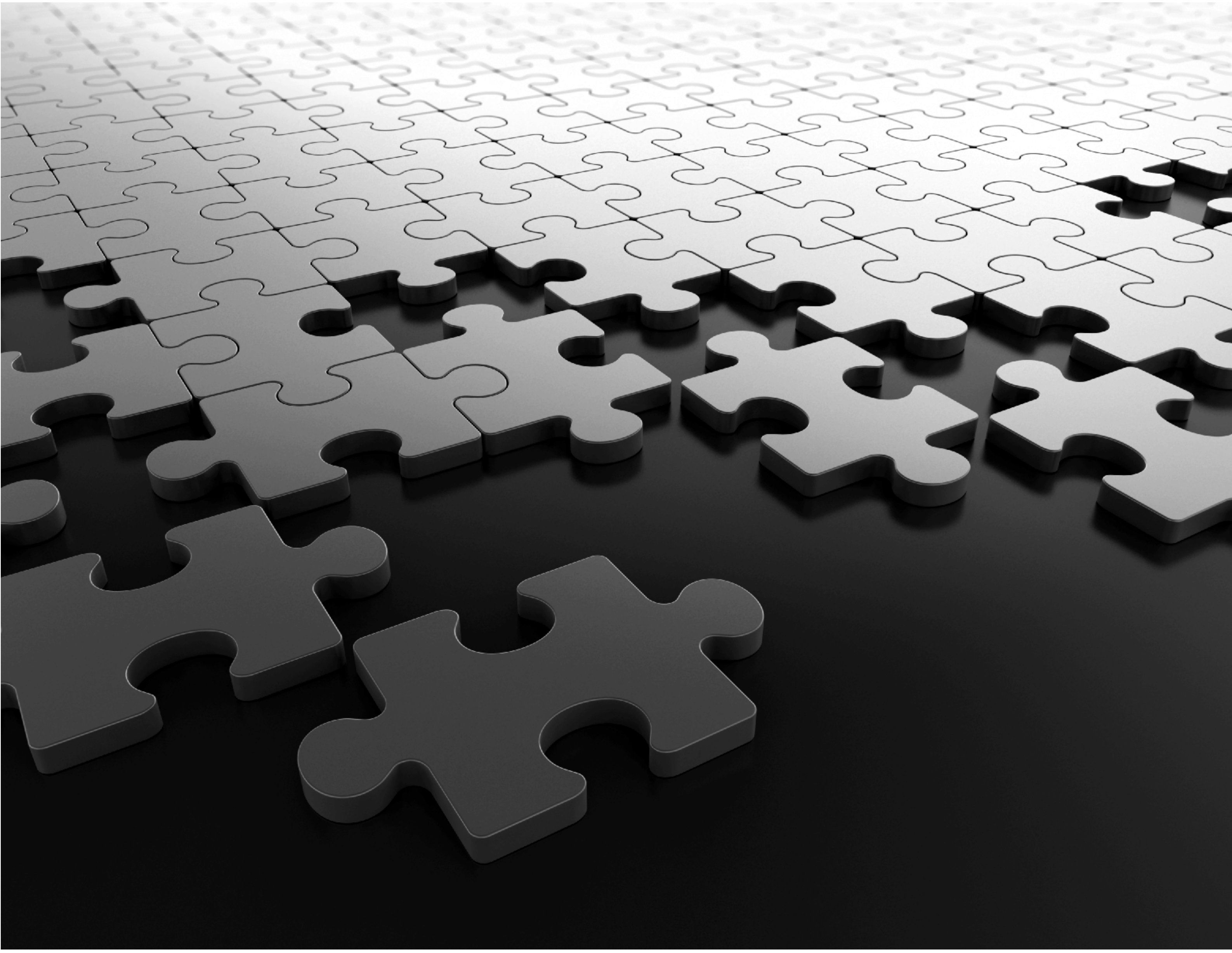
ETERNITY NE
System Manual



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ETERNITY NE

The Next Generation IP PBX for Small Businesses

System Manual

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This is a general documentation for all models of the product. The product may not support all the features and facilities described in the documentation.

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CHAPTER 1

Introduction

Welcome

Thank you for choosing Matrix ETERNITY NE! We hope you will make optimum use of this intelligent and versatile Unified Communication Server. Please refer this document carefully to get acquainted with the product before installing and operating it.

About this System Manual

ETERNITY NE has three variants built on the ETERNITY NENX platform, denoted as ETERNITY NENXIP50, ETERNITY NENX416 and ETERNITY NENX312.

This document provides detailed information and instructions for installing and configuring ETERNITY NENX. It also provides essential guidelines for configuring and accessing different features and facilities offered by it.

This document also aims to provide important informations and instructions related to ETERNITY NENX so that you can get properly acquainted with the Server and can use it effectively and efficiently.

You may also refer to ETERNITY NE Quick Start, for quick installation. To view or download the Quick Start and other related documents, scan the QR Code printed on the Product Label/Packaging Label.

You may also view or download the Quick Start from https://www.matrixtelesol.com/product-manuals.html

For instructions on using the features of the Server 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/product-registration-form.html

This is a common document for all the variants of ETERNITY NENX, that operate using the SARVAM UCS SOHO Application. This document is written with reference to ETERNITY NENXIP50.

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 system, 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 ETERNITY NENX.

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

- **System Administrators**, who will administer ETERNITY NENX. 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 configure the system 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**, persons/organizations who will use the resources of the ETERNITY NENX. They may be personnel of small and home businesses, and other commercial and public organizations/institutions.

Organization of this Document

This system manual is broadly divided into the following:

- **Introduction:** This chapter provides an overview of this document, its purpose, intended audience, organization, terms and conventions used to present information and instructions.

- **Product Description:** This chapter provides the description of the system features and benefits, the application scenarios, hardware overview and different interfaces supported by the system.

- **Installing ETERNITY NENX:** This chapter gives step-by-step instructions for preparing for and connecting various interfaces with the ETERNITY NENX such as connecting to VoIP network, connecting to CO network.

- **Configuring ETERNITY NENX:** This chapter contains description of how to configure ETERNITY NENX.

- **Basic Settings:** This chapter provides the instructions for configuring the basic parameters of the ETERNITY NENX, which are sufficient to get the system into operation.

- **Configuring Voice Mail system:** This chapter describes how to configure the Voice Mail System.

- **SMS Server:** This chapter contains the information regarding SMS Server feature supported in the Server.

- **Computer Telephone Integration:** This chapter provides information regarding the CTI feature supported in the Server.

- **Features and Facilities:** This chapter describes in detail, each feature and facility offered by the ETERNITY NENX. This includes a description of the feature/facility, how it works, and how to configure and use 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.

- **Voice Mail Features:** This chapter describes in detail, the voice mail features offered by the Server.

- **System Maintenance:** This chapter provides instructions for back-up, generating reports and debugging.
- **Status:** This chapter describes the indicators of the System, Network (Ethernet), SIP Trunks, SIP Extensions, Mobile and CO status.

Refer "Contents" for more details.

How to Read this System Manual

This document is organized in a manner to help you get familiar with ETERNITY NENX Platform and SARVAM UCS SOHO Application, help you with system installation, configuration and use of 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.

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 step-by-step format. 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. Example: Do Not Disturb, 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 access codes configured by the System Engineer. For more information, read the topic "Access Codes" in the document.

Notices

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

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 **'ETERNITY NE**', '**ETERNITY**', **'ETERNITY NENX**', **'SARVAM UCS'**, **'SARVAM UCS SOHO'** '**System**', '**PBX'**, **'Server'** are used interchangeably and synonymously to mean ETERNITY NENX.

- **'SIP Phone', 'IP Phone':** These words are used interchangeably and synonymously to mean IP Phones connected to ETERNITY NENX.
- **'SLT Port', 'FXS Port':** These words are used interchangeably and synonymously to mean the port to which a SLT Phone is connected.
- **Called party/Callee:** The person to whom the call is made.
- **Calling party/Caller:** The person who makes a call.
- **CO Network:** The public telephone exchange.
- **CO Lines:** The lines subscribed from the CO Network. In this document it refers to Analog, two-wire Trunk Lines.
- **CO Trunks:** Two-wire trunks, i.e. analog trunk lines from the POTS network.
- **Extension:** It is a telephone instrument, SLT, IP-Phone connected to ETERNITY NENX.
- **External Calls:** Calls made by users of ETERNITY NENX to subscribers of PSTN, PLMN, ITSPs, etc.
- **External Numbers:** It denotes the numbers of parties/individuals outside the PBX 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 ETERNITY NENX.
- **Internal numbers:** Same as extension numbers.
- **Mobile Extension:** A mobile/landline phone used as a remote extension of ETERNITY NENX. You can access all the features of an extension of ETERNITY NENX from the mobile/landline phone.
- **Port:** The physical interfaces for the trunk lines and extension lines.
- **Service Provider:** The providers of telecom network lines/Internet - POTS, PSTN, GSM, and Internet Telephony Service Providers (ITSP).
- **Single Line Telephone (SLT):** Any standard two-wire telephones attached as extensions of the ETERNITY NENX.

- **Station:** Same as extension.
- **System Administrator Commands/SA Commands:** Number strings dialed from the System Administrator mode to operate features or set/cancel features for other extensions.
- **System Commands/SE Commands:** Number strings dialed from the System Engineer mode to configure the system features/functions.
- **GSM (Global System for Mobile):** This is also referred to as the 2G network. Hence, 2G or GSM is used to denote the 2G network.

  The word ‘GSM’ is also used interchangeably and synonymously to mean 2G, 3G and 4G Mobile Network.
- **UMTS (Universal Mobile Telecommunications System):** This is also referred to as the 3G network. Hence, 3G or UMTS is used to denote the 3G network.
- **LTE (Long Term Evolution):** This is also known as the 4G network. Hence, 4G or LTE is used to denote the 4G network.

Using this Manual, you will be able to set up, operate and make optimum use of this feature-packed Server.

If you encounter any technical problems, please contact your Dealer/reseller or the Matrix Customer Care.

CHAPTER 2

Product Description

ETERNITY NENX is an integrated Unified Communication Server with seamless mobility. It caters to the communication requirements of small businesses.

ETERNITY NENX offers connectivity to analog and digital networks: CO, Mobile and VoIP. 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.

Intelligent features like Auto Attendant, CLI 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 enhances productivity.

ETERNITY NENX can route a VoIP call to GSM. In the same way, a call can be routed either on VoIP, GSM or CO. Further, you can select Fixed or Least Cost Routing to route outgoing calls. ETERNITY NENX can handle calls on all ports simultaneously.

ETERNITY NENX can work as an adjunct to your existing telephony infrastructure, as a Gateway, saving the cost of equipment replacement, wiring and installation, while giving you multiple network connectivity and a host of intelligent features. It is a unique convergence of innovative switching technology and intelligent software features.

The system is built on PCM/TDM, 100 percent non-blocking, digital technology, providing high density switching, and is powered by a 32-bit RISC processor. Ensuring reliable, efficient, and unrestricted simultaneous communication (incoming and outgoing) by all users.

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

- Auto Attendant
- Call Budget on Trunks and Extensions
- CDR Reports
- Class of Service
- CLI Based Routing
- Closed User Group
- Conference Dial-in
- Conversation Recording
- Fax over IP (FoIP)
- Hot Swapping of SIM Cards
- IP Trunks and Extensions
- Least Cost Routing
- Multi-Party Conference
- Power Fail Transfer Module
- Presence Indication and IM
- Return Call to Original Caller (RCOC)
- Scheduled Call Forward
- SIP Registrar and Proxy Server
- SMDR Reports
- System Activity Log
- System Fault Log
- Toll Control
- Video Calling
- Voice Applications
- Voice Mail
- Web-based Programming

ETERNITY NENX is easy to install and operate. The built-in web server *Jeeves* allows you to configure the system parameters and features on site and also from a remote location using any Internet browser.

The Interfaces

ETERNITY NENX supports the following interfaces for connecting to different telecom networks, standard telephones, IP Phones etc.

The CO Interface

The CO Interface enables ETERNITY NENX to be connected to the PSTN Network. The PSTN 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.

The versatile architecture of ETERNITY NENX 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
- Programmable Loop Current

The Mobile Interface

The Mobile Interface enables the ETERNITY NENX to be connected to GSM/UMTS/LTE network operators worldwide. ETERNITY NENX's Mobile Interface supports full Quad-Band Operation (GSM900, 1800, 1900MHz) for world-wide use, for Global, Inter and Intra country roaming.

The Mobile Interface supports the following features:

- GSM 2G, GSM 3G, GSM 4G network support.
- Hot Swapping of the SIM Card.
- Selectable GSM Frequency Bands - 900, 1800, 1900, 850 + 1900, 900+1800 MHz.
- Programmable Network Selection - Manual and Automatic.
- 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) and Personal Unlock Keyword (PUK).
- Wireless WAN support over 3G or 4G, which can be used to make VoIP calls (if VoIP module is installed).

ETERNITY NENX Mobile Interface does not support GPRS features, Fax and Data services, and network supported services, except CLIR and USSD.

The VoIP Interface

The Voice-over-IP (VoIP) Interface routes over the Internet, all the outgoing and incoming calls made or received by the extensions of the ETERNITY NENX and extensions of other System that are networked with the ETERNITY NENX.

The VoIP Interface supports Session Initiation Protocol (SIP), the industry standard VoIP.

The VoIP Interface supports SIP Trunks and SIP Extensions.

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

The VoIP Module has an in-built Registrar Server that allows any SIP enabled device like a Wi-Fi mobile handset, a PDA or an IP-Phone to be registered with it and function as the 'SIP Extension' of the ETERNITY NENX. The SIP Extension users can make and receive calls to any extension user of the ETERNITY as well as any external numbers over PSTN, GSM, VoIP. 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:

- 8 SIP Trunks - for Proxy as well as Peer-to-Peer (non-Proxy) calls
- Up to 50 SIP Extensions - Standard IP Phones, Matrix Extended IP Phones and UC Clients
- 8 Simultaneous Voice Calls
- Selectable Network Assignment (Connection Type) - Static IP, DHCP, PPPoE
- Selectable DNS - Automatic and Static
- Dynamic DNS for client
- STUN
- 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.
- Voice Codec Selection: G.723, G.729ab, 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
- Upto 10 Call Appearances on Extended IP Phone Extensions
- Registration of SIP Extensions from 3 different locations

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 ETERNITY NENX as extension phone.

The SLT Interface has the following features:

- Selectable Caller ID Presentation - DTMF, FSK
- Programmable Digit Pad Count for Caller ID
- 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)

ETERNITY NENX provides a full-fledged, 'in-skin' Voice Mail System. Thus, every extension of the ETERNITY NENX can be assigned a mailbox.

The key auto attendant and voice mail features supported by ETERNITY NENX'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
- 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 via email and call
- Redirecting messages

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

ETERNITY NENX'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.

LAN Interface

ETERNITY NENX has a single Ethernet port, to which you can connect a standalone computer or to a LAN Switch.

You will need to connect ETERNITY NENX to a computer to:

- access the web-based programming tool *Jeeves*.
- 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.

You will be able to use the VoIP interface only if the system is connected to the LAN network.

WAN Interface

ETERNITY NENX supports WAN interface over Ethernet port and Wireless WAN over the UMTS/LTE Mobile Port (Mobile Port1) with 3G/4G SIM.

When you use Wireless WAN over UMTS/LTE Mobile port, you can use the Ethernet Port as LAN.

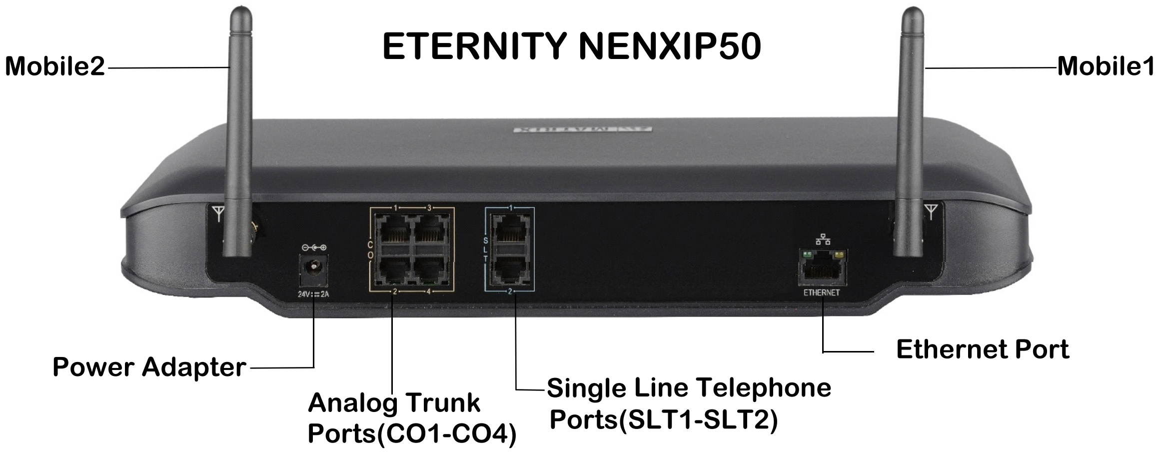
Know Your ETERNITY NENX

Matrix ETERNITY NENX supports three variants:

- ETERNITY NENXIP50
- ETERNITY NENX312
- ETERNITY NENX416

Illustrated below are design of the enclosure and position of connectors on different configurations of ETERNITY NENX.

ETERNITY NENXIP50



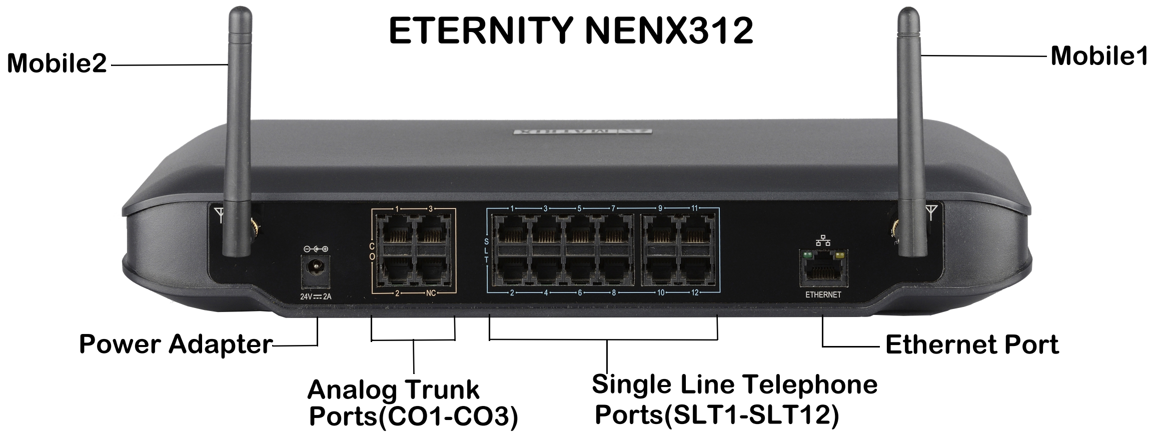
Built-in Interfaces

- 4 CO Trunks
- 2 SLT ports
- VoIP

Optional Interfaces

- Mobile
- Voice Mail System

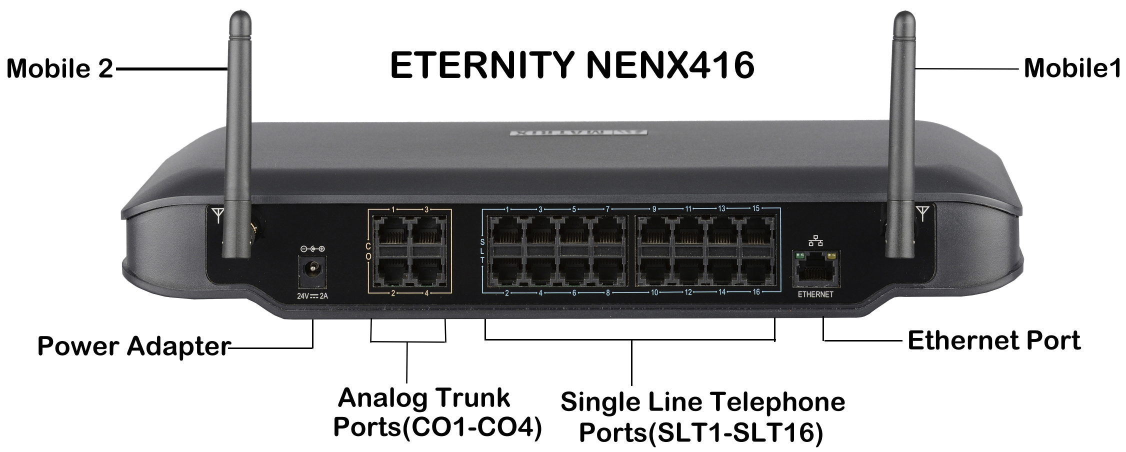
ETERNITY NENX312



Built-in Interfaces
- 3 CO Trunks
- 12 SLT ports

Optional Interfaces
- VoIP
- Mobile
- Voice Mail System

ETERNITY NENX416



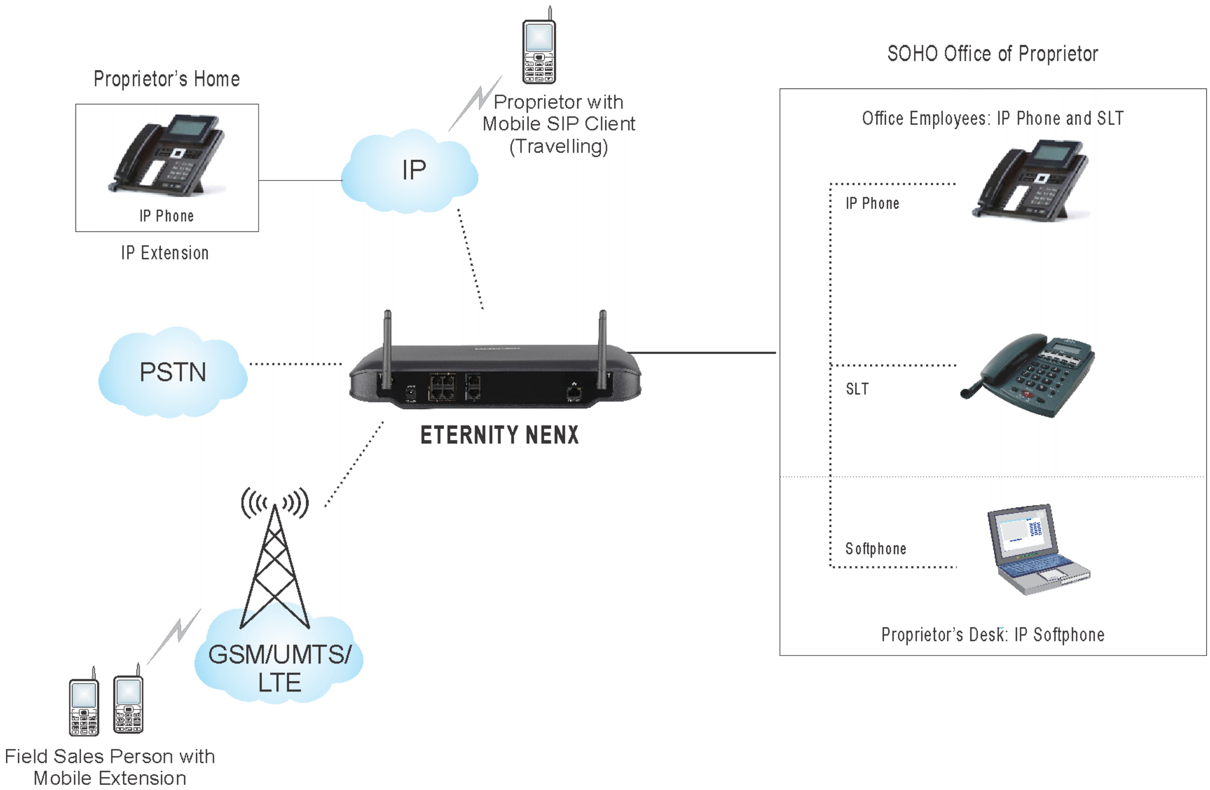
Built-in Interfaces
- 4 CO Trunks
- 16 SLT ports

Optional Interfaces
- VoIP
- Mobile
- Voice Mail System

Applications of ETERNITY NENX

ETERNITY NENX is designed for small and home offices, and nascent enterprises.

Illustrated below is an example of how ETERNITY NENX can be optimally used by small businesses, taking full advantage of its enterprise grade features and facilities.



Compatibility Versions of Extended Clients

Compatibility is an essential element for interactions between the Server and the clients.

The following table lists, version of the clients compatible with the System.

Clients	Version
VARTA ADR100	V01R03 and later
VARTA AMP100	V01R04 and later
VARTA WIN200	V01R07 and later
SPARSH VP248	V05R23 and later
SPARSH VP310	V01R13 and later
SPARSH VP330	V01R12 and later
SPARSH VP510	V01R12 and later
Extended SPARSH VP710	V01R03 and later
SPARSH VP210	V01R01 and later

It is recommended to use the above mentioned client version with System for the overall functionality.

Technical Specifications of ETERNITY NENX

This section enlists the technical specifications and other essential information of all the variants of ETERNITY NENX.

Specifications	ETERNITY NENXIP50	ETERNITY NENX312	ETERNITY NENX416
Application	SARVAM UCS SOHO		
CO Lines	4	3	4
Single Line Telephones	2	12	16
SIP Trunks	8	8	8
ETERNITY NE VS (VoIP Server Module)[a]	1	1	1
SIP Extensions[b]	50	50	50
GSM/UMTS/LTE Trunks*	2	2	2
Voice Mail Channels	4 Channels	4 Channels	4 Channels
VoIP Channels	8 Channels	8 Channels	8 Channels
Maximum IP to IP Call	25	25	25
Maximum IP to TDM Call	8	8	8
Ethernet (RJ45) Port	Auto MDIX (10/100 Base-T)		
Voice Mail System**	Pen drive (factory-fitted)		

a.By default, the VoIP module is factory fitted in ETERNITY NENXIP50. However, it is an optional module in ETERNITY NENX312 and ETERNITY NENX416. You must purchase the VoIP module to register and configure the SIP Extensions.

b.By default, ETERNITY NENXIP50 supports registration and configuration of 50 SIP Extensions and ETERNITY NENX312 and ETERNITY NENX416 supports registration and configuration of 10 SIP Extensions only. To configure and register additional SIP Extensions, you need to purchase and activate the IP Subscriber license as per your requirement. To know about IP Subscriber license, refer to "License Management".

* It is an optional module.
** It is an optional module, requires license activation.
To know other technical details of ETERNITY NENX, refer to "Appendix".

CHAPTER 3

Installing ETERNITY NENX

Preparing for Installation

Before you begin the installation of ETERNITY NENX, make sure you have the following items ready:

- A suitable location to install ETERNITY NENX.
- Necessary telecom wiring in place, with wall jacks for extension lines at the required locations.
- Standard, good quality, twisted pair telephone cables with RJ11 plug.
- A dedicated Power supply outlet close to the system.
- For the **SLT ports**, arrange for as many standard analog telephone instruments as required to connect as SLT extensions. You may select any standard telephone instrument like rotary phone, Pulse/Tone switchable push-button phone, Feature phone or Cordless phone.
- A fax machine, as required, to connect to SLT port.
- For the **CO ports**, arrange for one or more active Analog, two-wire trunk lines, as required.
- For the **Mobile ports**, a SIM Card to test mobile network connectivity, if GSM/UMTS/LTE module is present.
- SIP Account information to be configured in the system to test SIP calls.
- Any standard SIP Phone or Matrix Extended IP Phone to register as **SIP Extension** of ETERNITY NENX, if required.
- Insert the Pen Drive provided with the system. Keep the License Voucher ready (you need to purchase this from Matrix/Dealer), if you wish to use the Voice Mail System (VMS).
- A standalone computer or a computer connected in LAN to access **Jeeves**, the web-based configuration tool of ETERNITY NENX.

You will be able to use the VoIP interface only if the system is connected to the LAN network.

Installation Site

ETERNITY NENX may be mounted on a table or on the wall. Select an appropriate site to install ETERNITY NENX 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 or excessive cold.
- The site should not be near any source of electromagnetic noise such as any radio equipment, heavy transformers, faulty electric chokes of tube-lights, any device having faulty coil, etc.
- The site should be equidistant from all the extensions to simplify cabling network and reduce cabling costs.
- The site should allow installation of the system at a clearance of at least 3.5 feet off the ground. Installation at this height makes preventive or corrective maintenance tasks easy.
- If the system has GSM/UMTS/LTE module, the site should have sufficient network coverage available.

  For more details, refer “Protecting ETERNITY NENX and Yourself”.

Cables

- Select standard, good quality telephone cables 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.

Selecting Extensions

Select appropriate IP Phones to be connected as SIP extensions. You may connect:

- any model of the proprietary IP Phones
- any Standard SIP Phone

You may also select standard telephone instruments to be connected to the SLT ports. You may connect telephone instruments like rotary phone, Pulse/tone switchable push-button phone, Feature phone or Cordless phone.

Provide Power Supply

- The DC Adapter (24VDC-2.5 Amp) provided with ETERNITY NENX works with input voltage ranging between 100-240VAC.
- 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.
- The use of a UPS is recommended in case you experience frequent power failures.

Getting Started

- Unpack ETERNITY NENX and verify your package contents. In case any of the items is missing or damaged, contact your Dealer/Distributor.

Package Contents

- ETERNITY NENXIP50[1]
- ETERNITY NE VS (VoIP Server Module)
- Power Adapter
- 8 GB USB 2.0 Pen-Drive (Flash Drive)
- Power Cord as per country standard
- Wall Mounting Template
- Two self tapping screws and screw grips for wall mounting

Product documentation can be viewed by scanning the QR code printed on the Product Label/Packaging Label.

1. *For packaging contents of ETERNITY NENX312 and ETERNITY NENX416, refer to* *"Packing List".*

Protecting ETERNITY NENX and Yourself

ETERNITY NENX 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.

Take every safety precaution to reduce the risk of fire, electric shock and injury to persons. Read and understand the precautions, dos and don't s 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 locations that are close to a water source, on movable or unstable surfaces, near high frequency generating devices, and areas where it may be exposed to dust, direct sunlight, heat, excessive cold or humidity, where shocks or vibration are frequent or strong.
- 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. The DC Adapter (24VDC-2.5 Amp) provided with ETERNITY NENX works with input voltage ranging between 100-240 VAC.
- Protect the system from heavy voltages from the mains.
- ETERNITY NENX has trunk and extension interfaces. So there are chances of heavy voltages entering the system from trunk lines or from overhead extensions due to:
  - Heavy voltage line falling on the CO line or on the overhead extension cables. A dangerous surge can occur if a telephone line comes in contact with a power line.
  - Lightning/Thunderbolts.
- Protect ETERNITY NENX from lightning and electrical surges by installing Primary Protection/Surge Protectors on the trunk and long-distance or off-premise extension lines. The product warranty does not cover damages resulting from lack of primary protection on trunk lines.

Battery

ETERNITY NENX contains a 3VDC/18mAh (Li-Al) alloy-Manganese Dioxide Coin Battery (ML 1220 - Rechargeable) of diameter 12.5mm and height 2.0mm. 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.*

Shock and Fire Hazard

- Always wear a properly earthed, electrostatic discharge preventive wrist strap/belt while handling the system and its components to prevent damage to the system and harm to yourself.
- Do not open the system in power ON condition, to avoid shock hazards.
- The ventilation openings on the sides of the product's enclosure must not be blocked or covered to prevent overheating.
- Never insert or push objects of any kind into this product through the openings as they may touch dangerous voltage points or short out parts which may result in fire or electric shock.
- Do not overload wall outlets and extension cords as this can result in the risk of fire or electric shock.
- 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. If you are unable to insert the plug into the outlet, ask an electrician to replace the obsolete outlet. Do not defeat the purpose of the grounding type plug.
- Do not overload wall outlets and extension cords as this can result in the risk of fire or electric shock.
- 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 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

- Switch off power supply, and unplug the product from the power outlet before you clean.
- Do not use liquid cleaners or aerosol cleaners.
- Use a dry cloth for cleaning.

Service and Repair

- Do not disassemble this product. Incorrect reassembly may cause electric shock when the product is used. Take the product to a qualified technician when service or repair work is required.
- This product must be serviced by a qualified technician only. Call your dealer, if:
  - the power supply cord or plug is damaged or frayed.
  - liquid has been spilled into the product.
  - the product has been exposed to rain or water.
  - the product has been dropped or the cabinet has been damaged.
  - the product exhibits a distinct change in performance.

Disposal

- This product must be disposed according to the national laws and regulations prevailing in the country where it is installed. See “Disposal of Products/Components after End-Of-Life” for more details.

Warning for RF Safety:

The product complies with RF exposure guidelines as per standard FCC 47 CFR part 2. We recommend the following precautions:

- Make sure that the RF Antenna is installed at least 20cm away from other electronic and radio transmission devices.

- Make sure that the RF Antenna is installed at least 20cm 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.

Connecting CO Trunks

The CO ports of ETERNITY NENX provides the interface to connect the System with the CO Network. The CO interface supports different standards and features of networks across the world.

The number of CO Ports supported by the configurations of ETERNITY NENX are:

- **ETERNITY NENXIP50:** 4 CO Ports
- **ETERNITY NENX312:** 3 CO Ports
- **ETERNITY NENX416:** 4 CO Ports

Use standard, good quality, twisted-wire pair telephone cables with RJ11 plugs to connect the CO ports of ETERNITY NENX to the Trunk Lines from your exchange.

Connecting to Mobile Networks

The Mobile interface is optional in the ETERNITY NENX. The Mobile Port connects ETERNITY NENX to the GSM/UMTS/LTE network. It routes calls made and received over mobile networks, like a mobile handset[2].

The Mobile Port does not support Fax and network supported services, except CLIR and USSD.
The Mobile Port1 of ETERNITY NENX can also be used as the WWAN Interface to connect the system to the Internet.

For compatibility and use of Matrix GSM products (2G/3G/4G) in Russia and Iran Province connect with Matrix Sales or Technical Support Team.

Installing the GSM/UMTS/LTE Module

- Unpack the GSM/UMTS/LTE Module and verify package contents.

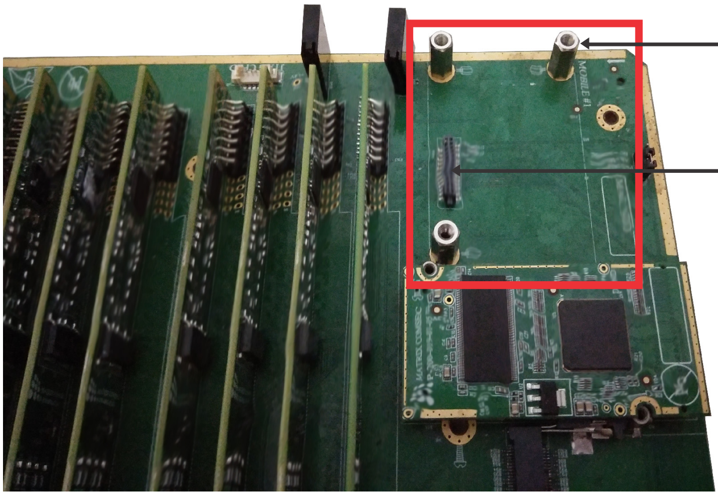
The 4G module will be supported only with PCB V3R1 onwards with CPLD version V3R2 or later.

- Make sure power supply is turned off before you begin installation.
- Unscrew and remove the top cover of the enclosure. Keep the screws and the cover aside.
- Select the Mobile Port for which you want to install the GSM/UMTS/LTE module on the mainboard. Locate the studs for GSM/UMTS/LTE module on the mainboard to install the GSM/UMTS/LTE module.

If you want to use Internet for SIP services, you must install 3G or 4G module in Mobile Port1 slot. Whether you install 2G or 3G or 4G module in Mobile Port2 slot, it will be used only for voice calls.

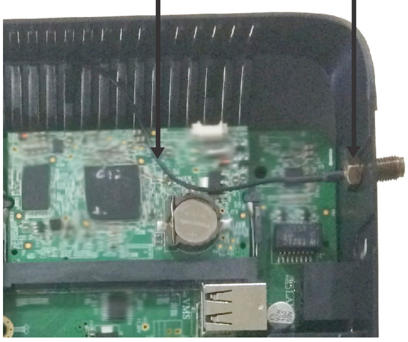
2. *Just like mobile handsets, each Mobile Port has a unique IMEI (International Mobile Equipment Identity) number, pasted on the mobile engine.*

- Gently seat the GSM/UMTS/LTE module on the studs on the mainboard. The connector pins on the module must make complete contact with those on the mainboard. Do not apply pressure.

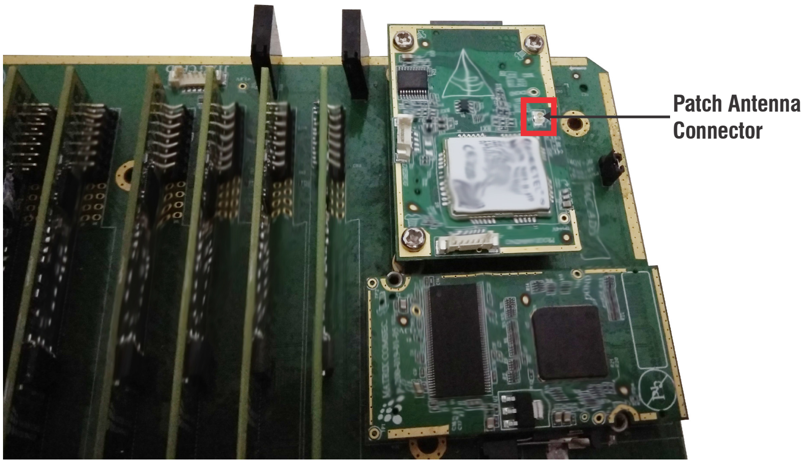


- When the module is seated firmly on the studs on the mainboard, secure the module with the screws.

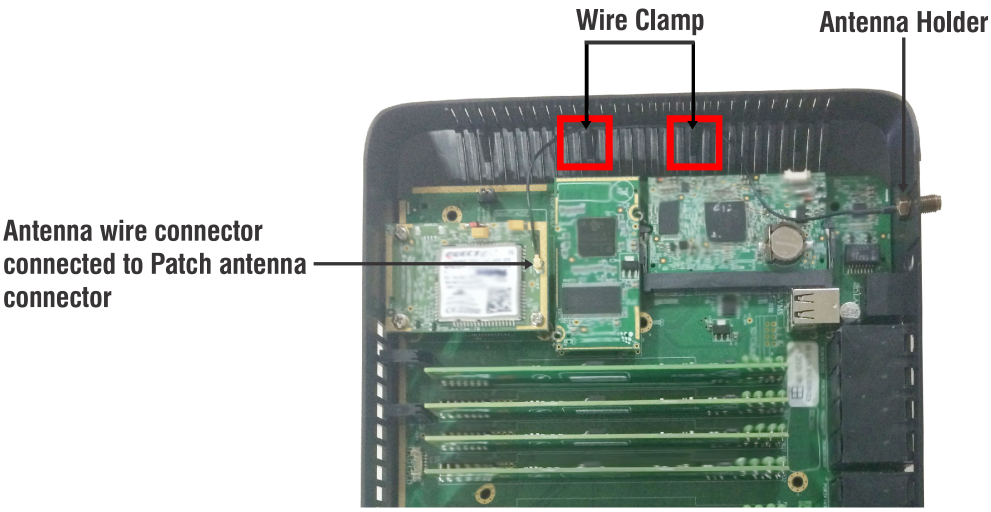
- Remove the rubber cap and fix the antenna holder of the Antenna wire onto the Antenna connector of the enclosure.



- Locate the patch antenna connector on the GSM/UMTS/LTE Module.



- Fix the antenna wire in the clamp and route the cable carefully to the GSM/UMTS/LTE module. Press the antenna wire connector onto the patch antenna connector on the GSM/UMTS/LTE Module as shown below.



- Similarly, you can install another GSM/UMTS/LTE module.

- If you do not want to install any other module, replace the top cover and secure the cover with the screws.

Personal Identification Number (PIN)

The SIM Cards can be protected from unauthorized use with a Personal Identification Number (PIN) on the SIM.

- Get the SIM Card from the GSM service provider of your choice ready.

- Protect the SIM Card from unauthorized use with a Personal Identification Number (PIN) on the SIM (in consultation with the customer/owner of the SIM).

SIM PIN Protection

To enable SIM PIN protection,

- get a mobile handset. Insert the SIM Card into the mobile handset.
- from the mobile handset, enable PIN Protection.
- change the SIM PIN to 1234 (this is the default PIN for both SIM Cards used in the system). You can change this SIM PIN later from ETERNITY NENX when configuring the Mobile Port.
- remove the SIM Card from the mobile handset.

If you do not want to use PIN protection, insert the SIM Card in the mobile handset and disable PIN protection. Remove the SIM Card from the mobile handset and insert it in the Mobile Port of the ETERNITY NENX.

- insert the SIM into the SIM slot of ETERNITY NENX, with its contact side facing up.
- push the SIM backwards into the slot. The SIM will be locked inside the slot.
- to unlock the SIM, push the protruded portion of the SIM backwards again.
- repeat the same steps to insert the other SIM Card.
- connect the Antenna to the antenna connector.

The UMTS/LTE Mobile Ports of ETERNITY NENX also provide Wireless WAN Interface. If you want to use wireless WAN over the Mobile Port, you must install UMTS/LTE module in the system and activate Internet services on the SIM.

- Switch On the power supply, if you have finished connecting all the required ports.

After each Power On, it takes about 3 minutes for the Mobile Ports to get registered with the network. Once registration with the mobile network is completed, the Mobile Port can be used.

ETERNITY NENX supports Hot Swapping of SIM Cards. You will be able to insert/remove SIM Card/s without switching off the system. However, while removing SIM Card in Power On condition, make sure the Mobile Port is idle or else it will lead to call disconnection.

Connecting to the VoIP Network

The VoIP Interface enables the extensions of ETERNITY NENX to connect to the IP network and make Proxy as well as Non-Proxy (Peer-to-Peer) VoIP calls.

By default, a VoIP module is factory fitted in ETERNITY NENXIP50. However, the VoIP interface is optional in the ETERNITY NENX312 and ETERNITY NENX416.

Voice Channels

ETERNITY NENX supports 8 Voice Channels.

SIP Extension-to-SIP Extension call and SIP Extension to SIP Trunk call will consume 2 VoIP channels, while a SIP Extension-to-SLT/ CO / Mobile Trunk will consume a single voice channel only.

If RTP Relay or Direct RTP is selected, maximum 25 Audio calls or 8 Video calls are supported.

SIP Trunks

ETERNITY NENX supports a maximum of 8 SIP Trunks[3].

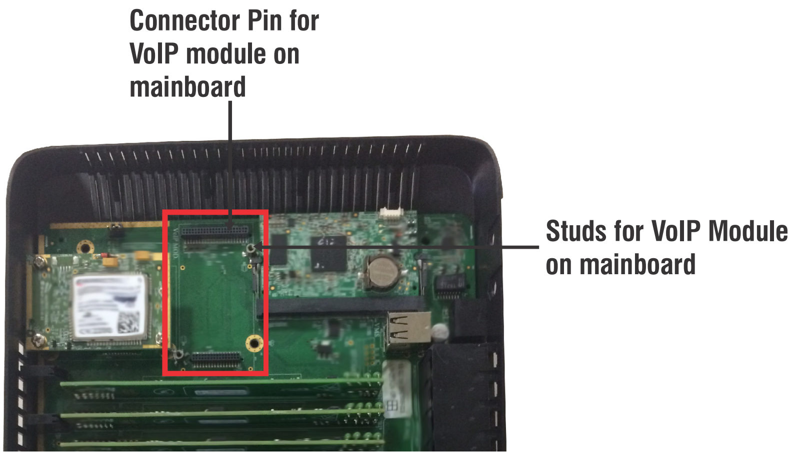
In countries, where the provision and use of Internet telephony services and products is prohibited and/ or subject to laws, regulations or licenses, the User is advised to comply with such laws and regulations when installing and using this product.

Installing the VoIP Module

- Unpack the VoIP module.
- Make sure power supply is turned off before you begin installation.
- Unscrew and remove the top cover of the enclosure. Keep the screws and the cover aside.
- Locate the studs and connector of the VoIP module on the mainboard.

3. *SIP Trunks are same as SIP Accounts. A SIP Account is an account you would get from your VoIP/Internet Telephony Service Provider much like you would get an email account from your Internet Service Provider.*

- Gently seat the VoIP module on the connector on the mainboard. The connector pins on the module must make complete contact with those on the mainboard. Do not apply pressure.



- When the module is seated firmly on the connector on the mainboard, secure the module with the screws on the studs.

- If you have no other modules to install, replace the top cover and secure the cover with the screws.

Before you connect the system to WAN, we recommend that you first connect a computer to the Ethernet Port of ETERNITY NENX, configure the Basic Settings, and then connect to WAN.

- You can connect ETERNITY NENX to WAN either over Ethernet port (Ethernet WAN) or over Mobile Port 1 (Wireless WAN).

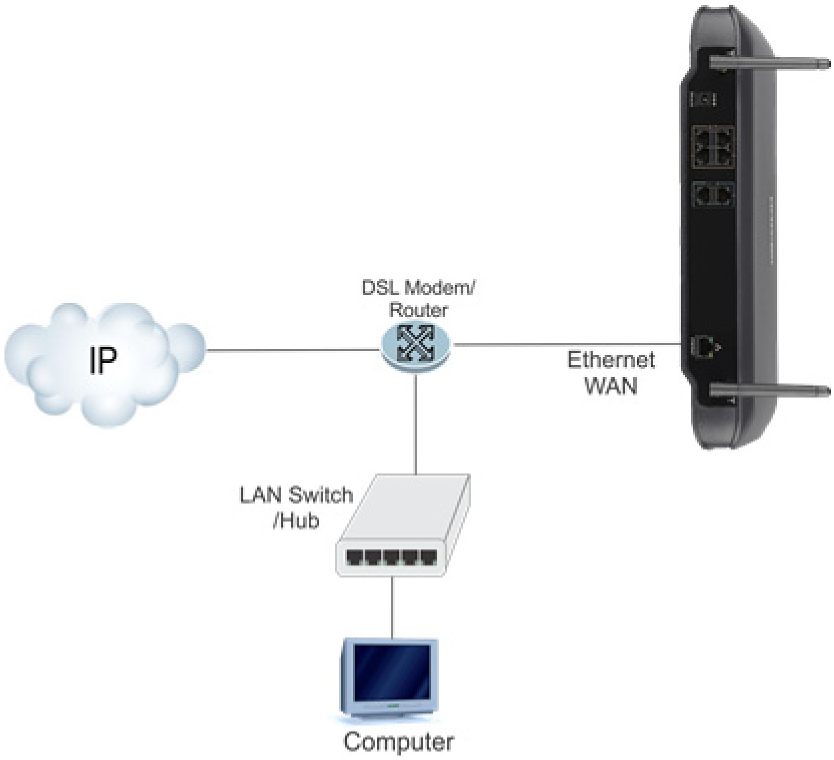
While several installation scenarios are possible, only three most common and most typical scenarios are depicted here.

Ethernet WAN

- Use the RJ45 Ethernet cable supplied for the Ethernet port of ETERNITY NENX to connect the system to the IP network, which may be Public Internet or a LAN.

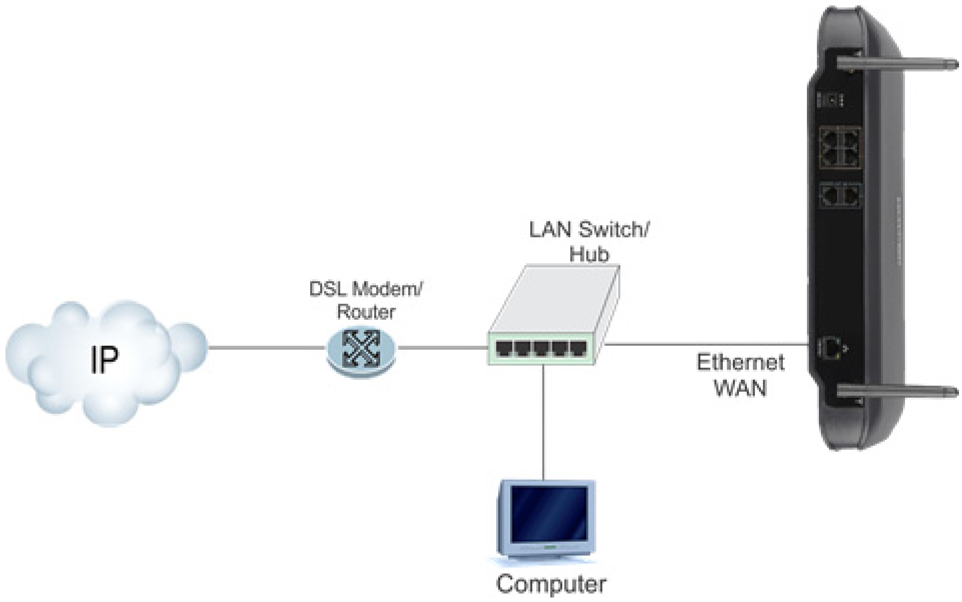
If connecting to the Public IP Network,

- Plug one end of the RJ45 Ethernet cable into the Ethernet Port of ETERNITY NENX and the other end into the Broadband Router/Modem.



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

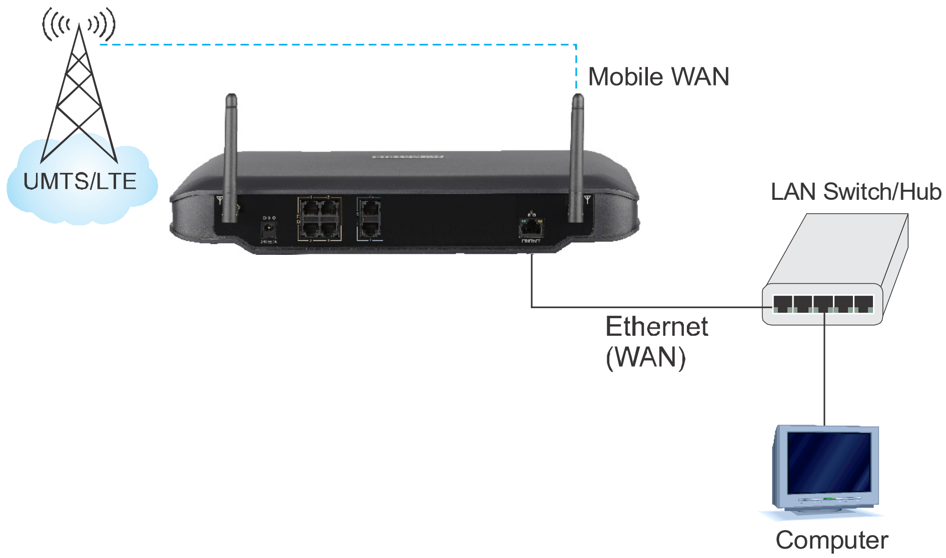
- Plug one end of the RJ45 Ethernet cable into the Ethernet Port of ETERNITY NENX and the other end into the LAN Switch/Hub.



Wireless WAN

- Make sure that:
  - you have installed a 3G UMTS/4G LTE module.
  - a SIM is present in the Mobile port.
  - Internet Services are activated on the SIM.

Wireless WAN in ETERNITY NENX is used only for making SIP Calls and not for configuring the system.



Connecting Single Line Telephones

The Single Line Telephone (SLT) ports provide 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.

A fax machine can also be connected to the SLT Port.

The number of SLT ports supported by the configurations of ETERNITY NENX are:

- **ETERNITY NENXIP50:** 2 SLT Ports
- **ETERNITY NENX312:** 12 SLT Ports
- **ETERNITY NENX416:** 16 SLT Ports

When connecting Single Line Telephones as extensions to your ETERNITY NENX,

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

   You may also connect a Fax machine, if required.

Use SLTs equipped with a 'Flash' key, as several of the features and facilities of the ETERNITY NENX 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. Use standard twisted wire pair the cables of good quality with RJ11 plugs to connect the analog single line telephone instruments to the SLT ports of ETERNITY NENX.

3. Place the SLTs, fax machine at the desired location.

4. Terminate the cables from the SLT ports of ETERNITY NENX on the wall jacks.

5. Connect the SLTs, Fax machine to the wall jacks.

Connecting SIP Extensions

ETERNITY NENX supports up to 50 SIP Extensions. SIP Extension users can make and receive calls to any extension user of the ETERNITY NENX and to external numbers over CO, Mobile and VoIP[4] networks.

ETERNITY NENX supports Video calling on SIP Extensions and SIP Trunks. You can make/receive video calls using SIP Extension/SIP Trunk. An Audio call can be converted to video call and vice versa.

You may register any SIP-enabled device—IP Phone, Soft Phone, Desktop Client, Analog Phone Adapter—as the SIP Extension of the ETERNITY NENX. You may also connect the Standard and Extended IP Phones of Matrix.

You can register the same SIP Extension from three different locations.

You may also connect/register the following as SIP Extensions:

- SPARSH VP248, The Extended IP Phone. For instructions, see “Connecting SPARSH VP248 as Extended SIP Extension”.
- SPARSH VP310, The Extended IP Phone. For instructions, see “Connecting SPARSH VP310 as Extended SIP Extension”.
- SPARSH VP330, Intuitive Touchscreen IP Phone. For instructions, see “Connecting SPARSH VP330 as Extended SIP Extension”.
- SPARSH VP510, Premium IP Phone. For instructions, see “Connecting SPARSH VP510 as Extended SIP Extension”.
- Extended SPARSH VP710, the Smart Video IP Phone. For instruction,“Connecting Extended SPARSH VP710 as Extended SIP Extension”.
- SPARSH VP210, Entry Level IP Phone. For instructions, see “Connecting SPARSH VP210 as Extended SIP Extension”

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.

To know the list of features supported, refer to “SARVAM UCS Features supported in Terminals”.

If you register the Extended IP Phone outside the Region/Country selected for ETERNITY NENX, 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 ETERNITY NENX. Also, Access Codes and Emergency Numbers will work according to the Region/Country selected for ETERNITY NENX.

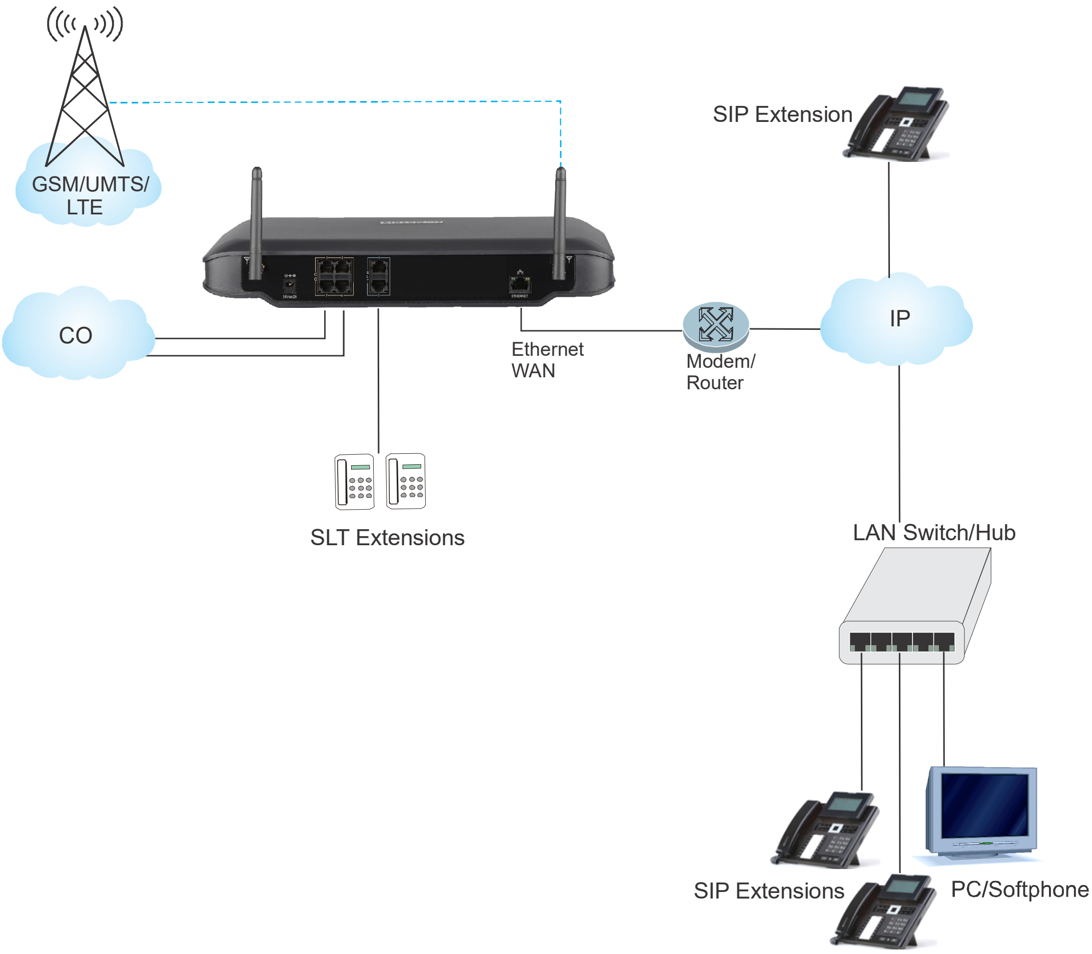
The SIP Extensions may be registered either over ***Ethernet WAN*** or over ***Wireless WAN (Mobile 1)***, according to your preference and your IP network installation scenario.

4. *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. Read “Logical Partition”.*

Ethernet WAN

If ETERNITY NENX is connected to a ***Public Network***,

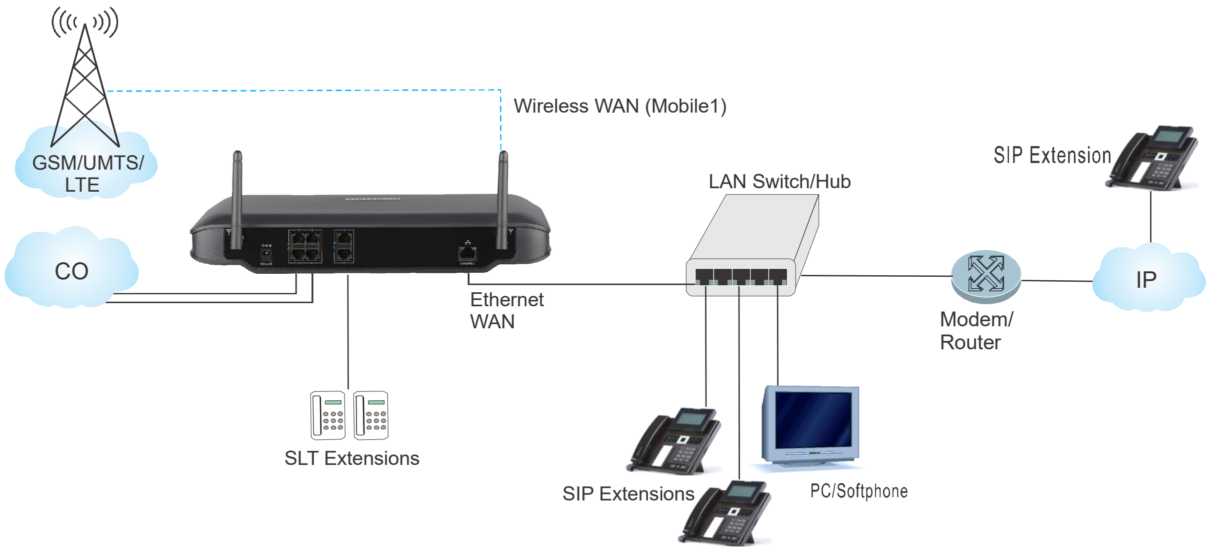
- Connect the Extended IP Phone, or any Standard IP SIP device to the LAN Switch.
- Register any SIP device—Extended IP phone/ Soft clients or Standard IP phone—on the public network as SIP Extension.



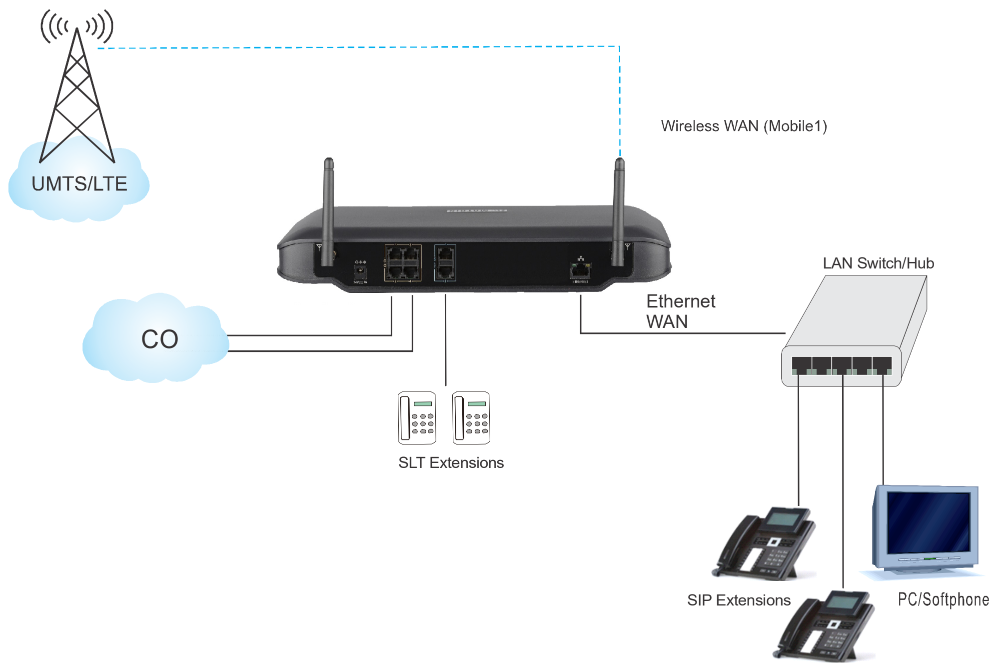
When you register any SIP device, other than the Extended IP Phone, on the public network as SIP Extension, you must configure the Registrar Server Address of ETERNITY NENX, the Registrar Server Port, the SIP ID, Authentication ID and Password in the SIP device.

If ETERNITY NENX is connected to a ***Private Network (Behind the NAT)***,

- Connect the Extended IP Phone, or any standard IP Phone to the LAN Switch.
- You may also register any SIP device—Extended IP Phone or Standard SIP phone—on the public network as SIP Extension. In this case, you must configure Port Forwarding for SIP and RTP on the Router.



Wireless WAN



- Connect the Extended IP Phone, or any standard IP Phone to the LAN Switch.
- You may also register any SIP device—Extended IP phone/ Soft clients or Standard IP phone—on the public network as SIP Extension.
- Wireless WAN in ETERNITY NENX is used only for making SIP Calls and not for configuring the system.

Connecting SPARSH VP248 as Extended SIP Extension

You are recommended to complete the following steps before connecting the Extended IP Phone to the ETERNITY NENX:

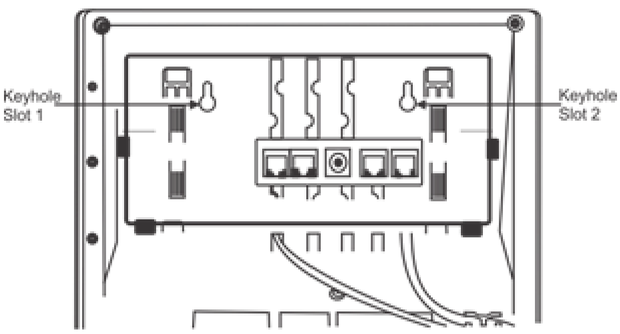
- Decide the location of the Extended IP Phone—within the same network or outside—according to your installation scenario.
- Log into Jeeves. See "System Configuration using the Web-based GUI".
- Configure **DHCP Server** on the "Network Parameters" page under *Basic Settings*.

*If you want to use the **DHCP Server** on your LAN for assigning IP Address to the Extended IP Phone, select **DHCP option 224** and **Data Type** as '**String**' and program the Ethernet Port IP Address /Domain Name and SPARSH Port in the format **"Ethernet IP Address: SPARSH Port"** in your LAN DHCP Server.*

- Assign an extension number to the Extended IP Phone on the "Extension and Feature Codes" page under *Basic Settings*.
- You must configure the necessary parameters in ETERNITY so that SPARSH VP248 can register as a SIP Extension. For instructions, see "SIP Extensions" under *Basic Settings*.

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.

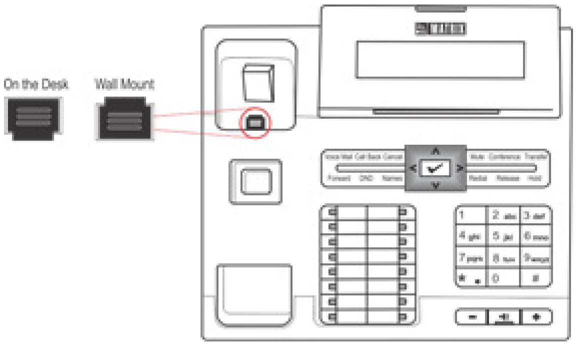
- Unpack the SPARSH VP248 box and verify package contents[5].
- Mount the phone on a desk or wall at a location convenient to you.
    - To mount the phone on the wall,
        - Use the mounting template to drill holes of appropriate size and distance. Fix the screw grips in the holes you drilled.
        - Fix two screws in the holes on the wall. Ensure 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.

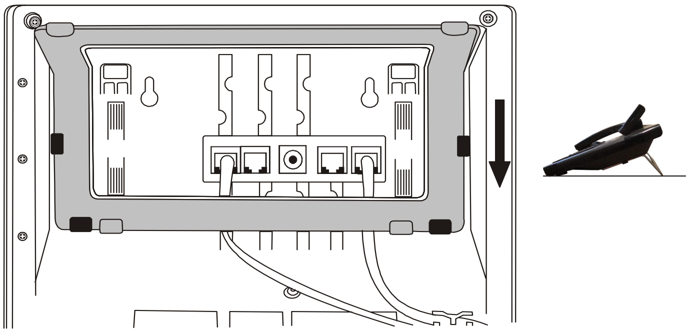
5. See "Packing List" in the Appendix topic.

- 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.

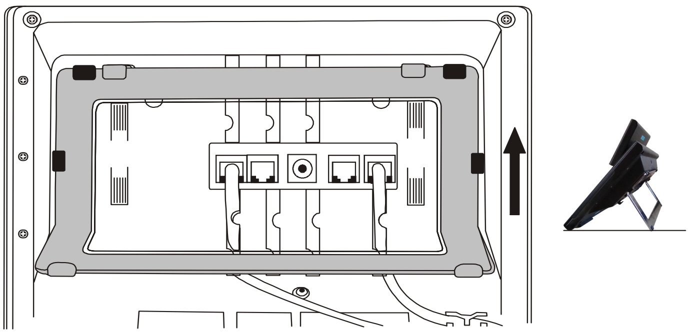


- When you mount the phone on a desk,
  - You can attach the Foot Stand in two ways as illustrated below.

Foot Stand attached at 30° Angle



Foot Stand attached at 50° Angle

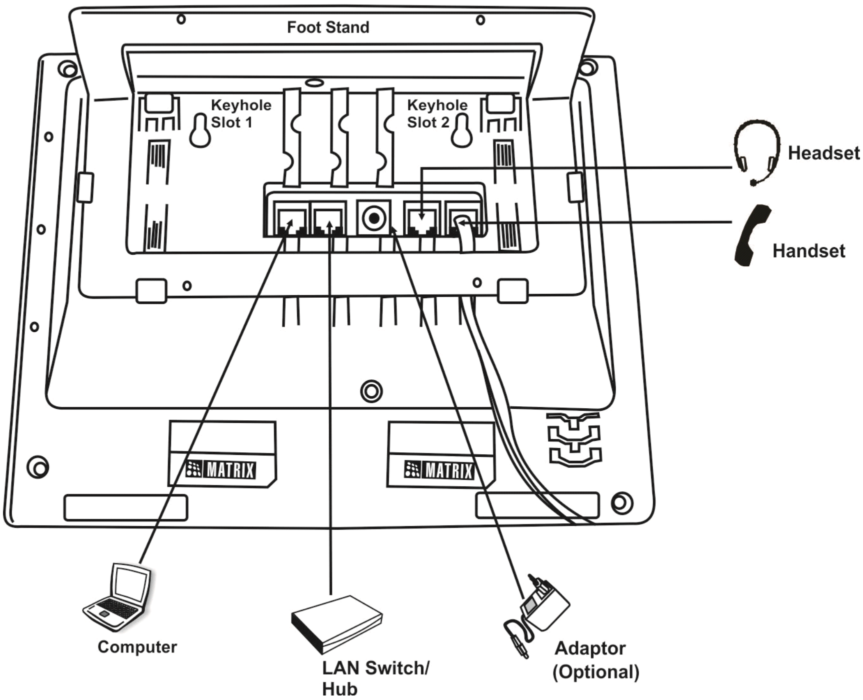


If you attach the Foot Stand at 50°, the phone will be placed in an almost upright position on your desk.

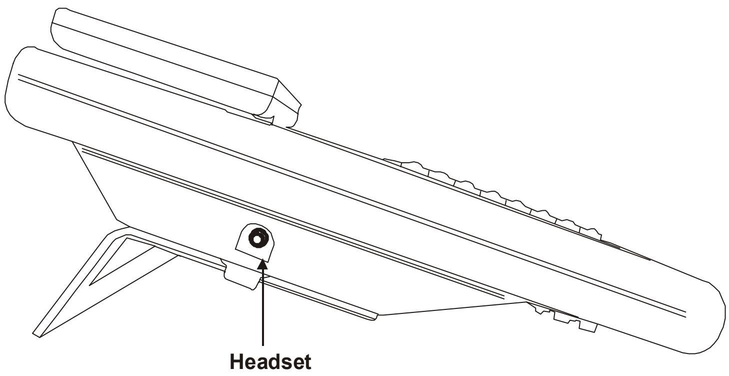
  - Decide which of these positions would work for you best and accordingly attach the Foot Stand.

- 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.



- If you want to use a Headset (not supplied) with your phone, you may 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, as illustrated in the figure below.



OR

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 .

- Connect the LAN Port of SPARSH VP248 to the LAN Switch/Hub or a Router, according to your installation scenario.

- 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 with LAN Port of the computer.

- 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.*

- Switch ON power supply.

  When you power the phone, the boot process will be initiated in the following sequence.

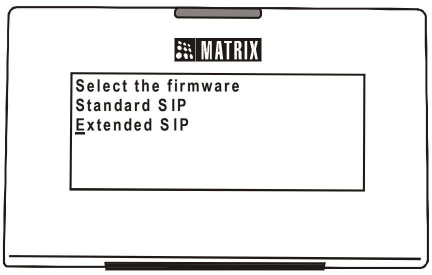
  - All keys with LED, including the Speaker key and the Ringer LED will glow.

  - The LCD display will light up and booting message appears.

  - As soon as the 'Loading...' message appears on the phone display, press **#** key.

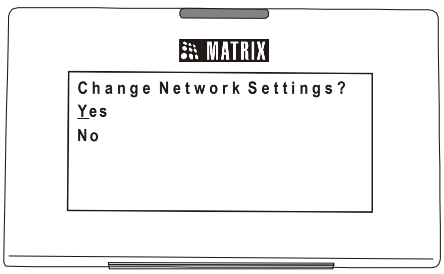
  - Select the firmware **Extended - IP Phone**. Move the cursor by pressing the DOWN navigation key **V**.

  - 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, 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, the phone initiates Auto Configuration to download the configuration files from ETERNITY NENX.

- 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 ETERNITY NENX.

- 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.

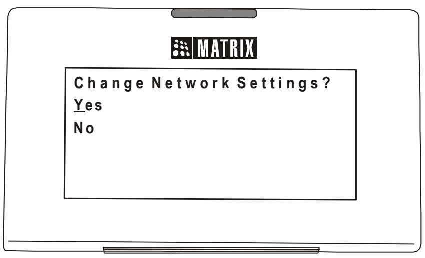
- The Enter key ✓ to make a selection or to complete an action.
- The Up key ∧ to move up the Menu.
- The Down key ∨ move down the Menu.
- The Forward key > move the cursor one character.
- The Back key < to move the cursor one character and to return from the submenu to the main menu.

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:

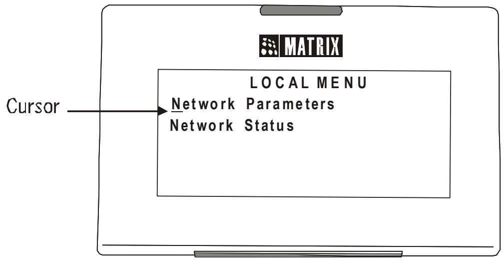
- During start-up, when the phone prompts you to change the network settings after loading the firmware, press the Enter Key to select Yes and access network settings.



- When the phone is making Network discovery, downloading configuration files, attempting registration, press the Enter Key ✓ to access network settings.

- When the phone is in idle state. You must press the DSS key assigned to 'Local Menu'.

- 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.

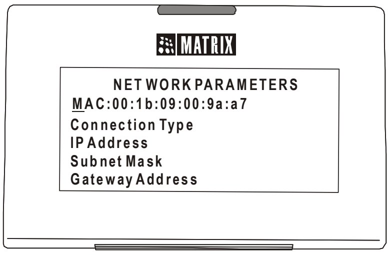


- You can configure Network Parameters and view Network status from the Local Menu.

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 and 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 the 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

- ETERNITY works as the Auto Configuration Server for the phone. Enter the IP Address or the Dynamic DNS Domain Name of the Ethernet Port of ETERNITY 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 the Ethernet Port of ETERNITY 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[6].

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.

6. 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 **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*.

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:

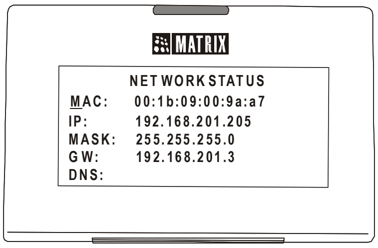
- Select **Enable?**.
- Enter the 802.1x Authentication **Identity** provided by your network administrator.
- Enter the 802.1x Authentication **MD5 Password** associated with the identity provided by your network administrator.

When you change the Network Settings, the phone will restart.

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 IP Address or Dynamic DNS Domain Name of the Ethernet Port of ETERNITY.
- **S. PORT:** The SPARSH Port of the Ethernet Port of ETERNITY.
- **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 Extended IP Phone to the ETERNITY NENX:

- Decide the location of the Extended IP Phone—within the same network or outside—according to your installation scenario.
- Log into the web-browser Jeeves. Read the topic “System Configuration using the Web-based GUI”.
- Configure **DHCP Server** on the “Network Parameters” page under *Basic Settings* of Jeeves.

*If you want to use the **DHCP Server** on your LAN for assigning IP Address to the Extended IP Phone, select **DHCP option 224** and **Data Type** as ‘**String**’ and program the Ethernet Port IP Address /Domain Name and SPARSH Port in the format **“Ethernet IP Address: SPARSH Port”** in your LAN DHCP Server.*

- Assign an extension number to the Extended IP Phone on the "Extension and Feature Codes" page under *Basic Settings* of Jeeves.

- You must configure the necessary parameters in ETERNITY so that SPARSH VP310 can register as a SIP Extension. For instructions, see "SIP Extensions" under *Basic Settings*.

Now, follow the steps described below to install the SPARSH VP310.

- Unpack the SPARSH VP310 box and verify package contents[7].

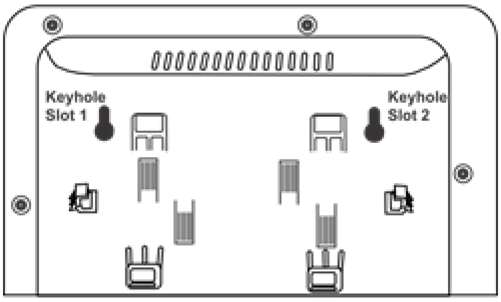
- You can mount the phone on a wall or on the desk.

- To mount the phone on a wall,

  - Use the mounting template to drill holes of appropriate size and distance.

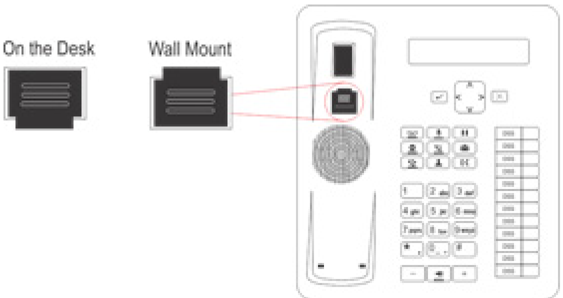
  - Fix the screw grips in the holes you drilled.

  - Fix two screws in the holes on the wall, ensuring that they are aligned with the Keyhole Slots 1 and 2 of 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.

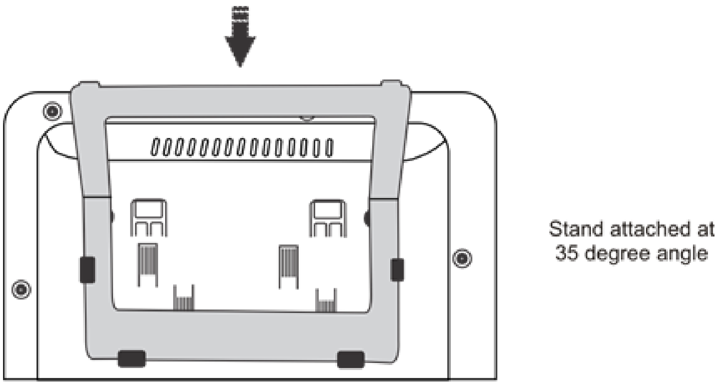
  - Reverse the handset wall mount tab to make sure that the handset remains intact when you mount the phone. Push the handset wall mount tab upwards to remove it from the slot. Rotate it 180 degrees clockwise and push it downwards into the slot.



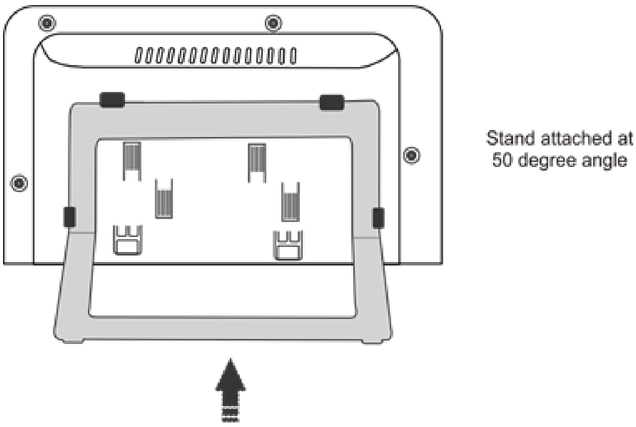
- When you mount the phone on a desk, you can attach the Foot Stand in two ways as illustrated in the following.

7. *See "Packing List" in the Appendix topic.*

Foot Stand attached at 35° Angle

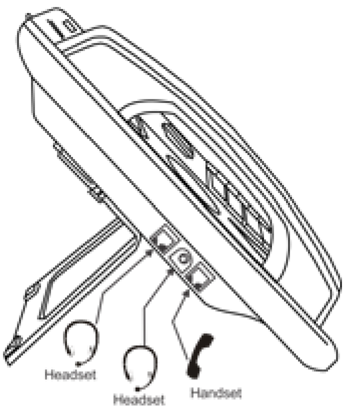


Foot Stand attached at 50° Angle



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.



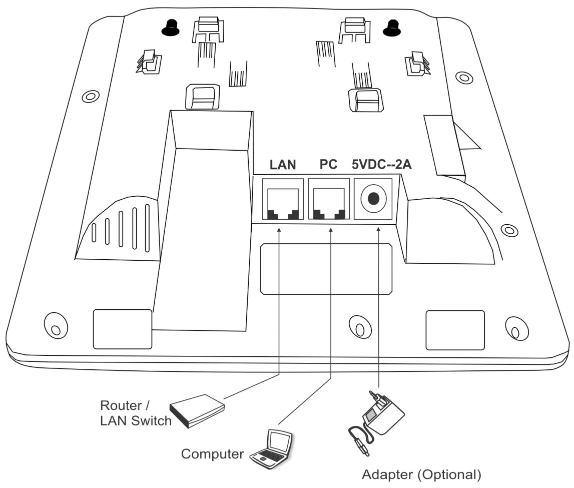
- Connect the Handset to the Phone body.
  - Plug the long straightened end of the phone cord into the handset jack on the left side panel of the phone marked with the handset symbol  .

- Plug the other (short straight) end of the phone cord into the jack at the bottom of the handset.

- 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 .



- Connect the LAN Port of SPARSH VP310 to the LAN Switch/Hub or a Modem, according to your installation scenario.

- 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 with LAN Port of the computer.

- 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.*

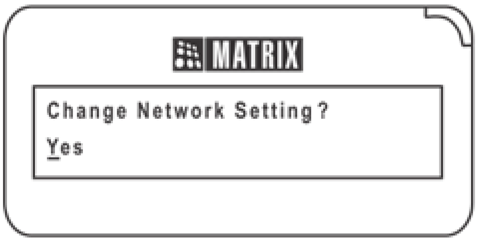
- Switch ON power supply.

  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, 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 ETERNITY NENX.

- 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 ETERNITY NENX.
- 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.

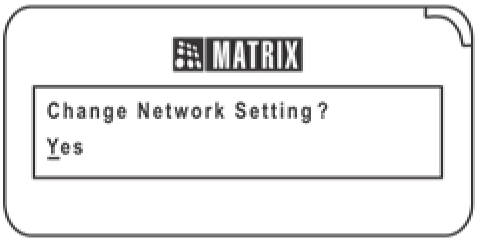
- The Enter key ✓ to make a selection or to complete an action.
- The Up key ∧ to move up the Menu.
- The Down key ∨ move down the Menu.
- The Forward key > move the cursor one character.
- The Back key < to move the cursor one character and to return from the submenu to the main menu.
- The Cancel key ✕ 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:

- 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.

- When the phone is making Network discovery, downloading configuration files, attempting registration.

  You must press the Down key V to access network settings.

- When the phone is in idle state. You must press the Down key V to access the Network Settings.

Configuring Network Parameters

- When the phone is in idle state. You must press the Down key V 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 and 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 the 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

- ETERNITY works as the Auto Configuration Server for the phone. Enter the IP Address or the Dynamic DNS Domain Name of the Ethernet Port of ETERNITY 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 the Ethernet Port of ETERNITY 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 fall back 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[8].

  The meaning of CoS bits with respect to traffic type is as follows:

CoS	Traffic Type
0	Best Effort
1	Background

8. *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 VP310 Extended IP Phone”, under *PCAP Trace*.

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 your network administrator.
- Enter the 802.1x Authentication **MD5 Password** associated with the 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 the 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 Ethernet Port IP Address / Domain Name of ETERNITY or the Secondary Server IP Address (if configured) or any Fallback Server IP Address.
  - **S. PORT:** This displays the port of the Active Server. It may be the SPARSH Port of ETERNITY 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 VP510 as Extended SIP Extension

You are recommended to complete the following steps before connecting the Matrix Extended IP Phone to ETERNITY NENX:

- Decide the location of the Extended IP Phone—within the same network or outside—according to your installation scenario.
- Log into the web-browser Jeeves. Read the topic "System Configuration using the Web-based GUI".
- Configure **DHCP Server** on the "Network Parameters" page under *Basic Settings* of Jeeves.

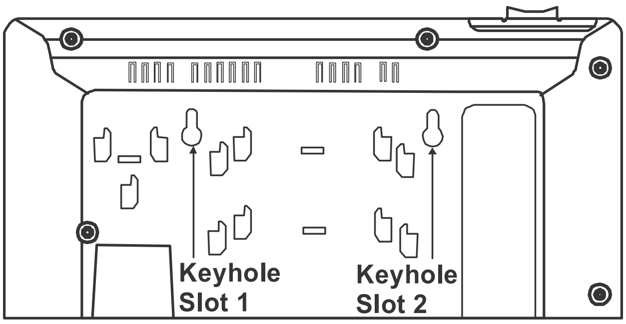
*If you want to use the **DHCP Server** on your LAN for assigning IP Address to the Extended IP Phone, select **DHCP option 224** and **Data Type** as '**String**' and program the Ethernet Port IP Address /Domain Name and SPARSH Port in the format **"Ethernet IP Address: SPARSH Port"** in your LAN DHCP Server.*

- Assign an extension number to the Extended IP Phone on the "Extension and Feature Codes" page under *Basic Settings* of Jeeves.
- You must configure the necessary parameters in ETERNITY so that SPARSH VP510 can register as a SIP Extension. For instructions, see "SIP Extensions" under *Basic Settings*.

You can also connect DSS532, the DSS Console with SPARSH VP510. To connect, follow the instructions given in "Installing DSS532 with SPARSH VP510".

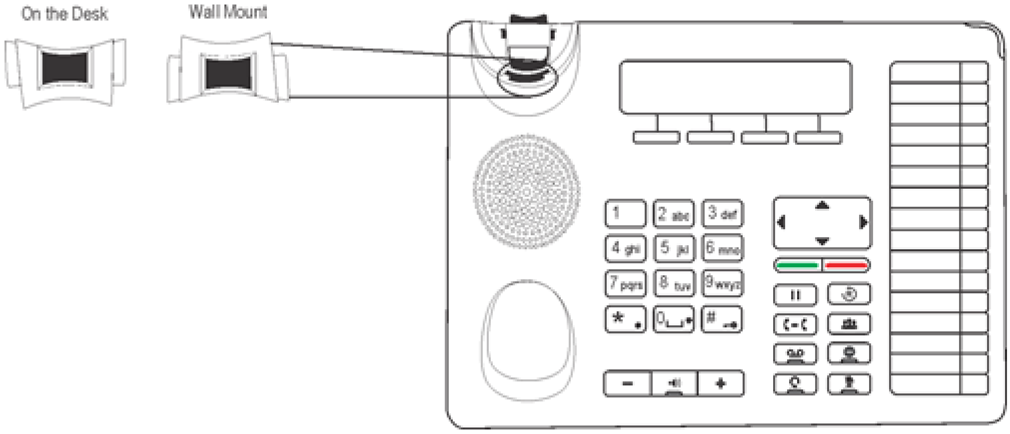
Now, follow the steps described below to install the Extended IP Phone.

- Unpack the SPARSH VP510 box and verify package contents[9].
- You can mount the phone on a wall or on the desk.
- 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.



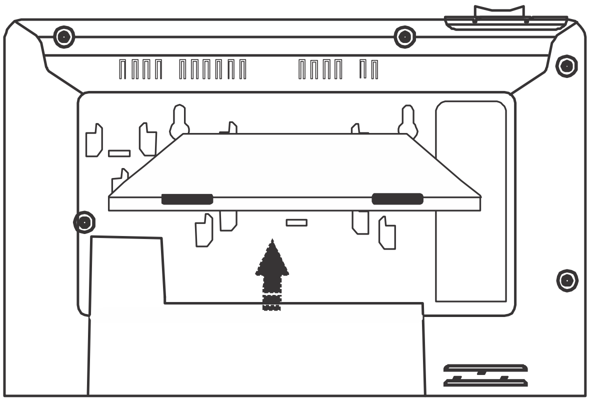
9. See "Packing List" in the Appendix topic.

- 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.

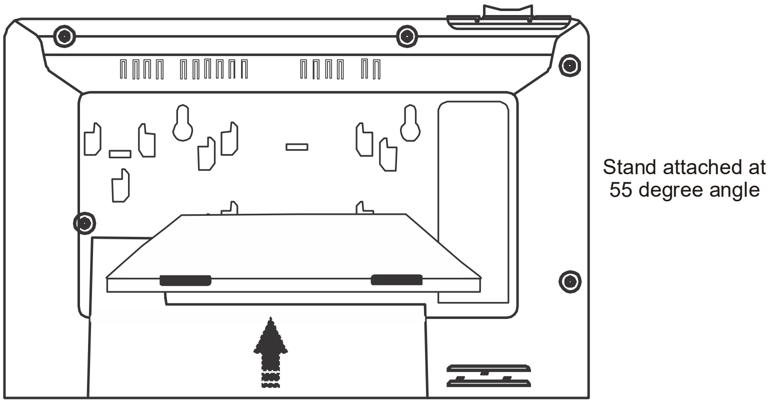


If you are unable to remove the wall mount tab, you may use a tool like a minus screw-driver to remove it.

- 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

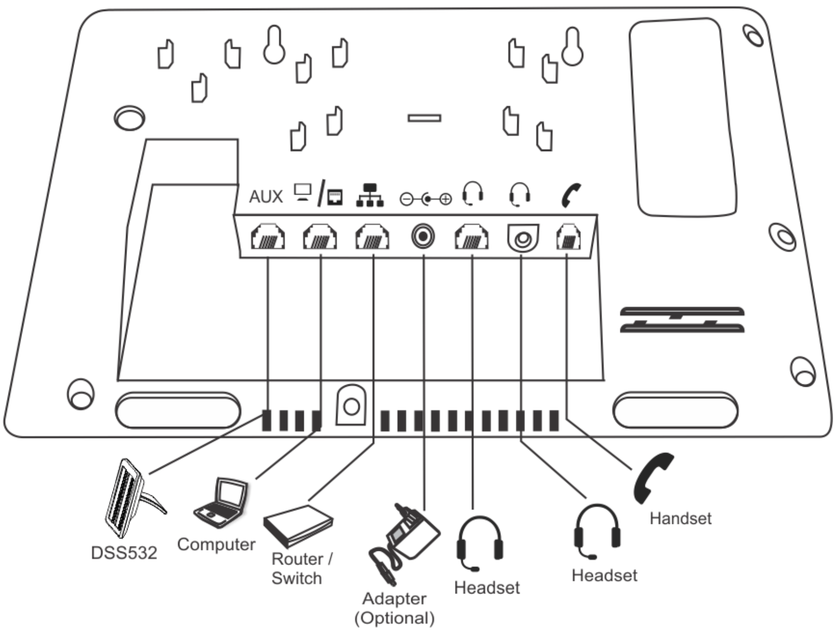


Stand attached at
45 degree angle



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

Refer to the diagram below for connectivity.



- 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.
- 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 .

- 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.
- 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.
- 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".
- 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.*
- 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 ETERNITY NENX.

- 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 ETERNITY NENX.

- 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 V 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:

- 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.

- When the phone is making Network discovery, downloading configuration files, attempting registration.

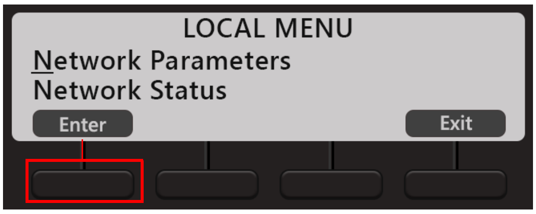
You must press the **Menu** key to access network settings.

- When the phone is in idle state, press the Down key ∨ .

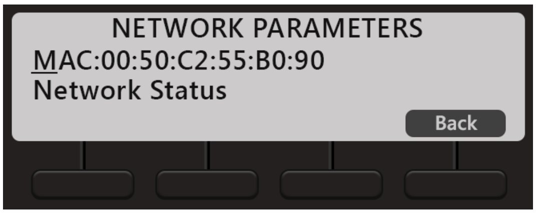
You can configure Network Parameters and view Network status from the Local Menu.

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

- ETERNITY works as the Auto Configuration Server for the phone. Enter the IP Address or the Dynamic DNS Domain Name of the Ethernet Port of ETERNITY 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 the Ethernet Port of ETERNITY 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 fall back 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[10].

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.

10. 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.

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.

- To start PCAP, when the cursor is placed on PCAP, press **Start** Key.
- To stop PCAP, press the **Stop** Key.

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 **V** to access the Network Settings.
- Again press Down key **V** 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.

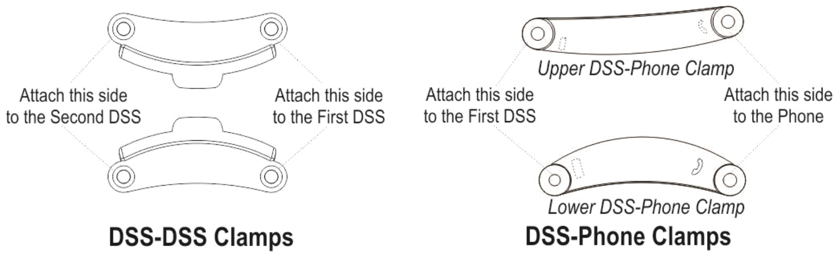
- **S. ADD:** This displays the IP address of the Active Server. It may be the Ethernet Port IP Address / Domain Name of ETERNITY or the Secondary Server IP Address (if configured) or any Fallback Server IP Address.
- **S. PORT:** This displays the port of the Active Server. It may be the SPARSH Port of ETERNITY 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 ETERNITY, you can install the DSS532 by following the steps given below:

- Unpack the box and verify the package contents[11].
- Four clamps are provided with the phone — 2 DSS-Phone Clamps and 2 DSS-DSS Clamp.

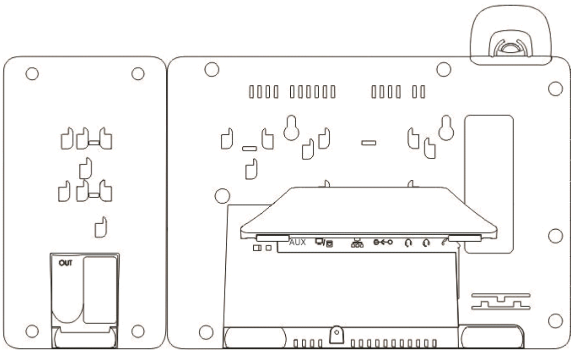


11. *See "Packing List" in the Appendix topic.*

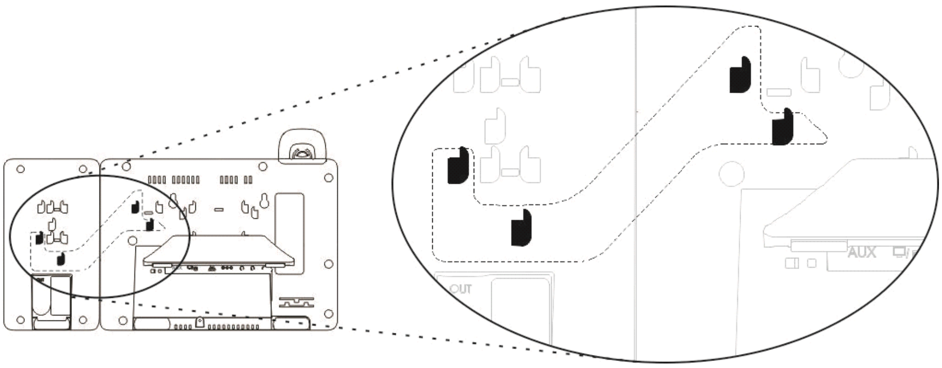
Connecting the First DSS532

Connecting the Extender

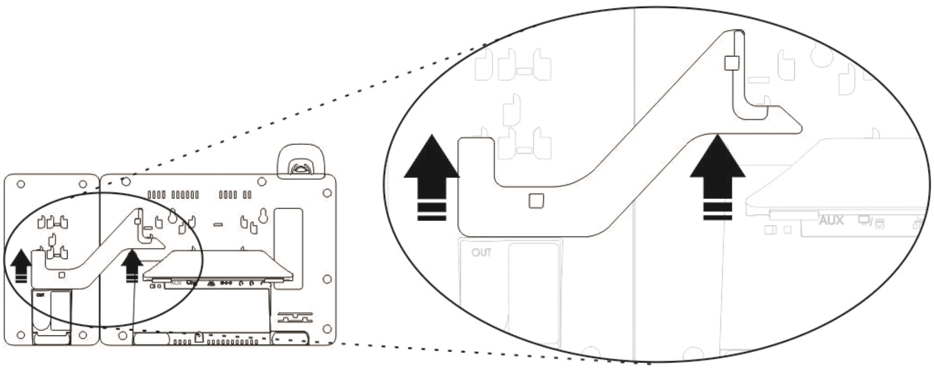
- Turn the phone upside down on the table and place the inverted DSS532 adjacent to it.



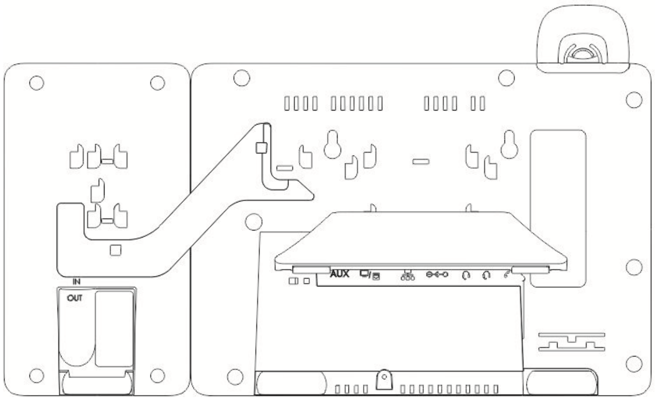
- To attach the DSS532 with the phone, place the DSS Extender as illustrated below.



- Insert the hooks on the Extender into the slots provided on the phone and the DSS532.

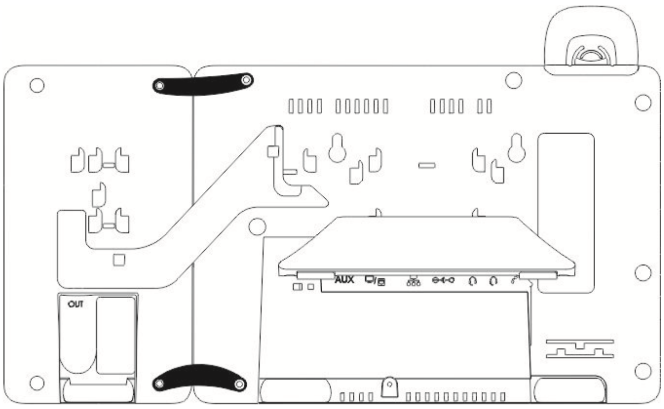


- Firmly slide the DSS Extender upwards to lock them in place.



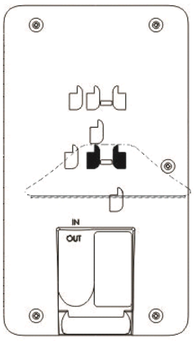
Attaching the Clamps

- 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.

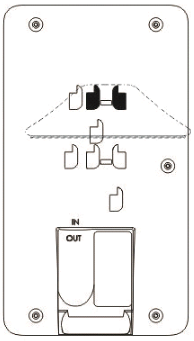


Attaching the Foot Stand

- 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

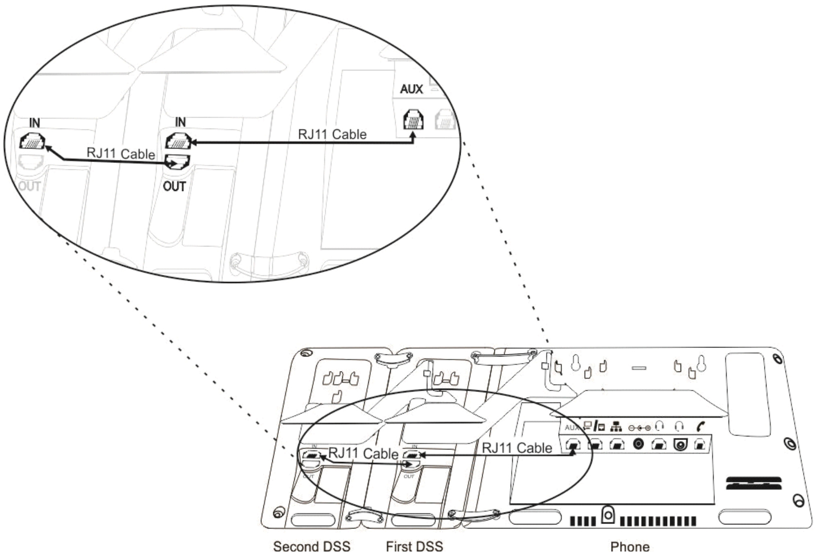


Stand attached at 55 degree angle

Make sure both, the DSS532 and phone are mounted at the same angle.

Connecting the Cables

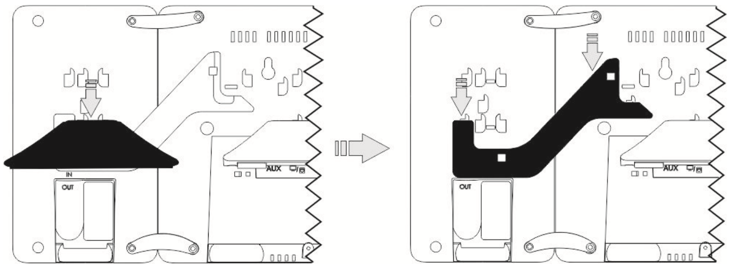
- To connect the DSS532 with phone, plug one end of RJ11 Cable into **Auxiliary (AUX) Port** of the phone and the other end into the **IN Port** of the DSS532.



Connecting Multiple DSS532

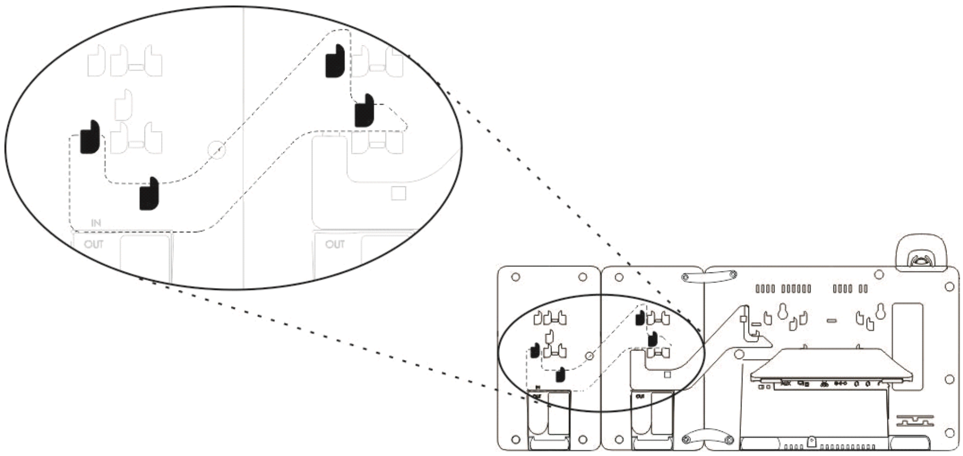
Remove the Foot Stand

- Remove the Foot Stand of attached DSS532. To do,
  - Firmly slide the Foot Stand of the attached DSS532 downward to unlock.
  - Now, slide down the attached DSS Extender in downward direction.

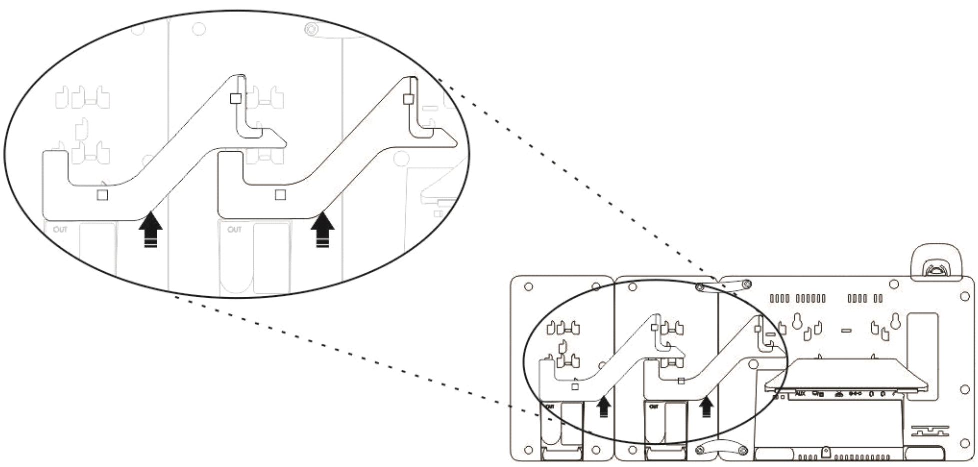


Attach the second DSS Extender

- 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.

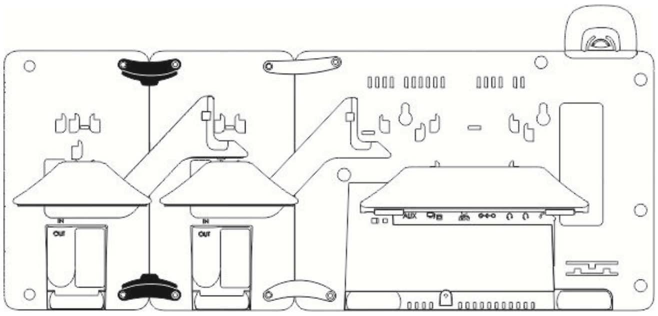


- Firmly slide both the DSS Extenders upward consecutively (attach the second extender first followed by the existing one attached to the phone) and lock them in place.



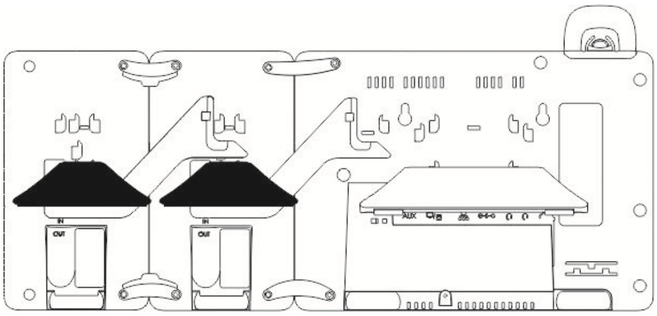
Attach the Clamps

- Attach the DSS-DSS Clamps between both the DSS532.



Attach the Foot Stand

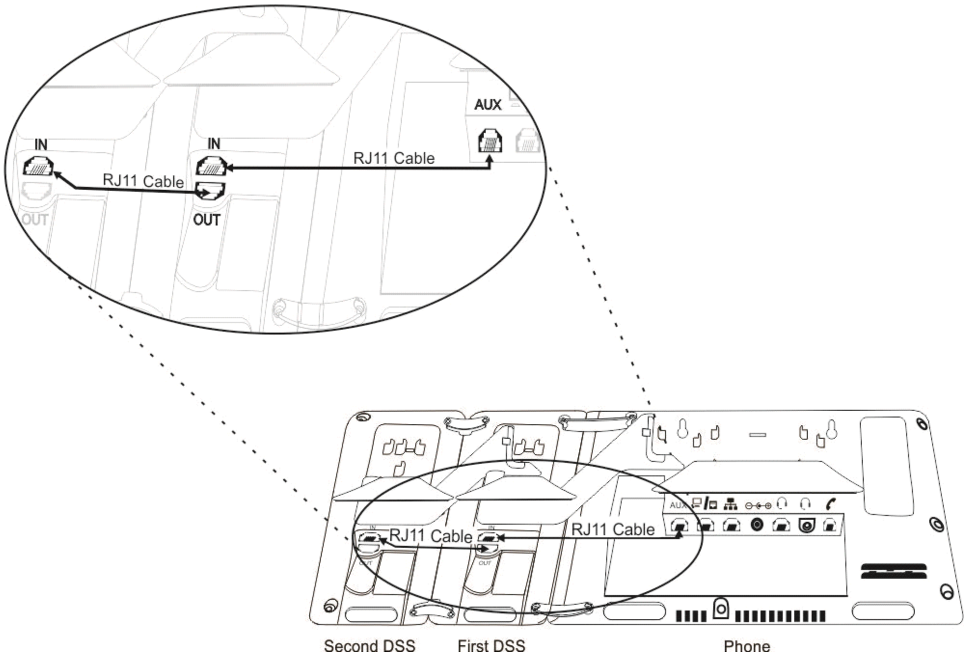
- Attach the Foot Stand of both the DSS532.



Make sure both, the DSS532 and the phone are mounted at the same angle.

Connect the second DSS532 to the existing assembly

- 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.

- After you have connected the DSS532 with the phone, you can configure the DSS Keys. For instructions, see "DSS Key Settings" under "Configuring Matrix SPARSH VP510".

  For detailed instructions to connect multiple DSS532, refer ***DSS532 Quick Installation Guide***.

Connecting SPARSH VP330 as Extended SIP Extension

You are recommended to complete the following steps before connecting the Matrix SPARSH VP330 to ETERNITY NENX:

- Decide the location where you want to place SPARSH VP330 within your LAN.
- By Default, in SPARSH VP330, the Connection Type selected is DHCP. See "Network Parameters" under *Basic Settings*.

*If you want to use the **DHCP Server** on your LAN for assigning IP Address to the Extended IP Phone, select **DHCP option 224** and **Data Type** as '**String**' and program the Ethernet Port IP Address /Domain Name and SPARSH Port in the format **"Ethernet IP Address: SPARSH Port"** in your LAN DHCP Server.*

- Log in to *Jeeves*. For instructions, see "System Configuration using the Web-based GUI".
- You must configure the necessary parameters in ETERNITY so that SPARSH VP330 can register as a SIP Extension. For instructions, see "SIP Extensions".

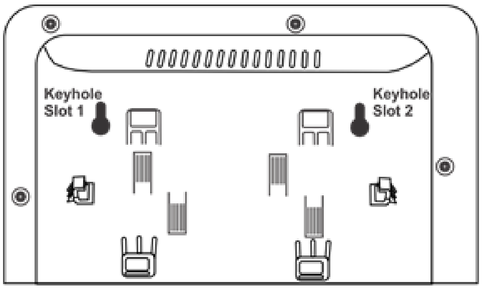
Now, follow the steps described below to install SPARSH VP330.

- Unpack the SPARSH VP330 box and verify package contents[12].

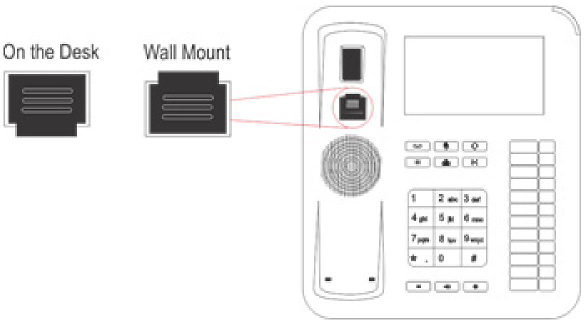
---

12. See "Packing List" in the Appendix topic.

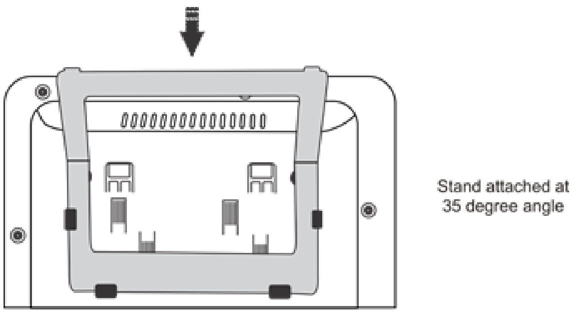
- Mount the phone on a desk or wall at a location convenient to you.
    - When mounting the phone on the wall,
        - Use the mounting template to drill holes of appropriate size and distance. Fix the screw grips in the holes you drilled.
        - Fix two screws in the holes on the wall, such that 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.

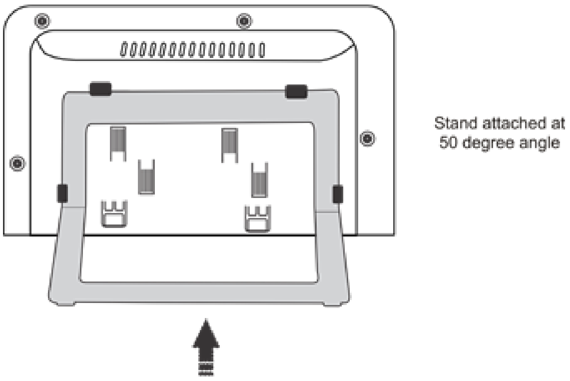


- Now, mount the phone on the wall, with the screws fitting into the Keyhole slots.
- Reverse the handset wall mount tab to make sure that the handset remains intact when you mount the phone. Push the handset wall mount tab upwards to remove it from the slot. Rotate it 180 degrees clockwise and push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

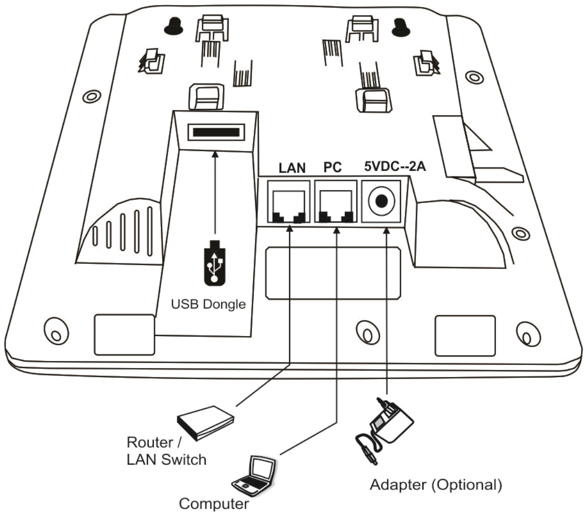


- When you mount the phone on a desk, you can attach the Foot Stand in two ways at **35° Angle** or at **50° Angle**.





- 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.



- 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.
- 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  .

- Connect the LAN Port of SPARSH VP330 to the IP Network — 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.

- 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.

- 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.*

- 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.
  - The loading message appears on the phone display, while loading the application.
  - 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 WiFi for connectivity, press Settings  .

Refer to the SPARSH VP330 User Guide,

- *To change the Network Settings of the phone and configure the network parameters.*
- *To use WiFi 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 ETERNITY NENX.

- On successful download of all configuration files, the phone attempts to register with ETERNITY NENX.

- On successful registration, the Home screen appears.

The phone will register successfully, only if the SIP Extension parameters in ETERNITY have been correctly configured as per your installation scenario.

Refer to the ***SPARSH VP330 User Guide*** to know more.

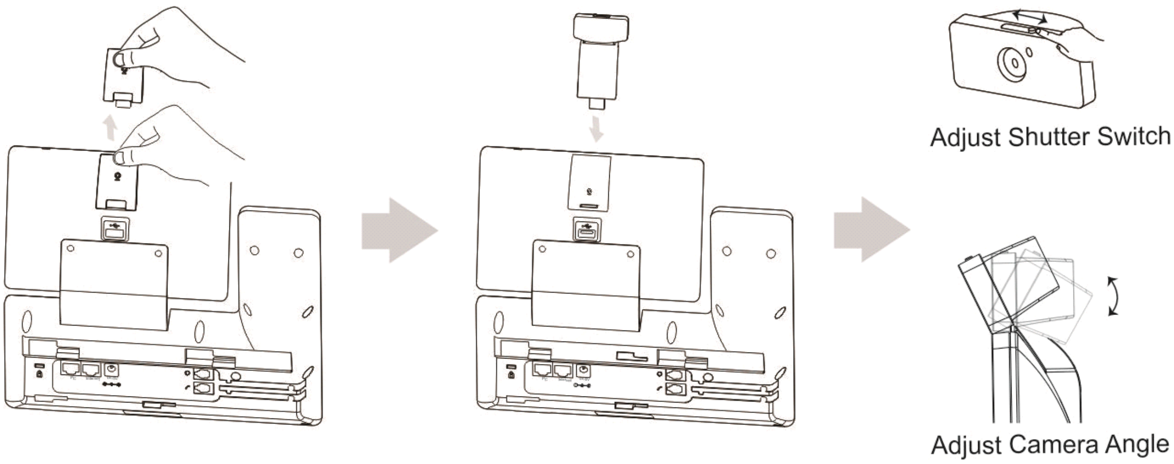
Connecting Extended SPARSH VP710 as Extended SIP Extension

You are recommended to complete the following steps before connecting the Matrix Extended SPARSH VP710 to ETERNITY NENX:

- 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 ETERNITY NENX so that Extended SPARSH VP710 can register as a SIP Extension. For instructions, see "Configuring Matrix Extended SPARSH VP710".

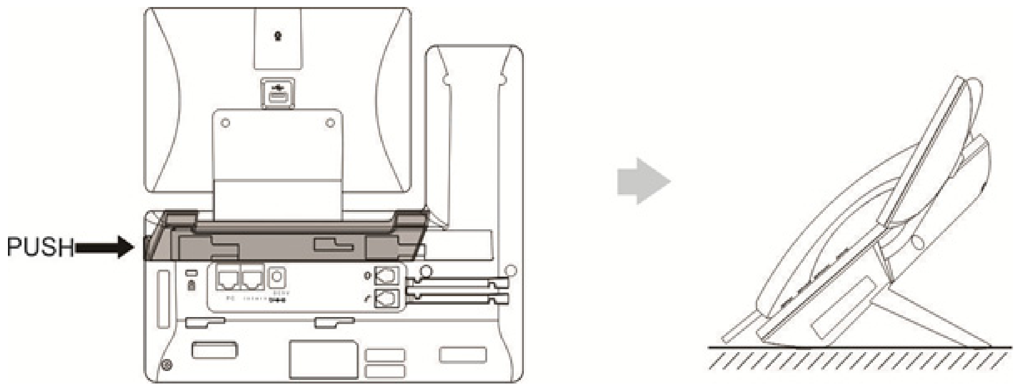
Now, follow the steps described below to install Extended SPARSH VP710.

1. **Inserting the camera**

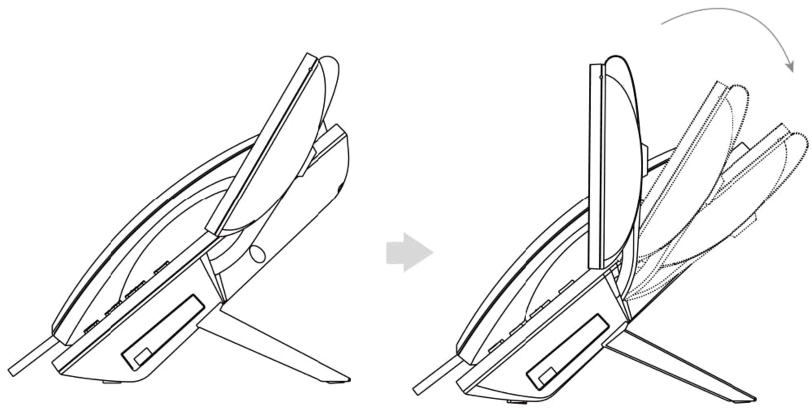


It is recommended to use only the Matrix original Camera, supplied with the IP Phone for video calling. The use of any third-party camera may cause damage to the phone. Damages to the phone caused by using third-party camera is not covered by Matrix warranty.

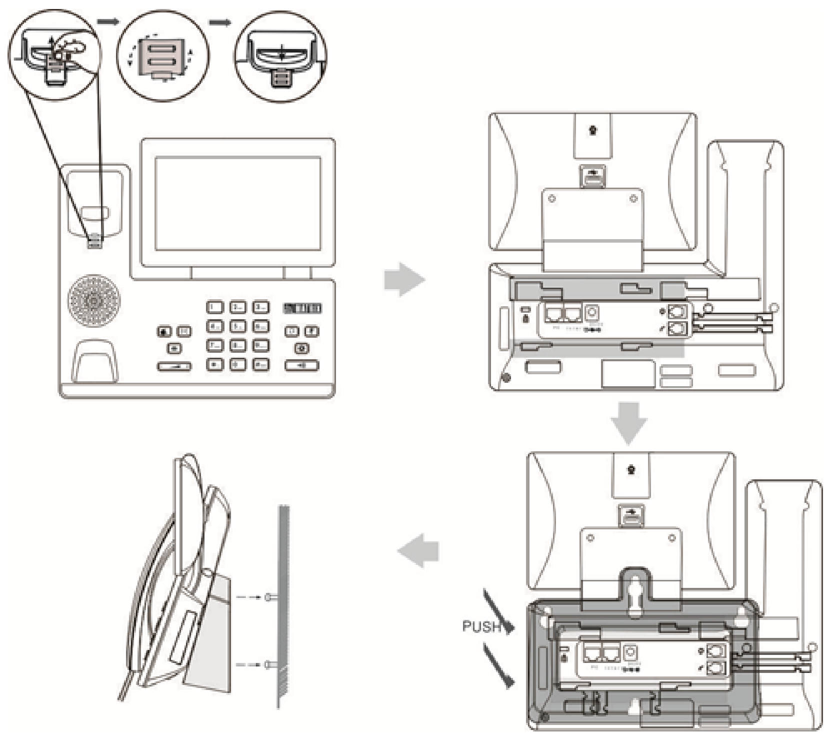
2. **Attaching the stand**



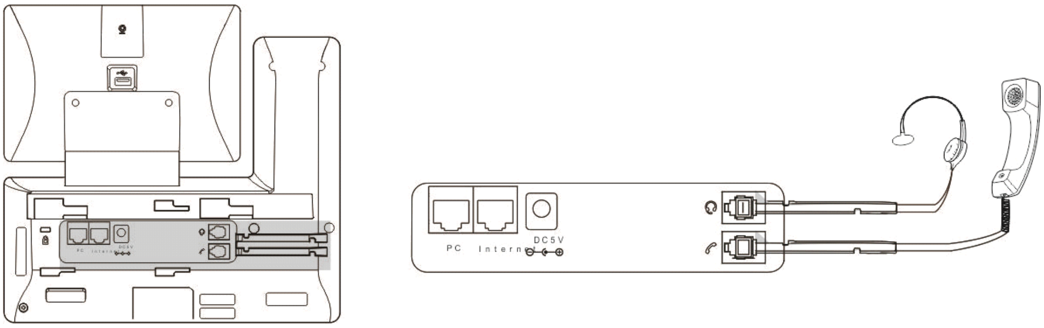
3. **Adjusting the angle of the touch screen.**



4. **Attaching the optional wall mounting bracket**

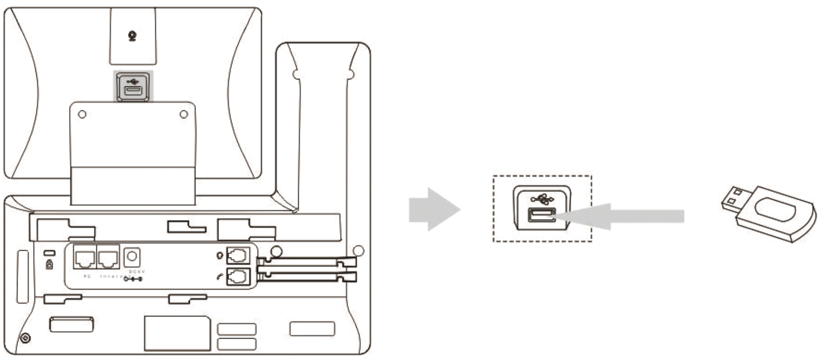


5. **Connect the handset and optional headset.**



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.**

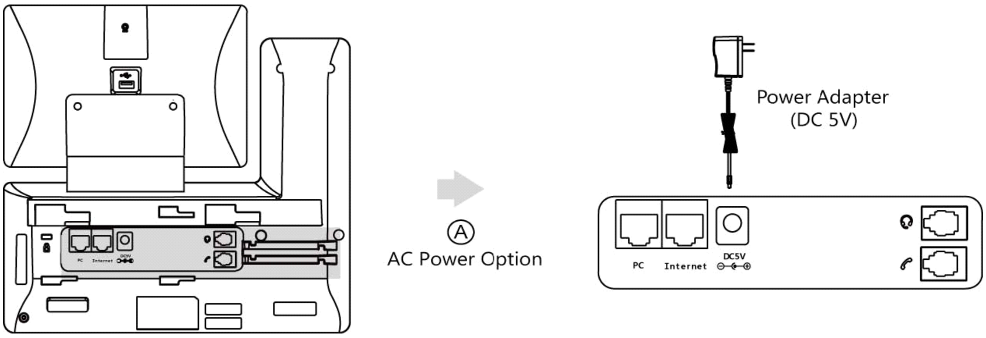
There are two options to connect the power and the network.

- 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.

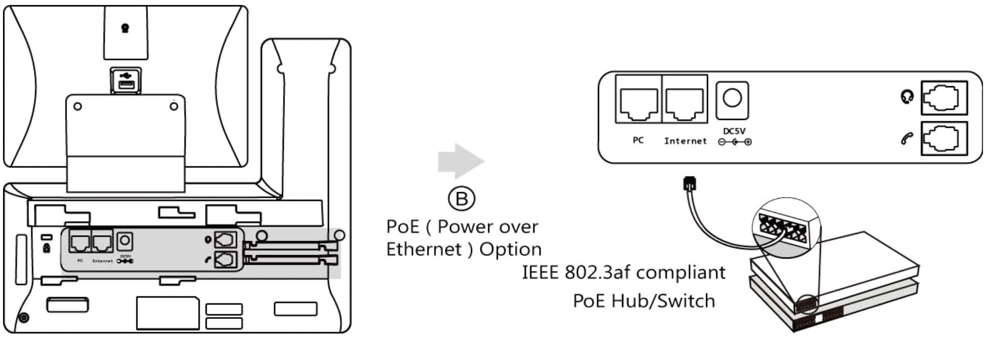


Power over Ethernet (PoE)

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 ETERNITY NENX.
- On successful download of all configuration files, the phone attempts to register with ETERNITY NENX.
- On successful registration, the Home screen appears.

The phone will register successfully, only if the SIP Extension parameters in ETERNITY NENX 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 Extended IP Phone to the ETERNITY NENX:

- Decide the location of the Extended IP Phone—within the same network or outside—according to your installation scenario.
- Log into the web-browser Jeeves. Read the topic “System Configuration using the Web-based GUI”.
- Configure **DHCP Server** on the “Network Parameters” page under *Basic Settings* of Jeeves.

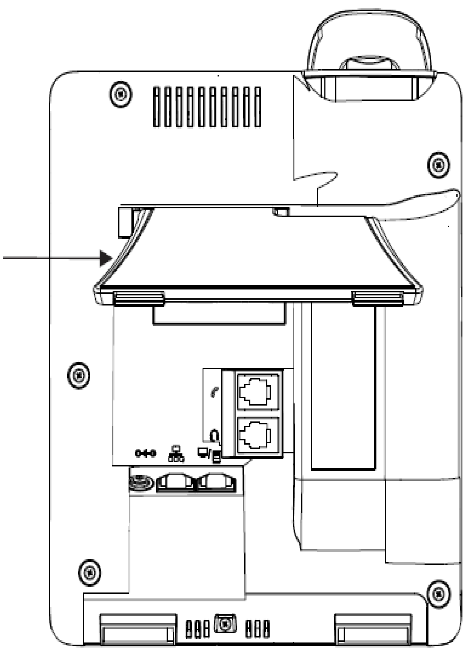
*If you want to use the **DHCP Server** on your LAN for assigning IP Address to the Extended IP Phone, select **DHCP option 224** and **Data Type** as ‘**String**’ and program the Ethernet Port IP Address /Domain Name and SPARSH Port in the format **“Ethernet IP Address: SPARSH Port”** in your LAN DHCP Server.*

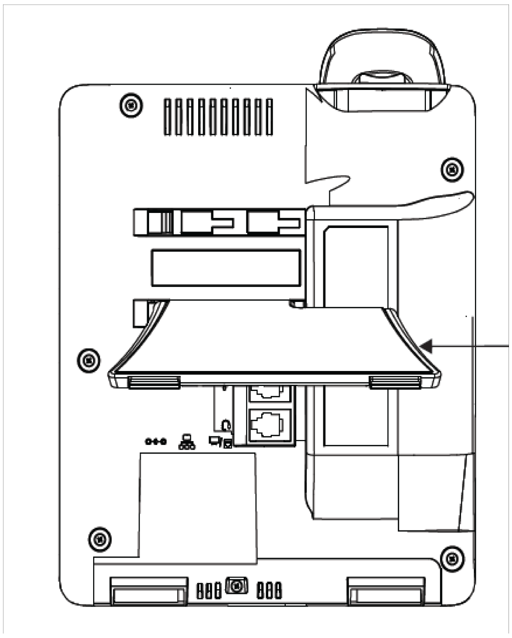
- Assign an extension number to the Extended IP Phone on the “Extension and Feature Codes” page under *Basic Settings* of Jeeves.
- You must configure the necessary parameters in ETERNITY so that SPARSH VP310 can register as a SIP Extension. For instructions, see “SIP Extensions” under *Basic Settings*.

Now, follow the steps described below to install the SPARSH VP210.

1. Unpack the SPARSH VP210 box and verify package contents.

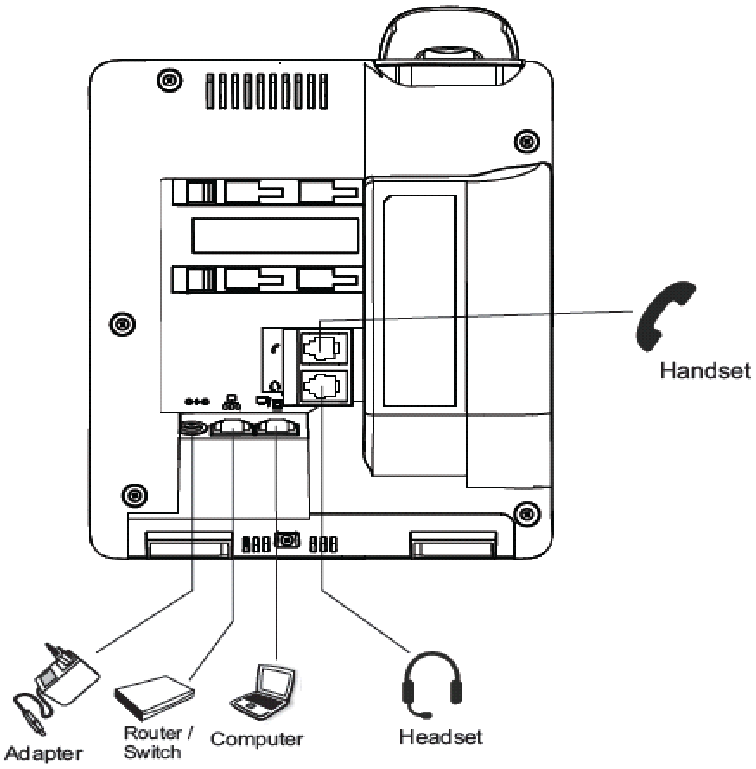
2. When you mount the phone on a desk, you can attach the Foot Stand in two ways at **45° Angle** or at **55° Angle**.





- 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 ETERNITY NENX.
- On successful download of all configuration files, the phone attempts to register with ETERNITY NENX.
- On successful registration, the Home screen appears.

The phone will register successfully, only if the SIP Extension parameters in ETERNITY NENX *have been correctly configured as per your installation scenario.*

Refer to the ***SPARSH VP210 (Extended) User Guide*** to know more.

Installing the Voice Mail System

The optional Voice Mail System (VMS) of the ETERNITY NENX provides mailbox facility to all extension users.

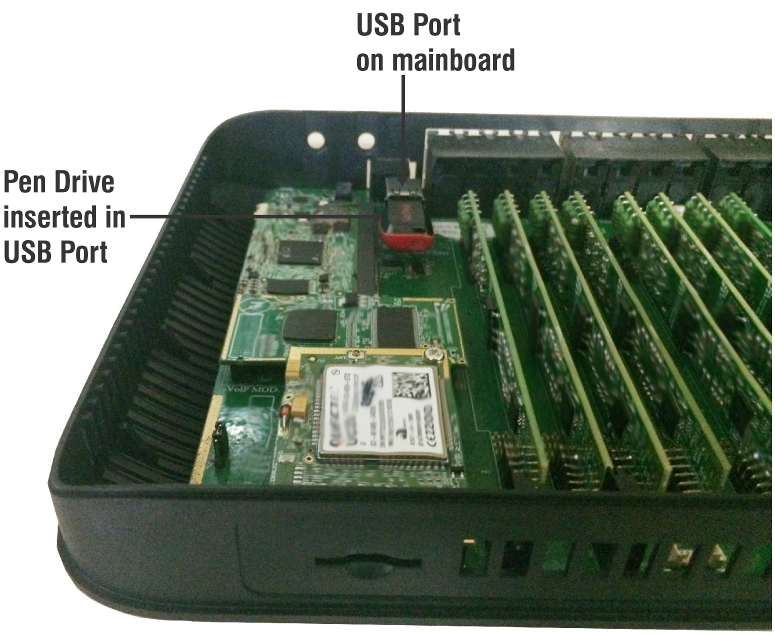
Each Mailbox has the capacity of storing 10,000 voice messages. The maximum size of each Mailbox is 60,000 minutes. By default, the size of each Mailbox is set to 5 minutes. The maximum Message Length for each Mailbox is 3600 seconds. By default, the Maximum Message Length for each Mailbox is set to 120 seconds.

The VMS in the ETERNITY NENX requires a license. The VMS in ETERNITY NENX is delivered as Pen Drive (factory -fitted), containing VMS configuration files, and voice messages for prompts and greetings. The Pen Drive is also the storage device for mailbox messages.

Matrix provides a 8GB Pen Drive. You can replace the Matrix's Pen Drive with another, if required and can use a Pen Drive of up to 32GB. To replace the Pen Drive, see "Replacing the Pen Drive" at the end of this topic.

Installing the VMS

- The Voice Mail System (VMS) is provided in a Pen Drive along with the system. This Pen Drive is factory fitted in the system. However to activate it, you must purchase the VMS License.
- In case you need to remove/insert the Pen Drive, you can follow the steps given below.
- Make sure power supply is turned off before you begin installation.
- Unscrew and remove the top cover of the enclosure. Keep the screws and the cover aside.
- Locate the USB port on the mainboard. Insert the Pen Drive into the upper USB port on the mainboard.



- If you have no other modules to install, replace the top cover and secure the cover with the screws.
- Connect a computer to the Ethernet Port of ETERNITY NENX with the Ethernet cable supplied for the port.
- Open a Web browser on the computer to access the embedded Web server, *Jeeves*.

- Activate the License Voucher for the VMS. See "License Management" under *Advanced Settings* for instructions.

- Configure VMS. See "Configuring Voice Mail System" for detailed instructions.

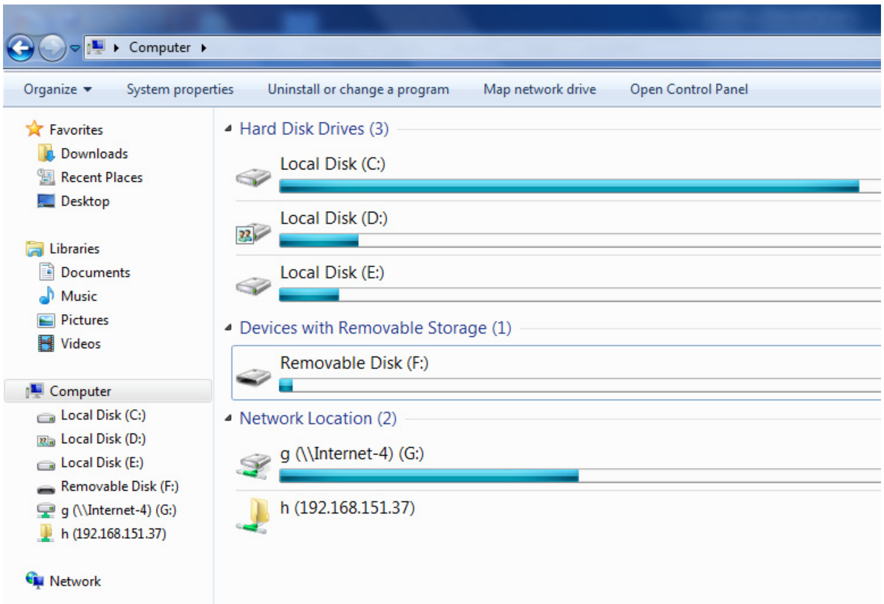
Replacing the Pen Drive

If you are replacing the factory supplied Pen Drive, you must:

- Format the new Pen Drive with FAT32 file system.
- Copy the contents of the factory fitted Pen Drive onto the new Pen Drive.

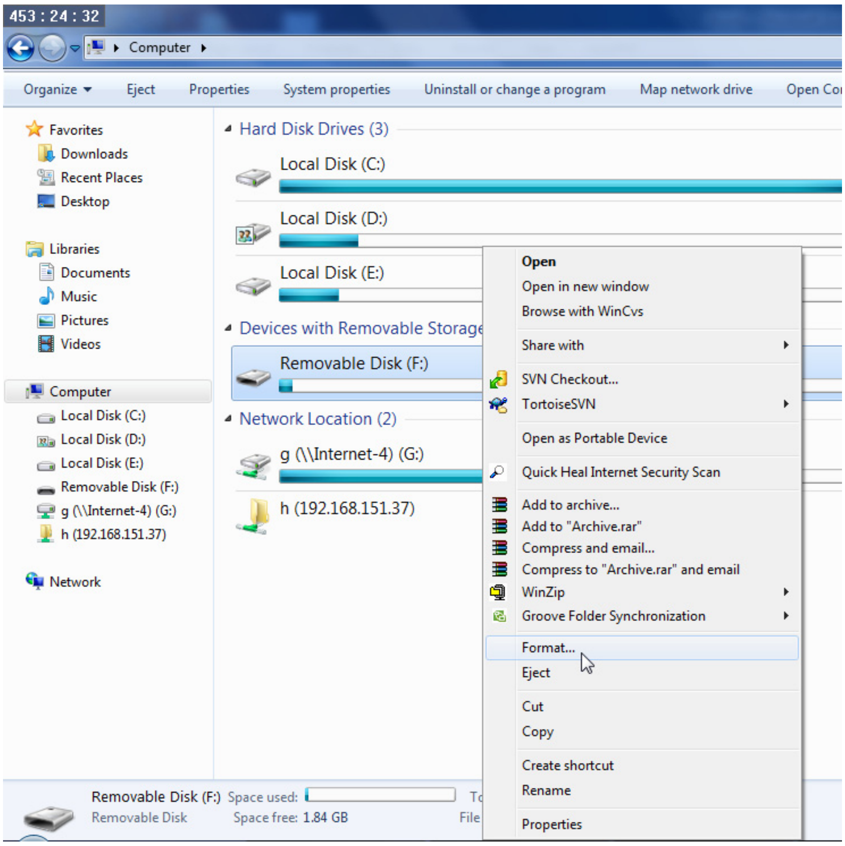
To format the Pen Drive with FAT32, follow the steps given below:

- Insert the Pen Drive in the USB port of your computer.
- Click **My Computer**.

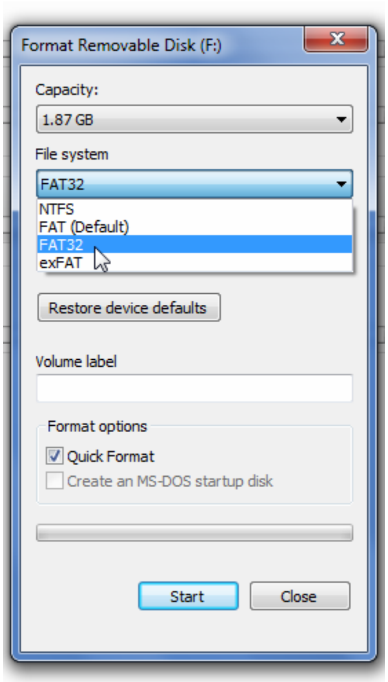


- Right-click the removable disk to which you have connected your Pen Drive, in this example **Removable Disk (F:)**.

- In the shortcut menu, select **Format**.

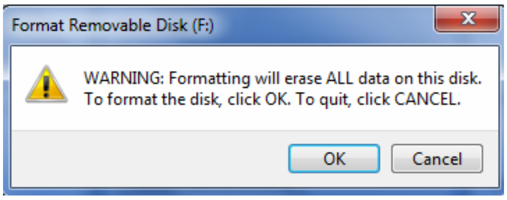


- The **Format Removable Disk (F:)** options appear on your screen. In **File Format** select **FAT32**.



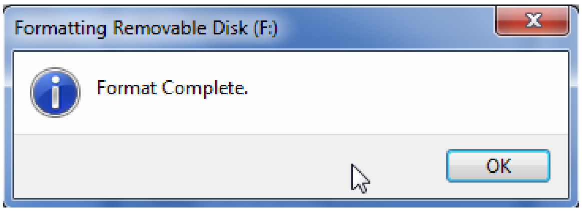
- Click **Start** to begin the formatting process.

- You will get an alert: *"WARNING: Formatting will erase ALL data on this disk. To format the disk, click OK. To quit, click CANCEL."*



- Click **OK** to format.

- When the formatting process is complete, the message "Format Complete" will appear on your screen.



- Click **OK** to close the formatting window.

- Now, copy the contents of the factory fitted Pen Drive onto the new Pen Drive.

Starting Up ETERNITY NENX

Power ON

- Connect the power adapter to ETERNITY NENX.
- Plug in the power adapter of ETERNITY NENX into the power outlet.
- Switch on the power supply and wait for the Reset Cycle to complete.

ETERNITY NENX supports a Power Fail Transfer Module, which enables you to receive and make calls even during power fail conditions. For more details, see *“Power Fail Transfer”.*

LED Indication

- At Power ON, Power LED will turn ON (Continuous Green).
- System LED (STS) will display following error/events/status.

LED Status	Colour	Comment
1sec On - 1sec Off (Continuous)	Green	ETERNITY NENX 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.

You may now access the web-based programming tool, ***Jeeves that is***, *the SARVAM UCS SOHO Application* and configure the same.

CHAPTER 4

Configuring SARVAM UCS

SARVAM UCS can be configured using the following tools:

- A web-based graphic user interface (GUI), called Jeeves - SARVAM UCS SOHO Application
- A telephone (specific parameters only)

Each of these is explained in detail, later in this chapter.

SARVAM UCS can be configured at two levels: System Engineer and System Administrator. A distinct set of features and facilities can be configured at each of these levels. SARVAM UCS can be configured both, on site and from a remote location without its functioning being affected.

System Configuration using the Web-based GUI

ETERNITY NENX provides a Graphic User Interface (GUI), Jeeves, the proprietary web-based configuration software of Matrix. It is an HTTP server built into the ETERNITY NENX.

Jeeves allows system configuration at two levels: System Engineer and System Administrator. Both these levels are protected by SE and SA password respectively. These passwords can be changed using Jeeves only. A distinct set of features and facilities can be configured at each of these levels.

It is possible to configure 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.

System Engineer Mode

At the System Engineer level, the entire system configuration 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.

Access to the SE mode is protected by means of a password, referred throughout this document as the SE Password. Four persons can simultaneously login and change the system settings from the SE mode.

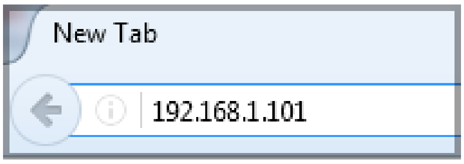
To be able to access Jeeves,

- the Ethernet Port of ETERNITY NENX must be connected with a stand-alone PC or in a LAN.
- a web-browser, either Internet Explorer 7 or later or Mozilla Firefox 3.5.1 or later, must be installed on the PC.

To log into the SE Mode,

- Open the browser (Internet Explorer/Mozilla Firefox) on the PC (Standalone or LAN PC) to which the ETERNITY NENX is connected.

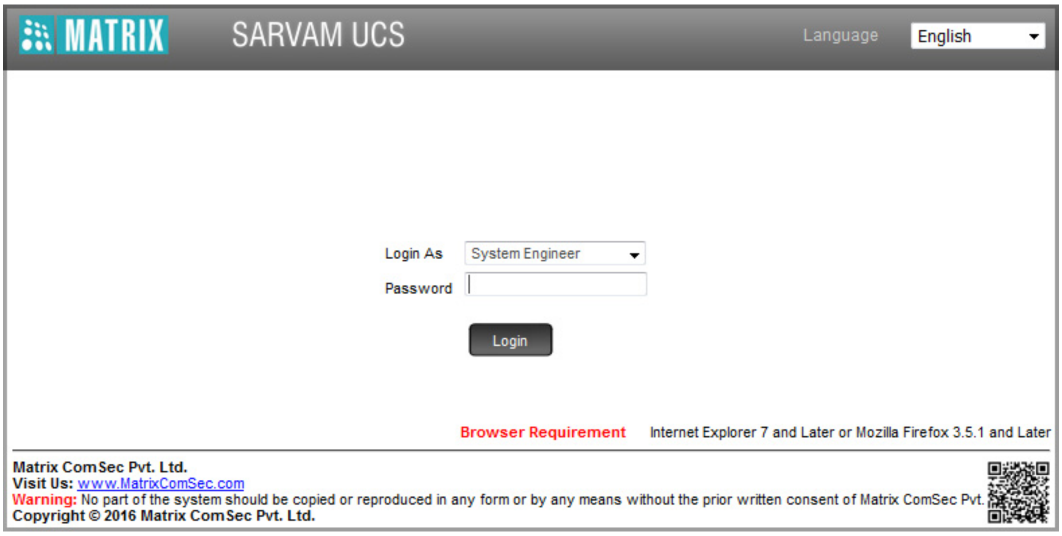
- Enter the IP address of the Ethernet Port of ETERNITY NENX in the address bar of the browser. Default IP Address is 192.168.1.101.



- You will be redirected to the HTTPS protocol for secure access.

- Click the https://192.168.1.101link.

- The **Login** page will open.



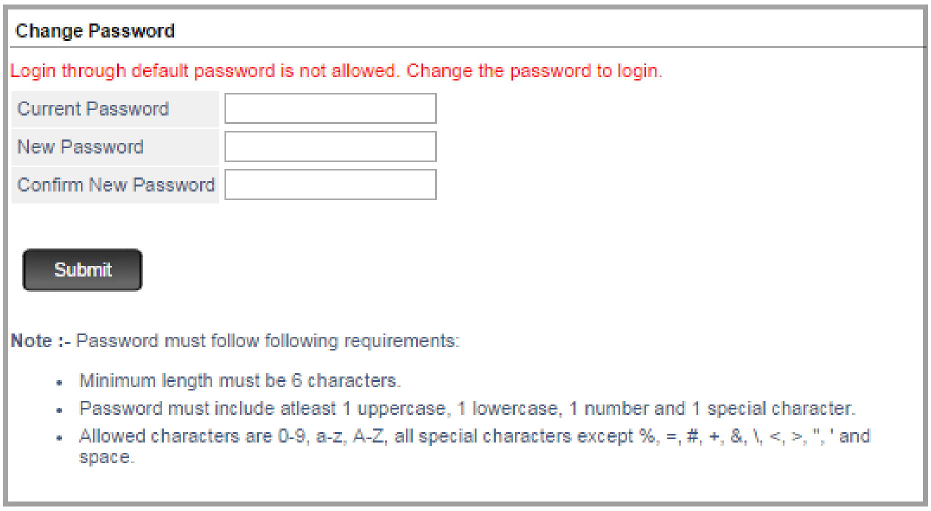
- In **Login as** select **System Engineer**.

- In **Password**, enter the default SE password, 1234.

- Click **Login**.

Before you start configuring the system, if you wish to view or download the ETERNITY NE Quick Start or any other related documents, you can click or scan the QR Code present on the login page of Jeeves.

- You will be prompted to change the default SE Password for accessing Jeeves.



- In **Current Password**, enter the default SE Password, 1234.

- In **New Password**, enter the desired 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.

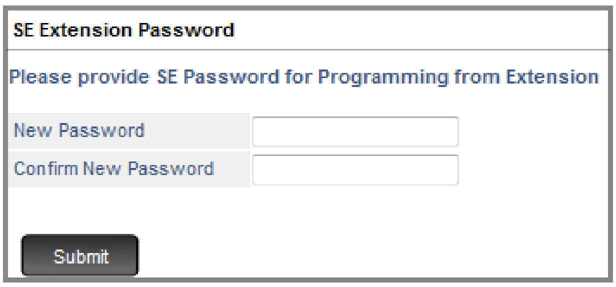
- In **Confirm New Password**, re-enter the new password to confirm.

- Click **Submit**.

  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**.



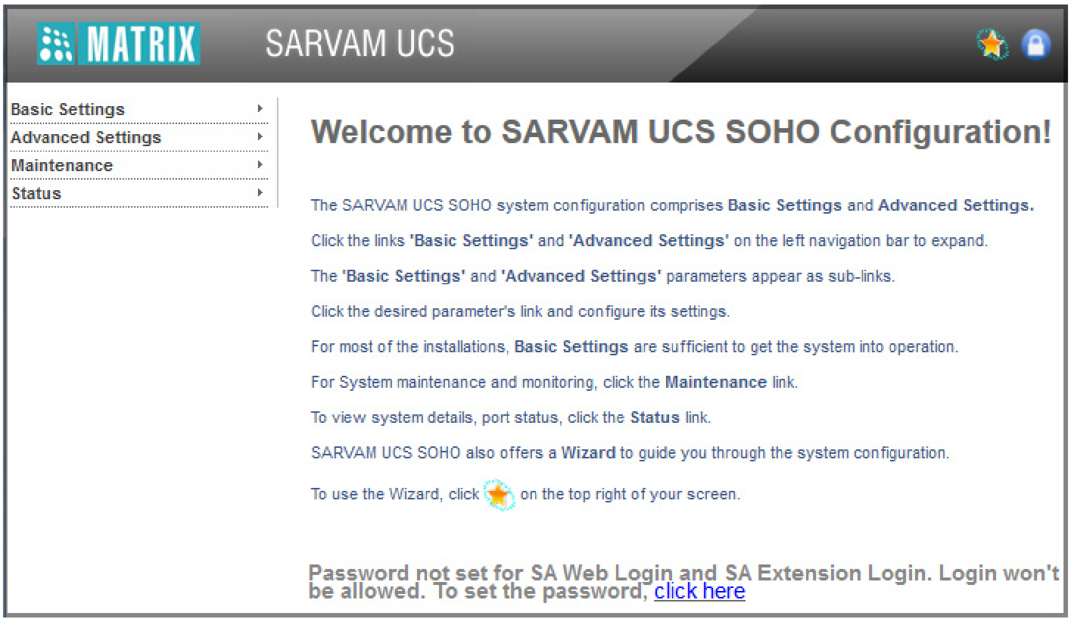
- Enter the **New Password.** The new password can be a minimum of 4 digits to a maximum of 12 digits. The valid digits are from 0 to 9.

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.**

  The Welcome page opens.



As this password is meant for restricting access to the SE mode, we strongly recommend you to:

- *Keep the password confidential.*
- *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.*

- *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 Wizard

For the ease of installation, as well as to simplify and speed up the process of setting up SARVAM UCS, the Jeeves offers **Quick Installation Wizard**. This wizard helps the Installer/System Engineer to quickly set-up the system for the standard PBX Application.

Using this Wizard, the Installer/System Engineer can configure as much as 80 percent of the system configuration, covering all the necessary parameters essential for the functioning of the system. For advanced configuration of features and facilities, the Installer/System Engineer must use Basic Settings and Advanced Settings links of the Jeeves.

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 Basic Settings and Advanced Settings links of the Jeeves.

As many as four System Engineers can simultaneously login into SE Mode and configure the system. However, it is recommended to avoid multiple login when using the Quick Installation Wizard. The use of the Wizard must be restricted to a single person only.

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 System reports, can be captured and printed.

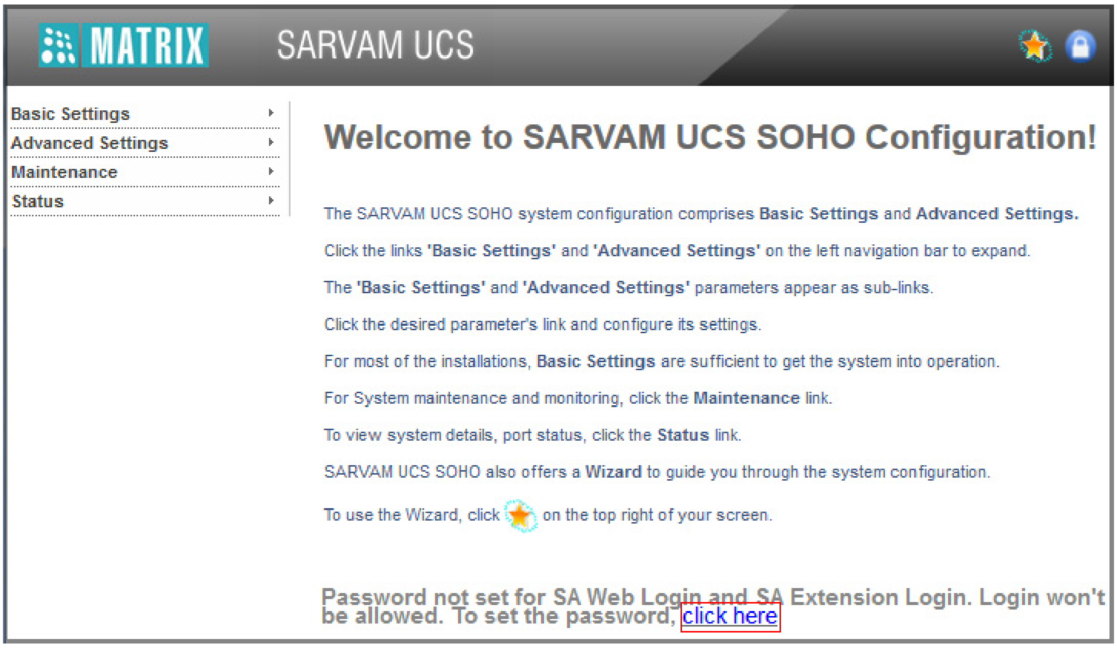
The System Administrator may be a operator or receptionist, or any one responsible for the operation and maintenance of the system.

The access to SA mode is protected by means of a password, referred throughout this documents as the SA Password. Four persons can simultaneously login and change the system settings from the SA mode.

You can log into the SA mode through Jeeves or from extension only after you have set the SA Password from the SE mode.

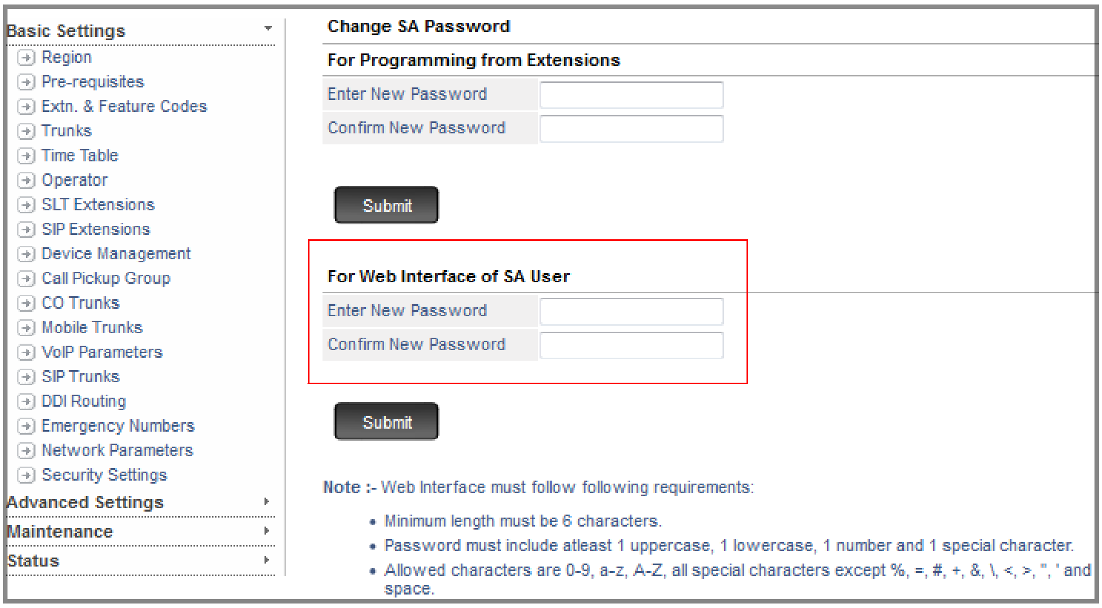
To log into the SA Mode,

- Login as System Engineer. The Welcome page opens.



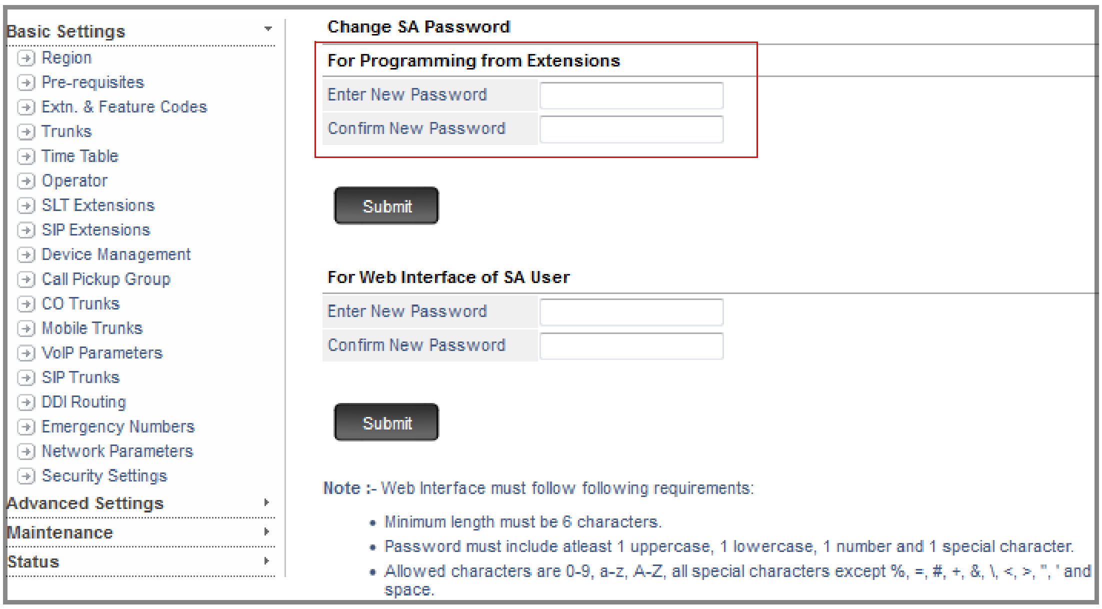
- To set the password for **SA Web Login** and **SA Extension Login**, click on the link.

Set SA Password for Accessing Jeeves



- Under **For Web Interface of SA 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**.

Set SA Password for Programming from Extensions

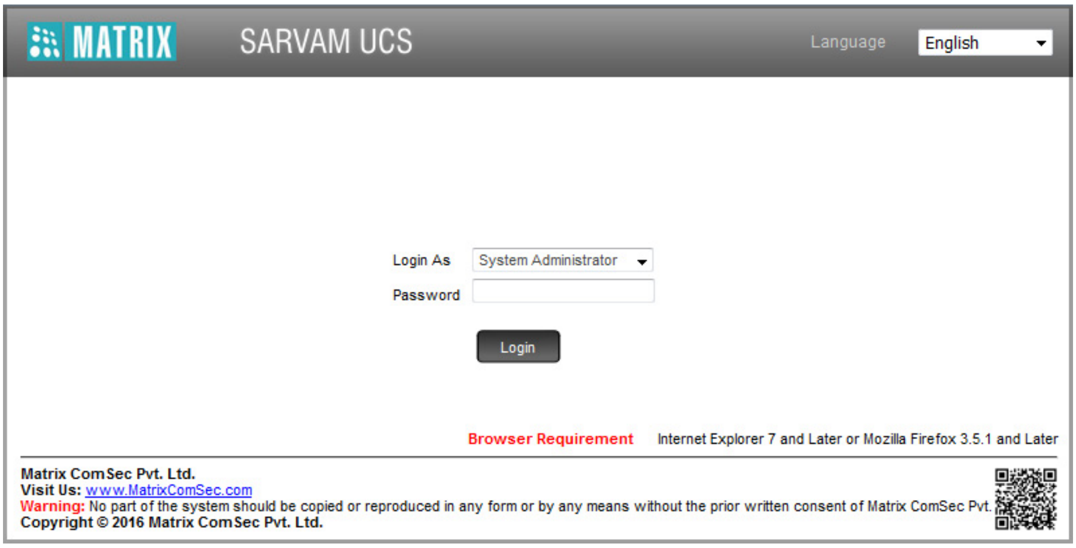


- Under **For Programming from Extensions**,
  - **Enter New Password.** The new password can be a minimum of 4 digits to a maximum of 12 digits. The valid digits are from 0 to 9.
  - In **Confirm New Password**, re-enter the new password to confirm.
  - Click **Submit**.

  Once the SA password is set, you can now log into the SA mode through Jeeves.

To enter the SA mode,

- Open the Login Page.



- In **Login as** select **System Administrator**.
- In **Password**, enter the New SA Password.
- Click **Login**.

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 confidential.*
- *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.*

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”.*

- *If you select the language on the Welcome page or on any of the Login pages, it is valid for the current session only. The default language will be applied on next login.*
- *When you select the "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. This default local language 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.*

System Configuration using a Telephone

A few important parameters of SARVAM UCS may be configured by dialing the relevant command strings from a telephone. This kind of system configuration using a telephone is useful in scenarios, where you do not want to access the Jeeves of the system.

By using a telephone you can set the essential network parameters of the system without connecting it to the Standalone PC or LAN. This can be achieved by connecting a SLT to the system and dialing the command strings by entering into the SE Mode. After you change the required network parameters, you can now connect the system directly in your network. To know more, see "Entering the SE Mode using a Telephone".

You can also configure the system using a telephone, in situations where Jeeves is not available. This can be done by connecting a Single Line Telephone (SLT) or an Extended IP Phone to the system.

For the ease of operation, you may use an Extended IP Phone instead of a SLT. Using an Extended IP Phone gives you the following advantages,

- you can view the command strings that you have keyed in on the phone's display.
- 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 a 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 be configured at each of these levels

- *It is possible to configure SARVAM UCS from any location using a Telephone. You can use a Telephone connected as an extension of SARVAM UCS to configure the system On site (where it is installed).*

- *You can also configure the system using a Telephone Off site, i.e. from a Remote location, using the "Direct Inward System Access (DISA)" feature of SARVAM UCS. You can access both the System Engineer mode as well as the System Administrator mode from the remote location.*

System Engineer Mode

At the System Engineer level, configuration can be done by dialing command strings referred to in this document as **SE Command** or **System Command**. SE Commands are unique to the feature/facility being configured. System Engineer mode is protected by a password referred to as SE Password.

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. See "Restoring Default SE Password" for instructions on restoring the default SE password.

Entering the SE Mode using a Telephone

.To enter the SE mode via Extended IP Phone / SLT (for all countries except 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.

To enter the SE mode via Extended IP Phone / 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.

To exit the SE mode:

- Dial ***00***.
- You will hear the dial tone of SARVAM UCS.

You can configure only the basic network parameters from your telephone using the default SE Extension Password, 1234. To know the detailed list of parameters which you can configured using the default SE Password, see *"Basic SE Commands".*

- *If you enter the wrong SE Password, an Error Tone will be played and an Error Message will be displayed on the LCD of your phone.*
- *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.*
- *There is no restriction on the number of persons who can simultaneously enter SE mode from a telephone and configure the system.*

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 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-configurable. See the topic “Access Codes” under *Features and Facilities*.

To know how to use/change feature settings with SA Commands, please refer the description of individual features under “Features and Facilities”.

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. You can log into the SA mode from extensions only after you have set the SA Password from the SE mode through Jeeves. 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.

- *When SARVAM UCS is used in the Standard PBX 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

To enter SA mode from an extension phone (for all countries except New Zealand),

- Dial ***1#92-SA Password***
- 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.
- 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***
- 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.
- 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 configure the SA Mode Timer:

- Dial ***1072-016-SA Mode Timer***
  Where,
  SA Mode Timer = 000 to 255 minutes.
  Default: 003 minutes.

- You can also exit the SA mode before the SA Mode Timer expires by dialing the command to exit the SA Mode. 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 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.

Logging Out Users from Jeeves

It is possible for four users to simultaneously login as System Engineers and use Jeeves. It is also possible to log out all these users at once or log out any of these users selectively. To do this,

- Dial ***1#91-SE Password*** to enter SE mode from a SLT.
- Dial ***2188*** to log out all users.
- Press *Enter* key to save setting.

CHAPTER 5

Basic Settings

Basic Settings help you with the basic configuration of the system in easy steps. The Basic Settings break down the complexities of configuration and cover as much as 80 percent of all your basic installation and configuration requirements.

Due to security concerns, the default settings of the system have been changed. For systems with Firmware later than V1R6.7 these settings will be applicable automatically. These country-specific default values assigned to the parameters in Basic Settings are sufficient for incoming calls to be placed on the system but for outgoing calls from the system you must change the settings as per your installation requirement. To know the changes in the default values, refer to "Modified default parameter values for Firmwares later than V1R6.7"

If you are upgrading the Firmware, refer "After updating Firmware Version later than V1R6.7" and "Modified default parameter values for Firmwares later than V1R6.7".

To configure Basic Settings,

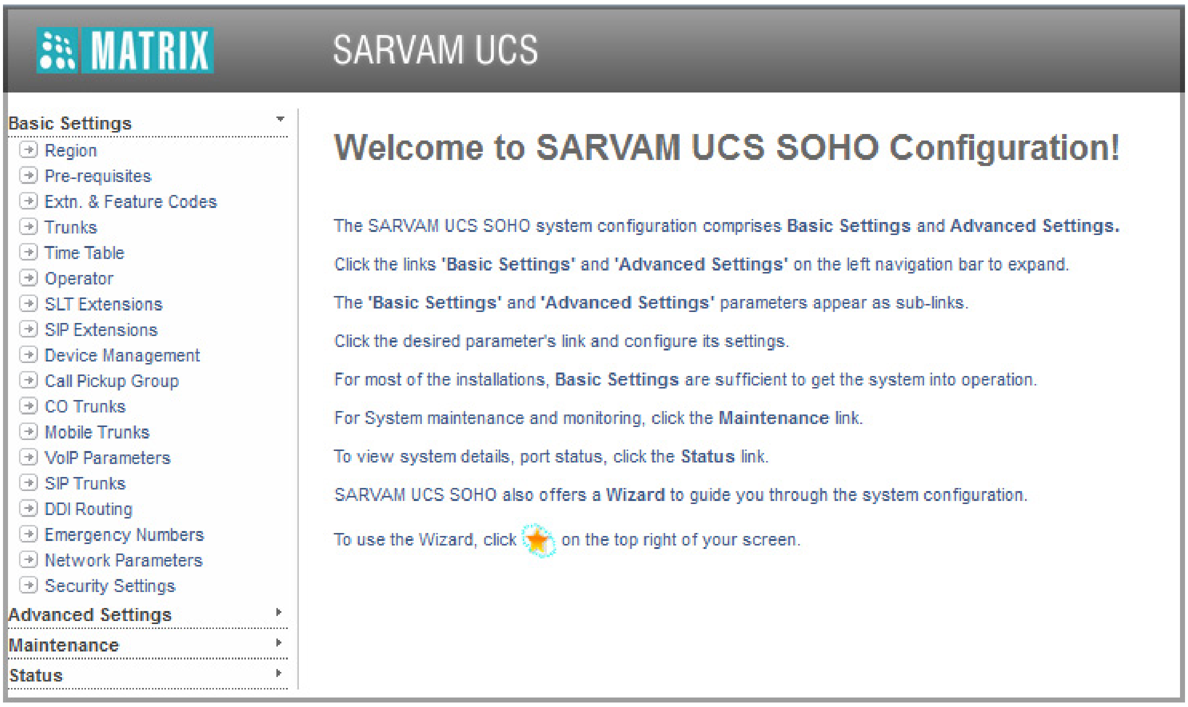
- Open Jeeves. See "System Configuration using the Web-based GUI" for instructions.

  The Login page will open.

- Login as **System Engineer**.

  On successful login, you reach the Home page. The **Basic Settings**, **Advanced Settings**, **Maintenance** and **Status** links appear on the left pane.

- Click **Basic Settings** on the left pane.



The sub-links to these basic parameters appear:

- Region
- Pre-requisites
- Extension & Feature Codes
- Trunks
- Time Table
- Operator
- SLT Extensions
- SIP Extensions
- Device Management
- Call Pickup Group
- CO Trunks
- Mobile Trunks
- VoIP Parameters
- SIP Trunks
- DDI Routing
- Emergency Numbers
- Network Parameters
- Security Settings

There are two ways to do the basic system configuration:

- using the *Wizard*. A special configuration Wizard will lead you logically, step-by-step through the configuration of the parameters listed above.

  Or

- using the *Basic Settings* links for selective configuration. You can choose the parameters you want to configure, and accordingly select the parameter links and configure the settings.

Using the Configuration Wizard

The configuration Wizard leads you step-by-step, through the configuration of the basic settings.

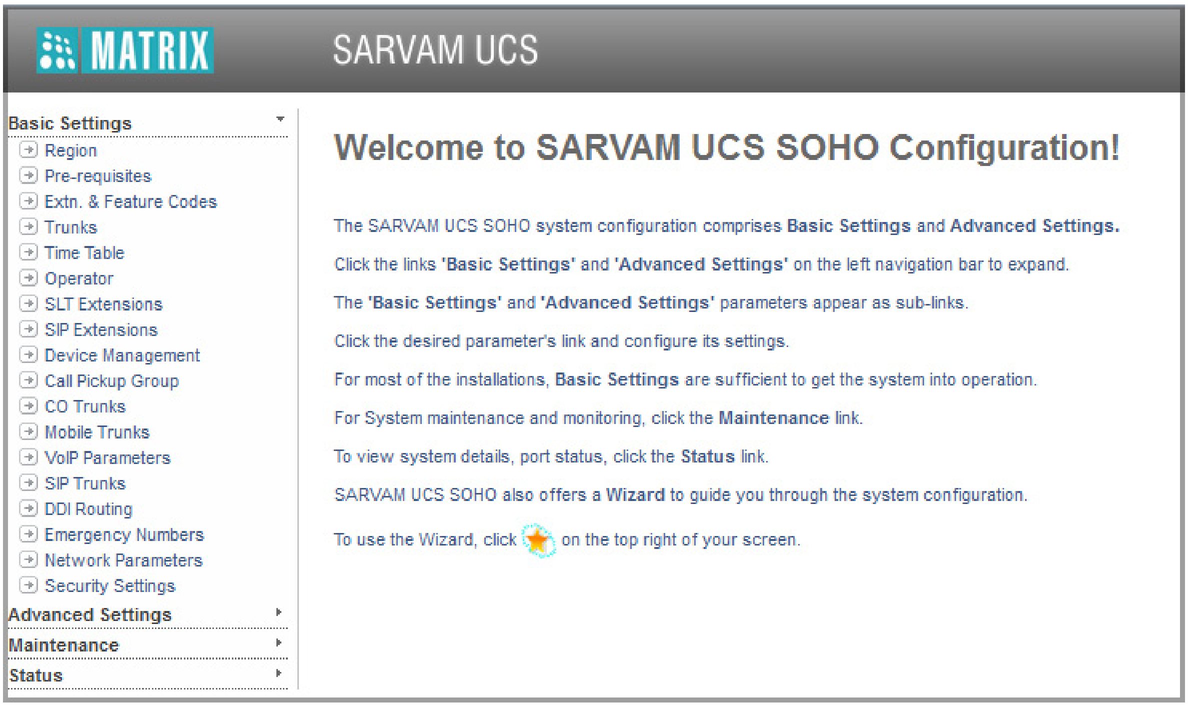
To use the Wizard,

- Click the **Wizard** icon on the top right of your screen. The home page of the Wizard will open.
- To navigate the Wizard pages use the **Next** and **Back** buttons.
- When you press the **Next** button, the changes on the current page are saved and the Wizard takes you to the next page.
- When you press the **Back** button, you will be prompted to save changes made on the current page.
- The **More** button and the **Less** button on the page allow you to expand and collapse respectively, the parameters on the page.
- You may exit the Wizard at any time by clicking the **Quit** button . The changes you made before you exit will be saved.

Using Basic Settings Links

You can choose the parameters you want to configure, and the order in which you want to configure. To do this,

- Click the **Basic Settings**. The parameter sub-links appear on the left pane.



- Click the parameter sub-link you wish to configure. The respective parameter page opens.
- Get familiar with the buttons and icons listed below before you begin to change the settings of the parameters on each page.

More: displays all the parameter links on the page.

Less: displays the essential parameter links on the page.

Expand: expands a link to display all parameters under the link.

Collapse: collapses a link. Hides all parameters under the link.

Copy: Click to copy the parameters under a specific link of a page to the same link under any other page.

Settings: allows you to configure the settings of a parameter further.

···· **More link:** displays all additional parameters on the page.

Logout: allows you to exit Jeeves.

- Set the desired values on this page and click **Submit** to save.

You may use the Wizard or selectively configure the Basic Settings pages, whichever works best for you.

When configuring the system for the first time, you may use the Wizard. When you want to make changes after the configuration has been done, you may selectively configure the Basic Settings parameters using the links and sub-links.

Do not allow simultaneous login and system configuration using the Basic Settings pages. As the system updates configuration changes last submitted, configuration changes made by one person may be overwritten by those made by another.

Region

The System can be operated anywhere in the world. It provides “Default Settings” to match country/region-specific requirements of users around the world.

The Default Settings are factory-set values for system and feature configuration.The system is designed to work efficiently in any country with these default settings.

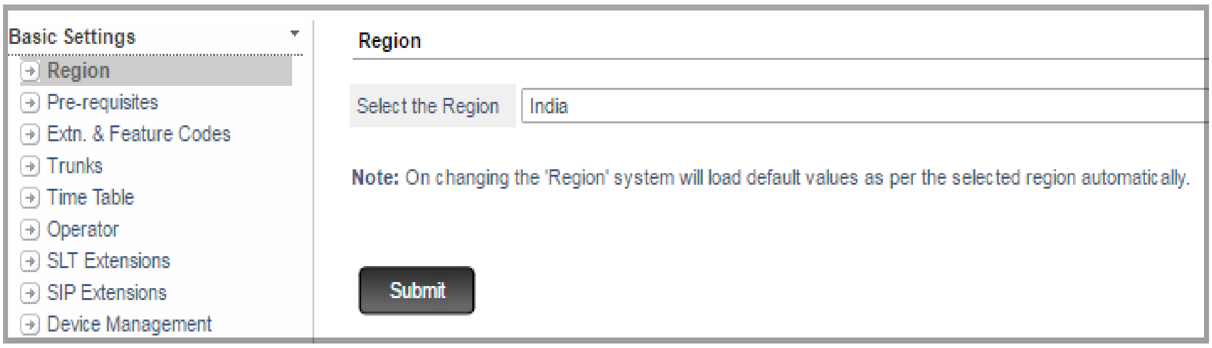
Default Settings also speed up the process of system configuration, as these are sufficient for getting the system into operation.

To load the country-specific Default Settings, you must select **Region** (the country/region) in which the system is installed. Certain countries like United States of America, Canada, Australia are divided into various regions due to different Date and Time and DST convention followed by that region. When you change the region within same country, only the DST and the Date and Time Settings will be changed as per the selected region. Other country-specific parameters will remain unchanged.

India is selected as the default Region. So, if you are installing the System in a country other than India, change the Region.

To configure,

- Click **Region**.



- **Select the Region** (the country) where the system is installed from the list.

- Click **Submit**.

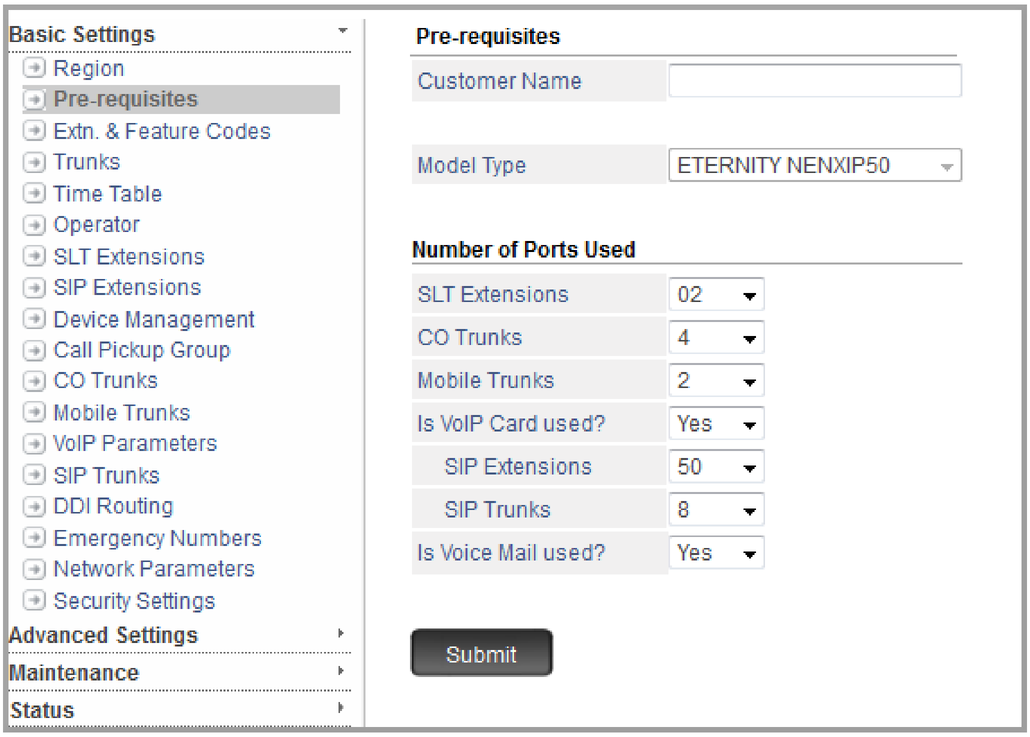
Pre-requisites

This page displays the system resources (number of trunk and extension ports supported) supported by 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 easier and more focused, SARVAM UCS allows you to define System Pre-requisites: the system configuration you are using (ETERNITY NENXIP50, ETERNITY NENX312, ETERNITY NENX416) the number of trunk and the number of extension ports you want to configure. Accordingly, Jeeves will show only as many trunk and extension ports that you have specified.

To configure,

- Click **Pre-requisites**.



- Enter **Customer Name**. Default: Blank.

  Customer Name is the name of the organization/enterprise using SARVAM UCS.

  The Customer Name you enter in this field will appear as header on the various System Reports generated and printed by the SARVAM UCS like SMDR Incoming, Outgoing and Internal Call Reports, Alarm Status Reports, etc.

  The Customer Name may consist of a maximum of 80 characters, including punctuation marks. So, you can also enter the organization's address along with the Customer Name.

You can assign Customer Name also on the "System Parameters" page. If you have entered the Customer Name on this page, the same Name will appear on the System Parameters page.

- **Model Type** is the name of the model of ETERNITY NENX detected, that is the model type which is on-board (connected). Accordingly, all pages of Jeeves display only those ports detected by the system for configuration as supported by that model.

  You can then define the number of ports you want to configure out of the existing port types.

- Define the **Number of Ports Used** for each Port Type: CO, SLT, Mobile, SIP from the respective combo boxes.

  For example, if you want 4 CO Trunks, select the same numbers from the combo box.

  If you want to use voice mail, select Yes in the respective combo box.

- Click **Submit**.

Extension and Feature Codes

The default Extension Numbers and Feature Access Codes[13] appear on this page.

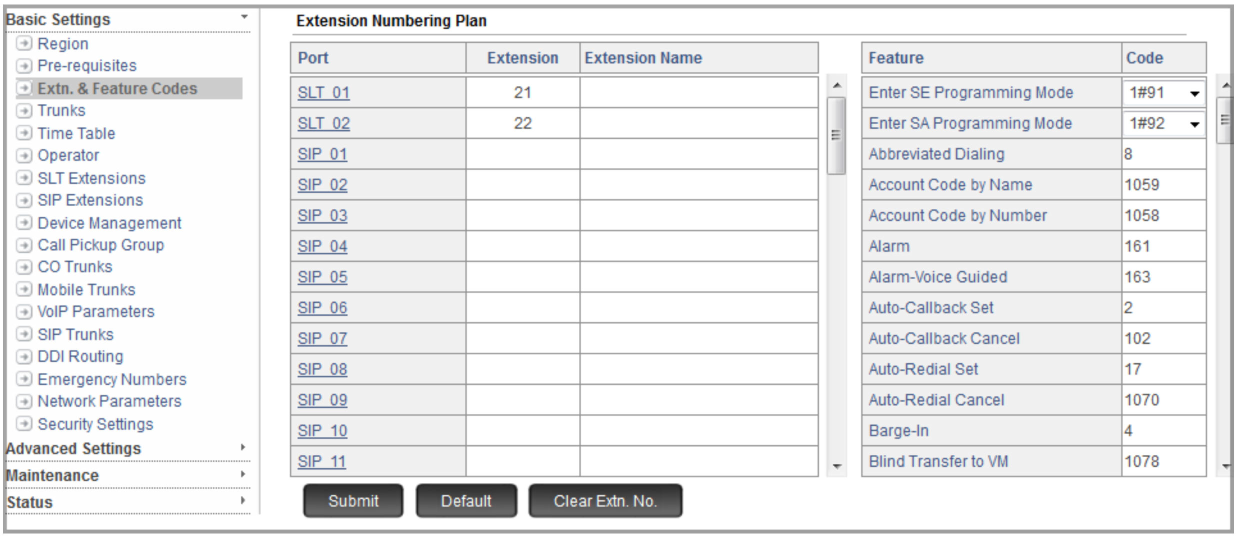
SARVAM UCS assigns default extension numbers to the extension port types:

- 21- 22 to SLT port number 01 to 02, depending on the model[14].
- By default, the Extension Numbers of SIP extensions are blank.

The default Feature Access Codes that appear on this page are country-dependent.

To configure,

- Click **Extn. & Feature Codes**.



- You may re-assign extension numbers to SLT and SIP Extensions.
  - The length of the extension number may be 6 digits. The digits 0 to 9, # and * are allowed. (*'*****' and '**#**' can be used only as the first digit.)*

When you change the extension numbers, make sure that they do not clash with any other Feature Access Codes in the dialing phase (codes starting with 1, 9, etc.). To know more, refer the "Access Codes" and "Conflict Dialing".

Each time there is a conflict of access codes, you will get an alert message prompting you to resolve the conflicting numbers. You must resolve the conflict first.

If you want to change the Feature Access Code that is conflicting with the extension number you have assigned, you can do so on this page.

- Assign names to extensions. Extension names may consist of a maximum of 18 characters.

13. *Feature access codes are short digit codes used to invoke a feature or function. See "Access Codes".*
14. *Here, the model type is ETERNITY NENXIP50. For ETERNITY NENX312 and ETERNITY NENX416, refer to "Technical Specifications of ETERNITY NENX".*

- *This page displays only the extension ports that are available in the system and the number of ports you have defined as **Number of Ports Used** in the Pre-requisites page.*
- *In the case of SIP Extensions, the number you configure as Extension Number will be considered as the SIP ID.*
- *You may configure the desired extension by clicking the respective link on this page.*

- If you have finished assigning numbers and names, click **Submit**.

  To assign the extension numbers and names all over again, click **Clear Extn No**.

  To default all the feature access codes and extension numbers, click **Default.**

- Reassign Feature Access Codes, if required. The new access code may be a single digit or a sequence of a maximum of 6 digits. Digits 0 to 9, # and * are allowed.

- Click **Submit.**

If you assign the same Access Code to more than one feature, Jeeves 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.

Trunks

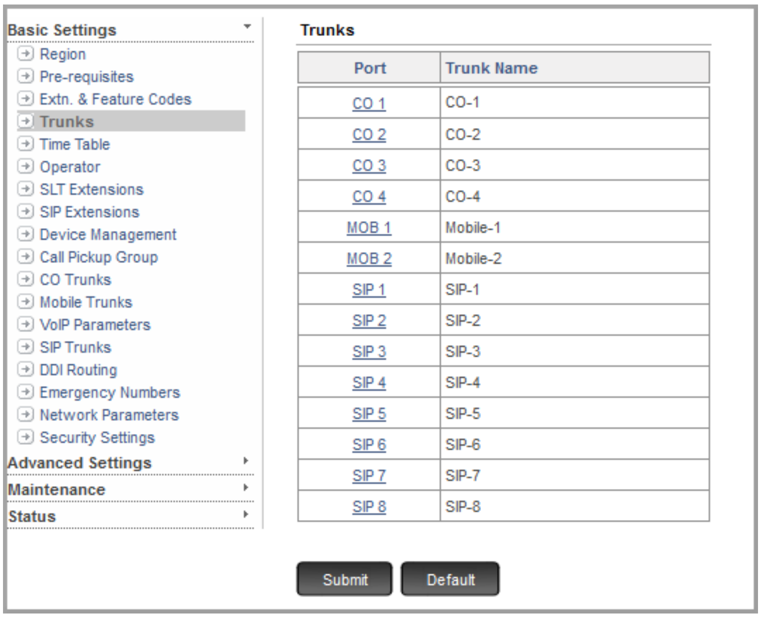
You can assign names to trunk ports for easy identification of the trunks. On this page, trunk ports appear with their default names and port numbers, in ascending order of the port number.

The default trunk name consist of the trunk type (CO, MOBILE, SIP) and port number. For example, the name of the Two-wire Trunk connected to port number 1 is displayed as CO 1.

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 Pre-requisites page.

To configure,

- Click **Trunks**.



- You may assign names to each trunk port type for easy identification. The name can be derived from the name of the Service Provider.

  The name may consist of a maximum of 18 alphanumeric characters.

- Configure the desired trunk by clicking the respective link on this page. For instructions, see “CO Trunks”, “Mobile Trunks”, “SIP Trunks”.

- Click **Submit**.

Time Tables

Certain features of SARVAM UCS like Operator, Class of Service, Toll Control, Outgoing Trunk Access, Direct Inward Dialing, Direct Inward System Access, Trunk Landing extensions, Call Routing, require its trunks and extensions 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 Day Time (Working hours), or access to outgoing long distance calls are to be denied during Night Time (Non-working hours), or the extension must play a different greeting message to the callers during Break Hours and Holidays (Non-working hours).

Time Tables can be assigned to extensions 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:

- Day Time (Working hours)
- Break Hours (Non-working hours)
- Night Time (Non-working hours)

The default Time Zones defined for each day are:

- Day Time (Working hours): 09:00 to 18:00
- Break Hours: 00:00 to 00:00
- Night Time (Non-working hours): 18:00 to 09:00

Day Time (Working hours), Break Hours and Night Time (Non-Working hours) are set to 00:00 for Sunday.

You can define a different Time Zone for your organization. Further, you can also program each day of a week with different time zones. For example, you may define the Day Time (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 Day Time (Working Hours) also for Sunday.

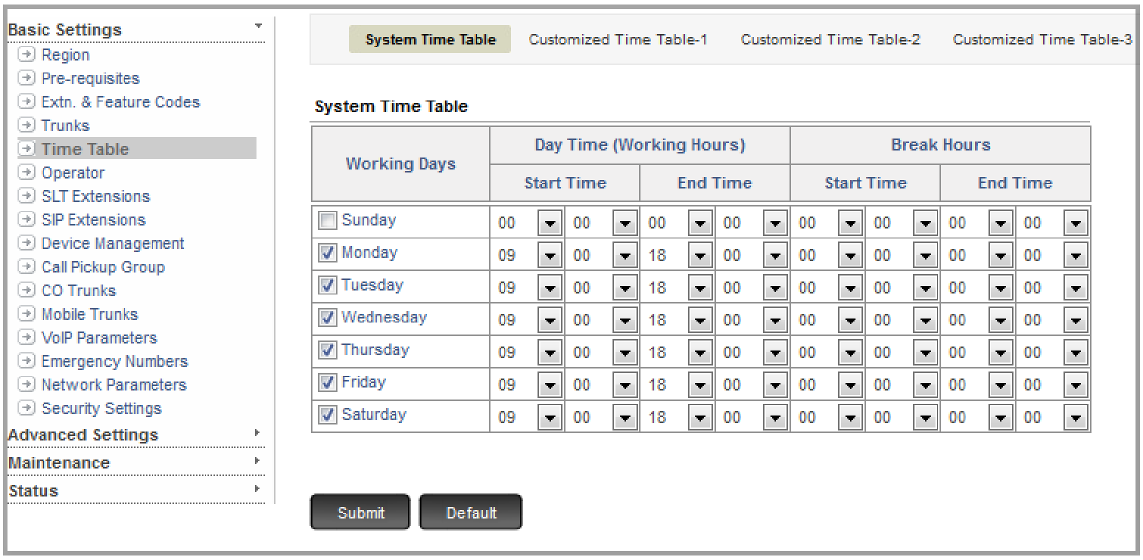
Time Tables

SARVAM UCS offers 4 different Time Tables: the System Time Table (applied to all trunks and extensions and Operator by default), and customized Time Tables - 1, 2, and 3 which you can set to your preferences and apply to trunks and extensions.

You can assign different Time Tables to different trunks and extensions. Refer the topic "Day Night Mode" to know more.

To configure,

- Click **Time Table**.



- Decide your Working days, Working hours and Break hours (24 hours format).

  By default, System Time Table has Monday to Saturday as Working days, Working hours are defined as: 09:00 to 18:00 for all Working days and Break hours are defined as: 00:00 for all Working days. Sunday is a Non-working day, with Working hours and Break hours set to 00:00.

  The system considers the time period other than the Day time (Working hours) and Break hours as Night time (Non-working hours).

  If the System Time Table suits you, retain it. You may redefine the Working days and Working hours. You can also define different Working hours for each day of the week.

- To define the **Working Day**, select the related check box.
- To define the **Day Time (Working Hours)**, select the **Start Time** and the **End Time** from the combo boxes.
- To define the **Break Hours**, select the **Start Time** and the **End Time** from the combo boxes.
- Click **Submit.**
- If you want to use a Customized Time Table, click the tab **Customized Time Table-1**.
- Define the Working days and the Day Time and Break Hours - start time and end time - for each Working day.
- Click **Submit.**

  If you want to use the other customized time table, click their tabs and follow the same instructions as above.

Operator

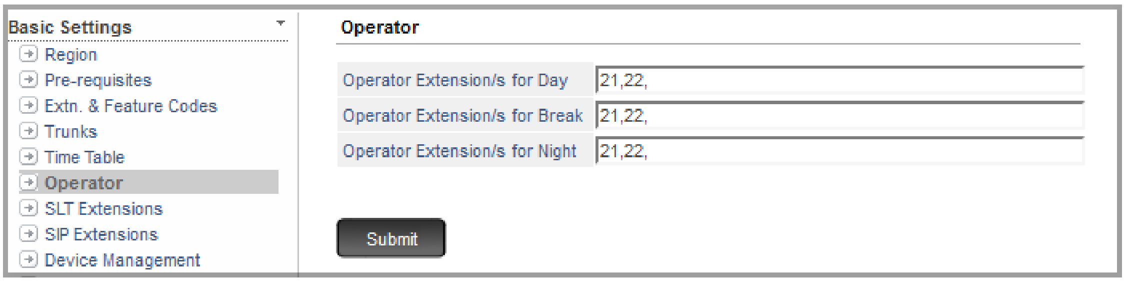
In the context of a PBX, users understand the term 'Operator' as a person who handles multiple simultaneous calls and functions as the link between callers and called parties.

For the PBX however, an 'Operator' is a Routing Group, i.e. a group of extensions to which calls made by extensions by dialing '9' are to be landed. This also includes Direct Inward Dialing calls on trunks during which the caller dials '9'.

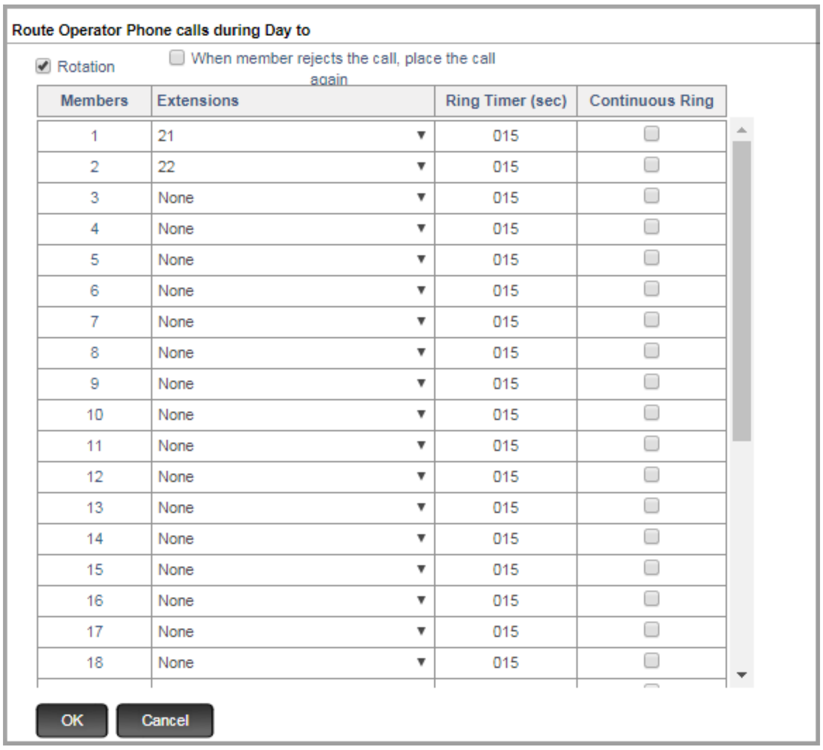
Depending on the amount of call traffic to be managed, there may be more than one Operator extension. Also, different Operator extensions may be assigned according to the time of the day. For instance, during the Day (Working hours) call may be landed on the extension of the Receptionists/Front Desk Personnel. During the Night (Non-working hours), calls may be landed on the extension of the Security Personnel.

To meet this requirement, SARVAM UCS allows you to configure the Operator for the Day (Working hours), Break hours and for the Night (Non-working hours).

- Click **Operator**.



- Double click **Operator Extension/s for Day**. The extensions to be selected in the routing group for Operator for the Day time appear in a new window.



- Select the **Extensions** which are to be used as Operator. These may be a SLT, SIP extension, Virtual extension or Voice Mail Auto Attendant Profile.

- For each extension you have selected, set the **Ring Timer**. This timer defines the time for which the extension, on which the call lands, should ring. Default: 015 seconds.

- For each extension you have selected, you may set **Continuous Ring**, if you want the extension to ring till the incoming call is answered. Default: Disabled.

  When Continuous Ring is selected, the first extension in the Operator group you have created will continue to ring, even as the system hunts for other extensions in the group to land the call. If the call still remains unanswered, the system will return the call to the first extension once again. This check box has no relevance, if there is only one Operator extension.

- Enable **Rotation**, if you have selected more than one extension as Operator. Default: Disabled.

  When you enable Rotation, each new call lands on the subsequent extension[15] in the group next to the one that received the last call. This ensures equal distribution of incoming calls to all the destination extensions in the Operator extension group.

  Rotation has no relevance if the Operator group has only one member extension.
- By default, **When member rejects the call, place the call again** is disabled. Therefore, if any SIP extension user rejects an incoming call, the system will not place the same call on this extension again while checking the routing group to land the call. You may enable this check box, if required.

---

15. The extension next to the one that received the previous call.

If this check box is disabled and you have selected the Continuous Ring 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.

- Click **OK**. All the extension numbers you selected will appear in **Operator Extension/s for the Day**. The extension numbers appear in the sequence you selected, separated by commas.
- Repeat the same steps to select **Operator Extension/s for Break** and **Operator Extension/s for Night**.
- Click **Submit**.

For each extension you have selected as Operator for the Day, Break and Night,
- *define the Class of Service*[16]*.*
- *set Toll Control*[17]*.*
- *select Outgoing Trunks*[18]*.*
- *set Priority*[19]*.*

You can set Class of Service, Toll Control, Outgoing Trunks and Priority for the Operator extension when you configure the parameters of different extension types: SLT and SIP Extensions.

16. *These are features allowed/denied to the Operator extension.*
17. *The type of calls to be allowed/denied to the Operator extension during the day, during break and at night: local calls, regional calls, national calls, international calls, limited calls.*
18. *The trunks through which outgoing calls are to be routed during the day, break and night.*
19. *Each extension of 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.*

SLT Extensions

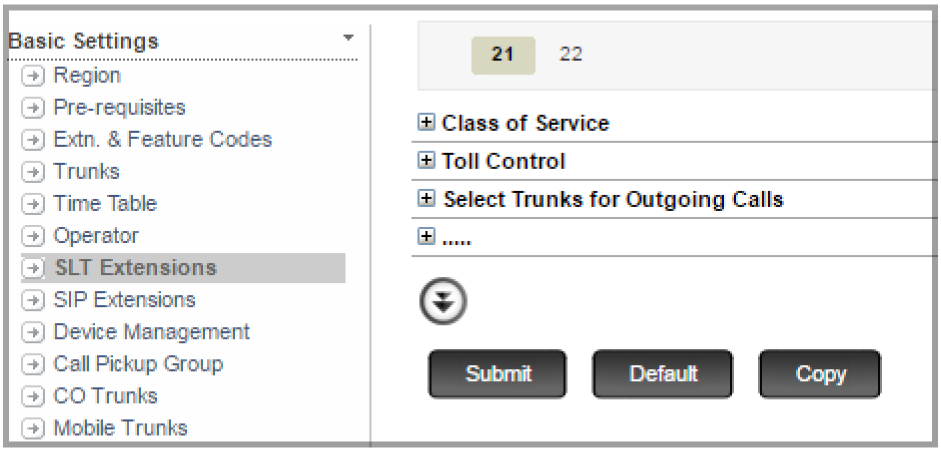
ETERNITY NENXIP50[20]supports 2 Single Line Telephone (SLT) extension ports, depending on the configuration you are using.

