, Exclusive Hold, Global Hold) 4. SIP (Consultation Hold, Exclusive Hold, Global Hold)

SARVAM UCS Features supported with RTP/Direct RTP

SARVAM UCS features that use Vocoder Channels

If RTP mode is set as RTP Relay or Direct RTP, system will use Vocoder channels for the following features:

- Call Taping
- DISA Call
- Voice Mail
- DID
- Trunk Auto Answer

SARVAM UCS features that use Vocoder channels when accessed from Extended IP Phones

If Extended IP Phones are connected as SIP Extensions and the RTP mode is set as RTP Relay or Direct RTP, system will use Vocoder channels for the following features:

- Conference
- Raid
- Interrupt Request
- Barge-In
- Conversation Recording
- Paging

SARVAM UCS features supported on Standard SIP Clients

If Standard SIP Phones are connected as SIP Extensions and the RTP mode is set as RTP Relay or Direct RTP, users can make internal calls, external calls as well as access the following features and facilities of SARVAM UCS:

- Abbreviated Dialing (both Personal and Global)
- Call Taping
- CUG Calling
- Operator
- Voice Mail
- Emergency Number

Features that need to be handled locally by Standard SIP Clients:

- Hold
- Transfer
- Conference
- Call Toggle

Troubleshooting

- All servicing to be undertaken ONLY by qualified service personnel. There are no user serviceable parts inside the unit.
- Always switch off "MAINS" and "BATTERY" marked switches of the system before opening the system and remove power cable from Mains plug, to avoid risk of electric shock.

ETERNITY GENX is not turning ON

AC Mains:

- Check the Mains Voltage.
- Check the Mains Switch.
- Check the Mains Fuse (6Amp Slow Blow, glass fuse provided in AC Mains socket of ETERNITY GENX Card).
- Check the MOV (275/20).
- Check for loose connection of PT3 connector (connecting AC Mains socket to ETERNITY GENX Card).
- Please contact authorized Matrix dealer.

One Station is not working

- Change the telephone instrument and check.
- Check wiring of that Station.
- Please contact authorized Matrix dealer.

When I call 2001 Station the call goes to 2002

- Dial 130 from 2001 (your call might have been forwarded).

Station not ringing

- Check ringer volume of the telephone instrument.
- Try replacing the telephone instrument.
- Dial 130 to disable call forward/call follow me feature.

Station found busy

- Check whether hand-set is properly kept on the cradle.
- Try replacing telephone instrument.
- Check wiring.

Station cannot dial

- Try replacing telephone instrument.
- Ensure dialing is not disabled through programming.

Incoming Call does not land correctly

- Ensure proper programming of trunk landing group for the trunk.
- Check for Call Privacy from incoming calls.
- Check the Time programmed in the System. This is a time sensitive feature.

CLI Number does not come on the Station

- Please check up with your Telephone Company (Service Provider) for CLI facility.
- Please check whether the Station where you are checking CLI function is programmed as CLI Phone.
- Please contact authorized Matrix dealer.

GSM Call not OK

- Please check antenna connection with antenna port of the GSM Card.
- Please check GSM network registration, with SIM card interface.
- Please check Mobile Trunks programming in the Routing Group.
- Select proper Routing Type selection as per Time Zone.

VoIP Call is not OK

- Please check registration of SIP with Server of Service Provider.
- Please check number dialing method is followed correctly.
- Please check SIP trunks are programmed correctly in the Routing Group.
- Please check proper Routing Type selection as per Time Zone.

Acronyms

ACB	Auto Call Back
AIS	Alarm Indication Signal
ANSI	American National Standard Institute
ANT	Automatic Number Translation
AO/DI	Always On/Dynamic ISDN
APM	Analog Personality Module
ATA	Analogue Telephone Adapter
AVD	Alternate Voice data signaling (Also called clear channel, out-of-band signaling)
BCC	Byte Check Code
BH	Break Hour
BI	Barge-In
BOS	Bit Oriented Signaling
BPS	Bits per second
BRI	Basic Rate Interface (2 B-Channels@64Kbps + D-Channel@64Kbps)
BSCs	Base Station Controllers
BSS	Base Station Subsystem
BTSs	Base Transceivers Stations
CAS	Channel Associate Signaling, Call Accounting Software
CCC	Call Cost Calculation
CCS	Common Channel Signaling
CCS	Compander Control Signal
CCWT	External Call Waiting Tone
CD	Carrier Detect
CDC	Call Duration Control
CDC	Call Duration Control
CDMA	Code Division Multiple Access
CDR	Call Detail Record
CESID	Customer Emergency Services Identification Dialing
CI	Call Incoming
CLIP	Calling Line Identification and Presentation
CLIR	Calling Line Identification Restriction
CO	Call Outgoing
CO	Central Office

COLP	Connected Line Identification Presentation
COLR	Connected Line Identification Restriction
COS	Class of Service
CPC	Calling Party Control
CPD	Call Progress Detection
CPTG	Call Progress Tone (Generation)
CPU	Call Pick-Up
CPW	Custom Pulse Width
CRC	Cyclic Redundancy Check
CTS	Clear to Send
CPE	Customer Premise Equipment
CUG	Closed User Group
CVT	Constant Voltage Transformer
DCD	Data Carrier Detected
DCE	Data Communication Equipment (Data circuit terminating equipment)
DDI	Direct Dialing-In
DHCP	Dynamic Host Configuration Protocol
DISA	Direct Inward System Access
DKP	Digital Key Phone
DLC	Digital Station Card
DND	Do Not Disturb
DNS	Domain Name System
T1E1	Digital Signal Level 1
DSR	Data Signal Ready
DSS64	Direct Station Selection Console
DST	Daylight Saving Time
DTE	Data Terminal Equipment
DTMF	Dual Tone Multi Frequency
DTR	Data Terminal Ready
E1	E-Carrier1 (30B+D)
EID	Exchange ID
ENQ	Enquiry
ETX	End of Text
E & M	Ear and Mouth Interface
FAS	Frame alignment signal

FCBC	Float cum Boost Charger
FIFO	First In First Out
FM	Frequency Modulation
FSK	Frequency Shift Keying
FTP	File Transfer Protocol
GDT	Gas Discharge Tube
GND	Ground
GSM	Global System for Mobile
GPRS	General Purpose Radio Service
HDSL	High Speed Digital Subscriber Line
HLR	Home Location Register
HOG	Hot Outward Dialing
IC	Incoming call
ICWT	Internal Call Waiting Tone
IMEI	International Mobile Equipment Identity
IP	Internet Protocol
IR	Interrupt
ISDN	Integrated Service Digital Network
ISP	Internet Service Provider
ITU	International Telecommunication Union
IVDT	Integral Voice/Data Terminal
LA	Left Align
LAN	Local Area Network
LAN NIC	Network Interface Card
LCD	Liquid Crystal Display
LCM	Line Coding Mechanism
LCR	Least Cost Routing
LCS	Live Call Screening
LE	Local Exchange
LED	Light Emitting Diode
LIFO	Last in First Out
LNR	Last Number Redial
LOF	Loss of Frame
LOS	Loss of signal
LSR	Least Cost Routing

MAC	Media Access Control Address
MCC	Mobile Country Code
MCI	Malicious Call Identification
MDF	Main Distribution Frame
MFA	Multi-Frame Alignment
MNC	Mobile Network Code
MOH	Music on Hold
MOV	Metal Oxide Varistor
MS	Mobile Station
MSC	Mobile Switching Center
MSN	Multiple Subscribers Numbers
MTBF	Mean Time between Failures
NH	Non Working Hour
NPI	Numbering Plan Identification, Number Planning Index
NSS	Network and Switching Subsystem
NT1/2	Network Termination (Type 1/2)
OG	Outgoing
OMC	Operation Maintenance Centers
OOF	Out of Frame
OSI	Open Systems Interconnect
OSS	Operation Support Subsystem
PAS	Public Address System
PBX	Private Branch Exchange
PC	Personal Computer
PCM	Primary Interface Compounding
PCOL	Personal Central Office Line
PFT	Power Fail Transfer Module
PIN	Personal Identification Number
PISN	Private Integrated Service Network
PLCC	Power Line Carrier Communication
PMS	Property Management Software
POTS	Plain Old Telephone Systems
PPDC	Pre PSTN Digit Count
PPM	Primary Protection Module
PPS	Pulse per second

PRI	Primary Rate Interface (23B+D/30B+2D)
PS	Power Supply
PSK	Phase Shift Keying
PSTN	Public Switched Telephone Network
PUK	Personal Unlock Key
QAM	Quadrature Amplitude Modulation
QSIG	Q-Signaling
RA	Right Align
RBT	Ring Back Tone
RF	Radio Frequency
RI	Ring Indicator
RLSD	Receive Line Signal Detector (also called Carrier Detect)
RTC	Real Time Clock
RTS	Request to Send
RXD	Receive data
SA	System Administrator
SAL	System Activity Log
SE	System Engineer
SEFS	Severely Eroded Framed Seconds
SES	Severely eroded seconds
SFL	System Fault Log
SID	Exchange Identity
SIP	Session Initiated Protocol
SIM	Subscriber Identity Module
SLIC	Subscriber Loop Integrated System
SLT	2 wire Analog Station, Single Line Telephone
SMDR	Station Message Detail Recording
SMPS	Switch Mode Power Supply
SP	Service Provider
STX	Start of Text
SPID	Service Provider Identifier
T1	T-Carrier (23B+D)
TA	Terminal Adaptor
TAC	Trunk Access Code
TAG	Trunk Access Group

TCM	Trellis Coding Modulation
TCP/IP	Transmission Control Protocol/Internet Protocol
TE1/2	Terminal Equipment (Type 1/2)
TEI	Terminal Endpoint Identifier
TLG	Trunk Landing Group
TON	Type of Numbering Plan (Caller/Called)
TON	Type of Number
TWT	Two Wire Trunks, 2 wire Analog Trunk Card
UART	Universal Asynchronous receiver/transmitter
UPS	Un-interrupted Power Supply
VIC	Voice Interface Card
VLR	Visitor Location Register
VMS	Voice Mail System
VMA	Voice Message Application
WAN	Wide Area Network
WH	Working Hour

Warranty Statement

Matrix warrants that its products will be free from defects in material and workmanship, under normal use and service for a period of twelve (12) months from the date of installation.

Matrix warranties the replacement or repair of any product or component(s) found to be defective during the applicable period and return the same, or grant a reimbursement credit with respect to the product or component. Parts repaired or replaced will be under warranty throughout the remainder of the original warranty period only. In case of software program design defect(s) that prevents the program from performing the specified functionality, affecting service and beneficial use of the product, Matrix reserves the right to incorporate solutions in its new release of the software and make it available to the customer within a reasonable period of time. The above said with regard to the software design defect, constitutes the sole obligation of Matrix and its authorized installer with respect to the product.

Matrix does not, however, affirm or stand for that the functions or features contained in the system will satisfy its end-user's particular purpose and /or requirements or that the operation of the program will be uninterrupted or error free.

This warranty is voidable by Matrix:

- If the product is used other than under normal use and is not properly serviced and maintained by qualified technicians.
- If the product is not maintained under proper environmental conditions.
- If the product is subjected to abuse, damage, misuse, neglect, fire, power flow, acts of God, accident.
- If the product is installed or used in combination or in assembly with the products that are not supplied or authorized by Matrix or are of inferior quality or design than Matrix supplied products, which may cause reduction or degradation in functionality.
- If the product is operated outside the product's specifications or used without designated protections.
- If the completely filled warranty cards have not been received by Matrix within 15 days of the installation.

In no event will Matrix be liable for any damages, including lost profits, lost business, lost savings, downtime or delay, labor, repair or material cost, injury to person, property or other incidental or consequential damages arising out of use of or inability to use such product, even if Matrix has been advised of the possibility of such damages or losses or for any claim by any other party.

Except for the obligations specifically set forth in this Warranty Policy Statement, in no event shall Matrix be liable for any direct, indirect, special, incidental or consequential damages, whether based on contract or any other legal theory, and where advised of the possibility of such damages.

Neither Matrix nor any of its channel partners makes any other warranty of any kind, whether expressed or implied, with respect to Matrix products. Matrix and its distributors, dealers or sub-dealers specifically disclaim the implied warranties of merchantability and fitness for a particular purpose.

This warranty is not transferable and applies only to the original user of the Product. All legal course of action subjected to Vadodara (Gujarat, India) jurisdiction only.

Disposal of Products/Components after End-Of-Life

Main components of Matrix products are given below:

- **Soldered Boards:** At the end-of-life of the product, the soldered boards must be disposed through e-waste recyclers. If there is any legal obligation for disposal, you must check with the local authorities to locate approved e-waste recyclers in your area. It is recommended not to dispose-off soldered boards along with other waste or municipal solid waste.

- **Batteries:** At the end-of-life of the product, batteries must be disposed through battery recyclers. If there is any legal obligation for disposal, you may check with local authorities to locate approved batteries recyclers in your area. It is recommended not to dispose off batteries along with other waste or municipal solid waste.

- **Metal Components:** At the end-of-life of the product, Metal Components like Aluminum or MS enclosures and copper cables may be retained for some other suitable use or it may be given away as scrap to metal industries.

- **Plastic Components:** At the end-of-life of the product, plastic components must be disposed through plastic recyclers. If there is any legal obligation for disposal, you may check with local authorities to locate approved plastic recyclers in your area.

After end-of-life of the Matrix products, if you are unable to dispose-off the products or unable to locate e-waste recyclers, you may return the products to Matrix Return Material Authorization (RMA) department.

Make sure these are returned with:
- proper documentation and RMA number
- proper packing
- pre-payment of the freight and logistic costs.

Such products will be disposed-off by Matrix.

"SAVE ENVIRONMENT SAVE EARTH"

E-Waste Management and Handling Rules

E-waste is a popular, informal name for electronic products nearing the end of their useful life. E-wastes are considered dangerous, as certain components of some electronic products contain materials that are hazardous, depending on their condition and density. The hazardous content of these materials pose a threat to human health and environment. Discarded electronics products such as circuit boards, batteries, wires and other electronic accessories if improperly disposed can leach lead and other substances into soil and groundwater. Many of electronic products can be reused, refurbished or recycled in an environmentally sound manner so that they are less harmful to the ecosystem.

Benefits of E-waste Recycling leach

Electronics Recycling Conserves Natural Resources

There are many materials that can be recovered from old electronic products. These materials can be used to make new products, thus reducing the need for the new raw materials. For instance, various metals can be recovered from circuit boards and other electronics can be recycled.

Electronics Recycling Supports the Community

Donating your old electronics plays an important role in the provision of refurbished products which can be of great help to certain industries, small organizations and non-profitable organizations. It also helps individuals gain access to technology that they could not have otherwise afforded.

Electronics Recycling Creates Employment Locally

Considering that around 90 percent of electronic equipment is recyclable, electronics recycling can play a significant role in creating employment. This is because new firms dealing with electronics recycling will form and existing firms will look to employ more people to recover recyclable materials. This can be triggered by the increase in the demand for electronics recycling.

Electronics Recycling Helps Protect Public Health and the Environment

Many electronics have toxic or hazardous materials such as mercury and lead, which can be harmful to the environment if disposed in trashcans. Reusing and recycling electronics safely helps in keeping the hazardous materials from harming humans or the environment. For example, certain electronic components and batteries are hazardous since they have lead in them. Printed circuit boards contain harmful materials such as cadmium, lead, mercury and chromium.

Instead of keeping old electronics or dumping them in landfills, recycling or reusing them is an appropriate option that should be supported by individuals and organizations. Considering the benefits of electronics recycling, it is very important that people in various parts around the world embrace this concept.

Creates Jobs

E-waste recycling creates new jobs for professional recyclers and creates a second market for the recycled materials.

Do's & Don'ts

Do's:

- Always look for information on the catalogue with your product for end-of-life equipment handling.
- Ensure that only Authorized Recyclers/Dismantler handle your electronic products.
- Always call at our toll-free No's to Dispose products that have reached end-of life.
- Always drop your used electronic products, batteries or any accessories, when they reach the end of their life at your nearest Authorized E-Waste Collection Points.
- Always disconnect the battery from product and ensure any glass surface is protected against breakage.

Don'ts:

- Do not dismantle your electronic Products on your own.
- Do not throw electronics in bins having "Do not Dispose" sign.
- Do not give e-waste to informal and unorganized sectors like Local Scrap Dealer/ Rag Pickers.
- Do not dispose your product in garbage bins along with municipal waste that ultimately reaches landfills.

E-Waste Management Plan

M/s. MATRIX COMSEC PVT LTD has partnered with **E-Waste Recyclers India (EWRI)** to comply with the new India E-Waste management and handling rules in providing drop-of centers and environmentally sound management of end of life electronics.

EWRI has obtained authorizations from the appropriate governmental agency for their processing facilities. EWRI will receive and recycle customer returned equipment, including all the e-waste. Customers can drop their e-waste in the drop-box provided at various collection centers of EWRI.

A list of collection centers along with the address is mentioned below.

The customers can also call on the following toll free number (1800-102-5679) from Monday to Friday between 10:00 AM to 5:30 PM to get details about the collection centers.

Collection Centers:

State/ City	Location	Logistic	Address	Toll-Free Number
Delhi	Rangpuri	Professional Logistics	Rangpuri, Milakpur Kohi Rangpuri, Rangpuri, New Delhi - 110037	1800-102-5679
Gurugram	Gurugram	Professional Logistics	295, LIG Colony, Sector 31, Gurugram, Haryana - 122022	1800-102-5679
Jharkhand	Dhanbad	Professional Logistics	Sardar Patel Nagar, Dhanbad, Jharkhand - 826004	1800-102-5679
Noida	Salarpur Khadar	Professional Logistics	2, Gejha Rd, Goyal Colony, Salarpur Khadar, Sector 102, Noida, Uttar Pradesh - 201304	1800-102-5679
Mumbai	Vashi	Professional Logistics	Plot-92,gala no 01,Sector 19C Vashi Navi, Mumbai - 400705	1800-102-5679

State/ City	Location	Logistic	Address	Toll-Free Number
Pune	Vallabh Nagar	Professional Logistics	No.3/20,Near Ashok Sah Bank, Vallabh Nagar, S.T.Stand Road, Pimpri, Pune - 302021	1800-102-5679
Odisha	Cuttack	Professional Logistics	Cuttack, Odisha	1800-102-5679
Hyderabad	Secunderabad	Professional Logistics	4,Block-3,4th Shatter at 179, MPR Estates Near Old Check Post Old Bowaenpally Secunderabad, Hyderabad - 500011	1800-102-5679
Bangalore	Yeshwanthpur	Professional Logistics	No.44 1st floor 2nd main D.D.U.T.T.L. Yeshwanthpur, Bangalore - 560022	1800-102-5679
Mangalore	Bhathery Road Boloor	Professional Logistics	Opp. Hindustan Lever Ltd, Sulthan, Bhathery Road Boloor, Mangalore (KA) - 575003	1800-102-5679
Jharkhand	Ranchi	Professional Logistics	Ranchi, Jharkhand	1800-102-5679
Chennai	Sennerkuppam	Professional Logistics	27,Sakthi Nagar Phase-II, Sennerkuppam, Near Bisleri Water Plant, Chennai - 600056	1800-102-5679
Rajasthan	Jaipur	Professional Logistics	A-81, 200 ft. By Pass, Heerapura, Jaipur, Rajasthan - 302021	1800-102-5679
Bokaro	Odisha	Professional Logistics	Cuttack, Odisha, India	1800-102-5679
Guwahati	Kundil	Professional Logistics	HN-34, Kundil Nagar Basistha Chariali, Near Parbhat Apartment, Guwahati - 781029	1800-102-5679
Lucknow	Kanpur Road	Professional Logistics	S-175,Ist Floor Transport Nagar Near RTO Kanpur Road Lucknow - 226004	1800-102-5679
Madhya Pradesh	Indore	Professional Logistics	284 AS-3 Scheme No.-78,Vijay Nagar, Indore, Madhya Pradesh	1800-102-5679
Ahmedabad	Pushp Penament	Professional Logistics	Shop No D-18, Pushp Penament, Behind Mony Hotel, Isanpur, Ahmedabad	1800-102-5679
Patna	Malyanil buddha	Professional Logistics	Dr. A.K Pandey (IPS) Malyanil buddha Colony, Patna (Bihar) - 800001	1800-102-5679
Andhra Pradesh	Vishakapatnam	Professional Logistics	Shop No.8, New Gajuwaka, Opp. High School Road, Vishakapatnam, Andhra Pradesh - 530026	1800-102-5679
Chandigarh	Pharbhat Road	Professional Logistics	Shop no:-19, Pharbhat Road, Opp:- Tennis Academy, Zirakpur, Chandigarh, Punjab	1800-102-5679