The number of SLT extensions available to you for configuration depends on the number of SLT ports supported by your configuration of ETERNITY NENX and the number of SLT ports you have specified on the "Pre-requisites" page.

To configure,

- Click **SLT Extensions**.

  On this page,



More: Click this button to view all parameter links on the page.

Less: Click this button to view only the essential parameter links on the page.

Expand: Click to expand a link to display all parameters under the link.

Collapse: Click to collapse a link. Hides all parameters under the link.

Copy: Click to copy the parameters under a specific link of a page to the same link under any other extension page.

Settings: Click to configure the settings of a parameter further.

···· **More link:** Click to view all additional parameters on the page.

To configure another SLT extension, click the SLT Extension name (number) tab.

To save the settings, click **Submit**.

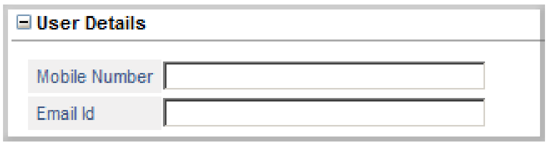
To assign default values to all the parameters of the SLT extension, click **Default**.

20. *ETERNITY NENX312 and ETERNITY NENX416 supports 12 and 16 SLT Ports respectively.*

To copy all SLT extension parameter values to another SLT extension, click **Copy**.

Follow the instructions provided below to configure the SLT extensions.

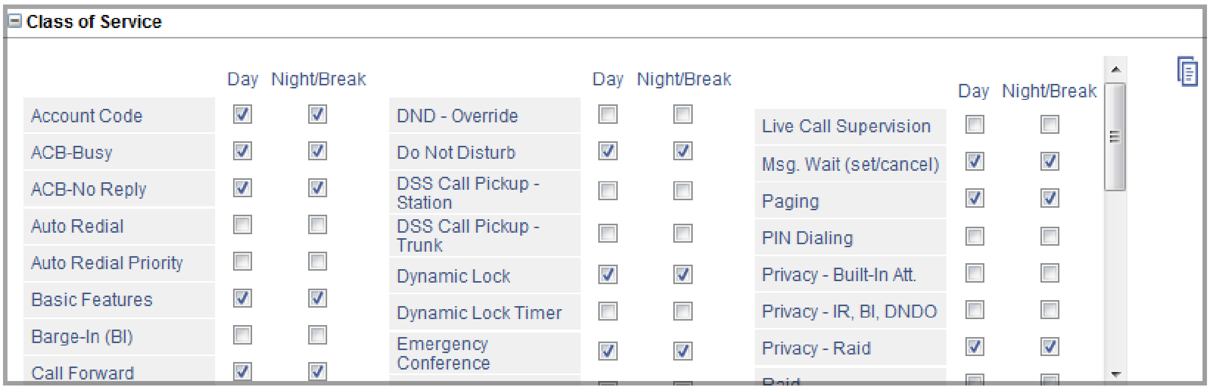
User Details



- Enter the **Mobile Number** of the extension user. The Number can be a maximum of 16 digits.Default: Blank.
- Enter the **Email ID** of the extension user. The Email ID can be a maximum of 64 characters. Default: Blank.
- Click **Submit.**

These parameters are used by the SMS Server application. To know more about this feature, see "SMS Server".

Class of Service

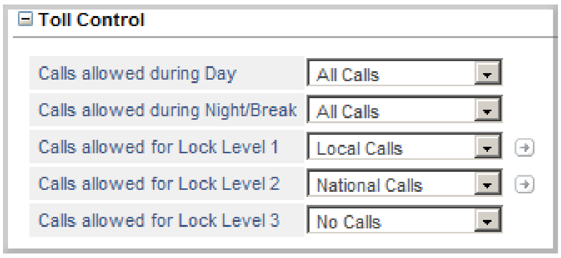


- Click **Class of Service** to expand.

  Define the Class of Service for the SLT extension for **Day** time and **Night/Break** time.
- To allow a feature, select the check box of the feature.
- To deny a feature, clear the check box.
- Click **Submit.**

  See "Class of Service (CoS)" to know more.

Toll Control



- Click **Toll Control** to expand. Set the desired Toll Control for the SLT extension for the Day and Night/ Break.
- Select the type of **Calls Allowed during Day**: All Calls, No Calls, Local Calls, Regional Calls, National Calls, International Calls, Limited Calls 1, Limited Calls 2, Limited Calls 3. Default: No Calls.
- Select the type of **Calls Allowed during Night/Break**: All Calls, No Calls, Local Calls, Regional Calls, National Calls, International Calls, Limited Calls 1, Limited Calls 2, Limited Calls 3. Default: No Calls.

The Toll Control levels on this page are based on the allowed and denied number lists of Local, Regional, National, International, and Limited Call numbers you configured in the "Toll Control - Allowed Denied Numbers".

- If you have not configured the allowed and denied number list for the Type of Calls you selected as Toll Control or if you want to add to the existing list, you may do it now.
  - Click the **Settings** icon.
  - The Toll Control - Allowed-Denied Numbers page will open in a new window.
  - Configure the Allowed and Denied Numbers.
  - Click **Submit**.
  - Close the window after you have configured the list.

The Settings icon does not appear when you select **No Calls** or **All Calls** option.

See "Toll Control" to know more about this feature.

Toll Control for Dynamic Lock

Dynamic Lock allows extension users to change the Toll Control Levels (Calling Permissions) of their extensions on their own by dialing a code. SARVAM UCS supports Toll Control Levels 0 to 3 for Dynamic Lock.

For each Toll Control Level from 0 to 3, you must assign 'Call Privilege'[21]. For each Call Privilege, you need to configure the corresponding number strings to be allowed and number strings to be denied. See "Dynamic Lock" to know more about this feature.

21. *The Call Privilege types are: No Calls, Local Calls, Regional Calls, National Calls, International Calls and Limited Calls.*

- Select the call privilege for **Calls allowed for Lock Level 1**: No Calls, Local Calls, Regional Calls, National Calls, International Calls, Limited Calls, All Calls. Default: No Calls.
- Select the call privilege for **Calls allowed for Lock Level 2**: No Calls, Local Calls, Regional Calls, National Calls, International Calls, Limited Calls, All Calls. Default: No Calls.
- Select the call privilege for **Calls allowed for Lock Level 3:** No Calls, Local Calls, Regional Calls, National Calls, International Calls, Limited Calls, All Calls. Default: No Calls.

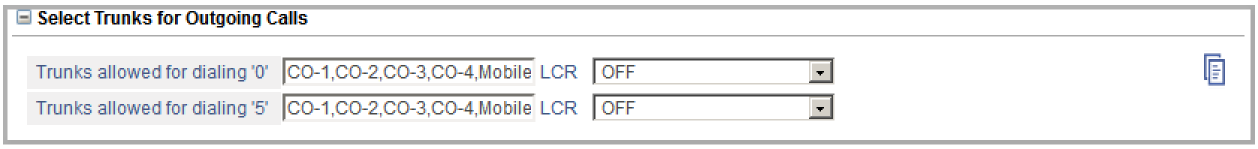
The Lock Levels on this page are based on the allowed and denied number lists of Local, Regional, National, International and Limited Call numbers you configured in the "Toll Control - Allowed Denied Numbers".

- If you have not configured the allowed and denied number list for the calls allowed/denied for the selected Lock Level or if you want to add to the existing lists, you may do it now.

  The **Settings** icon does not appear when you select **No Calls** or **All Calls** option.

  - Click the **Settings** icon.
  - The Toll Control - Allowed-Denied Numbers page will open in a new window.
  - Configure the Allowed and Denied Numbers for Local, Regional, National, International, and Limited Calls.
  - Click **Submit**.

Select Trunks for Outgoing Calls



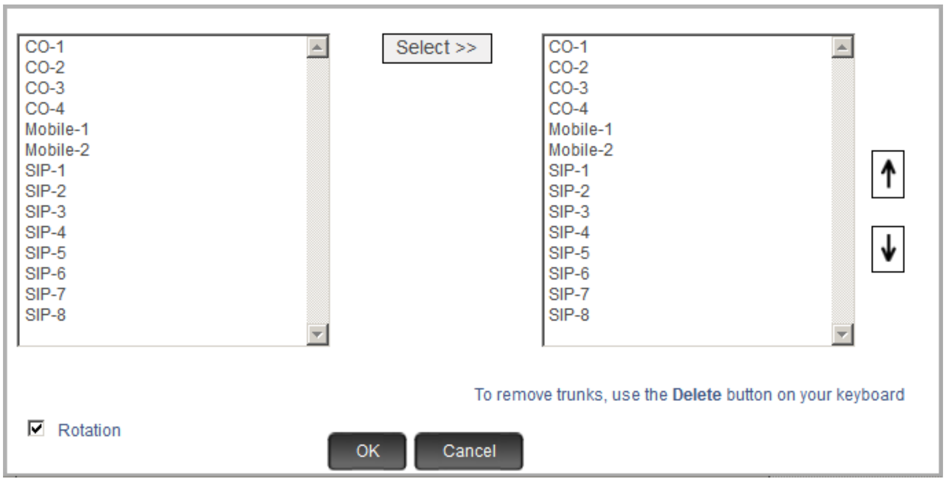
Outgoing calls (to external numbers) are made by dialing Trunk Access Codes (TAC). Default: 0 (TAC 1) and 5 (TAC 2). You may configure TAC 3 to TAC 6 as desired.

For each TAC, you need to select the Outgoing Trunks. All external calls made by dialing a particular TAC will be routed through the outgoing trunks you selected for that TAC.

You can also apply Least Cost Routing logic on the selected trunks, so that SARVAM UCS will route the outgoing call through the trunk that costs the lowest for the call.

- Click **Select Trunks for Outgoing Calls** to expand.
- Select **Trunks allowed for dialing '0'**. The outgoing call will be routed through the selected trunks when the extension user dials TAC '0'.

- Double-click the field. A multiple selection box opens.



On the left, the trunks appear with their names (if configured in "Trunks") and port numbers in a sequence, starting with CO trunks, followed by Mobile trunks and SIP trunks.

If you have not assigned any names to the trunks, they will appear with their default names (CO, MOBILE, SIP) and port numbers.

- To select a trunk, place your cursor on the desired trunk, and click **Select>>**.

  OR

- Press the **ctrl** key and click the left mouse button to select multiple trunks.
- You may change the sequence of the trunks you selected, if required, using the **Up** and **Down** arrow buttons on the right display box.
- To remove (delete) any trunks from the list, press the **Delete** key of your keyboard.
- You may enable **Rotation**, if you have selected more than one trunk. Default: Enabled.

  When you enable Rotation, each new outgoing call is routed through the subsequent trunk in the group[22]. This ensures equal distribution of outgoing call traffic on all trunks.

  When Rotation is disabled, calls are routed through the first trunk in the group. If this trunk is busy, the call is routed to the next trunk in the group.

  Rotation has no relevance if only one member trunk is selected.

- Click **OK**. The multiple selection box closes.

  All the trunks appear in Trunks allowed for dialing '0', in the sequence you selected, separated by commas.

- To apply **Least Cost Routing** on the Trunks allowed for 0 dialing, select the desired LCR method from the combo box:

22. *The first call through the first trunk, the second through the second, the third through the third trunk, and so forth. Thus each new call is routed through the trunk next to the one that routed the previous outgoing call.*

- **Number Based:** Choose this option if the service providers of the trunks you selected offers different tariffs according to area or distance, or phone numbers dialed.
- **Time Based:** Choose this option if the service providers of the selected trunks offer a different tariff according to the time of the day.
- **Time + Number Based:** Choose option if the service providers of the selected trunks offer different tariffs according to the time of the day as well as area/distance.
- **Service Provider Based:** Choose this option if the same service providers of the selected trunks offer different rates for calls made to numbers within their own network and for calls made to numbers of another Service Provider's network.

If you do not want to apply LCR, select OFF. The **Settings** icon does not appear if the LCR is OFF.

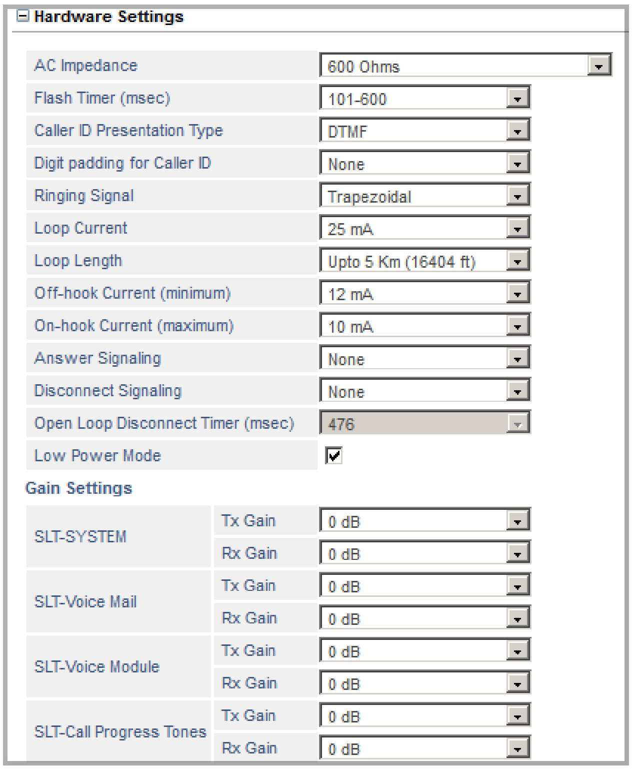
Configure LCR method that you selected for the trunk group.

- To configure Least Cost Routing method you selected, click the **Settings** icon. A new window opens.
  - Configure the LCR method. See “Least Cost Routing (LCR)” under Advanced Settings for instructions.
  - Click **Submit**. The window closes.
- Select **Trunks allowed for dialing ‘5’**. The outgoing call will be routed through the selected trunks when the extension user dials TAC ‘5’. Follow the same steps as described above.
  - Double-click the field. A multiple selection box opens.
  - Select trunks, placing your cursor on the desired trunk, and clicking **Select>>**.
  - Change the sequence of the trunks you selected, if required, using the **Up** and **Down** arrow buttons on the right display box. Delete trunks from the ones you have selected, if required.
  - Enable **Rotation**, if you have selected more than one trunk. Default: Disabled.
  - Click **OK**.

    All the trunks appear in Trunks allowed for dialing ‘5’, in the sequence you selected, separated by commas.
  - Choose a **Least Cost Routing** method, if you want to apply it on the trunks. Default: OFF.
  - Configure the Least Cost Routing method, by clicking **Settings** icon.

  Similarly, you will be able to select outgoing trunks for Trunk Access Codes 3, 4, 5 and 6, if you have assigned access codes to them in “Extension and Feature Codes”.
- Click **Submit**.

Hardware Settings



- Click **Hardware Settings** to expand.
- Set **AC Impedance** for the SLT port to match the AC impedance of the telephone instrument connected to it. Default: 600Ω.

  However, SARVAM UCS allows you to connect instruments with AC impedance other than 600Ω.
- Set the **Flash Timer (msec)** as per your requirement.

  Flash is generally required to enable the SLT user to use features like Hold, Transfer, etc. Flash can be dialed using the Flash key (if available) on the SLT or by tapping the hook switch.

  Flash is breaking of Loop Current for a specific time period. SARVAM UCS considers that an extension has dialed flash, if it detects breaking of the loop current for duration of the Flash Timer. The Flash timer is configurable. The range of this timer is from 70 (minimum) to 900 (maximum) milliseconds. By default, Flash Timer is set to 101-600 milliseconds.

  You can configure the Flash Timer according to the Flash Timer supported by the SLT instrument. If the SLT instrument breaks the loop current for more than the programmed Flash Timer, the system will consider it as a call disconnection.
- Select the appropriate **Caller ID Presentation Type** according to the CLIP Type supported by the telephone instrument connected to the SLT port.

  SARVAM UCS supports 3 signaling protocols for CLI on the SLT port: DTMF, FSK-V.23, and FSK-BellCore. Default: DTMF.

  Select None to disable CLIP on the SLT port.

- Define the **Digit padding for Caller ID** according to number of digits supported by the telephone instrument connected to the SLT port.

  Certain SLT instruments that support CLI require a minimum number of digits in the calling party's number to be able to identify and display it. The Minimum Caller ID Digits are the number of zeroes to be added to the calling party's number before displaying it on the called party's instrument.

- Select the appropriate **Ringing Signal** to match the type of ringing current expected by the telephone instrument connected to the SLT port.

  You may select from these Ring Types: Sinusoidal, Trapezoidal, Low Sinusoidal, Low Trapezoidal. Default: Trapezoidal.

- Increase or decrease the **Loop Current (mA)** according to the Loop Length[23], i.e. the length of the telephone cable between the wall jack (into which the SLT telephone instrument is plugged) and the SLT port.

  The SLT Port provides the telephone instrument connected to it Loop Current of 25, 30, 35 and 40 mA. Default: 25mA, which is sufficient to support Loop Length of 1 kilometer.

- Select the **Loop Length** as **Upto 5 Km** or **Above 5 Km** according to your installation scenario. The Loop Length is the distance between the Central Office and the telephone instrument connected to the SLT port.

- Set the value of the **OFF-Hook Current (minimum)** according to the current drawn by the SLT instrument connected to the SLT port.

  SARVAM UCS supports OFF-Hook detection for all types and brands of SLT instruments by providing for configurable values for threshold current for OFF-Hook detection: 10mA, 12mA, 14mA, 16mA and 18mA. Default: 12mA.

  When a SLT instrument draws current equal to or greater than the configured threshold value of current for OFF-Hook detection, SARVAM UCS will consider the SLT instrument as OFF-Hook and will offer dial tone to the SLT.

- Set the value of **ON-Hook Detection (maximum)** according to the current drawn by the SLT instrument connected to the SLT port.

  SARVAM UCS detects ON-Hook state of for all types and brands of SLT instruments by providing for configurable values for threshold current for ON-Hook detection: 10mA, 12mA, 14mA, 16mA and 18mA. Default: 10mA.

  When a SLT instrument draws current equal to or lower than the configured threshold value of current for ON-Hook detection, SARVAM UCS will consider the SLT instrument as ON-Hook and routes calls to this port.

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23. *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. Change the Loop Current according to the Loop Length of the SLT instrument connected to the port.*

SLT instruments also vary by the level of current they draw during the normal 'idle' state and when Flash is dialed[24] (the simulated idle state). So, when the Flash key of a 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 (i.e. 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.

- Select the appropriate **Answer Signaling**[25] option**:** Polarity Reversal or None. Select Polarity Reversal to generate Answer Signaling on the SLT port.

  Select None, if no Answer Signaling is to be generated on the SLT port. Default: None.

- Select the **Disconnect Signaling**[26] on the SLT port as Polarity Reversal or Open Look as appropriate. Select None if Disconnect Signaling is not generated on the SLT port. Default: None.

  - **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 configured 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.

- Set the **Open Loop Disconnect Timer (msec)**, if you selected Open Loop Disconnect as Disconnect Signaling 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 this timer is: 68 to 952 milliseconds. Default: 476 msec.

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24. *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.*

25. *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 a SLT port, during an outgoing call is made from that SLT port to any other port - CO/Mobile/SIP- 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.

26. *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 a SLT port, during an outgoing call is made from that SLT port to any other port - CO/Mobile/SIP - when the called party (remote party) disconnects, i.e. goes ON Hook, the Public Network provides a Disconnect Signal to trunk port indicate call disconnection. This signal can be generated 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.

- **Low Power Mode**: SARVAM UCS supports Low Power Mode option, using which the SLT instruments will consume less power in the On-Hook state. You may disable this check box, if required. Default: Enabled.

  SARVAM UCS also supports Thermal Shutdown. That is, the SLT instrument will stop functioning if the thermal temperature of the SLIC increases above the threshold level. To indicate the thermal shutdown, the respective SLT Port number will appear red in the "Extension and Feature Codes" page of the Jeeves. You must restart the system to resume functioning of the phone.

- **Gain Settings:** Configure the following Transmit and Receive Gain Settings.

  - **SLT - System (Tx-Gain and Rx-Gain)**: Configure the Gain Settings that you want the system to apply on the SLT 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: +6dB to -12dB and Rx Gain +6dB to -12dB. Default: 0 dB

  - **SLT- Voice Mail (Tx-Gain and Rx-Gain):** Configure the Gain Setting that you want the system to apply on the SLT port when the SLT user is accessing the Voice Mail System. Valid Range for Tx Gain: +6dB to -12dB and Rx Gain +6dB to -12dB. Default: 0 dB

  - **SLT - Voice Module (Tx-Gain and Rx-Gain):** Configure the Gain Settings that you want the system to apply on the SLT port when the system plays Voice Module to the SLT user. Valid Range for Tx Gain: +6dB to -12dB and Rx Gain +6dB to -12dB. Default: 0 dB

  - **SLT - Call Progress Tones (Tx-Gain and Rx-Gain):** Configure the Gain Setting that you want the system to apply on the SLT port while playing Call Progress Tones. Valid Range for Tx Gain: +6dB to -12dB and Rx Gain +6dB to -12dB. Default: 0 dB

  - **SLT - SLT (Tx-Gain and Rx-Gain)**: Configure the Gain Setting that you want the system to apply on the SLT when the FXS Port is connected to any other FXS Port of the system during an incoming or outgoing call. Valid Range for Tx Gain: +6dB to -12dB and Rx Gain +6dB to -12dB. Default: 0 dB.

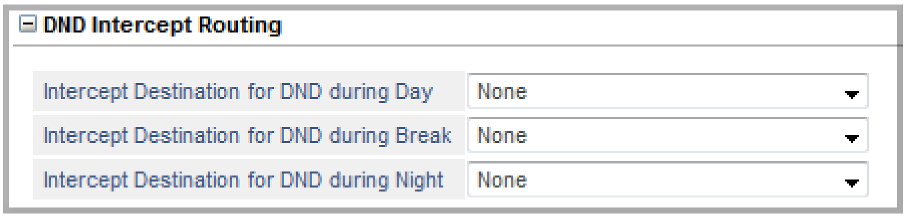
  - **SLT - CO (Tx-Gain and Rx-Gain)**: Configure the Gain setting that you want the system to apply on the SLT when the FXS Port is connected to CO Port of the system during an incoming or outgoing call. Valid Range for Tx Gain: +6dB to -12dB and Rx Gain +6dB to -12dB. Default: 0 dB.

  - **SLT - SIP (Tx-Gain and Rx-Gain)**: Configure the Gain Setting that you want the system to apply on the SLT when the FXS Port is connected to any SIP Trunk or SIP Extension of the system during an incoming or outgoing call. Valid Range for Tx Gain: +6dB to -12dB and Rx Gain +6dB to -12dB. Default: 0 dB.

  - **SLT - Mobile (Tx-Gain and Rx-Gain)**: Configure the Gain Setting that you want the system to apply on the SLT when the FXS Port is connected to any Mobile Port of the system during an incoming or outgoing call. Valid Range for Tx Gain: +6dB to -12dB and Rx Gain +6dB to -12dB. Default: 0 dB.

  - Click **Submit**.

DND Intercept Routing



When the DND set extension users want their calls to be attended by someone even if DND is set, they must select an Intercept Destination. Incoming calls landing on extension that has set DND will be routed to the Intercept Destination. This destination can be the users own mailbox or another extension (SLT, SIP). See "Do Not Disturb (DND)" for more details.

- Select the **Intercept Destination for DND during Day**. You may select:
  - None
  - any SLT Extension
  - any SIP Extension
  - Voicemail

  Default: None

  Similarly, you may select **Intercept Destination for DND during Break** and **Intercept Destination for DND during Night**.

SMDR Storage

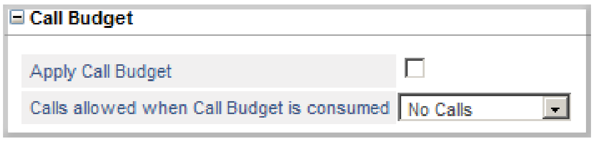
SARVAM UCS offers Station Message Detail Recording (SMDR) that enables you to record the details of Internal, Incoming (IC) and Outgoing (OG) calls made from/to all its extensions. To obtain SMDR as a report, you must enable SMDR Storage, and set filters. See "Station Message Detail Recording (SMDR)" to know more.

- Click **SMDR Storage** to expand.
- Select the type of internal calls to be stored from the combo box **Store Internal Calls**. You can select from the following options:
  - **Made/received by this extension**: the system will store all calls made to and received from this extension.
  - **Made by this extension**: the system will store outgoing calls made from this extension.
  - **Received by this extension**: the system will store only incoming calls from other extensions.
  - **Never**: the system will not store internal calls.
- To store details of incoming calls from external numbers, select **Store Incoming Calls**. Default: selected.

- To store details of outgoing calls made by the extension user to external numbers, select **Store Outgoing Calls**. Default: selected.

- Click **Submit.**

Call Budget



Using the Call Budget feature, you can allot a 'budget' limit for outgoing calls made by the extension. See “Call Budget on Extension” for more information. If you want to enable this feature on this extension,

- Click **Call Budget** to expand.

- Select the **Apply Call Budget** check box.

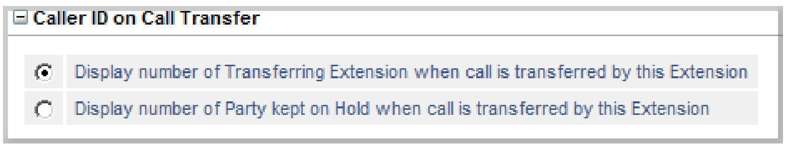
- You may define the calling permission for the extension, after it has consumed the call budget allotted to it. Select the type of **Calls** to be **allowed when Call Budget is consumed** from the following:
    - No Calls
    - Local Calls
    - Regional Calls
    - National Calls
    - International Calls
    - All Calls

  Default: No Calls.

*Click the Settings icon to view the **Toll Control - Allowed-Denied Numbers** page. This icon does not appear when you select **No Calls** or **All Calls** option.*

- Click **Submit.**

Caller ID on Call Transfer



This parameter is related to the CLIP feature. It allows you to choose whether the system should display the CLI of the ‘Held Party’ or the CLI of the ‘Transferring Party’ to the transfer destination extension while the call is being transferred.

See the feature description for “Calling Line Identification and Presentation (CLIP)” to know more.

- Click **Caller ID on Call Transfer** to expand.

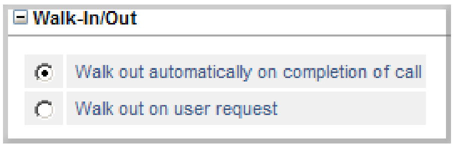
- Select the radio button of the desired option:

- Display Number of Transferring Extension when call is transferred on this extension.

- Display Number of Party kept on Hold when call is transferred on this extension.

- Click **Submit**.

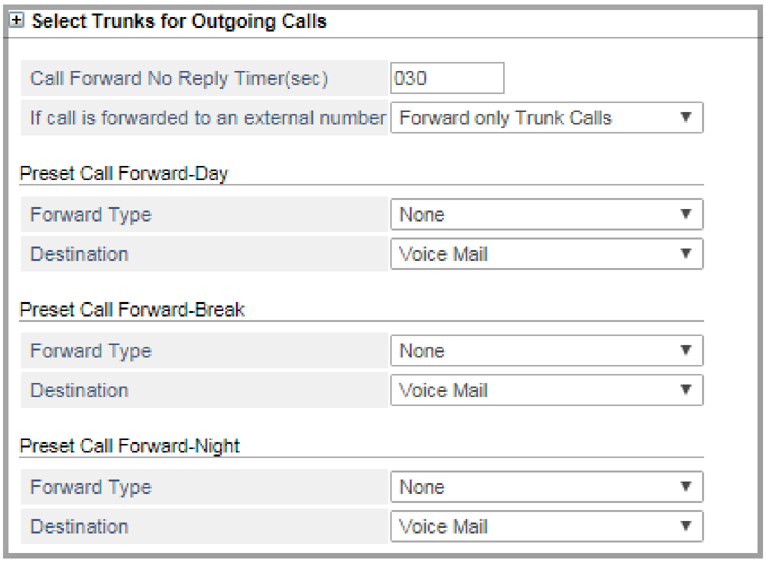
Walk-In/Walk-Out



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, where the extension user is automatically logged out after a call. ii) Walk-In until Logout, where the extension user remains logged on until s/he manually walks out or a second user walks into the same extension. To know more about this feature, see “Walk-In Class of Service”.

- Click **Walk-In/Walk-Out** to expand.
- Select the radio button of the type of Walk-Out mode you want to assign to the extension:
  - **Walk-Out automatically on completion of call:** Select this option, if you want to assign One-Call per Walk-In to the extension.
  - **Walk-Out on user request:** Select this option, if you want to assign Walk-In until Logout to the extension.
- Click **Submit**.

Call Forward



- Click **Call Forward** to expand.

- Set the **Call Forward No Reply Timer (sec)** to the desired value, if required. The range of this timer is 001-255. Default: 030 seconds.

  Call Forward No Reply Timer signifies the duration for which the system will wait for an extension to answer an incoming call, before forwarding the call to the programmed destination as Call Forward-No Reply. By default the Timer is set to 30 seconds. This timer is applicable for both Call Forward and Preset Call Forward. Refer “Call Forward” and “Preset Call Forward” to know more.

- Select the type of calls to be forwarded **If Call is forwarded to an external number**. You may select from the following options:
  - Forward only Internal calls
  - Forward only Trunk calls
  - Forward all calls (internal as well as trunk calls)

  Default: Forward only Trunk calls.

  This parameter is relevant for the features “Call Forward” and “Mobility Extension”.

- Select the **Forward Type** for **Preset Call Forward - Day**. You may select:
  - None
  - When Busy
  - When No Reply
  - When Busy or No Reply

  Default: None

- Select the **Destination** to which the calls are to be forwarded for Preset Call Forward - Day.

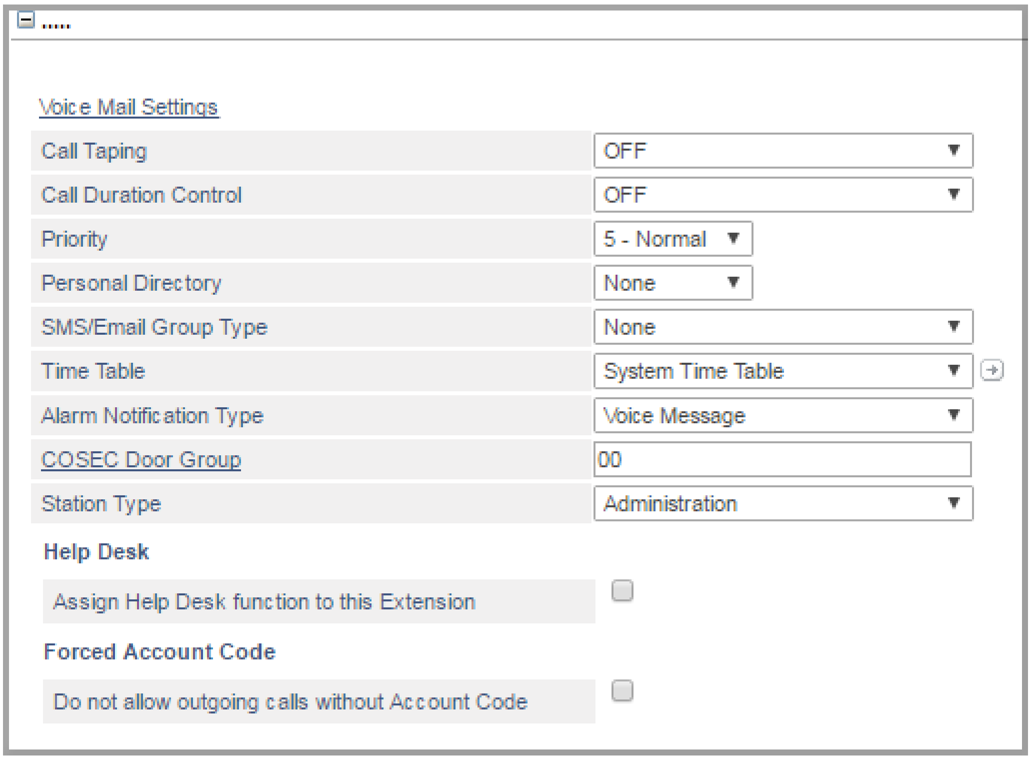
  Follow the instructions in steps 4 and 5 to configure **Preset Call Forward - Break** and **Preset Call Forward - Night**.

  See “Preset Call Forward” for more details.

- Click **Submit**

More Features

- Click **More...** link to expand.



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".

Call Taping

To use the "Call Taping" feature on the extension,

- Click **Call Taping** to expand.

- Select the type of **Call Taping** to be applied from the options:
  - OFF
  - Apply as per profile 1
  - Apply as per profile 2
  - Apply as per profile 3
  - Apply as per profile 4

  If you do not want to apply Call Taping, select OFF. Default: OFF.

- You can configure the Call Taping Profile number you selected for the extension. To do this,

  - Click the **Settings** icon. The page opens in a new window, displaying the parameters of the profile you selected for the extension. This icon does not appear when Call Taping is **OFF**.

  - In the new window,

- Select the **Tape Internal Calls** check box, if you want internal calls made and received by the extension to be taped.

- Select the **Tape Incoming Calls received without CLI** check box if you want incoming calls without CLI to be taped. Default: Disabled.

- In **Tape Incoming Calls received from following numbers**, type the numbers of the Incoming Calls that must be taped. You may type as many as 99 numbers.

- In **Tape Outgoing Calls made to following numbers,** type the external numbers that you want the system to tape. You may type as many as 99 numbers.

- Click **Submit**.

See "Call Taping" to know more.

Call Duration Control

To enable the "Call Duration Control (CDC)" feature on the extension,

- Click **Call Duration Control** to expand.

- Select the type of **Call Duration Control** to be applied from the options:
  - OFF
  - Apply as per CDC profile 1
  - Apply as per CDC profile 2
  - Apply as per CDC profile 3
  - Apply as per CDC profile 4

  If you do not want to apply Call Duration Control, select OFF. Default: Apply as per CDC profile 1.

- You can configure the Call Duration Control Profile number you selected for the extension. To do this,

  - Click the **Settings** icon. The page opens in a new window. The page displays the parameters of the profile you selected for the extension. This icon does not appear when Call Duration Control is **OFF**.

  - In the new window,

    - Enable **Apply CDC to Internal Calls,** if CDC is to be applied on internal calls. Default: Enabled.

    - Enable **Apply CDC for Incoming Calls received from trunk**, if CDC is to be applied on incoming external calls. Default: Enabled.

    - Enable **Apply CDC for Outgoing Calls made from trunk**, if CDC is to be applied to outgoing external calls. Default: Enabled.

    - If required, change the **CDC Timer** to the desired duration. The range of the timer is 0001 to 9999 seconds. Default: 160 seconds.

    - Enable the **Disconnect Call after CDC Timer** check box if you want calls to be disconnected on the expiry of the CDC Timer. Default: Enabled.

- In the **Apply CDC for calls matching with numbers** column, type the external numbers on which you want to apply CDC. You can enter as many as 99 numbers.

- In the **Do Not Apply CDC for calls matching with numbers** column, enter the numbers which you want to be exempt from CDC.

- Click **Submit**.

Priority

- Select a **Priority** from 1 to 9 for the SLT extension. Default: 5-Normal.

  Each extension of 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”.

Personal Directory

- Select a Personal Directory number from 01 to 50 that you want to assign to the SLT extension. Default: None.

  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. The Personal Directory is necessary for using the features “Abbreviated Dialing” and “Dial By Name”.

  When a Personal Directory is assigned to a SLT, it must also be configured first.

  The Personal Directory can be configured also by the extension user. Refer the topic “Abbreviated Dialing” for instructions on configuring the Personal Directory.

SMS/Email Group Type

- Select the **SMS/Email ID Group Type**, you want to assign to the extension user. Default: Blank.

  See “SMS/Email Group” under “SMS Server” to know more.

Time Table

- Select a Time Table for the SLT extension. Default: System Time Table.

  If you have not configured Time Table, you may do so now, by clicking the Time Table link. Define the Working days, and the start and end time of the Working hours and Break hours for each working day.

  See “Time Tables” to know more.

Alarm Notification Type

To configure this parameter and for the feature description, see “Alarms” to know more.

- Select the **Alarm Notification Type**, the system should use when the user answers an Alarm Call. Your options are:

- **Voice Message:** The extension user is played a message recorded in the Voice Module on answering the alarm call.
- **Music-on-Hold:** The extension user is played music-on-hold on answering the alarm call.
- **Voice Mail:** The extension user is greeted by the VMS on answering the alarm call.
- **Routing Group:** The alarm call is routed to the group of desired extensions configured as the Operator, so that the alarm request can be served. See “Operator” for instructions.

Default: Voice Message.

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

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 this is not an Operator extension, select the **Station Type** for the SIP Extension as **Administration**.

Help Desk

Configure this parameter if you want to define the extension as a “Help Desk”.

- Select the **Assign Help Desk function to this Extension** check box. Default: Disabled.

Forced Account Code

Configure this parameter, if you want to apply the feature Forced Account Code on the extension. When Forced Account code is enabled, the system will allow the extension user to dial an external number only after the user has entered the Account Code. To know more, see “Account Codes”.

- Select the **Do not allow Outgoing Calls without Account Code** check box.
- Click **Submit.**

Copy Parameter Values

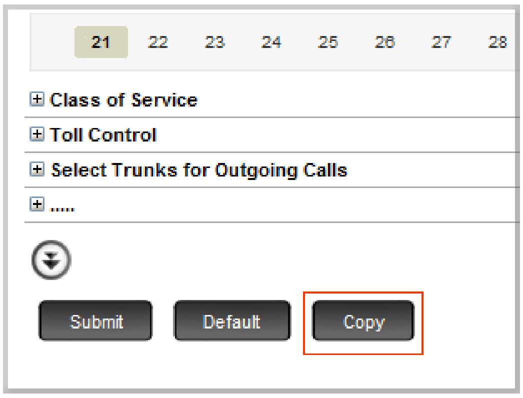
After you have finished configuring this SLT Extension, you may configure the next SLT Extension. To do so,

- Click the tab of the next SLT Extension.
- Follow the same steps as described above to configure another SLT Extension.

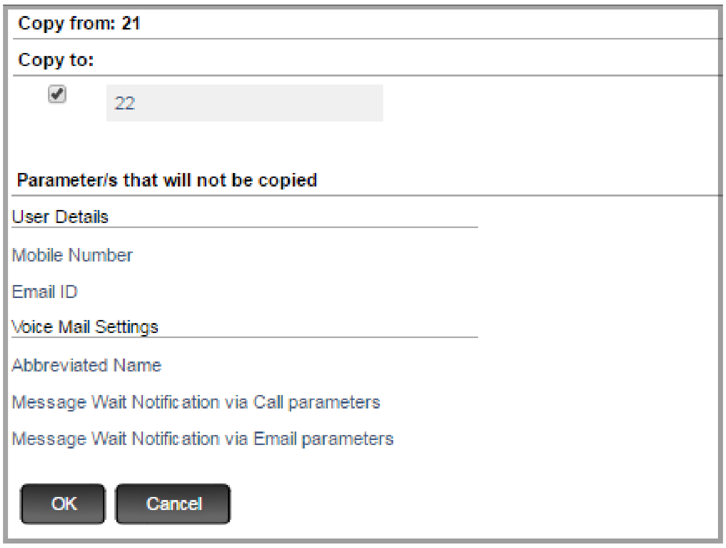
OR

You may use the **Copy** button at the bottom of the page to apply the same SLT Extension settings you configured for the current extension to another SLT Extension. To do so,

- Click **Copy**.



The Copy page opens in another tab.



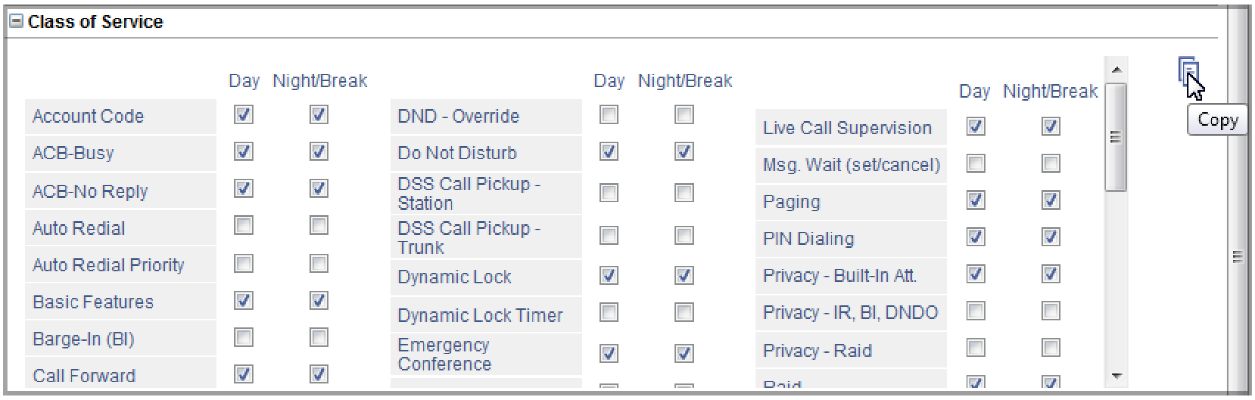
The copy page displays the number and/or name of the current extension in **Copy from:**. It also lists the numbers and/or names of the extensions to which you can copy the extension settings under **Copy To:**.

- Select the respective **Copy to** check boxes of the SLT Extensions to which you want to apply the current SLT Extension settings.
- Click **OK**.

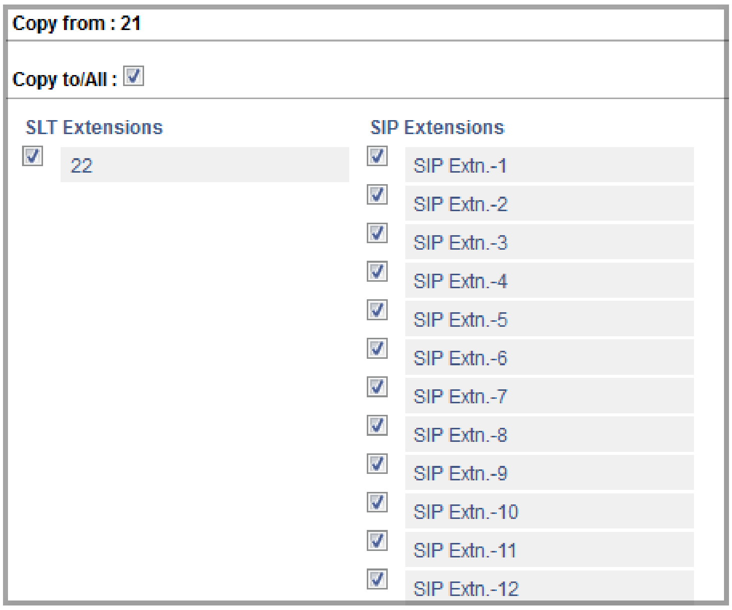
All the parameter values of the current SLT Extensions except those listed below **Parameter/s that will not be copied**, will be applied to the selected SLT Extensions.

You may also copy the parameter values of a specific link under current SLT Extension to the same link under any other extension page. To do so,

- Click the **Copy** icon besides the desired link.



The Copy page opens in another tab.



The Copy page displays the number and/or name of the current extension in **Copy from:**. It also lists the numbers and/or names of the extensions to which you can copy the extension settings under **Copy To/All:**.

- Select the **Copy to/All** check box, to apply the link settings to the same link under all other Extensions.

  OR

  You may also apply the link settings to the desired Extensions only, by selecting the respective check boxes.

- Click **OK**.

SIP Extensions

SIP Extensions function like any normal SLT extension of the System, allowing you to make and receive calls to any extension user of the System and to external numbers over PSTN, GSM, and VoIP lines, depending on the "Logical Partition" configured in the System.

SIP Extensions are a licensed feature. To know more, refer the "License Management" topic.

SARVAM UCS supports up to 50 SIP extensions. You can:

- Connect SPARSH VP248, the Extended IP Phone supplied by Matrix. To connect SPARSH VP248 with SARVAM UCS, see "Connecting SPARSH VP248 as Extended SIP Extension".
- Connect SPARSH VP310, the Executive IP Phone supplied by Matrix. To connect SPARSH VP310 with SARVAM UCS, see "Connecting SPARSH VP310 as Extended SIP Extension".
- Connect SPARSH VP510, the Premium IP Phone supplied by Matrix. To connect SPARSH VP510 with SARVAM UCS, see "Connecting SPARSH VP510 as Extended SIP Extension".
- Connect SPARSH VP330, the Intuitive Touch Screen Extended IP Phones supplied by Matrix. To connect SPARSH VP330 with SARVAM UCS, see "Connecting SPARSH VP330 as Extended SIP Extension".
- Connect Matrix Extended SPARSH VP710 as SIP Extensions. To connect SPARSH VP710 with SARVAM UCS, see "Connecting Extended SPARSH VP710 as Extended SIP Extension".
- Connect SPARSH VP210, the Entry Level IP Phone supplied by Matrix. To connect SPARSH VP210 with SARVAM UCS, see "Connecting SPARSH VP210 as Extended SIP Extension".
- 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 as SIP Extensions, for configuration instructions see "Configuring MATRIX VARTA WIN200 UC Client".
- If you have connected Standard SIP Phones or SIP enabled devices as SIP Extensions, for configuration instructions see "Configuring Standard SIP Phones".
- Any 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 Softphone clients and VARTA WIN200, see "Extended IP Phone/ VARTA UC Client - Operation".

- *You can register the Matrix IP Phone at three different locations as a single SIP Extension for Call Forking.*
- *The System supports interoperability with the standard IP Phones of Cisco, Polycom, Yealink, Snom and others. 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.*

The number of SIP extensions you have specified on the "Pre-requisites" page will be displayed on this page.

Configuring SIP Extensions using Jeeves

You need to configure the following parameters for SIP Extensions:

- **General Parameters**, see "Configuring SIP Extension General Parameters".

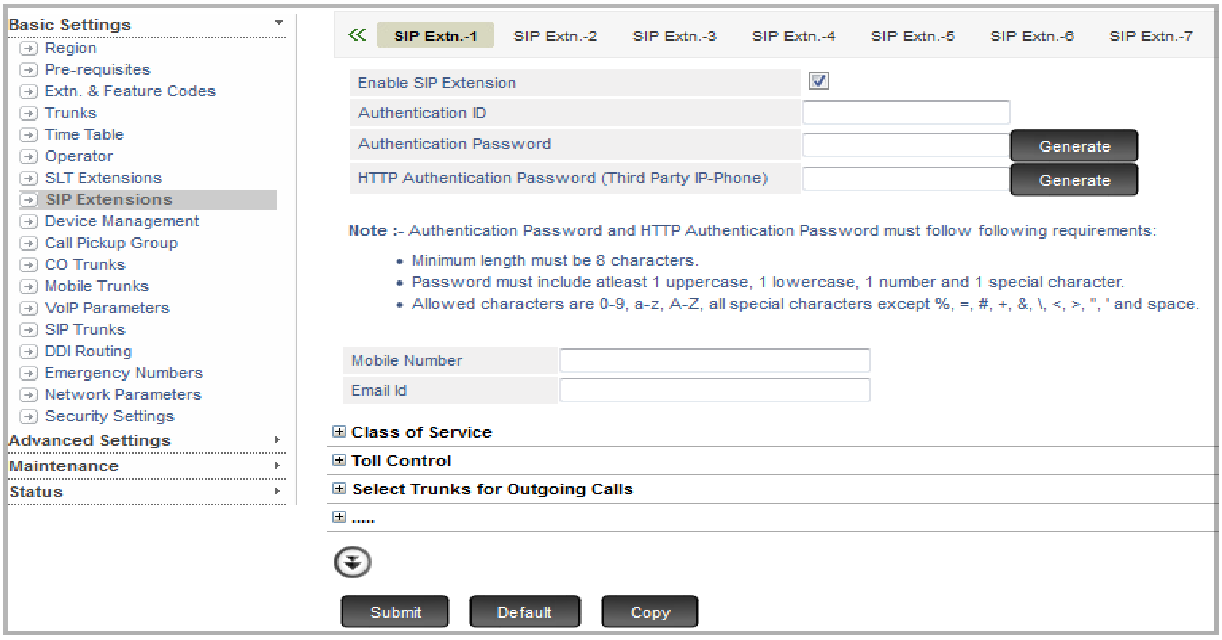
- **SIP Extension Settings** and the **Extended Phone Settings** as per the type of Extended Phone you have connected. See "Configuring Matrix SPARSH VP248" or "Configuring Matrix SPARSH VP310" or "Configuring Matrix SPARSH VP510" or "Configuring Matrix SPARSH VP330" or "Configuring Matrix Extended SPARSH VP710" or "Configuring Matrix SPARSH VP210" or "Configuring Matrix VARTA ADR100/AMP100 UC Clients" or "Configuring MATRIX VARTA WIN200 UC Client" or "Configuring Standard SIP Phones" respectively as per the type of Extended Phone you have connected.

Configuring SIP Extension General Parameters

To configure,

- Click **SIP Extensions.**

  On this page,



More: Click this button to view all parameter links on the page.

Less: Click this button to view only the essential parameter links on the page.

Expand: Click to expand a link to display all parameters under the link.

Collapse: Click to collapse a link. Hides all parameters under the link.

Copy: Click to copy the parameters under a specific link of a page to the same link under any other extension page.

Settings: Click to configure the settings of a parameter further.

···· **More link:** Click to view all additional parameters on the page.

To configure another SIP Extension, click the SIP Extension name (number) tab.

To save the settings, click **Submit**.

To assign default values to all the parameters of the SIP Extension, click **Default**.

To copy all SIP Extension parameter values to another SIP Extension, click **Copy**.

The default numbers and names of the SIP Extensions appears on the tabs on this page.

When you default the system, the configured IDs of all the SIP Extensions are cleared.

Follow the instructions provided below to configure the general parameters of SIP Extensions.

- By default the **Enable SIP Extension** check box is selected.

  If it is disabled, you will not be able to use this extension. You may clear the check box, when you want to deactivate the extension.

- Enter the **Authentication ID** for this extension. You cannot keep this field blank. Default: Blank.

  Authentication ID is the number which you want the VoIP module's Registrar Server to use for user authentication of the SIP messages received from this SIP Extension. The number may be a string of maximum 6 alphanumeric characters. All ASCII characters except < > and “ (double quote) are allowed.

Make sure the User ID configured in “Digest Authentication” does not conflict with the Authentication ID configured above.

- Enter the **Authentication Password** for this extension or click **Generate** to automatically generate a unique password. Authentication Password is the password to be used by the VoIP module’s Registrar Server to authenticate the SIP messages received from this 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 8 characters and can be a maximum of 12 characters.
  - 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. Default: Blank.

Make sure you note down or copy the Authentication Password in a confidential file.

To provide additional security, when the authentication fails for 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 Black List IP Address table manually. See “Black List IP Address - SIP Extensions” for more details. This activity will be logged in the “System Activity Log”.

You may also choose to configure simple Authentication Password, if required. However, it is recommended not to do so. For more details, see "Force Complex SIP/HTTP Authentication Password" under "Security Settings".

- Enter the **HTTP Authentication Password (Third Party IP-Phone)** or click **Generate** to automatically generate a unique password. This password will be used to authenticate the Standard SIP Phone connected to the SARVAM UCS.

  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.

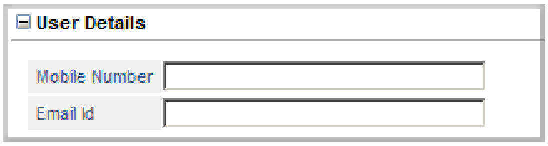
- To avoid unauthorized access, we recommend you to change the password regularly. Make sure it is strong and is kept confidential. When you enter the password manually, the password must:
  - be of minimum 8 characters and can be a maximum of 12 characters.
  - include at least one upper-case, one lower-case, 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. Default: Blank.

  To provide additional security, when the authentication fails for 5 times consecutively within 10 minutes due to wrong Authentication ID / Authentication Password, the system will block the IP Address for registration of this SIP Extension. However, you can register again after 10 minutes. This activity will be logged in the "System Activity Log".

  You may also choose to configure simple Authentication Password, if required. However, it is recommended not to do so. For more details, see "Force Complex SIP/HTTP Authentication Password" under "Security Settings".

Make sure, you have configured SIP ID as HTTP User Name and HTTP Authentication Password as HTTP Password in the Standard SIP Phone.

User Details and Auto Sign-In



- Enter the **Mobile Number** of the extension user. The Number can be a maximum of 16 digits. Default: Blank.

- 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.

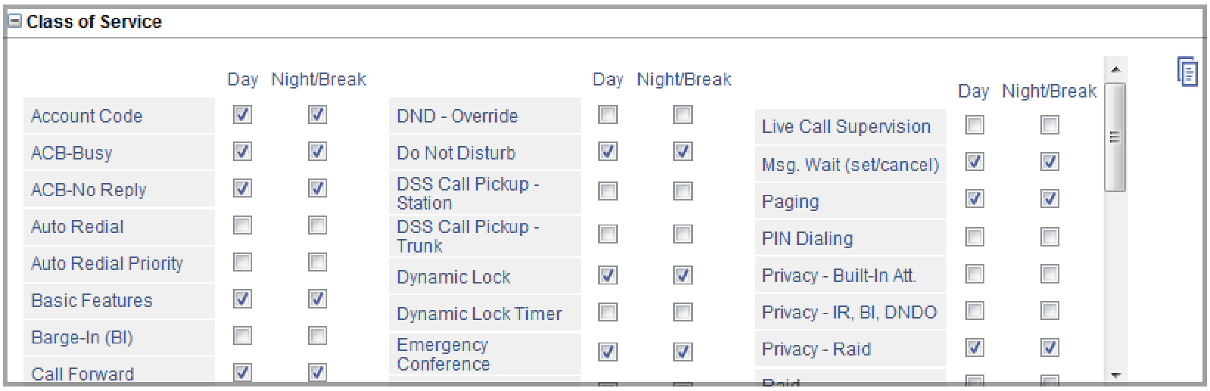
The Auto Sign-In Mail button will appear only after you have enabled the SIP Extension and configured the SIP ID, Authentication ID and Password.

Make sure the Auto Sign-In parameters have been configured. For details, refer to "Auto Sign-In Parameters".

- The **Mail Status** will display either sent, failed or sending.
- Click **Submit.**

These parameters are used by the SMS Server application. To know more about this feature, see “SMS Server”.

Class of Service



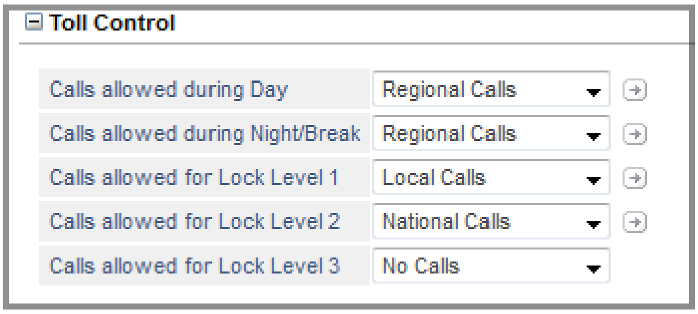
- Click **Class of Service** to expand.

  Define the Class of Service for the SIP extension for **Day** time and **Night/Break** time.

- To allow a feature, select the check box of the feature.
- To deny a feature, clear the check box.

See “Class of Service (CoS)” to know more.

Toll Control



- Click **Toll Control** to expand options. Set the desired Toll Control for the SIP extension for the Day and Night/Break.
- Select the type of **Calls Allowed during Day**: All Calls, No Calls, Local Calls, Regional Calls, National Calls, International Calls, Limited Calls 1, Limited Calls 2, Limited Calls 3. Default: No Calls.
- Select the type of **Calls Allowed during Night/Break**: All Calls, No Calls, Local Calls, Regional Calls, National Calls, International Calls, Limited Calls 1, Limited Calls 2, Limited Calls 3. Default: No Calls.

The Toll Control levels on this page are based on the allowed and denied number lists of Local, Regional, National, International, and Limited Call numbers you configured on the "Toll Control - Allowed Denied Numbers" page.

- If you have not configured the allowed and denied number list for the Type of Calls you selected as Toll Control, or if you want to add to the existing list, you may do it now.
  - Click the **Settings** icon.
  - The Toll Control - Allowed-Denied Numbers page will open in a new window.
  - Configure the Allowed and Denied Numbers.
  - Click **Submit**.
  - Close the window after you have configured the list.

  The Settings icon does not appear when you select **No Calls** or **All Calls** option.

See "Toll Control" to know more about this feature.

Toll Control for Dynamic Lock

Dynamic Lock allows extension users to change the Toll Control Levels (Calling Permissions) of their extensions on their own by dialing a code. SARVAM UCS supports Toll Control Levels 0 to 3 for Dynamic Lock.

For each Toll Control Level from 0 to 3, you must assign 'Call Privilege'[27]. For each Call Privilege, you need to configure the corresponding number strings to be allowed and number strings to be denied. See "Dynamic Lock" to know more about this feature.

- Click **Toll Control** to expand.
- Select the call privilege for **Calls allowed for Lock Level 1**: No Calls, Local Calls, Regional Calls, National Calls, International Calls, Limited Calls, All Calls. Default: No Calls.
- Select the call privilege for **Calls allowed for Lock Level 2**: No Calls, Local Calls, Regional Calls, National Calls, International Calls, Limited Calls, All Calls. Default: No Calls.
- Select the call privilege for **Calls allowed for Lock Level 3:** No Calls, Local Calls, Regional Calls, National Calls, International Calls, Limited Calls, All Calls. Default: No Calls.

The Lock Levels on this page are based on the allowed and denied number lists of Local, Regional, National, International, and Limited Call numbers you configured on the "Toll Control - Allowed Denied Numbers" page.

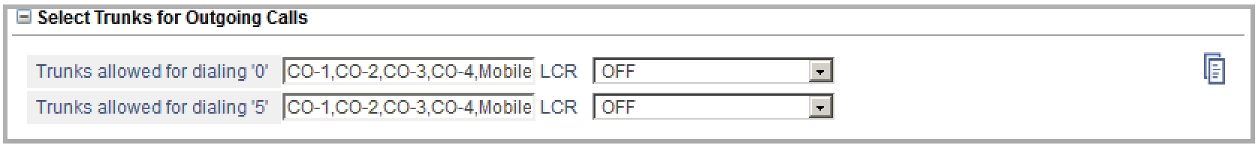
- If you have not configured the allowed and denied number list for the Calls allowed/denied for the selected Lock Level, or if you want to add to the existing lists, you may do it now.

  This icon does not appear when you select **No Calls** or **All Calls** option.

27. *The Call Privilege types are: No Calls, Local Calls, Regional Calls, National Calls, International Calls and Limited Calls.*

- Click the Settings icon.
- The Toll Control - Allowed-Denied Numbers page will open in a new window.
- Configure the Allowed and Denied Numbers for Local, Regional, National, International, and Limited Calls.
- Click **Submit**.
- Close the window after you have configured the list.

Select Trunks for Outgoing Calls

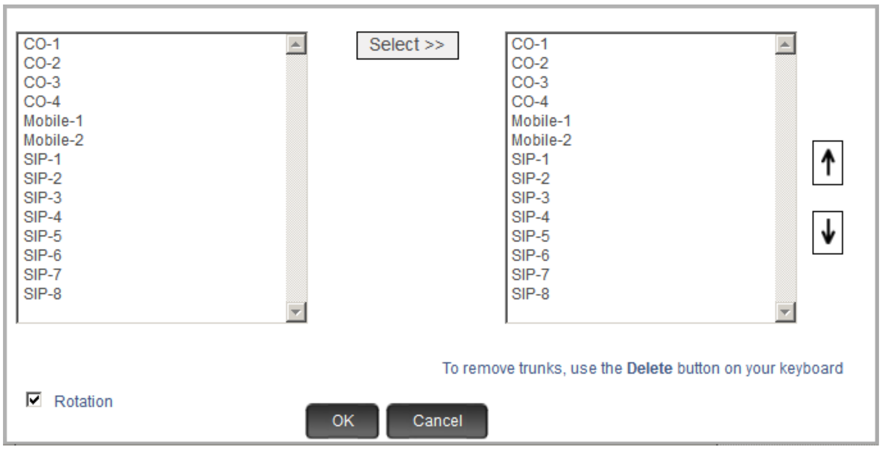


Outgoing calls (to external numbers) are made by dialing Trunk Access Codes (TAC). Default: 0 (TAC 1) and 5 (TAC 2). You may configure TAC 3 to TAC 6 as desired.

For each TAC, you need to select the Outgoing Trunks. All external calls made by dialing a particular TAC will be routed through the outgoing trunks you selected for that TAC.

You can also apply Least Cost Routing logic on the selected trunks, so that SARVAM UCS routes the outgoing call through the trunk that costs the lowest for the call.

- Click **Select Trunks for Outgoing Calls** to expand options.
- Select **Trunks allowed for dialing '0'**. The outgoing call will be routed through the selected trunks when the extension user dials TAC '0'.
    - Double-click the field. A multiple selection box opens.



On the left, the trunks appear with their names (if configured in "Trunks") and port numbers in a sequence, starting with CO trunks, followed by Mobile trunks and SIP trunks.

If you have not assigned any names to the trunks, they will appear with their default names (CO, MOB, SIP) and port numbers.

If you have enabled On-Site Configuration, only those trunks that are connected will appear in the box.

- To select a trunk, place your cursor on the desired trunk, and click **Select>>**.
  Or
- Press the **ctrl** key and click the left mouse button to select multiple trunks.

- You may change the sequence of the trunks you selected, if required, using the **Up** and **Down** arrow buttons on the right display box.

- You can also delete trunks from the ones you have selected by pressing **Delete** key on your keyboard.

- You may enable **Rotation**, if you have selected more than one trunk. Default: Enabled.

  When you enable Rotation, each new outgoing call is routed through the subsequent trunk in the group[28]. This ensures equal distribution of outgoing call traffic on all trunks.

  When Rotation is OFF, calls are routed through the first trunk in the group. If this trunk is busy, the call is routed to the next trunk in the group.

  Rotation has no relevance if only one member trunk is selected.

- Click **OK**. The multiple selection box closes.

  All the trunks appear in Trunks allowed for dialing '0', in the sequence you selected, separated by commas. For example, BSNL (CO-1), BSNL (CO-3), Reliance (CO-4) and Pulver (SIP1).

- To apply **Least Cost Routing** on the Trunks allowed for 0 dialing, select the desired LCR method from the combo box:

  - **Number Based:** Choose this option, if the service providers of the trunks you selected offer different tariffs according to area or distance, or phone numbers dialed.

  - **Time Based:** Choose this option, if the service providers of the selected trunks offer a different tariff according to the time of the day.

  - **Time + Number Based:** Choose this option, if the service providers of the selected trunks offer different tariffs according to the time of the day as well as area/distance/phone number.

  - **Service Provider Based:** Choose this option, if the same service providers of the selected trunks offer different rates for calls made to numbers within their own network and for calls made to numbers of another Service Provider's network.

  - If you do not want to apply LCR, select OFF.

  - Configure LCR method that you selected for the trunk group.

- To configure Least Cost Routing method you selected, click the Settings icon. A new window opens.

  - Configure the LCR method. See "Least Cost Routing (LCR)" under Advanced Settings for instructions.

  - Click **Submit**.

---

28. *The first call through the first trunk, the second through the second, the third through the third trunk, and so forth. Thus each new call is routed through the trunk next to the one that routed the previous outgoing call.*

  - Close the window.

This Settings icon does not appear if the LCR is OFF.

- Select **Trunks allowed for dialing '5'**. The outgoing call will be routed through the selected trunks when the extension user dials TAC '5'. Follow the same steps as described above.

  - Double-click the field. A multiple selection box opens.

  - Select trunks, placing your cursor on the desired trunk, and clicking **Select>>**.

  - Change the sequence of the trunks you selected, if required, using the **Up** and **Down** arrow buttons on the right display box. Delete trunks from the ones you have selected, if required.

  - Enable **Rotation**, if you have selected more than one trunk. Default: Enabled.

  - Click **OK**.

    All the trunks appear in Trunks allowed for dialing '5', in the sequence you selected, separated by commas.

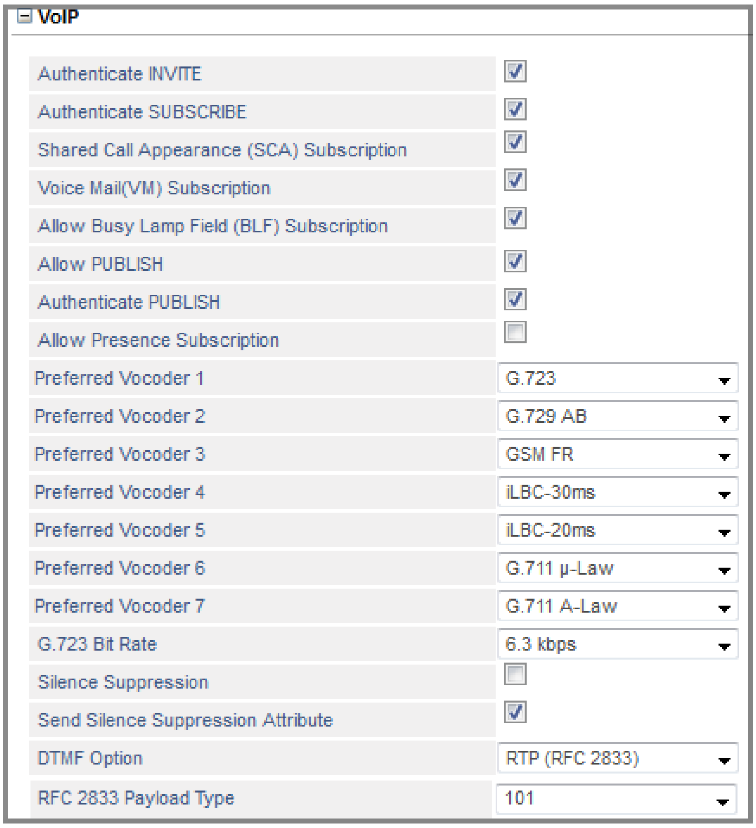
  - Choose a **Least Cost Routing** method, if you want to apply it on the trunks. Default: OFF.

  - Configure the Least Cost Routing method you selected by clicking the **Settings** icon.

  Similarly, you will be able to select outgoing trunks for Trunk Access Code 3, 4, 5 and 6, if you have assigned access codes to them in "Extension and Feature Codes".

- Click **Submit**.

VoIP



- Click **VoIP** to expand.
- Keep the **Authenticate INVITE** check box enabled for Authentication of INVITE Request in SIP Messages. Default: Enabled.
- Keep the **Authenticate SUBSCRIBE** check box enabled for Authentication of SUBSCRIBE Request in SIP Messages. Default: Enabled.

Make sure you have configured the Authentication ID for the SIP Extension.

- Keep the **Shared Call Appearance (SCA) Subscription** check box enabled, if you want to register this SIP extension with the same SIP ID at more than one location[29]. Default: Enabled.

  Shared Call Appearance provides notification on call states to all the phones registered with the same SIP ID at different locations. To know more about this feature, see “Shared Call Appearance” topic.

This feature is supported on Standard SIP Phones only.

- Keep the **Voice Mail (VM) Subscription** check box enabled, to provide the voice mail facility to the SIP Extension. Default: Enabled.

  You may clear this check box, if required.

---

29. *SARVAM UCS allows you to register SIP phones with the same SIP ID at three different locations.*

- Keep the **Allow Busy Lamp Field**[30] **(BLF) Subscription** check box enabled, to allow the SIP Extension to monitor the status of another extension or Trunk. Default: Enabled.

  Clear this check box, if you do not want the SIP Extension to monitor the status of another extension or Trunk.

- Select the **Allow PUBLISH** check box to allow the SIP extension user to choose whether or not to show their Presence Status to other SIP Extensions. Default: Enabled.

  If you allow PUBLISH subscription for a SIP Extension, you must also enable Authentication of PUBLISH.

- Keep the check box enabled, if you have enabled Allow Publish. Default: Disabled.

Make sure you have configured the Authentication ID for the SIP Extension.

- Keep the **Allow Presence Subscription** check box enabled, to allow the SIP Extension to view the presence status of other SIP-enabled Terminals, that is, whether they are available or not. 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.

- Select **Vocoders**[31] in the order of preference from the drop down box.
  The Vocoders supported by SARVAM UCS in the order of preference, i.e. 1st to 7th, by default are:
  - G.723
  - G.729 AB
  - GSM FR
  - iLBC - 30 ms
  - iLBC - 20 ms
  - G.711 μ – Law
  - G.711 A - Law

If you do not want to select any Vocoder, you can select the option 'None'. However, if all Vocoder Preferences from 1 to 7 are set to 'None', incoming and outgoing calls will be blocked.

- If you selected G.723 codec as a Preferred Vocoder, select **G.723 Bit Rate** as: Bit Rate: 5.3 Kbps or 6.3 Kbps. Default: 6.3kpbs.

  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.

- To suppress ‘Silence’ packets and allow only Voice packets through, select the check box **Silence Suppression**. Default: Disabled.

---

30. *Busy Lamp Field (BLF), a typical feature supported by PCM/TDM PBX and Key Telephone Systems, is also supported on SIP Extensions.*

    In PCM/TDM PBX 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.

31. *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.*

- *SARVAM UCS supports Silence Suppression for all Vocoders except GSM FR.*
- *Silence Suppression must be disabled, if you have selected 'Pass Through' as the "Fax Type".*

- Select the **Send Silence Suppression Attribute** check box, if you want to include "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 checkbox. By default, it is enabled**.**

- Select the appropriate **DTMF Option** from the three DTMF types supported by SARVAM UCS: RTP (RFC 2833) or SIP Info, or InBand. Default: RTP (RFC2833).

  The DTMF type determines how the DTMF digits will be sent over the IP Network, when a DTMF digit is pressed.

- **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.

- **Gain Settings:** To avoid noise or echo during speech on SIP Extensions, you must set the speech volume levels (Tx and Rx Gain) on the SIP Extension.

  - **SIP - System (Tx-Gain and Rx-Gain)**: Configure the Gain Settings that you want the system to apply on the SIP Extension with respect to the system (for example Call Conference). These will not be applicable for any other port type. Valid Range for Tx Gain: -31dB to +31dB and Rx Gain -31dB to +31dB. Default: 0dB.

  - **SIP- Voice Mail (Tx-Gain and Rx-Gain):** Configure the Gain Setting that you want the system to apply on the SIP Extension when the SIP Extension user is accessing the Voice Mail System. Valid Range for Tx Gain: -31dB to +31dB and Rx Gain -31dB to +31dB. Default: 0dB.

  - **SIP - Voice Module (Tx-Gain and Rx-Gain):** Configure the Gain Settings that you want the system to apply on the SIP Extension when the system plays Voice Module to the SIP Extension user. Valid Range for Tx Gain: -31dB to +31dB and Rx Gain -31dB to +31dB. Default: 0dB.

  - **SIP - Call Progress Tones (Tx-Gain and Rx-Gain):** Configure the Gain Setting that you want the system to apply on the SIP Extension while playing Call Progress Tones. Valid Range for Tx Gain: -31dB to +31dB and Rx Gain -31dB to +31dB. Default: 0dB.

  - **SIP - SLT (Tx-Gain and Rx-Gain)**: Configure the Gain Setting that you want the system to apply on the SIP Extension when it is connected to any FXS Port of the system during an incoming or outgoing call. Valid Range for Tx Gain: -31dB to +31dB and Rx Gain -31dB to +31dB. Default: 0dB.

  - **SIP - CO (Tx-Gain and Rx-Gain)**: Configure the Gain setting that you want the system to apply on the SIP Extension when it is connected to the CO Port of the system during an incoming or outgoing call. Valid Range for Tx Gain: -31dB to +31dB and Rx Gain -31dB to +31dB. Default: 0dB.

  - **SIP - SIP (Tx-Gain and Rx-Gain)**: Configure the Gain Setting that you want the system to apply on the SIP Extension when it is connected to the SIP Trunk or any other SIP Extension of the system during an incoming or outgoing call. Valid Range for Tx Gain: -31dB to +31dB and Rx Gain -31dB to +31dB. Default: 0dB.

- **SIP - Mobile (Tx-Gain and Rx-Gain)**: Configure the Gain Setting that you want the system to apply on the SIP Extension when it is connected to any Mobile Port of the system during an incoming or outgoing call. Valid Range for Tx Gain: -31dB to +31dB and Rx Gain -31dB to +31dB. Default: 0dB.

- SARVAM UCS supports **SRTP** (Secure Real Time Protocol) for secure conversations over SIP. The VoIP module of SARVAM UCS supports the following options:
  - **Disable:** Select **Disable** if you want SARVAM UCS to use normal RTP for transporting the speech packets.
  - **Optional**: Select **Optional** if you want SARVAM UCS to use 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**. Default: AVP.
  - **Forced:** Select **Forced** if you want SARVAM UCS to use 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.

- To apply **Echo Cancellation** for SIP to CO trunk calls, SIP to Digital Trunks (Mobile, SIP) and Extensions (SIP, SIP).
  - Keep the **Echo Cancellation** check box enabled. Default: Enabled.
  - Select Echo Cancellation **Tail Length (msec) for CO trunks**. It may be 32, 64, or 128 milliseconds. Default: 128. milliseconds.
  - Select Echo Cancellation **Tail Length (msec) for Extensions and Digital Trunks**. It may be 32, 64, or 128 milliseconds. Default: 32 milliseconds.

- Configure **Jitter Buffer**[32] to cut down on packet delays and improve voice quality.
  - Select the **Type** of Jitter Buffer you want to use: Static or Dynamic. Default: 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.
  - If you selected Static Jitter Buffer, configure **Minimum Delay**[33], from 10 to 280 milliseconds. Default 10 milliseconds.

    The value configured in the Minimum Delay determines the size of the Static Jitter buffer.

---

32. *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.*

33. *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 Static, the Minimum Delay defines the size of the Static Jitter Buffer. 'Minimum Delay' can be from 10 to 280 milliseconds. By default Minimum Delay is set to 10 milliseconds. The Static Jitter Buffer will store each received voice packets for the configured time and then it will send it to DSP for voice processing.*

- If you selected Dynamic Jitter Buffer, configure **Minimum Delay**[34] and **Optimization Factor**, from 1 to 13. Default: 10. By default Minimum Delay is set to 10 milliseconds.

  The Optimization Factor determines the rate of adaptation of the Dynamic Jitter Buffer to the jitter in the network. The minimum size of the Dynamic Jitter buffer depends on the 'Minimum Delay' configured.

  In networks with higher jitter, a higher value should be configured as Optimization Factor.

  The actual size of the Dynamic Jitter Buffer will be determined by the DSP on the basis of the Optimization Factor configured and actual network condition. Dynamic Jitter buffer can go up to maximum 300 milliseconds.

- If you are using **Fax over IP (FoIP)**[35], select the protocol as **Fax Type**. You may select:
  - T.38 (UDPTL)
  - T.38 (RTP)
  - Pass-through.

  Default: T.38 (UDPTL)

  - *'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.*

- If you have selected T.38 as Fax Type, configure **T.38 Fax Parameters**.

  Receiving good quality Fax over IP 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, do the following:

  - Configure the **Fax Version** 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 is set as 0.

  - Set the **Max Rate (kbps)** to: 2.4, 4.8, 7.2, 9.6,12, or 14.4.

    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. Default: 14.4 kbps.

  - Set the **Packet Period (msec)** to: 5, 10, 15, 20, 25, 30, 35, or 40. Default: 40.

    This parameter sets the sampling rate of TDM signal. If you cannot improve fax quality by lowering Fax Max Rate, you may reduce the Fax Packet Period.

    When you reduce the Fax Packet Period, the fax image will be sent at lower speed. EQM is inversely proportional to Fax Packet Period.
    Do not change the default settings unless it is required.

---

34. *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.*

35. *You can send/receive Fax, by connecting a fax machine to the SLT port of SARVAM UCS.*

- Set **Image Redundancy Level** to: None, 1, 2, or 3. Default: 1.

  The Image Redundancy Level is redundancy level for output Image, which can be from 0 to 3. 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, you will receive good quality fax.

  You may increase the Fax Image Level from 1 to 3 if the quality of fax does not improve with Fax Max Rate and Fax Packet Period.

  Level 0 means no redundancy.

- Set **Data Redundancy Level** to: None, 1, 2, 3, 4, 5, 6, or 7. Default: 3.

  This is a redundancy level for T.30 control data. Fax Data Level can be set from 0 to 7. Level 0 means no redundancy. Redundancy level increases from 1 towards 7. The higher the level set, 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.

- 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. Default: G.711 A-law.

- To improve the quality of Pass Through Fax reception on a Digital Trunk (Mobile), configure **Pass Through FAX Rx Gain (SIP to Digital Trunk Call)**. This parameter has relevance when you select 'Pass Through' as the Fax Type.

  - Select dB level for **Data Gain**, from -31 to 31. Default: -11 dB.

  - Select dB level for **Bypass Gain**, from -31 to 31. Default: -9 dB.

  Pass Through Fax packets are transported using RTP protocol. Normally, fax calls require less gain compared to voice calls. However, to improve fax reception, SARVAM UCS allows you to configure gain settings for fax (Data Gain and Bypass Gain).

  SARVAM UCS supports Fax Receive Gain for SIP to Digital Trunk calls as well as SIP to SLT Calls.

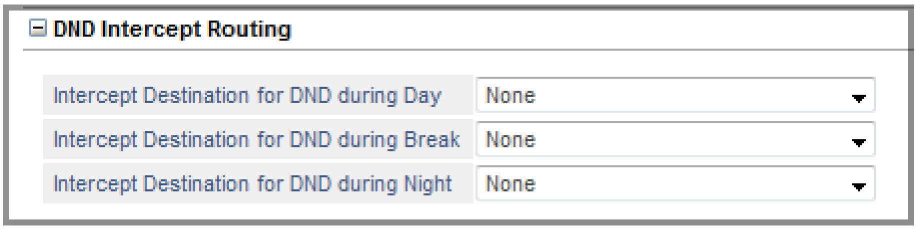
- Configure **Pass Through FAX Rx Gain (SIP to SLT Call)**, if you want to improve quality of Pass Through Fax received on a fax machine connected to a SLT port. This parameter has relevance only when you select ‘Pass Through’ as Fax Type.

  - Select dB level for **Data Gain**, from -31 to 31. Default: -11 dB.

  - Select dB level for **Bypass Gain**, from -31 to 31. Default: -9 dB.

- Click **Submit.**

DND Intercept Routing



When the DND set extension users wants their calls to be attended by someone even if DND is set, they must select an Intercept Destination. Incoming calls landing on extension that has set DND will be routed to the Intercept Destination. This destination can be the users own mailbox or another extension (SLT, SIP). See “Do Not Disturb (DND)” for more details.

- Select the **Intercept Destination for DND during Day.** You may select:
  - None
  - any SLT Extension
  - any SIP Extension
  - Voicemail

  Default: None

Similarly, you may select **Intercept Destination for DND during Break** and **Intercept Destination for DND during Night**.

SMDR Storage

The Station Message Detail Recording (SMDR) feature of SARVAM UCS enables you to record the details of Internal, Incoming (IC) and Outgoing (OG) calls made from/to all its extensions. To obtain SMDR as a report, you must enable SMDR Storage, and set filters. See “Station Message Detail Recording (SMDR)” to know more.

- Click **SMDR Storage** to expand.

- Select the type of internal calls to be stored from the combo box **Store Internal Calls**. You can select from the following options:

  - **Made/received by this extension**: the system will store all calls made to and received from this extension.

  - **Made by this extension**: the system will store outgoing calls made from this extension.

  - **Received by this extension**: the system will store only incoming calls from other extensions.

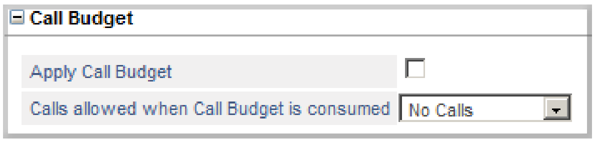
  - **Never**: the system will not store internal calls

- To store details of incoming calls from external numbers, select **Store Incoming Calls**. Default: selected.

- To store details of outgoing calls made by the extension user to external numbers, select **Store Outgoing Calls**. Default: selected.

- Click **Submit**.

Call Budget



Using the Call Budget feature, you can allot a 'budget' limit for outgoing calls made by the extension. See "Call Budget on Extension" for more information. If you want to enable this feature on this extension,

- Click **Call Budget** to expand.

- Select the **Apply Call Budget** check box.

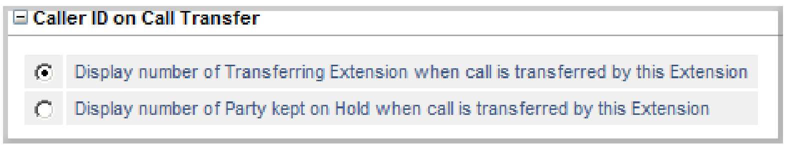
- You may define the calling permission for the extension, after it has consumed the call budget allotted to it. Select the type of **Calls** to be **allowed when Call Budget is consumed** from the following:
  - No Calls
  - Local Calls
  - Regional Calls
  - National Calls
  - International Calls
  - All Calls

  Default: No Calls.

*Click the Settings icon to view the **Toll Control - Allowed-Denied Numbers** page. This icon does not appear when you select **No Calls** or **All Calls** option.*

- Click **Submit**.

Caller ID on Call Transfer



This parameter is related to the CLIP feature. It allows you to choose whether the system should display the CLI of the 'Held Party' or the CLI of the 'Transferring Party' to the transfer destination extension while the call is being transferred.

See the feature description for "Calling Line Identification and Presentation (CLIP)" to know more.

- Click **Caller ID on Call Transfer** to expand.

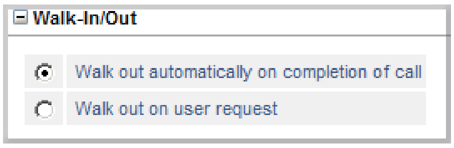
- Select the radio button of the desired option:

  - Display Number of Transferring Extension when call is transferred on this extension.

- Display Number of Party kept on Hold when call is transferred on this extension.

- Click **Submit**.

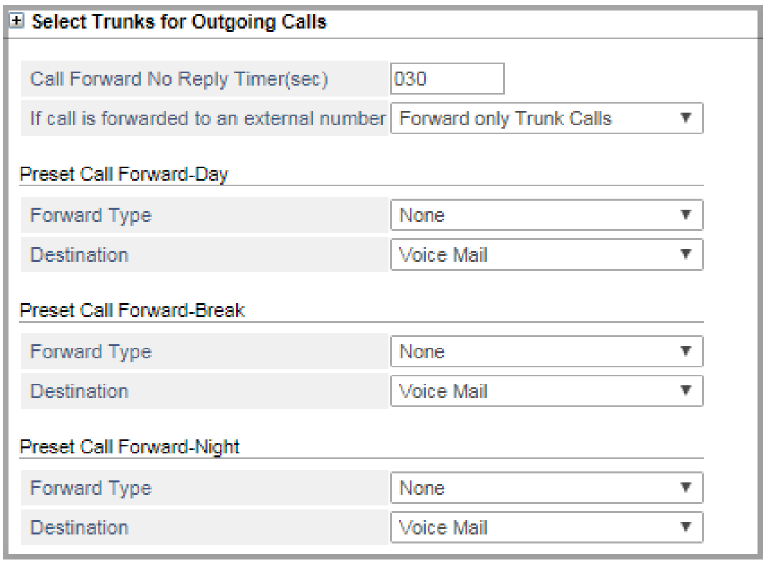
Walk-In/Walk-Out



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, where the extension user is automatically logged out after a call. ii) Walk-In until Logout, where the extension user remains logged on until s/he manually walks out or a second user walks into the same extension. To know more about this feature, see “Walk-In Class of Service”.

- Click **Walk-In/Walk-Out** to expand.
- Select the radio button of the type of Walk-Out mode you want to assign to the extension:
  - **Walk-Out automatically on completion of call:** Select this option, if you want to assign One-Call per Walk-In to the extension.
  - **Walk-Out on user request:** Select this option, if you want to assign Walk-In until Logout to the extension.
- Click **Submit**.

Call Forward



- Click **Call Forward** to expand.
- Set the **Call Forward No Reply Timer (sec)** to the desired value, if required. The range of this timer is 001-255. Default: 030 seconds.

Call Forward No Reply Timer signifies the duration for which the system will wait for an extension to answer an incoming call, before forwarding the call to the programmed destination as Call Forward-No Reply. By default the Timer is set to 30 seconds. This timer is applicable for both Call Forward and Preset Call Forward. Refer the feature description for "Call Forward" and "Preset Call Forward" to know more.

- Select the type of calls to be forwarded **If Call is forwarded to an external number**. You may select from the following options:
  - Forward only Internal calls
  - Forward only Trunk calls
  - Forward all calls (internal as well as trunk calls)

  Default: Forward only Trunk calls.

  This parameter is relevant for the features "Call Forward", "Mobility Extension" and "Call Forward - When Not Registered".
- Select the **Forward Type** for **Preset Call Forward - Day**. You may select:
  - None
  - When Busy
  - When No Reply
  - When Busy or No Reply

  Default: None
- Select the **Destination** to which the calls are to be forwarded for Preset Call Forward - Day.

  Follow the instructions in steps 4 and 5 to configure **Preset Call Forward - Break** and **Preset Call Forward - Night**.

  See "Preset Call Forward" for more details.
- Click **Submit**.

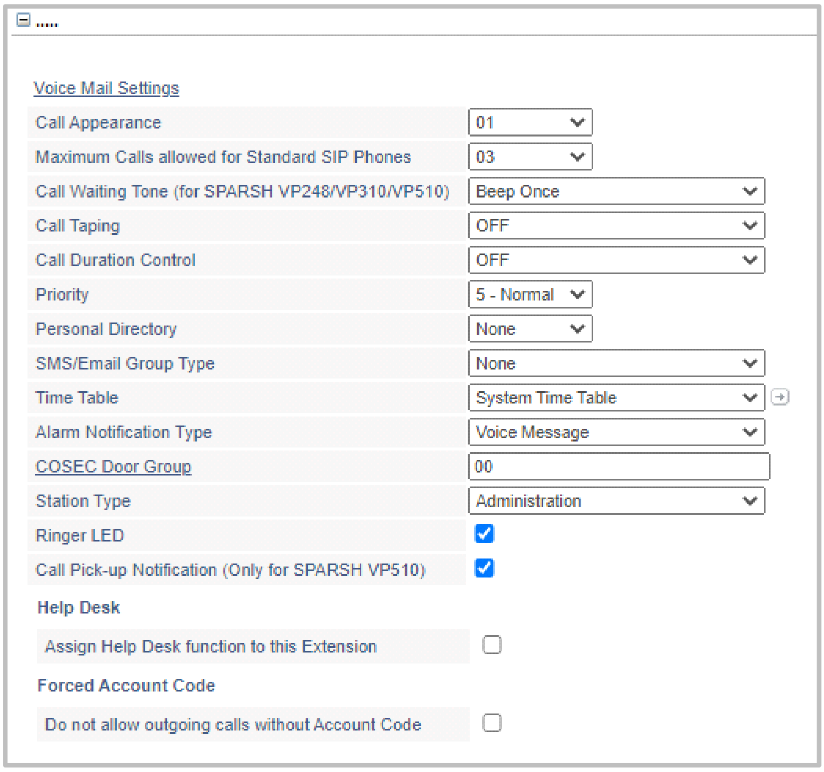
Configuring SIP Extension Settings as per the Device Type

- 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 VP310 as SIP Extensions, for configuration instructions, see "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 the Matrix SPARSH VP330 as SIP Extensions, for configuration instructions, see "Configuring Matrix SPARSH VP330".
- 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 SPARSH VP210 as SIP Extensions, for configuration instructions, see "Configuring Matrix SPARSH VP210".
- If you have registered the Matrix SPARSH Mobile Softphone Clients as SIP Extensions, for configuration instructions, see "Configuring Matrix VARTA ADR100/AMP100 UC Clients".

- If you have registered MATRIX VARTA WIN200 UC Client in any of the locations, refer to "Configuring MATRIX VARTA WIN200 UC Client"
- If you have connected Standard SIP Phones or SIP enabled devices as SIP Extensions, for configuration instructions, see "Configuring Standard SIP Phones".

More Features

- Click **More...** to expand.



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.

Call Appearances

- Set the number of **Call Appearances** for the SIP extension. Default: 2.

  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 SIP 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 more the number of Call Appearances assigned to an extension, the more the number of calls the extension user can handle simultaneously.

The SARVAM UCS supports a maximum of 8 Call Appearances on SIP extensions.

SIP extensions for Executives are usually assigned 2 Call Appearances, while the Operator extension is assigned 6 Call Appearances to handle 6 calls simultaneously.

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 **Call Waiting Tone (for SPARSH VP248/VP310/VP510)** beeps as per your requirement. You can select from the options:
  - OFF
  - Beep Once
  - Beep until Answered

  Default: Beep Once

  However, when a ongoing call is being taped or recorded, the call waiting tone for any new incoming call will not be played.

Call Taping

To use the “Call Taping” feature on the extension,

- Click **Call Taping** to expand options.

- Select the type of **Call Taping** to be applied from the options:
  - OFF
  - Apply as per profile 1
  - Apply as per profile 2
  - Apply as per profile 3
  - Apply as per profile 4

  If you do not want to apply Call Taping, select OFF.

  Default: OFF.

- Now, configure the Call Taping Profile number you selected for the extension. To do this,

  - Click the **Settings** icon. The page opens in a new window, displaying the parameters of the profile you selected for the extension. This icon does not appear when Call Taping is **OFF**.

  - In the new window,

    - Select the **Tape Internal Calls** check box, if you want internal calls made and received by the extension to be taped.

    - Select the **Tape Incoming Calls received without CLI** check box if you want incoming calls without CLI to be taped. Default: Disabled.

    - In **Tape Incoming Calls received from following numbers**, type the numbers of the Incoming Calls that must be taped. You may type as many as 99 numbers.

    - In **Tape Outgoing Calls made to following numbers,** type the external numbers that you want the system to tape. You may type as many as 99 numbers.

- Click **Submit**.

  See “Call Taping” to know more.

Call Duration Control

To enable the “Call Duration Control (CDC)” feature on the extension,

- Click **Call Duration Control** to expand options.
- Select the type of **Call Duration Control** to be applied from the options:
  - OFF
  - Apply as per CDC profile 1
  - Apply as per CDC profile 2
  - Apply as per CDC profile 3
  - Apply as per CDC profile 4

  If you do not want to apply Call Duration Control, select OFF. Default: Apply as per CDC profile 1.

- Configure the Call Duration Control Profile number you selected for the extension. To do this,
  - Click the **Settings** icon. The page opens in a new window. The page displays the parameters of the profile you selected for the extension. This icon does not appear when Call Duration Control is **OFF**.
  - In the new window,
    - Enable **Apply CDC to Internal Calls,** if CDC is to be applied on internal calls. Default: Enabled.
    - Enable **Apply CDC for Incoming Calls received from trunk**, if CDC is to be applied on incoming external calls. Default: Enabled.
    - Enable **Apply CDC for Outgoing Calls made from the trunk**, if CDC is to be applied to outgoing external calls. Default: Enabled.
    - If required, change the **CDC Timer** to the desired duration. The range of the timer is 0001 to 9999 seconds. Default: 160 seconds.
    - Enable **Disconnect Call after CDC Timer** check box if you want calls to be disconnected on the expiry of the CDC Timer. Default: Enabled.
    - In the **Apply CDC for calls matching with numbers** column, type the external numbers on which you want to apply CDC. You can enter as many as 99 numbers.
    - In the **Do Not Apply CDC for calls matching with numbers** column, enter the numbers which you want to be exempt from CDC.
  - Click **Submit**.

Priority

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”.

- Select a **Priority** from 1 to 9 for the SIP extension. Default: 5-Normal.

  If this SIP extension is assigned to Operator, you may want to set a higher priority for this extension.

Personal Directory

- Select a Personal Directory number from 01 to 50 that you want to assign to the SIP extension. Default: None.

  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 (). The Personal Directory is necessary for using the features "Abbreviated Dialing" and "Dial By Name".

  When a Personal Directory is assigned to an SIP, it must also be configured. The Personal Directory can be configured also by the extension user. Refer the "Abbreviated Dialing" topic for instructions on configuring the Personal Directory.

SMS/Email Group Type

- Select the **SMS/Email Group Type**, you want to assign to the extension user. Default: None.

  See "SMS/Email Group" under "SMS Server" to know more.

Time Table

- Select a Time Table for the SIP extension. Default: System Time Table.

  If you have not configured Time Table, you may do so now, by clicking the Time Table link. Define the Working days, and the start and end time of the Working hours and Break hours for each working day.

  See "Time Tables" to know more.

Alarm Notification Type

To configure this parameter and for the feature description, see "Alarms" to know more.

- Select the type of **Alarm Notification Type** the system should use when user answers Alarm Call**.** Your options are:

  - **Voice Message:** The extension user is played a message recorded in the Voice Module on answering the alarm call.

  - **Music-on-Hold:** The extension user is played music-on-hold on answering the alarm call.

  - **Voice Mail:** The extension user is greeted by the VMS on answering the alarm call.

  - **Routing Group:** The alarm call is routed to the group of desired extensions configured as the Operator, so that the alarm request can be served. See "Operator" for instructions.

  Default: Voice Message.

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

- 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 this is not an Operator extension, select the **Station Type** for the SIP Extension as **Administration**.

Ringer LED

This parameter decides whether the Ringer LED should glow or not on the desired extensions for incoming calls[36] and as missed call notification[37].

- By default, this check box is selected, that is, the LED will continue to blink until the missed call log is read or the call is answered/disconnected. Disable this check box if you do not want the Ringer LED to glow.

Call Pick-up Notification (Only for SPARSH VP510)

- 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 Pickup”, “Call Pickup Group” and “Class of Service (CoS)”. Call Pick-up Notifications will be displayed for SLT as well as SIP Extensions and for calls landing through CO as well as SIP Trunks. For details of the Notification, refer to the EON510_SPARSH VP510 V2 User Guide.

Help Desk

Configure this parameter if you want to define the extension as a “Help Desk”.

- Select the **Assign Help Desk function to this Extension** check box. Default: Disabled.

Forced Account Code

Configure this parameter, if you want to apply the feature Forced Account Code on the extension. When Forced Account code is enabled, the system will allow the extension user to dial an external number only after the user has entered the Account Code. To know more, see “Account Codes”.

- Select the check box **Do not allow Outgoing Calls without Account Code (Force Account Code)**.

- Click **Submit**.

Copy Parameter Values

After you have finished configuring this SIP Extension, you may configure the next SIP Extension. To do so,

- Click the tab of the next SIP Extension.

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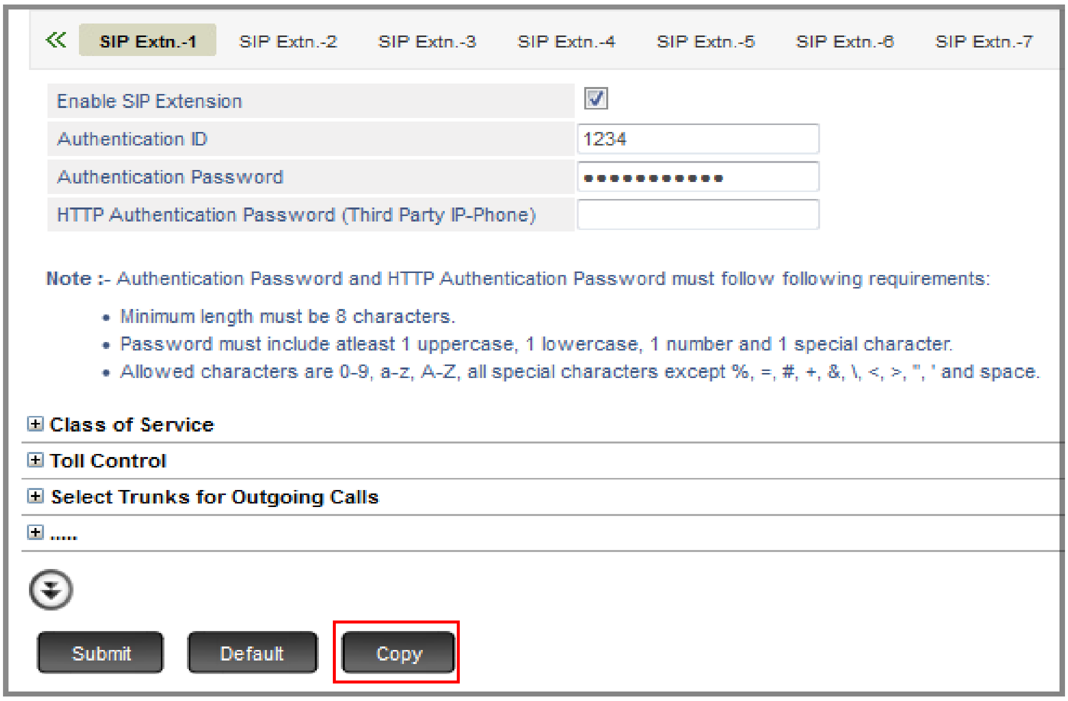
36. *Applicable only for SPARSH VP310 AND SPARSH VP510.*
37. *Applicable only for SPARSH VP510.*

- Follow the same steps as described above to configure another SIP Extension.

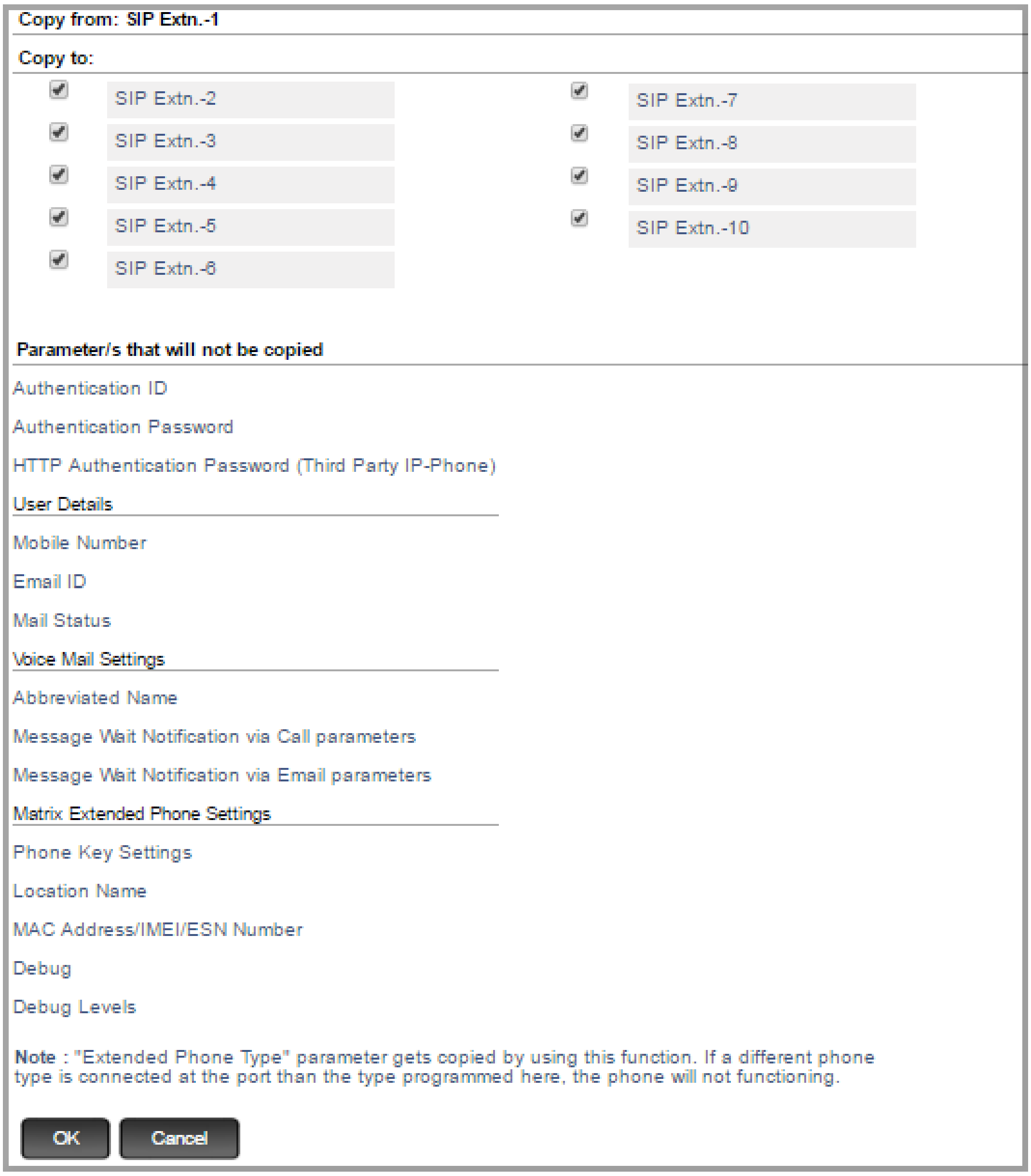
OR

You may use the **Copy** button at the bottom of the page to apply the same SIP Extension settings you configured for the current extension to another SIP Extension. To do so,

- Click **Copy**.



The Copy page opens in another tab.



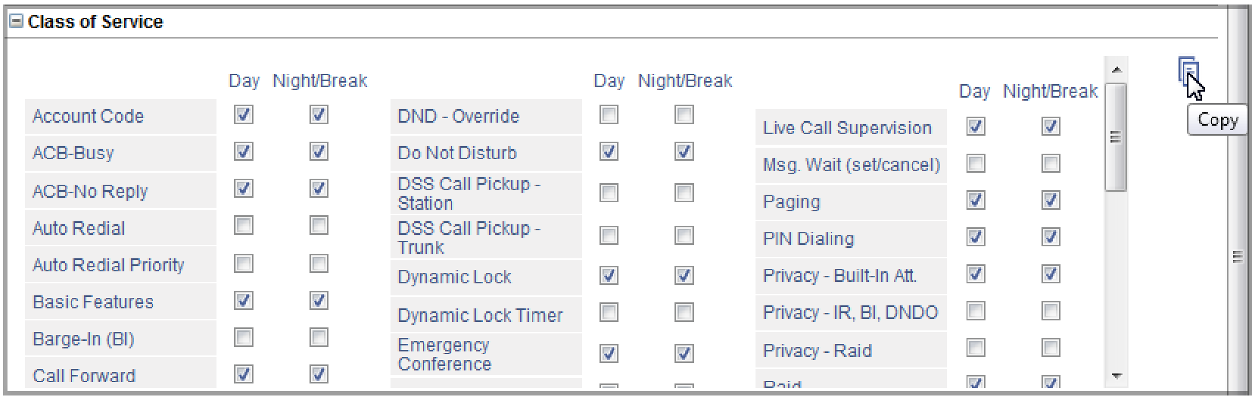
The copy page displays the number and/or name of the current extension in **Copy from:**. It also lists the numbers and/or names of the extensions to which you can copy the extension settings under **Copy To:**.

- Select the respective **Copy to** check boxes of the SIP Extensions to which you want to apply the current SIP Extension settings.
- Click **OK**.

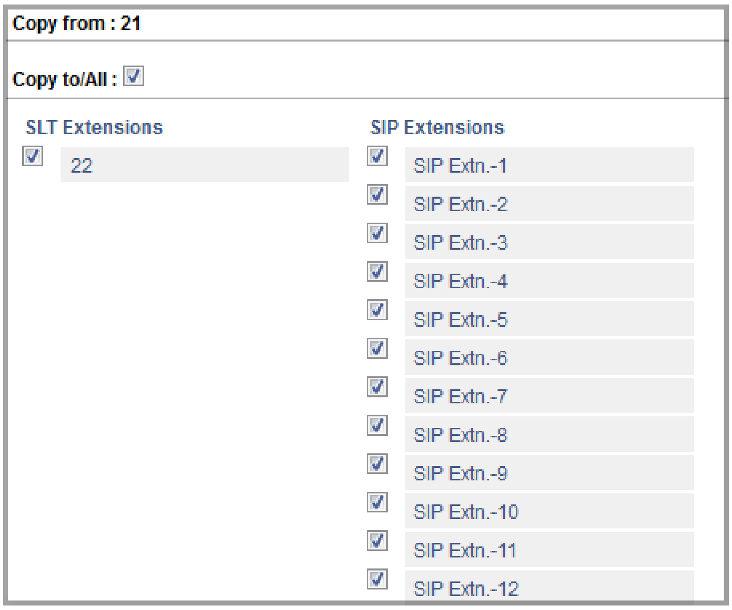
All the parameter values of the current SIP Extensions except those listed below **Parameter/s that will not be copied**, will be applied to the selected SIP Extensions.

You may also copy the parameter values of a specific link under current SIP Extension to the same link under any other extension page. To do so,

- Click the **Copy** icon besides the desired link.



The Copy page opens in another tab.



The Copy page displays the numbers and/or names of the extensions to which you can copy the extension settings under the **Copy To/All:**.

- Select the **Copy to/All** check box, to apply the link settings to the same link under all other Extensions.

  OR

  You may also apply the link settings to the desired Extensions only, by selecting the respective check boxes.

- Click **OK**.

Configuring Matrix SPARSH VP248

SPARSH VP248, the proprietary SIP-based IP Phone for SARVAM UCS, supplied by Matrix, is a feature-rich phone, providing voice communication over IP network. To know the list of features supported by Matrix SPARSH VP248, 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”.
- Extended IP Phone Settings, see “Configuring Matrix Extended Phone Settings using Jeeves”.

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 VP510, SPARSH VP330, Matrix Extended SPARSH VP710, SPARSH VP210, 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 as **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 VP510 at any of the locations, refer to “Configuring Matrix SPARSH VP510”.

If you have connected SPARSH VP330 at any of the locations, refer to “Configuring Matrix SPARSH VP330”.

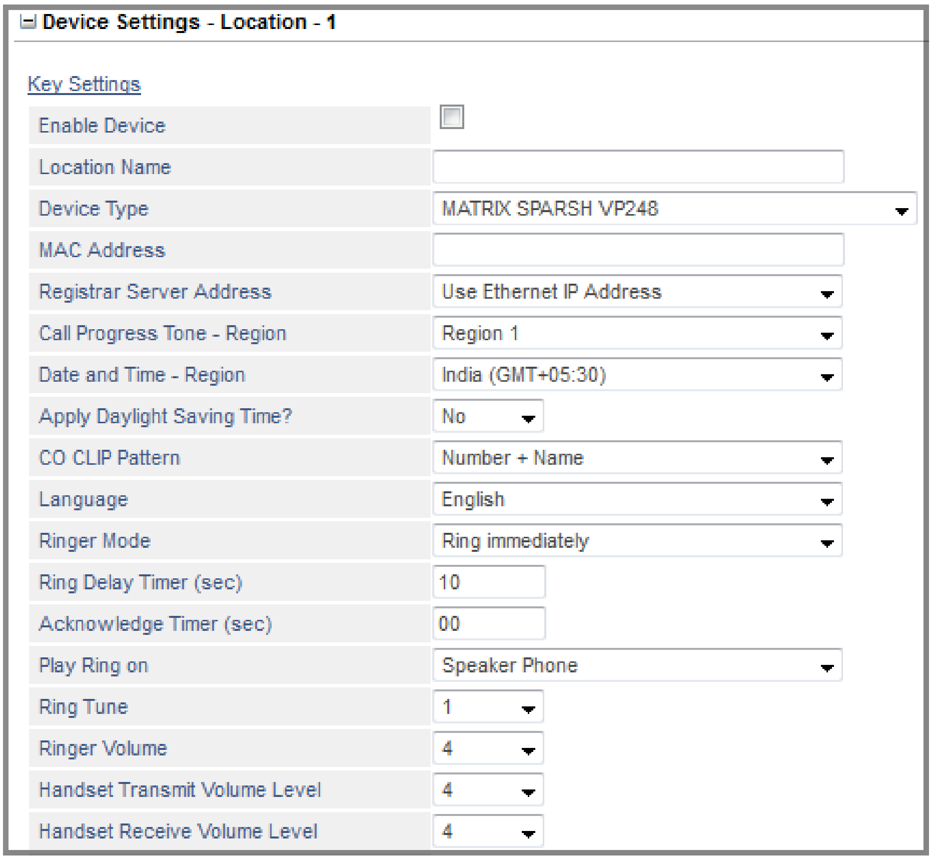
If you have connected Extended SPARSH VP710 at any of the locations, refer to “Configuring Matrix Extended SPARSH VP710”.

If you have connected 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 UC Client in any of the locations, refer to “Configuring MATRIX VARTA WIN200 UC Client”.

- Click **Device Settings - Location 1** to expand.



- 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 and click **Submit** to save.

- Enter the **MAC Address**[38] of the Matrix Extended IP Phone connected at this location in hexadecimal format, e.g. 00:50:C2:55:B0:10. Default: Blank.

  SARVAM UCS validates the IP-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 IP phone with the SIP Registrar of SARVAM UCS, according to your installation scenario:

  - If the Extended IP phone is in the same network (LAN) as SARVAM UCS, select **Use Ethernet IP Address** as Registrar Server Address.

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38. *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 Extended IP phone is in the Global Network and SARVAM UCS is connected to Internet over Mobile WAN, select **Use Mobile Port IP Address** as Registrar Server Address

- If the Extended IP 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.

- If you want SARVAM UCS to assign Registrar Server Address automatically, select **Automatic** as Registrar Server Address.

  By default, Use Ethernet IP Address is selected as the Registrar Server IP Address.

- To set the call progress tone generation standards of the country where the Matrix IP Phone is installed, select the **Call Progress Tone - Region**. Default: Region 1.

  See "Call Progress Tones" to know more.

- Select the region/country whose **Date and Time - Region** settings the IP Phone should follow. Default: India.

- If you want to enable Daylight Saving Time (DST) on the phone, set **Apply Daylight Saving Time?** 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 **CO CLIP Pattern** for the IP phone. 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 IP phone. Default: English.

  SARVAM UCS provides language support for English, French, German, Spanish, Portuguese, and Italian on the Matrix IP Phone. 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 delay (if the call is still not answered).
- Silent (silent mode).

Default: Ring Immediate.

- If you have 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 IP 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 IP 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 IP 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'.

- To assign the Ring Destination for the IP phone, select the desired destination for **Play Ring on**. You may choose
- **Speaker Phone:** The ring will be played on the Speakerphone.
- **Headset:** The ring will be played on the Headset.

  Default: Speaker Phone.

  When you select the Headset as the destination, make sure that you set the '*Headset Connected?'* parameter to Yes and connect a Headset to the IP Phone.

- Select the desired **Ring Tune** according to your/IP phone 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 IP phone, set the **Handset Transmit Volume** 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 IP phone, set the **Handset Receive Volume** 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 IP phone, set the **Headset Transmit Volume** 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 IP phone, set the **Headset Receive Volume** to the desired level, from 0 to 7. Default: 4.

- To change the Transmit Gain of the Speakerphone MIC Volume, set **Speaker Transmit Volume** to the desired level, from 0 to 7. Default: 4.

- To change the Receive Gain of the Speakerphone MIC Volume, set **Speaker Receive Volume** to the desired level, from 0 to 7. Default: 4.
- To use a Headset with the IP phone, set **Headset Connected?** to **Yes.** Default: No.

  Make sure that you connect a Headset to the IP phone, if you select Yes.
- Select the **Auto Answer** check box to enable this feature on the IP phone. Default: Disabled.

  When you set the "Auto Answer" feature on the IP phone, 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 IP phone 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.
- Define **RTP Port**:
  - **RTP Listening Port:** This is the port on which the IP phone listens for SIP messages over TCP. 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.
- Set the **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.
- If the IP phone is connected behind a NAT router, configure **NAT Keep Alive**.
  - Select the **Enable NAT Keep Alive** check box 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 9999 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.
- 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 sending 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.

- To debug using Syslog Client supported by the phone, configure **Debug** parameters:

  - Set **Enable Debug?** to Yes. Default: No.

    When the Debug parameter 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 of the remote Syslog Server as **Syslog Server Address**.

  - Enter the **Server Port**, i.e. the address of the Listening Port of the Syslog Server from 1025-65535 or 514. Default: 514.

    Syslog uses the UDP as transport protocol and listens on the port 514 (the default listening port).

  - Select **Debug Levels** from these options, by selecting the respective check box:
    - SIP
    - System
    - Hardware
    - Call
    - Network
    - VoPP

    The phone supports multiple debug levels. 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.

Phone Key Settings

You can customize the DSS keys of the Extended IP Phone, to match the requirements of the extension user.

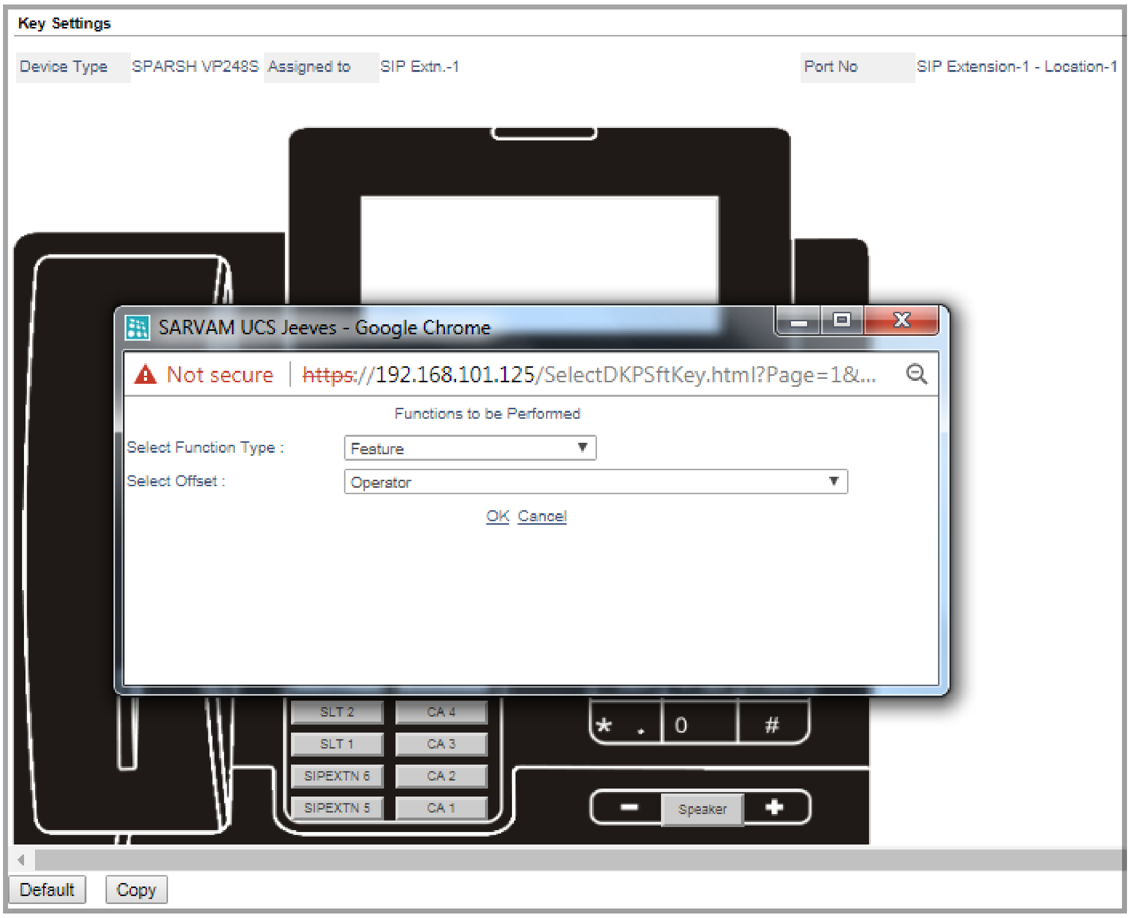
To customize the keys of the Extended IP Phone registered at this location,

- Click **Key Settings** link.

The default key map of the Extended IP Phone appears on your screen.

- Click the key you want to configure. For example, **CO3**.

  A dialog box opens, with the options for the **Functions to be Performed** by the key.



- In the **Select Function Type** list box, 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.

  In 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 box closes.

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 SIP extension, select **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 as Offset.

To assign direct access to a SIP Trunk, select **SIP Trunk** as Function Type and the desired 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

- Acknowledge
- Local Menu

If you have completed the configuration of the Matrix IP Phone Settings at location 1, follow the same steps as described above to configure the Matrix Extended IP Phone at the other two locations.

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.

- Enable SIP Extension
- SIP ID
- Authentication ID
- Authentication Password
- Registrar Server Address
- MAC Address
- Enable Device
- Device Type
- Phone Settings
- Call Progress Tone
- Date and Time
- Apply Daylight Saving Time?
- Quality of Service
- RTP Ports
- NAT Keep Alive
- Timers
- Authenticate INVITE
- The SE Password of SARVAM UCS is changed
- You restart the System
- Set the System to Default
- The Network Parameters are changed
- Specific parameters in the VoIP page are changed

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 by Matrix SPARSH VP310, 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 - Extended IP Phone, you must configure the following:

- SIP Extension General Parameters, see “Configuring SIP Extension General Parameters”.
- Extended IP Phone Settings, see “Configuring Matrix Extended Phone Settings using Jeeves”.

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, Matrix Extended SPARSH VP710, SPARSH VP210, Matrix VARTA UC Clients 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 as **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 VP510 at any of the locations, refer to “Configuring Matrix SPARSH VP510”.

If you have connected SPARSH VP330 at any of the locations, refer to “Configuring Matrix SPARSH VP330”.

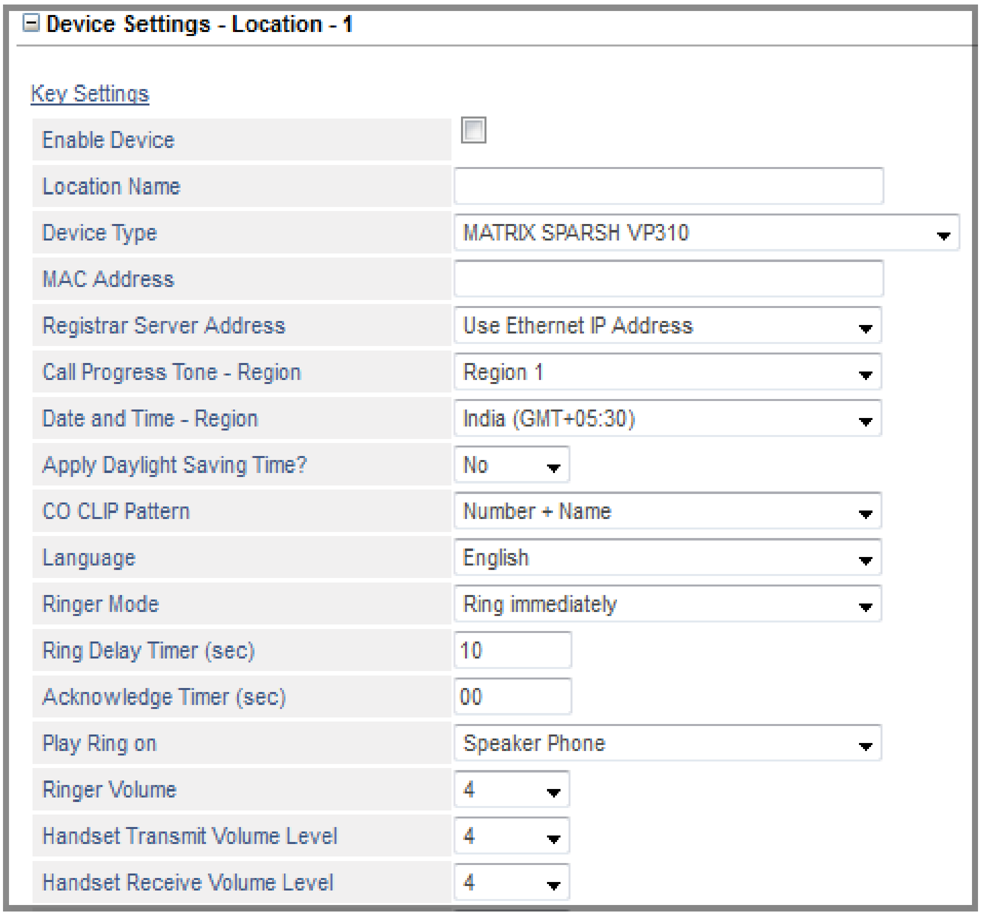
If you have connected Extended SPARSH VP710 at any of the locations, refer to “Configuring Matrix Extended SPARSH VP710”.

If you have connected 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 UC Client in any of the locations, refer to “Configuring MATRIX VARTA WIN200 UC Client”.

- Click **Device Settings - Location 1** to expand.



- 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 and click **Submit** to save.
- Enter the **MAC Address**[39] of the SPARSH VP310 connected at this location in hexadecimal format: 00:50:C2:55:B0:10. 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:

39. *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 Extended IP phone is in the same network (LAN) as SARVAM UCS, select **Use Ethernet IP Address** as Registrar Server Address.
- If the Extended IP phone is in the Global Network and SARVAM UCS is connected to Internet over Mobile WAN, select **Use Mobile Port IP Address** as Registrar Server Address
- If the Extended IP 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.
- If you want SARVAM UCS to assign Registrar Server Address automatically, select **Automatic** as Registrar Server Address.
- By default, Use Ethernet 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'.

- 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 delay (if the call is still not answered).
- Silent

Default: Ring Immediate.

- If you have 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'.

- 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 '*Headset Connected?'* parameter to Yes and connect a Headset to the IP Phone.

- 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 IP phone, set **Headset Connected?** to **Yes.** Default: No.

  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.
- Select **Transport Mode** and enable **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 is selected in VoIP Port Parameters.*
  - *If you select TLS, make sure the SIP Over TLS is selected in VoIP Port Parameters.*

  - For secure conversations over SIP, select the **Enable SRTP?** check box. The SIP messages will be transported over SRTP only.
- Define **RTP Port**.
  - **RTP Listening Port:** This is the port on which the IP phone listens for SIP messages over TCP. 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.
- Set the **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.
- If the IP phone is connected behind a NAT router, configure **NAT Keep Alive**.
  - Select the **Enable NAT Keep Alive** check box 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 9999 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.

- 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 sending 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.
- To debug using Syslog Client supported by the phone, configure the **Debug** parameters:
  - Set **Enable Debug?** to Yes. Default: No.

    When the Debug parameter 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 of the remote Syslog Server as **Syslog Server Address**.
  - Enter the **Server Port**, i.e. the address of the Listening Port of the Syslog Server from 1025-65535 or 514. Default: 514.

    Syslog uses the UDP as transport protocol and listens on the port 514 (the default listening port).
- Select **Debug Levels** from these options, by selecting the respective check box:
  - SIP
  - System
  - Hardware
  - Call
  - Network
  - VoPP

  The phone supports multiple debug levels. 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**.

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.

Phone Key Settings

You can customize the DSS keys of the Extended IP Phone, to match the requirements of the extension user.

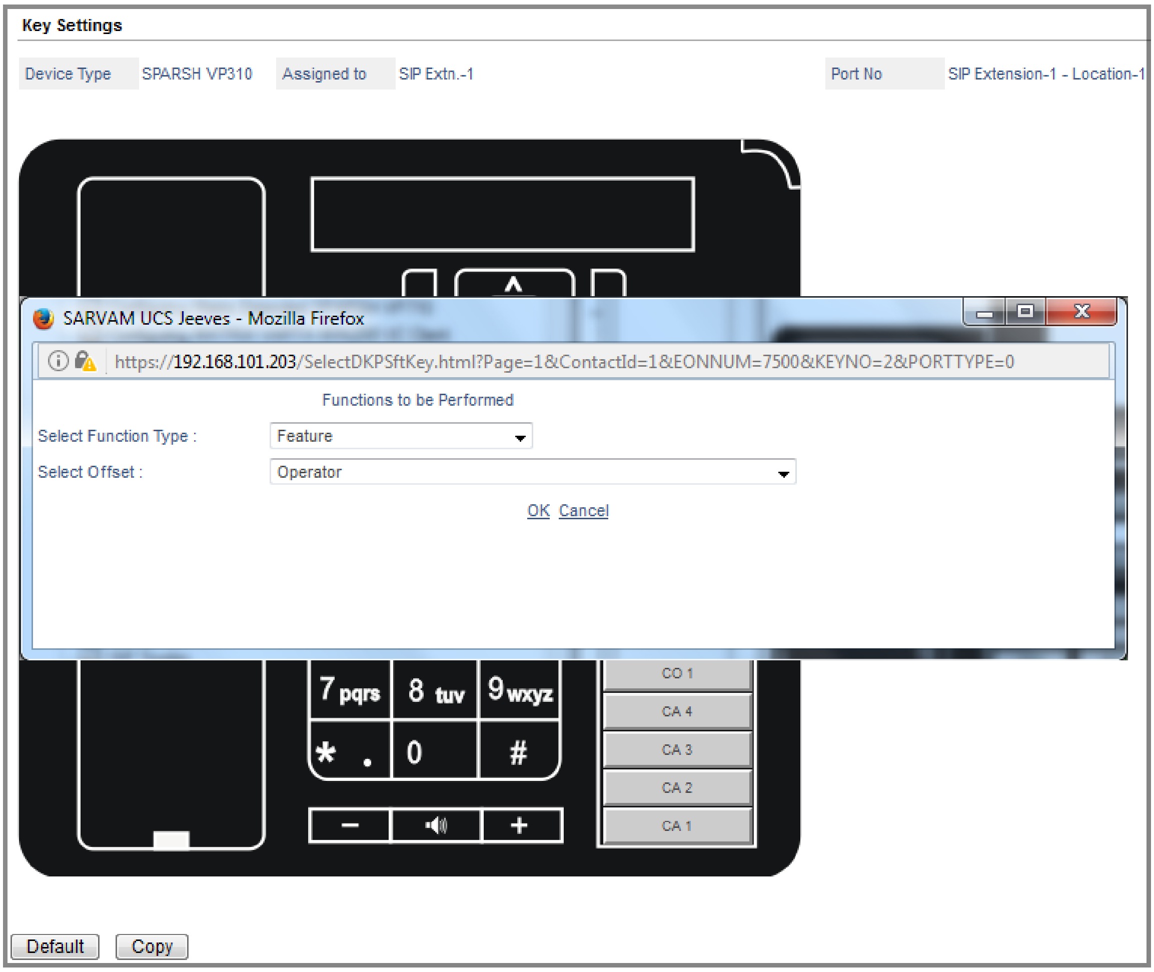
To customize the keys of the Extended IP Phone registered at this location,

- Click **Key Settings** link.

  The default key map of the Extended IP Phone appears on your screen.

- Click the key you want to configure. For example, **None**.

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**.

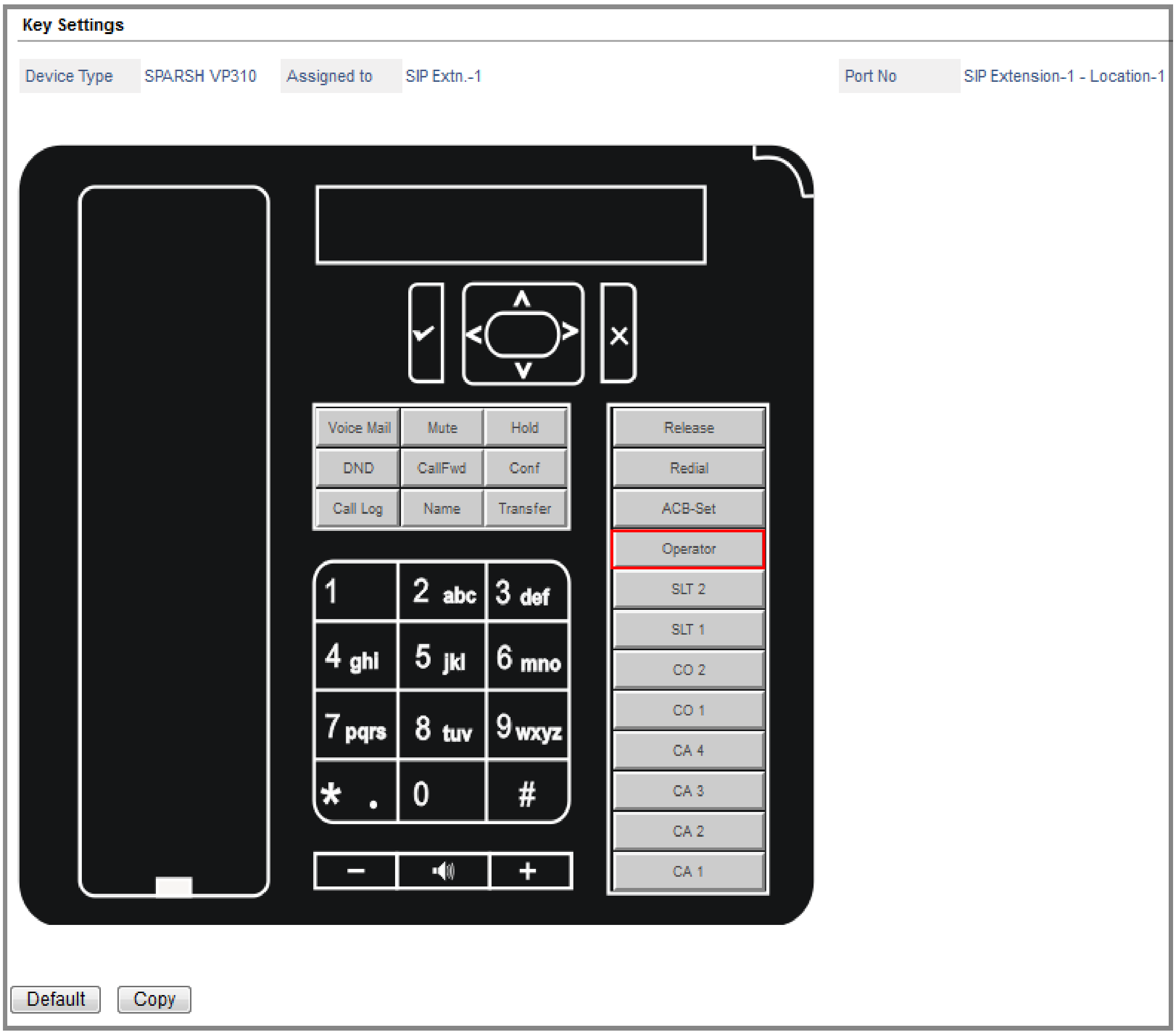
  The Operator function is a Feature, so select the option **Feature** from the **Select Function Type** list box.

  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. Select the appropriate Function Type and 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 SIP extension, select **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 as Offset.

  To assign direct access to a SIP Trunk, select **SIP Trunk** as Function Type and the desired 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
    - Acknowledge
    - Local Menu

If you have completed the configuration of the Matrix IP Phone Settings at location 1, follow the same steps as described above to configure the Matrix Extended IP Phone at the other two locations.

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.

- Enable SIP Extension
- SIP ID
- Authentication ID
- Authentication Password
- Registrar Server Address
- MAC Address
- Enable Device
- Device Type
- Phone Settings
- Call Progress Tone
- Date and Time
- Apply Daylight Savings Time?
- Quality of Service
- RTP Ports
- NAT Keep Alive
- Transport Mode
- Enable SRTP?
- Timers
- Authenticate INVITE
- The SE Password of SARVAM UCS is changed
- You restart the System
- Set the System to Default
- The Network Parameters are changed
- Specific parameters in the VoIP page are changed

Configuring Matrix SPARSH VP330

SPARSH VP330 is the proprietary Extended IP Phone with graphical touch-screen user interface, supplied by Matrix. To know the list of featured supported by Matrix SPARSH VP330, 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 as SIP Extension, you must configure the following:
- SIP Extension General Parameters, see“Configuring SIP Extension General Parameters”.
- SIP Extension Settings, see “Configuring Matrix Extended Phone Settings using Jeeves”.

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, Matrix Extended SPARSH VP710, SPARSH VP210, Matrix VARTA UC Clients 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 SPARSH VP248”.

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”.

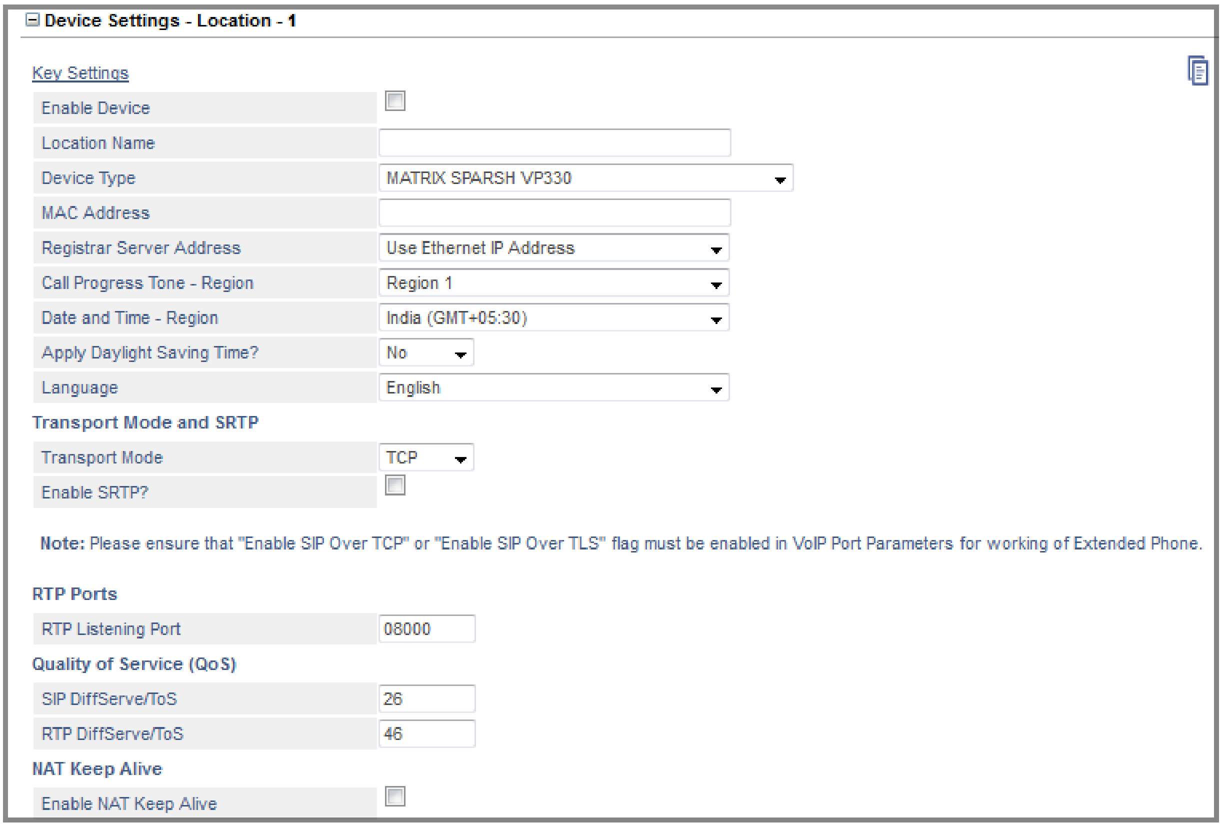
If you have connected SPARSH VP210 at any of the locations, refer to “Configuring Matrix SPARSH VP210”.

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 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 as **Location 1**, **Location 2** and **Location 3**.

- Click **Device Settings - Location 1** to expand.



- 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 and click **Submit** to save.

- Enter the **MAC Address**[40] of the SPARSH VP330 connected at this location in hexadecimal format: 00:50:C2:55:B0:10. 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 Extended IP phone is in the same network (LAN) as SARVAM UCS, select **Use Ethernet IP Address** as Registrar Server Address.

40. *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 Extended IP Phone is in the Global Network and SARVAM UCS is connected to Internet over Mobile WAN, select **Use Mobile Port IP Address** as Registrar Server Address.
- If the Extended IP 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.
- If you want SARVAM UCS to assign Registrar Server Address automatically, select **Automatic** as Registrar Server Address.

- To set the call progress tone generation standards of the country where the Matrix IP Phone is installed, select the **Call Progress Tone - Region**. Default: Region 1.

  See "Call Progress Tones" to know more.
- Select the region/country whose **Date and Time - Region** settings the IP Phone should follow. 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'.

- Select the **Language** for the IP phone. Default: English.

  SARVAM UCS provides language support for English, French, German, Spanish, Portuguese, and Italian on the Matrix IP Phone. 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 **Transport Mode** and enable **SRTP**:

  - Select the protocol to be used to transport the SIP messages. You can select the **Transport Mode** as **TCP** or **TLS**[41].

- *If you select TCP, make sure the SIP Over TCP is selected in VoIP Port Parameters.*
- *If you select TLS, make sure the SIP Over TLS is selected in VoIP Port Parameters.*

  - For secure conversations over SIP, select the **Enable SRTP?** check box. The SIP messages will be transported over SRTP only.

- Define **RTP Port**:

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41. *SPARSH VP330 supports TLS Version V1.0 only. To configure the TLS version, refer* *"Advance Options" in Security Settings.*

- **RTP Listening Port:** This is the port on which the IP phone listens for SIP messages over TCP. 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.

- 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.

- If the IP phone is connected behind a NAT router, configure **NAT Keep Alive**.
  - Select the **Enable NAT Keep Alive** check box 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 9999 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.

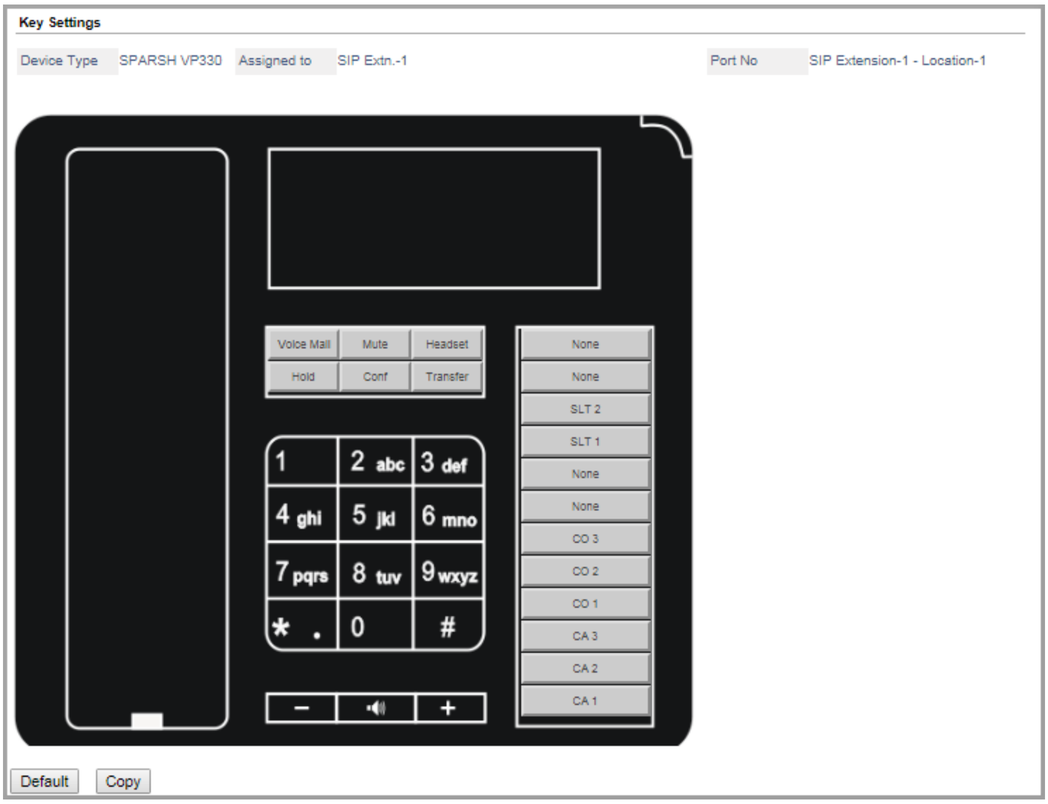
- Set the following **Timers** to the desired value, where required:
  - **SIP INVITE Timer (sec):** This is the time in seconds that the phone waits for a response from the called party after sending INVITE message. This timer starts after sending INVITE message to the called party and stops on receipt 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**.

Phone Key Settings

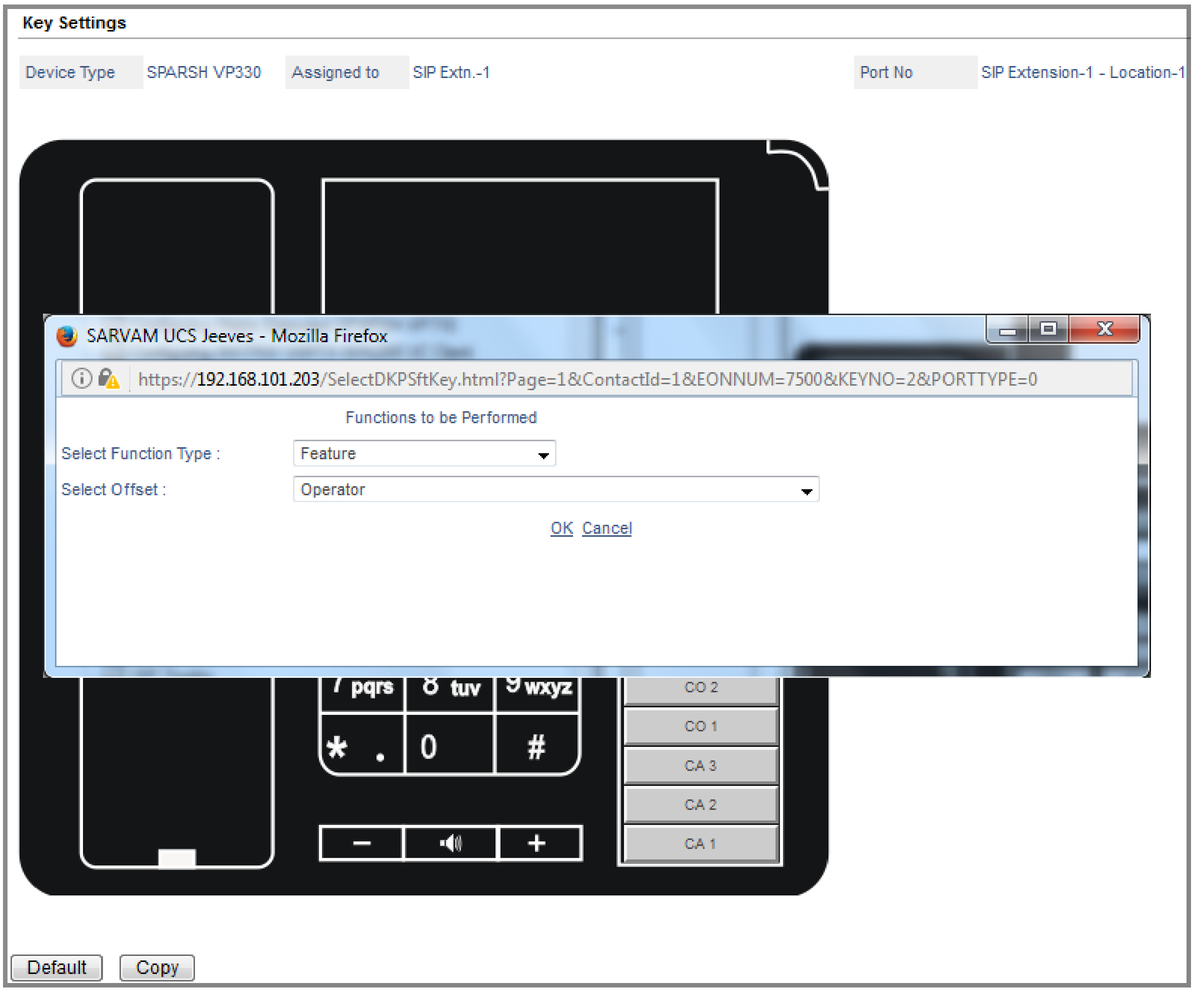
- To personalize the key map of the SPARSH VP330, click the **Key Settings** link.

- The key map of the Extended Phone opens in a new window on your screen.



- Click the key you want to configure. For example, **SLT 1**.

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 SIP extension, select **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 Trunk** as Function Type and the desired trunk number from **1** to **8** 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
	Emergency Conference
	Self Ring Test
	SA Command Prefix
	Department Group Call Forward

- 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.

  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:
  - Enable SIP Extension
  - SIP ID
  - Name
  - Authentication ID
  - Authentication Password
  - Authenticate INVITE
  - Registrar Server Address
  - MAC Address
  - Enable Device
  - Device Type
  - Phone Settings
  - Language
  - Transport Mode
  - Enable SRTP?
  - Call Progress Tone
  - Date and Time
  - Apply Daylight Saving Time?
  - Quality of Service (QoS)

- RTP Ports
- NAT Keep Alive
- 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
- You restart the System
- Set the System to Default
- The Network Parameters are changed
- Specific parameters in the VoIP page are changed

Configuring Matrix SPARSH VP510

SPARSH VP510, the Premium IP Phone sets the benchmark for quality performance with elegant design, crystal-clear voice and feature-rich capabilities at economical price. To know the list of featured supported by Matrix SPARSH VP510, 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".
- Extended IP Phone Settings, see "Configuring Matrix Extended Phone Settings using Jeeves".

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 VP510, SPARSH VP330, SPARSH VP210, Matrix Extended SPARSH VP710, Matrix VARTA UC Clients 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 want to use more than one SPARSH VP510 Extended IP Phones as a SIP Extension, configure their settings as **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".

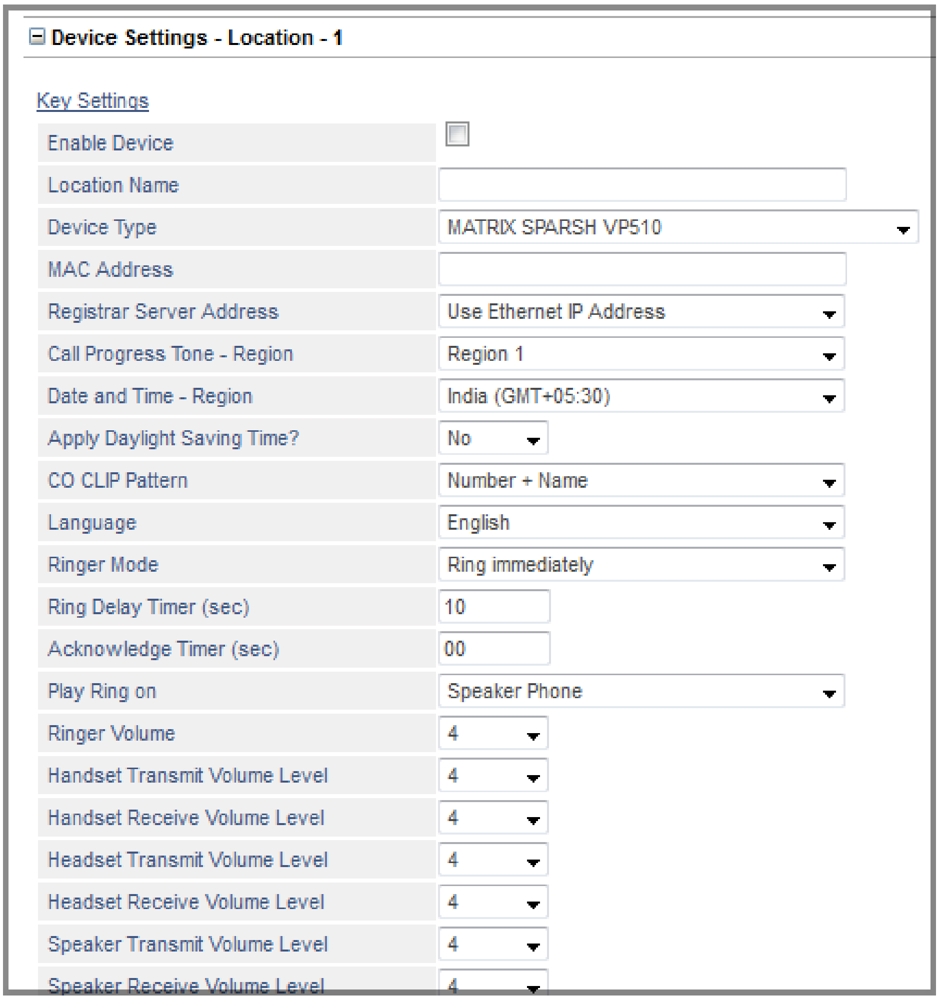
If you have connected Extended SPARSH VP710 at any of the locations, refer to "Configuring Matrix Extended SPARSH VP710".

If you have connected 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 UC Client in any of the locations, refer to "Configuring MATRIX VARTA WIN200 UC Client".

- Click **Device Settings - Location 1** to expand.



- 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 and click **Submit** to save.
- Enter the **MAC Address**[42] 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.

42. *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.*

- 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 Extended IP phone is in the same network (LAN) as SARVAM UCS, select **Use Ethernet IP Address** as Registrar Server Address.

  - If the Extended IP phone is in the Global Network and SARVAM UCS is connected to Internet over Mobile WAN, select **Use Mobile Port IP Address** as Registrar Server Address

  - If the Extended IP 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.

  - If you want SARVAM UCS to assign Registrar Server Address automatically, select **Automatic** as Registrar Server Address.

    By default, Use Ethernet 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'.

- 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 delay (if the call is still not answered).
    - Silent.

  Default: Ring Immediate.

- If you have 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'.

- 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 '*Headset Connected?'* parameter to Yes and connect a Headset to the IP Phone.

- 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 IP phone, set **Headset Connected?** to **Yes.** Default: No.

  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 9999 seconds. Default: 10 seconds.
- Set the **LCD Contrast Level** to a level from 1 to 4 that is comfortable to you. Default: 3.
- Select **Transport Mode** and enable **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 is selected in VoIP Port Parameters.*
  - *If you select TLS, make sure the SIP Over TLS is selected in VoIP Port 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.

- Define **RTP Port**.

- **RTP Listening Port:** This is the port on which the IP phone listens for SIP messages over TCP. 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.

- Set the **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.

- If the IP phone is connected behind a NAT router, configure **NAT Keep Alive**.

  - Select the **Enable NAT Keep Alive** check box 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.

- 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 sending 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.

- To debug using Syslog Client supported by the phone, configure the **Debug** parameters:

  - Set **Enable Debug?** to Yes. Default: No.

    When the Debug parameter 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 of the remote Syslog Server as **Syslog Server Address**.

  - Enter the **Server Port**, i.e. the address of the Listening Port of the Syslog Server from 1025-65535 or 514. Default: 514.

Syslog uses the UDP as transport protocol and listens on the port 514 (the default listening port).

- Select **Debug Levels** from these options, by selecting the respective check box:
  - SIP
  - System
  - Hardware
  - Call
  - Network
  - VoPP

  The phone supports multiple debug levels. 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**.

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.

Phone Key Settings

You can customize the DSS keys of the Extended IP Phone, to match the requirements of the extension user.

To customize the keys of the Extended IP Phone registered at this location,

- Click **Key Settings** link.

The default key map of the Extended IP Phone appears on your screen.

- Click the key you want to configure. For example, **CO4**.

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. Select the appropriate Function Type and 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 SIP extension, select **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 as Offset.

  To assign direct access to a SIP Trunk, select **SIP Trunk** as Function Type and the desired 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

- Acknowledge
- Local Menu

DSS Key Settings

You can connect four Direct Station Selection Consoles (DSS532) to SPARSH VP510, to increase the number of keys for providing more buttons, for direct single-touch access to extensions and features of SARVAM UCS. For connecting DSS Consoles, refer "Installing DSS532 with SPARSH VP510".

The system will automatically detect the console, if it is connected. If you have connected the DSS532 to SPARSH VP510, you may now configure the DSS Console keys.

To configure the DSS Keys,

- 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.

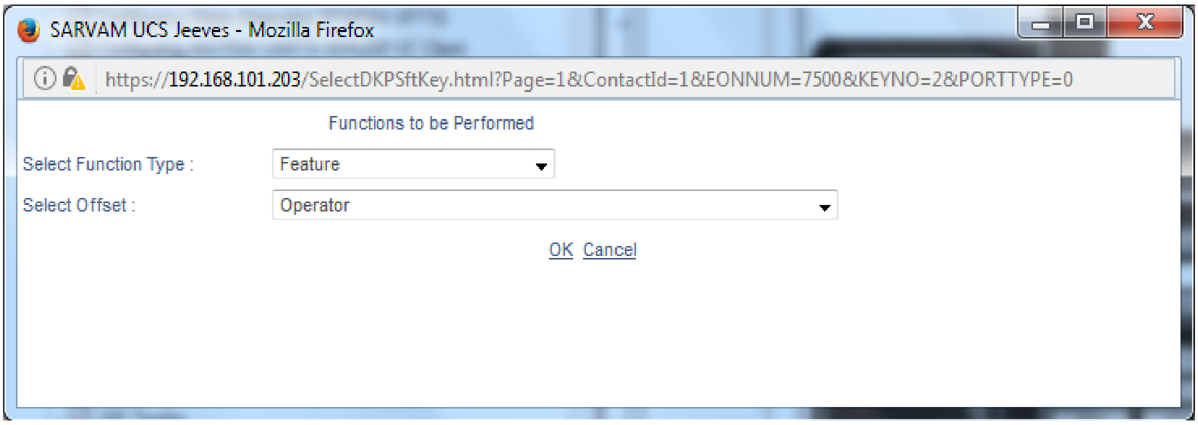
- Click **ADD DSS**.

- 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*
- *Acknowledge*
- *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” under “System Status”.

If you have completed the configuration of the Matrix IP Phone Settings at location 1, follow the same steps as described above to configure the Matrix Extended IP Phone at the other two locations.

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.

- Enable SIP Extension
- SIP ID

- Authentication ID
- Authentication Password
- Registrar Server Address
- MAC Address
- Enable Device
- Device Type
- Phone Settings
- Call Progress Tone
- Date and Time
- Apply Daylight Savings Time?
- Quality of Service
- RTP Ports
- NAT Keep Alive
- Transport Mode
- Enable SRTP?
- Timers
- Authenticate INVITE
- The SE Password of SARVAM UCS is changed
- You restart the System
- Set the System to Default
- The Network Parameters are changed
- Specific parameters in the VoIP page are changed

Configuring Matrix Extended SPARSH VP710

Extended SPARSH VP710[43], 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 SPARSH VP710 - Extended IP Phone, you must configure the following:

- SIP Extension General Parameters, see "Configuring SIP Extension General Parameters".
- Extended IP Phone Settings, see "Configuring Matrix Extended Phone Settings using Jeeves".

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 VP510, SPARSH VP330, SPARSH VP210, Matrix Extended SPARSH VP710, Matrix VARTA UC Clients 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 want to use more than one Extended SPARSH VP710 IP Phones as a SIP Extension, configure their settings as **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 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 UC Client in any of the locations, refer to "Configuring MATRIX VARTA WIN200 UC Client".

- Click **Device Settings - Location 1** to expand.

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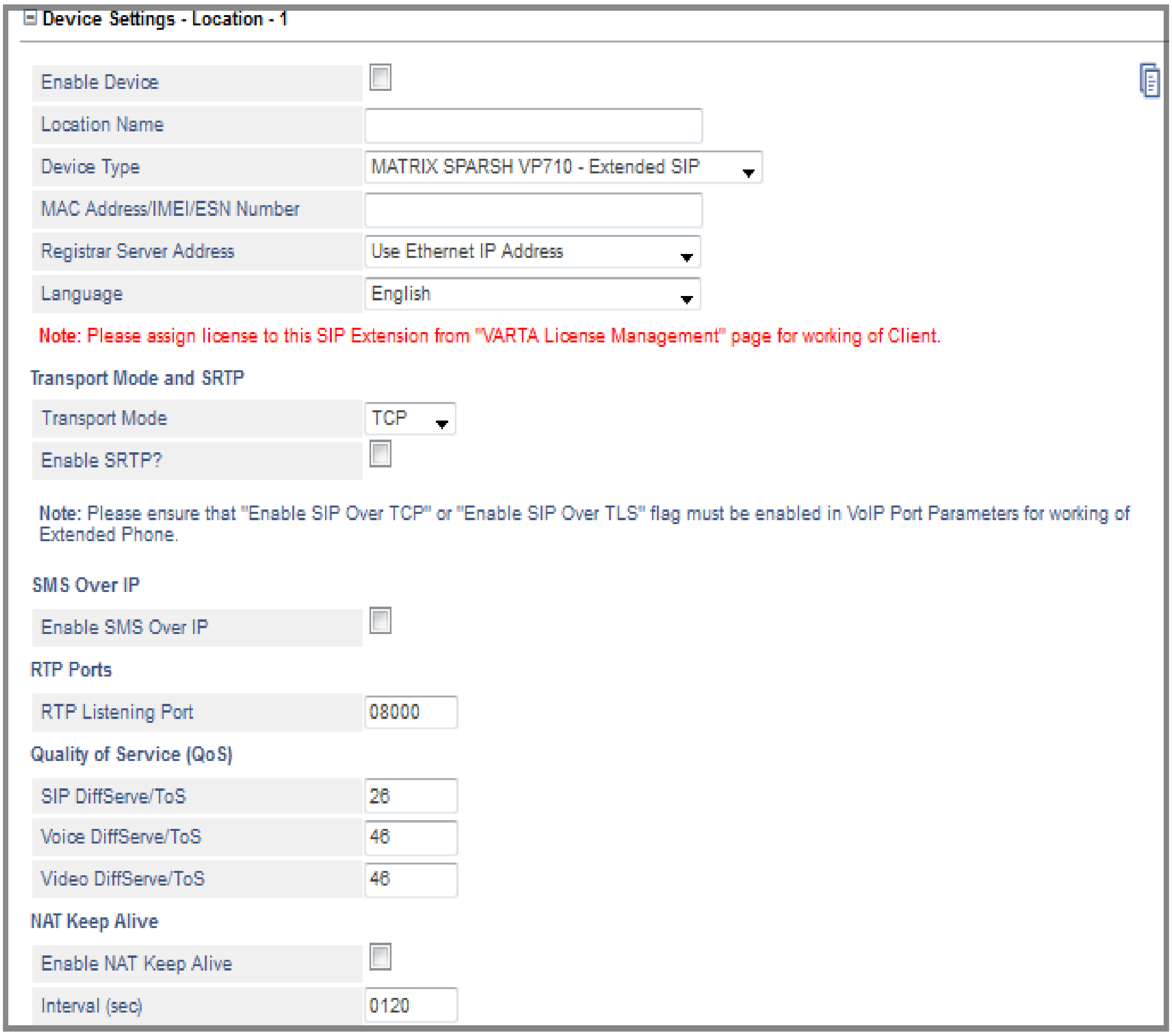
43. *Check for availability.*



- 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 and click **Submit** to save.

- Enter the **MAC Address**[44] 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.

44. *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.*

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 Extended IP phone is in the same network (LAN) as SARVAM UCS, select **Use Ethernet IP Address** as Registrar Server Address.
    - If the Extended IP phone is in the Global Network and SARVAM UCS is connected to Internet over Mobile WAN, select **Use Mobile Port IP Address** as Registrar Server Address
    - If the Extended IP 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.
    - If you want SARVAM UCS to assign Registrar Server Address automatically, select **Automatic** as Registrar Server Address.

      By default, Use Ethernet 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.*

- Select **Transport Mode** and enable **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 is selected in VoIP Port Parameters.*
- *If you select TLS, make sure the SIP Over TLS is selected in VoIP Port 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.

- 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"

- Define **RTP Port**.
    - **RTP Listening Port:** This is the port on which the IP phone listens for SIP messages over TCP. 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.
- Set the **Quality of Service (QoS)** for SIP signaling as:
    - **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
CS_5	101 000	40	5
EF	101 110	46	5

DSCP <=> IP Precedence Conversion Table			
DSCP Name	**DS Field Binary**	**Value Decimal**	**IP Precedence**
CS_6	110 000	48	6
CS_7	111 000	56	7
CS Class Selector (RFC_{2474})			
AFxy Assure Forwarding (x=class, y=drop precedence) (RFC_{2597})			
EF Expedited Forwarding (RFC_{3246})			

- If the IP phone is connected behind a NAT router, configure **NAT Keep Alive**.
  - Select the **Enable NAT Keep Alive** check box 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 9999 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.
- 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 sending 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**.

If you have completed the configuration of the Matrix IP Phone Settings at location 1, follow the same steps as described above to configure the Matrix Extended IP Phone at the other two locations.

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.
- Enable SIP Extension
- SIP ID

- Authentication ID
- Authentication Password
- Registrar Server Address
- MAC Address
- Enable Device
- Device Type
- Phone Settings
- Call Progress Tone
- Date and Time
- Apply Daylight Savings Time?
- Quality of Service
- RTP Ports
- NAT Keep Alive
- Transport Mode
- Enable SRTP?
- Timers
- Authenticate INVITE
- The SE Password of SARVAM UCS is changed
- You restart the System
- Set the System to Default
- The Network Parameters are changed
- Specific parameters in the VoIP page are changed

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 Extension, you must configure the following:

- SIP Extension General Parameters, see“Configuring SIP Extension General Parameters”.
- SIP Extension Settings, see “Configuring Matrix Extended Phone Settings using Jeeves”.

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, Matrix Extended SPARSH VP710, SPARSH VP210, Matrix VARTA UC Clients 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 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”.

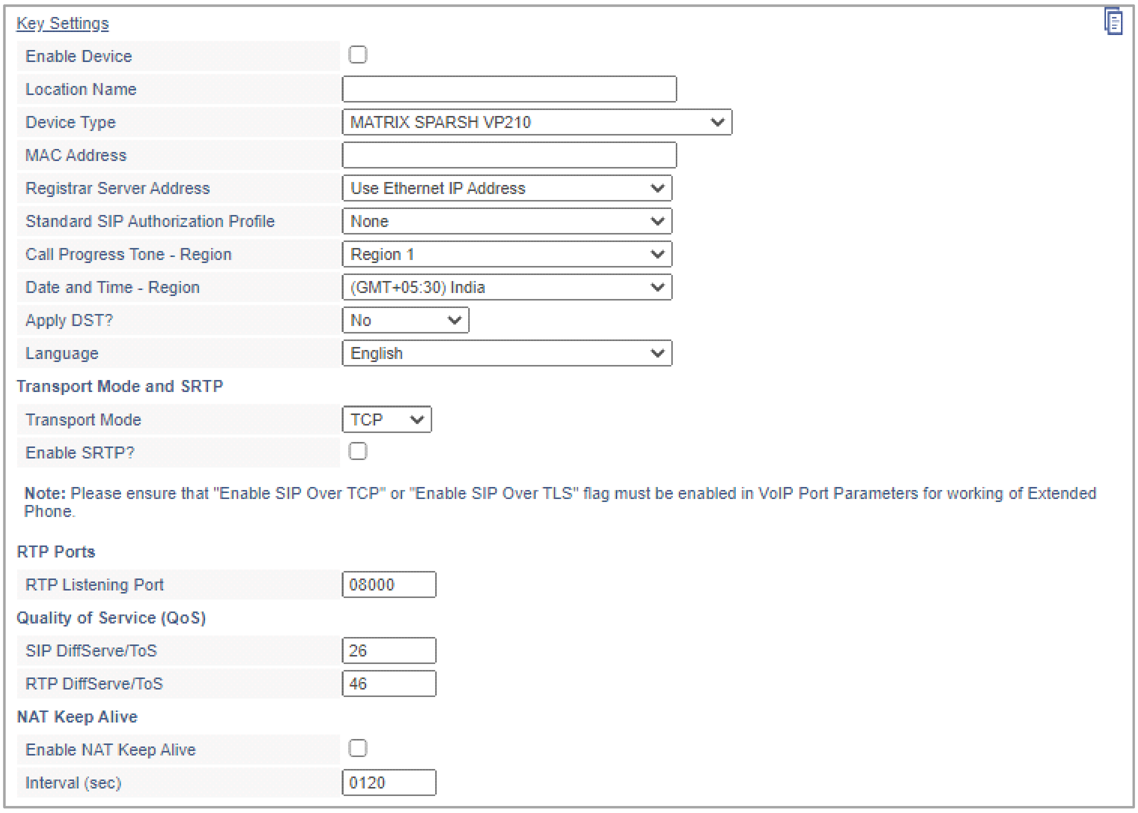
If you have connected Extended SPARSH VP710 at any of the locations, refer to “Configuring Matrix Extended SPARSH VP710”.

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 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 as **Location 1**, **Location 2** and **Location 3**.

- Click **Device Settings - Location 1** to expand.



- 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 and click **Submit** to save.
- Enter the **MAC Address**[45] of the SPARSH VP210 connected at this location in hexadecimal format: 00:50:C2:55:B0:10. 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 Extended IP phone is in the same network (LAN) as SARVAM UCS, select **Use Ethernet IP Address** as Registrar Server Address.

45. *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 Extended IP Phone is in the Global Network and SARVAM UCS is connected to Internet over Mobile WAN, select **Use Mobile Port IP Address** as Registrar Server Address.
- If the Extended IP 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.
- If you want SARVAM UCS to assign Registrar Server Address automatically, select **Automatic** as Registrar Server Address.

- To set the call progress tone generation standards of the country where the Matrix IP Phone is installed, select the **Call Progress Tone - Region**. Default: Region 1.

  See "Call Progress Tones" to know more.
- Select the region/country whose **Date and Time - Region** settings the IP Phone should follow. 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'.

- Select the **Language** for the IP phone. Default: English.

  SARVAM UCS provides language support for English, French, German, Spanish, Portuguese, and Italian on the Matrix IP Phone. 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 **Transport Mode** and enable **SRTP**.

  - Select the protocol to be used to transport the SIP messages. You can select the **Transport Mode** as **TCP** or **TLS**[46].

- *If you select TCP, make sure the SIP Over TCP is selected in VoIP Port Parameters.*
- *If you select TLS, make sure the SIP Over TLS is selected in VoIP Port Parameters.*

  - For secure conversations over SIP, select the **Enable SRTP?** check box. The SIP messages will be transported over SRTP only.

- Define **RTP Port**.

46. *SPARSH VP210 supports TLS Version V1.0 only. To configure the TLS version, refer "Advance Options" in Security Settings.*

- **RTP Listening Port:** This is the port on which the IP phone listens for SIP messages over TCP. 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.

- 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.

- If the IP phone is connected behind a NAT router, configure **NAT Keep Alive**.
    - Select the **Enable NAT Keep Alive** check box 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 9999 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.

- 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 sending 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.

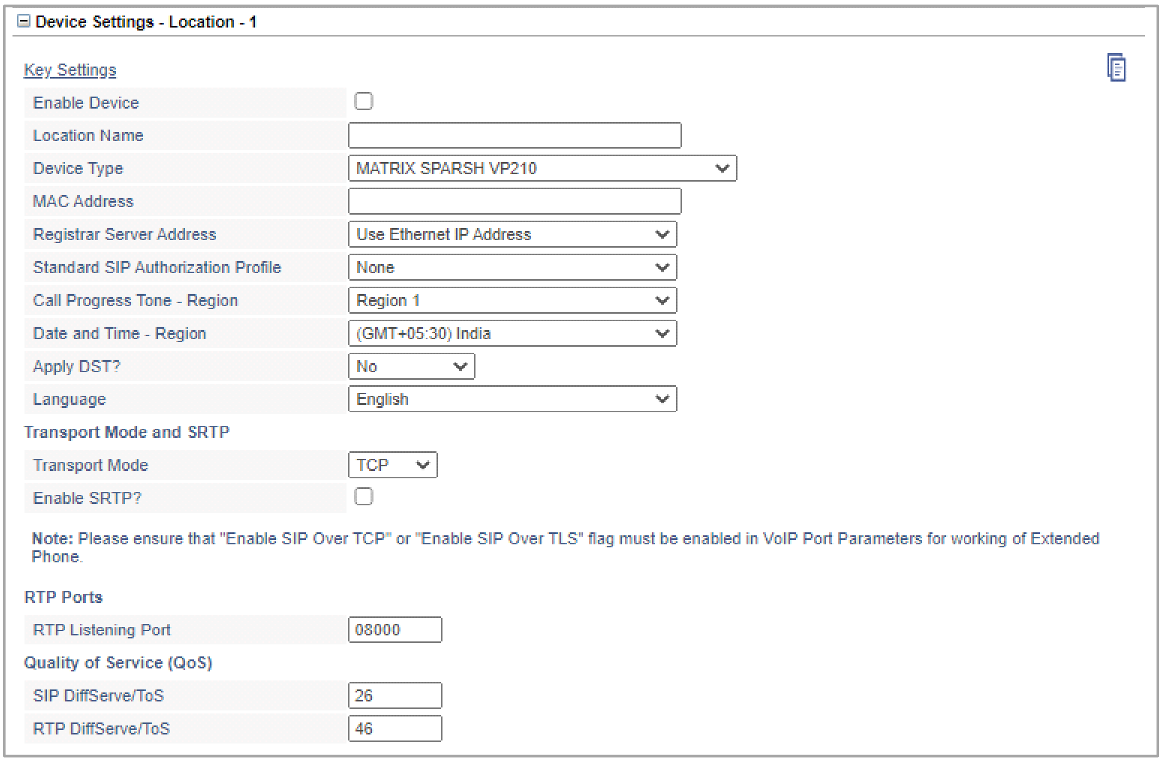
- Click **Submit** to save settings.

Phone Key Settings

You can customize the DSS keys of the Extended IP Phone, to match the requirements of the extension user.

To customize the keys of the Extended IP Phone registered at this location,

- Click **Key Settings** link.



- 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.

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.

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 VARTA WIN200 UC Client

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 premium features like Presence subscription and notification, Corporate Voicemail access to enhance your overall mobile experience. To know the list of featured supported, refer to “SARVAM UCS Features supported in Terminals”.

To use MATRIX VARTA WIN200 make sure you have:

- Purchased and activated the VARTA Essential or VARTA Professional license. For more details, see “License Management”.
- Assigned the desired license to the SIP Extension. For more details, see “VARTA License Management”.

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 MATRIX VARTA WIN200 for Windows - Desktop Client, you must configure the following:

- SIP Extension General Parameters, see “Configuring SIP Extension General Parameters”.
- Matrix Extended Phone Settings, see “Configuring Matrix Extended Phone Settings using Jeeves”.

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, Matrix Extended SPARSH VP710, SPARSH VP210, Matrix VARTA UC Clients or MATRIX VARTA WIN200 Desktop UC Client — at each location. In this case we assume that MATRIX VARTA WIN200 UC Client is registered at Location 1, 2 and 3.

If you want to use more than one MATRIX VARTA WIN200 for Windows - Desktop Client 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.*

If you have connected Matrix SPARSH VP248, refer to “Configuring Matrix SPARSH VP248”.

If you have connected Matrix SPARSH VP310, refer to “Configuring Matrix SPARSH VP310”.

If you have connected SPARSH VP510 at any of the locations, refer to “Configuring Matrix SPARSH VP510”.

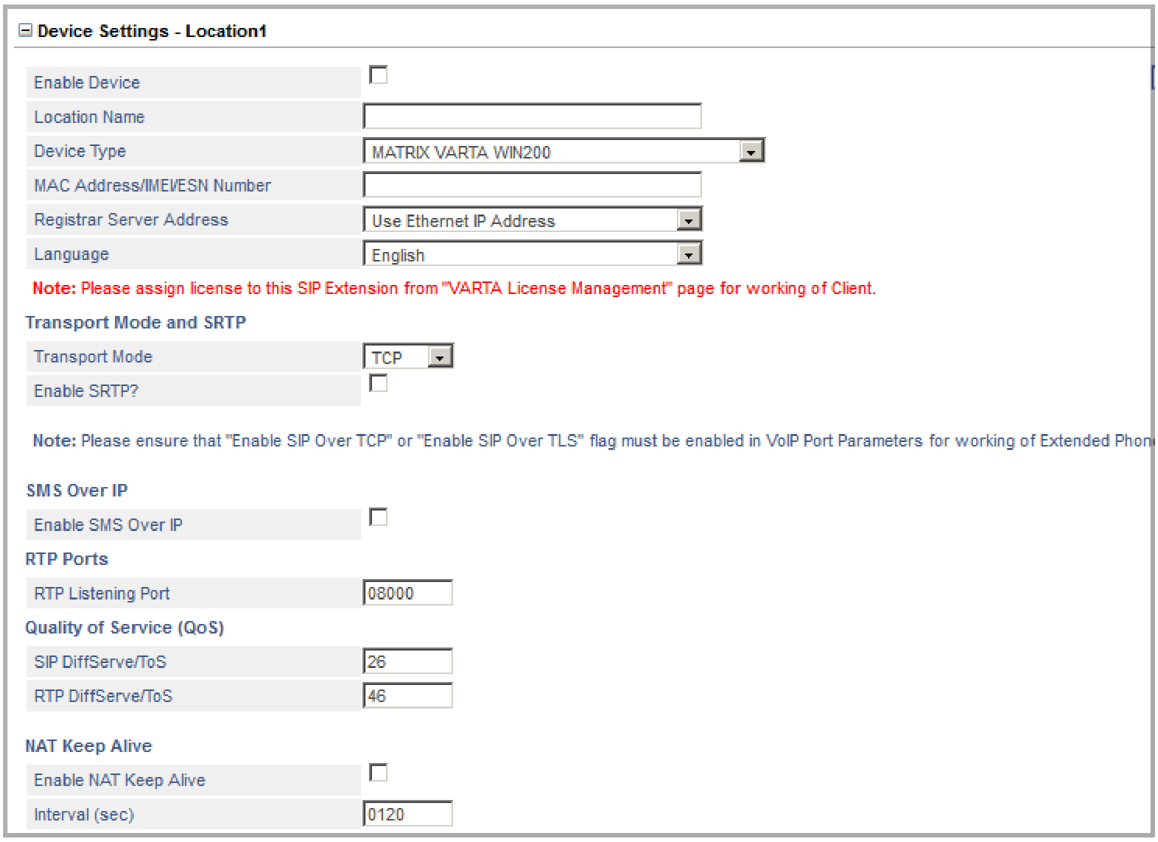
If you have connected Matrix SPARSH VP330, refer to “Configuring Matrix SPARSH VP330”.

If you have connected Extended SPARSH VP710 at any of the locations, refer to “Configuring Matrix Extended SPARSH VP710”.

If you have connected 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”.

- Click **Device Settings - Location 1** to expand.



- Select the **Enable Device** check box. Default: Disabled.

- Enter the **Location Name** of your choice. 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/IMEI/ESN Number**, enter the Device ID. 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 Ethernet IP Address** as Registrar Server Address.

- If the UC Client is in the Global Network and SARVAM UCS is connected to Internet over Mobile WAN, select **Use Mobile 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 the **VoIP Port Parameters**. See "VoIP Parameters".

- If Dynamic DNS is configured in the Network Parameters, select **Use Dynamic DNS Host Name** as External Registrar Server Address.

- If you want SARVAM UCS to assign Registrar Server Address automatically, select **Automatic** as Registrar Server Address.

  Default: Use Ethernet IP Address

- Select the **Language** for the application. Default: English.

  SARVAM UCS provides language support for English, French, German, Spanish, Portuguese, and Italian in the Application. When you select any of these languages, all the prompts and command strings will appear in the selected language.

- 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 Port Parameters.*
  - *If you select TLS, make sure the SIP Over TLS check box is selected in VoIP Port Parameters.*

- For secure conversations over SIP, select the **Enable SRTP?** check box. The SIP messages will be transported over SRTP only.

- If you want Desktop 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".

- 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.

- 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:

Traffic Type	DSCP Value (dec)
Best Effort	0
Background	8
Excellent Effort	40
AudioVideo	40
Voice	56
Control	56

- If the Matrix VARTA Desktop Clients is connected behind a NAT router, configure **NAT Keep Alive**.
  - Select the **Enable NAT Keep Alive** check box, 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 9999 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.
- 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 sending 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**.
- If you have completed the configuration of the UC Client at Location 1, follow the same steps as described above to configure the UC Client at Location 2 and Location 3.

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
- Language
- 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
- You restart the System
- Set the System to Default
- The Network Parameters are changed
- Specific parameters in the VoIP page are changed

Configuring Matrix VARTA ADR100/AMP100 UC Clients

Matrix offers the following Mobile UC Clients[47]:

- MATRIX VARTA ADR100 for Android Smartphones/Tablets
- MATRIX VARTA AMP100 for iPhones

To use MATRIX VARTA UC Clients for Mobile, you must make sure:

- Purchase and activate the VARTA Essential, VARTA Professional or VARTA Collaboration license. For more details, see “License Management”.
- Assign 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 “Auto Sign-In Parameters”
  - Configure the **General Parameters** in **SIP Extensions Settings**. For details, refer “SIP Extensions”
  - Make sure you send the **Auto Sign-In Configuration Mail**. For details, refer “SIP Extensions”.

    You can also view the status of Auto Sign-In Email in “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.*

47. *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 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.

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 as Matrix SPARSH Mobile Softphone Clients, you must configure the following:

- SIP Extension General Parameters, see "Configuring SIP Extension General Parameters".
- Matrix Extended Phone Settings, see "Configuring Matrix Extended Phone Settings using Jeeves".

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, Matrix Extended SPARSH VP710, SPARSH VP210, Matrix VARTA UC Clients or MATRIX VARTA WIN200 Desktop UC Client — at each location. In this case we assume that Matrix SPARSH Mobile Softphone Clients are registered at Location 1, 2 and 3.

If you want to use more than one Matrix VARTA UC Clients as SIP Extensions, configure their settings as **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.*

If you have connected SPARSH VP248 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 VP510 at any of the locations, refer to "Configuring Matrix SPARSH VP510".

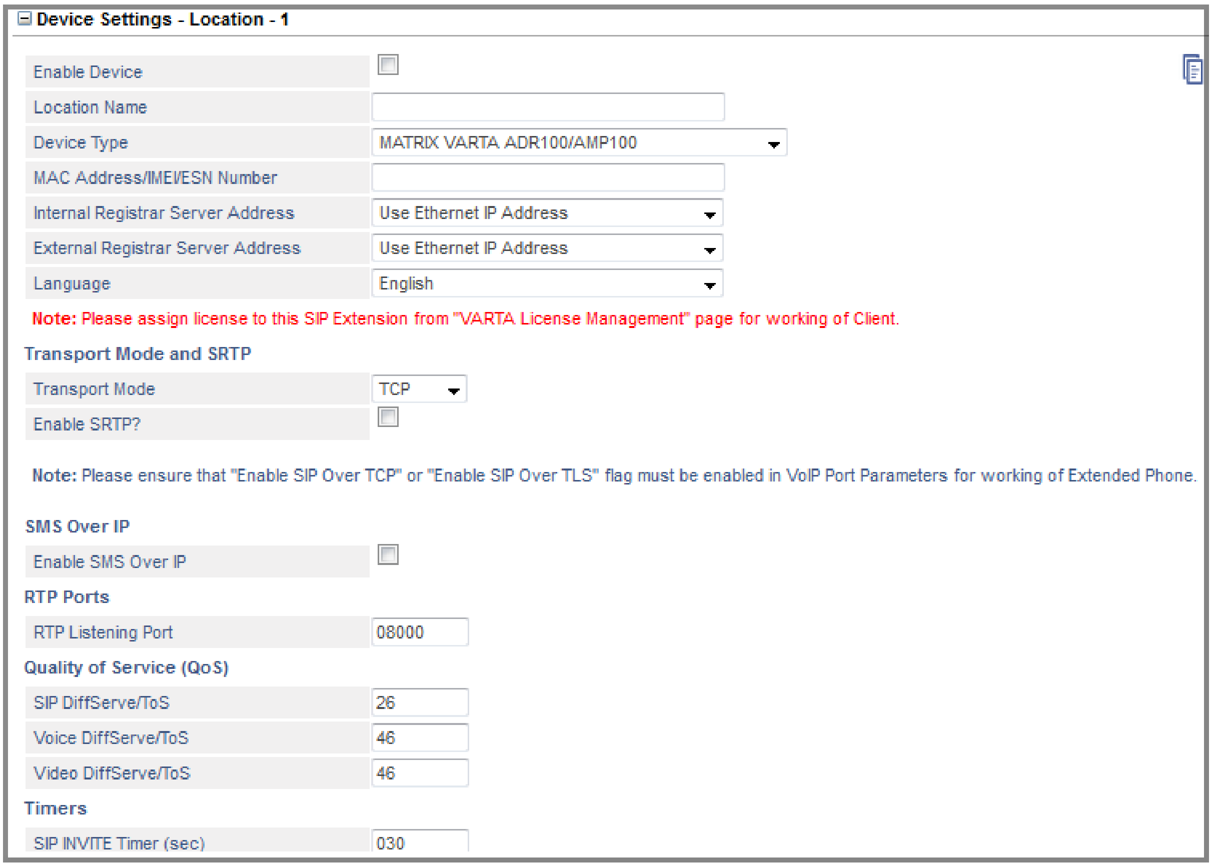
If you have connected SPARSH VP330 refer to "Configuring Matrix SPARSH VP330".

If you have connected Extended SPARSH VP710 at any of the locations, refer to "Configuring Matrix Extended SPARSH VP710".

If you have connected SPARSH VP210 at any of the locations, refer to "Configuring Matrix SPARSH VP210".

If you have registered MATRIX VARTA WIN200 UC Client in any of the locations, refer to “Configuring MATRIX VARTA WIN200 UC Client”.

- Click **Device Settings - Location 1** to expand.



- Select the **Enable Device** check box. Default: Disabled.

- Enter the **Location Name** of your choice. 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 and click **Submit** to save.

- Enter the **MAC Address/IMEI Number**[48]**/ESN Number** of the Matrix SPARSH Mobile Softphone Client.

  If you are using an iPhone, enter the **Device ID** here. Default: blank.

  SARVAM UCS validates the Matrix SPARSH Mobile Softphone Clients on the basis of the MAC Address/ IMEI Number/ESN number or 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 MAC Address/IMEI Number/ESN number or 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:

---

48. IMEI Number is the unique identification number of the GSM engine used in the Mobile handset.

- If you want the application to register using the LAN/WAN network, select **Use Ethernet 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 Internal Registrar Server Address.

- If you want SARVAM UCS to assign Internal Registrar Server Address automatically, select **Automatic** as the Internal Registrar Server Address.

  By default, Use Ethernet 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 Ethernet Port IP Address** as External Registrar Server Address.

  - If you want the application to register in the Global Network and SARVAM UCS is connected to Internet over Mobile WAN, select **Use Mobile Port IP Address** as External Registrar Server Address.

  - If you want the application to register 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 the **VoIP Port Parameters**. See "VoIP Parameters".

  - If Dynamic DNS is configured in the Network Parameters, select **Use Dynamic DNS Host Name** as External Registrar Server Address.

  - If you want SARVAM UCS to assign External Registrar Server Address automatically, select **Automatic** as the External Registrar Server Address.

    By default, Use Ethernet Port IP Address is selected as the External Registrar Server Address.

- Select the **Language** for the application. Default: English.

  SARVAM UCS provides language support for English, French, German, Spanish, Portuguese, and Italian in the Application. When you select any of these languages, all the prompts and command strings will appear in the selected language.

- 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 Port Parameters.*

  - *If you select TLS, make sure the SIP Over TLS check box is selected in VoIP Port Parameters.*

  For secure conversations over SIP, select the **Enable SRTP?** check box. The SIP messages will be transported over SRTP only.

- If you want Mobile 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".

- 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.

- 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 *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
CS_5	101 000	40	5

DSCP <=> IP Precedence Conversion Table			
DSCP Name	**DS Field Binary**	**Value Decimal**	**IP Precedence**
EF	101 110	46	5
CS_6	110 000	48	6
CS_7	111 000	56	7
CS Class Selector (RFC_{2474})			
AFxy Assure 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 sending 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.

  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:
  - Enable SIP Extension
  - SIP ID
  - Name
  - Authentication ID
  - Authentication Password
  - Authenticate INVITE
  - Enable Device
  - Device Type
  - Internal Registrar Server Address

- External Registrar Server Address
- MAC Address/IMEI/ESN Number
- Language
- Quality of Service
- Transport Mode
- Enable SRTP
- RTP Ports
- Timers
- Class of Service
- Trunk Access Code
- The SE Password of SARVAM UCS is changed
- You restart the System
- Set the System to Default
- The Network Parameters are changed
- Specific parameters in the VoIP page are changed

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 “SIP Extensions”.
- Make sure you send the **Auto Sign-In Configuration Mail**. For details, refer “SIP Extensions”.

You can also view the status of Auto Sign-In Email in “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 “SIP Extension Status”.

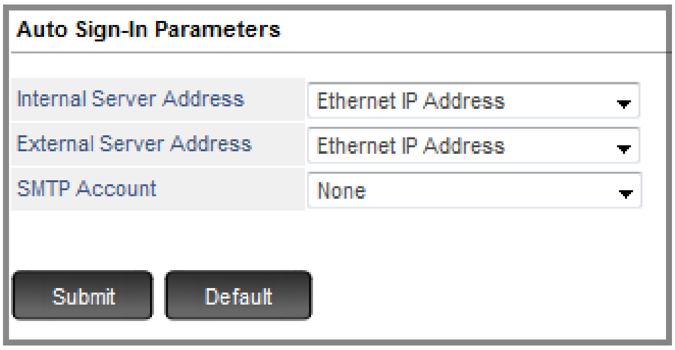
Configuring Auto Sign-In Parameters

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.

- Login as System Engineer.

- Under **Advanced Settings**, 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 LAN/WAN network, select **Ethernet Port IP Address** as the Internal Registrar Server Address.
  - If Dynamic DNS is configured in the Network Parameters, select **Dynamic DNS Host Name** as Internal Registrar Server Address.
  - If you do not want to configure the Internal Server Address, select **Don't Send**.

  By default, Ethernet 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 **Ethernet Port IP Address** as External Registrar Server Address.
  - If you want the application to register in the Global Network and SARVAM UCS is connected to Internet over Mobile WAN, select **Mobile Port IP Address** as External Registrar Server Address.
  - If you want the application to register in the Public Network and SARVAM UCS is located behind a Router, or behind a NAT Router and STUN is programmed, select **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 the **VoIP Port Parameters**. See "Network Parameters".
  - If Dynamic DNS is configured in the Network Parameters, select **Dynamic DNS Host Name** as External Registrar Server Address.
  - If you do not want to configure the External Server Address, select **Don't Send**.

    By default, Ethernet Port IP Address is selected as the External Server Address.

- Select the **SMTP Account**[49] though which you want the email to be sent.

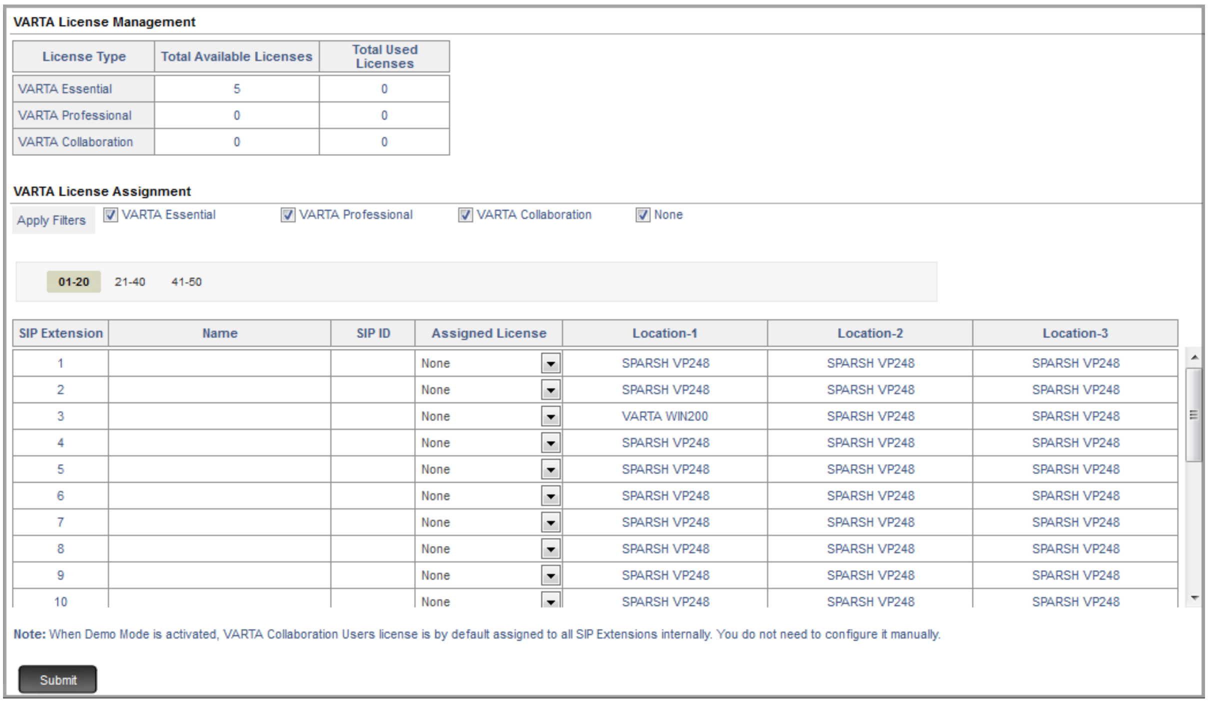
49. *Make sure that the SMTP settings are configured correctly.For more information, refer "SMTP Settings".*

*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.*

VARTA License Management

To view the VARTA License Status and to assign licenses to SIP Extensions,

- Login as System Engineer.
- Under **Advanced Settings**, click the **VoIP Configuration** link.
- Click **VARTA License Management**.



VARTA License Management

License Type	Total Available Licenses	Total Used Licenses
VARTA Essential	5	0
VARTA Professional	0	0
VARTA Collaboration	0	0

VARTA License Assignment

Apply Filters ☑ VARTA Essential ☑ VARTA Professional ☑ VARTA Collaboration ☑ None

01-20 21-40 41-50

SIP Extension	Name	SIP ID	Assigned License	Location-1	Location-2	Location-3
1			None	SPARSH VP248	SPARSH VP248	SPARSH VP248
2			None	SPARSH VP248	SPARSH VP248	SPARSH VP248
3			None	VARTA WIN200	SPARSH VP248	SPARSH VP248
4			None	SPARSH VP248	SPARSH VP248	SPARSH VP248
5			None	SPARSH VP248	SPARSH VP248	SPARSH VP248
6			None	SPARSH VP248	SPARSH VP248	SPARSH VP248
7			None	SPARSH VP248	SPARSH VP248	SPARSH VP248
8			None	SPARSH VP248	SPARSH VP248	SPARSH VP248
9			None	SPARSH VP248	SPARSH VP248	SPARSH VP248
10			None	SPARSH VP248	SPARSH VP248	SPARSH VP248

Note: 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.

Submit

When Demo Mode is activated, VARTA Collaboration license is assigned to all the SIP Extensions[50] 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.

50. *Applicable for ETERNITY NENXIP50.*

VARTA License Assignment

After registering and configuring MATRIX VARTA WIN200/VARTA ADR100/VARTA AMP100 and SPARSH VP710(Extended Mode) 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.
- **Name:** This displays the name of the Extensions.
- **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[51]
- Matrix SPARSH VP110
- Matrix SPARSH VP710 - Standard SIP
- Matrix SPARSH VP210
- 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.*

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 General Parameters**. For details, see "Configuring SIP Extension General Parameters" in "SIP Extensions".[52]

51. *Auto Provisioning is not supported in SPARSH VP248.*
52. *Some of the parameters may not be applicable depending on the Standard SIP Phone you have connected to the SARVAM UCS.*

- Configure specific parameters (for example, SIP ID/User Name, Authentication ID, Password, Server Address/Domain Name etc.) in the Standard SIP Phone/devices which are required to be registered with SARVAM UCS. 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
- SPARSH VP110
- SPARSH VP710 - Standard
- 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 **Device Settings - Location1** only.
- The phone will automatically fetch the configuration file(s) from SARVAM UCS and will get registered.

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.

Using the built-in DHCP Server of SARVAM UCS

- Make sure the Standard SIP Phone is connected in the same subnet as that of the **Ethernet** Port of SARVAM UCS.
- The ‘Auto Provisioning Server Address’ and ‘Server Port’ value will be provided to the Standard SIP Phone using **DHCP Option 66** in the format - **http://IP Address:SPARSH Port of SARVAM UCS**.

Using any third party DHCP Server in your LAN

- Make sure that the third party DHCP Server and your Standard SIP Phone, both are connected in the same subnet as that of the **Ethernet** 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 of SARVAM UCS**.

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 Phone

You are recommended to complete the following steps before connecting **Panasonic UTG200B**:

- Decide the physical location of the Standard SIP Phone.
- Log into Jeeves as the System Engineer.
- Configure "DHCP Server" of SARVAM UCS in the "Network Parameters". The **DHCP Server** of SARVAM UCS will be used for assigning the 'Auto Provisioning Server Address' and 'Server Port' to the Standard SIP Phone.

  If you are using third party DHCP Server in your LAN, see "Using any third party DHCP Server in your LAN".
- Configure the **SIP Extension General Parameters** in SARVAM UCS. For details, see "Configuring SIP Extension General Parameters" in "SIP Extensions".[53]
- Configure the device specific settings applicable to your **Panasonic UTG200B** at Location1 of the SIP Extension page. To do so,
  - Click **SIP Extensions**.
  - Click **Device Settings - Location1** to expand 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 and click **Submit** to save.	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

53. *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
Standard SIP Authorization Profile		Select the desired Standard SIP Authorization Profile from the list of profiles. Default: Panasonic	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 Ethernet IP Address.	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	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	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 Extended IP phone is in the same network (LAN) as SARVAM UCS, select Use Ethernet IP Address as Registrar Server Address.*

If the Extended IP phone is in the Global Network and SARVAM UCS is connected to Internet over Mobile WAN, select Use Mobile Port IP Address as Registrar Server Address

If the Extended IP Phone is connected in the Global Network and SARVAM UCS is located behind a NAT Router or 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.

If you want SARVAM UCS to assign Registrar Server Address automatically, select Automatic 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.
- Log into Jeeves as the System Engineer.
- Configure "DHCP Server" of SARVAM UCS in the "Network Parameters". The **DHCP Server** of SARVAM UCS will be used for assigning the 'Auto Provisioning Server Address' and 'Server Port' to the Standard SIP Phone.

  If you are using third party DHCP Server in your LAN, see "Using any third party DHCP Server in your LAN".
- Configure the **SIP Extension General Parameters** in SARVAM UCS. For details, see "Configuring SIP Extension General Parameters" in "SIP Extensions".[54]
- Configure the device specific settings applicable to your **Grandstream** at Location1 on the SIP Extensions page. To do so,
  - Click **SIP Extensions**.
  - Click **Device Settings - Location1** to expand 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

54. *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.	1. Grandstream GXP110x 2. Grandstream GXP2200 3. Grandstream GXP21xx/116x/14xx 4. Grandstream GXV3140/3175
Device Type	Select Device Type as any of the desired Grandstream Standard SIP Phone you want to connect and click Submit to save.	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 Ethernet 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. Default: Enabled	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 Extended IP phone is in the same network (LAN) as SARVAM UCS, select Use Ethernet IP Address as Registrar Server Address.*

If the Extended IP phone is in the Global Network and SARVAM UCS is connected to Internet over Mobile WAN, select Use Mobile Port IP Address as Registrar Server Address

If the Extended IP Phone is connected in the Global Network and SARVAM UCS is located behind a NAT Router or 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.

If you want SARVAM UCS to assign Registrar Server Address automatically, select Automatic 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.
- Log into Jeeves as the System Engineer.
- Configure "DHCP Server" of SARVAM UCS in the "Network Parameters". The **DHCP Server** of SARVAM UCS will be used for assigning the 'Auto Provisioning Server Address' and 'Server Port' to the Standard SIP Phone.

  If you are using third party DHCP Server in your LAN, see "Using any third party DHCP Server in your LAN".
- Configure the **SIP Extension General Parameters** in SARVAM UCS. For details, see "Configuring SIP Extension General Parameters" in "SIP Extensions".[55]
- Configure the device specific settings applicable to your **Yealink** at Location1 on the SIP Extensions page. To do so,
  - Click **SIP Extensions**.
  - Click **Device Settings - Location1** to expand 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 and click **Submit** to save.	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

55. *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 Ethernet 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. 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 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. Default: DST by Date.	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	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
	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 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
Time Display Format		Select the Time Display Format for the Standard SIP Phone. Default: 24Hr	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 Extended IP phone is in the same network (LAN) as SARVAM UCS, select Use Ethernet IP Address as Registrar Server Address.*

If the Extended IP phone is in the Global Network and SARVAM UCS is connected to Internet over Mobile WAN, select Use Mobile Port IP Address as Registrar Server Address

If the Extended IP Phone is connected in the Global Network and SARVAM UCS is located behind a NAT Router or 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.

If you want SARVAM UCS to assign Registrar Server Address automatically, select Automatic 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.
- Log into Jeeves as the System Engineer.
- Configure "DHCP Server" of SARVAM UCS in the "Network Parameters". The **DHCP Server** of SARVAM UCS will be used for assigning the 'Auto Provisioning Server Address' and 'Server Port' to the Standard SIP Phone.

  If you are using third party DHCP Server in your LAN, see "Using any third party DHCP Server in your LAN".
- Configure the **SIP Extension General Parameters** in SARVAM UCS. For details, see "Configuring SIP Extension General Parameters" in "SIP Extensions".[56]
- Configure the device specific settings applicable to your **Cisco** at Location1 on the SIP Extensions page. To do so,
  - Click **SIP Extensions**.

56. *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.*

- Click **Device Settings - Location1** to expand 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
Device Type		Select Device Type as any of the desired Cisco Phone you want to connect and click **Submit** to save.	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 Ethernet 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: 24Hr	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 Extended IP phone is in the same network (LAN) as SARVAM UCS, select Use Ethernet IP Address as Registrar Server Address.*

If the Extended IP phone is in the Global Network and SARVAM UCS is connected to Internet over Mobile WAN, select Use Mobile Port IP Address as Registrar Server Address

If the Extended IP Phone is connected in the Global Network and SARVAM UCS is located behind a NAT Router or 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.

If you want SARVAM UCS to assign Registrar Server Address automatically, select Automatic 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 Phone

You are recommended to complete the following steps before connecting the **Polycom** Standard SIP Phone:

- Decide the physical location of the Standard SIP Phone.
- Log into Jeeves as the System Engineer.
- Configure "DHCP Server" of SARVAM UCS in the "Network Parameters". The **DHCP Server** of SARVAM UCS will be used for assigning the 'Auto Provisioning Server Address' and 'Server Port' to the Standard SIP Phone.

  If you are using third party DHCP Server in your LAN, see "Using any third party DHCP Server in your LAN".
- Configure the **SIP Extension General Parameters** in SARVAM UCS. For details, see "Configuring SIP Extension General Parameters" in "SIP Extensions".[57]
- Configure the device specific settings applicable to your **Polycom** at Location1on the SIP Extensions page. To do so,
  - Click **SIP Extensions.**
  - Click **Device Settings - Location1** to expand 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

57. *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 Polycom IP Phone and click **Submit** to save.	Polycom IP Phone
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 Ethernet 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

Time Zone		Select the Time Zone of the region/country where the Standard SIP Phone is installed. Default: + 05:30 India (Calcutta)	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 Extended IP phone is in the same network (LAN) as SARVAM UCS, select Use Ethernet IP Address as Registrar Server Address.*

If the Extended IP phone is in the Global Network and SARVAM UCS is connected to Internet over Mobile WAN, select Use Mobile Port IP Address as Registrar Server Address

If the Extended IP Phone is connected in the Global Network and SARVAM UCS is located behind a NAT Router or 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.

If you want SARVAM UCS to assign Registrar Server Address automatically, select Automatic 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 Phone

You are recommended to complete the following steps before connecting the **Snom** Standard SIP Phone:

- Decide the physical location of the Standard SIP Phone.
- Log into Jeeves as the System Engineer.
- Configure "DHCP Server" of SARVAM UCS in the "Network Parameters". The **DHCP Server** of SARVAM UCS will be used for assigning the 'Auto Provisioning Server Address' and 'Server Port' to the Standard SIP Phone.

  If you are using third party DHCP Server in your LAN, see "Using any third party DHCP Server in your LAN".
- Configure the **SIP Extension General Parameters** in SARVAM UCS. For details, see "Configuring SIP Extension General Parameters" in "SIP Extensions".[58]
- Configure the device specific settings applicable to your **Snom** at Location1 on the SIP Extensions page. To do so,
  - Click **SIP Extensions**.

58. *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.*

- Click **Device Settings - Location1** to expand 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
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 and click Submit to save.	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 Ethernet 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: 24Hr	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 Extended IP phone is in the same network (LAN) as SARVAM UCS, select Use Ethernet IP Address as Registrar Server Address.*

If the Extended IP phone is in the Global Network and SARVAM UCS is connected to Internet over Mobile WAN, select Use Mobile Port IP Address as Registrar Server Address

If the Extended IP Phone is connected in the Global Network and SARVAM UCS is located behind a NAT Router or 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.

If you want SARVAM UCS to assign Registrar Server Address automatically, select Automatic 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 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. 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 "SIP Extensions".
- Configure the device specific settings applicable to **Htek** at Location1on the SIP Extensions page. To do so,
  - Click **SIP Extension Settings**.
  - 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 Ethernet 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: 11780	Htek 802 IP Phone
Max RTP Port	To define a range of RTP ports, configure the maximum local RTP port. Default: 11800	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 VP110 Standard SIP Phone

You are recommended to complete the following steps before connecting the **Matrix SPARSH VP110** Standard SIP Phone:

- Decide the physical location of the Standard SIP Phone.
- Log into Jeeves as the System Engineer.
- Configure "DHCP Server" of SARVAM UCS in the "Network Parameters". The **DHCP Server** of SARVAM UCS will be used for assigning the 'Auto Provisioning Server Address' and 'Server Port' to the Standard SIP Phone.

  If you are using third party DHCP Server in your LAN, see "Using any third party DHCP Server in your LAN".
- Configure the **SIP Extension General Parameters** in SARVAM UCS. For details, see "Configuring SIP Extension General Parameters" in "SIP Extensions".[59]
- Configure the device specific settings applicable to your **Matrix SPARSH VP110** at Location1on the SIP Extensions page. To do so,
  - Click **SIP Extensions**.
  - Click **Device Settings - Location1** to expand 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

59. *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.	Matrix SPARSH VP110
Device Type		Select Device Type as MATRIX SPARSH VP110 and click **Submit** to save.	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
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 phone with the SIP Registrar of SARVAM UCS. Default: Use Ethernet 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 Phone Key Settings		Select the Send Phone Key Settings check box to apply the Phone Key Settings to the phone. Default: Enabled	Matrix SPARSH VP110
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. For instructions, see “Dial Plan for SIP Extension”.	Matrix SPARSH VP110

Transport Mode		Select the protocol to be used to transport the SIP messages. Default: TCP	Matrix SPARSH VP110
Enable SRTP?		Select the Enable SRTP? check box for secure communication 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 to send SIP messages to the remote peer. Default: 5060	Matrix SPARSH VP110
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 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[c]		Select the type of contacts (Extension or/and Global Directory Contacts) the phone must download from the SARVAM UCS. These contacts 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 contacts 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[d] of the Standard SIP Phone. It can be maximum of 16 characters long. Default: user	Matrix SPARSH VP110
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 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 in the area where the Standard SIP Phone is connected. 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 Extended IP phone is in the same network (LAN) as SARVAM UCS, select Use Ethernet IP Address as Registrar Server Address.*

If the Extended IP phone is in the Global Network and SARVAM UCS is connected to Internet over Mobile WAN, select Use Mobile Port IP Address as Registrar Server Address

If the Extended IP Phone is connected in the Global Network and SARVAM UCS is located behind a NAT Router, or 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.

If you want SARVAM UCS to assign Registrar Server Address automatically, select Automatic as Registrar Server Address.

b. *You can also configure rules for the Dial Plan from each phone. To do so, refer to the SPARSH VP110 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.*

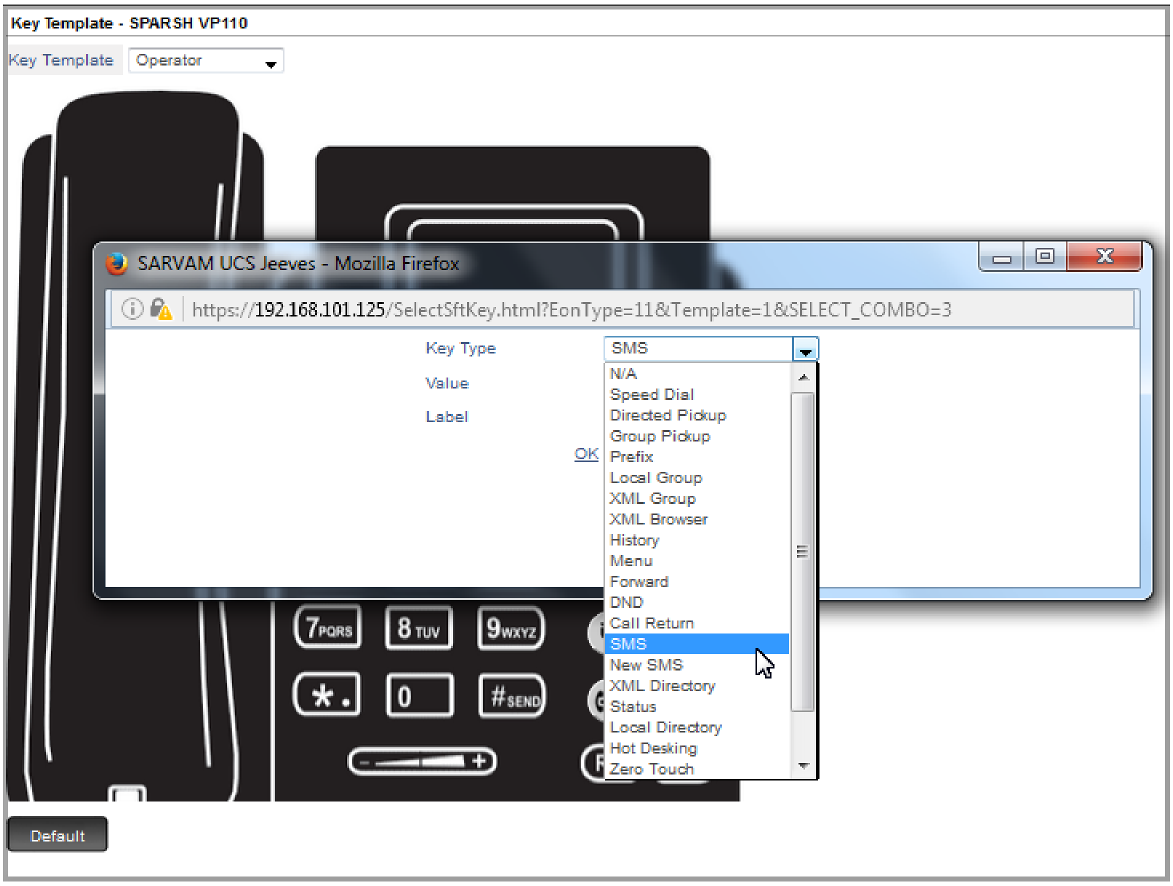
d. *To avoid unauthorized access, we recommend you to change the User Password regularly. Make sure it is strong and is kept confidential.*

e. *To avoid unauthorized access, we recommend you to change the Admin Password regularly. Make sure it is strong and is kept confidential.*

SPARSH VP110 Key Settings

- To personalize the key map of the SPARSH VP110, click SPARSH VP110 under **Key Template** in **Advanced Settings.**
- The key map of the SPARSH VP110 opens in a new window on your screen.

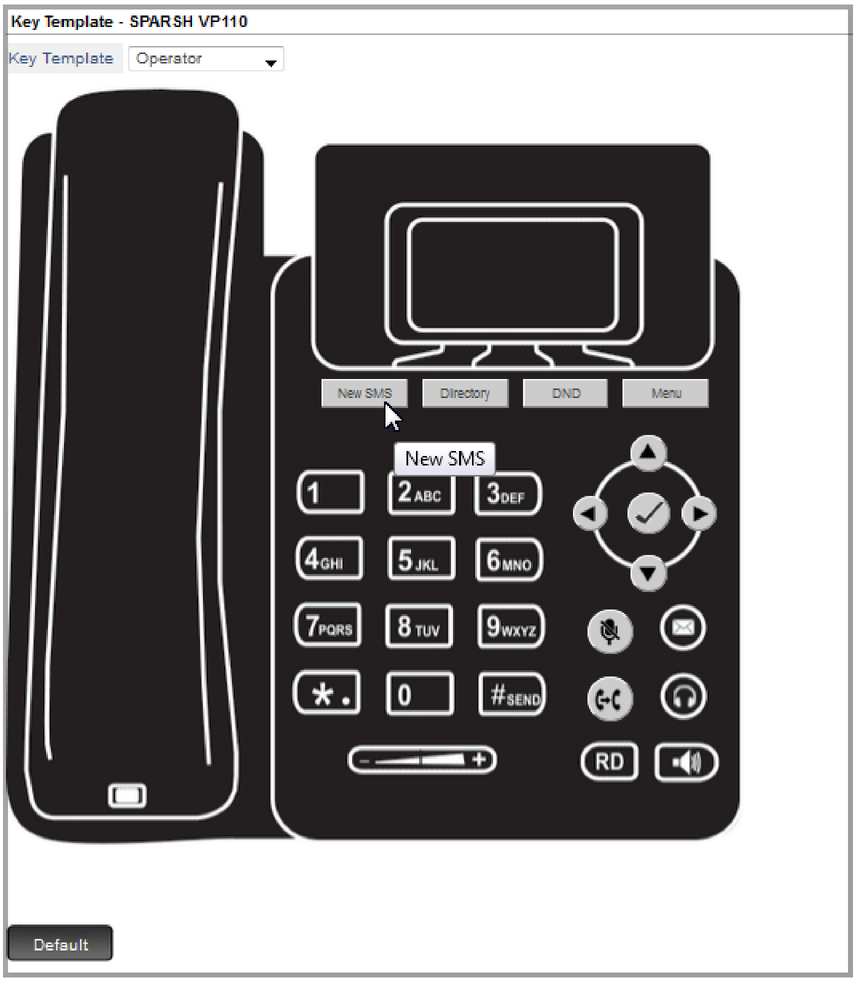
- For each **KeyType**, select the Type of function that is to be performed by the key. For example, to use the Softkey 1 to send a SMS, select Key Type as **SMS**.



You may also change the key Label, if required for Softkey 1 to 4.

- Click **OK**.

The *New SMS* feature appears on the key label.



Follow the same instructions to assign features to other Softkeys.

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 “SIP Extensions”.
- Configure the device specific settings applicable to your **Matrix SPARSH VP710- Standard SIP** at Location1 on the SIP Extensions page. To do so,
    - Click **SIP Extensions**.

- 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 Ethernet 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. For instructions, see "Dial Plan for SIP Extension".	Matrix SPARSH VP710
Transport Mode		Select the protocol to be used to transport the SIP messages. Default: TCP	Matrix SPARSH VP710
Enable SRTP?		Select the Enable SRTP? check box for secure conversations over SIP. Default: Disabled	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 Extended IP phone is in the same network (LAN) as SARVAM UCS, select Use Ethernet IP Address as Registrar Server Address.*

If the Extended IP phone is in the Global Network and SARVAM UCS is connected to Internet over Mobile WAN, select Use Mobile Port IP Address as Registrar Server Address

If the Extended IP Phone is connected in the Global Network and SARVAM UCS is located behind a NAT Router, or 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.

If you want SARVAM UCS to assign Registrar Server Address automatically, select Automatic 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.
- Log into Jeeves as the System Engineer.
- Configure "DHCP Server" of SARVAM UCS in the "Network Parameters". The **DHCP Server** of SARVAM UCS will be used for assigning the 'Auto Provisioning Server Address' and 'Server Port' to the Standard SIP Phone.

  If you are using third party DHCP Server in your LAN, see "Using any third party DHCP Server in your LAN".
- Configure the **SIP Extension General Parameters** in SARVAM UCS. For details, see "Configuring SIP Extension General Parameters" in "SIP Extensions".[60]

60. *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.*

- Configure the device specific settings applicable to your **Standard SIP Phone** at Location1on the SIP Extensions page. To do so,
    - Click **SIP Extensions**.
    - Click **Device Settings - Location1** to expand 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.*

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.

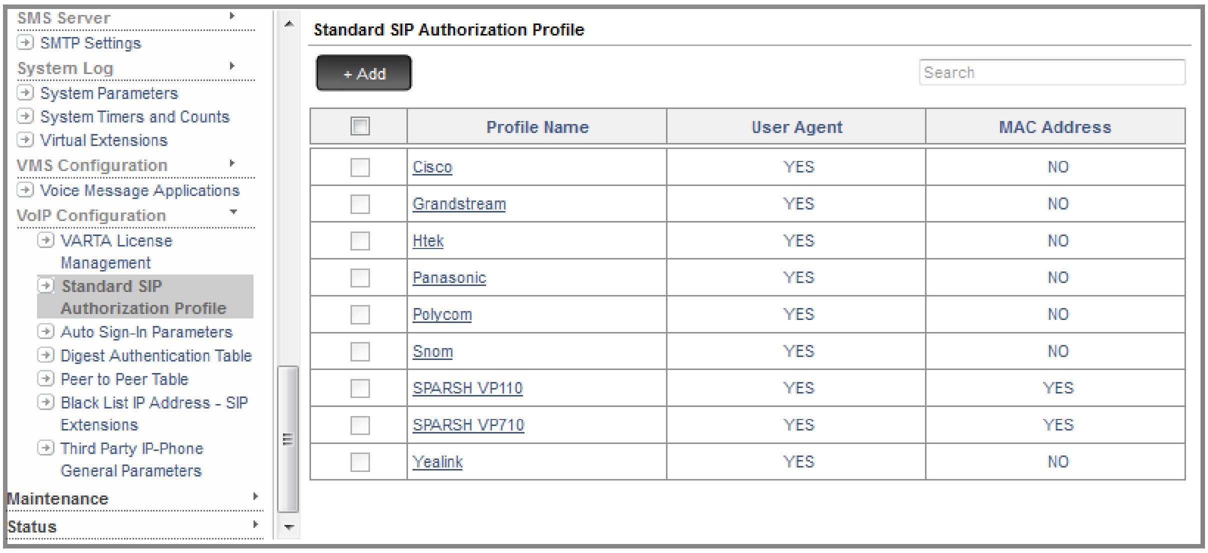
Using the Standard SIP Authorization Profile, you can register
- Matrix SPARSH VP110
- Matrix SPARSH VP710
- Third-party Standard SIP Phone
- Any Standard SIP Phone

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**.



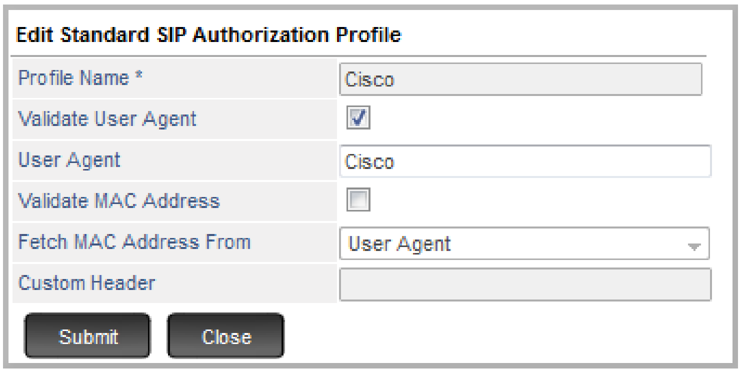
The list of default profiles supported by the system is displayed.

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 + Add and enter the details as per your requirement.
- Click **Submit**.

  The newly added Profile will get updated in the list.

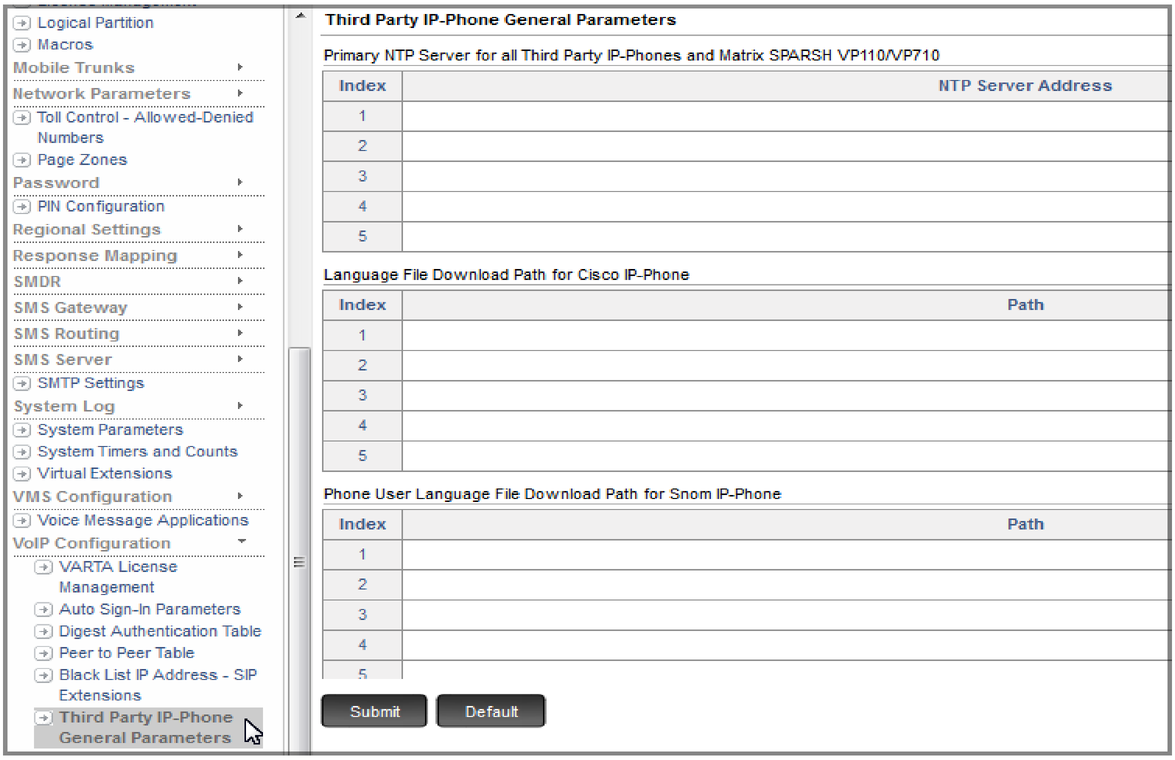
To delete a Profile,

- Select the corresponding check box and click .

Third Party IP-Phone General Parameters

To configure the Third Party IP-Phone General Parameters,

- Login as System Engineer.
- Under **Advanced Settings**, click **VoIP Configuration**.
- Click **Third Party IP-Phone General Parameters**.



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**.

The Server Addresses/Path you configure on this page will appear in the Combo box of the respective parameter on the SIP Extension page.

Black List IP Address - SIP Extensions

Black List IP Address allows you to restrict unauthorized users from accessing SARVAM UCS.

To use this feature, you must select the desired option in **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 the false credentials— Authentication ID or Authentication Password and the authentication attempt fails for 10 times consecutively, SARVAM UCS blacklist the IP Address and port used for registration.

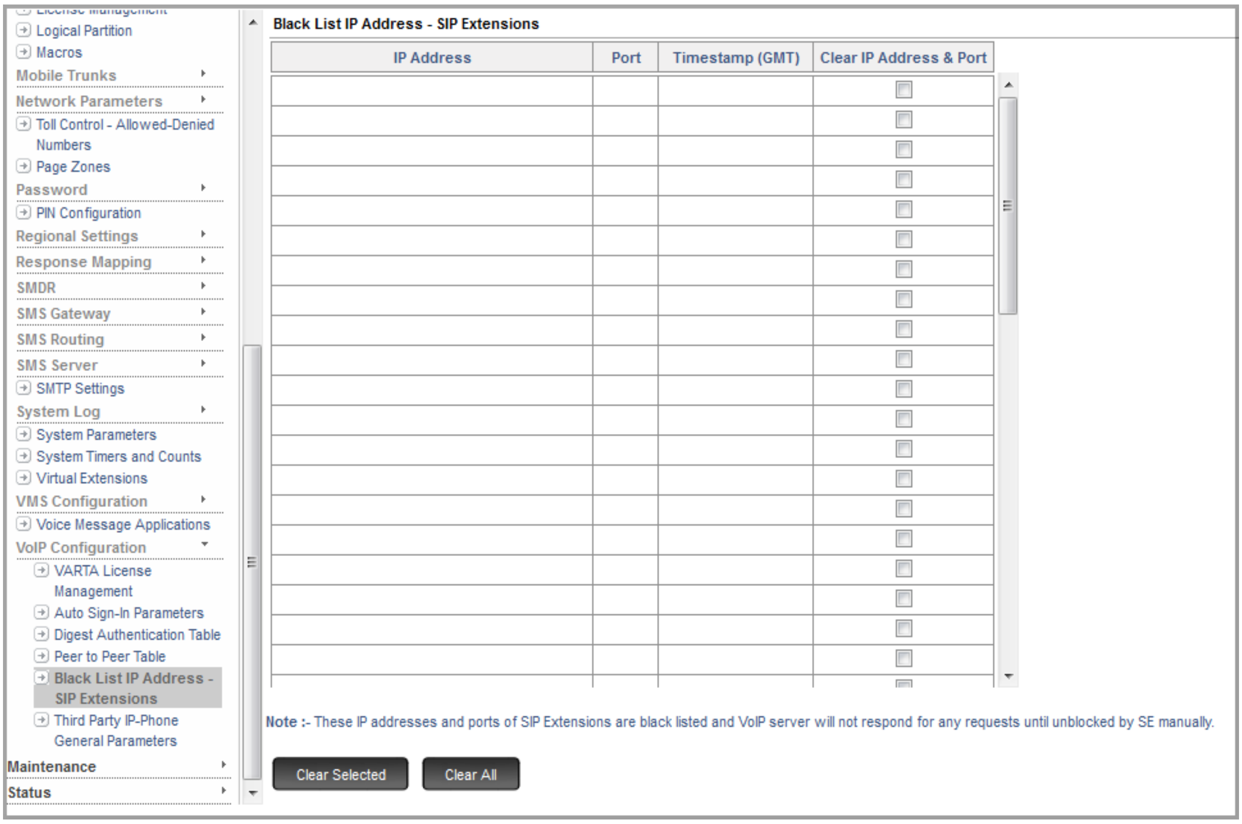
The blacklisted IP Address/es and ports are stored in the **Black List IP Address - SIP Extensions** table along with the date and time. This activity will also be logged in the “System Activity Log”.

If you want to allow access of SARVAM UCS to such blacklisted IP Address, you must remove it from the **Black List IP Address - SIP Extensions** table manually. If this IP Address is a Trusted IP Address, you can configure it in the Trusted IP Address/es table to avoid further blacklisting. See “Security Settings” for details.

Black List IP Address table stores upto 50 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,

- Login as System Engineer.
- Under **Advanced Settings**, click **VoIP Configuration**.
- Click **Black List IP Address - SIP Extensions**.



- 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 **Clear Selected**. The selected entries will be removed from the table.
- Click **Clear All** to clear all the entries stored in the table.

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 **Device Settings - 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,

- Click **Device Management**.
- The following information will be displayed for each connected phone:
  - **Device Name:** It displays the name of the phone connected in your LAN.
  - **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 the 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. Configure the following details to assign an extension to it:

    - **Select SIP Extension**: Select the SIP Extension Number you want to assign to the phone.
    - **Name**: Enter a name for the SIP Extension. You can enter the name of the person using the SIP Extension or his/her Department name. This name will appear as CLI on the Called Party's phone, when a call is made using this SIP Extension.

      The name may consist of a maximum of 18 alphanumeric characters.
    - **SIP ID:** Enter the SIP ID for the extension. It is the number which remote parties will use to call this 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 Authentication ID for the SIP Extension. You cannot keep this field blank. Authentication ID is the number which you want the VoIP module's Registrar Server to use for user authentication of the SIP messages received from this SIP Extension. The number may be a string of maximum 6 alphanumeric characters. All ASCII characters except < > and " (double quote) are allowed.

- **Authentication Password:** Enter the Authentication Password for the SIP Extension or click **Generate** to automatically generate a unique password. Authentication Password is the password to be used by the VoIP module's Registrar Server to authenticate the SIP messages received from this SIP Extension.

  Authentication Password is the password to be used by the VoIP module's Registrar Server to authenticate the SIP messages received from this 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 8 characters and can be a maximum of 12 characters.
  - 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. Default: Blank.

  To provide additional security, when the authentication fails for 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 Black List IP Address table manually. See "Black List IP Address - SIP Extensions" for more details. This activity will be logged in the "System Activity Log".

  You may also choose to configure simple Authentication Password, if required. However, it is recommended not to do so. For more details, see "Force Complex SIP/HTTP Authentication Password" under "Security Settings".

- **Select Location**: Always displays Location 1 as the Standard SIP Phone can be registered at Location 1 only for Auto Provisioning.

- **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 you configured will be displayed on the Device Management page.

- **Reboot:** Click this button to reboot the phone remotely.

The phone will reboot only if it supports remote reboot.

- Click **Clear All Assigned Entries**, to clear all the assignments.

- Click **Clear All Entries**, to clear all the entries on the page.

Call Pickup Group

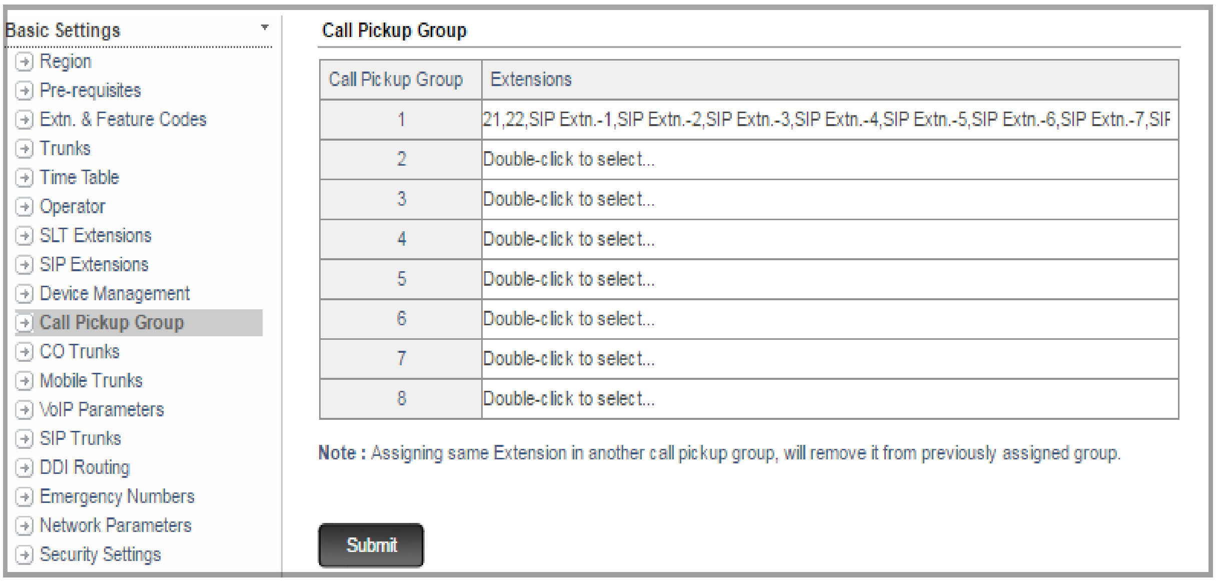
The Call Pickup feature of SARVAM UCS allows extension users to answer (internal and trunk) calls ringing on other extensions from their own extension; without physically going to the ringing extensions. For this extensions must be assigned to Call Pickup Groups.

See "Call Pickup" to know more about this feature.

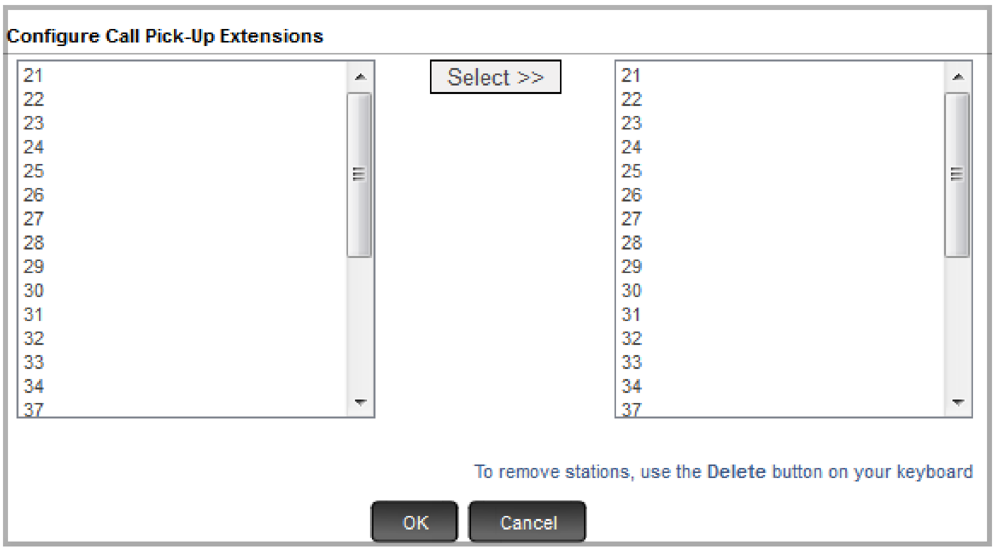
You can create as many as 8 Call Pickup Groups, and assign the extensions to these groups.

To create a Call Pickup Group,

- Click **Call Pickup Group**.



- For the desired Call Pickup Group Number, double click the **Extensions** field.



A multiple selection box opens. The extensions appear on the left, with names and numbers, as configured. If you have not configured names and numbers, the extensions will appear with their default access codes and extension numbers.

- Select the extensions which are to be assigned to the Call Pickup Group. These may be SLT or SIP extensions.

  Place your cursor on the desired extension and click **Select>>**.
  Or
  Press the **ctrl** key and click the left mouse button to select multiple extensions.

  You can also delete extensions from the ones you have selected using the Delete key of your keyboard.

- Click **OK**.

  All the extension numbers you selected will appear in **Extensions** for the Call Pickup Group number. The extension numbers appear in the sequence you selected, separated by commas.

You cannot select same extensions for another Call Pickup Group. Selecting same extension for another Call Pickup Group will delete the extension from the previous group and will allot it to the new group.

- To create another Call Pickup Group, double click the **Extensions** field against the desired group number (02 to 08) and select the desired extensions.

- Click **Submit** to save your Call Pickup Groups.

To know more about this feature, refer the topic "Call Pickup".

CO Trunks

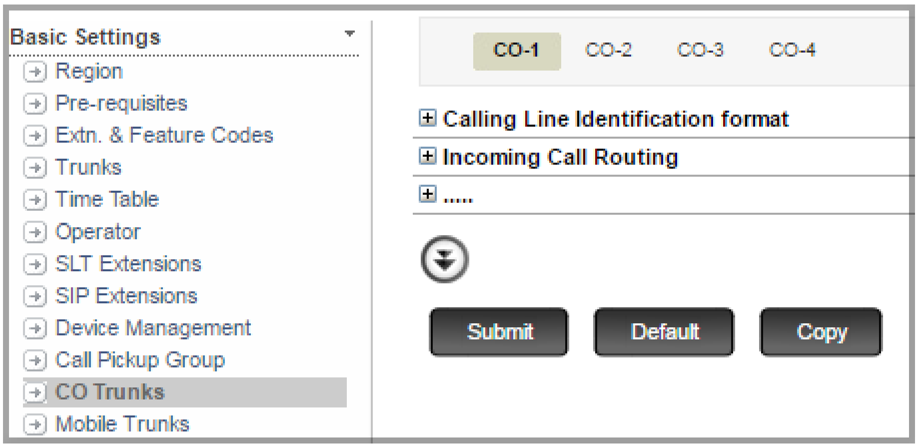
SARVAM UCS supports a maximum of 4 Analog Trunks[61]. The number of trunks available for configuration vary according to your configuration.

The number of CO Trunks you have specified on the "Pre-requisites" page will be displayed on this page.

To configure,

- Click **CO Trunks.**

  On this page,



More: Click this button to view all parameter links on the page.

Less: Click this button to view only the essential parameter links on the page.

Expand: Click to expand a link to display all parameters under the link.

Collapse: Click to collapse a link. Hides all parameters under the link.

Copy: Click to copy the parameters under a specific link of a page to the same link under any other trunk page.

Settings: Click to configure the settings of a parameter further.

···· **More link:** Click to view all additional parameters on the page.

To configure another CO Trunk, click the CO number (name) tab.

To save the settings, click **Submit**.

To assign default values to all the parameters of the CO Trunk, click **Default**.

To copy all CO Trunk parameter values to another CO Trunk, click **Copy**.

61. *ETERNITY NENX312 supports three Analog Trunk Ports.*

The CO Trunk numbers and the names (either default or the names you configured) appear on the tabs on this page, starting with CO-1 to CO-4.

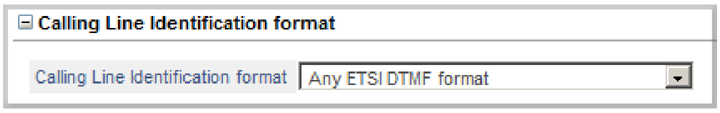
If you have assigned names to the CO Trunks on the "Trunks" page, the same names will appear on the tabs.

To change the name of the CO Trunk, you must go back to the "Trunks" page.

Follow the instructions provided below to configure the CO Trunks.

Calling Line Identification Format

- Click **Calling Line Identification format** to expand.

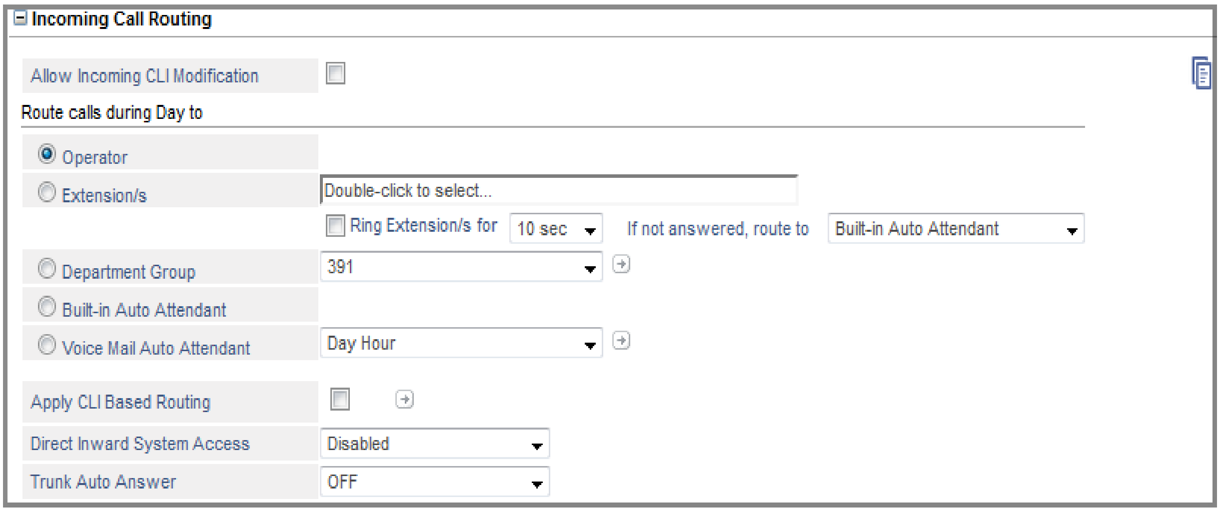


- Define the **Calling Line Identification Format** for the CO line. Select the format supported by your service provider from the below options. You may consult the service provider for this.
  - None
  - Any ETSI DTMF format
  - Any FSK V.23 format
  - Any FSK Bellcore format
  - 1st Ring, ETSI DTMF, 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)

  Default: Any ETSI DTMF format.

Incoming Call Routing

- Click **Incoming call Routing** to expand.



- 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".

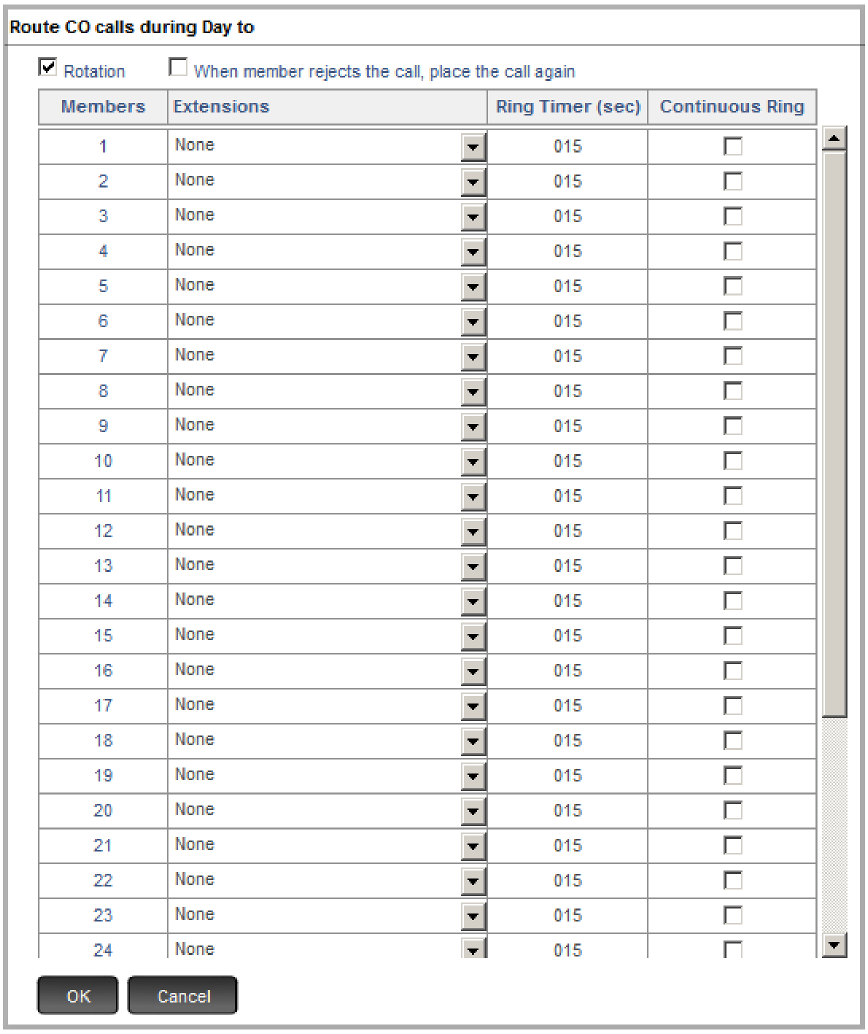
If you receive CLI in dialable format, there is no need to use this feature. In such case, keep the check box disabled.

*For an incoming call on the CO trunk, the Incoming CLI Modification is applicable only when both — the **Allow Incoming CLI Modification** check box and the **Enable Incoming CLI Modification** check box in System Parameters— are enabled.*

- Select the landing destination to **Route Calls during Day to**. You may select:
  - Operator
  - Extensions
  - Department Group
  - Built-in Auto Attendant
  - Voice Mail Auto Attendant (if available)

- If you select **Extensions** as the landing destination,

- Double click the **Extensions** field. A new window opens.



Members	Extensions	Ring Timer (sec)	Continuous Ring
1	None	015	
2	None	015	
3	None	015	
4	None	015	
5	None	015	
6	None	015	
7	None	015	
8	None	015	
9	None	015	
10	None	015	
11	None	015	
12	None	015	
13	None	015	
14	None	015	
15	None	015	
16	None	015	
17	None	015	
18	None	015	
19	None	015	
20	None	015	
21	None	015	
22	None	015	
23	None	015	
24	None	015	

- Select the extensions you want to assign as landing destination. The landing destination may be a SLT, SIP extension, Virtual extension or Voice Mail Auto Attendant Menu. If you select landing destination as Voice Mail Auto Attendant Menu, make sure you have configured the desired “Voice Mail Auto Attendant Menu”
- Set the **Ring Timer** for the extensions. This timer defines the time for which the extension, on which the call lands, should ring. Default: 015 seconds.
- Set **Continuous Ring**[62] for the extensions, if you want the extensions to ring till the incoming call is answered. Default: Disabled.
- Enable **Rotation**[63], if you have selected more than one extension. Default: Enabled.

If you select Voice Mail Auto Attendant Menu as landing destination, the parameters Ring Timer and Continuous Ring will not be applicable.

62. *When Continuous Ring is selected, the first extension in the group you have created will continue to ring, even as the system hunts for other extensions in the 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 has no relevance, if you select only one landing extension.*
63. *When Rotation is enabled, each new call lands on the extension next to the one that received the call last. This ensures equal distribution of incoming calls to all the destination extensions in this group.*

        - By default, **When member rejects the call, place the call again** is disabled. Therefore, if any SIP extension user rejects an incoming call, the system will not place the same call on this extension again while checking the routing group to land the call. You may enable this check box, if required.

          If this check box is disabled and you have selected the Continuous Ring 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.

        - Click **OK**. All the extensions you selected will appear in **Extension**, in the sequence you selected, separated by commas.

- You may route the incoming call to an auto attendant, if the selected extensions do not answer the call,
    - Select the **Ring Extension/s** check box and set the ring duration in **for___seconds**. Default: 10 seconds. This is the time for which the system will ring on the destination extensions you selected, and wait for an extension to answer. If the call remains unanswered on the expiry of this timer, SARVAM UCS will route the call to the auto attendant.
    - Select the destination for the unanswered call in **If not answered, route to**. You may select the **Built-In Auto Attendant** or the **Voice Mail Auto Attendant**. Default: Built-In Auto Attendant.
- If you select a **Department Group** as the landing destination,
    - Click the **Settings** icon. A new window opens.

Department Group

Department Group	Access Code	Name	Select Extension/s for Department Group	Voice Mail Settings
1	391		Double-click to select...	Voice Mail Settings
2	392		Double-click to select...	Voice Mail Settings
3	393		Double-click to select...	Voice Mail Settings
4	394		Double-click to select...	Voice Mail Settings
5	395		Double-click to select...	Voice Mail Settings

Submit Default Default One

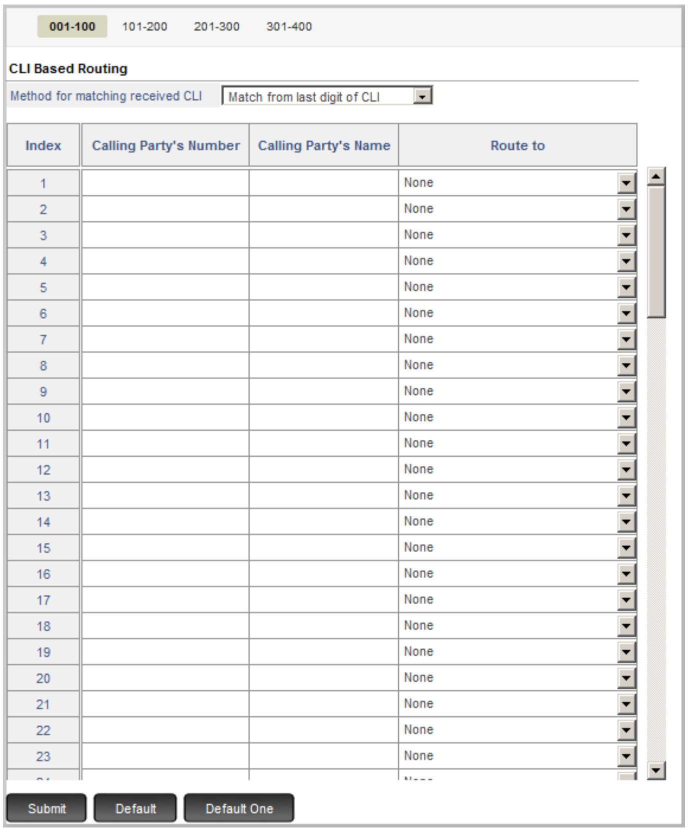
  To configure the Department Group parameters, see “Department Call”.

- If you select **Built-in Auto Attendant** as the landing destination for calls during the day,
    - the Voice Modules 02 to 13 will be played as “Built-In Auto Attendant Greeting Messages” and “Built-In Auto Attendant Guidance Messages”.
    - If you want to play a different message, make sure you record the desired messages in the Voice Modules after completing the installation with Basic Settings. Refer the “Voice Message Applications” topic to know more.

  See “Auto Attendant” to know more.

- If you select **Voice Mail Auto Attendant** as the landing destination for calls during the day, then configure the “Voice Mail Auto Attendant Menu”.
- To enable CLI Based Routing on the CO Trunk during day, select the **Apply CLI Based Routing** check box. Default: Disabled.

- Click the **Settings** icon. A new window opens.



To configure the CLI Based Routing Table parameters, see "CLI Based Routing".

- To enable "Direct Inward System Access (DISA)" on the CO Trunk, select the desired **Direct Inward System Access** option. Default: Disabled.
    - **PIN Authentication–Multiple calls:** Select this option if you want to enable DISA with PIN Authentication and allow multiple calls during the DISA login session.
    - **CLI Authentication–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.
    - **CLI Authentication–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.

If you select CLI Authentication Multiple Calls or CLI Authentication One Call as CLI Authentication, you must configure the DISA CLI Authentication Table under Advanced Settings. See "Direct Inward System Access (DISA)".

- To enable Trunk Auto Answer[64] feature on the trunk, select the type of **Trunk Auto Answer** you want:
    - **For all Calls:** the system will answer all incoming calls landing on the trunk line.
    - **When Busy:** the system will answer incoming calls on the trunk, if the landing destination is busy.

---

64. *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.*

- **Delayed:** the system will first land the call on the extension and if not answered before the expiry of the specified Delayed Trunk Auto Answer timer, it will be auto answered.

  On selecting this type of Trunk Auto Answer, you must configure **Delayed Trunk Auto Answer Timer**. You can enter the timer value within the range of 01 to 99 seconds. Default: 10 seconds.

- **OFF**: To disable Trunk Auto Answer, select **OFF**. Default: OFF.

- Select the Trunk Auto Answer **Greeting Message** you want the system to play when greeting the callers, from the following options:
  - Disabled
  - Greeting Message1
  - Greeting Message2
  - Greeting Message3
  - Greeting Message4

  Default: Disabled.

  If you do not want greeting message to be played, select **Disabled**.

When you select a Greeting Message, you must record a Voice Module with the desired Greeting Message and assign the Voice Module to Trunk Auto Answer Greeting application.

You can record 4 different Greetings Messages for Trunk Auto Answer and assign a different Greeting message for the Day, Break and Night. See the topic “Voice Message Applications” for instructions on recording and assigning voice modules to greeting messages.

- Select the Auto Answer **Ring Back Tone Message** you want the system to play to callers from the following options:
  - Do Not Play
  - Play Music on Hold
  - Play RBT Message1
  - Play RBT Message2
  - Play RBT Message3
  - Play RBT Message4

  If you do not want RBT message to be played, select **Do Not Play**. The system will play Ring Back Tone to the caller.

  Default: Do Not Play.

When you select an RBT Message (1–4), you must record a Voice Module with the desired RBT Message and assign the Voice Module to Trunk Auto Answer RBT Message application

You can record 4 different RBT Messages for Trunk Auto Answer and assign a different RBT message for the Day, Break and Night. See “Voice Message Applications” for instructions on recording and assigning voice modules for RBT messages.

- Select the Trunk Auto Answer **Busy Bye Message** you want the system to play to callers when the landing destination extension is busy, from the following options:
  - None
  - Bye Message1
  - Bye Message2
  - Bye Message3
  - Bye Message4

  If you do not want Busy Bye message to be played, select **None**. The system will not play Busy Tone to the caller. Default: None.

When you select a Bye Message (1–4), you must record a Voice Module with the desired Bye Message and assign the Voice Module to Trunk Auto Answer Bye Message application

You can record 4 different Bye Messages for Trunk Auto Answer and assign a different Bye message for the Day, Break and Night. See "Voice Message Applications" for instructions on recording and assigning voice modules for Bye messages.

- Follow the same instructions as described above to configure **Incoming Call Routing** to **Route Calls during Break to** and **Route Calls during Night to**.

Automatic Number Translation (ANT)

Automatic Number Translation (ANT)

Index	Dialed Number String	Strip Digits	Add Prefix
1		00	
2		00	
3		00	
4		00	
5		00	
6		00	
7		00	
8		00	
9		00	
10		00	
11		00	
12		00	
13		00	
14		00	
15		00	

The feature, Automatic Number Translation of SARVAM UCS modifies dialed numbers or part thereof to match the specific route numbering plan understood by the destination network—CO, GSM, VoIP—by adding or stripping country and area codes.

For example, you can configure Automatic Number Translation such that when an extension user dials a local landline number, the SARVAM UCS prefixes the number with the appropriate country-area code when it routes the call through the GSM network.

To apply Automatic Number Translation (ANT) on the CO Trunk,

- Click **Automatic Number Translation** to expand.

- The Automatic Number Translation table appears.

  The ANT table has three columns: Dialed Number String, Strip Digits, Add Prefix. Each number string is stored against an Index number, from 1 to 32. You can enter as many as 32 entries in the ANT table.

  - In the **Dialed Number String** column, 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 String before 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 a prefix to the Dialed Number String before the system dials out this number. The Prefix can be a maximum of 40 characters. Default: Blank.

- Click **Submit** to save your entries.

For more details, see the description for "Automatic Number Translation".

SMDR Storage

The Station Message Detail Recording (SMDR) feature of SARVAM UCS enables you to record the details of Internal, Incoming (IC) and Outgoing (OG) calls made from/to all its trunks and extensions. To obtain SMDR as a report, you must enable SMDR Storage, and set filters. See "Station Message Detail Recording (SMDR)" and "Station Message Detail Recording–Storage" to know more.

- Click **SMDR Storage** to expand.
- To store details of incoming calls landing on this trunk, select the check box **Store Incoming Calls**. Default: Enabled.
- To store details of outgoing calls made from this trunk, select the check box **Store Outgoing Calls**. Default: Enabled.
- Click **Submit**.

DSS Key Interface

This parameter determines what the system should do when an external called party in speech on this trunk is put on Hold by an extension user (Extended IP Phone user) by pressing a DSS key to dial another port.

For example, the SIP extension user on SIP1 port is in the middle of speech with an external party on CO2. If the extension user of SIP1 presses a DSS key to call another extension port, SIP2, two situations are possible, depending on the DSS Key Interface you configure:

i. The external caller on CO2 will be played music-on-hold, and the user on SIP1 will hear Ring Back Tone. The call will be placed on SIP2.

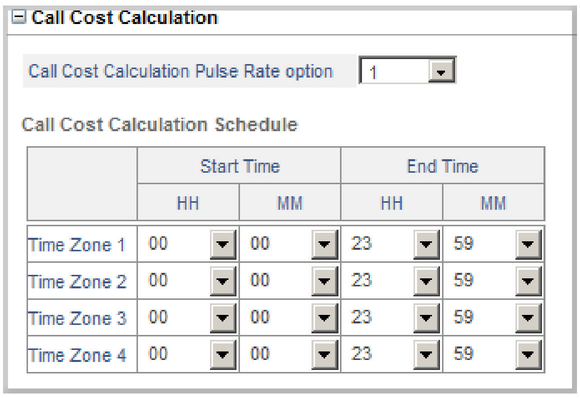
ii. The external caller on CO2 will be disconnected. SIP1 user will hear Ring Back Tone. The call will be placed on SIP2.

- Click **DSS Key Interface** to expand.
- Select the radio button of the desired DSS Key Interface option:
  - **Disconnect current call when Extended IP Phone user presses DSS key of another Trunk, while in speech on this Trunk.**
  - **Put current call on Hold, when Extended IP Phone user presses DSS key of another Trunk, while in speech on this Trunk** (Default).
- Click **Submit**.

Call Cost Calculation

If you want to Call Cost Calculation feature on this trunk,

- Click **Call Cost Calculation** to expand.



- Select a **Call Cost Calculation Pulse Rate Option**, from 1 to 4.

  Each Call Cost Calculation Pulse Rate option contains a **Pulse Rate Type**, which you must configure. Some service providers offer discounted rates for special days. In which case, you can configure the rates for normal days in the **Normal Pulse Rate** Table and the rates for special days in the **Discounted Pulse Rate** Table. See “Call Cost Calculation (CCC)”.

  Configure the Pulse Rate Type according to the billing pattern of the service provider of this trunk.

  Trunks following the same billing pattern should be assigned the same Pulse Rate Option.

- Define the **Call Cost Calculation Schedule** for the Pulse Rate option you selected.

The Pulse Rates offered by service providers may vary according to the time of the day. In such cases, you must divide the day into Time Zones, from 1 to 4, as required, to match the time of the pulse rates offered by your service provider.

Specify the **Start Time** and the **End time** (in 24 hours, HH:MM format) for the **Time Zone** Index in which the particular Pulse Rate will be applied.

The default Time Schedule (start and end time) for each Time Zone Index is as follows:

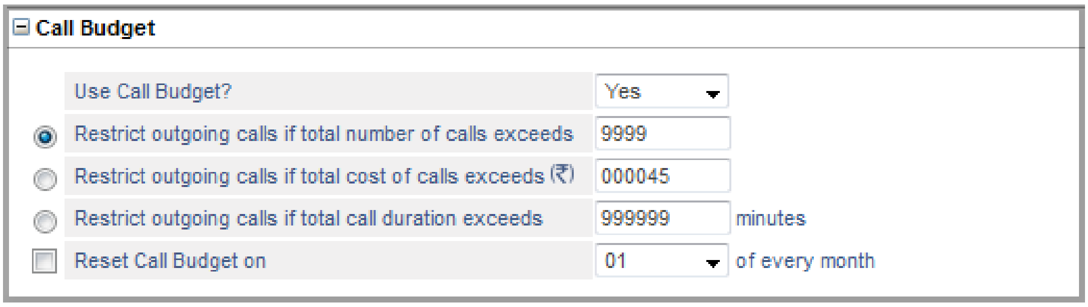
Time Zone Index	Start Time	End Time
T1	00:00	23:59
T2	00:00	23:59
T3	00:00	23:59
T4	00:00	23:59

If your service provider offers the same Pulse Rate for calling a number for the entire day, there is no need to change the default Time Schedule. The system will follow Time Zone 1.

See "Call Cost Calculation (CCC)"

Call Budget

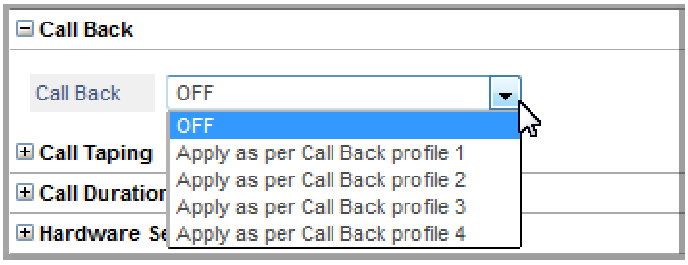
Configure this parameter if you want to control the cost of phone calls made from this trunk. See "Call Budget on Trunk" to know more.



- To disable call budget on the trunk, set **Use Call Budget?** to No. Default: Yes. By default the Call Budget type selected is **Restrict outgoing calls if total call duration exceeds**.
- Define the call budget in terms of number of calls in the **Restrict outgoing calls if total number of calls exceeds**. Default: 9999.
- Define the call budget in terms of amount in the **Restrict outgoing calls if total cost of the calls exceeds.** Default: 999999.
- Define the call budget in terms of duration in the **Restrict outgoing calls if total call duration exceeds**. Default: 300 minutes. By default this option is selected as the type of Call Budget.
- To have the system reset the Number of Calls/Amount/Minutes automatically on a particular date, select the desired date from the **Reset Call Budget on** combo box as 1st to 31st of the every month.

The consumed Call Budget Amount/Minutes/Number of Calls can be reset from SE and SA Mode, referred to as Manual Reset. Refer “Call Budget on Trunk” for more details.

Call Back



Configure this parameter if you want to enable the ‘Call Back on Trunk Port’ feature on the trunk. See “Call Back on Trunk Ports” to know more.

- Select a **Call Back Profile**, 1 to 4, for the trunk port from the combo box.
  Make sure you also configure the Call Back profile you select here.

  To configure the profile, click the **Settings** icon. This icon does not appear when the Call Back is **OFF**.
  - A new window opens.
  - Configure the Profile Number. See “Call Back on Trunk Ports” for instructions.
  - Click **Submit** to save your profile.
  - Close the window.

Call Taping

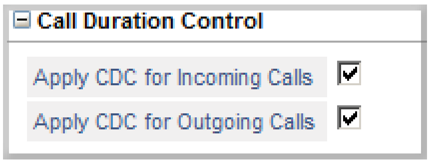
To use “Call Taping” feature on the trunk,

- Click **Call Taping** to expand.
- Select the check box to enable Call Taping on the trunk. Default: Enabled.

  To disable Call Taping on the trunk, clear the check box.
- Click **Submit**.

See “Call Taping” to know more.

Call Duration Control (CDC)



To use the "Call Duration Control (CDC)" feature on trunk,

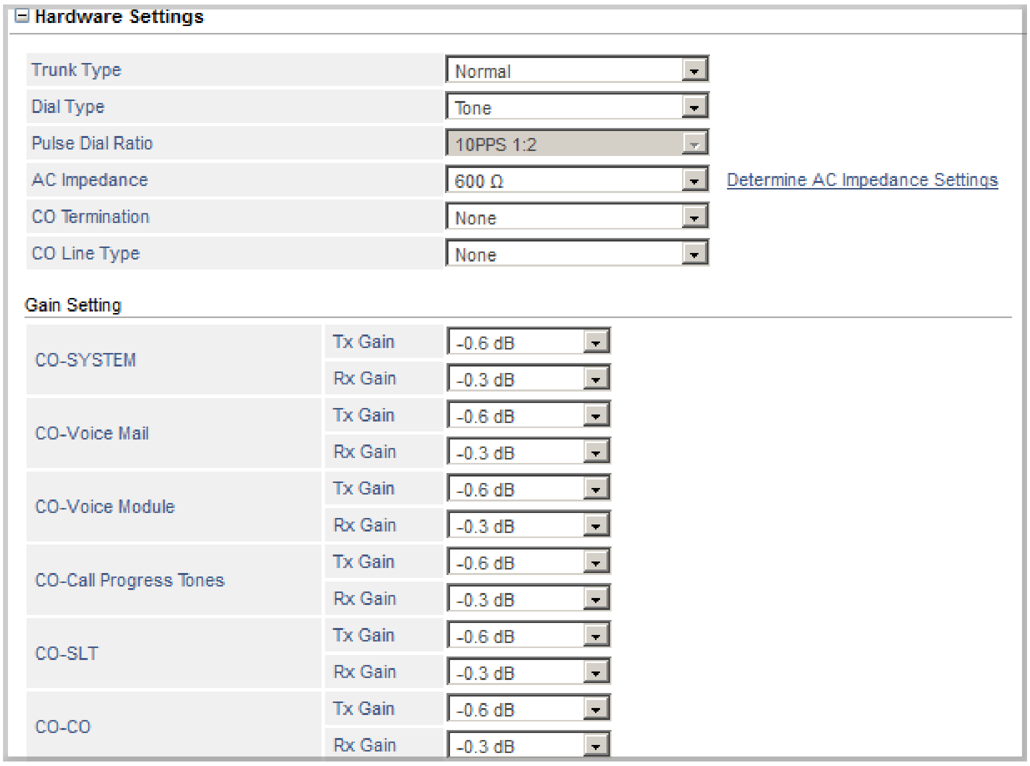
- Click **Call Duration Control** to expand options.
- Select the **Apply CDC for Incoming Calls** check box, to control the call duration for incoming calls on this trunk. Default: Enabled.

  Clear this check box, if you do not want to apply CDC for incoming calls on this trunk.
- Select the **Apply CDC for Outgoing Calls** check box, to control the call duration for outgoing calls on this trunk. Default: Enabled.

  Clear this check box, if you do not want to apply CDC for outgoing calls on this trunk.
- Click **Submit**.

See "Call Duration Control (CDC)" to know more.

Hardware Settings



- Click **Hardware Settings** to expand.
- Select your **Trunk Type**. Three types of CO trunks can be interfaced to a CO port of the SARVAM UCS:

- **Normal:** This is the conventional CO available from the PSTN.

- **Hotline:** The CO connecting two destinations immediately on grabbing the trunk.

- **Delayed Hotline:** A special CO available from the PSTN, which works as a normal dial type for some time and works as a hotline thereafter.

Default: Normal.

- Select **Dial Type** as **Tone** or **Pulse**. Default: Tone.

  You can configure 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.

- If you selected Pulse as Dial Type, select the appropriate **Pulse Dial Ratio** from the following options according to the type of Pulse Dialing Ration supported by your CO Network.
  - 10PPS, 1:2
  - 10PPS, 2:3
  - 10PPS, 1:1
  - 20PPS, 1:2
  - 20PPS,2:3
  - 20PPS, 1:1

  Default: 10 PPS 1:2.

- Select **AC Impedance**. The AC Impedance of the CO port must match with the AC Impedance supported by the PSTN network. By default, the AC Impedance is set as per the “Region” you have selected; however, you may select the impedance supported by your PSTN network from the following options.
  - 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

  Default: 600 ohms (for default Region ‘India’).

  You can also conduct the AC Impedance Test. To do this, click **Determine AC Impedance Settings** link. For more details, see “AC Impedance Test” topic.

- **CO Termination** 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 600Ω : 900Ω + 2.16µF. Default: None. 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 each CO Trunk port. 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 *"AC Impedance Test" topic.*

- **CO Line Type** allows you to select the Line model for the AC Impedance-CO Termination you 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). Default: None.

- 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. Default: -0.6dB for Tx Gain and -0.3dB for Rx Gain.

  - **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. Default: -0.6dB for Tx Gain and -0.3dB for Rx Gain.

  - **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. Default: -0.6dB for Tx Gain and -0.3dB for Rx Gain.

  - **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. Default: -0.6dB for Tx Gain and -0.3dB for Rx Gain.

  - **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. Default: -0.6dB for Tx Gain and -0.3dB for Rx Gain.

  - **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. Default: -0.6dB for Tx Gain and -0.3dB for Rx Gain.

  - **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. Default: -0.6dB for Tx Gain and -0.3dB for Rx Gain.

- **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. Default: -0.6dB for Tx Gain and -0.3dB for Rx Gain.

- Click **Submit**.

- Select the type of **Answer Supervision**[65] supported by your Network from the following options. Default: Pseudo Answer.

  - **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, i.e. 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.

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 selected Polarity Reversal or 12/16KHz as Answer Supervision Type, the call will be considered as matured and will not be stored in the Station Message Detail Record (SMDR) buffer. The same would be the case if you selected 12/16KHz when the CO supports Polarity Reversal.

- If you selected Pseudo Answer as Answer Supervision signal, configure the **Pseudo Answer Supervision Timer**.

  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. Default: 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 15 seconds, the call will be considered as matured after 15 seconds, even if it is answered after 20 seconds. Similarly, if this Timer is set to 80 seconds, but the call was answered after 20 seconds and disconnected after 40 seconds, this call will never be considered as matured as it ends before 80 seconds.

- Select the **Disconnect Supervision**[66] signal type supported by your network from the following options. Default: None.

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65. *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.

- **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 duration set in 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 you have set, 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 duration of 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.

- *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 selected '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 selected as the Disconnect Supervision for the port, the system will interpret this (Answer Signaling) signal as Disconnect Supervision signal and disconnect the call.*

- If you have selected Open Look Disconnect as Disconnect Supervision type, configure the **Open Loop Disconnect Timer**. The range of this timer is from 17 to 986 milliseconds. Default: 204 ms.

- If Call Disconnection is signaled by your CO Network in the form of Disconnect Tone, configure **Disconnect Tone Detection**.

  To enable the system to detect the Disconnect Tone accurately, select the **Enable Disconnect Tone Detection** check box and configure the Cadence (ON-OFF time) and Frequency of the Disconnect Tone, as supported by the CO network.

  - **Frequency 1 (Hz):** Frequency 1 is from 300 to 1400 Hz.

---

66. *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.

    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.

- **Operator:** This parameter has 3 options: No operator, Modulation (*), Addition (+)
  - If **No** operator is selected, frequency 2 will not be applicable.
  - If **Modulation (*)** is selected, frequency 1 and frequency 2 will be used as modulation i.e. F1* F2
  - If **Addition (+)** is selected, frequency 1 and frequency 2 will be used as addition i.e. 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 consists of Dual Frequency.

- **First Cycle Cadence - ON Time 1 (msec), OFF Time 1 (msec):** Configure the Cadence from 0000-9999 msec for ON Time 1 and OFF Time 1. Default: 0750.
- **Subsequent Cycles: ON Time 2 (msec), OFF Time 2 (msec):** Configure the Cadence from 0000-9999 msec for ON Time 2 and OFF Time 2. Default: 0750.
- Similarly, configure the Cadence from 0000-9999 msec for **ON/OFF Time 3** and **ON/OFF Time 4**. Default: 0000.

  When disconnect tone is detected on port, matches the Frequency and Cadences you have configured, the call will be disconnected and the CO port will be released.
- Configure DTMF Out Dial parameters. While dialing out the DTMF digits from the CO port, the following attributes of DTMF signal are critical.
  - **DTMF ON Time (msec):** It is the width of DTMF digit to be dialed out by the CO port. Set the ON time to the desired value. Default: 102 milliseconds.
  - **DTMF OFF Time (msec):** When the CO port dials out the DTMF digits on the CO, it waits for the DTMF OFF Time, while dialing the DTMF digits on the CO. Set the pause timer to the desired value. Default: 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.
- Configure the following **DTMF Detection** parameters, if necessary. 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 is the minimum level (dB) of the DTMF digit to be considered as valid. Default: -4.5dB.
  - **Minimum ON Time (msec):** This is 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: 34 milliseconds.
  - **Minimum OFF Time (msec):** This is the minimum time period between successive DTMF digits. The valid range of this time is 17 to 204 milliseconds. Default: 68.
- Set the following **Timers**, if necessary.
  - **Speech Delay Timer:** It is the time after which the system gives dial tone to the extension, when the extension user grabs the CO Trunk.

To understand the significance of this timer, let us consider a situation. Extension 201 does not have calling permission for long distance numbers. The user of extension 201 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 can be prevented by configuring the Speech Delay Timer to an appropriate value.

The range of this timer is from 000 to 255 seconds. Default: 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 (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 an incoming call on the CO port, if the CO gives ring with a long Ring OFF period, the system will consider that the ring has been stopped, and it will stop ringing the SLT port, even though the incoming call is still present.

  To get correct indication, the SARVAM UCS supports the Ring Cadence OFF timer on the 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 6 seconds. Default: 6 seconds.

- **Flash Time (msec):** This parameter is relevant for dialing out Flash on the CO Trunk to access some of the features of the CO Network. Configure the desired time of Flash to be generated on the CO Trunk. Default: 600 milliseconds.

- **On-hook Speed (msec):** This parameter allows you to set the duration of time for the line-side device to go on-hook.

  The ON-Hook speed you set here is measured from the time the ON-Hook bit is cleared until the loop current equals zero. Select the desired ON-Hook Speed from the following options:
  - <0.5msec
  - 3 msec
  - 26 msec

  Default: <0.5msec.

- **Off-hook Speed (msec):** 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

  Default: 8 ms.

- **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. Default: 002 seconds.

- To limit Loop Current, by default **Current Limiting** is enabled. When this check box is enabled, the Loop Current is limited to a maximum of 60mA.
- Set the **Minimum Loop Current** at which DAA module of the CO Trunk port can operate. Select the minimum operational loop current from the following options as per your requirement:
  - 10
  - 12
  - 14
  - 16

  Default: 10 mA.
- If required, you may adjust the **Tip Ring Voltage (Volts)** on the line side.

  In countries where Low voltage is required, you should set lower TIP/RING voltage in SARVAM UCS. 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

  Default: 3.5.
- Set the **Ringer Impedance** as **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.

  Default: 'High' (20Mohm).
- Set the **Ringer Threshold (Vrms)** level. This parameter defines the level below which the CO port would not validate the Ring signal, and the level above which it would validate the Ring signal. Set the Ringer Threshold to the desired value from the following options:
  - 13.5 - 16.5
  - 19.35 - 23.65
  - 40.5 - 49.5

  Default: 13.5 - 16.5 Vrms.
- If the CO trunk port you are configuring is located in a “Behind the System Application”, configure **Pre PSTN Digit Count.** 'Pre-PSTN Digit Count' or 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 PBX Applications, another PBX may be connected to the SARVAM UCS, with some of its CO Trunks terminating into the Extension ports of the other PBX, and other trunks being directly connected to the PSTN.

  If the CO Trunk port is directly connected to the PSTN, PPDC must be set to '0'.

For Trunk ports connected to extensions of another PBX, PPDC must be configured as per the number of digits in the Trunk Access Code defined for that PBX.

If the Trunk Access Code is a single digit, select '1'. If the Trunk Access Code is double or triple digit number, accordingly select '2' or '3' as PPDC.

To know more about this feature, refer “Behind the System Application”.

- To improve the quality of Fax over IP[67], configure **Pass Through Fax–Data Gain on SIP Trunk**, if you selected Pass Through Fax as Type of Fax over IP on SIP trunks, and if Pass Through Fax is to be received on a CO trunk. Select the dB Level for Data Gain. Default: -11 dB.

- Select the dB Level for **Pass Through Fax–Bypass Gain on SIP Trunk**. Default: -9 dB.

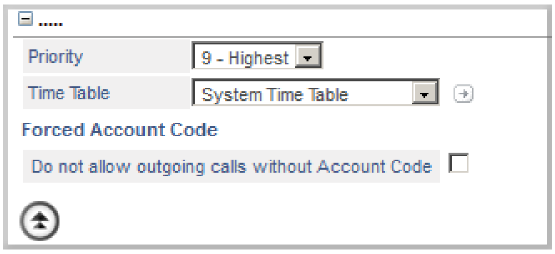
- *You can also set the Rx and Tx Gains for SIP to Digital Trunks and Extensions and SIP to SLT.*
- *To increase Rx and Tx Gain for SIP to SLT extensions, go to SLT parameters page. To increase Rx and Tx Gain for Digital Trunks/Extensions, go to “SIP Trunks” parameters page.*

- Click **Submit**

.

More Features

- Click **More...** link to expand.



Priority

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, see “Priority”.

- Select a **Priority** level from 1 to 9 for the trunk. Default: 9-Highest.

Time Table

- Select a **Time Table** for the CO trunk. Default: System Time Table.

  If you have not configured Time Table, you may do so now, by clicking the Time Table link. Define the Working days, and the start and end time of the Working hours and Break hours for each working day.

  See “Time Tables” to know more.

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67. *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.*

Forced Account Code

Configure this parameter if you want to apply the Forced Account Code feature on the trunk. See “Account Codes” to know more about this feature.

- Select the check box **Do not allow Outgoing Calls without Account Code** to enable Forced Account code on the trunk. Default: Disabled.

  When you enable this check box, the system will prompt extension users to dial the Account Code whenever they grab this trunk to dial out a number. The system will allow extension users to dial out numbers only after they have dialed the Account Code or Name.

Account Codes feature must also be enabled in the Class of Service of extension users who are to be allowed this feature.

Copy Parameter Values

After you have finished configuring this CO Trunk, you may configure the next CO Trunk. To do so,

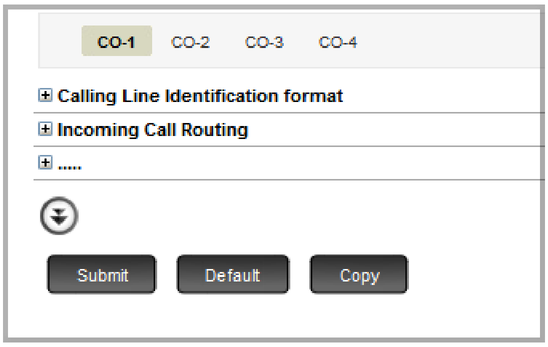
- Click the tab of the next CO Trunk.

- Follow the same steps as described above to configure another CO Trunk.

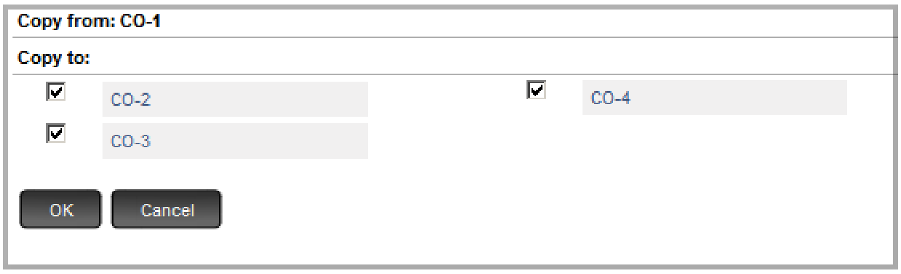
OR

You may use the **Copy** button at the bottom of the page to apply the same CO Trunk settings you configured for the current trunk to another CO Trunk. To do so,

- Click **Copy**.



The Copy page opens in another tab.

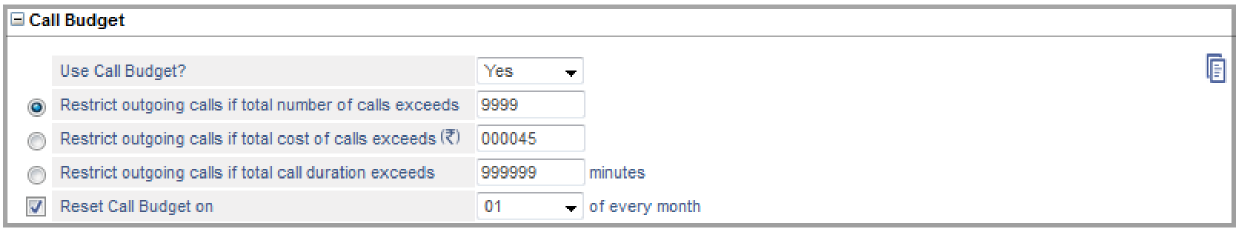


The copy page displays the number and/or name of the current trunk in **Copy from:**. It also lists the numbers and/or names of the trunks to which you can copy the trunk settings under **Copy To:**.

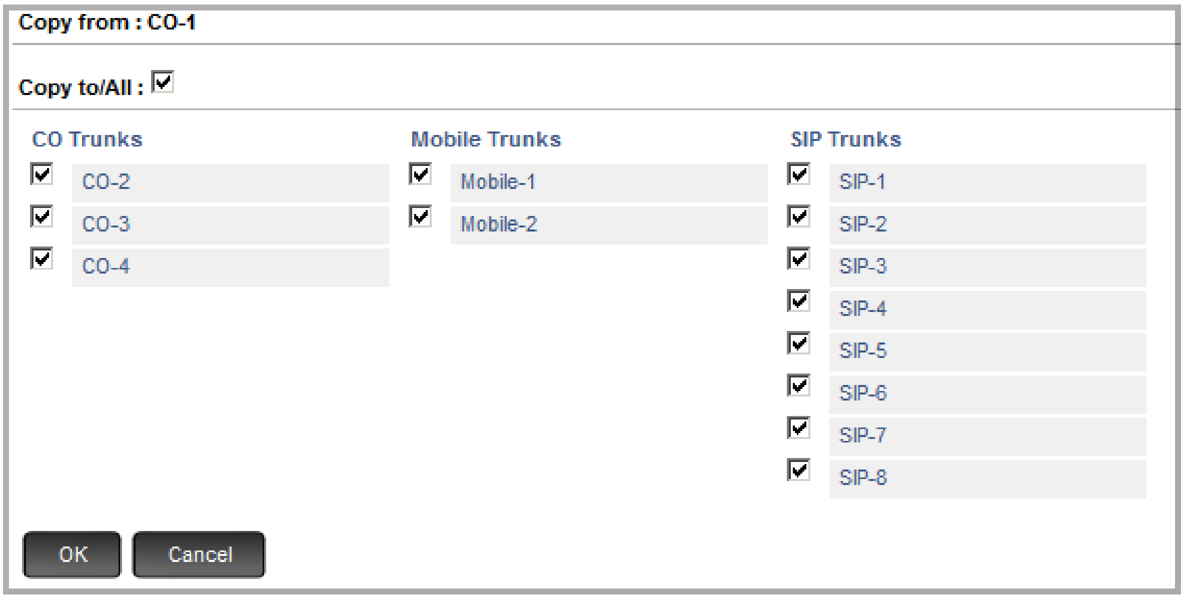
- Select the respective **Copy to** check boxes of the CO Trunks to which you want to apply the current CO Trunk settings.
- Click **OK**.

You may also copy the parameter values of a specific link under current CO Trunk to the same link under any other trunk page. To do so,

- Click the **Copy** icon besides the desired link.



The Copy page opens in another tab.



The Copy page displays the number and/or name of the current trunk in **Copy from:**. It also lists the numbers and/or names of the trunks to which you can copy the trunk settings under **Copy To/All:**.

- Select the **Copy to/All** check box, to apply the link settings to the same link under all other Trunks.
  OR
  You may also apply the link settings to the desired Trunks only, by selecting the respective check boxes.
- Click **OK**.

Mobile Trunks

SARVAM UCS supports 2 Mobile Trunks.

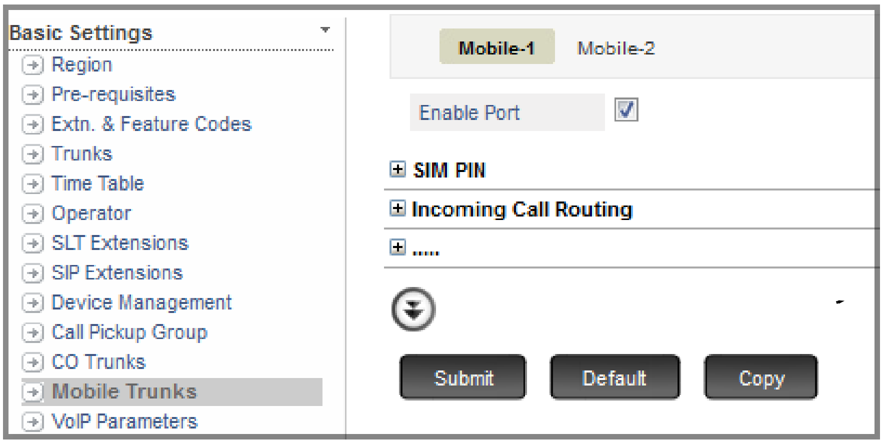
The number of Mobile Ports detected by the system on the "Pre-requisites" page will be displayed on this page.

For compatibility and use of Matrix GSM products (2G/3G/4G) in Russia and Iran Province connect with Matrix Sales or Technical Support Team.

To configure,

- Click **Mobile Trunks.**

  On this page,



More: Click this button to view all parameter links on the page.

Less: Click this button to view only the essential parameter links on the page.

Expand: Click to expand a link to display all parameters under the link.

Collapse: Click to collapse a link. Hides all parameters under the link.

Copy: Click to copy the parameters under a specific link of a page to the same link under any other trunk page.

Settings: Click to configure the settings of a parameter further.

···· **More link:** Click to view all additional parameters on the page.

To configure another Mobile Trunk, click the respective Mobile Trunk number (name) tab.

To save the settings, click **Submit.**

To assign default values to all the parameters of the Mobile Trunk, click **Default.**

To copy all Mobile Trunk parameter values to another Mobile Trunk, click **Copy.**

If you have assigned names to the Mobile Trunks on the "Trunks" page, it will appear on the tabs. For example, if you have named Mobile Trunk 1 as Vodafone, Mobile Trunk 2 as Airtel, these names will appear in place of Mobile 1 and Mobile 2. If you want to change the name of the Mobile Trunk, you must go back to the "Trunks" page. Follow the instructions provided below to configure the Mobile Trunks.

Select the **Enable Port** check box, to use the Mobile Trunk port. When the Mobile Trunk port is disabled, neither incoming nor outgoing calls can be made from that port. Default: Enabled.

SIM PIN

- Click **SIM PIN** to expand.
- You may change the **SIM PIN** for the SIM Card inserted in the Mobile Port.

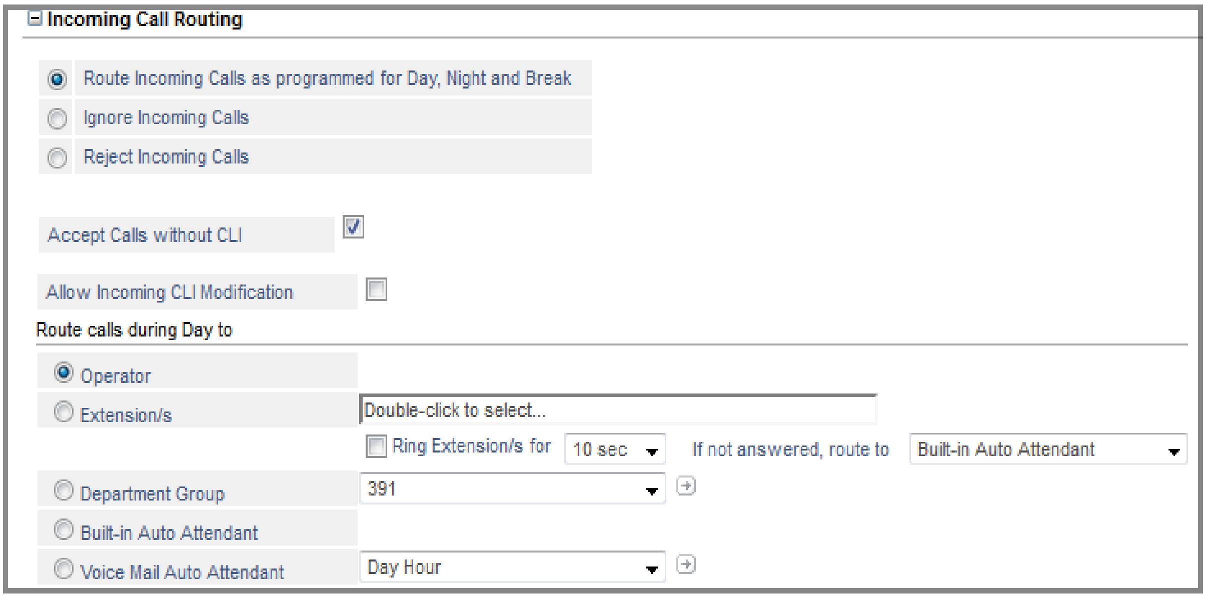
  Recall that you have changed the SIM PIN of the SIM Card using a Handset before installing it in the system to the default value '1234'. Now, you may change it.

  The SIM PIN may be up to 8 digits long. Only the digits from 0 to 9 are allowed.

*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.*

Incoming Call Routing

- Click **Incoming Call Routing** to expand.



- Select to route the incoming calls from the following options:
  - **Route Incoming Calls as programmed for Day, Night and Break:** Select this option, if you want incoming calls to be routed to specific destinations during the day, night and break. If you select this

option, you must also configure the landing destinations for incoming calls during the day, night and break.

- **Ignore Incoming calls**[68]
- **Reject Incoming calls**

- To allow incoming calls without CLI to land on the Mobile Port, select the **Accept Calls without CLI** check box.

- 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”.

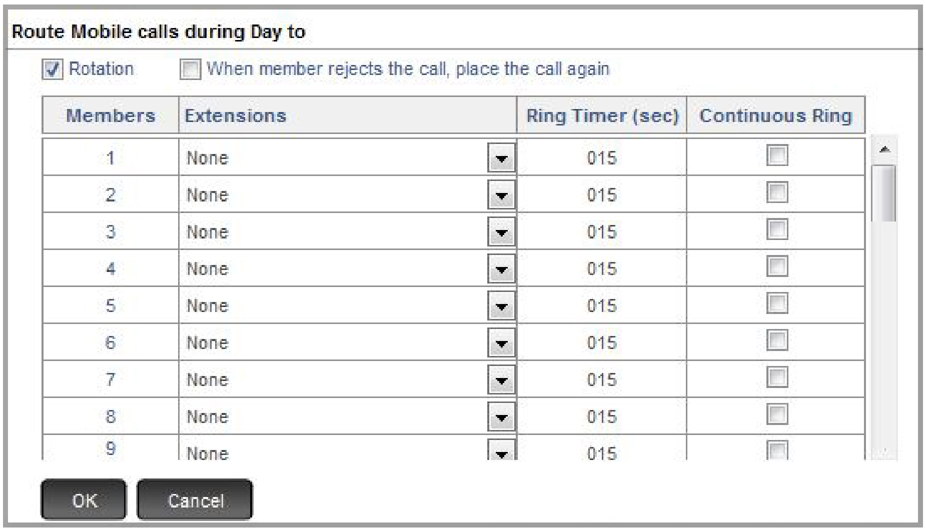
If you receive CLI in dialable format, there is no need to use this feature. In such case, keep the check box disabled.

*For an incoming call on the Mobile trunk, the Incoming CLI Modification is applicable only when both — the **Allow Incoming CLI Modification** check box and the **Enable Incoming CLI Modification** check box in System Parameters— are enabled.*

- Select the landing destination to **Route Calls during Day to**. You may select:
  - Operator
  - Extensions
  - Department Group
  - Built-in Auto Attendant
  - Voice Mail Auto Attendant (if available)

- If you select **Extensions** as the landing destination,

  - Double click the **Extensions** field. A new window opens.



  - Select the extensions you want to assign as landing destination. The landing destination may be a SLT, SIP extension, Virtual extension or Voice Mail Auto Attendant Menu. If you select landing

68. *In this case, the caller will continue to get ring back tone until the call is timed out by the network. No outgoing call can be made from the mobile trunk until the port is released.*

destination as Voice Mail Auto Attendant Menu, make sure you have configured the desired "Voice Mail Auto Attendant Menu".

- Set the **Ring Timer** for the extensions. This timer defines the time for which the extension, on which the call lands, should ring. Default: 15 seconds.
- Set **Continuous Ring**[69] for the extensions, if you want the extensions to ring till the incoming call is answered. Default: Disabled.
- Enable **Rotation**[70], if you have selected more than one extension. Default: Enabled.

If you select Voice Mail Auto Attendant Menu as landing destination, the parameters Ring Timer and Continuous Ring will not be applicable.

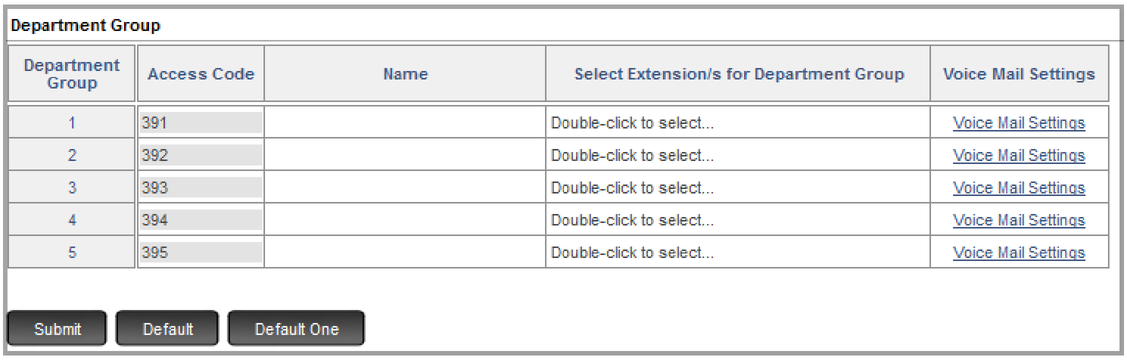
- By default, **When member rejects the call, place the call again** is disabled. Therefore, if any SIP extension user rejects an incoming call, the system will not place the same call on this extension again while checking the routing group to land the call. You may enable this check box, if required.

  If this check box is disabled and you have selected the Continuous Ring 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.
- Click **OK**. All the extensions you selected will appear in **Extension**, in the sequence you selected, separated by commas.

- You may route the incoming call to an auto attendant, if the selected extensions do not answer the call.
  - Select the **Ring Extension/s** check box and set the ring duration in **for___seconds**. Default: 10 seconds. This is the time for which the system will ring on the destination extensions you selected, and wait for the extension to answer. If the call remains unanswered on the expiry of this timer, SARVAM UCS will route the call to the auto attendant.
  - Select the destination for an unanswered call in the **If not answered, route to**. You may select **Built-In Auto Attendant** or **Voice Mail Auto Attendant**. Default: Built-In Auto Attendant.
- If you select a **Department Group** as the landing destination,

69. *When Continuous Ring is selected, the first extension in the group you have created will continue to ring, even when the system hunts for other extensions in the 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 has no relevance, if you select only one landing extension.*
70. *When Rotation is enabled, each new call lands on the extension next to the one that received the last call. This ensures equal distribution of incoming calls to all the destination extensions in this group.*

- Click the **Settings** icon. A new window opens.



Department Group

Department Group	Access Code	Name	Select Extension/s for Department Group	Voice Mail Settings
1	391		Double-click to select...	Voice Mail Settings
2	392		Double-click to select...	Voice Mail Settings
3	393		Double-click to select...	Voice Mail Settings
4	394		Double-click to select...	Voice Mail Settings
5	395		Double-click to select...	Voice Mail Settings

Submit Default Default One

To configure the Department Group parameters, see “Department Call”.

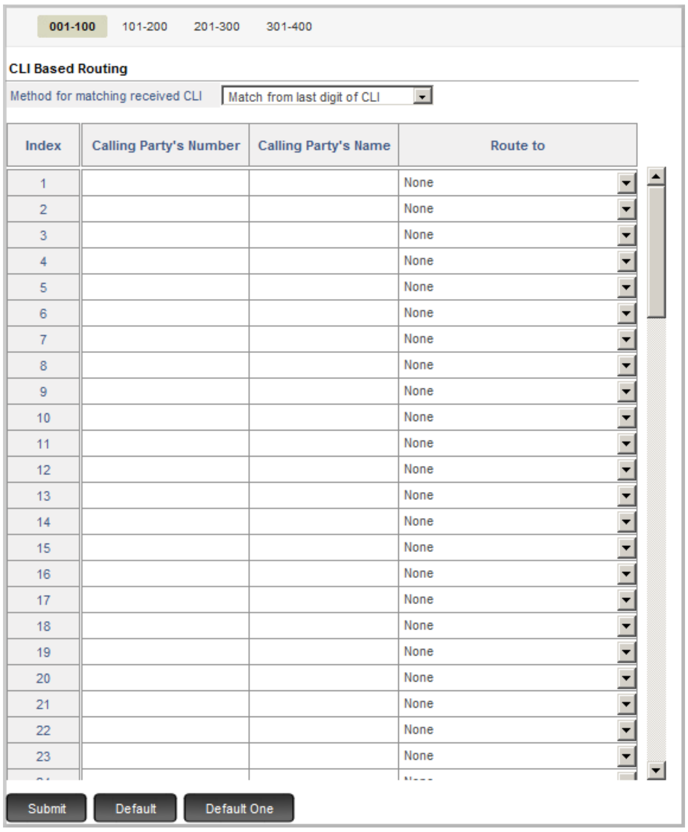
- If you select **Built-in Auto Attendant** as the landing destination for calls during the day,
  - the Voice Modules 02 to 13 will be played as “Built-In Auto Attendant Greeting Messages” and “Built-In Auto Attendant Guidance Messages”.
  - If you want to play a different message, make sure you record the desired messages in the Voice Modules after completing the installation with Basic Settings. Refer the “Voice Message Applications” topic to know more.

  See “Auto Attendant” to know more.

- If you select **Voice Mail Auto Attendant** as the landing destination for calls during the day, then configure the “Voice Mail Auto Attendant Menu”.

- To enable CLI Based Routing on the Mobile Trunk during day, select the **Apply CLI Based Routing** check box. Default: Disabled.

- Click the **Settings** icon. A new window opens.



To configure the CLI Based Routing Table parameters, see "CLI Based Routing".

- To enable "Direct Inward System Access (DISA)" on the Mobile Port, select the desired **Direct Inward System Access** option. Default: Disabled.
  - **PIN Authentication–Multiple calls:** Select this option if you want to enable DISA with PIN Authentication and allow multiple calls during the DISA login session.
  - **CLI Authentication–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.
  - **CLI Authentication–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.

If you select CLI Authentication Multiple Calls or CLI Authentication One Call as CLI Authentication, you must configure the DISA CLI Authentication Table under Advanced Settings. See "Direct Inward System Access (DISA)".

- To enable "Trunk Auto Answer"[71] on the trunk, select the type of **Trunk Auto Answer** you want:
  - **For all Calls:** the system will answer all incoming calls landing on the trunk line.
  - **When Busy:** the system will answer incoming calls on the trunk, if the landing destination is busy.

71. *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.*

- **Delayed:** the system will first land the call on the extension and if not answered before the expiry of the specified Delayed Trunk Auto Answer timer, it will be auto answered.

  On selecting this type of Trunk Auto Answer, you must configure **Delayed Trunk Auto Answer Timer**. You can enter the timer value within the range of 01 to 99 seconds. Default: 10 seconds.

- **OFF**: To disable Trunk Auto Answer, select **OFF**. Default: OFF.

- Select the Trunk Auto Answer **Greeting Message** you want the system to play when greeting the callers, from the following options:
  - Disabled
  - Greeting Message1
  - Greeting Message2
  - Greeting Message3
  - Greeting Message4

  Default: Disabled.

  If you do not want greeting message to be played, select **Disabled**.

When you select a Greeting Message, you must record a Voice Module with the desired Greeting Message and assign the Voice Module to Trunk Auto Answer Greeting application.

You can record 4 different Greeting Messages for Trunk Auto Answer and assign a different Greeting message for the Day, Break and Night. See the topic "Voice Message Applications" for instructions on recording and assigning voice modules to greeting messages.

- Select the Auto Answer **Ring Back Tone Message** you want the system to play to callers from the following options:
  - Do Not Play
  - Play Music on Hold
  - Play RBT Message1
  - Play RBT Message2
  - Play RBT Message3
  - Play RBT Message4

  If you do not want RBT message to be played, select **Do Not Play**. The system will play Ring Back Tone to the caller.

  Default: Do Not Play.

When you select an RBT Message (1–4), you must record a Voice Module with the desired RBT Message and assign the Voice Module to Trunk Auto Answer RBT Message application

You can record 4 different RBT Messages for Trunk Auto Answer and assign a different RBT message for the Day, Break and Night. See "Voice Message Applications" for instructions on recording and assigning voice modules for RBT messages.

- Select the Trunk Auto Answer **Busy Bye Message** you want the system to play to callers when the landing destination extension is busy, from the following options:
  - None
  - Bye Message1
  - Bye Message2
  - Bye Message3
  - Bye Message4

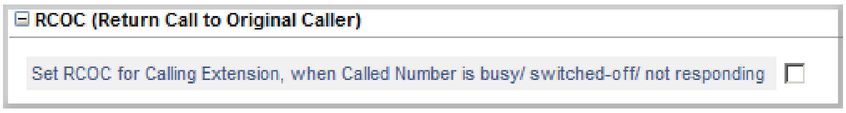
  If you do not want Busy Bye message to be played, select **None**. The system will not play Busy Tone to the caller. Default: None.

When you select a Bye Message (1–4), you must record a Voice Module with the desired Bye Message and assign the Voice Module to Trunk Auto Answer Bye Message application.

You can record 4 different Bye Messages for Trunk Auto Answer and assign a different Bye message for the Day, Break and Night. See “Voice Message Applications” for instructions on recording and assigning voice modules for Bye messages.

- Follow the same instructions as described above to configure **Incoming Call Routing** to **Route Calls during Break to** and **Route Calls during Night to.**

Return Call to Original Caller



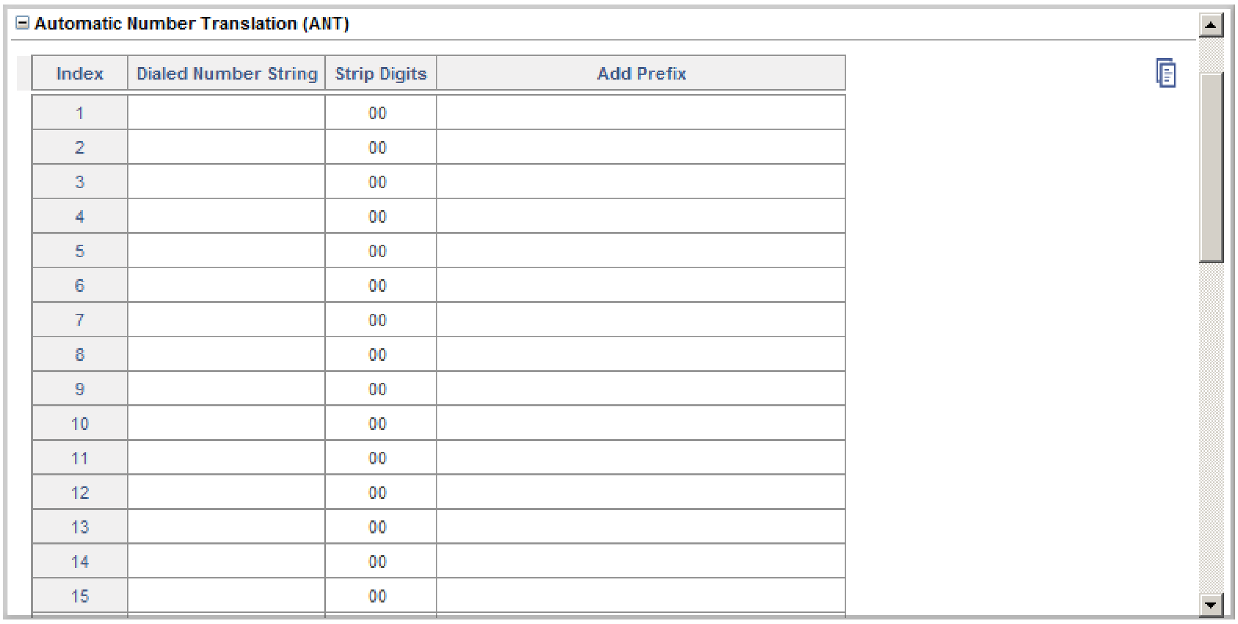
- Click **RCOC** to expand.

- To apply the **RCOC (Return Call to Original Caller)** feature on the mobile trunk, select the check box **Set RCOC for Calling Extension, when Called Number is busy/ switched-off / not responding.** Default: Disabled.

  When RCOC is enabled on the Mobile trunk, the system routes calls returned by remote parties back to the extensions that originally made the call from this trunk (i.e. the original callers' extensions).

  To know more, refer the “RCOC (Return Call to Original Caller)” feature.

Automatic Number Translation (ANT)



Automatic Number Translation (ANT)

Index	Dialed Number String	Strip Digits	Add Prefix
1		00	
2		00	
3		00	
4		00	
5		00	
6		00	
7		00	
8		00	
9		00	
10		00	
11		00	
12		00	
13		00	
14		00	
15		00	

The feature, Automatic Number Translation of SARVAM UCS modifies dialed numbers or part thereof to match the specific route numbering plan understood by the destination network—CO, GSM, VoIP—by adding or stripping country and area codes.

For example, you can configure Automatic Number Translation such that when an extension user dials a local landline number, the SARVAM UCS prefixes the number with the appropriate country-area code when it routes the call through the GSM network.

- To apply the Automatic Number Translation[72] feature on the Mobile Trunk,
  - Click **Automatic Number Translation** to expand.

  The ANT table has three columns: Dialed Number String, Strip Digits, Add Prefix. Each number string is stored against an Index number, from 1 to 32. You can enter as many as 32 entries in the ANT table.

  - In the **Dialed Number String** column, 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 String before 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 a prefix to the Dialed Number String before the system dials out this number. The Prefix can be a maximum of 40 characters. Default: Blank.

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72. *Automatic Number Translation modifies dialed numbers or part thereof to match the specific route numbering plan understood by the destination network—CO, GSM, VoIP—by adding or stripping country and area codes.For example, you can configure Automatic Number Translation such that when an extension user dials a local landline number, the SARVAM UCS prefixes the number with the appropriate country-area code when it routes the call through the GSM network.To know more about this feature, see the description for "Automatic Number Translation".*

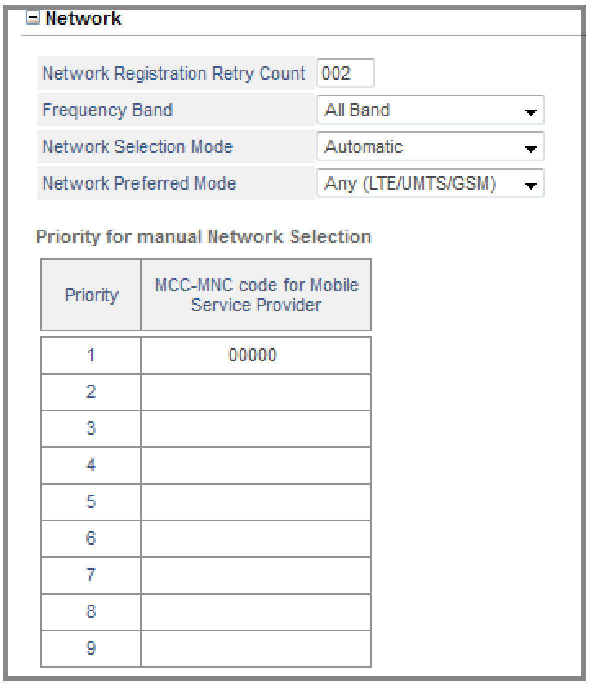
- Click **Submit**.

For more details, see the description for “Automatic Number Translation”.

SMDR Storage

- To record details and generate reports of Internal, Incoming and Outgoing calls made from/to the mobile trunk, enable SMDR Storage. See “Station Message Detail Recording (SMDR)” and “Station Message Detail Recording–Storage” to know more.
  - Click **SMDR Storage** to expand options.
  - Select the **Store Incoming Calls** check box to store details of incoming calls landing on this trunk. Default: selected.
  - Select the **Store Outgoing Calls** check box to store details of outgoing calls made from this trunk. Default: selected.

Network



- Click **Network** to expand.
  - Set the **Network Registration Retry Count** from 1 to 255. This is the number of attempts the Mobile Port will make to register with the network of the service provider whose SIM you have inserted in the port. Default: 02 attempts.

The Mobile Port is configured to automatically locate and register with the Network that supports the SIM Card installed on it. Also, at each power ON, the Mobile Port (i.e. 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[73]. On the expiry of this timer, the system will retry registration for the Count (i.e. number of times) configured and with each re-try attempt, the count will be decremented by one.

- Select the **Frequency Band** supported by the service provider network whose SIM Card is installed in the Mobile Port. Default: All Band.

- *Frequency Band selection is not required, if the Mobile Port has SIMCOM 3G or Quectel 3G/4G modules.*

- *When you change the Frequency Band, the change will be effected after the next system restart or the next Mobile Port restart.*

- Set the **Network Selection Mode** for the Mobile Port as **Automatic** or **Manual**. Default: Automatic.

  At each power on, the Mobile Port will automatically locate and register with the Network that supports the SIM Card present in it. If the Mobile port fails to register, it will restart the process of network selection on the expiry of the Network Registration Retry Timer and will make as many attempts to register as configured in the Network Registration Retry Count.

  Enable the Manual Network Selection mode, when there is more than one network operator available in the area[74], so as to prevent the SIM Card from registering with another available network and resulting in 'Roaming' charges.

- Select the **Network Preferred Mode** for the Mobile Port as **Any (LTE/UMTS/GSM), GSM only**, **UMTS only** or **LTE only**. Default: Any (LTE/UMTS/GSM).

If you select the Network Preferred Mode as LTE only, calling functionality will be possible only if the Service provider supports calling through VoLTE.

- If you have enabled Manual network selection mode, configure the table **Priority for Manual Network Selection**.

  In this table, enter the Network Operator Codes (MCC-MNC)[75] in the order of priority for the Mobile Port. The codes must not exceed 8 digits. You can store up to 9 Network Operator Codes in the order of priority.
  So, whenever you register with the network manually, the system will select the Network Operator that matches in order of priority. If the Mobile Port fails to register, it will restart the process of network selection on the expiry of the Network Registration Retry Timer.

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73. *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-configurable.*
74. *This may be the case in border areas.*
75. ***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.

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.

If you change the Network Selection Mode to 'Manual' and change the Network Operator Code manually, the change you made will come into effect only after the system restart.

DSS Key Interface

- Click **DSS Key Interface** to expand.

  This parameter determines what the system should do when an external called party in speech on this trunk is put on Hold by a Extended IP Phone extension user by pressing a DSS key to dial another port.

  For example, the SIP extension user on SIP1 port is in the middle of speech with an external party on a Mobile 1. If the extension user of SIP1 presses a DSS key to call another extension port, SIP2, two situations are possible, depending on the DSS Key Interface you configure:

  a. The external caller on Mobile 2 will be played music-on-hold, and the user on SIP1 will hear Ring Back Tone. The call will be placed on SIP2.

  b. The external caller on Mobile 1 will be disconnected. SIP1 user will hear Ring Back Tone. The call will be placed on SIP2.

  - Select the radio button of the desired DSS Key Interface option:

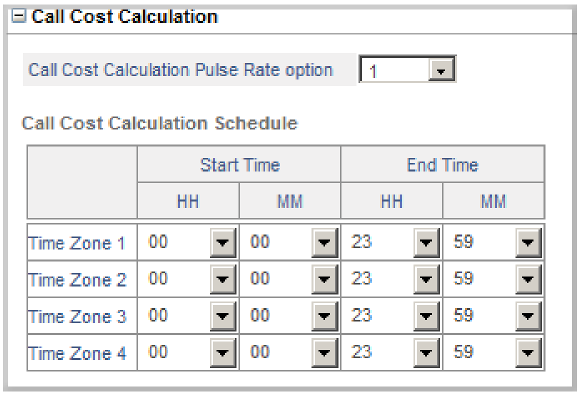
    - **Disconnect current call when Extended IP Phone user presses DSS key of another Trunk, while in speech on this Trunk**

    - **Put current call on Hold, when Extended IP Phone user presses DSS key of another Trunk, while in speech on this Trunk.** (Default).

Call Cost Calculation

Configure this parameter, if you want to apply Call Cost Calculation feature on the trunk. See "Call Cost Calculation (CCC)".

- Click **Call Cost Calculation** to expand.



- Select a **Call Cost Calculation Pulse Rate Option**, from 1 to 4.

  Each Call Cost Calculation Pulse Rate option contains a **Pulse Rate Type**, which you must configure. Some service providers offer discounted rates for special days. In which case, you can configure the rates for normal days in the **Normal Pulse Rate** Table and the rates for special days in the **Discounted Pulse Rate** Table. See "Call Cost Calculation (CCC)"

  Configure the Pulse Rate Type according to the billing pattern of the service provider of this trunk.

  Trunks following the same billing pattern should be assigned the same Pulse Rate Option.

- Define the **Call Cost Calculation Schedule** for the Pulse Rate option you selected.

  The Pulse Rates offered by service providers may vary according to the time of the day. In such cases, you must divide the day into Time Zones, from 1 to 4, as required, to match the time of the pulse rates offered by your service provider.

  Specify the **Start Time** and the **End time** (in 24 hours, HH:MM format) for the **Time Zone** Index in which the particular Pulse Rate will be applied.

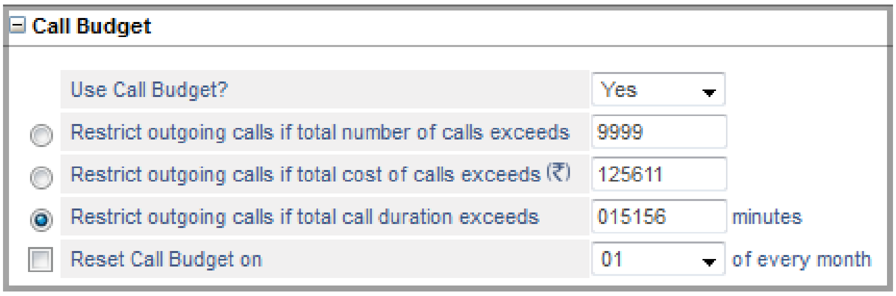
  The default Time Schedule (start and end time) for each Time Zone Index is as follows:

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 calling a number for the entire day, there is no need to change the default Time Schedule. The system will follow Time Zone 1.

  See "Call Cost Calculation (CCC)".

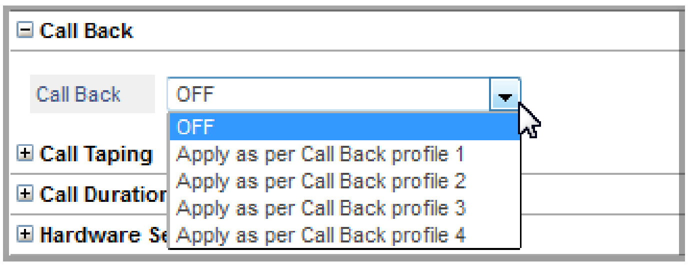
Call Budget



- To disable call budget on the trunk, set **Use Call Budget?** to No. Default: Yes. By default the Call Budget type selected is **Restrict outgoing calls if total call duration exceeds**.
  - Define the call budget in terms of number of calls in the **Restrict outgoing calls if total number of calls exceeds.** Default: 9999.
  - Define the call budget in terms of amount in the **Restrict outgoing calls if total cost of the calls exceeds.** Default: 999999 (local currency).
  - Define the call budget in terms of duration in the **Restrict outgoing calls if total call duration exceeds.** Default: 300 minutes. By default this option is selected as the type of Call Budget.
  - To have the system reset the Number of Calls/Amount/Minutes automatically on a particular date, select the desired date from the **Reset Call Budget on** combo box as 1st to 31st of the every month.

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



- Click **Call Back** to enable the 'Call Back on Trunk Port' feature on the mobile trunk. See "Call Back on Trunk Ports" to know more about this feature.
  - Select a **Call Back Profile**, 1 to 4, for the trunk port from the combo box.
    Make sure you also configure the Call Back profile you select here.

  To configure the profile, click the **Settings** icon. This icon does not appear when Call Back is **OFF**.
  - A new window opens.
  - Configure the Profile Number. See "Call Back on Trunk Ports" for instructions.
  - Click **Submit** to save your profile.
  - Close the window.

Call Taping

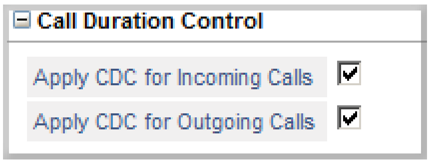
To use "Call Taping" feature on the trunk,

- Click **Call Taping** to expand.
- Select the check box to enable Call Taping on the trunk. Default: Enabled.

  To disable Call Taping on the trunk, clear the check box.
- Click **Submit**.

See "Call Taping" to know more.

Call Duration Control (CDC)



To use the "Call Duration Control (CDC)" feature on trunk,

- Click **Call Duration Control** to expand.
- Select the **Apply CDC for Incoming Calls** check box, to control the call duration for incoming calls on this trunk. Default: Enabled.

  Clear this check box, if you do not want to apply CDC for incoming calls on this trunk.
- Select the **Apply CDC for Outgoing Calls** check box, to control the call duration for outgoing calls on this trunk. Default: Enabled.

  Clear this check box, if you do not want to apply CDC for outgoing calls on this trunk.
- Click **Submit**.

See "Call Duration Control (CDC)" to know more.

Gain Settings

- Click **Gain Settings** to expand. Configure the following Transmit and Receive Gain Settings.
  - **Mobile - System (Tx-Gain and Rx-Gain)**: Configure the Gain Settings that you want the system to apply on the Mobile 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 and Rx Gain is: Very Low, Low, Medium, High, Very High. Default: Medium.
  - **Mobile - Voice Mail (Tx-Gain and Rx-Gain)**: Configure the Gain Setting that you want the system to apply on the Mobile port when the incoming calls on the Mobile port are being answered by the VMS. Valid Range for Tx Gain and Rx Gain is: Very Low, Low, Medium, High, Very High. Default: Medium.
  - **Mobile - Voice Module (Tx-Gain and Rx-Gain)**: Configure the Gain Settings that you want the system to apply on the Mobile port when incoming calls are answered using Auto Attendant or Trunk Auto Answer. Valid Range for Tx Gain and Rx Gain is: Very Low, Low, Medium, High, Very High. Default: Medium.
  - **Mobile - Call Progress Tones (Tx-Gain and Rx-Gain)**: Configure the Gain Setting that you want the system to apply on the Mobile port while playing Call Progress Tones. Valid Range for Tx Gain and Rx Gain is: Very Low, Low, Medium, High, Very High. Default: Medium.
  - **Mobile - SLT (Tx-Gain and Rx-Gain):** Configure the Gain Setting that you want the system to apply on the Mobile Port when the Mobile Port is connected to any FXS Port of the system during an incoming or outgoing call. Valid Range for Tx Gain and Rx Gain is: Very Low, Low, Medium, High, Very High. Default: Medium.
  - **Mobile - CO (Tx-Gain and Rx-Gain)**: Configure the Gain setting that you want the system to apply on the Mobile Port when the Mobile Port is connected to another CO Port of the system during an incoming or outgoing call. Valid Range for Tx Gain and Rx Gain is: Very Low, Low, Medium, High, Very High. Default: Medium.

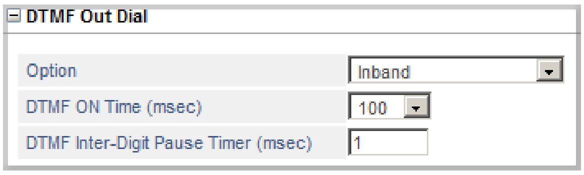
- **Mobile - SIP (Tx-Gain and Rx-Gain)**: Configure the Gain Setting that you want the system to apply on the Mobile Port when the Mobile Port is connected to any SIP Trunk or SIP Extension of the system during an incoming or outgoing call. Valid Range for Tx Gain and Rx Gain is: Very Low, Low, Medium, High, Very High. Default: Medium.

- **Mobile - Mobile (Tx-Gain and Rx-Gain)**: Configure the Gain Setting that you want the system to apply on the Mobile Port when the Mobile Port is connected to any other Mobile Port of the system during an incoming or outgoing call. Valid Range for Tx Gain and Rx Gain is: Very Low, Low, Medium, High, Very High. Default: Medium.

- Click **Submit**.

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.

DTMF Out Dial



- Click **DTMF Out Dial** to expand.

  - Select the DTMF Dial Out **Option**; whether to send the DTMF digits from the Mobile Ports **In-band** or through signaling, that is, **Using AT Command.** Default: 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.

  - Set **DTMF ON Time (msec)** to the desired value. Default: 100ms.

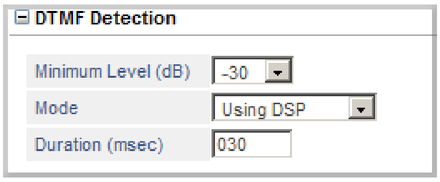
    DTMF ON Time is the time for which the DTMF digit will remain ON when being dialed out by the SARVAM UCS. DTMF ON Time finds its application in the “Multi-Stage Dialing” feature.

  - Set the **DTMF Inter-Digit Pause Timer**. Default: 1 second.

    The Pause Time provides the delay in number dialing from the mobile port. The Pause Time 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 configured as 3 seconds, the SARVAM UCS will out dial the digit 2 after 9 seconds, i.e. after a delay of individual P i.e. 3+3+3 =9. The range of this time is from 1 to 9. The Pause Time is used in the “Multi-Stage Dialing” feature.

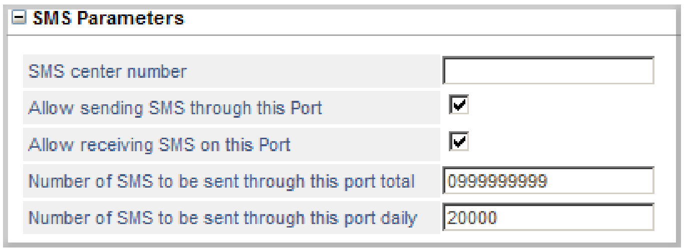
DTMF Detection



- Click **DTMF Detection** to expand.
  - Select the required **Minimum Level** value from the drop down list; which can be 0dB, -3dB, -6dB, -9dB, -12dB, -15dB, -18dB, -21dB, -24dB, -27dB and -30dB. Default: -30dB.
  - Select the required **Mode**; whether to detect the DTMF digits **Using Modules** or **Using DSP**. Default: Using DSP.

    When you select **Using Module**, the length of the DTMF digits will be determined by the Timer you set.
  - Enter the minimum **Duration** of the timer. The valid range of this timer is 20 to 100ms. Default: 30ms.

SMS Parameters



- Click **SMS Parameters** to expand.
  - In **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 +
  - Enable the **Allow sending SMS through this Port** check box, if you want to use this Mobile Trunk Port to send SMS.
  - Enable the **Allow receiving SMS on this Port** check box, if you want to use this Mobile Trunk Port to receive SMS.
  - In **Number of SMS to be sent through this port total**, enter the maximum number of SMS that can be sent using this Mobile Trunk Port. Valid Range: 1 to 4294967296. Default: 0999999999.
  - In **Number of SMS to be sent through this port daily,** enter the maximum number of SMS that can be sent using this Mobile Trunk Port daily. Valid Range: 1 to 20000. Default: 20000.

VoLTE Configuration

- *VoLTE Configuration will be visible only when 4G module is installed.*
- *The 4G module will be supported only with PCB V3R1 onwards with CPLD version V3R2 or later.*

The feature, VoLTE (Voice over Long Term Evolution) Configuration, 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 network coverage and added security.

SARVAM UCS gives you the facility to access VoLTE network depending upon the 4G GSM module installed in the system. 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 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.

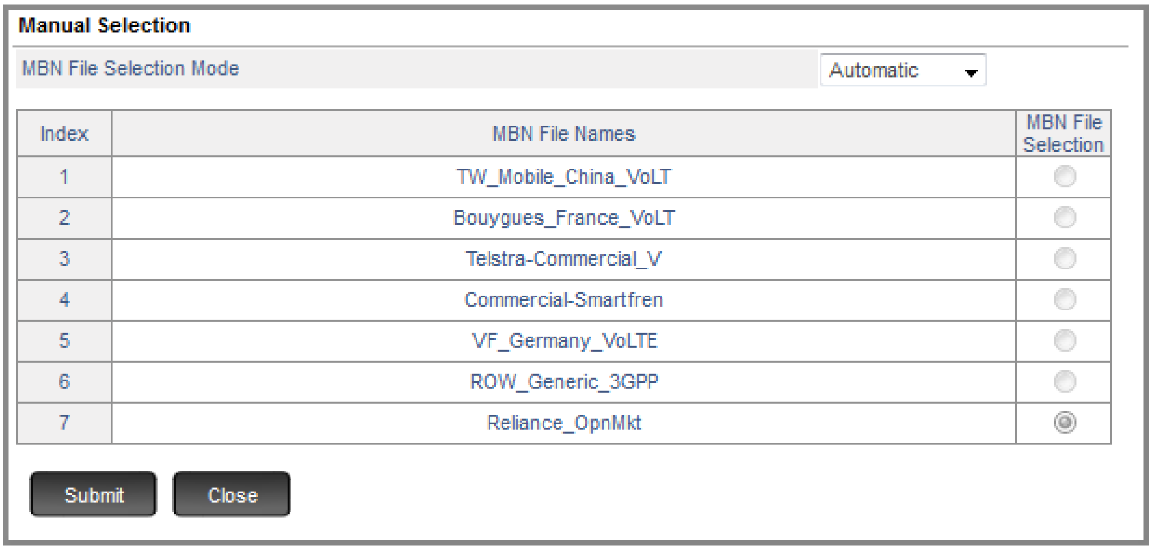
*Make sure that you have configured **Network Preferred Mode** as Any (LTE/UMTS/GSM) or LTE only. For more details, refer to “Network”.*

- Click **VoLTE Configuration** to expand.
  - **MBN File Selection Mode** displays the current mode.

  To change the Selection Mode,

    - Click .

A window pops up which displays the **Manual Selection** Table.



Index	MBN File Names	MBN File Selection
1	TW_Mobile_China_VoLT	
2	Bouygues_France_VoLT	
3	Telstra-Commercial_V	
4	Commercial-Smartfren	
5	VF_Germany_VoLTE	
6	ROW_Generic_3GPP	
7	Reliance_OpnMkt	

The list of MBN Files will be visible only when 4G SIM is inserted.

- In **MBN File Selection Mode**, select the desired option - Automatic or Manual. 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 select **Automatic** mode, 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 select **Manual** mode, 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.

- The module stores varies MBN files. **MBN File Used** displays the name of the MBN file being used by the particular Mobile Port.

MBN File Upload

To upload a new MBN file,

- Click **MBN File Upload** button. A new window pops up.

- In **Select MBN File,** click **Browse** button to reach the location where the 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 Idle state or no SIM is inserted.*
- *you check all the ports for the updated MBN files.*

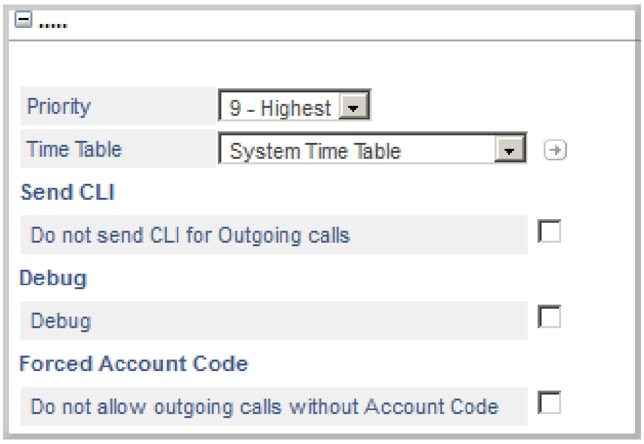
On successful upload a message is displayed.

To view uploaded file in the list of MBN files,

- In **MBN File Selection Mode,** click .

More Features

- Click **More...** link to expand.



Priority

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, see "Priority".

- Select a **Priority** level from 1 to 9 for the trunk. Default: 9-Highest.

Time Table

- Select a **Time Table** for the Mobile trunk. Default: System Time Table.

If you have not configured Time Table, you may do so now,
- Clicking the **Settings** icon.
- The Time Table page opens.

- Define the Working days, and the start and end time of the Working hours and Break hours for each working day.
- Click **Submit**.
- Close the window.

See "Time Tables" to know more.

Send CLI

- To hide the subscriber number of the Mobile Port from being displayed to the called party, select the check box **Do not send CLI for Outgoing Calls.**

This feature will work only if subscribed/supported by your mobile service provider.

Debug

- To initiate debug for the Mobile Port, select the **Debug** check box. Default: Disabled.

Forced Account Code

- To apply the Forced Account Code feature on the Mobile Trunk, select the **Do not allow Outgoing Calls without Account Code** check box (Forced Account Code). Default: Disabled.

  When you enable this check box, the system will prompt extension users to dial the Account Code whenever they grab this trunk to dial out a number. The system will allow extension users to dial out numbers only after they have dialed the Account Code or Name.

Account Codes feature must also be enabled in the Class of Service of extension users who are to be allowed this feature. See "Account Codes" to know more about this feature.

Copy Parameter Values

After you have finished configuring this Mobile Trunk, you may configure the next Mobile Trunk. To do so,

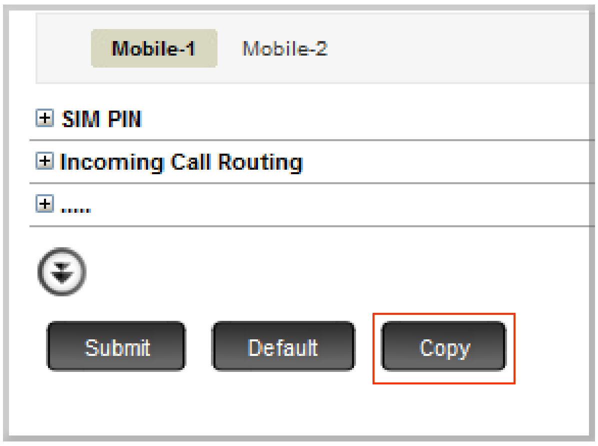
- Click the tab of the next Mobile Trunk.

- Follow the same steps as described above to configure another Mobile Trunk.

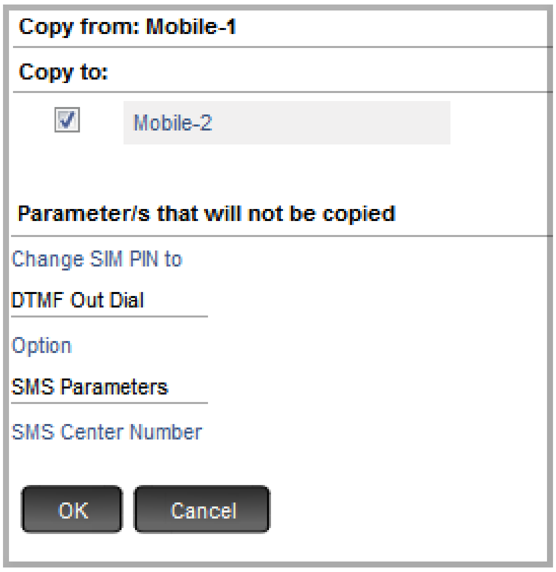
OR

You may use the **Copy** button at the bottom of the page to apply the same Mobile Trunk settings you configured for the current trunk to another Mobile Trunk. To do so,

- Click **Copy**.



The Copy page opens in another tab.



The copy page displays the number and/or name of the current trunk in **Copy from:**. It also lists the numbers and/or names of the trunks to which you can copy the trunk settings under **Copy To:**.

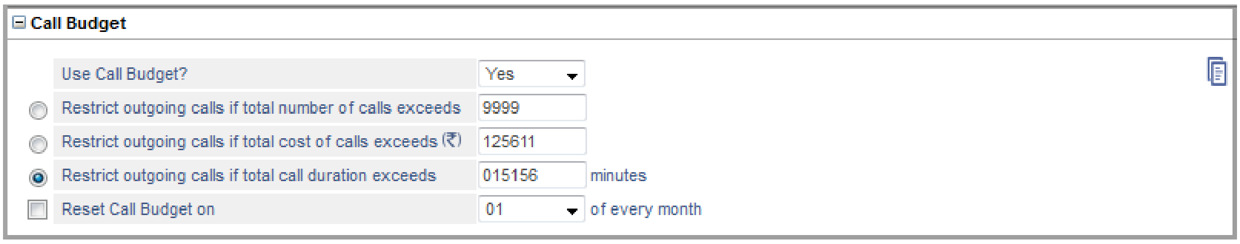
- Select the respective **Copy to** check boxes of the Mobile Trunks to which you want to apply the current Mobile Trunk settings.

- Click **OK**.

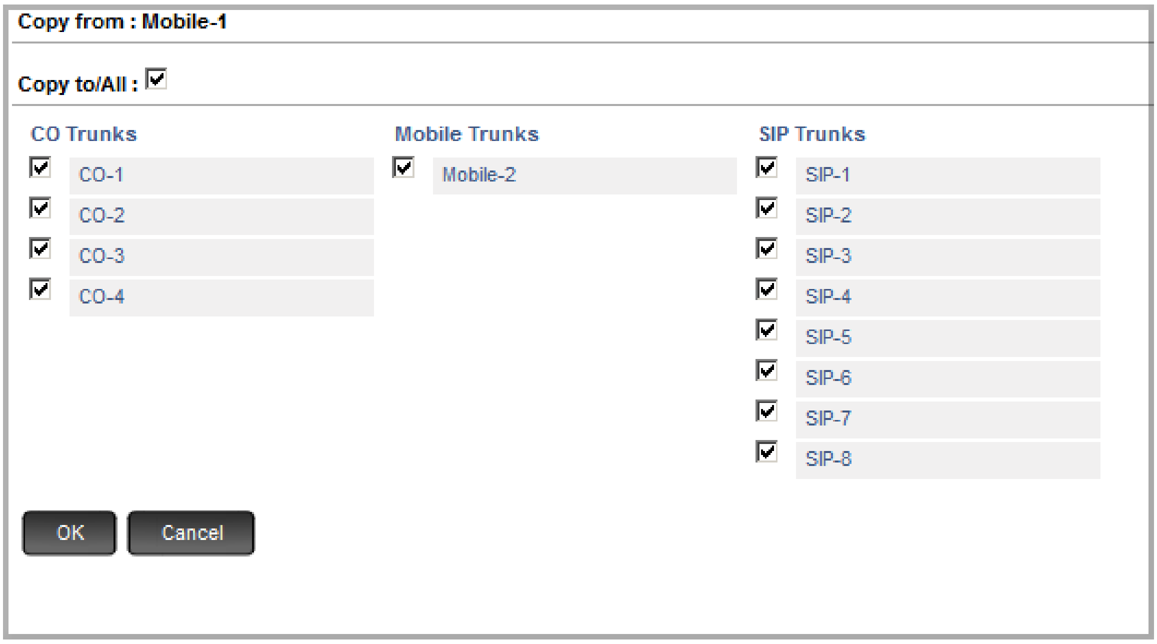
All the parameter values of the current Mobile Trunk except those listed below **Parameter/s that will not be copied**, will be applied to the selected Mobile Trunk.

You may also copy the parameter values of a specific link under current Mobile Trunk to the same link under any other trunk page. To do so,

- Click the **Copy** icon besides the desired link.



The Copy page opens in another tab.



The Copy page displays the number and/or name of the current trunk in **Copy from:**. It also lists the numbers and/or names of the trunks to which you can copy the trunk settings under **Copy To/All:**.

- Select the **Copy to/All** check box, to apply the link settings to the same link under all other Trunks.

  OR

  You may also apply the link settings to the desired Trunks only, by selecting the respective check boxes.

- Click **OK**.

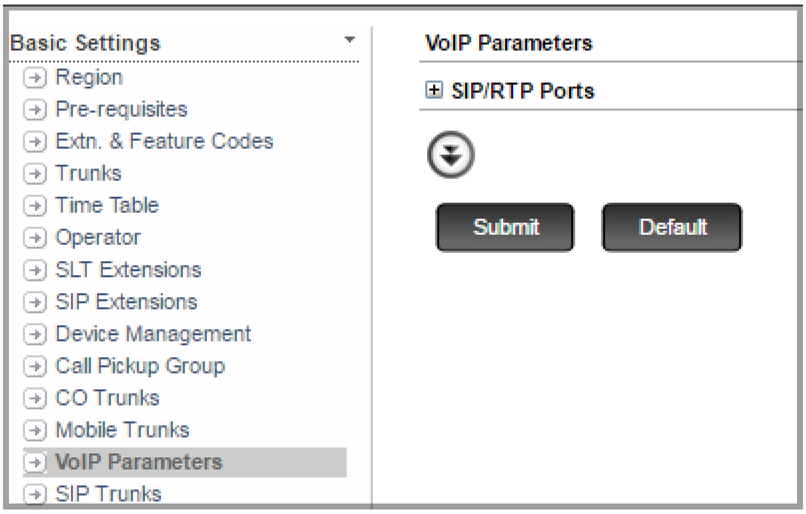
VoIP Parameters

You will reach this page if you have set 'Is VoIP used?' to 'Yes' on the "Pre-requisites" page.

To configure,

- Click **VoIP Parameters**.

  On this page,



More: Click this button to view all parameter links on the page.

Less: Click this button to view only the essential parameter links on the page.

Expand: Click to expand a link to display all parameters under the link.

Collapse: Click to collapse a link. Hides all parameters under the link.

Settings: Click to configure the settings of a parameter further.

···· **More link:** Click to view all additional parameters on the page.

To save the settings, click **Submit.**

To assign default values to all the parameters of the VoIP Port, click **Default.**

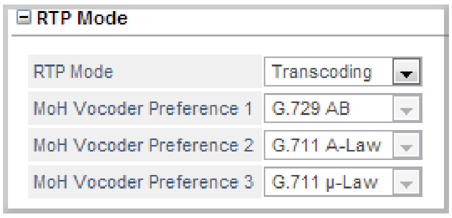
VoIP Server Domain

- Click **VoIP Server Domain**[76] to expand.

- Enter the **VoIP Server Domain**, if you want the SIP clients to register with the Registrar Server of the VoIP application using the domain handled by the SARVAM UCS.

The VoIP module is capable of maintaining a domain for registering SIP clients (any SIP-enabled device) as SIP Extensions.

RTP Mode



- Click **RTP Mode** to expand.

- 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 VoIP module and DSP channels will be used 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 DSP channels for SIP to SIP calls. Thus the maximum number of SIP to SIP calls is equal to the number of DSP channels divide by 2.

  - **RTP Relay:** When this option is selected, RTP packets will be routed through the VoIP module but no DSP channels will be used for SIP to SIP calls. The system will use DSP channels to route SIP to TDM calls and vice versa. System will use DSP 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 DSP 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 DSP channels to route SIP to TDM calls and vice versa. System will use DSP channels for some

---

76. *SIP clients can be registered with the VoIP application either using the domain handled by the VoIP module or using the Ethernet Port IP Address.*

    If domain is programmed, VoIP module will listen for the SIP message which is redirected to the programmed domain as well as will listen for SIP messages on the Ethernet IP address.

    But if domain is not programmed, the VoIP module will listen for SIP messages only on the Ethernet Port IP address.

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.

Default: Transcoding.

Direct RTP is not supported on SIP Trunks, VARTA Clients and Extended SPARSH VP710. System will use RTP Relay for SIP Trunks, VARTA Clients and Extended SPARSH VP710.

- *If RTP Relay or Direct RTP is selected,*
  - *maximum 25 Audio calls or 8 Video calls are supported.*
  - *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.*

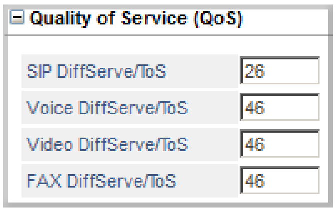
- If RTP Mode is set as **RTP Relay** or **Direct RTP**, the MoH will be played when the system holds any SIP end point. The MoH is played as per the configured MoH Vocoder Preference.

  Select the Vocoders in the order of their preference, for **MoH Vocoder Preference 1**, **MoH Vocoder Preference 2** and **MoH Vocoder Preference 3**.

  Default: MoH Vocoder Preference 1 is G.729 AB, MoH Vocoder Preference 2 is G.711 A-Law, MoH Vocoder Preference 3 is G.711 μ – Law.

  If required, you can customize the MoH to be played as per your requirement. For detailed instructions, see "Voice Message Applications".

Quality of Service



Quality of Service (QoS) refers to priority of IP packets on network layer. QoS is configured for signaling (SIP) and media (Voice, Video and Fax). SARVAM UCS sends all the SIP signaling messages and the Media packets according to the QoS settings. To configure QoS for SIP (signaling) and Media (Voice, Video and Fax):

- Click **Quality of Service (QoS)** to expand.

- Define the priority bits for SIP messages in the **SIP Diffserve/ToS**. The VoIP module sends all the SIP signaling messages with this QoS setting. Valid range is 00 to 63. Default: 26.

- Define the priority bits for Voice packets in the **Voice DiffServe/ToS.** The VoIP module sends all the Voice packets with this QoS setting. Valid range is 00 to 63. Default: 46.

- Define the priority bits for Video packets in the **Video DiffServe/ToS.** The VoIP module sends all the Video packets with this QoS setting. Valid range is 00 to 63. Default: 46.

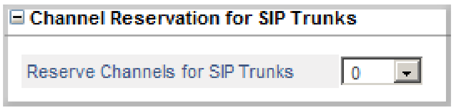
- Define the priority bits for Fax packets in the **Fax DiffServe/ToS.** The VoIP module sends all the Fax packets with this QoS setting. Valid range is 00 to 63. Default: 46.

Disconnection on Silence Detection

You can have any matured incoming or outgoing call to be disconnected automatically, if silence (No RTP Packets) is detected for more than a specified duration of time. To use this feature,

- Click **Disconnection on Silence Detection** to expand.
- Select the **Disconnection on Silence Detection** check box. Default: enabled.
- Define **Detection Time (sec).** This Timer defines the duration for which silence must be detected continuously, for the system to consider it as silence detection and disconnect the call. The valid range of this Timer is from 001 to 999 seconds. By default, it is set to 999 seconds.

Channel Reservation for SIP Trunks

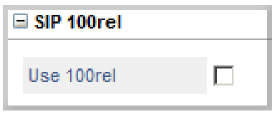


- Click **Channel Reservation for SIP Trunks** to expand.
- Define the number of voice channels you would like to reserve for SIP trunk calls in the **Reserve Channels for SIP Trunks**. Default: 0.

  SARVAM UCS supports 8 voice channels, which can be used by SIP Extensions and SIP trunks.

  It may happen that SIP Extension users use up most of the voice channels, leaving few or none for making/receiving SIP Trunk calls. This can be avoided by reserving some voice channels exclusively for SIP trunk calls.

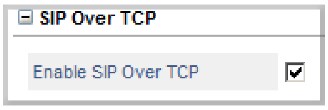
SIP100rel



This parameter is to be configured if you want to support reliable transmission of (SIP) provisional responses and to use PRACK (Provisional Acknowledgement).

- Click **SIP 100rel** to expand.
- To enable SIP 100rel, select the check box **Use 100rel.** Default: Disabled.

SIP Over TCP

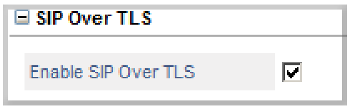


SARVAM UCS supports transporting of SIP messages over User Datagram Protocol (UDP), Transfer Control Protocol (TCP) as well as Transport Layer Security connection. Despite the advantages that SIP over TCP and SIP over TLS offer, it is more common to use UDP to transport SIP messages. To receive SIP messages over TCP,

- Click **SIP Over TCP** to expand.
- Select the check box **Enable SIP over TCP**. Default: Enabled.

To be able to send SIP messages over TCP, you must configure ‘TCP’ as the ‘Default Transport for Outgoing Messages’ on the SIP Trunks page.

SIP Over TLS

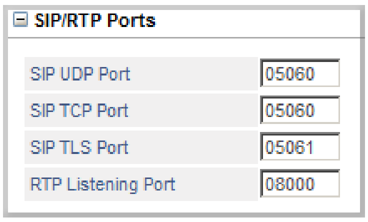


SARVAM UCS supports transporting of SIP messages over Transport Layer Security (TLS). TLS offers secure SIP signaling. To receive SIP messages over TLS,

- Click **SIP Over TLS** to expand.
- Select the check box **Enable SIP over TLS**. Default: Enabled.

To be able to send SIP messages over TLS, you must configure ‘TLS’ as the ‘Default Transport for Outgoing Messages’ on the SIP Trunks page.

SIP/RTP Ports



- Click **SIP/RTP Ports** to expand.
- Enter the **SIP UDP Port** address.

  This port defines the port on which the 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.

- Enter the **SIP TCP Port** address.

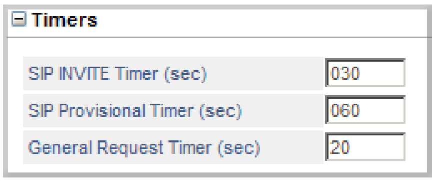
This port defines the port on which the VoIP Port of SARVAM UCS listens for SIP messages transported over TCP. This port is also used as the source port for sending SIP messages to the remote peer. The valid range for this port is 1025-65535. The default SIP TCP Port is 05060.

- Enter the **SIP TLS Port** address.

  This port defines the port on which the VoIP Port of 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.

- Enter the **RTP Listening Port** address. This port defines the port on which the VoIP Port of 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 1025-65278. The default RTP Listening Port is 08000.

Timers



Configure the following SIP message related timers:

- Click **Timers** to expand.

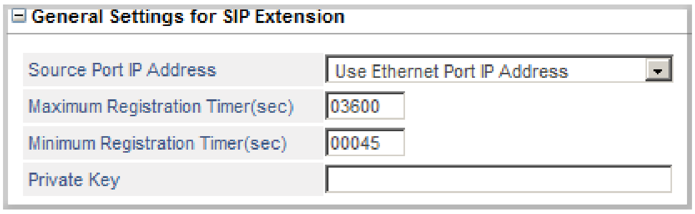
- Set the **SIP Invite Timer.** This is the time in seconds that SARVAM UCS waits for a response from the called party after ending the INVITE message. This timer starts after sending INVITE message to the called party and stops on receipt of the provisional response or the final response or when the 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 10-180 seconds. Default: 30 seconds.

- Set the **SIP Provisional Timer**. This is the time in seconds that the SARVAM UCS 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, SARVAM UCS terminates the call process and plays error tone to the user. The range of SIP provisional Timer is 10-180 seconds. Default: 60 seconds.

- Set the **General Request Timer** to the desired duration. This is the time in seconds for which SARVAM UCS waits for response for a transaction request. This timer starts on the initiation of a transaction. This timer stops on the receipt of a response for the request. On expiry of the timer, SARVAM UCS clears the transaction. This timer is used for Registration request. The range of the General Request Timer is 10-60 seconds. Default: 20 seconds.

The Timers will be applicable only after System Restart.

General Settings for SIP Extensions



If you are connecting SIP Extensions to SARVAM UCS, configure the following general parameters for SIP extensions:

- Click **General Settings for SIP Extensions** to expand.
- Select the **Source Port IP Address**. This parameter specifies the NAT Traversal mechanism for SIP messages. You can select any of these options:
  - **Use Ethernet Port IP Address:** Select this parameter if your SARVAM UCS is directly connected to the public IP network (not behind a NAT Router).
  - **Use Router's Public IP Address:** Select this parameter if your SARVAM UCS is located behind a NAT Router, and you have configured 'Router's Public IP Address' in the “Network Parameters”.
  - **Use IP Address fetched using STUN:** Select this parameter if your SARVAM UCS is located behind a NAT Router, and you have configured 'Use IP Address fetched using STUN' in the “Network Parameters”.
- Set the **Maximum Registration Timer (sec)** to the desired value. This is the maximum expiry timer, which SARVAM UCS will accept in the REGISTER request received. If the system receives more than this value, the configured value will be sent in the SIP message. The same Timer is used to handle SUBSCRIBE requests. The valid range of this timer is from 10 to 99999 seconds. Default: 03600 seconds.
- Set the **Minimum Registration Timer (sec)** to the desired value. 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. Default: 00045 seconds.

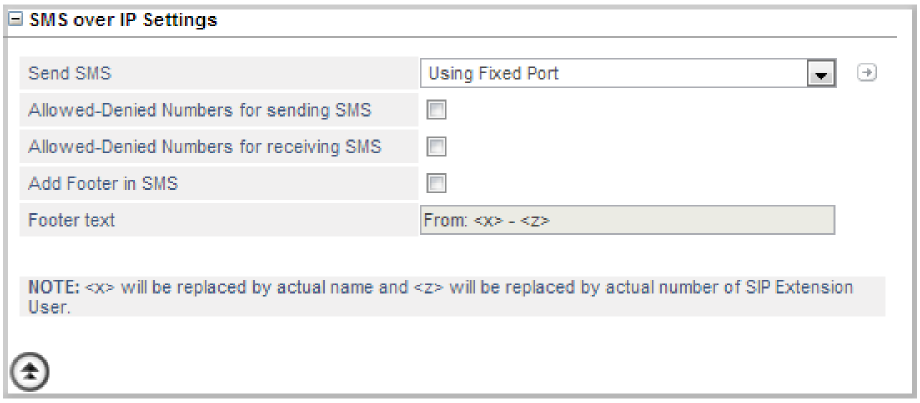
The Timers will be applicable only after System Restart.

- Enter the **Private Key**. It may consist of a maximum of 24 characters (all ASCII characters allowed). Default: Blank.

  Private key is a security mechanism used by SARVAM UCS to authenticate SIP messages. It is the MD5 authentication key used by the system to encrypt/decrypt the SIP messages.

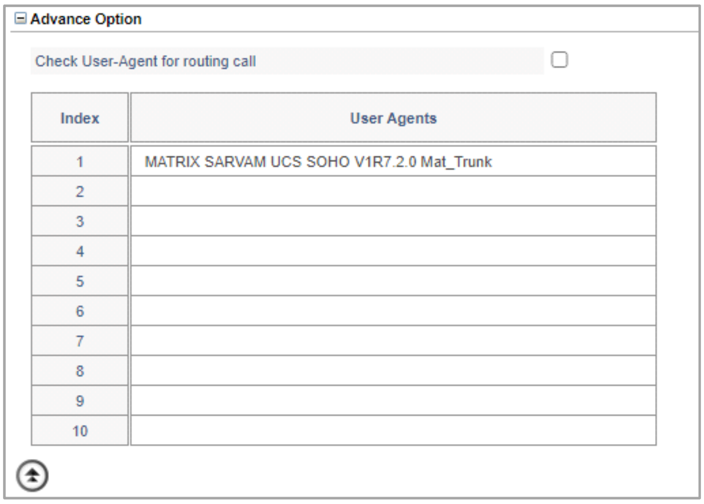
- Click **Submit** to save your VoIP port settings.

SMS over IP Settings



To configure SMS over IP parameters, see "How to configure" under "SMS over IP".

Advance Option



- Click **Advance Option** to expand.

- Select the **Check User-Agent for routing call** check box to enable. Default: Disabled. 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.

SIP Trunks

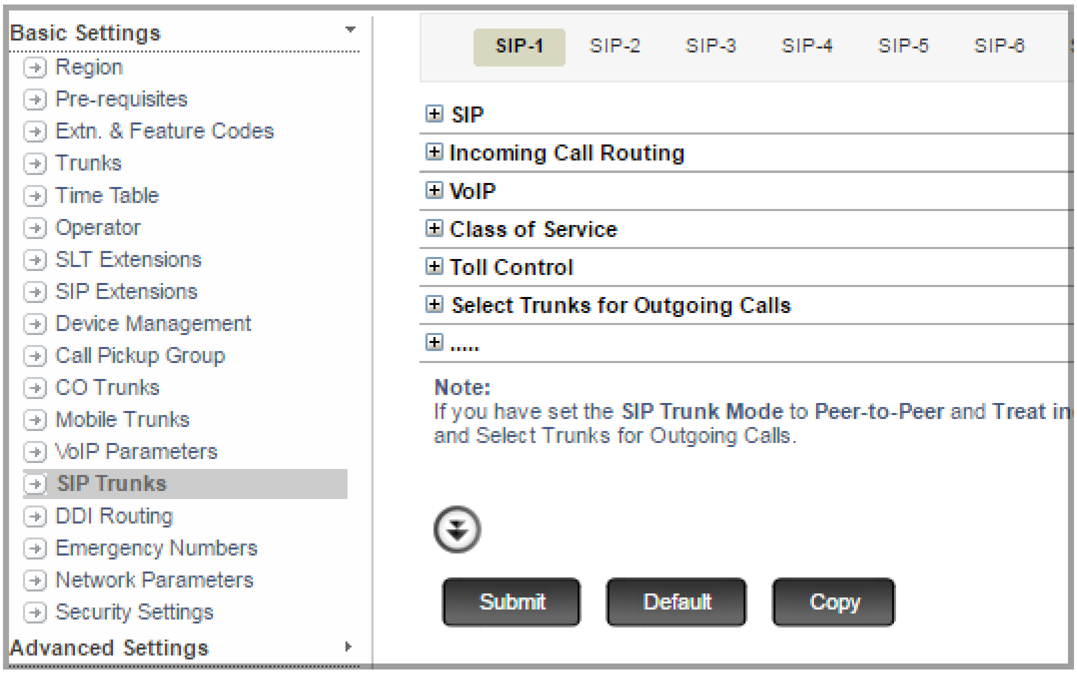
SARVAM UCS supports 8 SIP Trunks. On this page, the number of SIP Trunks you have specified on the "Pre-requisites" page will be displayed.

- *SIP Trunk page is displayed only if you have specified the Number of SIP Trunks Used on the 'Pre-requisites' page*.
- *SIP Trunks are to be configured only if you are using the services of Internet Telephony Service Providers for VoIP calls.*

To configure,

- Click **SIP Trunks**.

  On this page,



More: Click this button to view all parameter links on the page.

Less: Click this button to view only the essential parameter links on the page.

Expand: Click to expand a link to display all parameters under the link.

Collapse: Click to collapse a link. Hides all parameters under the link.

Copy: Click to copy the parameters under a specific link of a page to the same link under any other trunk page.

Settings: Click to configure the settings of a parameter further.

.... **More link:** Click to view all additional parameters on the page.

To configure another SIP Trunk, click the respective SIP Trunk number (name) tab.

To save the settings, click **Submit**.

To assign default values to all the parameters of the SIP Trunk, click **Default**.

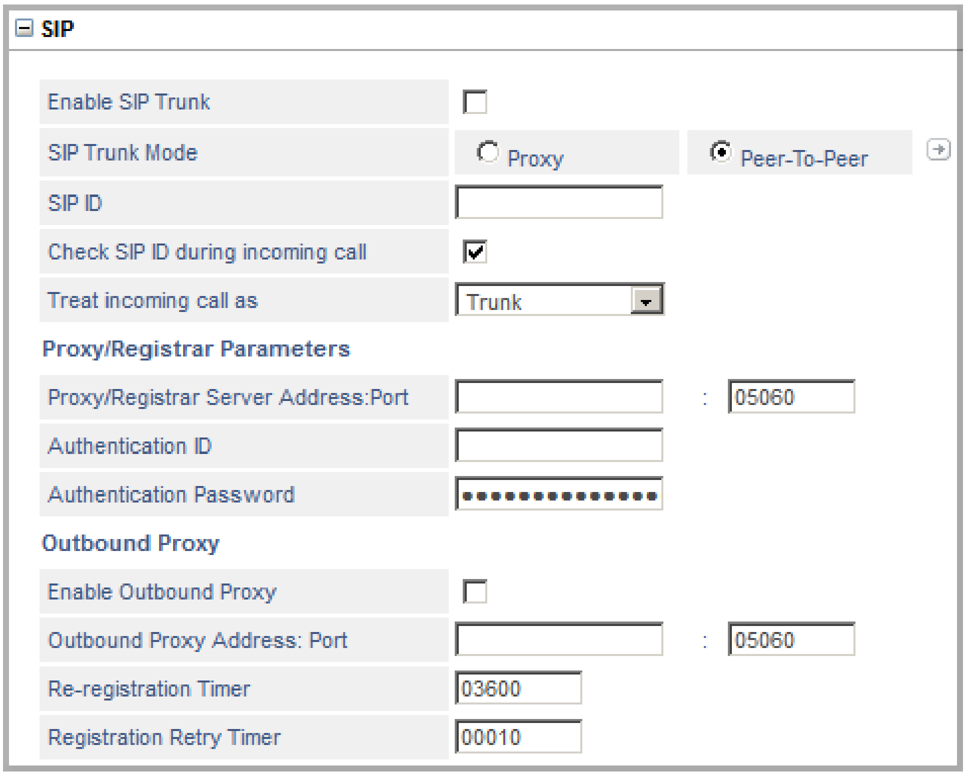
To copy the same SIP Trunk parameter values to the another SIP Trunks, use **Copy**.

If you have assigned names to the SIP Trunks on the "Trunks" page, it will appear on the tabs. For example, if you have named SIP 1 as GlobalTalk, SIP 2 as Pulver, these names will appear in place of SIP Trunk1 and SIP Trunk2.

If you want to change the name of the SIP Trunk, you must go back to the "Trunks" page.

Follow the instructions provided below to configure the SIP Trunks.

SIP



- Click **SIP** to expand.

- Select the **Enable SIP Trunk** check box to activate the SIP Trunk.

- Set the **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 the services of an ITSP, select **Peer-to-Peer** and click **Settings** icon to configure the Peer-to-Peer Table. See "Peer-to-Peer Calling" feature for instructions.

- Enter the **SIP ID** for the trunk.

  This is the ID which remote parties will use to call this SIP Trunk. The SIP ID may be a number or text consisting of a maximum of 40 characters.

  If you have defined the trunk mode as Proxy, enter the User ID provided by your ITSP. For example, if SIP URI provided by the ITSP is 12345@abc.com, enter 12345 in this field.

  If you have defined the trunk mode as Peer-to-Peer, enter the desired User ID.

- To select a SIP Trunk during an Incoming Call Routing, 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.

- Set the **Treat Incoming call as Trunk** or **Station**, according to your requirement.

  If you select **Trunk**, the **Incoming Call Routing** configured for the SIP Trunk will be applied.

  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 using the corresponding trunks.

  - 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.

  - If a Trunk Access Code and External Number are received in the ‘To:” field of the INVITE message, the call will be dialed out using the outgoing trunks you configure in **Select Trunks for Outgoing Calls**.

    By default, Trunk is selected.

  If you select Station, you must also configure the parameters—“Class of Service”, “Caller ID on Call Transfer”, “Toll Control”, “Select Trunks for Outgoing Calls”, “Call Taping”, “Call Duration Control”.

- *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*

Proxy/Registrar Parameters

If you have defined the SIP Trunk mode as Proxy, configure the Proxy/Registrar Server parameters.

- Enter the **Proxy/Registrar Server Address** and the Registrar Server **Port** provided by your ITSP.

  The Registrar Server Address can be an IP Address or domain.

  The Registrar Server Listening Port ranges from 1025 to 65535. Default: 05060.

- Enter the authentication ID (user ID) provided by your ITSP in the **Authentication ID**.

- Enter the password provided by your ITSP for the Authentication ID in the **Authentication Password.**

Outbound Proxy

If the ITSP of this SIP Trunk has a SIP outbound server to handle voice calls, configure outbound proxy settings.

- Select the check box **Enable Outbound Proxy**. Default: Disabled.

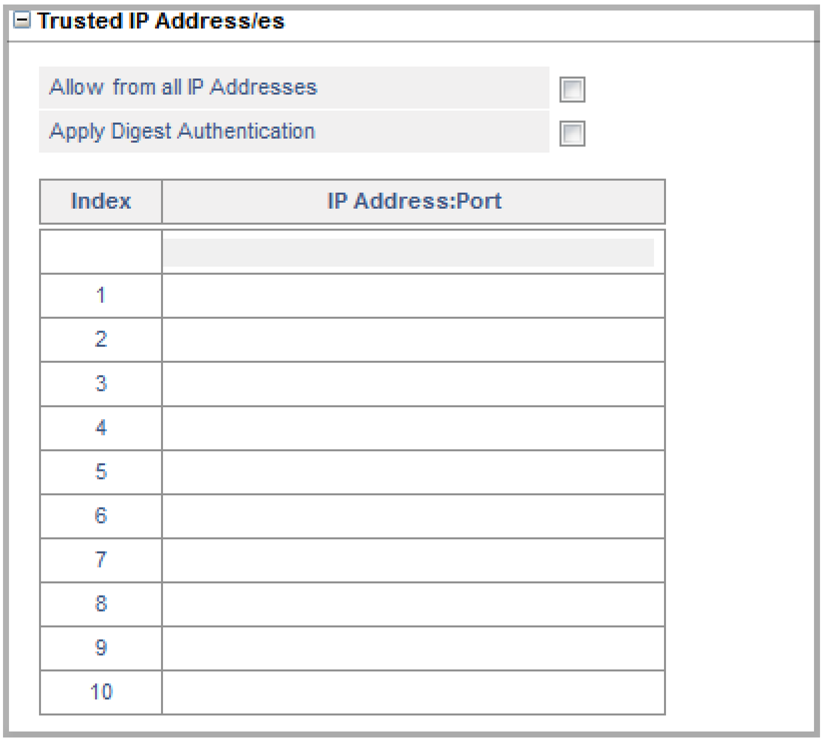
- Enter the IP address or domain name of the outbound proxy server and the server port address in the **Outbound Proxy Server Address: Port**. The valid range is 1025 to 65535. Default Server Port: 05060.

- Set the **Re-registration Timer**. This is the time period after which the SARVAM UCS will send registration request to maintain registration binding with the Registrar server[77]. The valid range of this timer is 00001-65535. Default: 3600 seconds.

- Define the **Registration Retry Timer.** This Timer stands for the period between retries for registration with the Registrar Server. 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 with the Registrar Server. The valid range of this timer is from 00001- 65535. Default: 10 seconds.

77. *The Registrar Server deletes an entry of its client from its database on expiry of a fixed timer. This timer is set by the Registrar Server. The Re-registration Timer of SARVAM UCS enables it to send a registration request before the Timer of the Registrar Server expires to remain registered on the Server.*

Trusted IP Address/es



You must configure the **Trusted IP Address/es** table to allow incoming calls from specific IP addresses on this SIP Trunk.

If you select SIP Trunk Mode as **Peer to Peer** and you do not configure this table, by default, all incoming calls on this SIP Trunk will be rejected.

If you select SIP Trunk Mode as **Proxy** and you do not configure this table, by default, incoming calls will be allowed only from the Proxy/Registrar Server Address or Outbound Proxy Address configured for this SIP Trunk. All other calls on this SIP Trunk will be rejected.

- Click **Trusted IP Address/es** to expand.

If you have selected SIP Trunk mode as Peer to Peer, configure the following.

- Enable **Allow from all IP Addresses** check box, if you want to allow incoming calls on this SIP Trunk from all IP Addresses. Default: Disabled.
- Enable **Apply Digest Authentication** check box, if you want to allow incoming calls from callers only after the callers authenticate 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 **Settings** icon. The Digest Authentication table opens in a new window. For instructions, see "Digest Authentication".

- Enable **Consider Peer to Peer Table for Trusted IP Address** 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/es** table.

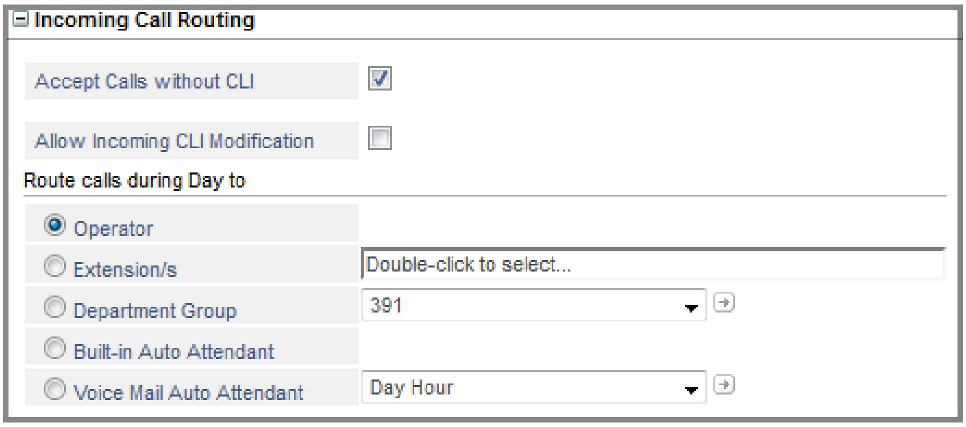
- The first entry in the table will display the *Proxy/Registrar Server Address:Port* or *Outbound Proxy Address: Port* as configured for this SIP Trunk, when SIP Trunk mode is selected as Proxy.

- 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.

Incoming Call Routing



- Click **Incoming Call Routing** to expand.

- To allow incoming calls without CLI to land on the SIP Trunk, select the **Accept Calls without CLI** check box.

- Select the **Allow Incoming CLI Modification** check box if you want to apply Incoming CLI Modification on the SIP 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".

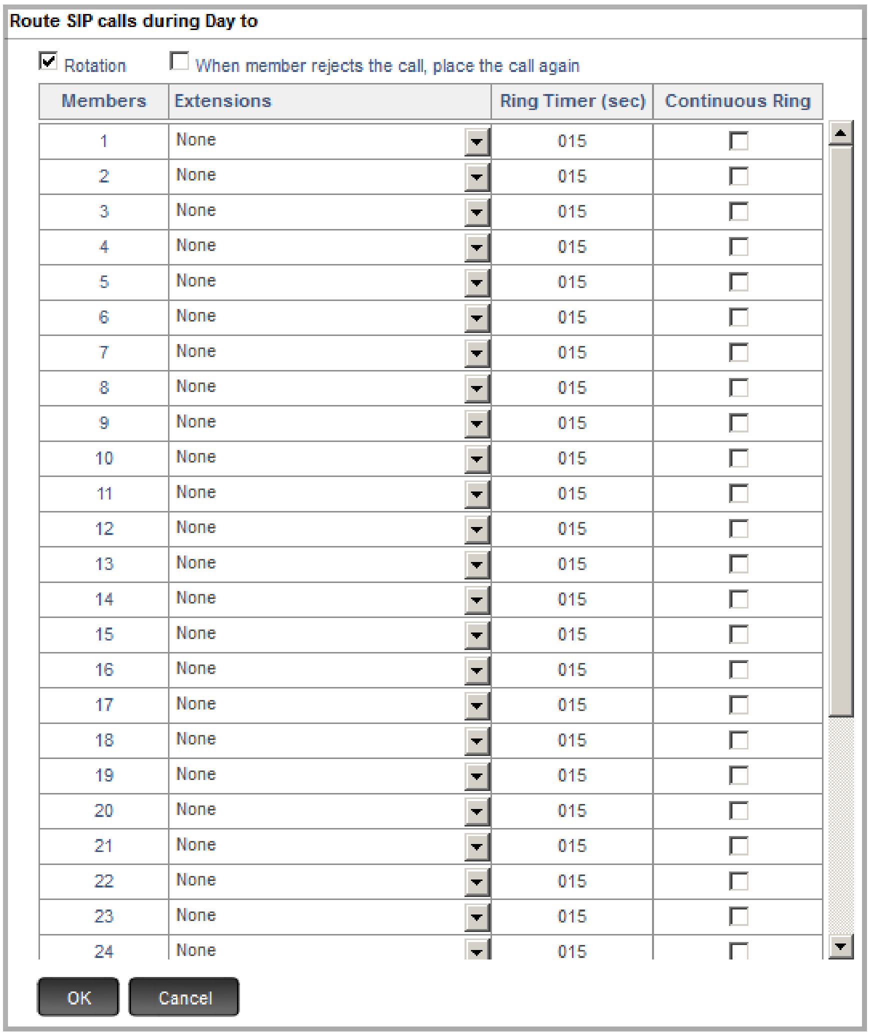
If you receive CLI in dialable format, there is no need to use this feature. In such case, keep the check box disabled.

*For an incoming call on the SIP trunk, the Incoming CLI Modification is applicable only when both — the **Allow Incoming CLI Modification** check box and the **Enable Incoming CLI Modification** check box in System Parameters— are enabled.*

- Select the landing destination to **Route Calls during Day to**. You may select:
  - Operator
  - Extensions
  - Department Group
  - Built-in Auto Attendant
  - Voice Mail Auto Attendant (if available).

- If you select **Extensions** as the landing destination,

  - Double click the **Extensions** field. A new window opens.



Route SIP calls during Day to

☑ Rotation ☐ When member rejects the call, place the call again

Members	Extensions	Ring Timer (sec)	Continuous Ring
1	None	015	☐
2	None	015	☐
3	None	015	☐
4	None	015	☐
5	None	015	☐
6	None	015	☐
7	None	015	☐
8	None	015	☐
9	None	015	☐
10	None	015	☐
11	None	015	☐
12	None	015	☐
13	None	015	☐
14	None	015	☐
15	None	015	☐
16	None	015	☐
17	None	015	☐
18	None	015	☐
19	None	015	☐
20	None	015	☐
21	None	015	☐
22	None	015	☐
23	None	015	☐
24	None	015	☐

OK Cancel

  - Select the extensions you want to assign as landing destination. The landing destination may be a SLT, SIP extension, Virtual extension or Voice Mail Auto Attendant Menu. If you select landing destination as Voice Mail Auto Attendant Menu, make sure you have configured the desired “Voice Mail Auto Attendant Menu”

  - Set the **Ring Timer** for the extensions. This timer defines the time for which the extension, on which the call lands, should ring. Default: 015 seconds.

  - Set **Continuous Ring**[78] for the extensions, if you want the extensions to ring till the incoming call is answered. Default: Disabled.

- Enable **Rotation**[79], if you have selected more than one extension. Default: Enabled.

If you select Voice Mail Auto Attendant Menu as landing destination, the parameters Ring Timer and Continuous Ring will not be applicable.

- Click **OK.** All the extensions you selected will appear in **Extension**, in the sequence you selected, separated by commas.

- You may route the incoming call to an auto attendant, if the selected extensions do not answer the call,
  - Select the **Ring Extension/s** check box and set the ring duration in **for___seconds**. Default: 10 seconds. This is the time for which the system will ring on the destination extensions you selected, and wait for the extension to answer. If the call remains unanswered on the expiry of this timer, SARVAM UCS will route the call to the auto attendant.
  - Select the destination for an unanswered call in the **If not answered, route to**. You may select the **Built-In Auto Attendant** or the **Voice Mail Auto Attendant**. Default: Built-In Auto Attendant.
- If you select a **Department Group** as the landing destination,
  - Click the **Settings** icon. A new window opens.

Department Group

Department Group	Access Code	Name	Select Extension/s for Department Group	Voice Mail Settings
1	391		Double-click to select...	Voice Mail Settings
2	392		Double-click to select...	Voice Mail Settings
3	393		Double-click to select...	Voice Mail Settings
4	394		Double-click to select...	Voice Mail Settings
5	395		Double-click to select...	Voice Mail Settings

Submit | Default | Default One

To configure the Department Group parameters, see “Department Call”.

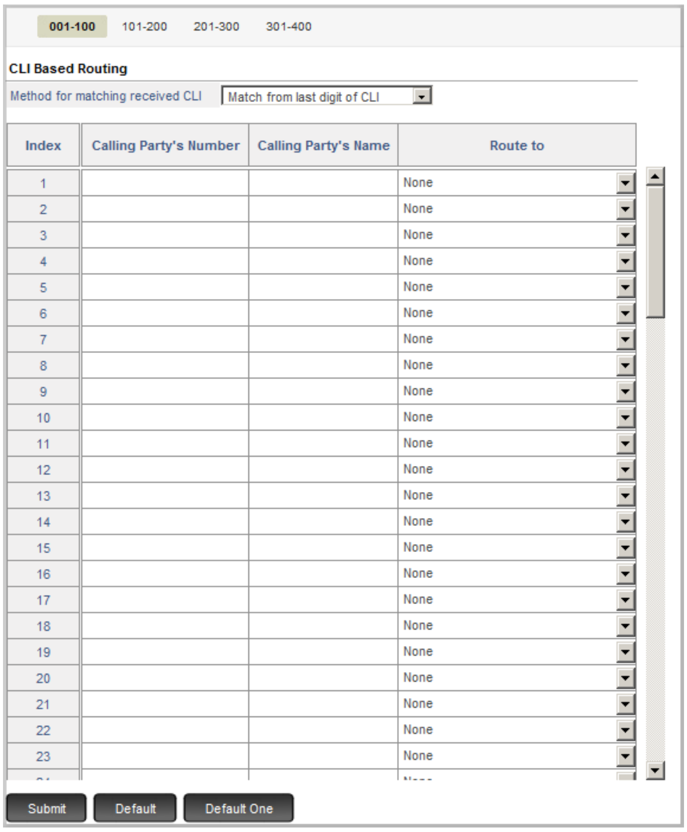
- If you select **Built-in Auto Attendant** as the landing destination for calls during the day,
  - the Voice Modules 02 to 13 will be played as “Built-In Auto Attendant Greeting Messages” and “Built-In Auto Attendant Guidance Messages”.
  - If you want to play a different message, make sure you record the desired messages in the Voice Modules after completing the installation with Basic Settings. Refer the “Voice Message Applications” topic to know more.

  See “Auto Attendant” to know more.

- If you select **Voice Mail Auto Attendant** as the landing destination for calls during the day, then configure the “Voice Mail Auto Attendant Menu”.

78. *When Continuous Ring is selected, the first extension in the group you have created will continue to ring, even as the system hunts for other extensions in the 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 has no relevance, if you select only one landing extension.*
79. *When Rotation is enabled, each new call lands on the extension next to the one that received the call last. This ensures equal distribution of incoming calls to all the destination extensions in this group.*

- To enable CLI Based Routing on the SIP Trunk during day, select the **Apply CLI Based Routing** check box. Default: Disabled.
    - Click the **Settings** icon. A new window opens.



To configure the CLI Based Routing Table parameters, see “CLI Based Routing”.

- To enable “Direct Inward System Access (DISA)” on the SIP Trunk, select the desired **Direct Inward System Access** option. Default: Disabled.
    - **PIN Authentication–Multiple calls:** Select this option if you want to enable DISA with PIN Authentication and allow multiple calls during the DISA login session.
    - **CLI Authentication–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.
    - **CLI Authentication–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.

If you select CLI Authentication Multiple Calls or CLI Authentication One Call as CLI Authentication, you must configure the DISA CLI Authentication Table under Advanced Settings. See “Direct Inward System Access (DISA)”.

- To enable “Trunk Auto Answer”[80] on the trunk, select the type of **Trunk Auto Answer** you want:
    - **For all Calls:** the system will answer all incoming calls landing on the trunk line.

80. *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.*

- **When Busy:** the system will answer incoming calls on the trunk, if the landing destination is busy.

- **Delayed:** the system will first land the call on the extension and if not answered before the expiry of the specified Delayed Trunk Auto Answer timer, it will be auto answered.

  On selecting this type of Trunk Auto Answer, you must configure **Delayed Trunk Auto Answer Timer**. You can enter the timer value within the range of 01 to 99 seconds. Default: 10 seconds.

- **OFF**: To disable Trunk Auto Answer, select **OFF**. Default: OFF.

- Select the Trunk Auto Answer **Greeting Message** you want the system to play when greeting the callers, from the following options:
  - Disabled
  - Greeting Message1
  - Greeting Message2
  - Greeting Message3
  - Greeting Message4

  Default: Disabled.

  If you do not want greeting message to be played, select **Disabled**.

When you select a Greeting Message, you must record a Voice Module with the desired Greeting Message and assign the Voice Module to Trunk Auto Answer Greeting application.

You can record 4 different Greeting Messages for Trunk Auto Answer and assign a different Greeting message for the Day, Break and Night. See the topic “Voice Message Applications” for instructions on recording and assigning voice modules to greeting messages.

- Select the Auto Answer **Ring Back Tone Message** you want the system to play to callers from the following options:
  - Do Not Play
  - Play Music on Hold
  - Play RBT Message1
  - Play RBT Message2
  - Play RBT Message3
  - Play RBT Message4

  If you do not want RBT message to be played, select **Do Not Play**. The system will play Ring Back Tone to the caller.

  Default: Do Not Play.

When you select an RBT Message (1–4), you must record a Voice Module with the desired RBT Message and assign the Voice Module to Trunk Auto Answer RBT Message application

You can record 4 different RBT Messages for Trunk Auto Answer and assign a different RBT message for the Day, Break and Night. See “Voice Message Applications” for instructions on recording and assigning voice modules for RBT messages.

- Select the Trunk Auto Answer **Busy Bye Message** you want the system to play to callers when the landing destination extension is busy, from the following options:
  - None
  - Bye Message1
  - Bye Message2
  - Bye Message3
  - Bye Message4

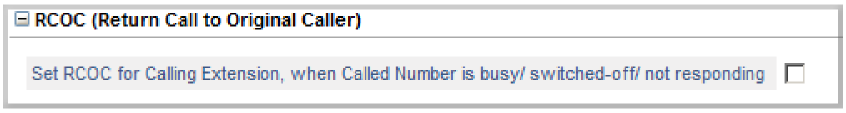
  If you do not want Busy Bye message to be played, select **None**. The system will not play Busy Tone to the caller. Default: None.

When you select a Bye Message (1–4), you must record a Voice Module with the desired Bye Message and assign the Voice Module to Trunk Auto Answer Bye Message application

You can record 4 different Bye Messages for Trunk Auto Answer and assign a different Bye message for the Day, Break and Night. See “Voice Message Applications” for instructions on recording and assigning voice modules for Bye messages.

- Follow the same instructions as described above to configure **Incoming Call Routing** to **Route Calls during Break to** and **Route Calls during Night to**.

RCOC (Return Call to Original Caller)



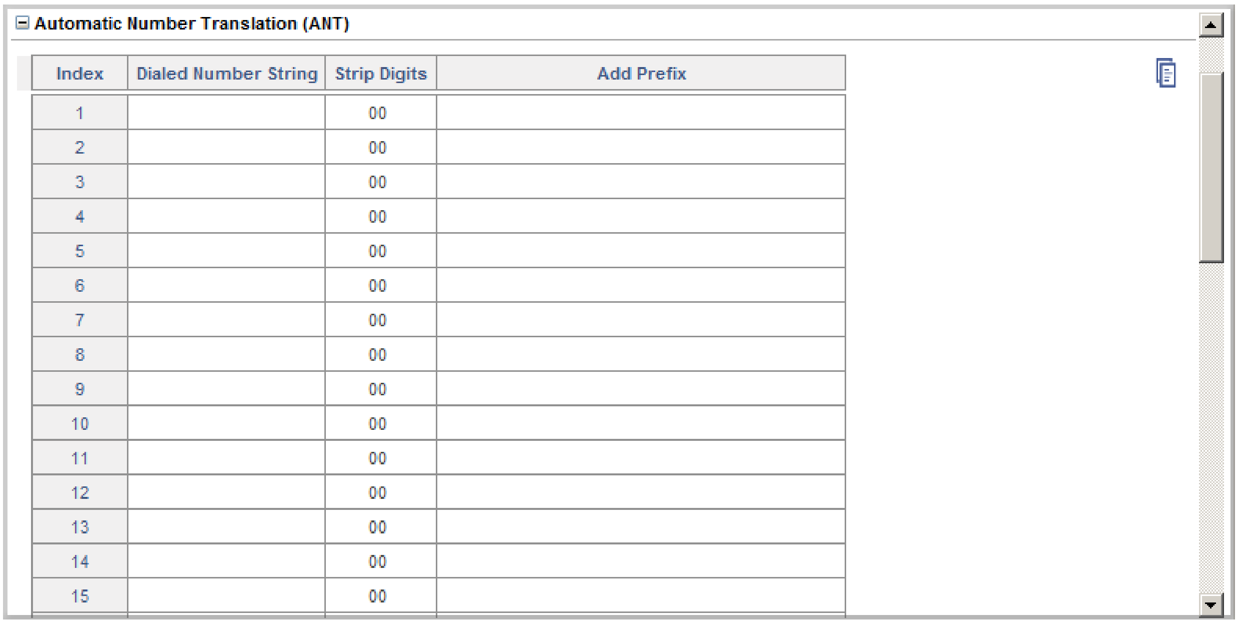
When RCOC is applied on the SIP Trunk, the system routes calls returned by remote parties back to the extensions that originally made the call from this port (i.e. the original callers' extensions). To know more, see “RCOC (Return Call to Original Caller)”.

To apply RCOC (Return Call to Original Caller) on the SIP Trunk,

- Click **RCOC** to expand.

- Select the check box **Set RCOC for Calling Extension, when Called Number is busy/ switched-off / not responding**. Default: Disabled.

Automatic Number Translation (ANT)



Automatic Number Translation (ANT)

Index	Dialed Number String	Strip Digits	Add Prefix
1		00	
2		00	
3		00	
4		00	
5		00	
6		00	
7		00	
8		00	
9		00	
10		00	
11		00	
12		00	
13		00	
14		00	
15		00	

The Automatic Number Translation feature of SARVAM UCS modifies dialed numbers or part thereof to match the specific route numbering plan understood by the destination network—PSTN, GSM, VoIP—by adding or stripping country and area codes.

For example, you can configure Automatic Number Translation such that when an extension user dials a local landline number, the SARVAM UCS prefixes the number with the appropriate country-area code when it routes the call through the GSM network. To know more about this feature, see the description for “Automatic Number Translation”.

To apply Automatic Number Translation (ANT) on the SIP Trunk,

- Click **Automatic Number Translation** to expand.
- The Automatic Number Translation table appears.
  The ANT table has three columns: Dialed Number String, Strip Digits, Add Prefix. Each number string is stored against an Index number, from 1 to 32. You can enter as many as 32 entries in the ANT table.
  - In the **Dialed Number String** column, 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 String before 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 a prefix to the Dialed Number String before the system dials out this number. The Prefix can be a maximum of 40 characters. Default: Blank.
- Click **Submit.**

For more details, see the description for “Automatic Number Translation”.

SMDR Storage

The Station Message Detail Recording (SMDR) feature of SARVAM UCS enables you to record the details of Internal, Incoming (IC) and Outgoing (OG) calls made from/to all its trunks and extensions. To obtain SMDR as a report, you must enable SMDR Storage, and set filters. See “Station Message Detail Recording (SMDR)” and “Station Message Detail Recording–Storage” to know more.

- Click **SMDR Storage** to expand.
- To store details of incoming calls landing on this SIP trunk, select the **Store Incoming Calls** check box. Default: selected.
- To store details of outgoing calls made from this SIP trunk, select the **Store Outgoing Calls** check box. Default: selected.
- Click **Submit**.

DSS Key Interface

This parameter determines what the system should do when an external called party in speech on this trunk is put on Hold by an extension user by pressing a DSS key to dial another port.

For example, the Extended IP Phone user (SIP1) is in the middle of speech with an external party on a CO2.

If the SIP1 user presses a DSS key to call another extension port, SIP2, two situations are possible, depending on the DSS Key Interface you configure:

i. The external caller on CO2 will be played music-on-hold, and the user on SIP1 will hear Ring Back Tone. The call will be placed on SIP2.

ii. The external caller on CO2 will be disconnected. SIP1 user will hear Ring Back Tone. The call will be placed on SIP2.

- Click **DSS Key Interface** to expand options.
- Select the radio button of the desired DSS Key Interface option:
  - **Disconnect current call when Extended IP Phone user presses DSS key of another Trunk, while in speech on this Trunk.**

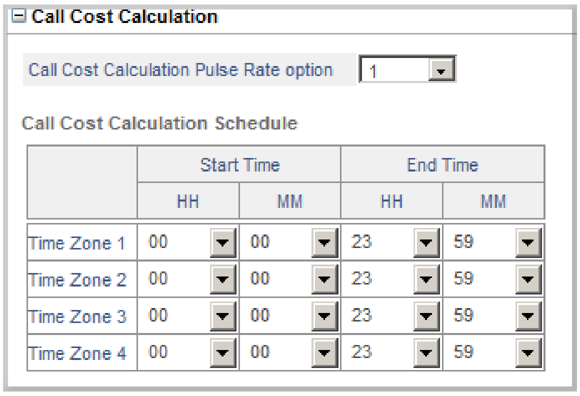
- **Put current call on Hold, when Extended IP Phone user presses DSS key of another Trunk, while in speech on this Trunk.** (Default).

- Click **Submit.**

Call Cost Calculation

Configure this parameter if you want to apply Call Cost Calculation feature on the trunk. See “Call Cost Calculation (CCC)”.

- Click **Call Cost Calculation** to expand.



- Select a **Call Cost Calculation Pulse Rate option**, from 1 to 4.

  Each Call Cost Calculation Pulse Rate option contains a **Pulse Rate Type**, which you must configure. Some service providers offer discounted rates for special days. In which case, you can configure the rates for normal days in the **Normal Pulse Rate** Table and the rates for special days in the **Discounted Pulse Rate** Table. See “Call Cost Calculation (CCC)”.

  Configure the Pulse Rate Type according to the billing pattern of the service provider of this trunk.

  Trunks following the same billing pattern should be assigned the same Pulse Rate Option.

- Define the **Call Cost Calculation Schedule** for the Pulse Rate option you selected.

  The Pulse Rates offered by service providers may vary according to the time of the day. In such cases, you must divide the day into Time Zones, from 1 to 4, as required, to match the time of the pulse rates offered by your service provider.

  Specify the **Start Time** and the **End time** (in 24 hours, HH:MM format) for the **Time Zone** Index in which the particular Pulse Rate will be applied.

  The default Time Schedule (start and end time) for each Time Zone Index is as follows:

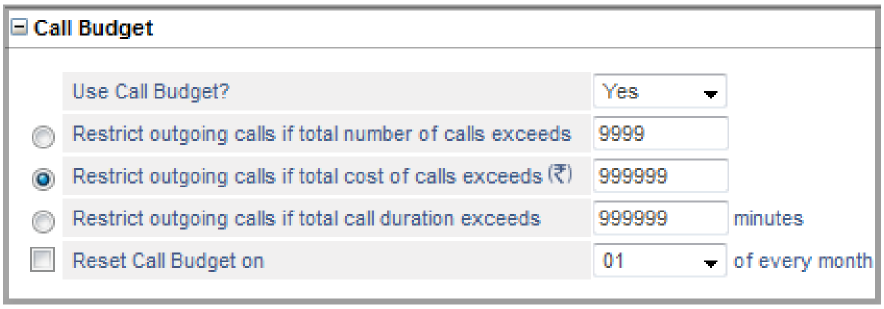
Time Zone Index	Start Time	End Time
T1	00:00	23:59
T2	00:00	23:59
T3	00:00	23:59
T4	00:00	23:59

If your service provider offers the same Pulse Rate for calling a number for the entire day, there is no need to change the default Time Schedule. The system will follow Time Zone 1.

See "Call Cost Calculation (CCC)".

Call Budget

- Click **Call Budget** to expand.

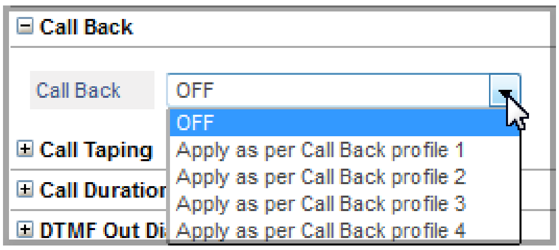


Configure this parameter if you want to control the cost of phone calls made from this trunk. See "Call Budget on Trunk" to know more.

- To disable call budget on the trunk, set **Use Call Budget?** to No. Default: Yes. By default the Call Budget type selected is **Restrict outgoing calls if total call duration exceeds**.
- Define the call budget in terms of number of calls in the **Restrict outgoing calls if total number of calls exceeds**. Default: 9999.
- Define the call budget in terms of amount in the **Restrict outgoing calls if total cost of the calls exceeds.** Default: 999999 (local currency).
- Define the call budget in terms of duration in the **Restrict outgoing calls if total call duration exceeds.** Default: 300 minutes. By default this option is selected as the type of Call Budget.
- To have the system reset the Number of Calls/Amount/Minutes automatically on a particular date, select the desired date from the **Reset Call Budget on** combo box as 1st to 31st of the every month.

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



Configure this parameter if you want to enable the 'Call Back on Trunk Port' feature on the trunk. See "Call Back on Trunk Ports" to know more.

- Click **Call Back** to expand.
- Select a **Call Back Profile**, 1 to 4, for the trunk port from the combo box. Make sure you also configure the Call Back profile you select here.

  To configure the profile, click the **Settings** icon.
  - A new window opens.
  - Configure the Profile Number. See “Call Back on Trunk Ports” for instructions.
  - Click **Submit** to save your profile.
  - Close the window.

Call Taping

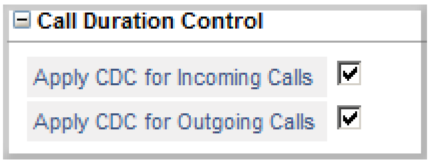
To use “Call Taping” feature on the trunk,

- Click **Call Taping** to expand option.
- Select the check box to enable Call Taping on the trunk. Default: Enabled.

  To disable Call Taping on the trunk, clear the check box.
- Click **Submit**.

See “Call Taping” to know more.

Call Duration Control (CDC)



This option is applicable when you select SIP Trunk Mode as Proxy and have selected Treat incoming call as Trunk. To use the “Call Duration Control (CDC)” feature on trunk,

- Click **Call Duration Control** to expand options.
- Select the **Apply CDC for Incoming Calls** check box, to control the call duration for incoming calls on this trunk. Default: Enabled.

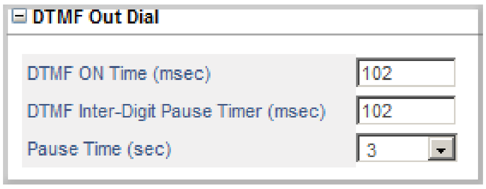
  Clear this check box, if you do not want to apply CDC for incoming calls on this trunk.
- Select the **Apply CDC for Outgoing Calls** check box, to control the call duration for outgoing calls on this trunk. Default: Enabled.

  Clear this check box, if you do not want to apply CDC for outgoing calls on this trunk.

- Click **Submit.**

See “Call Duration Control (CDC)” to know more.

DTMF Out Dial



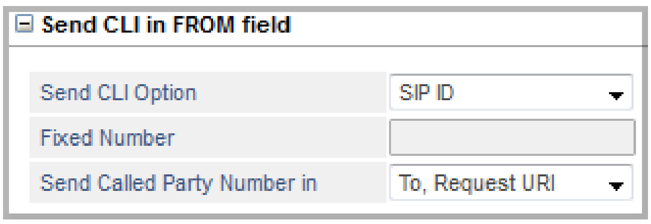
This parameter is applicable for the “Multi-Stage Dialing” feature.

- Click **DTMF Out Dial** to expand.
- Set the time for which the DTMF digit should remain ON while being dialed out by the SARVAM UCS as **DTMF ON Time**.

  The range of this timer is from 051 to 255 milliseconds. Default: 102 msec.
- Set the time for which SARVAM UCS should wait before dialing the successive DTMF digits as **DTMF Inter- Digit Pause Timer (msec)**. The range of this timer is from 051 to 255 milliseconds. Default: 102 msec.
- Set the time **Pause Time (sec)**. It is used for the “Multi-Stage Dialing” feature. 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 Pause Time is set to 3 seconds.

  For example, if 'PPP7' is to be out dialed and Pause time is programmed as 3 seconds, the SARVAM UCS will out dial the digit 7 after 9 seconds, i.e. after a delay of individual P (i.e. 3+3+3 =9).

Send CLI in FROM field



- Click **Send CLI in FROM field** to expand.
- Select the CLI Option to be sent to the remote party, when outgoing calls are made using this SIP trunk. You may select any of the following **Send CLI Option**:
  - **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 (i.e. the number of the extension making the outgoing call through the SIP trunk) as CLI.

  The DDI number assigned to the calling extension will be sent, instead of its extension number.

  If the calling extension has disabled the parameter 'Send DDI as CLI', then the Pilot number configured in the DDI Routing table for this SIP Trunk will be sent as CLI.

  If calling extension has enabled CLIR, no CLI will be sent by this 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 want 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.

  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 Option 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.

- 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.

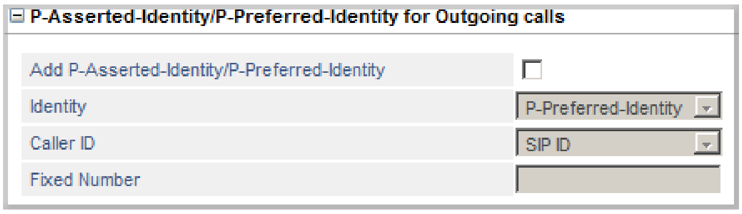
  - If you select *To, Request URI*, then the called party number will be sent in both the fields.

    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.

P-Asserted-Identity/P-Preferred-Identity for Outgoing calls

- Click **P-Asserted-Identity/P-Preferred-Identity for Outgoing calls** to expand.



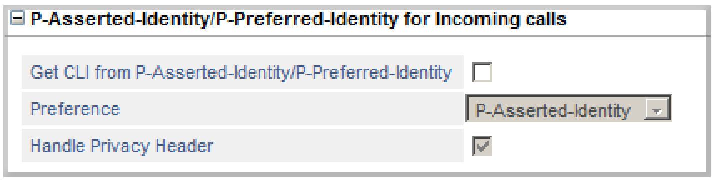
- Configure the following **P-Asserted-Identity/P-Preferred-Identity for Outgoing calls** parameters as provided to you by your Service Provider.
  - Select **Add P-Asserted-Identity/P-Preferred-Identity** check box, if supported by your Service Provider.
  - Select **P-Preferred-Identity** or **P-Asserted-Identity** as **Identity**.
  - Select the desired **Caller ID** option from the following:
    - **SIP ID:** Select this option, if you want the SIP ID configured on the SIP Trunk to be displayed as the Caller ID.
    - **Calling Party Wise:** Select this option, 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:** Select this option, 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 (+).

      Default: SIP ID.

P-Asserted-Identity/P-Preferred-Identity for Incoming calls

- Click **P-Asserted-Identity/P-Preferred-Identity for Incoming calls** to expand.



- 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.

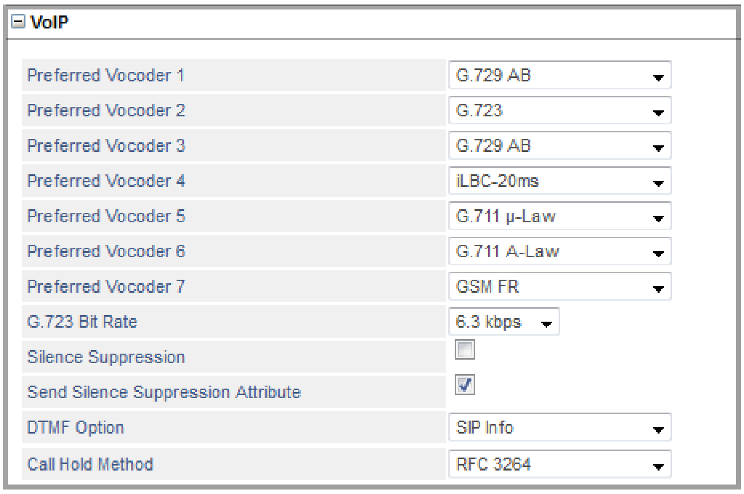
- Select **P-Preferred-Identity** or **P-Asserted-Identity** as the **Preference**.
- By default **Handle Privacy Header** is enabled. If the system receives Privacy Header as:
  - **Privacy: ID**, anonymous will be displayed as the CLI to the called extension.
  - **Privacy: None**, the CLI received 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.

VoIP

- Click **VoIP** to expand.

Vocoders



- Select **Vocoders**[81] in the order of preference from the drop down box.
  The Vocoders supported by SARVAM UCS in the order of preference, i.e. 1st to 7th, by default are:
  - G.723
  - G.729 AB
  - GSM FR
  - iLBC - 30 ms
  - iLBC - 20 ms
  - G.711 µ – Law
  - G.711 A - Law

If you do not want to select any Vocoder, you can select the option 'None'. However, if all Vocoder Preferences from 1 to 7 are set to 'None', incoming and outgoing calls will be blocked.

- If you have selected G.723 codec as a Preferred Vocoder, select **G.723 Bit Rate** as: Bit Rate: 5.3 Kbps or 6.3 Kbps. Default: 6.3kpbs.

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81. *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.*

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.

- Select the check box **Silence Suppression** to suppress the 'Silence' packets, and to allow only the Voice packets through. SARVAM UCS supports Silence Suppression for all Vocoders except GSM FR. Default: Disabled.

- Select the check box **Send Silence Suppression Attribute**, if you want the system to 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 this check box.By default it is enabled.

Silence Suppression Attribute is not dependent on the Silence Suppression option set.

- 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.

  SARVAM UCS supports the DTMF options: **RTP (RFC 2833)**, **SIP Info**, and **In-Band**. Select the appropriate option. Default: RTP (RFC 2833).

- 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.

Gain Settings

To avoid noise or echo during speech on SIP Trunks, you must set the speech volume levels (Tx and Rx Gain) on SIP Trunk with reference to other ports.

- **SIP - System (Tx-Gain and Rx-Gain)**: Configure the Gain Settings that you want the system to apply on the SIP Trunk with respect to the system (for example Call Conference). These will not be applicable for any other port type. Valid Range for Tx Gain: -31dB to +31dB and Rx Gain -31dB to +31dB. Default: 0dB.

- **SIP - Voice Mail (Tx-Gain and Rx-Gain):** Configure the Gain Setting that you want the system to apply on the SIP Trunk when the incoming calls on the SIP Trunk are being answered by the VMS. Valid Range for Tx Gain: -31dB to +31dB and Rx Gain -31dB to +31dB. Default: 0dB.

- **SIP - Voice Module (Tx-Gain and Rx-Gain):** Configure the Gain Settings that you want the system to apply on the SIP Trunk when incoming calls are answered using Auto Attendant or Trunk Auto Answer. Valid Range for Tx Gain: -31dB to +31dB and Rx Gain -31dB to +31dB. Default: 0dB.

- **SIP - Call Progress Tones (Tx-Gain and Rx-Gain):** Configure the Gain Setting that you want the system to apply on the SIP Trunk while playing Call Progress Tones. Valid Range for Tx Gain: -31dB to +31dB and Rx Gain -31dB to +31dB. Default: 0dB.

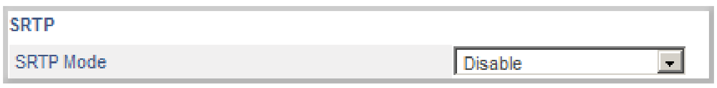
- **SIP - SLT (Tx-Gain and Rx-Gain)**: Configure the Gain Setting that you want the system to apply on the SIP Trunk when the SIP Trunk is connected to any FXS Port of the system during an incoming or outgoing call. Valid Range for Tx Gain: -31dB to +31dB and Rx Gain -31dB to +31dB. Default: 0dB.

- **SIP - CO (Tx-Gain and Rx-Gain)**: Configure the Gain setting that you want the system to apply on the SIP Trunk when the SIP Trunk is connected to the CO Port of the system during an incoming or outgoing call. Valid Range for Tx Gain: -31dB to +31dB and Rx Gain -31dB to +31dB. Default: 0dB.

- **SIP - SIP (Tx-Gain and Rx-Gain)**: Configure the Gain Setting that you want the system to apply on the SIP Trunk when the SIP Trunk is connected to any other SIP Trunk or SIP Extension of the system during an incoming or outgoing call. Valid Range for Tx Gain: -31dB to +31dB and Rx Gain -31dB to +31dB. Default: 0dB.

- **SIP - Mobile (Tx-Gain and Rx-Gain)**: Configure the Gain Setting that you want the system to apply on the SIP Trunk when the SIP Trunk is connected to any Mobile Port of the system during an incoming or outgoing call. Valid Range for Tx Gain: -31dB to +31dB and Rx Gain -31dB to +31dB. Default: 0dB.

SRTP



SARVAM UCS supports **SRTP** (Secure Real Time Protocol) for secure conversations over SIP. The VoIP module of SARVAM UCS supports the following options:

- **Disable:** Select **Disable** if you want SARVAM UCS to use normal RTP for transporting the speech packets.