State/ City	Location	Logistic	Address	Toll-Free Number
Kolkata	B.T. ROAD DUNLOP	Professional Logistics	156A/73, Northern Park, B.T. Road Dunlop, Kolkata -700108	1800-102-5679
Odisha	Bhubaneswar	Professional Logistics	Acharya Vihar - jaydev Vihar Rd, Bhubaneswar, Odisha	1800-102-5679
West Bengal	Asansol	Professional Logistics	Shop No-4 Asansol Station Bus Stand Road, Munshi Bazar, Asansol, West Bengal - 713301	1800-102-5679

Regulatory Information

Customer Information-ACTA

Federal Communications Commission Statement[359]

Part 15:

Note: This equipment has been tested and found to comply with the limits for a Class A digital device, pursuant to Part15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference when the equipment is operated in a commercial environment. This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with the instruction manual, may cause harmful interference to radio communications. Operation of this equipment in a residential area is likely to cause harmful interference in this case the user will be required to correct the interference at his own expense.

Using Automatic Dialers:

This equipment supports automatic dialing of Emergency number for using the feature 'Security Dialing and Reporting'. Hence the user should take care about following point:
WHEN PROGRAMMING EMERGENCY NUMBERS AND (OR) MAKING TEST CALLS TO EMERGENCY NUMBERS:

- Remain on the line and briefly explain to the dispatcher the reason for the call.
- Perform such activities in the off-peak hours, such as early morning or late evenings.

Using Direct Inward Dialing ("DID"):

The equipment supports a feature 'direct inward dialing' (DID). Hence while operating it, following points must be considered:
ALLOWING THIS EQUIPMENT TO BE OPERATED IN SUCH A MANNER AS TO NOT PROVIDE FOR PROPER ANSWER SUPERVISION IS A VIOLATION OF PART 68 OF THE FCC'S RULES. PROPER ANSWER SUPERVISION IS WHEN:

a. This equipment returns answer supervision to the public switched telephone network (PSTN) when DID calls are:

   - Answered by the called station
   - Answered by the attendant
   - Routed to a recorded announcement that can be administered by the customer premises equipment (CPE) user.
   - Routed to a dial prompt

b. This equipment returns answer supervision on all DID calls forwarded to the PSTN. Permissible exceptions are:

   - A call is unanswered
   - A busy tone is received
   - A reorder tone is received

359. This is applicable for ETERNITY LENX, MENX, GENX and PENX.

Equal Access:

This equipment is capable of providing the end user equal access to the carrier of the user's choice. This equipment is capable of providing users access to interstate providers of operator services through the use of access codes. Modification of this equipment by call aggregators to block access dialing codes is a violation of the Telephone Operator Consumers Act of 1990.

Electrical Safety:

Telephone companies report that electrical surges, typically lightning transients, are very destructive to customer terminal equipment connected to AC power sources. This has been identified as a major nationwide problem.

However Matrix provides all protection against lightning transients in the equipment; the user must provide a suitable surge arrestor while integrating the equipment with other networking equipments.

Using FAX Capability:

The Telephone Consumer Protection Act of 1991 makes it unlawful for any person to use a computer or other electronic device, including FAX machines, to send any message unless such message clearly contains in a margin at the top or bottom of each transmitted page or on the first page of the transmission, the date and time it is sent and an identification of the business or other entity, or other individual sending the message and the telephone number of the sending machine or such business, other entity, or individual. (The telephone number provided may not be a 900 number or any other number for which charges exceed local or long-distance transmission charges.) In order to program this information into your FAX machine, Refer user's Guide for the software of the Fax operation, such as 'Win Fax'.

Installation and Repairs

Before installing this equipment, users should ensure that it is permissible to be connected to the facilities of the local telecommunications company. The equipment must also be installed using an acceptable method of connection. Repairs to certified equipment should be coordinated by a representative designated by the dealer/ supplier. Contact the support at Tech.Support@matrixcomsec.com

Any repairs or alterations made by the user to this equipment, or equipment malfunctions, may give the telecommunications company cause to request the user to disconnect the equipment.

Module Supported : Quectel UC20-G		
Standards and directive	**Applied / Complied Harmonized Standards**	
	RE Directive 2014/53 EU, Article 3(1)(a) ● Safety	EN 60950-1:2006+A11:2009+A1:2010+A12:2011+A2:2013
	RE Directive 2014/53 EU, Article 3(1)(a) ● Health	EN 62311:2008
	RE Directive 2014/35 EU, Article 3(1)(b) ● EMC	ETSI EN 301 489-1 V2.1.1, ETSI EN 301-408-52 V1.1.0 ETSI EN 301 489-19 V2.1.0
	RE Directive 2014/53 EU, Article 3(2) ● Radio	EN 301 908-1 V11.1.1, EN 301 908-2 V11.1.1 EN 301 511 V12.5.1*, Draft EN 303 412 V1.1.0*
	* Note: This is non-harmonized radio standard accepted by the RED (Radio Equipment Directive)	
FCC Identifier	XMR201510UC20	
Moducations Supported	GSM: GMSK, 8PSK, WCDMA: BPSK, QPSK, 16QAM GPS: BPSK GLONASS: OFDM	

Module Supported: Quectel M95	
Standards and directive	2014/53/EU Radio Equipment Directive ETSI EN 301 489-1 V1.9.2 (2011-09), ETSI EN 301 489-7 V1.3.1 (2005-11) ETSI EN 301 511 V9.0.2 (2003-03), 3GPP TS 51.010-1 V9.1.0 (2010-03) EN 62311:2008 EN 60950-1:2006+A11:2009+A1:2010+A12:2011+A2:2013
FCC Identifier	XMR201512M95
Modulations Supported	GMSK(EGSM), GMSK(DCS)

TEC Certificate:

ETERNITY LENX27SDC

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दूरसंचार विभाग,भारत सरकार

DEPARTMENT OF TELECOMMUNICATIONS, GOVERNMENT OF INDIA

दूरसंचार अभियांत्रिकी केंद्र

TELECOMMUNICATION ENGINEERING CENTRE

CERTIFICATE OF PROVISIONAL MANDATORY CONFORMANCE

No.: 672900053 **Dated:** 2020-01-01

This is to certify that the product described below conforms to the Essential Requirement issued by TEC under Mandatory Testing and Certification of Telecom Equipment as notified vide Indian Telegraph (Amendment) Rules, 2017.

APPLICANT:
MATRIX COMSEC PRIVATE LIMITED, 394, GIDC, MAKARPURA, BARODA 390010, GUJARAT, INDIA

ORIGINAL EQUIPMENT MANUFACTURER:
MATRIX COMSEC PRIVATE LIMITED, 394, GIDC, MAKARPURA, BARODA 390010, GUJARAT, INDIA

PRODUCT NAME:
Private Automatic Branch Exchange

PRODUCT VARIANT:
Private Automatic Branch Exchange

MODEL NO. :
ETERNITY LENX27SDC

ESSENTIAL REQUIREMENT NO. :
TEC67291906

SOFTWARE VERSION:
SARVAM UCS ENT V01R02

HARDWARE VERSION:
V1R3

VALID FROM:
2020-01-01

VALID TO:
2021-12-31

FAMILY NAME:
NA

QR CODE:

ASSOCIATED MODEL NO & INTERFACES TESTED:
as given overleaf (if any)

PRADEEP KUMAR MISRA Digitally signed by PRADEEP KUMAR MISRA Date: 2020.01.01 17:09:57 +05'30'

(P. K. MISRA)
DIRECTOR (TA)

1. Terms and Conditions mentioned at MTCTE portal shall be applicable
2. Bill of Material is annexed with this certificate.

List of Associated Models

List of Interfaces Tested

1. 2 Wire Trunk
2. ISDN PRI
3. Gigabit Ethernet Electrical

BILL OF MATERIAL							
Module Name	Module Code	Module Hardware Version	Module Software Version	Sub Unit Name	Sub Unit Model No.	Remarks	Module present in Main Model
Model No.							ETERNITY LENX27SDC
ETERNITY MENX-LENX CARD CPU	3000000229	V1R3	V1R2			CPU Card	Y
ETERNITY LE Card PS48VDC-500 Watts	3000000175	V1R4	-			DC Power Supply Card	Y
NX DBM VOCODER64	3000000301	V1R2	V1R6			VOCODER Daughter-board Module (Hardware) for ETERNITY MENX-LENX CPU card to Support Maximum 64 Simultaneous VOCODING Channels.	Y
NX DBM VMS64	3000000300	V1R3	V7R1			VOICE MAIL	Y

BILL OF MATERIAL							
Module Name	Module Code	Module Hardware Version	Module Software Version	Sub Unit Name	Sub Unit Model No.	Remarks	Module present in Main Model
						Daughter-board Module (Hardware) to Support Maximum 64 Simultaneous VOICE MAIL Sessions For ETERNITY MENX-LENX CPU CARDS.	
ETERNITY LE CARD SLT48	3000000176	V1R5	V1R11			Expansion Card for Analog Extensions (SLT)	Y
ETERNITY ME CARD SLT32	3000000215	V2R8	V4R12			Expansion Card for Analog Extensions (SLT)	Y
ETERNITY ME CARD SLT16	3000000214	V2R8	V4R12			Expansion Card for Analog Extensions (SLT)	Y