- **Optional**: Select **Optional** if you want SARVAM UCS to use 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**. Default: AVP.

- **Forced:** Select **Forced** if you want SARVAM UCS to use 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.

Echo Cancellation

SARVAM UCS supports Echo Cancellation for SIP to CO trunk calls and SIP to Digital Trunks (Mobile, SIP) and Extensions (SIP). If you want to apply Echo Cancellation,

- Select the **Echo Cancellation** check box. Default: Enabled.

- Set the tail length for echo cancellation for SIP to CO trunks as **Tail Length for CO (msec)**. The echo cancellation Tail Length for SIP to CO trunks can be 32/64/128 milliseconds. Default: 128 milliseconds.

- Set the tail length for echo cancellation for SIP to Mobile/SIP trunks and SIP to SIP extensions as **Tail Length (msec) for Extensions/Digital Trunks (msec).** The echo cancellation Tail Length for SIP to Digital Trunks/Extensions can be 32/64/128 milliseconds. Default: 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.

- Select the Jitter Buffer **Type** as **Static** or **Dynamic**.

  If you select **Static** Jitter Buffer, configure the '**Minimum Delay**'. The value configured in the Minimum Delay determines the size of the Static Jitter buffer.

  If you select **Dynamic** Jitter Buffer, configure the '**Optimization Factor**' and '**Minimum Delay**'. The minimum size of the Dynamic Jitter buffer depends on the 'Minimum Delay' configured. The Optimization Factor determines the rate of adaptation of the Dynamic Jitter Buffer to the jitter in the network.

- Set the **Minimum Delay (msec)** (for both Static and Dynamic Jitter Buffer) to the desired value from 10 to 280 milliseconds. Default: 10.

The Minimum Delay determines the size of the Static Jitter Buffer. 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 packet for the configured time 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.

- Set the **Optimization Factor** (only if you selected Dynamic Jitter Buffer) to the desired value from 01 to 13. Default: 10.

  In networks with higher jitter, a higher value should be configured as Optimization Factor.

  The actual size of the Dynamic Jitter Buffer will be determined by the DSP on the basis of the Optimization Factor configured and actual network condition. Dynamic Jitter buffer can go up to maximum 300 milliseconds.

Fax over IP

Fax Over IP	
Fax Type	T.38 (UDPTL)

- You can send/receive Fax over IP (FoIP) from a Fax machine connected to the SLT port of the SARVAM UCS. Select as **Fax Type**, the protocol of Fax over IP.

  The SARVAM UCS supports the fax options: **T.38 (UDPTL)**, **T.38 (RTP)**, and **Pass Through**. Default: T.38 (UDPTL)

- *'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

T.38 Fax Parameters	
Fax Version	0
Max Rate (kbps)	14.4
Packet Period (msec)	40
Image Redundancy Level	1
Data Redundancy Level	3

If you selected T.38 protocol as Fax Type, you may configure Eye Quality Monitor (EQM) related parameters to improve quality of Fax. The higher the EQM, the better the Fax quality. To improve the quality of T.38 fax reception,

- Select the T.38 **Fax Version** for the desired SIP Trunk as per your requirement. Default: 0. You can configure the Fax Version from 0 to 3.

- Set the **Max Rate (Kbps)** to the desired speed. Default: 14.4 kbps.

  The Max Rate controls the Fax image transfer speed. As EQM is inversely proportional to fax Max Rate, if you receive poor quality fax, the Max Rate should be reduced.

- Set the **Packet Period (msec)** to the desired value. Default: 40msec.

  The packet period sets the sampling rate of TDM signal. If you cannot improve fax quality by lowering Fax Max Rate, you may reduce the Fax Packet Period.

  When you reduce the Fax Packet Period, the fax image will be sent at lower speed. EQM is inversely proportional to Fax Packet Period.

  Do not change the default settings unless it is required.

- Set the **Image Redundancy Level** to the desired value. The Fax Image Level is redundancy level for output Image, which can be from 0 to 3. Default: 1.

  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.

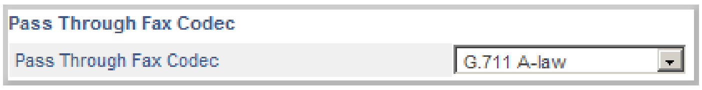
  You may increase the Fax Image Level from 1 to 3 if the quality of fax does not improve with Fax Max Rate and Fax Packet Period.

  Level 0 means no redundancy.

- Set the **Data Redundancy Level** to the desired value. Default: 3. This is a redundancy level for T.30 control data. Fax Data Level can be set from 0 to 7. Level 0 means no redundancy. Redundancy level increases from 1 towards 7. The higher the level set, 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.

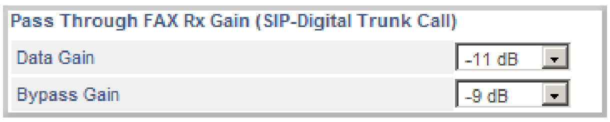
Pass Through Fax Codec



- 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 m-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.

Pass Through FAX RX Gain (SIP to Digital Trunk calls)

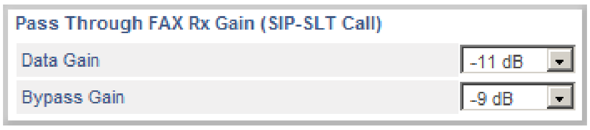


Configure this parameter if you selected Pass Through[82] as Fax Type, and the fax is to be received on a digital trunk.

- Select the dB level for **Data Gain (dB)** from -31 to 31 dB. Default: -11 dB.

- Select the dB Level for **Bypass Gain (dB)** from -31 to 31 dB. Default: -9 dB.

Pass Through FAX Rx Gain (SIP to SLT calls)

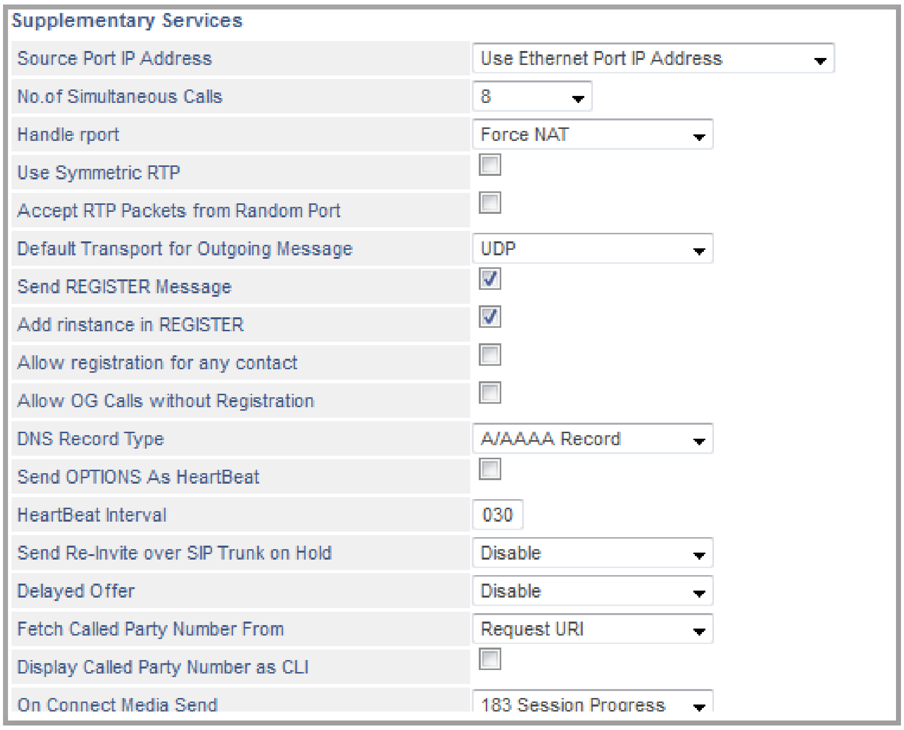


Configure this parameter if Pass Through Fax is to be received on a fax machine connected to a SLT port.

- Select the dB level for **Data Gain (dB)** from -31 to 31 dB. Default: -11 dB.

- Select the dB level for **Bypass Gain (dB)** from -31 to 31 dB. Default: -9 dB.

Supplementary Services



- In **Source Port IP Address** list, select the Source Port IP Address according to your VoIP installation scenario.

  - Select **Use Ethernet Port IP Address**, if SARVAM UCS is connected directly to the public internet.

  - Select **Use IP Address Fetched Using STUN**, if SARVAM UCS is located behind a NAT router other than Symmetric.

    If you select this option, keep the **Outbound Proxy** disabled.

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82. *Pass Through Fax packets are transported using RTP protocol. Normally, fax calls require less gain compared to voice calls. However, to improve fax reception, SARVAM UCS enables you to configure gain settings for fax. Fax gain settings consist of Data Gain and Bypass Gain. SARVAM UCS supports Fax Receive Gain for SIP to Digital Trunk calls as well as SIP to SLT Calls.*

- Select **Use Router's Public IP Address**, if SARVAM UCS is located behind a NAT Router (any type).

  If you select this option, keep the **Outbound Proxy** disabled.

- In the **No. of Simultaneous Calls**[83] list, select the number of simultaneous calls you want to allow on this SIP Trunk. Default: 8.

  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.

  Configure this parameter only if the ITSP supports less than 8 simultaneous calls.

- In **Handle report**, select from the following options:
  - **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

- 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.

  Select the **Use Symmetric RTP** check box, if the SARVAM UCS 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 **Accept RTP Packets from Random Port** is disabled, that is, the System will accept RTP packets from the negotiated RTP port only.

  Enable this check box, 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 check box in PCAP is disabled. For details, see "PCAP Trace".*
    - *RTP Mode is set as Transcoding or RTP Relay. For details, see "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.*

- Select the **Default Transport for Outgoing Messages** from the following options:
  - **UDP:** Outgoing messages are transported using UDP.
  - **TCP:** Outgoing messages are transported using TCP.
  - **TLS**: Outgoing messages are transported using TLS. If you select this option, you must enable **SIP Over TLS** on the "VoIP Parameters" page.

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83. *This parameter is only applicable if you have set the RTP Mode as Transcoding Mode in "VoIP Parameters".*

Default: UDP.

- *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.*

- Select whether or not the system should **Send REGISTER Message** from the SIP trunk. Default: Send REGISTER Message is enabled.

  Clear the check box if you do not want REGISTER Message to be sent.

- **'rinstance'** is any random value which can be used by the Ethernet Port to fetch its own contact binding, i.e. to know the Registration Expiry Timer assigned by the server.
  Select if the system should **Add 'rinstance' in REGISTER** message. Default: 'rinstance' is enabled.

  If you want to remove 'rinstance' in REGISTER, clear the check box.

  By default, the system does not allow outgoing calls to be made if the status of the SIP trunk is 'not registered'.

- Enable **Allow registration for any contact** check box, if you wish to register the SIP trunk with an ITSP. In this case even if you do not receive proper contact header from the ITSP, the SIP trunk will get registered.

- To allow outgoing calls to be made from the SIP Trunk even when it is not registered, select the **Allow OG Calls without Registration** check box. Default: Disabled.

- Set **DNS Record Type** 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.

  - **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.

    Default: A/AAAA Record

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.*

- You can select whether or not the system should send the OPTIONS message periodically to the Proxy Server to monitor its availability. Select the **Send OPTIONS as Heartbeat** check box to send the OPTIONS message. 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.

- 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.

- 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 **Send Re-INVITE over SIP Trunk on Hold** 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”.

- Enable the **Delayed Offer** check box, if you want the SIP Trunk to generate INVITE without SDP. Default: Disabled.

- SARVAM UCS extensions may be assigned DDI numbers 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. Set the **Fetch Called Party Number From** option as per the requirement of your ITSP.

  Default: Request URI.

- 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 an Extended IP Phone or a Mobile Softphone 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 configured 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.

- If Built-In Auto Attendant or DISA is enabled on the SIP Trunk, select the response you want the system to send in **On Connect Media send**, that is 200 OK or 183 Session Progress.

- 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 in **Answer Source Trunk on Receiving**. You can select **Early Media** or **200 OK**.

- If your ITSP requires “tel” in the URI so that it can handle/ route calls to/from global numbers, in **Use “tel” URI type in** set the option as required by your ITSP from the following:
  - TO, Request URI
  - FROM
  - TO, Request URI and FROM

  Default: None.

  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.

- Select **Send "user=phone"** 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.

- Select the **Include ptime header in SDP** check box, if you want to add ptime header in SDP offer and answer. Default: Disabled.

- When an incoming call is diverted/forwarded to an external number using the same SIP Trunk, configure the following **Redirection** 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** checkbox. Default: Disabled.

  - If you have selected **Generate new Call** as the **Call Redirection Type**, you may select the **Include "Diversion" while diverting the call** checkbox. 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** checkbox. 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's Identity and if you select **Received** the system will send the received CLI to the target.

- 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:

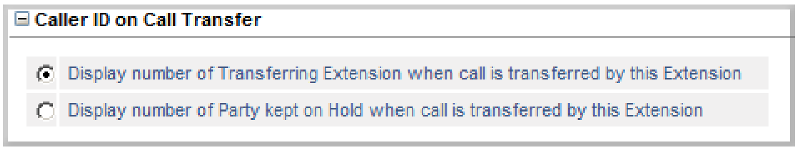
- **System:** In this option, the SARVAM UCS will handle the Call Transfer locally.
- **Network:** In this option, the Call Transfer is handled by the Network.

Default: System.

If you have configured the above parameters, click **Submit** at the bottom of the page to save your settings.

Caller ID on Call Transfer

If you set the SIP Trunk mode as **Peer-to-Peer** and selected **Treat Incoming Call** as **Station**, you may configure Caller ID on Call Transfer.



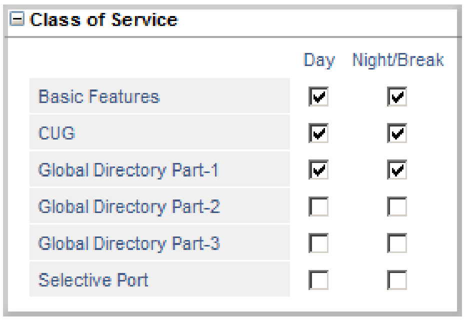
You can choose whether the system should display the CLI of the ‘Held Party’ or the CLI of the ‘Transferring Party’ to the transfer destination extension while the call is being transferred.

See the feature description for “Calling Line Identification and Presentation (CLIP)” to know more.

- Click **Caller ID on Call Transfer** to expand.
- Select the radio button of the desired option:
  - **Display number of Transferring Extension when call is transferred by this Extension.**
  - **Display number of Party kept on Hold when call is transferred by this Extension**

Class of Service

If you set the SIP Trunk mode as **Peer-to-Peer** and selected **Treat Incoming Call** as **Station**, configure class of service.



- Click **Class of Service** to expand options.

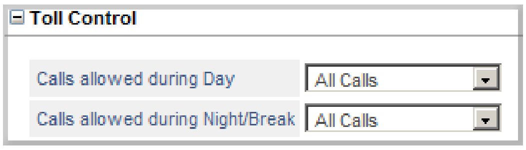
  Define the Class of Service for the SLT extension for **Day** time and **Night/Break** time.

- To allow a feature, select the check box of the feature.

- To deny a feature, clear the check box.

Toll Control

If you set the SIP Trunk mode as **Peer-to-Peer** and selected **Treat Incoming Call** as **Station**, configure Toll Control.



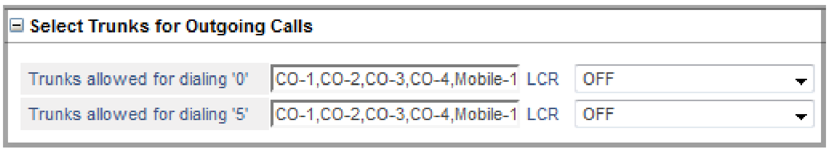
- Click **Toll Control** to expand options. Set the desired Toll Control for the SIP trunk for the Day and Night/ Break.
- Select the type of **Calls Allowed during Day**: All Calls, No Calls, Local Calls, Regional Calls, National Calls, International Calls, Limited Calls 1, Limited Calls 2, Limited Calls 3. Default: No Calls
- Select the type of **Calls Allowed during Night/Break**: All Calls, No Calls, Local Calls, Regional Calls, National Calls, International Calls, Limited Calls 1, Limited Calls 2, Limited Calls 3. Default: No Calls.

You can configure the Toll Control - Allowed-Denied Numbers for each type. To do this,

- Click the **Settings** icon. This opens **Toll Control - Allowed-Denied Numbers** page in a new window. This icon does not appear when Toll Control is selected as **All Calls** or **No Calls**.

Select Trunks for Outgoing Calls

If you set the SIP Trunk mode as **Peer-to-Peer** and selected **Treat Incoming Call** as **Station**, select the Trunks for Outgoing Calls.



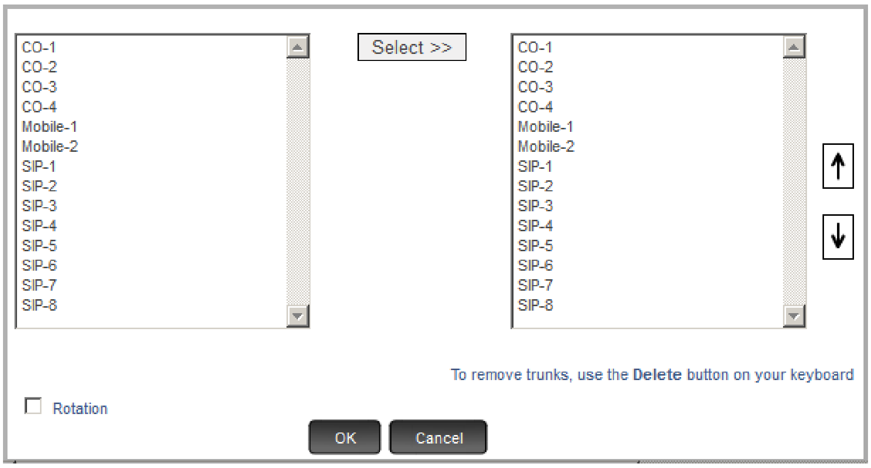
Outgoing calls (to external numbers) are made by dialing Trunk Access Codes (TAC). Default: 0 (TAC 1) and 5 (TAC 2). You may configure TAC 3 to TAC 6 as desired.

For each TAC, you need to select the Outgoing Trunks. All external calls made by dialing a particular TAC will be routed through the outgoing trunks you selected for that TAC.

You can also apply Least Cost Routing logic on the selected trunks, so that SARVAM UCS routes the outgoing call through the trunk that costs the lowest for the call.

- Click **Select Trunks for Outgoing Calls** to expand.
- Select **Trunks allowed for dialing '0'**. The outgoing call will be routed through the selected trunks when the extension user dials TAC '0'.

- Double-click the field. A multiple selection box opens.



On the left, the trunks appear with their names (if configured in "Trunks") and port numbers in a sequence, starting with CO trunks, followed by Mobile trunks and SIP trunks.

If you have not assigned any names to the trunks, they will appear with their default names (CO, MOB, SIP) and port numbers.

If you have enabled On-Site Configuration, only those trunks that are connected will appear in the box.

- To select a trunk, place your cursor on the desired trunk, and click **Select>>**.

  Or

- Press the **ctrl** key and click the left mouse button to select multiple trunks.
- You may change the sequence of the trunks you selected, if required, using the **Up** and **Down** arrow buttons on the right display box.
- You can also delete trunks from the ones you have selected by pressing **Delete** key on your keyboard.
- You may enable **Rotation**, if you have selected more than one trunk. Default: Disabled.

  When you enable Rotation, each new outgoing call is routed through the subsequent trunk in the group[84]. This ensures equal distribution of outgoing call traffic on all trunks.

  When Rotation is disabled, calls are routed through the first trunk in the group. If this trunk is busy, the call is routed to the next trunk in the group.

  Rotation has no relevance if only one member trunk is selected.

- Click **OK**. The multiple selection box closes.

  All the trunks appear in Trunks allowed for dialing '0', in the sequence you selected, separated by commas. For example, BSNL (CO-1), BSNL (CO-3), Reliance (CO-4) and Pulver (SIP1).

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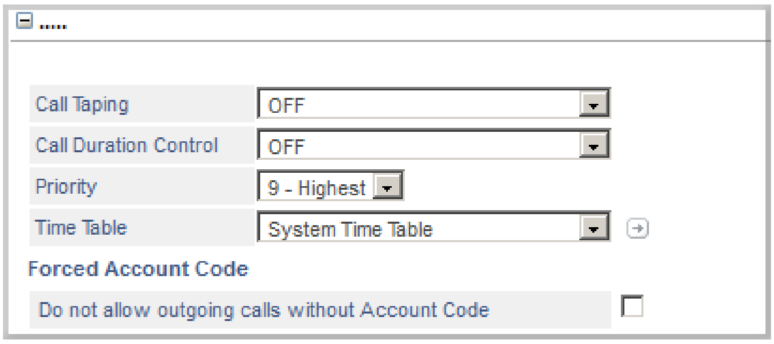
84. *The first call through the first trunk, the second through the second, the third through the third trunk, and so forth. Thus each new call is routed through the trunk next to the one that routed the previous outgoing call.*

- To apply **Least Cost Routing** on the Trunks allowed for 0 dialing, select the desired LCR method from the combo box:
  - **Number Based:** Choose this option if the service providers of the trunks you selected offer different tariffs according to area or distance, or phone numbers dialed.
  - **Time Based:** Choose this option if the service providers of the selected trunks offer a different tariff according to the time of the day.
  - **Time + Number Based:** Choose option if the service providers of the selected trunks offer different tariffs according to the time of the day as well as area/distance/phone number.
  - **Service Provider Based:** Choose this option if the same service providers of the selected trunks offer different rates for calls made to numbers within their own network and for calls made to numbers of another Service Provider's network.
  - If you do not want to apply LCR, select OFF. The **Settings** icon does not appear if the LCR is OFF.
  - Configure LCR method that you selected for the trunk group.
- To configure Least Cost Routing method you selected, click the **Settings** icon. A new window opens.
  - Configure the LCR method. See "Least Cost Routing (LCR)" under Advanced Settings for instructions.
  - Click **Submit**.
- Select **Trunks allowed for dialing '5'**. The outgoing call will be routed through the selected trunks when the extension user dials TAC '5'. Follow the same steps as described above.
  - Double-click the field. A multiple selection box opens.
  - Select trunks, placing your cursor on the desired trunk, and clicking **Select>>**.
  - Change the sequence of the trunks you selected, if required, using the **Up** and **Down** arrow buttons on the right display box. Delete trunks from the ones you have selected, if required.
  - Enable **Rotation**, if you have selected more than one trunk. Default: Disabled.
  - Click **OK**.

    All the trunks appear in Trunks allowed for dialing '5', in the sequence you selected, separated by commas.
  - Choose a **Least Cost Routing** method, if you want to apply it on the trunks. Default: OFF.
  - Configure the Least Cost Routing method you selected by clicking the **Settings** icon. See "Least Cost Routing (LCR)" under Advanced Settings for instructions.

  Similarly, you will be able to select outgoing trunks for Trunk Access Code 3, 4, 5 and 6, if you have assigned access codes to them in "Extension and Feature Codes".
- Click **Submit** to save the settings.

More Features



Call Taping

If you have set the SIP Trunk Mode to Peer-to-Peer and Treat incoming call as Station, you may also enable Call Taping.

- Click **Call Taping** to expand options.

- Select the type of **Call Taping** to be applied from the options:
  - OFF
  - Apply as per profile 1
  - Apply as per profile 2
  - Apply as per profile 3
  - Apply as per profile 4

  If you do not want to apply Call Taping, select OFF.

  Default: OFF.

- You can configure the Call Taping Profile number you selected for the extension. To do this,

  - Click the **Settings** icon. The page opens in a new window, displaying the parameters of the profile you selected for the extension. This icon does not appear when Call Taping is **OFF**.

  - In the new window,

    - Select the **Tape Internal Calls** check box, if you want internal calls made and received by the extension to be taped.

    - Select the **Tape Incoming Calls received without CLI** check box if you want incoming calls without CLI to be taped. Default: Disabled.

    - In **Tape Incoming Calls received from following numbers**, type the numbers of the Incoming Calls that must be taped. You may type as many as 99 numbers.

    - In **Tape Outgoing Calls made to following numbers,** type the external numbers that you want the system to tape. You may type as many as 99 numbers.

    - Click **Submit** to save the Profile.

See "Call Taping" to know more.

Call Duration Control

If you have set the SIP Trunk Mode to Peer-to-Peer and Treat incoming call as Station, you may also enable Call Duration Control.

- Click **Call Duration Control** to expand options.
- Select the type of **Call Duration Control** to be applied from the options:
  - OFF
  - Apply as per CDC profile 1
  - Apply as per CDC profile 2
  - Apply as per CDC profile 3
  - Apply as per CDC profile 4

  If you do not want to apply Call Duration Control, select OFF. Default: Apply as per CDC profile 1.
- You can configure the Call Duration Control Profile number you selected for the extension. To do this,
  - Click the **Settings** icon. The page opens in a new window. The page displays the parameters of the profile you selected for the extension. This icon does not appear when Call Duration Control is **OFF**.
  - In the new window,
  - Enable **Apply CDC to Internal Calls,** if CDC is to be applied on internal calls. Default: Enabled.
  - Enable **Apply CDC for Incoming Calls received from trunk**, if CDC is to be applied on incoming external calls. Default: Enabled.
  - Enable **Apply CDC for Outgoing Calls made from trunk**, if CDC is to be applied to outgoing external calls. Default: Enabled.
  - If required, change the **CDC Timer** to the desired duration. The range of the timer is 0001 to 9999 seconds. Default: 160 seconds.
  - Enable **Disconnect Call after CDC Timer** check box if you want calls to be disconnected on the expiry of the CDC Timer. Default: Enabled.
  - In the **Apply CDC for calls matching with numbers** column, type the external numbers on which you want to apply CDC. You can enter as many as 99 numbers.
  - In the **Do Not Apply CDC for calls matching with numbers** column, enter the numbers which you want to be exempt from CDC.
  - Click **Submit**.

Priority

- Select a **Priority** level from 1 to 9 for the trunk. Default: 9-Highest.

  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, see "Priority".

Time Table

- Select a **Time Table** for the SIP trunk. Default: System Time Table.

If you have not configured Time Table, you may do so now,
- Clicking the Settings icon.
- The Time Table page opens.
- Define the Working days, and the start and end time of the Working hours and Break hours for each working day.
- Click **Submit**.
- Close the window.

See “Time Tables” to know more.

Forced Account Code

Configure this parameter if you want to apply the Forced Account Code feature on the SIP trunk. See “Account Codes” to know more about this feature.

- Select the **Do not allow Outgoing Calls without Account Code** check box to enable Forced Account code on the trunk. Default: Disabled.

  When you enable this check box, the system will prompt extension users to dial the Account Code whenever they grab this trunk to dial out a number. The system will allow extension users to dial out numbers only after they have dialed the Account Code or Name.

Account Codes feature must also be enabled in the Class of Service of extension users who are to be allowed this feature.

Copy Parameter Values

After you have finished configuring this SIP Trunk, you may configure the next SIP Trunk. To do so,

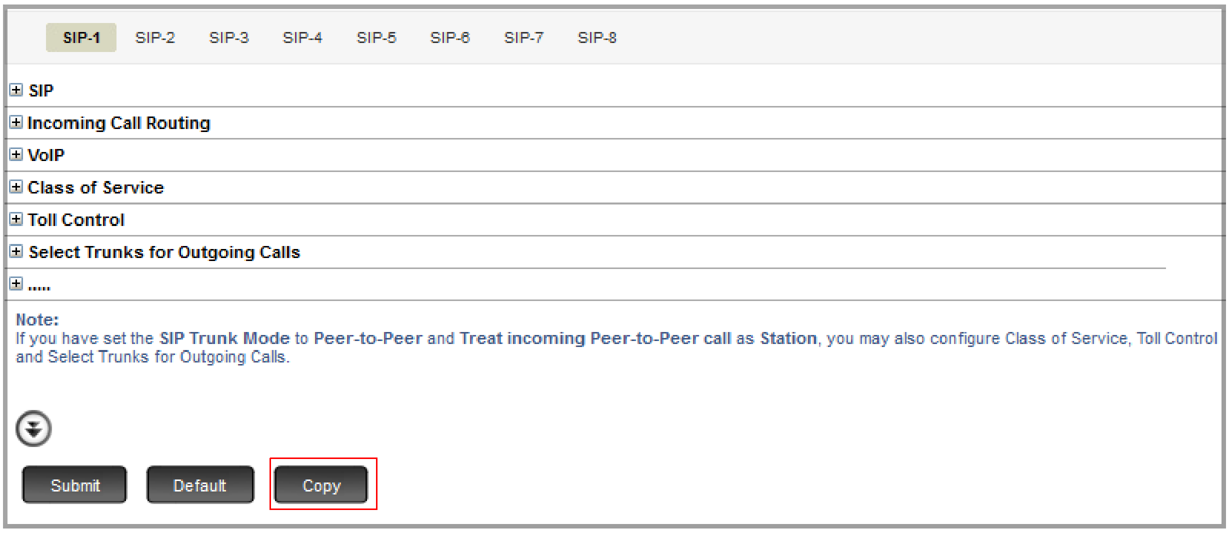
- Click the tab of the next SIP Trunk.

- Follow the same steps as described above to configure another SIP Trunk.

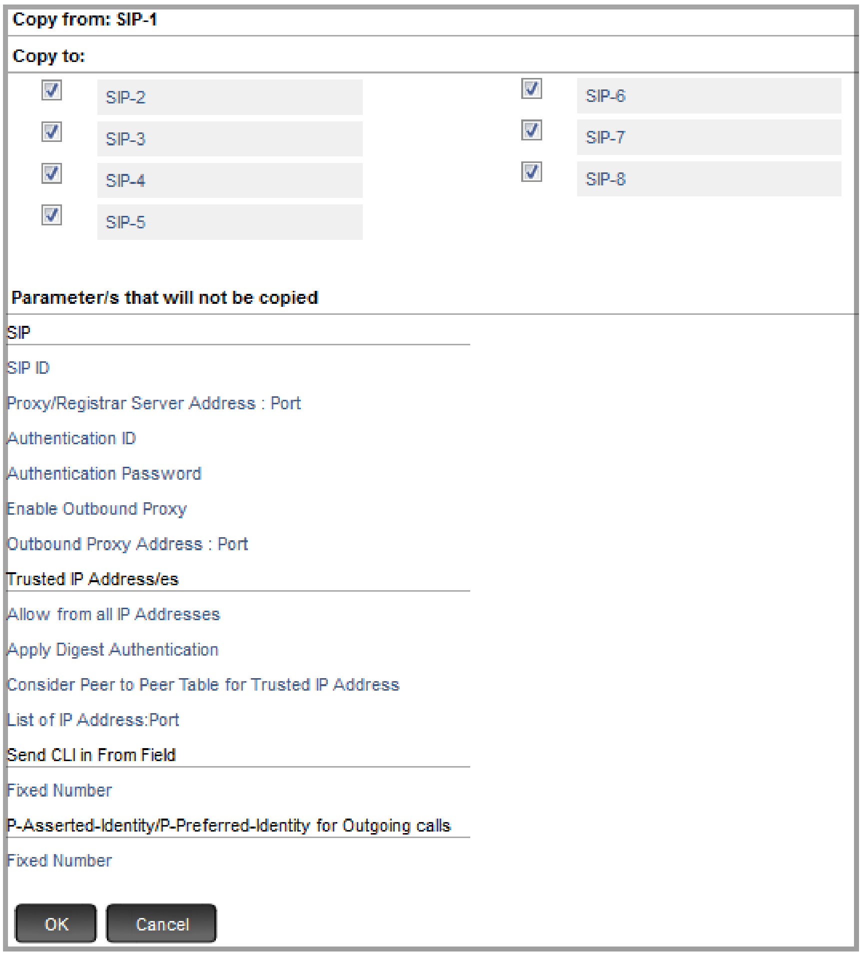
OR

You may use the **Copy** button at the bottom of the page to apply the same SIP Trunk settings you configured for the current trunk to another SIP Trunk. To do so,

- Click **Copy**.



The Copy page opens in another tab.



The copy page displays the number and/or name of the current trunk in **Copy from:**. It also lists the numbers and/or names of the trunks to which you can copy the trunk settings under **Copy To:**.

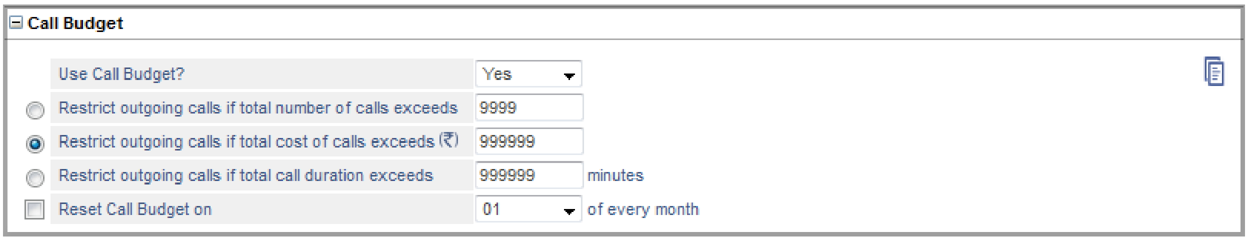
- Select the respective **Copy to** check boxes of the SIP Trunks to which you want to apply the current SIP Trunk settings.

- Click **OK**.

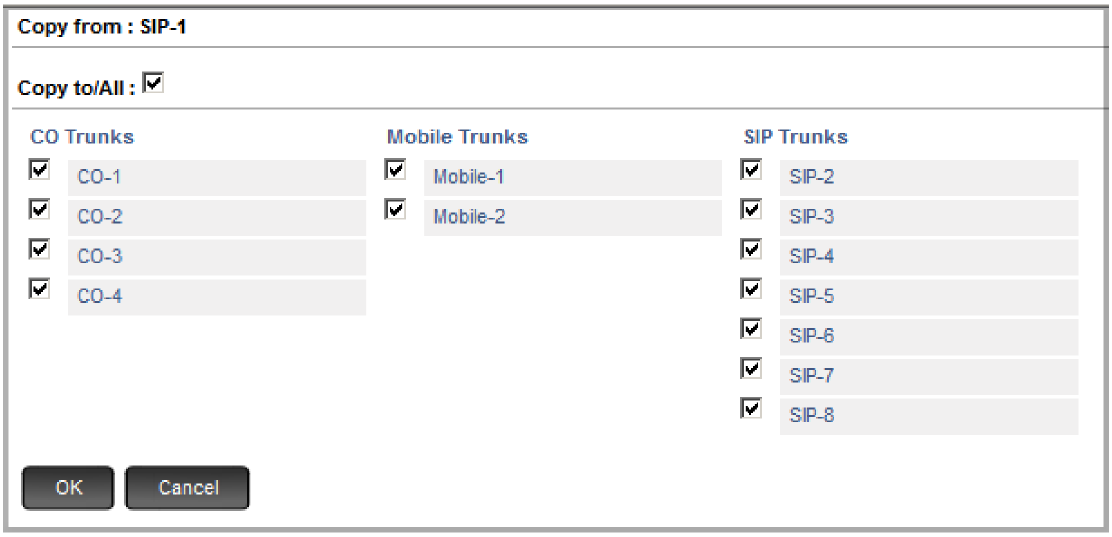
All the parameter values of the current SIP Trunk except those listed below **Parameter/s that will not be copied**, will be applied to the selected SIP Trunk.

You may also copy the parameter values of a specific link under current SIP Trunk to the same link under any other trunk page. To do so,

- Click the **Copy** icon besides the desired link.



The Copy page opens in another tab.



The Copy page displays the number and/or name of the current trunk in **Copy from:**. It also lists the numbers and/or names of the trunks to which you can copy the trunk settings under **Copy To/All:**.

- Select the **Copy to/All** check box, to apply the link settings to the same link under all other Trunks.
  OR
  You may also apply the link settings to the desired Trunks only, by selecting the respective check boxes.

- Click **OK**.

DDI Routing

Direct Dial-In (DDI) is a service feature provided by ITSPs over SIP Trunks to its customers. These are additional numbers associated with the main number (Pilot Number).

This is useful in organizations where the System is connected and each extension user needs to be assigned a separate direct number. When there is an incoming call on a specific DDI number, it is routed directly to the extension user who is assigned the number without the intervention of the operator.

Before we understand how DDI routing works, we must understand the following terms:

- **Pilot Number:** It is a unique number assigned by the service provider to a SIP Trunk. The Pilot Number can be a maximum of 16 digits.

- **DDI Numbers:** These are additional numbers allotted by the Service Provider. Each extension user can be assigned a separate DDI number.

To use this feature make sure you have:

- Configured the Pilot Numbers for each SIP Trunk
- Assigned DDI Numbers to desired extension users.

Let us understand this with the help of an example:

- SIP Trunk1 is connected to SARVAM UCS.

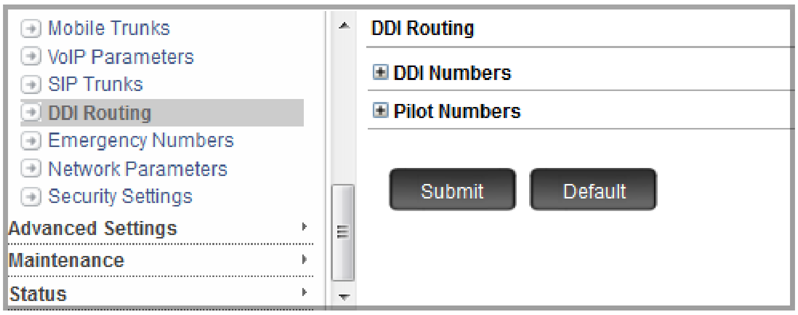
- The Pilot Number 2630555 is assigned by the service provider along with 50 DDI numbers.

- Extension User1 is assigned the DDI number 2630556 and Extension User2 is assigned the DDI number 2630557. When there is an incoming call on 2630556 it is routed to Extension User1, similarly calls on 2630557 are routed to Extension User2 directly.

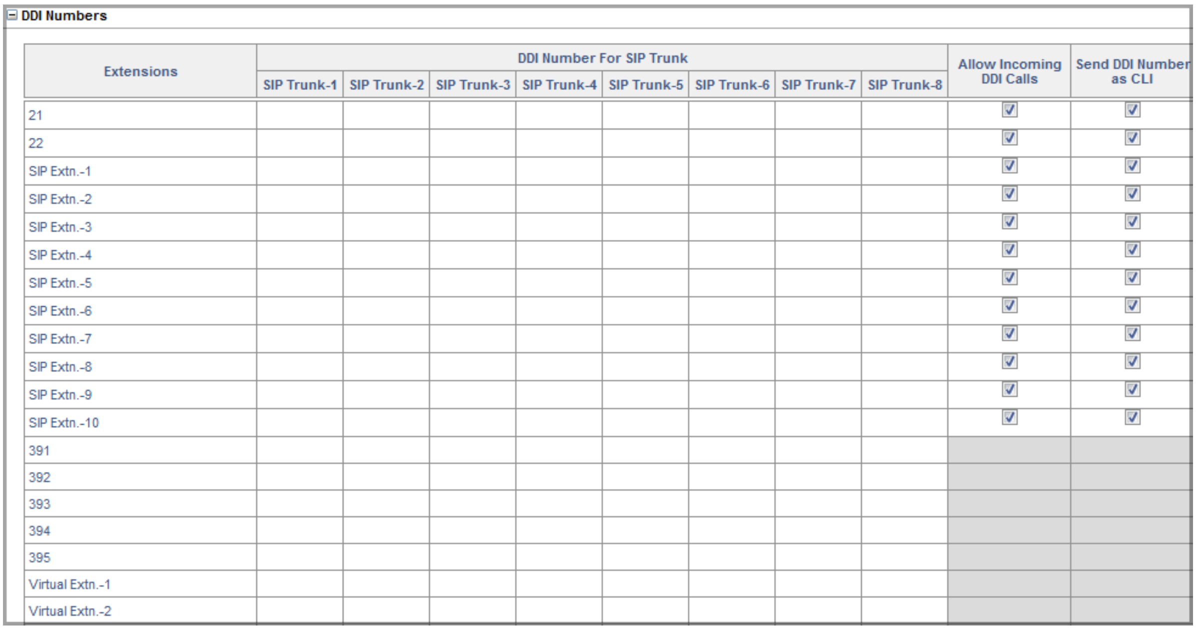
- If outgoing calls are made by Extension User1 then the CLI sent to the called party will be the DDI number assigned to Extension User1, if **Send DDI Number as CLI** option is enabled. If not then the Pilot number will be sent.

To configure,

- Click **DDI Routing**.



- Click **DDI Numbers** to expand.



DDI Numbers

Extensions	DDI Number For SIP Trunk								Allow Incoming DDI Calls	Send DDI Number as CLI
	SIP Trunk-1	SIP Trunk-2	SIP Trunk-3	SIP Trunk-4	SIP Trunk-5	SIP Trunk-6	SIP Trunk-7	SIP Trunk-8		
21									☑	☑
22									☑	☑
SIP Extn.-1									☑	☑
SIP Extn.-2									☑	☑
SIP Extn.-3									☑	☑
SIP Extn.-4									☑	☑
SIP Extn.-5									☑	☑
SIP Extn.-6									☑	☑
SIP Extn.-7									☑	☑
SIP Extn.-8									☑	☑
SIP Extn.-9									☑	☑
SIP Extn.-10									☑	☑
391										
392										
393										
394										
395										
Virtual Extn.-1										
Virtual Extn.-2										

- In **Extensions**, the access codes and the names of the Extension users as configured by you in the *Extn. & Feature Codes* page will be displayed.

- In **DDI Number for SIP Trunk**, for each SIP Trunk, enter the DDI Numbers provided by the ITSP of that trunk.

  You can assign these DDI numbers to SLT, SIP, Virtual extension users, Department Groups, VMAA.

  For example, in **SIP Trunk 1**, enter the DDI numbers among the DDI numbers provided by the ITSP with whom you have registered the SIP Trunk 1. Each number can be assigned to the desired SLT, SIP or Virtual extension users, Department Group, VMAA.

  Similarly, assign the DDI numbers for other SIP Trunks.

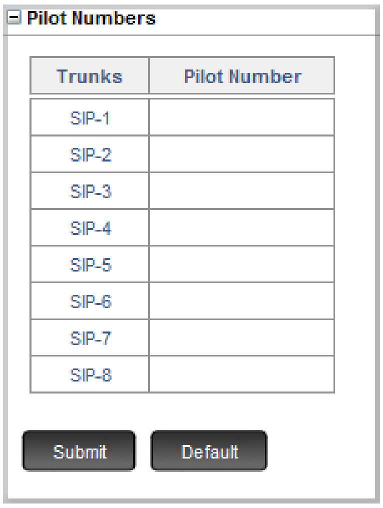
- Keep the **Allow Incoming DDI Calls** check box enabled, to allow the system to land incoming calls with the same DDI number on the extension user who is assigned this number. Default: Enabled.

  Clear the check box, if you want to disable.

- Keep the **Send DDI Number as CLI** check box enabled, to allow the system to send the DDI number as CLI when outgoing calls are made by the extension user who is assigned this number. Default: Enabled.

  If you clear the check box, the system will send the Pilot number of the respective SIP Trunk as CLI for outgoing calls made from this extension.

- Click **Pilot Numbers** to expand.



Enter the DDI **Pilot Number** provided by the ITSP with whom you have registered each SIP Trunk.

- Click **Submit**.

Emergency Numbers

- *This page displays the default Emergency Numbers for the 'Region' selected in the system. In other words, it displays the Emergency Numbers specific to the country/region where the ETERNITY NENX is installed.*

- *The default Emergency Numbers that appear on this page are non-editable. All you need to do is to select the Outgoing Trunks for each default Emergency Number.*

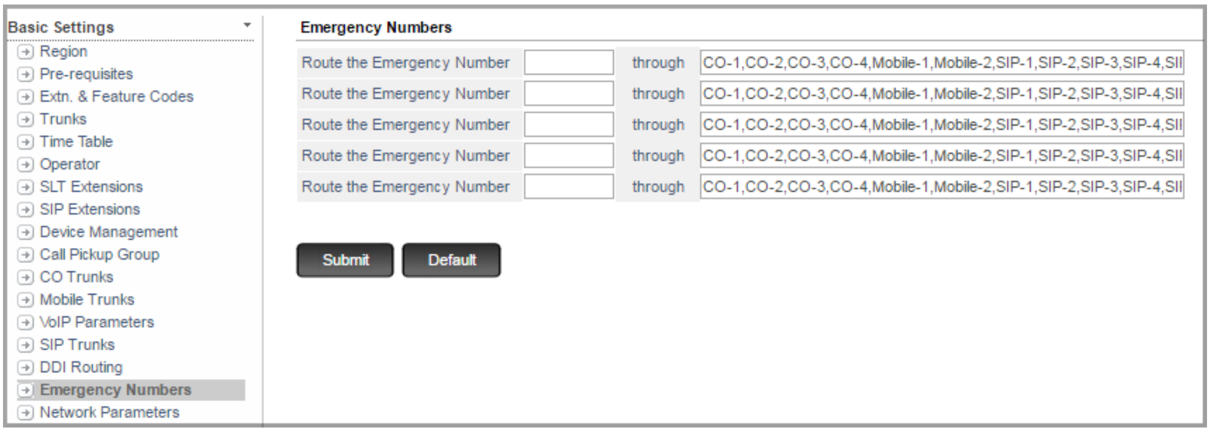
  For example, if '112' is the default Emergency Number for the Mobile network. So, you may select the Mobile Trunk for dialing this number.

- *You may add additional Emergency Numbers to the existing default numbers in the edit boxes and select the Outgoing Trunks for these numbers.*

- *If you upgrade the firmware make sure you click **Default**. The system will upload the default Emergency Numbers as per the Region.*

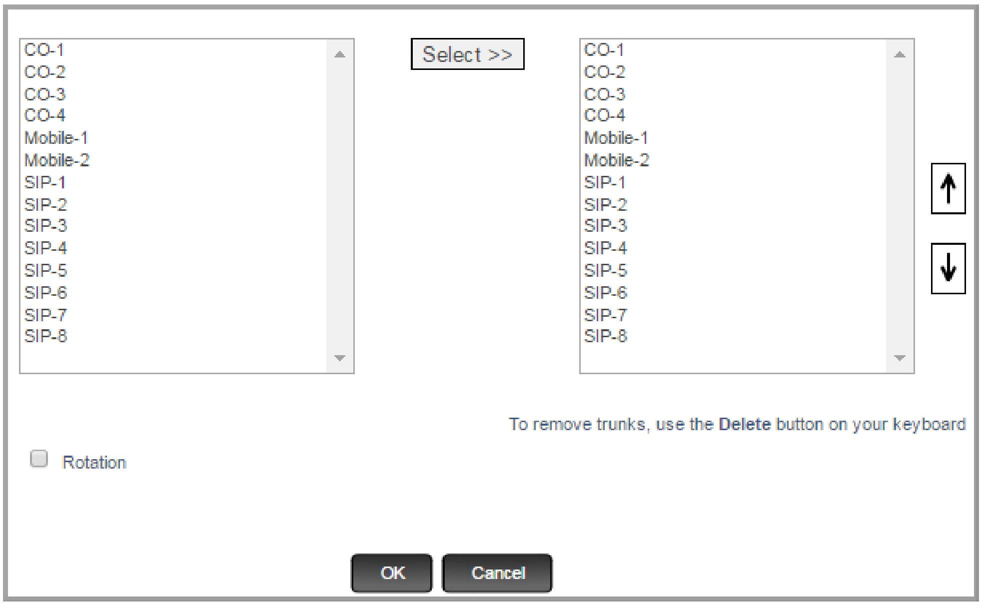
Disclaimer: *Matrix Comsec is not responsible for incorrect configuration of Emergency Numbers.*

- To add an Emergency Number, click the **Emergency Numbers**.



- Enter the desired number in the **Route the Emergency Number**.

- To select Outgoing Trunks to route the emergency number, double click the **Through** field.



A multiple selection window opens.

- On the left pane of this multiple selection window, the trunks[85] with their names (if configured in "Trunks") and port numbers appear in a sequence.

- To select a trunk, place your cursor on the desired trunk, and click **Select>>**.
  OR
  Press the **ctrl** key and click the left mouse button to select multiple trunks.

  Make sure that the trunks selected for each Emergency number route the Emergency call to the right network.

  To change the sequence of the trunks you selected, use the **Up** and **Down** arrow buttons on the right display box.

  To delete trunks from those you have selected, use **Delete** key on your keyboard.

- Select **Rotation**, if you have selected more than one trunk. Default: Disabled.

  When you enable Rotation, each new outgoing call is routed through the subsequent trunk in the group,[86] ensuring equal distribution of outgoing call traffic on all the trunks.

  When Rotation is OFF, calls are routed through the first trunk in the group. If this trunk is busy, the call is routed to the next trunk in the group.

  Rotation has no relevance if only one trunk is selected.

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85. *If you have not assigned names to the trunks, they will appear with their default names (CO, Mobile, SIP) and port numbers. If you have enabled On-Site Configuration, only those trunks that are connected will appear in the box.*

86. *The first call through the first trunk, the second through the second, the third through the third trunk, and so forth. Thus each new call is routed through the trunk next to the one that routed the previous outgoing call.*

- Click **OK** to save the selected trunks.

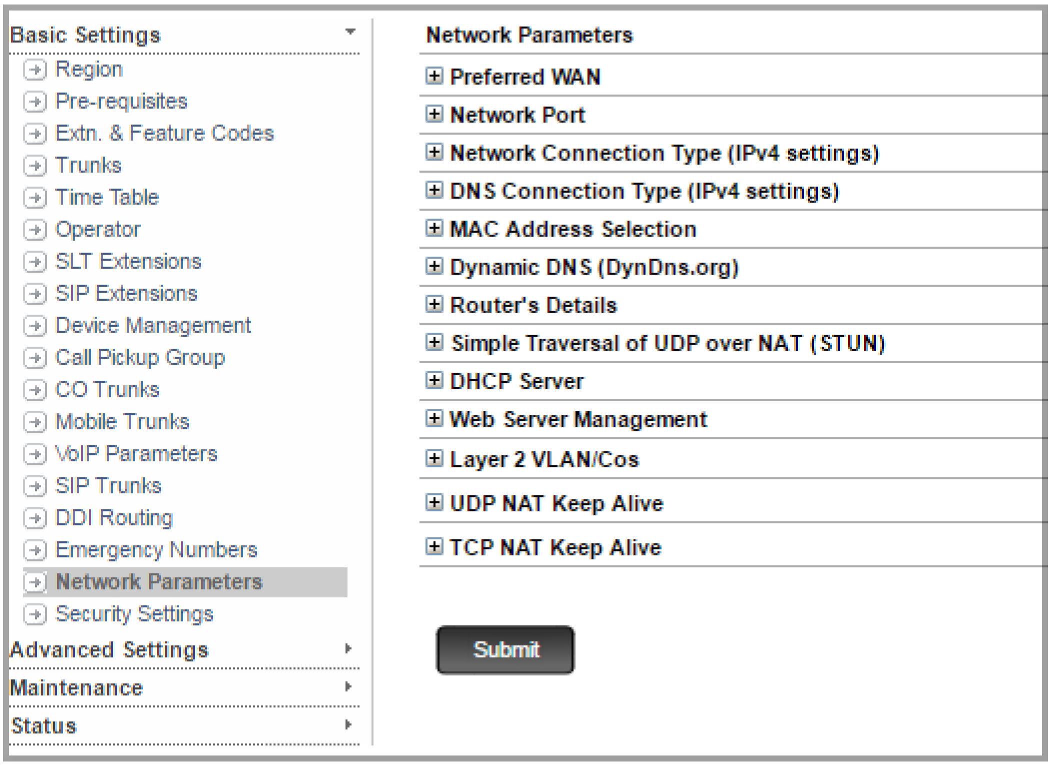
  All the trunks you selected appear in **Through**, in the sequence you selected, separated by a comma.

- Click **Submit**.

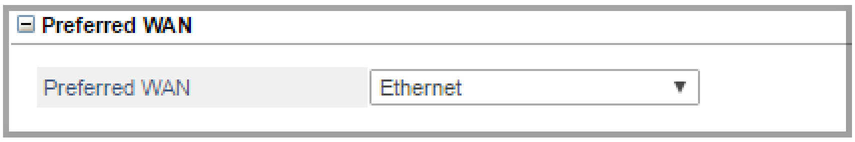
Network Parameters

- Click **Network Parameters**.

  The network parameters page opens.



Preferred WAN



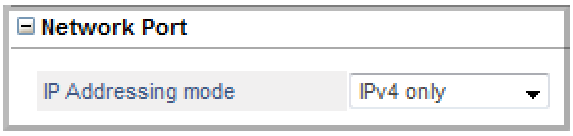
- Click **Preferred WAN** to expand.
- Set the **Preferred WAN** according to the WAN interface you have used for your VoIP installation, which may be:
  - **Ethernet WAN**
  - **Wireless WAN** (Mobile1)

  Default: Ethernet

  If you select Ethernet as Preferred WAN, you must configure the parameters: Network Connection Type, DNS Connection Type, Dynamic DNS, Router's Public IP Address, and Simple Traversal of UDP through NAT (STUN).

  If you select Wireless WAN (Mobile 1), configure Wireless WAN parameters.

Network Port

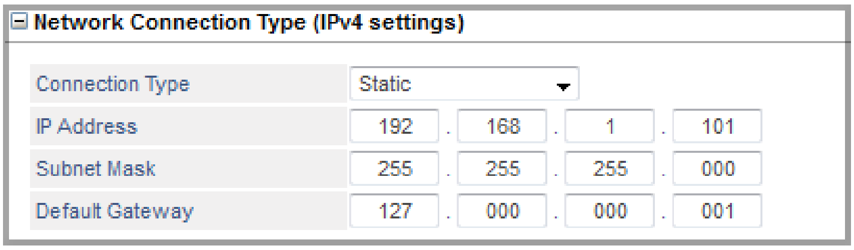


- Click **Network Port** to expand.
- **IP Addressing mode:** Select the IP version you want the system to use. You may select — IPv4 only or IPv4 and IPv6. Default: IPv4 only.

  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

Network Connection Type (IPv4 settings)



- Click **Network Connection Type (IPv4 Settings)** to expand.
- Select the network connection type, i.e. the IP Addressing Scheme used by your network to assign the WAN IP address to the Ethernet Port: Static, DHCP, PPPoE. Default: Static.
  - **Static:** If your network uses Static IP addresses, select Static and configure the following parameters.
    - **IP Address:** Enter the IP Address you obtained from your network Administrator for the Ethernet Port of SARVAM UCS in this field. Make sure that the IP Address does not conflict with that of any other device on the LAN.
    - **Subnet Mask:** Enter the Subnet Mask you obtained from your network Administrator for the Ethernet Port in this field.
    - **Default Gateway:** Enter the IP Address of the Router's LAN Interface as the Default Gateway IP Address.
  - **DHCP:** If your network uses DHCP addressing, the DHCP server will dynamically assign an IP Address, the Subnet Mask, the Gateway Address to the Ethernet Port whenever SARVAM UCS is restarted. You have to configure the Domain Name Server (DNS) Address only, if not already provided by your Internet Service Provider.

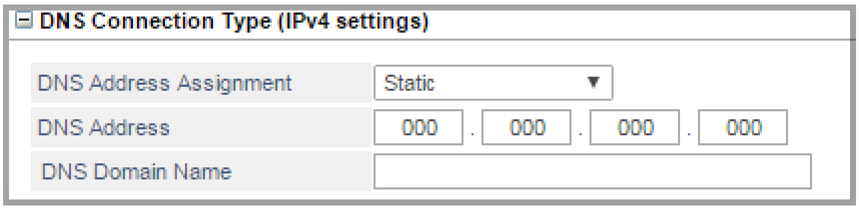
- **PPPoE:** If your network uses PPPoE addressing, the PPPoE server will automatically assign an IP Address, Subnet Mask and Gateway Address to the Ethernet Port of SARVAM UCS. You need to configure the following parameters provided by your Internet Service Provider:

  - **User ID:** Enter the User Name provided by the Internet Service Provider. The User Name 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 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 provided, leave this field blank.

DNS Connection Type (IPv4 settings)



Configure the DNS related parameters as provided by your Internet Service Provider. You may consult your LAN administrator in this regard.

- Click **DNS Connection Type (IPv4 Settings)** to expand.

- Select **DNS Address Assignment** as **Static** or **Auto** according to the Connection Type (IP Addressing scheme) used by the network.

  - Select **Static** if:
    - your network uses Static IP Addressing.
    - your network uses DHCP or PPPoE, but the DHCP/PPPoE server does not provide DNS Address automatically.

    If your network does not assign DNS Address automatically, set DNS Address Assignment as **Static** and enter the DNS Server Address in the **DNS Address**.

    Select **Automatic** if:
    - your network uses DHCP or PPPoE IP Addressing.
    - the DHCP/PPPoE server of your network assigns the DNS Address automatically.

  - **DNS Address:** This field will be editable only if you have selected DNS Address Assignment as 'Static'. Enter the DNS Server IP Address here. The DNS Address can be a maximum of 15 characters.

    If you selected DNS Address Assignment as 'Automatic', the DNS Address assigned by the DHCP/PPPoE server will appear here.

  - **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 16 characters. Default: Blank.

Network Connection Type (IPv6 settings)

- Click **Network Connection Type (IPv6 settings)** to expand.

  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: 64.

      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 Ethernet 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).

- **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).

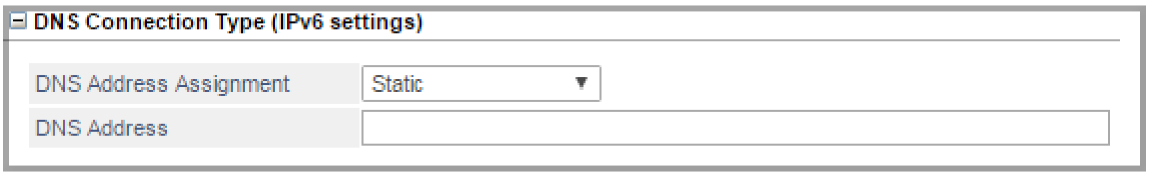
- **Stateless Auto-Configuration:** Select this option as the connection type, if your network uses DHCP to obtain various necessary parameters from DHCP Servers so the DHCP clients can operate in an Internet Protocol (IP) network. DHCPv6 for stateless configuration parameters allows a stateless or statefull 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).

DNS Connection Type (IPv6 settings)



- Click **DNS Connection Type (IPv6 settings)** to expand.

    - **DNS Address Assignment:** If you selected 'Static' as your IPv6 connection type, you can select only 'Static' as the DNS Address Assignment.

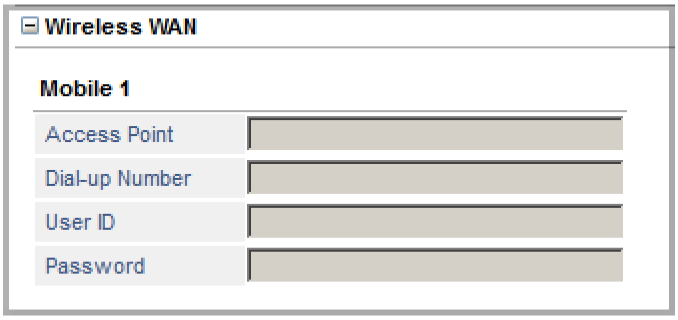
If you have selected PPPoE as your IPv6 connection type, and the PPPoE server provides DNS Address, set the DNS Address Assignment as 'Auto'. If the PPPoE server does not provide DNS Address, set the DNS Address Assignment as 'Static' and configure the DNS Server Address provided by your ISP.

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.

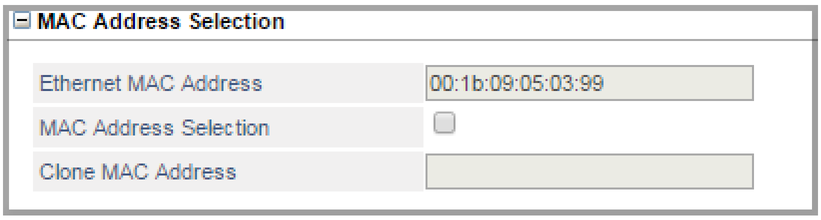
Wireless WAN



If you select Wireless WAN as your preferred WAN, configure the Wireless WAN for the Mobile Port1. Make sure you have Internet Services enabled on the SIM present in the Mobile Port.

- Click **Wireless WAN** to expand.
- In **Access Point**, enter the access point provided by your Service Provider.
- Enter the **Dial-up Number** provided to you for the internet service by your Service Provider.
- Enter the **User ID** provided to you for accessing the internet service by your Service Provider.
- In **Password**, enter the authentication password for the User ID provided by your Service Provider.

MAC Address Selection



- Click **MAC Address Selection** to expand.

- **Ethernet MAC Address:** This non-editable field displays the MAC Address of the Ethernet port.

- **MAC Address Selection:** Select the **MAC Address Selection** check box if you want the Ethernet Port to use a MAC Address other than its own unique MAC Address as source MAC Address.

  When **MAC Address Selection** is disabled, the Ethernet Port will use its unique MAC Address as the source MAC Address on all Ethernet Frames. When **MAC Address Selection** is enabled, the Ethernet Port will use the cloned MAC Address on all Ethernet frames.

  Select the check box to enable cloning of the MAC Address of the Ethernet Port. Default: Disabled.

- **Clone MAC Address:** If you have enabled **MAC Address Selection**, enter the MAC Address to be cloned here.

Dynamic DNS (DynDns.org)

SARVAM UCS supports Dynamic DNS Server client of the Service Provider Dynamic DNS.org. Dynamic DNS (DDNS) is a service that maps internet domain names to IP addresses. Dynamic DNS may be required if you have registered SIP extensions with the Registrar Server of SARVAM UCS.

When you register IP extensions with SARVAM UCS and your network uses dynamic IP addressing (DHCP or PPPoE) to assign the WAN IP Address[87], all the SIP clients registered with SARVAM UCS will need to be updated with the new WAN IP Address to be able to function. Dynamic DNS resolves this by mapping a host name (domain name) (host name) and the preferred WAN IP Address. Each time a new WAN IP Address is assigned by the network, SARVAM UCS updates the mapping on Dynamic DNS server.

You may register with DynamicDNS.org server to use this service. If you have registered with Dynamic DNS.org, do the following:

- Click **Dynamic DNS (DynDns.org)** to expand.

- Select the **Enable Dynamic DNS** check box. Default: Disabled.

  The Dynamic DNS server stores the mapping between hostname and IP Address, which can be updated periodically. However, if the SARVAM UCS frequently sends IP Address update requests to the DDNS server, the server is likely to block the hostname in its database and terminate the DDNS services provided to SARVAM UCS.

---

87. According to your installation scenario and Preferred WAN you configured.

So, if you restart the SARVAM UCS frequently, chances are that DDNS server may block the hostname of your system. This will in turn affect the ability of the system to receive the calls using DDNS hostname since the entry (mapping between hostname and IP Address) in the DNS server will be deleted in such scenarios.

SARVAM UCS updates the WAN IP Address in the DDNS server at each Power ON. You can also update the WAN IP Address at any time you want, as required.

- Enter the User ID you created on DynDNS.org in the **User ID**. A maximum of 40 characters, including all ASCII characters are allowed.

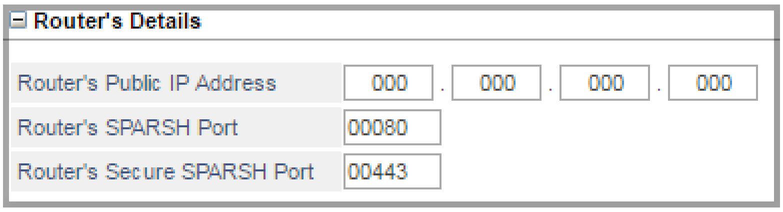
- Enter the password you created for the User ID on DynDNS.org in the **Password**. The password may be not more than 24-characters long.

- Enter the **Host Name** you created on DynDNS.org. The host name must be under 40 characters. All ASCII characters except < > and “ (double quote) are allowed.

- Define the **Retry Trials.** This is the number of attempts SARVAM UCS will make to send the WAN 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 5.

- Whenever you want to update the WAN IP Address on the DDNS server, press **Click to Update IP Address**.

Router’s Details



This parameter is of relevance if the WAN port of SARVAM UCS is located behind a NAT Router and configuration requests are to be forwarded to the public internet.

Router‘s Public IP Address is also used for web access and auto configuration of the phones. It specifies the fixed IP Address of your NAT router required for NAT Traversal in SIP messages.

- Click **Router’s Details** to expand.

- Enter the **Router’s Public IP Address**.

- Enter the **Router’s SPARSH 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.

When the system is set to default, the Router’s SPARSH Port value will not be set to default.

- Enter the **Router's Secure SPARSH 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.

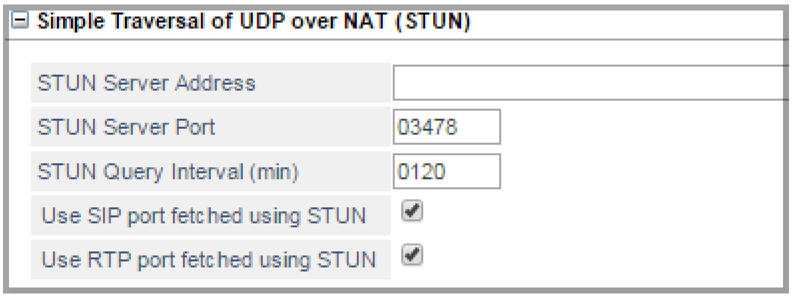
If you want the SARVAM UCS and VARTA AMP100 to communicate using secure protocol, make sure you have enabled the option Enable Secure Connection with Server in the client.

When the system is set to default, the Router's Secure SPARSH Port value will not be set to default.

You also need to select 'Router's Public IP Address' as the 'Source Port IP Address' in the 'SIP Extension General Parameters', when you configure SIP Extensions. Refer the "VoIP Parameters".

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 configure this parameter.

Simple Traversal of UDPs through NAT (STUN)



STUN is to be configured only if the Ethernet WAN of SARVAM UCS is located behind a NAT Router and SIP Messages need to be forwarded to the public internet.

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*** configure this parameter.

If your router as asymmetric NAT, configure the following STUN parameters:

- Click **Simple Traversal of UDPs through NAT (STUN)** to expand.
- In the STUN Server Address: Port,
  - Enter the **STUN Server Address**, a maximum of 40 characters.
  - Enter the Listening Port of the STUN Server in the **STUN Server Port**. The valid range for this field is from 1024-65535. The default STUN Port is 03478.

- In **STUN Query Interval (min)**, enter the time interval between each STUN query for the Public IP Address of the NAT Router. The range of this interval is from 0001 to 9999 minutes. By default, it is set to 120 minutes.

- To allow the SIP Port number to be fetched using STUN in the SIP message, you must select **Use SIP Port fetched using STUN** check box. Default: enabled.

  If you are using Port-Forwarding in the Router for SIP messages, disable Use SIP Port fetched using STUN by clearing the check box.

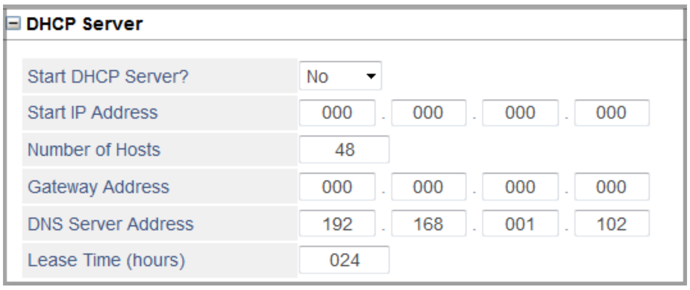
- To allow the RPT Port number to be fetched using STUN in the SIP message, you must select **Use RTP Port fetched using STUN** check box. Default: enabled.

  If you are using Port-Forwarding in the Router for SIP messages, disable Use RTP Port fetched using STUN by clearing the check box.

You also need to select 'Use IP Address fetched using STUN' as the 'Source Port IP Address' in the 'SIP Extension General Parameters', when you configure SIP Extensions'. Refer “VoIP Parameters”.

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 configure this parameter.

DHCP Server



SARVAM UCS has an in-built DHCP server for registering the Matrix Extended IP Phones. Configure DHCP Server settings if you are connecting Extended IP Phones.

- Click **DHCP Server** to expand.

- To enable the DHCP Server, set **Start DHCP Server?** to **Yes**. Default: No.

- In the **Start IP Address**, define the starting IP Address to be assigned to DHCP client. SARVAM UCS will allocate the first available free IP Address to DHCP clients.

- In the **Number of Hosts**, define the maximum number of DHCP Clients to be assigned IP Addresses by the DHCP server. You can define the Number of Hosts from 2 to 48. Default: 48.

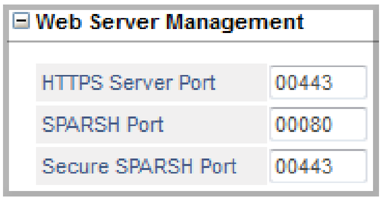
- In the **Gateway Address**, enter the Gateway IP Address to be assigned to the DHCP Clients.

- In the **DNS Server Address**, enter the DNS address to be assigned to the DHCP Clients.

- Define the maximum **Lease Time (hours)** for which the IP Addresses should be allocated to the DHCP Clients. The Lease Time may be from 1 to 192 hours. Default: 24 hours.

DHCP client should renew its lease within the Lease Timer. If the client does not renew its lease before the expiry of the configured Lease Time, SARVAM UCS will free the IP Address. This IP Address may be assigned to another DHCP client.

Web Server Management



- Click **Web Server Management** to expand.
- Enter the **HTTPS Server Port**. Valid range: 443 or any value ranging from 1025 to 65535. Default: 443.
- 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.

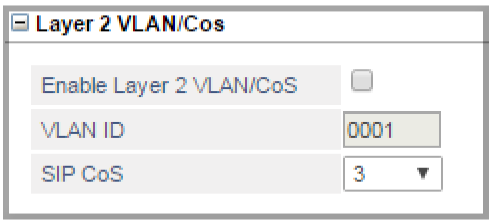
  Valid range: 80 or any value ranging from 1025 to 65535. Default: 80.
- Enter the **Secure SPARSH Port number**, if you want to auto configure the VARTA AMP100/VARTA ADR100/ VARTA WIN200 with the server using a secure protocol. The system will listen for the configuration request from the VARTA clients 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 to communicate using secure protocol, make sure you have enabled the option Enable Secure Connection with Server in the client.

When the system is set to default, the port values of HTTPS Server Port, SPARSH Port Number and Secure SPARSH Port value will not be set to default.

Layer 2 VLAN/Cos



- Click **Layer 2 VLAN/Cos** to expand.

  Layer 2 VLAN/CoS is to be configured if the Ethernet 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[88].

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:

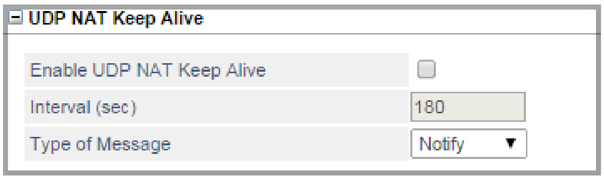
COS	Traffic Type
0	Best Effort
1	Background
2	Spare
3	Excellent Effort
4	Controlled Load
5	Video
6	Voice
7	Network Control

- **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.

- **SIP CoS:** Define the CoS (priority) bits in all SIP packets. The range of CoS bits is from 0 to 7. Default: 3.

UDP NAT Keep Alive



- Click **UDP NAT Keep Alive** to expand.

UDP NAT Keep Alive is to be configured when the Ethernet port is connected behind a NAT router[89] and SIP messages are transported over UDP. UDP NAT Keep Alive messages must be sent to refresh the UDP binding in the NAT router.

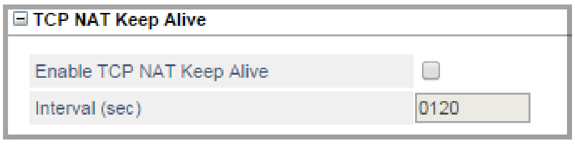
88. *The IEEE 802.1P standard allows Layer2 switches to prioritize the traffic, thus providing Quality of Service (QoS), better handling of data that pass over a network, thereby resulting in greater reliability and quality. Quality of Service (QoS) on Layer2 is referred to as Class of Service (CoS) which is defined by IEEE 802.1P.*
89. *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.*

- **Enable UDP NAT Keep Alive:** Select this check box 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 Ethernet 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 is to configured when the Ethernet 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:** Select this check box 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 Ethernet 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.

Click **Submit** to save settings.

Security Settings

The feature Security Settings enables you to restrict unauthorized access to ETERNITY NENX using Web, Third Party Auto Configuration, Force complex SIP Authentication Password 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.

Force Complex SIP/HTTP Authentication Password

By default, system enforce complex SIP and HTTP Authentication Password to avoid unauthorized system access. However, you may choose to keep simple Authentication Password, if required. If you allow simple Authentication Password then your system security might be at risk. See "How to configure" below to change password complexity. This activity will be logged in the "System Activity Log".

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, ETERNITY 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 10 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.

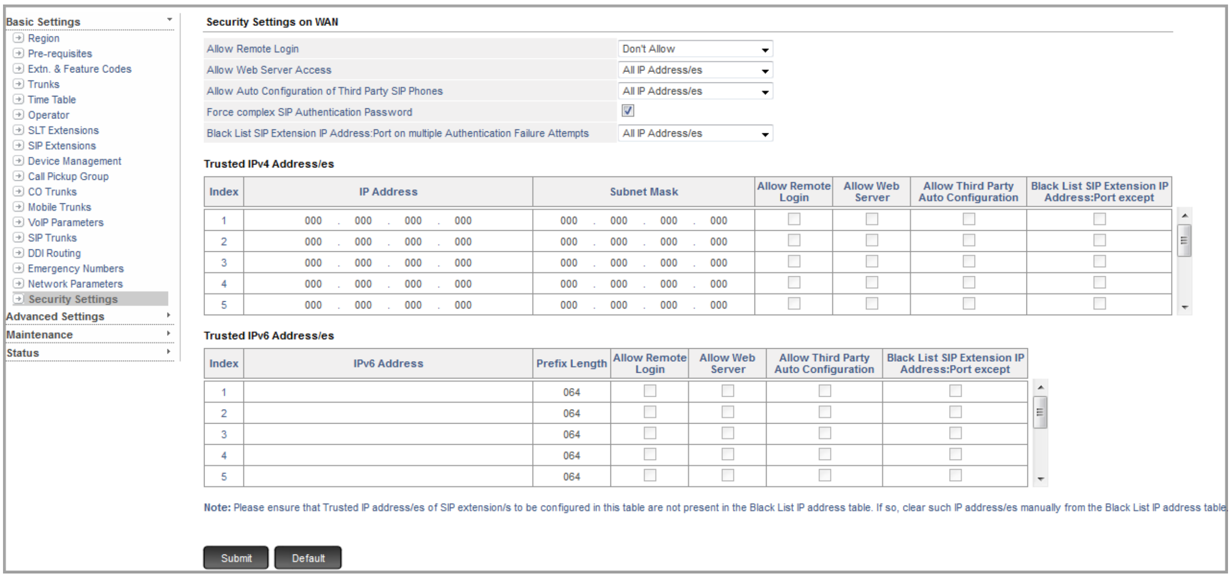
- with this table configured,
    - access to Remote Login, Web Server, Third Party Auto Configuration will be allowed only to the configured Trusted IP Address/es.
    - Trusted IP Addresses configured for the registration of the SIP Extension, will not be blacklisted.

The successful attempt to access SARVAM UCS using Remote Login or Web will be logged in the System Activity Log. Also, Enable/Disable of Force complex SIP Authentication Password will be logged in the System Activity Log.

When you change the default settings of the jumper (J7) on the CPU module, the parameters—Allow Remote Login, Allow Web Server Access From—will also be set to default.

How to configure

- Log in as System Engineer.
- Click **Basic Settings**.
- 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 All IP Address/es or Only Trusted IP Address/es option. By default, **All IP Address/es** 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, **All IP Address/es** option is selected for **Black List SIP Extension IP Address:Port on multiple Authentication Failure Attempts** . 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 All IP Address/es or Only Trusted IP Address/es. By default, **All IP Address/es** 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.

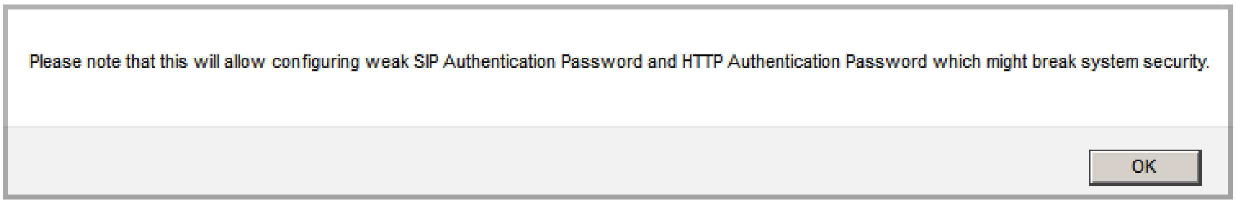
- By default, **All IP Address/es** option is selected for **Black List SIP Extension IP Address:Port on multiple Authentication Failure Attempts** . If you do not want to include the Trusted IP Addresses, select **Except Trusted IP Address/es** option.

  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.

- **Force complex SIP Authentication Password:** By default, this check box is enabled. You must configure complex SIP/HTTP Authentication Password for the "SIP Extensions". Password must be of minimum 8 characters and can be 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.

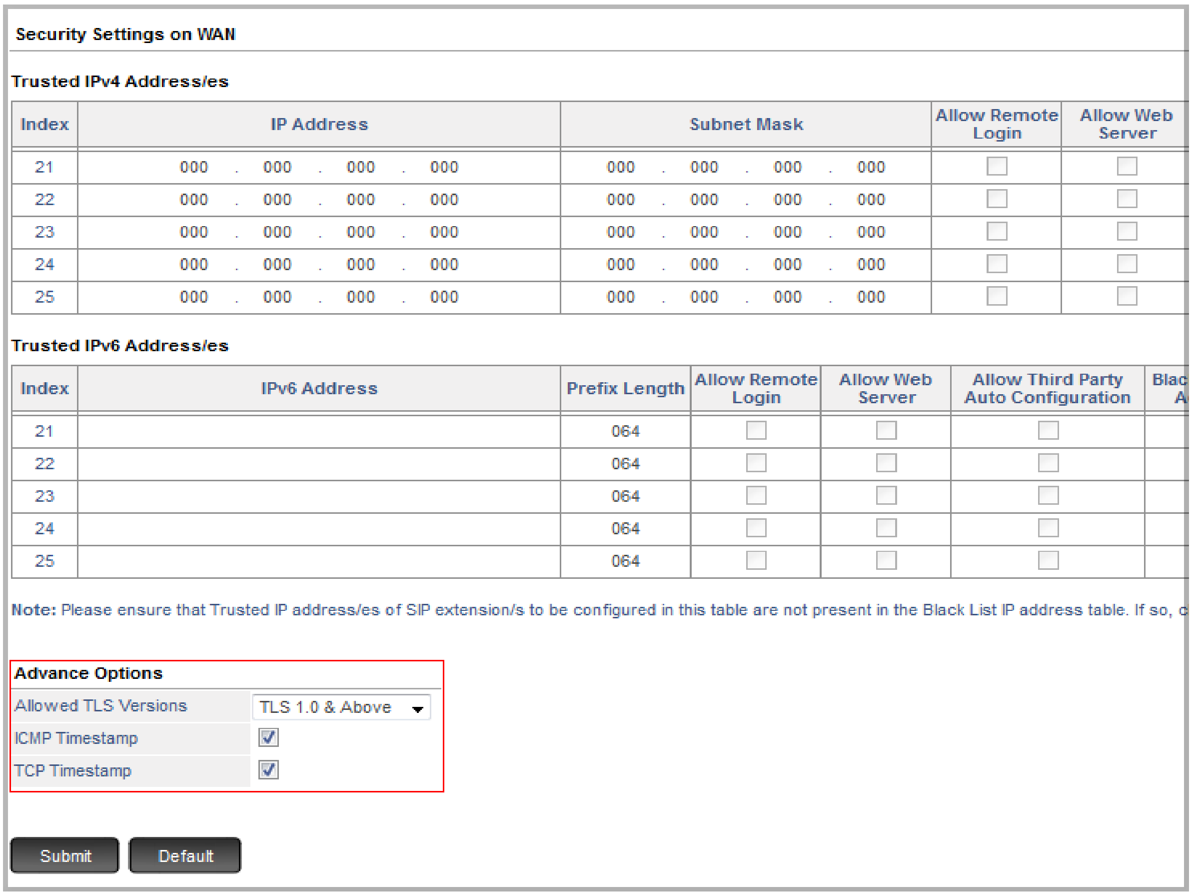
  Disable this check box, if you want the system to allow configuration of simple SIP/HTTP Authentication Password. When you disable this check box, the following alert message appears.



  Click OK. You will be able to configure Authentication password of minimum 2 characters. Allowed characters are 0-9, a-z, A-Z, all special characters except %, =, #, +, &, \, <, >, ", ' and space.

- Click **Submit**.

Advance Options



- In **Allowed TLS Versions,** select the TLS Version[90] 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.

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.

90. *SPARSH VP330 supports TLS Version 1.0 only.*

CHAPTER 6

Configuring Voice Mail System

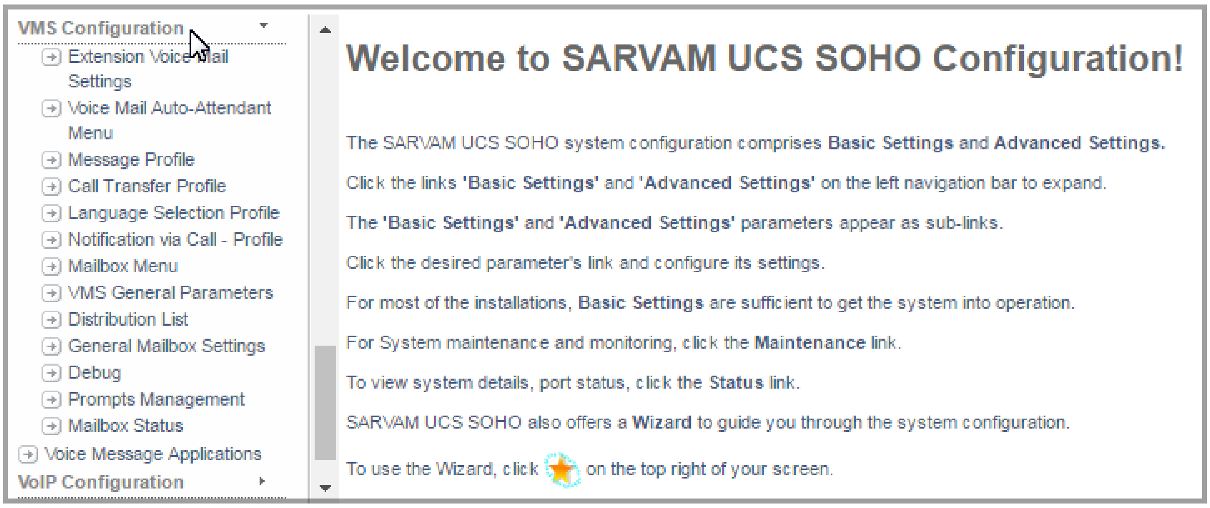
Before you begin configuration of the VMS related parameters, consider the following points:

- SARVAM UCS offers a Voice Mail System (VMS) with VMS Auto Attendant features. The VMS is supplied with a Pen Drive (factory-fitted). To use Voice Mail System, you must purchase and activate the VMS license. Refer to "License Management".
- 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 port number access codes. Configure the VMS parameters for these extensions. Refer to "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 (CO, Mobile, SIP) and 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 day, break and night for the desired trunk ports on which you want to use the VMS Auto Attendant. Refer to "Voice Mail Auto Attendant Menu".
  - assign the desired Voice Mail Auto Attendant (VMAA) Menu to the trunks. Refer to "CO Trunks", "SIP Trunks", "Mobile Trunks".
- The prompts used to route the call using the Voice Mail Auto Attendant can be customized as per your requirement. Refer to "Prompts Management".

To configure Voice Mail Settings,

- Login as System Engineer.

- Under **Advanced Settings**, click **VMS Configuration.**



- Configure the VMS Parameters using the different sub-links as required.

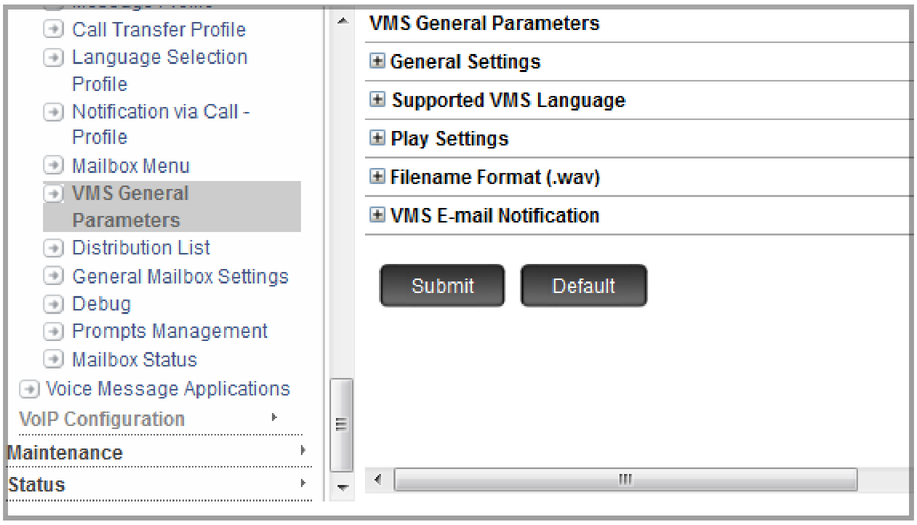
Configuring VMS General Parameters

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

- Login as System Engineer.
- Under **Advanced Settings**, 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 check box 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.

- **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[91]:** 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, 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 transfer* or *Single File before and after call transfer* as per your requirement.

  If you select *Individual File before and after call transfer*, 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.

91. *This parameter will not function if you have selected None as the* ***Use SMTP Account*** *option. You may configure this parameter for future use.*

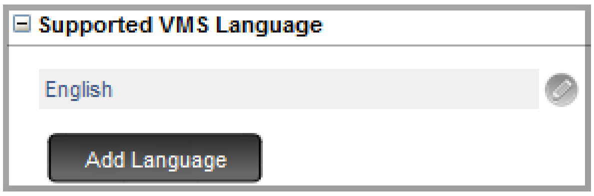
- **Date Format:** Select the Date Format — DD-MM-YYYY or MM-DD-YYYY. The Date format selected here will be applicable for all the VMS features, VMS E-Mail Notifications, etc.

*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

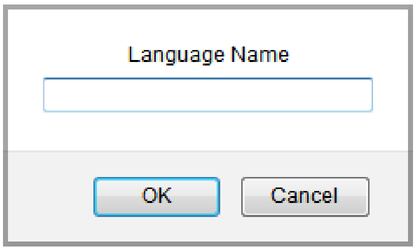
You may add a new language, edit or delete the existing languages.



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.

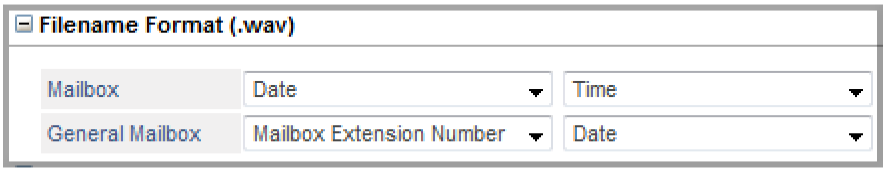
To delete a language, click .

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[92]— 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.

92. *The **Mailbox Extension Number** parameter is only applicable for General Mailbox Filename Format.*

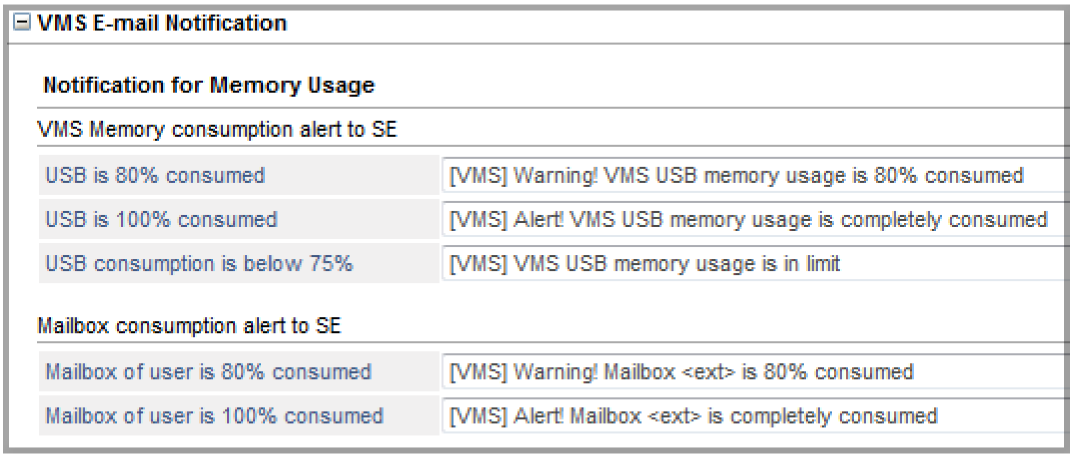
Option	Meaning
Calling Number	Select this option if you want the system to add the calling number. If calling number is Blank, the system will add ‘No Number’.
Called Number	Select this option if you want the system to add the called number. If Called number is not available, system will ignore this 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.

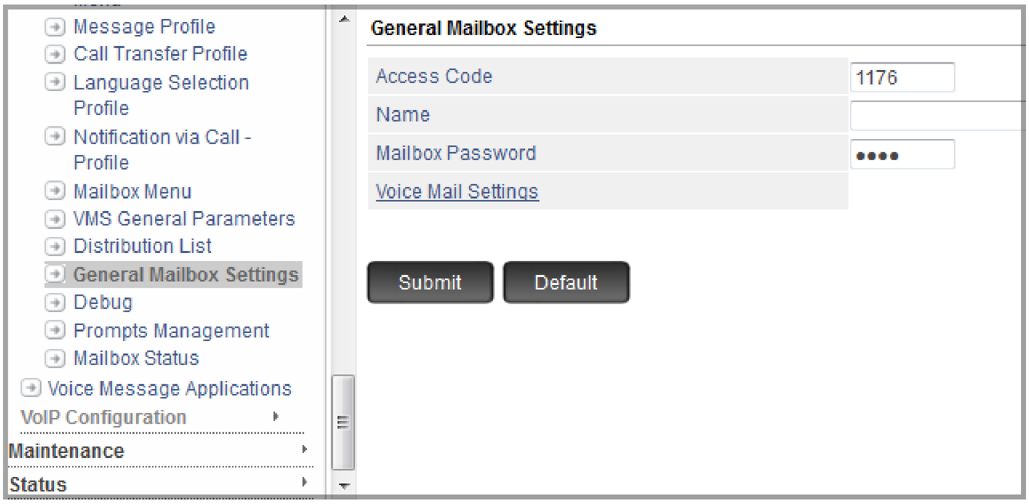
To access the General Mailbox, the extension users must have the feature General Mailbox enabled in their CoS.

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

- Login as System Engineer.
- Under **Advanced Settings**, 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.

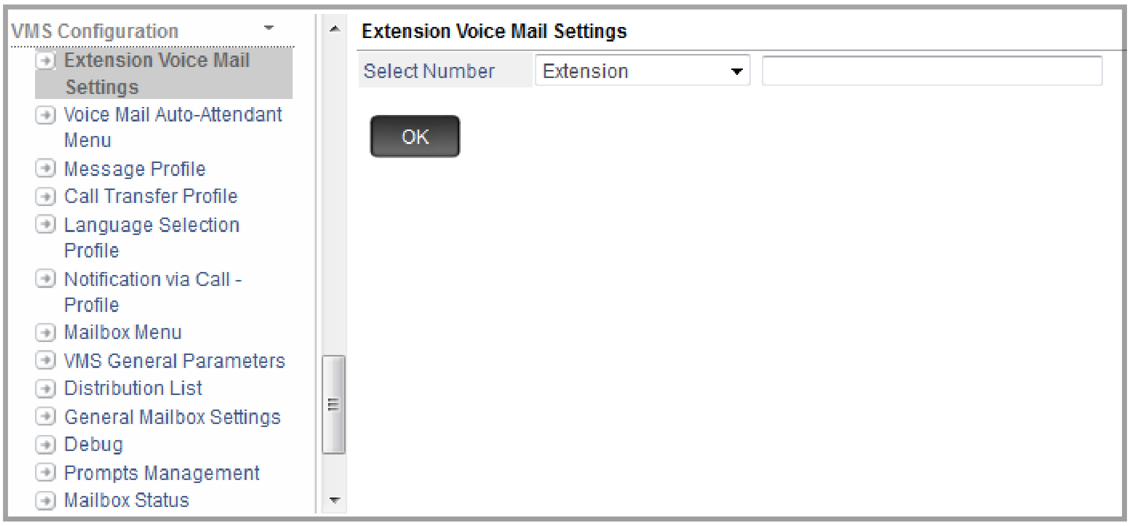
For more information and instructions on how to access the General Mailbox, see “Accessing General Mailbox”.

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.

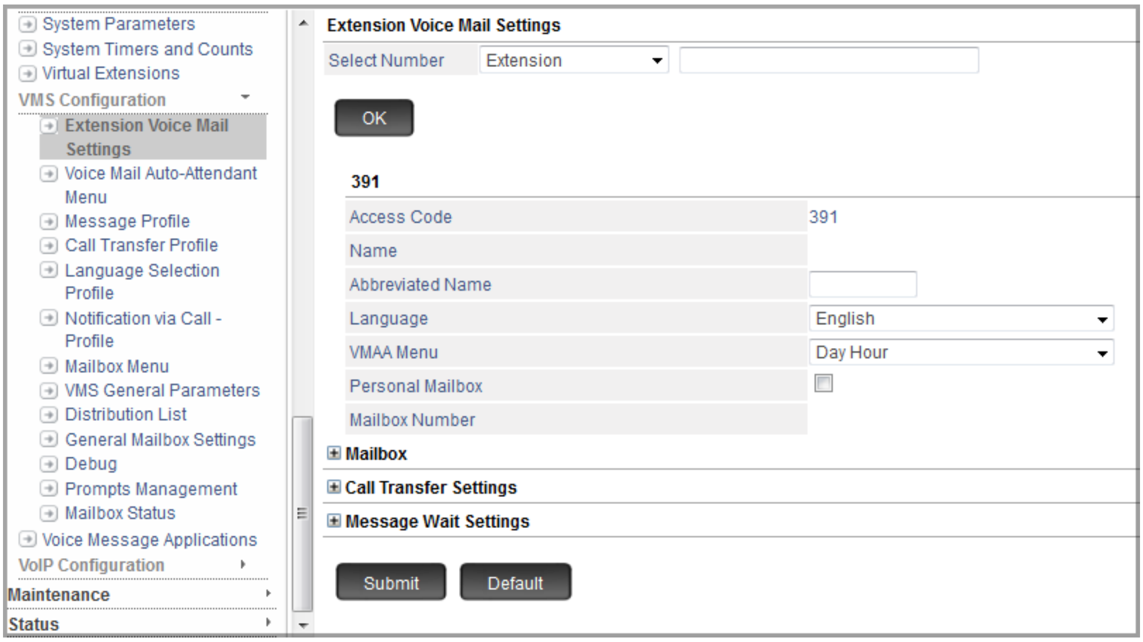
Configuring Extension Voice Mail Settings

- Login as System Engineer.
- Under **Advanced Settings**, 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.
- 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.

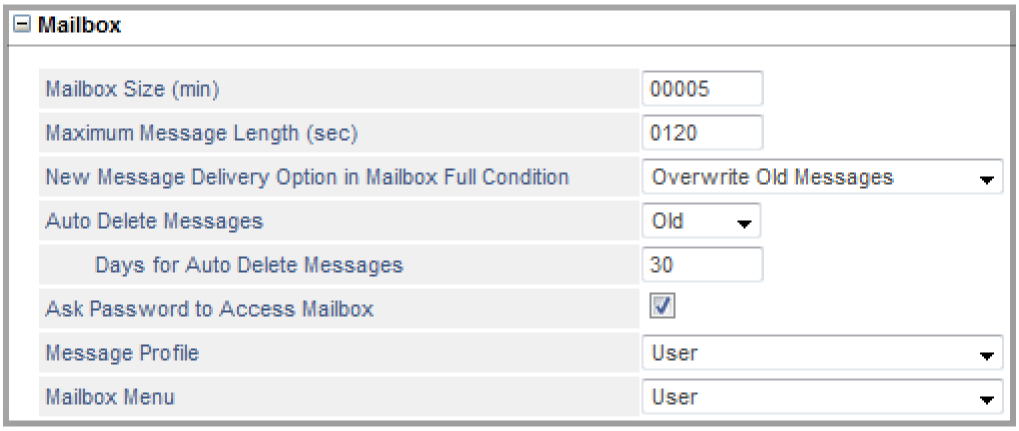
- **VMAA Menu:** Select the VMAA Menu Number which you want to assign to the respective Extension or Department Group. For more information, see “Voice Mail Auto Attendant Menu”.

- **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[93]



- 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.

    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.

93. *Not applicable when you select **Operator** as the “Select Number” option.*

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.

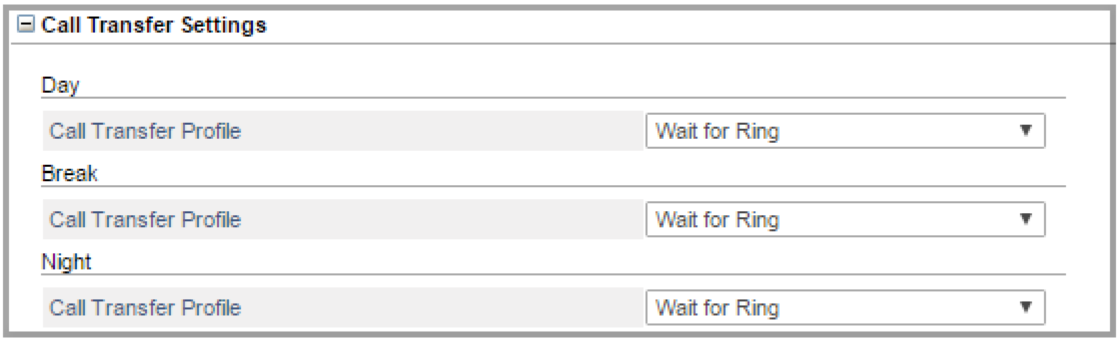
  If you want to remove password protection, clear this check box.

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".

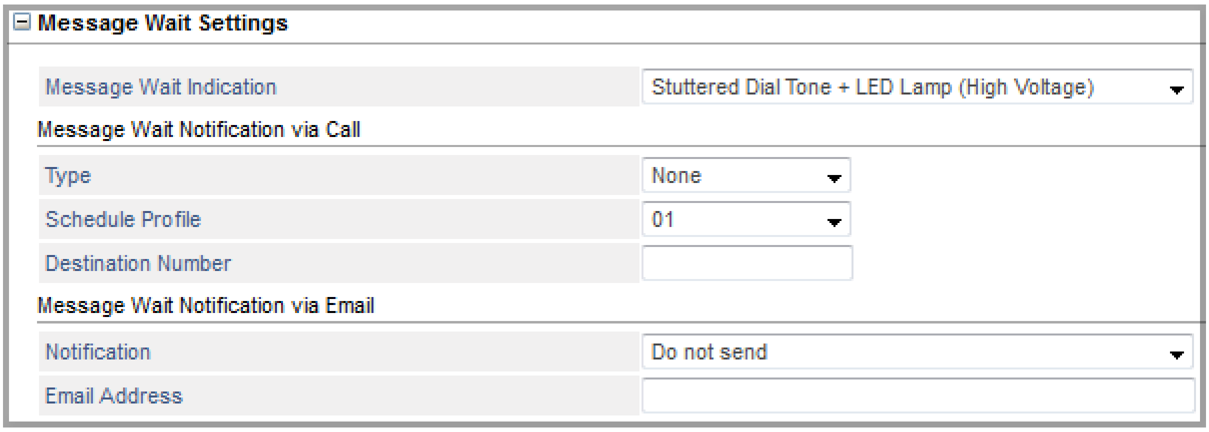
- In **Mailbox Menu**, select the Mailbox Menu you want to assign. For more information, see "Mailbox Menu".

Call Transfer Settings[94]



- Under **Day**, 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**, 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 **Night**, 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[95]



- 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.

94. *Not applicable when you select **General Mailbox** as the “Select Number” option.*
95. *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 4 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[96]

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.

96. *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[97]

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.

---

97. *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, refer to Incoming Call Routing in “CO Trunks”, “Mobile Trunks” and “SIP Trunks”.

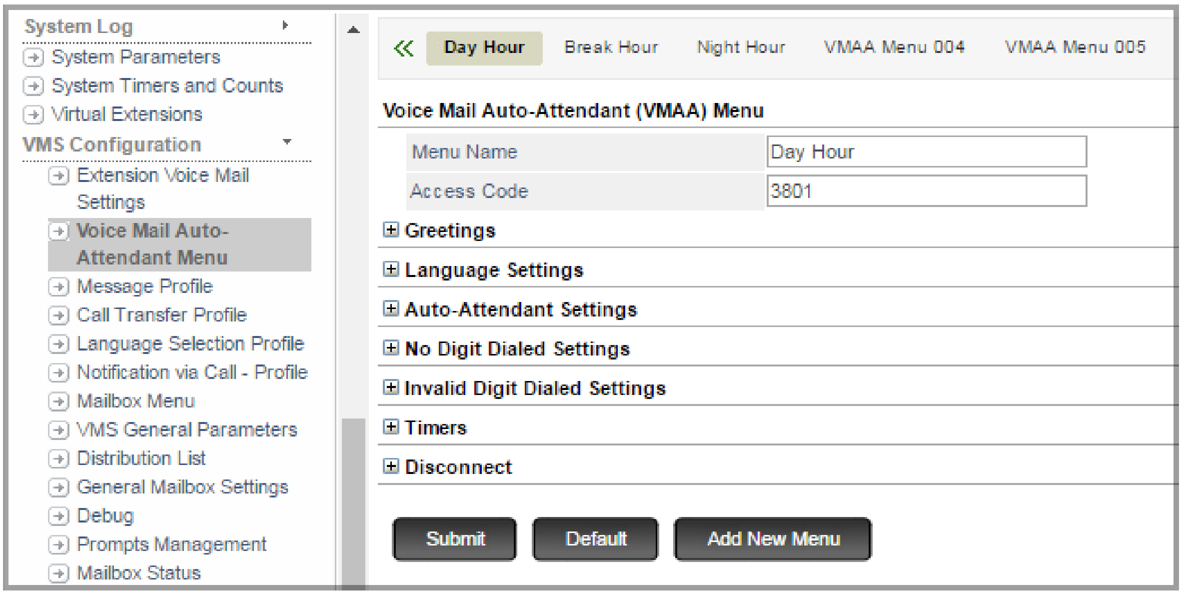
If VMAA Menu assigned to any action/trunk is deleted, the system will consider the first VMAA Menu for that action/ trunk by default.

VMS supports a maximum of 64 VMAA Menu.

By default, three VMAA Menus — Day Hour, Break Hour and Night 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

- Login as System Engineer.
- Under **Advanced Settings**, 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.

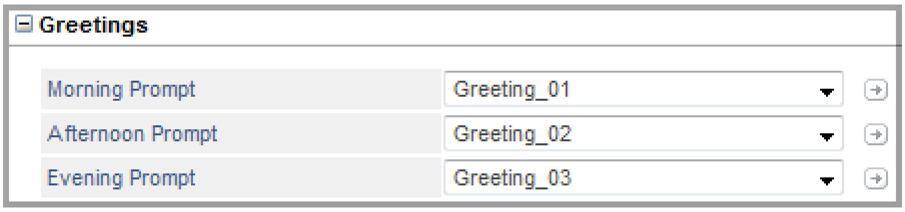
To edit a VMAA Menu,

- 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 01 to 64.
- 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.

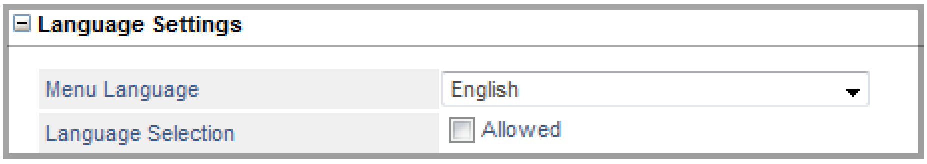
- Click **Settings** . The **Prompts Management** page opens.

  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.

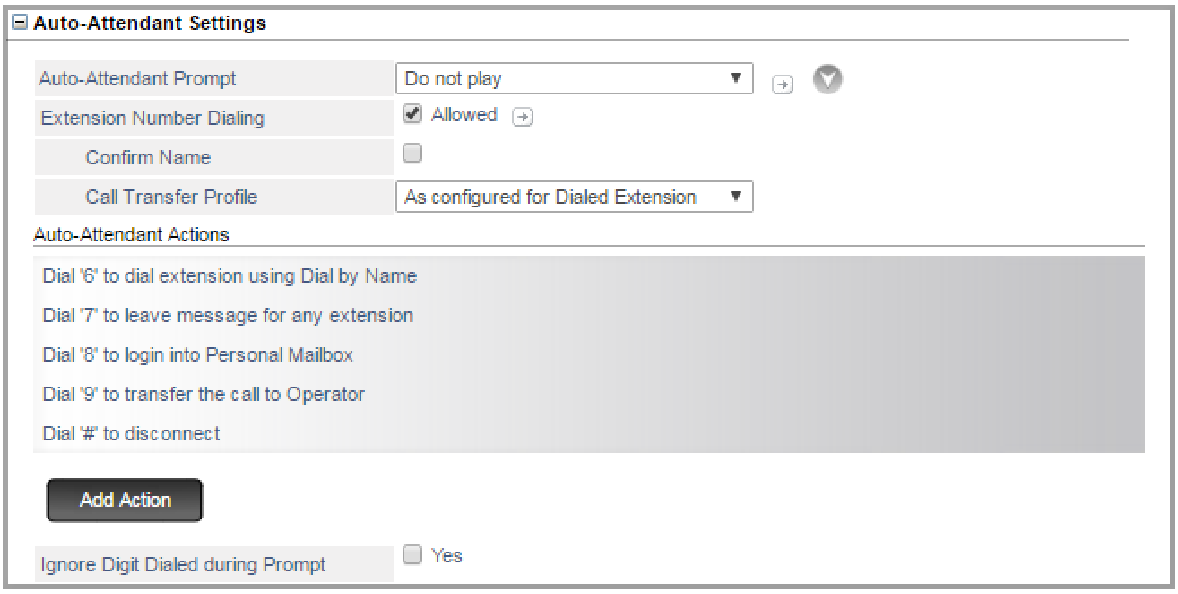
  - In **Language Selection Profile**, select the language profile you want to assign to the respective VMAA Menu.

    - 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 3 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 Department Group** or **Transfer to Extension**, select the respective 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 ask for the name of the called party after dialing the 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 name stored in the Global Directory.

*The options — **Confirm Name** and **Confirm Name Prompt** are only applicable when **Extension Number Dialing** is enabled.*

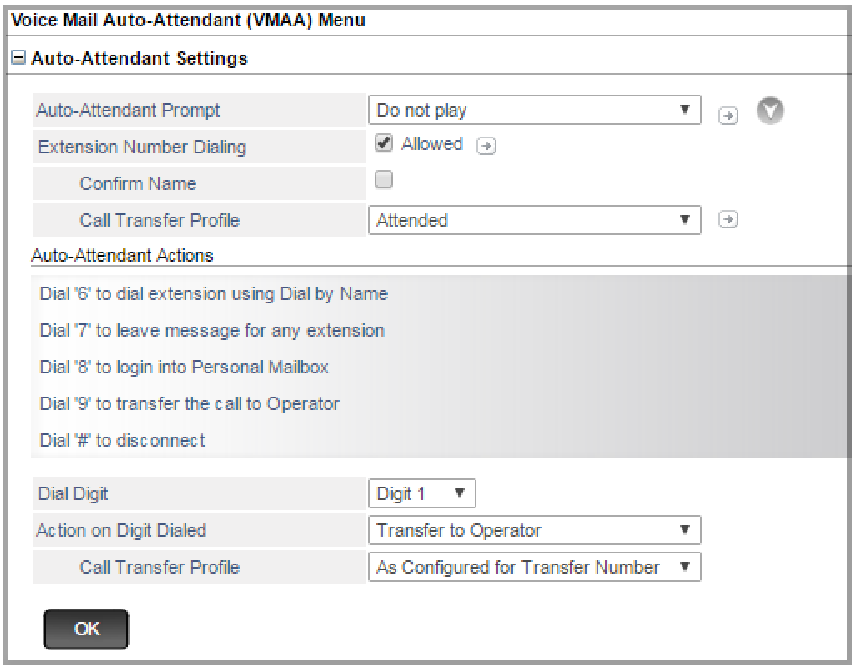
- In **Call Transfer Profile**, select the desired Call Transfer Profile you want the system to use. Make sure you have configured the parameters for this profile. By default, *As configured for Dialed Extension* is selected.
  - Select **As configured for Dialed Extension** if you want the VMS Auto 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 Extension** or **Transfer to Department Group**, select the 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".

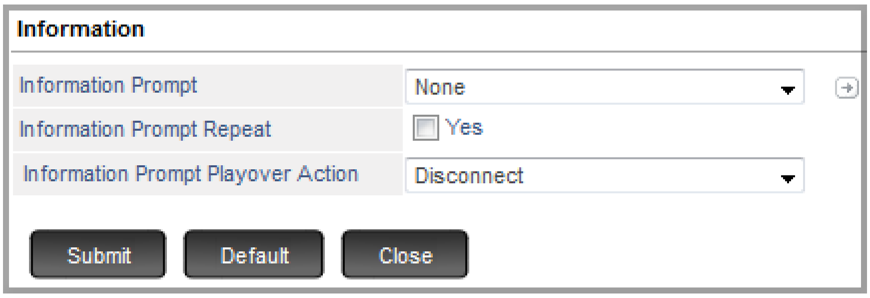
      All the VMS prompts henceforth will be played in the language selected here.
    - If you select **Repeat prompt**, the system will repeat all the Auto-Attendant prompts configured in sequence.

- 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 Department Group** or **Transfer to Extension**, select the respective Department Group number or enter the desired extension number.

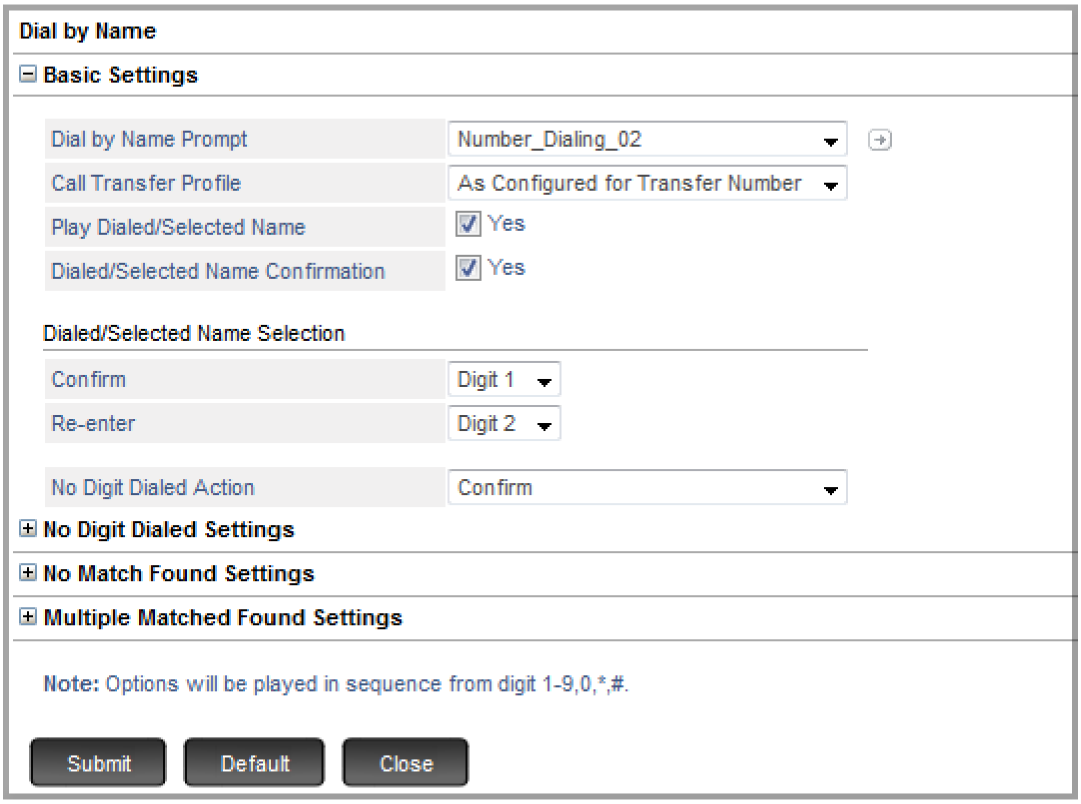
- In **Call Transfer Profile**, select the desired Call Transfer Profile as per your requirement. By default, *As configured for Transfer Number* is selected.

  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 name dialed.

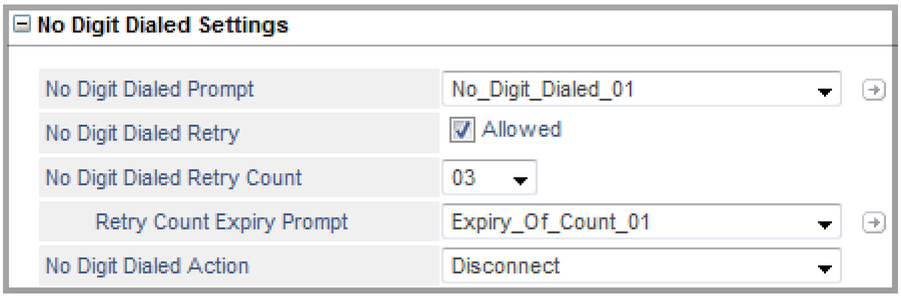
    - In **Confirm**, assign the Digit (0-9, *, # or None) that you want the system to play to the caller for confirming the Name dialed before transferring the call. If you assign *None*, the option will not be played.

    - 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 name 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 Department Group** or **Transfer to Extension**, select the respective Department Group number or enter the desired extension number.

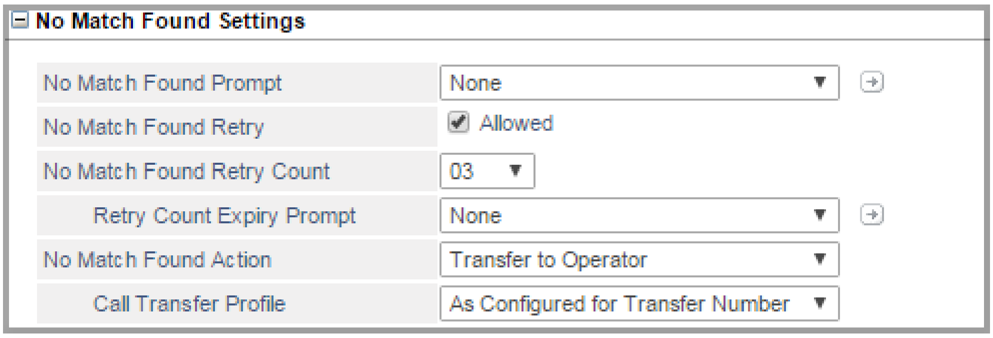
    In **Call Transfer Profile**, select the desired Call Transfer Profile as per your requirement. By default, *As Configured for Transfer Number* is selected.

    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 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 No Match Found Prompt.

- Select the **No Match Found Retry** check box to prompt the caller to dial the Extension name 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 Department Group** or **Transfer to Extension**, select the respective Department Group number or enter the desired extension number.

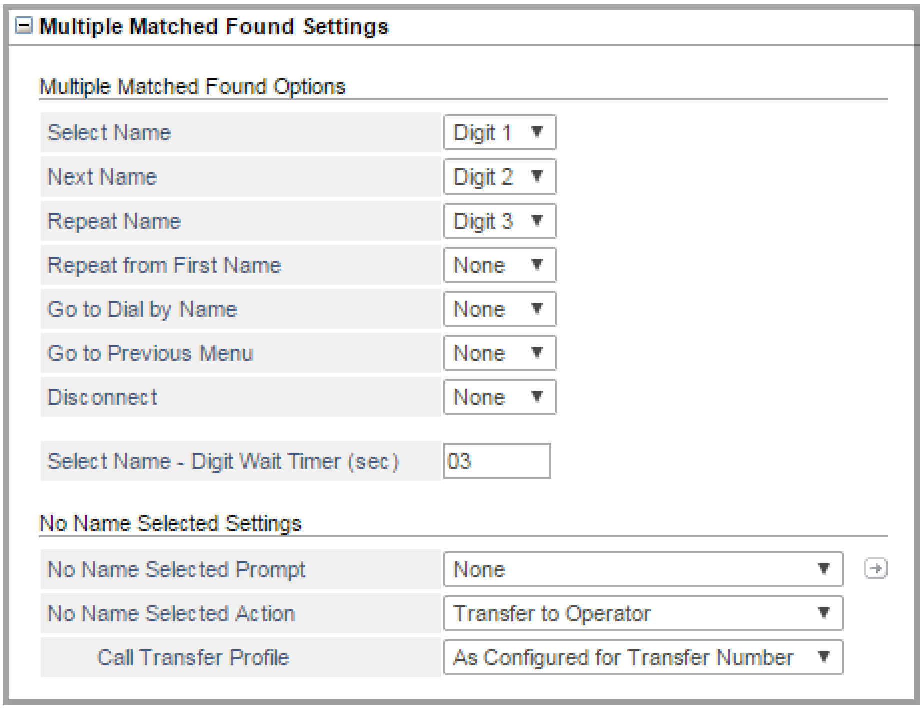
    In **Call Transfer Profile**, select the desired Call Transfer Profile as per your requirement. By default, *As Configured for Transfer Number* is selected

    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 Department Group** or **Transfer to Extension**, select the respective 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 Department Group** or **Transfer to Extension**, select the respective 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 Department Group** or **Transfer to Extension**, select the respective 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.

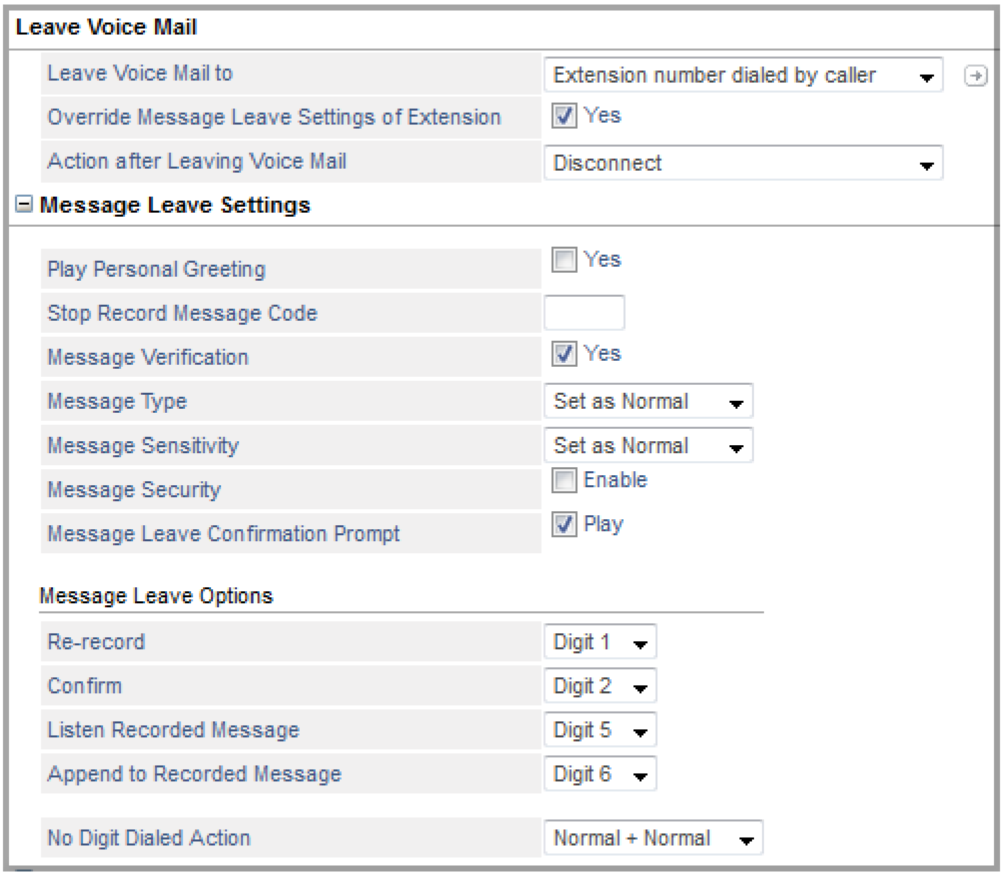
Click **Settings** to configure the parameters of the selected profile. For more information, see "Call Transfer Profile".

- 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 Department Group** or **Transfer to Extension**, select the respective 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.

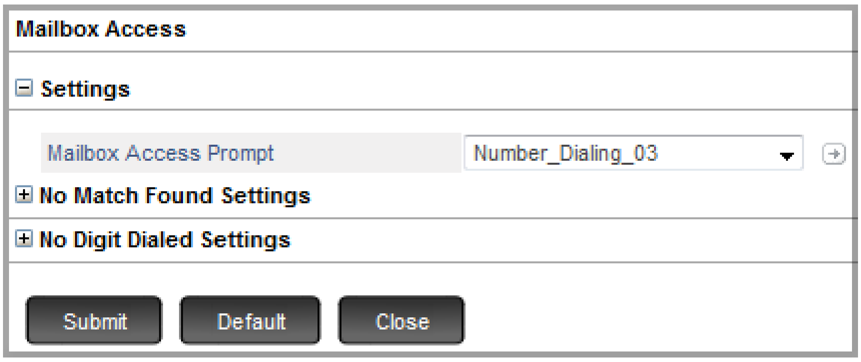
    Click **Settings** to configure the parameters of the selected profile. For more information, see "Call Transfer Profile".

- 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.

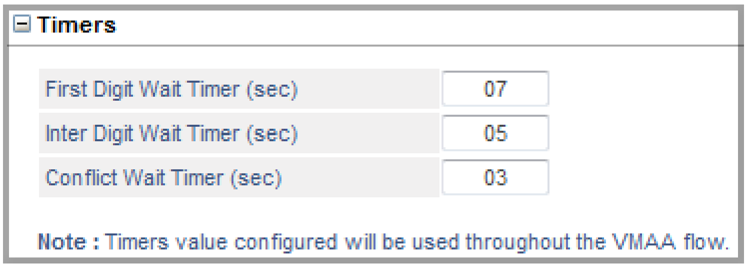
No Digit Dialed Settings

- For details, refer "No Digit Dialed Settings".

Invalid Digit Dialed Settings

- For details, refer "Invalid Digit Dialed Settings".

Timers

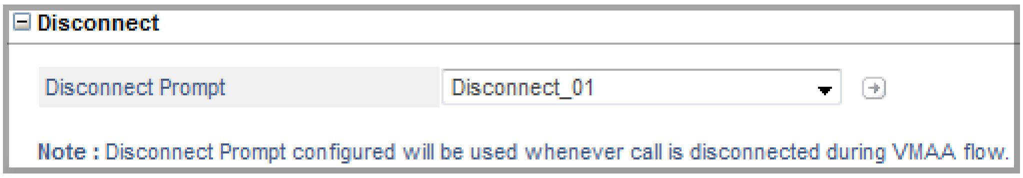


- In **First Digit Wait Timer**, enter the time for which you want the system to wait for the caller to dial the first digit for menu option selection or for entering the Extension or Department Group number. On the expiry of the First Digit Wait Timer, the system will apply "No Digit Dialed Settings". 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.

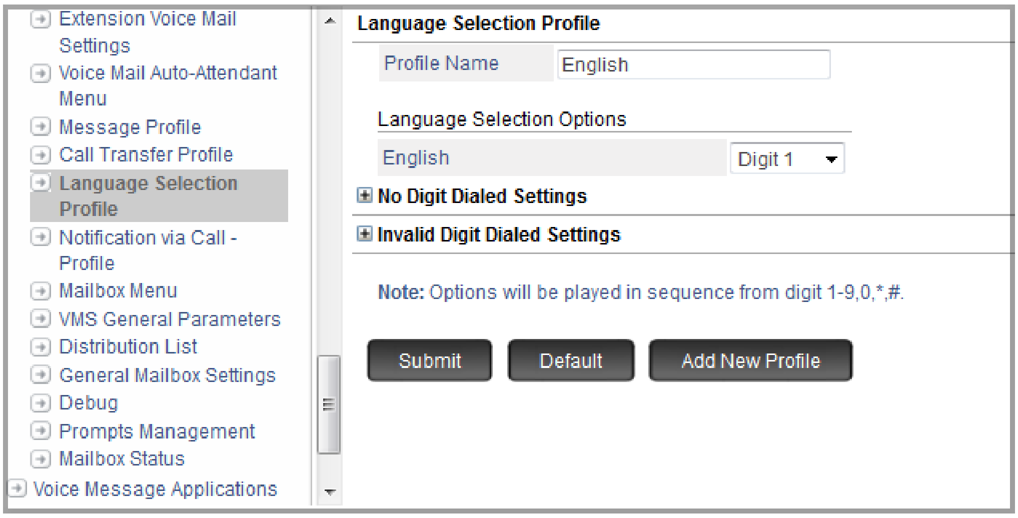
Language Selection Profile

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

- Login as System Engineer.
- Under **Advanced Settings**, 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.

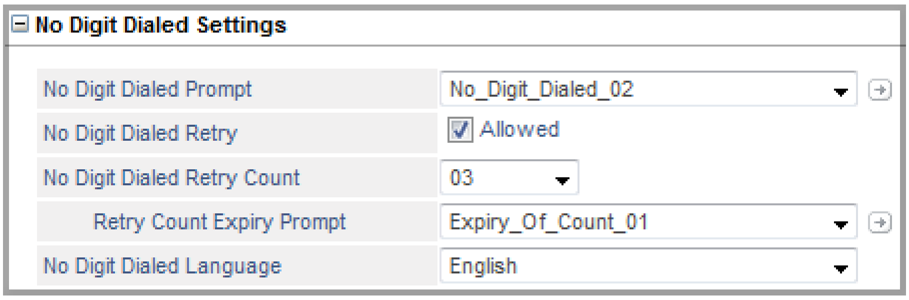
The digit you configure will be played as the digit to be dialed by the caller to select the respective language. The options will be played in the sequence — 1-9, 0, *, # — as per the Digits configured. The language for which the Digit selected is None will not be played.

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.

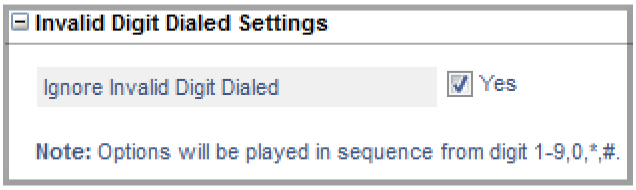
You may add a new prompt from here. For further details, see “Prompts Management”.

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

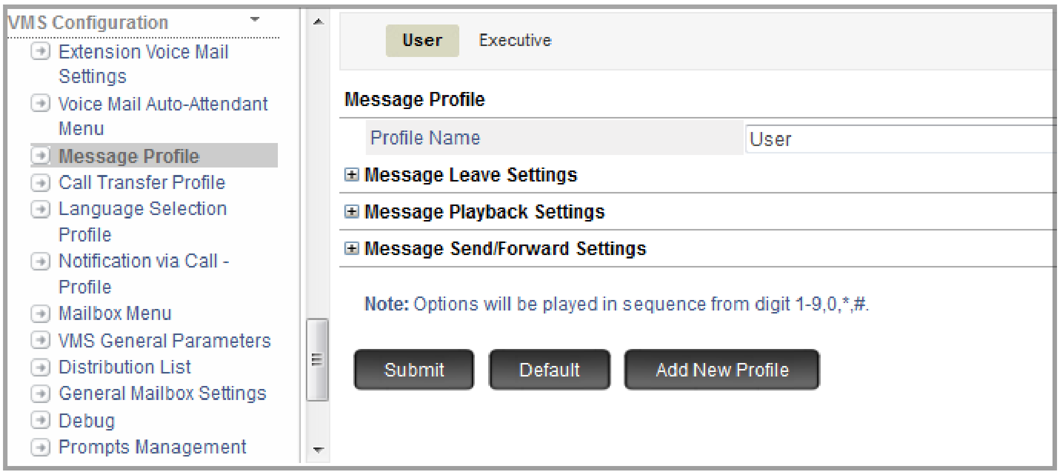
Message Profile allows you to customize the Message Leave Settings, Message Playback Settings and Message Sent/Forward Settings.

VMS supports a maximum of 12 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

- Login as System Engineer.
- Under **Advanced Settings**, 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 12.

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.

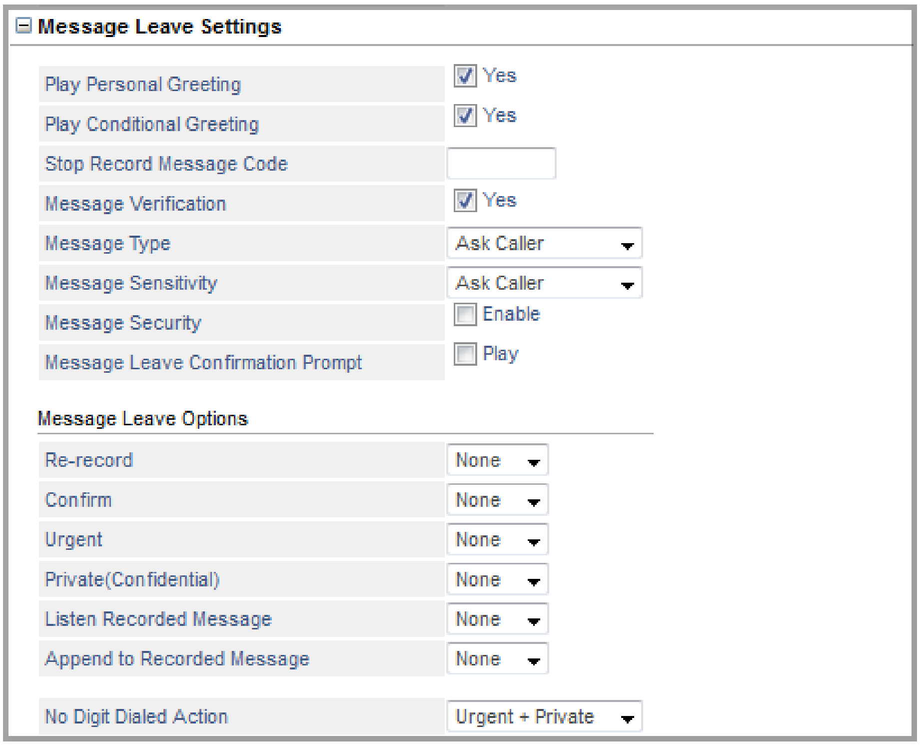
- Select **Set as Private**, if you want the system to store the messages received in the mailbox as Private. The message left by the external callers or replied by the internal callers will be considered as confidential and forwarding the message will be restricted.
- Select **Ask Caller**, if you want 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.

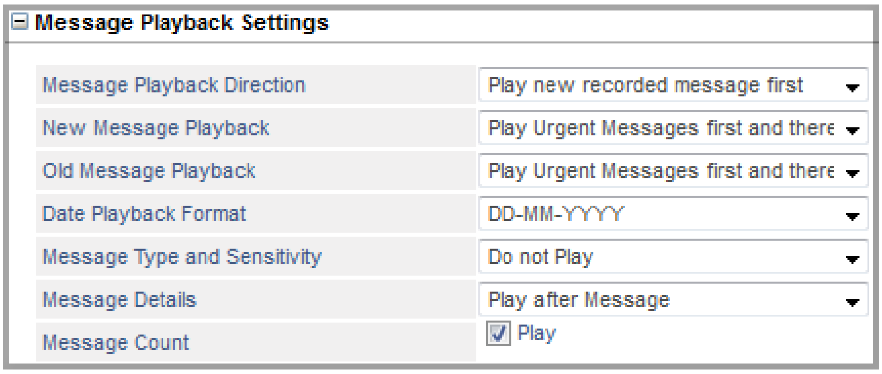
- In **No Digit Dialed Action**, if no digit is dialed by the caller, the system will take action as per the Message Type and Message Sensitivity configured.

  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.

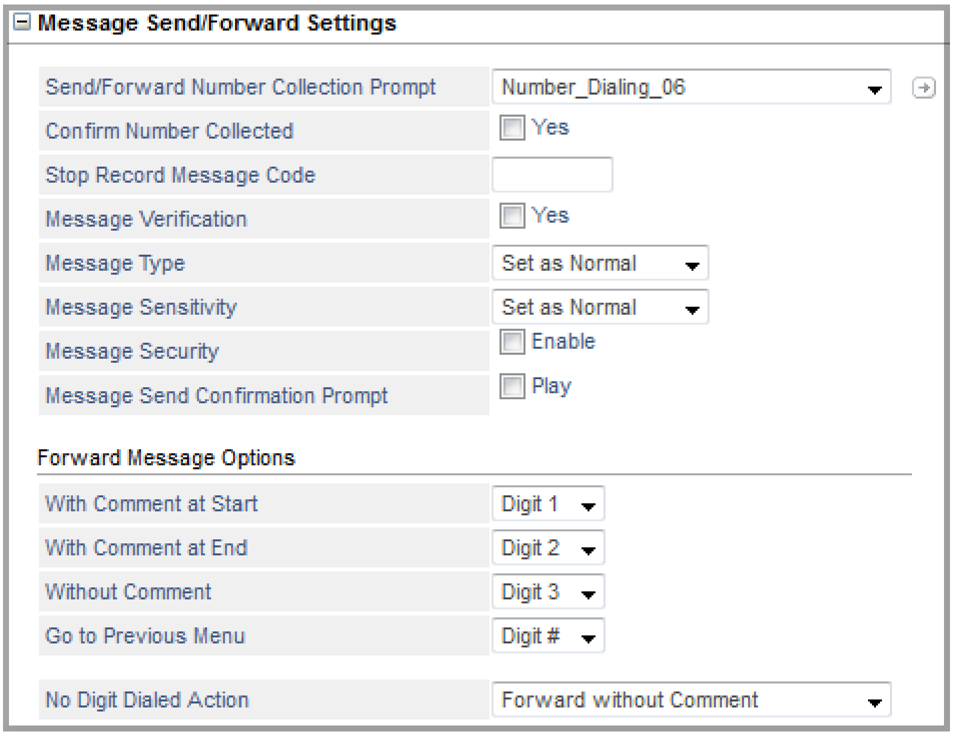
- Select the **Message Count** check box if you want the system to play the Message sequence number to the extension user before playing the message. The message count is just to differentiate between various messages played to the caller.

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 dialed by 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".

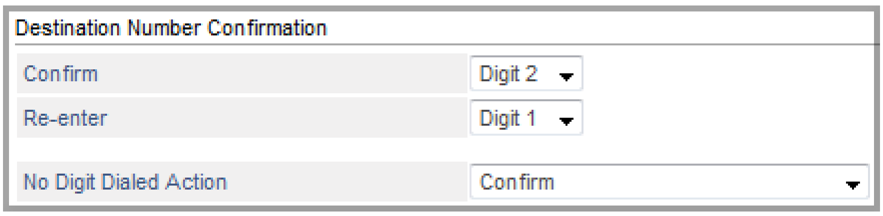
- The **Message Security** parameter is reserved for future use. Keep this check box disabled.

- 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.

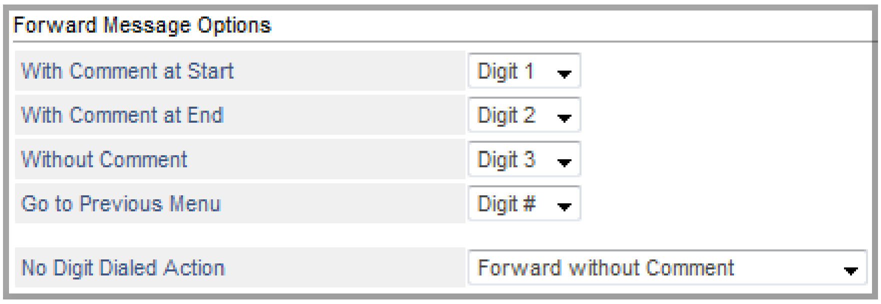
Make sure you do not assign the same digit to both the options.

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 enter 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:

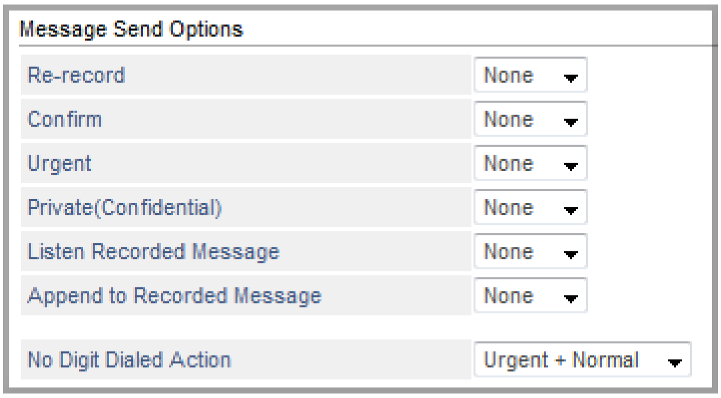
- **With Comment at Start**: The system will provide an option to the extension user for recording a message in addition to the existing recorded message before forwarding the message to the destination number/s. The additional message will be added as the initial part of the message i.e. will be added 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.

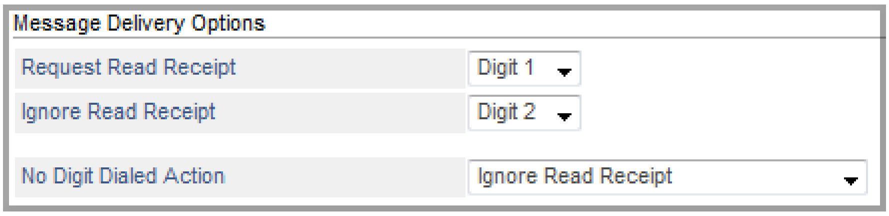
The Message, that is, the recorded message along with the appended message cannot exceed the Maximum Message Length. Hence, the system will allow recording of the appending message only till it reaches the Maximum Message Length.

- 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.

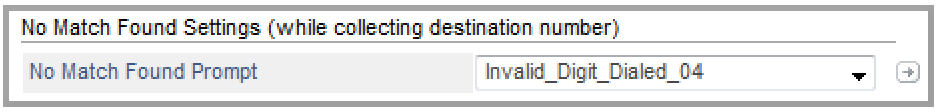
  As soon as the destination number reads the message, the extension user will receive a read receipt in form of a voice mail. To retrieve the read receipt, access your voice mail. The read receipt includes the name, destination number and the first five seconds of the message sent or forwarded.

In case of multiple destination numbers, the extension user will receive multiple read receipts in form of a voice message after the message has been read by the destination numbers.

- **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.

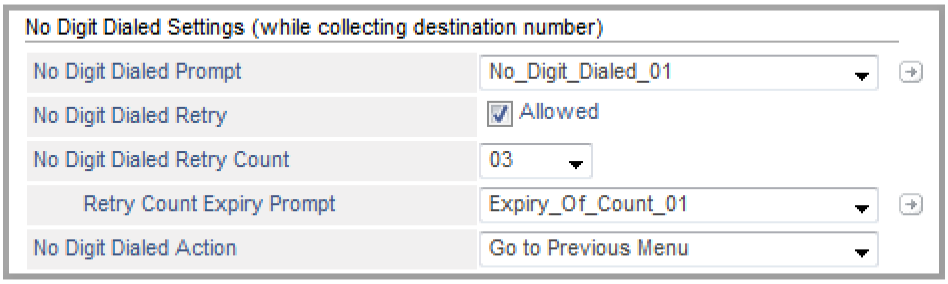
  - Click **Settings** . The **Prompts Management** page opens.

    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

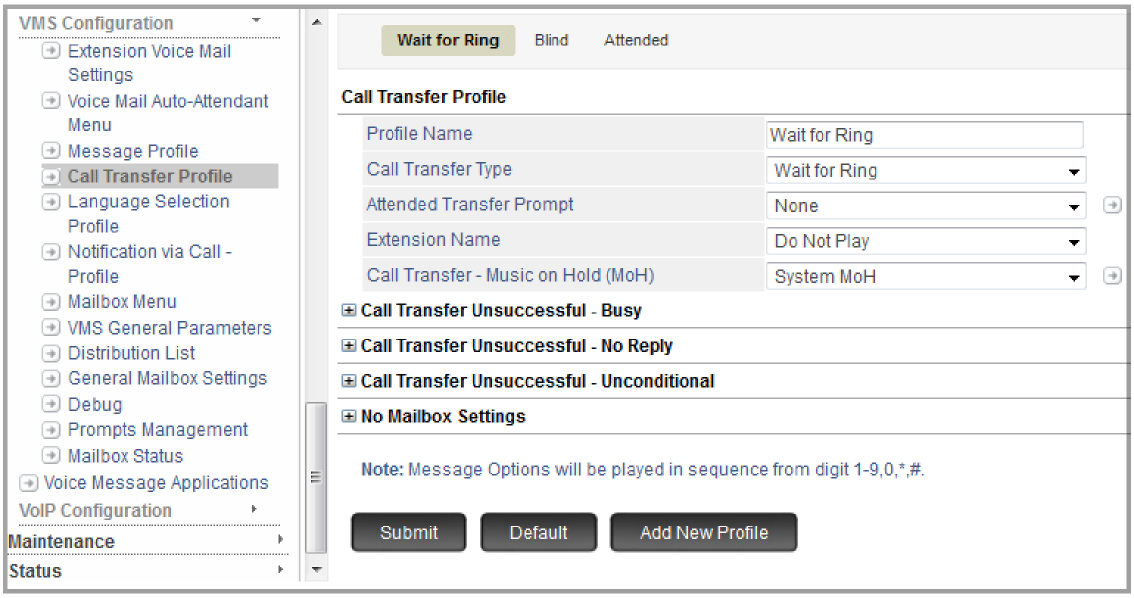
Call Transfer Profile allows you can select the Call Transfer Type — Blind, Wait for Ring, Wait for Answer, Screened, None — and customize parameters related to it as per your requirement.

VMS supports a maximum of 32 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

- Login as System Engineer.
- Under **Advanced Settings**, 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 transfer target'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 Transfer target 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**,

    - 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 **Transfer to Department Group** or **Transfer to Extension**,

    - Select the respective Department Group number or enter the desired extension number.

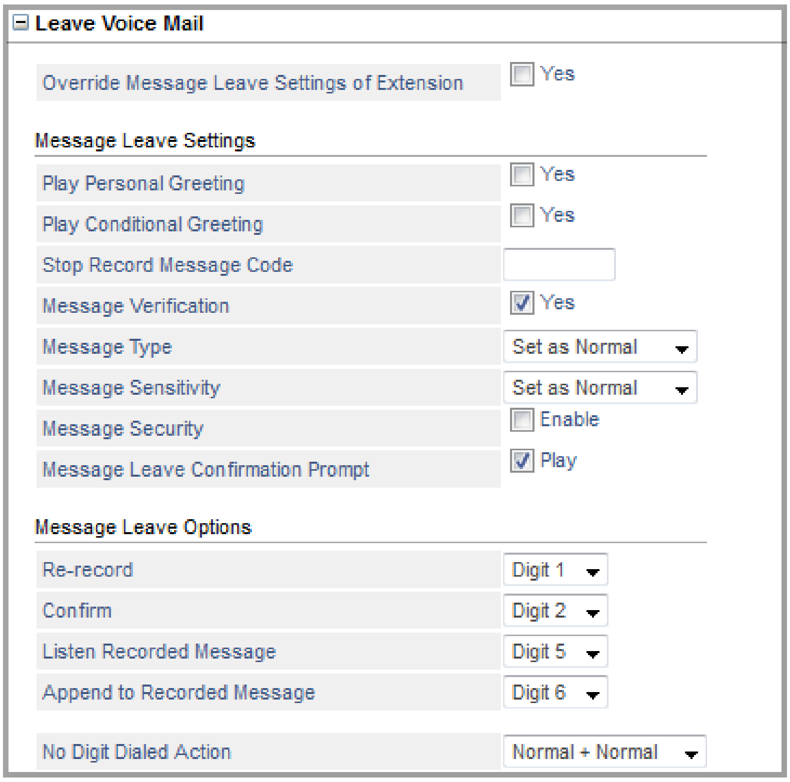
    - In Call Transfer Profile, select the desired Call Transfer Profile as per your requirement. By default, *As configured for Transfer Number* is selected.

  - 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:

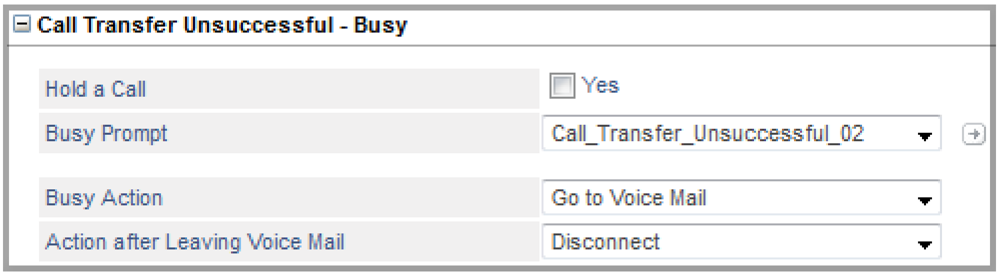
- process the call immediately for the busy condition as per the options you select.
  or
- process the call after being held for a certain duration as per the options you select.

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 **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 Transfer Number** 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 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 Department Group** or **Transfer to Extension**, select the respective 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 Department Group** or **Transfer to Extension**, select the respective 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.

  - 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 Department Group** or **Transfer to Extension**, select the respective 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 Department Group** or **Transfer to Extension**, select the respective 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.

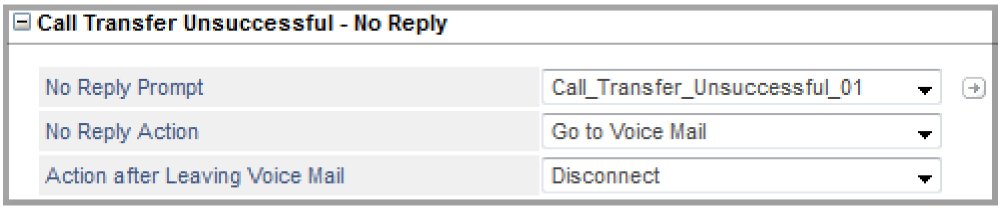
    Click **Settings** to configure the parameters of the selected profile.

  - 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 Department Group** or **Transfer to Extension**, select the respective 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.

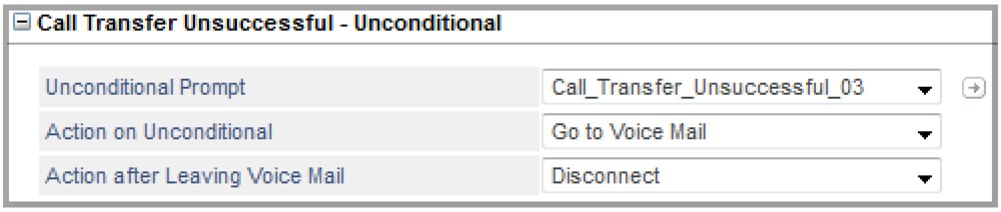
    Click **Settings** to configure the parameters of the selected profile.
  - If you select **Go to VMAA Menu**, select the desired VMAA Menu.
  - If you select **Disconnect**, the system will disconnect the call.
  - If you select **Give Advanced Option**, see "Advanced Options"
- In **Action after Leaving 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 Department Group** or **Transfer to Extension**, select the respective 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 Department Group** or **Transfer to Extension**, select the respective 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.

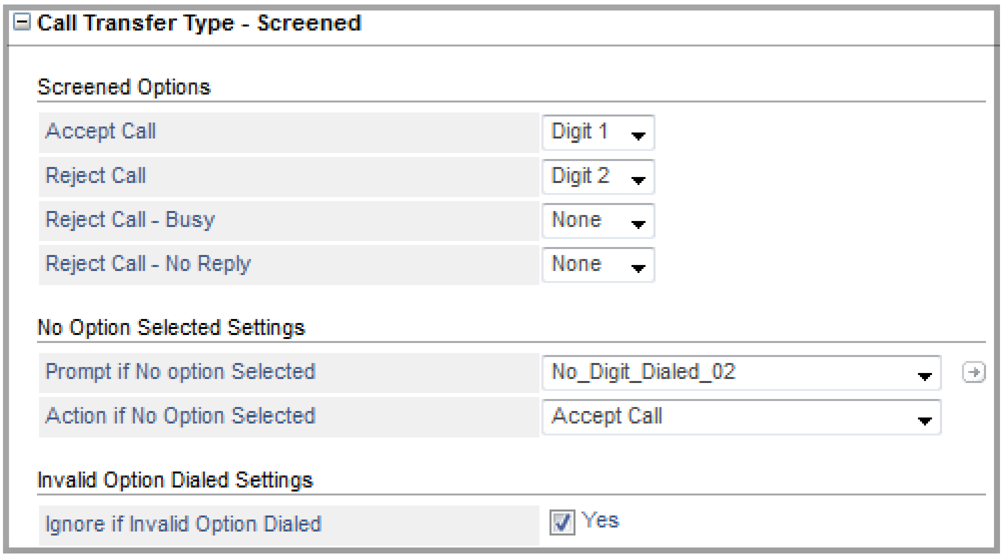
    Click **Settings** to configure the parameters of the selected profile.
  - If you select **Go to VMAA Menu**, select the desired VMAA Menu.
  - If you select **Disconnect**, the system will disconnect the call.
  - If you select **Give Advanced Option**, see "Advanced Options"
- In **Action after Leaving 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 Department Group** or **Transfer to Extension**, select the respective 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 Invalid Option Dialed,** 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.

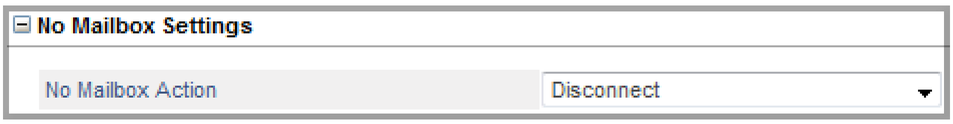
      You may add a new prompt from here. For further details, see "Prompts Management".

    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 Department Group** or **Transfer to Extension**, select the respective Department Group number or enter the desired extension number.

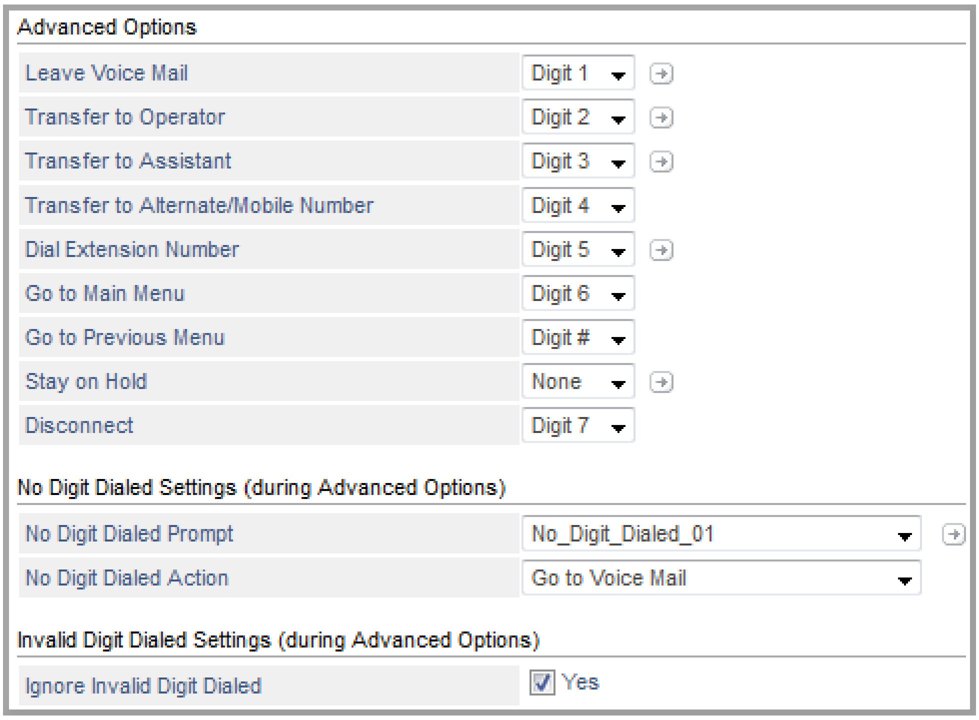
    In **Call Transfer Profile**, select the desired Call Transfer Profile as per your requirement. By default, *Wait for Ring* is selected.

    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.



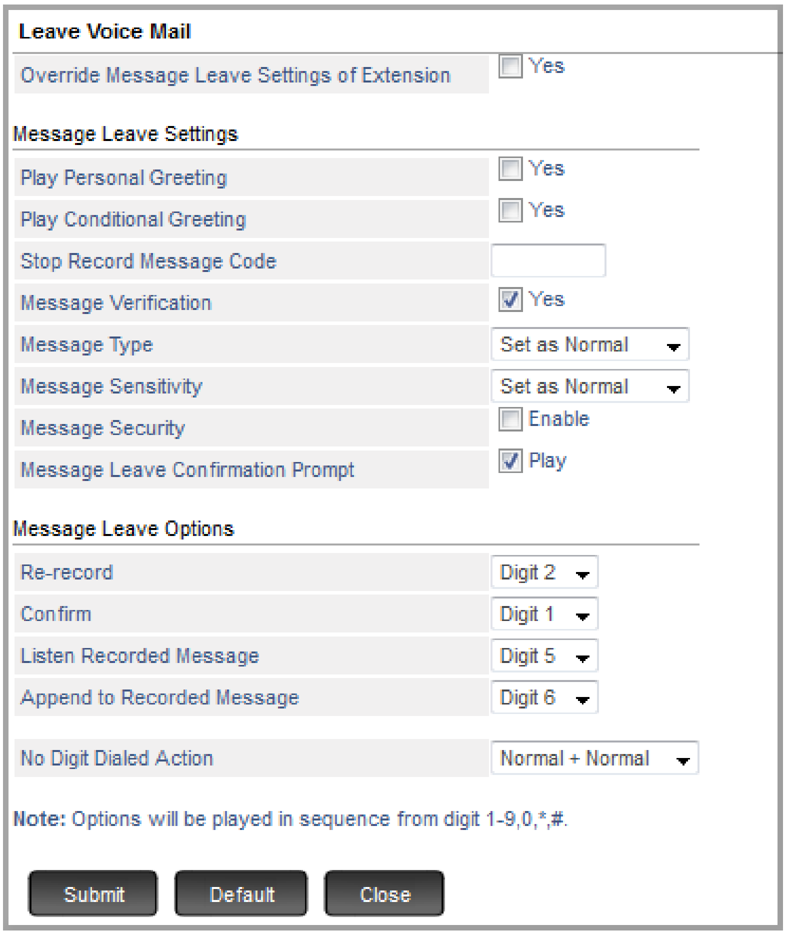
Make sure you do not assign the same digit to multiple options.

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

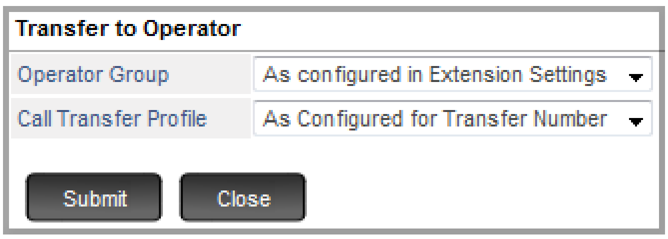


After you assign the digit, click **Settings** to configure the parameters.

- 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

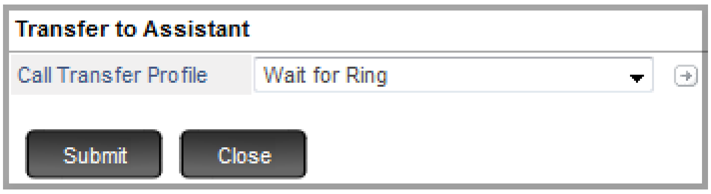


After you assign the digit, click **Settings** to configure the parameters.

- In **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.

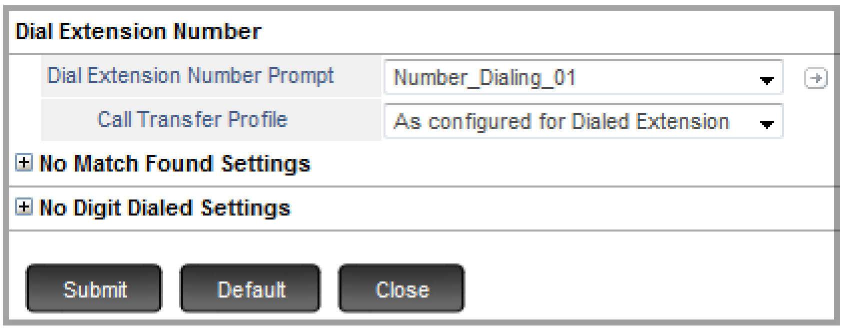
  - In **Call Transfer Profile**, select the desired Call Transfer Profile as per your requirement. By default, *Wait for Ring* is selected.

    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 Department Group** or **Transfer to Extension**, select the respective 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 Department Group** or **Transfer to Extension**, select the respective 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

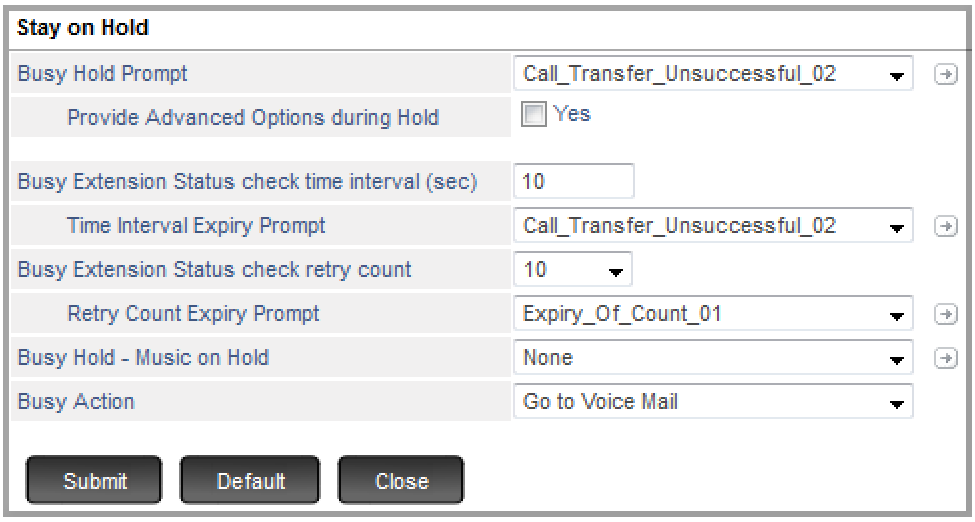
Assign a digit for this option if you want the system to re-direct the caller to the Main Menu.

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 **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 Department Group** or **Transfer to Extension**, select the respective 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 Department Group** or **Transfer to Extension**, select the respective 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 Department Group** or **Transfer to Extension**, select the respective 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.

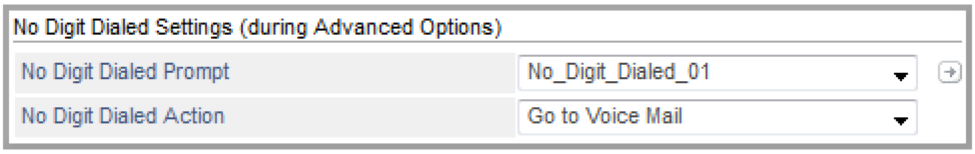
  - If you select **Give Advanced Option**, see "Advanced Options".

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.

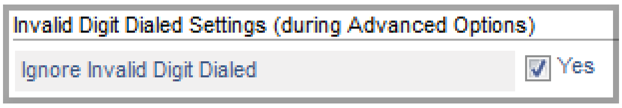
    You may add a new prompt from here. For further details, see "Prompts Management".

  Once the files are uploaded these appear as options for No Digit Dialed Prompt.

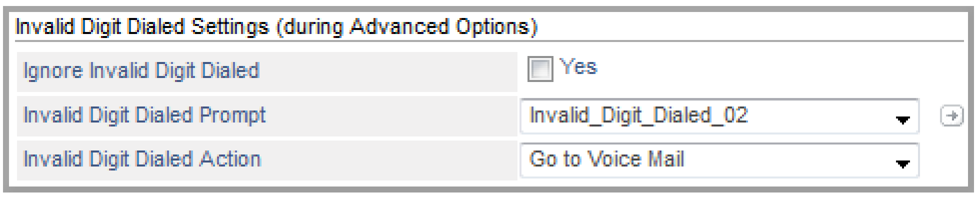
- In **No Digit Dialed Action**, you can select — Disconnect, Go to Voice Mail, Transfer to Operator, Transfer to Assistant, Transfer to Alternate Number, Go to Main Menu, Go to Previous Menu, Stay on Hold. Refer to the details given above under "Advanced Options".

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”.

Notification via Call-Profile

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".

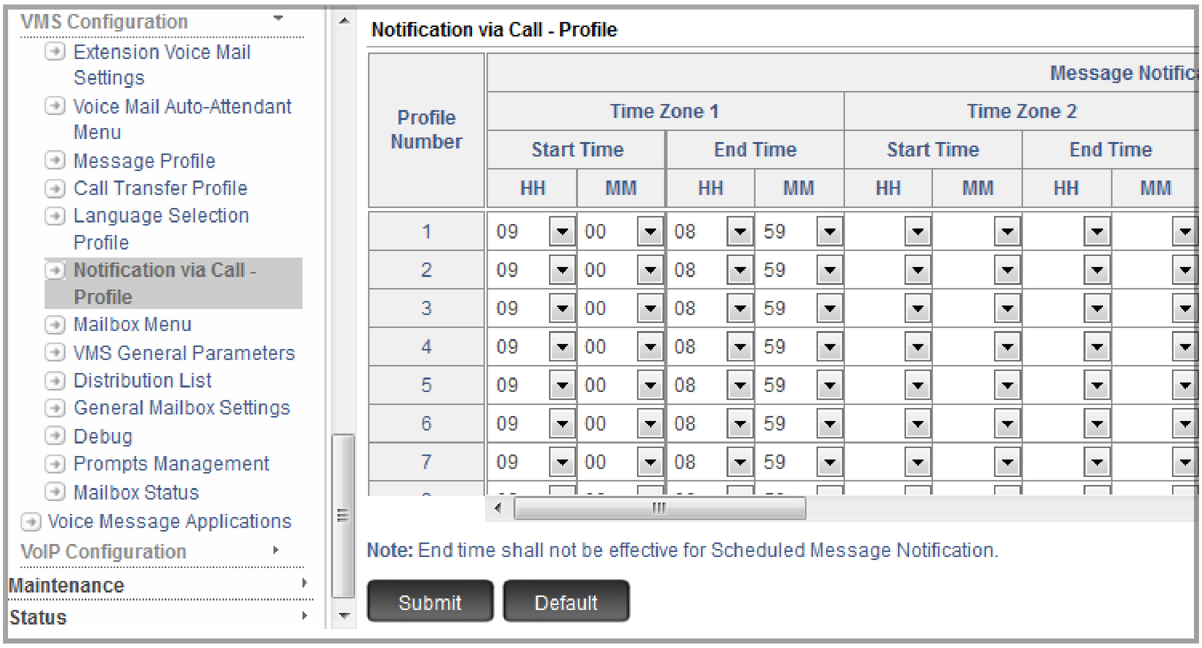
- if required, the Message Notification Retry Count, Message Notification Interval and Message Notification Ring. See "System Timers and Counts".

Configuring Notification via Call - Profile

- Login as System Engineer.

- Under **Advanced Settings**, click **VMS Configuration**.

- Click **Notification via Call - Profile**.



- 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 8 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**
  - 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.

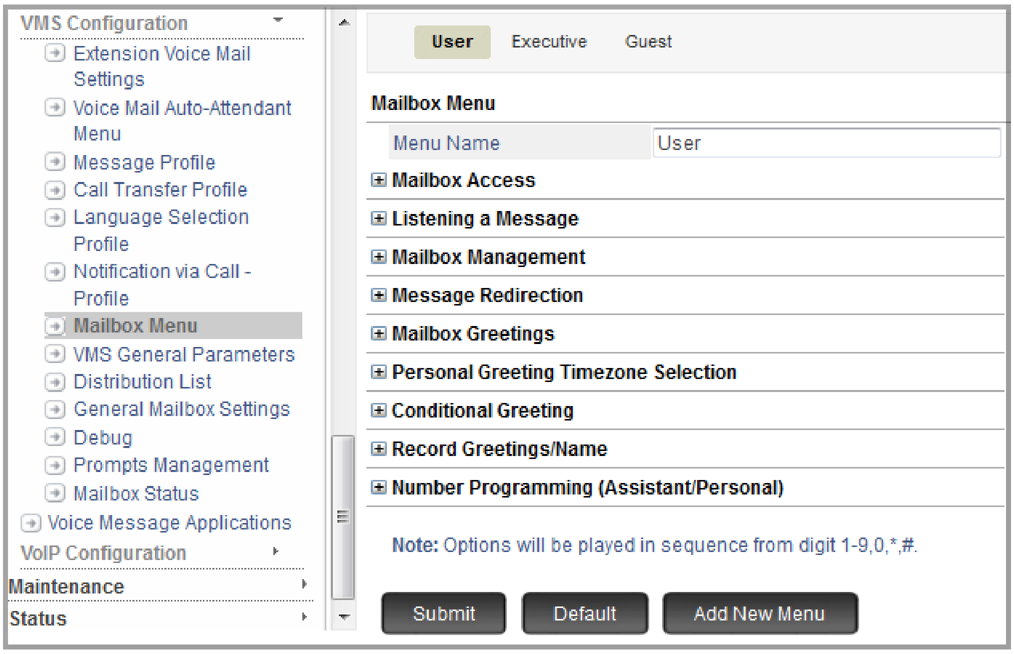
Mailbox Menu

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

- Login as System Engineer.
- Under **Advanced Settings**, 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 xx* 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.

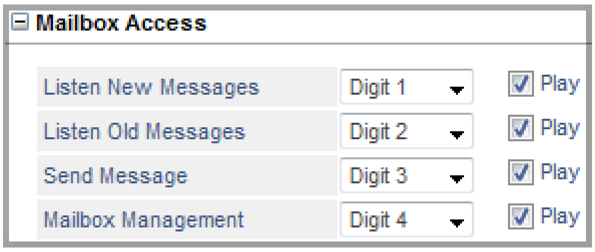
If you assign the digit to an option but do not select the Play check box for the same, then the option will not be played to the caller.

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".

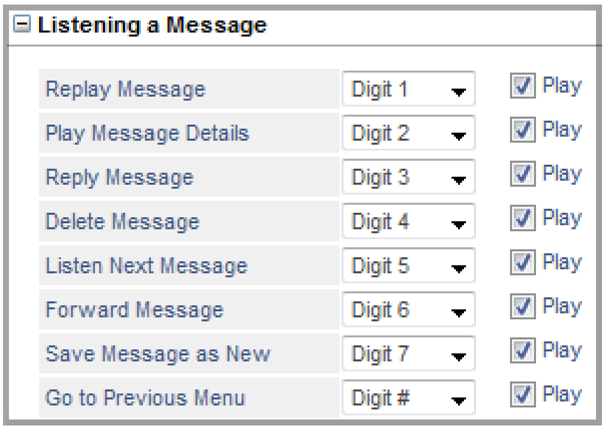
- **Listen Old Messages**: The system will play the old(read) messages present in the caller's personal mailbox. The old messages will be played as per the *Message Playback Settings* configured in the Message Profile. For details, refer "Message Playback Settings".

- **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".

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  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.

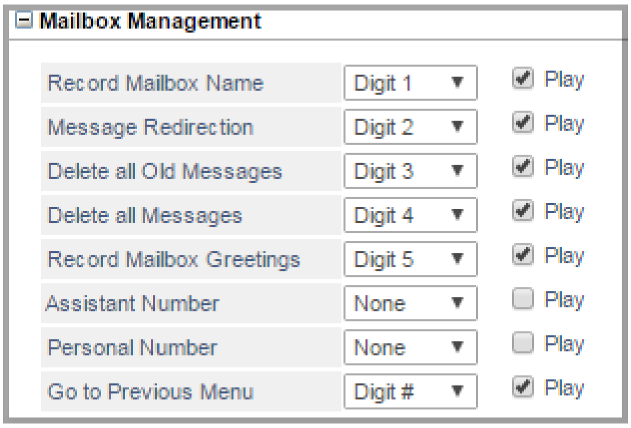
The message will not be forwarded if the *Message Sensitivity* is set as *Private*. For details, refer “Message Profile”.

- **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.

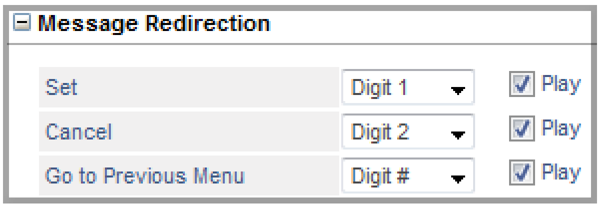
  The Personal Number is used as an alternate number. The caller can call this number when the user is not available to answer the call.

  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.

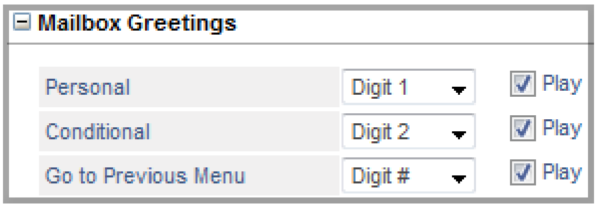
  The Message Redirect Number must be a valid Extension Number, Department Group Number. A Mailbox must be assigned to this Number.

- **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.



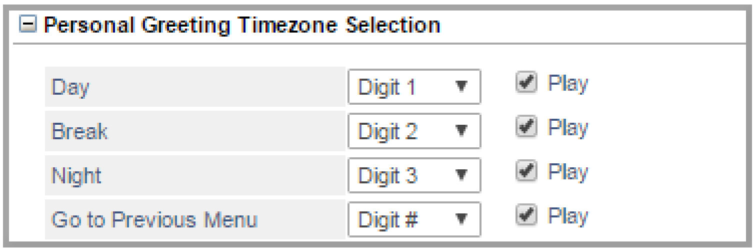
- **Personal**: The system will play the *Personal GreetingTimezone Selection* options. The caller must select the timezone — Working Hour, Break Hour or Non-Working Hour for which s/he wishes to Record, Play or Erase the Personal greetings. For details, refer “Personal Greeting Timezone Selection”.

- **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:

- **Day**: 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**: 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.

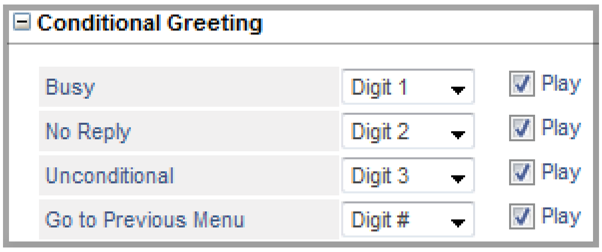
- **Night**: The system will play the *Record Greetings/Name* options and allow the caller to Record, Play or Erase a personal greeting for the Non-Working Hour Timezone.

- **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.

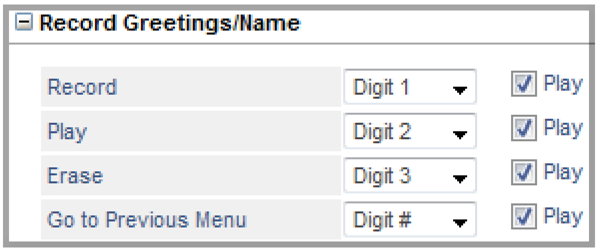
- **Unconditional**: The system will play the *Record Greetings/Name* options and allow the caller to Record, Play or Erase the Conditional Greeting of type — Unconditional.

- **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.

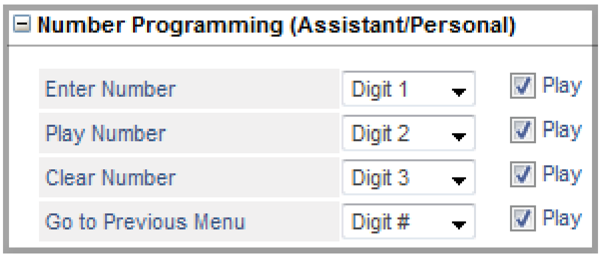
- **Play**: The system will Play the Mailbox Name or the Personal/Conditional Greeting to the caller.

- **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

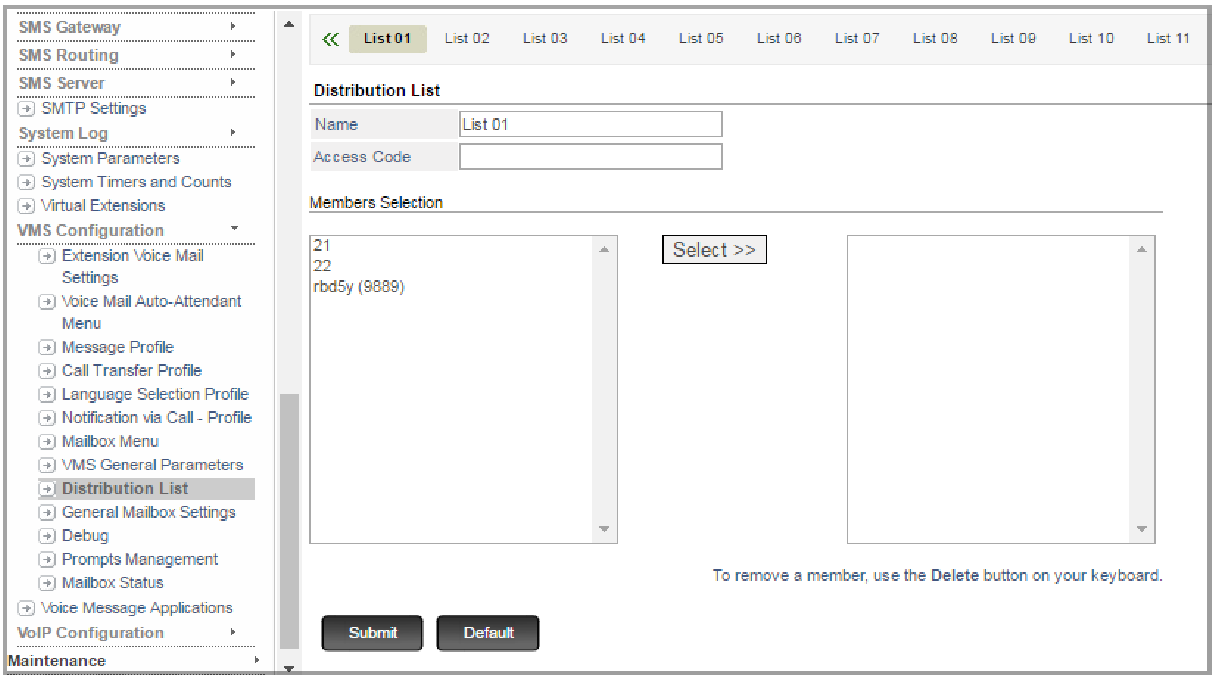
A Distribution List enables extension users to send the same message to a group of extensions at the same time.

Any extension with a mailbox can be included in a Distribution List. You can create 30 Distribution Lists of 50 members each.

How to configure

To configure Distribution List using Jeeves,

- Login as System Engineer.
- Under **Advanced Settings**, 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,

        - Select 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**.

Recording Voice Messages

The VMS of SARVAM UCS supports voice messages for different functions, which are broadly classified as:

- **System Greetings**: These are voice messages played 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 voice guidance messages played to the 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, i.e. 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 are voice guidance messages played to the 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 (i.e. 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. 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.

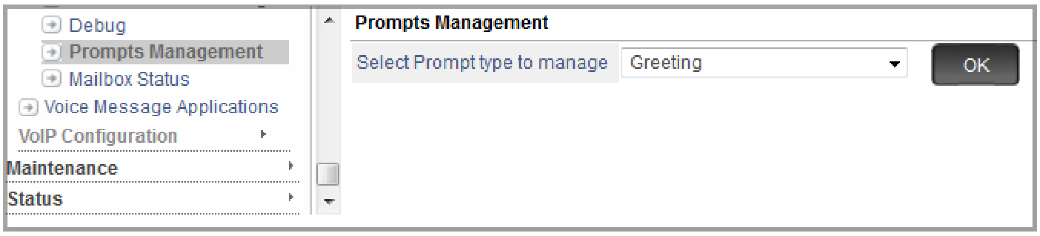
The following table defines the maximum number of prompt files supported in each folder.

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

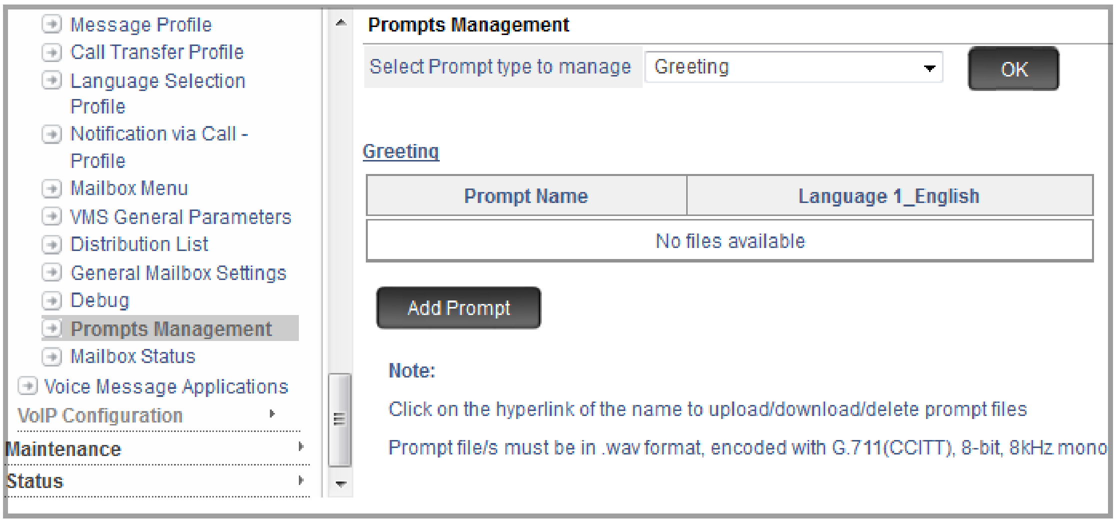
How to configure

- Login as System Engineer.
- Under **Advanced Settings**, 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.

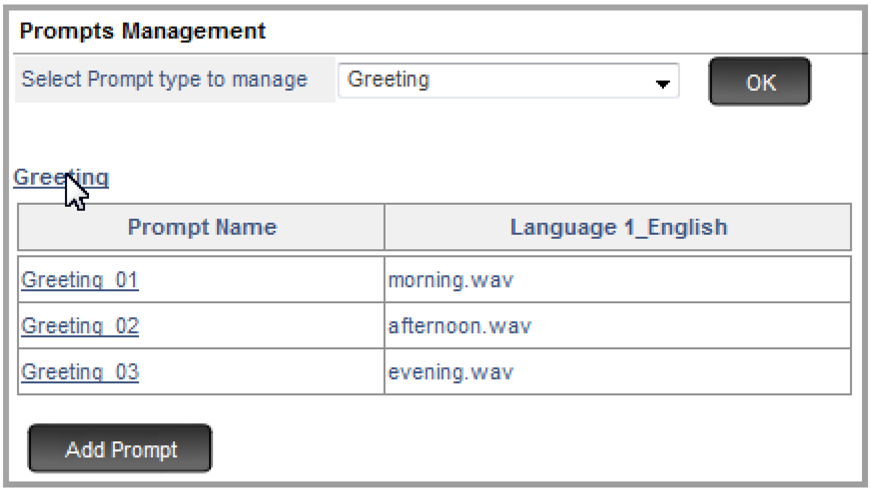
- Click to upload.

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 **Prompt Type**.



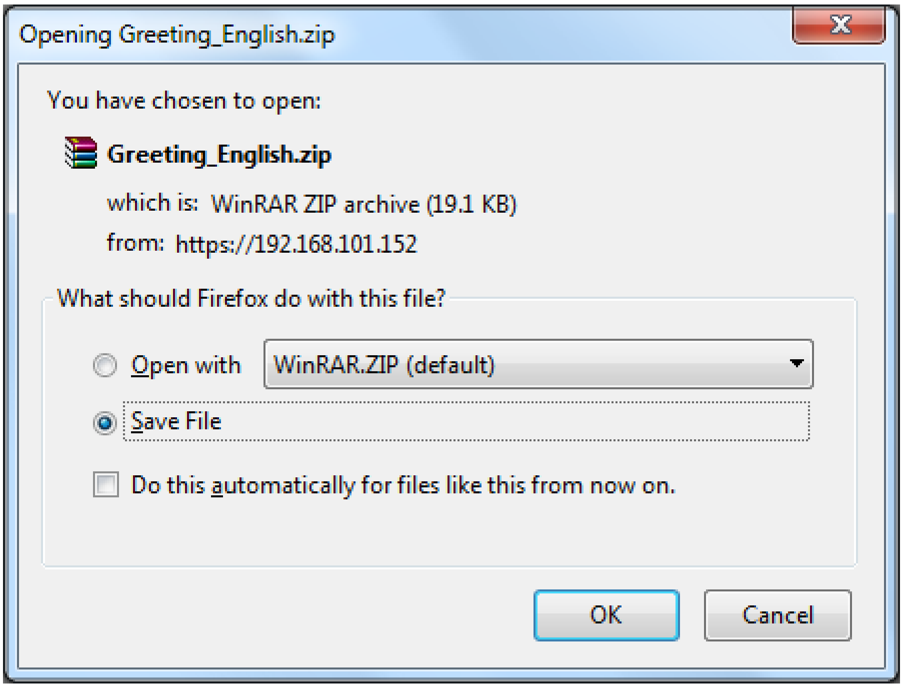
The **Upload/Download Prompt Type** Window opens.

- 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.

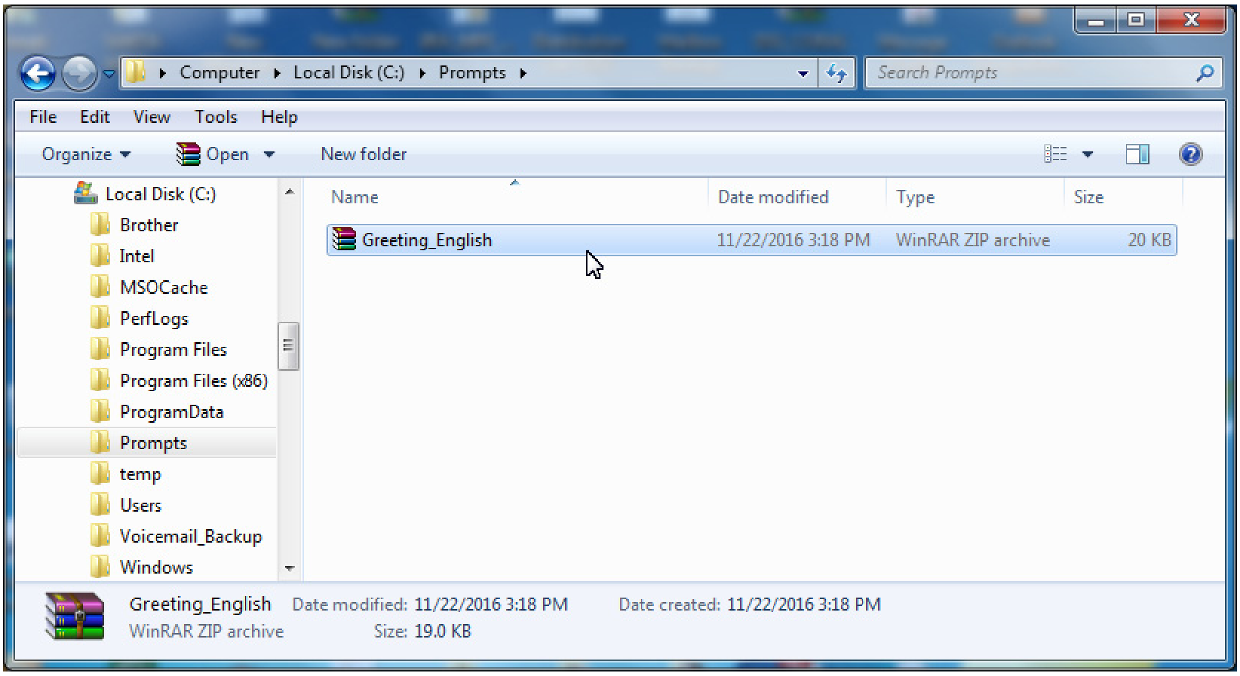


- You can either open the zip file or save the file to a location.

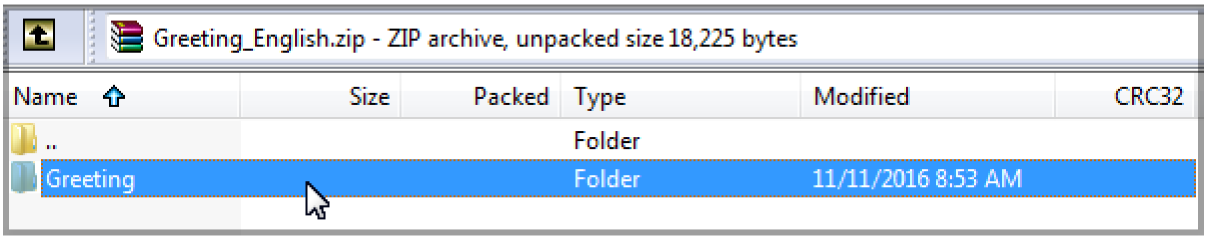
The above window display depends upon the browser you are using. Check the ***Download Settings*** *of your browser and set the Download path accordingly.*
OR
If your browser does not ask you for the location you want to save your file, it saves it in the default location according to the download path 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****.*

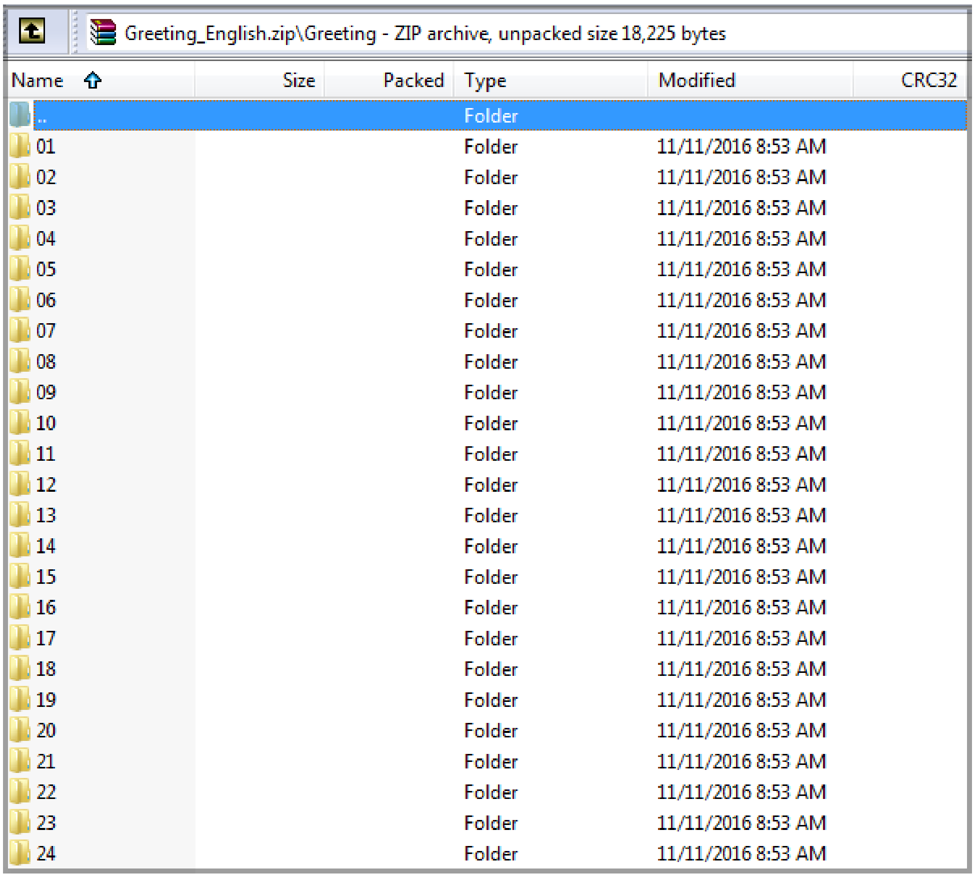
- Save the file on the local disk.



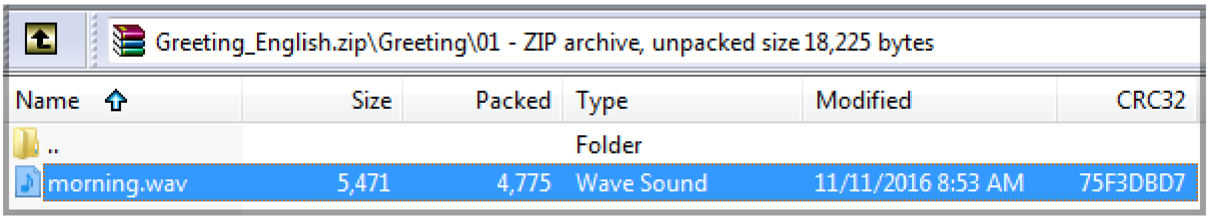
- To view the files,
  - Open the **Prompt Type_Language** folder.



- Now, open the **Prompt Type folder** to view all the subfolders containing the prompts.



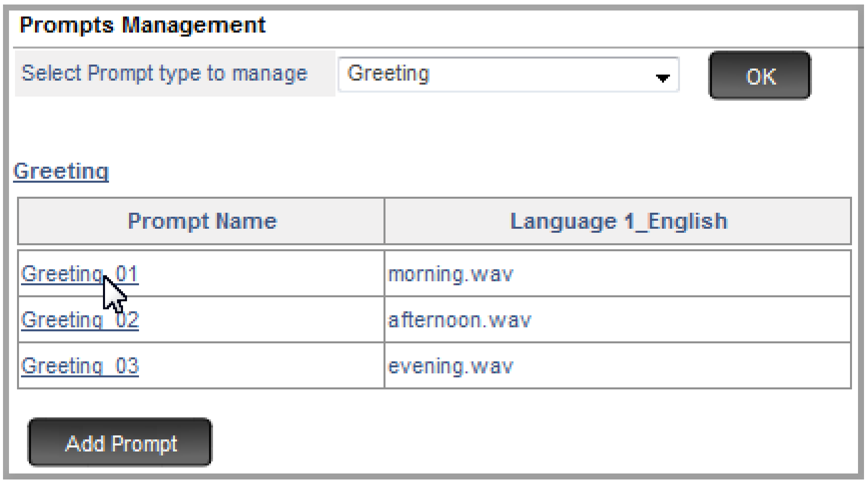
- Open the subfolder containing the prompt.



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.



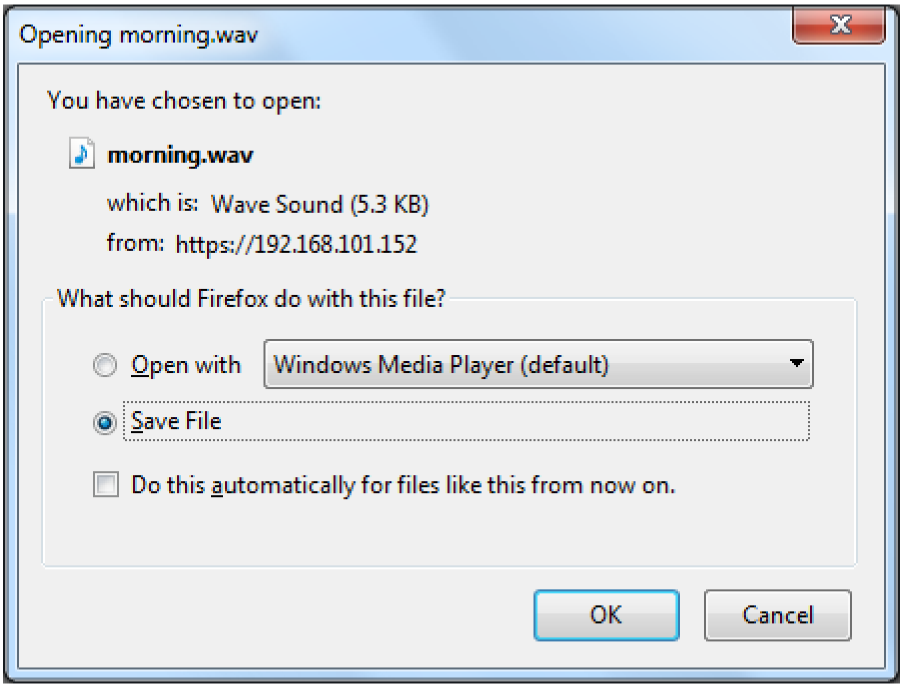
The **Add/Edit Prompt** window will open.

- 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.

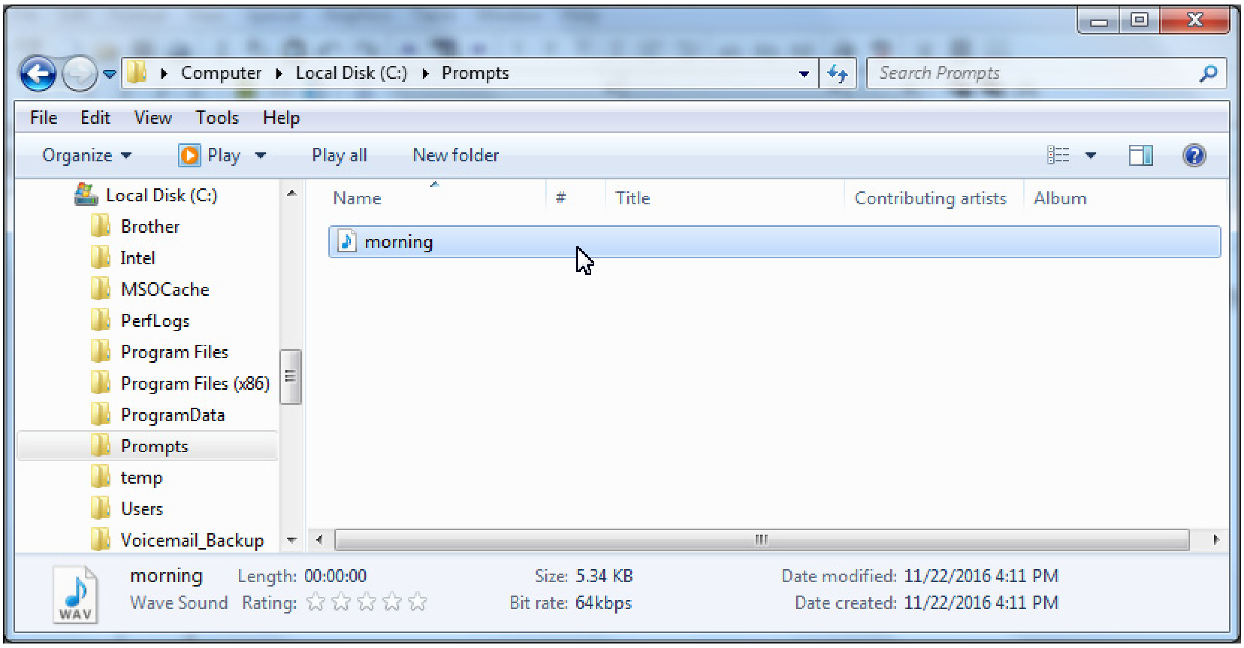


- You can either open the wav file or save the file to a location.