BILL OF MATERIAL							
Module Name	Module Code	Module Hardware Version	Module Software Version	Sub Unit Name	Sub Unit Model No.	Remarks	Module present in Main Model
ETERNITY ME CARD SLT8	3000000216	V2R8	V4R12			Expansion Card for Analog Extensions (SLT)	Y
ETERNITY ME CARD DKP32	3000000197	V2R8	V4R5			Expansion Card for Digital Key Phones (DKP)	Y
ETERNITY ME CARD DKP16	3000000196	V2R8	V4R5			Expansion Card for Digital Key Phones (DKP)	Y
ETERNITY ME CARD DKP8	3000000198	V2R8	V4R5			Expansion Card for Digital Key Phones (DKP)	Y
ETERNITY ME CARD DKP32LL	3000000499	V2R8	V4R5			Expansion Card for Digital Key Phones (DKP) with Long Loop	Y
ETERNITY ME CARD DKP16LL	3000000498	V2R8	V4R5			Expansion Card for Digital Key Phones (DKP) with Long Loop	Y
ETERNITY ME	3000000194	V2R7	V4R10			Expansion Card	Y

BILL OF MATERIAL							
Module Name	Module Code	Module Hardware Version	Module Software Version	Sub Unit Name	Sub Unit Model No.	Remarks	Module present in Main Model
CARD CO8+SLT24						for Analog Trunks (CO) and Analog Extensions (SLT)	
ETERNITY ME CARD CO16	3000000192	V2R8	V4R11			Expansion Card for Analog Trunks (CO)	Y
ETERNITY ME CARD CO8	3000000193	V2R8	V4R11			Expansion Card for Analog Trunks (CO)	Y
ETERNITY ME CARD T1E1PRI DUAL	3000000218	V1R1	V5R1			Expansion Card for T1/E1/PRI Ports	Y
ETERNITY ME CARD T1E1PRI SINGLE	3000000219	V1R1	V5R1			Expansion Card for T1/E1/PRI Ports	Y

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दूरसंचार विभाग,भारत सरकार

DEPARTMENT OF TELECOMMUNICATIONS, GOVERNMENT OF INDIA

दूरसंचार अभियांत्रिकी केंद्र

TELECOMMUNICATION ENGINEERING CENTRE

CERTIFICATE OF PROVISIONAL MANDATORY CONFORMANCE

No.: 672900042 **Dated:** 2019-11-21

This is to certify that the product described below conforms to the Essential Requirement issued by TEC under Mandatory Testing and Certification of Telecom Equipment as notified vide Indian Telegraph (Amendment) Rules, 2017.

APPLICANT:
MATRIX COMSEC PRIVATE LIMITED, 394, GIDC, MAKARPURA, BARODA 390010, GUJARAT, INDIA

ORIGINAL EQUIPMENT MANUFACTURER:
MATRIX COMSEC PRIVATE LIMITED, 394, GIDC, MAKARPURA, BARODA 390010, GUJARAT, INDIA

PRODUCT NAME:
Private Automatic Branch Exchange

PRODUCT VARIANT:
Private Automatic Branch Exchange

MODEL NO. :
ETERNITY MENX16SAC

ESSENTIAL REQUIREMENT NO. :
TEC67291906

SOFTWARE VERSION:
SARVAM UCS ENT V01R02.01.02

HARDWARE VERSION:
V1R3

VALID FROM:
2019-11-21

VALID TO:
2021-11-20

FAMILY NAME:
NA

QR CODE:

ASSOCIATED MODEL NO & INTERFACES TESTED:
as given overleaf (if any)

PRADEEP KUMAR MISRA
Digitally signed by PRADEEP KUMAR MISRA
Date: 2019.11.21 15:57:12 +05'30'

(P. K. MISRA)
DIRECTOR (TA)

1. Terms & Conditions mentioned at TEC website shall be applicable.
2. Bill of Material is annexed with this certificate.

List of Associated Models

List of Interfaces Tested

1. 2 Wire Trunk
2. ISDN PRI
3. Gigabit Ethernet Electrical

BILL OF MATERIAL							
Module Name	Module Code	Module Hardware Version	Module Software Version	Sub Unit Name	Sub Unit Model No.	Remarks	Module present in Main Model
Model No.							ETERNITY MENX16SAC
ETERNITY MENX-LENX CARD CPU	3000000229	V1R3	V1R2			CPU Card	Y
ETERNITY ME CARD PSUNI-500 Watts	3000000211	V2R6	-			AC Power Supply Card	Y
NX DBM VOCODER64	3000000301	V1R2	V1R6			VOCODER Daughter-board Module (Hardware) for ETERNITY MENX-LENX CPU card to Support Maximum 64 Simultaneous VOCODING Channels.	Y
NX DBM VMS64	3000000300	V1R3	V7R1			VOICE MAIL	Y

BILL OF MATERIAL							
Module Name	Module Code	Module Hardware Version	Module Software Version	Sub Unit Name	Sub Unit Model No.	Remarks	Module present in Main Model
						Daughter-board Module (Hardware) to Support Maximum 64 Simultaneous VOICE MAIL Sessions For ETERNITY MENX-LENX CPU CARDS.	
ETERNITY ME CARD SLT32	3000000215	V2R8	V4R12			Expansion Card for Analog Extensions (SLT)	Y
ETERNITY ME CARD SLT16	3000000214	V2R8	V4R12			Expansion Card for Analog Extensions (SLT)	Y
ETERNITY ME CARD SLT8	3000000216	V2R8	V4R12			Expansion Card for Analog Extensions (SLT)	Y

BILL OF MATERIAL							
Module Name	Module Code	Module Hardware Version	Module Software Version	Sub Unit Name	Sub Unit Model No.	Remarks	Module present in Main Model
ETERNITY ME CARD DKP32	3000000197	V2R8	V4R5			Expansion Card for Digital Key Phones (DKP)	Y
ETERNITY ME CARD DKP16	3000000196	V2R8	V4R5			Expansion Card for Digital Key Phones (DKP)	Y
ETERNITY ME CARD DKP8	3000000198	V2R8	V4R5			Expansion Card for Digital Key Phones (DKP)	Y
ETERNITY ME CARD DKP32LL	3000000499	V2R8	V4R5			Expansion Card for Digital Key Phones (DKP) with Long Loop	Y
ETERNITY ME CARD DKP16LL	3000000498	V2R8	V4R5			Expansion Card for Digital Key Phones (DKP) with Long Loop	Y
ETERNITY ME CARD CO8+SLT24	3000000194	V2R7	V4R10			Expansion Card for Analog Trunks (CO) and Analog Extensions	Y

BILL OF MATERIAL							
Module Name	Module Code	Module Hardware Version	Module Software Version	Sub Unit Name	Sub Unit Model No.	Remarks	Module present in Main Model
						(SLT)	
ETERNITY ME CARD CO16	3000000192	V2R8	V4R11			Expansion Card for Analog Trunks (CO)	Y
ETERNITY ME CARD CO8	3000000193	V2R8	V4R11			Expansion Card for Analog Trunks (CO)	Y
ETERNITY ME CARD T1E1PRI DUAL	3000000218	V1R1	V5R1			Expansion Card for T1/E1/PRI Ports	Y
ETERNITY ME CARD T1E1PRI SINGLE	3000000219	V1R1	V5R1			Expansion Card for T1/E1/PRI Ports	Y

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दूरसंचार विभाग,भारत सरकार

DEPARTMENT OF TELECOMMUNICATIONS, GOVERNMENT OF INDIA

दूरसंचार अभियांत्रिकी केंद्र

TELECOMMUNICATION ENGINEERING CENTRE

CERTIFICATE OF PROVISIONAL MANDATORY CONFORMANCE

No.: 672900073 **Dated:** 2020-05-21

This is to certify that the product described below conforms to the Essential Requirement issued by TEC under Mandatory Testing and Certification of Telecom Equipment as notified vide Indian Telegraph (Amendment) Rules, 2017.

APPLICANT:
MATRIX COMSEC PRIVATE LIMITED, 394, GIDC, MAKARPURA, BARODA 390010, GUJARAT, INDIA

ORIGINAL EQUIPMENT MANUFACTURER:
MATRIX COMSEC PRIVATE LIMITED, 394, GIDC, MAKARPURA, BARODA 390010, GUJARAT, INDIA

PRODUCT NAME:
Private Automatic Branch Exchange

PRODUCT VARIANT:
Private Automatic Branch Exchange

MODEL NO. :
ETERNITY MENX16SDC

ESSENTIAL REQUIREMENT NO. :
TEC67291906

SOFTWARE VERSION:
SARVAM UCS ENT V01R02.01.02

HARDWARE VERSION:
V1R3

VALID FROM:
2020-05-21

VALID TO:
2022-05-20

FAMILY NAME:
NA

QR CODE:

ASSOCIATED MODEL NO & INTERFACES TESTED:
as given overleaf (if any)

PRADEEP KUMAR MISRA
Digitally signed by PRADEEP KUMAR MISRA
Date: 2020.05.21 11:45:15 +05'30'

(P. K. MISRA)
DIRECTOR (TA)

1. Terms and Conditions mentioned at MTCTE portal shall be applicable
2. Bill of Material is annexed with this certificate.