*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,

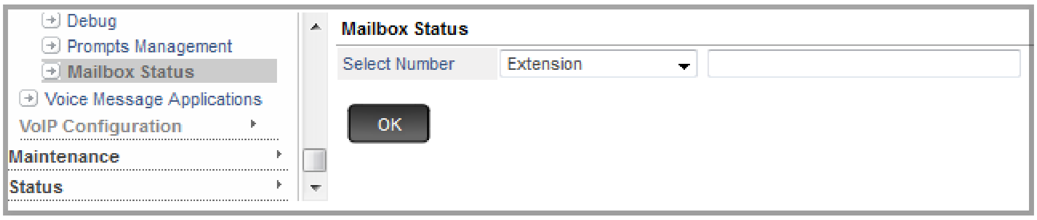
- *If you wish to upload customized prompts in the Miscellaneous folder, follow the same instructions as mentioned above to download the folder and copy the customized prompts in the desired folders.*
- *You also need to make sure that the Folder number 01 has only one file you need, delete the other file. Only then you will be able to upload the folder again with customized prompts.*

Mailbox Status

The Mailbox Status Page displays the mailbox details of the respective Extension, Department Group or General Mailbox. It also allows you to delete the voice messages from the mailbox assigned to these Extensions, Department Groups or General Mailbox.

How to configure

- Login as System Engineer.
- Under **Advanced Settings**, click **VMS Configuration**.
- Click **Mailbox Status**.



OR

- Login as System Administrator.
- Under **Voice mail**, click **Mailbox Status**.

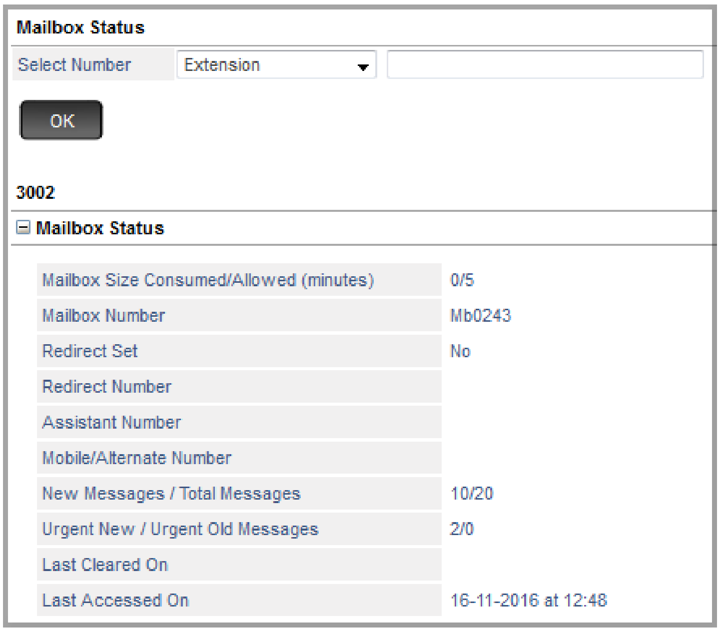
- **Select Number:** You may select — Extension, Department Group or General Mailbox — whose mailbox status you wish to view.
  - For **Extension**, enter the Extension Number/Name and then select the same from the drop down list.

- For **Department Group**, select the desired Department Group Number from the drop down list.

- Click **OK**.

Mailbox Status

- 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.

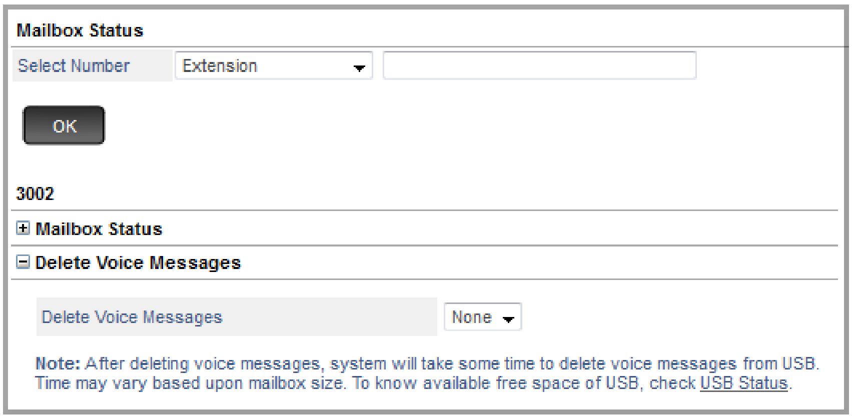
- **Last Cleared On**: It displays the date and time when all the Mailbox Messages (old and new) were last cleared by the user or SE.

- **Last Accessed On**: It displays the date and time when the Mailbox was last accessed.

Delete Voice Messages

The option to delete the voice messages for Extension, Department Group or General Mailbox will appear only when the mailbox assigned to these contains messages.

- Click **Delete Voice Messages** to expand.



- **Delete Voice Messages**: Select the type of voice message, you wish to delete — None, All, New, Old. By default, None is selected.
- Click **Delete**.

  A confirmation message appears. Click **OK** to delete the messages.

  After deleting the voice messages, you can now click **USB Status** to check the space available.

- *Depending upon the size of the mailbox you delete, the system may take some time to clear the deleted messages from the USB. The USB Status will be updated only after these messages are cleared.*
- *You can also view the Mailbox Status or delete the voice messages from the SE Mode. To do so, click **Mailbox Status** under **Status** and following the steps described above.*

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”.

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

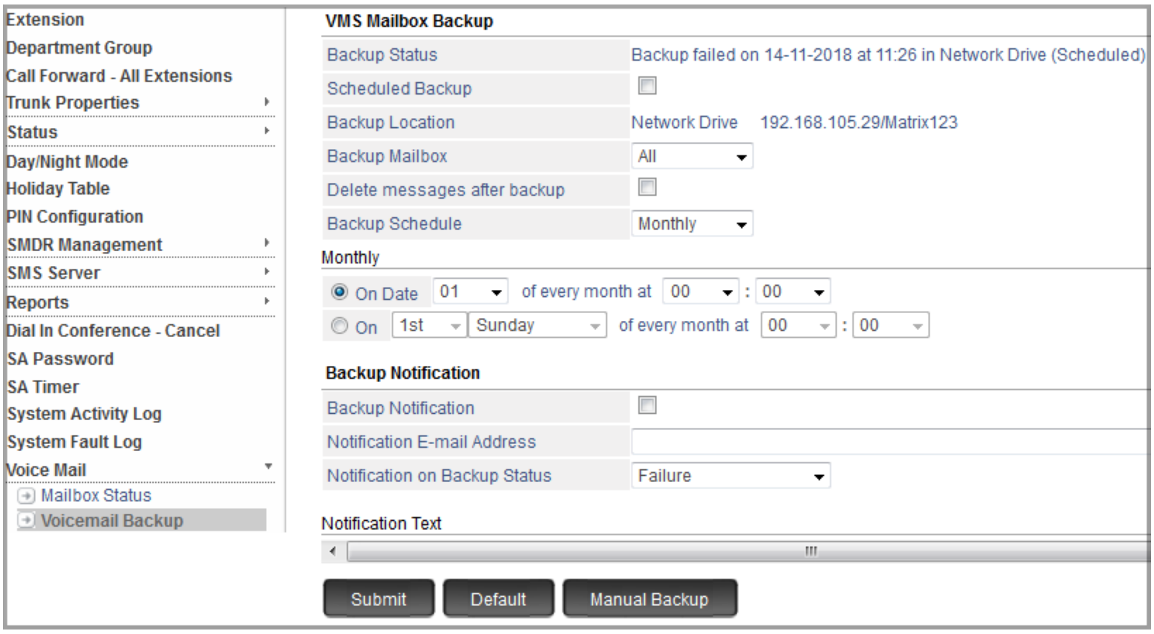
- Manually: The backup is taken whenever you want.
  Or
- as per Schedule: The backup is taken on a preset Day, Date and Time.

- *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

- Login as System Administrator.
- Click **Voice Mail**.
- 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.

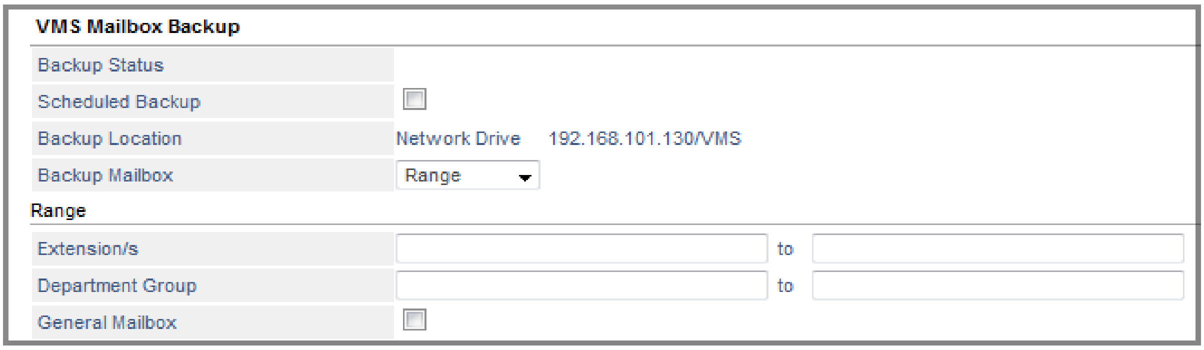
If storage capacity of the Network drive is full, then backup will not be taken and an error is displayed.

- In **Backup Mailbox,** 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**.

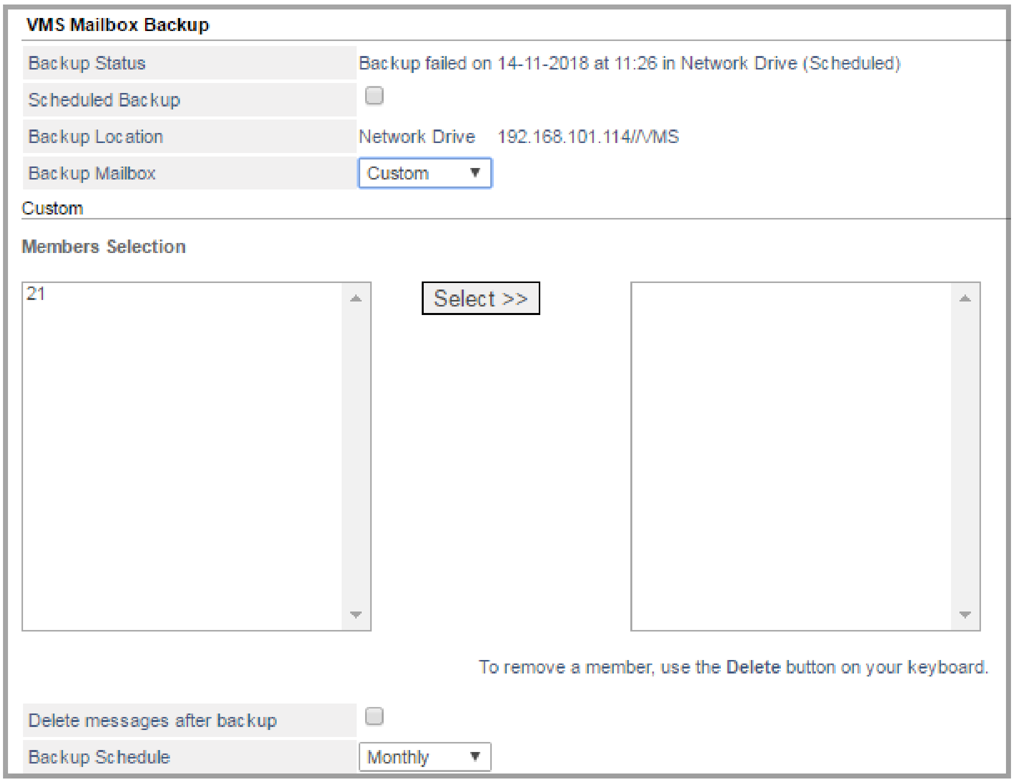


- To backup voicemail of a range of Personal mailboxes, enter the starting and ending **Extension** Numbers and then select the same from the drop down list.

- 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 on the right side panel.

  - To remove a selected member, click on the desired extension on 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 on 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>.**

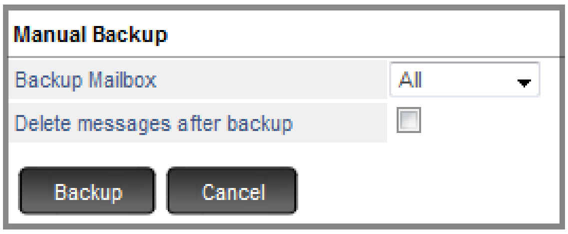
  Date-Time will be replaced with the date and time of the system when manual/scheduled Voicemail Backup was taken.

Manual Backup

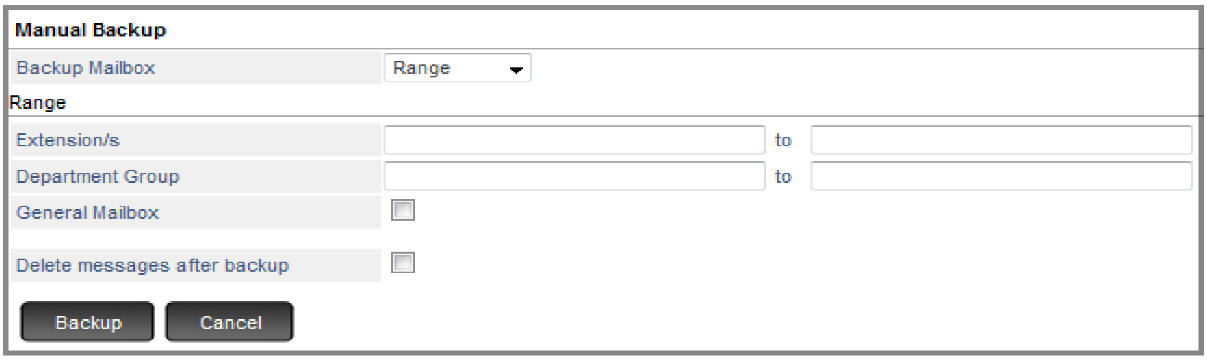
If the Network drive is not configured, then Manual Backup option will not be displayed.

To take the Voicemail backup manually,

- Click on **Manual Backup** button. A new window for Manual Backup is opened.



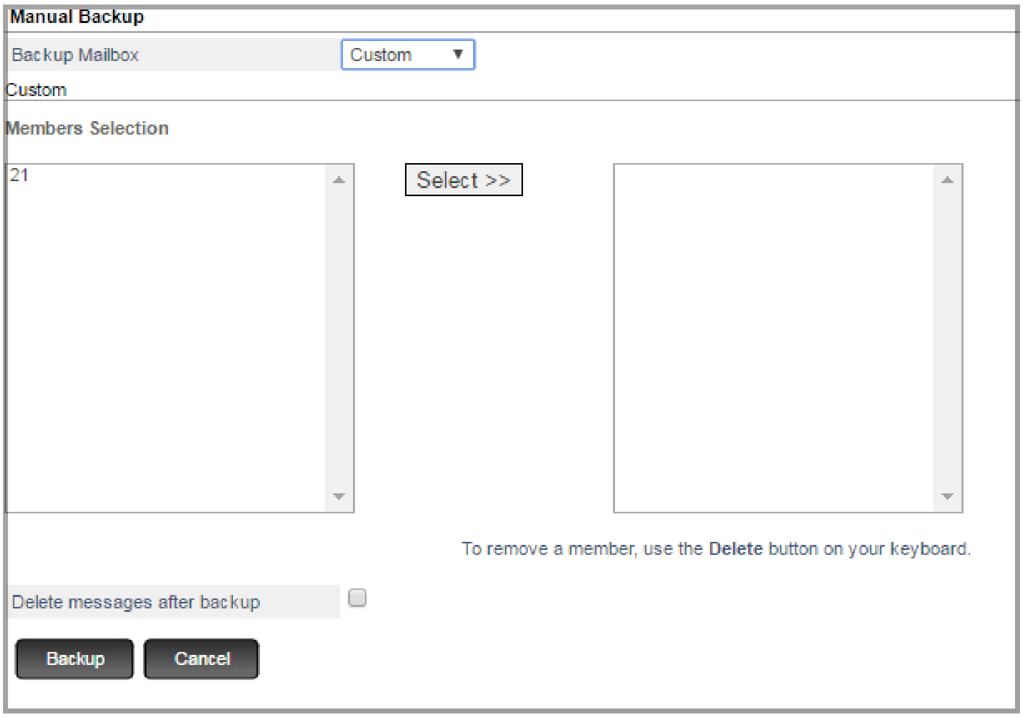
- **Backup Mailbox** allows you to select the mailbox/es for which the backup needs to be taken.
  - To backup voicemail of all extensions, select **All** Extensions. This would take the backup of all mailboxes — Extensions, Department Groups and General mailbox. By default, All option is selected.
  - 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.

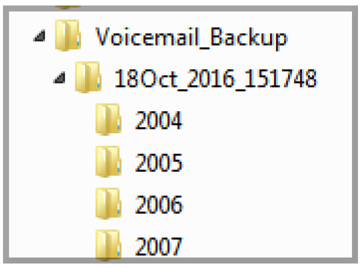
- Select the **Delete messages after backup** check box if you want the system to delete the respective extension/s voicemail messages after their backup has been stored on the network drive.

*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 contain the respective extension's voicemail files.

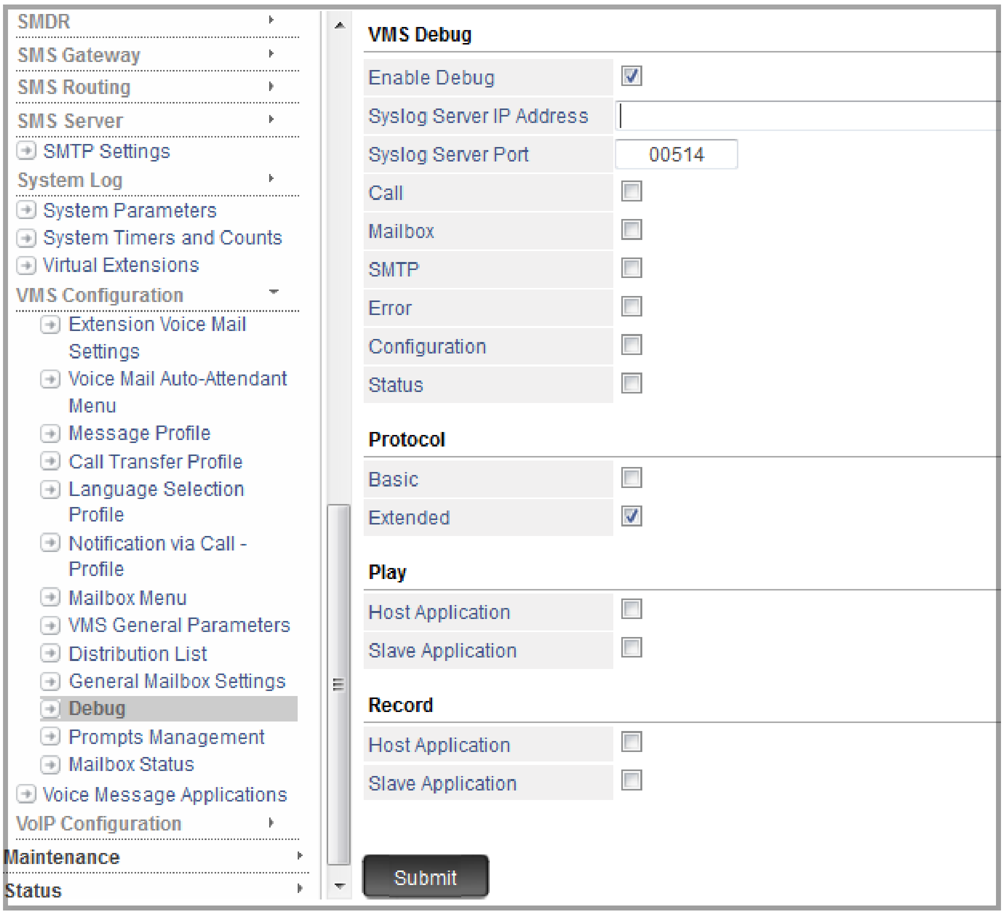
VMS Debug

The VMS supports debug for the VMS Application and SMTP. You can view debug messages on the Syslog Server[98].

To be able to use Syslog for debug, you will need to configure the Syslog Server Address and the Server Port on which Syslog will listen for the debug messages.

How to configure

- Login as System Engineer.
- Under **Advanced Settings**, click **VMS Configuration**.
- Click **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.

98. *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.*

- In **Syslog Server Port**, configure the address of the Listening Port of the Syslog Server. By default, the remote server port address is 514. Valid range of the port is from 1025 to 65535 and 514.

If you disable the check box, the system will not send the VMS Debug to the Syslog Server.
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**.

You can also configure the VMS Debug from the Maintenance link. To do so, click VMS Debug under Maintenance.

CHAPTER 7

SMS Server

SMS Server application of SARVAM UCS, 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”.

The SMS Server application works as an intermediary between the GSM Short Message Service and the SARVAM UCS. The Server supports multi-part, 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 to forward IM from SIP Extensions as SMS, see “SMS over IP”.*

How it works

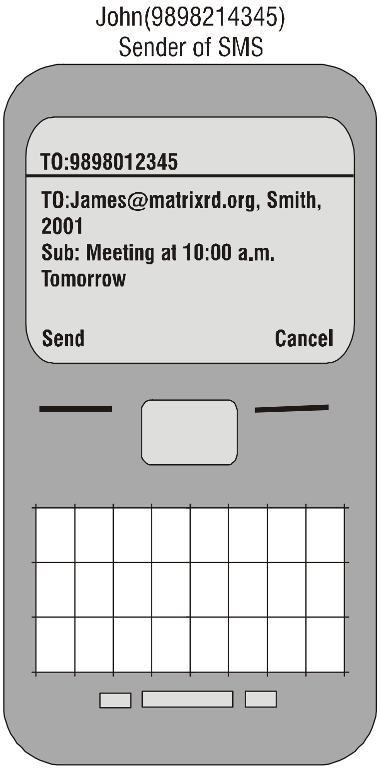
For this feature to work,

- 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 “SLT Extensions”, “SIP Extensions”. “Configuring Personal Directory” and “Configuring Global Directory”.
- 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 “Mobile Trunks”.

- you must configure the SMS Server parameters as well as the multi-part 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 ETERNITY, 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.

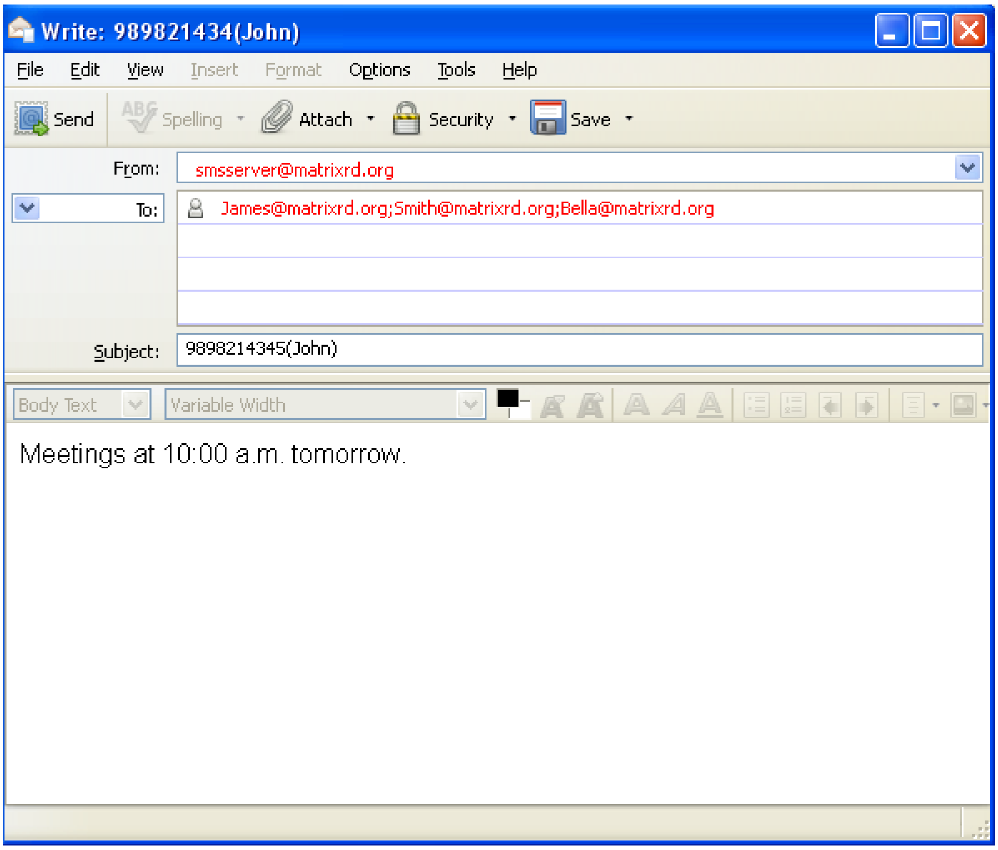
The destination can be the recipients Extension number, Name/Group Name or Email ID. The **To** field in the message body specifies the destination. To send the same message to multiple destinations, enter the destinations separating them with a comma or semicolon.

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.

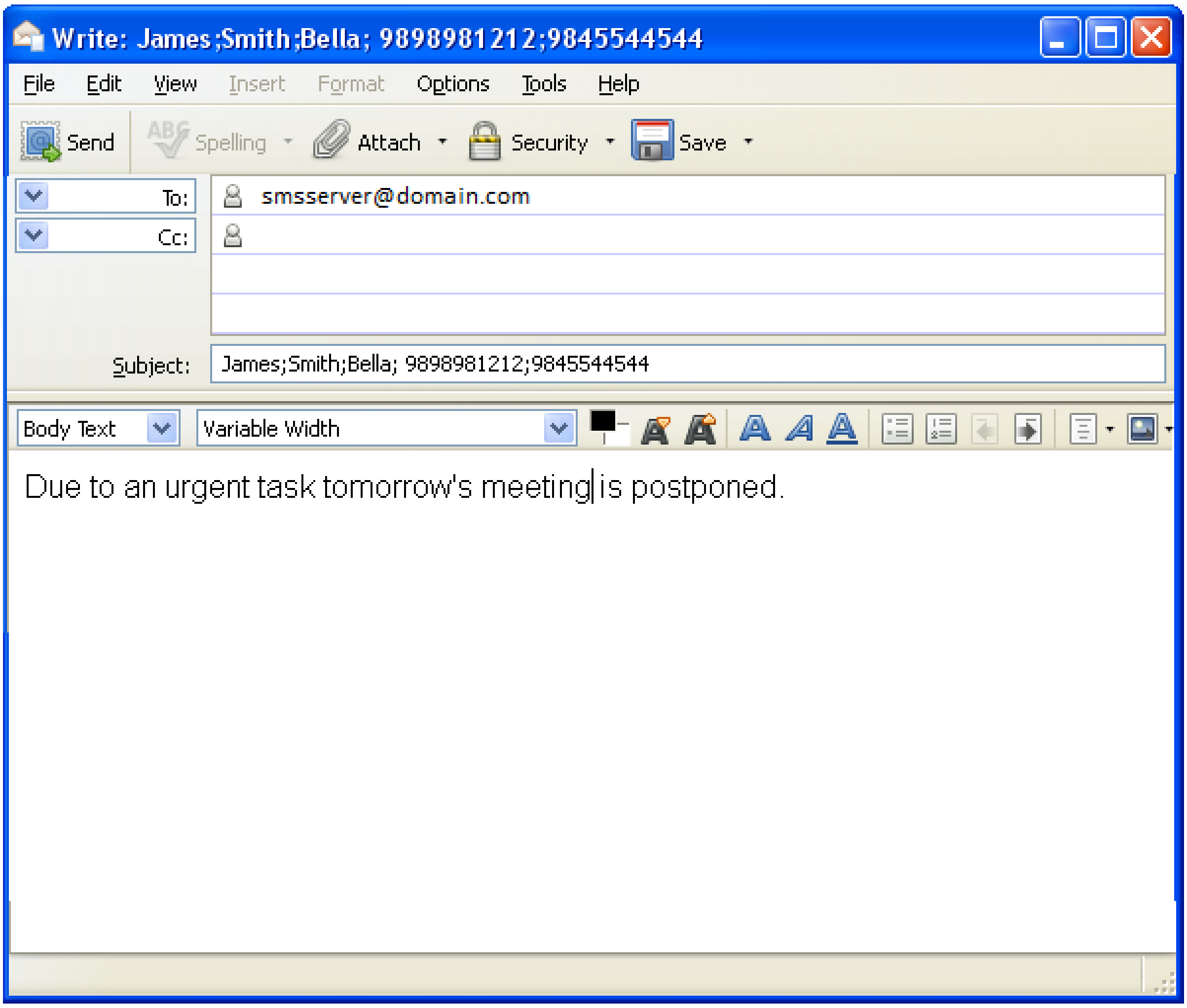


In this Email, the **From** field contains the Email address of the SMS Server. The **To** field contains the Email IDs of the recipients, the **Subject** contains the Name and/or Number of the sender and the **Body** of the message contains the message for the recipients.

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,

- The sender of the Email is John, John@matrixrd.org.
- The **To** field must be the Email ID of the SMS Server, smsserver@domain.com.
- 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.
- The **body** must contain the message to be sent to the recipients, that is ‘Due to an urgent task tomorrow’s meeting is postponed.’
- 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.

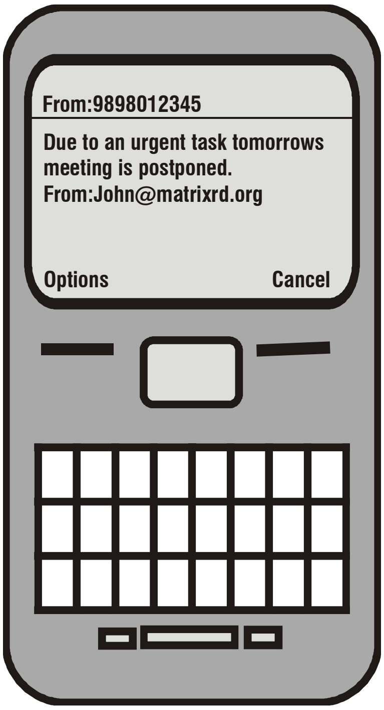
  If the destination is a Number, the server will check the number in the Denied list. If a match is found the email will be rejected. If a match is not found, the Server will send the SMS directly to the number using the Mobile Port of ETERNITY NENX.

  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".
- Minimum time delay 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 List for sending Bulk SMS**, only if you have enabled the **Allowed Email IDs to send Bulk SMS** check box. If the **Allowed Email IDs to send 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.

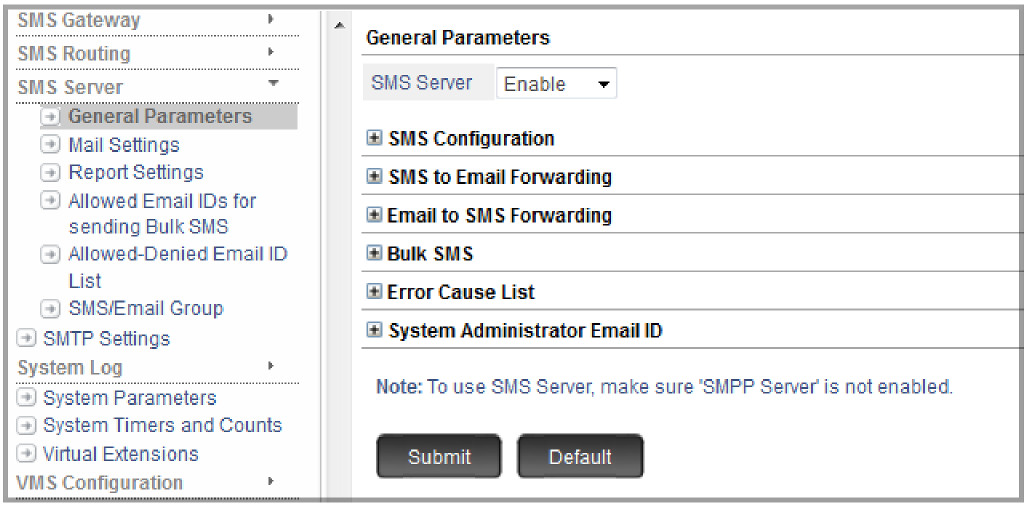
How to configure

For the SMS Server to function, you need to configure the following parameters,

General Parameters

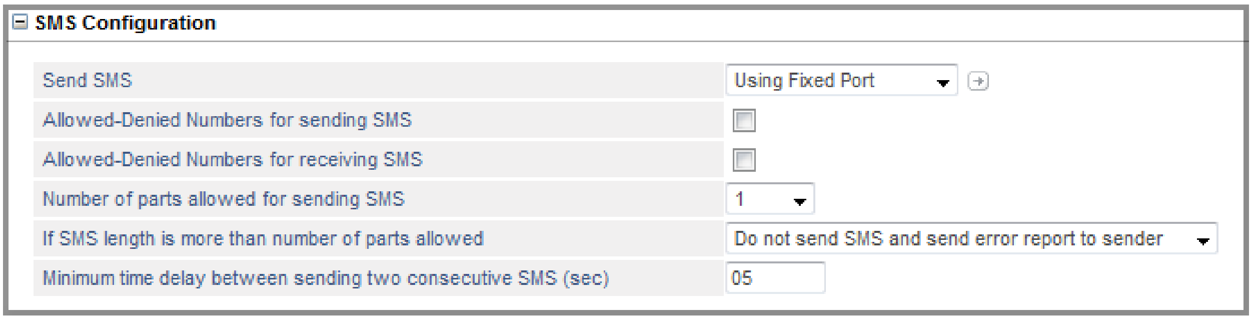
- Login as System Engineer.
- Under **Advanced Settings**, click **SMS Server**.
- Click **General Parameters**.

- 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 a fixed Mobile Port or through different Mobile Ports according to specific numbers and time.

  To send messages through a fixed Mobile Port, 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 through certain preferred Mobile Port/s during a defined time interval, 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 up to 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 up to 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 multi-part 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.

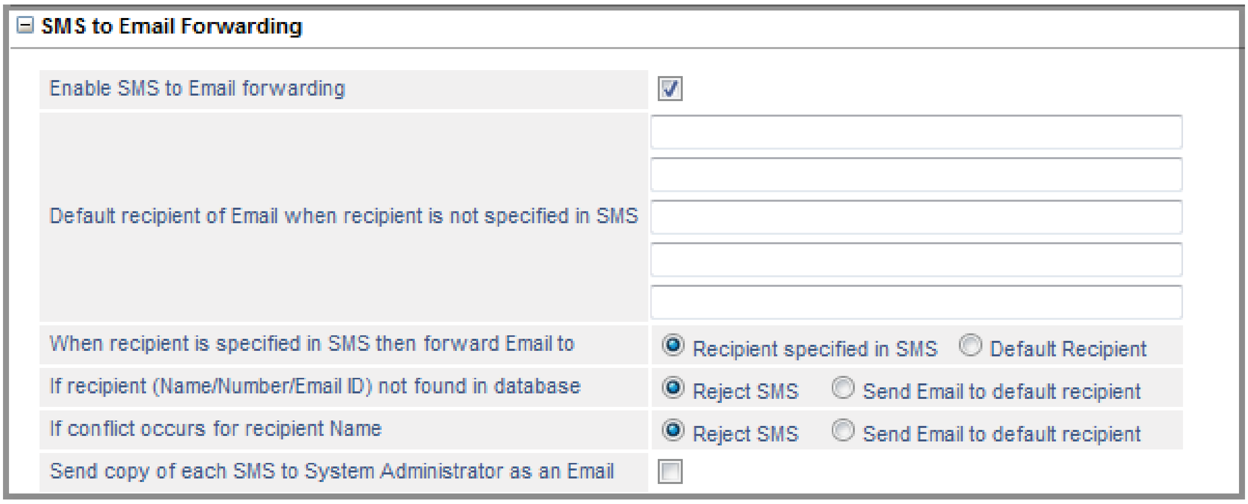
  - If you enable **Send error report to sender when SMS length is more than allowed parts**, enter the message you want to send to the sender in the email in **Reply error report as an Email to sender containing text.** Default text: Some texts of message are ignored as length of mail is more than allowed characters.

- Configure the **Minimum time delay between sending two consecutive SMS (sec)** as supported by the network. Default: 05.

- Click **Submit.**

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 up to 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.

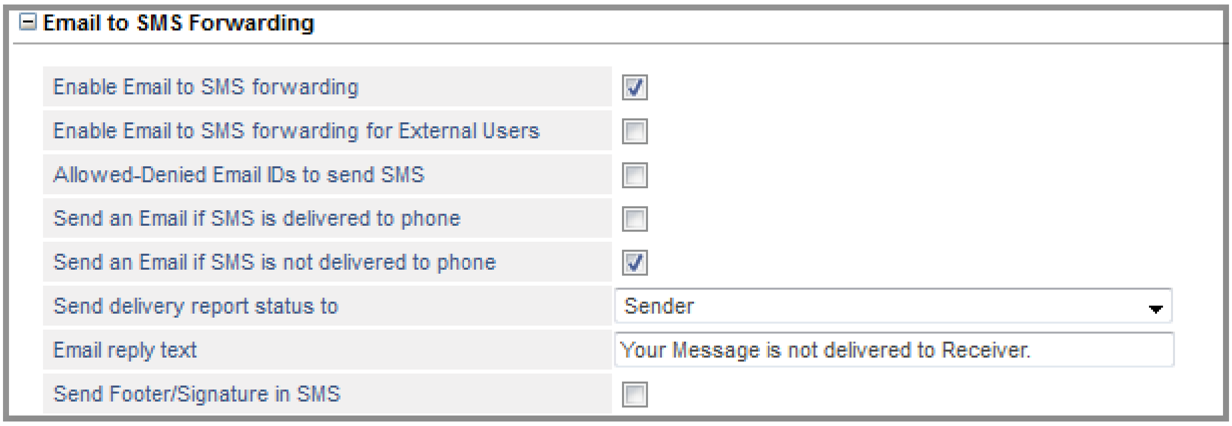
- Enable the **Send copy of each SMS to System Administrator as an Email** check box, if you want to send a copy of SMS as an Email to the System Administrator. Default: Disabled.

  If you enable this check box, you must configure at least one Email address of the System Administrator under “System Administrator Email ID”.

- Click **Submit.**

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 up to 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** 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: Enabled.

- 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.

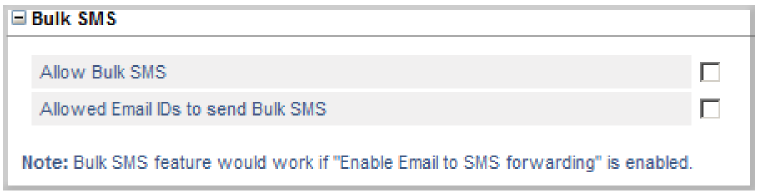
- Enable the **Send Footer/Signature in SMS** check box and select the desired option to be sent:
  - Specific Text: Enter the text/message to be sent to the recipient as signature in the SMS.
  - Send Email ID of Sender: Enter the Email ID of the Sender to be sent to the recipient as signature in the 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**.

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. 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 List for sending Bulk SMS** table opens. You can configure up to 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 List for sending Bulk SMS** link under **SMS Server**

  Default: Disabled.

- Click **Submit**.

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 **Error Cause List** to expand.

Error Cause List

Error Cause No.	Email reply text
1	You are not allowed to access SMS Server features.Please consult your system administrator.
2	You are not allowed to send SMS on this Number - <X>. Please consult your system administrator.
3	Name - <X> is not found in Directory.
4	Conflict occurred for Name - <X> in Directory. Please check the name.
5	Number is not found for Name - <X> in Directory.
6	No port is programmed or available to send SMS. Consult your system administrator.
7	No contact is added. Please add at least one contact.
8	Your mail is large to send the SMS, please shorten your mail or contact your system administrator.
9	SMS limit exceeds. Please consult your system administrator.
10	Daily SMS limit exceeds. Please consult your system administrator.

Note: <X> or <x> will reflect actual Name or Number received in Email and System will send reply mail containing error message to sender. E.g. if Error Cause 5 is programmed as 'Number is not found for Name - <x>'. Let's say SMS is received for name 'Joseph' and number is not programmed in directory then reply mail will be sent to sender with text 'Number is not found for Name - Joseph'

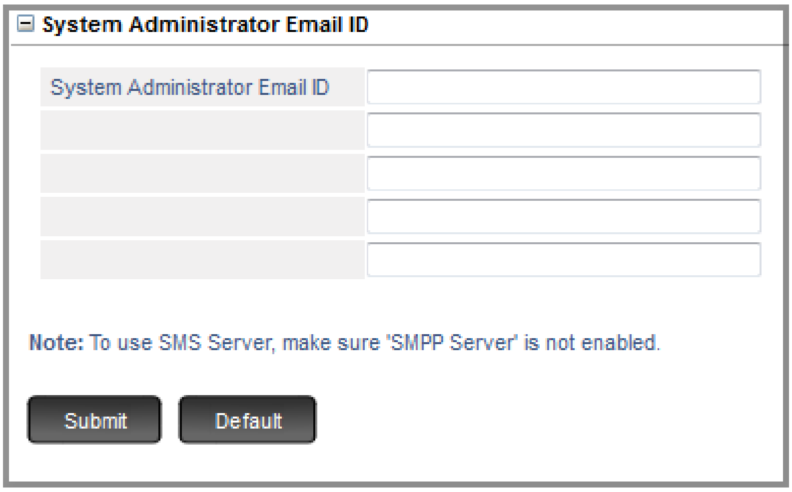
The Error Cause table displays the Error Cause number with the corresponding Email reply text.

- 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 **System Administrator Email ID** to expand.



Configure the following parameters:

- 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** checkbox under "SMS to Email Forwarding".

You can configure up to 5 Email ID’s. The Email IDs can be of maximum 64 characters.

- Click **Submit**.

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 utilize the benefits offered by the service providers, such as 1000 free SMS in a month, reduced rates to send messages etc.

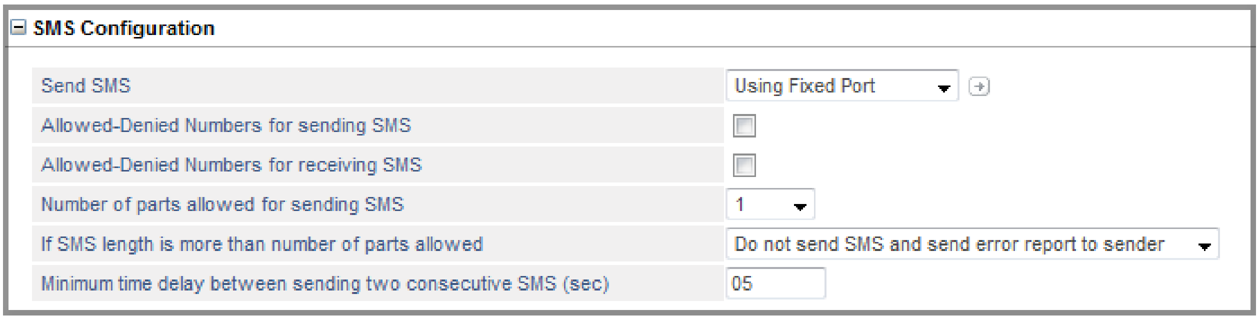
The system allows you to configure different Fixed Port Routing Tables for the SMS Server application and SMS over IP application. But, the Least Cost Routing table is common for both these applications.

How to configure

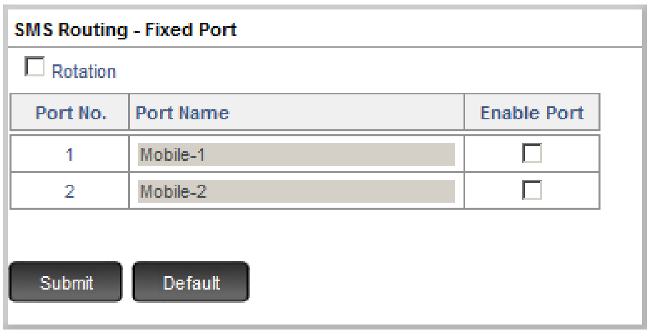
Fixed Port Routing (SMS Server)

You can route the SMS through a fixed mobile port. To do this, you must determine the port through which you want to send the SMS and then configure the SMS Routing-Fixed Port table.
To configure Fixed Port Routing,

- Login as System Engineer.
- Under **Advanced Settings**, click **SMS Server.**
- Click **General Parameters**.
- Click **SMS Configuration** to expand.
    - In **Send SMS**, select **Using Fixed Port**.



- Click **Settings**, the **SMS Routing-Fixed Port** table opens.



- 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 second Mobile Port and so on. For example, if there is a request to send 7 SMS. Then, the first SMS will be sent through MOB Port 1, second SMS through MOB Port 2, third SMS through MOB Port 1, fourth SMS through MOB Port 2 and so on.

    If the Rotation check box is cleared, all the SMS will be sent using the Mobile Port enabled by you. For example, if you have enabled Ports MOB Port 1 and there is a request to send 5 SMS. Then, all the SMS will be sent through MOB Port 1 only.

  - **Port No.:** This is the Mobile Port number.

  - **Port 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 “Mobile Trunks”.

  - **Enable Port:** 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 “Mobile Trunks”.*

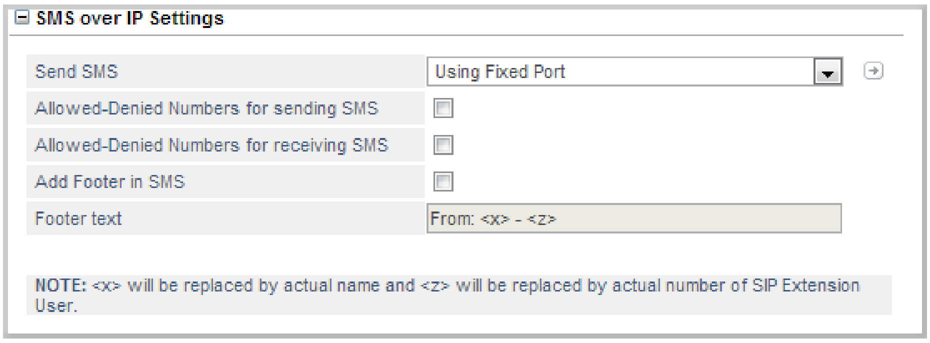
Fixed Port Routing (SMS over IP)

You can route the SMS through a fixed mobile port. To do this, you must determine the port through which you want to send the SMS and then configure the SMS over IP-Fixed Port table.

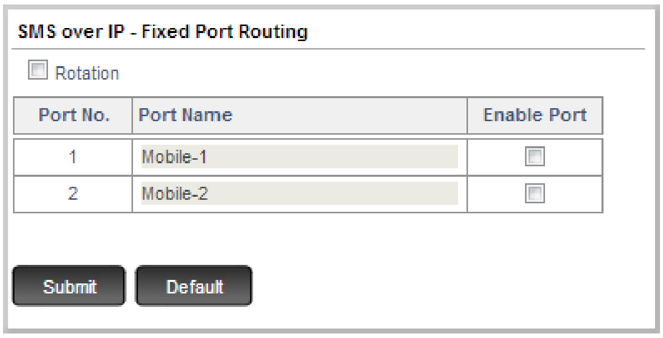
To configure Fixed Port Routing,

- Login as System Engineer.
- Under **Basic Settings**, click **VoIP Parameters**.
- Click **SMS over IP Settings** to expand.

- As the **Send SMS** option select **Using Fixed Port**.



- Click on **Settings** . The **SMS over IP-Fixed Port Routing** table opens.



- 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 second SMS through the second Mobile Port and so on. For example, if there is a request to send 7 SMS. Then, the first SMS will be sent through MOB Port 1, second SMS through MOB Port 2, third SMS through MOB Port 1, fourth SMS through MOB Port 2 and so on.

    If the Rotation check box is cleared, all the SMS will be sent using the Mobile Port enabled by you. For example, if there is a request to send 5 SMS. Then, all the SMS will be sent through MOB Port 1 only.
  - **Port No.:** This is the Mobile Port number.
  - **Port 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 “Mobile Trunks”.
  - **Enable Port:** 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 “Mobile Trunks”.*

Least Cost Routing

Least Cost Routing (also referred to as Automatic Route Selection) is an expense control feature of ETERNITY.

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.

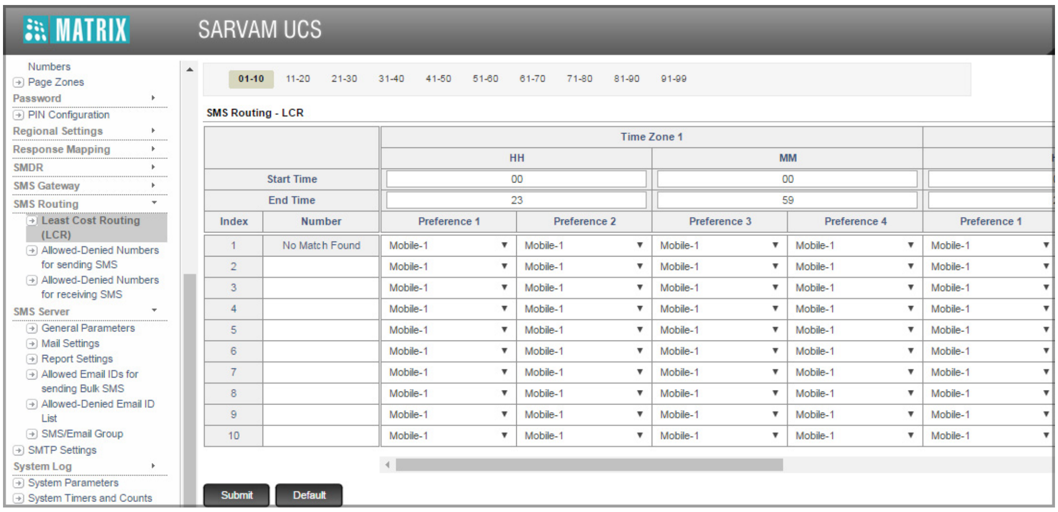
The system can be configured to select the most cost effective trunk for the time of the day when the SMS is sent, or to select the most cost effective trunk for the destination number to which the SMS is sent, or to select the most cost effective trunk considering both time of the day and destination 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

- Login as System Engineer.
- Under **Advanced Settings**, click **SMS Routing**.
- Click **Least Cost Routing (LCR)**. The SMS Routing-LCR table opens.



- Configure the following parameters:

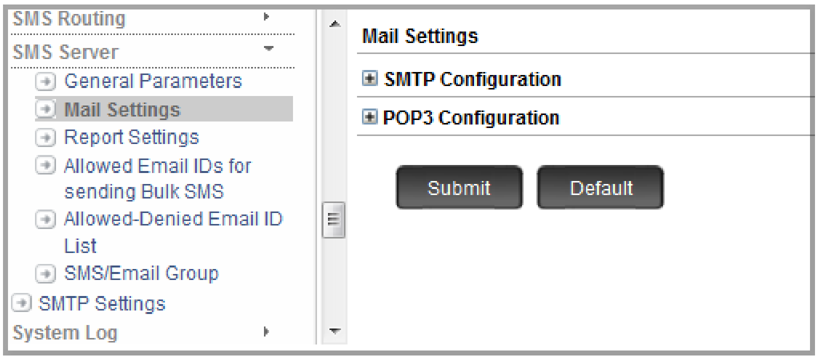
- **Time Zone 1** to **4:** Configure the **Start Time** and **End Time** for each time zone. You can configure four different time zones—Time Zone1, 2, 3 and 4 as per your requirement.
- **Number:** Configure the destination numbers to which the messages are to be sent.
- **Preference1** to **4**: Select the trunks in the order of preference through which you want to send the SMS. You can select upto 4 preferences.
- Click **Submit**.

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

- Login as System Engineer.
- Under **Advanced Settings**, click **SMS Server**.
- Click **Mail Settings**.



SMTP Configuration

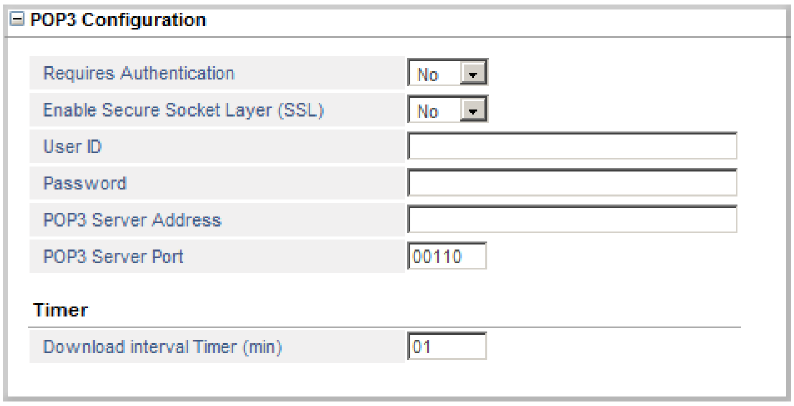
Contact your Network Administrator for the following information and configure the parameters as per the configurations done in the Email Server to register the SMS Server as an SMTP Client.

- Click **SMTP Configuration** 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**.

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** 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. 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. The Server Address can be a maximum of 46 characters.

- Click **Submit**.

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'**, 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 **OK**.

The message *"Please refresh the web browser after few seconds to check the test mail status"* appears. Click **OK**.

Refresh the web browser after a few seconds. The Test Result will be displayed in the 'Test Status'.

- **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

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 500 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:

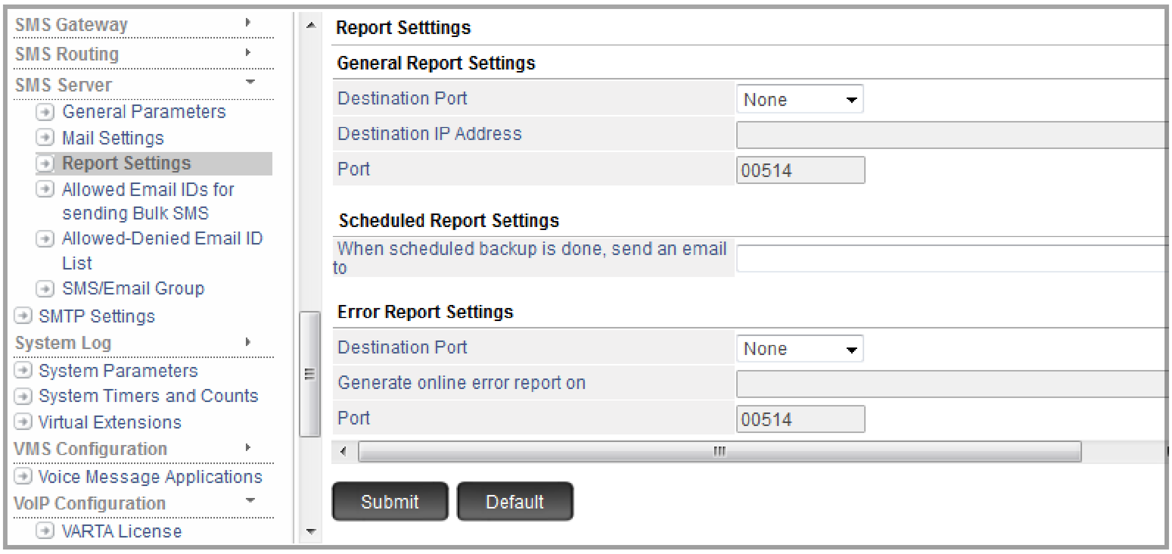
- Index
- Email
  - Direction
  - Email Address
  - Status
  - Date
  - Time
- SMS
  - Direction
  - Number
  - Status
  - Date
  - Time
  - Part of SMS
- Mobile Port
- Text

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

- Login as System Engineer.
- Under **Advanced Settings**, click **SMS Server**.
- Click **Report Settings**.



- Under **General Report Settings**,
  - Select **Ethernet** as the **Destination Port**.
  - In the **Destination IP Address:Port**, enter the IP Address and Port of the remote Syslog Server.
- 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 the **Destination IP Address:Port**, enter the IP Address and Port of the remote Syslog Server.

How to use

You can print Reports whenever you want or schedule printing of the report from the System Administrator mode.

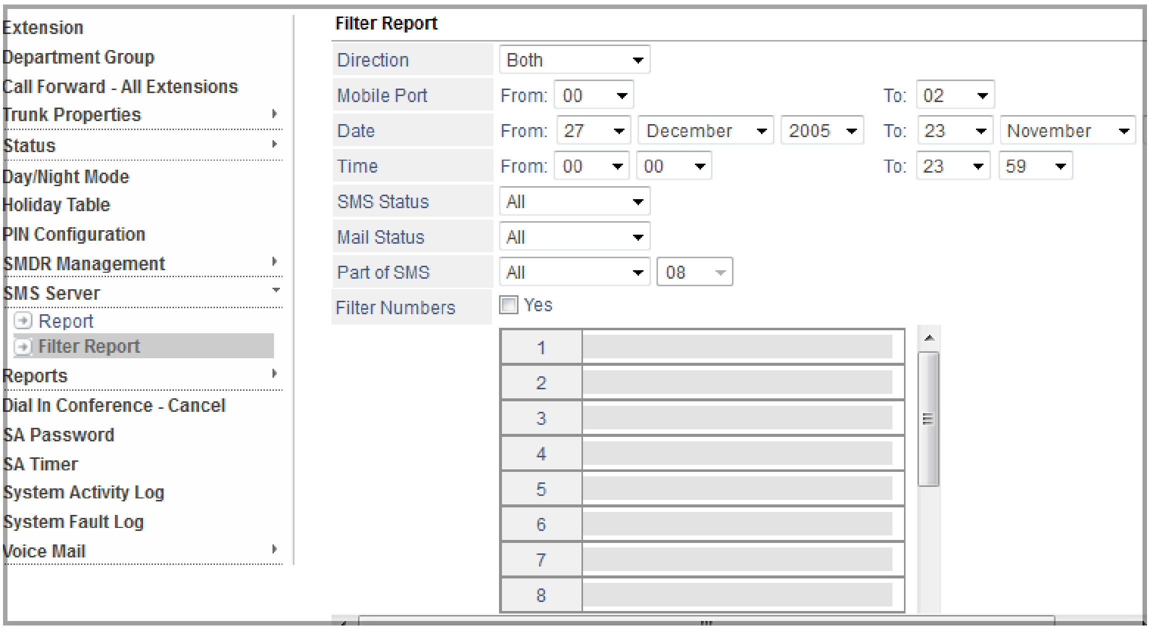
You must set the filters as per your requirement before you print the Report. To do this,

- Login as System Administrator.

- Click **SMS Server** to expand.

Settings Report Filters

- Under **SMS Server**, 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 **Date** 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.

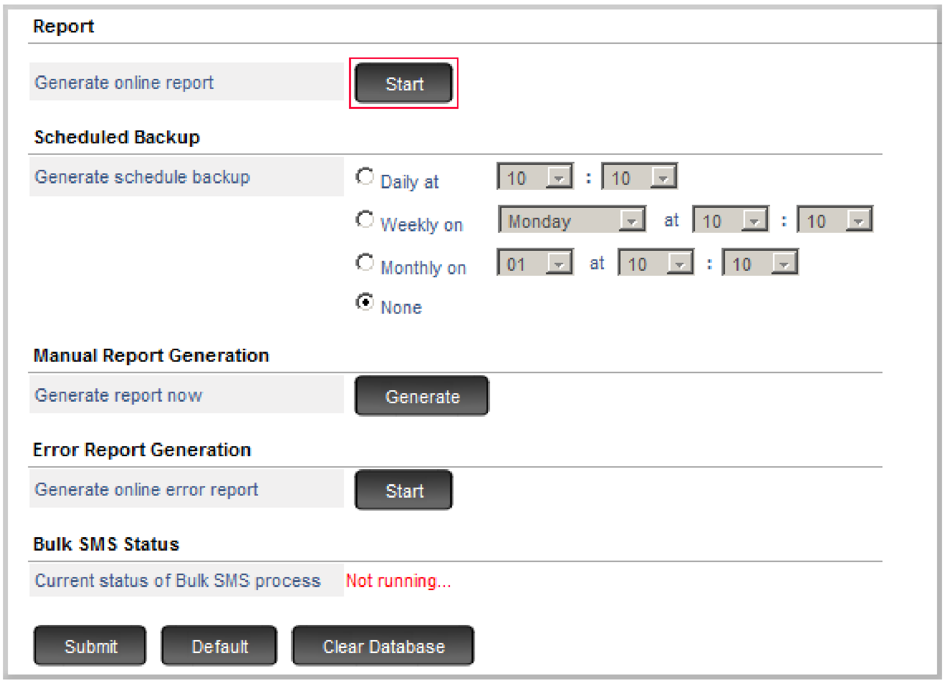
  - Click **Submit**.

- To view the report, click **Generate Report**. The report is displayed in a new window.

Online Report Generation

- Under **SMS Server**, click **Report**.

  - To **Generate online report,** click **Start**.

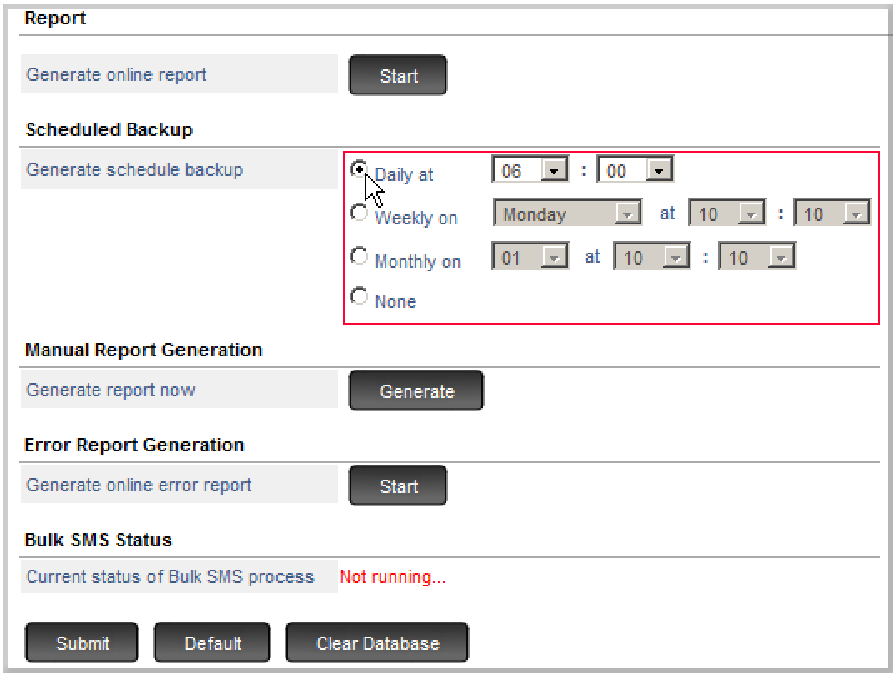


  - To stop printing, click **Stop**.

Scheduled Report Generation

- Under **SMS Server**, click **Report**.

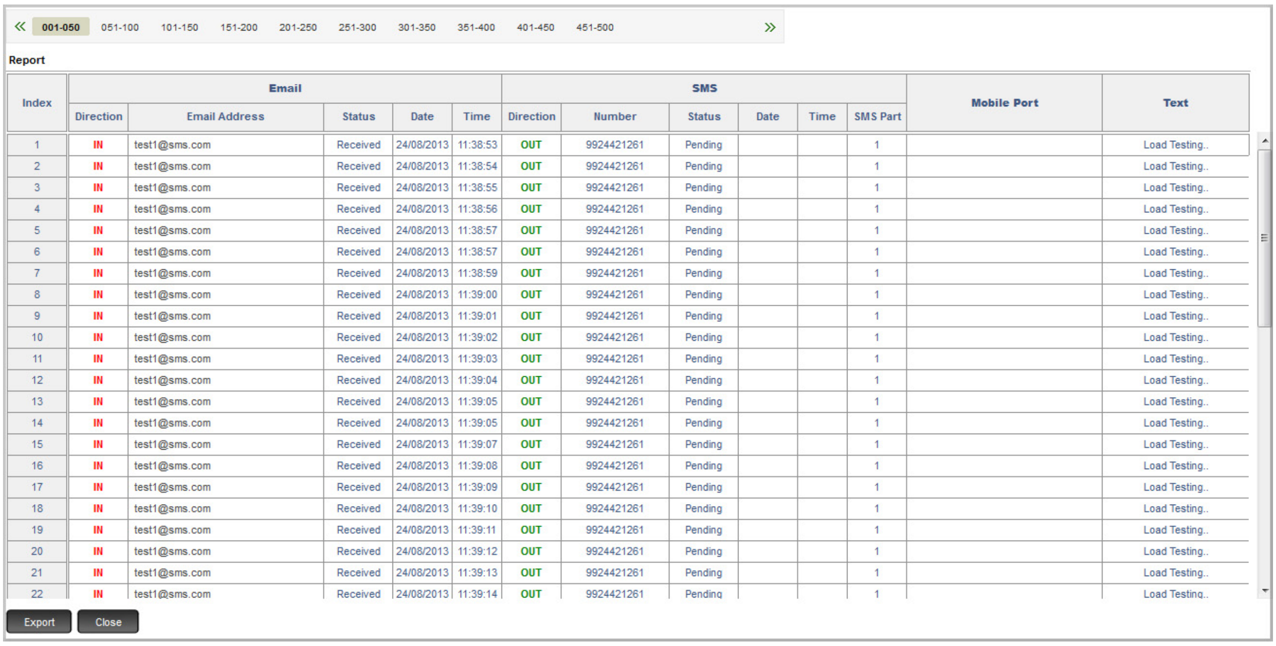
- To **Generate schedule backup**, select the desired option a particular day, day of the week, or day of the month.



- Click **Submit**.

Manual Report Generation

- To generate the report manually, under **SMS Server**, click **Report**.
  - To **Generate report now**, click **Generate**.
  - The report is generated and appears in a new window.



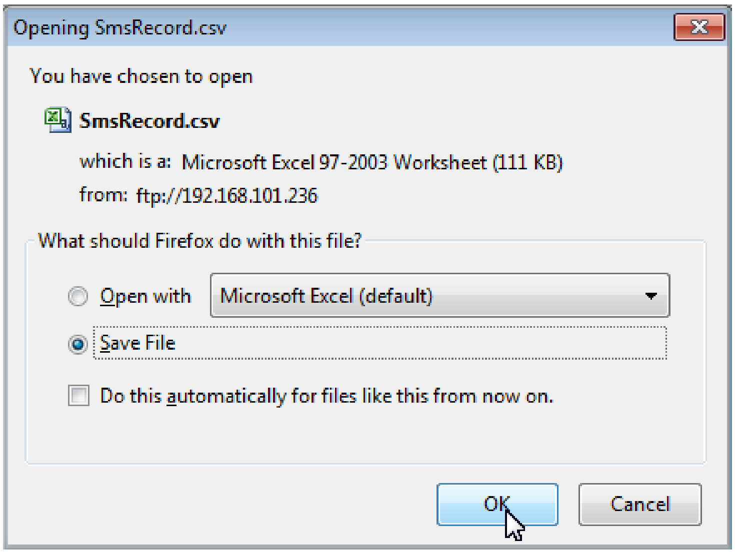
« 001-050 051-100 101-150 151-200 201-250 251-300 301-350 351-400 401-450 451-500 »

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

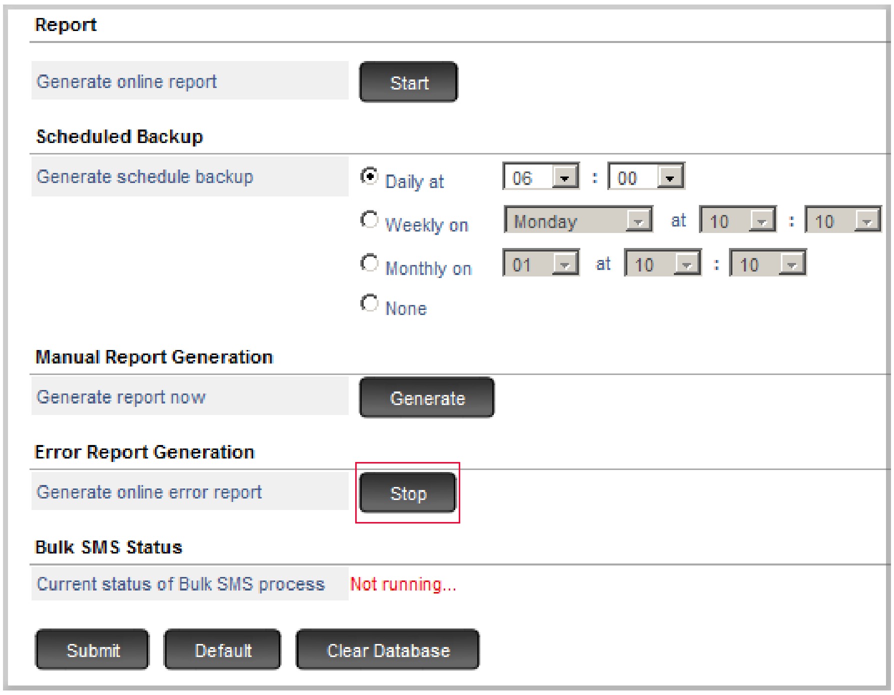
- Click **Export**, if you wish to save the report at the desired location.



The report will be saved in CSV format.

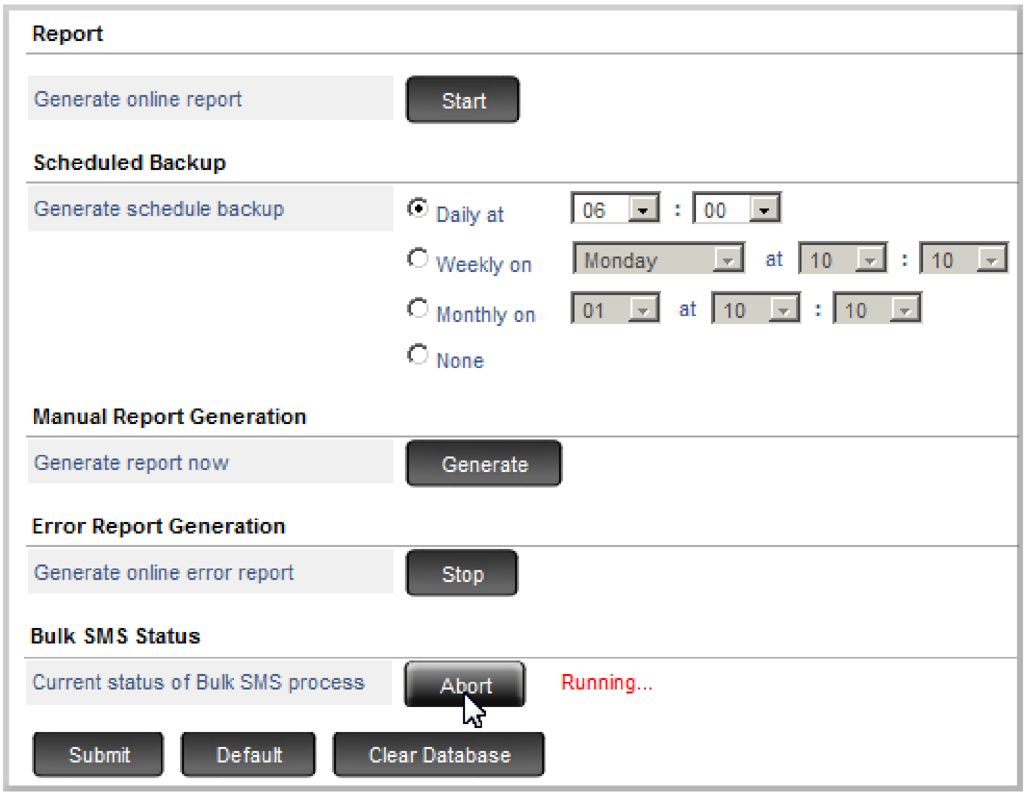
Online Error Report Generation

- Under **SMS Server**, click **Report**.
  - To **Generate Online error report,** click **Start**.
  - To stop printing, click **Stop**.



Bulk SMS Status

- Under **SMS Server**, click **Report**.
  - 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 **Abort** to stop the ongoing process midway.
- To clear all the database, click **Clear Database.**

SMS over IP

SMS over IP of SARVAM UCS 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 "Mobile Trunks".
- enable **SMS over IP** for SIP extension 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".

Wherever the SMS over IP Application handover the handling of the SMS to the SMS Server Application, make sure you have the SMS Server license to achieve the desired functionality. 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 ETERNITY NENX, 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 will have to send the same message to individual recipients.

- 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.

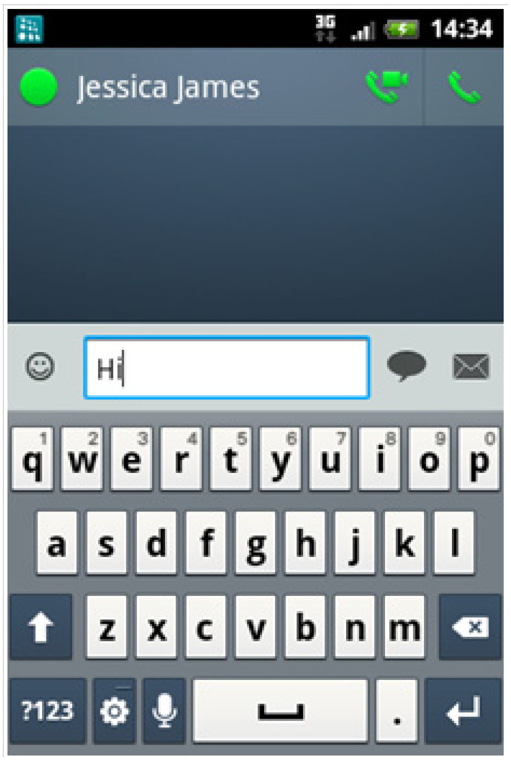
  When the destination is an Extension Number other than SIP extension, the further handling of the SMS will be done by the SMS Server Application. See "SMS Server" for more details.

- 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,

- Login as System Engineer.

- Under **Basic Settings**, click **VoIP Parameters**.

- Click **SMS 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 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 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 **Settings** . 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 **Settings** . The **Allowed-Denied Numbers for receiving** 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 Routing**.

  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**.

SMS/Email Group

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

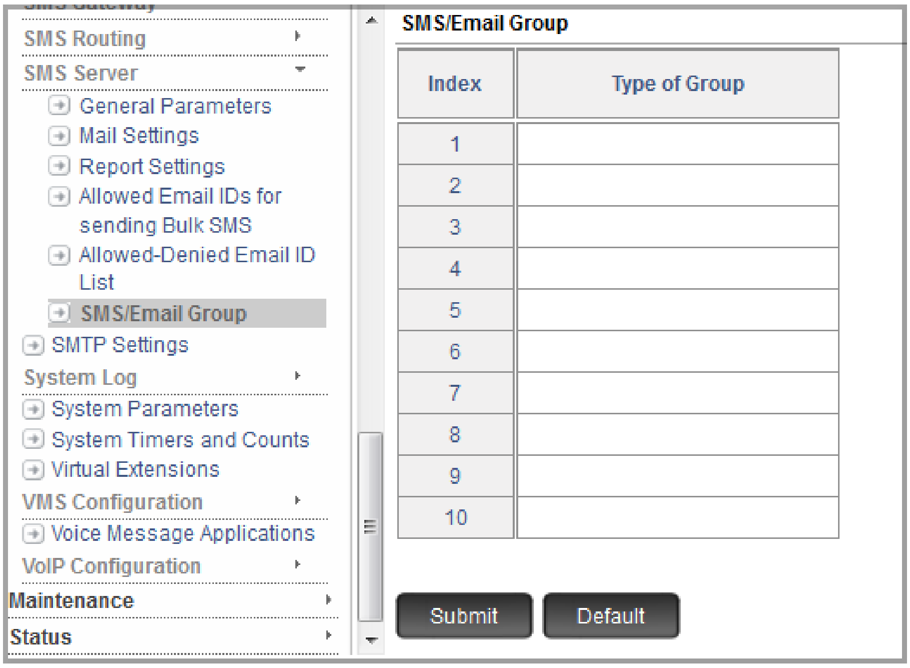
To use SMS/Email Group feature,

- 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 “More Features” under “SLT Extensions” and “More Features” under “SIP Extensions”.
- Assign a Global/Personal Directory contact to the SMS/Email Group you created as per your requirement. For instructions, see “Configuring Personal Directory” and “Configuring Global Directory”.

Configuring SMS/Email Group table

- Login as System Engineer.
- Under **Advanced Settings**, click **SMS Server**.

- Click **SMS/Email Group**.



You can configure maximum 10 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**.

CHAPTER 8

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.matrixtelesol.com/product-manuals.html*

How it works

Using CTI, only following features and facilities of SARVAM UCS can be accessed:

Features	SLT	Extended IP Phone	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	✓	✓	✓
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.matrixtelesol.com/product-manuals.html
- Configure the Matrix TSP parameters. For instructions refer to the TAPI User Guide.
- Configure the CTI parameters in SARVAM UCS. See "How to configure".
- Make sure the necessary third party TAPI applications are installed.

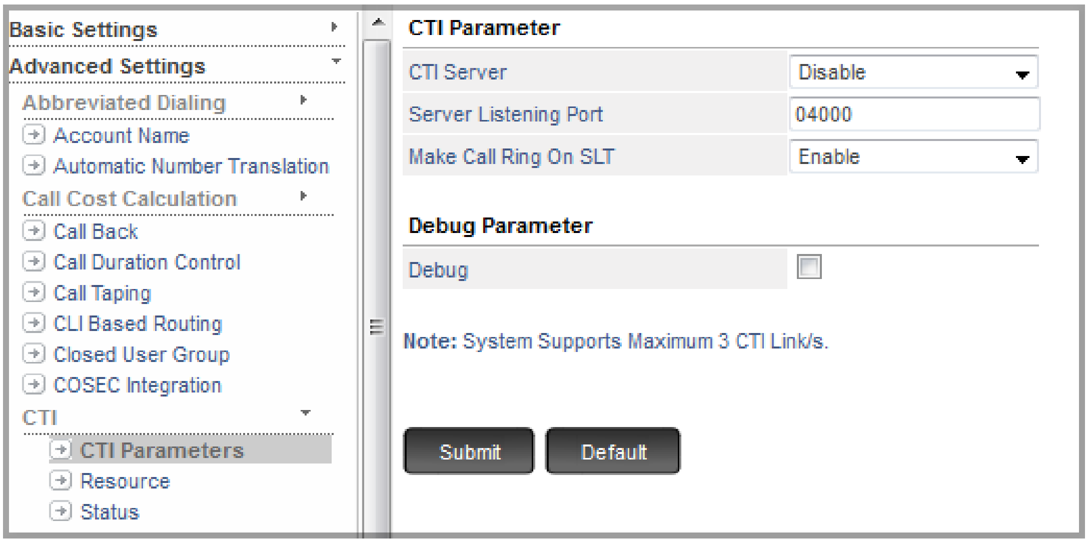
This is how CTI works,

- 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 the 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.

How to configure

You must configure the following CTI parameters. To do so,

- Login as System Engineer.
- Under **Advanced Settings**, 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 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 9

Features and Facilities

If you have CDMA Mobile SIM is installed in your system, it is recommended to avoid using Voice Mail features as the DTMF Detection might not work efficiently.

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

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 frequently called, long-digit numbers quickly and easily.

This feature requires you to store the frequently called, long-digit numbers[99] 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 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 configured and assigned to groups of extensions. The use of Personal Directories is limited to the extensions to which they are assigned.

A single personal directory accommodates 25 numbers. Each number may be up to 16 digits long. A personal directory has Index numbers from 1 to 25 against which the frequently dialed telephone numbers are stored along with their corresponding Names and Trunk Access Code (TAC).

As many as 50 different personal directories, numbered from 01 to 50 can be created and assigned to SLT and SIP Extensions.

99. These may be numbers of your branch offices, your clients, numbers of emergency services such as fire, police.

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 21. The number 02652630555 is stored at Index number 16 of this directory. The user of extension 21 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 (TAC) specified for this number in the personal directory.

- *When an extension user dials an abbreviated number from the Personal Directory, the system first checks the Outgoing trunks and Toll Control Level assigned to the extension and then dials out the number.*
- *Each extension can access only the personal directory assigned to it.*
- *Personal Directory can be configured 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.*

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, referred to 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 999 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 an extension user of SARVAM UCS wants to use Global Abbreviated Dialing, all that the user 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 SARVAM UCS 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 selected for routing Global Directory’s numbers.

- *Extensions can use Global Abbreviated Dialing only if this feature is enabled 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 21 is allowed Global Directory Part 1 in its CoS, the user of extension 21 can dial out only those numbers contained in Global Directory Part 1.*
- *Extensions must be assigned all three parts of the directory in their Class of Service to be able to access the entire Global Directory. By default, Global Directory Part 1, 2 and 3 are disabled in the CoS of all extensions. Therefore, all extension users cannot dial numbers stored in the Global Directory.*

How to configure

For both Personal and Global Abbreviated Dialing to work, the System Engineer must:

- Configure the Personal Directories and the Global Directory.
- Assign Personal Directory to the desired extensions, which may be SLT and SIP extensions.
- Enable Global Directory Part 1, Part 2 or Part 3 as desired in the Class of Service allowed to the extensions.

It is possible for extension users to configure their Personal Directories using their own phones.

It is also possible for Extended IP Phone users to add, delete and edit contacts in Global Directory Part 1 from their extension phones. For this, Global Directory Part 1 and Global Directory Programming must be allowed in their CoS by the System Engineer.

Preparing Numbers Lists for Personal and Global Directories

In consultation with the extension users, you may:

- Determine the number of personal directories that need to be configured.
- Make a list of numbers frequently dialed by the extension.
- Ask the extension users the numbers they want to include in the personal directory of their extension.
- Make separate lists of numbers along with their corresponding names, email id, group and trunk access codes, for each personal directory. You may draw tables on paper and enter the numbers and corresponding names, email Ids, group and trunk access codes against each Index number.

Personal Directory 01

Index No.	Number	Name	Email ID (optional)	Group (optional)	TAC
1					
2					
:	:	:			:
:	:	:			:

Index No.	Number	Name	Email ID (optional)	Group (optional)	TAC
25					

Personal Directory 02

Index No.	Number	Name	Email ID (optional)	Group (optional)	TAC
1					
2					
:	:	:			:
:	:	:			:
25					

You may assign the same personal directory to more than one extension.

- 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 table on paper and enter the telephone numbers along with their names, email Ids, group. Also decide the outgoing trunks to route the Global Directory numbers.

Global Directory

Memory Location	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 file".

For Global directory contacts, the format of the CSV file must be as follow: For example 121, 9867985489, Sean Gilbert, Sean@hotmail.com, 1, 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".
000 is the Alternate Number Group (optional).

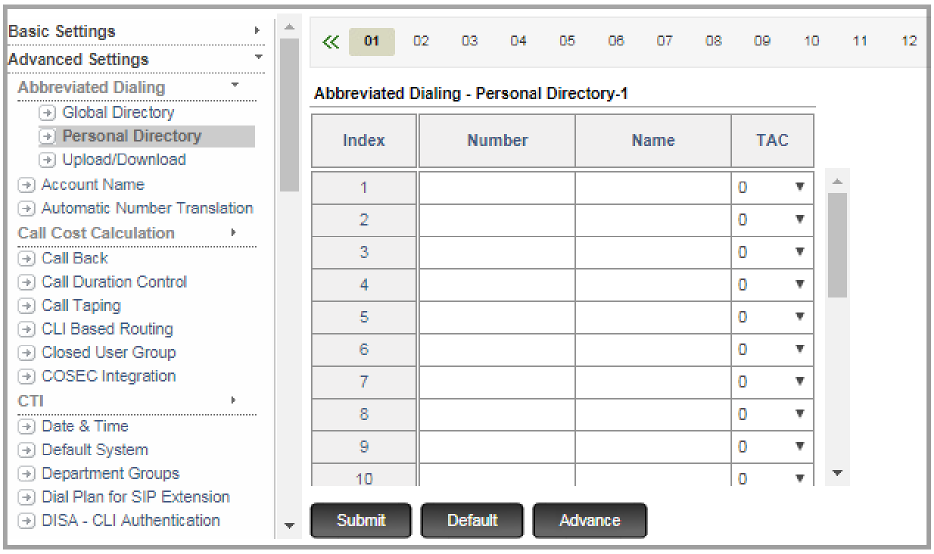
To upload Global Contacts CSV file using Jeeves, see "Upload Global Directory CSV file".

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

- Login as System Engineer.
- Under **Advanced Settings,** click **Abbreviated Dialing**.
- Click **Personal Directory**.



You can configure up to 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. The number may consist of 16 digits (maximum).
- Enter the **Name** against the Number. The Name may contain up to 12 alphanumeric characters. All ASCII characters except < > and " (double quote) are allowed.
- Select the trunk access code **TAC**, as required. Default: 0.
- 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**.
- Follow the same steps as described above to configure another Personal Directory.

Keep a print out of each personal directory for your record and for the record of the extension user. 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.

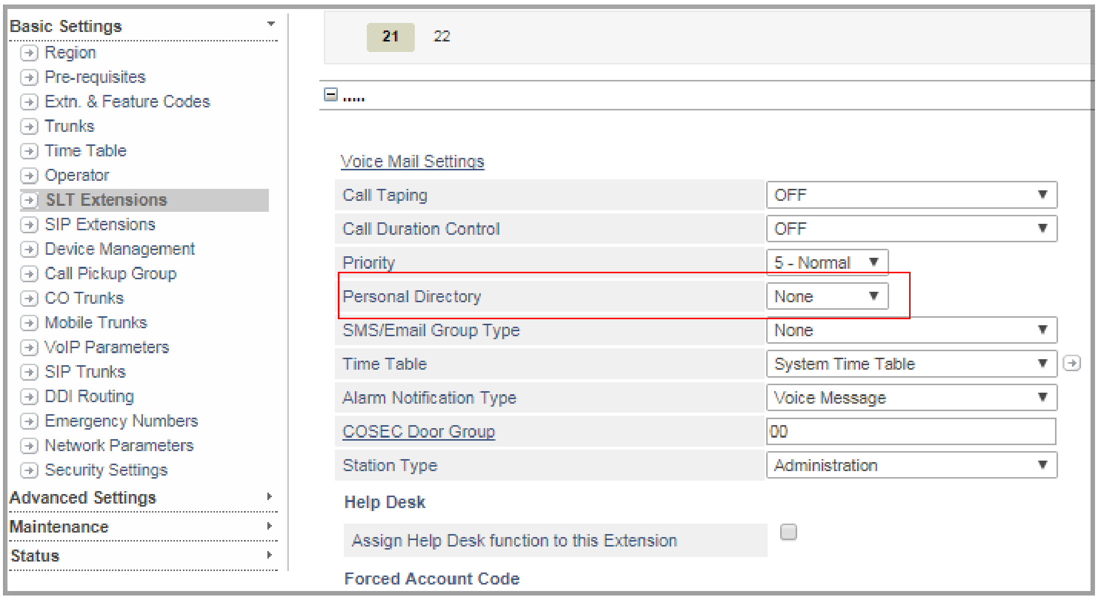
Assigning Personal Directories to Extensions

After you have configured the Personal Directories, assign a personal directory to each extension. The extension may be a SLT or SIP extension.

To assign Personal Directory to SLT extensions,

- Under **Basic Settings**, click **SLT Extensions**.
- Click the **More ...** link to expand options.

- Select the **Personal Directory** number to be assigned to the SLT extension. For example, to assign Personal Directory 02 to SLT 21, select '02'.



- Click **Submit.**

- To assign Personal Directory to another SLT, click the tab of the desired SLT and follow the same steps as described above to assign the Personal Directory.

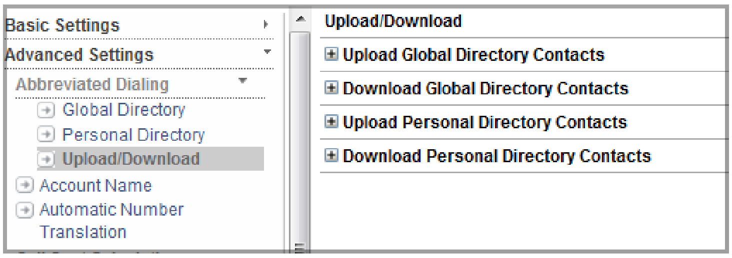
Similarly, you can assign a Personal Directory to a SIP extension.

Upload Personal Directory CSV file

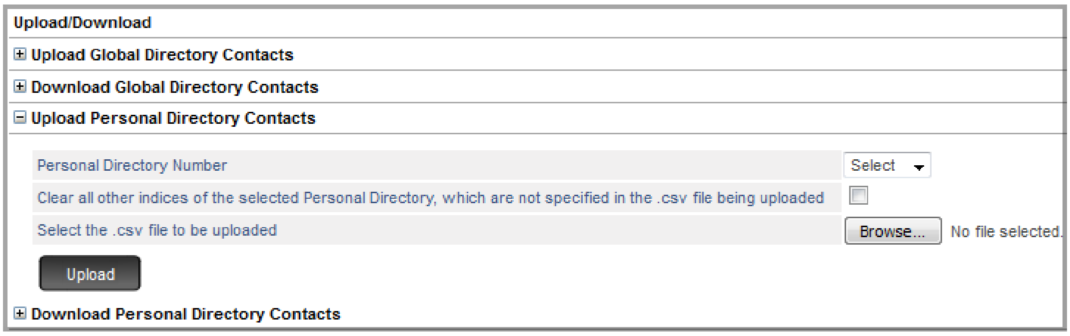
- Login as System Engineer.

- Under **Advanced Settings,** click **Abbreviated Dialing**.

- Click **Upload/Download**.



- 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 **Browse** to **Select the .csv file to be uploaded** from the location on the local disk.

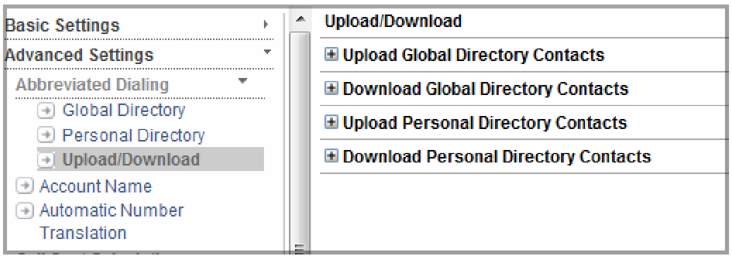
- Click **Upload**.

- All the contacts of the CSV file will be uploaded in the selected Personal Directory. To view, click the respective Personal Directory link.

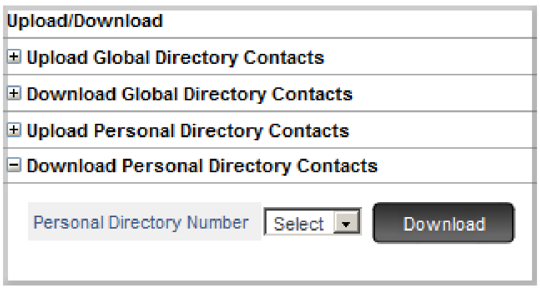
Download Personal Directory CSV file

- Under **Advanced Settings,** click **Abbreviated Dialing**.

- Click **Upload/Download**. The Upload/Download page opens.



- 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 **Download**.

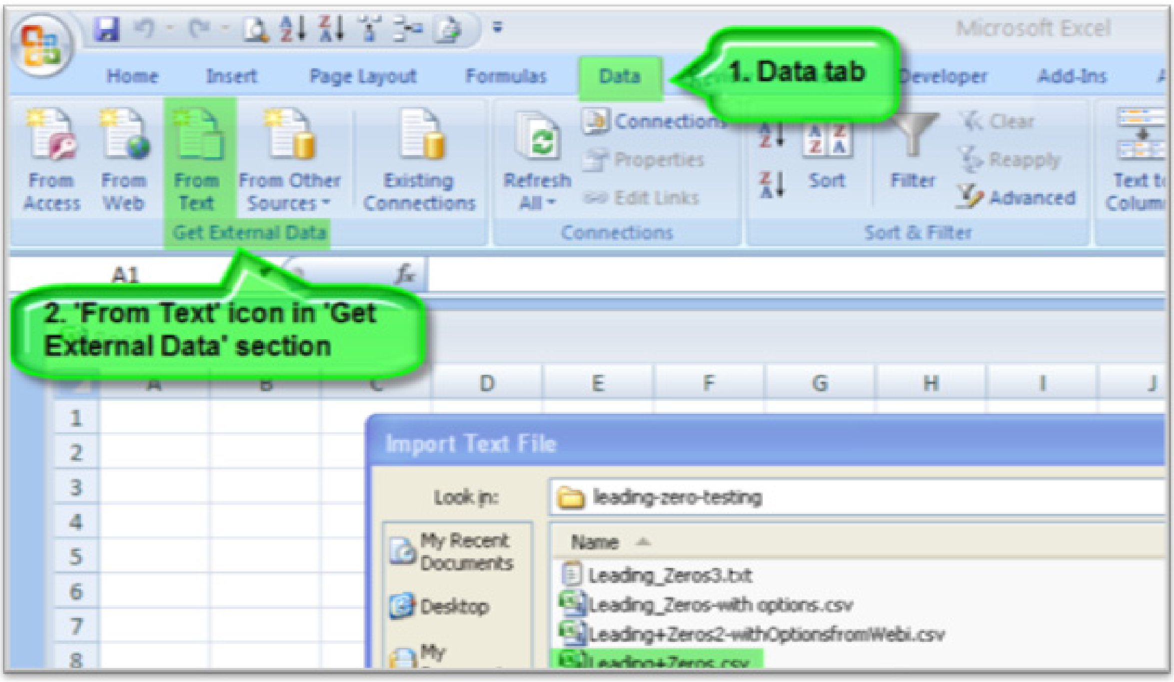
- You will get a prompt with an option to open the **Opening Personal_Directory_01.csv** file or save the file to a location. Save the file on the local disk.

- To open the **Personal_Directory_01.csv** file from the location on the local disk, follow the steps given below:

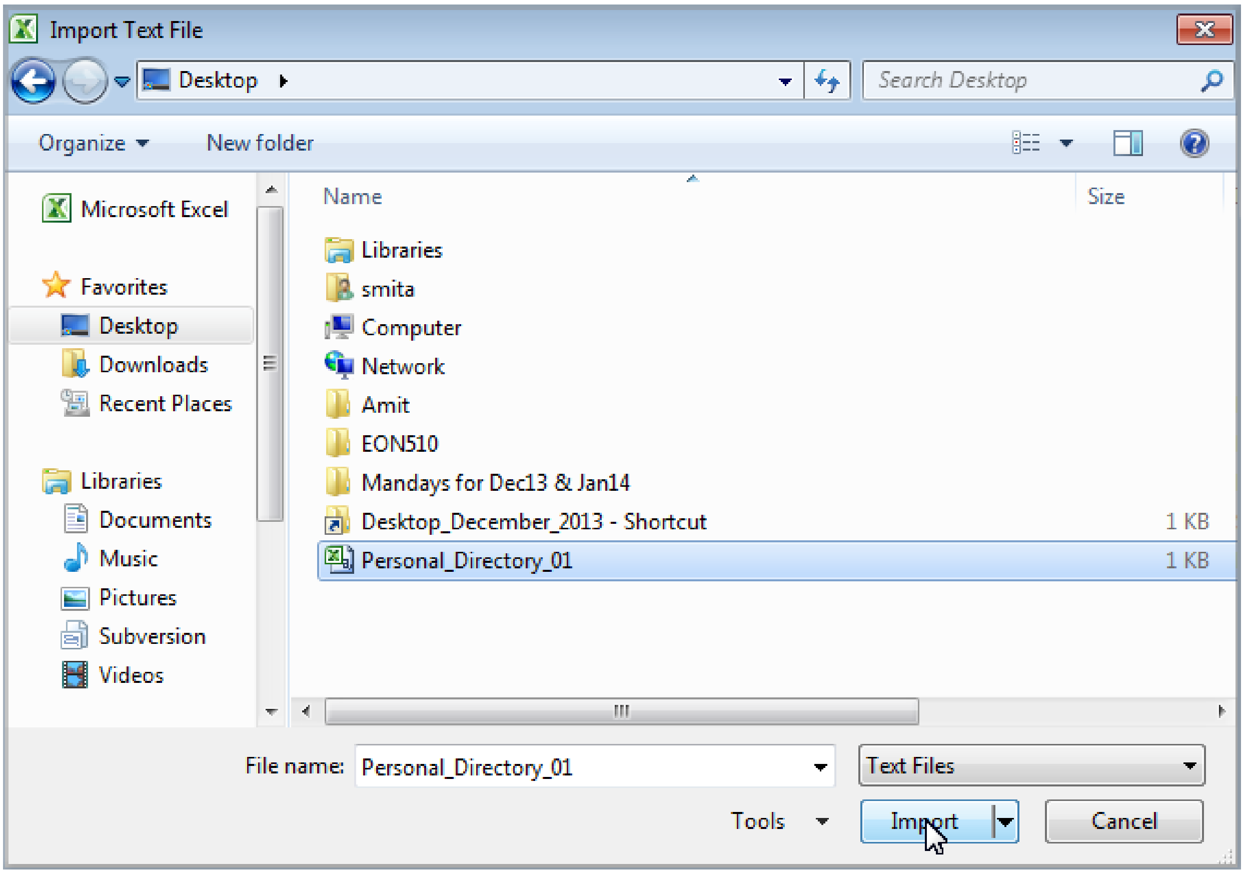
    - **DO NOT OPEN THE CSV FILE DIRECTLY WITH EXCEL**!

    - Open a **New** worksheet in Excel.

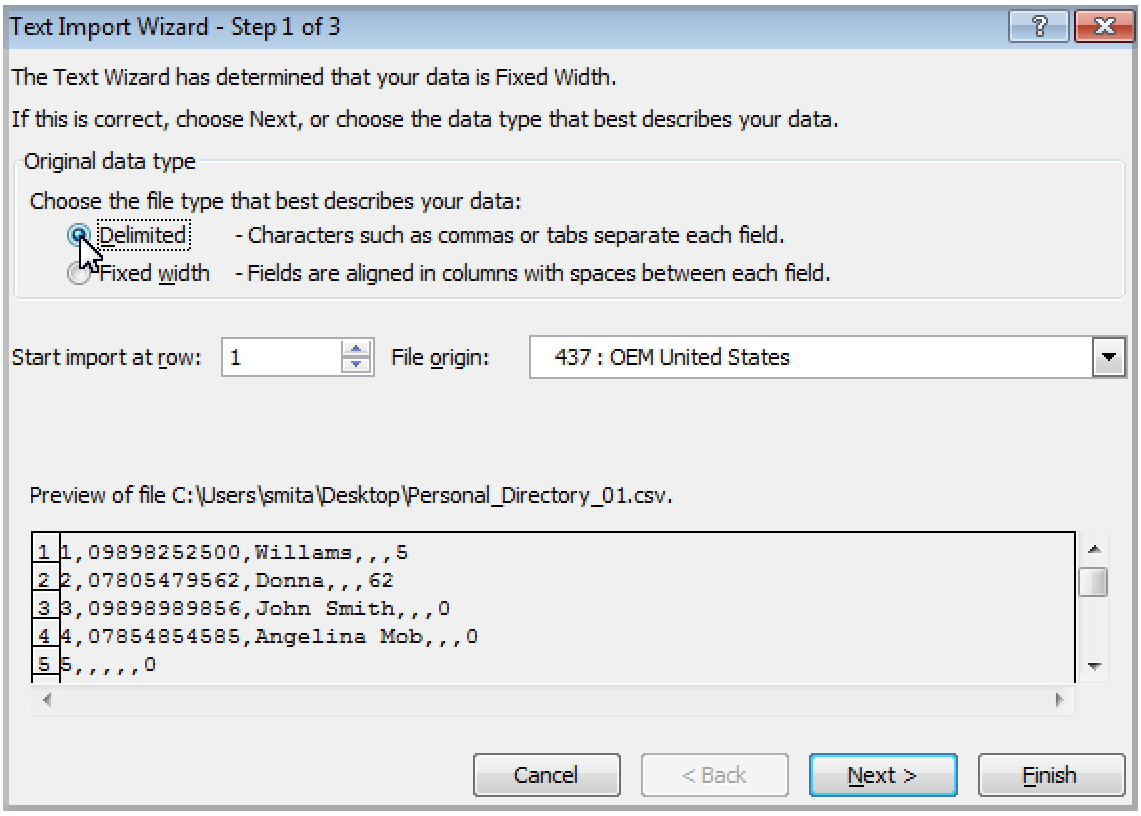
    - Open the **Data** tab and select **From text button in the Get External Data**.



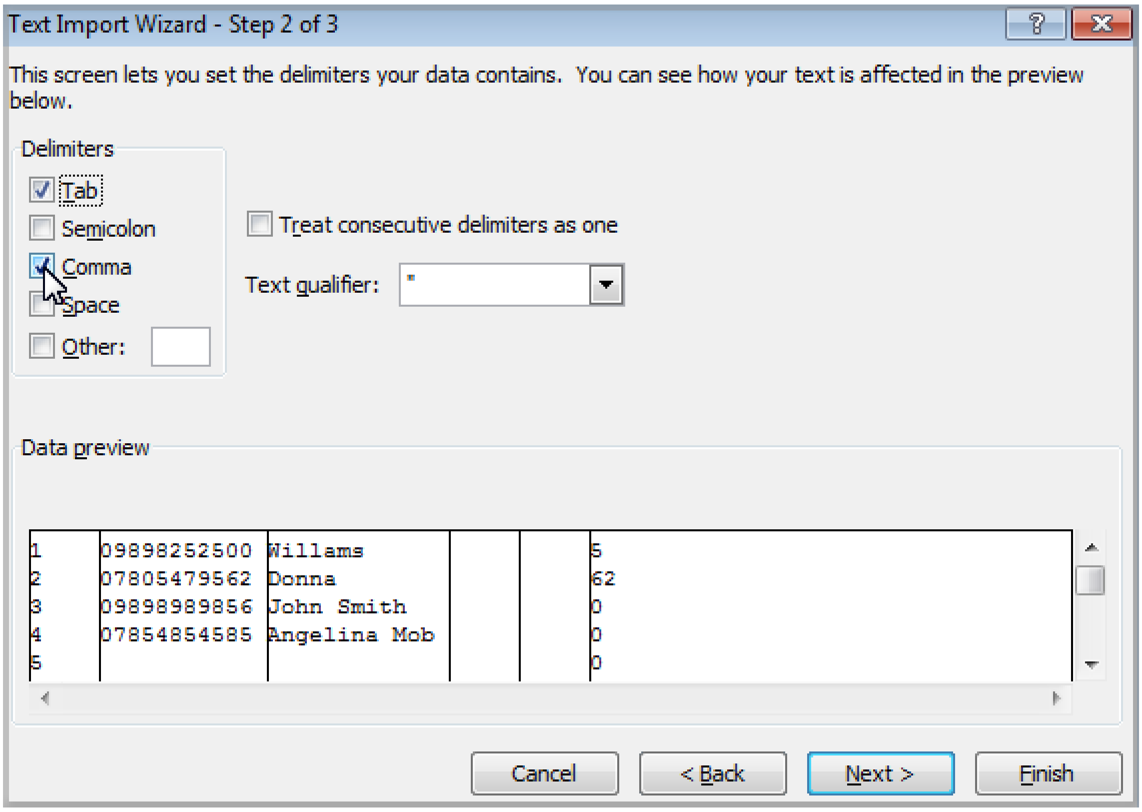
- Select your CSV file from the location on the local disk to import.



- In **Original data type** section, select **Delimited** radio button and click **Next**.

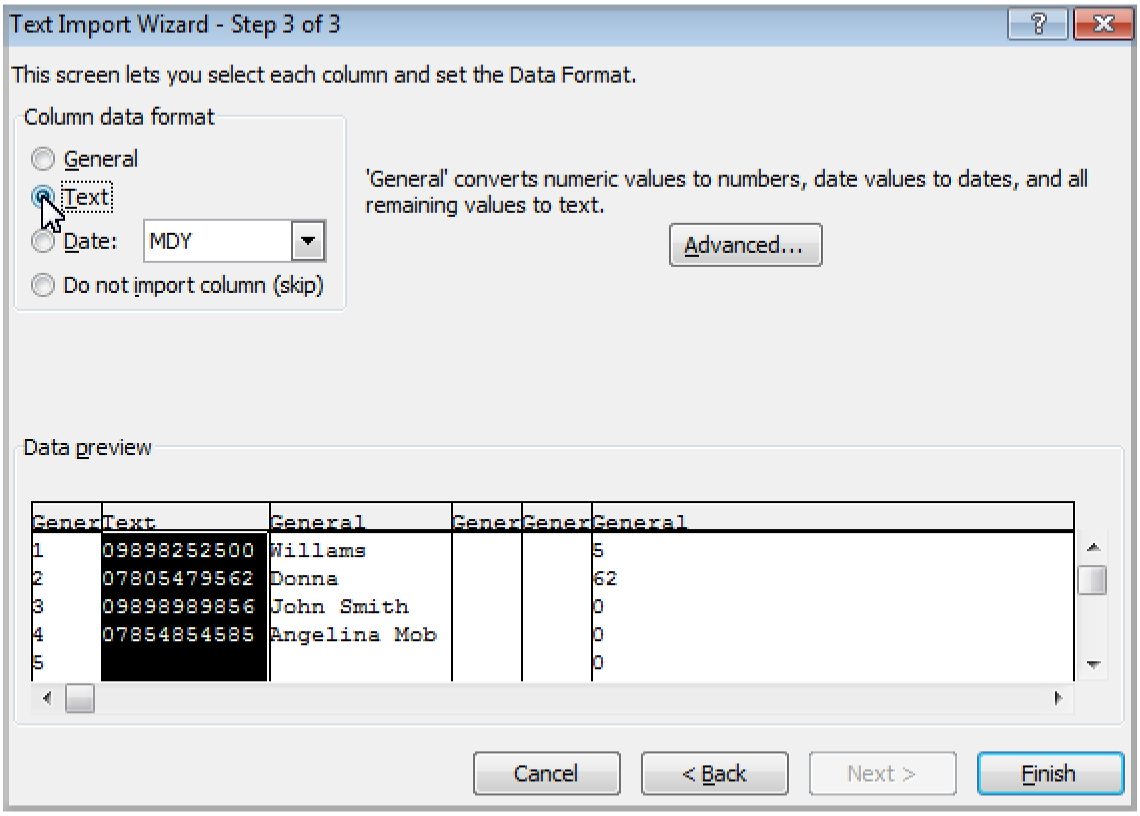


- In **Delimiters** select the **Comma** check box (column dividers will appear in preview) and click **Next**.



- 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 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 1 to 25 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-XX*.

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 1 to 25 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-XX, 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.

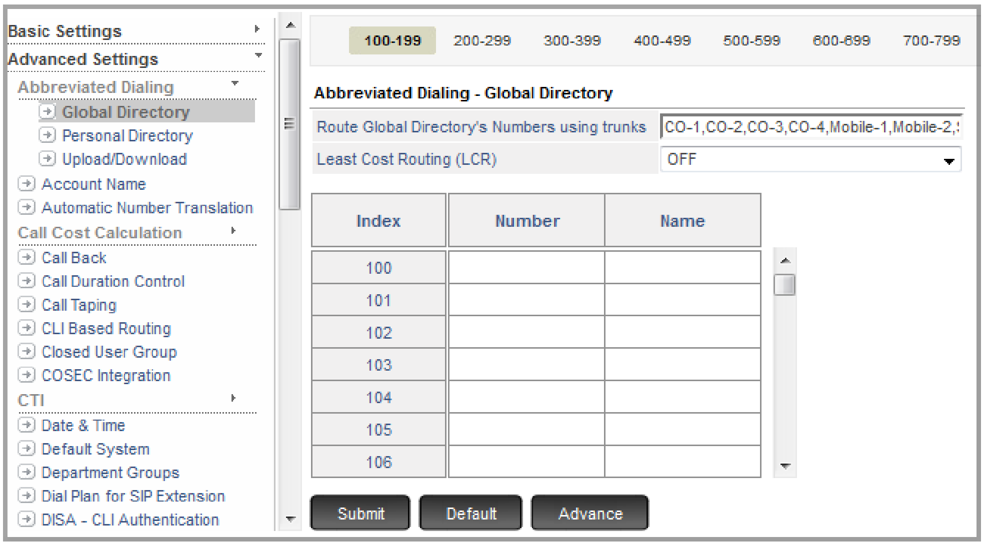
Modify/Edit a Contact

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 Global Directory

- Login as System Engineer.
- Under **Advanced Settings,** click **Abbreviated Dialing**.
- Click **Global Directory**.



Each page has 100 entries. To go to the next 100 entries click the links above the table '0200-0299'

For each directory configure the following parameters:

- Enter the **Number** you wish to store against the memory location. The number may consist of 16 digits (maximum).
- Enter the **Name** against the Number. The Name may contain up to 12 alphanumeric characters. All ASCII characters except < > and “ (double quote) are allowed.
- 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.

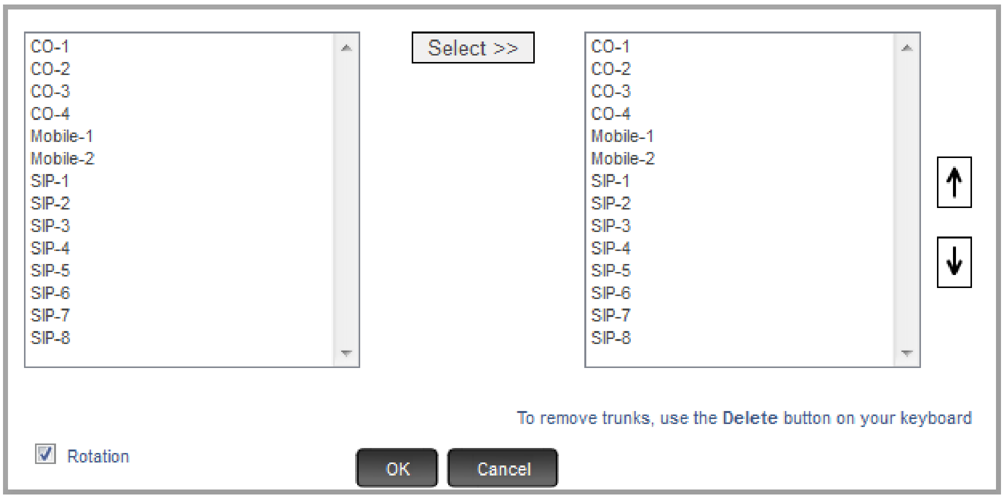
- You can assign the contact to a Group. Select the desired **Group** Type from the list. The system clubs together contacts assigned the same Group. Default: None. For details, see "SMS/Email Group".

- Click **Submit**.

- To configure more numbers, click the next tabs 200-299, 300-399, 400-499, 500-599, and so on. Follow the same steps as described above.

- Select the trunks to be used for routing the numbers of the Global Directory dialed out by extensions.

Double click the **Route Global Directory's Numbers using Trunks** field.



A multiple selection box appears.

- To select trunks, place your cursor on the desired trunk listed on the left box and click **Select>>**.

- You may change the sequence of the trunks using the **Up** and **Down** buttons on the right display box.

- You may enable **Rotation**, if you have selected more than one trunk.

- Click **OK**.

- If you want to apply the Least Cost Routing logic on the trunks, select the desired **LCR** type from the list box: Time based, Number based, Time+Number based, Service Provider based. See "Least Cost Routing (LCR)" to know more.

- Click **Submit**.

Applying Global Directory to Extensions

- Now, enable Global Directory in the "Class of Service (CoS)" of the extensions. To do this,

  - Click **Basic Settings** and select the desired extension type:
    - "SLT Extensions"
    - "SIP Extensions"

  - On the respective extension page, click **Class of Service** to expand.

Enable the part of the Global Directory you want to assign to the extension in its Class of Service.

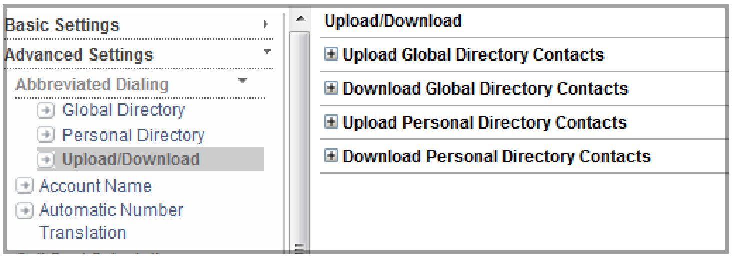
By default, Global Directory Part 1, 2 and 3 are not allowed to all extensions in their CoS.

- Click **Submit**.

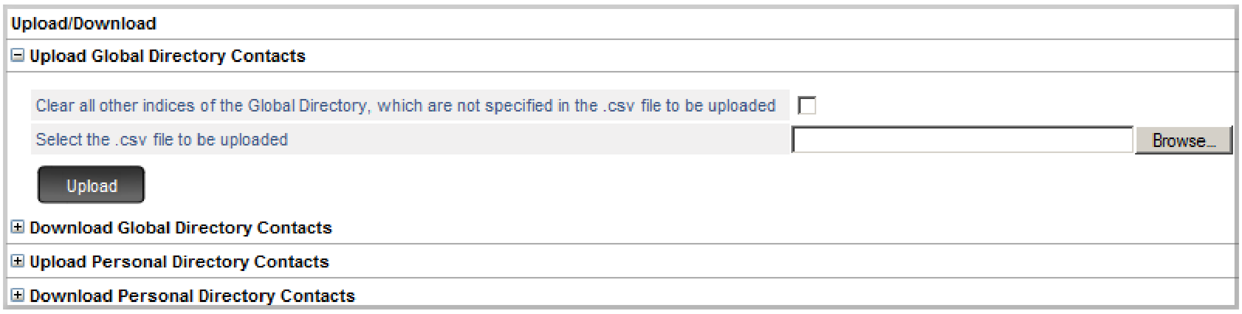
- *If Global Directory Part 1, 2 or 3 is assigned to an extension user, SARVAM UCS will not check for Toll Control.*
- *When you assign Global Directory Programming to a SIP extension user, the user can configure any number in Global Directory Part 1, this includes numbers denied to the extension user 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 configuring 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 SIP extension users.*

Upload Global Directory CSV file

- Login as System Engineer.
- Under **Advanced Settings**, click **Abbreviated Dialing**.
- Click **Upload/Download**.



- 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 **Browse** to **Select the .csv file to be uploaded** from the location on the local disk.

- Click **Upload**.

- All the contacts of the CSV file will be uploaded in the Global Directory. To view, click the Global Directory link.

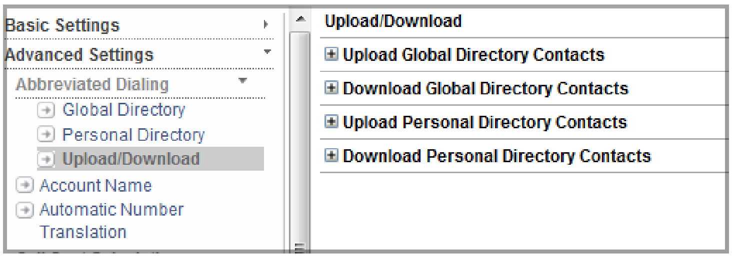
- Configure **Route Global Directory's Numbers using trunks** and **Least Cost Routing (LCR)** parameters as per your requirement. See "Configuring Global Directory".

- *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 with LDAP.*

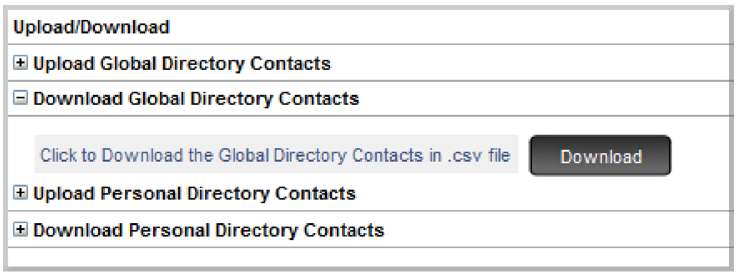
Download Global Directory CSV files

- Under **Advanced Settings**, click **Abbreviated Dialing**.

- Click **Upload/Download**.



- Click **Download Global Directory Contacts** to expand.



- Click **Download**.

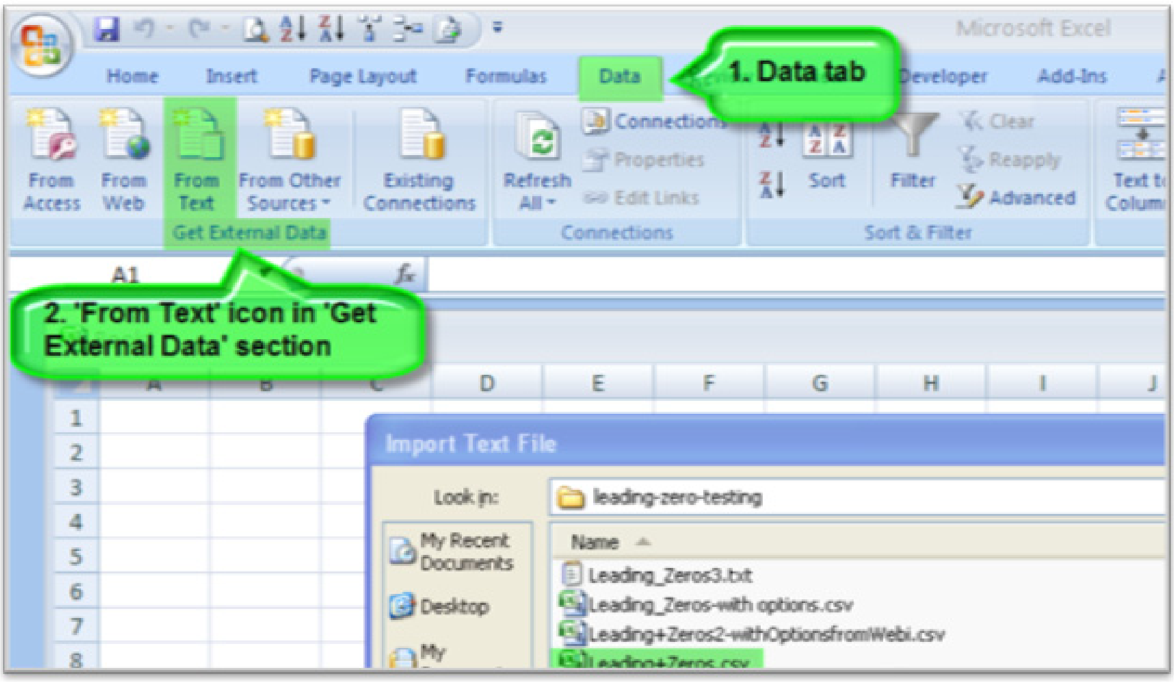
- You will get a prompt with an option to open the **Opening Global_Directory.csv** file or save the file to a location. Save the file on the local disk.

- To Open the **Global_Directory.csv** file from the location on the local disk, make sure you follow the steps given below:

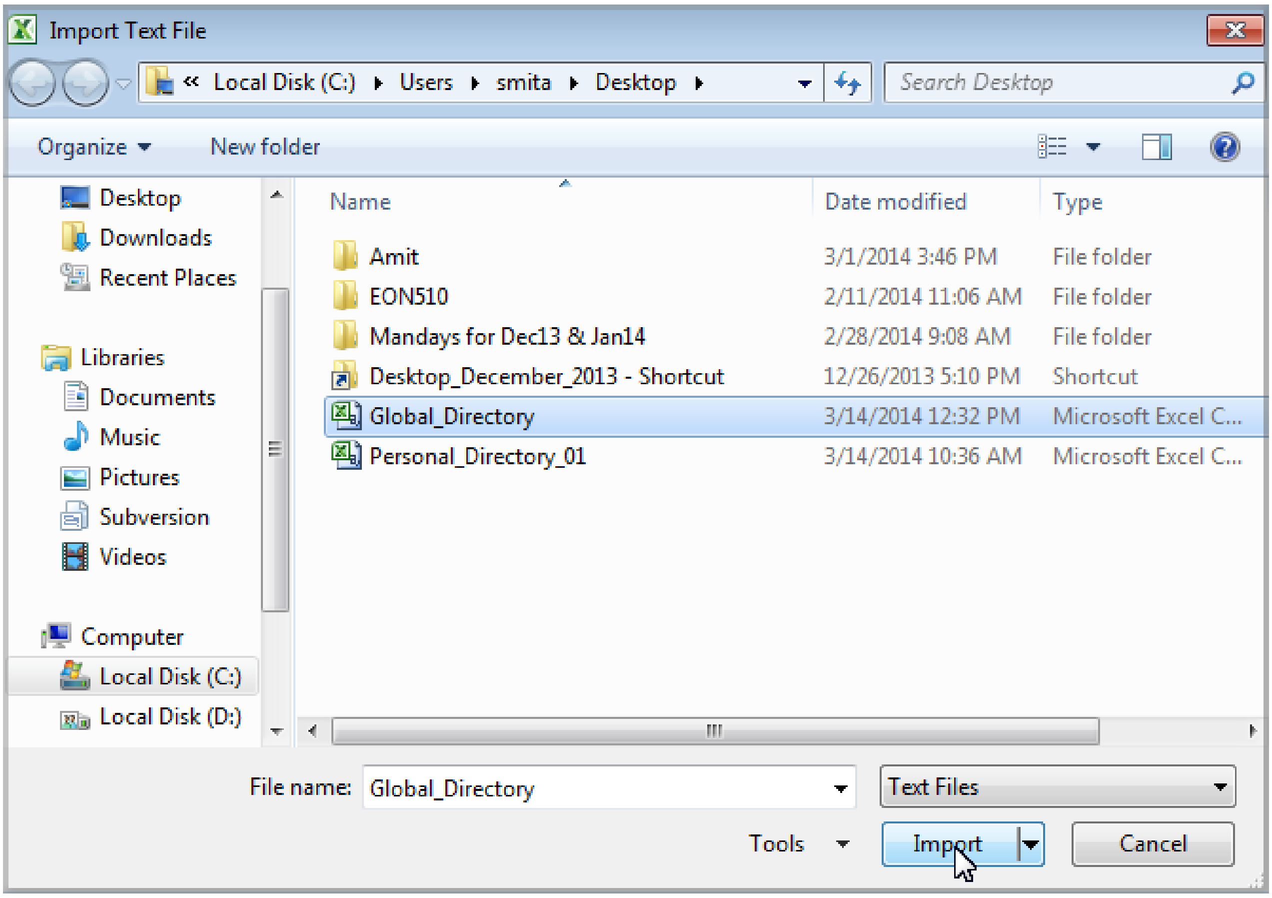
  - **DO NOT OPEN THE .CSV FILE DIRECTLY WITH EXCEL**!

  - Open a **New** worksheet in Excel.

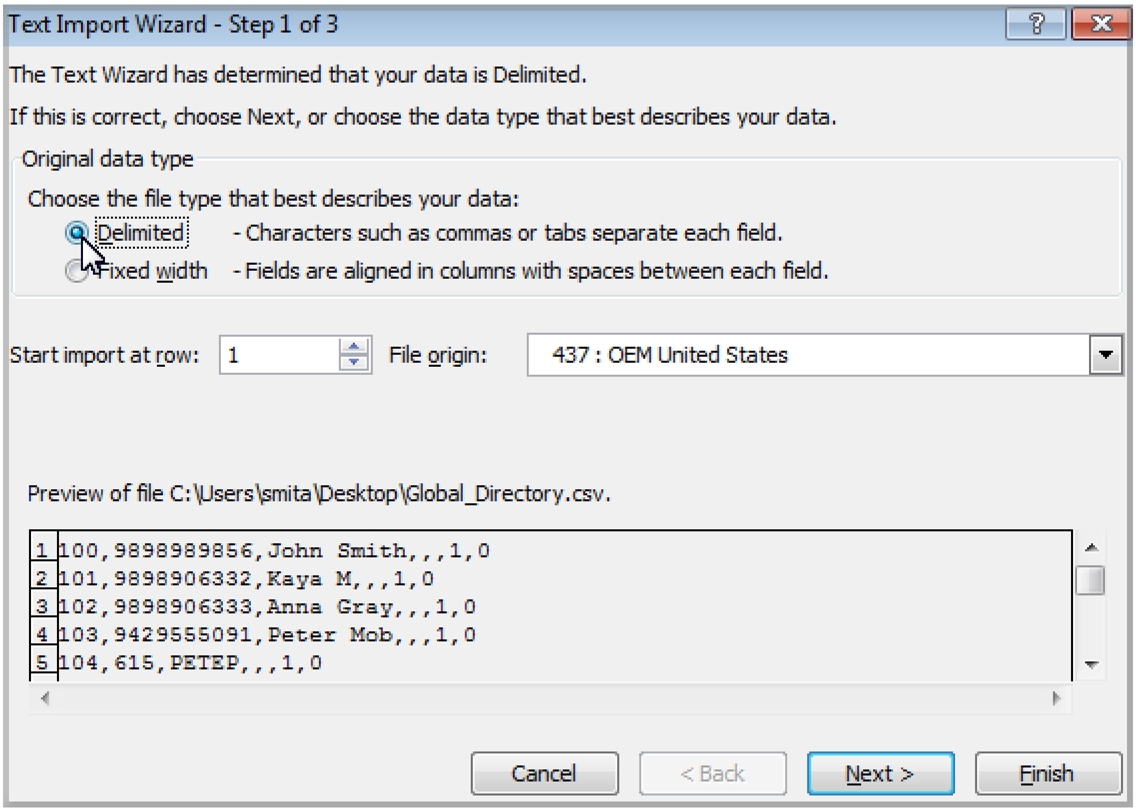
- Open the **Data** tab and select **From text button in the Get External Data**.



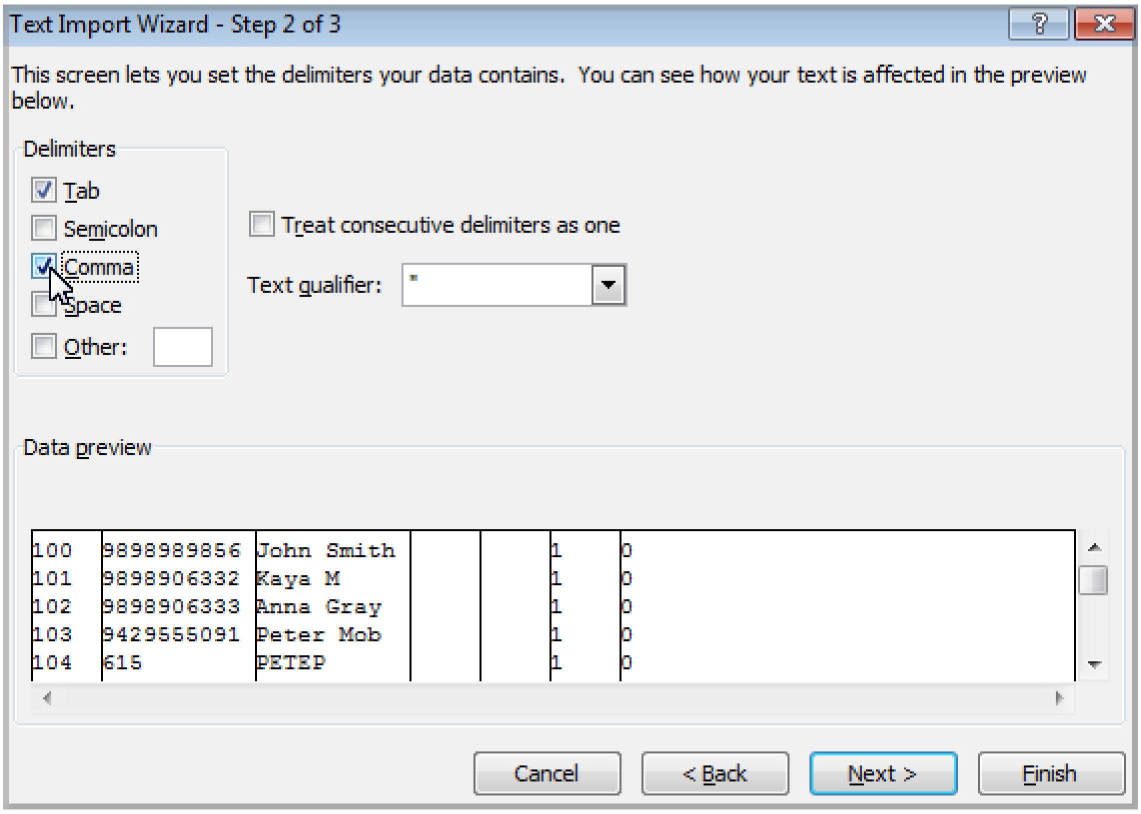
- Select your CSV file from the location on the local disk to import.



- In **Original data type** section, select **Delimited** radio button and click **Next**.

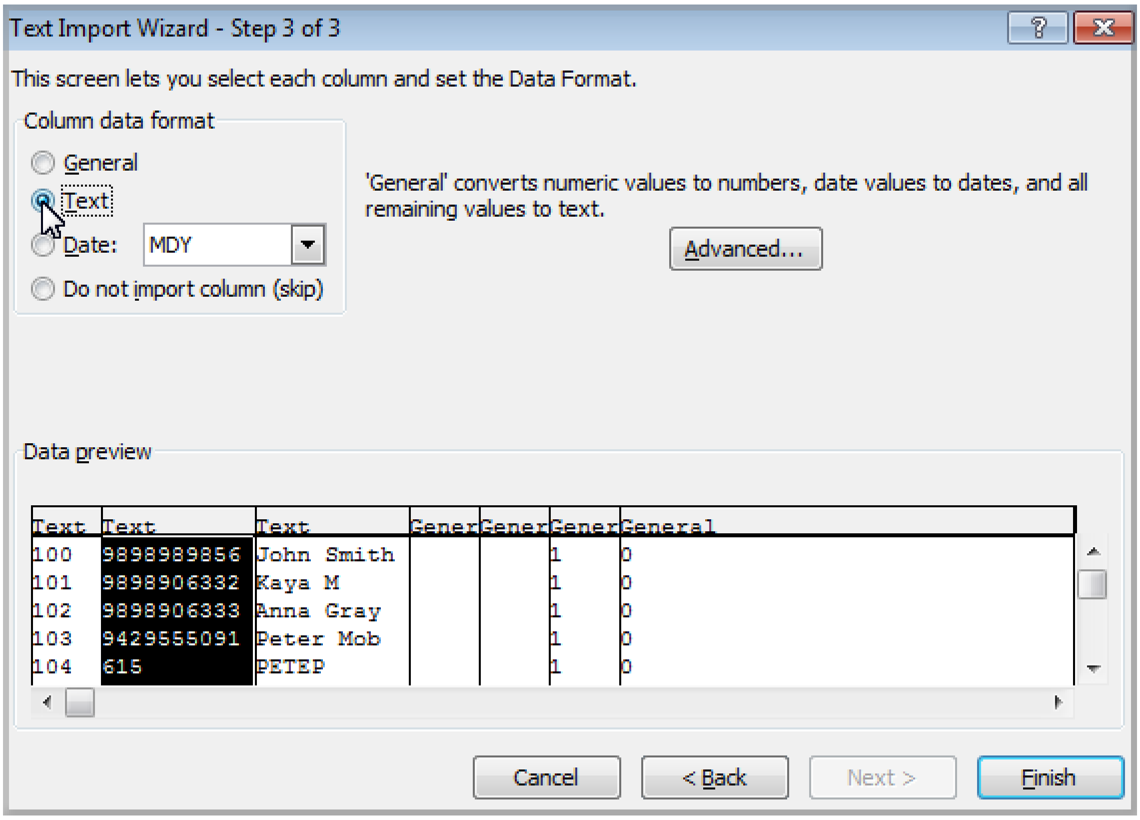


- In **Delimiters** select the **Comma** check box (column dividers will appear in preview) and click **Next**.



- 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.

Configuration of Personal Directory by Extension Users

Extension users can configure their own Personal Directories using their extension phones. The personal directory may be configured using SLT or Extended IP Phone.

If you are using a SLT, you will not be able to configure the Name of the contact in the directory.

For Extended IP Phone Users

- Press DSS Key assigned to Personal Directory (if programmed).
  OR
- Dial 1071.
- Enter Personal Memory Index (001 to 025)
- Enter Number of the contact (max. 16 digits).
- Press 'Enter' key.
- Enter Name of the contact.
- 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 (001 to 025).
- Dial Number of the contact (max. 16 digits).
- Press #*.
- Dial Trunk Access Code.
- You get confirmation tone.
- Replace handset

Configuration of Global Directory by Extension Users

Extension users can add, delete and edit contacts in Global Directory Part 1using their extension phones, provided that:

- their phone is an Extended IP 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 select the outgoing trunks for the contact numbers 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 configures Global Directory Part 1, the system will automatically assign the number and name to a free Memory Location. The system will use the trunks selected by the System Engineer to dial out the number added by the extension user.*

To configure Global Directory Part 1 from the Matrix 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 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.
- 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.
- The desired number will be dialed out.

Global Abbreviated Dialing

For 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 Global Directory Index number: 100 to 999.
- 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 Global Directory Index number: 100 to 999.
- The desired number will be dialed out.

Access Codes

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.
- Invoking a feature. Activating or deactivating a feature.

Accordingly Access Codes are classified into:

- **Extension Codes:** Codes used for calling SLT and SIP extensions. These codes are also commonly referred to as Extension numbers.

  Default Extension codes[100] are:
  - SLT: 21 to 22
  - SIP: Blank

- **Logical Group Codes:** Codes used for calling a group of extensions as in a Department group, a group of trunks.

  Default logical group codes:
  - Department groups: 391 to 395
  - Trunks access codes: 0 and 5

- **Feature Codes:** Codes used for invoking a feature.

  Default feature codes: there are different feature codes for every feature and function of SARVAM UCS. For example, '4' for Barge-In, '5' for Raid, '13' for Call Forward.

  Access codes may consist of single digits or a sequence of a maximum of 6 digits.

  You may change the default access codes, if required. For example: the default Operator code '9' can be changed to '0'.

How it works

Whenever an access code is dialed from an extension, the system matches each digit in the code with the access codes configured within the system to determine the instruction, like whether it is an extension it must call, or a trunk line it must grab, a port it has to activate. 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.

100. In reference to ETERNITY NENXIP50.

- When the second digit '3' is dialed, the system finds a match for '13'.
- As '13' is common for all Call Forward options[101], 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.

The different phases that a call typically passes through are:

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. Extension 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, with one held party.

'Idle' phase is when no code is dialed. In the 'Denied Phase' no code is allowed to be dialed.

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.

101. *Call forwarding options: Unconditional, When Busy, When No Reply, When Busy or No Reply.*

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, Extension and Logical Group Codes too must be unique and should not match with any of the features invoked in the 'Dial' phase.

How to configure

SARVAM UCS provides default Access Codes for extensions, 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 Codes table also indicates the call phase in which each feature is invoked.

Feature	Access Code	Call Phases					
		Dial	Routing	Blocked	Placed	Matured 2-Way	Matured 3-Way
Enter SE Programming Mode	1#91	Y				Y	
Enter SA Programming Mode	1#92	Y				Y	
Abbreviated Dialing	8	Y				Y	
Account Code by Name	1059	Y				Y	
Account Code by Number	1058	Y				Y	
Alarm	161	Y				Y	
Alarm Voice-guided	163						
Auto Call Back Set	2			Y	Y		
Auto Call Back Cancel	102	Y				Y	
Auto Redial Set	17	Y				Y	
Auto Redial Cancel	1070	Y				Y	
Barge-In	4			Y			
Blind Transfer to Voicemail	1078					Y	
Call Chaining	1050					Y	
Call Cost Display	1075	Y					
Call Forward	13	Y				Y	
Call Forward - When Not Registered	*13	Y					
Call Park	115					Y	
Call Park Retrieve	116	Y				Y	
Call Pick Up-Selective	12	Y				Y	
Call Pickup - Group	4	Y				Y	

Feature	Access Code	Call Phases					
		Dial	Routing	Blocked	Placed	Matured 2-Way	Matured 3-Way
Call Toggle	1						Y
Cancel All Features	1051	Y				Y	
Change User Password	114	Y				Y	
CLI Restriction	103	Y					
Conference	*3						Y
Conversation Recording	1095					Y	
COSEC Door Open	*7	Y					
Department Group Call Forward	1179	Y				Y	
Dial-In Conference	*19	Y				Y	
DISA Login	1079	Y					
DND	18	Y				Y	
DND Override	4		Y				
Dynamic Lock	14	Y				Y	
Emergency Conference	1177	Y					
Flashing on Trunk	*	Y				Y	
Forced Answer	5				Y		
Forced Release	#*			Y			
Transfer	Flash					Y	
Hot Desk	1091	Y					
Hotline	15	Y				Y	
Intercom	*5	Y				Y	
Interrupt Request	3			Y			
Invoke RCOC	**	Y					
Keypad Lock	-	Y					
Last Caller Recall	1092	Y				Y	
Leave Temp. / Rejoin Conf.	191						Y
Live Call Supervision	1098	Y				Y	
Meet Me Paging	1093	Y				Y	
Message Wait Set/Cancel	1076	Y				Y	
Presence	1097	Y					
Mute	1052	Y				Y	
Operator	9	Y				Y	Y
Paging	1074	Y				Y	
Personal Directory Program	1071	Y				Y	

Feature	Access Code	Call Phases					
		Dial	Routing	Blocked	Placed	Matured 2-Way	Matured 3-Way
PIN Dialing	*2	Y				Y	
Raid	5			Y			
Redial	7	Y				Y	
Reminder	162	Y				Y	
Reminder Voice-guided	164						
Retrieve New Message	1077	Y				Y	
Room Monitor	1073	Y				Y	
SA Mode	1072	Y				Y	
Scheduled Call Forward	1175	Y				Y	
Selective Port Access	69	Y				Y	
Self Ring Test	1057	Y					
Terminate Conference	190						Y
Trunk Reserve	6			Y			
User Absent/Present	104	Y				Y	
Voice Help	1090	Y				Y	
Voice Mail	390	Y				Y	
Walk-In Class Of Service	111	Y				Y	
Trunk Access Code (TAC) 1	0	Y				Y	
Trunk Access Code (TAC) 2	5	Y				Y	
Trunk Access Code (TAC) 3		Y				Y	
Trunk Access Code (TAC) 4		Y				Y	
Trunk Access Code (TAC) 5		Y				Y	
Trunk Access Code (TAC) 6		Y				Y	
Department Group 1	391	Y				Y	
Department Group 2	392	Y				Y	
Department Group 3	393	Y				Y	
Department Group 4	394	Y				Y	
Department Group 5	395	Y				Y	

You can either use the default Access Codes or change them to suit your preferences.

To change Access Codes, see “Extension and Feature Codes” under *Basic Settings* for instructions.

To change the Access Code of General Mailbox, see “General Mailbox Settings” under *Voice Mail System*.

Account Codes

Account Codes are a very useful feature for business consultants, law firms, advertising and media agencies, and the like, which interact 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, which is entered in the Account Name List.

With an Account Code and Name assigned, 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 extension user may dial the Account Code *before* dialing the external number,
  Or
  when *in speech* with the client/third party (by putting the party on hold).
- Call details for these calls are recorded 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.

  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

SARVAM UCS can be configured 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 SIP Phone, 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, extensions must be allowed this feature in their Class of Service (CoS).*
- *If you want to use Account Names, you must configure the Account Name List.*
- *When the Forced Account Code is enabled on a trunk, the system will ask the extension user to enter the account code irrespective of the method of dialing, whether 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 SIP Phone.

How to configure

For Account Code to work, the System Engineer must:

- Enable 'Account Codes' feature in the Class of Service (CoS) of the extensions to which this feature is to be allowed.
- Prepare and configure the Account Name List, if you are going to assign names to the account codes.
- If Forced Account Code is to be used, enable 'Forced Account Code' check box on the trunks through which calls using account codes are to be made.

Preparing Account Name List

In consultation with the User, the System Engineer may:

- 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, i.e., 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

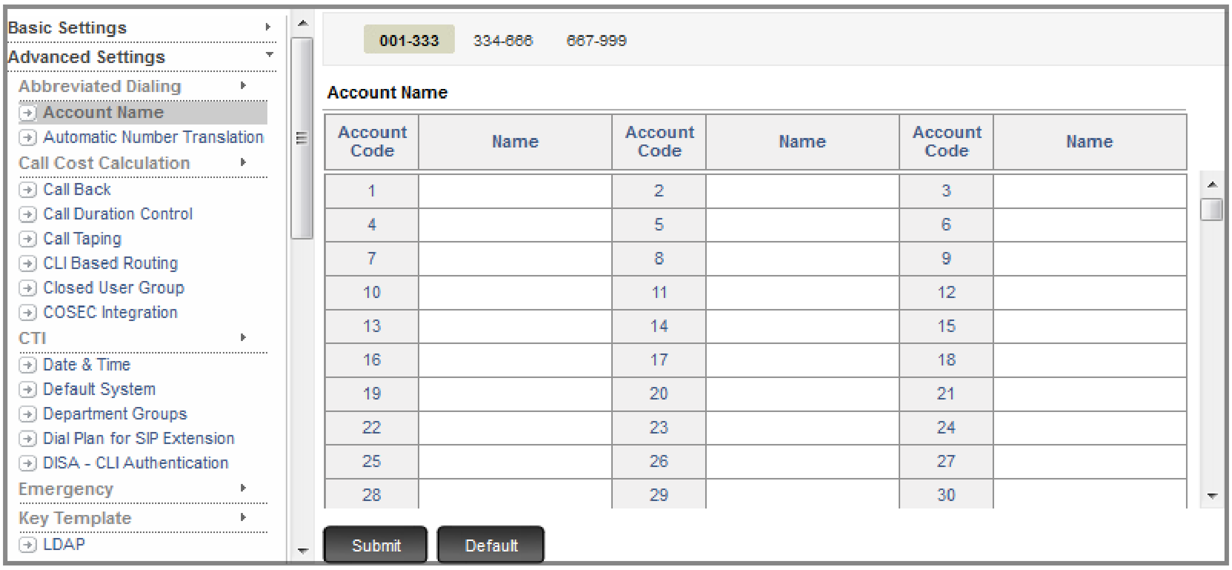
  You need not follow a cardinal numbering sequence when assigning Account Codes.

  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

To configure Account Name list,

- Login as System Engineer.
- Under **Advanced Settings**, click **Account Name**.

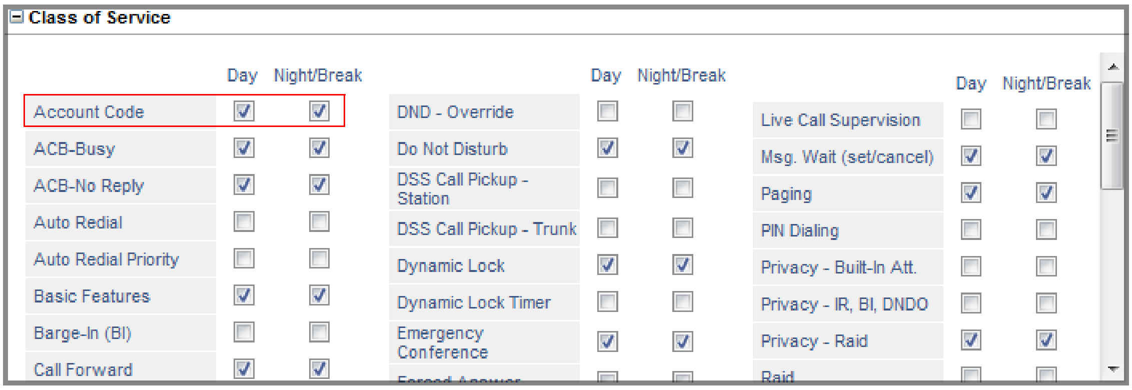


- Enter the Names of the clients against the account codes you have assigned to them. Refer the paper with the two-column table you created.
- Click **Submit**.

To enable Account Code on extensions,

- Click **Basic Settings**.
- Click the extension type—SLT, SIP—to which you want to allow this feature.
- Select the extension number to which you want to allow this feature.

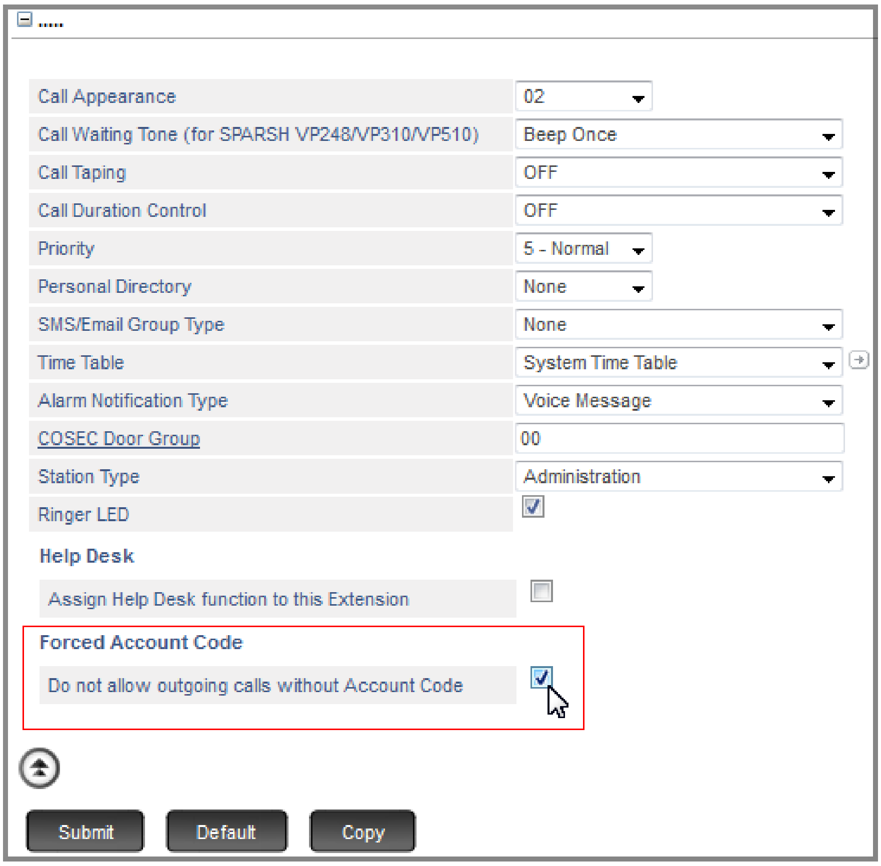
- Click **Class of Service** to expand.
- Enable **Account Code** in the **Class of Service** for the Day and Night/Break.



- Click **Submit.**

To enable **Forced Account Code** on the extension,

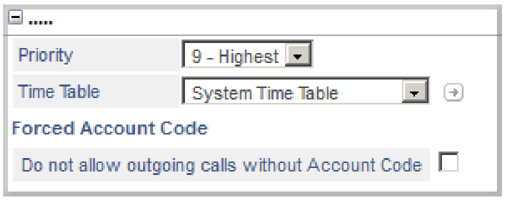
- Click the **More...** link to expand.



- Select the **Do not allow outgoing calls without Account Code** check box to enable this feature on the extension.
- Click **Submit.**

To enable **Forced Account Code** on trunks,

- Click the link of the trunk type—CO, Mobile, SIP—to which you want to allow this feature.
- Select the trunk you on which you want to assign this feature.
- On the respective trunk port page, click the **More...** link.
- Click **Forced Account Code** on the trunk.



- Select the **Do not allow outgoing calls without Account Code** check box to enable this feature on the trunk.
- Click **Submit**.

How to use

Account Codes can be dialed in two ways: by Number and by Names.

Account codes, i.e. number and names, can be dialed:

- before making the call,
- during the call,
- when grabbing a trunk (if Forced Account Code check box 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

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 check box is enabled:

For 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 an Extended IP Phone.

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 Extended IP Phone, 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 'Transfer' Key.
- 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 Extended IP 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 dialing 'flash' 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.

See “Station Message Detail Recording–Report”, for more detailed instructions on printing reports using filters.

AC Impedance Test

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.

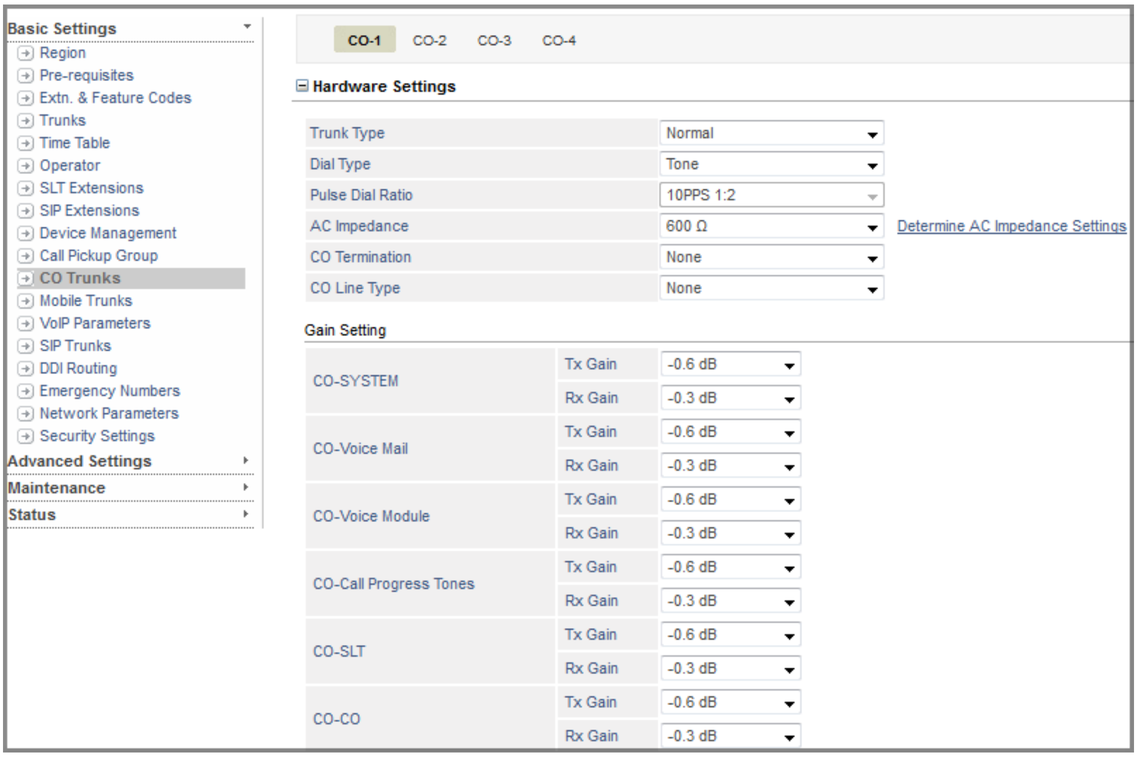
⚠ *The DSS Key assigned to this CO Trunk will display status as busy to all the user, for the duration of the test.*

Configuring Using Jeeves

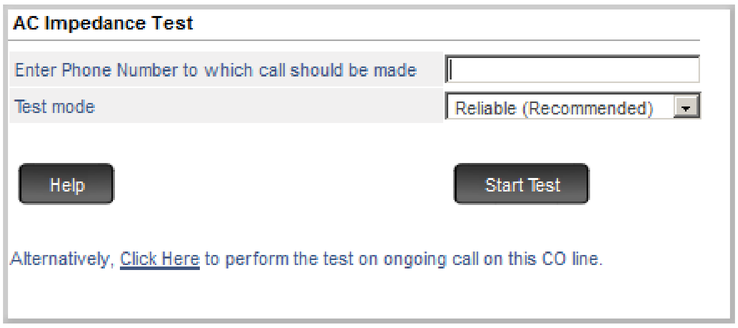
You can conduct the test after you have set the relevant test parameters. To do this,

- Login as System Engineer.
- Under **Basic Settings**, click **CO Trunks**. Select the desired CO Trunk.

- Click **Hardware Settings** to expand.



- Click **Determine AC Impedance Settings** link to view the **AC Impedance Test** window.



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.

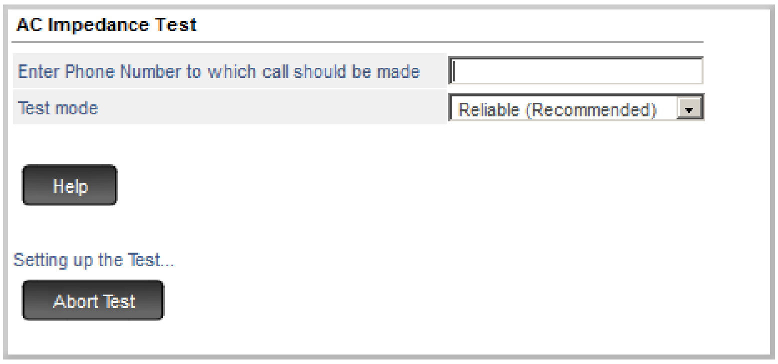
If you are using a mobile phone number, be sure the handset of the configured number supports the Mute function.

- 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 **Start Test**. The system calls the phone number you configured. The message '*Setting up the Test....*' appears on your screen.



While the test is being conducted, in SARVAM UCS you will hear pulsating tone on all the 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 on the CO trunk you are testing.

  By default, the Answer Supervision selected as 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.

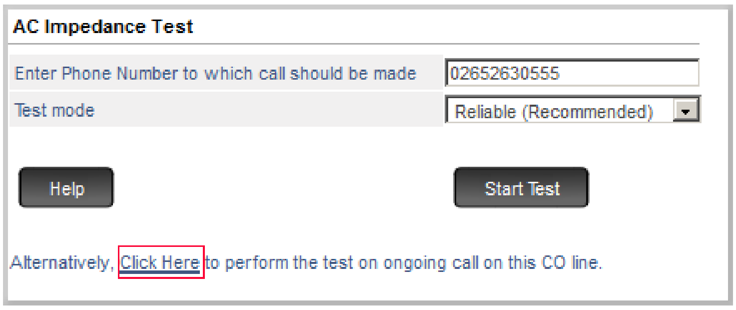
- After 5 seconds of Music-on-Hold, you will hear the test signals being transmitted by the system for the duration of the test. The message '*Test running successfully...*' appears on your screen.

- On completion of the test, the system will automatically disconnect the call. The message '*Test completed*' appears on your screen.

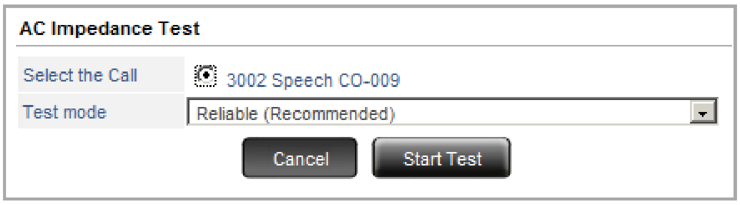
  However, if you wish to abort the test midway, you may click **Abort Test**.

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.

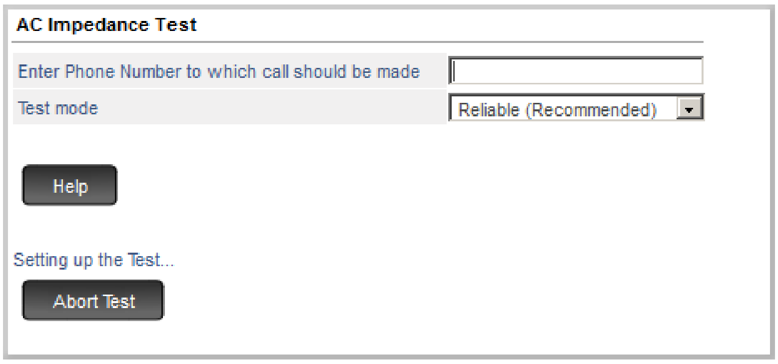


- Select the **Test Mode**. You may select **Reliable (Recommended)** or **Accurate**.

  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.

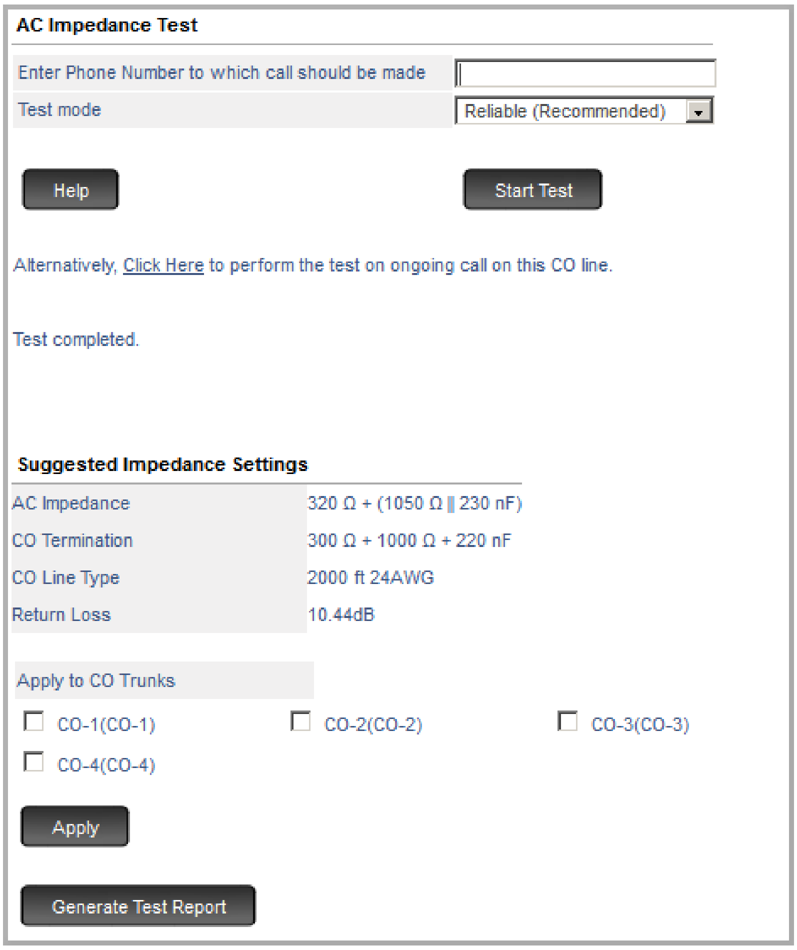
- Click **Start Test**. The message '*Setting up the Test....*' appears on your screen.



While the test is being conducted, in SARVAM UCS you will hear pulsating tone on all the ports of the system.

- 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 **Abort Test**.



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 **Apply**.
- Verify the settings by making a trial call from an Extended IP Phone. 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 Hardware Settings for each of these trunks.

- To generate the detailed test report, click **Generate Test Report**.

Suggested Impedance Settings

AC Impedance	320 Ω + (1050 Ω ‖ 230 nF)
CO Termination	300 Ω + 1000 Ω + 220 nF
CO Line Type	2000 ft 24AWG
Return Loss	10.44dB

Apply to CO Trunks

☐ CO-1(CO-1) ☐ CO-2(CO-2) ☐ CO-3(CO-3)

☐ CO-4(CO-4)

Apply

Generate Test Report

The detailed test report appears in a new window.

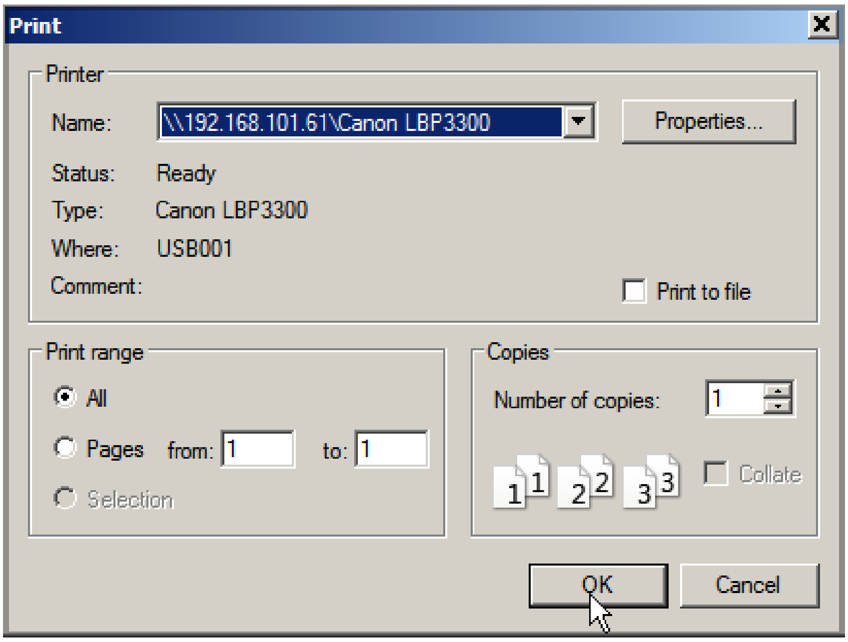
AC Impedance Test Detail Report

AC Impedance	CO Termination	CO Line Type	Return Loss
600 Ω	None	2000 ft 22AWG	6.41dB
600 Ω	150 Ω + 510 Ω + 47 nF	2000 ft 22AWG	9.61dB
600 Ω	220 Ω + 820 Ω + 150 nF	2000 ft 22AWG	8.86dB
600 Ω	**600 Ω**	**2000 ft 22AWG**	**10.10dB**
600 Ω	600 Ω + 1.5 µF	2000 ft 22AWG	0.00dB
600 Ω	900 Ω + 2.16 µF	2000 ft 22AWG	0.00dB
600 Ω	1200 Ω + 376 Ω + 112 nF	2000 ft 22AWG	0.00dB
270 Ω + (750 Ω ‖ 150 nF)	220 Ω + 120 Ω + 115 nF	2000 ft 22AWG	0.00dB
220 Ω + (820 Ω ‖ 115 nF)	220 Ω + 820 Ω + 115 nF	2000 ft 22AWG	0.00dB
370 Ω + (620 Ω ‖ 310 nF)	220 Ω + 820 Ω + 120 nF	2000 ft 22AWG	0.00dB
370 Ω + (620 Ω ‖ 310 nF)	370 Ω + 620 Ω + 310 nF	2000 ft 22AWG	0.00dB
320 Ω + (1050 Ω ‖ 230 nF)	200 Ω + 560 Ω + 100 nF	2000 ft 22AWG	0.00dB
320 Ω + (1050 Ω ‖ 230 nF)	270 Ω + 750 Ω + 150 nF	2000 ft 22AWG	0.00dB
320 Ω + (1050 Ω ‖ 230 nF)	300 Ω + 1000 Ω + 220 nF	2000 ft 22AWG	0.00dB
320 Ω + (1050 Ω ‖ 230 nF)	370 Ω + 620 Ω + 310 nF	2000 ft 22AWG	0.00dB
600 Ω	None	2000 ft 24AWG	0.00dB
600 Ω	150 Ω + 510 Ω + 47 nF	2000 ft 24AWG	0.00dB
600 Ω	220 Ω + 820 Ω + 150 nF	2000 ft 24AWG	0.00dB
600 Ω	600 Ω	2000 ft 24AWG	0.00dB
600 Ω	600 Ω + 1.5 µF	2000 ft 24AWG	0.00dB
600 Ω	900 Ω + 2.16 µF	2000 ft 24AWG	0.00dB

Print

- 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.

Alarms

SARVAM UCS offers Alarm feature on all extensions.

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.

Alarms can be voice-guided, if the Voice Mail System (VMS) is present in your ETERNITY NENX. SARVAM UCS can register as many as 60 Alarms set by the Operator and extension users.

How it works

Personalized Alarm

When the Alarm serving mechanism is configured as 'Personalized',

- The Operator phone rings first[102], 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.
- 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.
- If the extension is busy[103], the Operator phone will display a text message notifying that the extension number is 'Busy'.

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102. *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.*
103. *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'.*

- The Operator can now choose to:
  - inform the extension user about the alarm in person or send someone to do it.
    OR
  - try the busy extension again.
    OR
  - set “Auto Call Back (ACB)”.

Personal Alarms will work even if the extension user has set DND or Call Forward.

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 the Matrix Extended IP Phone, 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 the Voice Mail, or routed to the Operator, depending upon the Alarm Notification Type configured by the System Engineer.

  The System Engineer may consult with the Enterprise to decide which of these options is to be configured 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.
- 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.
- If the extension phone is busy the system will continue to make Alarm Attempts at the Alarm Interval configured. 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 configured, 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. The Alarm Acknowledgement Code is not configurable.

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.

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, 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.

- *SARVAM UCS can register as many as 60 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, i.e. 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 or the extension user attempts to cancel alarms from the phone, the system will cancel all the 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 configurable.*
- *Alarms can be set for all extensions of SARVAM UCS, including the Operator phone also.*
- *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.

- **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.

- **Alarm Ring Timer:** This is 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. The range of this timer is 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.

- **Number of Alarm Attempts:** This is the number of attempts the system makes 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.

- **Alarm Attempt Interval:** This is the time period between each Alarm Call attempt. By default, the Alarm Attempt Interval is set to 5 minutes. The range of this interval is from 1 to 9.

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- **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 **Configurable Alarm Type** check box.

  When this check box 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 check box is disabled.

- **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 you want to offer only **Automated** Alarms to 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** check box.

  When this check box 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 check box is disabled.

- *When both check boxes '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' check box is disabled, but the 'Configurable Alarm Category' check box is enabled, the system will set 'Once Only' alarm calls, but will give the option of selecting 'Automated' or 'Personalized' as the serving mechanism.*
- *Similarly, if 'Configurable Alarm Type' is enabled, but the 'Configurable Alarm Category' check box is disabled, the system will allow both 'Once Only' and 'Daily' alarms to be set, but the serving mechanism will be 'Automated'.*

- **Voice Guided Alarm Verification:** For Voice-guided Alarms, the VMS of the system allows you to enable/disable the Alarm Verification, allowing extension users to confirm the Time set for an alarm and Date and time set as a reminder. By default, this check box is enabled.

The check boxes '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 check boxes are disabled.

- **Alarm Notification Type:** This is the means of notifying the extension user about the Alarm call. SARVAM UCS supports the following types of Alarm Notifications:
  - **Voice Mail:** The extension user is connected to the Voice Mail System.
  - **Voice Message:** The extension user is played the message recorded in the Voice Module assigned to Alarm.
  - **Music-On-Hold:** The extension user is played music on hold.
  - **Routing Group:** The alarm call is routed to the group of desired extensions configured as the Operator, so that the alarm request can be served.
- **Macros:** These are short codes for simulating the Alarm call. These are used for SLTs which have special function keys to 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, a SLT has a special function key for Alarm calls which sends the string **53** to the system. The system can be configured to translate **53** received from the SLT in to the feature access code for Voice-guided Alarm calls, **163**.

How to configure

To configure the Alarms feature, do the following:

- Select the **Alarm Notification Type** for the extensions.
- Configure, as required, the **Alarm Call related parameters**: Number of Attempts, Attempt Interval, Configurable Alarm Type and Category, and Snooze.
- Configure **Macros**, if the SLT extension has special function keys, and you want to a function key for the Alarm feature.

Now,

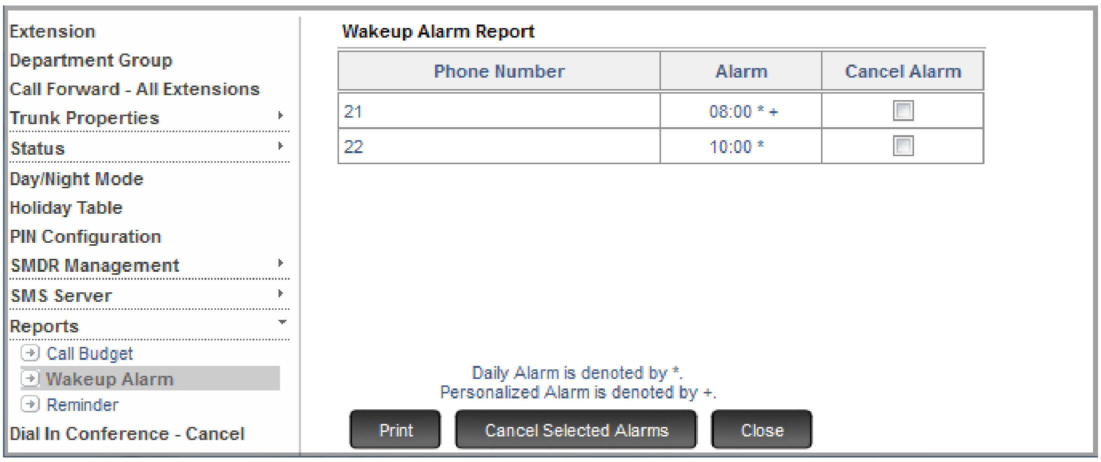
- Login as System Engineer.
- To configure **Alarm Notification Type** for the extension, see “Alarm Notification Type” in “SLT Extensions” and “Alarm Notification Type” in “SIP Extensions” under Basic Settings.
- To configure Alarm parameters, see “Alarm” in “System Parameters”.
- If you select Music-On-Hold as Alarm Notification type, no further configuration is required.
- 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 Voice Mail as the Alarm Notification Type, make sure you have installed the Voice Mail System module in the system.
- If you have selected Routing Group as Alarm Notification Type, you must create a Routing Group for Operator. See “Operator” for instructions.
- To program SLTs with special Alarm function key and to create macro for a Extended IP Phone key, see “Macros”.

Viewing and Printing Alarm Status

The Operator can view the status of Alarms that are yet to be served from the System Administrator pages of Jeeves.

To view Alarm Status,

- Login as System Administrator.
- Click **Reports**.
- Click the **Wakeup Alarm**.



The unserved Alarm calls appear on the page.

- To cancel any of the unserved Alarm calls,
    - select the **Cancel Alarm** check box of the extension number for which you want to cancel the alarm.
    - click **Cancel Selected Alarms** at the bottom of the page.
- To print this page, click **Print**.

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 Extended IP Phone, SLT and from System Administrator mode.

Operator using Extended IP Phone

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

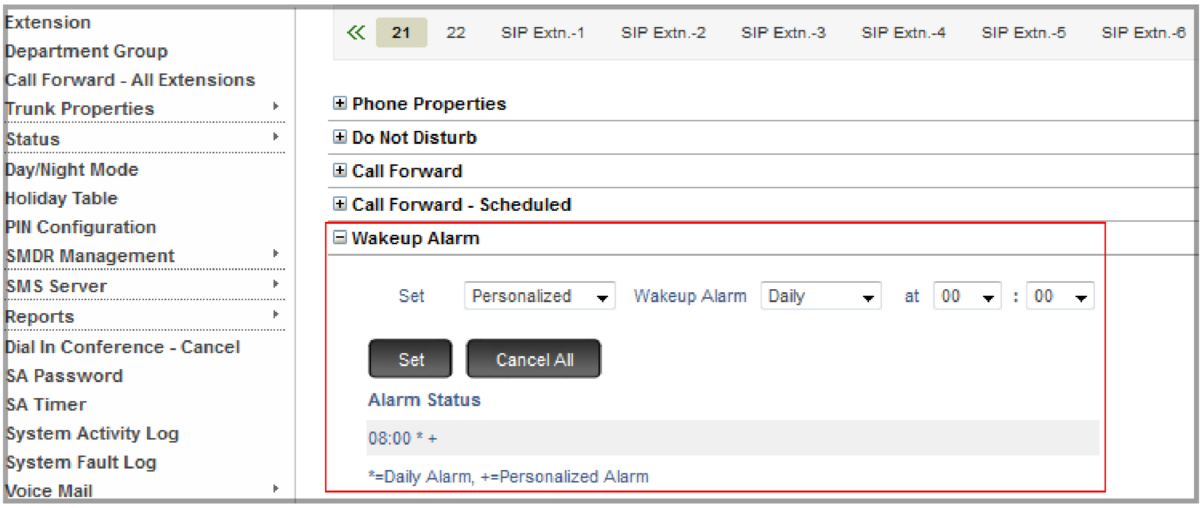
- Pick up the handset.
- Dial ***1072-003***
- Enter Extension number.
- Enter Time in HH:MM
- Dial 1 for Once Only or Dial 2 for Daily
- Dial 1 for Personalized or Dial 2 for Automated.
- Press 'Enter' key to set Alarm.
- You get a confirmation tone and a text message with the phone number for which 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:

- Login as System Administrator.
- Click **Extensions**.
- Click the desired Extension tab. The Extension page 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.

Alarms set/cancel by Extension Users

Extension Users using Extended IP Phone

If the extension user has Extended IP Phone, 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 Extended IP Phone or a SLT to print Alarm Status Reports.

For 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.

Alternate Number Dialing

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 configure, the Alternate Number Groups in the Global Directory. With the alternate numbers configured 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 configure 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[104], until the call gets through.
- When any of the alternate numbers gets through, the system will give a ring on your extension.

104. The number of auto redial attempts depends on the Auto Redial Count configured 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.

- *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, 100-399 (Part 1), 400-699 (Part 2), and 700-999 (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, SARVAM UCS will try the alternate numbers related to 8100.*

How to configure

To use Alternate Number Dialing, you must do the following:

1. Make a List of Alternate Numbers.
2. Create Alternate Number Groups.
3. Configure 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.

- *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 configure the numbers in the directory, before you begin creating Alternate Number Groups. Refer the topic "Abbreviated Dialing" for instructions on configuring 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 are allowed in the CoS of the extensions.*

Preparing Alternate Number List

In consultation with the extension users, 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

Name of the Contact	Alternate Numbers
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

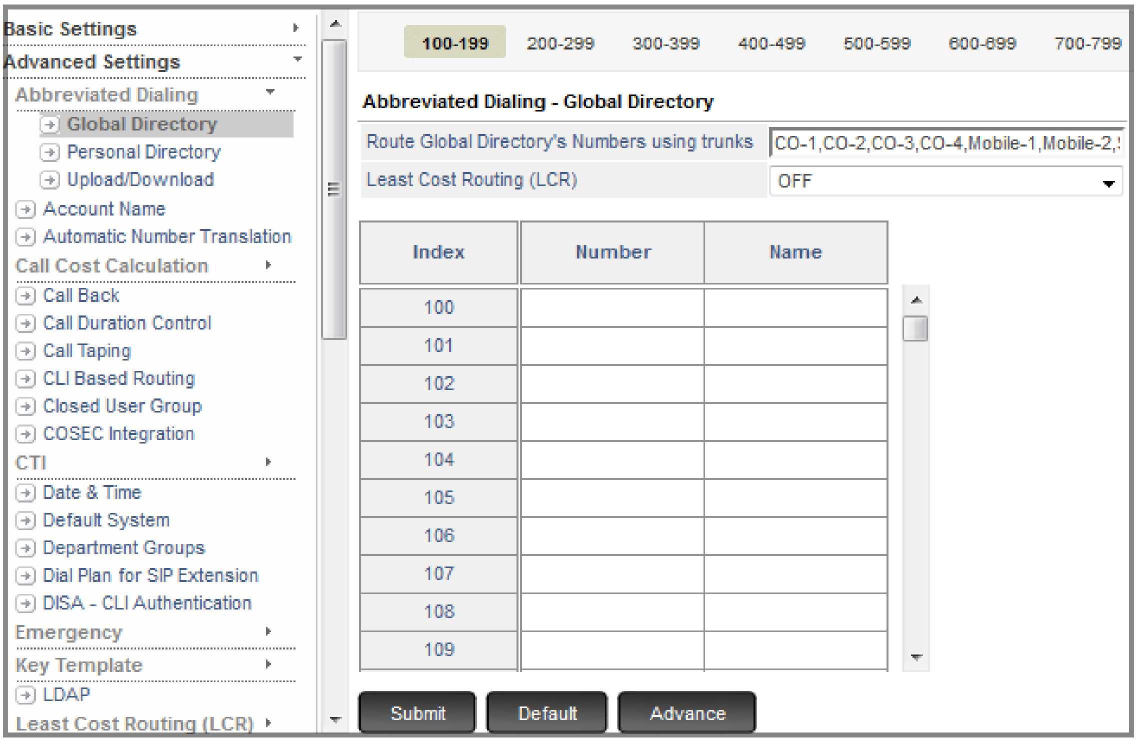
Configuring Alternate Number Groups in Global Directory

Make sure that the numbers on the list are also configured in the Global Directory. If any of these numbers do not exist in the Global Directory, configure them in the Global Directory first. Refer the topic “Abbreviated Dialing” for instructions on configuring the Global Directory.

To create Alternate Number Groups and configure them in the Global Directory,

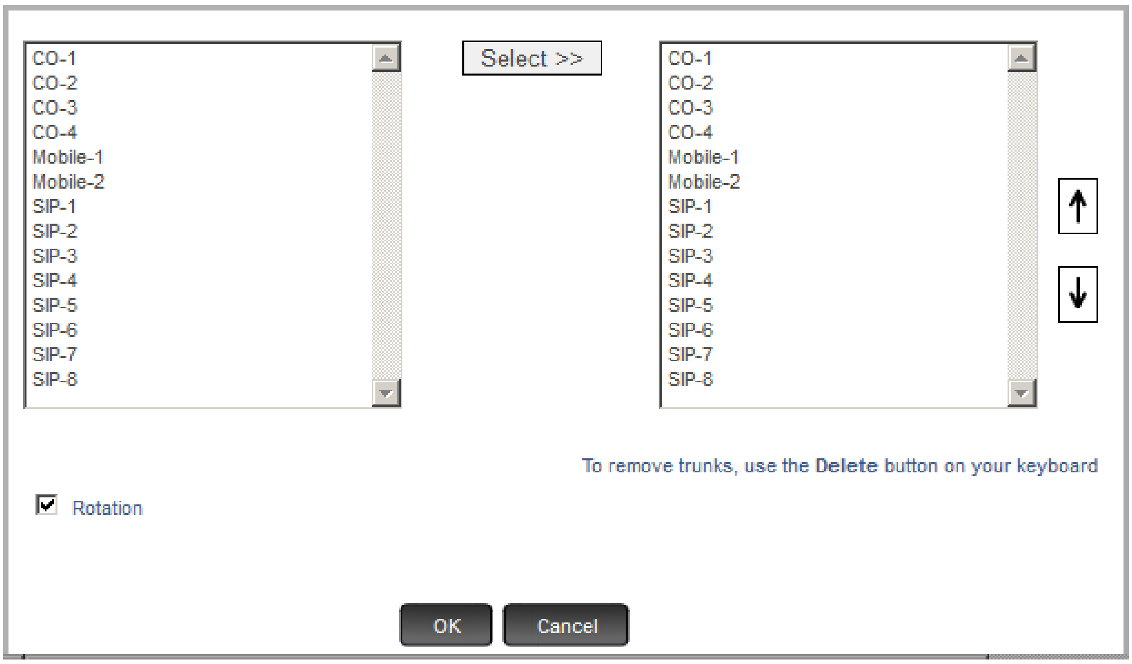
- Login as System Engineer.
- Under **Advanced Settings**, click **Abbreviated Dialing**.
- Click **Global Directory**.

- Click **Advance** on this page.



- If you have not configured the Global Directory, you may do so now.

- Double click the **Route Global Directory's numbers using Trunks** field, and select the trunks to be used for routing the numbers of the Global Directory dialed out by extensions.



- You may enable **LCR** on the selected trunks.

- Enter the contact's **Number** you wish to store. The number may consist of 16 digits (maximum).

- Enter the contact **Name** against the Number. The Name may contain up to 12 characters (maximum).
- In the **Alternate Number Group** column, enter the number of the Alternate Number Group you assigned to the numbers.

  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	Number	Name	Alternate Number Group
100	2640459	Midas Biz	001
101	2631235	Midas Biz	001
102	2635589	Midas Biz	001
103	2565590	Midas Biz	001
104	2788856	GoodLife Inn	004
105	022281110001	Jet Set	002
106	022281110002	Jet Set	002
107	033298765432	R. Mendez	000
108	2640075	Bacchus	003
109	2640076	Bacchus	003
:	:	:	:
129	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 configured at different memory locations in the Global Directory.

  In the above example, one number of the GoodLife Inn is entered 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** to save entries.
- Enable the features 'Last Number Redial' and 'Global Directory', in the Class of Service 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, all extensions are allowed Last Number Redial in their Class of Service. However, only Global Directory Part 1 is enabled by default in their CoS.

  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 configured in Global Directory Part 2, extensions must have Global Directory Part 2 in their Class of Service.

Refer the topic “Class of Service (CoS)” for instructions on configuring CoS of extensions.

How to use

Confirm with your System Engineer that

- Alternate Number Groups are configured 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”.

Apple Push Notification Service Support

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.

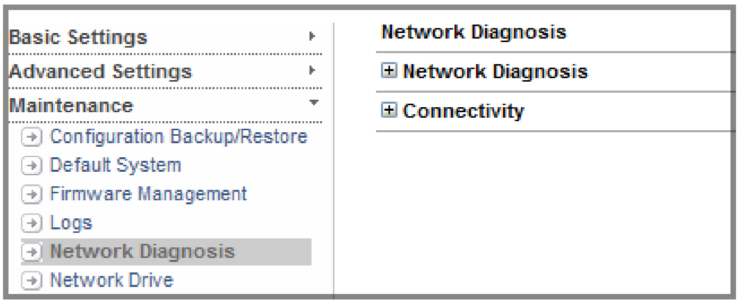
APNS Connectivity

A connectivity between the system and the APNS Server is required so that the Push notifications can be sent to the clients.

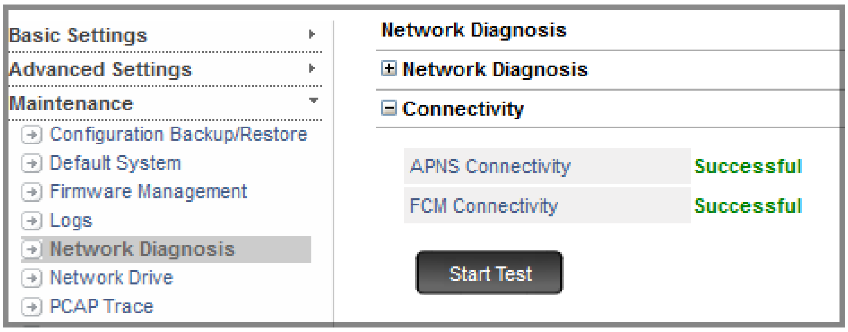
To check the APNS connectivity status,

- Login as System Engineer.

- Under **Maintenance**, click **Network Diagnosis**.



- 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 Answer

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 Extended IP Phones only.

How it works

With Auto Answer set on an Extended IP Phone, whenever a call lands on the Extended IP Phone extension,

- the extension rings for the duration of the Auto Answer Timer[105]. This timer is configured, and by default it is set to 1 second.
- on the expiry of the Auto Answer Timer, the system plays a beep[106] 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:

- Enable Auto Answer in the Extended IP Phone Extensions.
- 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.
- Enable Headset Connectivity, if headset is to be used for Auto Answer.

You can do the above configuration using Jeeves, or an extension users can also configure the above parameters using the Phone Menu of Extended IP Phone. See “How to use” later in this topic.

Configuring Auto Answer using Jeeves

For instructions, see “Configuring Matrix SPARSH VP248” and “Configuring Matrix SPARSH VP310” and “Configuring Matrix SPARSH VP510”.

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105. This timer defines the time in seconds that the phone should wait before going OFF-Hook to answer incoming calls.
106. Beep is not supported on Extended IP Phones.

How to use

Extension users can set/cancel Auto Answer and enable Headset connectivity from their Extended IP Phone by navigating the Menu of the phone.

To set Auto Answer:

- Press the DSS Key assigned to Auto Answer function[107].
  OR
- Press 'Enter' Key.
- Scroll down to select 'Phone Settings'; press Enter Key.
- Enter Your User Password.
- 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.
  OR
- Press 'Enter' Key.
- Scroll down to select 'Phone Settings'; press Enter Key.
- Enter Your User Password.
- 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.

107. *This function must have been programmed by the System Engineer on a DSS Key of the phone.*

Auto Attendant

Auto attendant feature 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* of SARVAM UCS, greets the caller and prompts the caller to dial the desired extension number. The call is then transferred to the extension number dialed by the caller.

SARVAM UCS also offers *Delayed Built-In Auto Attendant*, whereby incoming calls to Extensions defined as the landing destinations, 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.

Auto Attendant can be used by regular callers who know extension numbers to reach the desired extensions without Operator assistance. Thus, Auto Attendant reduces call traffic on the Operator extension, saves call set-up and transfer time for the callers. It is particularly useful during non-working hours and holidays, and projects a professional image of the organization.

*Delayed Built-In Auto Attendant works only when you have selected **Operator** or **Extensions** as the landing destination for incoming calls on the trunk.*

*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, enable **Privacy from Built-In Auto Attendant** in their Class of Service.* *See "Privacy".*

How it works

Auto Attendant can be configured on all trunk types—CO, Mobile, SIP—for the Day (working hours), Break hours and Night (non-working hours).

When configuring Auto Attendant on a trunk, you may select either the Built-In Auto Attendant or the Voice Mail Auto Attendant as the destination for incoming calls.

Built-In Auto Attendant

When Built-In Auto Attendant of SARVAM UCS is selected as the destination for incoming calls on a trunk, this is how Built-In Auto Attendant 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. The caller gets Ring Back Tone from the CO network during this period.
- The system greets the caller with the pre-recorded voice message called the *Built-In Auto Attendant Welcome Greeting* for the current time zone. A Voice Module must be assigned for the Built-In Auto Attendant Welcome Greeting.
- The Built-In Auto Attendant Welcome Greeting message is played once.

  If no voice module is assigned as Welcome Greeting, the system will play music-on-hold after answering the call 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 routing it to the Operator. For this, you need to enable the ***Disconnect Built-In Auto Attendant call, when caller does not dial any digit*** *check box in the System Parameters. When this check box 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 *Wrong Dial* voice message to the caller. This message is played once. The system waits for the duration of the *Built-In Auto Attendant Error Tone Timer* (default: 5 seconds).

  If Wrong Dial Voice Message is not configured, 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 configured, the caller will hear Busy Tone, played for duration of the *Built-In Auto Attendant Busy Tone Timer* (default: 15 seconds), followed by the Built-In Auto Attendant Dial Prompt.

You may enable the ***Disconnect Built-In Auto Attendant Call, when dialed number is busy*** *check box in the System Parameters to disconnect the call, if the dialed extension is busy.*

- The dialed extension is free. The system calls the extension and plays *Built-In Auto Attendant Ring Back Tone Message* (if configured) 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 configured) or plays Ring Back Tone to the caller.

If there is no reply from the dialed extension, you can disconnect the call instead of routing it to the Operator by enabling the ***Disconnect Built-In Auto Attendant call, when dialed number is not responding*** *check box in the System Parameters.*

Voice Mail Auto Attendant

SARVAM UCS supports 4 simultaneous VMS calls. 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 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 and processes the call further according to the “Voice Mail Auto Attendant Menu” you assigned to the trunk.

Delayed Auto Attendant

You can use Delayed Auto Attendant for incoming calls that are not answered by the landing destination (Extensions) within a certain time period, to be handled either by the Built-In or the Voice Mail Auto Attendant.

When you use Delayed Auto Attendant,

- As a call lands on a trunk, the system checks the Incoming Call Route configured for the current time zone for the trunk.
- On finding *Operator* or *Extensions* as the landing destination, the system rings on the destination extension for the duration of time defined for ringing the extension (default: 10 seconds).

- If no reply is received from the extensions, the system routes the call to the auto attendant you selected—Built-In or Voice Mail Auto Attendant.
- The call is processed further by the auto attendant you selected.

How to configure

To use **Voice Mail Auto Attendant** on trunks, do the following:

- Make a list of the trunks by their port type (“CO Trunks”, “Mobile Trunks”, “SIP Trunks”) and port number on which you want to use the Voice Mail Auto Attendant.
- Configure Welcome and Greeting messages. You may either use the default, pre-recorded welcome messages of the VMS, or record in .WAV format, the custom welcome messages that meet your requirements. See “Recording Voice Messages”.
- Configure **Voice Mail Auto Attendant Menu**. You may configure a different menu’s for different trunk types.
- Select the option **Voice Mail Auto Attendant** to **Route Incoming Calls** during the Day, Break and the Night on the desired trunks. For instructions see “CO Trunks”, “Mobile Trunks” and “SIP Trunks”
- For each trunk port, configure **Voice Mail Auto Attendant**; select the Voice Mail Auto Attendant Menu number.

To use the **Built-In Auto Attendant** on trunks, do the following:

- Make a list of the trunks by their port type (CO, Mobile, SIP) and port number on which you want to use the Built-In Auto Attendant.
- Select the option ‘Built-In Auto Attendant’ to Route Incoming Calls during the Day, Break and the Night on the desired CO, SIP, Mobile trunks. For instructions see “CO Trunks”, “Mobile Trunks” and “SIP Trunks”
- 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 Auto Attendant greetings and voice prompts at each stage of the Built-In Auto Attendant call, you need to assign Voice Modules for the following 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 once after answering the call. A different welcome greeting can be configured for Working Hours, Break Hours and Non-working Hours.
    - **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	Morning Greeting (Built-In Auto Attendant /Trunk Auto Answer)	Good Morning!
03	Afternoon Greeting (Built-In Auto Attendant/Trunk Auto Answer	Good Afternoon!
04	Evening Greeting (Built-In Auto Attendant/Trunk Auto Answer)	Good Evening!
05	Built-In Auto Attendant- Welcome Greeting for Day Time (Working Hours)	Welcome!
06	Built-In Auto Attendant - Welcome Greeting for Night/Break 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, 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.

- Configure the following Built-In Auto Attendant related Timers, as required:
  - **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)

  See “System Timers and Counts” for instructions.

- Configure the following Built-In Auto Attendant related check boxes, as required:

  - **Disconnect Built-In Auto Attendant call, when dialed number is busy:** when this check box is enabled, if the dialed extension is found busy, the system will disconnect the Built-In Auto Attendant call instead of routing it to the Operator. Default: disabled.

  - **Disconnect Built-In Auto Attendant call, when dialed number is not responding:** when this check box is enabled the system, if there is no reply from the landing destination extension, the system will disconnect the Built-In Auto Attendant 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 check box is enabled, if the caller fails to dial a digit within the Built-In Auto Attendant Beeps Timer, the system will disconnect the Built-In Auto Attendant call instead of routing it to the Operator. Default: disabled.

  See “System Parameters” for instructions.

Auto Call Back (ACB)

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 to you set a maximum of 50 Auto Call Backs.

How it works

When you set Auto Call Back,

- 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 configurable.
- 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 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.*
- *SARVAM UCS has the capacity to serve 50 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 cancelled by the system. Also, the system will not serve any more requests if all the 50 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, i.e. 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, i.e. 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.

How to configure

Auto Call Back is a "Class of Service (CoS)" dependent feature. An extension user can set/cancel Auto Call Back only if it is enabled in the extension's Class of Service.

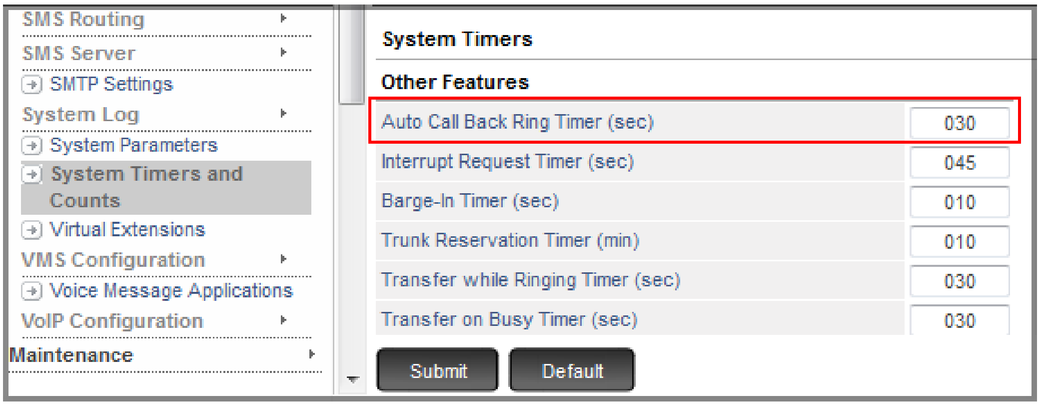
The only configuration involved in this feature is enabling/disabling Auto Call Back in the Class of Service and changing the duration of the Auto Call Back Ring Timer, if required.

By default, all extensions of SARVAM UCS are allowed Auto Call Back feature for the day and night/break in their Class of Service. So, all extensions of 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, you may disable this feature in the class of service of these extensions. For instructions on enabling/disabling this feature on the different extension types, refer the topics "SLT Extensions" and "SIP Extensions"under *Basic Settings.*

If you want to increase or decrease the duration of the of the Auto Call Back ring on both extensions, i.e. the extension requesting Auto Call Back and the destination extension, configure the 'Auto Call Back Ring Timer'.

- Login as System Engineer.
- Under **Advanced Settings**, click **System Timers and Counts**.
- Scroll to reach '**Other Features**'



- Change the value to the desired duration.
- Click **Submit.**

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 Extended IP Phone Users

To set Auto Call Back on Busy:

- Press the 'Call Back' Key on Busy Tone or press the DSS Key assigned to Auto Call Back on Busy Tone.
  OR
- Dial 2 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.

To cancel Auto Call Back on Busy:

- Press the 'Call Back' Key again or press the DSS Key assigned to Auto Call Back again.
  OR
- Dial 102
- 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.

For SLT Users

To set Auto Call Back on Busy:

- On Busy Tone.
- Dial 2.
- You get confirmatory tone
- Replace handset.

To cancel Auto Call Back:

- Pick up the handset.
- Dial 102.
- You get confirmatory tone.
- Replace handset.

Auto Call Back on No Reply

For Extended IP Phone Users

To set Auto Call Back on No Reply:

- Press the 'Call Back' Key / DSS Key assigned to Auto Call Back on Ring Back Tone.
  OR
- 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 will be turned on.
- Go idle.

To cancel Auto Call Back on No Reply:

- Press the 'Call Back' Key / DSS Key assigned to Auto Call Back again.
  OR
- Dial 102.
- 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.

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 50 pending requests and is unable to accept yours.

Auto Redial

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 the 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 'Auto Redial-Ring Back Tone Wait Timer[108] 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 'Auto Redial-Ring Timer[109]. This timer is programmable and is set to 45 seconds (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 configured 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 to match the user preference, e.g.: decreasing the number of attempts to 3 and increasing the interval to 60 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 configured to make 20 redial attempts at intervals of 20 seconds. The number of attempts as well as duration of the interval are configurable, e.g.: number of attempts can be set to 30 and the interval to 15 seconds.

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108. 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.

109. Time for which the extension that has requested Auto Redial should ring. Valid range of the timer: 000 to 255 seconds. Default: 045 seconds.

To change the number of redial attempts and the interval between them, you must configure the Auto Redial Count and the Auto Redial Timer respectively. In addition to these, the system has two other related timers, which can be configured 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 on the same number simultaneously.*
- *The system uses the same trunk access code you used for dialing the number. If you dialed '0', the system grabs one of the free trunks selected for dialing '0'.*
- *If the number was dialed the first time using selective trunk access, the system will use the same trunk for Auto Redial.*
- *If the extension has 'Dynamic Lock', and you have set the 'Auto Redial', the system will check the Toll control as per dynamic lock level.*

Auto Redial may not work well on CO trunks, as its functioning greatly depends on line condition.

How to configure

For Auto Redial to work, you must:

- Enable the features 'Auto Redial' and 'Auto Redial Priority' in the "Class of Service (CoS)" of the extensions to which this feature is to be allowed.

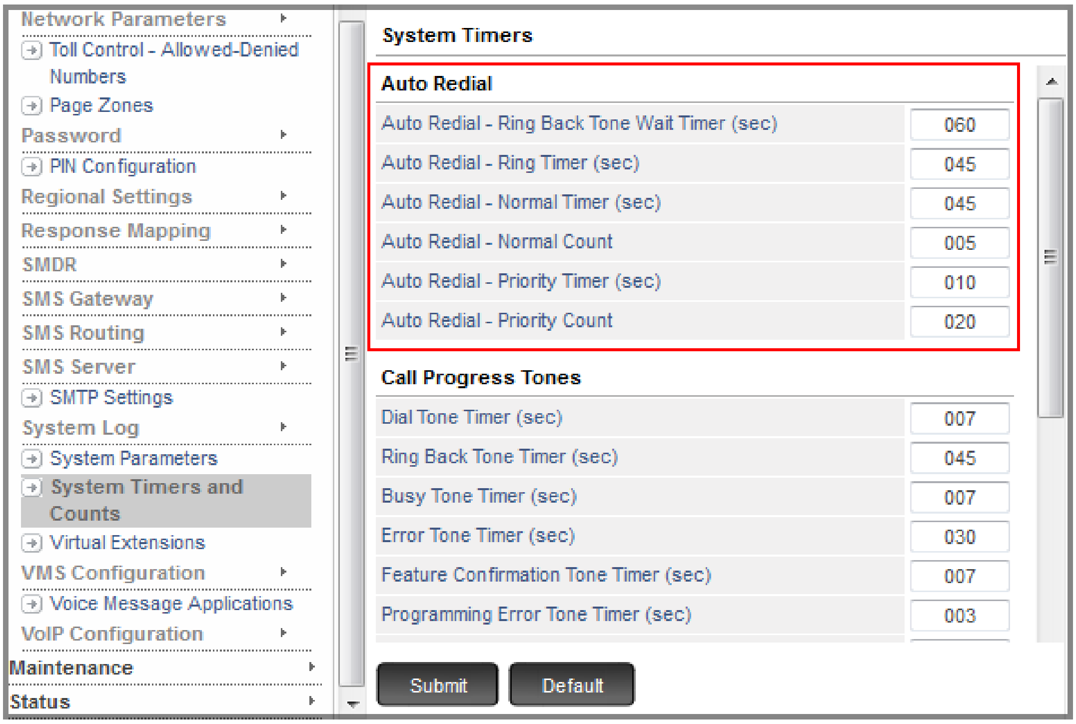
  By default, none of the extensions of SARVAM UCS are allowed Auto Redial and Auto Redial Priority in their Class of Service. You must enable it to allow these features to the extensions.