List of Associated Models

List of Interfaces Tested

1. 2 Wire Trunk
2. ISDN PRI
3. Gigabit Ethernet Electrical

BILL OF MATERIAL							
Module Name	Module Code	Module Hardware Version	Module Software Version	Sub Unit Name	Sub Unit Model No.	Remarks	Module present in Main Model
Model No.							ETERNITY MENX16SDC
ETERNITY MENX-LENX CARD CPU	3000000229	V1R3	V1R2			CPU Card	Y
ETERNITY ME CARD PS48VDC-500 Watts	3000000209	V3R6	-			DC Power Supply Card	Y
NX DBM VOCODER64	3000000301	V1R2	V1R6			VOCODER Daughter-board Module (Hardware) for ETERNITY MENX-LENX CPU card to Support Maximum 64 Simultaneous VOCODING Channels.	Y

BILL OF MATERIAL							
Module Name	Module Code	Module Hardware Version	Module Software Version	Sub Unit Name	Sub Unit Model No.	Remarks	Module present in Main Model
NX DBM VMS64	3000000300	V1R3	V7R1			VOICE MAIL Daughter-board Module (Hardware) to Support Maximum 64 Simultaneous VOICE MAIL Sessions For ETERNITY MENX-LENX CPU CARDS.	Y
ETERNITY ME CARD SLT32	3000000215	V2R8	V4R12			Expansion Card for Analog Extensions (SLT)	Y
ETERNITY ME CARD SLT16	3000000214	V2R8	V4R12			Expansion Card for Analog Extensions (SLT)	Y
ETERNITY ME CARD SLT8	3000000216	V2R8	V4R12			Expansion Card for Analog Extensions	Y

BILL OF MATERIAL							
Module Name	Module Code	Module Hardware Version	Module Software Version	Sub Unit Name	Sub Unit Model No.	Remarks	Module present in Main Model
						(SLT)	
ETERNITY ME CARD DKP32	3000000197	V2R8	V4R5			Expansion Card for Digital Key Phones (DKP)	Y
ETERNITY ME CARD DKP16	3000000196	V2R8	V4R5			Expansion Card for Digital Key Phones (DKP)	Y
ETERNITY ME CARD DKP8	3000000198	V2R8	V4R5			Expansion Card for Digital Key Phones (DKP)	Y
ETERNITY ME CARD DKP32LL	3000000499	V2R8	V4R5			Expansion Card for Digital Key Phones (DKP) with Long Loop	Y
ETERNITY ME CARD DKP16LL	3000000498	V2R8	V4R5			Expansion Card for Digital Key Phones (DKP) with Long Loop	Y
ETERNITY ME CARD CO8+SLT24	3000000194	V2R7	V4R10			Expansion Card for Analog Trunks (CO) and Analog	Y

BILL OF MATERIAL							
Module Name	Module Code	Module Hardware Version	Module Software Version	Sub Unit Name	Sub Unit Model No.	Remarks	Module present in Main Model
						Extensions (SLT)	
ETERNITY ME CARD CO16	3000000192	V2R8	V4R11			Expansion Card for Analog Trunks (CO)	Y
ETERNITY ME CARD CO8	3000000193	V2R8	V4R11			Expansion Card for Analog Trunks (CO)	Y
ETERNITY ME CARD T1E1PRI DUAL	3000000218	V1R1	V5R1			Expansion Card for T1/E1/PRI Ports	Y
ETERNITY ME CARD T1E1PRI SINGLE	3000000219	V1R1	V5R1			Expansion Card for T1/E1/PRI Ports	Y

सत्यमेव जयते

दूरसंचार विभाग,भारत सरकार

DEPARTMENT OF TELECOMMUNICATIONS, GOVERNMENT OF INDIA

दूरसंचार अभियांत्रिकी केंद्र

TELECOMMUNICATION ENGINEERING CENTRE

CERTIFICATE OF PROVISIONAL MANDATORY CONFORMANCE

No.: 672900003 **Dated:** 2019-09-13

This is to certify that the product described below conforms to the Essential Requirement issued by TEC under Mandatory Testing and Certification of Telecom Equipment as notified vide Indian Telegraph (Amendment) Rules, 2017.

APPLICANT:
MATRIX COMSEC PRIVATE LIMITED, 394, GIDC, MAKARPURA, BARODA 390010, GUJARAT, INDIA

ORIGINAL EQUIPMENT MANUFACTURER:
MATRIX COMSEC PRIVATE LIMITED, 394, GIDC, MAKARPURA, BARODA 390010, GUJARAT, INDIA

PRODUCT NAME:
Private Automatic Branch Exchange

PRODUCT VARIANT:
Private Automatic Branch Exchange

MODEL NO. :
ETERNITY GENX

ESSENTIAL REQUIREMENT NO. :
TEC67291906

SOFTWARE VERSION:
SARVAM UCS SME V01R02

HARDWARE VERSION:
V1R3

VALID FROM:
2019-10-01

VALID TO:
2021-09-30

FAMILY NAME:
NA

QR CODE:

ASSOCIATED MODEL NO & INTERFACES TESTED:
as given overleaf (if any)

PRADEEP KUMAR MISRA Digitally signed by PRADEEP KUMAR MISRA Date: 2019.09.13 14:31:39 +05'30'

(P. K. MISRA)
DIRECTOR (TA)

1. Terms & Conditions mentioned at TEC website shall be applicable.
2. Bill of Material is annexed with this certificate.

List of Associated Models

S.No.	Certificate No.	Associated Model
1	672900003.1	ETERNITY GEN(12SAC

List of Interfaces Tested

1. 2 Wire Trunk
2. ISDN PRI
3. Gigabit Ethernet Electrical

BILL OF MATERIAL								
Module Name	Module Code	Module Hardware Version	Module Software Version	Sub Unit Name	Sub Unit Model No.	Remarks	Module present in Main Model	Module present in Associated model-1
Model No.							ETERNITY GENX	ETERNITY GENX12SAC
ETERNITY GENX CARD CPU	3000000169	V1R3	V1R2			CPU Card	Y	Y
ETERNITY GENX CARD PSUNI	3000000137	V1R6	-			AC Power Supply Card	Y	Y
NX DBM VOCODER64	3000000301	V1R2	V1R6			VOCODER Daughter-board Module (Hardware) for ETERNITY GENX CPU card to Support Maximum 64 Simultaneous VOCODING Channels.	Y	Y

BILL OF MATERIAL								
Module Name	Module Code	Module Hardware Version	Module Software Version	Sub Unit Name	Sub Unit Model No.	Remarks	Module present in Main Model	Module present in Associated model-1
NX DBM VMS64	3000000300	V1R3	V7R1			VOICE MAIL Daughter-board Module (Hardware) to Support Maximum 64 Simultaneous VOICE MAIL Sessions For ETERNITY GENX CPU CARDS.	Y	Y
ETERNITY GE CARD SLT20	3000000140	V2R5	V3R0			Expansion Card for Analog Extensions (SLT)	Y	Y
ETERNITY GE CARD SLT16	3000000139	V2R5	V3R0			Expansion Card for Analog Extensions (SLT)	Y	Y

BILL OF MATERIAL								
Module Name	Module Code	Module Hardware Version	Module Software Version	Sub Unit Name	Sub Unit Model No.	Remarks	Module present in Main Model	Module present in Associated model-1
ETERNITY GE CARD SLT8	3000000141	V2R5	V3R0			Expansion Card for Analog Extensions (SLT)	Y	Y
ETERNITY GE CARD DKP16	3000000125	V1R1	V3R0			Expansion Card for Digital Key Phones (DKP)	Y	Y
ETERNITY GE CARD DKP8	3000000127	V1R1	V3R0			Expansion Card for Digital Key Phones (DKP)	Y	Y
ETERNITY GE CARD DKP16LL	3000000663	V1R1	V3R0			Expansion Card for Digital Key Phones (DKP) with long loop	Y	Y
ETERNITY GE CARD DKP4+SLT16	3000000126	V2R5	V3R0			Expansion Card for Digital Key Phones (DKP) and	Y	Y

BILL OF MATERIAL								
Module Name	Module Code	Module Hardware Version	Module Software Version	Sub Unit Name	Sub Unit Model No.	Remarks	Module present in Main Model	Module present in Associated model-1
						Analog Extensions (SLT)		
ETERNITY GE CARD CO4+DKP2+SLT12	3000000119	V2R5	V3R0			Expansion Card for Analog Trunks (CO), Digital Key Phones (DKP) and Analog Extensions (SLT)	Y	Y
ETERNITY GE CARD CO4+DKP2+SLT8	3000000120	V2R5	V3R0			Expansion Card for Analog Trunks (CO), Digital Key Phones (DKP) and Analog Extensions (SLT)	Y	Y
ETERNITY GE	3000000118	V2R5	V3R0			Expansion	Y	Y

BILL OF MATERIAL								
Module Name	Module Code	Module Hardware Version	Module Software Version	Sub Unit Name	Sub Unit Model No.	Remarks	Module present in Main Model	Module present in Associated model-1
CARD CO2+DKP2+SLT16						Card for Analog Trunks (CO), Digital Key Phones (DKP) and Analog Extensions (SLT)		
ETERNITY GE CARD CO4+SLT16	3000000121	V2R5	V3R0			Expansion Card for Analog Trunks (CO) and Analog Extensions (SLT)	Y	Y
ETERNITY GE CARD CO16	3000000117	V1R3	V3R0			Expansion Card for Analog Trunks (CO)	Y	Y
ETERNITY GE CARD CO8	3000000122	V1R3	V3R0			Expansion Card for Analog Trunks	Y	Y

BILL OF MATERIAL								
Module Name	Module Code	Module Hardware Version	Module Software Version	Sub Unit Name	Sub Unit Model No.	Remarks	Module present in Main Model	Module present in Associated model-1
						(CO)		
ETERNITY GE CARD T1E1PRI SINGLE	3000000142	V1R5	V1R0			Expansion Card for T1/E1/PRI Port	Y	Y

सत्यमेव जयते

दूरसंचार विभाग,भारत सरकार

DEPARTMENT OF TELECOMMUNICATIONS, GOVERNMENT OF INDIA

दूरसंचार अभियांत्रिकी केंद्र

TELECOMMUNICATION ENGINEERING CENTRE

CERTIFICATE OF PROVISIONAL MANDATORY CONFORMANCE

No.: 672900074 **Dated:** 2020-05-21

This is to certify that the product described below conforms to the Essential Requirement issued by TEC under Mandatory Testing and Certification of Telecom Equipment as notified vide Indian Telegraph (Amendment) Rules, 2017.