  For instructions on enabling/disabling this feature on the different extension types, refer the topics "SLT Extensions" and "SIP Extensions"under *Basic Settings*.

- 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.
- If required, also change other related Timers such as Auto Redial Ring Back Tone (RBT) Wait Timer, Auto Redial Ring Timer.

To change Auto Redial Counts and Timers,

- Login as System Engineer.
- Click **System Timers and Counts**.



- Change the Count and Timer of the type of Auto Redial - Normal or Priority - you have set.
- You may change any of the related timers - Auto Redial Ring Back Tone (RBT) Wait Timer, Auto Redial Ring Timer - as per your preferences on this page.
- Click **Submit.**

If SARVAM UCS is installed in Australia, please ensure that:

- *The Timer for Auto Redial Normal as well as Priority must be set to more than 5 seconds.*
- *The Auto Redial Priority Count should be set to less than 15.*

How to use

For 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.

To cancel Auto Redial:

- Pick up the handset.
- Dial 1070.
- Replace handset.

Automatic Number Translation

SARVAM UCS offers connectivity to different networks—CO, GSM/UMTS/LTE, VoIP—each having a different numbering plan. For example, the GSM/UMTS/LTE network requires area codes to be dialed for local numbers, whereas CO requires dialing of area codes for long distance calls.

When SARVAM UCS is connected to more than one network, outgoing calls may be routed through any of these networks, depending on the outgoing call routing 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 feature, Automatic Number Translation of SARVAM UCS takes care of this. It modifies the dialed numbers or part thereof to match with the specific route numbering plan understood by the destination network (CO, GSM/UMTS/LTE, VoIP). This includes adding or stripping of country codes and area codes.

For example, when an extension user dials a local landline number, Automatic Number Translation can be configured such that SARVAM UCS prefixes the number with the appropriate country-area code when it routes the call through the GSM/UMTS network.

How it works

Automatic Number Translation makes use of the Automatic Number Translation (ANT) Table. You can configure upto 32 entires in each ANT table.

The ANT Table consists of three columns, namely:

- **Dialed Number String:** This column contains the numbers you expect the users to dial.
- **Strip Digits:** 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.

You can configure the ANT table for each trunk separately.

Here is an example of how this table is to be configured and used:

Say, 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’.
- first 2 digits to be stripped off from all numbers beginning with 91 and ‘0’ to be added as a 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 Number String	Strip Digits	Add Prefix
XXXXXXXXXX	0	1

Dialed Number String	Strip Digits	Add Prefix
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 Number String column. Since no digit is to be stripped off, '0' is entered in the Strip Digits column. As the prefix '1' is to be added before the Dialed Number String, it 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.

- First 2 digits to be stripped off from all numbers beginning with 91 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 then add 0. For example, when a user dials the number 919925801882, the system will dial out 09925801882.

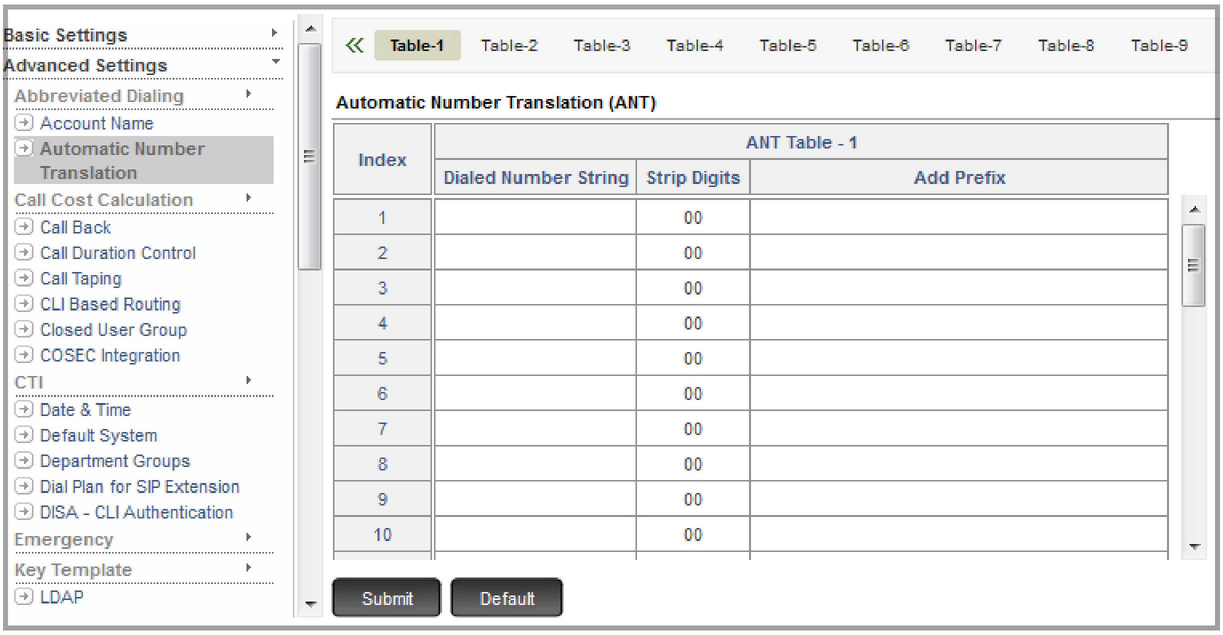
  Automatic Number Translation also forms the basis of "Multi-Stage Dialing".

How to configure

To configure Automatic Number Translation table,

- Login as System Engineer.

- Under **Advanced Settings**, click **Automatic Number Translation**.



The Automatic Number Translation table appears.

The ANT table has three columns: Dialed Number String, Strip Digits, Add Prefix. Each number string is stored against an Index number, from 1 to 32. You can enter as many as 32 entries in the ANT table.

- In **Dialed Number String**, 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 **Strip Digit**, enter the number of digits you want the system to strip off from the Dialed Number String before dialing out this number. The valid range is from 00 to 16. Default: 0.

- In **Add Prefix**, enter the number you want the system to add as a prefix to the Dialed Number String before the system dials out this number. The Prefix can be a maximum of 40 characters. Default: Blank.

* Click **Submit**.

You may also configure the Automatic Number Translation (ANT) Table on the desired trunks. See ***Automatic Number Translation (ANT)*** under "CO Trunks", "Mobile Trunks" and "SIP Trunks" for instructions.

Barge-In

Barge-In allows you to break into an on-going conversation between two extension users, between an extension user and an external caller.

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, i.e. 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 Extended IP Phone, 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. By default, all extension port types have Barge-In and Privacy from Barge-In disabled in their Class of Service.

While it makes sense to offer all extensions Barge-In, providing Privacy from Barge-In also to all extensions will not serve the purpose of Barge-In.

Decide which extensions are to be allowed 'Barge-In', and which are to be allowed 'Privacy from Barge-In' and enable these features in their Class of Service for the Day and Night/Break.

Refer the topic “Class of Service (CoS)” for instructions.

Barge-In Timer

Barge-In Timer is the time after which the caller gets connected to the called party after dialing Barge-In code or pressing the Barge-In key. By default the Timer is set to 10 seconds. To configure the Barge-In Timer, see **Other Features** under “System Timers and Counts”.

How to use

For 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[110].

- 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.

110. *This default feature access code can be changed to suit your preference. Refer the topic* *"Access Codes".*

BCCH Selection

BCCH Selection feature enables you to lock the Mobile Port of SARVAM UCS to a particular cell or channel or BTS (Base Transceiver Station) to ensure better network availability and minimize call drops due to bad signal/ network failure.

*This feature is supported only when **GSM**[111] or **LTE** is selected as the **Network Preferred Mode** for the Mobile Port. See* *"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.

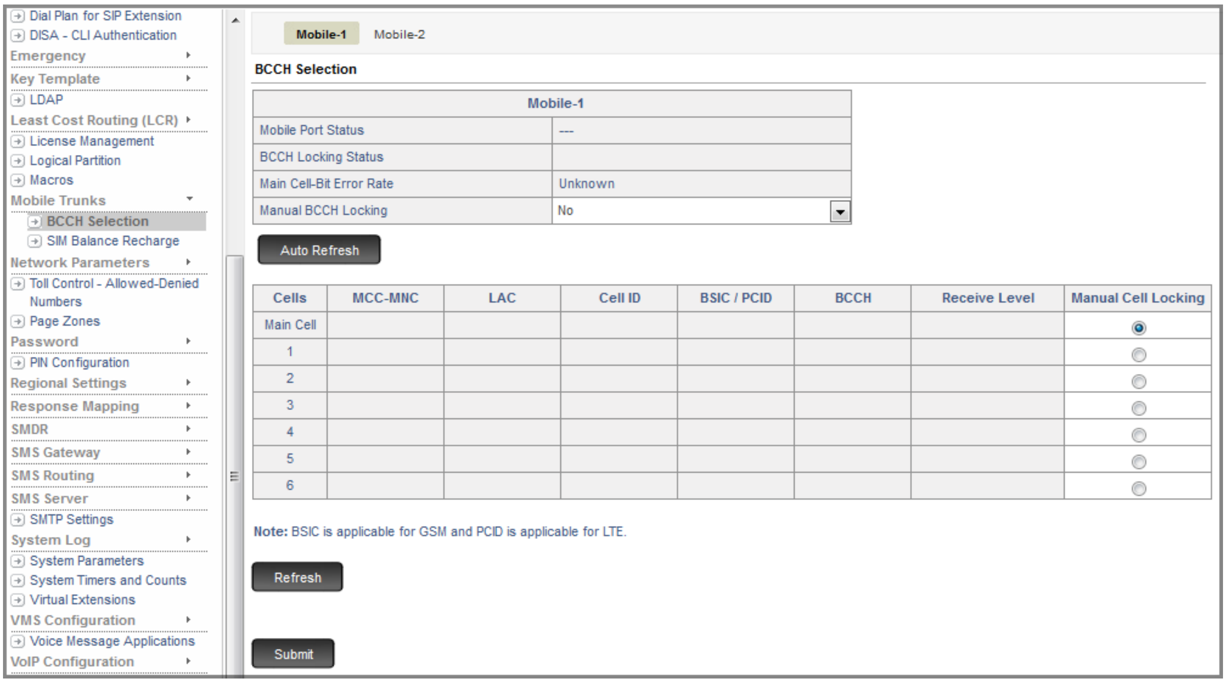
How to configure

To lock the Mobile Port to a cell or a channel,

- Login as System Engineer.
- Under **Advanced Settings**, click **Mobile Trunks**.

111. BCCH Selection will not be supported if QUECTEL UC20G module is installed in your system.

- Click **BCCH Selection.**



- The page displays the following parameters:

  - **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 i.e. 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.

STATUS	DESCRIPTION
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:** By clicking this button, BCCH Selection page is refreshed automatically and all its parameters are 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 the link '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. For example, LAC is displayed in hexadecimal characters for SIMCOM-2G engine whereas, 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. For example, cell ID is displayed in hexadecimal characters for SIMCOM-2G engines which ranges from 0000 to FFFF whereas, 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.

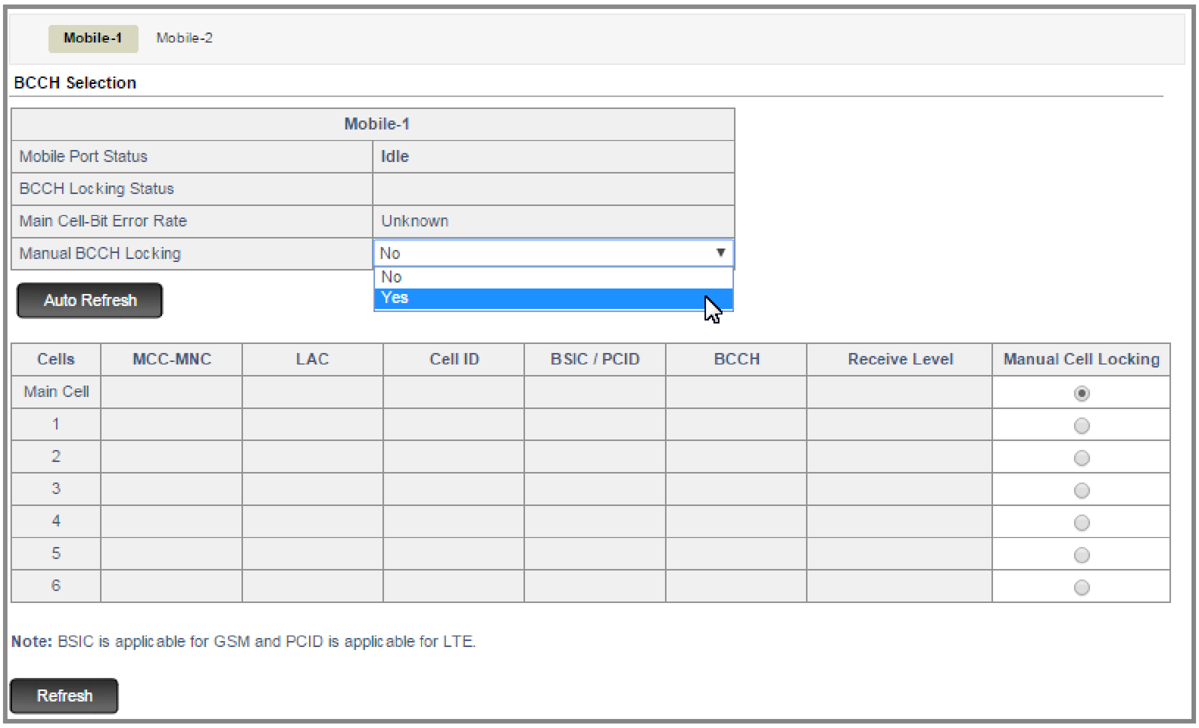
- **BCCH (Broadcast Control Channel):** In this field, the BCCH value of the cell is displayed. BCCH defines the frequency channel number.

- **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,

- Set the parameter **Manual BCCH Locking** to **Yes**.



- Go to the Cell to which you want to lock the Mobile Port.

- Select the radio button **Manual Cell Locking** of that Cell.

- Click **Submit**.

- The BCCH Locking for the 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 Mobile port.

- You may now log out of Jeeves.

Example: Consider the following example when using this feature.

Problem:

- The System is installed in roaming area, where more than one network is available, network A and network B.
- Mobile Network Selection is set to 'Manual' mode, network A is selected as the first priority and network B is selected as the second priority.
- The Mobile Port gets registered with network A. After registration, the user locks the Mobile Port to one of the cells of network A.
- After registration, if the module or the system restarts or gets de-registered from the network, module starts registration process again.
- While re-registering, the system 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.

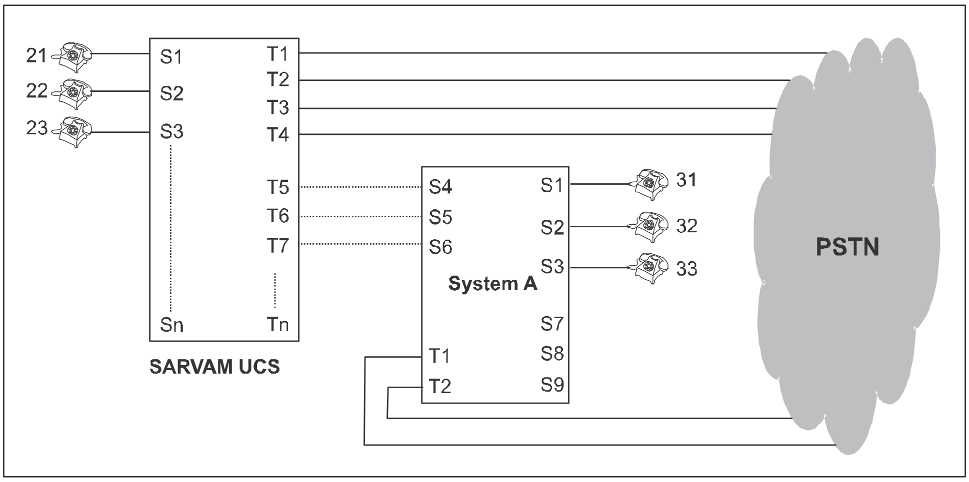
Behind the System Application

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 extension ports 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 Extensions (SLT) S4, S5, S6 of System-A.

However, Trunk lines T1, T2 of System-A are connected directly to the PSTN.

In such application scenarios, implementing toll control restrictions for the trunks becomes difficult 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' (or the Trunk Access Code assigned) and gets PSTN dial tone. When the user dials the number, SARVAM UCS applies Toll Control.

Similarly, when the user of Extension 21 tries to grab a trunk T5, T6 or T7 (which are connected to extensions of System-A) by dialing Trunk Access Code '0', the user gets the dial tone of System-A. This means, the user of Extension 21 must dial '0' again to grab CO dial tone of the T1/ T2 connected to System-A.

However, when the user dials '0' again, SARVAM UCS applies Toll Control. It detects the dialed number as ‘00’ and interprets this as an attempt to dial a long distance number. Since Extension 21 is not allowed long distance dialing in its Toll Control, SARVAM UCS rejects dialing on the trunk and plays an error tone to Extension 21.

To resolve this, SARVAM UCS provides a configurable Pre-PSTN Digit Count (PPDC) for each CO trunk.

The PPDC defines the number of digits to be dialed by the extensions to reach the PSTN. The system applies Toll Control check on extensions only after checking the configured PPDC.

On trunks that are connected to another System, in this case, T5, T6, and T7. PPDC must be configured with the same number of digits as the Trunk Access Codes assigned for System-A. For example, if the Trunk Access Code is a single digit number '0', the PPDC on must be configured as '1'. If the Trunk Access Code is a two-digit number, 61, the PPDC should be configured as '2'.

On trunks directly connected to the PSTN, i.e. T1, T2, T3, T4 of SARVAM UCS, the PPDC must be configured as ‘0’.

To take the above example further, when PPDC is configured as ‘1’ on T5, T6 and T7, when the user of Extension 21 dials ‘0’ followed by another ‘0’ to grab T1 or T2 trunk of System-A, the system will check the PPDC configured on the trunk. On finding ‘1’ the system will ignore the first 0 dialed by the extension, and let the extension user grab T1 or T2 by considering the second ‘0’. The Extension user will get the PSTN dial tone from T1 or T2 of System-A.

How to configure

The PPDC should be configured only for 'Behind the PBX 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.

For CO Trunks that are directly connected to the PSTN, PPDC must be configured as ‘0’.

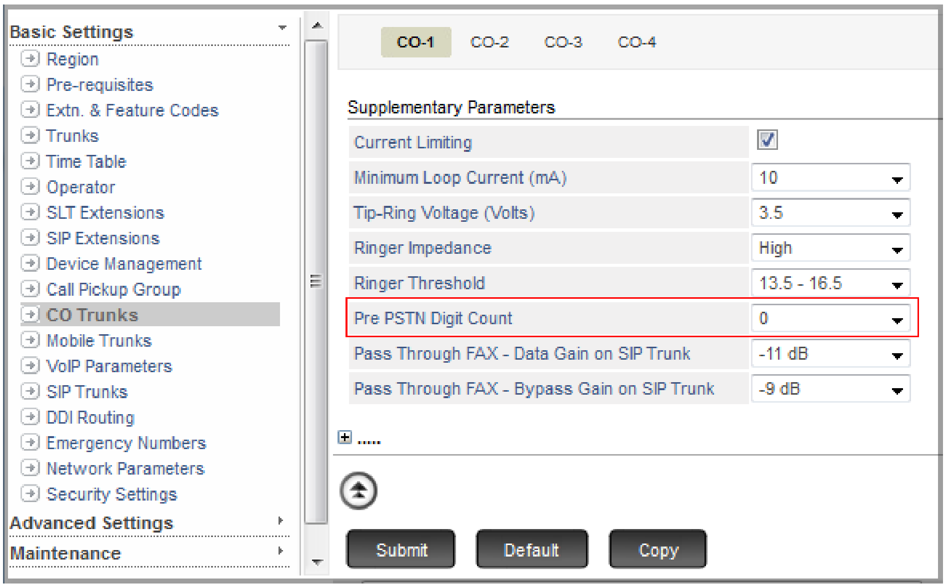
For CO Trunks that are connected to the extensions of another System, PPDC must be configured as per the number of digits in the Trunk Access Codes defined for the second System.

To configure PPDC on a CO Trunk,

- Login as System Engineer.
- Under **Basic Settings**, click **CO Trunks**.
- Click the desired CO trunk number tab.

  The parameters of the selected CO trunk tab appears.
- Click **Hardware Settings** to expand.

- Scroll to **Supplementary Parameters**.



- Select 0 in **Pre PSTN Digit Count**.

- Click **Submit.**

- Similarly, click the desired CO Trunk tab to which you want to assign PPDC count i.e. trunks connected to the extensions of another System, and assign the appropriate PPDC count to the trunk. This would depend on the number of digits in the Trunk Access Code defined for the trunks in other System. If the TAC is single digit, select '1'. If TAC is double or triple digit, select '2' or '3' as applicable.

- Click **Submit.**

Busy Lamp Field for Trunks

On SIP extensions, SARVAM UCS supports the Busy Lamp Field (BLF) Subscription feature 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, Mobile and SIP.

To provide this feature to SIP extension users, 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 call 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

- A 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-1, BLF subscription is enabled on extension 3301 and CO-1 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-1 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-1 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, 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-1 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 be indicated after the previous call is terminated.

How to configure

To provide BLF to SIP extension users, you must do the following:

- Enable **Allow Busy Lamp Field Subscription** under “VoIP” for the desired SIP Extensions. See “SIP Extensions” for more details.

- 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[a]
CO	CO-xxx	Here, x should be a valid CO port number, that is, from 1 to 4.
Mobile	Mobile-xxx	Here, x should be a valid Mobile port number, that is, from 1 to 2.
SIP	SIP-xx	Here, x should be a valid SIP Trunk number, that is, from 1 to 8.

a.The number of ports mentioned here are for ETERNITY NENXIP50. For details regarding the number of ports supported for ETERNITY NENX312/ETERNITY NENX416, see “Technical Specifications of ETERNITY NENX”.

Call Back on Trunk Ports

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: CO, Mobile and SIP trunks.

When the Call Back feature is enabled on a trunk port, and there is a missed call on that trunk port, 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 configured for that number, either from the port on which it was received or from a different port, depending on the configuration. SARVAM UCS can be configured 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 to call them back, thereby saving on charges (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 ‘Call Back Timer’ may be configured. 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 from which the call was received or an alternative number.
- The number on which call back is to be made must be configured in a List. Also, if you want the call back to be made to an alternative number, the alternative number must also be configured in a list.
- You must select the trunks through which the call back should be made. If necessary, you may enable Least Cost Routing on the outgoing trunks you selected for call back.
- Select a ‘Call Back Mode’, i.e. how the call should be routed when the call back is answered by the remote party; whether it should be routed through Operator, Built-In Auto Attendant, or DISA.

Following is an example of a Call Back on a Mobile Port, when the above parameters are configured.

- Caller A calls Mobile Port 1.
- The system checks if the Call Back is enabled on Mobile Port 1.
- If enabled, the system matches the CLI of A with the Call Back profile assigned to the Mobile Port 1 to determine if the calling number is eligible for a call back.
- A match is found in the list configured in the Call Back profile.[112]
- The system waits for the period of the Call Back Timer (configurable, 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.

112. *If the system does not find a match for the CLI of the caller in the list, the 'Call Back' feature will not be applicable and the call will be processed according to the normal incoming call logic.*

- If A disconnects within the Call Back Timer, the system applies Call Back for A’s number.
- The system checks the ‘Call Back to’ parameter in the Call Back profile, to determine whether it has to call back the same number or an alternative number.
- If alternative number is configured as ‘Call Back to’, the system checks the list of alternative numbers. As the CLI of A matches with the number in the list, the system checks the corresponding alternative number configured for A’s CLI number.
- The system checks if the number is to be called from the same trunk port or a different group of trunks.
- If the same trunk port is configured, the system will make a call to the number using Mobile Port 1.
- If a different group of trunks has been assigned, the system will check if Least Cost Routing is enabled on these trunks and accordingly makes the call back.
- When A answers the call, the system checks the type of Call Back Mode enabled in the Call Back Profile assigned to Mobile Port 1 (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 1.
     - 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 1.
     - A gets dial tone of SARVAM UCS.
     - A can now reach any extension or trunk of SARVAM UCS from DISA Mode.
  3. 'CLI Authentication - Single Call' is enabled as Call Back Mode on Mobile Port 1.
     - A gets dial tone of another trunk of SARVAM UCS.
     - A can make calls from the trunk.
  4. ‘Operator’ is enabled as Call Back Mode on Mobile Port 1.
     - A gets Ring Back Tone.
     - The system lands the call on the Operator extension assigned to Mobile Port 1.

  Read the topics “Auto Attendant” and “Direct Inward System Access (DISA)” to know more about the call respective call logic.

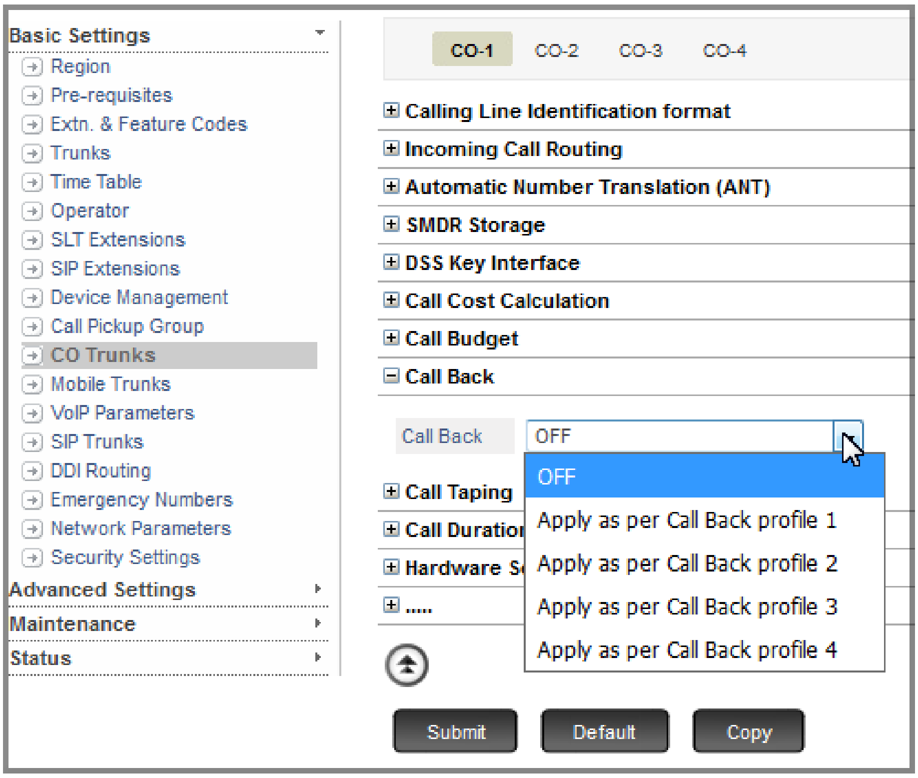
- *Since this feature is essential for callers, they must be aware of its functioning, to be able to use it, that is, disconnect the call within the Call Back Timer. If the caller does not disconnect within the Call Back Timer, the call will be processed according to the normal incoming call logic.*

- *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 a group of trunks, when any of these trunks is busy, SARVAM UCS will support only the last call back request in the group. Previous requests will be processed as per the normal incoming call management logic.*
- *For call back requests made from a group of trunks, when any of these trunks is in the restricted category call back request will not be served by SARVAM UCS. See* *"Logical Partition" topic for more details.*

How to configure

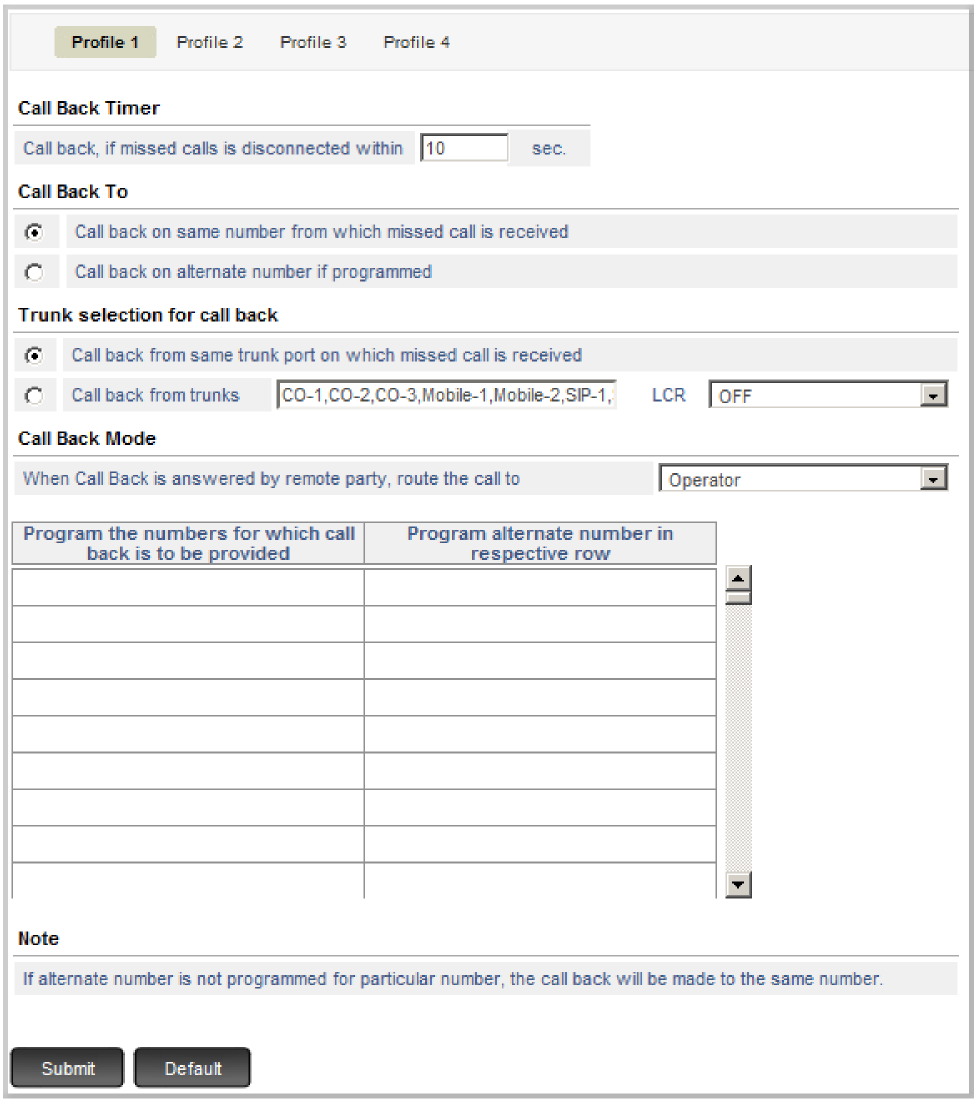
For this feature to function, you must configure the Call Back Profiles and assign them to each Trunk port type (CO, Mobile, SIP) on which you want to use this feature.

- Login as System Engineer.
- Click **Basic Settings.**
- Select the desired Trunk port type. For example CO trunks.
- The CO trunk page opens. Select the number of the CO trunk you want to configure by clicking the trunk number tab.
- Click **Call Back.**
- Select the Call Back Profile you want to apply to this trunk.



- Now, configure the Call Back Profile you selected by clicking the Settings icon.

- The **Call Back Profile** opens in a new window.
- Configure the profile parameters in this window.



- Set the **Call Back Timer** to the desired duration. 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 list you configured.

  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.

- Select the number which the system must **Call Back To**. You may select:
  - **Call Back on same number from which missed call is received.** If you select this option, you must configure the numbers eligible for call back in the **Program the numbers for which call back is to be provided** list.

- **Call back on alternative number if programmed.** If you select this option, for each number you configured as eligible for call back, you must configure an alternative number in the **Program alternate number in respective row** list.

  When a missed call is eligible for call back (matches with list you configured), the 'Call Back to' parameter determines the number on which the call back is to be made, i.e. 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 to' option.

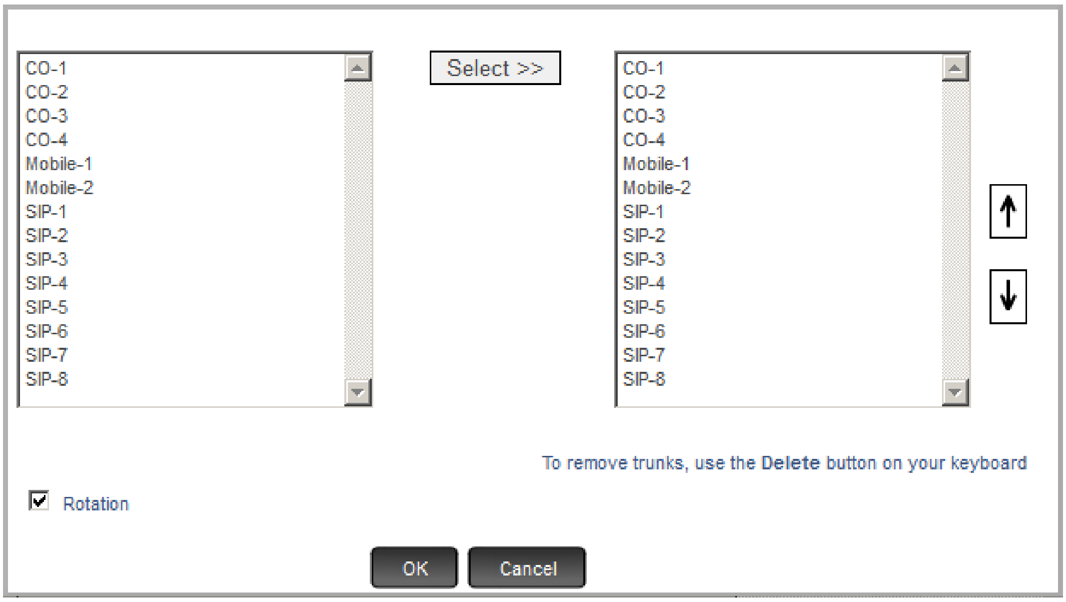
  In countries where CLI received on trunks cannot be dialed without modification, you may select "Alternate Number' as the 'Call Back to' option. You may also select 'Alternate Number' as Call Back on when you want the call back to be made to a different number.

- Configure **Trunk Selection on Call Back.** This parameter determines the trunk port to be used to make call back.The call back can be made using the same port or a group of trunks.

  - Select **Call Back from same port from which missed call is received**, to use the same port to make the call back.

  - To make the call back using a group of trunks, select **Call Back from trunks**, and double click this field to select the trunks.

    A multiple selection box opens.



On the left, the trunks appear with their names (if configured in "Trunks") and port numbers in a sequence, starting with CO trunks, followed by Mobile trunks and SIP trunks.

- To select a trunk, place your cursor on the desired trunk, and click **Select>>**.
  Or
- Press the **ctrl** key and click the left mouse button to select multiple trunks.

When selecting a trunk port type, make sure the trunk you select is not under restricted category in Logical Partition.

- You may change the sequence of the trunks you selected, if required, using the **Up** and **Down** arrow buttons on the right display box.
- You can also delete trunks from the ones you have selected.
- You may enable **Rotation**, if you have selected more than one trunk. Default: Disabled.

  Rotation has no relevance if only one member trunk is selected.
- Click **OK**. The multiple selection box closes.
- To apply Least Cost Routing on the trunks, select desired **LCR** method from the combo box: Number-based, Time-based, Time+Number based, Service Provider based. Configure LCR method that you selected for the trunk group, if not already done.

- Select the **Call Back Mode**, from the options; how a 'Call Back' call answered by the remote party should be routed:
  - **Operator:** When the remote party answers the Call Back call, the system will route the call to the Operator[113].
  - **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-Answer Signaling:** 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)".

  By default, Operator is selected as the Call Back Mode.
- Configure the call back number lists.
  - In the column **Program the numbers for which call back is to be provided**, enter the 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.

    Number strings configured in this list are compared with the actual received CLI.

    The number string configured in this list may be shorter than the number string received as CLI, but only if the configured number string completely matches with the received CLI from the right towards left, the system will consider it as a complete match.

113. *'Operator' is the extension which is assigned to the Mobile port.*

- In the column **Program alternate number in respective rows**, enter the numbers that the system must call back. 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 to' option you selected and the alternate number list.

  For each number string you entered in the previous list, enter a corresponding alternative number, *if*:

  - you have selected 'Call back on same CLI number on which missed call is received' as the 'Call Back to' option, BUT the CLI received cannot be dialed out without modification. In such a case, modify the CLI received and enter the string in the alternate number List.[114]

    OR

  - you have selected 'Call Back on alternate number' as 'Call Back to' option. In this case, configure the alternative number for the CLI received in the alternate number list.

    The modified CLI or the Alternate number should be entered in the same row as the corresponding received CLI number. For example, for the received CLI number string entered in row 2, enter the corresponding modified CLI/Alternate number string in row 2.

    When the CLI received matches with the number string entered in row 2 of the alternate number list, the call back will be made using the (modified/Alternate) number stored at row 2 of this List.

If you have selected 'Alternate Number' as 'Call Back to' option, but do not want to provide alternative numbers to call back particular callers (i.e. CLI received), configure the CLI of these callers in the first list but keep the corresponding rows in the alternative number list blank.

- Click **Submit** to save Call Back on Trunk.

- To configure Call Back on other trunk port types, go to "Mobile Trunks" and "SIP Trunks" under Basic Settings, and follow the same instructions as above.

114. *Where the CLI received can be dialed out without any modification, you do not need to configure any number string in the Alternate number List.*

Call Budget on Extension

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 calls 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.
  - Until a new Call Budget is allocated to the extension, the system allows extension user to make calls as configured in **Calls allowed when Call Budget is consumed** under Call Budget for that 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 *Calls allowed When Call Budget is 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 a fresh budget to the extensions from the System Administrator (SA) mode, at any time. The Call Budget allotted by the System Administrator will be reloaded in the following month.

- *Call Budget is not based on real time (Online) call cost calculation. 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 be disconnected, though the budget is exceeded. To prevent this from occurring, you may configure the "Call Duration Control (CDC)" feature.*
- *Call Budget is dependent on precise Call Cost Calculation. So, SMDR parameters and long distance codes must be configured correctly to prevent errors in calculation.*
- *This feature works independent of any Call Accounting Software (CAS) installed with SARVAM UCS.*
- *SARVAM UCS will calculate cost of phone calls made by extension phones even when no call budget is allocated[115].*

How to configure

To be able to use this feature, you must do the following configuration:

- **Apply Call Budget** on the Extensions on which you want to use this feature.
- Select the types of **Calls allowed when Call Budget is consumed** under each extension as required.

For instructions, see "SLT Extensions" and "SIP Extensions" under *Basic Settings*.

How to use

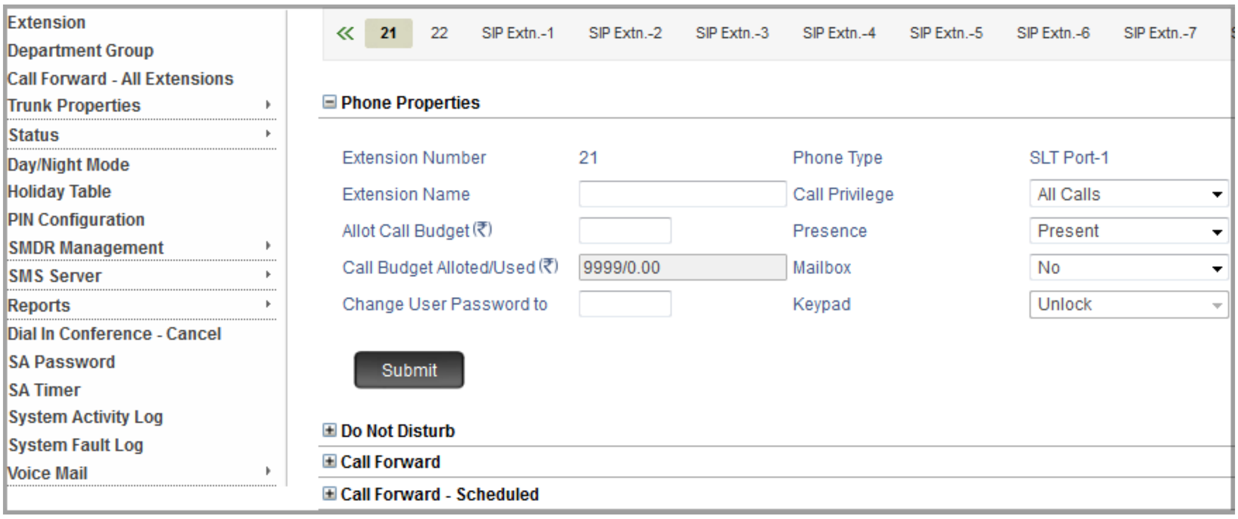
Call Budget amount can be allotted to extensions from the System Administrator mode, using Jeeves or by dialing SA commands from an extension phone.

Assigning Call Budget to Extensions using Jeeves

- Login as System Administrator.
- Click **Extension**.
- The Extension numbers appear on the tabs, starting with SLT extensions. To select a particular extension number or type, click the tab.

115. *Based on the feature 'Call Cost Calculation'.*

- 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**.
- To allot call budget to another extension, click the extension number tab, and follow the same instructions as above.

For 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

Call Budget on Trunks is an expense control feature of SARVAM UCS that enables you to keep track of the cost of phone calls made from the different Trunks of SARVAM UCS.

With this feature, each trunk can be allotted a 'budget' limit for outgoing calls. This budget limit can be configured to be reloaded manually each time it is exceeded or on a scheduled date, which may be either daily or a particular date of the month.

There are three types of Call Budget limits 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 free every month. The customer can avail of this offer using 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, no calls will be routed through this trunk.

The consumed Budget can be reset, after which it 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.

How it works

Call Budget can be enabled on all trunk port types - CO, Mobile, SIP. 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 also 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 configured. 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 configured 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 configured, 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 increments, 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, if outgoing calls are restricted from such trunks.
- 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. 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 be disconnected, even 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 configured properly to prevent errors in calculation.*

- *This feature works independent of any Call Accounting Software (CAS) installed with SARVAM UCS.*

- *SARVAM UCS will calculate cost of phone calls made by the trunks even when no call budget is allocated*[116].

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116. Based on the feature 'Call Cost Calculation'.

How to configure

To be able to use this feature, you must, configure the Call Budget parameters on the trunk on which you want to use this feature.

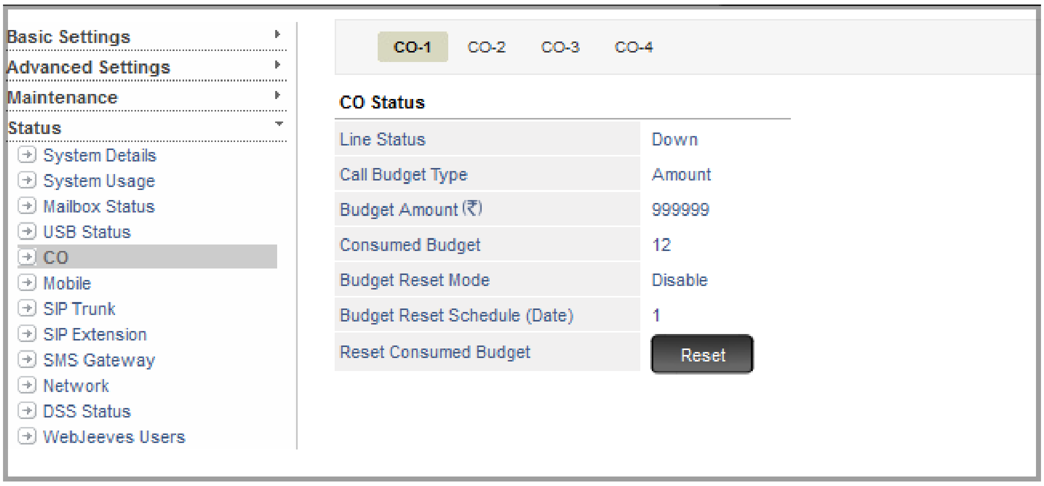
For instructions, see "CO Trunks", "Mobile Trunks" and "SIP Trunks" under *Basic Settings*.

Resetting Call Budget on Trunks

The consumed Call Budget can be reset manually anytime as required, or on a scheduled date either daily or on a particular date of the month.

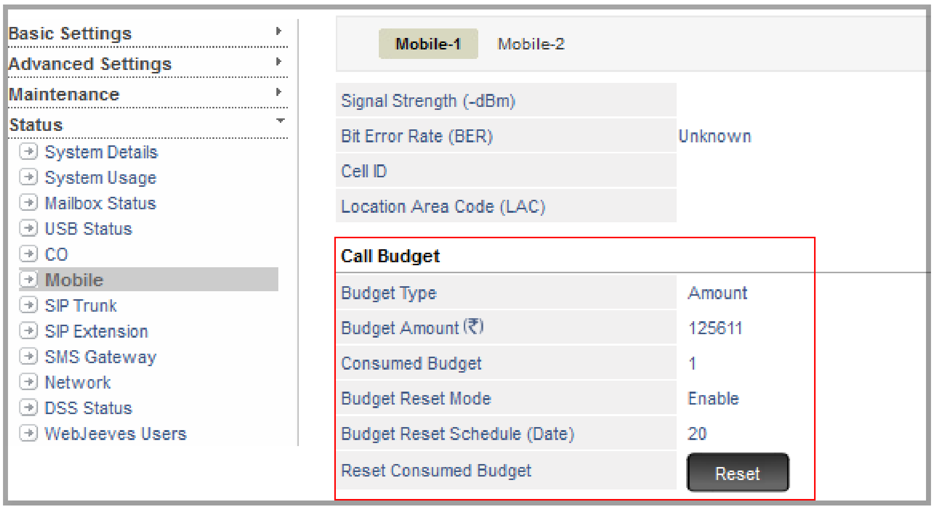
To reset the Call Budget on trunks,

- Login as System Engineer.
- Under **Status**, click **CO**.



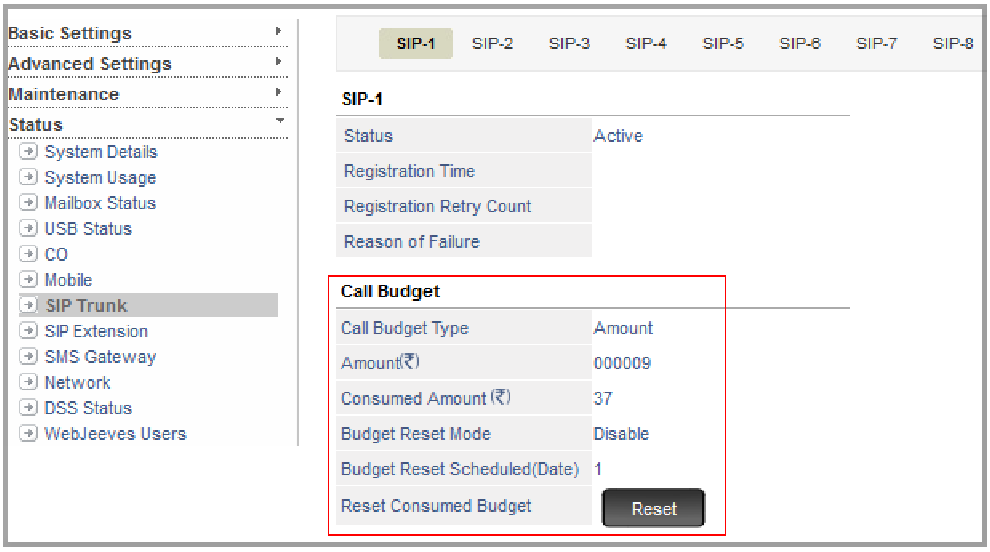
- If you have set Call Budget to reset on a scheduled date, the date will appear in the **Budget Reset Schedule (Date)**.
- To reset Call Budget manually, click the **Reset** button assigned to **Reset Consumed Budget**.

- To reset call Budget on Mobile Trunks, click **Mobile** under **Status**.



If you have set Call Budget to be reset on a scheduled date, the date will appear in the **Budget Reset Schedule (Date)**.

- To reset Call Budget manually, click the **Reset** button assigned to **Reset Consumed Budget**.
- To reset Call Budget on SIP Trunks, click **SIP Trunk** under **Status**.



If you have set Call Budget to be reset on a scheduled date, the date will appear in the **Budget Reset Scheduled (Date)**.

- To reset Call Budget manually, click the **Reset** button assigned to **Reset Consumed Budget**.

Similarly, you can reset Call Budget on Trunks from the SA mode.

Call Chaining

In Call Chaining, an external/internal call transferred by the Operator to another extension or external number is returned back 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 in 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 configuration involved in the functioning of this feature is assigning this feature to a DSS key which has LED and changing, if necessary, the Call Park Release Timer.

For instructions on assigning a feature to DSS keys, see Phone Key Settings under “Configuring Matrix SPARSH VP248”, “Configuring Matrix SPARSH VP310”, “Configuring Matrix SPARSH VP510”, “Configuring Matrix SPARSH VP210” and “Configuring Matrix SPARSH VP330”.

To change the Call Park Release Timer, see “System Timers and Counts”.

How to use

For 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)

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

The cost of a call depends on:

- Trunk Port (CO, Mobile, SIP) used for making the outgoing call.
- Number dialed, whether local, long distance or international.
- Time and day when the call was made, whether daytime, break time, night time, or on a Special Day.
- Duration of the call.

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 calculates 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 as 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 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 by following example:

- A Outgoing Call is made on 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

    - Area Code Table is configured as:

Index	Area Code	Name	Ignore Digit Count	Pulse Rate Type for Pulse Rate Option - 1	Pulse Rate Type for Pulse Rate Option - 2	Pulse Rate Type for Pulse Rate Option - 3	Pulse Rate Type for Pulse Rate 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 1.00 (As ` or $ as per applicable currency) for the duration of 30 seconds and for the additional unit also, the cost will be considered as 1.00 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,

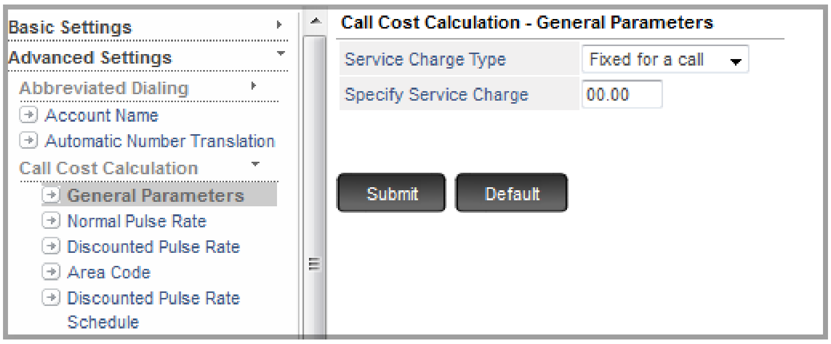
- Login as System Engineer.
- Under **Advanced Settings**, click **Call Cost Calculation**.

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**.



- In the **Service Charge Type**, 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 **Specify Service Charge**.
  - **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 **Specify Service Charge**.
  - **% 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 **Specify Service Charge in %** which appears.
- Click **Submit**.

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 *Basic Settings*, and configure these on the different trunk port types. See “CO Trunks”, “Mobile Trunks”, and “SIP Trunks” under *Basic Settings* for instructions.

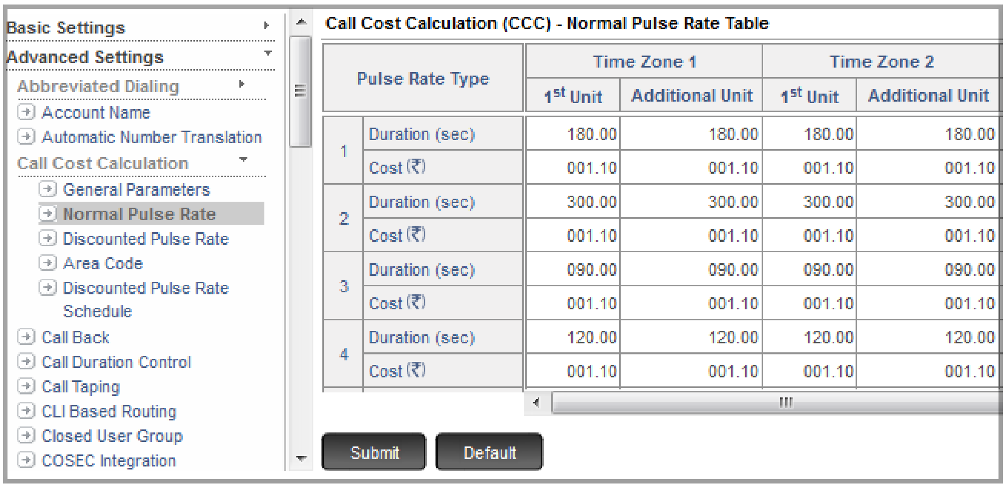
Configuring Pulse Rate Types

Generally service providers offer discounted call rates for holidays. 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 for holidays, you may skip configuring the Discounted Pulse Rate table.

To configure the Normal Pulse Rate table,

- Click **Normal Pulse Rate**.



- 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.

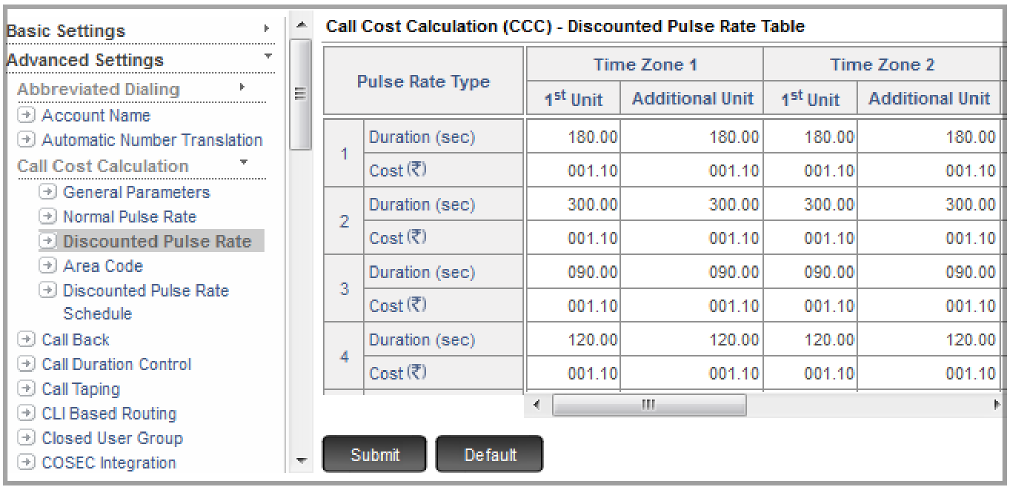
  The Pulse Rates offered by service providers may vary according to the time of the day. In such cases, you will need to configure the **Call Cost Calculation Schedule** for the trunk, by dividing the day into Time Zones, from 1 to 4, as required, to match the time of the pulse rates offered by your service 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.

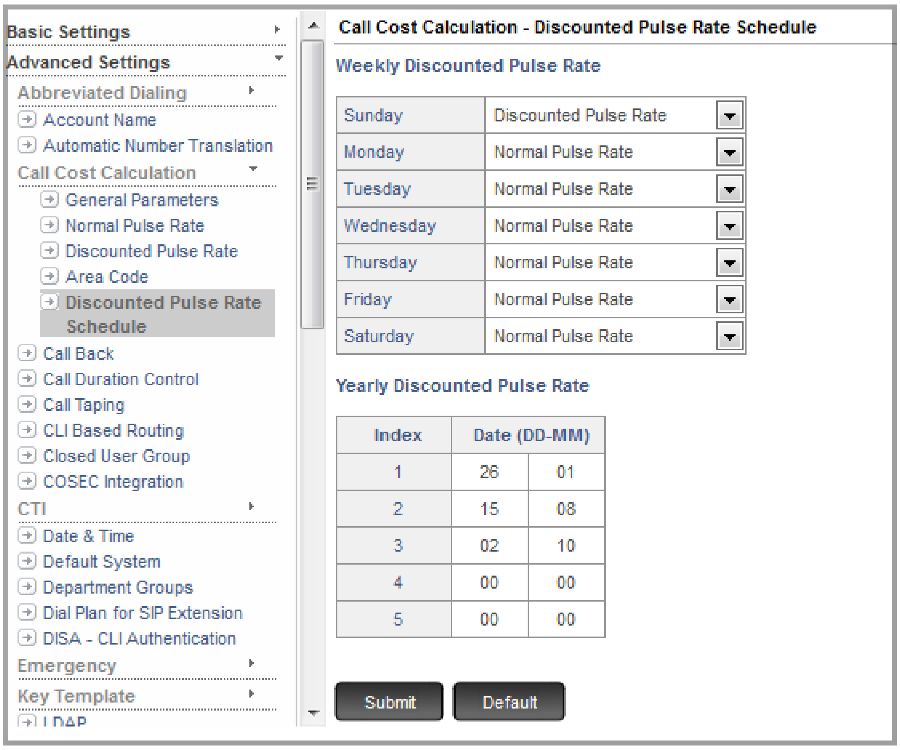
- Click **Submit**.

To configure the Discounted Pulse Rate table,

- Click **Discounted Pulse Rate**.



- Configure the **Discounted Pulse Rate Type** with rates for the **First Unit** and the **Additional Unit**, as required.
- Click **Submit.**
- Click **Discounted Pulse Rate Schedule.**



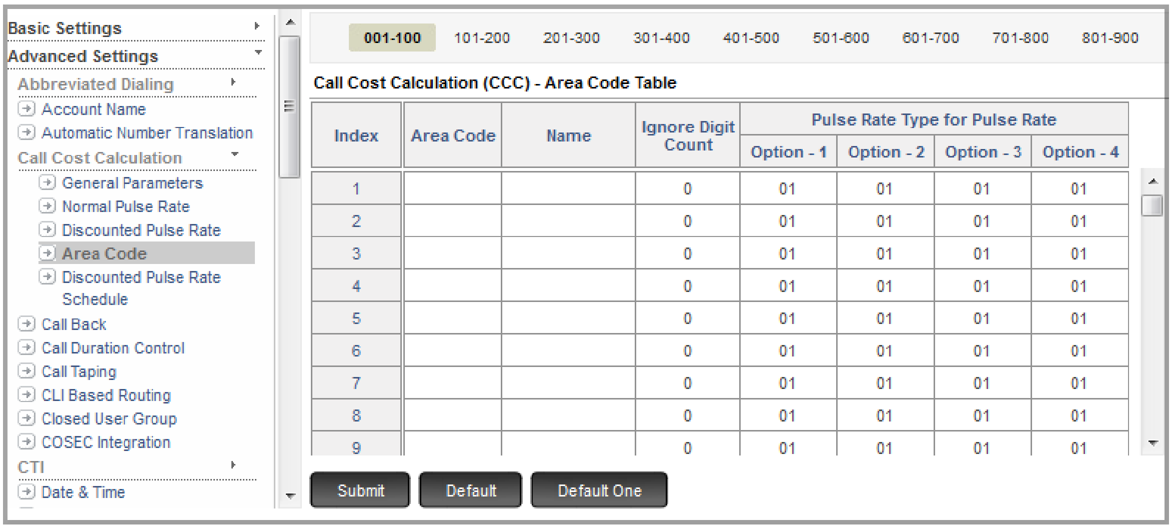
- Define **Weekly** and **Yearly** Discounted Pulse Rates on this page.
- Click **Submit.**

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 **Area Code**.



- 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**.

Call Cost Display

If your extension is a Extended IP Phone, you can use the Call Cost Display feature to view the cost of the last 10 external calls made from your extension.

The system will display the dialed numbers and the call cost for each number that it has calculated on the LCD of your 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-Extension-Flag***
  Where,
  Extension is the Number of the Extended IP Phone
  Flag is
  0 for Disable
  1 for Enable
  By default Call Cost Display is enabled.

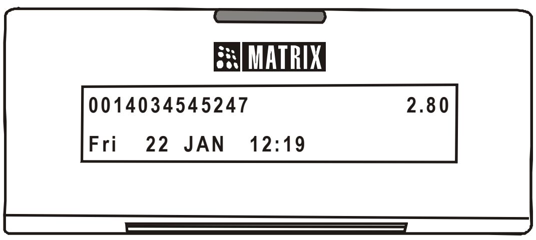
- Exit SA mode.

How to use

For Extended IP Phone Users Only

- Press the 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, the display will show the number as:



- The display is retained till the extension remains OFF-Hook.
- Go Idle.

Call Duration Control (CDC)

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.

How it works

- A is an extension user. B is an external party.

External-Outgoing Calls

- A dials B's number.
- SARVAM UCS will check whether the CDC is applicable on Extension 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 Profile** applied to A. In the Profile,
    - 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 **Apply CDC for calls matching with numbers** list and **Do Not Apply CDC for calls matching with numbers** list in the CDC profile.

To check whether CDC is applicable on the Trunk

- The system checks **Apply CDC for outgoing calls made from Trunk** check box.

The following results are possible:

- 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 **Apply CDC for Outgoing calls** check box is enabled on the Trunk. CDC is applied on the call.
- 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 **Apply CDC for Outgoing Calls** check box is disabled on the Trunk. CDC is not applied on the call.
- 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 in the CDC profile. The **Apply CDC for Outgoing Calls** check box is enabled on the Trunk. CDC is not applied on the call.
- 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 in the CDC profile. The **Apply CDC for Outgoing calls** check box is disabled on the Trunk. CDC is not applied on the call.

- The **Apply CDC for outgoing calls made from Trunk** check box is enabled and a match is found in both these lists. The system gives precedence to the Do Not Apply CDC for calls matching with numbers list. The **Apply CDC for Outgoing Calls** check box is enabled on the Trunk. CDC is not applied on the call.

- The **Apply CDC for outgoing calls made from Trunk** check box is enabled and a match is found in both these lists. The system gives precedence to the Do Not Apply CDC for calls matching with numbers list. The **Apply CDC for Outgoing Calls** check box is disabled on the Trunk. CDC is not applied on the call.

- When CDC is applied on 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 configured for the desired time limit.
- At the end of the default/programmed time limit of the CDC Timer, the Beep Timer starts (5 seconds; non-configurable) 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 Call after CDC Timer** check box is enabled.
- If this check box is disabled, the call will not be disconnected.
- Instead, the **CDC Warn Timer** will be loaded again. 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 trunks** check box is enabled in the Profile assigned to A.
  - the **Disconnect Call after CDC Timer** check box is enabled in the Profile assigned to A.
  - a match is found for the B’s number in the **Apply CDC for calls matching with numbers** List.
  - the **Apply CDC for Incoming Calls** check box is enabled on the Trunk on which the call is received.

  CDC is applied on the call.
- A will get beeps and then the call will be disconnected.

Internal Calls

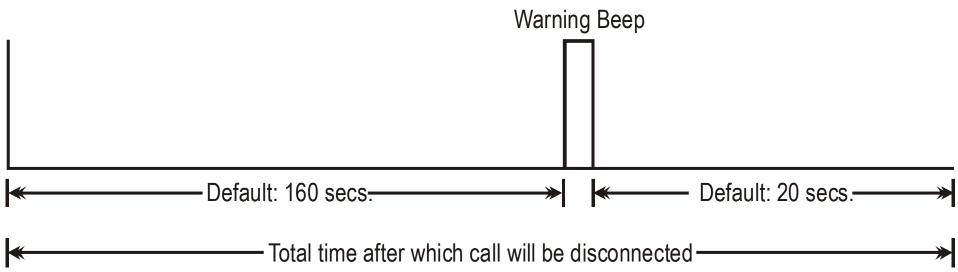
A and B are extension users.

- A calls B.

- The system checks whether the check box, **Apply CDC to Internal Calls**, is enabled in the CDC profile assigned to either A or B.

- 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 and Raid:** CDC is treated as turned 'ON'.

- **Emergency Number Dialing:** Emergency calls are not affected by this feature, i.e. CDC will not be applied on the dialing of Emergency Numbers.

How to configure

To use this feature on extensions,

- Configure Call Duration Control Profiles. You may configure up to 4 different Profiles.

- Assign a CDC Profile to the extensions on which you want to apply this feature.

- Enable CDC for incoming calls received on and outgoing calls made from the Trunks on which you want to apply this feature.

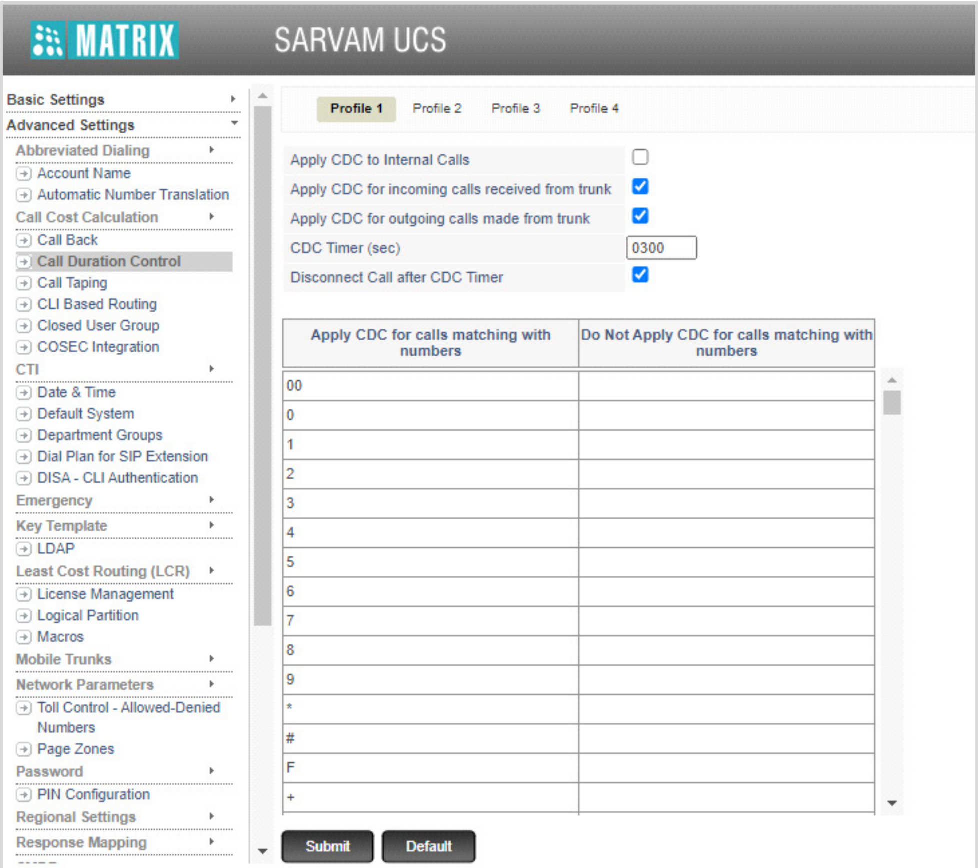
To configure Call Duration Control Profiles,

- First, 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, and a list of numbers on which CDC is not to be applied.

- Make a list of extensions on which CDC is to be applied and the CDC Profile number you want to assign to these extensions.

- Login as System Engineer.
- Under **Advanced Settings**, click **Call Duration Control**.



- The CDC **Profile 1** opens on this page.
- By default, **Apply CDC to Internal Calls** check box is selected, that is, CDC is applied on internal calls. Clear the check box to disable. Default: Enabled.
- By default, **Apply CDC for Incoming Calls received from trunk** check box is selected, that is, CDC is applied on incoming external calls. Default: Enabled.
- By default, **Apply CDC for Outgoing Calls made from trunk**, check box is selected, that is, CDC is applied to outgoing external calls. Default: Enabled.
- If required, change the **CDC Timer** to the desired duration. The range of the timer is 0001 to 9999 seconds. Default: 160 seconds.
- By default, **Disconnect Call after CDC Timer** check box is selected, that is calls will be disconnected on the expiry of the CDC Timer. Clear the check box to disable.Default: Enabled.
- In the **Apply CDC for calls matching with numbers** column, enter the numbers on which CDC should be applied. You may refer to the list you prepared.
- In the **Do Not Apply CDC for calls matching with numbers** column, enter the numbers on which CDC should not be applied. You must configure this list with numbers which you want to be exempt from CDC.
- Click **Submit** at the bottom of the page to save changes.

- To configure another CDC Profile, click the Profile number tab and follow the same steps described above.

Assign the CDC Profile to the desired extensions. For instructions, see “SLT Extensions” and “SIP Extensions”, under **Basic Settings**.

Enable CDC on the desired trunks. For instructions, see **Call Duration Control** under “CO Trunks”, “Mobile Trunks” and “SIP Trunks” under **Basic Settings**.

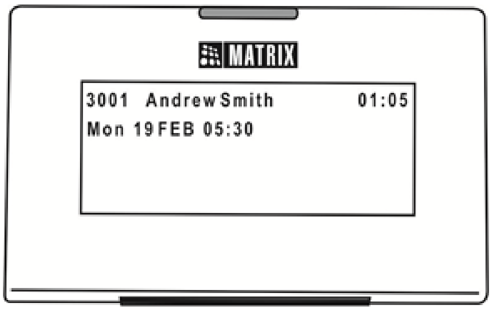
Call Duration Display

By invoking Call Duration Display, extension users can view the duration of the current call instantly.

The current call may be incoming or outgoing, internal or external call.

How it works

- The Extended IP Phone 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

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 usual workplace. The 'Call Forward' feature of SARVAM UCS ensures this.

Using this feature, calls landing on an extension can be forwarded to another extension, external number, Voice Mail Group, or 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 desks.

You can also set Call Forward for all the extensions from the SA Mode. See “Settings Call Forward for an Extensions using SA Mode”.

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 configuring 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** - when calls are forwarded to another phone number. Both phones, i.e. 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 find themselves shuttling between two places frequently. As it is cumbersome to forward the calls from one extension to another and cancel it repeatedly, 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. To know more, see “Preset Call Forward”.

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 then 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 tone 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 tone from D, it waits for the Call Forward No-Reply timer to expire.
- The system forwards the call to the Voice Mail.

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*** *check box 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 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.

- *You can select the types of calls, i.e. internal calls only, or trunk calls, or both, to be forwarded to external numbers. You can configure 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* ***If call is forwarded to an external number*** *must be configured on the extensions. See “Call Forward” in “SIP Extensions”.*
- *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 the extension.*

How to configure

To provide this feature to an extension, you need to do the following configuration:

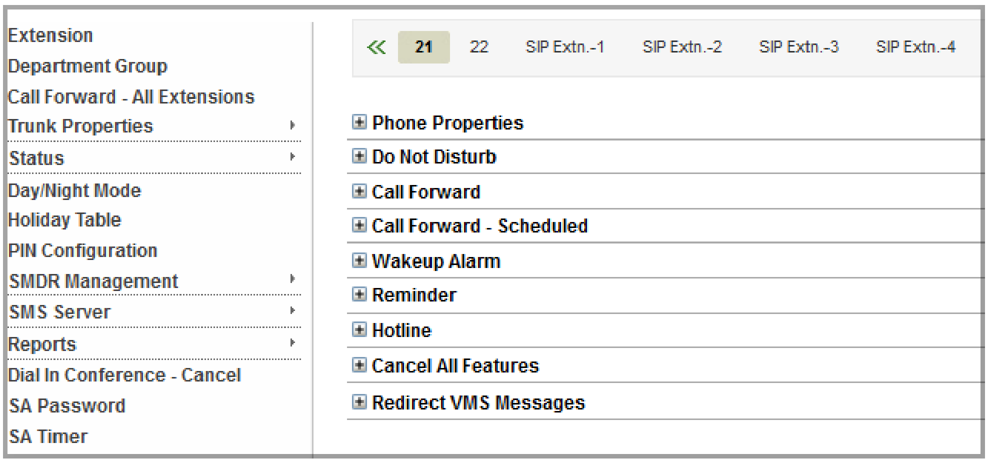
- Enable Call Forward in the "Class of Service (CoS)" of the Extensions. By default, Call Forward is enabled in the Class of Service of all extensions of SARVAM UCS for the Day and Night/Break.
- If required, change the duration of the Call Forward No-Reply Timer on the extension. By default, the timer is set to 30 seconds.
- Select the type of call—internal, trunk, all calls— to be forwarded when call is forwarded to an external number on the extension. By default, Forward only Trunk Calls is selected.

  For instructions on configuring Call Forward related parameters for different extension types, see "SLT Extensions" and "SIP Extensions" under *Basic Settings*.

You can set Call Forward for each Extension from the SA Mode.

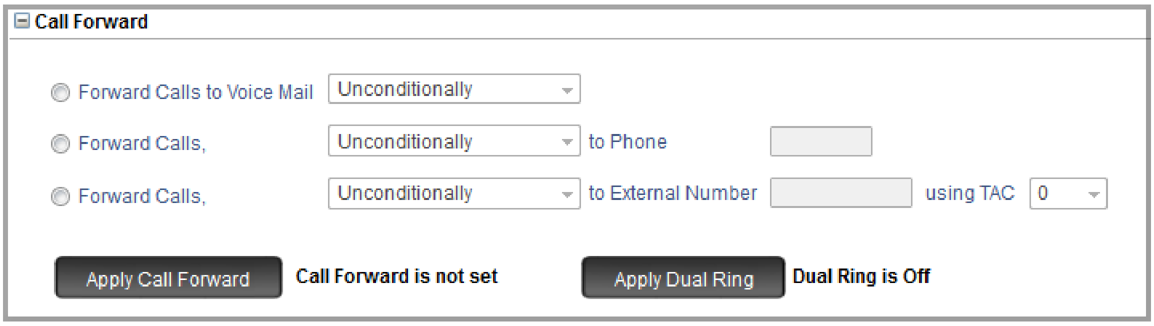
Settings Call Forward for an Extensions using SA Mode

- Login as System Administrator.
- Click **Extension**.
- The Extension numbers appear on the tabs on the top pane, starting with SLT extensions.



- To select a particular extension number or type, click on the tab.

- Click **Call Forward** to expand.

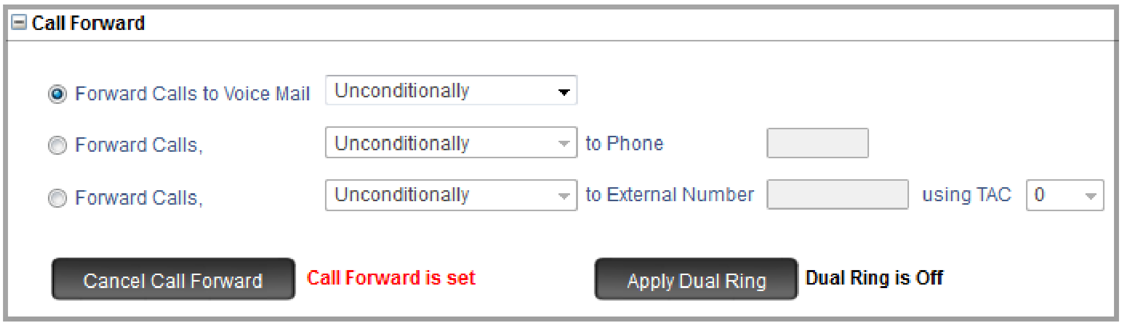


- Select the type of Call Forward you want to set for the extension:
  - To forward calls to voice mail, select the radio button **Forward Calls to Voice Mail** and the type of call forward. Default: Unconditionally.
  - To forward calls of this extension to another extension, select **Forward Calls to Phone** radio button 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** radio button, and select the type of call forward. Default: Unconditionally.

    Enter the external number to which calls must be forwarded.

    By default, TAC 0 is assigned for routing external calls. You can change the TAC as per your requirement.
- Click **Call Forward** to set Call Forward.

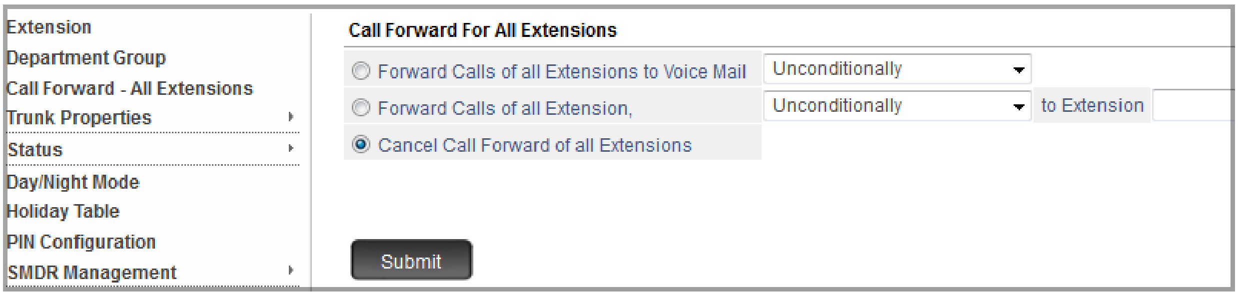


  The label of the Call Forward button changes color and the message “Call Forward is set” appears.
- Click **Dual Ring** to set Call-Forward Remote with Dual Ring.

  The label of the Dual Ring button changes color and the message “Dual ring is On” appears.
- To set Call Forward on another extension, click the extension number tab, and follow the same instructions as above.

You can also forward the calls of all extensions at one go to the same destination from the SA mode. To do this,

- Click **Call Forward- All Extensions**.



  - 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.**

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 Extended IP Phone Users

To set Call Forward

- Press the 'Call Forward' key.
  OR
- Dial 13.
- Scroll to select the desired Call Forward Type.
- Press 'Enter' key.
- Enter destination Phone Number/Voice Mail/ Department Group Number.
- You get a confirmatory text message and confirmation tone.
- Go Idle.

To set Call Forward-Dual Ring

- First, set the desired Call Forward type.
- Press the 'Call 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.

- *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.*
  - *For users in USA, TAC for dialing external numbers are: 9, 5.*
- *If call is to be forwarded on voice mail, dial the Access Code for the Voice Mail. The default Access Code is 390. 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 'Call 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/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-Scheduled

Extension users may want their calls to be automatically forwarded to a desired destination number during Day (Working hours) or Night/Break (Non-working hours).

For 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 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 and non-working 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 the 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 Time Zones: Day (Working hours) and Night/Break (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 “Call Forward” topic, 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 (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 “Call Forward” topic.
- 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.*
- *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 canceled 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 configure 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 Call Forward option* ***If call is forwarded to an external number*** *must be configured on 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 configuration of this feature involves the same parameters as in “Call Forward”.

- Call Forward must be enabled in the Class of Service (CoS) of the extensions to which this feature is to be allowed.
- If Call Forward No-Reply is to be set, and if required, the Call Forward No-Reply Timer may be configured.
- You may select the types of calls—internal, trunks, all calls—to be forwarded to the external number by configuring the option **Call Forward** option **If call is forwarded to an external number** on the extensions.
- 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 Class of Service. 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 System Administrator mode.

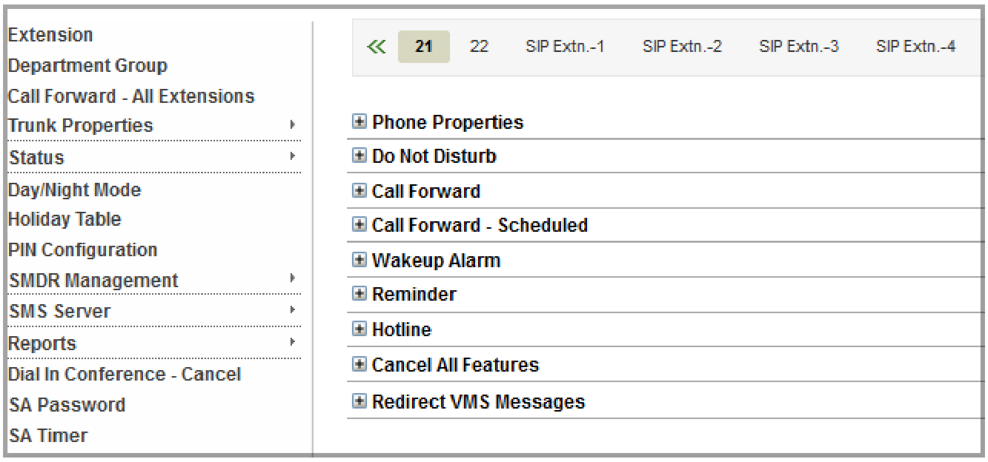
- *The destination number for forwarding calls can be a maximum of 16 digits. Terminate the command with #* if destination number is less than 16 digits.*
- *If the destination number is an external number, enter the Trunk Access Code followed by the destination number.*

Setting Call Forward-Scheduled for Extensions 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.

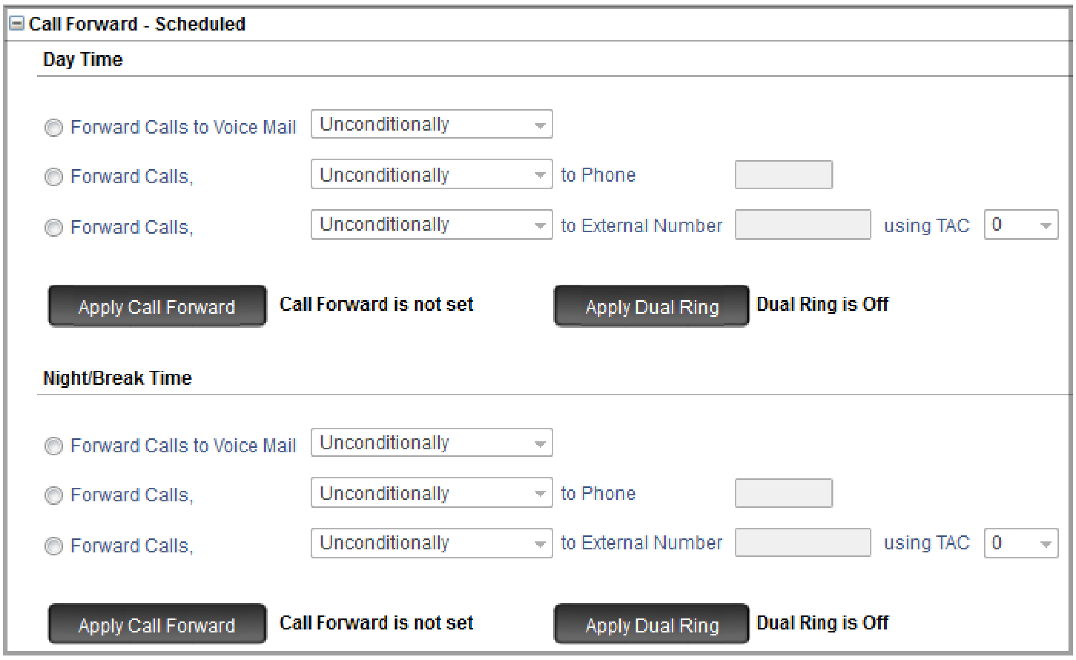
Setting Call Forward-Scheduled using Jeeves

- Login as System Administrator.
- Click **Extension**. The Extension numbers appear on the tabs on the top pane, starting with SLT extensions.



- To select a particular extension number or type, click the tab.

- Click **Call Forward-Scheduled** to expand.



- You can set **Call Forward - Scheduled** for Day Time and Night/Break Time.

  To set Call Forward for the extension for the Day Time, under **Day Time,** select the type of Call Forward:

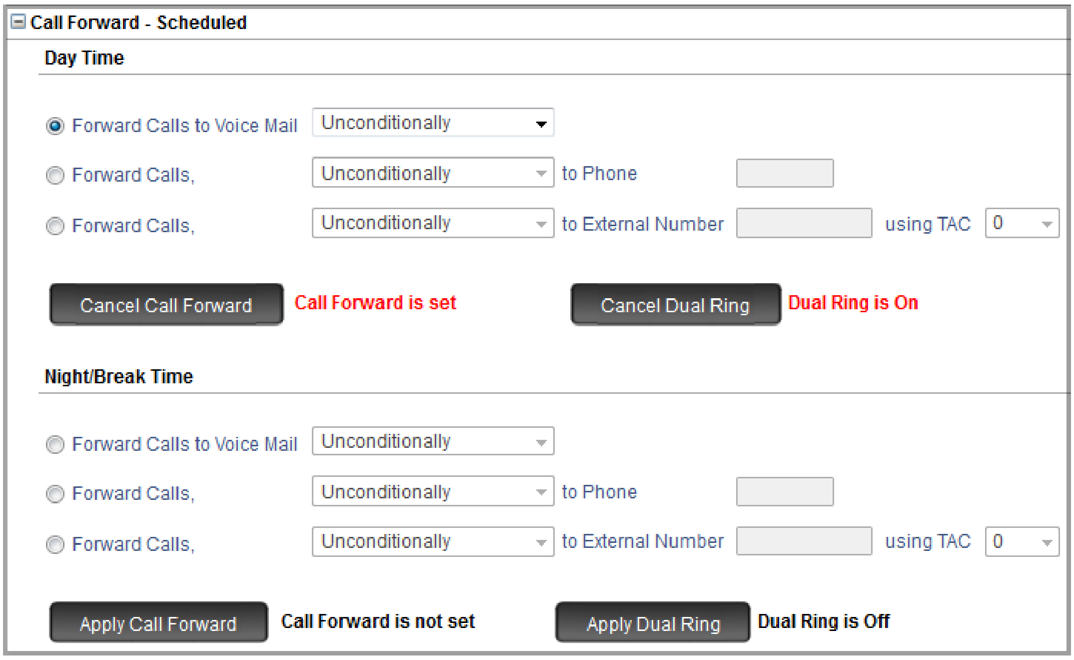
  - To forward calls to voice mail, select **Forward Calls to Voice Mail** radio button and the type of call forward. Default: Unconditionally.
  - To forward calls of this extension to another extension, select **Forward Calls to Phone** radio button and the type of call forward. Default: Unconditionally.

    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** radio button, and select the type of call forward. Default: Unconditionally.

    Enter the external number to which calls must be forwarded.

    By default, TAC 0 is assigned for routing external calls. You can change the TAC as per your requirement.

- Click **Apply Call Forward** to set Call Forward -Scheduled.



The confirmation message “Call Forward is set” appears.

- Click **Dual Ring** to set Call-Forward Remote with Dual Ring.

  The confirmation message “Dual ring is On” appears.

- To set call forward for Night/Break Time, 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

For Extended IP Phone Users

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 from SA mode

To set CF-Scheduled for the Day

- Lift handset.
- Enter SA Mode.
- 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.
- Exit SA mode
- Replace handset.

To set Call Forward-Scheduled for the Night/Break

- 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.
- Replace handset.

To cancel Call Forward-Scheduled for all Time Zones

- Lift handset.
- Dial ***1072-223-Extension Number-0***
- Replace handset.

Call Forward-Remote

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.

Call Forward-Remote is possible only from the System Administrator (SA) mode. When set/canceled from the SA mode, it will not depend on the assigned CoS.

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 (CoS)” of extensions that are to be allowed this feature.

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.

By default, SA Mode is disabled on all extension types.

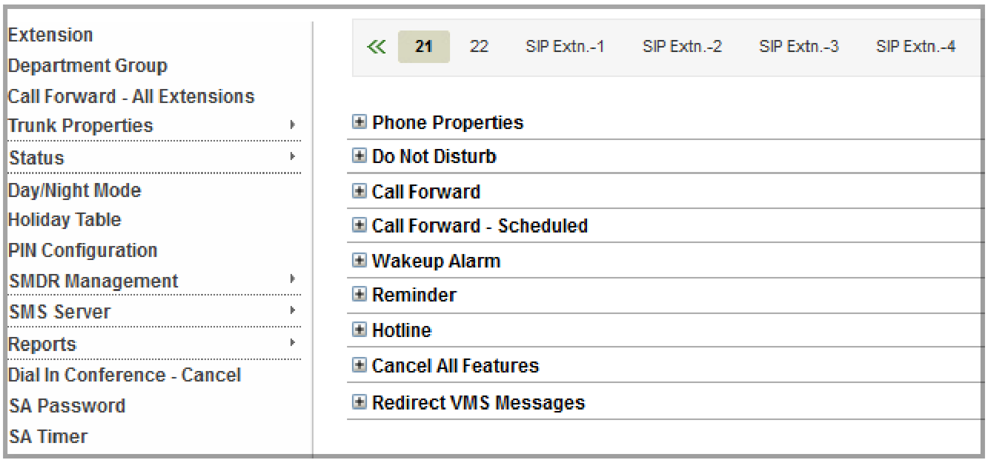
You may decide which extensions should be allowed Call Forward-Remote feature. Generally, only few extensions are allowed this feature.

For instructions on configuring SA Mode and SA Extension in the Class of Service of different extension types, see “SLT Extensions” and “SIP Extensions” under *Basic Settings*.

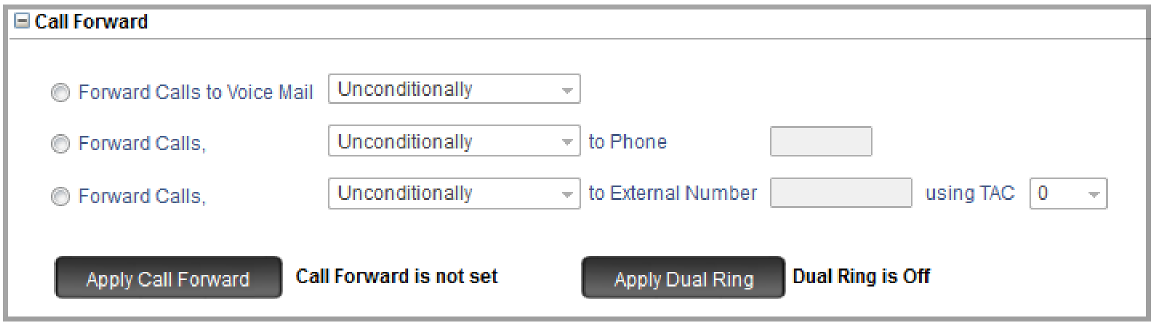
How to use

Setting Call Forward - Remote from Jeeves

- Login as System Administrator.
- Click **Extension**.
- The Extension numbers appear on the tabs on the top pane, starting with SLT extensions.



- To select a particular extension number or type, click on the tab.
- Click **Call Forward** to expand.



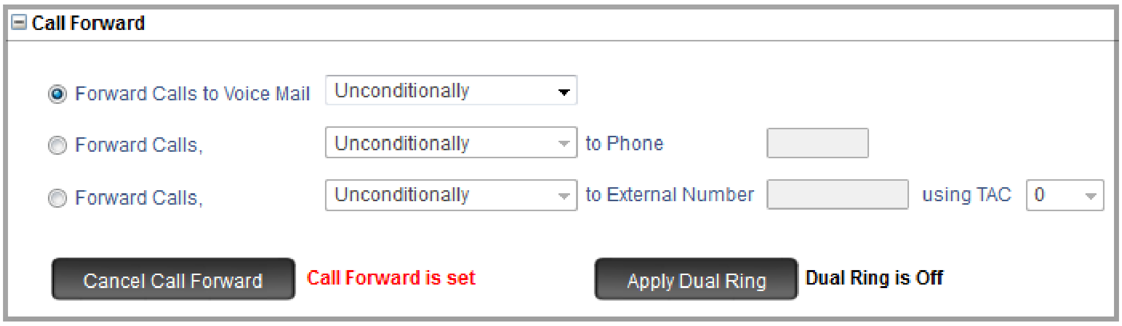
- Select the type of Call Forward you want to set for the extension:
  - To forward calls to voice mail, select the radio button **Forward Calls to Voice Mail** and the type of call forward. Default: Unconditionally.
  - To forward calls of this extension to another extension, select **Forward Calls to Phone** radio button 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** radio button, and select the type of call forward. Default: Unconditionally.

    Enter the external number to which calls must be forwarded.

    By default, TAC 0 is assigned for routing external calls. You can change the TAC as per your requirement.

- Click **Call Forward** to set Call Forward.



The label of the Call Forward button changes color and the message “Call Forward is set” appears.

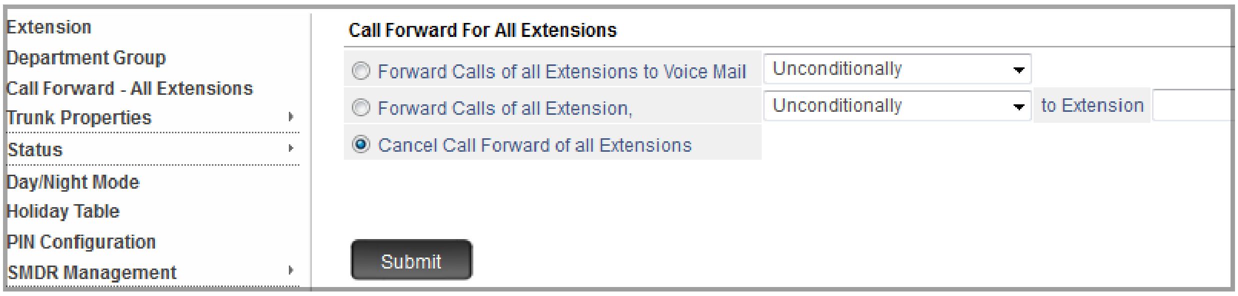
- Click **Dual Ring** to set Call-Forward Remote with Dual Ring.

  The label of the Dual Ring button changes color and the message “Dual ring is On” appears.

- To set Call Forward on another extension, click the extension number tab, and 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 **Forward Calls of all Extensions** radio button and select the Call Forward type.
- In **to Extension**, enter the Call Forward destination number.
- Click **Submit**.
- To cancel, select the **Cancel Call Forward of all extensions** radio button.
- Click **Submit**.

Using Call Forward - Remote from a Telephone

For 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[117]/Voice Mail[118]/Department Group.
- You get a confirmation tone and a text message for the Call Forward type set.
- Go Idle.

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.

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
  - 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.
- 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.

---

117. *If call is to be forwarded to an extension of 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. For users in USA, TAC for dialing external numbers are: 9, 5.

118. *If call is to be forwarded on voice mail, dial the Access Code for the Voice Mail. The default Access Code is 390.*

Call Forward - When Not Registered

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 a time zone—Day Time and Break/Night Time, by setting *Call Forward-When Not Registered - Scheduled*.

Call Forward - When Not Registered-Scheduled can be set for Day Time and Break/Night Time on the same SIP Phone. It can be canceled individually for a desired time zone, or both time zones at once. A different destination number can be set for forwarding calls for Day Time, Break Time and Night Time. For example, the destination number for Night Time can be a mobile number and the destination number for Day Time 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,

- make sure the Call Forward feature is enabled in the Class of Service (CoS) of the SIP extensions. For instructions, see “Class of Service” under the topic “SIP Extensions”.

- if required, select 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 **If Call is forwarded on external number** on the SIP Extension. For instructions, see “Call Forward” under “SIP Extensions”.

If you want to allow Call Forward-When Not Registered to be set only by the System Administrator (SA) for the extension users, you must disable the ‘Call Forward’ feature in the CoS assigned to the SIP extensions.

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.

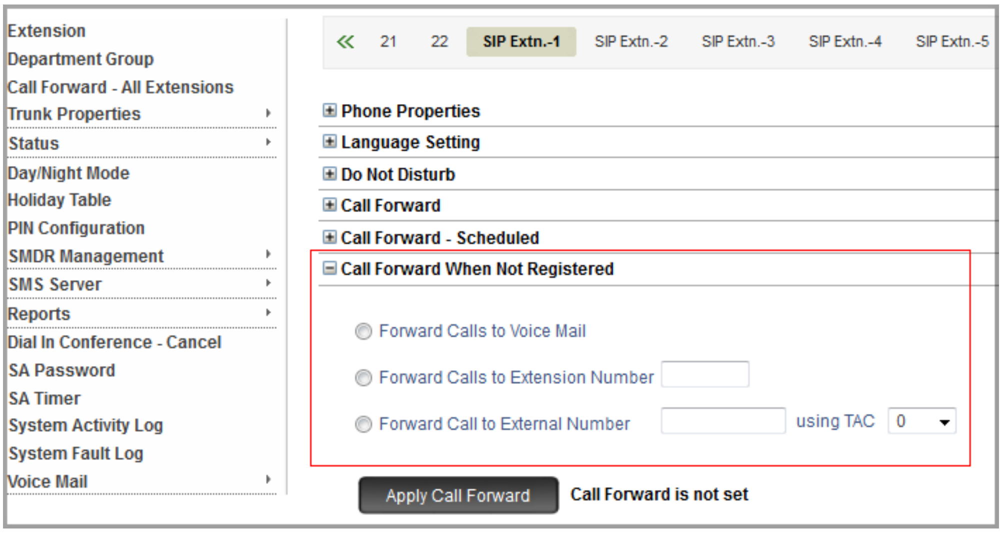
Setting Call Forward-When Not Registered

Call Forward-When Not Registered can be set from

- the SA mode from Jeeves.
- the SIP phones connected as extensions.

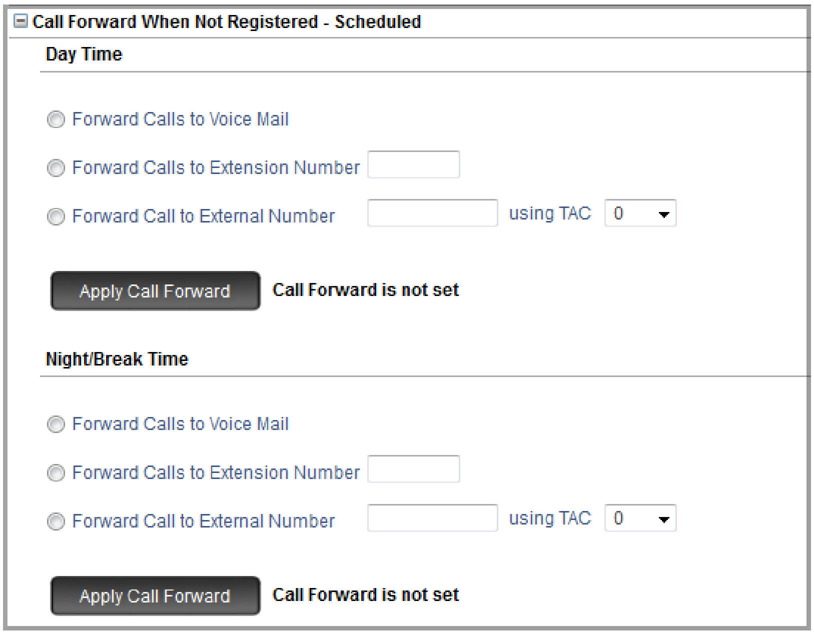
Call Forward-When Not Registered set by SA

- Login as System Administrator.
- Click **Extension**.
- Click the desired SIP Extension number on which you want to set this feature.
- 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 **Apply Call Forward**. 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 the day, under **Day Time**, 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 **Apply Call Forward**. The message “Call Forward is set” appears.
- To set call forward for **Night/Break Time**, 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.

For Matrix Extended IP Phone Users

- 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 'Day Time/ Break Time/ Night Time', 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 'Day Time/ Break Time/ Night Time', 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.

For Standard IP Phone Users

- 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 Day Time.
- Dial ***13-3-1-Destination Number*** for Break Time.
- Dial ***13-4-1-Destination Number*** for Night Time.
- Replace handset.

To cancel Call Forward - When Not Registered

- Dial ***13-1-0***.

To cancel Call Forward When Not Registered - Scheduled

- Dial ***13-2-0*** for Day Time
- Dial ***13-3-0*** for Break Time.
- Dial ***13-4-0*** for Night Time.
- Replace handset.

To cancel All Call Forward-When Not Registered

- Lift handset.
- Dial ***13-0.***
- Replace handset.

Call Hold

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 an Extended IP Phone and is retrieved from the same Extended IP Phone that put it on hold.
- **Global Hold:** An on-going conversation is put on hold from an Extended IP Phone and is retrieved from any 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 only on SIP Extensions.*
- *Consultation Hold is supported on the SLTs and SIP 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 that is put on hold.*

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 Extended IP Phone that has put the call on hold. The Extended IP Phone rings and the user may answer the call.
- The returned call is disconnected, if the Extended IP Phone is not in the idle state, or if the call is not answered by the Extended IP Phone user.

A call placed on Exclusive Hold can be retrieved in the following ways:

- Pressing the Hold key again (when the Extended IP Phone is idle).
- Pressing the Call Appearance key of the call put on hold (when the 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

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.

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 Extended IP Phone that has put the call on hold. The 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 Extended IP Phone is not in idle state, or if the call is not answered by Extended IP Phone user.

A call placed on Exclusive Hold can be retrieved in the following ways:

- Pressing the DSS key (when the Extended IP Phone is idle).
- Pressing the Call Appearance key of the call put on hold (when the 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 Extended IP Phone connected to 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 Extended IP Phone which put it on hold. The Extended IP Phone rings and the user may answer the call.
- The returned call is disconnected, if the Extended IP Phone is not in idle state, or if the call is not answered by Extended IP Phone user.

To be able to place calls on Global Hold, you must select **Global Hold** as the **Default Call Hold Type** in the System Parameters. The 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 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)”

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 this feature to work, you must configure the following parameters:

- **Class of Service:** Call Hold must be enabled in the Class of Service (CoS) of the Extended IP Phone/SLT users you want to allow this feature.

  Call Hold is included in the 'Basic Features' in the Class of Service assigned to the users. So, all extensions of SARVAM UCS can use this feature.

  Refer the topics “Class of Service (CoS)” 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 check box on the SIP Trunk. See “VoIP” in “SIP Trunks” to know more.

- **DSS Keys:** Configure DSS Keys for Trunks and Extensions on the Extended IP Phones, which are allowed to retrieve calls on Global Hold. Configure the DSS Keys for Exclusive Hold, if required. Refer the topic “Phone Key Settings” in “Configuring Matrix SPARSH VP248”, “Phone Key Settings” in “Configuring Matrix SPARSH VP310”, “Phone Key Settings” in “Configuring Matrix SPARSH VP330”, “Phone Key Settings” in “Configuring Matrix SPARSH VP210” and “Phone 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” topic.

For *Consultation Hold* to work, Call Hold must be enabled in the “Class of Service (CoS)” of the Extended IP Phones you want to allow this feature.

How to use

Exclusive Hold

For 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 Extended IP Phone.
- Press DSS Key of the Trunk/extension you put on hold from your 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 Extended IP Phone.
- Press DSS Key of the Trunk/extension you put on hold from your Extended IP Phone.

To put a call on Exclusive Hold using DSS key assigned to Exclusive Hold

- You are in speech with 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 Extended IP Phone.
- Press DSS Key of the Trunk/extension you put on hold from your 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 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 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 Extended IP Phone, press the DSS Key of the Trunk/extension put on Global hold.

Consultation Hold

For 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 Logs

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 Extended IP Phone extensions only.

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, i.e. the latest call detail will replace the record of the oldest call detail.

Given the limited Call Log capacity, the system allows you to choose if you want internal calls to be displayed or not in the Missed, Answered and Dialed Call Logs, and accordingly stores the 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 configuration, except:

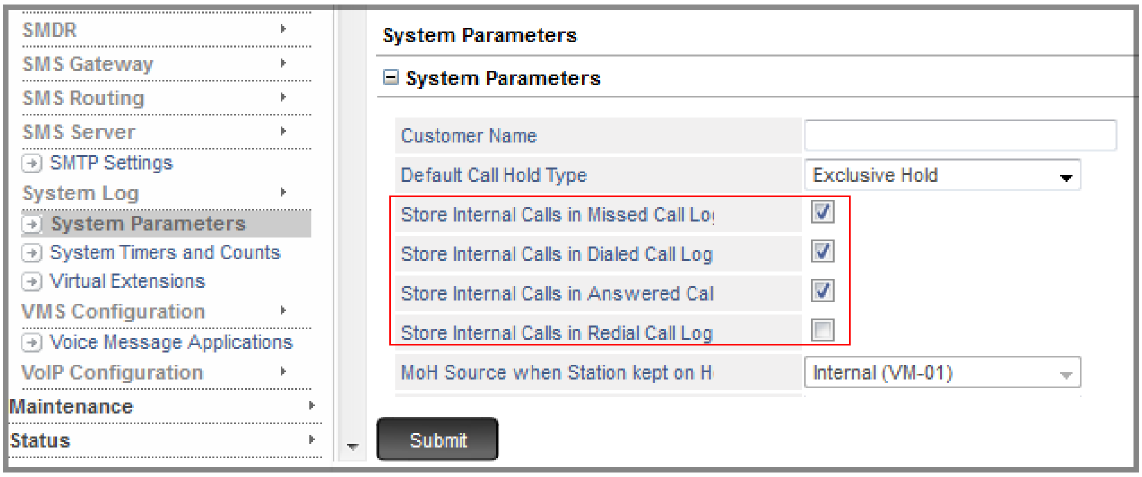
- Selecting whether internal calls should be logged in the Missed, Answered and Dialed Call Logs. This can be done on the '“System Parameters”' page under *Advanced Settings*.

- Programming of a DSS key for the Call Logs feature. See “Phone Key Settings” under “Configuring Matrix SPARSH VP248”, “Configuring Matrix SPARSH VP310”, “Phone Key Settings” under “Configuring Matrix SPARSH VP210”, “Phone Key Settings” under “Configuring Matrix SPARSH VP330” and “Phone Key Settings” under “Configuring Matrix SPARSH VP510”.

Configuring Internal Call Logging using Jeeves

- Login as System Engineer.

- Under **Advanced Settings**, click **System Parameters**.



- Under General Parameters, you may enable any or all of the following check boxes:
  - Store Internal Calls in Missed Call Log
  - Store Internal Calls in Dialed Call Log
  - Store Internal Calls in Answered Call Log
  - Store Internal Calls in Redial Call Log

- Click **Submit**.

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 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"[119].
- Enter the name of the contact.
- Press Enter key.
- 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.*

119. Only if there is a free Memory Index in the Personal Directory.

Call Park

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.

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 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.
- 2-9 are the 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 on-hold music.
- 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 be able to use this feature, you must do the following configuration:

- Enable Call Park in the "Class of Service (CoS)" of the extensions which you want to allow this feature. For instructions see "SLT Extensions" and "SIP Extensions" under *Basic Settings*.

- Assign DSS key for 'General Call Park - Auto'.
  OR
  Assign a separate DSS key for each General Orbit (2-9).

  For instructions, see **Phone Key Settings** under "Configuring Matrix SPARSH VP248", "Configuring Matrix SPARSH VP310" and "Configuring Matrix SPARSH VP330", "Configuring Matrix SPARSH VP210" and "Configuring Matrix SPARSH VP510".

- If required, change the value of the Call Park Timer and the Call Park Release Timer to the desired duration. See "System Timers and Counts" for instructions.

How to use

For 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 (indicated through LED) General Orbit:

- 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 only

- You are in speech with extension/external caller.
- Press DSS Key/CSF Key[120] assigned to ‘General Call Park - Auto[121]’
  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'.
- Dial 116
  OR
- Enter Orbit Number where you parked the call (1-9)
  (Personal Orbit:1; General: 2-9).

For SLT Users

To park a call

- 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.

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120. This is applicable only for SPARSH VP510.
121. This feature is not applicable for SPARSH VP330/SPARSH VP210.

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 Pickup

Call Pickup 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.

SARVAM UCS offers two types of Call Pickup:

- **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 Pickup 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 calls 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 SIP Extensions of SARVAM UCS, the system will send information for the call present in the first call loop only.*

How it works

Call Pickup Group

- Extensions must be assigned to Call Pickup Groups. The extensions in a Call Pickup group may be SLT and SIP Extensions.
- As many as 8 such groups may be formed.
- Each group is assigned a number 1 to 8.
- For example, extensions 21, 22, 23, and 33, 34, 35 are assigned to Pick-Up Group number 3.
- When an extension in this group rings, any extension in the group 3 can pick up the call by dialing '4' the (default) feature access code for "Call Pickup Group".
- The ringing extension should be in the same Pick-Up Group.

Call Pickup Selective

- Extensions need not be in Call Pickup 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 the ringing extension.

When more than one extension in a Pick-Up Group are ringing, you can choose which one to answer first, using Call Pickup Selective.

Feature Interactions:

- **Call States:** Call Pickup will fail if the ringing extension goes into idle state just when you are dialing the pick-up access code.
- **Auto Call Back:** Call Pickup will fail if the call ringing on the extension is an Auto Call Back request.
- **Alarms:** Call Pickup will fail if the call ringing on the extension is an Alarm Call.

How to configure

Creating Call Pickup Groups

On a sheet of paper, list the extensions that are to be grouped into a Call Pickup Group. Make as many Call Pickup Groups as required. Assign each group a number.

Call Pickup Group Number	SLT Extensions	SIP Extensions
1	201, 202, 203	321, 322, 323
2	204, 205, 206	324, 325, 326
3		
:		
8		

The numbering of Call Pickup Groups must start from 1 and end at 8.

You cannot select same extensions for two Call Pickup Group. Selecting same extension for another Call Pickup Group will delete the extension from the previous group and will allot it to the new group.

Configuring Extensions in Call Pickup Groups

For this feature to function, go to *Basic Settings* of Jeeves,

- Configure "Call Pickup Group". You may refer to the sheet of paper you prepared.
- For each SLT/SIP extension assigned to a CPU Group, make sure **Call Pickup** is enabled in the "Class of Service (CoS)" of the extensions for the Day and Night/Break. Default: Enabled. See "Basic Settings" for instructions on configuring CoS of different Extension types: SLT, SIP.

  If you wish to receive Call Pickup Notification (only applicable to SPARSH VP510), make sure you have enabled the **Call Pick-up Notification (Only for SPARSH VP510)** check box. See "SIP Extensions" as well as refer to the refer to the EON510_SPARSH VP510 V2 User Guide.

How to use

For Extended IP Phone Users

To pick up a ringing extension in your Group

- Press DSS Key assigned to Call Pickup 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 Pickup 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

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.

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.	Tooooooo......... 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 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 System 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”.

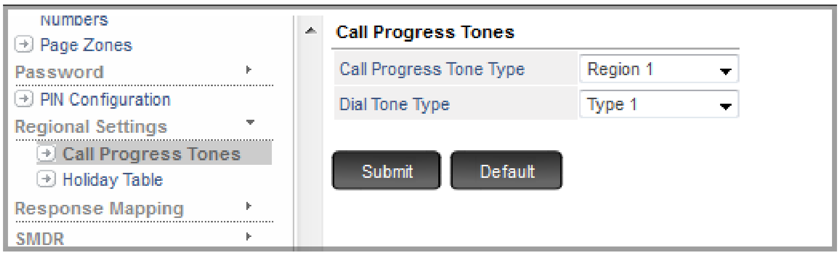
For countries that use different frequencies and cadences for the same tone, e.g. USA, only one frequency/cadence among the group is considered. See “Default CPTG Type” at the end of this topic.

How to configure

Configuration of Call Progress Tones involves 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, you may change the CTPG type set by the system. To do this,

- Login as System Engineer.
- Under **Advanced Settings**, click **Regional Settings**.
- Click **Call Progress Tones**.



- Select the desired **Region** from the drop down list.
- Select the desired **Dial Tone Type** as **Type 1** or **Type 2** from the combo box.
- Click **Submit.**

To change CPT-related Timers,

- Click the **System Timers and Counts**.
- Go to **Call Progress Tones**.
- Change the values of the CPT-related Timers as desired.
- Click **Submit.**

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.

Demonstration of Tones

It is possible to demonstrate Call Progress Tones to users by dialing the SE commands from a 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.

To demonstrate call progress tones:

- Enter SE mode from a 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	400*25	Continuous	400*25	Continuous	400*25	.4on .2off .4on 2.0off	425	0.380on 0.380off
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	400*50	Continuous	400*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
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 1off	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

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)
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	0.38on 0.38off
36	UK	350+440	Continuous	350+440	Continuous	400+450	0.4on 0.2off 0.4on 2.0off	400	0.38on 0.38off
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	350+440	Continuous	440	Continuous	350+440	1.0on 3.0off	440	0.75on 0.75off
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
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	0.380on 0.380off	425*25	0.1on 0.1off	425* 25	0.1on 0.9off	425	0.2on 0.2off 0.2on 4.4off	425	0.1on 2.9off
6	Brazil	425	0.25on 0.25 off	425	0.1on 0.1off	425	0.1on 0.9off	425	0.1on 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

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)
9	Egypt	450	0.5on 0.5off	425*50	0.1on 0.1off	425* 50	0.1on 0.9off	400*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	0.2on 0.3off 0.2on 1.5off
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	0.2on 4.9off 0.2on 4.9off
14	Indonesia	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
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 0.3on 0.1off 0.2on 5.0off	425	0.1on 2.9off
19	Japan	400	0.25on 0.25off	400	0.1on 0.1off	400	0.1on 0.9off	400*25	0.5on 2.0off 0.1on 0.5off 0.1on 3.5off	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	0.3on 0.3off 0.3on 3.3off	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

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)
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	0.2on 0.2off 0.2on 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	0.4on 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	0.2on 0.2off 0.2on 10.0off	425	0.1on 2.9off
30	Singapore	425	0.25on 0.25off	425	0.13on 0.13off	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	0.2on 0.3off 0.2on 1.5off
32	Spain	425	0.25on 0.25off	425	0.1on 0.1off	425	0.1on 0.9off	425	0.2on 0.2off 0.2on 3.5off	425	0.1on 2.9off
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 0.23on 0.53off	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 0.23on 0.53off	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

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)
39	Belgium	440	0.25on 0.25off	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
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 (Regionwise)

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

When the Ethernet port of the system is connected to a public IP network, it may be necessary to restrict traffic to and from a particular IP address only.

With the feature 'Call Restriction based on IP Address', SARVAM UCS makes it possible to entertain requests on its Ethernet port for 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. See “Trusted IP Address/es” under “SIP Trunks” for instructions.

Call Taping

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.

This feature is useful for keeping records of important conversations. For this feature to work, the system must have the Voice Mail System present.

You can tape:
- Incoming and outgoing external calls.
- Incoming and outgoing internal calls.

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 be configured in the **Call Taping Profile**. This Profile is assigned to the extensions on which Call Taping is to be used.

- **Call Taping** must be enabled on 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 Incoming Calls Received without CLI** must be enabled in the Call Taping Profile assigned to the extensions on which Call Taping is to be used.

To be able to record internal calls, the **Tape Internal Calls** check box must be enabled in the Call Taping profile 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 Call Taping Profile 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.

A calls C

- The system matches the dialed number with the numbers configured in the **Tape Outgoing calls made to following numbers** in the Call Taping Profile assigned to A. 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 configured in the **Tape Incoming Calls received from following numbers** in the Call Taping Profile assigned to B. The system finds a match.

- When speech is established, the system records the speech between D and B in E's mailbox.

- Call Taping Beeps will be played to D and B, if **Play Beep when Raid/Call Taping/Conversation Recording starts** is enabled in the System Parameters.

*The system will tape incoming calls without CLI, only if the parameter **Tape Incoming Calls received without CLI** is enabled in the Call Taping profile assigned to the extension.*

A calls B

- The system checks, if the **Tape Internal Calls** is enabled in the Call Taping Profile 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, if **Play Beep when Raid/Call Taping/Conversation Recording starts** is enabled in the System Parameters.

  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

To provide this feature to extensions,

- **Call Taping Profile** must be configured. A Call Taping Profile consists of the following parameters:

  - List of Incoming and Outgoing phone numbers for which the system should initiate Call Taping.

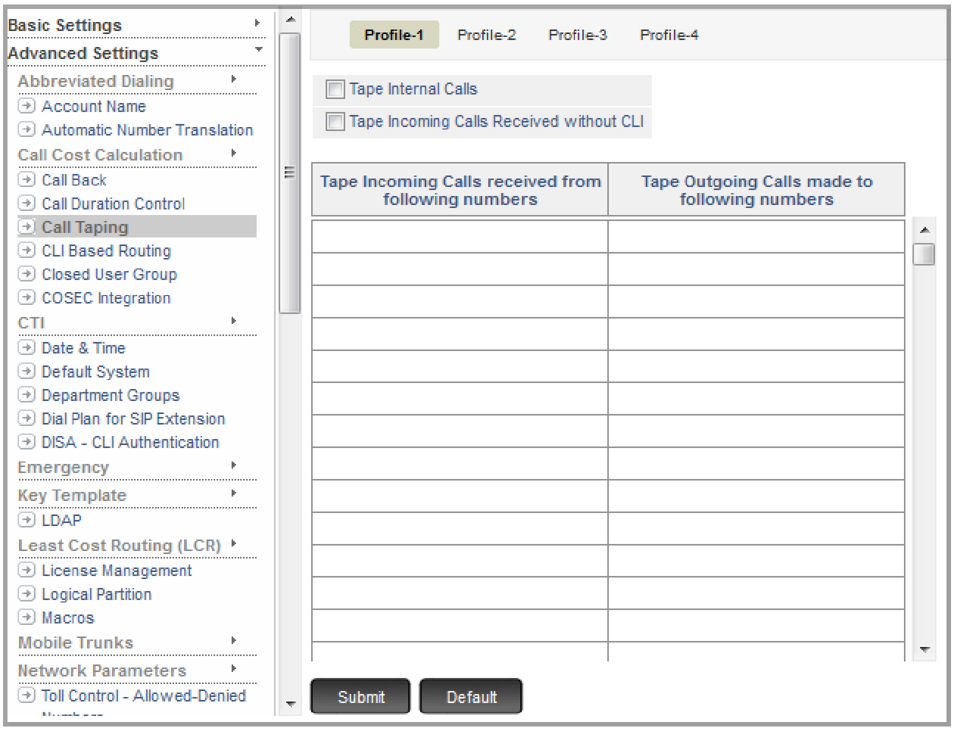
- Tape Internal Calls check box.
- Tape Incoming Calls Received without CLI check box.

You can configure four different Profiles and assign these profiles to different extensions.

- Specify where to save Call Taping files—**Common Mailbox** or **Personal Mailbox**.
- If Call Taping files are to be saved in a **Common Mailbox**, you must enter the extension number whose mailbox is to be used for Call Taping.
- If you select common mailbox, then select the type of file you want the system to generate for saving the taped conversation in **Save Call Tapping Files as**. You may select — *Individual File before and after call transfer* or *Single File before and after call transfer* as per your requirement**.**
- Call Taping must be enabled on the **Trunks** assigned to the extension users for making/receiving external calls.
- By default, **Play Beep when Raid/Call Taping/Conversation Recording starts** is enabled in the System Parameters. If required, you may disable these beeps.

To configure **Call Taping Profile**,

- Login as System Engineer.
- Under **Advanced Settings**, click **Call Taping**.
- **Call Taping Profile 1** appears.



- In **Tape Incoming Calls received from following numbers**, enter the numbers of external callers whose speech you want to tape.

- In **Tape Outgoing Calls made to following numbers**, enter the numbers of external parties dialed by extension users whose speech you want to tape.

- To tape calls between extensions, select **Tape Internal Calls** check box. Default: Disabled.

- To tape incoming calls received without a CLI, select **Tape Incoming calls received without CLI** check box. Default: Disabled.

- Click **Submit**.

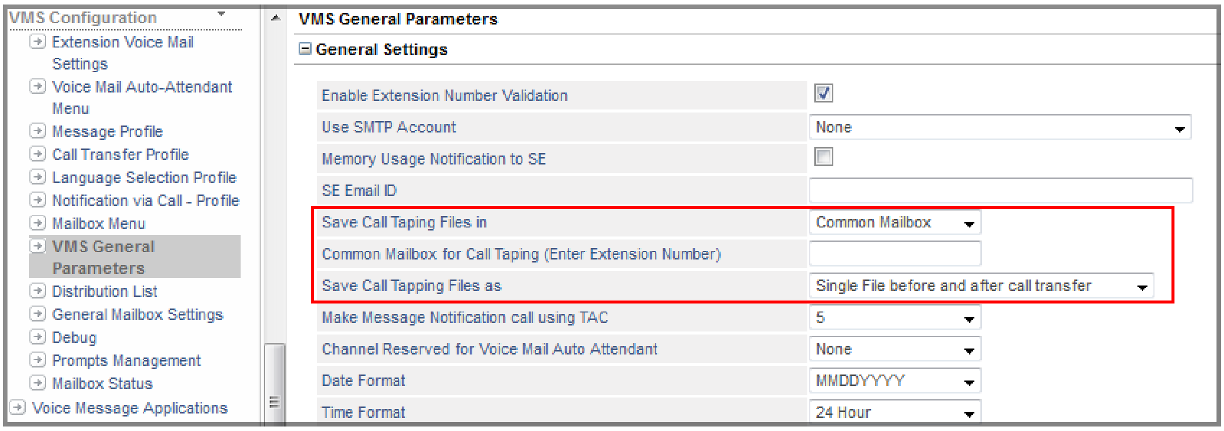
- To configure another Call Taping Profile, click the desired Profile number tab. Follow the same instructions as above to configure the Call Taping Profile.

- Now, assign the Call Taping Profiles you configured to the desired extension types. For instructions, see “SLT Extensions” and “SIP Extensions” under Basic Settings.

For Call Taping Mailbox,

- Under **VMS Configuration**, click **VMS General Parameters**.

  The **VMS General Parameters** page opens.



- Click **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, 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 taped 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 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 one single file for saving the conversation taped before and after the call is transferred

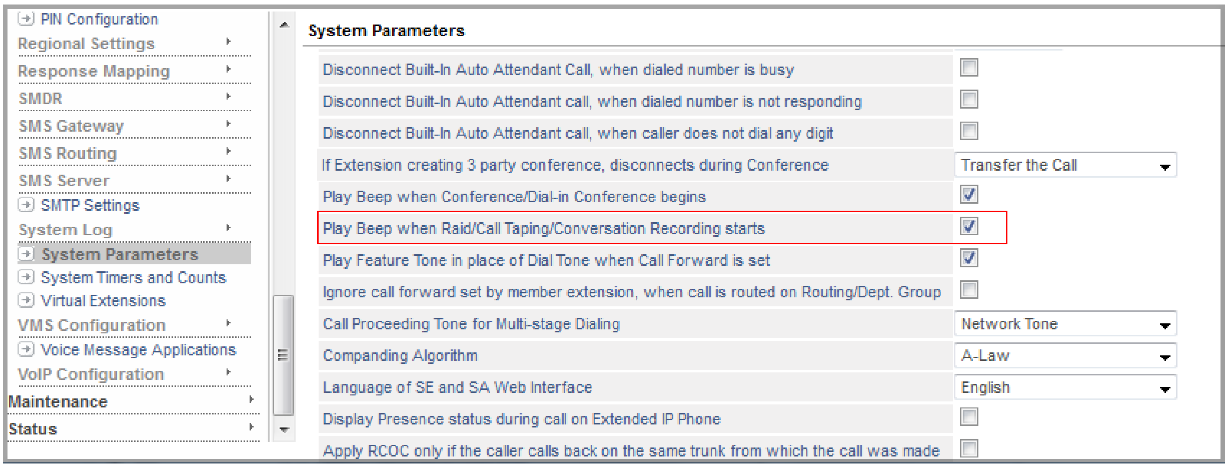
- Click **Submit.**

To enable **Call Taping on Trunks**,

- See **Call Taping** under "CO Trunks", "Mobile Trunks" and "SIP Trunks" under **Basic Settings** for instructions.

To enable/disable **Beeps during Call Taping**,

- Under **Advanced Settings**, click **System Parameters**. The System Parameters page opens.



- If required, disable the Beeps played when Call Taping starts by clearing the **Play Beep when Raid/Call Taping/ Conversation Recording starts** check box. Default: Enabled.

- Click **Submit.**

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

Extended IP Phone Users

- Press Voice Mail Key.
  OR
- Dial 390 -Your Mailbox Password[122]

- Follow Voice Mail Prompts to listen to new messages.

SLT Users

- Lift the handset.
- Dial 390-Your Mailbox Password[123]
- Follow Voice Mail Prompts to listen to new messages.
- Replace handset.

Accessing Common Mailbox

- Press Voice Mail Key.
  OR
- Dial 390 (default Voice Mail Access Code).
- Dial 0 to go to Home Position (default Access Code).
- Dial 8 (default Access Code).
- Dial the number of the extension to which the common mailbox has been assigned.
- Enter the password for the common mailbox[124].
- Follow Voice Mail Prompts to listen to the new/old messages.
- Replace handset, after you have finished listening to the messages.

If the common mailbox is password protected, make sure that you provide the password to all extension users who are to be provided access to this mailbox.

---

122. *Only if the mailbox is password protected, you will be prompted to enter the mailbox password.*
123. *Only if the mailbox is password protected, you will be prompted to enter the mailbox password.*
124. *Only if the mailbox is password protected, you will be prompted to enter the mailbox password.*

Call Toggle

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 (i.e. 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)" dependent feature. By default, Call Toggle is included in the 'Basic Features' allowed to all extensions of SARVAM UCS. You cannot disable this feature selectively in the CoS of extensions, without disabling the entire set of Basic Features.

No other configuration is required for this feature, except for programming a DSS key for Call Toggle, if required, on the Extended IP Phone extensions. See **Phone Key Settings** under "Configuring Matrix SPARSH VP248", "Configuring Matrix SPARSH VP310", "Configuring Matrix SPARSH VP330", "Configuring Matrix SPARSH VP210" and "Configuring Matrix SPARSH VP510".

How to use

For 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 Transfer

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 to him.
- **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" are allowed in 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 Hang-up 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 Consultation 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 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[125].
- 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 (Trunk to Trunk Transfer) 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:

- **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.

125. *The process of Trunk-to-Trunk transfer takes place outside of the PBX. So, the PBX 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.*

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 in the VMS of the system. For instructions, refer to “Configuring Voice Mail System”.

Call Transfer in Class of Service

By default Call Transfer is assigned to all the extensions of SARVAM UCS, as it is included in the 'Basic Features' assigned to the extensions. You cannot disable 'Call Transfer' selectively without disabling the entire set of 'Basic Features'.

By default Trunk-to-Trunk Transfer is disabled in the default CoS assigned to the extensions.

Refer to the topic “Class of Service (CoS)” 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.

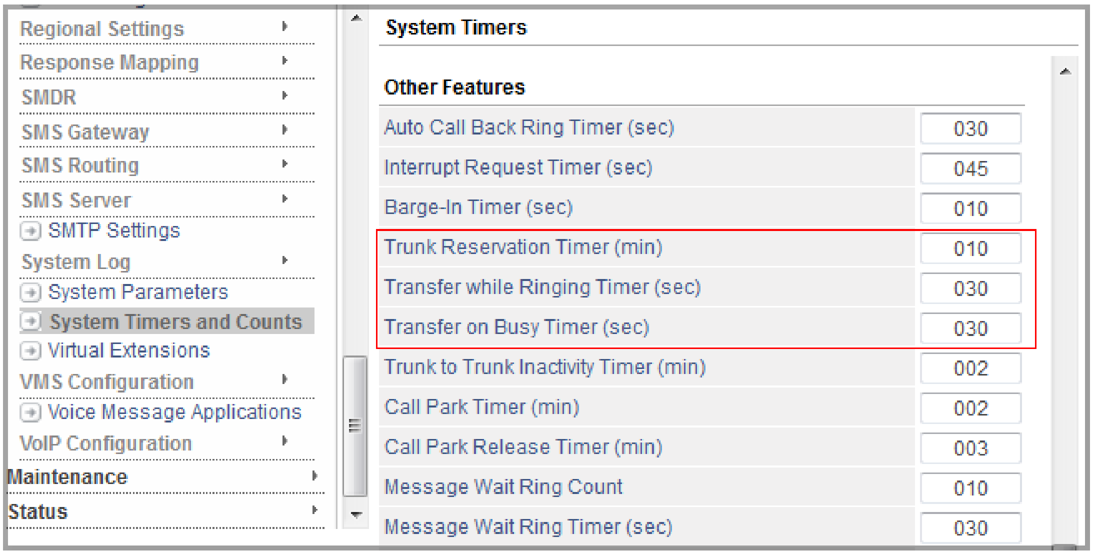
- **Transfer on Busy Timer:** This timer is related to Call-Transfer on Busy. It is the time for which the system waits for the busy extension to respond to the intrusion tone. By default the timer is set to 30 seconds. At the end of the timer the call is returned to the transferring extension.

- **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 only.

Changing Call Transfer Related Timers using Jeeves

- Login as System Engineer.

- Under **Advanced Settings**, click **System Timers and Counts**.



- Scroll to reach the Call Transfer related timers and change the values as required.

- Click Submit.

How to use

For 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[126]- External Number.
- Go ON-Hook or press ‘Transfer’ Key.

Trunk to Extension Transfer:

126.Trunk Access Code (TAC): Users worldwide may dial a code from 0, 5. Users in USA may dial a code from 9, 5.

- 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.
  OR
- Speech with Trunk.
- Press ‘Transfer’ Key and dial Trunk Access Code[127]-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.

127. Trunk Access Code (TAC): Users worldwide may dial a code from 0, 5. Users in USA may dial a code from 9, 5.

External Number to Extension Transfer:

- Speech with Trunk.
- Press Flash.
- Dial desired party's extension number.
- Replace handset.

Trunk to Trunk Transfer:

- Speech with External Number1.
- Dial Flash.
- Dial Trunk Access Code
- Dial External Number2.
- Speech with the External Number2.
- 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)

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 the Extended IP Phones. The calling number can also be presented 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 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 the network.
- 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 mobile 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.

CLIP and Caller ID Presentation on Call Transfer will work only if CLIR is not enabled on the extension that has transferred the call. Refer the topic "Calling Line Identity Restriction (CLIR)".

How to configure

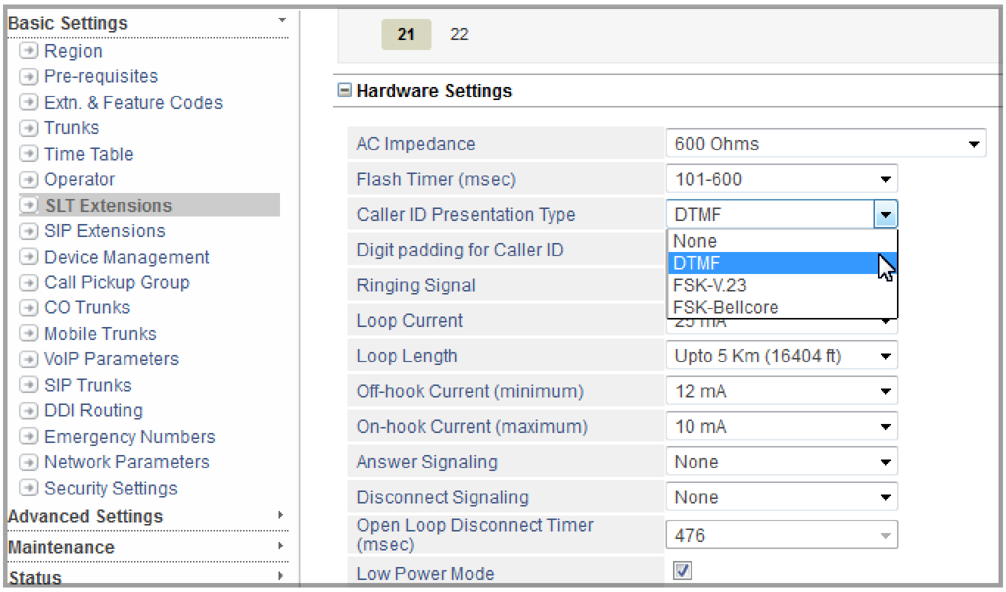
The functioning of this feature is controlled by two parameters: **CLIP Type** and **Caller ID on Call Transfer**.

If you want to replace '+' characters received as CLI on telephones that do not support CLI prefixed with this character, you may configure the **Replace '+' from CLI** option in the "System Parameters".

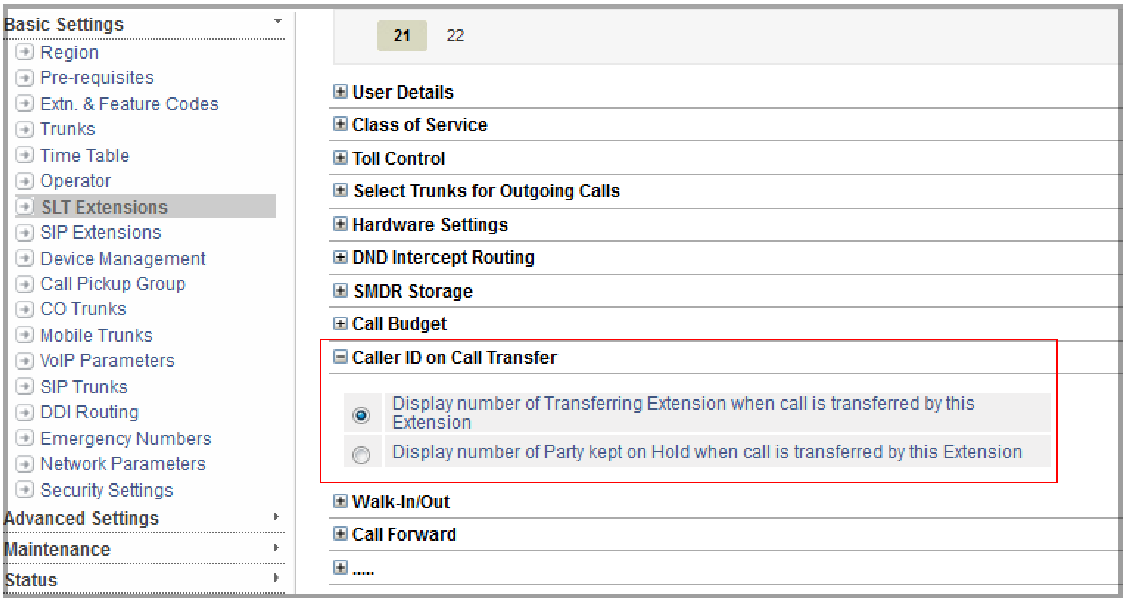
If SLTs supporting CLI are connected to SARVAM UCS, you must select a signaling protocol for CLI for the SLT extensions. **DTMF** is selected as CLIP Type for all SLT extensions.

To select CLIP type for SLT extensions,

- Login as System Engineer.
- Under **Basic Settings**, click **SLT Extensions**.
- Select the tab of the desired SLT.
- Click **Hardware Settings** to expand.



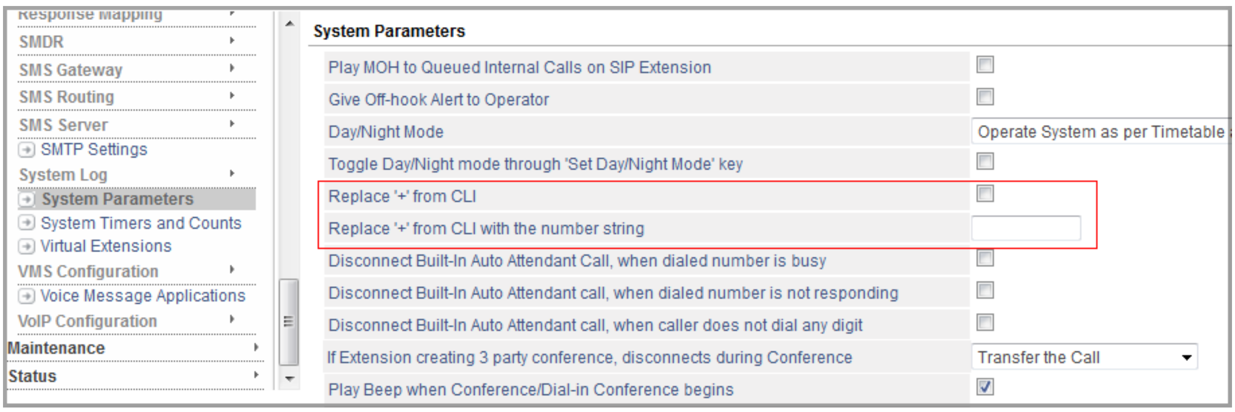
- As **Caller ID Presentation Type**, select the CLIP Type supported by the SLT: DTMF, FSK V.23, FSK-Bellcore. Default: DTMF.
- Click **Submit.**
- Now, click **Caller ID on Call Transfer** to expand.



- Select the desired option for displaying Caller ID to this extension when a call is being transferred to this extension:
  - **Display number of Transferring Extension when call is transferred by this Extension**
  - **Display number of Party kept on Hold when call is transferred by this Extension**
- Click **Submit.**
- To configure CLIP Type and Caller ID on Call Transfer on another SLT extension, click the desired extension number tab and follow the instructions as above.

To replace '+' characters received as CLI on telephones that do not support CLI prefixed with this character,

- Under **Advanced Settings**, click **System Parameters**.
- Click **System Parameters** to expand.



- Select **Replace '+' from CLI** check box.

- In **Replace '+' from CLI with the number string**, enter the desired number string that the system should replace the ‘+’ in the string with.
- Click **Submit.**

Calling Line Identity Restriction (CLIR)

SARVAM UCS allows extension users to suppress their identity, i.e. extension number and name, when they call other extensions. This feature is called Calling Line Identity 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 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 SLT that supports CLI.

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)".

By default, both these features are disabled on all extensions of SARVAM UCS. Thus, none of the extensions of SARVAM UCS can suppress their CLI or force any other extension to display its CLI.

Decide which extensions should be allowed CLIR and which should be allowed CLIR Override and accordingly configure their CoS. For instructions, see **Class of Service** under "SLT Extensions" and "SIP Extensions".

How to use

For Extended IP Phone Users

To enable CLIR

- Press DSS Key assigned to CLIR[128].
  OR
- Dial 1031 to enable CLIR.

128. You are 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.

- You get confirmatory tone and message on the phone's display.
- Go idle.

To disable CLIR:

- Press DSS Key assigned to CLIR again[129].
  OR
- Dial 1030.
- You get confirmatory tone and message on the phone's display.
- Go idle.

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.
- Replace handset.

129. *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.*

Cancel All Extension Features

For each feature of the SARVAM UCS that an extension user has set/enabled on the extension, the system provides access code for cancellation of the feature.

Extension users can 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 the extension user dials 'Cancel All Extension Features' command, the following features, if set, are canceled from the extensions:

- Auto Answer
- Auto Call Back
- Auto Redial
- Call Forward
- Call Forward - Scheduled
- Do Not Disturb
- Hot Line
- Trunk Reservation
- Walk-In Class of Service

Auto Redial, Call Forward and Call Follow-Me, Do Not Disturb, Hotline and Trunk Reservation are Class of Service dependent features. These features can be set/canceled from an extension only if these are included in its *“Class of Service (CoS)”**.*

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 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.

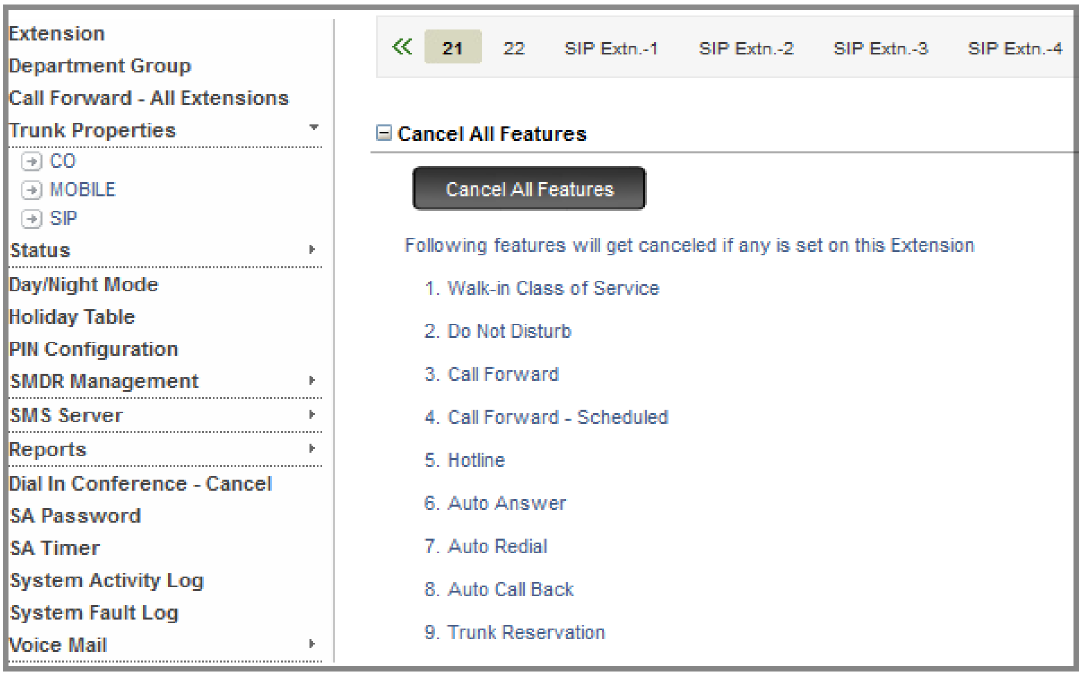
For SLT Users

- 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. To do this,

- Login as System Administrator.
- Click **Extension**.
- Click the desired Extension tab.
- The searched extension users details appear on your screen.
- Click **Cancel All Features** to expand.



- Click **Cancel All Features.** 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

Class of Service (CoS)

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 Day time (Working hours) may not be required during Night/ Break (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 Class of Service (CoS).

How it works

Each extension port of the System has an associated CoS for the Day and Night/Break, that indicates which features of the System the port is allowed to access during working hours and during non-working hours.

A feature can be allowed or denied to an extension by enabling or disabling it in the CoS of that extension.

The CoS of all extensions may be uniform, or different CoS can be assigned to different extensions, according to the time of the day. Doing so, each extension can access a different set of features during the Day (working hours) and at Night/Break (non-working hours).

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 the CoS of all extensions.

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 list of features supported on the extensions for the Day and Night/Break.

Default CoS

Feature	Allowed During	
	Day	Night/Break
Account Code	✓	✓
Auto Call Back-Busy	✓	✓
Auto Call Back-No Reply	✓	✓
Auto Redial		
Auto Redial Priority		
Basic Features[a]	✓	✓

Feature	Allowed During	
	Day	Night/Break
Barge-In		
Call Forward	✓	✓
Call Park	✓	✓
Call Pickup	✓	✓
Call Recording		
CLIR		
CLIR Override		
CUG (Closed User Group)		
Conference	✓	✓
Continued Dialing	✓	✓
Department Group - Call Forward	✓	✓
DISA		
Do Not Disturb - Override		
Do Not Disturb	✓	✓
DSS Call Pickup-Station		
DSS Call Pickup-Trunk		
Dynamic Lock	✓	✓
Dynamic Lock Timer		
Emergency Conference	✓	✓
Forced Answer		
Forced Release		
General Mailbox	✓	✓
Global Directory Part-1		
Global Directory Part-2		
Global Directory Part-3		
Global Directory Programming		
Hot Desk		
Hotline		
Intercom		
Interrupt Request	✓	✓
Live Call Supervision		
Message Wait (Set/Cancel)	✓	✓
Paging	✓	✓
PIN Dialing		

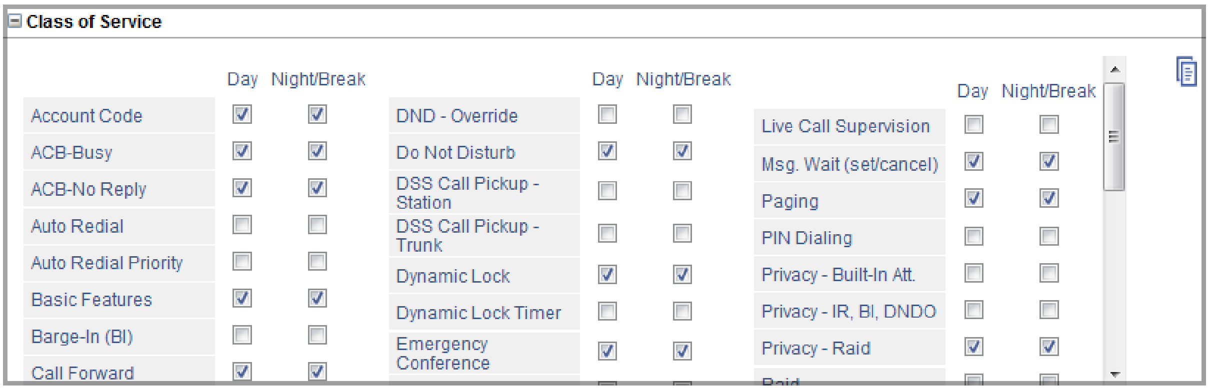
Feature	Allowed During	
	Day	Night/Break
Privacy from Built-In Auto Attendant		
Privacy from Interrupt Request, Barge-In and DND-Override		
Privacy - Raid	✓	✓
Raid		
Return Call to Original Caller (RCOC)	✓	✓
Room Monitor		
SA Extension		
SA Mode		
Selective Port		
Trunk Call Waiting		
Trunk Reservation		
Trunk-Trunk Transfer		

a. *Basic Features includes: Internal Call, Call Hold, Call Toggle, Call Transfer, Department Call, Operator Access, Redial, Mute.*

Assigning CoS to Extensions

- Prepare a list of SLT and SIP extensions by their names/numbers.
- Read the list of features supported on the extensions (see above Table 'Default CoS Groups for Features').
- Against each extension name/number on the list, write the features needed for the Day and Night/Break. 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 SLT and SIP extensions.
- Are there any other features, in addition to those on the common list, which you want to allow to select extensions?
- If yes, extend the common list you prepared by adding the features to be allowed to the select extensions.
- Now, to configure the CoS features to be allowed or denied to extensions, go to “Basic Settings” of Jeeves.
    - Click the extension settings link for SLT and SIP Extension.
    - Select the desired extension number.

The default CoS group features appear under *Class of Service*.



- The check boxes of features that are allowed for the Day and Night/Break are selected.

  The default CoS fulfills the requirements of most extension users. Check if the default CoS features you want to allow are enabled and features you want to deny are disabled.

- To allow a feature for the Day and Night/Break, select its corresponding check box.

- To deny a feature for the Day and Night/Break, clear its check box.

- Click **Submit**.

The following features can be set/canceled 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*

CLI Based Routing

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 number on a particular extension or group of extensions. This is known as CLI Based Routing.

How it works

Let us understand CLI Based Routing with an example.

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 and CLI Based Routing should be enabled on the desired trunks for each Time Zone.

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 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 first checks if CLI Based Routing is enabled on the trunk for the current time zone.
    - If enabled, the system checks whether 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 and names of the calling parties and the corresponding destination extensions in the CLI Based Routing Table. You can store up to 400 numbers in the CLI Routing Table.
- enable CLI Based Routing on the desired trunks for Day, Break and Night time.

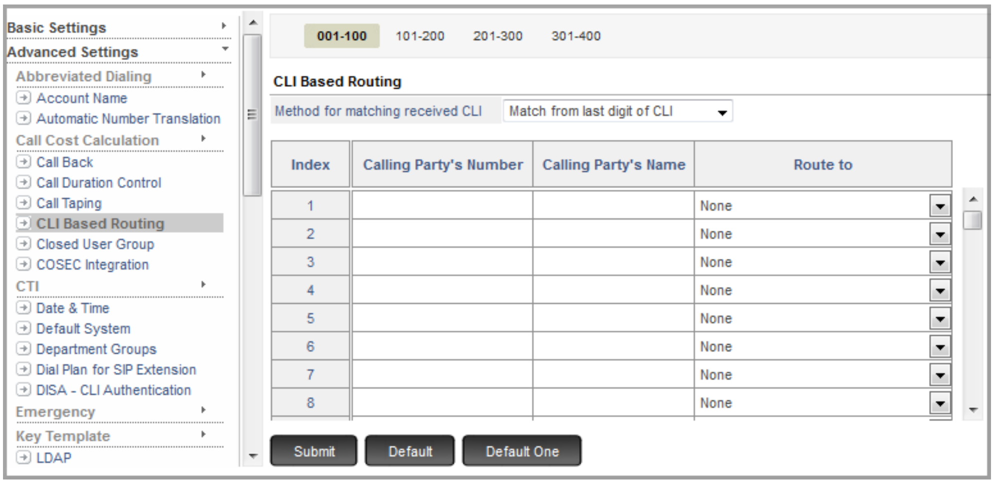
Creating CLI Routing Table

- On a sheet of paper, create a 3-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 access codes of the destination.
- 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.

Index	Calling Party's Number	Calling Party's Name	Route to (Access Code of Destination Extension)
001	2640459	MidasBiz	21
002	022281110001	Jet Set	37
:			
010	2640075	Bacchus	SIP Extn.-2

To configure the **CLI Based Routing Table**,

- Login as System Engineer.
- Under **Advanced Settings**, 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.

- Each number is to be stored at a Location Index numbered from 001 to 400.

  There are 100 entries on each page. To go to the next 100 Index, click the numbered links at the top of the CLI table (101-200, 201-300, 301-400).

- Refer to the table you created on paper, and configure the following parameters.

- In **Calling Party's Number**, enter the number of the calling party, not exceeding 16 digits. You can also enter '+' in the number string.

- In **Calling Party's Name**, enter the name of the calling party. You can enter a maximum of 8 characters in this field.

- In **Route to**, select the landing destination extension number, which may be a SLT, SIP extension, Virtual extension, Department Group or Voice Mail Auto Attendant Profile.

- Click **Submit**.

To enable CLI Based Routing on the Trunks,

- See **Incoming Call Routing** under "CO Trunks", "Mobile Trunks" and "SIP Trunks" for instructions.

Closed User Group (CUG)

A Closed User Group is a network of PBXs to provide seamless connectivity. The PBXs connected in the network behave as a single unit. Extension users of one PBX can reach the extension users of the other PBX without dialing any access code, as if they were dialing extension numbers of their own PBX.

You can connect SARVAM UCS and other PBXs over Analog trunks as well as over the IP Network.

How it works

Let us understand this application with the help of this illustration:

- Say, SARVAM UCS is installed at Location A and is networked with PBX A at Location B.
- To create a **CUG over Analog Trunks**,
  - the CO port (FXO) of SARVAM UCS is connected to the SLT (FXS) port of PBX A.
  - the CO port (FXO) of PBX A is connected to a SLT(FXS) port of SARVAM UCS.
- To create a **CUG over IP Network**, both SARVAM UCS and PBX A may be connected directly to the Public network, in Peer-to-Peer mode.
- The Extension Numbers of SARVAM UCS and PBX A must be unique.
- Closed User Group (CUG) must be enabled in the Class of Service assigned to the extension users of SARVAM UCS.
- Seamless connectivity can be provided to the extension users at both locations with suitable configuration of SARVAM UCS and PBX A.
- **At Location A**, you need to configure the *Closed User Group Table* in SARVAM UCS.

  The CUG Table stores up to 32 entries. You must configure the following parameters in the table:

  - **Route Code:** The Route Code is the truncated number string for the extension numbers of the other PBX. The truncated number string must start with the starting digit of the extension numbers. In case, all extension numbers of PBX A start with ‘3’, then you can configure the Route Code as a single digit, ‘3’.
  - **Strip Digit Count:** Keep the Strip Digit Count as ‘0’.
  - **Self Route:** Keep the Self Route check box disabled.
  - **Dialed Digit Count**: This is the digit length of the extension numbers of PBX A[130]. Configure the Dialed Digit Count as ‘3’.
  - **LCR:** This parameter is not relevant for this application.
  - **Route using Trunks**: This is the number of the Tie Trunk that the system should use for routing the dialed number string that matches with the Route Code and Dialed Digit Count.

130. *When digits are dialed on a trunk, the system waits for the Inter Digit Wait Timer after the last digit is dialed. The Dialed Digit Count helps avoid the delay caused by the Inter Digit Wait Timer. When you configure the digit length, the system will match the dialed number string with the digit length you have configured and will route the number without waiting for the Inter Digit Wait Timer, if a match is found.*

It the CUG is connected over **Analog Trunks**, you must select the CO Trunk connected to the SLT (FXS) port of PBX A.

If the CUG is connected **over IP network**, you must select the SIP Trunk to be used for routing the calls.

- **Apply Toll Control**: When Self Route check box is disabled, system will check this 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. See “Toll Control” for more details.

- **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 may also generate a report according to the Call Cost. For details, see “Station Message Detail Recording–Report”.

  The CUG Table you configure on SARVAM UCS would look like this:

Route Index	Route Code	Strip Digit Count	Self Route	Dialed Digit Count	LCR	Route using Trunks	Apply Toll Control	Apply Call Cost
1	3	0	Disabled	3	Off	CO2	Disabled	Disabled
2								
:								
32								

- **At Location B,** you may do a suitable configuration of the PBX. If the PBX at Location B were SARVAM UCS, the CUG Table you configure would look like this:

Route Index	Route Code	Strip Digit Count	Self Route	Dialed Digit Count	LCR	Route using Trunks	Apply Toll Control	Apply Call Cost
1	2	0	Disabled	3	Off	CO2	Disabled	Disabled
2								
:								
32								

- With the necessary configuration done at both locations, when an extension user 201 of SARVAM UCS at Location A dials extension 301 of PBX A, the system 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 Tie Trunk defined for routing the Route Code, i.e. CO2.

*If the CUG is connected **over IP network**, you must also configure the SIP Trunk as Peer-to-Peer trunk and configure the Peer-to-Peer Table. See “Peer-to-Peer Calling” for instructions.*

How to configure

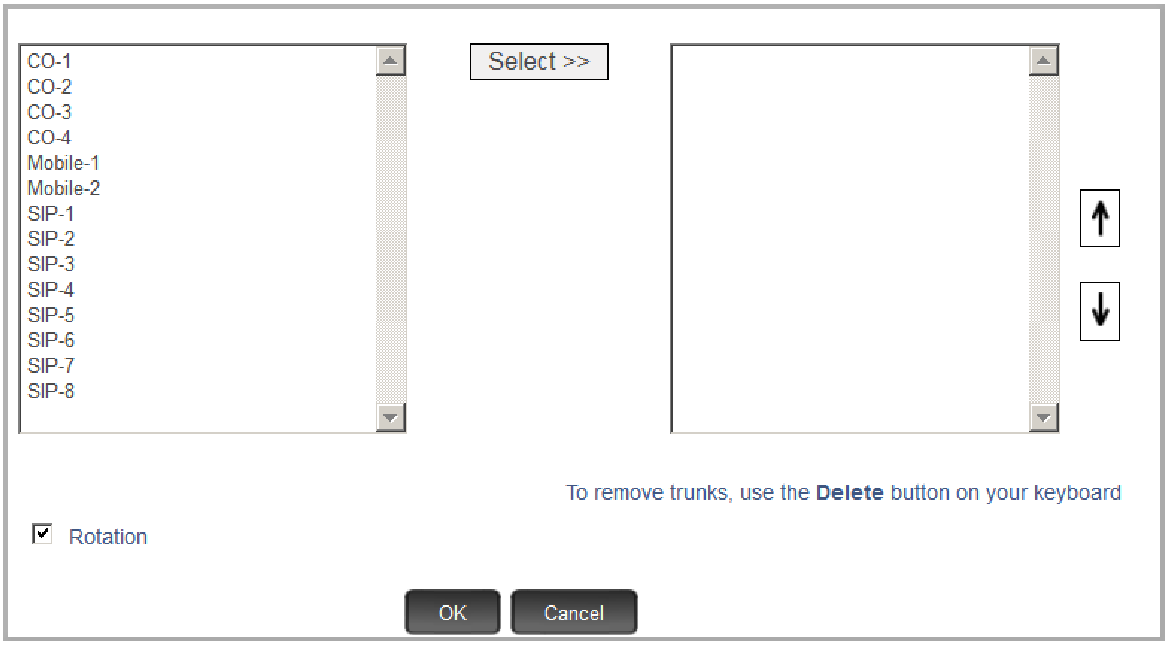
To configure Closed User Group,

- Login as System Engineer.
- Under **Advanced Settings**, click **Closed User Group**.
- In **Route Code**, enter the extension numbers of the other PBX in the CUG. Instead of entire number strings, you can configure a single digit, the starting digit of the extension numbers as Route Code. A maximum of 16 characters can be configured as Route Code. Default: Blank.
- Keep the **Strip Digit Count** as ‘**0**’. Default: 0
- Clear the **Self Route** check box. Default: Disabled.
- In the **Dialed Digit Count** field, enter the digit length of the extension numbers of the other PBX in the CUG. The digit length may be up to 99. Default: 99.
- Keep the **LCR** option as OFF. Default: OFF.

  To know more about LCR, see “Least Cost Routing (LCR)”.

When there are trunks connected directly to the network and you want to allow extension users to access these trunks without dialing the Trunk Access Codes for these trunks, enable LCR to select the lowest cost trunk from these trunks.

- Double click the **Route using Trunks** field. A multiple selection box opens.



  - On the left, the trunks appear with their names (if configured in “Trunks”) and port numbers in a sequence, starting with CO trunks, followed by Mobile trunks and SIP trunks.

    If you have not assigned any names to the trunks, they will appear with their default names (CO, MOB, SIP) and port numbers.

    If you have enabled On-Site Configuration, only those trunks that are connected will appear in the box.

- To select a trunk, place your cursor on the desired trunk, and click **Select>>**.
  Or
- Press the **ctrl** key and click the left mouse button to select multiple trunks.

- You may change the sequence of the trunks you selected, if required, using the **Up** and **Down** arrow buttons on the right display box.

- To de-select a trunk, place your cursor on the desired trunk and press the **Delete** key on your keyboard.

- You may enable **Rotation**, if you have selected more than one trunk. Default: Disabled.

  When you enable Rotation, each new outgoing call is routed through the subsequent trunk in the group[131]. This ensures equal distribution of outgoing call traffic on all trunks.

  When Rotation is OFF, calls are routed through the first trunk in the group. If this trunk is busy, the call is routed to the next trunk in the group.

  Rotation has no relevance if only one member trunk is selected.

- Click **OK**. The multiple selection box closes.

- All the trunks appear in the **Route using Trunks** field, in the sequence you selected, separated by commas.

- Clear the **Apply Toll Control** check box, if you do not want to apply toll control to the CUG numbers dialed by you. By default, this check box is enabled and therefore the system will apply toll control to all the outgoing calls.

- **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**.

131. *The first call through the first trunk, the second through the second, the third through the third trunk, and so forth. Thus each new call is routed through the trunk next to the one that routed the previous outgoing call.*

Conference-3 Party

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 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

SARVAM UCS supports two simultaneous 3-Party Conferences.

If a call put on Consultation Hold is to be included in a Conference, you must first retrieve it.

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 B'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[132] 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.

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132. The total Beep Time = Beep Cadence + Wait Timer. The Wait Timer(non-programmable) is fixed as 2 seconds by the system.

- *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 Extension creating 3 party conference, disconnects during Conference** in the System Parameters.*
- *The Conference can be broken only by the 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 Hang-up.*

How to configure

For this feature to work, the 'Conference' feature 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 **Play beeps when Assistant present in 3-Party Conference** check box. See “System Parameters”, for instructions.
- Enable **Play beeps when Assistant leaves the Conference** check box. See “System Parameters”, for instructions.
- Configure the desired interval time in **Conference – Assistant Present Beep Interval (sec)**. See “System Timers and Counts”.
- Make sure you select **Station Type** as **Assistant** for the Operator Extension. For instructions, see See “SLT Extensions” and “SIP Extensions” under “Basic Settings”.

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

By default, Conference is enabled in the CoS assigned to all the extensions of SARVAM UCS. So, all extensions of SARVAM UCS can make conference calls.

If you want to deny 3-Party Conference to selected extensions, follow these steps:

1. Click **SLT Extensions** and select the particular SLT extension.
2. Click the **Class of Service** of this SLT extension.
3. Clear the **Conference** check box to disable/deny 3-Party Conference on that SLT extension.

   Similarly, you can deny 3-Party Conference to SIP Extensions.

Refer **Class of Service** under “SLT Extensions” and “SIP Extensions” for detailed instructions.

The 'Conference' feature 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 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

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 and the Multi-party Conference feature code.

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, i.e. 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[133] will be played:
- when the operator is present in the conference at regular intervals
- when the operator has left the conference

Extension users can initiate multiparty conference from a remote location using “Direct Inward System Access (DISA)”.

SARVAM UCS supports 6 participants in a Multi-party Conference. The system supports 2 simultaneous Multi-party Conferences.

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

Initiating a Multiparty Conference

- A has initiated a 3-party conference with B and C.
- 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).

133. The total Beep Time = Beep Cadence + Wait Timer. The Wait Timer(non-programmable) is fixed as 2 seconds by the system.

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 Extended IP Phone users can remove another participant from the Multiparty Conference. The 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 configurable, 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.

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 extension. 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 Conference/ Dial-in Conference begins**. See "System Parameters", for instructions.

For confidential conferences, you must:

- Enable **Play beeps when Assistant present in Multiparty Conference** check box. See "System Parameters", for instructions.

- Enable **Play beeps when Assistant leaves the Conference** check box. See "System Parameters", for instructions.

- Configure the desired interval time in **Conference – Assistant Present Beep Interval (sec)**. See "System Timers and Counts".

- Make sure you select **Station Type** as **Assistant** for the Operator Extension. For instructions, see "SLT Extensions" and "SIP Extensions" under "Basic Settings".

- 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 extension users are to be allowed to initiate or join the Conference from a remote location, "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. See **Phone Key Settings** under "Configuring Matrix SPARSH VP248", "Configuring Matrix SPARSH VP310" and "Configuring Matrix SPARSH VP330", "Configuring Matrix SPARSH VP210" and "Configuring Matrix SPARSH VP510".

How to use

For 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 music on-hold.
- 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.

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.

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.

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, see the instructions "Dial-In Conference using DISA" under "Conference Dial-In".

Conference Dial-In

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[134] will be played:

- when the operator is present in the conference at regular intervals
- when the operator has left the conference

SARVAM UCS supports only two simultaneous Dial-In conferences. One Dial-In conference can have maximum 6 participants. However, if two Dial-In Conference is to be conducted simultaneously, each conference can have upto 3 participants.

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 either **1** or **2**.

  - 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 informed only to the participants.

- All the other participants must be informed about the conference.

  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 and C about the conference and provides them conference number, for example, '1' and the password, '4040'.

134. *The total Beep Time = Beep Cadence + Wait Timer. The Wait Timer (non-programmable) is fixed as 2 seconds by the system.*

- *If C wants to schedule a conference, C must log into his extension from DISA mode.*
- *The conference password cannot be* ***1111****.*
- *The Conference Number must correspond with the number of simultaneous Dial-In Conferences supported by SARVAM UCS, in this case: 1 and 2. If a user dials a conference number other than this, system will play an Error Tone.*

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 conference number and the password, for instance: '1' and '4040'. In this case, A initiates the conference.

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 conference number and password.

  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 Extended IP Phone users can remove another participant from the Dial-In conference. The 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. By default, this feature is enabled on all extensions, so all extensions can use this feature. See "SLT Extensions" and "SIP Extensions" under "Basic Settings".

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 Conference/ Dial-in Conference begins**. See "System Parameters", for instructions.

For confidential conferences,

- Enable **Play beeps when Assistant present in Multiparty Conference** check box. See "System Parameters", for instructions.
- Enable **Play beeps when Assistant leaves the Conference** check box. See "System Parameters", for instructions.
- Configure the desired interval time in **Conference – Assistant Present Beep Interval (sec)**. See "System Timers and Counts".
- Make sure you select **Station Type** as **Assistant** for the Operator Extension. For instructions, see See "SLT Extensions" and "SIP Extensions" under "Basic Settings".
- 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 configure a DSS key for Dial-In Conference, Terminating a Conference, Temporarily Leave/Rejoining a Conference, if required.
- You can configure a DSS key for Dial-In Conference, Terminating a Conference, Temporarily Leave/ Rejoining a Conference, if required. See **Phone Key Settings** under "Configuring Matrix SPARSH VP248", "Configuring Matrix SPARSH VP310", "Configuring Matrix SPARSH VP510", "Configuring Matrix SPARSH VP210" and "Configuring Matrix SPARSH VP330".

How to use

For 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 ‘Include Party in Conf’ option 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 Conference/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.

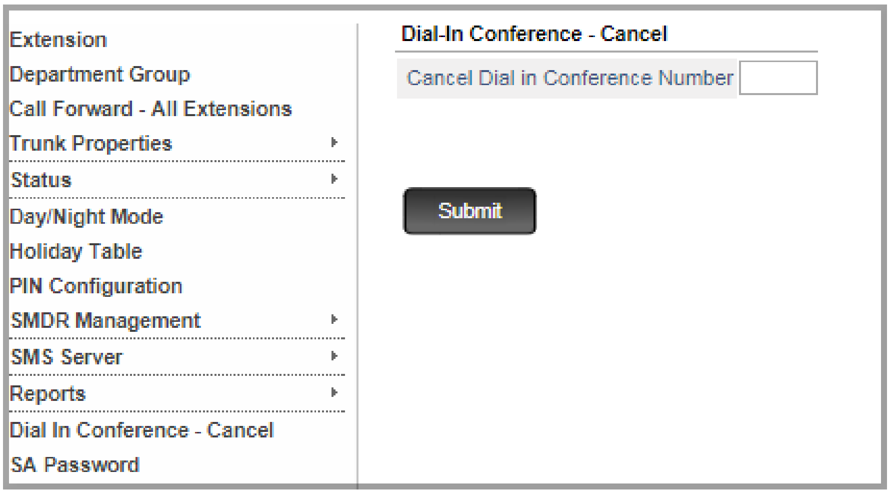
To cancel a Dial-In Conference from the System Administrator (SA) Mode:

- Enter SA mode from a 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,

- Login as System Administrator.

- Click **Dial-In Conference - Cancel**.



- Enter the conference number (1 or 2) which you want to cancel in the **Cancel Dial-In Conference Number**.

- Click **Submit**.

For SLT Users

The instructions are the same as for Extended IP Phone users, except you will not get any confirmatory or prompts as text messages on your phone.

- Lift the handset.
- You get dial tone.

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 Flash-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 VMAA”. After joining the conference users can Temporary Leave, Rejoin or Permanently Leave the Conference also.

Conflict Dialing

You may recall that “Access Codes” are dialed at different call phases. No two Access Codes must be 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, 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 Extension Codes (Extension Numbers). One Feature Access Code can be a part of (subset) another code, e.g. 4, 41, 412, etc.; extension numbers 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. 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, the system will 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.

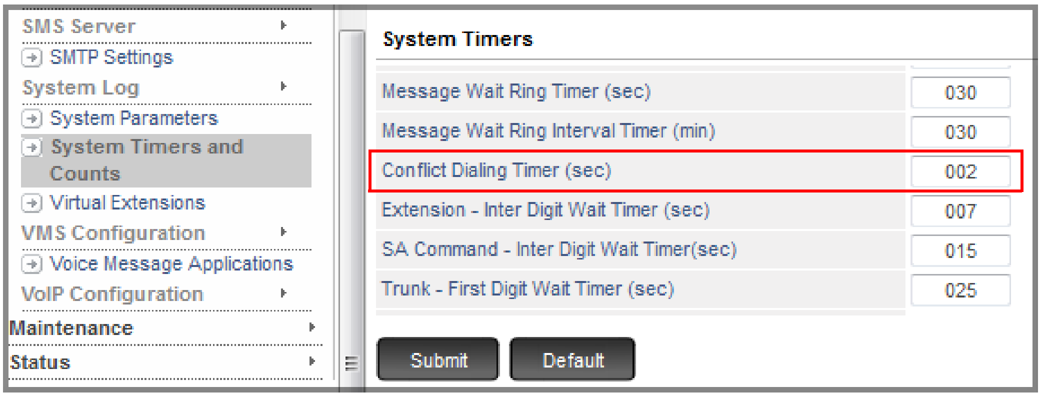
If the duration of the Conflict Dialing Timer is long, it may cause delay in the system's response to the feature. If the duration is less, the system may misinterpret the access codes. Ensure that the value of the Timer is programmed optimally (i.e. at least the default value).

To change the Conflict Dialing Wait Timer,

- Login as System Engineer.

- Under **Advanced Settings**, click **System Timers and Counts**.

- Scroll to **Other Features**. Set **Conflict Dialing Timer** to the desired value.



- Click **Submit.**

Conversation Recording

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.

For this feature to work, the Voice Mail System must be present in the system and extensions must have a mailbox assigned to them for recording conversations.

- *Use this feature in accordance with the local privacy laws.*
- *Matrix Comsec is not responsible for any misuse/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 on 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.

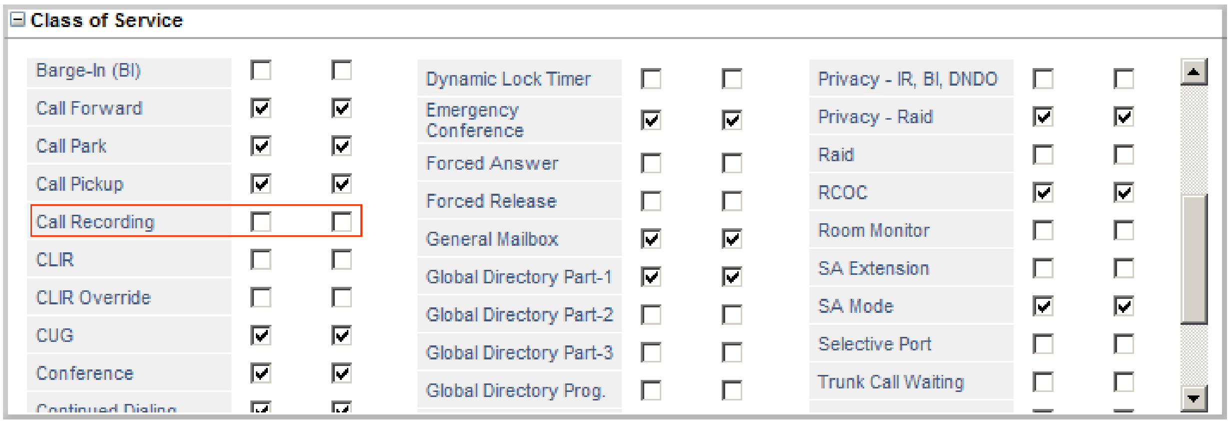
- A calls C.
- C answers the call.
- A dials the command for Conversation Recording in mid-speech.
- C is put on Consultation Hold.
- The system sends a string of digits to the Voice Mail System to initiate Conversation Recording.
- A and C are in speech again.
- The conversation recording starts in 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.

How to configure

To provide this feature to extensions,

- Conversation Recording must be enabled in the “Class of Service (CoS)” of the extensions to which this feature is to be allowed. By default, none of the extensions have this feature in their CoS. Decide which extensions should be allowed Conversation Recording and enable the **Call Recording** feature in their CoS.



For instructions on configuring Conversation Recording related parameters on different extension types, see “SLT Extensions” and “SIP Extensions” under *Basic Settings*.

- Extensions that are to be allowed Conversation Recording must also have a mailbox. By default all extensions are assigned a mailbox in their **Voice Mail Settings**.

If required, you may disable the Beeps played when Conversation Recording starts. Decide whether you want Beeps to be played and accordingly enable/disable the **Play Beep when Raid/Call Taping/ Conversation Recording starts** in the “System Parameters” under *Advanced Settings.*

How to use

For 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 390[135]

- Follow Voice Mail Prompts.
- Go Idle.

135. This is the default Voice mail Feature Access Code. If this has been changed, use the new code.

Conversations are recorded as New Messages. So, follow the voice mail prompts to listen 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 390
- Follow Voice Mail Prompts.
- Replace handset.

COSEC Integration

SARVAM UCS supports integration with COSEC to support Matrix COSEC Door Controllers. These door controllers can be used in VIP Apartments and Villas, wherein ETERNITY NENX is installed.

With this integration the users — SLT, 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.

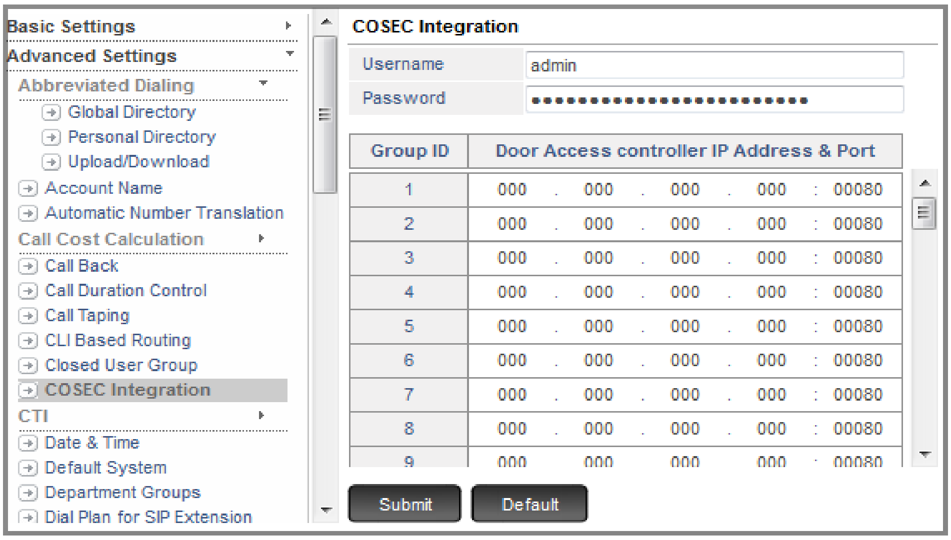
How it works

- 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.*

How to configure

- Login as System Engineer.
- Under **Advanced Settings**, click **COSEC Integration**.



- 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 "SLT Extensions".

To assign Group IDs to SIP users, see parameter COSEC Door Group in "SIP Extensions".

To assign a DSS key to COSEC Door Open, refer the topic **Phone Key Settings** under "Configuring Matrix SPARSH VP248", "Configuring Matrix SPARSH VP310", "Configuring Matrix SPARSH VP330", "Configuring Matrix SPARSH VP210" and "Configuring Matrix SPARSH VP510".

Customer Name

Customer Name is the name of the organization/enterprise that has deployed SARVAM UCS. As the User, you can program the name of your company/organization in the system.

When Customer Name is programmed 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, Alarm Status reports, etc.

The Customer Name may consist of a maximum of 80 alphanumeric characters, including punctuation marks. So, you can add a contact address to the Customer Name.

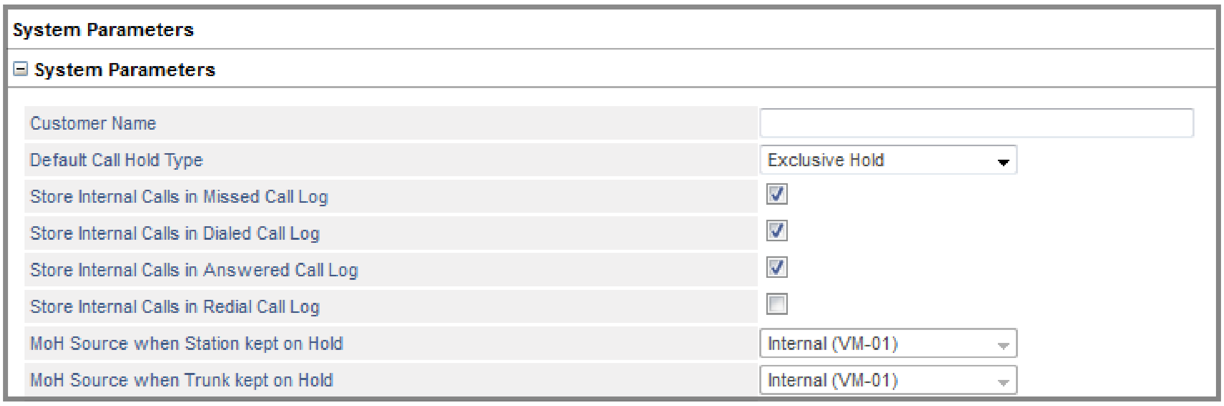
How to configure

Customer Name can be programmed at the time of installation on the “Pre-requisites” page under Basic Settings, or any time thereafter on the “System Parameters” page under Advanced Settings.

You can also change or correct the Customer Name you have assigned any time.

If you have not assigned the customer Name to the system,

- Login as System Engineer.
- Under **Advanced Settings**, click **System Parameters**
- Click **System Parameters** to expand.



- Under General Parameters, enter the name (and address, if desired) of the organization/enterprise in **Customer Name**. For example: Prudent Investment, 701 Sunshine Boulevard, Bangalore.
- Click **Submit**.

Day Night Mode

Certain features of the SARVAM UCS like Operator, Class of Service, Toll Control, Built-In Auto Attendant, Direct Inward System Access (DISA) etc, require extensions and trunks to behave differently according to the Day Time (Working hours), Break hours and Night Time (Non-working hours), which are referred to as Time Zones.

These Time Zone dependent features and facilities are operated automatically according to the Time Tables configured in the system. In a Time Table, the Time Zones — Day Time (Working Hours), Break Hours and Night Time (Non-Working Hours) are defined for the entire week. Time Table is assigned to trunks, extensions and other time zone dependent 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 of 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/Break 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 dependent 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.

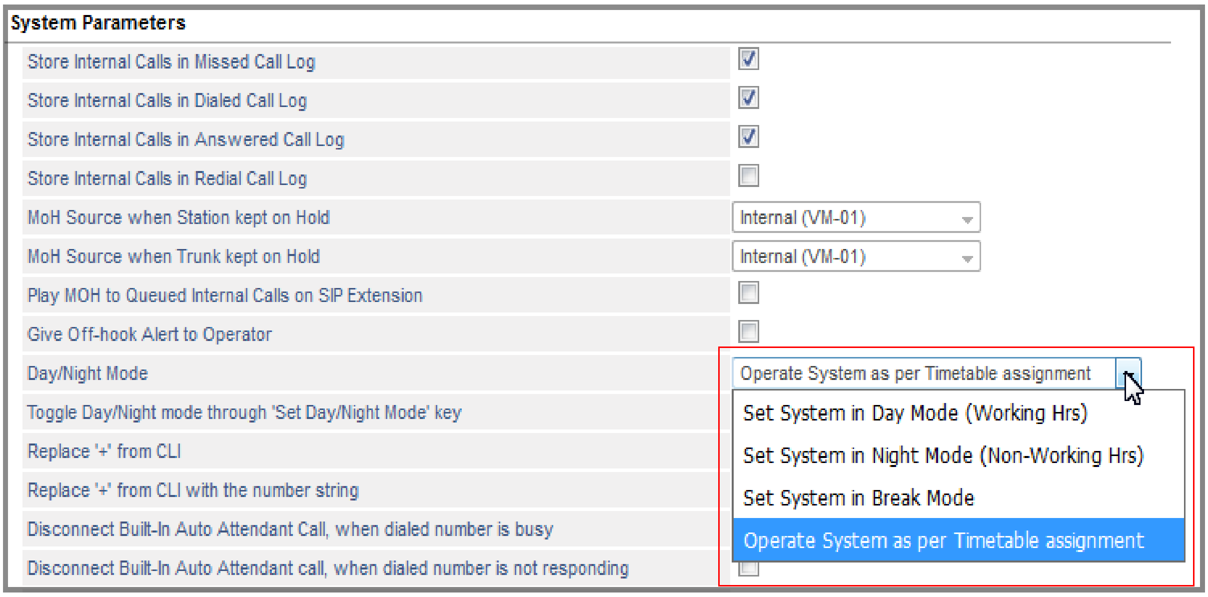
Thus, Day Night Mode forces the system to work in a particular Time Zone, until it is changed again, manually.

How to Configure

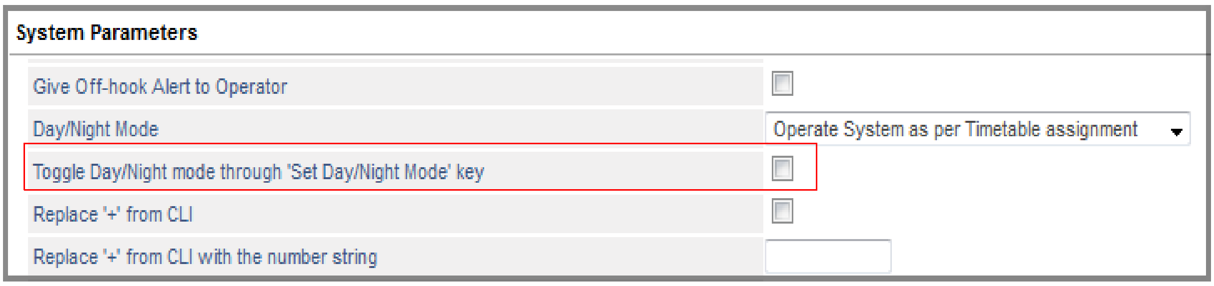
Day Night Mode can be set by the System Engineer (SE Mode) as well as the System Administrator (SA Mode).It can be done using Jeeves or by dialing a command from a Telephone.

Setting Day Night Mode from SE Mode using Jeeves

- Login as System Engineer.
- Under **Advanced Settings**, click **System Parameters**.



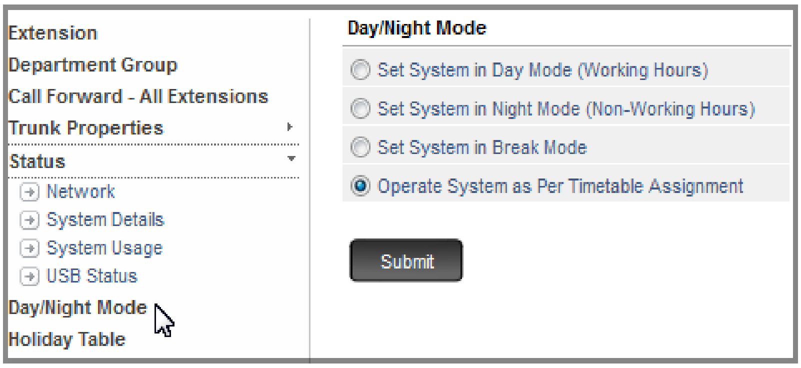
- Click **System Parameters** to expand.
- Go to **Day Night Mode** and select the desired option:
  - Set System in Day Mode (Working Hrs)
  - Set System in Night Mode (Non-Working Hrs)
  - Set System in Break Mode
  - Operate System as per Timetable assignment (default).



- Select the check box for **Toggle Day/Night mode through 'Set Day/Night Mode' key** to switch to Day Mode (Working Hrs) or Night Mode (Non-Working Hrs) on pressing the DSS Key. By default, this is disabled.
- Click **Submit.**

Setting Day Night Mode from SA Mode using Jeeves

- Login as System Administrator.
- Click **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 Mode
  - Operate System as per Timetable assignment (default).
- Click **Submit**.

Setting Day Night Mode from SA mode using a Telephone

- Enter SA Mode from a SLT/Extended IP Phone.
  - Dial ***1072-018-Code***
    Where,
    Code is from 1 to 4
    1 is for Day Mode
    2 is for Night Mode
    3 is for Break Mode
    4 is for Operate system as per Time Table.
- Exit SA Mode.

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, see **Phone Key Settings** under "Configuring Matrix SPARSH VP248", "Configuring Matrix SPARSH VP310", "Configuring Matrix SPARSH VP330" and "Configuring Matrix SPARSH VP510".

- Select the check box **Toggle Day/Night mode through 'Set Day/Night Mode Key'** to enable the toggle functionality for the DSS 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 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 an Extended 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
SPARSH VP248/SPARSH VP310/ SPARSH VP510	Day Mode Set	Blue	Continuous ON
	Night Mode Set	Red	Continuous ON
	Break Hours Mode Set	Violet	Continuous ON
	System set to work as per Time Table	--	OFF

If you are setting Day Night Mode from an Extended 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 on pressing Day Night Mode DSS Key	Color after toggle	Cadence
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

Daylight Saving Time (DST)

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 (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 SARVAM UCS is advanced and set backward automatically according to the DST convention of the country/region where 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 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
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

DST Type	DST Timings		Applicable in Countries
	Start Time	End Time	
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 in 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 SARVAM UCS is advanced and set backward manually according to the DST convention of the country/region where 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, i.e. 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:

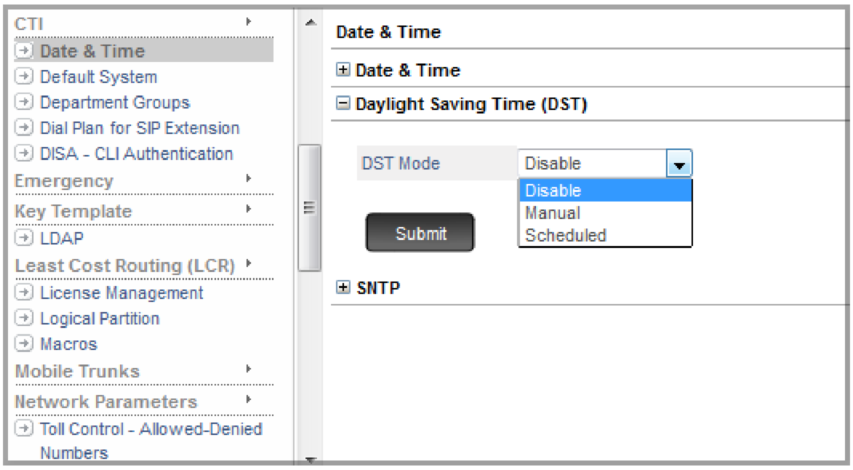
1. The 'Date and Month' method, which specifies a date of the month that DST will start or end. For example: starting on March 11 and ending on November 4.

2. 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.

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

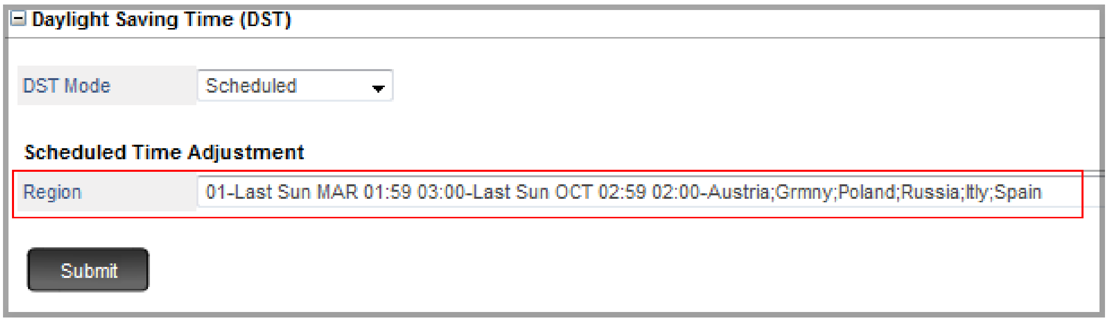
- Login as System Engineer.
- Under **Advanced Settings**, click **Date and Time**.
- Click **Daylight Savings Time** to expand.



- Set **DST Mode** to **Manual** or **Scheduled** as per your requirement.

Scheduled DST Adjustment

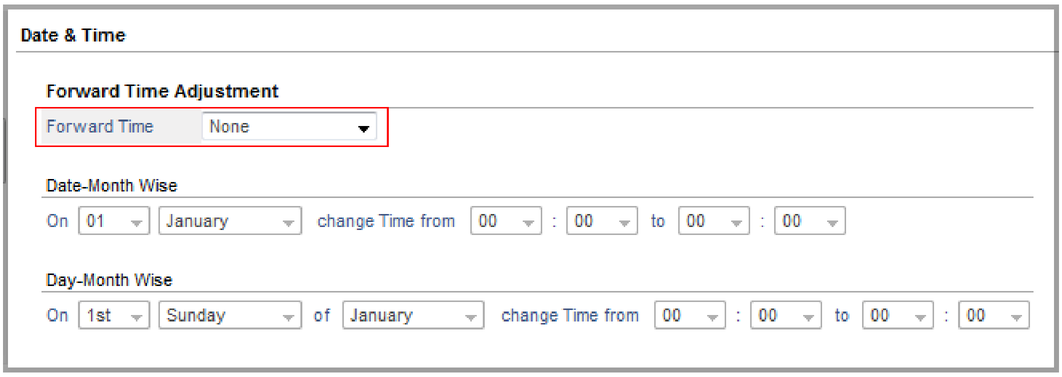
- If you have selected **Scheduled** DST mode, in the **Region**, select the country/region where your system is installed.



- Click **Submit** 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:



  - **Date-Month Wise** to specify the date of the month DST will start.

    OR

  - **Day-Month Wise** to specify the day of the month DST will start.

If you select Day-Month Wise option, the Date-Month Wise option will be disabled, and vice versa.

Date-Month Wise

- If you select the **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.

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, i.e. 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.

- Now, go to the option 'Backward Time Adjustment' to set the time back (i.e. end DST and begin standard time).

- Follow the same steps described above 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** 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, i.e. 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, e.g. 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.*

Department Call

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 an 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 5 Department groups. The *member* extensions of a department group may be Single Line Telephones (SLT), SIP extensions, Virtual Extensions.

The **Department Call** feature is included in the **Basic Features** in the CoS of all extensions of SARVAM UCS. So, all extensions can make Department calls. You cannot deny this feature to any extension, without denying the entire set of Basic Features.

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 391.

Internal Calls

- Extension E dials 391 to call the Department.
- The system checks E's Class of Service for the Department Call feature.
- The feature is enabled. Then the system checks if Rotation is enabled for the extensions in the Department.
- If the Rotation checkbox 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.

External Calls

Department Calls can be made using Built-In Auto Attendant. For example, a company may use 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 391 using Built-In Auto Attendant.
- The system checks if Rotation is enabled in the group of extensions in the Department.
- As the Rotation check box 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 check box 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 391.
- The same process as described above will be repeated.
- But the system will land the call on extension C first, because Rotation check box is enabled for this group of extensions.
- 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 department extension group. The extensions will ring for the duration of the Ring Timer, either continuously or one-by-one (as per the Continuous check box 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.

Voice Mail

A Department Group can be assigned a common mailbox for Voice Mail, called the *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 391 further,

- Extensions A, B, C and D are all members of Department Group 1 with the Access Code 391.

- 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.[136]

- To the first extension that answers the notification call, for example, Extension A, the Voice Mail System informs about the new messages 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."*

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136. This will depend on the type of Message Wait Indication configured for the extension in its Voice Mail Settings.

- 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*' (30 seconds). If none of the member extensions answers the call before the expiry of this timer, the call is forwarded to the destination.

  You may set the *Call Forward No Reply Timer for Department* to the desired value. This Timer is 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 check box 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 Direct Inward Dialing or Direct Inward System Access) dials the Access Code 391 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.
- select member extensions for the Department groups.
- assign appropriate access codes to the Department groups.
- assign a name to the department group.
- enable *Department Call (Basic Features)* in the Class of Service of the extensions that are to be allowed to make Department calls.

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, for instance: 3 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	SIP Extension
1	391		201, 202, 203	321
2	392		204, 205, 206	322, 323
:				
5	395		211, 212, 213, 214	336

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.

Enabling Department Call in the Class of Service

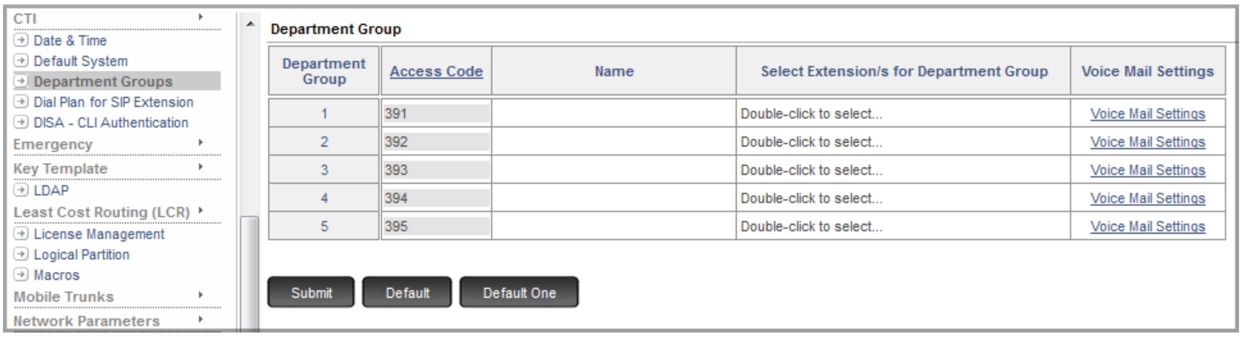
The *Department Call* feature is included in the *Basic Features* in the CoS for Day and Night/Break, of all extensions of SARVAM UCS. So, all extensions can make Department calls. You cannot deny this feature to any extension, without denying the entire set of Basic Features.

By default, *Department Group Call Forward* is enabled for day and night/break time in the CoS of all extensions of SARVAM UCS. 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 in the CoS group of member extensions, and disable this feature in the CoS group of all other extensions. For instructions see "SLT Extensions" and "SIP Extensions" under *Basic Settings*. Also see the topic "Class of Service (CoS)".

Configuring Department Groups using Jeeves

- Login as System Engineer.
- Under **Advanced Settings**, click **Department Groups**.



Creating Department Groups

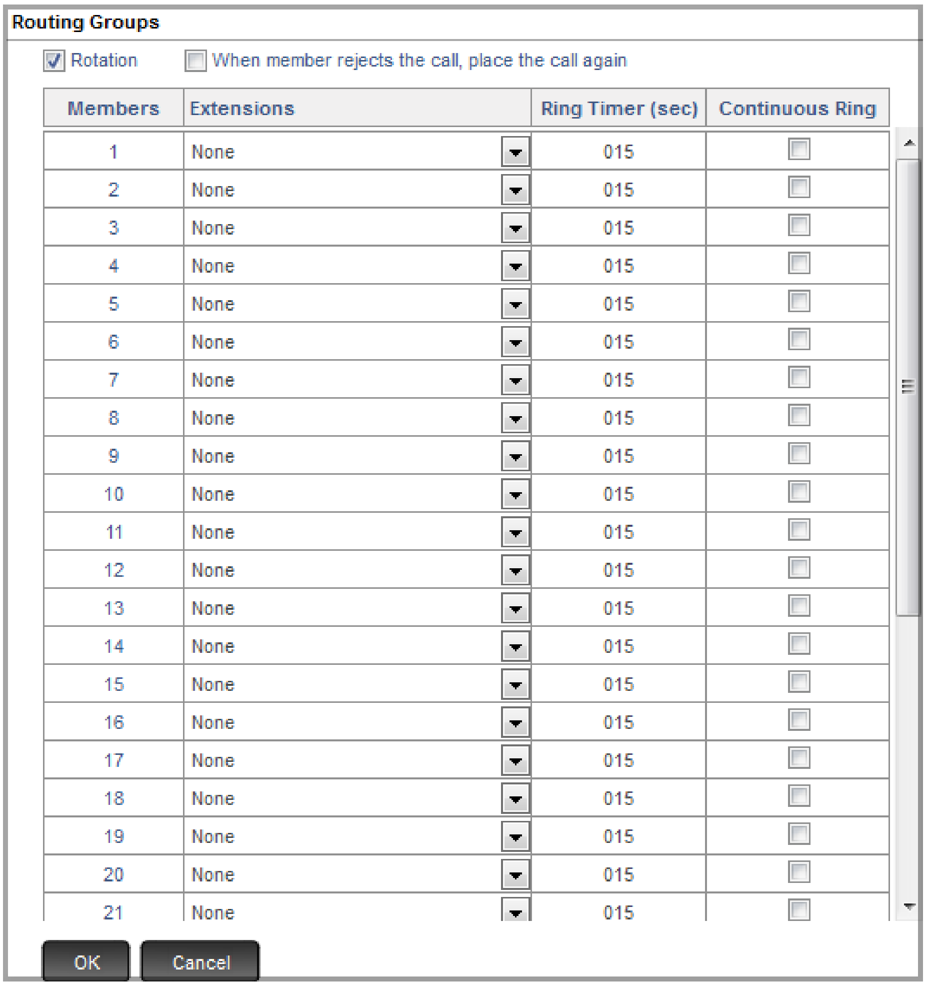
- To create a Department group, assign an **Access Code** for the department group against the Index Number.

  By default, the Access Codes assigned to Department groups are from 391 to 395.

  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 or SIP access code or any feature access code of the Dial Phase. Refer the topic "Access Codes" to know more.

- 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.

- Now, double click the **Select Extension/s for Department Group** field.



The Routing Groups window opens. Select the extensions you want to include in this Department group from the list box.

- Configure **Ring Timer (sec)**. This timer defines the time for which the extension, on which the call lands, should ring. Default: 15 seconds.

- Select the **Continuous Ring** check box, if you want the 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 of no relevance, if you have selected only one member extension in the department group.

- Select the **Rotation** check box, if you want to enable rotation of calls in the department group having 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 group.

  Rotation has no relevance if the department group has only one member extension.

- By default, **When member rejects the call, place the call again** is disabled. Therefore, if any SIP extension user rejects an incoming call, the system will not place the same call on this extension again while checking the routing group to land the call. You may enable this check box, if required.

If this check box is disabled and you have selected the Continuous Ring 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.