APPLICANT:
MATRIX COMSEC PRIVATE LIMITED, 394, GIDC, MAKARPURA, BARODA 390010, GUJARAT, INDIA

ORIGINAL EQUIPMENT MANUFACTURER:
MATRIX COMSEC PRIVATE LIMITED, 394, GIDC, MAKARPURA, BARODA 390010, GUJARAT, INDIA

PRODUCT NAME:
Private Automatic Branch Exchange

PRODUCT VARIANT:
Private Automatic Branch Exchange

MODEL NO. :
ETERNITY GENX12SDC

ESSENTIAL REQUIREMENT NO. :
TEC67291906

SOFTWARE VERSION:
SARVAM UCS SME V01R02

HARDWARE VERSION:
V1R3

VALID FROM:
2020-05-21

VALID TO:
2022-05-20

FAMILY NAME:
NA

QR CODE:

ASSOCIATED MODEL NO & INTERFACES TESTED:
as given overleaf (if any)

PRADEEP KUMAR MISRA Digitally signed by PRADEEP KUMAR MISRA Date: 2020.05.21 11:47:40 +05'30'

(P. K. MISRA)
DIRECTOR (TA)

1. Terms and Conditions mentioned at MTCTE portal shall be applicable
2. Bill of Material is annexed with this certificate.

List of Associated Models

List of Interfaces Tested

1. 2 Wire Trunk
2. ISDN PRI
3. Gigabit Ethernet Electrical

BILL OF MATERIAL							
Module Name	Module Code	Module Hardware Version	Module Software Version	Sub Unit Name	Sub Unit Model No.	Remarks	Module present in Main Model
Model No.							ETERNITY GENX12SDC
ETERNITY GENX CARD CPU	3000000169	V1R3	V1R2			CPU Card	Y
ETERNITY GE CARD PS48VDC	3000000134	V3R3	-			DC Power Supply Card	Y
NX DBM VOCODER64	3000000301	V1R2	V1R6			VOCODER Daughter-board Module (Hardware) for ETERNITY GENX CPU card to Support Maximum 64 Simultaneous VOCODING Channels.	Y
NX DBM VMS64	3000000300	V1R3	V7R1			VOICE MAIL Daughter-board	Y

BILL OF MATERIAL							
Module Name	Module Code	Module Hardware Version	Module Software Version	Sub Unit Name	Sub Unit Model No.	Remarks	Module present in Main Model
						Module (Hardware) to Support Maximum 64 Simultaneous VOICE MAIL Sessions For ETERNITY GENX CPU CARDS.	
ETERNITY GE CARD SLT20	3000000140	V2R5	V3R0			Expansion Card for Analog Extensions (SLT)	Y
ETERNITY GE CARD SLT16	3000000139	V2R5	V3R0			Expansion Card for Analog Extensions (SLT)	Y
ETERNITY GE CARD SLT8	3000000141	V2R5	V3R0			Expansion Card for Analog Extensions (SLT)	Y
ETERNITY GE	3000000125	V1R1	V3R0			Expansion Card	Y

BILL OF MATERIAL							
Module Name	Module Code	Module Hardware Version	Module Software Version	Sub Unit Name	Sub Unit Model No.	Remarks	Module present in Main Model
CARD DKP 16						for Digital Key Phones (DKP)	
ETERNITY GE CARD DKP8	3000000127	V1R1	V3R0			Expansion Card for Digital Key Phones (DKP)	Y
ETERNITY GE CARD DKP 16LL	3000000663	V1R1	V3R0			Expansion Card for Digital Key Phones (DKP) with long loop	Y
ETERNITY GE CARD DKP4+SLT16	3000000126	V2R5	V3R0			Expansion Card for Digital Key Phones (DKP) and Analog Extensions (SLT)	Y
ETERNITY GE CARD CO4+DKP2+SLT12	3000000119	V2R5	V3R0			Expansion Card for Analog Trunks (CO), Digital Key Phones (DKP) and Analog Extensions (SLT)	Y

BILL OF MATERIAL							
Module Name	Module Code	Module Hardware Version	Module Software Version	Sub Unit Name	Sub Unit Model No.	Remarks	Module present in Main Model
ETERNITY GE CARD CO4+DKP2+SLT8	3000000120	V2R5	V3R0			Expansion Card for Analog Trunks (CO), Digital Key Phones (DKP) and Analog Extensions (SLT)	Y
ETERNITY GE CARD CO2+DKP2+SLT16	3000000118	V2R5	V3R0			Expansion Card for Analog Trunks (CO), Digital Key Phones (DKP) and Analog Extensions (SLT)	Y
ETERNITY GE CARD CO4+SLT16	3000000121	V2R5	V3R0			Expansion Card for Analog Trunks (CO) and Analog Extensions (SLT)	Y
ETERNITY GE	3000000117	V1R3	V3R0			Expansion Card	Y

BILL OF MATERIAL							
Module Name	Module Code	Module Hardware Version	Module Software Version	Sub Unit Name	Sub Unit Model No.	Remarks	Module present in Main Model
CARD CO16						for Analog Trunks (CO)	
ETERNITY GE CARD CO8	3000000122	V1R3	V3R0			Expansion Card for Analog Trunks (CO)	Y
ETERNITY GE CARD T1E1PRI SINGLE	3000000142	V1R5	V1R0			Expansion Card for T1/E1/PRI Port	Y

Eternity PENX6SAC

दूरसंचार विभाग, भारत सरकार
DEPARTMENT OF TELECOMMUNICATION, GOVERNMENT OF INDIA
दूरसंचार अभियांत्रिकी केंद्र
TELECOMMUNICATION ENGINEERING CENTRE

CERTIFICATE OF MANDATORY CONFORMANCE
(Regular)

Certificate No.: 672900123 **Dated:** 2021-04-07

This is to certify that the product described below conforms to the Essential Requirement issued by TEC under Mandatory Testing and Certification of Telecom Equipment as notified vide Indian Telegraph (Amendment) Rules, 2017.

APPLICANT:
MATRIX COMSEC PRIVATE LIMITED, 394, CIDC, MAKARPURA, BARODA 390010, GUJARAT, INDIA

ORIGINAL EQUIPMENT MANUFACTURER:
MATRIX COMSEC PRIVATE LIMITED, 394, CIDC, MAKARPURA, BARODA 390010, GUJARAT, INDIA

PRODUCT NAME:
Private Automatic Branch Exchange

PRODUCT VARIANT:
Private Automatic Branch Exchange

PREVIOUS CERTIFICATE NO. :
NA

PREVIOUS CERTIFICATE ISSUED DATE:
NA

MODEL NO. / FAMILY NAME:
ETERNITY PENX6SAC

ESSENTIAL REQUIREMENT NO. :
TEC67292002

SOFTWARE VERSION:
V1R6.6.0

HARDWARE VERSION:
V1R2

VALID FROM:
2021-04-07

VALID UPTO:
2026-04-06

ASSOCIATED MODEL NO. & INTERFACES TESTED:
As given overleaf (if any)

QR CODE:

IMI / IMC CLASS:
A

VENKATA RAMA RAJU CHELLE
Digitally signed by VENKATA RAMA RAJU CHELLE
Date: 2021.04.07 11:52:16 +05'30'

DIRECTOR (TC-1)

1. Terms & Conditions mentioned at MTCTE portal (http://www.mtcte.tec.gov.in/) shall be applicable.
2. Bill of Material is annexed with this certificate

List of Associated Models

List of Interfaces Tested

1. 2 Wire Trunk
2. ISDN PRI
3. Gigabit Ethernet Electrical (for PABX)

BILL OF MATERIAL							
Module Name/Attribute Name	Module Code	Module Hardware Version	Module Software Version	Sub Unit Name	Sub Unit Model No.	Remarks	Module present in Main Model
Module No.							ETERNITY PENX6SAC
ETERNITY PENX6SAC	3000001321	V1R2	V1R6.6.0			CPU Card	Y
ETERNITY PE CARD PSUNI	3000000271	V1R2	-			AC Universal Power Supply	Y
ETERNITY PE CARD SLT8	3000000273	V1R6	V3R0			Expansion Card for Analog Extensions (SLT)	Y
ETERNITY PE CARD SLT4	3000000272	V1R6	V3R0			Expansion Card for Analog Extensions (SLT)	Y
ETERNITY PE CARD DKP8	3000000266	V1R6	V3R0			Expansion Card for Digital Key Phones (DKP)	Y
ETERNITY PE CARD DKP2+SLT6	3000000265	V1R6	V3R0			Expansion Card for Digital Key Phones (DKP) and Analog	Y

BILL OF MATERIAL							
Module Name/Attribute Name	Module Code	Module Hardware Version	Module Software Version	Sub Unit Name	Sub Unit Model No.	Remarks	Module present in Main Model
						Extensions (SLT)	
ETERNITY PE CARD CO2+DKP2+SLT4	3000000260	V1R6	V3R0			Expansion Card for Analog Trunks (CO), Digital Key Phones (DKP) and Analog Extensions (SLT)	Y
ETERNITY PE CARD CO2+SLT6	3000000261	V1R6	V3R0			Expansion Card for Analog Trunks (CO) and Analog Extensions (SLT)	Y
ETERNITY PE CARD CO4+SLT4	3000000263	V1R6	V3R0			Expansion Card for Analog Trunks (CO) and Analog Extensions (SLT)	Y
ETERNITY PE	3000000274	V1R5	V1R0			Expansion Card	Y

BILL OF MATERIAL							
Module Name/Attribute Name	Module Code	Module Hardware Version	Module Software Version	Sub Unit Name	Sub Unit Model No.	Remarks	Module present in Main Model
CARD T1E1PRI SINGLE						for T1/E1/PRI Port	
NX DBM VOCODER64	3000000301	V1R4	V1R6			Expansion Card for VoIP with Server and Client Personalities	Y
NX DBM VMS64	3000000300	V1R4	V7R2			Expansion Card for Voice Mail Applications	Y
ETERNITY PE CARD CO8	3000000264	V1R6	V3R0			Expansion Card for Analog Trunks (CO)	Y
ETERNITY PE CARD GSM4	3000000269	V2R2	V1R0			Expansion Card for GSM Applications	Y
ETERNITY PE CARD GSM4 3G	3000000270	V2R2	V1R0			Expansion Card for GSM Applications	Y
ETERNITY PE CARD GSM4 4G	3000001342	V2R2	V1R0			Expansion Card for GSM Applications	Y

CE Certificates:

ETERNITY LENX

MATRIX
TELECOM | SECURITY

Declaration of Conformity

Manufacturer's Name:	Matrix Comsec Pvt. Ltd.
Manufacturer's Address :	15 & 19-GIDC, Waghodia, Dist: Vadodara 391760 Gujarat, India
Declares that the product/s Product:	IP-PABX with MEDIA GATEWAY (Unified Communication –Server Based)
Model Type	ETERNITY LENX
Trade Name:	MATRIX
Product Options:	This declaration covers all options of the above products

Conforms to the following product specification;

EMI/EMC:

CISPR 22 : 2008
CISPR 24 : 2010
IEC 61000-4-2 : 2008
IEC 61000-4-3 : 2010
IEC 61000-4-4 : 2012
IEC 61000-4-5 : 2014
IEC 61000-4-6 : 2013

SAFETY:

IEC 60950-1: 2005 (Second Edition) + Am 1:2009 + Am 2:2013

Supplementary information:
The Product herewith complies with the following directives ;

EMC	2014/30/EC
Low Voltage Directive	2014/35/EC
RoHS Recast (RoHS 2)	2011/65/EU (as per standard EN 50581:2012)

Mr. Ganesh Jivani
Director
Date: 24.02.2017
Vadodara

Matrix Comsec Pvt. Ltd. MATRIX Vadodara

CE

MATRIX COMSEC PVT. LTD.
Head Office: 394-GIDC, Makarpura, Vadodara-390 010, India. Ph: +91 265 2630555, Email: Inquiry@MatrixComSec.com • www.MatrixComSec.com
Factory: 19-GIDC, Waghodia, Dist. Vadodara-391 760, India. Ph: +91 2668 263172/73
Registered Office: C-1/394, GIDC, Makarpura, Vadodara-390 010, India. • **CIN:** U72200GJ1998PTC034047

MATRIX
TELECOM | SECURITY

Declaration of Conformity

Manufacturer's Name	: MATRIX COMSEC PVT LTD
Manufacturer's Address	: 15 & 19 - GIDC, WAGHODIA, VADODARA - 391760 (GUJARAT, INDIA)

Declares that the product/s

Product	: IP-PABX
Model/Type	: ETERNITY GENX
Trade Name	: MATRIX

Confirms to the following product specification:

EMI/EMC Standard(s):

EN 55032	: 2015
EN 61000-3-2	: 2014
EN 61000-3-3	: 2013
EN 55024	: 2010 + A1: 2015
IEC 61000-4-2	: 2008
IEC 61000-4-3	: 2006 + A1: 2007 + A2: 2010
IEC 61000-4-4	: 2012
IEC 61000-4-5	: 2014 + A1: 2017
IEC 61000-4-6	: 2013
IEC 61000-4-8	: 2009
IEC 61000-4-11	: 2004 + A1: 2017

SAFETY Standard:

EN 60950-1: 2006 + Am11:2009 + Am 1: 2010 + Am12: 2011 + Am 2: 2013

Supplementary Information:

The product herewith complies with the following directives:

EMC Directive	2014/30/EU
Low Voltage Directive	2014/35/EU
RoHS Directive (RoHS2)	2011/65/EU (as per EN 50581:2012)

Matrix Comsec Pvt. Ltd. MATRIX Vadodara

Mr. Ganesh Jivani
Director
Date: 13/06/2018

CE

MATRIX COMSEC PVT. LTD.
Registered/Head Office: 394-GIDC, Makarpura, Vadodara-390 010, India. Ph: +91 265 2630555, Email: Inquiry@MatrixComSec.com • **www.MatrixComSec.com**
Manufacturing Unit: 15 & 19-GIDC, Waghodia, Dist. Vadodara-391 760, India. Ph: +91 2668 263172/73 • **CIN:** U72200GJ1998PTC034047

ETERNITY PENX

MATRIX
TELECOM | SECURITY

EU DECLARATION OF CONFORMITY

EU DECLARTION OF CONFIRMITY

Manufacture : : MATRIX COMSEC PVT LTD

Manufacture Address : 15 & 19- GIDC , Waghodia, Vadodara-391760 (Gujarat, India)

Trade Name : **MATRIX**

Declare that the DoC is issued under our sole responsibility and belongs to the following products;

Product : **ETERNITY PENX**

Model/ TYPE : ETERNITY PENX6SAC

Essential Requirements /Directives		Applied Specifications/ Standards
EMC	2014/30/EU	EN 55032: 2015+A11:2020; EN 55035:2017+A11:2020; EN 61000-3-2: 2019; EN 61000-3-3: 2013+A1:2019; EN 61000-4-2: 2009; EN 61000-4-3: 2006+A2:2010;EN 61000-4-4: 2012; EN 61000-4-5: 2014+A1:2017; EN 61000-4-6: 2014; EN 61000-4-8: 2010; EN 61000-4-11: 2020
LVD/SAFETY	2014/35/EU	IEC 62368-1: 2018
RoHS (RoHS2)	2011/65/EU	EN 50581: 2012

I hereby declare that the equipment named above has been designated to comply with the relevant section of the above reference standards and meet all essential requirements of the specified directives.

CE

Matrix Comsec Pvt. Ltd. MATRIX Vadodara

Mr Ganesh Jivani
Managing Director
Date: 04/11/2020

MATRIX COMSEC PVT. LTD.
Registered/Head Office: 394-GIDC, Makarpura, Vadodara-390 010, India. Ph: +91 265 2630555, Email: Inquiry@MatrixComSec.com • **www.MatrixComSec.com**
Manufacturing Unit: 15 & 19-GIDC, Waghodia, Dist. Vadodara-391 760, India. Ph: +91 2668 263172/73 • **CIN:** U72200GJ1998PTC034047

Regulatory Information for Terminals

FCC Class B Information [360]

This equipment has been tested and found to comply with the limits for a Class B digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation.

This equipment generates, uses and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communications.

However, there is no guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one of the following measures:

- Reorient or relocate the receiving antenna.
- Increase the separation between the equipment and receiver.
- Connect the equipment into an outlet on a circuit different from that to which the receiver is connected.
- Consult the dealer or an experienced radio/TV technician for help.

This device complies with Part 15 of the FCC Rules. Operation is subject to the following two conditions:

1. this device may not cause harmful interference, and
2. this device must accept any interference received, including interference that may cause undesired operation.

360.Common for EON48, EON310 and EON510.

EON48

MATRIX
TELECOM | SECURITY

Declaration of Conformity

Manufacturer's Name:	Matrix Comsec Pvt. Ltd.
Manufacturer's Address :	15 &19-GIDC, Waghodia, Dist: Vadodara 391760 Gujarat, India
Declares that the product/s Product:	Digital Key Phone
Model Type:	EON 48
Trade Name:	MATRIX
Product Options:	This declaration covers all options of the above product, conforms to the following product specifications:

EMI/EMC Standard:

CISPR 22	: 2004-06 (Edition 4.1)
CISPR 24	: 2010
IEC 61000-4-2	: 2001-04 (Edition 1.2)
IEC 61000-4-4	: 2004-07 (Edition 2.0)
IEC 61000-4-5	: 2005-11(Edition 2.0)
IEC 61000-4-6	: 2004-11(Edition 2.2)
IEC 61000-4-8	: 2001-03(Edition 1.1)

SAFETY Standard:

IEC 60950-1 : 2001 (Edition 1.0)

Supplementary Information's:
The Product herewith complies with the following directives :

EMC	2014/30/EU
Low Voltage Directive	2014/35/EU
RoHS Recast (RoHS 2)	2011/65/EU (as per standard EN 50581:2012)

Matrix Comsec Pvt. Ltd. Vadodara

Mr. Ganesh Jivani
Director
Date: 28.09.2017

CE

MATRIX COMSEC PVT. LTD.
Registered/Head Office: 394-GIDC, Makarpura, Vadodara-390 010, India. Ph: +91 265 2630555, Email: Inquiry@MatrixComSec.com • **www.MatrixComSec.com**
Manufacturing Unit: 15 & 19-GIDC, Waghodia, Dist. Vadodara-391 760, India. Ph: +91 2668 263172/73 • **CIN:** U72200GJ1998PTC034047

EON310

MATRIX
TELECOM | SECURITY

Declaration of Conformity

Manufacturer's Name	:	Matrix Comsec Pvt. Ltd.
Manufacturer's Address	:	15 &19-GIDC, Waghodia, Dist: Vadodara 391760 Gujarat, India
Declares that the product/s Product	:	Digital Key Phone
Model Type	:	EON310
Trade Name	:	MATRIX
Product Options	:	This declaration covers all options of the above product, conforms to the following product specifications:

EMI/EMC Standard:

EN 55022 : 2010
EN 55024 : 2010
EN 61000-4-2 : 2009
EN 61000-4-3 : 2006(with A1:2008 and A2:2010)
EN 61000-4-4 : 2012(with A1:2010)
EN 61000-4-5 : 2006(with A1:2000)
EN 61000-4-6 : 2009
EN 61000-4-8 : 2010

SAFETY Standard:

IEC 60950-1 : 2005 + A1:2009 + A2:2013

Supplementary Information's:
The Product herewith complies with the following directives :

EMC	2014/30/EU
Low Voltage Directive	2014/35/EU
RoHS Recast (RoHS 2)	2011/65/EU (as per standard EN 50581:2012)

Mr. Ganesh Jivani
Director
Date: 28.09.2017

CE

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EON510

MATRIX
TELECOM | SECURITY

Declaration of Conformity

Manufacturer's Name:	Matrix Comsec Pvt. Ltd.
Manufacturer's Address :	15 &19-GIDC, Waghodia, Dist: Vadodara 391760 Gujarat, India
Declares that the product/s Product:	Premium Digital Key Phone
Model Type:	EON 510
Trade Name:	MATRIX
Product Options:	This declaration covers all options of the above product, conforms to the following product specifications:

EMI/EMC Standards:

EN 55022	: 2010
EN 55024	: 2010
EN 61000-4-2	: 2009
EN 61000-4-3	: 2010
EN 61000-4-4	: 2012
EN 61000-4-5	: 2014
EN 61000-4-6	: 2014
EN 61000-4-8	: 2010

SAFETY Standard:

IEC 60950-1 : 2005 (2nd Edition) + A1:2009 + A2:2013

Supplementary Informations :
The Product herewith complies with the following directives :

EMC	2014/30/EU
Low Voltage Directive	2014/35/EU
RoHS Recast (RoHS 2)	2011/65/EU (as per standard EN 50581:2012)

Matrix Comsec Pvt. Ltd. Vadodara

CE

Mr. Ganesh Jivani
Director
Date: 17.07.2017

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FCC Part 15B ID : 2ADHNVP310 for SPARSH VP310

FCC Part 15B ID : 2ADHNVP330 for SPARSH VP330

FCC Part 15B ID : 2ADHNVP510 for SPARSH VP510

FCC Class B Information [361]

This equipment has been tested and found to comply with the limits for a Class B digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation.

This equipment generates, uses and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communications.

However, there is no guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one of the following measures:

- Reorient or relocate the receiving antenna.
- Increase the separation between the equipment and receiver.
- Connect the equipment into an outlet on a circuit different from that to which the receiver is connected.
- Consult the dealer or an experienced radio/TV technician for help.

This device complies with Part 15 of the FCC Rules. Operation is subject to the following two conditions:

3. this device may not cause harmful interference, and
4. this device must accept any interference received, including interference that may cause undesired operation.

361.Common for SPARSH VP310, SPARSH VP330, SPARSH VP510, SPARSH VP248, SPARSH VP210.

SPARSH VP248

MATRIX
TELECOM | SECURITY

Declaration of Conformity

Manufacturer's Name:	Matrix Comsec Pvt. Ltd.
Manufacturer's Address :	15 & 19-GIDC, Waghodia, Dist: Vadodara 391760 Gujarat, India
Declares that the product/s Product:	SPARSH VP
Model Type:	SPARSH VP248
Trade Name:	MATRIX
Product Options:	This declaration covers all options of the above products

Conforms to the following product specification.

EMI/EMC Standard:

CISPR 22	: 2005-04 Edition 5.0
IEC 61000-3-2	: 2004-11 Edition 2.2
IEC 61000-3-3	: 2002-03 Edition 1.1
CISPR 24	: 1997 AMD2:2002
IEC 61000-4-2	: 2001-04 Edition 1.2
IEC 61000-4-3	: 2002-09 Edition 2.1
IEC 61000-4-4	: 2004-07 Edition 2.0
IEC 61000-4-5	: 2001-04 Edition 1.1
IEC 61000-4-6	: 2004-11 Edition 2.1
IEC 61000-4-8	: 2001-03 Edition 1.1
IEC 61000-4-11	: 2004-03 Edition 2.0

SAFETY
IEC 60950-1: 2001 Edition 1.0

Supplementary information:
The Product herewith complies with the following directives ;

EMC	2014/30/EU
Low Voltage Directive	2014/35/EU
RoHS Recast (RoHS 2)	2011/65/EU (as per standard EN 50581:2012)

Mr. Ganesh Jivani
Director
Date: 26.05.2016

Matrix Comsec Pvt. Ltd. MATRIX Vadodara

CE

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Factory: 39-GIDC, Waghodia-391 760, Dist. Vadodara, India. Ph: +91 2668 262056/57, Fax: +91 2668 262631

SPARSH VP310

MATRIX
TELECOM | SECURITY

Declaration of Conformity

Manufacturer's Name:	Matrix Comsec Pvt. Ltd.
Manufacturer's Address :	15 &19-GIDC, Waghodia, Dist: Vadodara 391760 Gujarat, India
Declares that the product/s Product:	SPARSH VP
Model Type:	SPARSH VP310
Trade Name:	MATRIX
Product Options:	This declaration covers all options of the above product, conforms to the following product specifications:

EMI/EMCStandard:

EN 55022 : 2010 (Consolidated with Corrigendum (AC) :2011)
IEC 61000-3-2 : 2014
IEC 61000-3-3 : 2013
EN 55024 : 2010
IEC 61000-4-2 : 2008 (Edition 2:2008-12)
IEC 61000-4-3 : 2006 (Edition 3:2006 Consolidated with Am1:2007 & A2:2010)
IEC 61000-4-4 : 2012
IEC 61000-4-5 : 2014
IEC 61000-4-6 : 2013
IEC 61000-4-8 : 2009 (Edition 2: 2009-09)
IEC 61000-4-11 : 2004 (Edition 2:2004)

SAFETY Standard:

IEC 60950-1 : 2005(Edition 2) + A1:2009 + A2:2013

Supplementary Information's:
The Product herewith complies with the following directives :

EMC	2014/30/EU
Low Voltage Directive	2014/35/EU
RoHS Recast (RoHS 2)	2011/65/EU (as per standard EN 50581:2012)

Mr. Ganesh Jivani
Director
Date: 28.09.2017

Matrix Comsec Pvt. Ltd. MATRIX Vadodara

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SPARSH VP330

MATRIX
TELECOM | SECURITY

Declaration of Conformity

Manufacturer's Name:	Matrix Comsec Pvt. Ltd.
Manufacturer's Address :	15 &19-GIDC, Waghodia, Dist: Vadodara 391760 Gujarat, India
Declares that the product/s Product:	SPARSH VP(INTUITIVE TOUCH-SCREEN IP PHONE)
Model Type:	SPARSH VP330
Trade Name:	MATRIX
Product Options:	This declaration covers all options of the above product, conforms to the following product specifications:

EMI/EMC Standard:

EN 55022	: 2010 (Consolidated with Corrigendum (AC) :2011)
IEC 61000-3-2	: 2014
IEC 61000-3-3	: 2013
EN 55024	: 2010
IEC 61000-4-2	: 2008 (Edition 2:2008-12)
IEC 61000-4-3	: 2006 (Edition 3:2006 Consolidated with Am1:2007 & A2:2010)
IEC 61000-4-4	: 2012
IEC 61000-4-5	: 2014
IEC 61000-4-6	: 2013
IEC 61000-4-8	: 2009 (Edition 2: 2009-09)
IEC 61000-4-11	: 2004 (Edition 2:2004)

SAFETY Standard:

IEC 60950-1 : 2005 + A1:2009 + A2:2013

Supplementary Information's:
The Product herewith complies with the following directives :

EMC	2014/30/EU
Low Voltage Directive	2014/35/EU
RoHS Recast (RoHS 2)	2011/65/EU (as per standard EN 50581:2012)

Mr. Gaurav Jivani
Director
Date: 28.09.2017

CE

MATRIX COMSEC PVT. LTD.
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Manufacturing Unit: 15 & 19-GIDC, Waghodia, Dist. Vadodara-391 760, India. Ph: +91 2668 263172/73 • **CIN:** U72200GJ1998PTC034047

SPARSH VP510

MATRIX
TELECOM | SECURITY

Declaration of Conformity

Manufacturer's Name:	Matrix Comsec Pvt. Ltd.
Manufacturer's Address :	15 &19-GIDC, Waghodia, Dist: Vadodara 391760 Gujarat, India
Declares that the product/s Product:	Premium IP Phone
Model Type:	SPARSH VP510
Trade Name:	MATRIX
Product Options:	This declaration covers all options of the above product, conforms to the following product specifications:

EMI/EMC Standard:

EN 55022	: 2010
IEC 61000-3-2	: 2014
IEC 61000-3-3	: 2013
EN 55024	: 2010
IEC 61000-4-2	: 2008 (Edition 2:2008-12)
IEC 61000-4-3	: 2010
IEC 61000-4-4	: 2012
IEC 61000-4-5	: 2014
IEC 61000-4-6	: 2013
IEC 61000-4-8	: 2009 (Edition 2: 2009-09)
IEC 61000-4-11	: 2004 (Edition 2:2004)

SAFETY Standard:

EN/IEC 60950-1 : 2006(Edition 2.0) + A1:2009 + A2:2013

Supplementary Information's:
The Product herewith complies with the following directives :

EMC	2014/30/EU
Low Voltage Directive	2014/35/EU
RoHS Recast (RoHS 2)	2011/65/EU (as per standard EN 50581:2012)

Mr. Ganesh Jivani
Director
Date: 28.09.2017

Matrix Comsec Pvt. Ltd. Vadodara

CE

MATRIX COMSEC PVT. LTD.
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SPARSH VP210

MATRIX
TELECOM | SECURITY

EU DECLARATION OF CONFORMITY

EU DECLARTION OF CONFIRMITY

Manufacture : : MATRIX COMSEC PVT LTD

Manufacture Address : 15 & 19- GIDC , Waghodia, Vadodara-391760 (Gujarat, India)

Trade Name : **MATRIX**

Declare that the DoC is issued under our sole responsibility and belongs to the following products;

Product : **SPARSH VP**

Model/ TYPE : SPARSH VP210

Essential Requirements /Directives		Applied Specifications/ Standards
EMC	2014/30/EU	EN 55032: 2015+A11:2020; EN 55035:2017+A11:2020; EN 61000-3-2: 2019; EN 61000-3-3: 2013+A1:2019; EN 61000-4-2: 2009; EN 61000-4-3: 2006+A2:2010; EN 61000-4-4: 2012; EN 61000-4-5: 2014+A1:2017; EN 61000-4-6: 2014; EN 61000-4-8: 2010; EN 61000-4-11: 2020
LVD/SAFETY	2014/35/EU	IEC 62368-1: 2018
RoHS (RoHS2)	2011/65/EU	EN 50581: 2012

I hereby declare that the equipment named above has been designated to comply with the relevant section of the above reference standards and meet all essential requirements of the specified directives.

CE

Matrix Comsec Pvt. Ltd. MATRIX Vadodara

Mr Ganesh Jivani
Managing Director
Date: 04/11/2020

MATRIX COMSEC PVT. LTD.
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Manufacturing Unit: 15 & 19-GIDC, Waghodia, Dist. Vadodara-391 760, India. Ph: +91 2668 263172/73 • CIN: U72200GJ1998PTC034047

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  R&D Team
  Matrix Comsec Pvt Ltd
  394, Makarpura GIDC,
  Vadodara - 390 010
  Gujarat
  India.
  Customer shall bear the shipping and handling charges.

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Version 2, June 1991

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