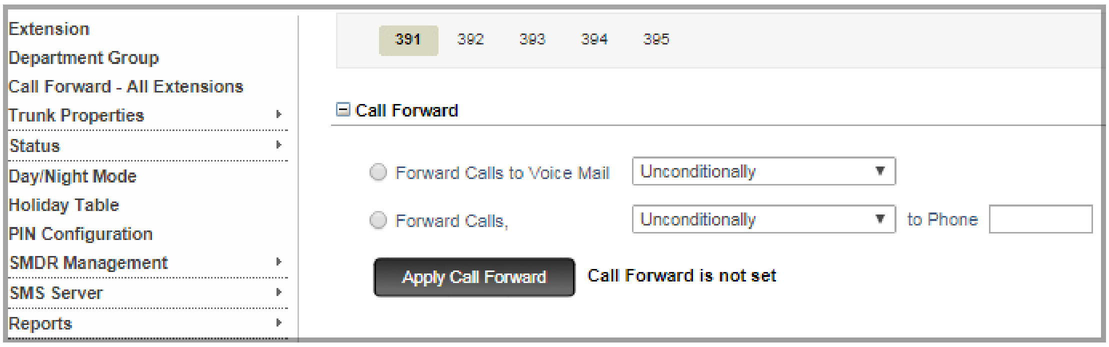
- In **Voice Mail Settings**, click the Voice Mail Settings link. The respective Extension Voice Mail Settings window will open. You may edit the parameters. For details, see “Extension Voice Mail Settings”.
- Repeat the same steps to configure another Department Group number.
- Click **Submit**.

Setting Call Forward for Department Group

You can set Call Forward for Department Group from SA Mode only.

To set Call Forward,

- Login as System Administrator.
- Click **Department Group**.



- 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.
  - 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 **Apply Call Forward**.

  The color of the text indicating that Call Forward is set will change to red.

To cancel Call Forward,

- Click **Cancel Call Forward**.

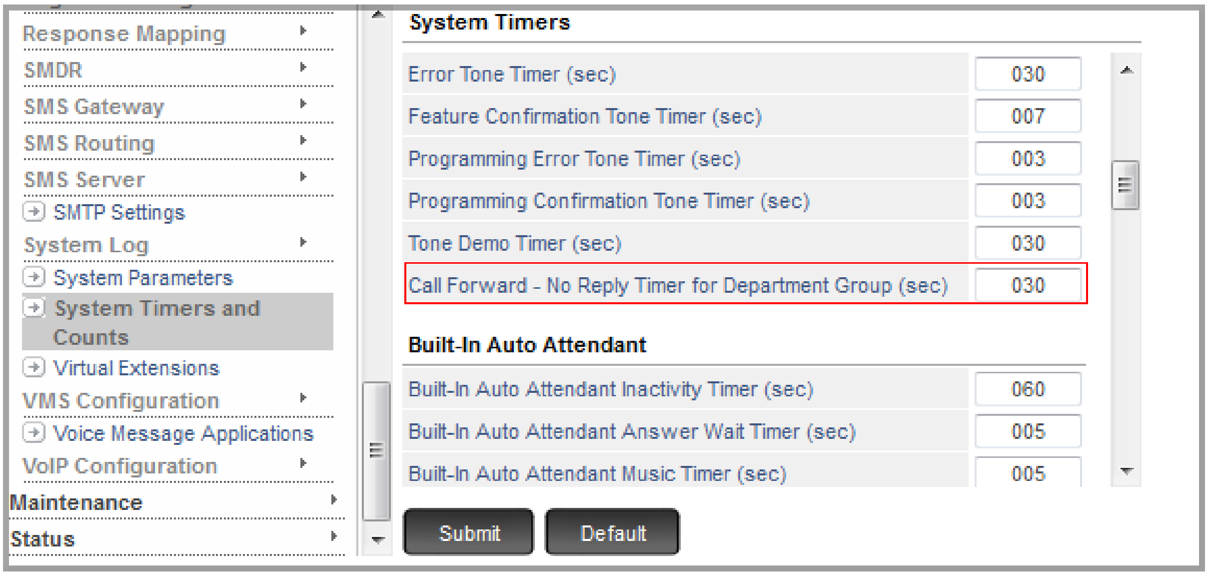
  The color of the text indicating that Call Forward is not set will change to black.

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,

- Login as System Engineer.
- Under **Advanced Settings**, click **System Timers and Counts**.



- Scroll to **Call Forward No-Reply Timer Department Group (sec)**.
- Set the timer to the desired value.
- Click **Submit**.

How to use

Making a Department Call

Making a department call is the same as calling another extension.

For Extended IP Phone Users

- Press DSS Key assigned to the desired Department Group.
  OR
- Dial the desired Department Group Number (default: 391-395).
- You get Ring Back Tone as the call lands on an extension within the department group.
- Talk when the call is answered.
- Go Idle.

For SLT Users

- Lift handset.
- Dial the desired Department Group Number (default: 391-395).
- 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 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 messages 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 Indication set on your extension phone.

When the LED Lamp of the Message Waiting Key glows,
OR
When your phone plays Message Wait Ring[137],

- Lift the handset.
- Dial 390.
- 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.

If you hear Stuttered Dial Tone/Voice Message when you go OFF-Hook,

- Dial 390.
- 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 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 Access Code of the Department Group whose calls are to be forwarded.

137. Refer “Distinctive Rings” for description.

- 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 Access Code of the Department Group 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
  - Dial 0 to Cancel Call Forward
- Dial Destination number (number of the desired extension or voice mail)
- You get confirmation tone.
- Replace handset.

Dial Plan for SIP Extension

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 routes the call.

How to configure

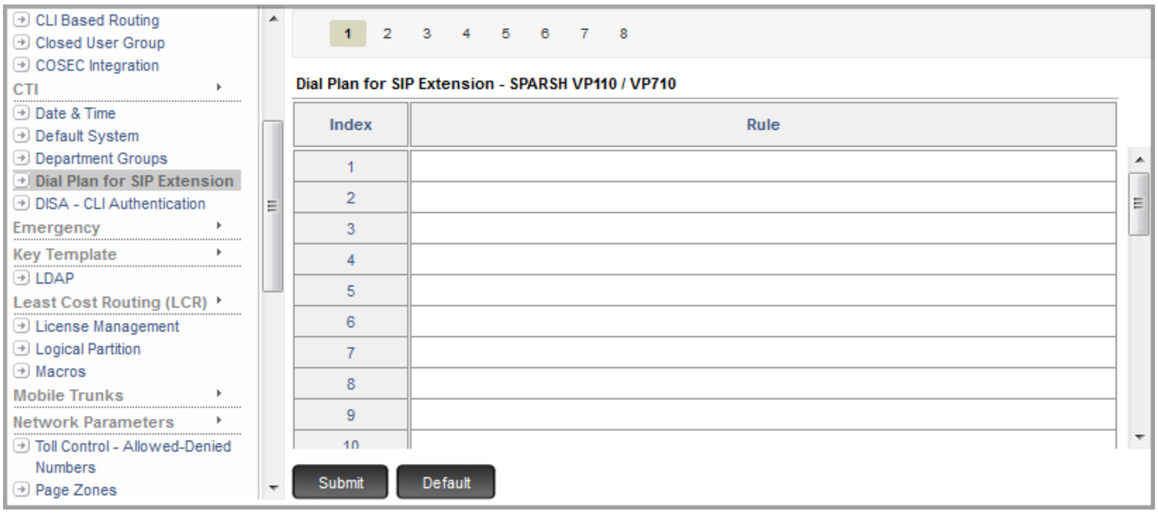
Configuring the Dial Plan involves the following steps:

- Selecting a Dial Plan and configuring the rules in the Dial Plan.
- Assigning the Dial Plan to the desired SIP Extensions.

Configuring Dial Plan

- Login as System Engineer.

- Under **Advanced Settings**, 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.

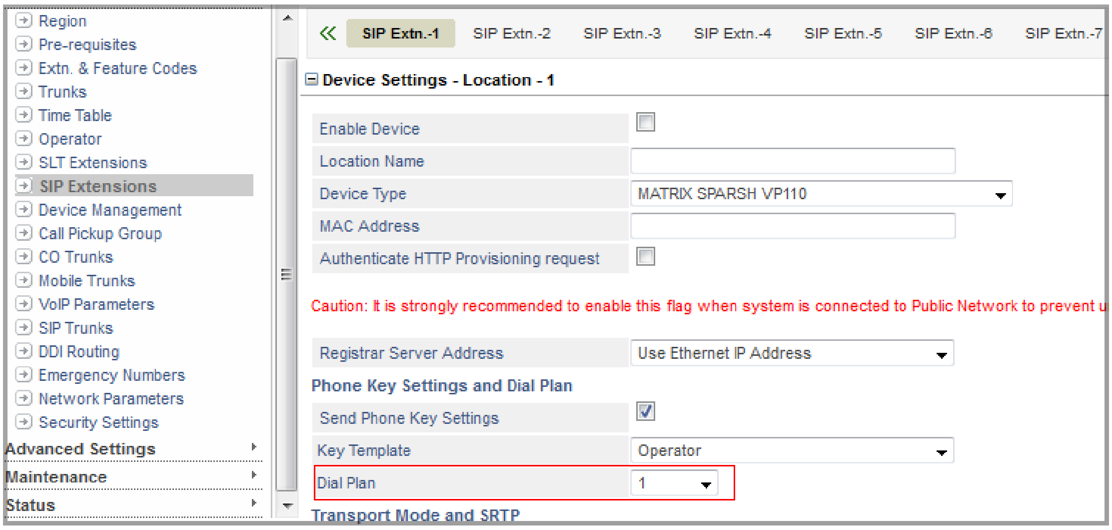
  For example, if you want that users should be able to dial extension numbers from 3000 to 3999 without any delay, configure the Rule as 3XXX.
  For more details to configure the rules, refer to the topic *Dial Plan* in the *SPARSH VP110 User Guide*.

- Click **Submit**.

Assigning Dial Plan to SIP Extensions

To assign the Dial Plan you configured for VP110,

- Under **Basic Settings**, click **SIP Extensions.**
- Click the tab of the Extension number at which you have registered VP110.
- Now, click the location at which you have registered SPARSH VP110, for example, **Device Settings - Location 1**.
- Scroll to **Phone Key Settings and Dial Plan**.
- Select the **Dial Plan** number you configured.
- Click **Submit.**



To assign the Dial Plan you configured for VP710,

- Under **Basic Settings**, click **SIP Extensions.**
- Click the tab of the Extension number at which you have registered VP710 as a Standard SIP.
- Now, click the location at which you have registered SPARSH VP710, for example, **Device Settings - Location 1**.
- Scroll to **Dial Plan.**
- Select the **Dial Plan** number you configured.
- Click **Submit.**

Dial By Name

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 number.

This feature is accessible only to extension users of the Extended IP Phone.

With Dial By Name, extension users need not remember the desired party's telephone number or short codes, i.e. “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.
- **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. Upto 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/ department groups. Their names are assigned to SLT 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

- Press the DSS Key assigned to 'Dial By Name' feature.
- On SPARSH VP248, press the 'Names' key. On SPARSH VP310, press the ‘Contacts’ key.
- The prompt <Name:XXX> appears on the phone display.
- Enter the name of the desired party[138].
- For example, you wants to call Midas Biz, enter 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 your extension and the Extension Names configured in the system.
- Scroll the list using the Up/Down navigation keys to reach the desired contact's name.
  OR
  Instead of scrolling the entire list, you can enter 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.
- Select the desired name by pressing 'Enter' Key.
- The system dials out the number stored under the selected name. The number is displayed on the your phone.

138. *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.*

How to configure

For this feature to work, the following must be configured:

- **DSS Key:** A direct station selection (DSS) key must be configured for the Dial by Name feature. Without the DSS Key this feature will not be accessible.

  SPARSH VP248 has DSS Key labeled as 'Names' whereas SPARSH VP310 has the Fixed Function Key labeled as ‘Contacts’.

- **Global Directory:** The names of the external parties must be configured against their respective telephone numbers in the directory. Refer the topic “Abbreviated Dialing” for instructions on configuring Global Directories.

- **Personal Directory:** The names of the external parties must be configured against their respective numbers in the Personal Directory. Refer the topic “Abbreviated Dialing” for instructions on configuring Personal Directories.

  The Personal Directory must be assigned to the Extensions.

- **Extension Names:** Extensions may be SLTs and SIP extensions. Refer “Extension and Feature Codes” under *Basic Settings* for instructions on assigning names to extensions.

- **Class of Service:** Dial By Name is allowed to all Extended IP Phone users. However, the use of this feature is related to the following features, which must be enabled in the Class of Service of the 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 directory parts in the CoS of the SIP extensions. See “Class of Service (CoS)” for instructions.

The system will display the names exactly as they have been configured in the Personal and Global Directories.

How to use

For 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 'Names' key again.
- Enter the name/initial letters of the contact's name.

Dialed Number Directory

Dialed Number Directory is an Extended IP Phone feature.

It is the list of numbers dialed out from the Extended IP 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 Extended IP 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 an Extended IP Phone extension user makes an outgoing external 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 Extended IP 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 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[139].

How to configure

No specific programming required.

How to use

For Extended IP Phone Users

- Press DSS Key assigned to Redial function.
  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.

139. Recall that the system does not check for Toll Control when Abbreviated Dialing is used.

- 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.

Digest Authentication

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 authorized 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 Password of the trusted/authorized calling party SIP devices are stored. 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, i.e. 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

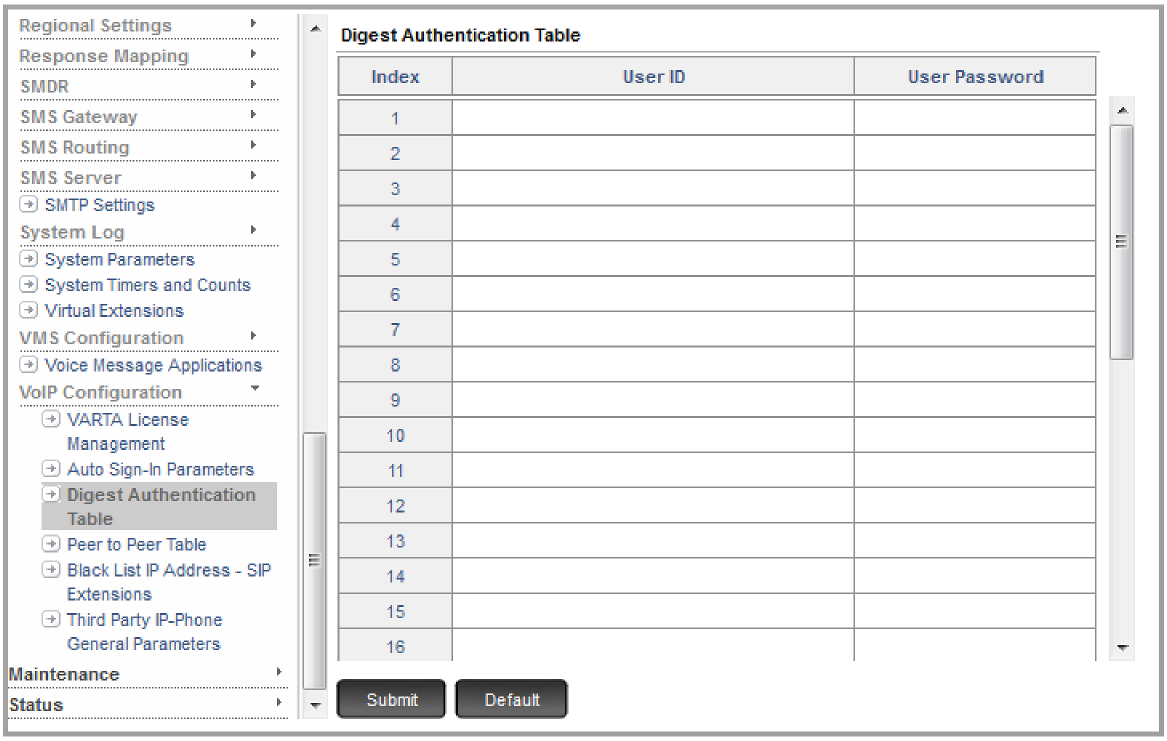
To use this feature,

- make a list of devices whose incoming calls (SIP requests), you want to allow after authentication.
- enable Digest Authentication on the desired SIP Trunks. For instructions, see “Trusted IP Address/es” under “SIP Trunks”.
- configure the Digest Authentication Table.

To configure Digest Authentication Table,

- Login as System Engineer.
- Under **Advanced Settings**, click **VoIP Configuration**.

- Click **Digest Authentication Table**.



- In **User ID**, enter the User ID to be authenticated. The User ID must be within 40 characters.
- In **Password**, 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**.

Direct Inward System Access (DISA)

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 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 consisting of:

- the DISA Feature Access Code.
- the extension number they want to access.
- the User Password of that extension.

The 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 as long as they are logged into 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 the caller is authenticated on the basis of CLI, the system plays the (internal system) Dial Tone.

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 as long as they are logged in to the DISA mode.

To end the DISA login session, caller 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 dialing '0'. 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 SARVAM UCS from the remote location.

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 only to those users who need to access the system using DISA.

How it works

For this feature to work, you must enable the desired DISA variant on the desired trunks: CO, Mobile, SIP.

- A call lands on a DISA enabled Trunk.
- The system checks if a DISA variant is enabled on the trunk for the current time, that is Day, Break or Night.
- 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[140].
  - 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[141].
  - 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.

140. If no voice message is recorded, the system plays music-on-hold to the caller.
141. If the caller does not go Off-hook within this timer, the system releases the call.

- If the caller dials an external number using a CO trunk, the system starts the *DISA Inactivity Timer* (configurable; default: 2 minutes).
- 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.
- 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[142].
  - The caller gets logged into the Auto Login extension and gets the dial tone of SARVAM UCS.
  - The caller dials the desired number to make the call.
  - 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).
  - 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** 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[143].

142. *If no match is found for the CLI of the caller in the Table, the call will be routed as per the Incoming Call Route configured in SARVAM UCS.*

- The caller gets logged into the Auto Login extension and gets dial tone of the outgoing trunks selected for dialing '0'.

- If the caller dials an external number using a CO trunk, the system starts the *DISA Inactivity Timer* (configurable; 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 the variants of DISA, PIN Authentication and CLI Authentication, the caller can use all the features allowed in the "Class of Service (CoS)" of the extension the caller is logged into.*

- *DISA Inactivity Timer is not applicable for SIP and Mobile trunks.*

- *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 Built-In Auto Attendant logic, if Built-In Auto Attendant is enabled. If DISA and Built-In Auto Attendant both are disabled, the incoming call will be routed as per the Incoming Call Route logic. To know more, see "Auto Attendant".*

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"[144].

How to configure

To provide DISA to remote users you need to do the following configuration:

- Select the DISA variant for the Trunks in the **Route Incoming Calls** option on the trunk port parameters. See "CO Trunks", "Mobile Trunks", and "SIP Trunks" under *Basic Settings* for instructions.

- Enable DISA in the"Class of Service (CoS)" of the extensions which you want to allow callers to access using DISA. See "SLT Extensions" and "SIP Extensions" under *Basic Settings* for instructions on configuring Class of Service for the different Extension types.

143. *If no match is found for the CLI of the caller in the Table, the call will be routed as per the Incoming Call Route configured in SARVAM UCS.*
144. *This feature is supported in Firmware Version V10R10 and later.*

- Change the User Password of the DISA extensions, if you have selected *DISA PIN Authentication-Multiple Calls* on a trunk. The default User Password (1111) will not work. See "User Password" and "System Security" for 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* on a trunk, you must configure the **CLI Authentication Table**.

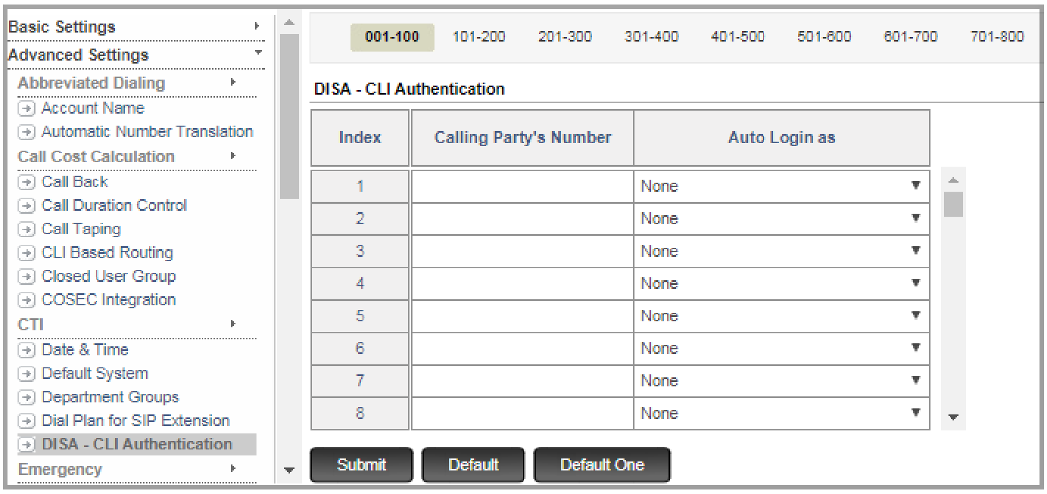
To configure the **DISA CLI Authentication Table**,

- 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 SARVAM UCS you want to allow this extension user to log in.

- Login as System Engineer.

- Under **Advanced Settings**, click **DISA - CLI Authentication.**



- 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 **Calling Party's Number**, 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 **Auto Login as**, select the extension you want to allow access to after the Calling Party Number is authenticated.

- Click **Submit.**

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: *DISA Idle State Timer* and *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, which it can interpret if these are received during DISA session.

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, i.e. Day, Break or Night.
- 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 **#1** to go Off-hook, **#0** to go On-hook, and 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 at the end of music/Built-In Auto Attendant Voice Message.
- 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** 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 (DSS532) is an add-on module with tri-color LEDs for 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.

You can attach up to four DSS532 to SPARSH VP510, connected with SARVAM UCS.

DSS532

A maximum of four DSS532 can be attached to SPARSH VP510. With each DSS532, 32 additional keys are at your disposal to be used as DSS keys. Similarly, if four DSS532 are attached, 128 additional keys can be used as DSS keys.

For instructions:

- to install the DSS532 with SPARSH VP510, see "Installing DSS532 with SPARSH VP510".
- to configure the DSS keys of the Console connected to a SPARSH VP510, see "DSS Key Settings" under "Configuring Matrix SPARSH VP510".

DSS Keys

You can assign Extension 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.
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 that extension blinks Blue to indicate ringing at that 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

Distinctive Rings are ringing patterns used for distinguishing between different types of call events.

SARVAM UCS supports the following types of call events:

Call Event	Ring Type set 1 (T1)	Ring Type set 2 (T2)	Ring Type set 3 (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
Alarm Call	Long Fast	Long Fast	Long Fast
Emergency	Long Fast	Long Fast	Long Fast
Operator Alarm	Long Fast	Long Fast	Long Fast
Message Wait	Short Fast	Short Fast	Short Fast
Programming Ring	Continuous	Continuous	Continuous
Ring Test	Short Slow	Short Slow	Short Slow
Priority	Triple	Triple	Triple
Call Supervision	Continuous	Continuous	Continuous
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 configurable.

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
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 configurable. You can change the Ring Text, if required.

How to configure

At the time of installation, when the System Engineer selects the “Region” (as per the geographical location of the system), 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.

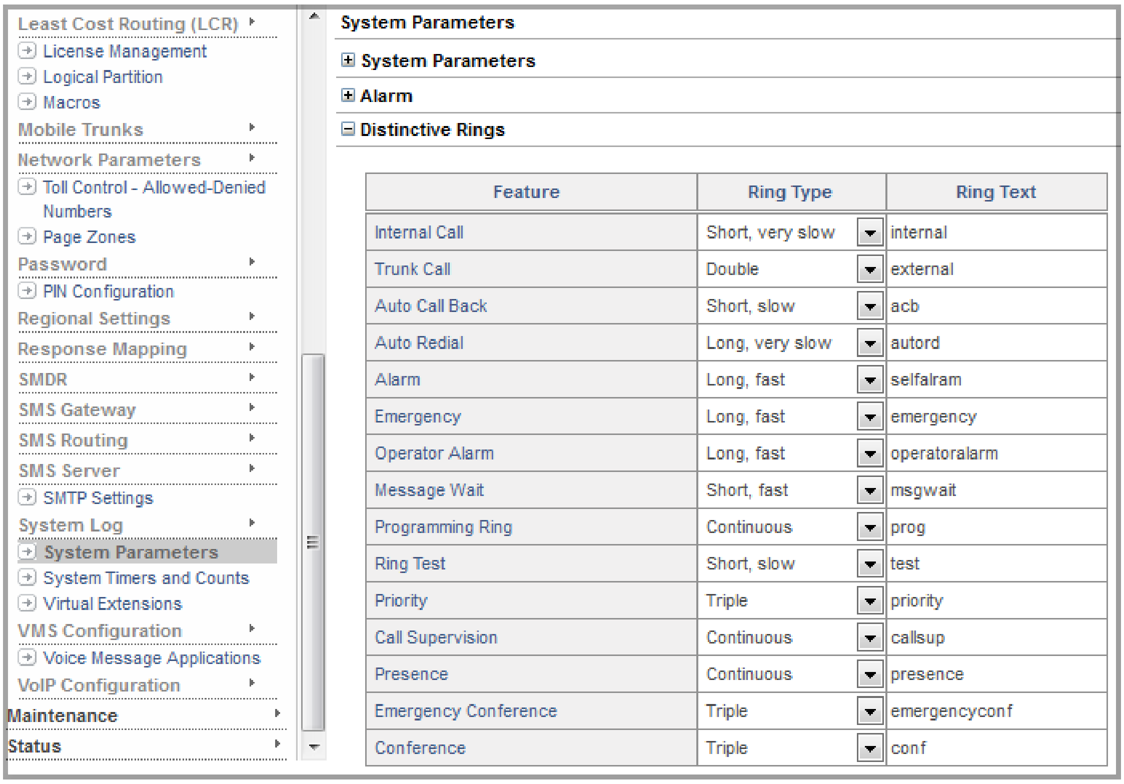
When you change the Ring Texts for the Ring Types in SARVAM UCS, you must configure the same Ring Text in the SIP phones.

Configuring Distinctive Rings using Jeeves

The country-specific Distinctive Ring Pattern is set automatically by the system when you select the Region Code. However, if required, you may change the default Ring Pattern loaded by the system.

To configure Distinctive Rings,

- Login as System Engineer.
- Under **Advanced Settings**, click **System Parameters**.
- Click **Distinctive Rings** to expand.



Feature	Ring Type	Ring Text
Internal Call	Short, very slow	internal
Trunk Call	Double	external
Auto Call Back	Short, slow	acb
Auto Redial	Long, very slow	autord
Alarm	Long, fast	selfalram
Emergency	Long, fast	emergency
Operator Alarm	Long, fast	operatoralarm
Message Wait	Short, fast	msgwait
Programming Ring	Continuous	prog
Ring Test	Short, slow	test
Priority	Triple	priority
Call Supervision	Continuous	callsup
Presence	Continuous	presence
Emergency Conference	Triple	emergencyconf
Conference	Triple	conf

- Select the desired Ring Pattern (Ring Type) for each call event/feature 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**.

Do Not Disturb (DND)

Extension users may restrict incoming calls landing on their extensions, in order to work without disturbance. 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. Intercept Destination can be configured only by the System Engineer. 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 PBX.
- 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 restricts calls to the phone on which DND is set. The 'DND-Override' feature breaks this restriction 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, if enabled. For instructions, see "SLT Extensions" and "SIP Extensions" under *Basic Settings*.
- 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, 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 an Extended IP Phone.

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 call that extension will be played an error tone. Callers who are using 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 Extended IP Phone, while C has a 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'[145].
- 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 a 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 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.

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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145. While DND and DND Text Message can be set from any phone, DND Text Message can be viewed on 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.

Configuring DND in Class of Service

- By default, all extension port types have **DND** enabled for the Day and Night/Break in their Class of Service. So, all extensions of SATRVAM UCS can set and cancel DND.

  DND-Override and **Privacy from DND-Override** are disabled in the CoS of all extensions by default. 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 only to selected extensions, disable this feature in the CoS of all other extensions other than these extensions.

  Similarly, if 'DND-Override' is to be to be allowed to the Operator and a few other extensions, enable this feature in the CoS of the Operator and other extensions.

  If 'Privacy from DND-Override' is to be allowed to certain extensions only, enable this feature in the CoS of these extensions.

  For instructions on configuring DND in the “Class of Service (CoS)” of the extensions which you want to allow this feature, see “SLT Extensions” and “SIP Extensions” under *Basic Settings*.

Configuring DND Text Messages

- By default, 9 DND Text Messages are supported 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

Message #	DND Message
6	On Vacation
7	On Business Trip
8	Out of Office
9	With a Guest

You can use these default message options or customize the DND Text Messages from 2 to 9, see "System Parameters" for instructions.

Configuring Stuttered Dial Tone

- If required, you may configure the system to play Stuttered Dial Tone on the extension that has set DND. For instructions, see "System Parameters".

Configuring Intercept Destination for DND

- You may set Intercept Destination for DND for each timezone. For instructions, see **DND Intercept Routing** under "SLT Extensions" and "SIP Extensions".

Configuring Voice Message for DND Notification

- You may also configure the Voice Message for DND Notification, as per user requirement.

  To be able to play a voice message to callers for DND notification, you must first record a Voice Module with the desired message.

  SARVAM UCS supports 16 Voice Modules of **16** seconds duration each.

  Record a Voice Module with the message "The dialed extension has activated Do Not Disturb" (recommended).

  Assign the Voice Module to the Voice Message Application defined for 'DND Notification'.

  See the topic "Voice Message Applications" for instructions.

How to use

DND set/canceled by Extension Users

For 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 the '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 the '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 the 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 the handset.

To select DND Message

- Lift the 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 the handset.

To cancel DND:

- Lift the handset.
- Dial 18-0
- Replace the handset.

DND-Remote

For 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 the '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 the '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 the '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 the 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 the 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 the handset.

DND-Override

For Extended IP Phone 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 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

DSS Call Pick-Up allows 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.

SARVAM UCS offers two types of DSS Call Pick-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 the user's Extended IP 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 the user's Extended IP Phone.

- *For SIP Trunks, DSS keys can be assigned for each Call Appearance.*
- *If you have assigned a DSS key to 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 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 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 the Class of Service of the desired extensions. For instructions, see *Class of Service* under "SIP Extensions".

- configure the DSS Keys for the desired extensions and trunks on their Extended IP Phone. See "Phone Key Settings" under "Configuring Matrix SPARSH VP248" and "Phone Key Settings" under "Configuring Matrix SPARSH VP310", "Phone Key Settings" under "Configuring Matrix SPARSH VP210" and "Phone Key Settings" under "Configuring Matrix SPARSH VP330" and "Phone Key Settings" in "Configuring Matrix SPARSH VP510".

How to use

For 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

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.

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'[146] is to be assigned and corresponding number strings to be allowed and number strings to be denied for each Call Privilege are to be configured.

- **Toll Control - Level 0** is Time Zone based, wherein the Call Privilege Type must be defined for the **Day** (working hours) and the **Night/Break** (non-working hours). For instance, you may define 'All Calls' as Call Privilege for the Day, and 'No Calls' as Call Privilege for the Night.

  By default, Call Privilege 'No Calls' is selected for the Day and Night/Break.

- **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 redefined for allowing all types of calls by selecting 'All Calls' as Call Privilege Type for **Calls allowed for Lock Level 3**. Similarly, you can configure Level 0 to allow Local Calls and configure the Local Number strings in the list 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 users change the Toll Control Level of the extension whenever they want, by dialing the feature access code.

    For example, an extension user having Toll Control Level 2 (configured as National calls) can restrict long distance dialing on the extension by setting the Toll Control Level to 1 (configured as Local calls) before leaving the workplace. On return, the user can restore the previous Toll Control Level, by setting it back to Level 2.

146. The Call Privilege types are: No Calls, Local Calls, Regional Calls, National Calls, Limited Calls, All Calls.

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, the user 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 extension user to dial the four-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 Calls allowed for Lock Levels 0 (allowed for Day and Night/Break), Level 1, Level 2, and Level 3 are configured in the Toll Control 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 the Toll Control Level assigned to the extension.
- 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, i.e. Day and Night/Break. 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 '*Dynamic Lock Timer'* feature in the Class of Service 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 '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, but the system will not check it 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 a separate configuration of Emergency Numbers, which remain unaffected by Dynamic Lock set on the phones. Refer the “Emergency Dialing” topic to know more about this feature.

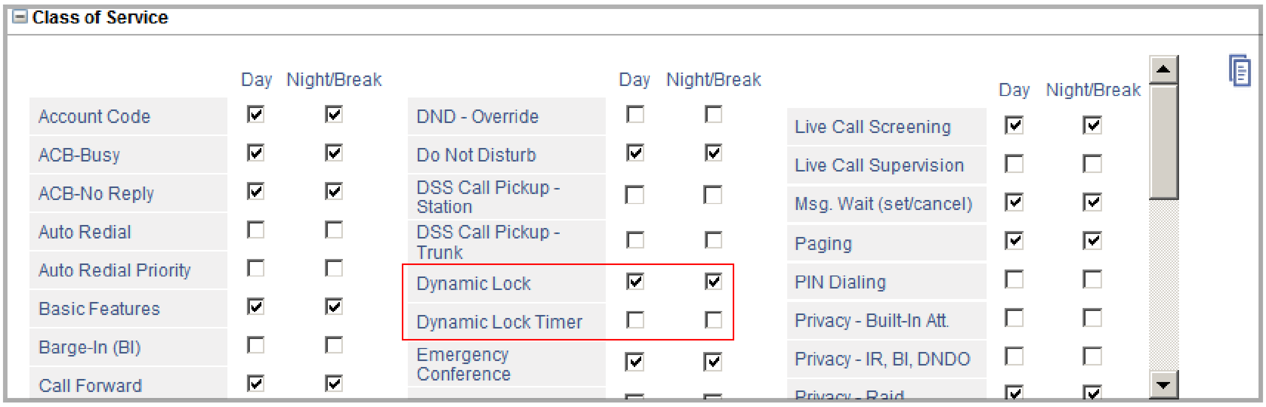
How to configure

For this feature to work, it must be enabled in the Class of Service of the extensions.

By default, Dynamic Lock enabled on all extension types during the Day and the Night/Break. So, all extensions of SARVAM UCS can set Dynamic Lock.

The 'Dynamic Lock Timer' is disabled in the Class of Service of all extension types.

To allow the extension to use Automatic Dynamic Lock, you must enable this timer in the Class of Service. See "Class of Service (CoS)" for instructions.



Toll Control Level must be configured on the extension.

To configure Toll Control Levels on the extension, refer the topic "Toll Control" for instructions.

How to use

Extension users can set Dynamic Lock—Manual and Automatic—by themselves or have it set by the Operator or any other extension user for their extension.

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.***

- *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 users must change the default User Password to be able to set the Dynamic Lock on their extension. Refer the topic "User Password" for instructions on changing the password.

Changing Dynamic Lock by Extension Users

For 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.

Change your User Password first, before you use this feature. 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.

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

For Extended IP Phone Users

To set Dynamic Lock-Manual

- Press DSS key assigned to Dynamic Lock.
  OR
- Dial 1072-002[147].
  OR
- Enter Phone Menu.
- 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[148].
  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[149].
  OR

147. *The feature 'SA Extension' must be enabled in the "Class of Service (CoS)" of the extension from which this code is being dialed. If the feature 'SA Mode' is enabled in the Class of Service of this extension, the extension user will be prompted for the SA password.*
148. *The feature 'SA Extension' must be enabled in the "Class of Service (CoS)" of the extension from which this code is being dialed. If the feature 'SA Mode' is enabled in the Class of Service of this extension, the extension user will be prompted for the SA password.*

- Enter Phone Menu.
- 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

- Press 'Enter' key.
- You get confirmation message and 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.

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149. *The feature 'SA Extension' must be enabled in the “Class of Service (CoS)” of the extension from which this code is being dialed. If the feature 'SA Mode' is enabled in the Class of Service of this extension, the extension user will be prompted for the SA password.*

Emergency Conference

Emergency Conference enables you to establish a Conference between a pre-defined group of extensions using a feature access code. Make sure, Emergency Conference is enabled in the Class of Service assigned to your extension.

This feature can be used to call and consult a group of people in emergency situations.

SARVAM UCS supports up to 6 parties in an Emergency conference. You can merge up to two Emergency Conferences with 3 members in each conference.

For more details, see "Conference-3 Party" and "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. However, for Emergency Conference you can make a Department Group with five members only.

  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, B and C as members. The Access Code assigned to the Department Group is 391. A 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 established among the members of Department Group 392. This Department Group has extensions X, Y and Z as members. X is the initiator of the conference as Emergency Conference is enabled in the Class of Service assigned to his/her extension.

Now, extension A wants to initiate an Emergency Conference.

This is how the feature will work:

Initiating an Emergency Conference

- A dials the feature access code for Emergency Conference, followed by access code of the Department Group (391).
- All extensions in the Department Group (extensions B and C) which are free will start ringing. The system will play *Emergency Conference* ring (default: Triple Ring) on the Extensions. Extension that is busy will not be included in the call.

  If there are 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.
- Extension B goes Off-Hook to answer the call first. B gets connected to the initiator of the conference, extension A.

- Two-way speech is established with extension A and B. C will continue to ring.
- When extension C goes Off-Hook to answer the call, A and B get a beep, and three-way speech is established between A, B, C.
- Whenever a new member joins the conference, all other extensions already in conference will get a beep, if the check box *Play Beep when Conference/Dial-In Conference Starts* is enabled in the "System Parameters".
- If the extension A, conference initiator disconnects, other two extensions in the conference will be connected and two way speech will be established between them.

Merging Emergency Conferences

Only two Emergency conferences, with 3 participants in each conference can be merged in SARVAM UCS.

- While in Conference, A presses the 'Conference' Key.
- From the Conference Menu, A selects Merge Conference to merge with another ongoing Emergency Conference.
- A selects 392 from the list of ongoing Emergency Conferences. Both the Emergency Conferences (391 and 392) are merged.

Canceling an Emergency Conference

- Only the initiator of the conference, extension A, 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 A 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 "SLT Extensions" and "SIP Extensions". 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 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 check box *Play Beep when Conference/Dial-In Conference Starts*. For instructions, see "System Parameters".

  This check box is common for other features like "Conference-Multiparty", "Conference Dial-In".
- 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

You can initiate an Emergency Conference also using "Direct Inward System Access (DISA)".

For 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.

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 the 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

When an emergency call is made from an extension, the system dials out the number using any of the free trunks selected for routing Emergency Numbers. Since the number is dialed out by the System, the Emergency Service that attends 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 extension 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 extension or the selected Emergency Reporting Group.

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

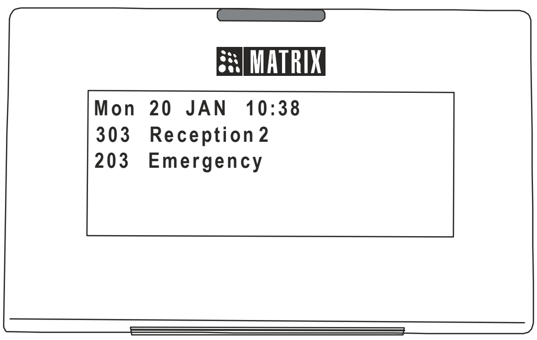
How it works

When an extension of SARVAM UCS makes an emergency call,

- The system hunts for the outgoing trunks selected for routing emergency numbers, and dials out the number from a free trunk.

- Simultaneously, the system informs the Emergency Reporting Group/Operator that an extension has dialed an emergency number, by ringing on the Emergency Reporting Group/Operator extension for the duration of the Emergency Reporting Call -Ring Timer (configurable; default: 10 minutes).

- If Emergency Reporting Group/Operator is 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 203.

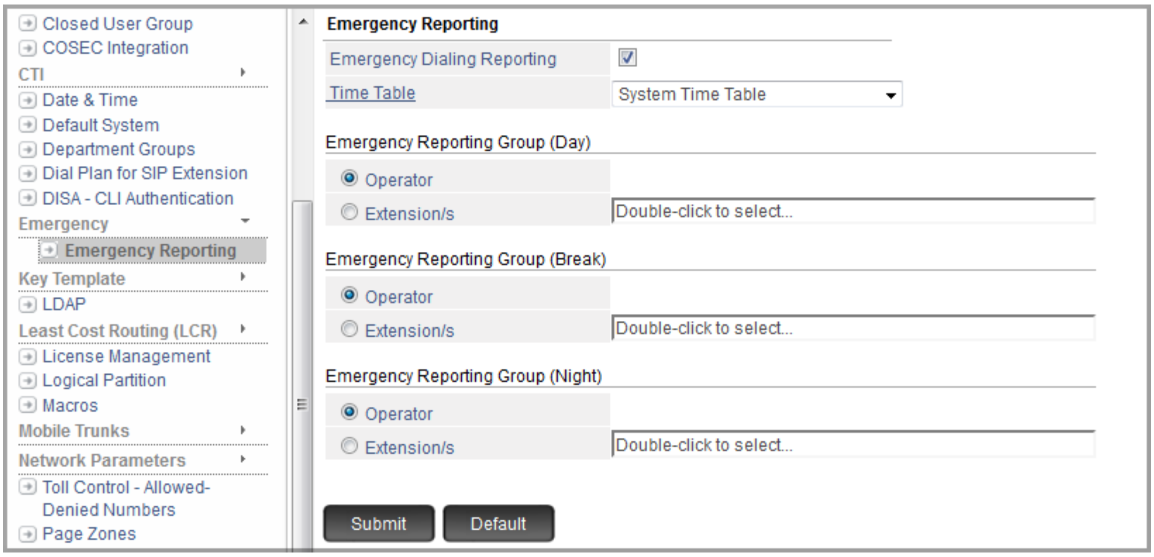
- To acknowledge the Emergency call, the Emergency Reporting Group/Operator must press the enter key. The acknowledged Emergency calls are logged into the System Activity Log.

- 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.

How to configure

To configure Emergency Reporting,

- Login as System Engineer.
- Under **Advanced Settings**, click **Emergency**.
- Click **Emergency Reporting**.



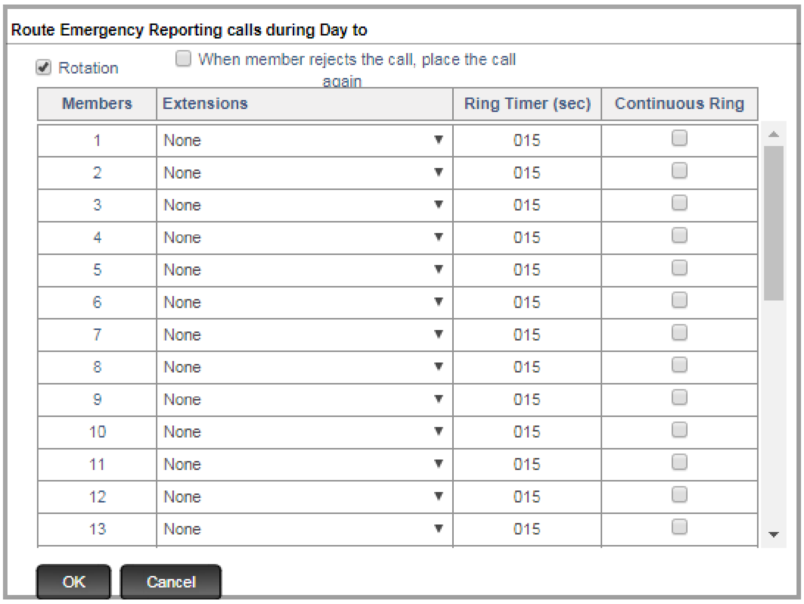
- **Emergency Dialing Reporting:** By default this check box is enabled. The system will detect the extension that makes an emergency call.
- **Time Table:** A Time Table is a schedule for the three Time Zones, namely Working hours and Break hours. You can define and select the Time Table as per you requirement.

  There are 4 different Time Table templates to select from. By default, System Time Table is selected. In System Time Table, six days of the week - Monday to Saturday -have working hours from 9:00-18:00 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 System Time Table OR customize and assign a different Time Table. Refer to “Time Tables” for more details.
- **Emergency Reporting Group:** You can either select **Operator** or can define an **Emergency Reporting Group** as the landing destination for placing the emergency reporting calls.

  By default, Operator is selected as the landing destination.

  To customize the Emergency Reporting Group (Day), double click the **Extension/s** field. The extensions to be selected in the routing group for the Day time appear in a new window.



- Select the **Extensions** for the Emergency Reporting Group. These may be a SLT, SIP extension, Virtual extension or Voice Mail Auto Attendant Menu.

- For each extension you have selected, set the **Ring Timer**. This timer defines the time for which the extension, on which the call lands, should ring. Default: 015 seconds.

- For each extension you have selected, you may set Continuous Ring, if you want the extension to ring till the incoming call is answered. Default: Disabled.

  When Continuous Ring is selected, the first extension in the Emergency Reporting Group group you have created will continue to ring, even as the system hunts for other extensions in the group to land the call. If the call still remains unanswered, the system will return the call to the first extension once again. This check box has no relevance, if there is only one extension in the Emergency Reporting Group.

- Enable **Rotation**, if you have selected more than one extension in the Emergency Reporting Group. Default: Enabled.

  When you enable Rotation, each new call lands on the subsequent extension[150] in the group next to the one that received the last call. This ensures equal distribution of incoming calls to all the destination extensions in the Emergency Reporting Group.

  Rotation has no relevance if the Emergency Reporting Group has only one member extension.

- By default, **When member rejects the call, place the call again** is disabled. Therefore, if any extension user rejects an incoming call, the system will not place the same call on this extension again while checking the routing group to land the call. You may enable this check box, if required.

  If this check box is disabled and you have selected the Continuous Ring 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.

150. The extension next to the one that received the previous call.

- Click **OK**.

  All the extension numbers you selected will appear in **Extension/s**. The extension numbers appear in the sequence you selected, separated by commas.

- 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. This log can be viewed by any 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 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.

  Also, see the “Emergency Dialing” and “Emergency Numbers” topics.

Emergency Dialing

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 trunks selected for the emergency number. See "Emergency Numbers".

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 Number will always be out dialed through the outgoing trunks you selected for dialing these 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, 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 SARVAM UCS fails.

How to configure

The Emergency numbers are fixed as per the Region where SARVAM UCS is installed. See the topic "Emergency Numbers" under *Basic Settings* for instructions on configuring Emergency Numbers.

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 “Operating VARTA AMP100”.
- MATRIX VARTA WIN200, UC Client for Windows. For detailed description, see “MATRIX VARTA WIN200”.

  To know the list of features 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
- Menu based operation of SARVAM UCS features
- Multiple Language support

SARVAM UCS Features

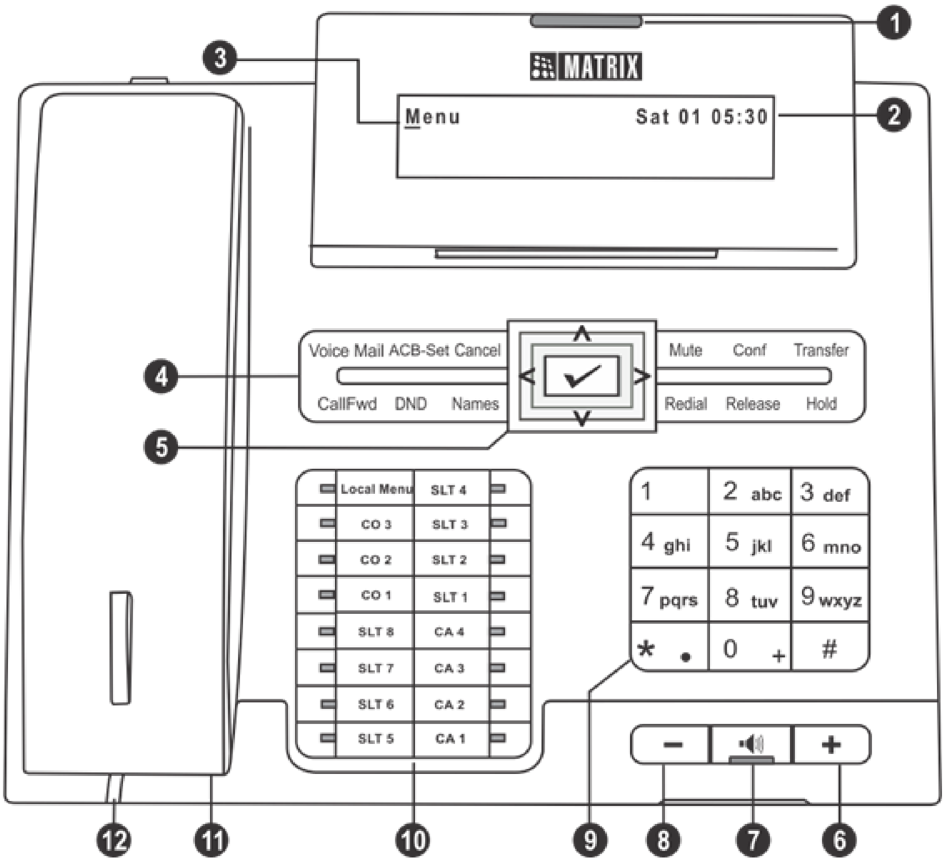
The SPARSH VP248 supports SARVAM UCS features, a few of these are listed below:

- 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.

∨ 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. For instructions on programming these keys, see “Phone Key Settings” under “Configuring Matrix SPARSH VP248”. 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.

For instructions on programming these keys, see “Phone 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.

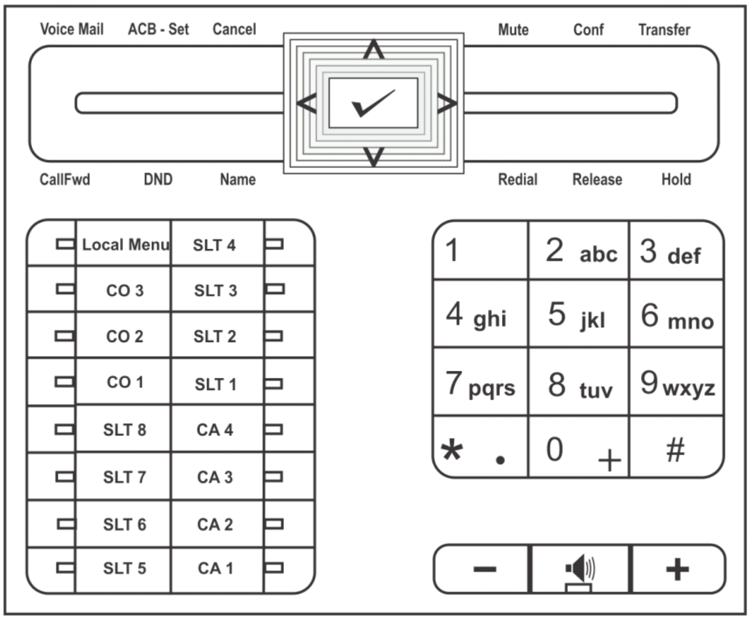
So you can use any stereo headset of standard make with a 2.5 mm single connector or a stereo headset with an RJ9 connector.

You can also configure any of the DSS keys to function as the Headset key. For instructions on programming the key, see “Phone 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 “Phone Key Settings” under “Configuring Matrix SPARSH VP248”.

Matrix Extended IP Phone, SPARSH VP248 Key Template (default)



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 the 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 “SIP Extensions” for instructions.

Connecting SPARSH VP248

For detailed instruction to connect SPARSH VP248, see “Connecting SPARSH VP248 as Extended SIP Extension” in “Connecting SIP Extensions”.

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.

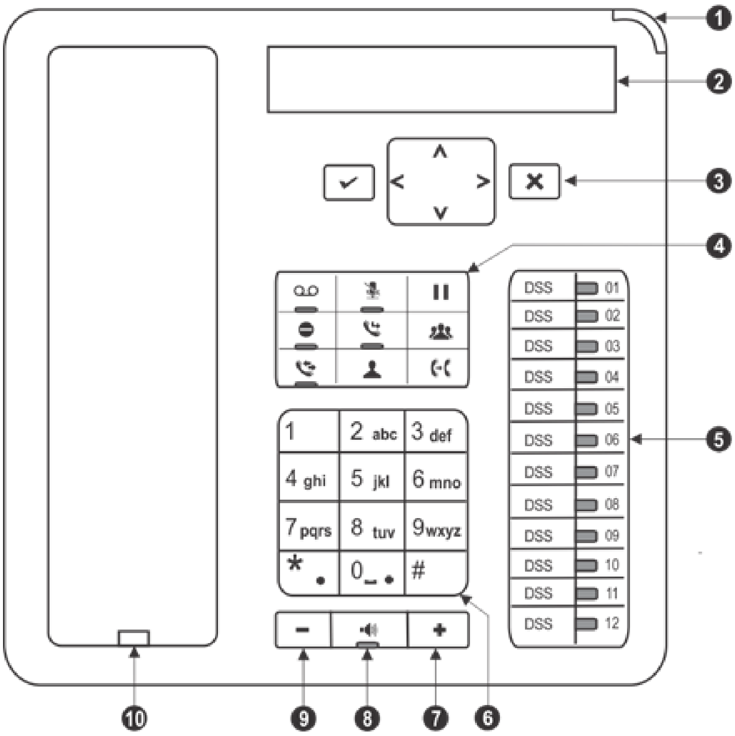
Key Features

- 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 is Off-Hook, Off-Hook Alert will be provided to SPARSH VP310 if it is the Operator phone.

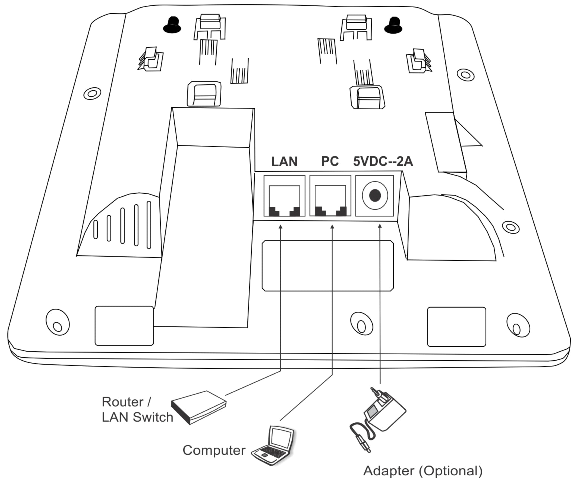
SPARSH VP310

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 Feature keys are programmable. For instructions on programming these keys, see “Phone Key Settings” under “Configuring Matrix SPARSH VP310”. 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 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 "Phone 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 body.

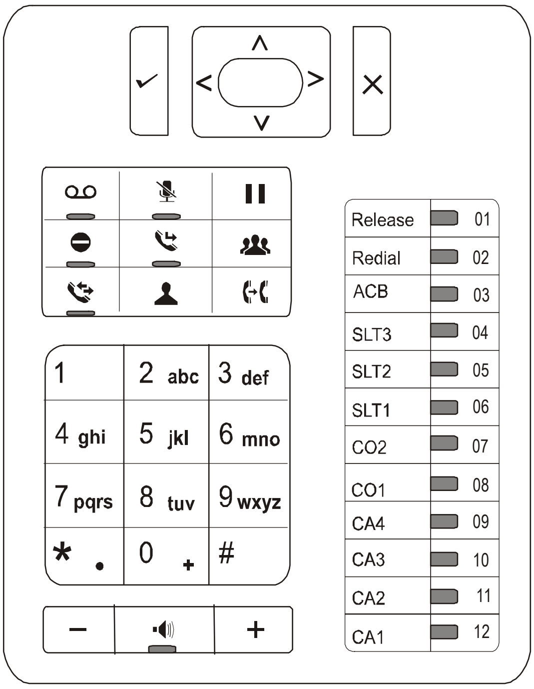
So you can use any stereo headset of standard make with a 3.5 mm single connector or a stereo headset with an RJ9 connector.

You can also program any of the DSS keys to function as the Headset key. For instructions on programming the key, see “Phone 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 “Phone Key Settings” under “Configuring Matrix SPARSH VP310”.

Matrix Extended IP Phone, SPARSH VP310 Key Template (default)



Phone Menu

Press the Enter Key to access the Phone Menu.

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.)
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 the 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).

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.

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 "SIP Extensions" for instructions.

Connecting SPARSH VP310

For detailed instruction to connect SPARSH VP310, see "Connecting SPARSH VP310 as Extended SIP Extension" in "Connecting SIP Extensions".

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 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 PBX 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.

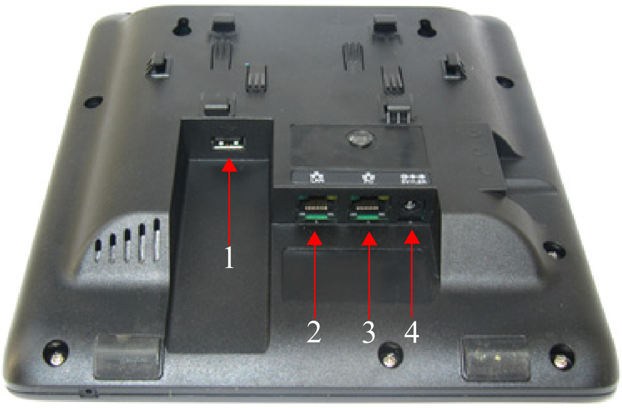
- **Improved Audio:** High Definition (HD) Audio output that delivers crystal clear voice and life like conversations over HD handset and hands-free speaker.

- **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

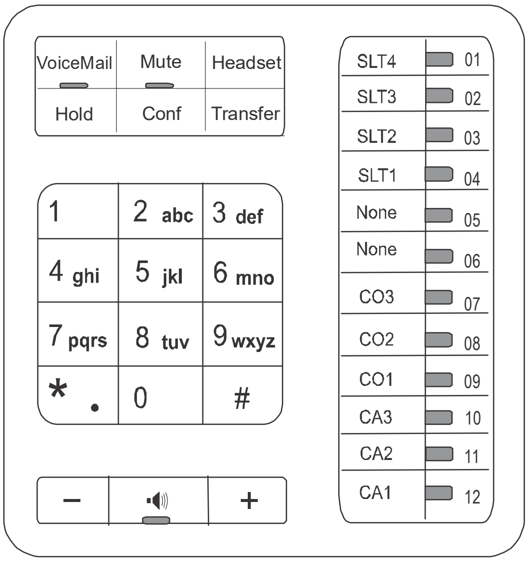


1	USB Port for Wi-Fi Adapter
2	LAN Port
3	PC Port
4	Power Port for the Power Adapter (Optional)

Left Side View

1	RJ11 Headset Port
2	Headset Port
3	Handset Port

SPARSH VP330 Key Template (default)



Feature Keys and DSS Keys

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, see “Phone Key Settings” under “Configuring Matrix SPARSH VP330” for instructions.

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. Features/facilities assigned to these keys can be changed by the System Engineer.

These Keys have dual color LEDs - blue and red. For instructions on programming these keys see, "Phone 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 instruction to connect SPARSH VP330, see "Connecting SPARSH VP330 as Extended SIP Extension" in Connecting SIP Extensions.

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 speaker-phone 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 Desk-phone 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

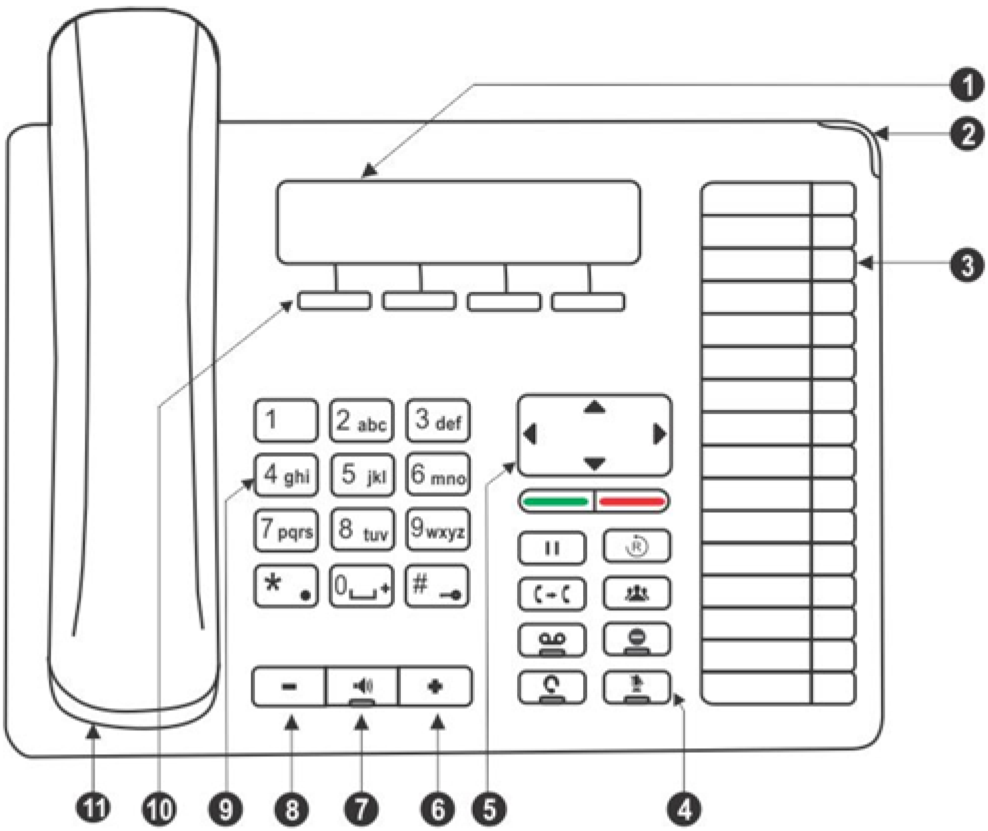
- Minimum 240*64 pixel and above graphical LCD with Back light
- Message Wait and Ringer Lamp
- Built-in 16 DSS Keys for Feature, Line, Extension
- Add -on DSS module facility
- HD Voice, HD Handset, HD Speaker
- 4 Context Sensitive Keys
- 5 Features Keys (With LED) -Voice mail, Headset, Mute, DND, Speaker
- 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 is Off-Hook, Off-Hook Alert will be provided to SPARSH VP510, if it is the Operator phone.

SPARSH VP510

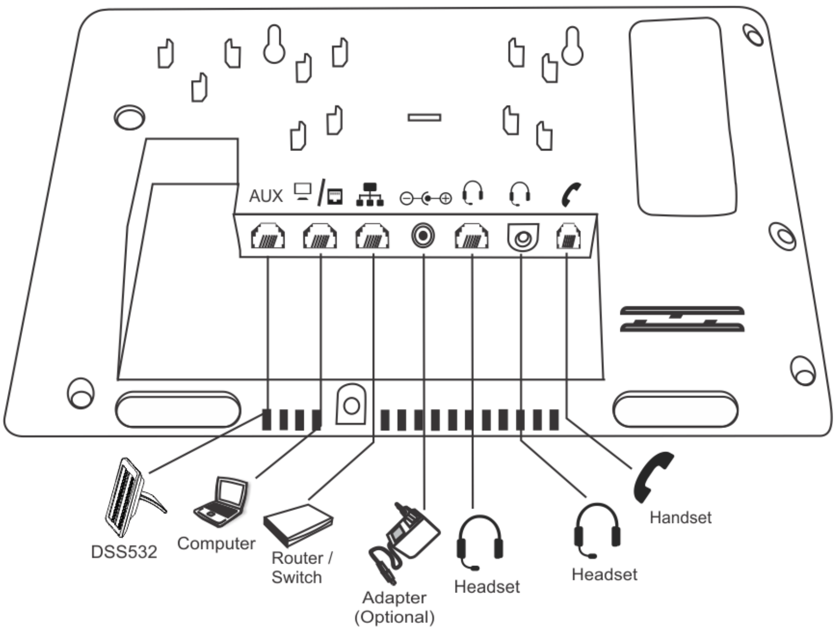
Front View



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 Feature keys are programmable. For instructions on programming these keys, see “Phone Key Settings” under “Configuring Matrix SPARSH VP510”. 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.

For instructions on programming these keys, see "Phone 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.

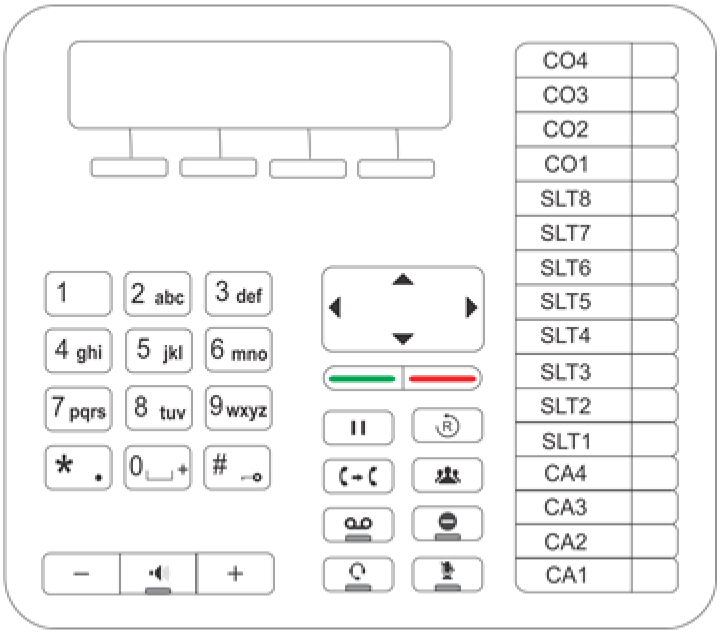
To use the Headset, a Headset Key is assigned on the phone. The Headset Key on the phone is equipped with a single color LED which glows Blue when pressed to indicate that the Headset mode is turned on and is turned off, when you press it again to indicate that you have exit the headset mode.

You can also program any of the DSS keys to function as the Headset key. For instructions on programming the key, see “Phone 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 “Phone Key Settings” under “Configuring Matrix SPARSH VP510”.

Matrix Extended IP Phone, SPARSH VP510 Key Template (default)



Phone Menu

To access the Phone Menu, press the Enter Key. You can access the following PBX 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.
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 V to access the Network Settings.
- Press the Up key Λ , if you wish to change the Ringtone.

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 "SIP Extensions" for instructions.

Connecting SPARSH VP510

For detailed instruction to connect SPARSH VP510, see "Connecting SPARSH VP510 as Extended SIP Extension" in "Connecting SIP Extensions".

Configuring SPARSH VP510

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.

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.

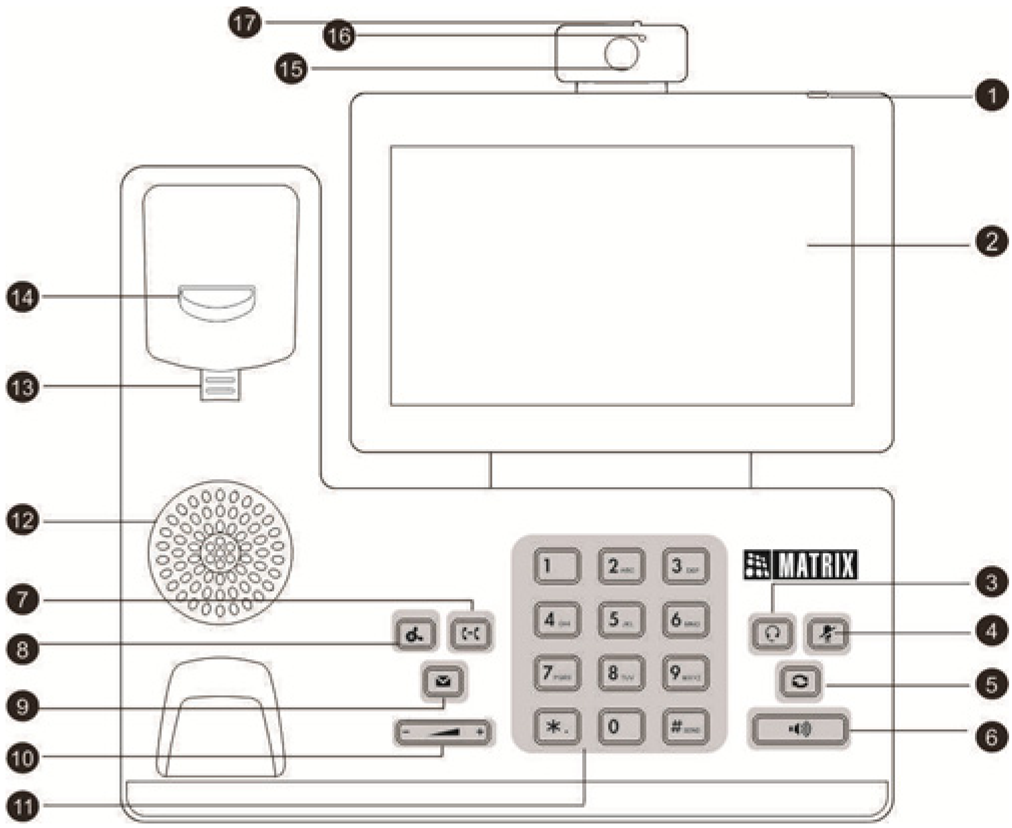
- **Cost Effective Calling:** If you are using the enterprise Wi-Fi or Ethernet network to register the IP Phone with the System, calls made will be almost free.

- **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.

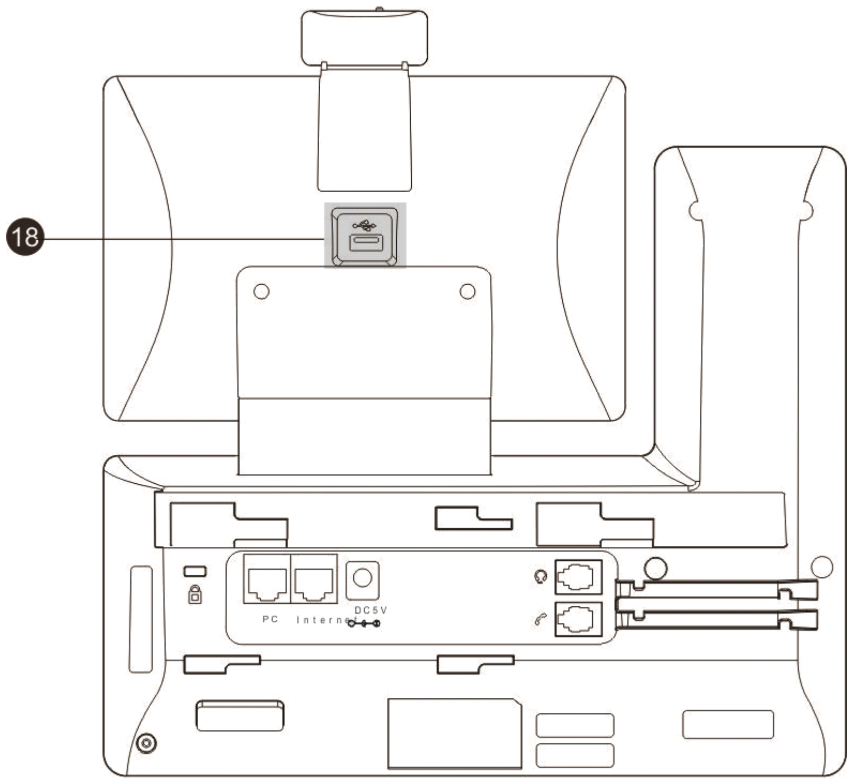
- **Advanced Call Capabilities:** Provides access to the features such as Callback, Dial-in Conference, Conversation Recording and many more.

Extended SPARSH VP710

Front View



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.
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 “Configuring Matrix Extended SPARSH VP710”.

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.

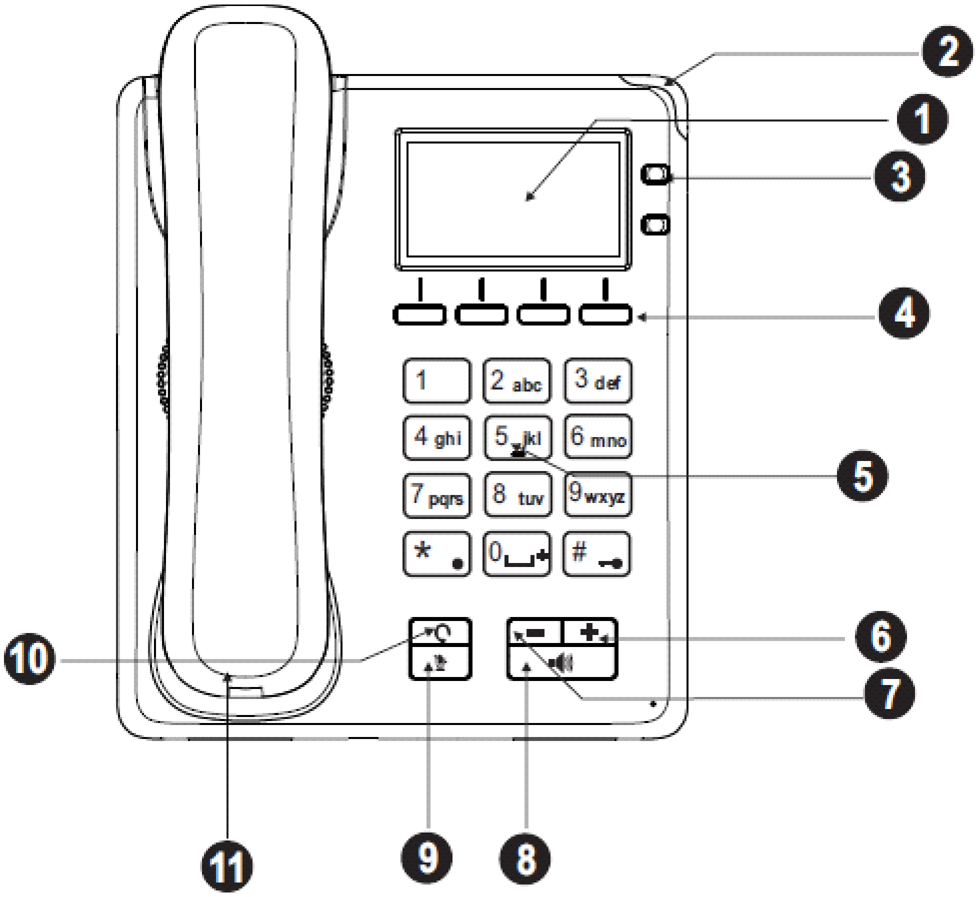
The State-of-the-art Deskphone is best suited for usage in lobbies, cafeterias, conference centers wherein the basic level endpoint security is sufficient. It can also be used by Administrative Staff, Hospitality guest rooms, knowledge workers etc.These phones offer flexibility to streamline communication 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

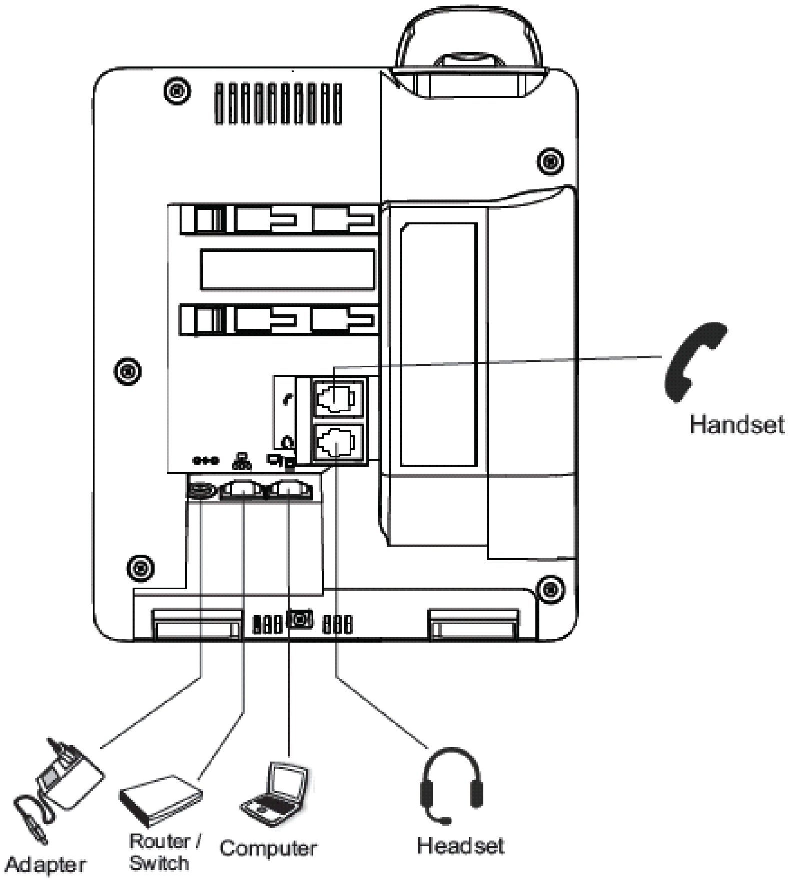
Front View



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.

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.

Menu option	Description
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.

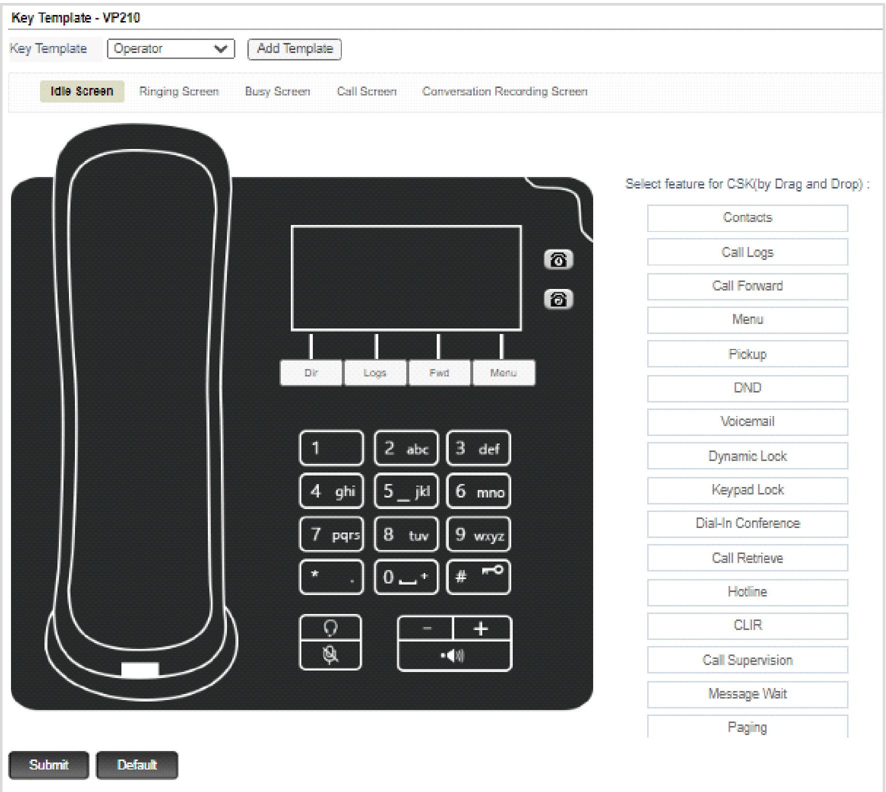
- In the Idle Screen you can assign the desired feature/function to the Context Keys as well as set their priorities as per your requirement.
- In the other Screens you can only set the priorities of the features.

Refer to "Key Maps" as well as "Phone 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, refer to "Phone 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 "Connecting SIP Extensions".

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 V2.3.3 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 “MATRIX VARTA User Licenses”.

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 **iOS7**.

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*.

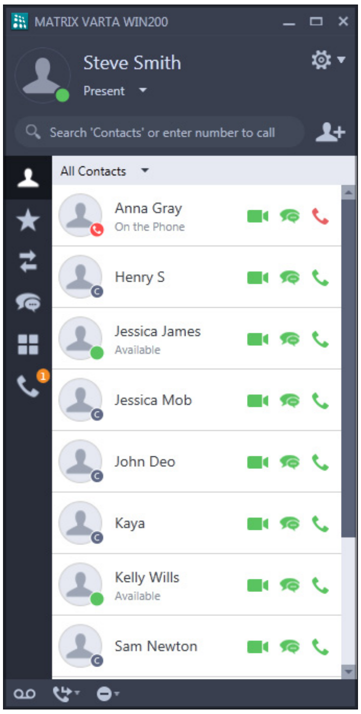
Configuring VARTA AMP100

For detailed instructions on how to configure VARTA AMP100, see 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

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

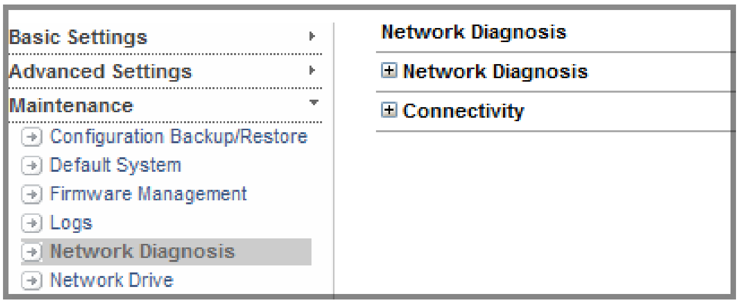
After System Restart the VARTA Client Inactivity Timer will be reset.

FCM Connectivity

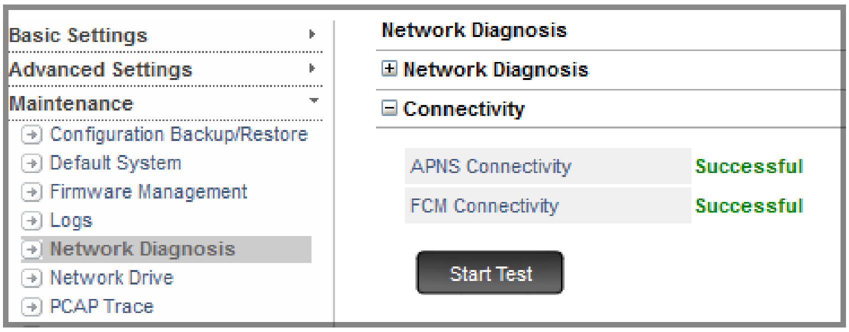
A connectivity between the system and the FCM Server is required so that the Push notifications can be sent to the clients.

To check the FCM connectivity status,

- Log into Jeeves.
- Click **Maintenance.**
- Under **Maintenance**, click **Network Diagnosis**



- 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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Flashing on Trunks (Continued Dialing)

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)". For instructions see "SLT Extensions" and "SIP Extensions" under *Basic Settings*.

How to use

For 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 configured).
- 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 from a 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.

Follow Me

Using this feature, you can make your calls follow you wherever you go. You can receive your calls on another extension, whenever you want.

How it works

- A's extension number is 201.
- B's extension number is 203.
- A is currently at B's extension.
- A wants to receive calls from extension 201 on extension 203.
- A sets Follow Me on extension 203.
- All calls landing on A's extension 201 will be forwarded to extension 203.
- When A returns to extension 201, A cancels 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 canceled.*
- *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 "Do Not Disturb (DND)" and "Call Forward".

How to configure

To be able to use Follow Me, extension users must have Call Forward feature enabled in their "Class of Service (CoS)" for the Day and Night/Break time, as required. See "SLT Extensions" and "SIP Extensions" under "Basic Settings" for instructions.

How to use

For 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

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 a SLT or an Extended IP Phone. However, the called extension (being forced to answer) must be an Extended IP Phone.

- Extension user A (SLT) calls extension user B (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 is turned on (goes OFF-Hook).
- A may now talk to B.

Forced Answer can be used only when the called extension is 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 (CoS)" for the Day and Night/Break time, as required. See "SLT Extensions" and "SIP Extensions" under "Basic Settings" for instructions.

How to use

For 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

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.*

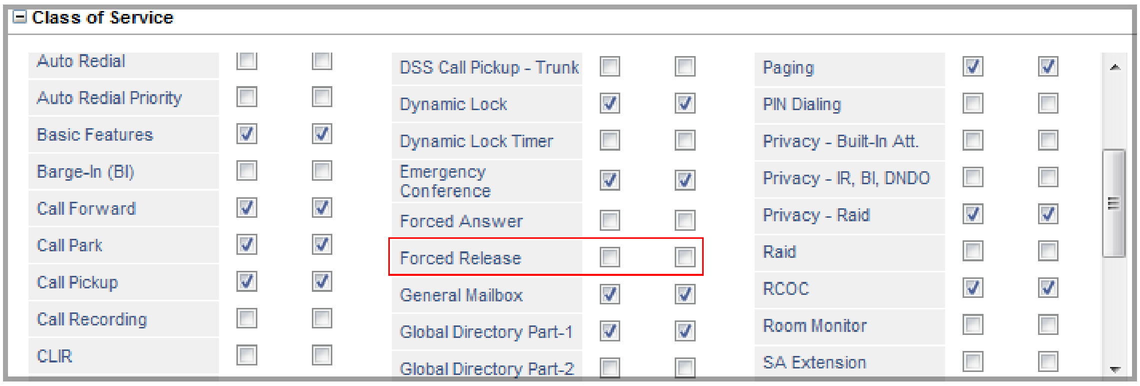
Forced Call Disconnection is not supported on SIP Trunks.

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.

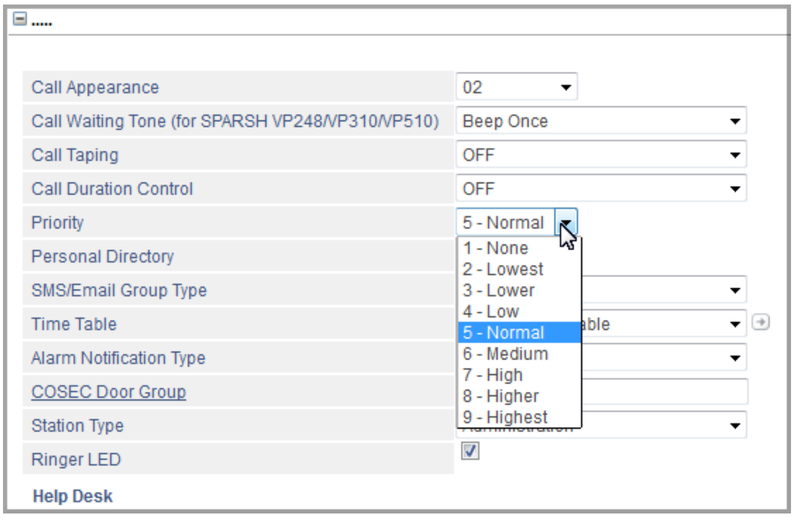
How to configure

To be able to use Forced Call Disconnection, extension users must have:

- **Forced Release** feature enabled in their "Class of Service (CoS)" for the Day and Night/Break time, as required.



- Higher "Priority" assigned than other extensions.



*As Forced Call Disconnection on a busy trunk is possible only if the extension user has grabbed that trunk using "Selective Port Access", the feature **Selective Port** must be enabled in the "Class of Service (CoS)" for the Day and Night/Break time, as required. See "Basic Settings" for instructions on configuring CoS for the different extension port types.*

How to use

For Extended IP Phone Users

To forcibly disconnect a busy extension/trunk[151]

- 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.

151. *You must have grabbed this trunk using Selective Port Access.*

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.

Gain Settings

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—SLT, Mobile, SIP. 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

- Login as System Engineer.
- Click **Basic Settings**.
- To configure Gain Settings for CO Trunks, see “Hardware Settings” under “CO Trunks”.
- To configure Gain Settings for Mobile Trunks, see “Gain Settings” under “Mobile Trunks”.
- To configure Gain Settings for SIP Trunks, see “Gain Settings” under “SIP Trunks”.
- To configure Gain Settings for SLT Extensions, see “Hardware Settings” under “SLT Extensions”.
- To configure Gain Settings for SIP Extensions, see “VoIP” under “SIP Extensions”.

Handover and Handoff

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 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* in "SIP Extensions".

Help Desk

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 202 is defined as Help Desk for HR policies and general rules.
- Extension 216 calls the Help Desk extension 202.
- 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 216.

Also see "Auto Call Back (ACB)"

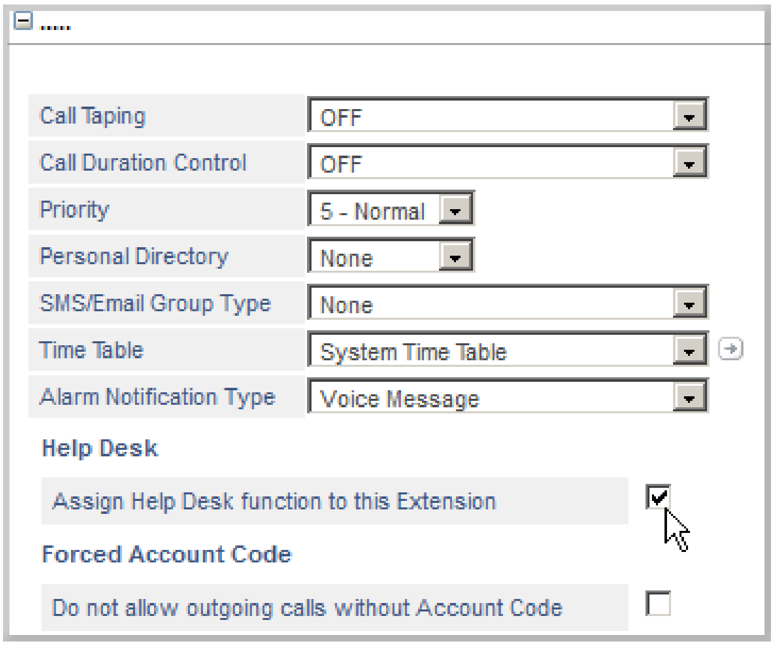
How to configure

To enable Help Desk feature on any extension,

- Login as System engineer.
- Click **Basic Settings**.
- Click the desired extension port type: SLT, SIP to which you want apply this feature.
- Select the extension number to which you want to apply this feature, by clicking the tab.

  The selected extension tab opens.
- Click the **More** button to view all parameters on this page.
- Click the **More...**link on the extension page.

- Scroll to **Help Desk**.



- Select the **Assign Help Desk function to this Extension** check box.

  The selected extension will be designated as Help Desk.

- Click **Submit**.

Holiday Table

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 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 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.

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 Night (Non-working hours),
  - the system checks for and routes the incoming call as defined for the trunk during Night (Non-Working hours).
  - customized Holiday Message, if configured, is played to the caller. The Holiday message is played in place of the Welcome message.
  - Extensions work according to the Class of Service, Toll Control and Trunks assigned for outgoing calls, assigned to them during the Night (Non-working hours).

Similarly, if the Time Zone for Holiday is defined as Day (Working Hours), the system will operate the time-zone based features of the trunks and extensions according to the Day (Working Hours).

- When Time Zone for Holiday is As per Timetable,
  - the system routes the call as defined for the current time zone.
  - 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 Trunks assigned for outgoing calls to them for the current time zone.

Feature Interaction:

Day/Night Mode: You can set SARVAM UCS 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. See “Day Night Mode” for more details.

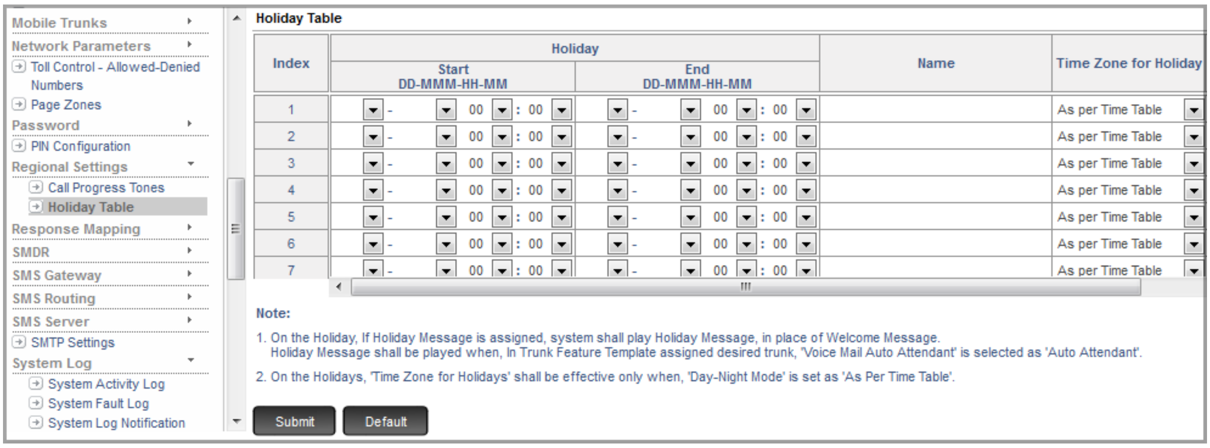
Set the parameter Day/Night Mode to ***Operate System as per Timetable Assignment*** in the System Parameters, if you want the system to operate as per the Time Zone you set for the Holiday. See “System Parameters” for more details.

How to configure

You can configure the Holiday Table from the SE as well as the SA mode.

To configure the Holiday Table,

- Login as System Engineer.
- Under **Advanced Settings**, 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.

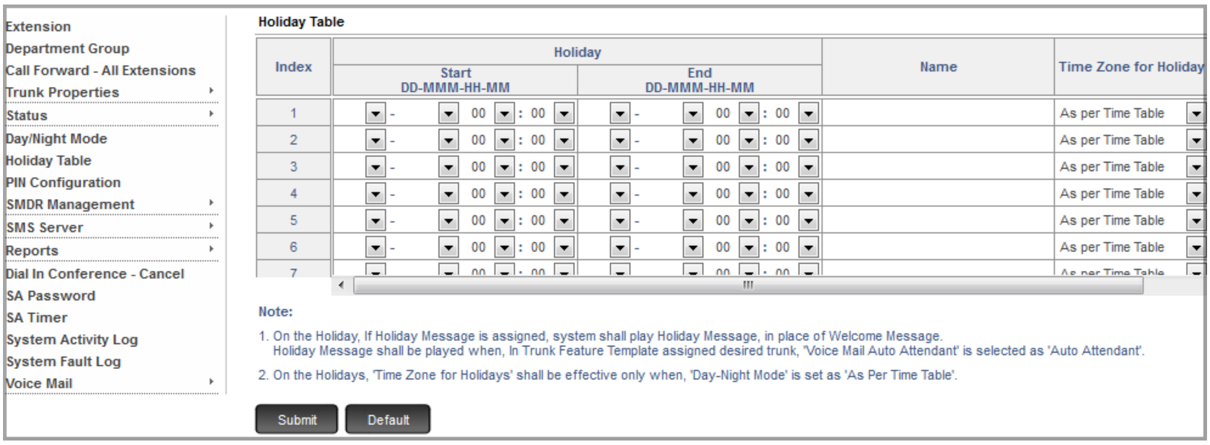
  Default, Start and End Date are Blank. Valid Range is from 01 to 31.
  Start and End Month are Blank. Valid Range is from January to December.
  Start and End Time is 00:00 (Hours:Minutes). Valid Range is from 00:00 to 23:59

- You can assign a **Name** to each time period you have defined as Holiday. For example, Christmas, Independence Day, Thanksgiving. Default: Blank
- In the **Time Zone for Holiday,** select the time zone according to which you want the trunks, extensions and time dependent features and facilities to work during the period you defined as Holiday. You may select Day (Working Hours), Night (Non-working Hours) or As Per Time Table. Default: As Per Time Table.
- If the incoming calls are routed to the Voice Mail Auto Attendant, select the **Holiday Message** number that you want the system to play to the callers. Default: Auto_Attendant_01.
- Click **Submit**.

You can also configure the Holiday Table from the SA mode also. To do this,

- Login as System Administrator.

- Click the **Holiday Table**.



- Follow the same steps as given above to configure the Holiday Table.

- Click **Submit**.

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 **Incoming Call Routing**. For instructions, see “CO Trunks”, “Mobile Trunks” and “SIP Trunks” under Basic Settings.

- 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

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 SLT extensions only.

Hot Desking is possible only from one SLT to another SLT extension.

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 canceling 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 (CoS)” of the Host Extension and the Hot Desk Extension. By default, this feature is disabled on all extensions, so none of the extensions can use this feature. To enable this feature in the CoS of the extensions, see “SLT Extensions” and “SIP Extensions” under “Basic Settings”.

How to use

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

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 a SLT or an Extended IP Phone, 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, 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.

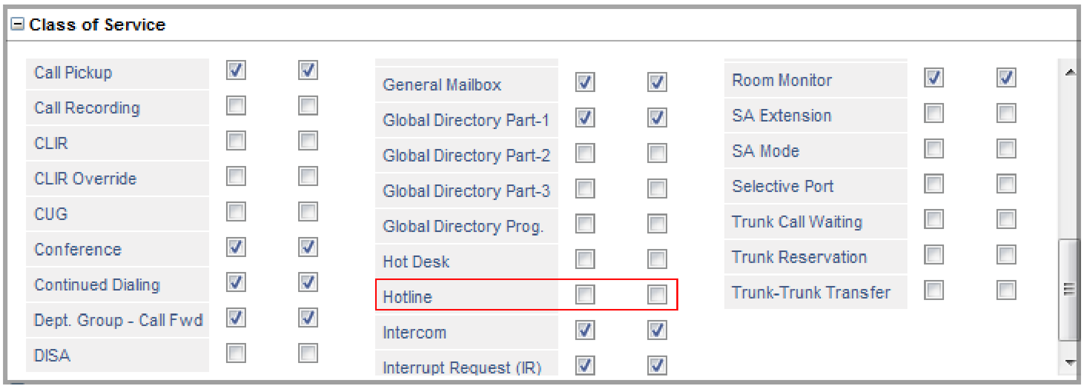
  - If A sets delayed Hot Outward Dialing for a Trunk or an External Number, the system will play dial tone to A and wait for the duration of the Hotline Timer for A to dial digits. If A does not dial any digits within this timer, the system connects A to the Trunk/External Number.

- 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 Day and Night/Break time, as required. See “Basic Settings” for instructions on configuring the different extension port types.



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.

How to configure

The Operator or any extension user having access to System Administrator mode can set or cancel Hotline for other extension users from the SA mode. To do this,

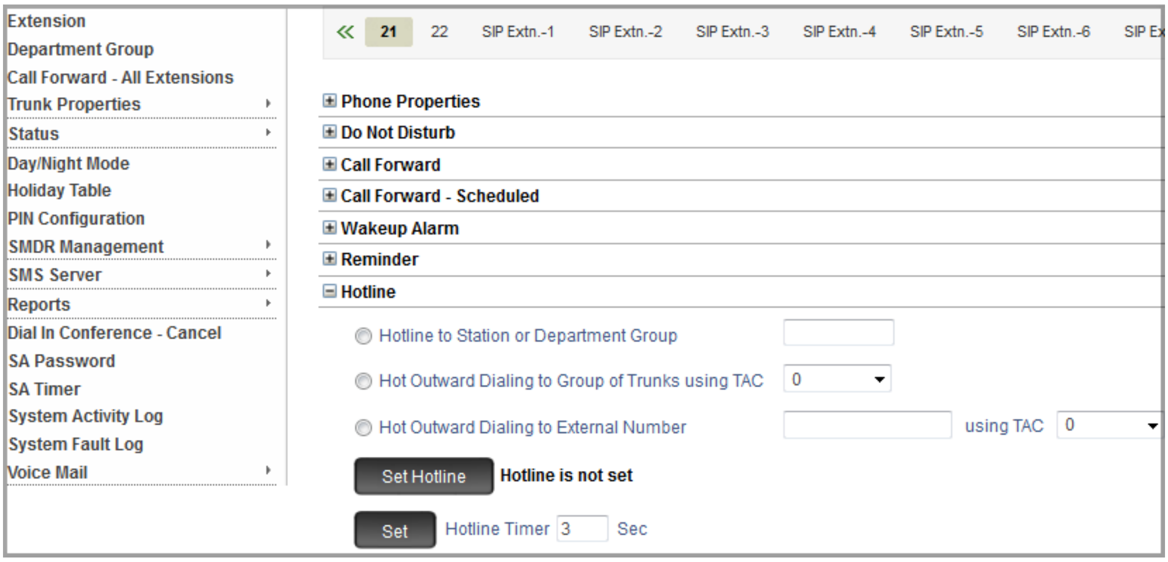
- Login as System Administrator.

- Click **Extension**.

- Now, click on the desired **Extension Number** tab.

- The 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 dialed out by the system.
- Click **Set Hotline** 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 **Set** to set Delayed Hotline.
- To cancel Hotline, click **Cancel Hotline**.

Set/Cancel Hotline by Extension Users

For Extended IP Phone Users

To set Hotline on an Extension / Department Group

- Press DSS key assigned to Hotline.
- Scroll to select 'Set Hotline Station (extension) / Department', press enter key.
- Enter Extension/ Department Group Number.
  OR
- Dial 151-Extension Number / Department Group Number.

To set Hot Outward Dialing for Trunk

- Press DSS key assigned to Hotline.
- Scroll to select 'Set Hotline Outgoing Trunk', press enter key.
- Enter TAC.
  OR
- Dial 152-TAC.

To set Hot Outward Dialing for External Number

- Press DSS key assigned to Hotline.
- Scroll to select 'Set Hotline Outgoing Trunk: 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 with 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

For PBX 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' configured, 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 modifying the CLI manually.

How it works

- Incoming CLI Modification parameters must be configured in the system considering the dialing pattern supported by the local public network.
- Accordingly, SARVAM UCS matches the CLI received with the configured parameters.
- It detects whether it is an international, national or local number.
- It modifies the CLI according to the Modification parameters configured.
- It presents the modified CLI to the extension; stores the modified CLI in the SMDR and in the Call Logs of the extension.
- When the received CLI is dialed out by the extension user from Call Log, SARVAM UCS dials out the same number.

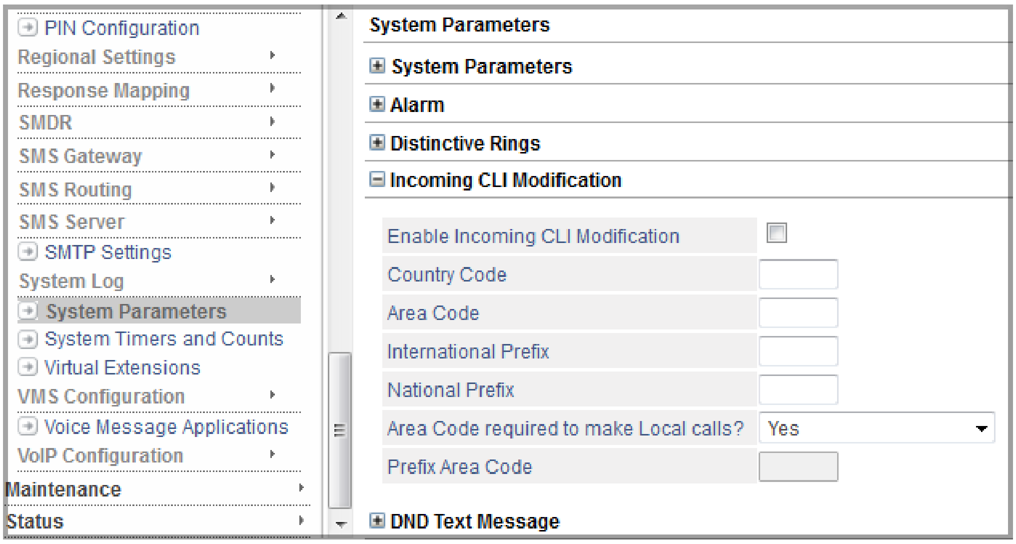
How to configure

For this feature to work, you must configure **Incoming CLI Modification** parameters in the "System Parameters" and **Allow Incoming CLI Modification** in the respective trunk of SARVAM UCS. To know more, refer to "CO Trunks", "Mobile Trunks" and "SIP Trunks".

- Login as System Engineer.

- Under **Advanced Settings**, click **System Parameters**.

- On the System Parameters page, click **Incoming CLI Modification** to expand.



- **Enable Incoming CLI Modification:** Select this check box, if you want to use the Incoming CLI Modification feature. By default, this check box is disabled.

If you receive CLI in dial-able format, there is no need to use this feature. In such case, keep the check box disabled. You do not need to configure 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. By default the Country Code is '91' (India).

  For example, if the 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'.

- **Area Code:** Enter the Area Code of the place where 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. By default, Area Code is '265' (Vadodara city).

  For example, if you want to enter Area Code for Mumbai, enter only '22'. Do not enter the prefix '0' to the area code.

- **International Prefix:** Enter the digits that are required as Prefix for dialing International Numbers. The prefix may be upto 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 upto 5 digits, with numbers from 00000 to 99999. By default, it is Blank.

- **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:** select this option, if your public telephone network does not require the dialing of Area Code for local numbers.

- **Yes:** select this option, if your public telephone network requires the dialing of Area Code for local numbers.

- **Yes, with Prefix Digit:** select this option, if you public telephone network requires the dialing of Area Code with a particular Prefix for local numbers. If you select this option, you must also configure the Prefix digits for the Area Code.

  By default, the option, ‘No’ (Area Code not required) is selected.

- **Prefix Area Code:** If you have selected **Yes, with Prefix Digit** option for **Area Code required to make local calls?** parameter, enter the Prefix digits for the Area Code for local calls in this field.
- Click **Submit.**

Interrupt Request (IR)

Interrupt Request allows you to break into an ongoing conversation after intimating the extension user about the interruption.

In case of an 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

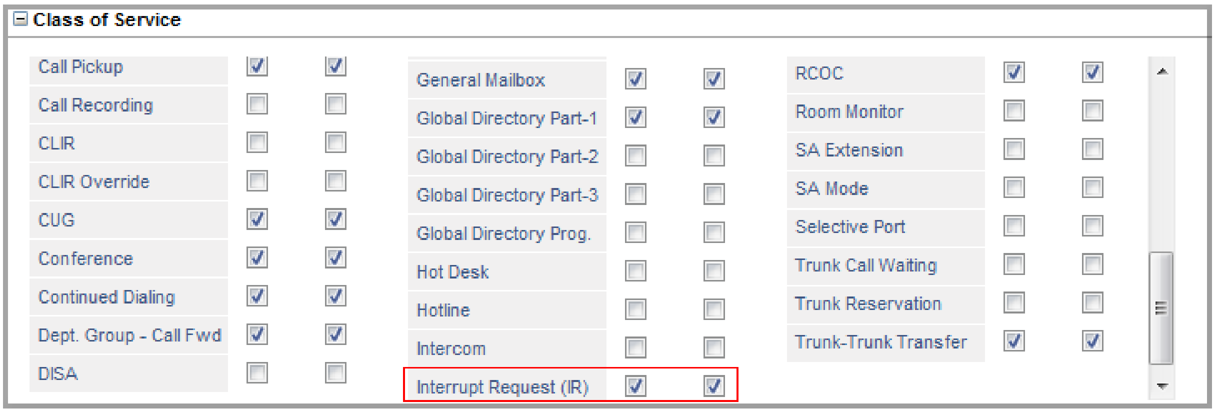
- A, B and C are extension users.
- A and B are talking to each other.
- C calls A.
- C gets busy tone.
- C dials Interrupt Request feature code.
- C gets Ring Back tone (RBT) and A gets beeps indicating a new call.
- To answer C’s call, A must dial **Flash** before the expiry of the Interrupt Request 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 PBX.
- **“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.
- **“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)" for the Day and Night/Break time, as required.



See "Basic Settings" for instructions on configuring the different extension port types.

Interrupt Request Timer

Interrupt Request Timer is the time for which the extension on which interrupt request is made gets the beeps. By default the Timer is set to 45 seconds. Valid Range: 001 to 255 seconds. Default: 045 seconds.

Changing Interrupt Request Timer using Jeeves

- Login as System Engineer.
- Under **Advanced Settings**, click **System Timers and Counts.**
- Scroll to reach **Other Features** and go to the parameter **Interrupt Request Timer**

- Set the desired value for the Timer.
- Click **Submit.**

How to use

For 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

Intercom

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 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 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. See *Class of Service* in the "SLT Extensions" and "SIP Extensions" topics under "Basic Settings" for instructions

How to use

For 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.

Last Caller Recall

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 201.
- A wants to know who made the last call to extension 201.
- A dials Last Caller Recall feature access code (Default:1092).
- The extension that last called 201 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, enabling interoperability with IP Phones of Cisco and Polycom. 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 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 Handset.
- Dial 1092.
  The system dials out the extension number that last called your extension.

Last Number Redial

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 call 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 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 this feature.

How to use

For Extended IP Phone Users

- Press Redial Key.
  Or
- Dial 7
  A List of last 16 numbers dialed by you will appear on your phone’s display.
- Scroll to select the desired number.
- Press Enter key.
- The system dials out the selected number.

For SLT Users

- Lift Handset
- Dial 7
- The system dials out the number last dialed from your extension.

Least Cost Routing (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 SARVAM UCS, LCR recognizes where the call is going to. It selects the lowest cost trunk among all the trunks allotted to that extension to make outgoing calls, 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 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 configured:

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 9 am to 8 pm, while Service Provider 2 offers a lower tariff for calls made between 8 pm to 9 am.

   When Time-based LCR is configured, 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 lower calling rates for calls made from City A to City B, than Service Provider 2 and Service Provider 3.

3. **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 Mobile services and applies different charges for fixed line and Mobile services.

   When Service Provider-based LCR is configured, 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 lowest cost trunk configured for that number

4. **Time and Number based LCR:** This type of LCR is a combination of number and time based LCR, i.e. 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 9 am to 8 pm, as compared to Service Provider 2, whereas Service Provider 2 offers lower rates for calls made from City A to City B during off peak hours (8 pm to 9 am).

When Time+Number based LCR is configured, 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 lowest cost trunk.

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 *"Least Cost Routing - Carrier Pre-Selection"* *topic 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.

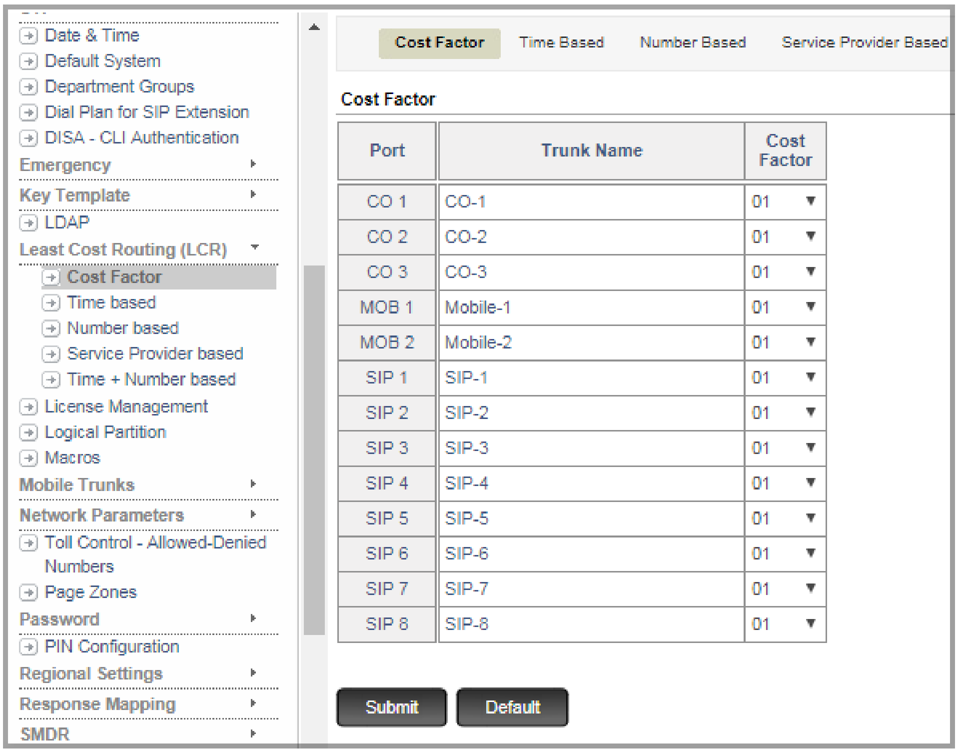
By default, all trunks are assigned Cost Factor number 01.

After assigning Cost Factor to Trunks, you must configure the Type of LCR to be used on Trunks in the **Select Trunks for Outgoing Trunks** allotted to the extensions for making calls.

Assigning Cost Factor to Trunks

- Login as System Engineer.
- Under **Advance Settings**, click **Least Cost Routing (LCR)**.

- Click **Cost Factor**.



- The **Cost Factor** table displays all the **Port Type-Number** of the system and the **Trunk Names**.

- In **Cost Factor** column of the table, assign a cost factor to each trunk type. For instance, assign Cost Factor 01 to CO1 and Cost Factor 02 to CO2. Similarly, assign Cost Factor 03 to Mobile Trunk 1, Cost factor 04 to Mobile Trunk 2, Cost Factor 05 to SIP trunk 1.

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 preferences. 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 9 am to 8 pm 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 Mobile trunk of BSNL. When this line is busy, you want the system to attempt to route calls through the Mobile trunk of Reliance.
- You want calls made between 8 pm to 9 am to be routed through BSNL CO1 trunk 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 Mobile-01 (07) and Mobile-02 (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 CO1, i.e. 01 as Preference 1, 2, 3, and 4. When calls are made during this time period, they will be routed through CO1 only.

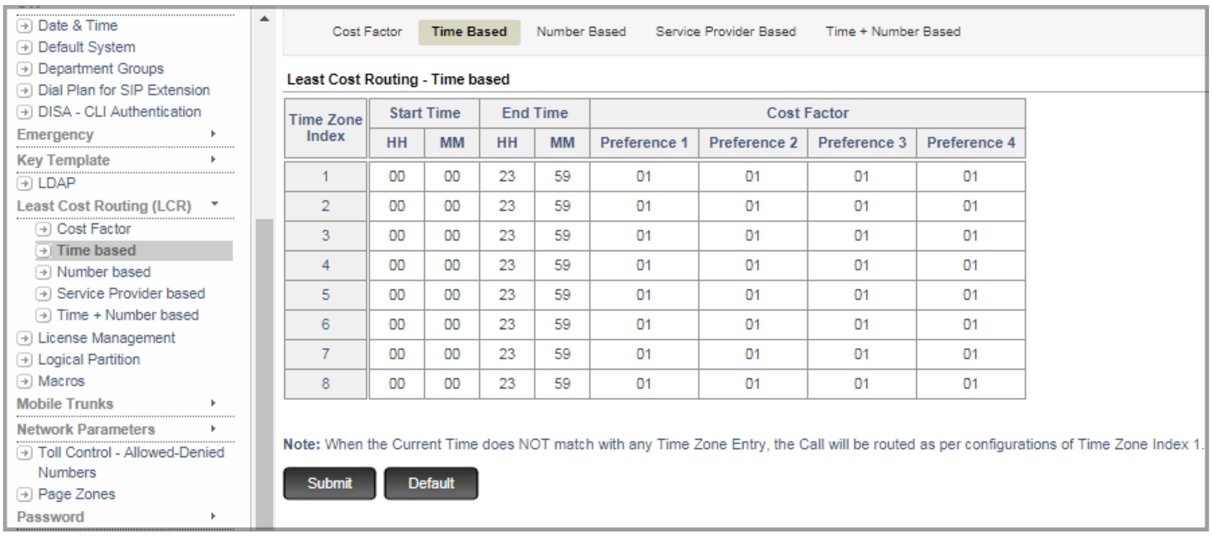
- If you have finished defining Time Zones and the preferred trunks for the time zones, configure the Time-based LCR using Jeeves.

Configuring Time based LCR using Jeeves

- Login as System Engineer.

- Under **Advanced Settings**, click **Least Cost Routing (LCR)**.

- Click **Time based**.



- Enter the values of the Time based LCR you prepared on the sheet of paper in the appropriate fields.

- Click **Submit.**

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 1 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 free trunk configured.

- 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.

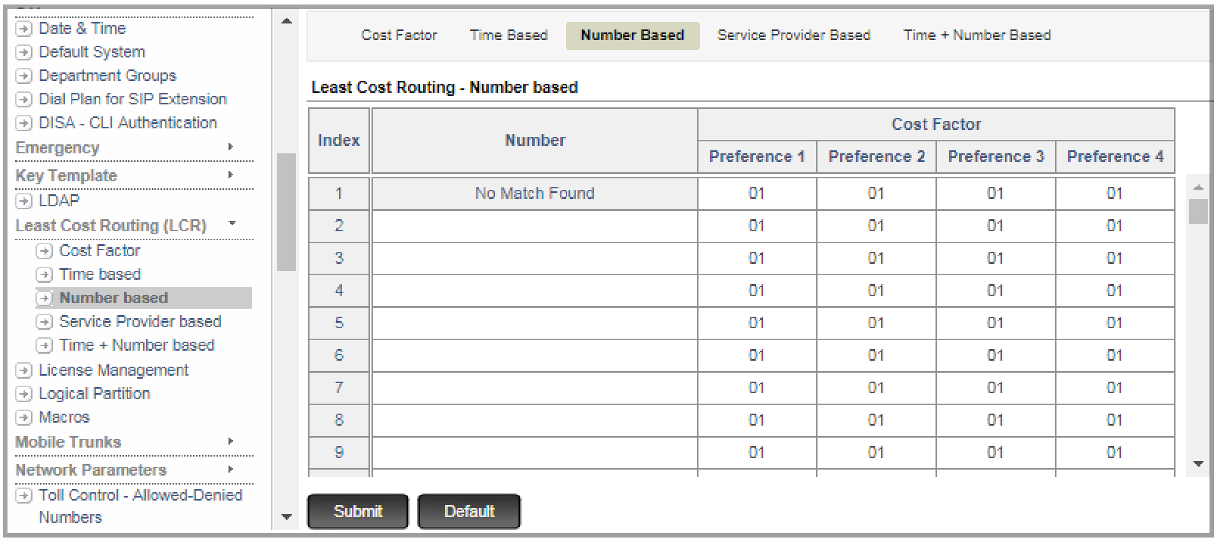
  Similarly, all local numbers start with 2, so enter this number in the number string column, and select the CO trunk in the order of preference. As in this example, you have only two CO trunks, so you may keep the same two trunks as your preference.

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					
:					
:					
99					

- If you have finished entering the number strings, and selecting the preferred trunks for the numbers, configure the Number based LCR using Jeeves.

Configuring Number based LCR using Jeeves

- .Login as System Engineer.
- Under **Advanced Settings**, click **Least Cost Routing (LCR)**.
- Click **Number based**.



- Enter the values of the Number based LCR you prepared on the sheet of paper in the appropriate fields.
- In Number, you can enter 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.**

Wildcard Characters

ETERNITY 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-Destination Port Determination-Destination Number Based* table 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 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 lowest cost trunk configured for that number. For this, you must configure 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	05	06	01	02
02	6	022	3	01	01	02	04
:	:		:				
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 configure as many as 99 different numbers which are stored against Index numbers from 1 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 who offers the lowest tariff to that number/area code. Refer the table you prepared for assigning Cost Factor to trunks.

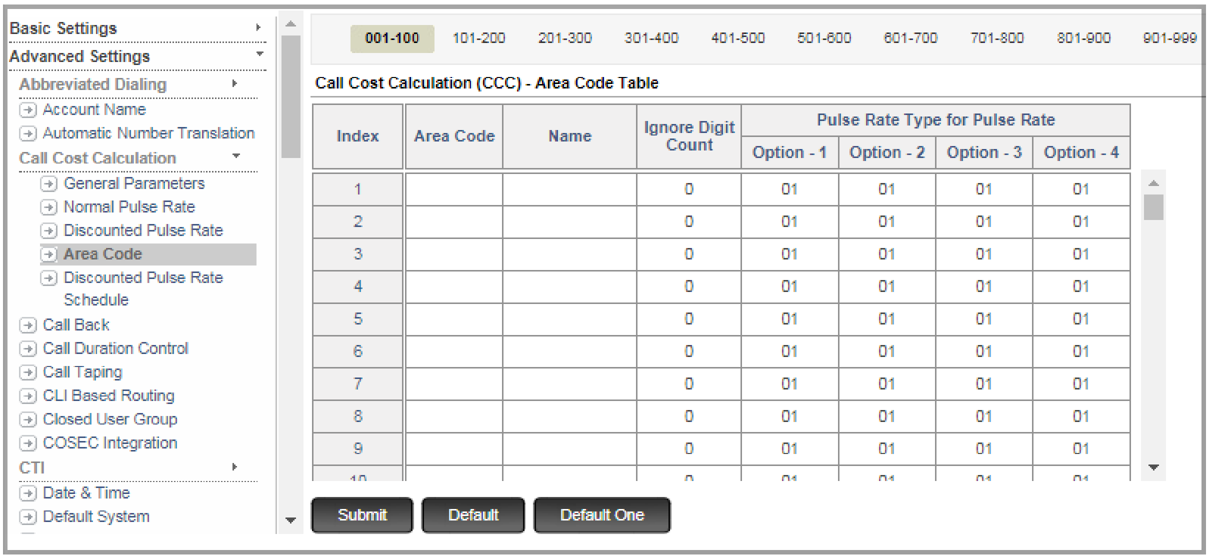
- If you have finished entering the number strings, their corresponding area codes and the Ignore Digit Count, and the preferred trunks, configure Service Provider based LCR using Jeeves.

Configuring Service Provider based LCR using Jeeves

- Login as System Engineer.

- Under **Advanced Settings**, click **Call Cost Calculation**.

- Click **Area Code**.



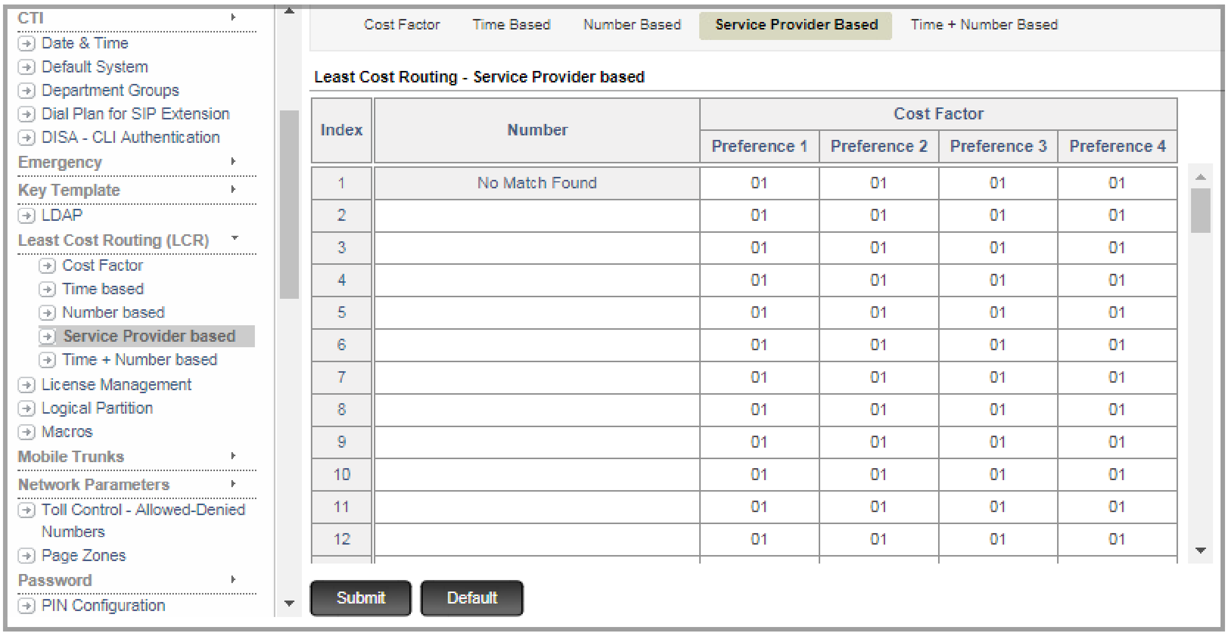
Each area code is stored at an index number.

- 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.**

- Now, click **Least Cost Routing (LCR)**.

- Click **Service Provider Based**.



- Enter the values of the Service Provide based LCR you prepared on the sheet of paper in the appropriate fields.

- In Number, you can enter 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.**

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 + Wildcard characters). 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 preferences. 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 8 am to 12 pm, 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 08:00 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.

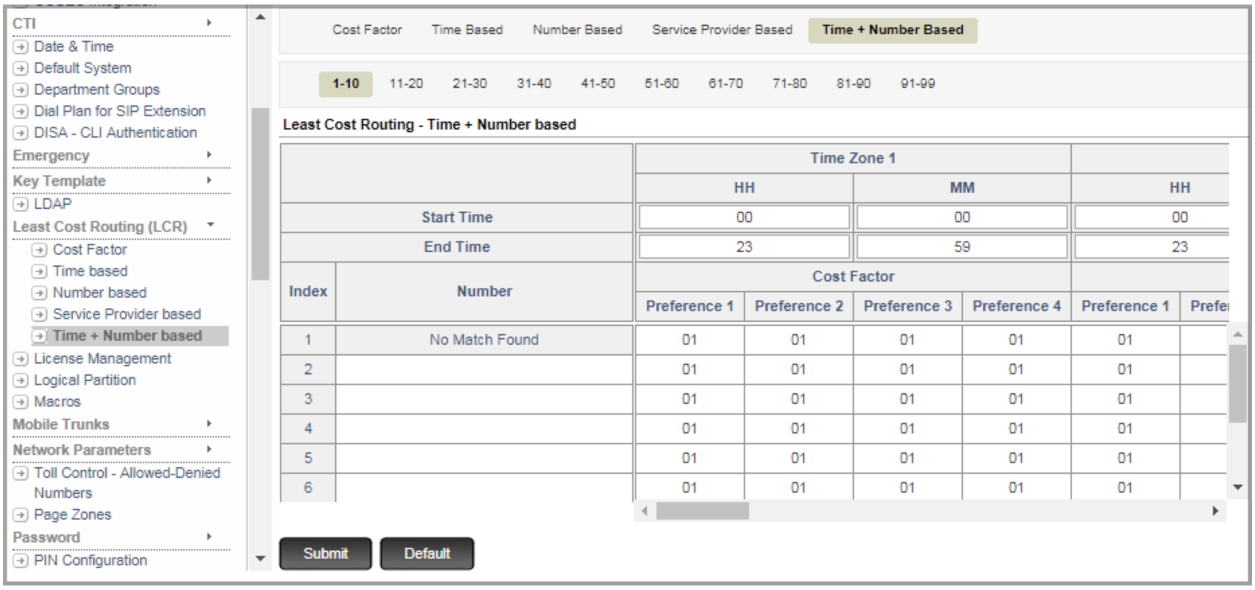
		Time Zone1				Time Zone2				Time Zone3				:	Time Zone 8
		HH		MM		HH		MM		HH		MM			
Start Time		08		00		12		00		09		00			
End Time		12		00		18		00		20		00			
		Cost Factor				Cost Factor				Cost Factor					
Index	Number	Preference 1	Preference 2	Preference 3	Preference 4	Preference 1	Preference 2	Preference 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.

Configuring Time+Number based LCR using Jeeves

- Login as System Engineer.
- Under **Advanced Settings**, click **Least Cost Routing (LCR)**.

- Click **Time+Number Based.**



- Enter the values of the Time+Number based LCR you prepared on the sheet of paper in the appropriate fields.

- In Number, you can enter 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**.

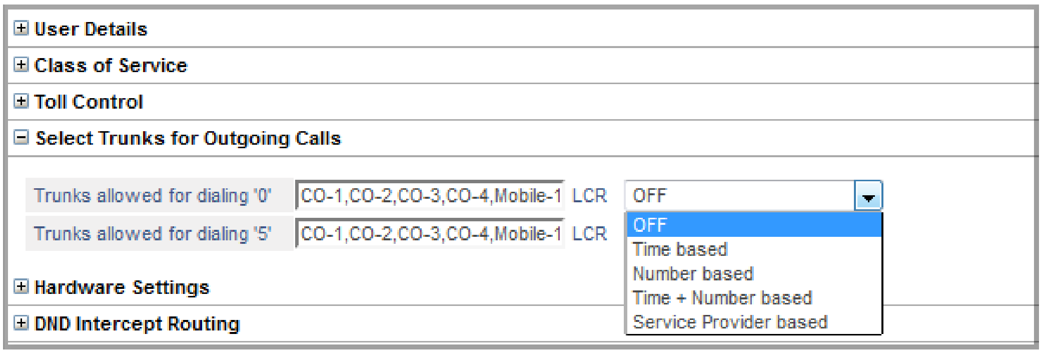
Configuring LCR Type on Trunks

After assigning Cost Factor to trunks and configuring the LCR Type - Time based, Number based, Service Provider based, Time and Number based you must now apply the desired LCR Type on the outgoing trunks selected for the extension types: SLT and SIP.

- Under "Basic Settings", open the following Extension type:
  - "SLT Extensions"
  - "SIP Extensions"

- Select the desired Extension number, for example SLT 201.

- Click **Select Trunks for Outgoing Calls** option for the Extensions.



- For the trunks you selected for Dialing the Trunk Access Codes 0 and 5, select the desired LCR Type from the combo box: Time-based, Number based, Service Provider based (Cost Factor), Time+Number based.

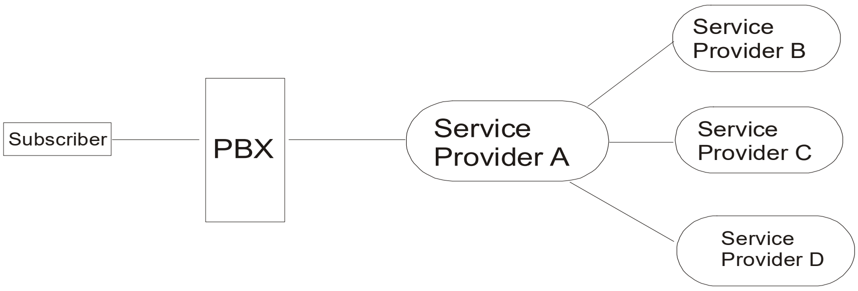
  Since you have already configured the LCR Type, you do not need to configure the LCR settings further on this page.

- Click **Submit.**

Least Cost Routing - Carrier Pre-Selection

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,



A 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 configure the Automatic Number Translation (ANT) table on the desired trunks—CO, Mobile or SIP. See **Automatic Number Translation (ANT)** under "CO Trunks", "Mobile Trunks" and "SIP Trunks" for instructions.

License Management

The application SARVAM UCS and the features that it supports, require the purchase of licenses. When you buy the ETERNITY NENX Platform, you get a unique default license key for the platform.

This section lists the features of SARVAM UCS for which you require licenses. It also provides guidelines for activating different licenses supported in SARVAM UCS. To know the name of the licenses, you need to purchase and activate, see “Supported Licenses”.

IP Subscriber License (For SIP Extensions)

SARVAM UCS supports 50 SIP Extensions. With this license, you can register SIP-enabled devices with SARVAM UCS. Without a license you cannot register any SIP Extension.

By default, ETERNITY NENXIP50 supports registration and configuration of 50 SIP Extensions and ETERNITY NENX312[152] and ETERNITY NENX416[153] supports registration and configuration of 10 SIP Extensions only. To configure and register additional SIP Extensions, you need to purchase and activate the IP Subscriber license. To know about the IP Subscriber license supported in SARVAM UCS, refer to “Supported Licenses” and “Pre-activated Licenses”.

To register a Standard SIP Phone/Extended IP Phone, you only need to purchase an IP Subscriber License whereas to register a VARTA UC Client, you need to purchase both the IP Subscriber and relevant VARTA Licenses.

Voice Mail System

SARVAM UCS offers a Voice Mail System (VMS) with VMS Auto Attendant features. The VMS is provided in a Pen Drive. To use Voice Mail System, you must purchase and activate the VMS license. Refer to “Supported Licenses” and “Pre-activated Licenses”

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. Refer to “Supported Licenses” and “Pre-activated Licenses”

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 SARVAM UCS. The Server supports multipart, 7 bit text messages as well as UNICODE messages.

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152. Make sure you have purchased and installed the VoIP Module. Refer to “Installing the VoIP Module”.
153. Make sure you have purchased and installed the VoIP Module. Refer to “Installing the VoIP Module”.

The Server functions as a SMTP Client to send emails and as a POP3 Client to receive mails. Refer to “Supported Licenses” and “Pre-activated Licenses”

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. Refer to “Supported Licenses” and “Pre-activated Licenses”

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.

If you want to register additional VARTA Clients, you must purchase the VARTA licenses as per your requirement. To know the license details, refer to “Supported Licenses” and “Pre-activated Licenses”

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		
	WIN200	ADR100	AMP100	WIN200	ADR100	AMP100	WIN200	ADR100	AMP100
Making Calls	✓	✓	✓	✓	✓	✓	✓	✓	✓
Receiving 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 *VARTA WIN200 User Guide* and for VARTA UC Clients for Mobile refer to the respective User Guide — *VARTA ADR100 User Guide* and *VARTA AMP100 User Guide*.

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 IPSUB5	License for 5 IP Subscribers to create 5 SIP Users.
SARVAM VMS SOHO	License to attend 4 Simultaneous calls with Mailboxes for Individual 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 USER5P	License for 5 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 SMS SERVER SOHO	License to enable SMS Server functionality to connect to any Email Client and send/receive Email to SMS and vice versa over GSM Interface.
SARVAM SMS GATEWAY SOHO	License to enable SMS Gateway functionality to connect to third party SMS Gateway Clients and send/receive SMS over GSM Interface.
SARVAM CTI SOHO	License to enable CTI functionality to connect to third party CTI application.

Pre-activated Licenses

Pre-activated licenses	ETERNITY NENXIP50 purchased after January 1, 2021 and later	ETERNITY NENXIP50 purchased before January 1, 2021	ETERNITY NENX312	ETERNITY NENX416
SARVAM UCS SOHO	Yes	Yes	Yes	Yes
SARVAM VARTA USER5E	40	5	0	0
SARVAM VARTA USER5P	5	0	0	0
SARVAM VARTA USER5C	5	0	0	0
SARVAM VMS SOHO	Yes	No	No	No
SARVAM SMS SERVER SOHO	Yes	No	No	No
SARVAM SMS GATEWAY SOHO	Yes	No	No	No
SARVAM CTI SOHO	Yes	No	No	No

Demo Provision

Demo provision for licensed features is useful when the customer’s system having licensed features cannot be repaired on-site and a standby system needs to be installed or when end users demand to use certain licensed features on trial basis before actually purchasing the license.

Demo Provision allows you to access and use all the licensed features and functionalities[154] supported by the application free of cost for a period of 60 days.

To avail this facility,

- Login as System Engineer.
- Under **Advanced Settings**, click **License Management**.

- Under **Demo Period**, click the **Start** button. The demo period for using the licensed feature in your system starts and the Start button will change to Pause. The demo period is of 60 days.

  After the demo period starts, you can click on the **View Profile** button to view the list of supported features and functionalities.

  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 respective license and activate the license key in your system.

When you default SARVAM UCS, the demo period will not be reset.

How to activate your License

Instructions for Matrix Channel Partners

Your license voucher is 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.

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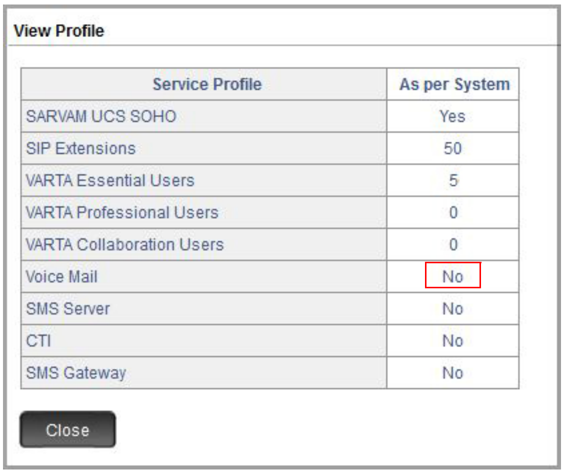
154. In Demo Mode, VARTA Collaboration license will be assigned to all the VARTA clients.

- Login as System Engineer.

- Under **Advanced Settings**, click **License Management**.

  The License Management Page displays the following:

  If you wish to view the features and functionalities currently available to your system, click the **View Profile** button.



- Click **Close**.
- Now, note down or copy the current **License Key** on this page.

- Keep your Current License Key and the License Voucher ready.

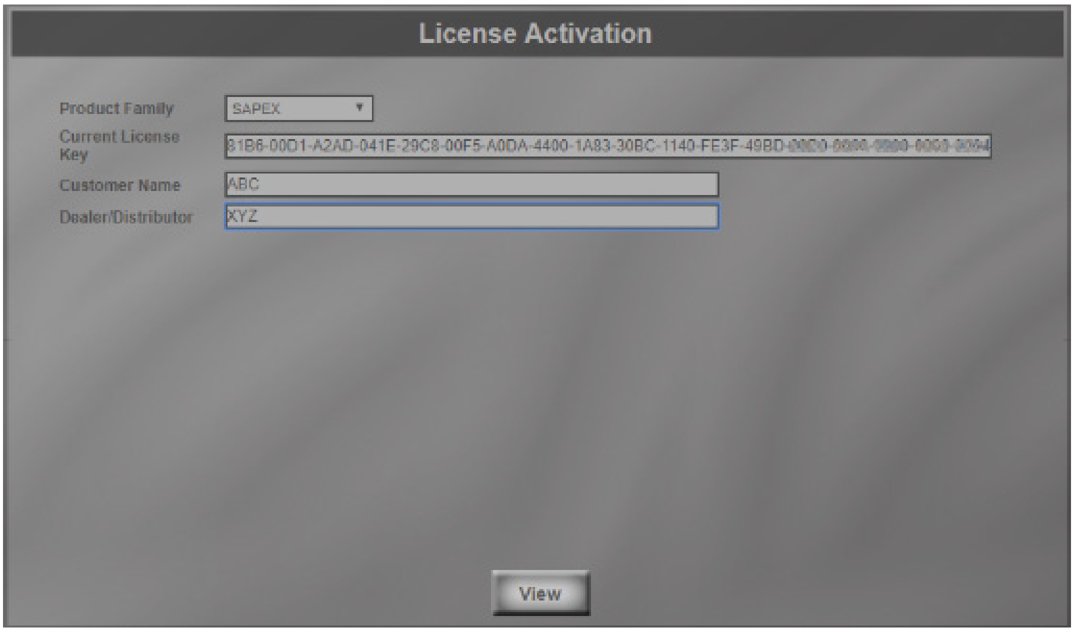
- Open a new window on your browser.

- Enter **http://www.matrixcomsec.com/MatrixLicense/** in the address bar.

- Click **Login to Access**. The **Login to Access** page will open.

- Enter your **User Name** and **Password** provided by Matrix and click **Login**.

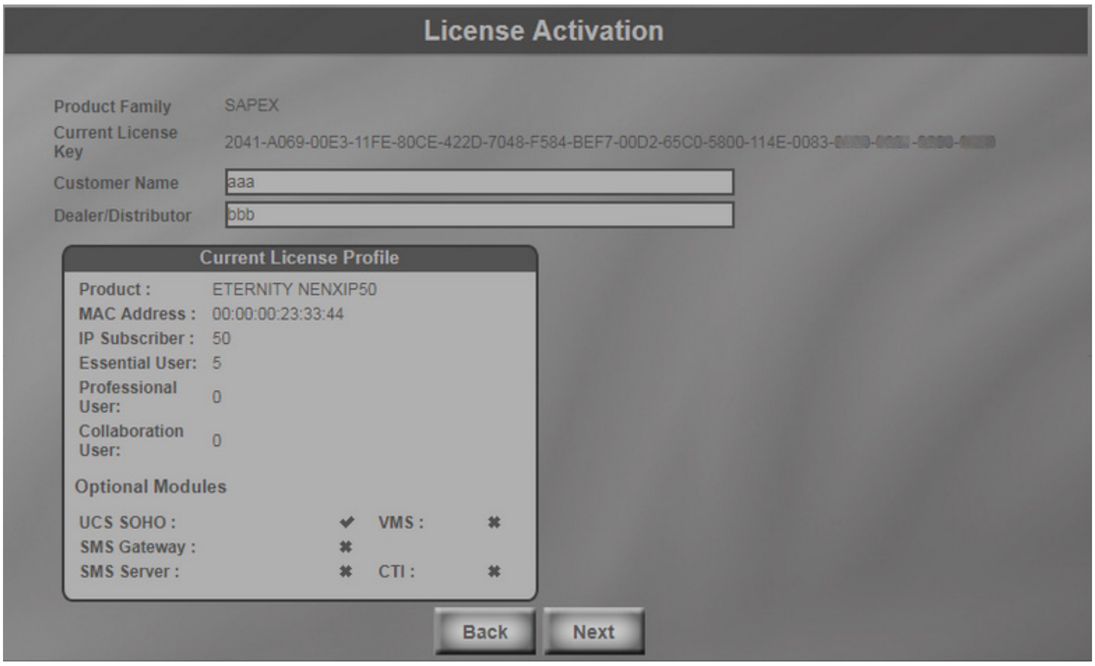
On successful login, the **License Activation** page will open.



- As **Product Family**, select the option **SAPEX**.

- In **Current License Key**, type the Current License Key you noted or paste the key you copied from the License Management page of Jeeves.

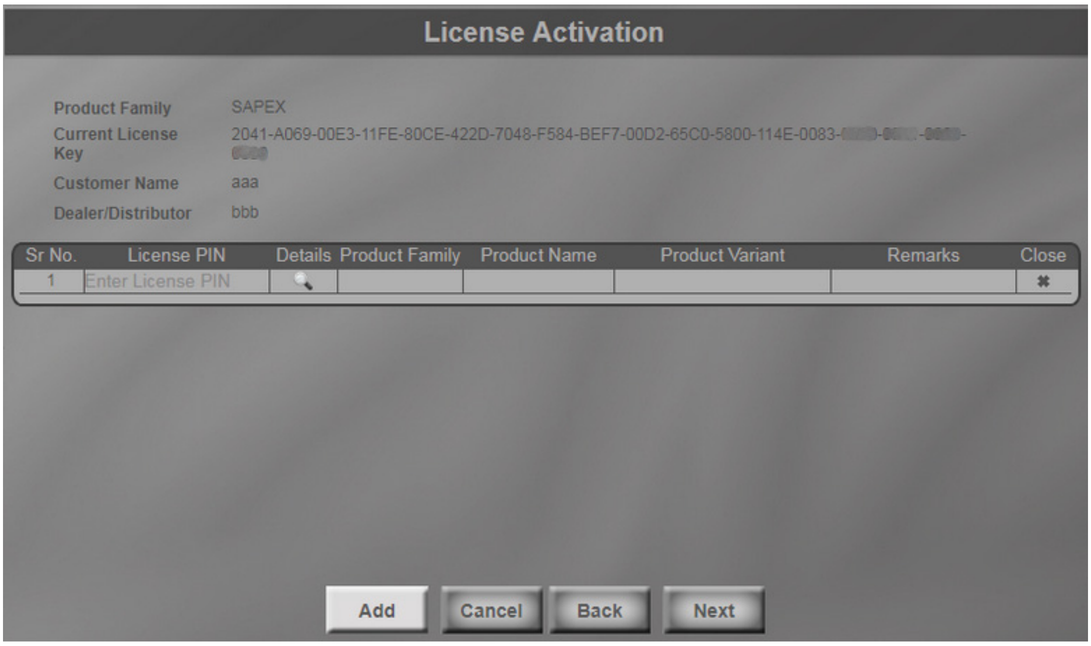
- Click **View**.



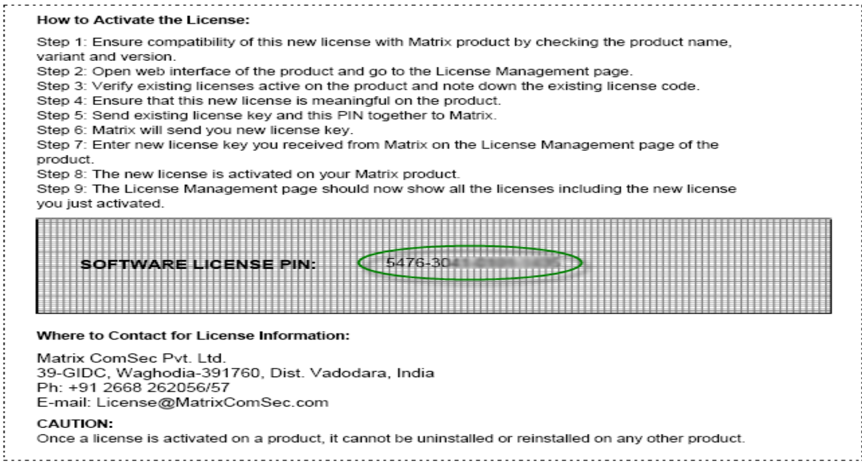
The page will show the current License Profile of the system.

- Click **Next** to continue.

  The **License Activation** page will open.



- In **License PIN** on this page, enter or paste 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.

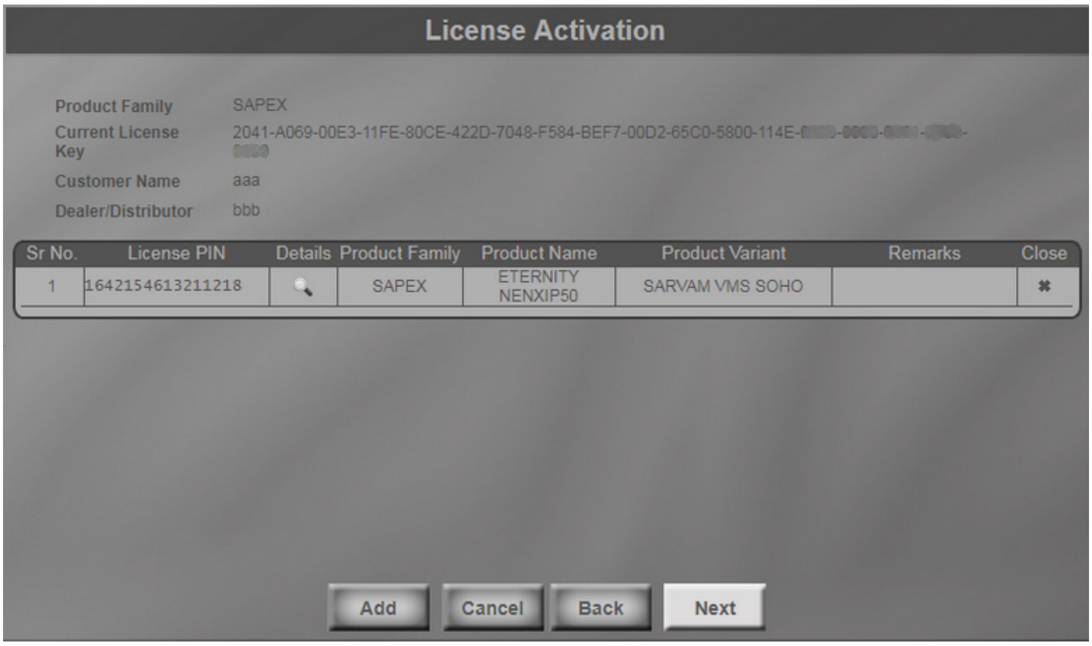
SOFTWARE LICENSE PIN: 5476-30

Where to Contact for License Information:

Matrix ComSec Pvt. Ltd.
39-GIDC, Waghodia-391760, Dist. Vadodara, India
Ph: +91 2668 262056/57
E-mail: License@MatrixComSec.com

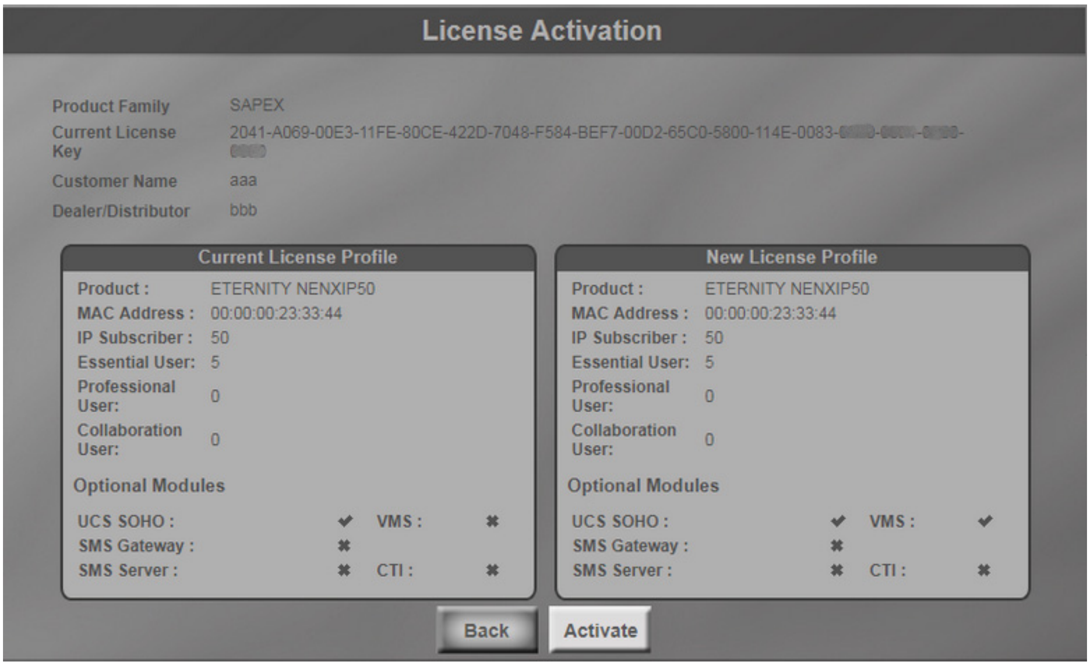
CAUTION:
Once a license is activated on a product, it cannot be uninstalled or reinstalled on any other product.

- Click **Details.**



The details appear in the fields **Product Family**, **Product Name**, **Product Variant**.

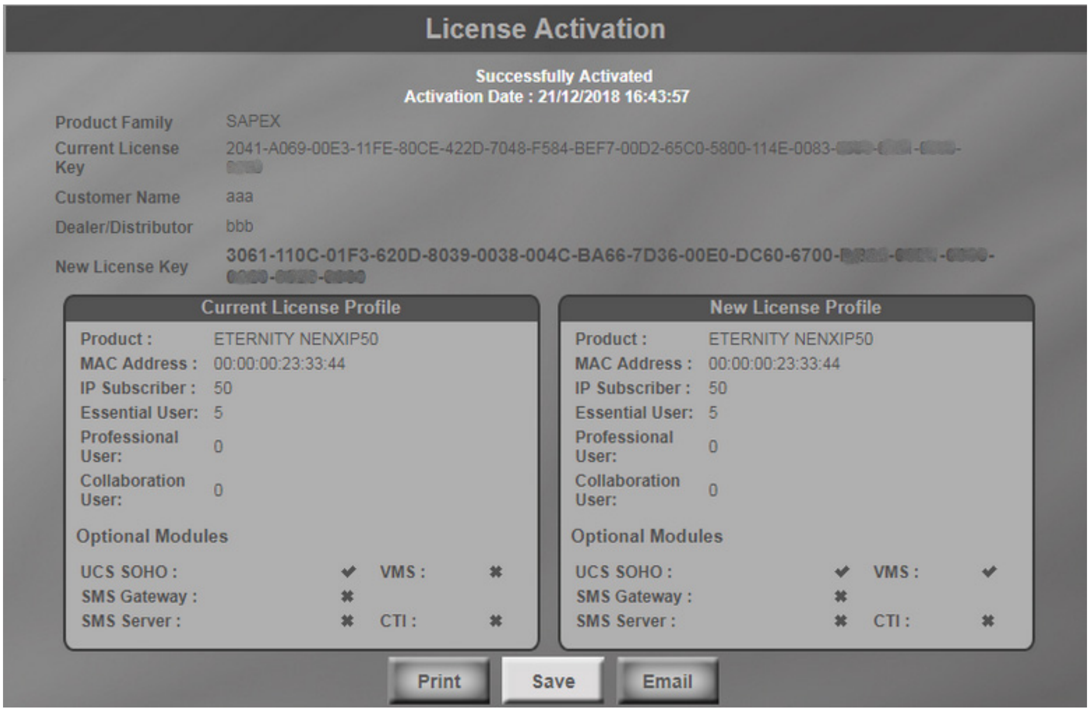
• Click **Next**. Your **Current License Profile** and **New License Profile** will appear on this page.



• Click **Activate** 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.

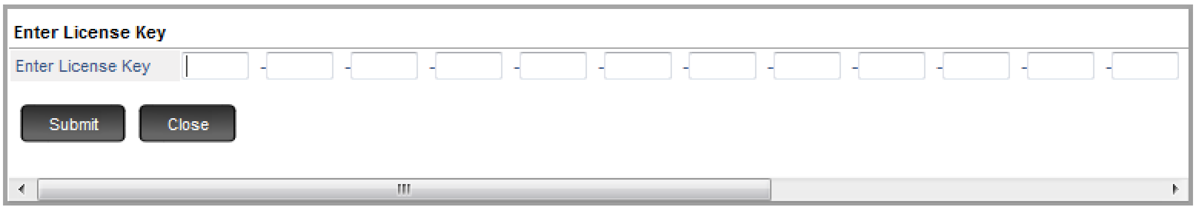
A confirmation mail will also be sent to your e-mail ID (registered with Matrix).



You may **Save**, **Print** or **Mail** this information for your records, by clicking the relevant button.

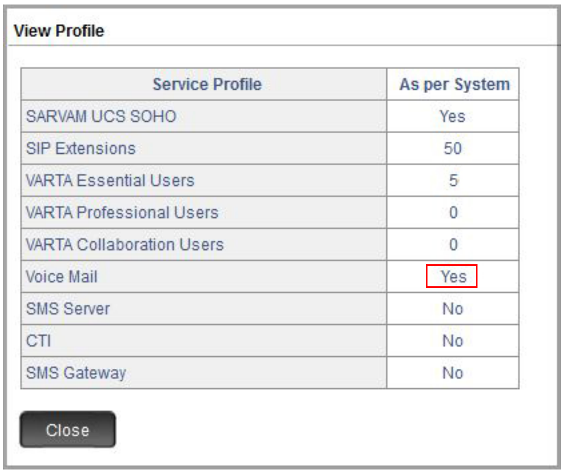
- Note down or copy the New License Key generated on this page.
- Go back to the Jeeves window (or login as System Engineer again, if your session has ended).
- Under **Advanced Settings**, click **License Management**.

- Click the **Enter License Key** button. A new window opens.



- In **Enter License Key**, paste or enter this New License Key.
- Click **Submit**.

  To view the status of licenses activated by you, click the **View Profile** button again.


View Profile

Service Profile	As per System
SARVAM UCS SOHO	Yes
SIP Extensions	50
VARTA Essential Users	5
VARTA Professional Users	0
VARTA Collaboration Users	0
Voice Mail	Yes
SMS Server	No
CTI	No
SMS Gateway	No

Close

  A new window opens which displays the updated service profile.

- To log off, click **Logout**.

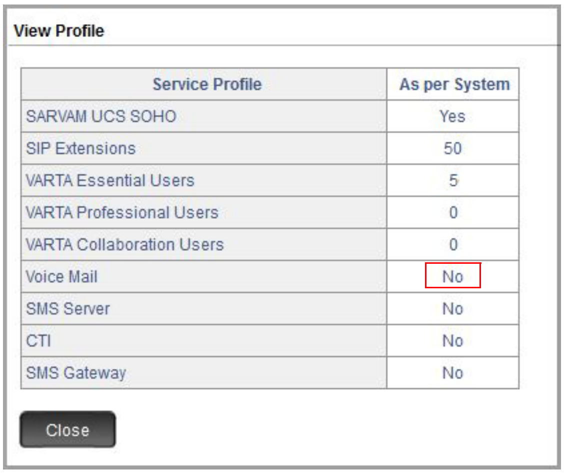
If you are unable to use Online Activation of the License Key or have no internet access, contact the Matrix License Support Centre for assistance in generating the New License key.

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.
- Login as System Engineer.
- Under **Advanced Settings**, click **License Management**.

If you wish to view the features and functionalities currently available to your system, click the **View Profile** button.


View Profile

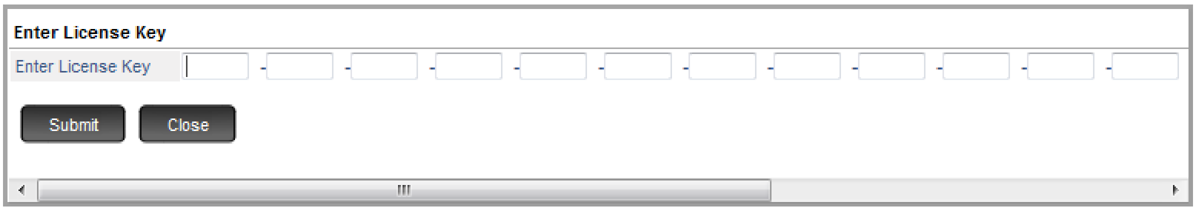
Service Profile	As per System
SARVAM UCS SOHO	Yes
SIP Extensions	50
VARTA Essential Users	5
VARTA Professional Users	0
VARTA Collaboration Users	0
Voice Mail	No
SMS Server	No
CTI	No
SMS Gateway	No

Close

- Click **Close**.
- Note down or copy the Current **License Key** on this page.
- 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.
- Login as System Engineer.

- Under **Advanced Settings,** click **License Management**.

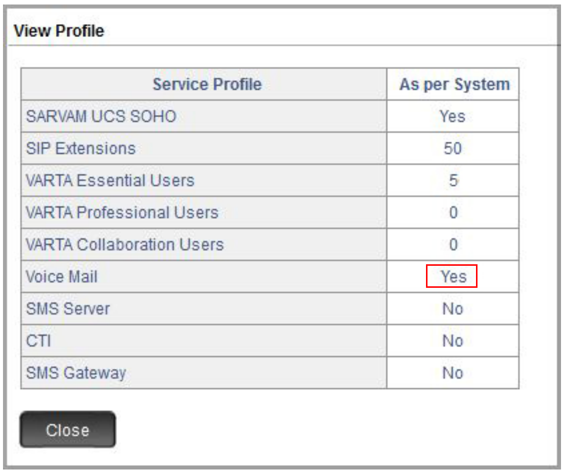
- Click **Enter License Key** button. A new window opens.



- In **Enter License Key**, paste or enter this New License Key as obtained from Matrix.

- Click **Submit.**

  To view the status of licenses activated by you, click the **View Profile** button again.


View Profile

Service Profile	As per System
SARVAM UCS SOHO	Yes
SIP Extensions	50
VARTA Essential Users	5
VARTA Professional Users	0
VARTA Collaboration Users	0
Voice Mail	Yes
SMS Server	No
CTI	No
SMS Gateway	No

Close

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)

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, see “How to configure”.

- 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 LDAP Server Directory Administrator 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

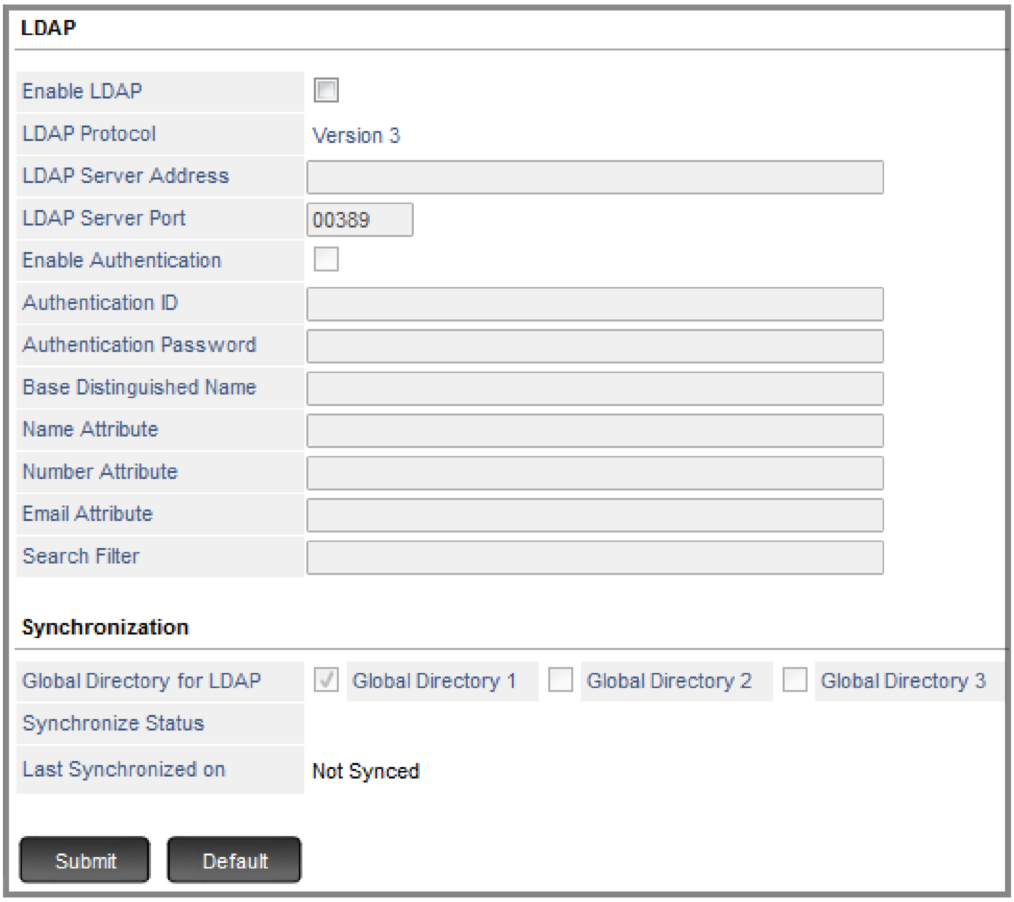
LDAP database consists of various details — first name, last name, common name, multiple phone numbers etc. To distinguish it, each type of detail is assigned a specific attribute in LDAP database. Contact your LDAP Server Directory Administrator to know the detailed list of attributes maintained.

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

- Login as System Engineer.
- Under **Advanced Settings**, click **LDAP**.



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.

- ***Sync Now** and **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**.

How to use

To make an outgoing call to an LDAP contact from the synchronized Global Directory, refer to “Global Abbreviated Dialing”.

Live Call Supervision

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 Extended IP Phone extensions and on SLT extensions having CLI phone.

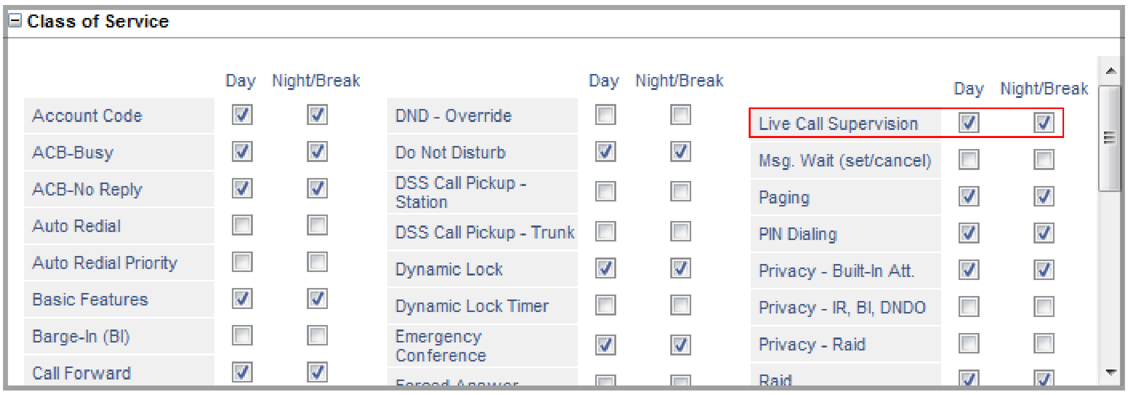
How it works

- A is the supervisor of B.
- A wants to know where B is calling, B 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)" for the Day and Night/Break time, as required.



See "Basic Settings" for instructions on configuring *Class of Service* for different extension port types: "SLT Extensions" and "SIP Extensions".

How to use

For 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

Logical Partitioning is used to restrict the flow of call traffic between PSTN and Private Networks as well as between PSTN and 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 is a legal offence. SARVAM UCS supports Logical Partitioning for this purpose.

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 arising related to this.

How it works

Logical Partition is applied on the Trunk ports. You can change the calling permission, that is, whether to 'allow' or 'restrict' the calls between different trunks, depending on the regulations prevalent in your country.

Depending on the calling permission programmed between the trunks, the system will allow or deny the calls on the trunks.

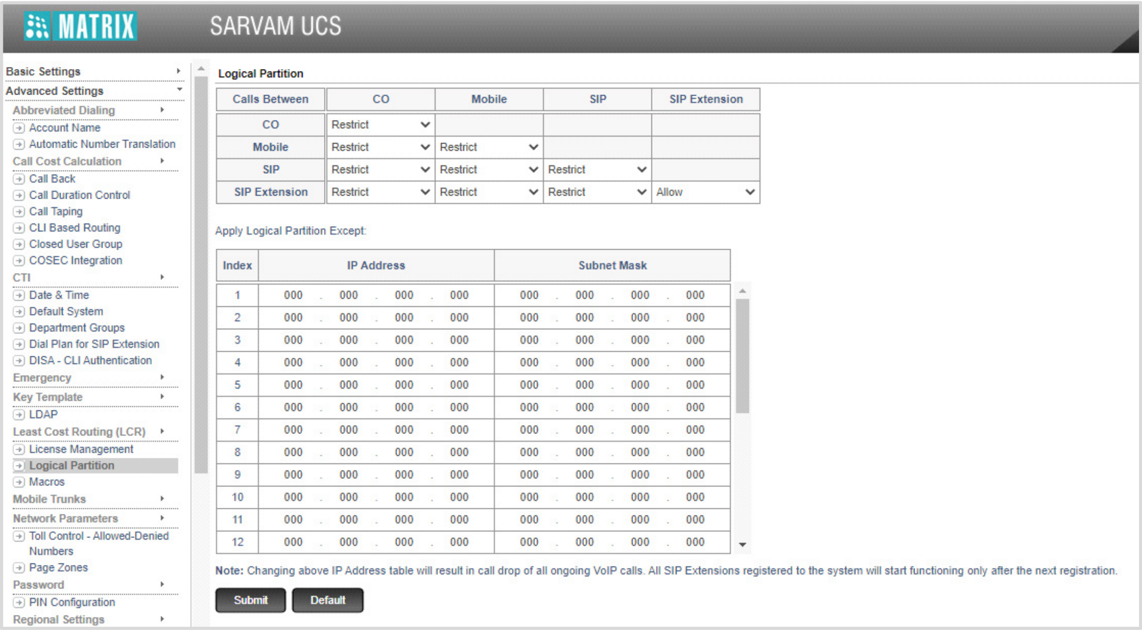
To remove the logical partition for local SIP Extensions within office premises, that is, to allow calls between Internal Extension (SLT, Local SIP Extension and Remote Extension) and External Trunks, you can configure the table defining the IP address range of those local SIP Extensions. These local SIP Extensions will be considered similar to the PBX extensions such as 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 SLT/Local SIP Extensions, but Logical Partition will be applied for the external SIP Extension calls and trunk calls.

How to configure

- Login as System Engineer.

- Under **Advanced Settings**, click **Logical Partition.**



Now, allow or restrict the calls between the different Trunk Port types ans SIP Extensions.

- In the table, configure the IP addresses and subnet masks of those local SIP Extensions/SIP Trunks for which you want to allow the Internal Extension and External (Trunk) calls without checking the logical partition. These local SIP Extensions will be considered similar to the PBX extensions such as SLT. You can configure up to 25 entries in this table.

- Click **Submit.**

- *By default, if India is selected as the Region, calls within and between all types of trunks are restricted, except within SIP (SIP Trunks) and SIP Extensions.*

- *When call permission is restricted between trunks and/or between a type of trunk to SIP/SIP Extension, the 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 Built-In 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, Call Forwarding will not be applied.*

- ***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.*

Macros

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 Extended IP Phone.

You can also assign Macros on SLTs that have special keys.

How it works

- Extension 201, frequently sets Call Forward-All Calls to an external number 26550333.
- To do this, each time, Extension 201 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 201 is an Extended IP Phone, the Macro can be assigned to a DSS key on the phone.
- Instead of dialing this number string, the user of Extension 201 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, a 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.

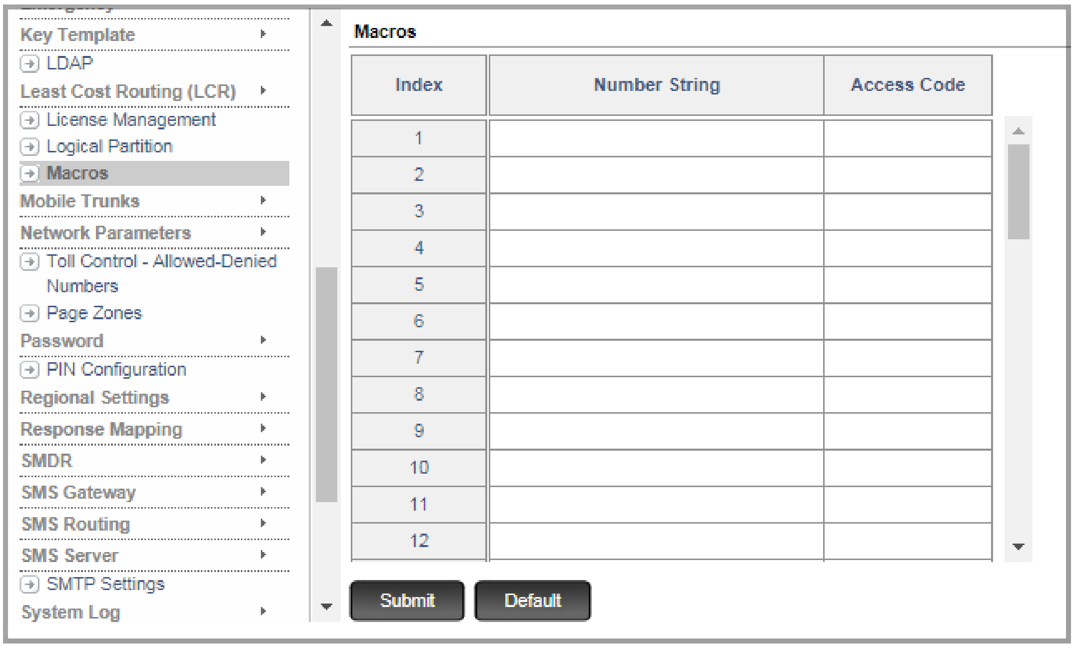
You can create Macros for the strings dialed out using the special keys, whereby the string dialed by each of these special keys is associated with a particular function. For example, the special key for dialing *50 is associated with Call Forward - All Calls to an external number. So, when the extension user presses *50, the system receives this string and takes appropriate action, i.e. 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.

- Login as System Engineer.
- Under **Advanced Settings**, click **Macros.**



Each macro is stored against an index number.

- When creating a Macro for the **Extended IP Phone**,
  - In **Number String**, enter the strings the system should consider as command when the DSS Key on the Extended IP Phone is pressed.
  - Click **Submit**.
  - Now, assign this Macro to a DSS Key on the Extended IP Phone.
  - When you assign a Macro to a DSS Key on the 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 instructions on assigning a keyboard macro to a DSS key of your Extended IP Phone, see “Phone Key Settings” under “Configuring Matrix SPARSH VP248”, “Phone Key Settings” under “Configuring Matrix SPARSH VP310” and “Phone Key Settings” under “Configuring Matrix SPARSH VP330”and “Phone Key Settings” under “Configuring Matrix SPARSH VP510”.

- When creating a Macro for a special Key on a **SLT**,

- In **Number String**, enter the strings the system should consider as command when the special key on the SLT is pressed.

- In **Access Codes**, enter the strings sent by the SLT on pressing the special function key.

  For example, if the SLT sends the string '*53' to SARVAM UCS, when the function key for Alarms is pressed, enter the string **163** (the feature access code for Voice-guided Alarms) in **Number String**, 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.**

Meet Me Paging

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

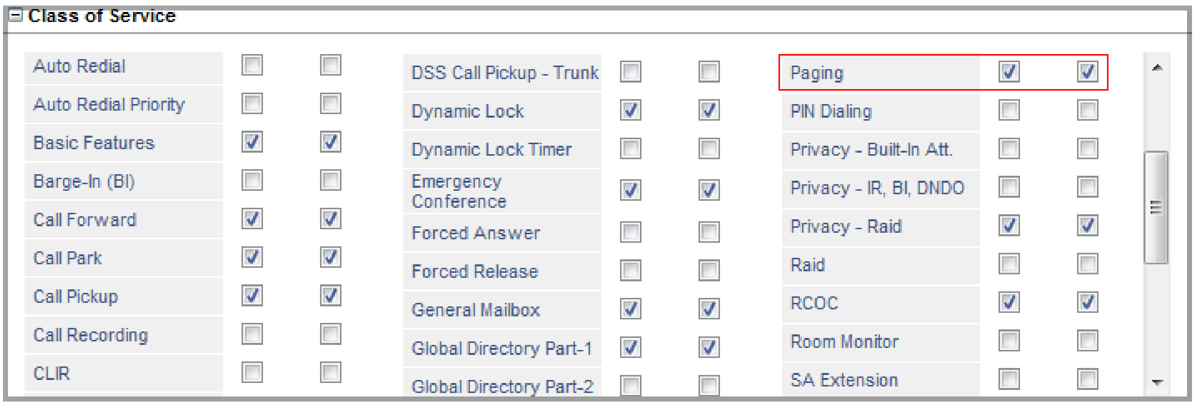
- A calls B’s extension, but B is away from the desk.
- A uses Paging and makes an announcement asking B to call A’s extension.
- To get connected to A’s extension, B may use Meet Me Paging from any extension, while the announcement is being made.
- B dials Meet Me Paging code and A’s number during the 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 “Paging” feature allowed in their Class of Service.



How to use

For 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

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 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 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 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 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 listens 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.

***Voice Mail** has priority over extension **Message Wait** set by extensions. If an extension user has both **Message Wait** and **new Voice mail**, and when the extension user presses the Retrieve Message Wait key or dials the feature access code to Retrieve Message Wait, the call will first be placed to the Voice Mail System. When the extension user presses the Retrieve Message Wait key again, the call will be placed to the extension that set Message Wait.*

Types of Message Wait Indication

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 Indication 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 4 Voice Modules simultaneously. The Voice Module for Message Wait Notification will not be played if there are already 4 being played simultaneously. In this case, Stuttered Dial Tone will be played for Message Wait Indication, when the extension user goes OFF-Hook.

LED Lamp

When the extension user has a 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 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. See "Basic Settings" for instructions on configuring the CoS of different Extension types: SLT, SIP. By default, this feature is enabled in the CoS of all extension types for the Day and Night/Break.

- Select the desired **Message Wait Indication** for each extension in its **Voice mail Settings**. See "Basic Settings" for instructions on configuring the different Extension types: SLT, SIP.

- If you have 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 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 configure the features 'Message Wait' and 'Retrieve Message Wait' on Extended IP Phone extensions. For instructions, see **Phone Key Settings** under "Configuring SIP Extension Settings as per the Device Type" in *SIP Extensions*.

How to use

For 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

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 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 extension is provided on their desks.
- **Virtual Users:** they are extension users of SARVAM UCS who share a physical extension, or may not have any physical extension 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, follow the steps described below.

- List out the Extension Users and Virtual Users and configure them first.
- Make sure that extensions which are to be provided Mobility Extensions have the features Call Forward and DISA enabled in their "Class of Service (CoS)".
- Make a list of External numbers to which the Mobility Extension users will forward their calls. Configure these numbers in the 'Allowed List' of Local, Regional, National and International Numbers, as appropriate.
- The Toll Control assigned to the extension will be applied when a call is forwarded to an external number. Make sure that the extensions which are to be provided Mobility Extension have the required "Toll Control" level for Call Forward to the External Numbers (the numbers you have configured in the Allowed List). Toll Control level is to be configured on the Mobility Extensions.
- Configure the parameter 'External Call Forward for' on the Mobility Extensions. This parameter defines the types of calls 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.
- Configure the parameter 'DISA' on the trunk lines on 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 port.

- Make a list of numbers which the Mobility Extension users will use to access SARVAM UCS from DISA mode. Configure this list of numbers in the "DISA-CLI Authentication Table".

  Configure this list of numbers in the 'Calling Number' field of the Authentication Table. Configure the 'Port Type' and 'Port Number' of the Extension 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 its features from a remote location as described below.

Receiving calls

To receive calls, the Mobility Extension User must set Call Forward on his extension with an external number (mobile number, landline number, etc.) as the destination number.

To make calls ring on the extension and the external number simultaneously, the Mobility Extension Users must activate the Call Forward-Dual Ring feature on their extensions.

The Mobility Extension Users can also choose where they want to receive the calls during a particular time of the day. For example, they can receive calls during a particular time of the day, i.e. Time Zone on their external number and have their calls received by their Voice mail or the Operator or any other number during another Time Zone. To do this, they must set **"Call Forward-Scheduled"** on their extension. 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 Extension.

The Mobility Extension User can also dial codes of the Personal directory and Global directory numbers to use the feature Abbreviated Dialing.

Accessing Features

Mobility Extension Users 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 their 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 extension number [to make call on an extension]
- 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 extension number [to make call on extension]
- Dial ***#0*** on receiving Ring Back Tone (RBT).

To perform Call Transfer-On Busy Extension

- 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 Extension 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 Pickup

To pick up the call of same Call Pickup-Group

- Make a call on DISA enabled Trunk from Mobility Extension User.
- After receiving dial tone, dial ***4.***
- The call ringing on the extension of the same Call Pick up-Group will get connected with Mobility Extension User.

To pick up the call of a selected extension

- Make a call on DISA enabled Trunk from Mobility Extension User.
- After receiving dial tone, dial ***12-Extension Number.***
- The call ringing on the dialed extension 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 Extension is busy

To make a call on an internal extension 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-Extension/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-Extension/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-Extension/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-Extension/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.*

Music on Hold (MOH)

The music played to extension users and external callers who are put on hold is called Music on Hold (MoH).

How it works

A Voice Module assigned to 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.

*Music-on-Hold will also be played to an extension, when the option **Route to Operator** is selected as the **Alarm Notification Type** for the extension, and the Operator extension is busy, when the extension goes Off-hook to answer the Alarm call.*

*If your 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 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.

Multi-Stage Dialing

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,

- If the Extension users are allowed to make international calls using calling card from the trunk CO 1, you must configure “Automatic Number Translation” table on the CO 1 trunk.
- The Automatic Number Translation table consists of Dialed Number Strings, Strip Digits and Add Prefix columns.
- In Dialed Number Strings, 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 (W) between the calling card server number and the PIN number.

  You must also insert a delay by configuring the Pause Timer after the PIN 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 - 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 only the desired destination number, 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

To be able to use Multi-Stage Dialing, you must configure the following:

- Automatic Number Translation table on the desired trunks. For instructions on configuring ANT table on different Trunk types, see ***Automatic Number Translation (ANT)*** under “CO Trunks”, “Mobile Trunks” and “SIP Trunks”.

- Configure the **DTMF Out Dial** parameter on the desired trunks. For instructions, on configuring DTMF Out Dial on different Trunk types, see ***Hardware Settings*** under “CO Trunks”, “Mobile Trunks” and “SIP Trunks”.

- If required, you may also configure the **Call Proceeding Tone for Multi-stage Dialing**. For instructions, see “System Parameters”.

Mute

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 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 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

OFF-Hook Alert

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, 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 text 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 inform the extension user to place the handset of the phone correctly.
- If the extension phone is a SLT, OFF-Hook Alert will be given to the Operator phone only. (The Operator phone must be an Extended IP Phone).
- If the extension phone that is OFF-Hook is Extended IP Phone, 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

- Login as System Engineer.
- Under **Advanced Settings**, click **System Parameters**.

- Select the **Give Off-hook Alert to Operator** check box.

- Click **Submit.**

One Touch Transfer

One Touch Transfer feature of SARVAM UCS enables you to transfer an ongoing call from one extension to another 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 SIP Extension (Matrix Extended IP Phones or Mobile Clients) only. The fixed destination extension number can be a SLT or a SIP Extension.

How it works

- There is an ongoing call between extension A and B.
- A wants to transfer the ongoing call to extension C.
- A presses the DSS Key assigned to One Touch Transfer (Extension C's number is configured as the fixed destination extension number for One Touch Transfer).
- Extension C starts ringing. A answers the call ringing on extension C.
- Speech is established between extension C and B.
- Extension A is disconnected.

- *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

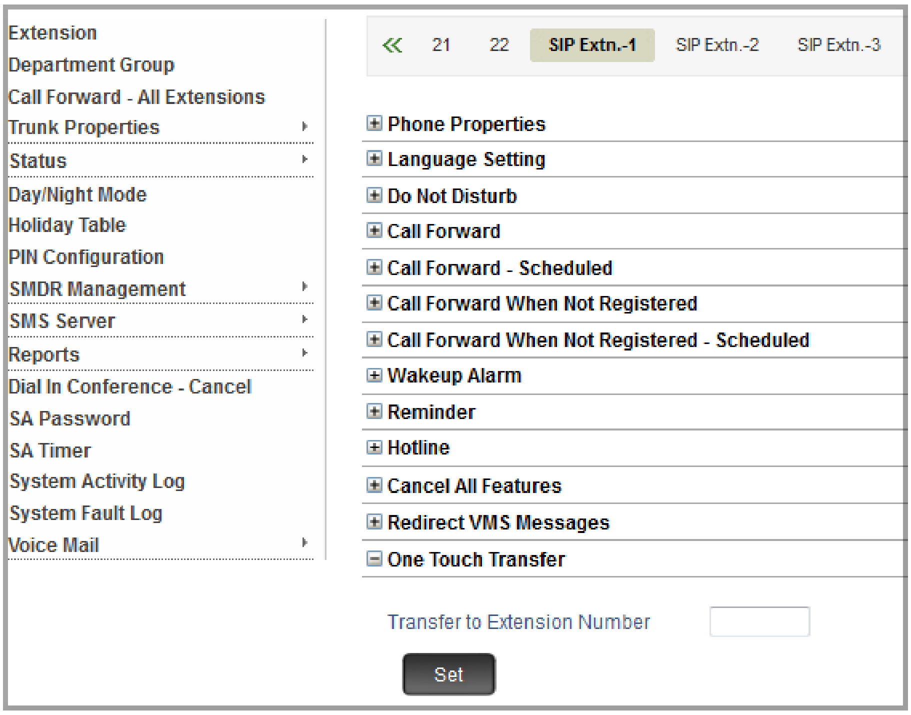
To access this feature,

- Make sure the Basic Features are enabled in the Class of Service assigned to you. For instructions, see *Class of Service* in "SLT Extensions" and "SIP Extensions".
- Make sure you have configured a DSS Key for One Touch Transfer. For instructions on configuring DSS Key, see the topic "Phone Key Settings" in "Configuring Matrix SPARSH VP248", "Phone Key Settings" under "Configuring Matrix SPARSH VP310", "Phone Key Settings" in "Configuring Matrix SPARSH VP330", "Phone Key Settings" under "Configuring Matrix SPARSH VP210" and "Phone Key Settings" under "Configuring Matrix SPARSH VP510".
- Configure the fixed 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

- Login as System Administrator.
- Click **Extension**.

- Now, click on the desired **Extension Number** tab.
- The extension users details appear on your screen.
- Click **One Touch Transfer** to expand.



- In **Transfer to Extension Number**, enter the fixed 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 Fixed 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 fixed destination extension number on which you want the call to be transferred.
- You hear Confirmation tone.

How to use

For Extended IP Phone Users

To use One Touch Transfer

- During an ongoing call, press the DSS Key assigned to One Touch Transfer.
- The fixed destination extension number configured for One Touch Transfer starts ringing.
- Answer the ringing call.
- You will be in speech with the remote party.

Paging

Paging allows you to make announcement to a group of extension users by just lifting the handset of your phone and dialing a code. In this way you can deliver a message to a mass of people at once.

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 the Matrix Extended IP Phone, or any Standard SIP Phone as their extensions, you can use paging instead of calling them up one by one.

- *You can start paging from SLT or any SIP Extension. However, Paging is possible only on SIP Extensions, which may be Extended IP Phones 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 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. You can create 12 different Page Zones.
- The SIP extensions which are to be paged must be included in the Page Zones. You can create 12 different Page Zones. Each Page Zone accommodates up to 16 SIP extensions.
- Paging must be enabled in the Class of Service allowed to the SLT and SIP extension from which this feature is to be used.

The Process

- A SLT/SIP extension user having Paging in the Class of Service, dials the Access code for Paging and the Number of the Page Zone to which s/he wants to make the announcement.
- The system hunts for the IP Phones in the Page Zone which are free and activates the speaker of these SIP Phones.
- The calling SLT/SIP extension user makes his/her announcement.
- All SIP extensions in the Page Zone can hear the announcement. But they cannot speak to the calling SLT/SIP extension user, as the mic of their phones is muted.
- 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 SLT/SIP extension goes ON-Hook after the announcement.

- The system deactivates the mic of the SIP extensions.

How to configure

For this feature to work, you must create Page Zones and enable this feature in the “Class of Service (CoS)” of the extensions which are to be allowed to page other extensions.

Creating Page Zones

Decide how many Page Zones you want to create. You can create upto 12 different Page Zones. Configure desired name for each Page zone. You can include maximum 16 SIP extensions in each Page Zone.

For each Page Zone, decide and assign the SIP extensions.

On a sheet of paper, create a two-column table, as shown below. Write the numbers of the SIP extensions you want to include in the page zone in the member column.

Page Zone 1

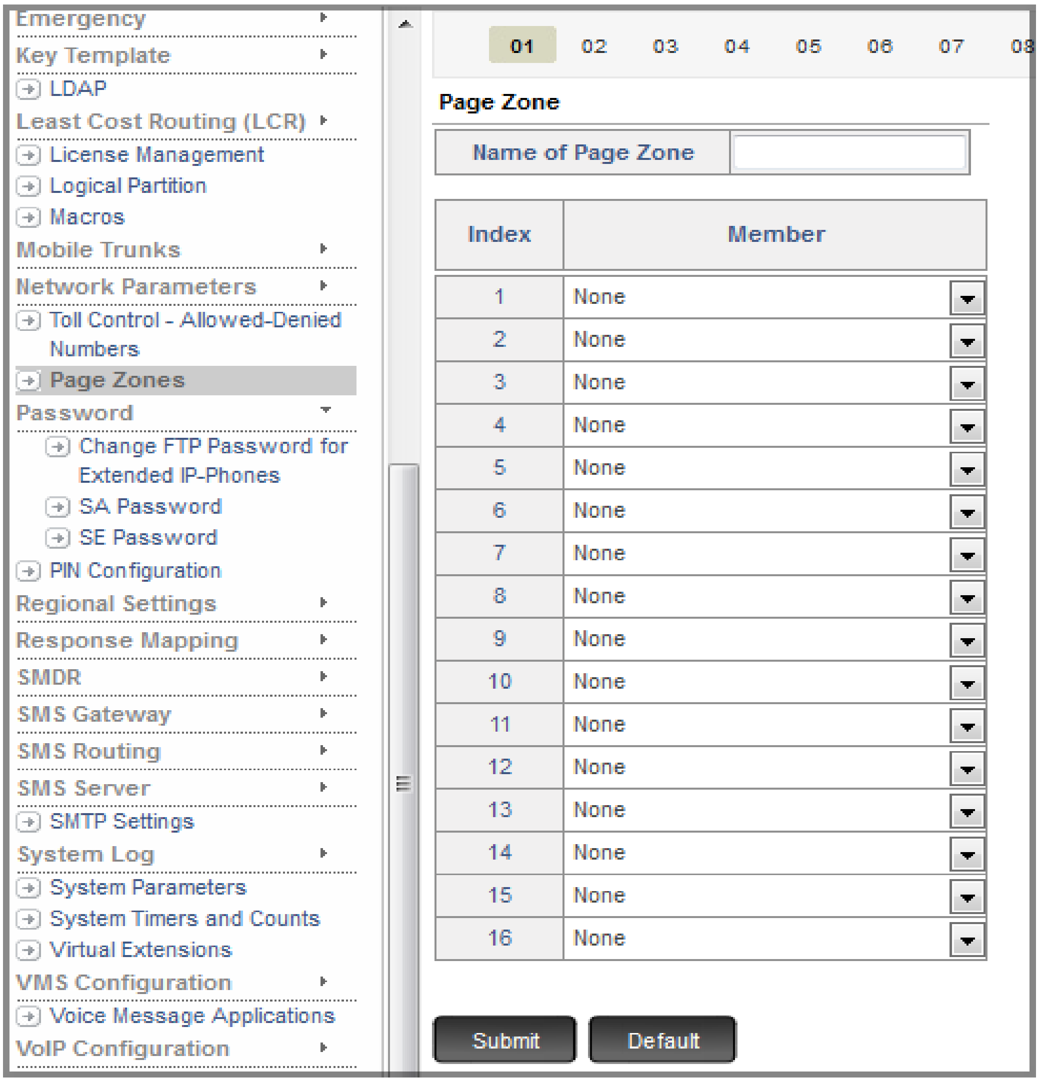
Index	Member
1	SIP Ext-2
2	SIP Ext.-4
3	SIP Ext.-5
:	
:	
:	
16	

Now, program Page Zone using Jeeves.

To configure Page Zones,

- Login as System Engineer.

- Under **Advanced Settings**, click **Page Zones**.

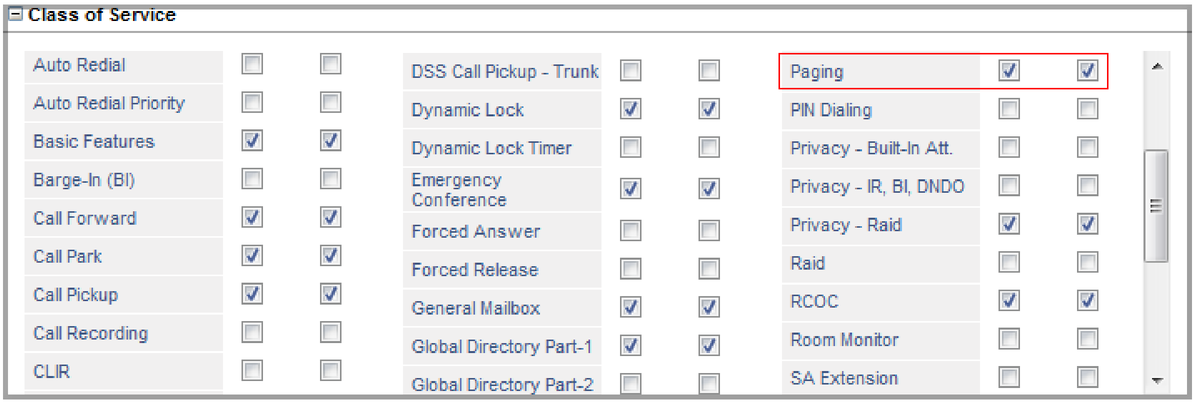


- The Page zone tabs from 1 to 12 appear on this page.
- Configure **Page Zone 01**.
- Enter desired name in the **Name of Page Zone**. Default: Blank.

  Name can be of maximum 16 characters. All ASCII characters (except Less than **<,** Greater than **>** and Double Quote **"**) are allowed.
- For each Index number, select a **Member** extension. Select the extension number you want to include in this Page Zone. Default: None.
- Click **Submit**.
- Similarly, you can create another Page Zone.

Configuring Paging in Class of Service

By default, Paging is enabled in the Class of Service for the Day and Night/Break. So, all extensions of SARVAM UCS can page.



How to use

It is possible to page from SLT and SIP Extensions.

For 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.
- On the prompt ‘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.

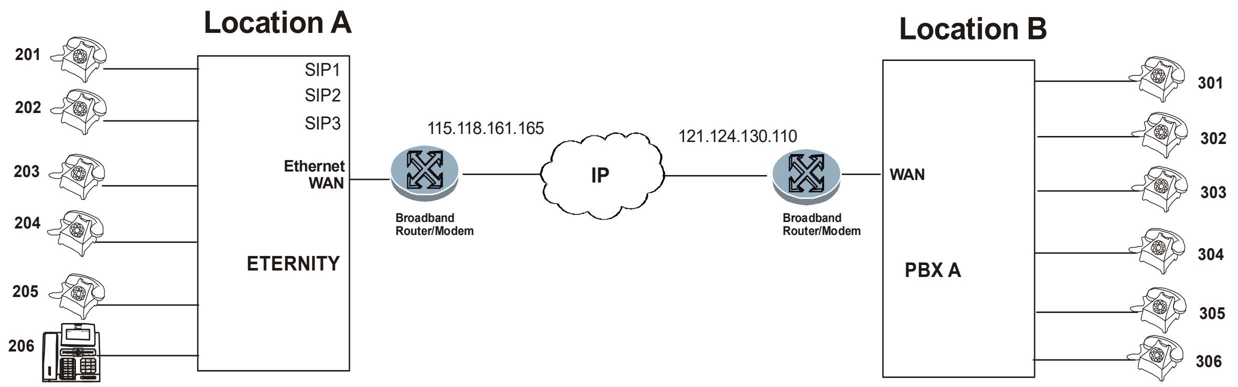
Peer-to-Peer Calling

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 an IP network.
- SARVAM UCS is installed at Location A.
- The PBX 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 PBX.
- **At Location A**, you need to do the following configuration in SARVAM UCS:
    - Select a SIP Trunk to be used for this application and enable it. For example, SIP 1.
    - 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 instructions on configuring the above parameters, see "SIP" under "SIP Trunks".

In the Router, you must configure the same SIP and RTP Ports as configured in 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 “Trusted IP Address/es” under “SIP Trunks”.

- Set the **Send CLI** option on the SIP trunk as **Calling Party Wise**. See “Send CLI in FROM field” under “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:

Closed User Group

Index	Route Code	Strip Digit Count	Self Route	Dialed Digit Count	LCR	Route using Trunks	Apply Toll Control	Apply Call Cost
1	3	0	[ ]	3	OFF	SIP-1,	[ ]	[ ]
2		0	[ ]	99	OFF	Double-click to select...	[x]	[x]
3		0	[ ]	99	OFF	Double-click to select...	[x]	[x]
4		0	[ ]	99	OFF	Double-click to select...	[x]	[x]
5		0	[ ]	99	OFF	Double-click to select...	[x]	[x]
6		0	[ ]	99	OFF	Double-click to select...	[x]	[x]
7		0	[ ]	99	OFF	Double-click to select...	[x]	[x]
8		0	[ ]	99	OFF	Double-click to select...	[x]	[x]
9		0	[ ]	99	OFF	Double-click to select...	[x]	[x]
10		0	[ ]	99	OFF	Double-click to select...	[x]	[x]

Submit | Default | Default One

  - In **Route Code**, enter the extension numbers of the PBX 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, ‘**3**’.

  - In **Route using Trunks**, select the SIP trunk number to be used for routing the Peer-to-Peer calls. In this case, SIP 1.

  - 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.

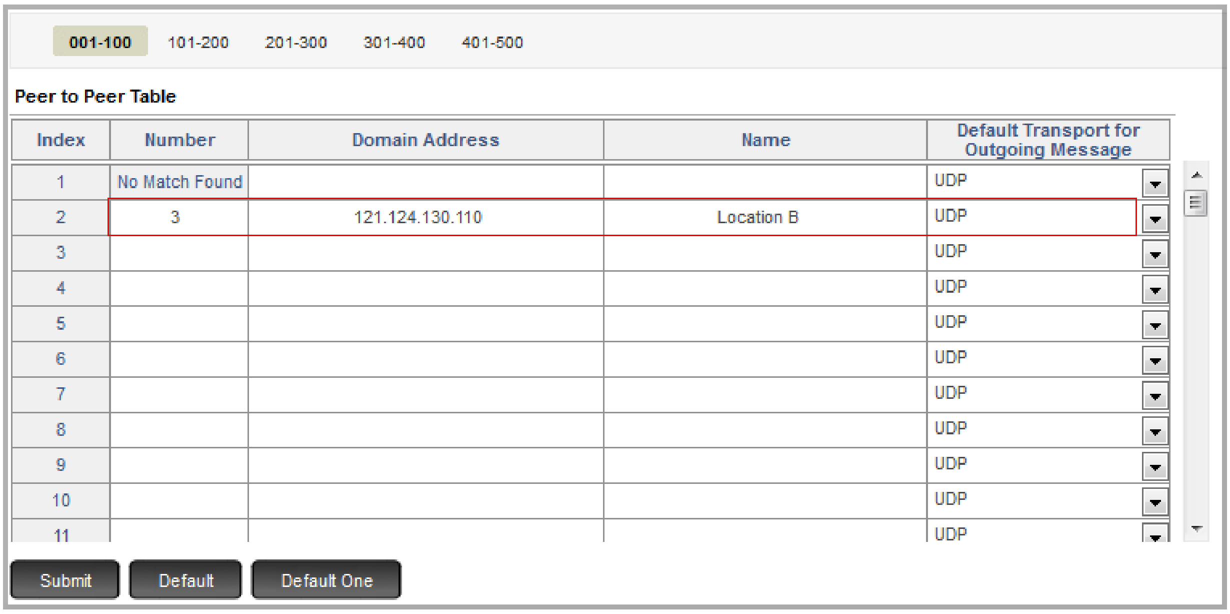
  - By default, **Apply Call Cost** check box is enabled. Clear this check box as for Peer to Peer calls you do not require the call cost calculation.

    See “Closed User Group (CUG)” for more details.

- 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, Default Transport for Outgoing Message.

  At location A, you would have to configure the Peer-to-Peer table as follows:



001-100 101-200 201-300 301-400 401-500

Peer to Peer Table

Index	Number	Domain Address	Name	Default Transport for Outgoing Message
1	No Match Found			UDP
2	3	121.124.130.110	Location B	UDP
3				UDP
4				UDP
5				UDP
6				UDP
7				UDP
8				UDP
9				UDP
10				UDP
11				UDP

Submit Default Default One

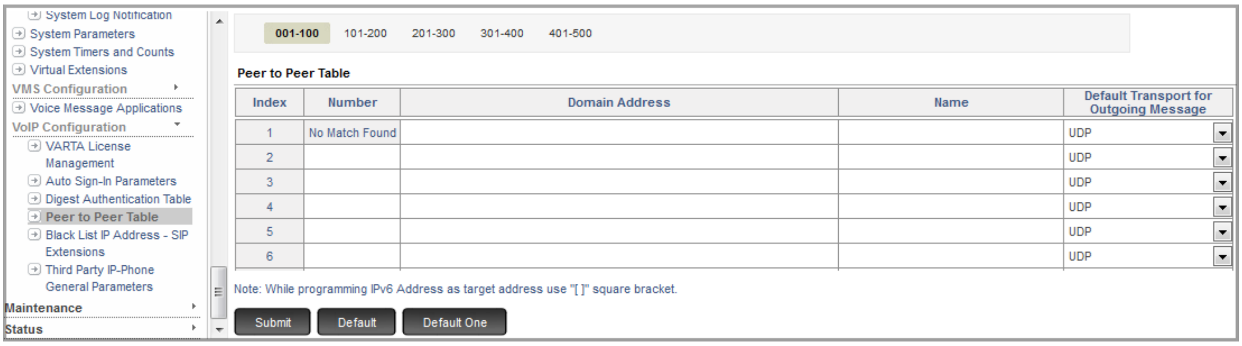
  - In the **Number** column of the Peer-to-Peer table, enter the digit you configured as **Route Code** in the CUG Table for calling the extension numbers of the PBX at Location B. In this case ‘3’.
  - In **Domain Address**, enter the **IP Address** of the WAN Port of the Router at Location B.
  - In **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 PBX.
- When an extension user 201 of SARVAM UCS at Location A dials 301, an extension number of PBX B, the system 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 the PBX B. With suitable configuration done in PBX B, the call will be routed to the desired destination i.e. extension 301.

How to configure

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 “SIP Trunks”.

To configure the Peer-to-Peer Table,

- Login as System Engineer.
- Under **Advanced Settings**, click **VoIP Configuration**.
- Click **Peer-to-Peer Table**.



The first entry is reserved for No Match Found.

- In **Number**, enter the number string—prefix or entire number—that will be dialed. In this case, it would be the Route Code in the CUG Table for calling the remote extension numbers. The number string must not exceed 8 characters. 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 Address may consists of up 48 characters (maximum). Default: Blank.

  The Domain Address can also be in the form of Address: Port number.
- In **Name**, enter a name to identify the number string you configured. The name may consist of 24 characters (maximum). Default: Blank.

  The name you configure here will not be used in SIP signaling.
- In **Default transport for Outgoing Message**, select the option for transporting outgoing SIP messages. You can select UDP, TCP or TLS. Default: UDP.
- Click **Submit**.

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 50 PINs for each type of toll control.

Calls made using PIN can be logged in 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 followed by his/her PIN. As soon as SARVAM UCS detects a valid feature access code and PIN, it gives dial tone to A. Now, user A can dial the desired international number.

2. User A can directly dial an international number from any extension that has PIN Dialing enabled in its CoS. 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.

   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. To know more, refer to "System Timers and Counts".*

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. For instructions, see "Class of Service (CoS)".

- Make sure, LCR is enabled on the Trunks assigned to the Extensions from where you want to make outgoing calls using PIN. For instructions, see *Select Trunks for Outgoing Calls* in the "SLT Extensions" and "SIP Extensions" under *Basic Settings*.

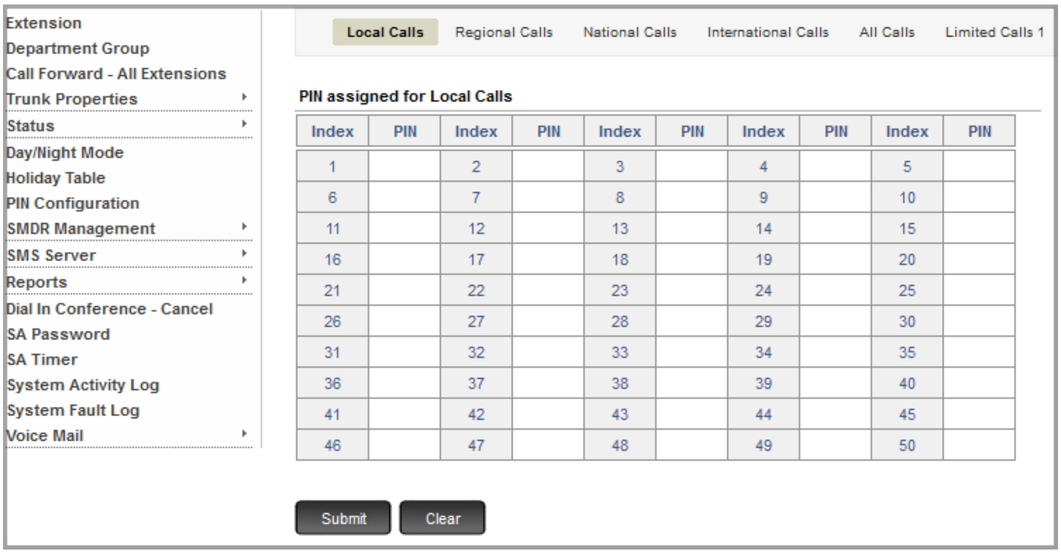
- If you want to log outgoing calls made using PIN, make sure the *Store Outgoing Calls* parameter is enabled on the desired extensions. For instructions, see *SMDR Storage* in the "SLT Extensions" and "SIP Extensions" under *Basic Settings*.

- Configure the PIN table for each type of Toll Control level from the SA Mode.

- Select the Outgoing Trunks and the type of LCR for the PIN's, for each Toll Control Level from the SE Mode. See "To configure Trunks and LCR".

To configure PIN Table,

- Login as System Administrator.
- Click **PIN Configuration**.



- Select the desired Toll Control level.

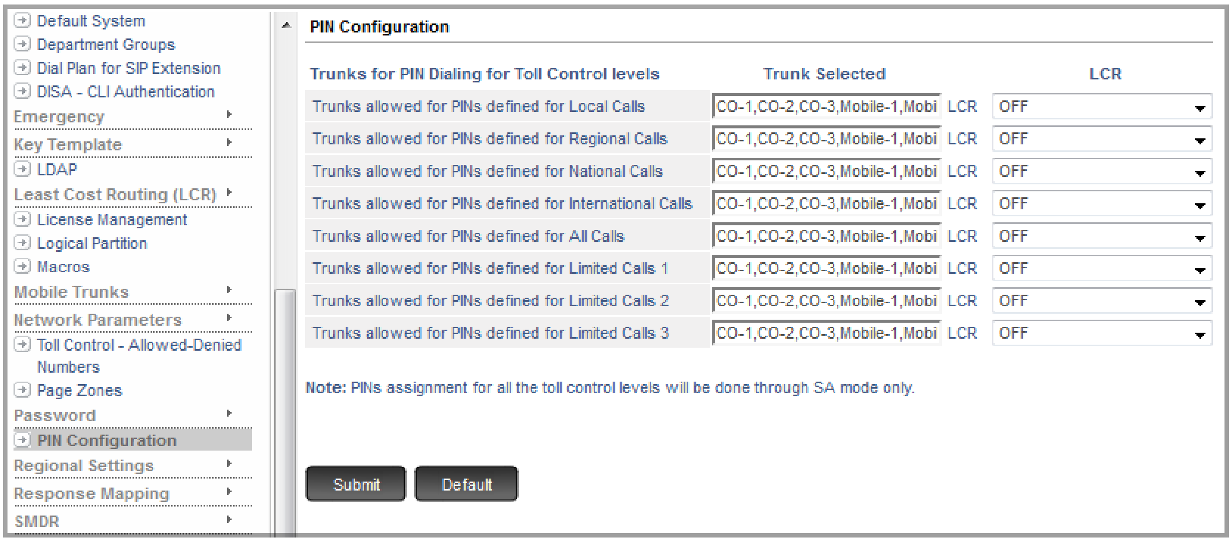
  Each Toll Control level includes the Allowed and Denied numbers as configured by the SE. See "Toll Control" for more details.

- Configure the PIN at a desired index. 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.*

To configure Trunks and LCR

- Login as System Engineer.
- Under **Advanced Settings**, click **PIN Configuration**.



Under **Trunks for PIN Dialing for Toll Control levels**,

- For each Toll Control level, select the desired Trunks.
- For each Toll Control level, select the desired LCR type.

  Make sure, you have configured the LCR table for the selected LCR type. See “Least Cost Routing (LCR)” for more details.

How to use

For Extended IP Phone Users

To use PIN Dialing from any extension:

- Go OFF-Hook.
- Press DSS Key assigned to the ‘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 SMDR Storage of the desired extension.
- enable **Print Calls made using PIN** and set the PIN range in **Calls made using PIN** under **Calls** in the **Outgoing Call Report** from the SA Mode.
- 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 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.

Power Fail Transfer

When the power supply to SARVAM UCS fails, SLT 1 is automatically connected to the CO1. The PBX switches from normal functioning mode to the Power Failure Connection mode. Only CO1 trunk can be used during a power failure. Outgoing calls can be made from SLT1 extension as well as incoming calls on the CO1 trunk can land on this extension. When power is resumed and normal functioning of the system is restored, calls on the Power Failure Connection remains unaffected until they are terminated.

Feature Interactions

Forced Call Disconnection: If the extension user is in conversation over PFT port, when the power supply to SARVAM UCS fails and the power supply is restored during the ongoing conversation, then other extension users cannot invoke Forced Call Disconnection on the PFT SLT or CO port.

If an extension, that has Forced Call Disconnection enabled in its CoS and a higher priority than the called extension, invokes Forced Call Disconnection on PFT port, then the system will provide error tone and display the message *Forced Call Rel Not Alwd* on the phone LCD.

ACB (Auto Call back): If the extension user is in conversation over PFT port, when the power supply to SARVAM UCS fails and the power supply is restored during the ongoing conversation, then the system will allow other extension users to set ACB on the PFT SLT or CO port.

RAID: If the extension user is in conversation over PFT port, when the power supply to SARVAM UCS fails and the power supply is restored during the ongoing conversation, then the system will not allow other extension users to invoke RAID on the called PFT SLT or CO port.

If the calling extension has RAID enabled in the CoS and has a higher priority than the called PFT extension/trunk and it invokes RAID, then the system will give error tone and display message *Raid Not Allowed* on the phone LCD.

Barge - In: If the extension user is in conversation over PFT port, when the power supply to SARVAM UCS fails and the power supply is restored during the ongoing conversation, then the system will not serve Barge-In request on the busy PFT SLT.

Interrupt request: If the extension user is in conversation over PFT port, when the power supply to SARVAM UCS fails and the power supply is restored during the ongoing conversation, then the system will not serve Interrupt Request on the busy PFT SLT.

Trunk reservation: If the extension user is in conversation over PFT port, when the power supply to SARVAM UCS fails and the power supply is restored during the ongoing conversation, then the system will allow other extension users to set Trunk reservation on the busy PFT CO port.

Presence

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/her desk for an indefinite period, but does not want to use Call Forward or set Do Not Disturb. S/he wants to indicate to callers about his/her 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 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, 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 “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 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 SIP Phone, 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 a 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 SIP extension user sets 'Present' the letter 'A' will disappear from the phone's display.

   When any other 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 other extensions.

4. **On the Phone:** When an extension user sets 'On the Phone', the system will display this message to 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 SIP extension users call another extension, the name of the called extension is displayed on the calling SIP extension. Now, if the check box **'Display Presence status during call on Extended IP Phone**' is enabled in the System Parameters, when Extended IP Phone extension users call another extension, the calling Extended IP Phone extensions will be displayed the 'presence' status message published by the called extension[155].
- 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 ETERNITY 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 “SIP Extensions”.

How to configure

This feature involves the configuration of the following parameters:

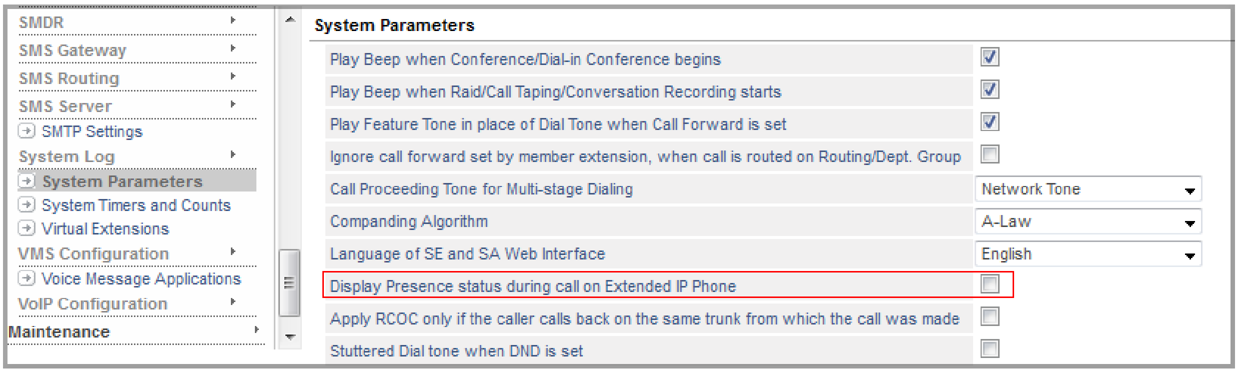
- **'Display Presence status during call on Extended IP Phone' check box:** Extended IP Phone extension users will be able to view the presence status for the called extension only if this check box is enabled in the System Parameters.
- **PUBLISH:** SIP extension users who want to publish their presence 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.
- **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.

155. *SIP extension users can also dial a feature access code and the number of the extension to see the status of that extension on their SIP Phone. But this would not be required, if the 'Display Presence status during call on Extended IP Phone ' check box is enabled in the System Parameters.*

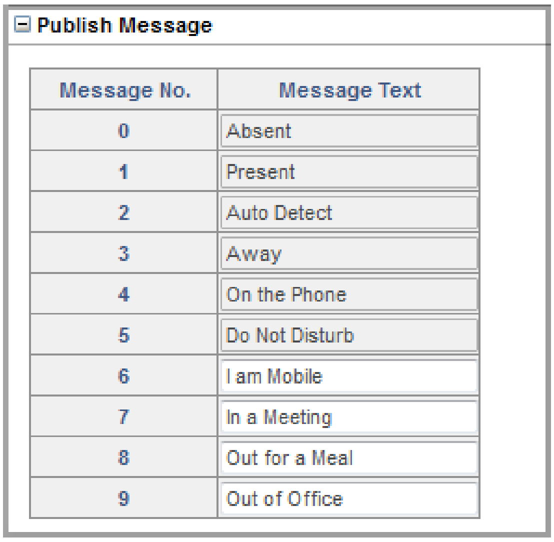
- **Publish Messages:** It is possible to customize the Publish Messages listed above from 6 to 9 viz.: 'I am Mobile', 'In Meeting', 'Out for Meal', 'Out of Office'.

To configure,

- Login as System Engineer.
- Under **Advanced Settings**, click **System Parameters**.
- Click **System Parameters** to expand.
- Enable the check box **Display Presence status during call on Extended IP Phone**.



- Click **Submit**.
- Click **Publish Message** to expand.

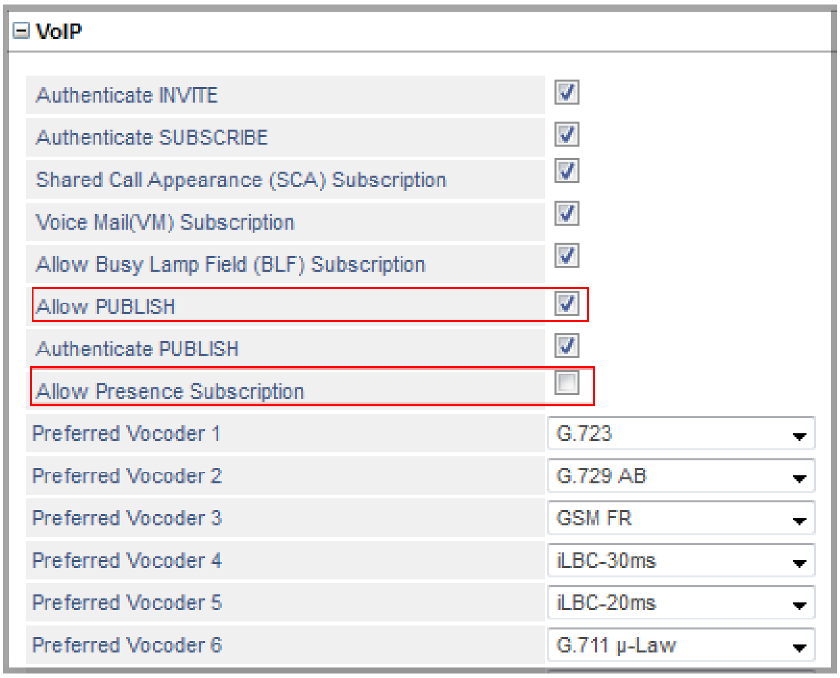


Publish Message

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 a Meeting
8	Out for a Meal
9	Out of Office

- 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**.
- Now, click **SIP Extensions** under **Basic Settings**.

- Go to the desired SIP Extension number for which you want to enable the features **PUBLISH** and **Presence Subscription**.
- Click **VoIP** to expand.



By default Allow Presence Subscription is disabled. To enable PUBLISH and Presence Subscription, enable **Allow PUBLISH** and **Allow Presence Subscription** check boxes.

- Click **Submit**.

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 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, i.e. 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

Index No.	Meaning
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
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

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, 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 will be applicable automatically.

The Preset Call Forward feature of SARVAM UCS offers the following forwarding options:

- **When Busy** - calls are forwarded to the selected destination when the called party's phone is busy.
- **When No Reply** - calls are forwarded to the selected destination when the called party does not answer the phone before the expiry of Call Forward No-Reply Timer.
- **When Busy or No Reply** - calls are forwarded to the selected destination 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 will wait for the Call Forward No-Reply Timer to expire and then forward all incoming calls to A’s Voicemail.

A has set Preset Call Forward When Busy to B’s extension.

- The system will forward the A’s calls 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.

- The Preset Call Forward request will be served and the call will land on A after expiry of No-Reply Timer.

*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. See “Call Forward” for instructions. In this case, the Preset option will not be applicable. As soon as A cancels Call Forward from his/her extension, the Preset option will be applicable.

- *Preset Call Forward cannot be canceled 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

To provide this feature to an extension, you need to do the following configuration:

- Select the **Preset Call Forward type** and the **Destination** for each timezone. For instructions, see “SLT Extensions” and “SIP Extensions” under *Basic Settings*.
- If required, you may change the duration of the **Call Forward No-Reply Timer** on the extension. By default, the timer is set to 30 seconds.

Priority

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 higher 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

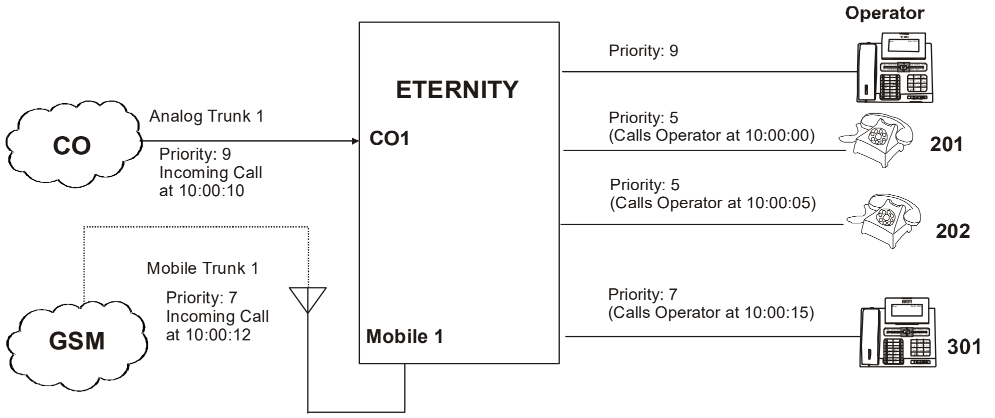
Highest Priority can be assigned to Extensions of important or higher ranking persons in an organization; for example, calls from senior managers or top executives in an organization can be allowed to be answered first by the destination extension.

Highest Priority can be assigned to particular Trunks, so that when there are incoming calls on different trunks at the same time, the call on the trunk with the highest priority gets answered first by the destination extension. For instance, special or private trunk lines, trunk lines dedicated as help lines or emergency trunks, or trunks designated as hotlines can be assigned greater priority.

How it works

Priority can be assigned to all Trunk types (Analog, Mobile, SIP) and Extension types (SLT, SIP).

To understand how this feature works, consider this illustration:



Here,

- There are two incoming calls, one on the Analog Trunk 1 and the Mobile Trunk 1 at the same time.
- Analog Trunk 1 has priority, '9', the Mobile Trunk 1 is assigned priority '7'.
- Three extensions, 201, 202 and 302 are calling the Operator. Extension 301 has priority '7', while extensions 201 and 202 have the same priority, '5'.
- Now, on the Operator, 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 201	10:00:00	5
SLT 202	10:00:05	5
Analog Trunk 1	10:00:10	9
Mobile Trunk 1	10:00:12	7
SIP 301	10:00:15	7

- These incoming calls, however, will appear on the Display of Operator's phone (Extended IP Phone) in the order of priority:
    - Analog Trunk 1
    - Mobile Trunk 1
    - SIP 301
    - SLT 201
    - SLT 202
- Now, when the Operator goes Off-hook (pressing speaker key or picking up the handset), the call on Analog Trunk 1 will be answered first, as Analog 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 SIP 301 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 SIP 301 will be placed on the Operator phone with a *Priority Ring* (configurable; default: Triple Ring).

- When the Operator goes Off-hook, the call from SIP 301 is answered.

- When the Operator goes On-hook and then Off-hook after answering the call from SIP 301, the call from SLT 201 will get answered first, though both 201 and 202 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 will be indicated by *Priority Ring* on the landing destination.

- *Priority Ring will be played only if the Extended Phone is SPARSH VP248.*
- *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 a 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

You can assign Priority to Trunks and Extension by configuring this parameter on the desired Trunk or Extension port type under “Basic Settings”.

You can also configure, *Priority Ring*, if required. See “Distinctive Rings”.

To assign Priority for Trunks and Extensions,

- Login as System Engineer.

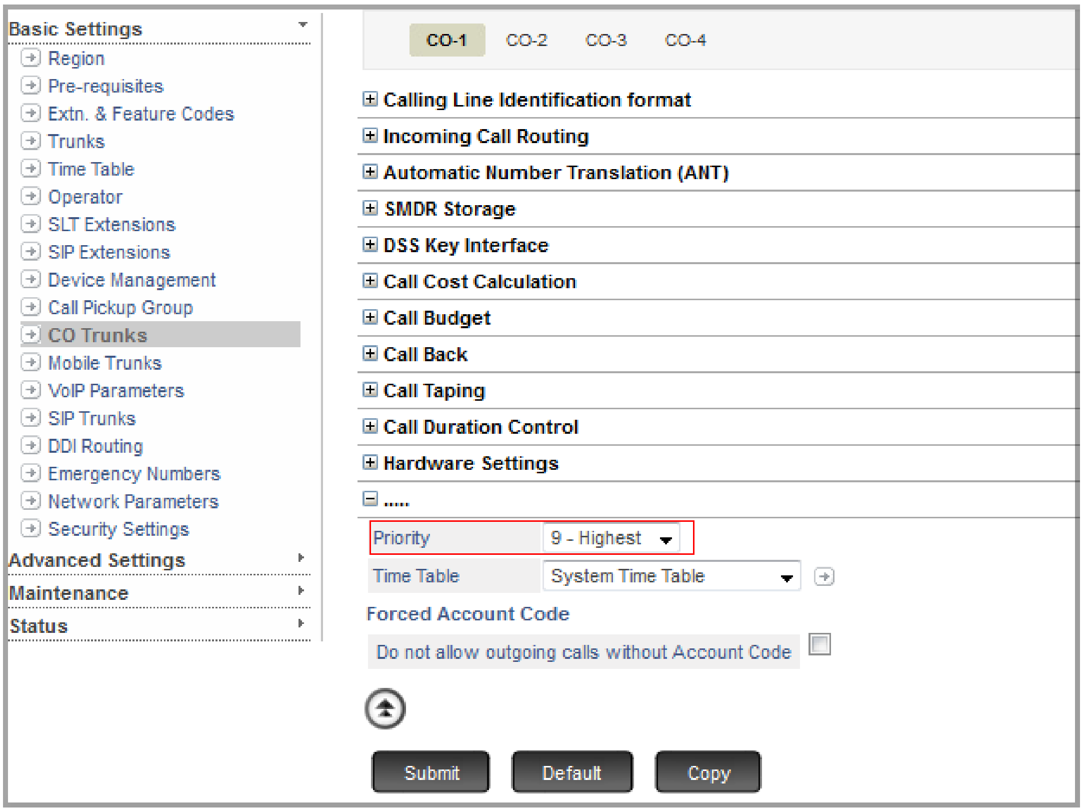
- Click **Basic Settings**.

Assigning Priority to Trunks

- Click the desired Trunk type: CO, Mobile, SIP Trunks.

- When the page of the respective Trunk type opens, select the Trunk number on which you want to set Priority.

- On this page, click **More....** to view additional parameters on this page.

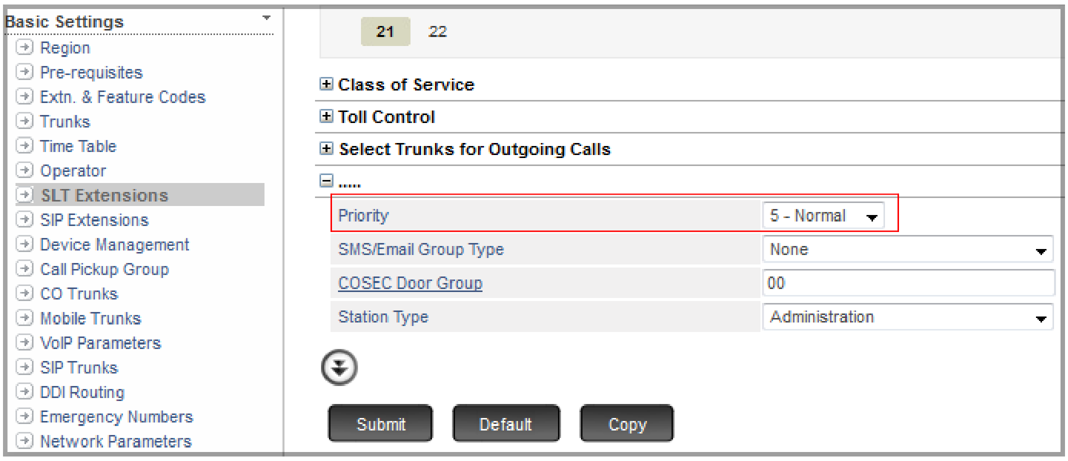


- Select a Priority level from 1 to 9 for the Trunk. Default: 9-Highest for all Trunk Types.
- Click **Submit.**

Assigning Priority to Extensions

- Click the desired Extension type: SLT, SIP Extensions (for Extended IP Phone Settings).
- When the page of the respective Extension type opens, click the tab of the Extension number on which you want to set Priority.

- On this page, click the **More...** to view additional parameters on this page.



- Select a Priority level from 1 to 9 for the Extension. Default: 5-Normal for all Extensions. Extensions designated as Operator can be assigned higher priority than other extensions.

- Click **Submit**.

Privacy

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 by using 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 trunk landing destinations 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 landing destination 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 landing destination for calls on Trunk 1.

- **Privacy from Built-In-Auto Attendant**: 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 Auto Attendant to call extension A. Extension C's call will be blocked and C will get error tone.

How to configure

To provide Privacy to extensions, you must enable this feature in their "Class of Service (CoS)", for the Day and Night/Break.

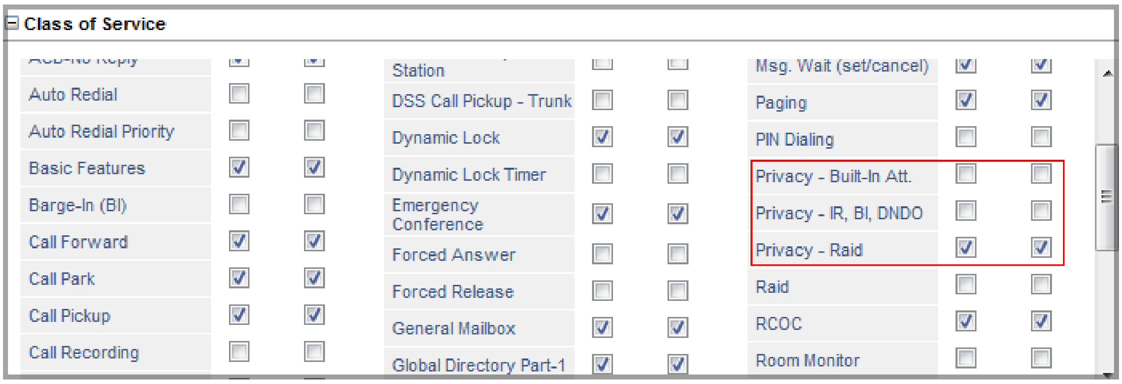
By default, Privacy from Raid is enabled in the Class of Service of all Extension types: SLT, SIP. So, none of the extensions can raid the other. You may disable this feature on extension, if required.

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 for all Extension types. You may keep this feature disabled on extensions on which you want to provide Privacy from Trunk Call intrusion beeps.

To be able to use Privacy feature, extension users must have this feature enabled in their "Class of Service (CoS)" for the Day and Night/Break time, as required.



See "Basic Settings" for instructions on configuring Class of Service for different extension port types: "SLT Extensions" and "SIP Extensions".

Quick Dial

Quick Dial provides 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 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 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: 01 to 25.
- 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 Extended IP Phone.

  For instructions on configuring Quick Dial on the Matrix Extended IP Phone, see "Phone Key Settings" under "Configuring Matrix SPARSH VP248", "Phone Key Settings" under "Configuring Matrix SPARSH VP310", "Phone Key Settings" under "Configuring Matrix SPARSH VP510" and "Phone Key Settings" under "Configuring Matrix SPARSH VP330".
- 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 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

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 into 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.

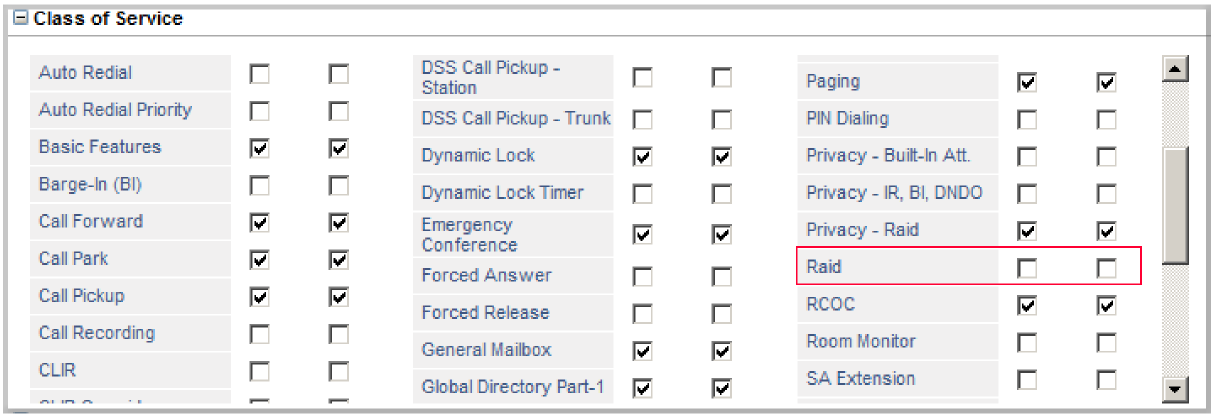
Feature Interactions

- 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.
- 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.

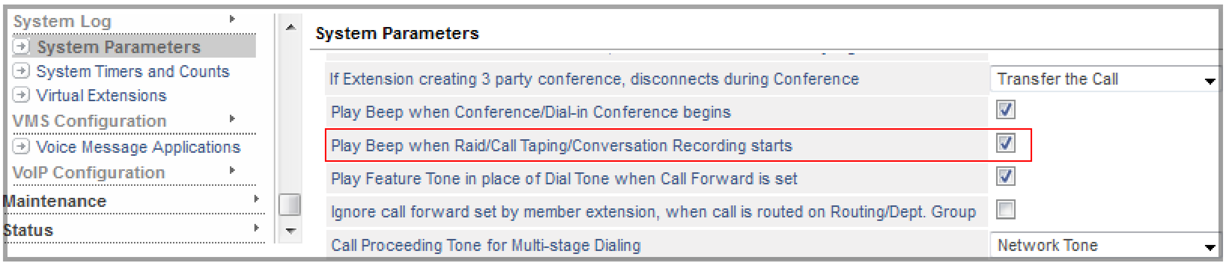
How to configure

To be able to use Raid, extension users must have this feature enabled in their "Class of Service (CoS)". See "Basic Settings" for instructions on configuring the different Extension types: SLT, SIP.



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/Call Taping/ Conversation Recording starts** check box in the **System Parameters**. See "System Parameters" for instructions.



How to use

For 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)

Extensions users of the PBX 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 PBX 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 mostly lands 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 the Operator cannot directly 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 PBX 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 the Mobile Ports and the SIP Trunks.

RCOC is not supported when calls are made from Analog trunks due to the signaling limitations of Analog trunks.

How it works

The Prerequisites

- RCOC must be enabled on the desired Trunks: Mobile and SIP.
- RCOC must be enabled in the Class of Service assigned to the (original caller) extension.

RCOC Table

RCOC table is maintained internally by SARVAM UCS and it is non-programmable.

- SARVAM UCS can keep a record of 255 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.

- 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 outgoing 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[156]:

    - 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 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.

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*** *check box 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*** *check box 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.

- 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 (configurable; default 45 seconds). If the original caller does not answer the call within this Timer, the call is routed to the destination configured 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 in the system.

---

156. The Trunk from which the outgoing call is made.

- *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 an extension 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" and "Enabling RCOC in Class of Service" below for instructions.

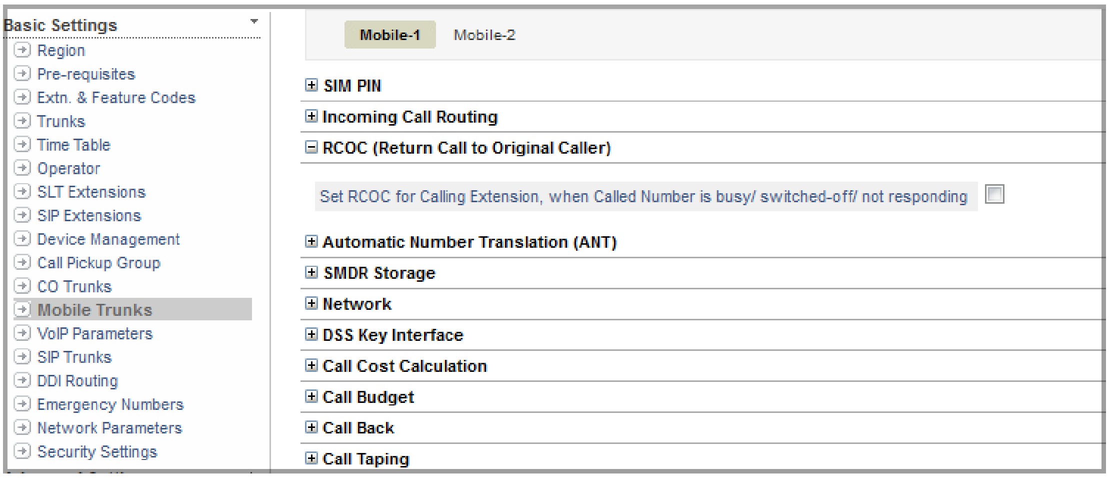
If desired, the related Timers, the RCOC Record Delete Timer and the Ring Back Tone Timer may also be changed. See "System Timers and Counts" for instructions.

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** check box in the System Parameters. See "System Parameters" for instructions.

Enabling RCOC on Trunk

- Login as System Engineer.
- Click the trunk type on which you want to enable this feature. For example: Mobile.

- Click **RCOC (Return Call to Original Caller)** to expand.



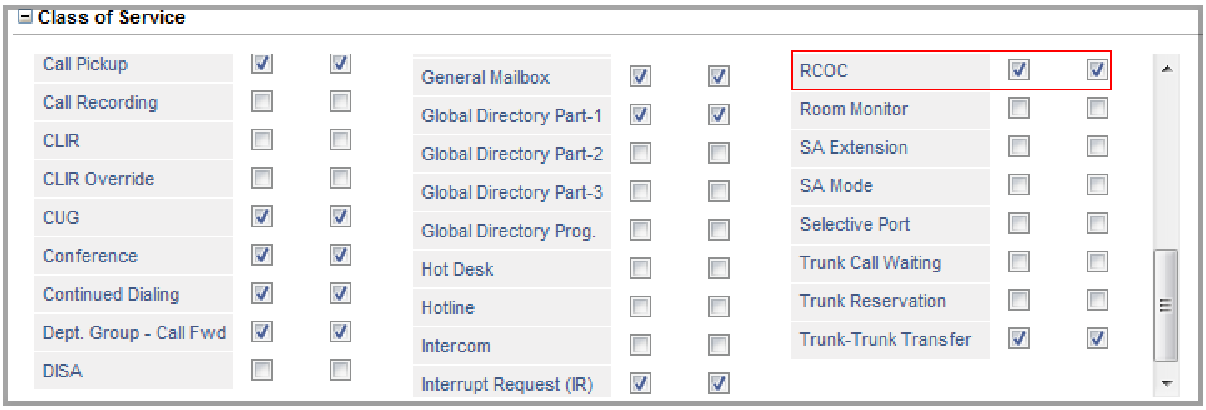
- Select the **Set RCOC for Calling Extension, when Called Number is busy/ switched-off/ not responding** check box to enable this feature on the desired trunk port.

- Click **Submit.**

  Similarly, you can enable this feature on SIP trunks.

Enabling RCOC in Class of Service

By default the 'RCOC' feature is enabled in the "Class of Service (CoS)" of all extensions of SARVAM UCS. So, all extensions of SARVAM UCS are by default allowed this feature.



Real Time Clock (RTC)

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.

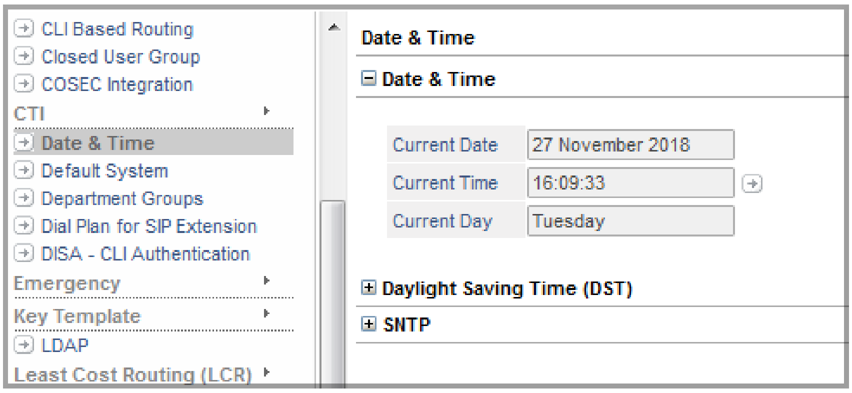
SARVAM UCS has a built-in Real Time Clock (RTC) circuit that maintains date and time. When you select Region, the RTC is automatically set to the current date and time of the country/region where SARVAM is installed.

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.

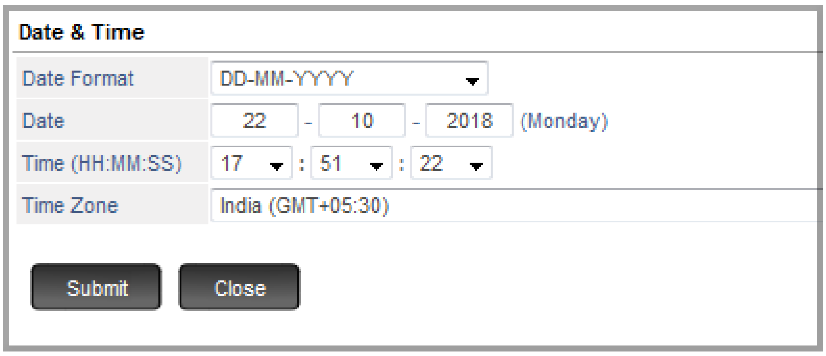
How to Configure

- Login as System Engineer.
- Under **Advanced Settings**, click **Date & Time**.

  The Date & Time page opens.
- Click **Date and Time** to expand.



- Click the **Settings** icon and set the following parameters to the desired values:



  - **Date Format:** Select the format to display the Date as **DD-MM-YYYY** or **MM-DD-YYYY**.
  - **Date:** Enter the current Date in DD-MM-YYYY format.

- **Time:** Enter the current Time in HH-MM-SS format.

- **Time Zone:** By default the Time Zone applicable for your country is selected. If required you may select a different Time Zone.

- Click **Submit**.

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

Time Zone	Index
(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
(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

Time Zone	Index
(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
(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

Time Zone	Index
(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
(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

Time Zone	Index
(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

Room Monitor

This feature enables the extension users to listen to the conversations taking place in another location where an 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 misuse/abuse of this feature by users.*

How it works

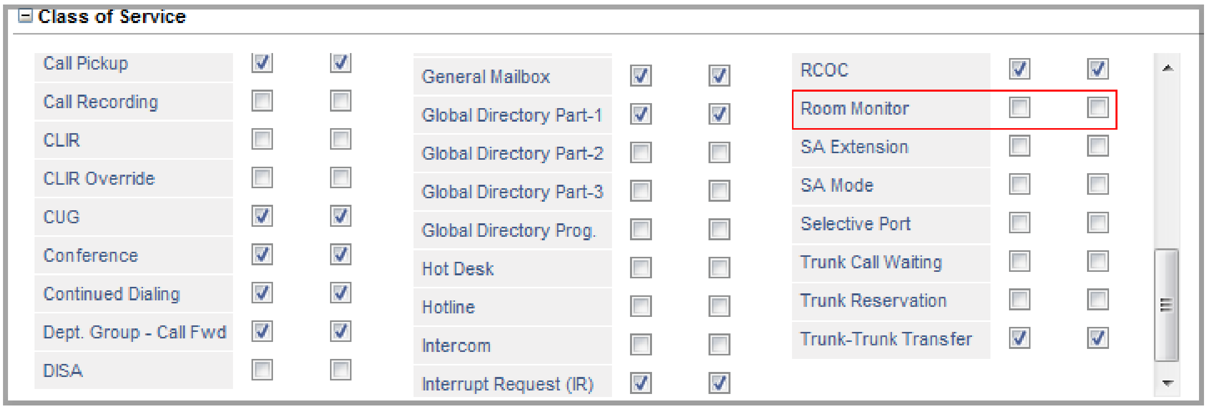
- A is a supervisor in a Manufacturing unit.
- A's room is on the second floor. The manufacturing area is on the ground floor.
- To keep track of the activities in the plant on the ground floor, there must be an Extended IP Phone at the place where the activities need to be monitored, and A's extension must have higher "Priority" than the extension at the monitored location.
- If there is an Extended IP Phone at the desired location, A can activate Room Monitor.

  A can activate Room Monitor only if the Extended IP Phone at the desired location is idle.
- When A activates Room Monitor, the microphone of the Extended IP Phone on the ground floor is turned on. A can now listen to all the sounds/ talking 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 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 an Extended IP Phone.

How to configure

To be able to use Room Monitor, extension users must have this feature enabled in their "Class of Service (CoS)" for the Day and or Night/Break time (depending on their requirement). See "Basic Settings" for instructions on configuring the different extension port types: SLT, SIP.



How to use

For Extended IP Phone Users

- 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.

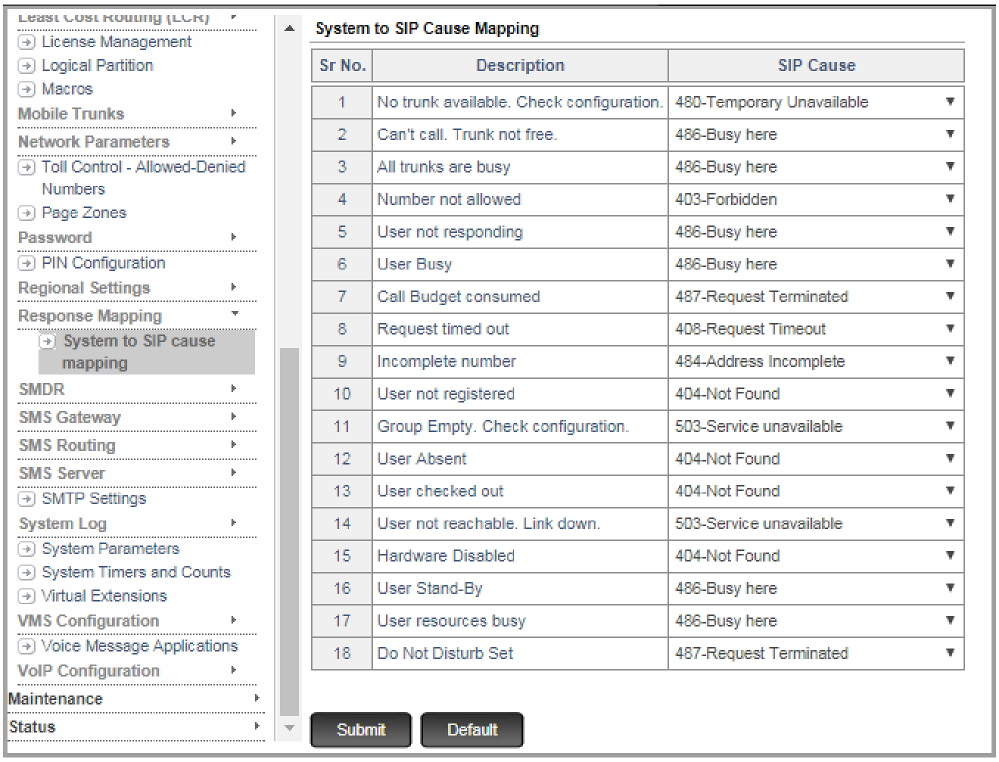
Response Mapping

All SIP messages are either requests from a server or a client, or responses to a request. For calls to be established in a SIP network, the mapping of system disconnect/release events to that of SIP codes and events and vice-versa is very important.

SARVAM UCS supports configurable System to SIP Cause Mapping. You can select the code and event the system must send as the SIP response to remote SIP Peer, when the system disconnect/release the SIP call.

How to configure

- Login as System Engineer.
- Under **Advanced Settings**, click **Response Mapping**.
- Click **System to SIP cause mapping**.



Sr No.	Description	SIP Cause
1	No trunk available. Check configuration.	480-Temporary Unavailable
2	Can't call. Trunk not free.	486-Busy here
3	All trunks are busy	486-Busy here
4	Number not allowed	403-Forbidden
5	User not responding	486-Busy here
6	User Busy	486-Busy here
7	Call Budget consumed	487-Request Terminated
8	Request timed out	408-Request Timeout
9	Incomplete number	484-Address Incomplete
10	User not registered	404-Not Found
11	Group Empty. Check configuration.	503-Service unavailable
12	User Absent	404-Not Found
13	User checked out	404-Not Found
14	User not reachable. Link down.	503-Service unavailable
15	Hardware Disabled	404-Not Found
16	User Stand-By	486-Busy here
17	User resources busy	486-Busy here
18	Do Not Disturb Set	487-Request Terminated

- Select the SIP cause for each system disconnect/release cause that you want the system to send as the SIP response to the remote SIP Peer.
- Click **Submit**.

Reminder

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 Reminder calls can be configured as 'Personalized' or 'Automated'.

- Reminders can be voice-guided, if SARVAM UCS has a Voice Mail System (VMS) in it.

- SARVAM UCS can register as many as 48 Reminders 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[157], displaying the number of the extension to which the reminder call is to be served.

- When the Operator answers this call, the 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.

- 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.

157. The Operator phone rings for the duration of the Alarm Ring Timer. If the Operator does not answer the call, SARVAM UCS will make two more Attempts at an Alarm Attempt Interval of 5 minutes to call the Operator.

- If the extension is busy[158], 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 the Matrix Extended IP Phone, Reminder 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 the Voice Mail, or be 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, 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, 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, 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'.

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 Acknowledgment Code **0** is non-configurable.

158. *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'.*

- *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 status of Reminders—details of Reminders that have not been served—on the *Reminder* page of Jeeves, from the System Administrator (SA) mode, 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.

  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.

SARVAM UCS offers voice-guided Reminders using VMS to extension users and the Operator.

Voice-guided reminders lead users through a menu, helping them set the reminder in a step-by-step manner.

Voice-Guided Reminders set/canceled by Operator

The Operator can set voice-guided Reminders for extension users

For 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 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, s/he can set the Reminder 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 Extended IP Phones

To set Reminder for the extension user

- Press the 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' check box 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 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.

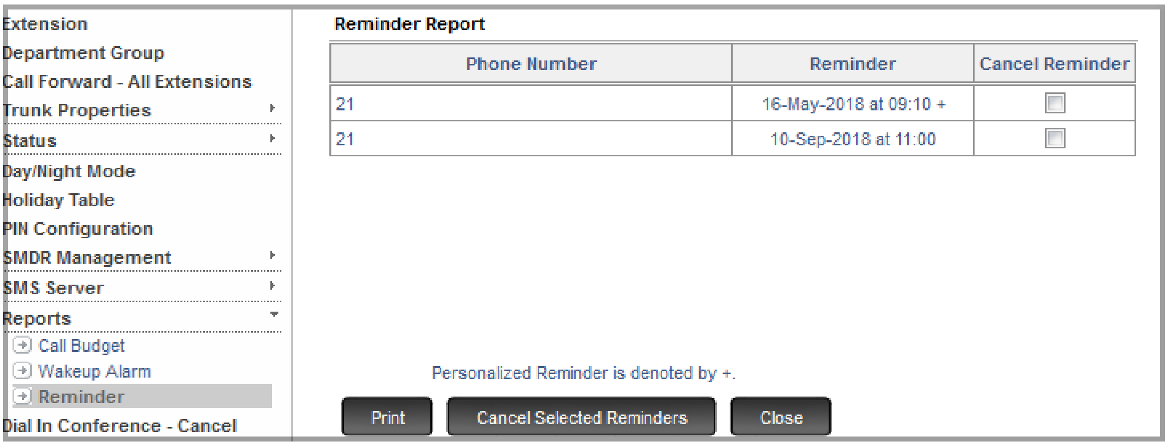
To cancel Reminder

- Pick up the handset.
- Dial 162
- Dial #
- You get confirmation tone.
- Replace handset.

Viewing and Printing Reminder Status

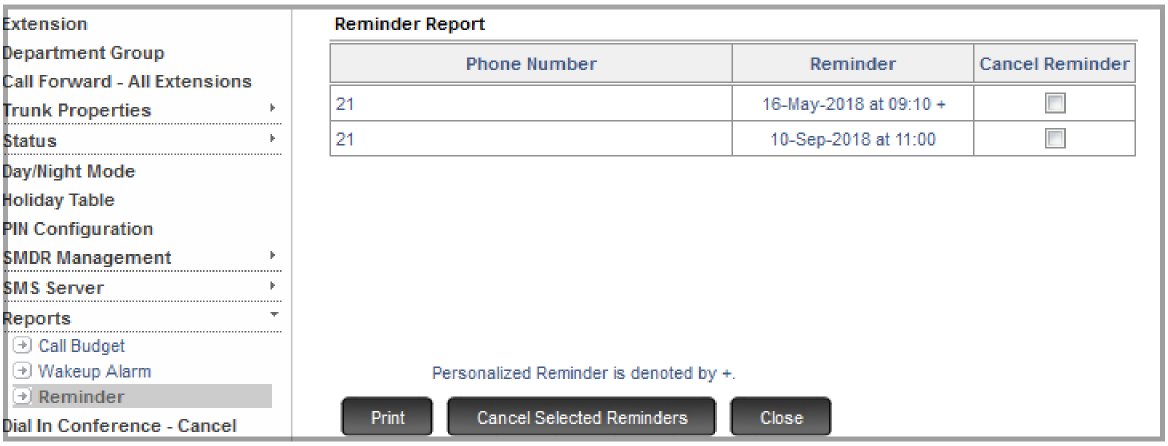
The Operator can view the status of Reminders that are yet to be served from the System Administrator pages of Jeeves. To view Reminder Status,

- Login as System Administrator.
- Click **Reports**.
- Click **Reminders**.



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 **Cancel Selected Reminders** at the bottom of the page.
- To print this page, click **Print**.
- 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.

Selective Port Access

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.
- By default, the following access codes are assigned to each Port Type:

Port Type	Port Type Code	Port Number[a]
SLT	01	01 to 2
CO	03	01 to 04
Mobile	25	01 to 02
SIP Trunk	26	01 to 08
SIP Extension	34	01 to 50
Virtual Extensions	36	01 to 20

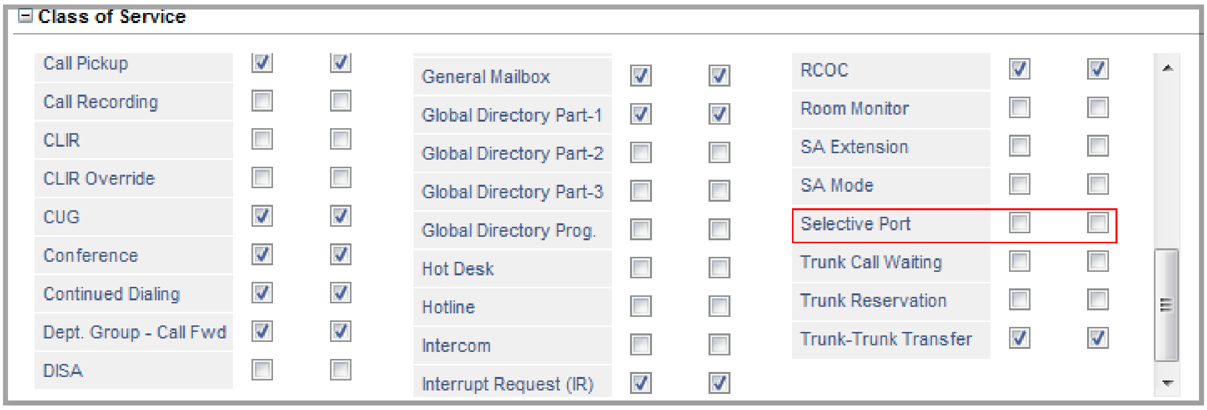
a. The number of ports mentioned here are for ETERNITY NENXIP50. For details regarding the number of ports supported for ETERNITY NENX312/ETERNITY NENX416, see “Technical Specifications of ETERNITY NENX”.

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 access SIP Extension 10, A must dial ***69-34-10***.

How to configure

To be able to use Selective Port Access, extension users must have this feature enabled in their “Class of Service (CoS)” for the Day and Night/Break time, as required. See “Basic Settings” for instructions on configuring CoS for the different extension port types.



How to use

For 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-Port Type-Port Number
- Dial 89-Port Type-Port Number (for users in USA)

For SLT Users

- Dial 69-Port Type-Port Number
- Dial 89-Port Type-Port Number (for users in USA)

Self Ring Test

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 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

Shared Call Appearance (SCA) allows 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.*
- *If a call is put on 'Consultation Hold' from Location -1 of SIP Extension (binding -1), then the SIP Extension registered at Location-2/3 (another binding) cannot 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 check box 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 under **VoIP** on the SIP Extension Settings. For instructions, see "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

SARVAM UCS supports Balance Inquiry and Recharging of the SIM Card installed in its Mobile Ports[159].

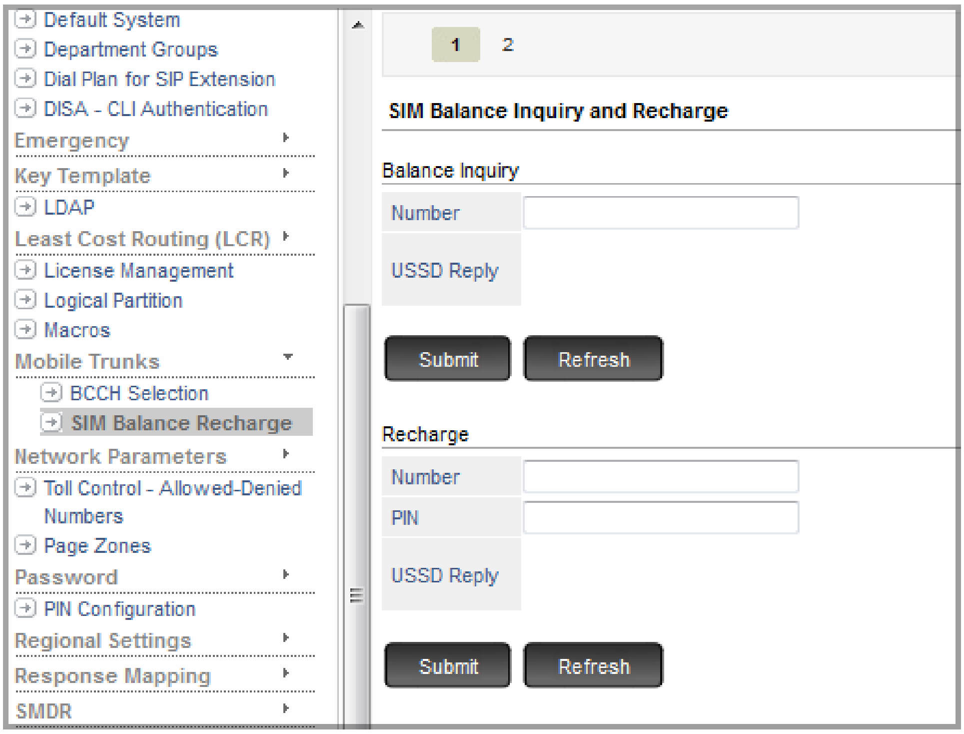
To be able to use this feature, first get information about the following from your Network Operator:

- **Balance Inquiry Number:** This is the number provided by the Network Operator to their subscribers for checking Balance. Different Network Operators have different numbers
- **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.

SARVAM UCS allows you to check your SIM balance from Jeeves.

How to check

- Login as System Engineer.
- Under **Advanced Settings**, click **Mobile Trunks**.
- Click **SIM Balance Recharge**.
- Select the Mobile Port on which you want to request SIM Balance/Recharge.



159. *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.*

Balance Inquiry

To make 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 Card

To recharge the SIM Card,

- Enter the Recharging Service Number provided by the Network Operator.

  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 #.

- Enter the **PIN** number which is printed on the Recharge Voucher/Coupon. Your Recharge PIN number may consist 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.

- **USSD Reply**: 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.

- 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**.

- *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 both the SIM Cards present 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 Mobile network.*

- *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

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

- SMS Gateway license.
- SMPP Clients installed on a computer connected in the same LAN as SARVAM UCS.
- configure the required parameters in the SMPP Clients to register itself with SARVAM UCS.
- define the Mobile port through which the messages are to be sent/received in the SMPP Clients.
- configure the SMPP Client parameters in SARVAM UCS.

This is how SARVAM UCS SMS Gateway works,

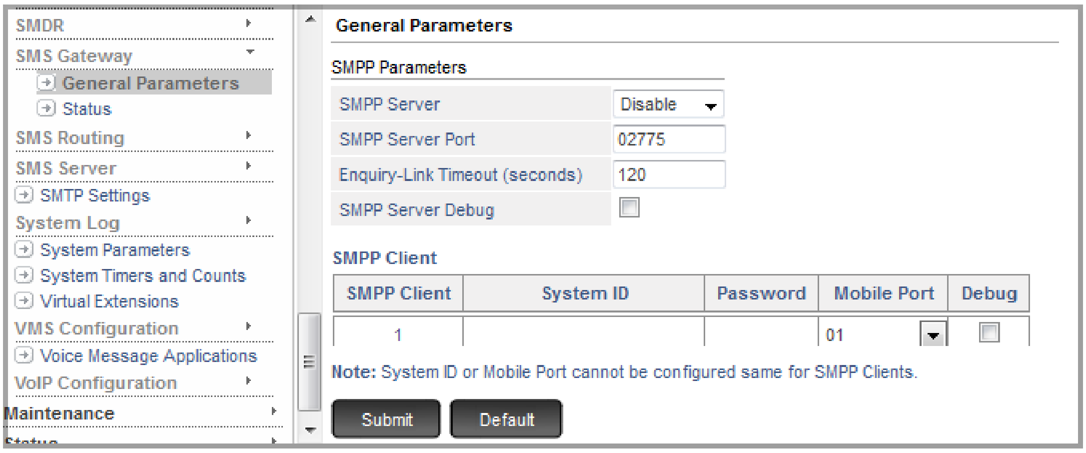
- On successful registration of the SMPP Client with SARVAM UCS, a binding is established between the SMPP Client and SARVAM UCS.
- The SMPP Client can bind itself with SARVAM UCS as a Receiver, Transmitter or Transceiver.
  - As a Transmitter, the SMPP Client will only be able to send messages using 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 in any other state, the SMS will not be sent.*
- *If the Mobile port is disabled/unregistered, the messages will not be sent.*

- 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

- Login as System Engineer.
- Under **Advanced Settings**, click **SMS Gateway**.



- Click **General Parameters**.
  - **SMPP Server:** Select **Enable**, if you wish to use the SMS Gateway feature. By default, the SMPP Server is disabled
  - **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 troubleshooting 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 2.

- **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. SARVAM UCS supports a maximum of 2 Mobile ports[160].

Make sure you do not assign the same Mobile Port to multiple SMPP Clients.

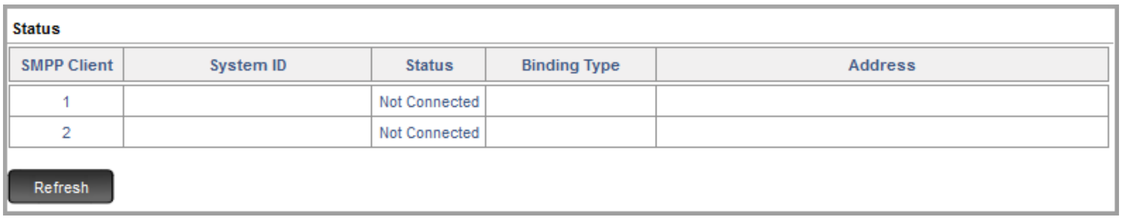
- **Debug:** If you want to monitor the events and processes of the SMPP Client, for troubleshooting and identifying faults and errors, select the **Debug** check box. The debug messages are sent to the remote Syslog Server. For detailed instructions, for configuring the Destination Port, Syslog Server IP Address and Port, see "System Debug".

Viewing SMPP Server Status

- Login as System Engineer.

- Under **Advanced Settings**, click **SMS Gateway**.

- Click **Status**.



Status

SMPP Client	System ID	Status	Binding Type	Address
1		Not Connected		
2		Not Connected		

Refresh

- 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 displays any one of the following:

Status	Description
Not Connected	When the SMPP Server does not receive any request from the SMPP Client.

160. *Depends on the model you have. Please refer the Appendix for an overview of the system resources and maximum expansion capacity.*

Status	Description
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 displays the type of SMPP Client Binding, that is Transmitter, Receiver or Transceiver.
- **Address:** This displays the SMPP Client's IP Address and Port.

SMTP Settings

The SMTP Settings must be configured if you wish to send the mails for VMS applications, auto sign-in related mails and/or register the SMS Server as a 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**[161] 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.

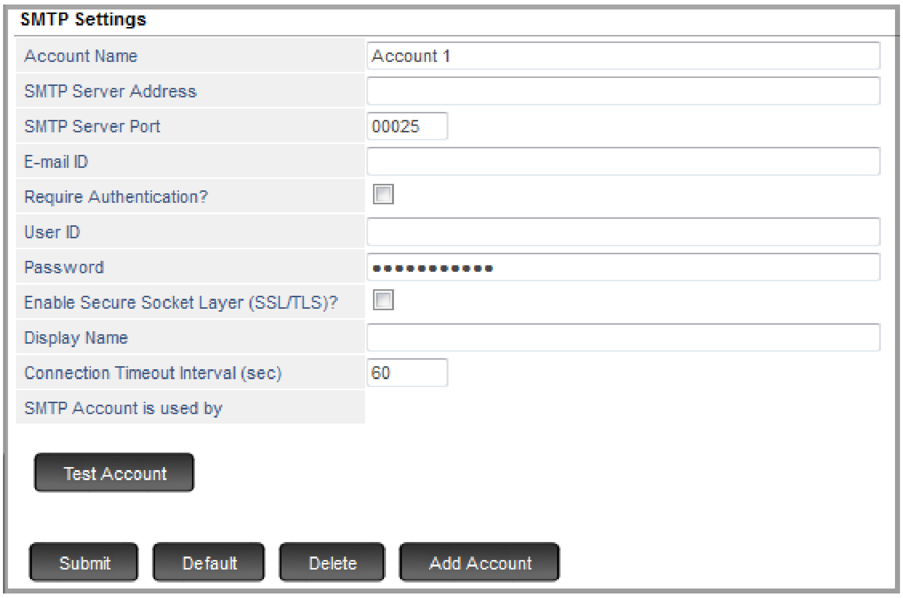
How to configure

To configure SMTP Settings,

- Login as System Engineer.
- Under **Advanced Settings**, click **SMTP Settings**.

161. You can also have the new voice message mailed as an attachment with the message wait notification.

By default no account is configured. You can configure a maximum of three accounts. Click **Add Account** and then configure the following parameters:



- 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 **Password** as provided 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 Account is used by** 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**.

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'.

- **Test Status:** Any one of the results listed below may appear:

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.
"Sending Test Mail Failed"	When connection to SMTP has been established successfully but there is no acknowledgement for the test mail sent.
"Test Mail Sent Successfully"	When acknowledgement 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 Time Protocol - SNTP

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.

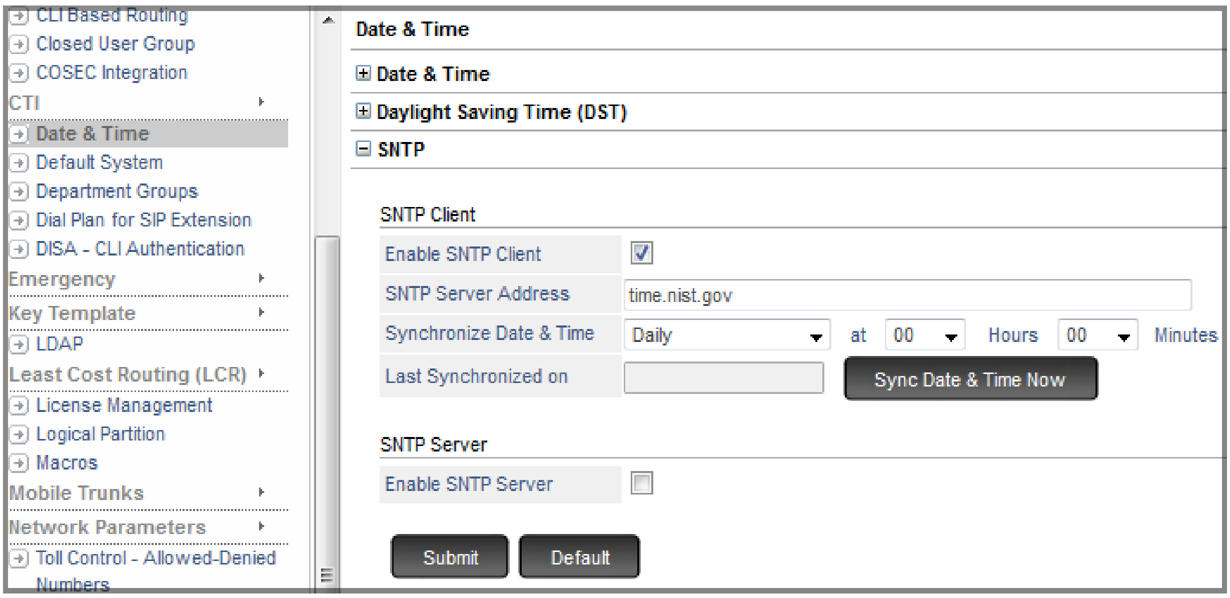
SNTP Server

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.

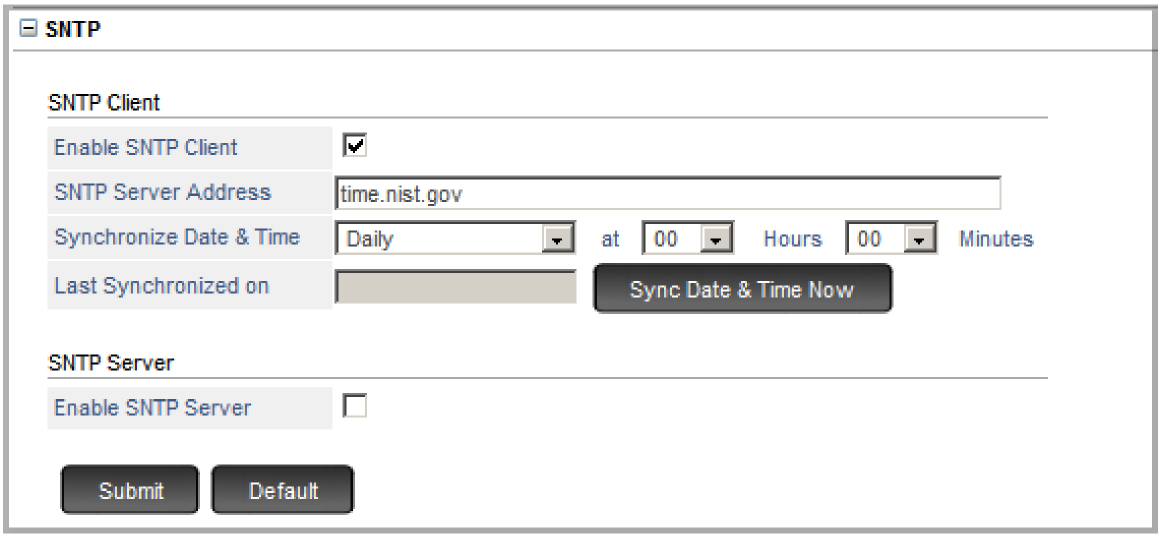
How to configure

- Login as System Engineer.
- Under **Advanced Settings**, click **Date and Time**.
- Click **SNTP** to expand.



SNTP Client

- Configure the following parameters:



- **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 SNTP Server Address. SNTP Server address can be of maximum 64 characters. Default: time.nist.gov

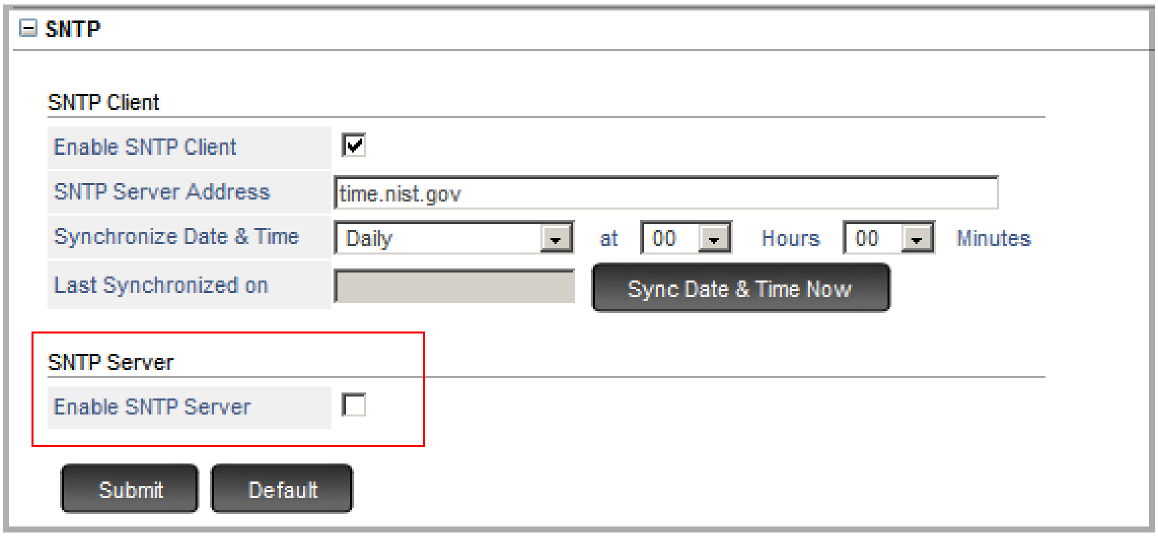
- **Synchronize Date and Time:** You can Synchronize Date and Time with the SNTP Server on daily or weekly basis. If you select weekly, the system will sync the date and time with the SNTP Server on every Monday. Select the desired option as per your requirement. Default: Daily.

- **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

- Configure the following parameters:



  - **Enable SNTP Server:** Select this check box, if you want the system to also act as a **SNTP Server** and the extensions (IP Phones) registered with it behave as the clients. Default: Disabled.

- Click **Submit.**

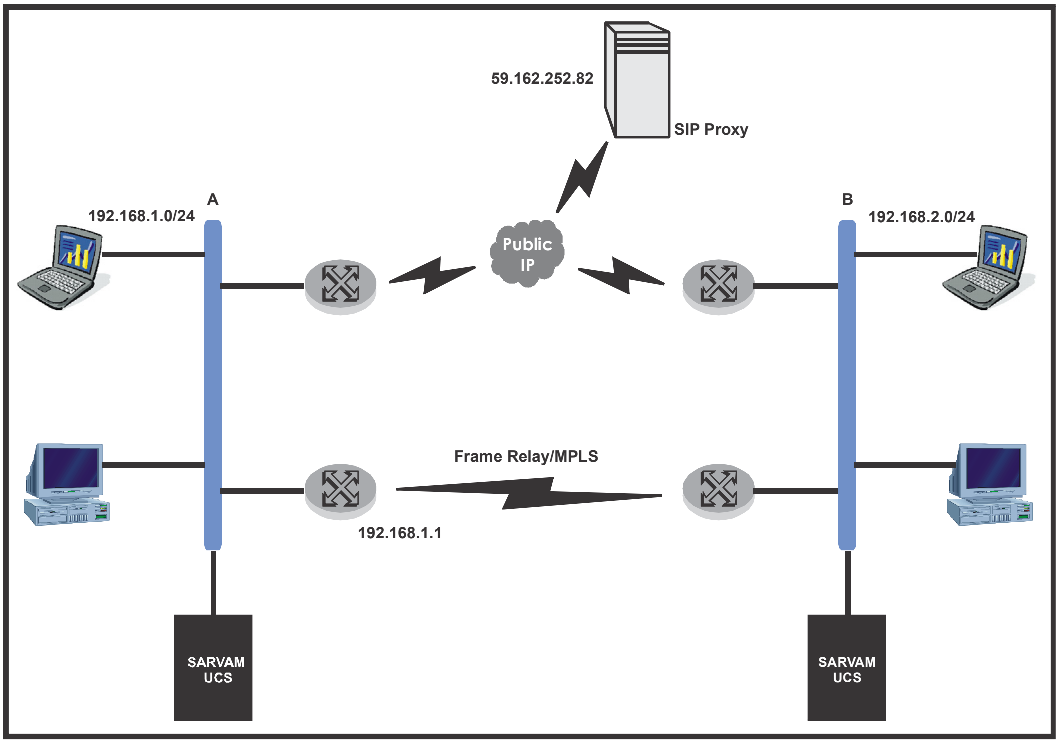
Static Routing Table

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.

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

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.

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.

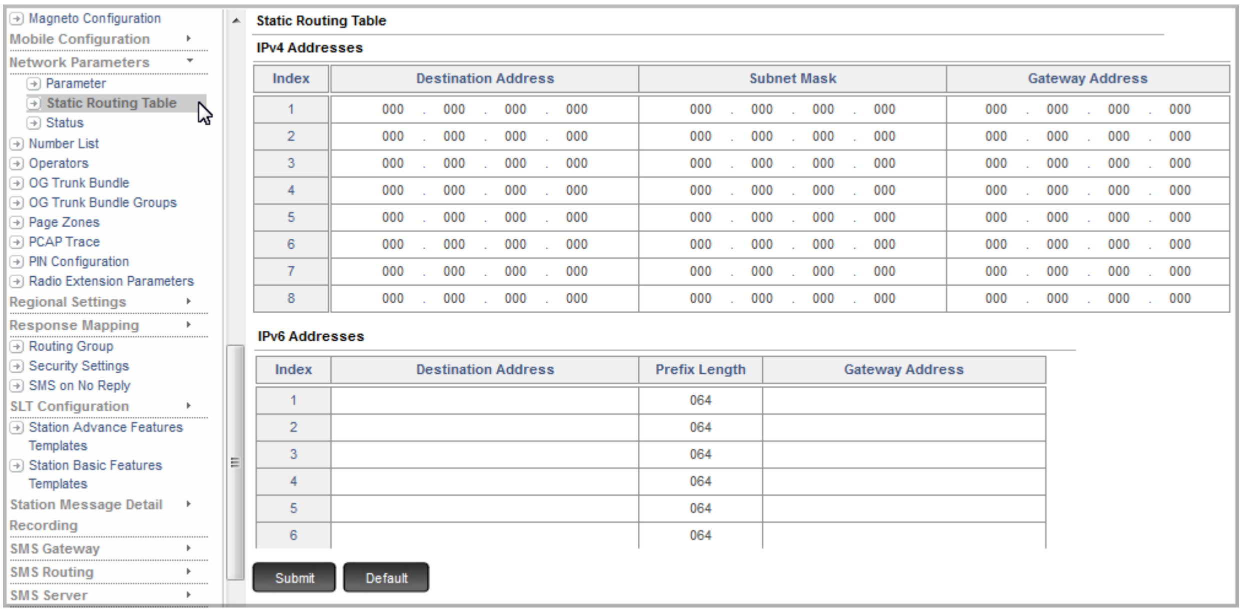
If no match is found, SARVAM UCS will send the IP Packets to the **Default Gateway Address** (Network Connection Type) you configured in the Network Parameters. For detailed instructions, see "Network Parameters".

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

- Login 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.**

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.**

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**.

Station Message Detail Recording (SMDR)

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).

SARVAM UCS has a total SMDR buffer capacity of 12000 calls. It stores:

- 6000 outgoing calls.
- 5000 incoming calls.
- 1000 internal calls.

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 configured to enable the storing of the Incoming Calls, Outgoing Calls and Internal calls using filters. See “Station Message Detail Recording–Storage”.

- **SMDR - Report:** These parameters are configured to assign destination port for getting report of Incoming, Outgoing and Internal calls to get offline report. 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). 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. See “Station Message Detail Recording–Posting”.

Station Message Detail Recording–Storage

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, 5000 incoming calls, and 1000 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.

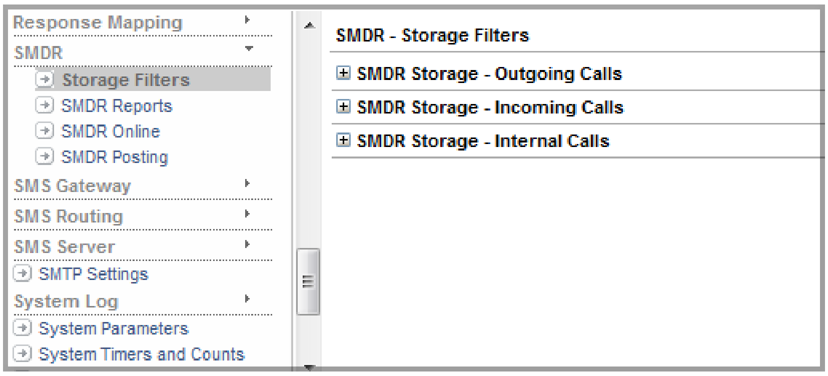
The SMDR buffer data is maintained even during power failures. However it is advisable to take frequent printouts of the calls to avoid accidental loss of the data.

How to configure

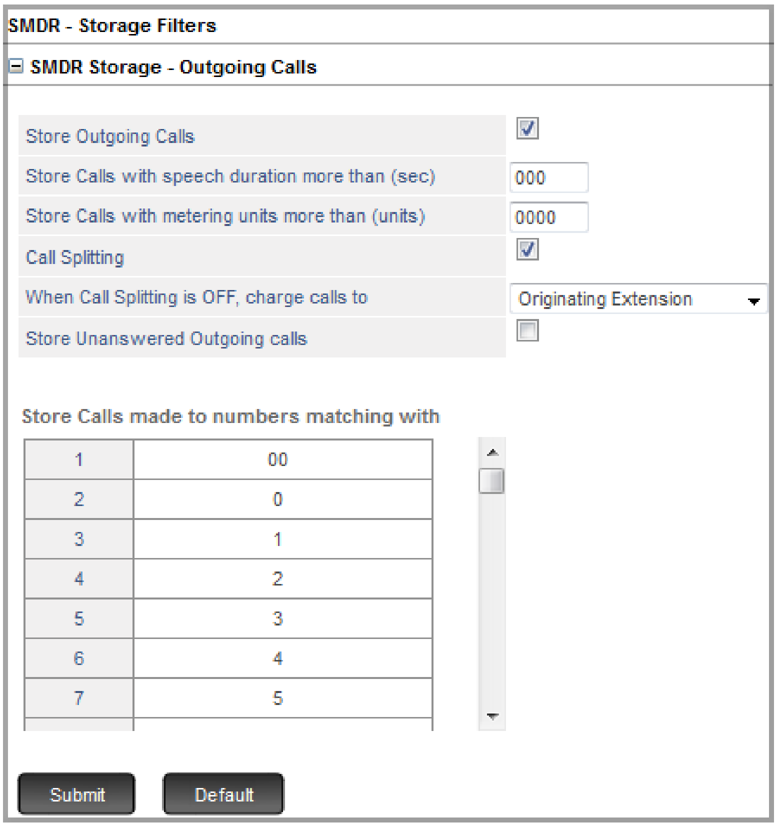
To enable storage of SMDR of Outgoing, Incoming and Internal Calls, you must enable this feature in the system and set the storage filters as per your requirement. To do this,

To configure SMDR-Storage,

- Login as System Engineer
- Under **Advanced Settings**, click **SMDR**.
- Click **Storage Filters**.



- 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.

- 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 **When Call Splitting is OFF, charge calls to**, 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.

- Select the **Apply Number List** check box, if you wish to store calls made to particular numbers. Then you must configure **Store Calls made to numbers matching with.**

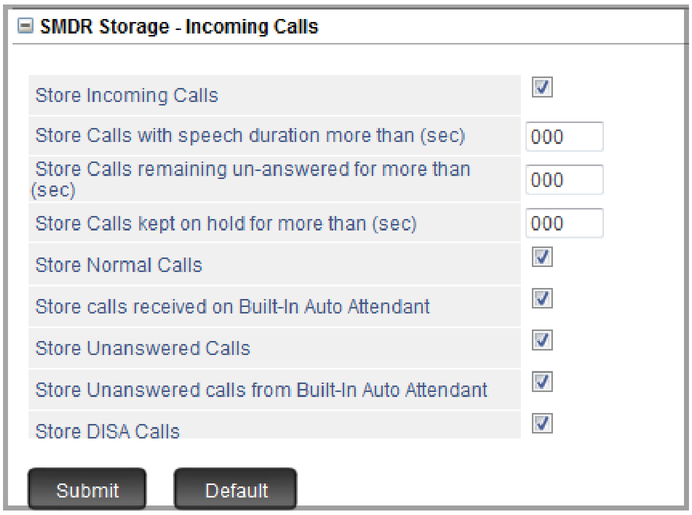
  As the SMDR Outgoing call storage buffer is limited, you may limit the storage of calls to a certain numbers only by configuring **Store Calls made to numbers matching with**.

  You may configure as many as 250 numbers. The system will store records of outgoing calls made to these numbers only.

  SARVAM UCS uses the best match logic for storing these numbers. For example, if you configure 0 here, the system will store all outgoing calls made to numbers starting with 0.

- Click **Submit**.

- To configure storage filters for incoming calls, click **SMDR Storage - Incoming Calls** to expand,



- Select the **Store Internal Calls** check box to enable storage of internal 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.

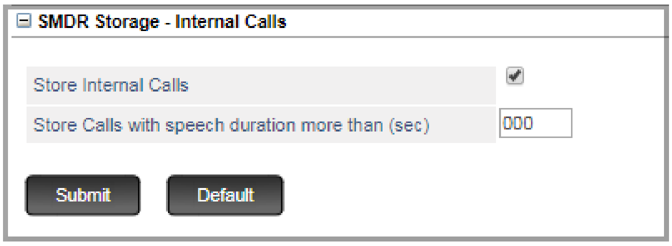
- If you want incoming calls that were kept on hold for a certain duration to be stored, set the filter **Store Calls kept on hold for more than (sec)** to the desired duration.

- 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 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 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**.

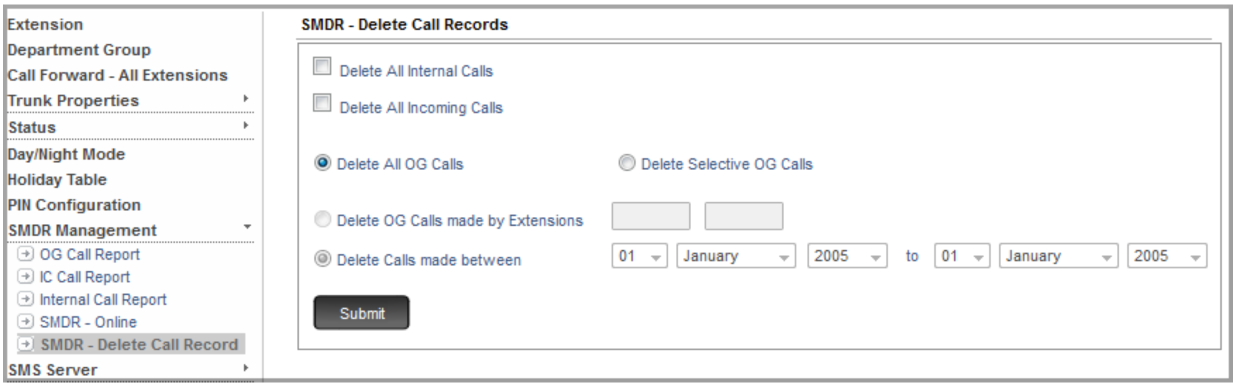
How to use

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,

- Login as System Administrator.
- Click **SMDR Management** to expand.

- Click **SMDR - Delete Call Record.**



- 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 Extensions** radio 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.**

  The SMDR buffer will be cleared according to the settings you enabled on this page.

To delete SMDR records from an extension phone,

- Enter SA mode from an 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 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.

Station Message Detail Recording–Online

SARVAM UCS can generate report for the calls as and when the call is made, and send the report to the printer or a computer.

SARVAM UCS supports 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 the following fields:
    - Extension that 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 **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.
- Each **Outgoing call** is stored with the 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.).

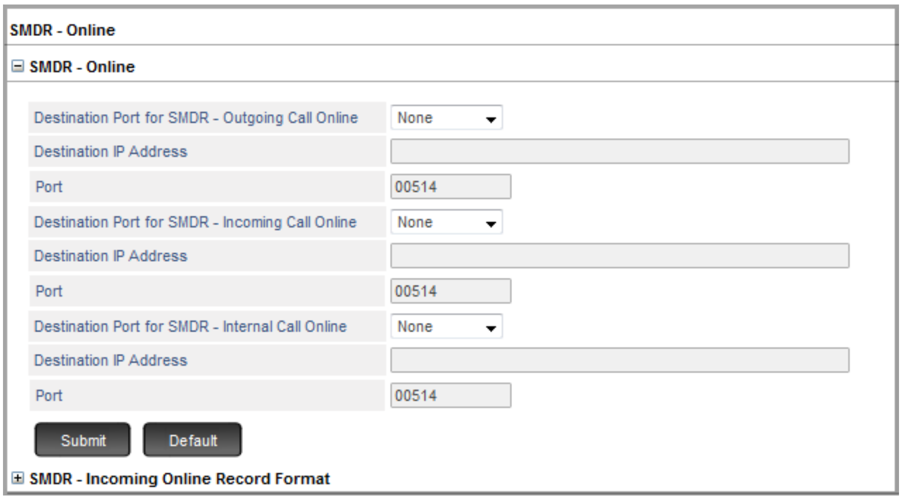
How to configure

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**.

To configure SMDR- Online,

- Login as System Engineer.
- Under **Advanced Settings**, click **SMDR**.
- Click **SMDR - Online**.
- To configure SMDR- online parameters, click **SMDR Online** to expand.



- For **SMDR - Outgoing Call Online**,
  - Select 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: 514, 1025 to 65535.
- For **SMDR - Incoming Call Online**,
  - Select 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: 514, 1025 to 65535.
- For **SMDR - Internal Call Online**,
  - Select 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: 514, 1025 to 65535.

You can also configure Call Record Format for Incoming Calls as per your requirement.

- To configure Call Record Format for Incoming Calls, click **SMDR - Incoming Online 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					

Submit Default

- 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 is 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 check box 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 check box of Date Format. The global check box 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 check box 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 check box.

- **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, 'Do not 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, 'Do not 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.

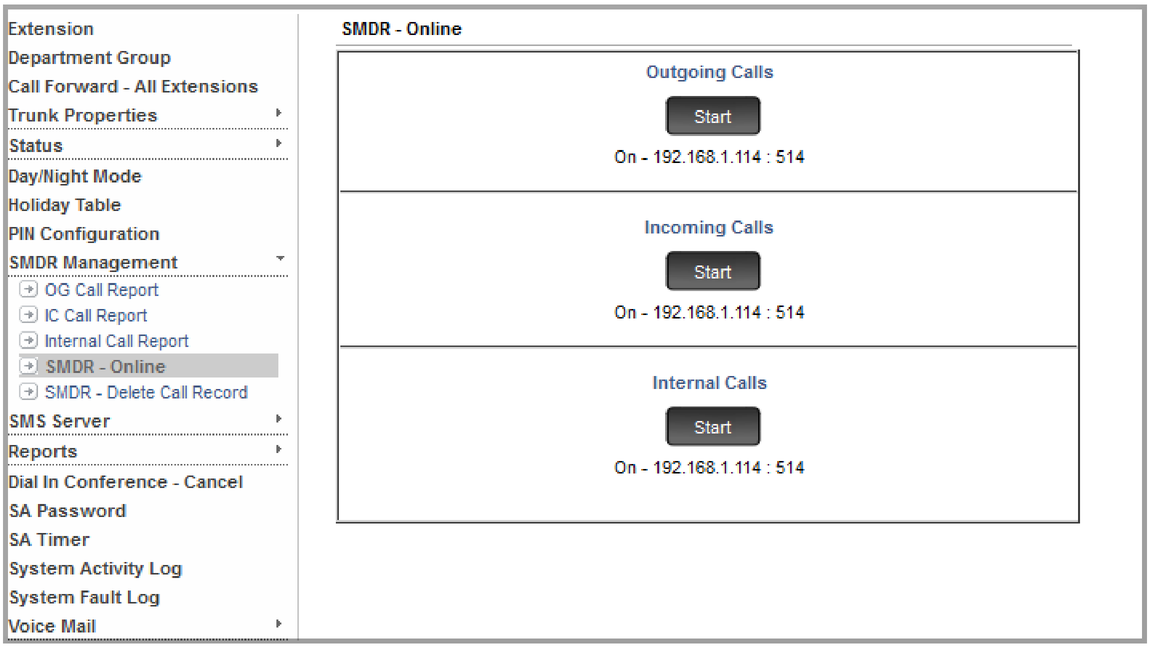
- Click **Submit.**

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,

- Login as System Administrator.
- Click **SMDR Management** to expand.
- Click **SMDR-Online**.



- To start SMDR Online for Outgoing Calls, Incoming Calls and Internal Calls, click **Start**.
- To stop SMDR Online for any of these call types, click **Abort**.

To generate SMDR Online report from an extension phone,

- Enter SA mode from a SLT/Extended IP Phone.

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

To start/stop Online report for Internal Calls:

- Dial **1072-136-1** to start
- Dial **1072-136-0** to stop

- Exit SA mode.

Station Message Detail Recording–Report

SARVAMUCS 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 Syslog[162] 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 and print.

SARVAM UCS supports SMDR only on TCP.

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 Syslog Server IP address as Destination address for Incoming, Outgoing and Internal calls.

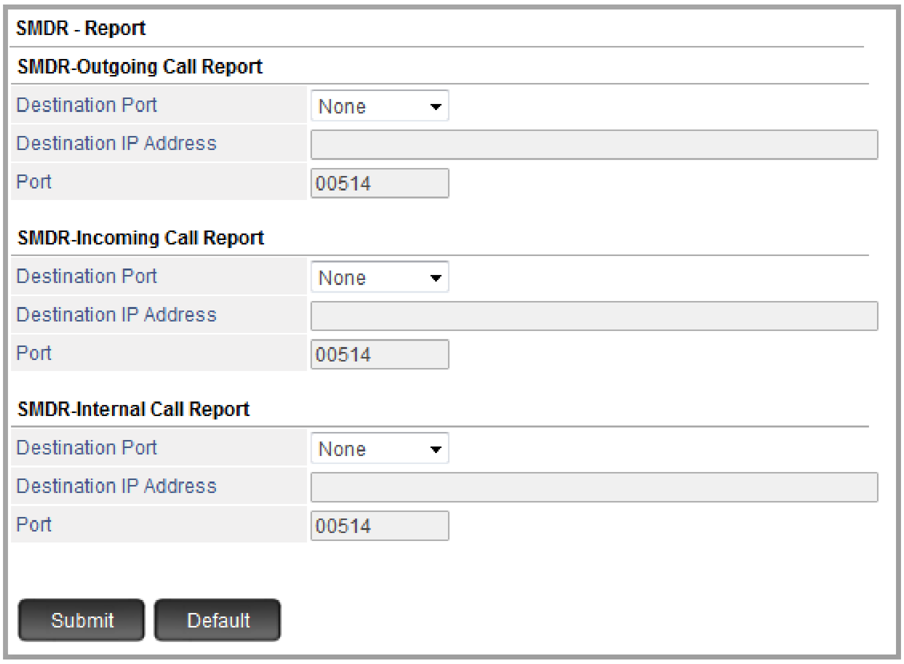
To configure SMDR-Report,

- Login as System Engineer.

- Under **Advanced Settings**, click **SMDR**.

162. Syslog is one of the protocols used extensively for sending debug messages, and is defined in RFC 3164.

- Click **SMDR Reports**.



- For **SMDR - Outgoing Call Report**,
    - Select the **Destination Port**.
    - In the **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: 514, 1025 to 65535.
- For **SMDR - Incoming Call Report**,
    - Select the **Destination Port**.
    - In the **Destination IP Address**, enter the IP Address of the remote Syslog Server.
    - In **Port**, enter the port of the remote Syslog Server. Valid port range: 514, 1025 to 65535.
- For **SMDR - Internal Call Report**,
    - Select the **Destination Port**.
    - In the **Destination IP Address**, enter the IP Address of the remote Syslog Server.
    - In **Port**, enter the port of the remote Syslog Server. Valid port range: 514, 1025 to 65535.
- Click **Submit**.

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.

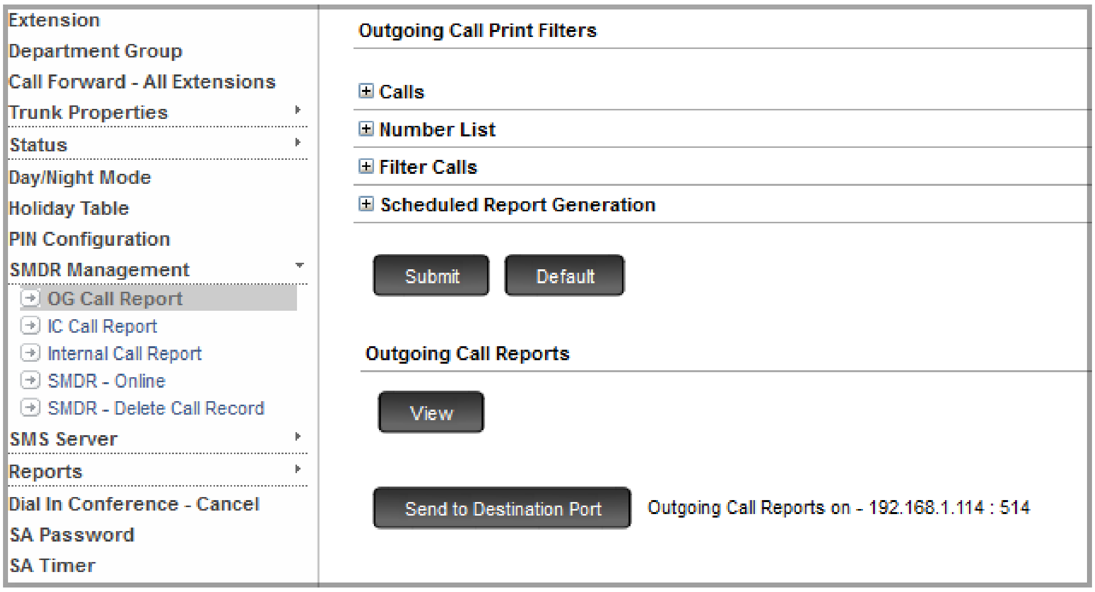
Print SMDR-Report using Jeeves

- Login as System Administrator.

- Click **SMDR Management**.

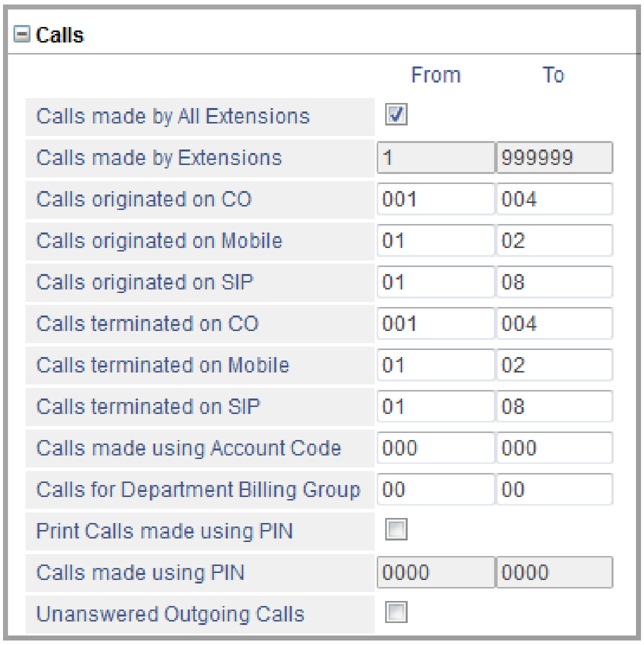
Outgoing Calls

- To print Outgoing Calls with filters, click **OG Call Report**.



Setting Print Filters

- Click **Calls** to expand.



- Set the following filters as desired. You can print reports 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 reports of specific extensions, clear the **Calls made by All Extensions** check box and enter the desired extension range in **From** and **To**.

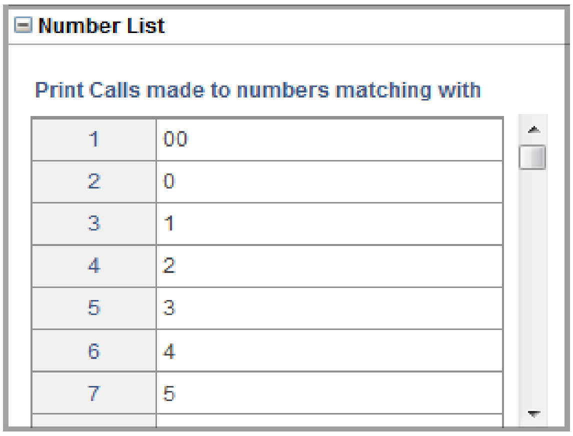
  You can also print reports of outgoing calls originating and terminating on specific trunks: **CO**, **Mobile**, **SIP**.

  You can also print outgoing calls made using **Account Code**, **PIN, Calls for Department Billing Groups** and **Unanswered Outgoing Calls**.

*If you have extension numbers beginning with # or *, make sure the range you assign for “Calls made by Extensions” in the From and To fields must also have # or *. A mix of both will not work. For example, if you have extension numbers from *1 to *99 and you want to generate report of outgoing calls made by these extensions, you must enter *1 in the From field and *99 in the To field.*

- Click **Submit.**

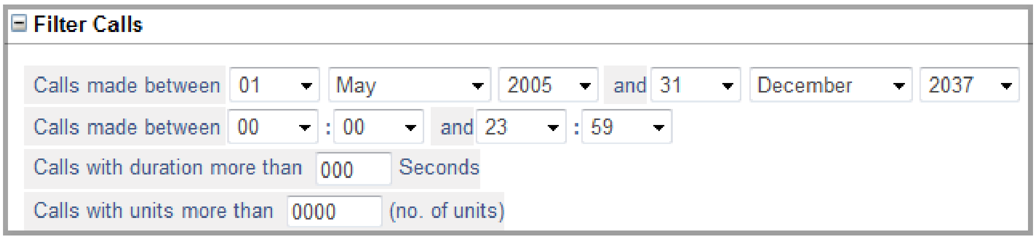
- Click **Number List** to expand.



  - To print outgoing calls made to certain numbers, enter the desired numbers in the **Print Calls made to numbers matching with** fields.

- Click **Submit**.

- 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**.

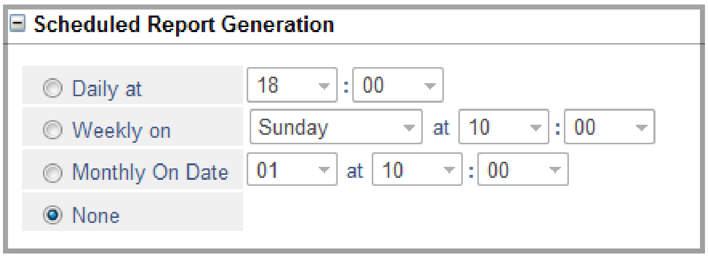
  - If you want outgoing calls that exceed as certain duration to be printed, set the filter **Calls with duration more than (sec)** to the desired duration. All outgoing calls with duration greater than this value, will be printed.

  - 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**.

Scheduled Report Generation

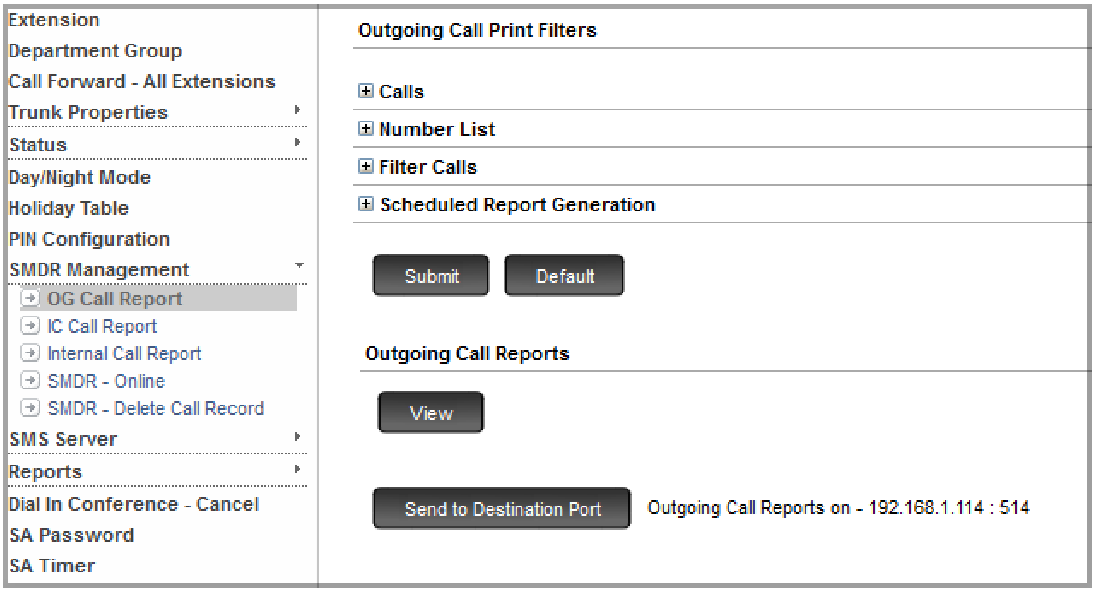
- Click **Scheduled Report Generation** to expand.



- 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**.

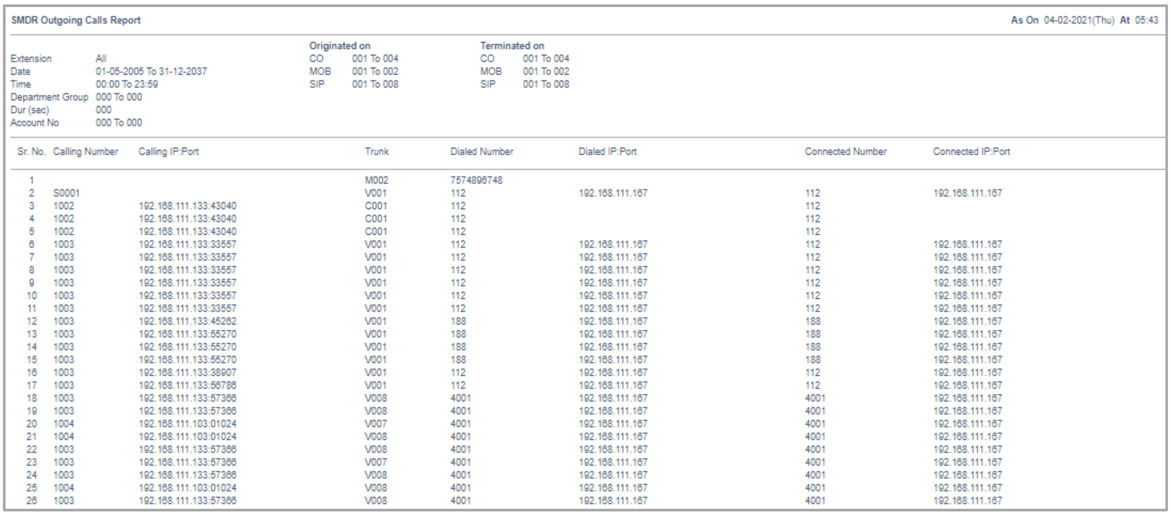
Manual Report Generation



- 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 the report /print the report on the local printer,

- Click the **View** button.



SMDR Outgoing Calls Report — As On 04-02-2021(Thu) At 05:43

Extension	All		Originated on		Terminated on		
Date	01-05-2005 To 31-12-2037	CO	001 To 004	CO	001 To 004		
Time	00:00 To 23:59	MOB	001 To 002	MOB	001 To 002		
Department Group	000 To 000	SIP	001 To 008	SIP	001 To 008		
Dur (sec)	000						
Account No	000 To 000						

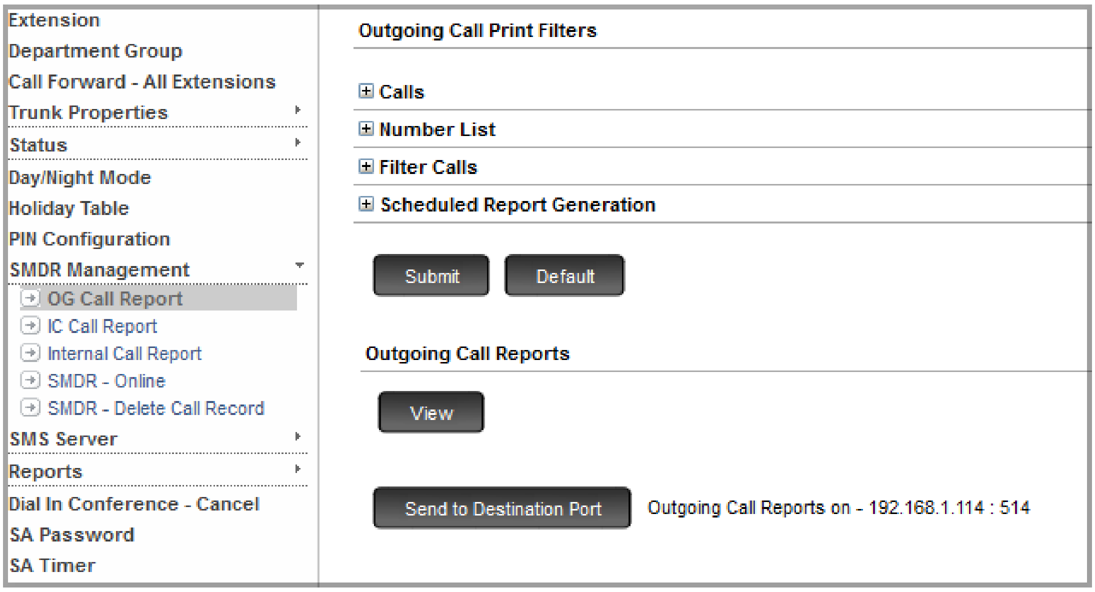
Sr. No.	Calling Number	Calling IP:Port	Trunk	Dialed Number	Dialed IP:Port	Connected Number	Connected IP:Port
1			M002	7574896748			
2	S0001		V001	112	192.168.111.167	112	192.168.111.167
3	1002	192.168.111.133:43040	C001	112		112	
4	1002	192.168.111.133:43040	C001	112		112	
5	1002	192.168.111.133:43040	C001	112		112	
6	1003	192.168.111.133:33557	V001	112	192.168.111.167	112	192.168.111.167
7	1003	192.168.111.133:33557	V001	112	192.168.111.167	112	192.168.111.167
8	1003	192.168.111.133:33557	V001	112	192.168.111.167	112	192.168.111.167
9	1003	192.168.111.133:33557	V001	112	192.168.111.167	112	192.168.111.167
10	1003	192.168.111.133:33557	V001	112	192.168.111.167	112	192.168.111.167
11	1003	192.168.111.133:33557	V001	112	192.168.111.167	112	192.168.111.167
12	1003	192.168.111.133:45262	V001	188	192.168.111.167	188	192.168.111.167
13	1003	192.168.111.133:55270	V001	188	192.168.111.167	188	192.168.111.167
14	1003	192.168.111.133:55270	V001	188	192.168.111.167	188	192.168.111.167
15	1003	192.168.111.133:55270	V001	188	192.168.111.167	188	192.168.111.167
16	1003	192.168.111.133:38907	V001	112	192.168.111.167	112	192.168.111.167
17	1003	192.168.111.133:56786	V001	112	192.168.111.167	112	192.168.111.167
18	1003	192.168.111.133:57366	V008	4001	192.168.111.167	4001	192.168.111.167
19	1003	192.168.111.133:57366	V008	4001	192.168.111.167	4001	192.168.111.167
20	1004	192.168.111.103:01024	V007	4001	192.168.111.167	4001	192.168.111.167
21	1004	192.168.111.103:01024	V008	4001	192.168.111.167	4001	192.168.111.167
22	1003	192.168.111.133:57366	V008	4001	192.168.111.167	4001	192.168.111.167
23	1003	192.168.111.133:57366	V007	4001	192.168.111.167	4001	192.168.111.167
24	1003	192.168.111.133:57366	V008	4001	192.168.111.167	4001	192.168.111.167
25	1004	192.168.111.103:01024	V008	4001	192.168.111.167	4001	192.168.111.167
26	1003	192.168.111.133:57366	V008	4001	192.168.111.167	4001	192.168.111.167

- A detailed report is displayed on the screen.

- Click **Print** to print the report.

To print the report on the Destination Port,

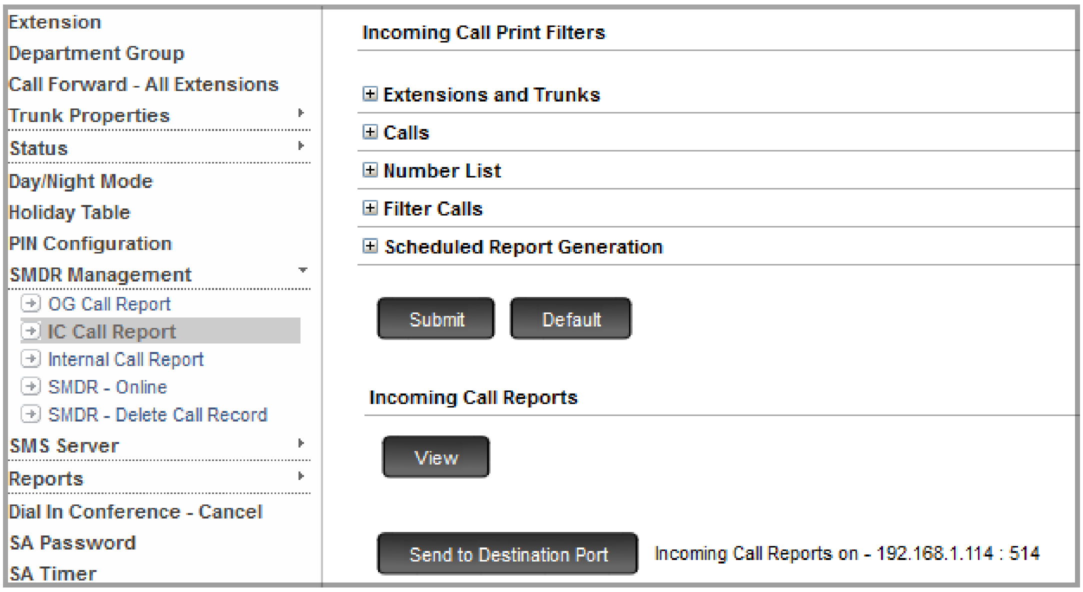
- Click **Send to Destination Port**.



- To stop printing, click **Abort**.

Incoming Calls

- To print Incoming Calls with filters, click **IC Call Report**.



Setting Print Filters

- Click **Extensions and Trunks** to expand.

- Set the filters as desired. You can print reports of incoming calls received on all extensions, a specific extension or a range of extensions and trunk ports: **CO**, **Mobile**, **SIP**.

  To print reports 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.

- Click **Submit.**

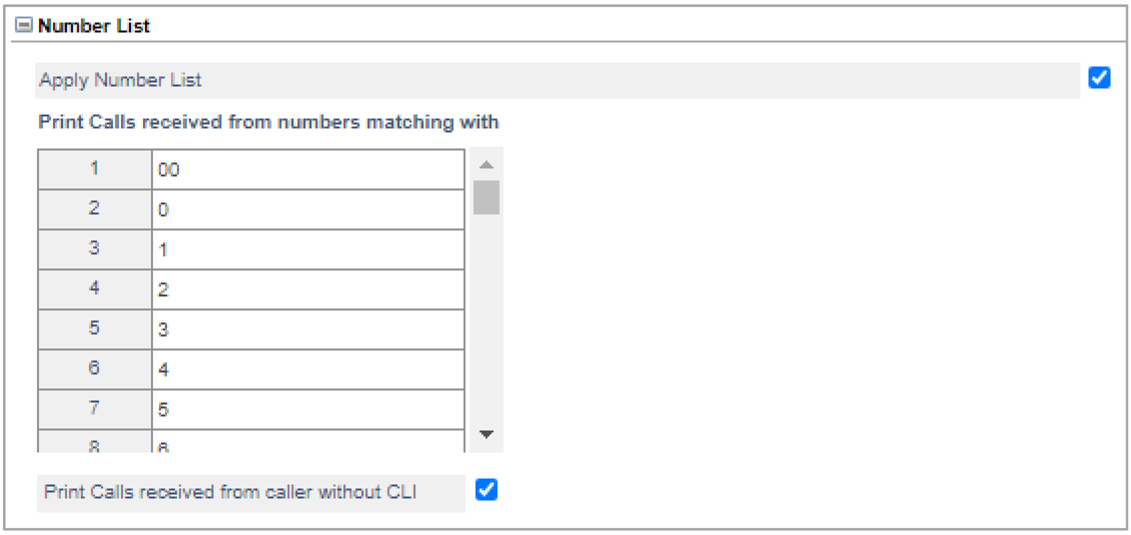
- Click **Calls** to expand.

  - 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**.

    By default, the report will contain all types of calls. To disable a particular option, clear its respective check box.

- Click **Submit.**

- Click **Number List** to expand.



  - Select the **Apply Number List** check box to print calls from certain numbers and configure **Print Calls received with numbers matching with** and P**rint Calls received from caller without CLI.**

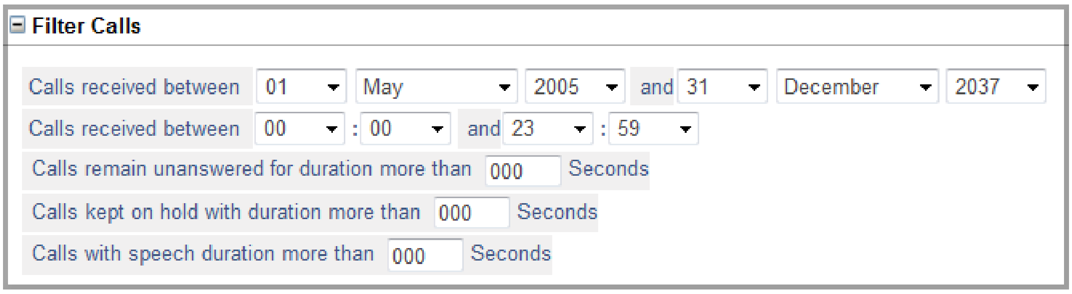
    To print incoming calls received from certain numbers, enter the CLI of these numbers in the **Print Calls received with numbers matching with** fields.

    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.*

- Click **Submit.**

- 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)**.

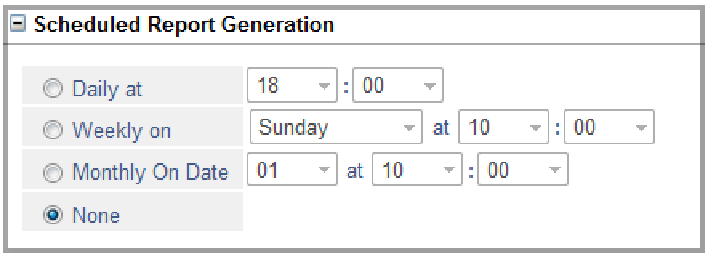
- To print calls that were kept on hold for more than a certain duration, set the filter **Calls kept on hold with duration more than (seconds)** to the desired value.

- To print calls with speech duration of a certain duration, set the filter **Calls with speech duration more than (seconds)**

- Click **Submit.**

Scheduled Report Generation

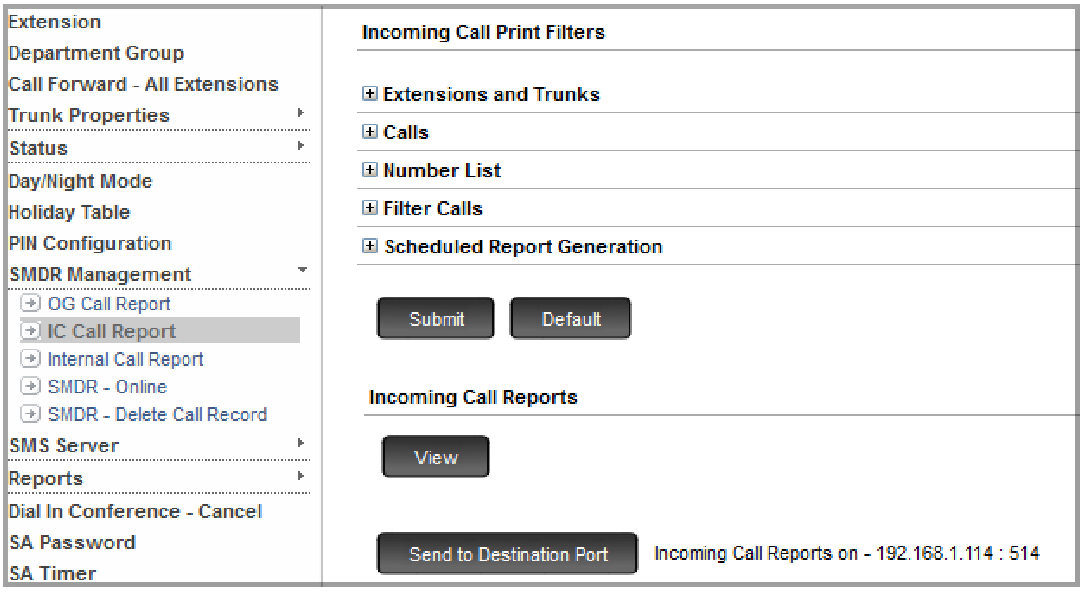
- Click **Scheduled Report Generation** to expand.



- 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.**

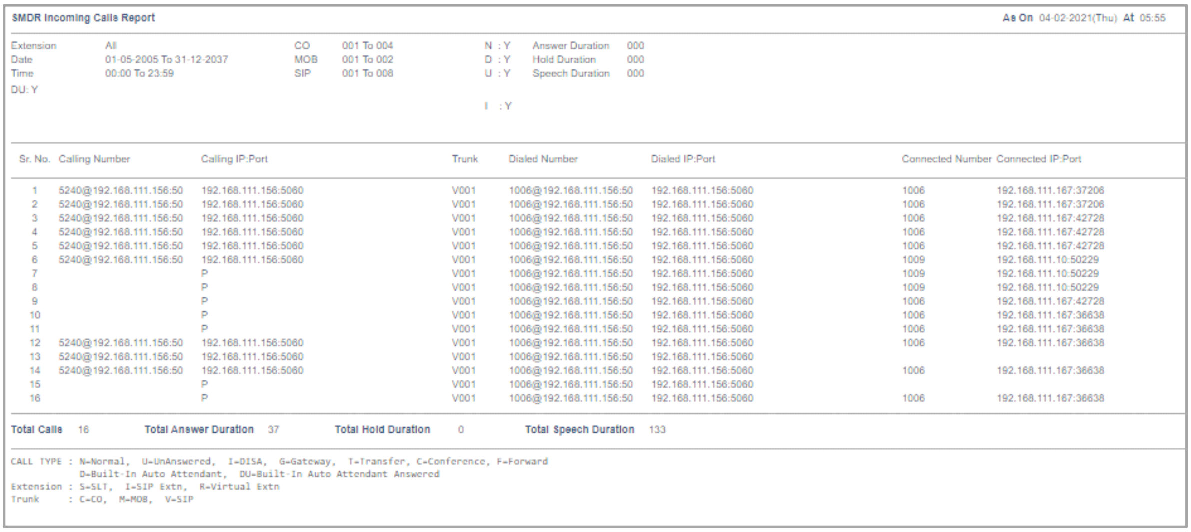
Manual Report Generation



- 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 the report /print the report on the local printer,

  - Click the **View** button.



SMDR Incoming Calls Report — **As On** 04-02-2021(Thu) **At** 05:55

Extension	All	CO	001 To 004	N	: Y	Answer Duration	000
Date	01-05-2005 To 31-12-2037	MOB	001 To 002	D	: Y	Hold Duration	000
Time	00:00 To 23:59	SIP	001 To 008	U	: Y	Speech Duration	000
DU: Y							
				I	: Y		

Sr. No.	Calling Number	Calling IP:Port	Trunk	Dialed Number	Dialed IP:Port	Connected Number	Connected IP:Port
1	5240@192.168.111.156:50	192.168.111.156:5060	V001	1006@192.168.111.156:50	192.168.111.156:5060	1006	192.168.111.167:37206
2	5240@192.168.111.156:50	192.168.111.156:5060	V001	1006@192.168.111.156:50	192.168.111.156:5060	1006	192.168.111.167:37206
3	5240@192.168.111.156:50	192.168.111.156:5060	V001	1006@192.168.111.156:50	192.168.111.156:5060	1006	192.168.111.167:42728
4	5240@192.168.111.156:50	192.168.111.156:5060	V001	1006@192.168.111.156:50	192.168.111.156:5060	1006	192.168.111.167:42728
5	5240@192.168.111.156:50	192.168.111.156:5060	V001	1006@192.168.111.156:50	192.168.111.156:5060	1006	192.168.111.167:42728
6	5240@192.168.111.156:50	192.168.111.156:5060	V001	1006@192.168.111.156:50	192.168.111.156:5060	1009	192.168.111.10:50229
7		P	V001	1006@192.168.111.156:50	192.168.111.156:5060	1009	192.168.111.10:50229
8		P	V001	1006@192.168.111.156:50	192.168.111.156:5060	1009	192.168.111.10:50229
9		P	V001	1006@192.168.111.156:50	192.168.111.156:5060	1006	192.168.111.167:42728
10		P	V001	1006@192.168.111.156:50	192.168.111.156:5060	1006	192.168.111.167:36638
11		P	V001	1006@192.168.111.156:50	192.168.111.156:5060	1006	192.168.111.167:36638
12	5240@192.168.111.156:50	192.168.111.156:5060	V001	1006@192.168.111.156:50	192.168.111.156:5060	1006	192.168.111.167:36638
13	5240@192.168.111.156:50	192.168.111.156:5060	V001	1006@192.168.111.156:50	192.168.111.156:5060		
14	5240@192.168.111.156:50	192.168.111.156:5060	V001	1006@192.168.111.156:50	192.168.111.156:5060	1006	192.168.111.167:36638
15		P	V001	1006@192.168.111.156:50	192.168.111.156:5060		
16		P	V001	1006@192.168.111.156:50	192.168.111.156:5060	1006	192.168.111.167:36638

Total Calls 16 **Total Answer Duration** 37 **Total Hold Duration** 0 **Total Speech Duration** 133

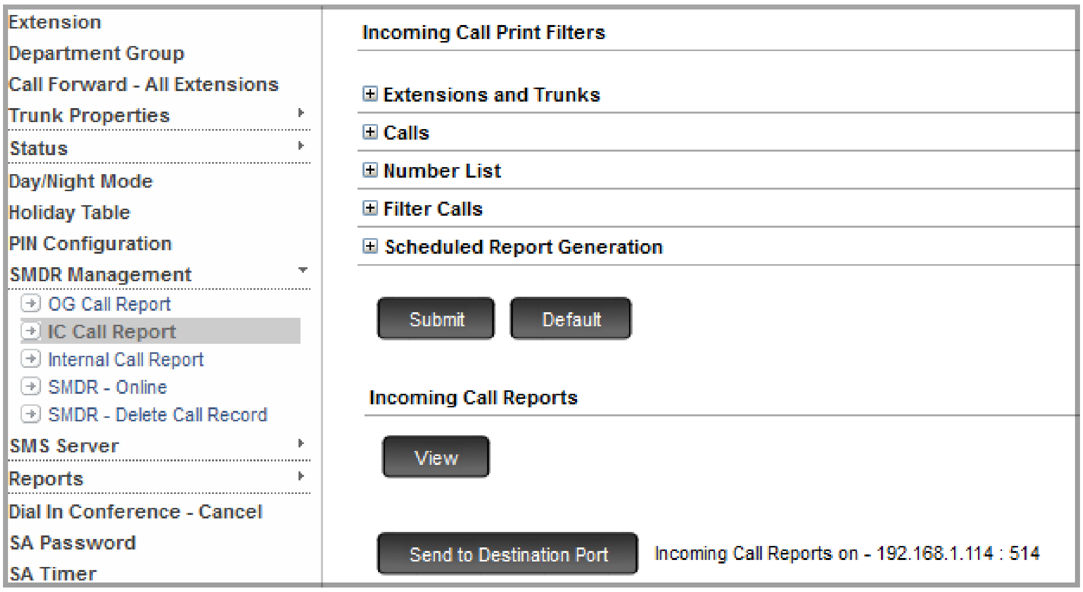
```
CALL TYPE : N=Normal,  U=UnAnswered,  I=DISA,  G=Gateway,  T=Transfer, C=Conference, F=Forward
            D=Built-In Auto Attendant,  DU=Built-In Auto Attendant Answered
Extension : S=SLT,  I=SIP Extn,  R=Virtual Extn
Trunk     : C=CO,  M=MOB,  V=SIP
```

  - A detailed report is displayed on the screen.

  - Click **Print** to print the report.

To print the report on the Destination Port,

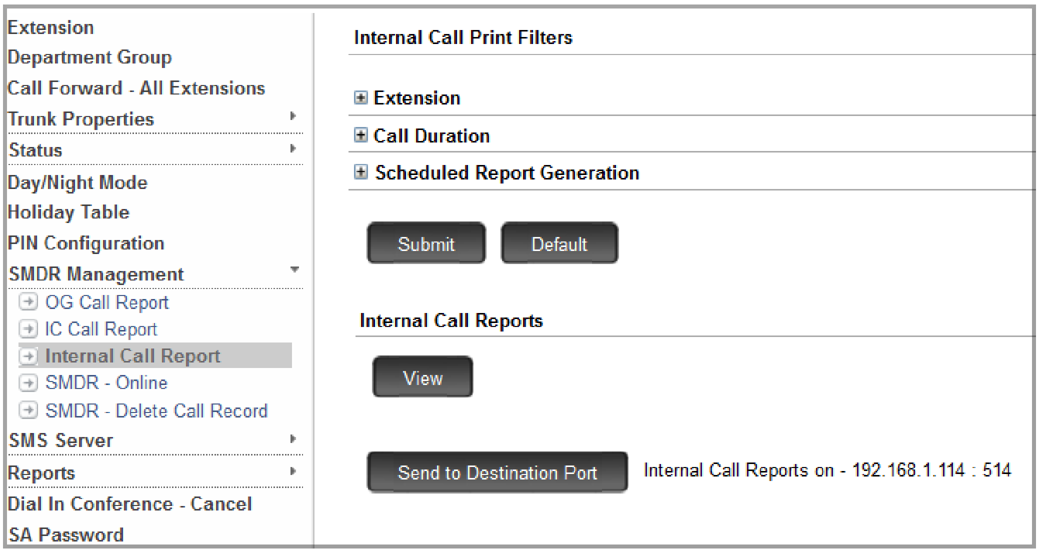
- Click **Send to Destination Port**.



- To stop printing, click **Abort**.

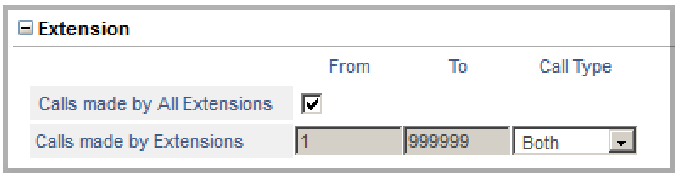
Internal Calls

- To print Internal call Report with filters, click **Internal Call Report** link.



Setting Print Filters

- Click **Extension** 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.

  You can also print calls made and calls received by extensions by selecting the **Call Type**.

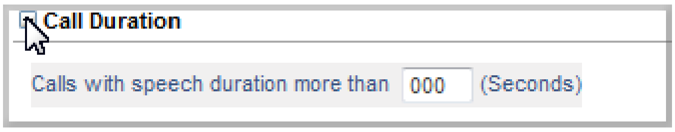
  - Select **Both** as Call Type, to print calls made and received by the extensions.
  - Select **Caller** as Call Type, to print only those calls that were made by the extension.
  - Select **Receiver** as Call Type, to print only those calls that were received by the extension.
  - Select **None**, if you do not want to use the Call Type filter.

  To print calls made by a particular extension or a range of extensions, clear the **Calls made by All Extension** 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**.

- Click **Call Duration** to expand.

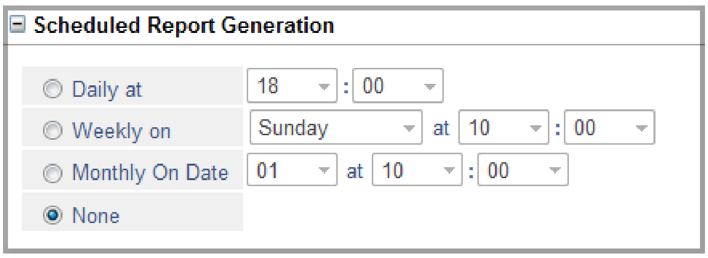


  - To print calls with speech duration of a certain duration, set the filter **Calls with speech duration more than (seconds).**

- Click **Submit**.

Scheduled Report Generation

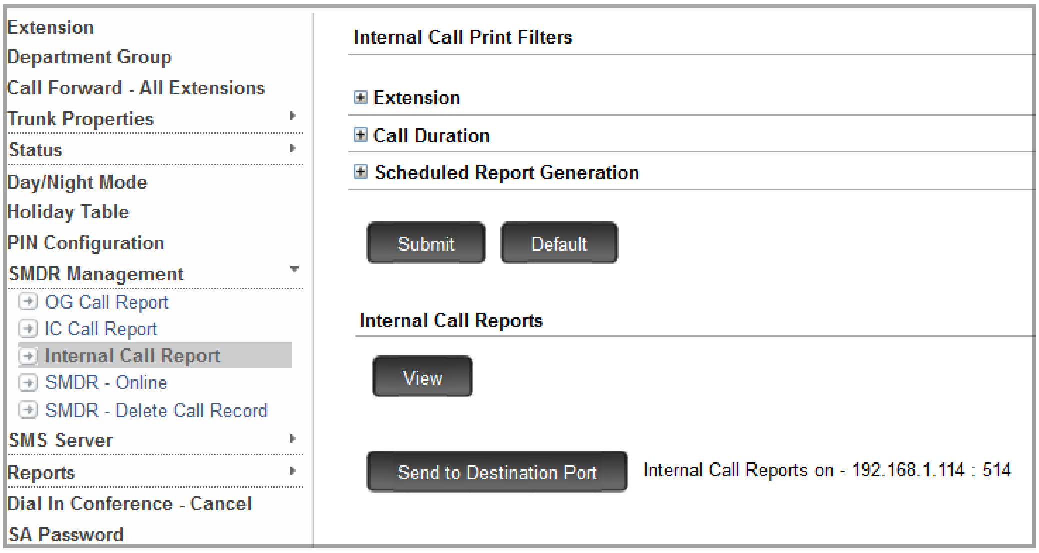
- Click **Scheduled Report Generation** to expand.



- 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**.

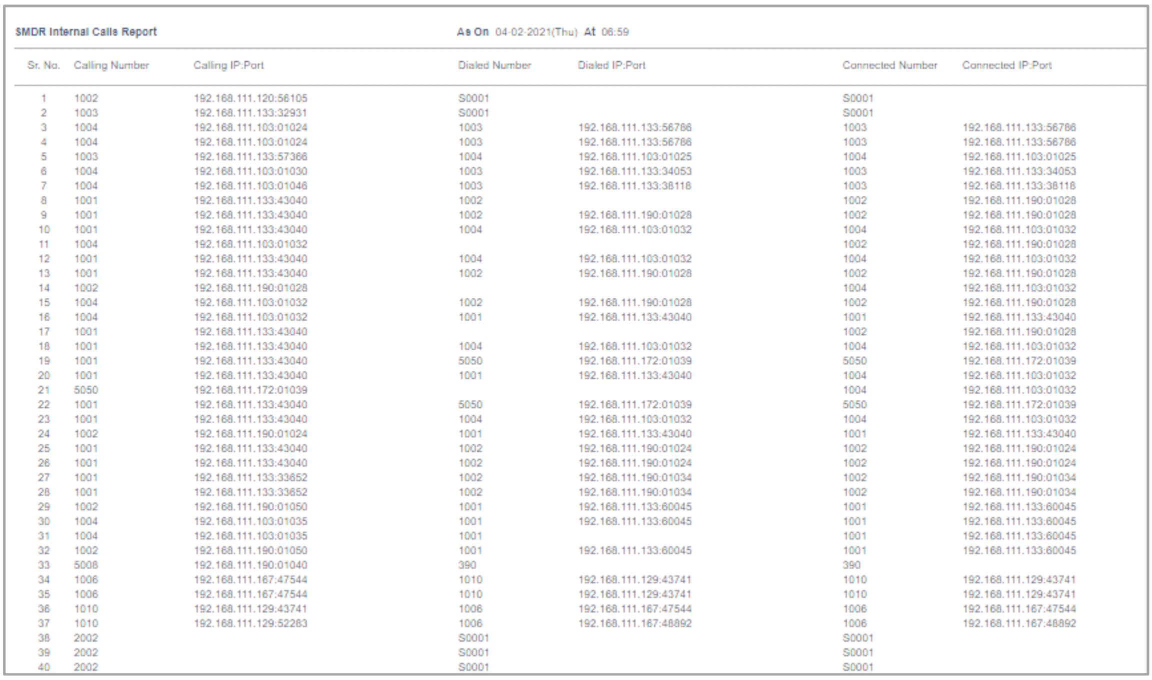
Manual Report Generation



- 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 the report /print the report on the local printer,

  - Click the **View button**.



SMDR Internal Calls Report **As On** 04-02-2021(Thu) **At** 06:59

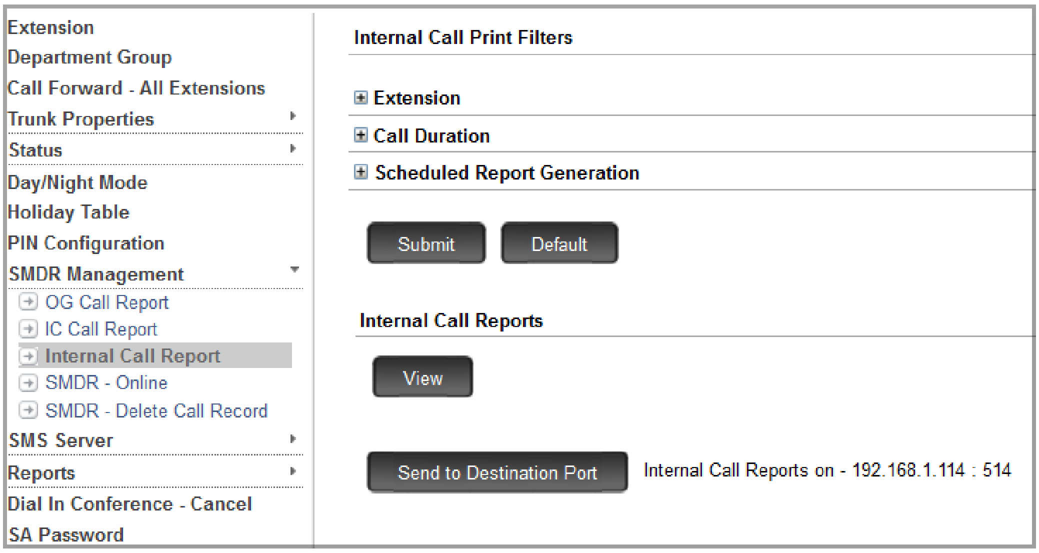
Sr. No.	Calling Number	Calling IP:Port	Dialed Number	Dialed IP:Port	Connected Number	Connected IP:Port
1	1002	192.168.111.120:56105	S0001		S0001	
2	1003	192.168.111.133:32931	S0001		S0001	
3	1004	192.168.111.103:01024	1003	192.168.111.133:56786	1003	192.168.111.133:56786
4	1004	192.168.111.103:01024	1003	192.168.111.133:56786	1003	192.168.111.133:56786
5	1003	192.168.111.133:57366	1004	192.168.111.103:01025	1004	192.168.111.103:01025
6	1004	192.168.111.103:01030	1003	192.168.111.133:34053	1003	192.168.111.133:34053
7	1004	192.168.111.103:01046	1003	192.168.111.133:38118	1003	192.168.111.133:38118
8	1001	192.168.111.133:43040	1002		1002	192.168.111.190:01028
9	1001	192.168.111.133:43040	1002	192.168.111.190:01028	1002	192.168.111.190:01028
10	1001	192.168.111.133:43040	1004	192.168.111.103:01032	1004	192.168.111.103:01032
11	1004	192.168.111.103:01032			1002	192.168.111.190:01028
12	1001	192.168.111.133:43040	1004	192.168.111.103:01032	1004	192.168.111.103:01032
13	1001	192.168.111.133:43040	1002	192.168.111.190:01028	1002	192.168.111.190:01028
14	1002	192.168.111.190:01028			1004	192.168.111.103:01032
15	1004	192.168.111.103:01032	1002	192.168.111.190:01028	1002	192.168.111.190:01028
16	1004	192.168.111.103:01032	1001	192.168.111.133:43040	1001	192.168.111.133:43040
17	1001	192.168.111.133:43040			1002	192.168.111.190:01028
18	1001	192.168.111.133:43040	1004	192.168.111.103:01032	1004	192.168.111.103:01032
19	1001	192.168.111.133:43040	5050	192.168.111.172:01039	5050	192.168.111.172:01039
20	1001	192.168.111.133:43040	1001	192.168.111.133:43040	1004	192.168.111.103:01032
21	5050	192.168.111.172:01039			1004	192.168.111.103:01032
22	1001	192.168.111.133:43040	5050	192.168.111.172:01039	5050	192.168.111.172:01039
23	1001	192.168.111.133:43040	1004	192.168.111.103:01032	1004	192.168.111.103:01032
24	1002	192.168.111.190:01024	1001	192.168.111.133:43040	1001	192.168.111.133:43040
25	1001	192.168.111.133:43040	1002	192.168.111.190:01024	1002	192.168.111.190:01024
26	1001	192.168.111.133:43040	1002	192.168.111.190:01024	1002	192.168.111.190:01024
27	1001	192.168.111.133:33652	1002	192.168.111.190:01034	1002	192.168.111.190:01034
28	1001	192.168.111.133:33652	1002	192.168.111.190:01034	1002	192.168.111.190:01034
29	1002	192.168.111.190:01050	1001	192.168.111.133:60045	1001	192.168.111.133:60045
30	1004	192.168.111.103:01035	1001	192.168.111.133:60045	1001	192.168.111.133:60045
31	1004	192.168.111.103:01035	1001		1001	192.168.111.133:60045
32	1002	192.168.111.190:01050	1001	192.168.111.133:60045	1001	192.168.111.133:60045
33	5008	192.168.111.190:01040	390		390	
34	1006	192.168.111.167:47544	1010	192.168.111.129:43741	1010	192.168.111.129:43741
35	1006	192.168.111.167:47544	1010	192.168.111.129:43741	1010	192.168.111.129:43741
36	1010	192.168.111.129:43741	1006	192.168.111.167:47544	1006	192.168.111.167:47544
37	1010	192.168.111.129:52263	1006	192.168.111.167:48892	1006	192.168.111.167:48892
38	2002		S0001		S0001	
39	2002		S0001		S0001	
40	2002		S0001		S0001	

  - A detailed report is displayed on the screen.

- Click **Print** to print the report.

To print the report on the Destination Port,

- Click **Send to Destination Port**.



- To stop printing, click **Abort**.

You may print the Online report captured on the Syslog Server after suitable modification of the format.

Print SMDR-Report using a Telephone

To generate SMDR-Report from an extension phone,

- Enter SA mode from SLT/Extended IP Phone.

Print Filter Settings - Incoming Calls

To set Print Filters for incoming Calls,

- Dial the Commands listed in the table for the desired filter.

Filter	Commands	Values
Calls received on CO Port	**1072-161-CO-CO**	CO: 001 - 004
Calls received on Mobile Port	**1072-165-Mobile-Mobile**	Mobile: 01-02
Calls received on SIP Trunk	**1072-166-SIP-SIP**	SIP: 01 - 08
Print Normal Calls	**1072-152-Flag**	Flag: 0 = Disable 1 = Enable
Print Built-In Auto Attendant Calls	**1072-153-Flag**	Flag: 0 = Disable 1 = Enable

Filter	Commands	Values
Print Unanswered Calls	**1072-154-Flag**	Flag: 0 = Disable 1 = Enable
Print Built-In Auto Attendant Unanswered Calls	**1072-155-Flag**	Flag: 0 = Disable 1 = Enable
Print DISA Calls	**1072-156-Flag**	Flag: 0 = Disable 1 = Enable
Calls received between Date	**1072-167-DD-MM-YYYY-DD-MM-YYYY**	DD: 01-31 MM: 01-12 YYYY: 0000-9999
Calls received between Time	**1072-168-HH-MM-HH-MM**	HH: 00-23 MM: 00-59
Calls remain unanswered for duration more than	**1072-158-Seconds**	Seconds: 000-999
Calls kept on hold with duration more than	**1072-159-Seconds**	Seconds: 000-999
Calls with speech duration more than	**1072-157-Seconds**	Seconds: 000-999
Set Time for Scheduled report Daily	**1072-173-HH-MM**	HH: 00-23 MM: 00-59
Set Time for Scheduled report Weekly	**1072-174-Day-HH-MM**	Day: 1 - 7 HH: 00 - 23 MM: 00 - 59
Set Time for Scheduled report Monthly	**1072-175-Date-HH-MM**	Date: 01 - 31 HH: 00 - 23 MM: 00 - 59
Set filters to Default	**1072-170**	

- Exit SA mode.

Manual Report Generation

- Enter SA mode from SLT/Extended IP phone.

  To start/stop Report for **Incoming Calls**:

  - Dial **1072-171-1** to start
  - Dial **1072-171-0** to stop

- Exit SA mode.

Scheduled Report Generation

- Enter SA mode from SLT/Extended IP phone.

To enable Scheduled Report for **Incoming Calls**:

- Dial **1072-172-1** to enable Daily Scheduled Report.
- Dial **1072-172-2** to enable Weekly Scheduled Report.
- Dial **1072-172-3** to enable Monthly Scheduled Report.
- Dial **1072-172-0** to disable Scheduled Report.

- Exit SA mode.

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

To set Print Filters for Internal Calls,

- Dial the Commands listed in the table for the desired filter.

Filter	Command	Values
Calls with speech duration more than (seconds)	**1072-138-Seconds**	Seconds: 000-999
Set Time for Scheduled report Daily	**1072-143-HH-MM**	HH: 00-23 MM: 00-59
Set Time for Scheduled report Weekly	**1072-144-Day-HH-MM**	Day: 1 - 7 HH: 00 - 23 MM: 00 - 59
Set Time for Scheduled report Monthly	**1072-145-Date-HH-MM**	Date: 01 - 31 HH: 00 - 23 MM: 00 - 59

- Exit SA mode.

Manual Report Generation

- Enter SA mode from SLT/Extended IP phone.

  To start/stop Report for **Internal Calls**:

  - Dial **1072-141-1** to start
  - Dial **1072-141-0** to stop

- Exit SA mode.

Scheduled Report Generation

- Enter SA mode from SLT/Extended IP phone.

  To enable Scheduled Report for **Internal Calls**:

  - Dial **1072-142-1** to enable Daily Scheduled Report.

- Dial **1072-142-2** to enable Weekly Scheduled Report.
- Dial **1072-142-3** to enable Monthly Scheduled Report.
- Dial **1072-142-0** to disable Scheduled Report.

- Exit SA mode.

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

To set Print Filters for Outgoing Calls,

- Dial the Commands listed in the table for the desired filter.

Filter	Commands	Value
Calls originated on CO	**1072 - 183 - CO - CO**	CO: 001 - 004
Calls originated on Mobile	**1072 - 187 - Mobile- Mobile**	Mobile: 01-02
Calls originated on SIP	**1072 - 188 - SIP- SIP**	SIP: 01 - 08
Calls terminated on CO	**1072-103-CO-CO**	CO: 001 - 004
Calls terminated on Mobile	**1072-107-Mobile-Mobile**	Mobile: 01-02
Calls terminated on SIP	**1072-108-SIP-SIP**	SIP: 01 - 08
Calls made using Account Code	**1072-115-Account Code-Account Code**	Account Code: 000 - 999.
Calls for Department Billing Group	**1072-109-Dept Group No.-Dept. Group No.**	Dept. Group: 01 - 05
Calls made on or from date between	**1072-110-DD-MM-YYYY-DD-MM-YYYY**	DD: 01-31 MM: 01-12 YYYY: 0000-9999
Calls made at or between time	**1072-111-HH-MM-HH-MM**	HH: 00-23 MM: 00-59
Calls with duration more than	**1072-113-Seconds**	Seconds: 000-999
Calls with metering units more than	**1072-114-Units**	Units: 0000-9999
Time for Scheduled report Daily	**1072-123-HH-MM**	HH: 00-23 MM: 00-59
Time for Scheduled report Weekly	**1072-124-Day-HH-MM**	Day: 1 - 7 HH: 00 - 23 MM: 00 - 59
Time for Scheduled report Monthly	**1072-125-Date-HH-MM**	Date: 01 - 31 HH: 00 - 23 MM: 00 - 59
Set filters to Default	**1072-120**	

- Exit SA mode.

Delete calls made from an extension

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 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

- Enter SA mode from SLT/Extended IP phone.

  To start/stop Report for **Outgoing Calls**:

  - Dial **1072-121-1** to start
  - Dial **1072-121-0** to stop

- Exit SA mode.

Scheduled Report Generation

- Enter SA mode from SLT/Extended IP phone.

  To enable Scheduled Report for **Outgoing Calls**:

  - Dial **1072-122-1** to enable Daily Scheduled Report.
  - Dial **1072-122-2** to enable Weekly Scheduled Report.
  - Dial **1072-122-3** to enable Monthly Scheduled Report.
  - Dial **1072-122-0** to disable Scheduled Report.

- Exit SA mode.

*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.*

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, SARVAM UCS offers the flexibility to send call detail records using the protocol supported by CAS. SARVAM UCS supports various CAS protocols such as, Holidex, Hobic, Micros A, Micros B, Comm One, Call-Inn, Bell-HOBIC, XIOX, RSI and others. Each of these are described below.

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 the 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 TCP/IP Ethernet port. The CAS is interfaced on the Ethernet port of the SARVAM UCS. For every outgoing call, call detail record is posted on the Ethernet port.

SMDR-Posting sends outgoing call records only.

SMDR-Posting Protocols

SARVAM UCS supports different posting protocols from the system to CAS. The flow of messages between the SARVAM UCS and various 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

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, 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

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, 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.	

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, SARVAM UCS will log this message in the System Fault Log and look for new message to be sent 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 Call Detail Record formats for various protocols 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	
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	Yes	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	Continuou s	Left	NA	NA	
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						
Currency Symbol (Enter Decimal Value)	000 000 000 000 000 000 000 000						

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	13	04	Fixed	Left	Yes	032	As per the Programmed String
Extension Number	06	06	Fixed	Right	Yes	032	
Trunk Number	17	05	Matrix Format	Left	Yes	032	
Date	39	10	DD-MM-YYYY	Right	Yes	032	MM/DD
Time	50	08	HH:MM:SS	Right	Yes	032	HH:MM
Duration	059	005	Seconds	Right	Yes	032	Duration is in Minutes.
Units	064	005	Fixed	Right	Yes	032	
Amount	069	008	Currency with Decimal Point	Right	Yes	032	Format is DDD.CC
Currency	000	001	Fixed	Right	Yes	032	$
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.

Parameter	Start Column Number	Field Length	Format	Alignment	Filler Char. Required?	Filler Char. Decimal Value	Remarks
Location	000	005	Fixed	Right	NA	NA	
Called Number	22	16	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	83	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						
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	
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	Yes	032	Country Specific
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	
PIN	00	04	Fixed	Right	Yes	032	--
Account Code	00	04	Fixed	Right	Yes	032	--

Parameter	Start Column Number	Field Length	Format	Alignment	Filler Char. Required?	Filler Char. Decimal Value	Remarks
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						
Currency Symbol (Enter Decimal Value)	000 000 000 000 000 000 000 000						

- *The Called Number will be left aligned.*
- *If the actual dialed number is less than the specified field width, the number will be sent as per the programmed alignment.*
- *If the dialed number length is greater than the width of the dialed number field, the trailing number digits will be removed. For example: if the number dialed is 15134036508 (11 digits) and the width of the called number field is 8, then the first 8 digits will be sent.*

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 will need the following functional components to make the interface work:

- A computer with a spare Ethernet port (not supplied by Matrix) Or any free Ethernet Port of the LAN Switch on which the CAS server application software is running.
- The CAS Software (not supplied by Matrix).
- SARVAM UCS.

Now, connect the Ethernet port of SARVAM UCS to the computer (on which CAS server application is running) or to one of the Ethernet ports of the LAN Switch, if the CAS server is in the same LAN.

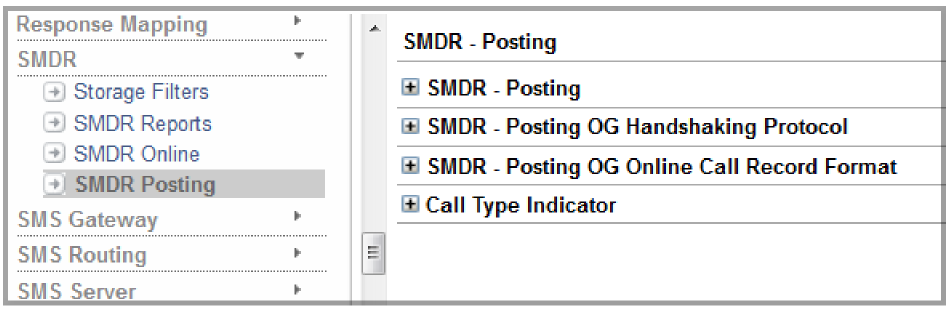
How to configure

Configuring the SMDR-Posting feature involves the following steps:

- Enabling storage of SMDR for Outgoing (OG) Calls. 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.
    - If SMDR-Posting is through TCP/IP (i.e. the CAS Interface is to be set up on the Ethernet 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.

To configure SMDR posting,

- Login as System Engineer.
- Under **Advanced Settings**, click **SMDR**.
- Click **SMDR Posting**.



Selecting SMDR-Posting Protocol for CAS

- Click **SMDR Posting** to expand.

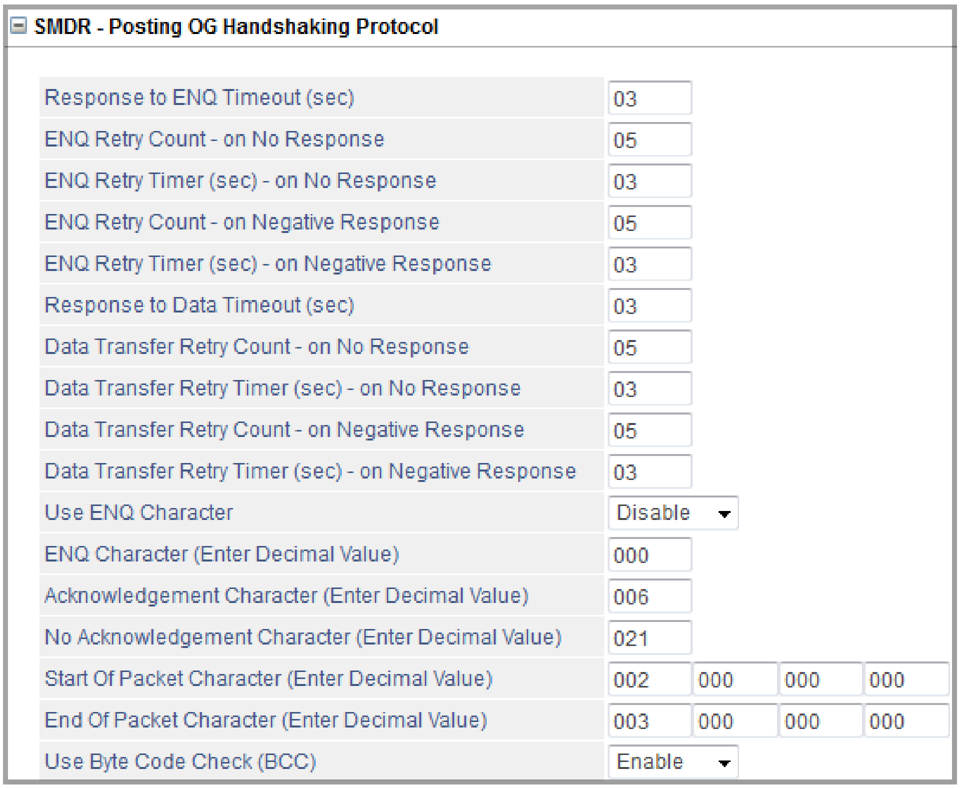
- In the **SMDR-OG Posting Protocol** (Handshaking and OG Call Record Format) drop down list, select the appropriate protocol to be used. Default: Matrix.
- In the **Destination Port**, select **Ethernet**. This is Ethernet Port of SARVAM UCS on which SMDR Posting is set up. Default: None.
- In the **Destination IP Address**, enter the IP Address of the PC on which the CAS server application software is running, i.e. where SARVAM UCS should post SMDR.
- 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: 5000, 514, 1025 to 65535.
- To start SMDR posting, select **Process** as **Start**.
- Click **Submit**.

Refining Handshake Parameters

You may need to refine some of the Handshake parameters of the selected SMDR-Posting protocol, i.e. 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** option to expand.



- 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. Default: 03 seconds.
  - **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 Timer (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 (Enter Decimal Value):** The ASCII character (Single Character) used to send ENQUIRE (ENQ) signal to the receiver.

- **Acknowledgement Character (Enter Decimal Value):** The ASCII character (Single Character) used by the receiver to acknowledge the receipt of the Link Control Character/Message Data.

- **No Acknowledgement Character (Enter Decimal Value):** 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 (Enter Decimal Value):** 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 which case the string should be completed by programming remaining three characters with ASCII Null Character (000).

- **End of Packet Character (Enter Decimal Value):** 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 which 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 check box is to be enabled when the protocol uses BCC Signal.

- Click **Submit**.

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** option to expand.

SMDR - Posting OG Online Call Record Format

Parameter	Start Column No.	Field Length	Format	Alignment	Filler Char. Required?	Filler Char. (Decimal Value)
Serial Number	001	004	Fixed	Right	Yes	032
Increment Counter	000	001	Fixed	Left	N/A	N/A
Property Code	000	004	Fixed	Left	Yes	032
Extension Number	006	006	Fixed	Right	Yes	032
Trunk Number	017	005	Matrix Format	Left	Yes	032
Date	041	008	DD-MM-YY	Right	Yes	032
Time	050	008	HH:MM:SS	Right	Yes	032
Duration	059	005	Seconds	Right	Yes	032
Units	065	004	Fixed	Right	Yes	032
Amount	070	007	Currency with Decimal Point	Right	Yes	032
Currency	000	001	Fixed	Right	Yes	032
Call Type Indicator	000	001	Fixed	Right	N/A	N/A
Location	000	005	Fixed	Right	N/A	N/A
Called Number	022	018	Continuous	Left	N/A	N/A
PIN	000	004	Fixed	Right	Yes	032
Account Code	000	004	Fixed	Right	Yes	032
Remarks	078	002	Fixed	Left	N/A	N/A
Reset Serial Number to 001	Do not Reset					
Starting Character - Increment Counter	A					

The Call Detail Format for OG-SMDR Posting Protocols consists of the following parameters. 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.

Refine the following format parameters according to the type of posting protocol you have selected and the requirement of the CAS being used by the organization.

- **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.*

- **Date:** The date on which the call was made. The date fill check box 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 check box of Date Format. The global check box 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 check box 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'.*

- **Duration:** The duration of each call. Program the duration unit and the duration fill check box.

- *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 check box.

- *Filler Character field is applicable for both the sub fields of Amount viz. Rupees/Paisa i.e. 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.*

- **Call Type Indicator:** This indicates the type of call made, i.e. whether local, national or international. Define the Call Types in the "Call Type Indicator".

  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.

- **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 (e.g. 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 check box is to be programmed if the prefix string 0ac1 is to be sent when interfacing with OG-SMDR Posting Protocol.

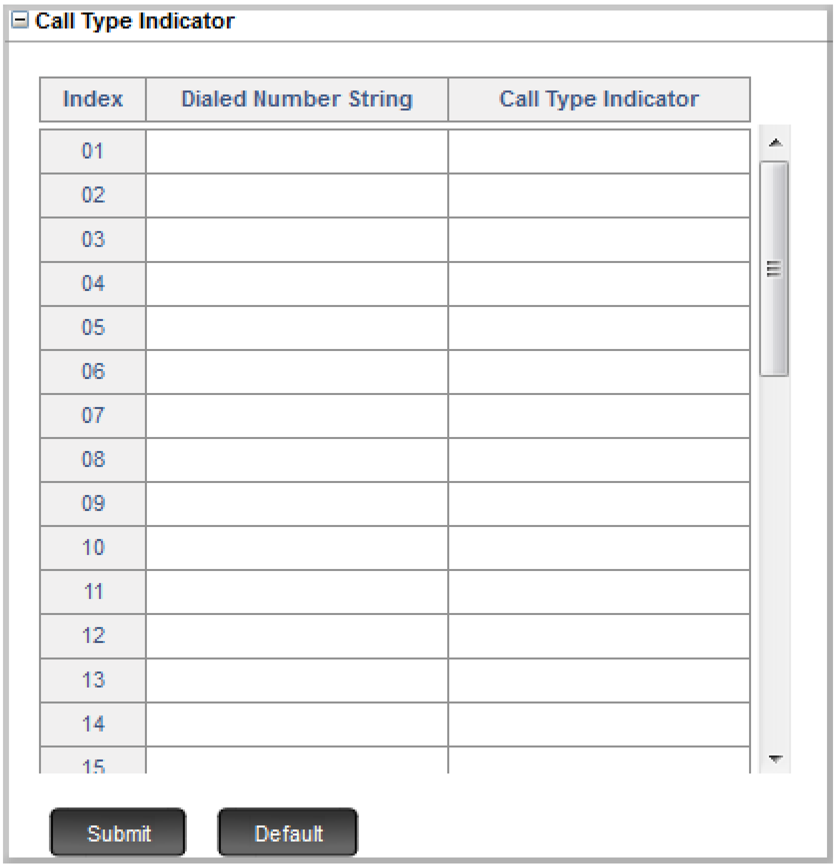
- **Property Code:** Enter the property code required by the CAS.

- **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 the **Call Type Indicator** option to expand.



- In the **Dialed Number String** column, enter the number strings for each Call Type. You can enter the prefix, example: 0 for long distance calls, 2 for local numbers, etc.

- For each **Dialed Number String**, define as **Call Type Indicator**, this is an abbreviation of the Call Type, example: 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.**

System Activity Log

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 printer or downloaded on a computer in the form of a report. The report 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.

The System Administrator can print or download System Activity Log, online or offline.

SARVAM UCS 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 can also send notifications for all the activities, as and when they occur to the concerned person via SMS and/or Email. For more details, refer to “System Log Notification”.

How it works

- The Syslog Server address must be assigned to which the system can send the activity log.
- If the System Administrator extension is 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 Extended IP Phone extension, is turned ON.
- The System Administrator can view the activity log by pressing the DSS key (if assigned). The Extended IP Phone of the System Administrator will display the activity in this format:

  DD-MM HH:MM <Activity Index>

*Format of 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	MATRIX SARVAM VxxRyy Started	The Version Revision of the system when the system starts, where xx = 2 digit Version number and yy = 2 digit Revision number
02	Default Configuration Loaded	After the system has completed loading the default configurations.
03	Card Present: Slot=xx, Type: CARD Type	where, xx = Slot number and CARD Type= Card type with version revision
04	SLT Normal: nnnnnn, Slot=00, Port=yy	The SLT port restored to idle state from stand-by mode where, nnnnnn = max. 6 digit flexible number assigned to SLT and yy = Port number
08	SMDR-IC buffer deletion: nnnnnn	The date and time when the SMDR of the IC calls buffer was deleted, where nnnnnn = max. 6 digit flexible number of the station
09	SMDR-OG buffer deletion:nnnnnn	The date and time when the SMDR of the OG calls buffer was deleted, where nnnnnn = max. 6 digit flexible number of the station
10	SMDR-Internal buffer deletion: nnnnnn	The date and time when the SMDR of the internal calls buffer was deleted, where nnnnnn = max. 6 digit flexible number of the station
11	SE Access From: nnnnnn	The extension number of the user, who accessed the SE mode using commands, where nnnnnn = max. 6 digit flexible number of the station
12	SA Access From: nnnnnn	The extension number of the user, who accessed the SA mode using commands, where nnnnnn = max. 6 digit flexible number of the station
13	Emergency Number Dialed: nnnnnn	When any emergency number is dialed by the system, where nnnnnn = max. 6 digit flexible number of the station
14	Login nnnnnn <Station name>	User Status (Present) via Access Codes - Date, Time and Extension Number, where nnnnnn = max. 6 digit flexible number of the station
15	Logout nnnnnn <Station name>	User Status (Absent) via Access Codes - Date, Time and Extension Number, where nnnnnn = max. 6 digit flexible number of the station
16	Reg Fail, Authentication PW Invalid, SIPTrk=xx	SIP Trunk registration failure due to invalid Authentication Password, where xx=SIP trunk number.
17	Reg Fail, Config Parameters Invalid, SIPTrk=xx	SIP Trunk registration failure due to invalid configuration of SIP parameters, where xx=SIP trunk number.
18	Stack Construct Authentication Fail, Slot=xx, Port=yy	Stack Construct Authenticate Fail, where xx=slot number and yy=port number.

Event Index	Activity	Description
19	Stack Construct IP Address Invalid, Slot=xx, Port=yy	Stack Construct Fails due to invalid IP Address, where xx=slot number and yy=port number.
20	Call Budget exhausted, xx - yyy, NAME	Assigned Call Budget exhausted for the particular trunk. where xx=port type, yyy=port offset and NAME=name of the trunk
23	Personal Mailbox is full for Extension nnnnnn mmmm	Personal Mailbox of the extension user/department group/general mailbox is full, along with the extension number/department group number/general mailbox number and extension name/general mailbox name where, nnnnnn = max. 6 digit flexible number of the station, mmmm= name
24	VMS USB Memory is Full	When 100% of the USB memory of VMS has been utilized.
25	Restart due to change Network Para Slot=xx, Port=yy	When the VoIP module restarts due to change in Network Parameters. where xx = Slot number and yy = Port number
26	Restart due to change SIP Para Slot=xx, Port=yy	When the VoIP module restarts due to change in SIP Parameters. where xx = Slot number and yy = Port number
27	Restart due to Master request Slot=xx, Port=yy	When the VoIP module restarts after a restart request is received from the CPU module.where xx = Slot number and yy = Port number
31	CPU - Ethernet link up	When the Ethernet port is connected and the network link is functioning.
32	CPU - Ethernet link down	When the Ethernet port is connected and the network link is not functioning.
36	VMS USB Memory 80% Full	When 80% of the USB memory of VMS has been utilized.
37	VMS USB Memory Usage in Limit	When less that 80% of the USB memory of VMS has been utilized.
38	DSP Frame Sync error, Slot=xx	Internal system DSP error, where xx=slot number
39	DSP Drop error, Slot=xx	Internal system DSP error, where xx=slot number
40	DSP Time Out error, Slot=xx	Internal system DSP error, where xx=slot number
41	DSP DMA Address error, Slot=xx	Internal system DSP error, where xx=slot number
42	Message Notification Retry Count over for nnnnnn	When Message notification via call is set as immediate and notification is not successful after retry count is over, where nnnnnn = max. 6 digit flexible number of the station
44	Call between nnnnnn and COxx dropped as prone to fake	Fraudulent call from CO Trunk, where nnnnnn = max. 6 digit flexible number of the station. xx=port number

Event Index	Activity	Description
45	Emergency Call of xxxxxx is Acknowledged by yyyyyy.	Emergency Call of an extension is acknowledged by another extension where xxxxxx = max. 6 digit flexible number of the station from which the Emergency Call was made and yyyyyy = max. 6 digit flexible number of the station which had acknowledged the Emergency call.
53	Forward DST Applied at: DD-MMM HH:MM	Forward DST is applied.
54	Backward DST Applied at: DD-MMM HH:MM	Backward DST is applied.
55	DST Config Change: Enabled/ Disabled	Change in DST Settings, that is settings are changed from Manual to Scheduled or vice versa or any changes are made in the Manual or Scheduled DST Settings.
56	System VxxRyy Started with System Command App	At every Power-On, when the System is up using System Application.
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_ADDR	The Date and Time on which the Login mode (SE/SA) and IP Address is blocked. (IP Address is the address from which the last failure attempt was made)
59	IP=IP_ADDR blocked for Auto Prov. of User SIP_ID	The Date and Time on which the IP Address is blocked for registration of the Standard SIP Phone. (IP Address and Port from which the last failure attempt was made.)
60	IP=IP_ADDR:PORT blocked	The Date and Time on which the IP Address is blocked for registration of the SIP extension. (IP Address and port from which the last failure attempt was made)
61	Send to Network Error for Connection ID XXX	XXX is the connection id for which the call is released by VoIP
62	Shadow+ file removed	The Date and Time on which the system removed the Shadow+ file.
63	General Mailbox is Full	
66	XY Login IP= Address	When anyone logs into the SE or SA mode. XY is the login mode- SE or SA
67	System Restart by YY	When the system is restarted from Web GUI or extension number using commands where, YY=IP address of extension number.
70	Extended Firmware is not upgraded	
71	Internal USB not detected	While firmware upgrade internal USB is not detected, hence upgrade will not be possible.

Event Index	Activity	Description
72	Mobile SIM:XX Removed/Inserted	When SIM card is removed or inserted. SIM Hot-swap.
73	LDAP Test completed successfully	When link establishment between LDAP server and client is tested successfully.
74	LDAP Sync done for Global Directory X, Y and Z	X, Y and Z=1/2/3 as per the Global directory synced with LDAP
75	Complex SIP Password - Enable/ Disable by IP_Address	The IP address from which Complex SIP password is enabled/disabled.
77	Validity expires in 0 days - CertificateFriendlyName	
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

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.

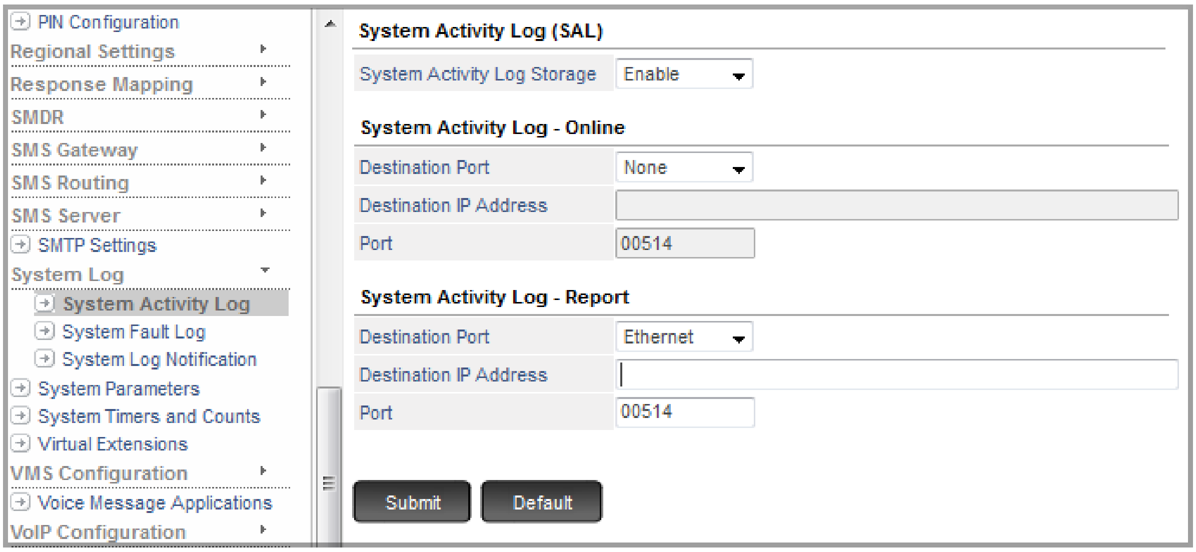
If you want to use Syslog Server, you must assign the IP Address of the remote Syslog Server as the Destination Port for the System Activity Log.

If the System Administrator extension is an Extended IP Phone, you may assign a DSS key for System Activity Log. For instructions, see **Phone Key Settings** under “Configuring SIP Extension Settings as per the Device Type” in *SIP Extensions*.

To configure System Activity Log settings,

- Login as System Engineer.
- Under **Advanced Settings**, click **System Log**.

- Click **System Activity Log**.



- Enable **System Activity Log Storage**.

- To generate **System Activity Log - Online**, i.e. as and when the activity is recorded, select **Destination Port**. Default: None.

  - In **Destination IP Address: Port**, enter the IP Address and the port of the remote Syslog Server. The valid range of this port is 514 or 1025 to 65535.

- To generate **System Activity Log - Report**, i.e. offline, whenever desired, select **Destination Port**. Default: None.

  - In the **Destination IP Address: Port**, enter the IP Address and the port of the remote Syslog Server. The valid range of this port is 514 or 1025 to 65535.

- Click **Submit**.

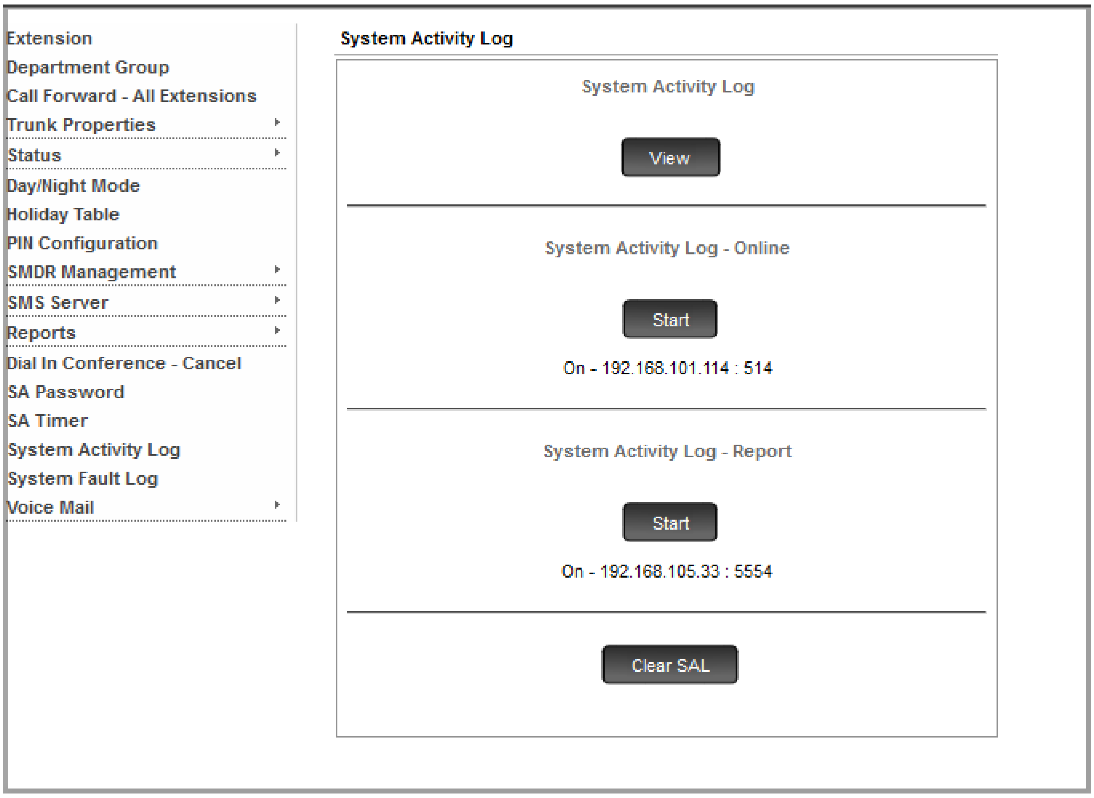
How to use

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,

- Login as System Administrator.

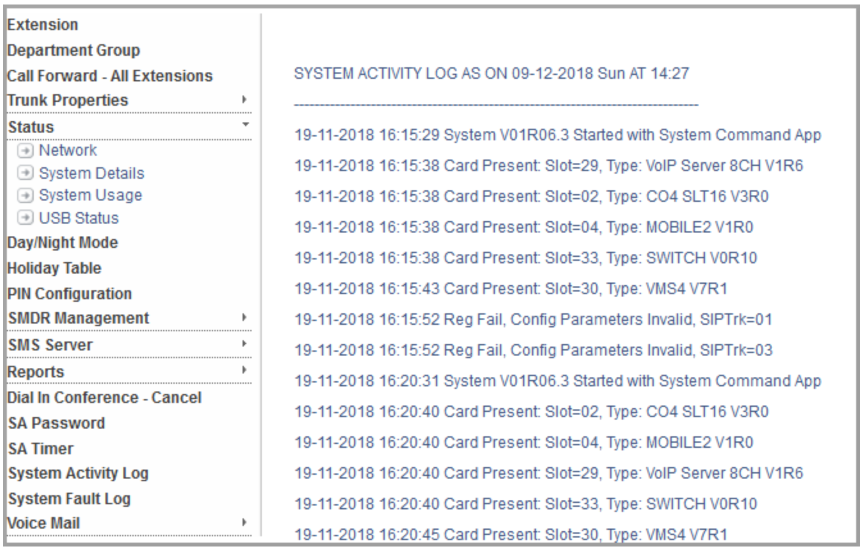
- Click **System Activity Log**.



- To start capturing the **System Activity Log - Online**, click **Start**.
- To stop capturing the **System Activity Log - Online**, click **Abort**.
- To start capturing the **System Activity Log - Report**, click **Start**.
- To stop capturing the **System Activity Log - Report**, click **Abort**.
- To clear System Activity Logs from the buffer, click **Clear SAL**.

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 **View.**



System Activity Logs appears on your screen.

- To print this report on the local printer connected to your computer, click **Print**. An alert message appears.

- Click **OK**. 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.

To generate Activity logs from an extension phone,

- Enter SA Mode from a 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.

System Activity Log Display

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 "System Activity Log".

How to use

To be able to use this feature optimally, the System Administrator extension phone must be an Extended IP Phone, and a DSS Key must be assigned on the phone to System Activity Log Display. For instructions, see **Phone Key Settings** under "Configuring SIP Extension Settings as per the Device Type" in *SIP Extensions*.

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. To know the index details, refer to "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

SARVAM UCS maintains a log of all the 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 the 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.

The System Administrator can print or download System Activity Log, online or offline.

SARVAM UCS 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 can also send notifications for the faults that occur in the system to the concerned person via SMS and/or Email. For more details, refer to "System Log Notification".

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 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 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.

The following table summarizes the different activities that are logged into system fault log.

Event Index	Event	Description
04	SLT Standby-Open: nnnnnn, Slot=xx, Port=yy	When the SLT reaches Open state from the Stand-by mode, where nnnnnn = max. 6 digit flexible number assinged to SLT, xx = Slot number and yy = Port number
09	RTC Failure	Reserved for future use
10	VoPP Program Download Fail, Mod=1	Internal failure, DSP fails to load/start the VoIP module
11	VoPP Fail, Mod=1	Internal failure, DSP fails to sync the VoIP module

Event Index	Event	Description
12	Registration Timer Fail, SIP Trunk=xx	SIP Trunk registration failed due to expiry of the Registration Timer, where xx=SIP trunk number
15	HPI Queue Fail, Slot=xx	The events in the queue fail for a particular module type, where xx=slot number
16	Slave DSP Alive Error, Slot=xx	DSP is not Alive, where xx=slot number
17	SMTP Error Code - xx[a]	VMS fails to send emails, where xx=error code number
18	VOPP-Failed to Open Kernel File, Mod=1	VoIP DSP Error
19	VOPP-Failed to Boot, Mod=1	VoIP DSP Error
24	Scheduled backup file of SMS Server is deleted	SMS Server Scheduled backup file is deleted
25	SLT Thermal Shutdown: nnnnnn, Slot=xx, Port=yy	SLT instrument is not functioning due to thermal shutdown, where nnnnnn= 6 digit flexible number, xx=slot number and yy=port number
31	LDAP Test Failed - Reason	When LDAP Test has failed.
32	LDAP Sync Failed - Reason	When LDAP synchronization with Global directory has failed.
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, SARVAM UCS VoIP module 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 SARVAM UCS 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 functioning normally. You need to restart the system to resolve the problem.

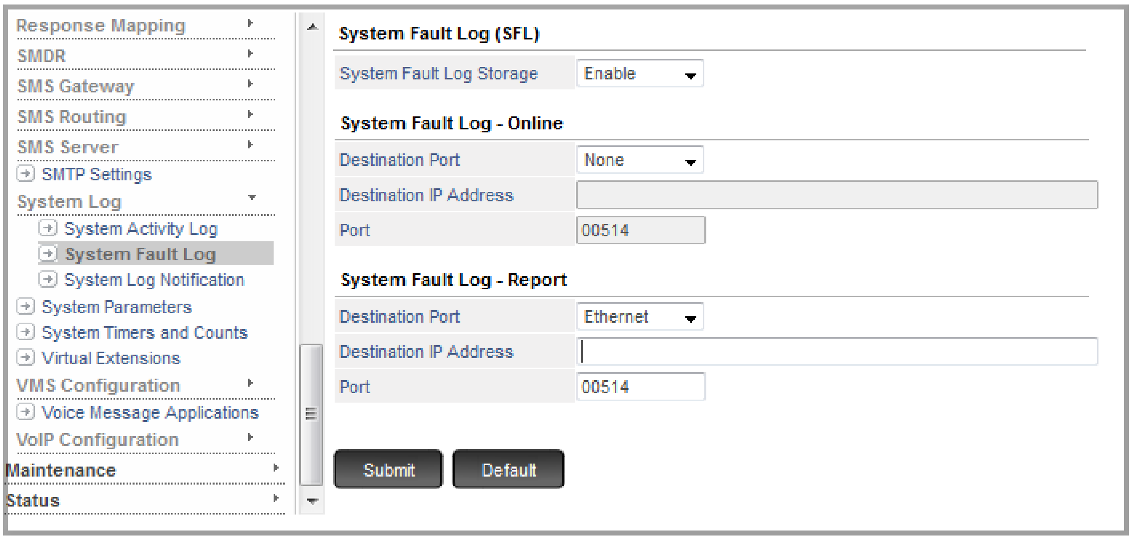
How to configure

To be able to use this feature, you must enable storage of Fault Logs, and assign a Destination Port for the Fault Logs.

If the System Administrator extension is an Extended IP Phone, you may assign a DSS key for System Fault Logs. For instructions, see **Phone Key Settings** under "Configuring SIP Extension Settings as per the Device Type" in *SIP Extensions*.

To configure System Fault Log settings,

- Login as System Engineer.
- Under **Advanced Settings**, click **System Log**.
- Click **System Fault Log**.



- Enable **System Fault Log Storage**.
- To generate **System Fault Log - Online**, i.e. as and when the fault occurs, select **Destination Port**. Default: None.
  - In **Destination IP Address: Port**, enter the IP Address and the port of the remote Syslog Server. The valid range of this port is 514 or 1025 to 65535.
- To generate **System Fault Log - Report**, i.e. offline, whenever desired, select **Destination Port**. Default: None.
  - In **Destination IP Address: Port**, enter the IP Address and the port of the remote Syslog Server. The valid range of this port is 514 or 1025 to 65535.
- Click **Submit**.

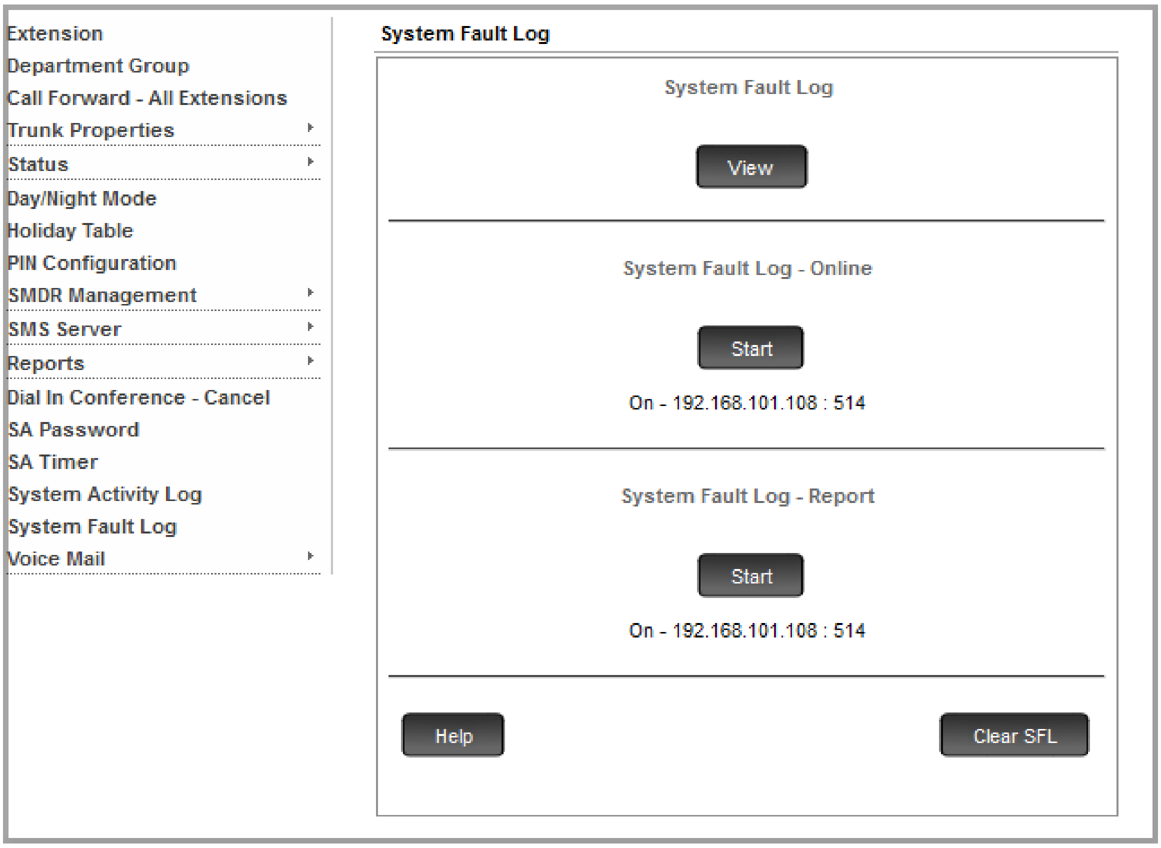
How to use

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 report generation using Jeeves,

- Login as System Administrator.

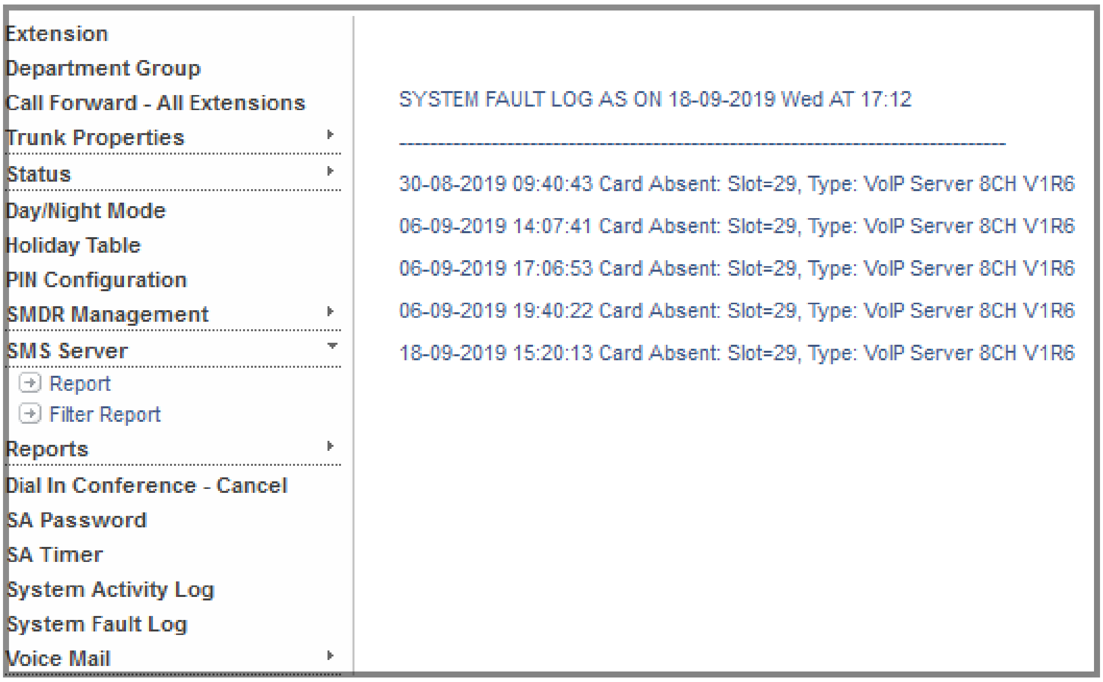
- Click **System Fault Log**.



- To start capturing the **System Fault Log - Online**, click **Start**.
- To stop capturing the **System Fault Log - Online**, click **Abort**.
- To start capturing the **System Fault Log - Report**, click **Start**.
- To stop capturing the **System Fault Log - Report**, click **Abort**.
- To clear System Fault Logs from the buffer, click **Clear SFL**.

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 **View**.



System Fault Log appears on your screen.

- To print this report on the local printer connected to your computer, click **Print**. An alert message appears.
- Click **OK**. 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.

To generate Fault logs from an extension phone,

- Enter SA mode from 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
Time (HH:MM:SS)	12
Even Index No. (two characters)	21
Event Description	24

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"
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

Error Code	Error Description
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

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 "System Fault Log".

How it works

To be able to use this feature optimally, the System Administrator extension phone must be 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. To know the index details, refer to "System Fault Log".

System Log Notification

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,

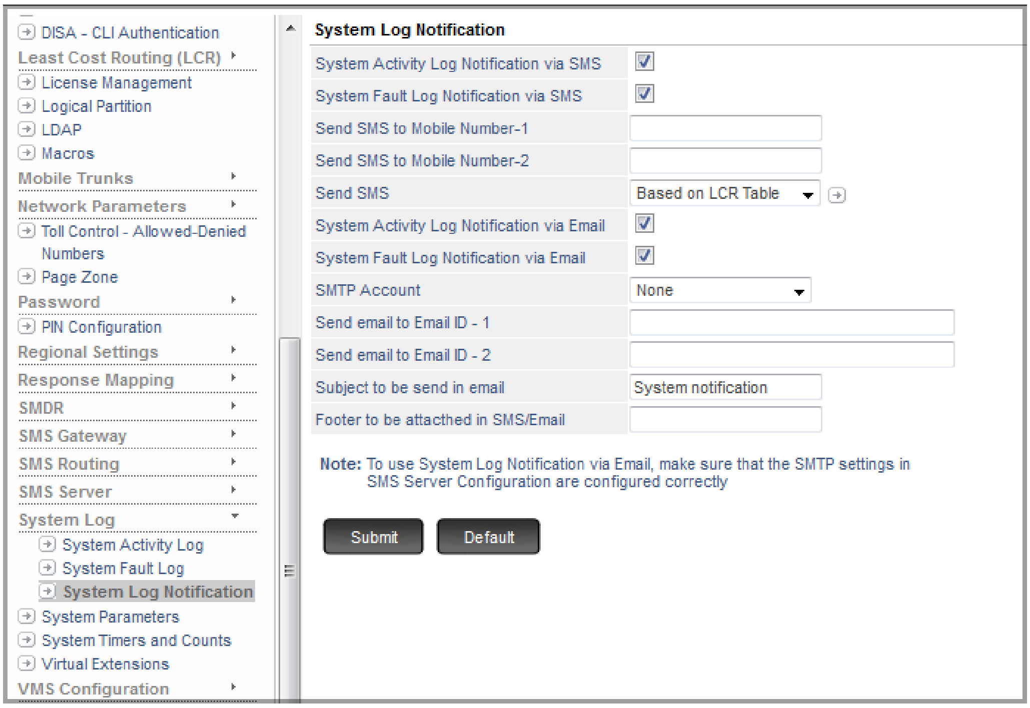
- as a message to the specific mobile number.
  or/and
- as an email to the specific email id.

SARVAM UCS maintains a log of all the system activities and faults. These logs can be printed on a printer or downloaded on a computer in the 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.

- Login as System Engineer.
- Under **Advanced Settings**, click **System Log**.
- Click **System Log Notification.**



- By default, **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 a fixed Mobile Port, select **Using Fixed Port** and click **Settings** , 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** and click **Settings** , the **SMS Routing - LCR** table opens. Configure the **SMS Routing - LCR** table. For detailed information, see "Least Cost Routing".
- By default, **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, **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 **SMTP Account,** select the desired SMTP account.
- 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.**

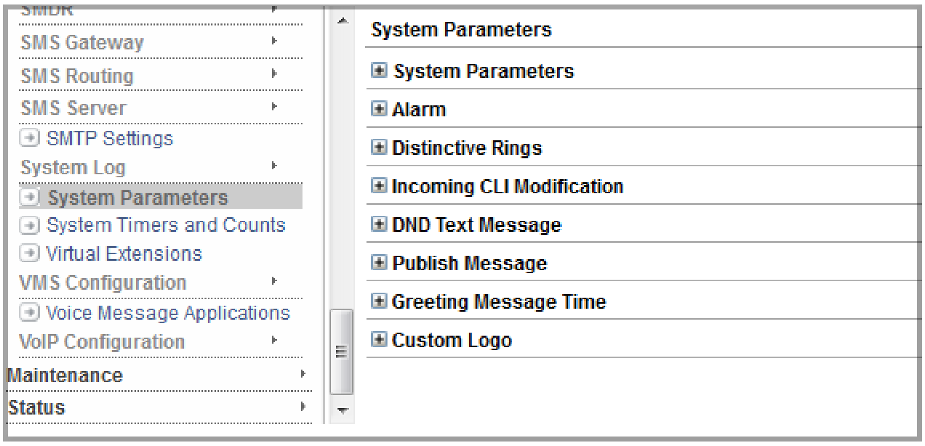
System Parameters

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 here along with the instructions for configuring them using Jeeves.

How to configure

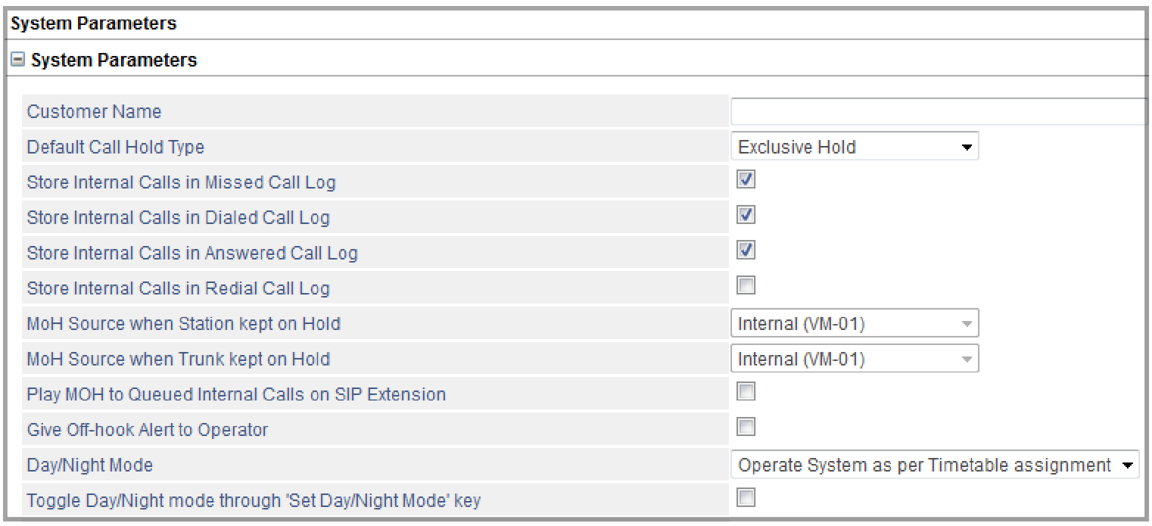
To configure System Parameters,

- Login as System Engineer.
- Under **Advanced Settings**, click **System Parameters**.



System Parameters

- Click **System Parameters** to expand. To view more parameters, use the vertical scroll bar on your right.



- **Customer Name:** You can assign the name of your 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 SARVAM UCS.

You can assign Customer Name also on the "Pre-requisites" page. If you have entered the Customer Name on this page, the same Name will appear on the Pre-requisites page.

- **Default Call Hold Type:** This parameter is related to the "Call Hold" feature of SARVAM UCS. You can select **Exclusive Hold** or **Global Hold** as per your requirement.

- **Store Internal Calls in Missed Call Log:** SARVAM UCS stores call logs of external calls only. You may instruct the system to store internal calls in the Missed Call log by enabling this parameter. See "Call Logs".

- **Store Internal Calls in Dialed Call Log:**You may instruct the system to store internal calls in the Dialed Call log by enabling this parameter. See "Call Logs".

- **Store Internal Calls in Answered Call Log:** You may instruct the system to store internal calls in the Answered Call log by enabling this parameter. See "Call Logs".

- **Store Internal Calls in Redial Call Log**: To have the system store internal calls in the Redial Call Log, select this check box. To know more about this feature, see "Last Number Redial". Default: Disabled.

- **MoH Source when Station kept on Hold**: Select the option Internal (VM-01), 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). to know more, refer to "Music on Hold (MOH)".

- **MoH Source when Trunk kept on Hold**: Select the option Internal (VM-01), 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) to know more, refer to "Music on Hold (MOH)".

- **Play MOH to Queued Internal Calls on SIP Extension:** The Extended IP Phone 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 is busy, select this check box. For more information, see "Music on Hold (MOH)".

- **Give Off-hook Alert to Operator:** When this check box is enabled, 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. To know more, refer to "OFF-Hook Alert".

- **Day/Night Mode:** You can set the Time Zone of the system as Day Mode (Working-Hours) or Break Mode or Night Mode (Non-Working hours) at 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 Timetable assignment**[163]. Default: Operate System as per Timetable assignment.

  For more details see "Day Night Mode" and "Time Tables".

163. *Certain features of 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.*

- **Toggle Day/Night mode through 'Set Day/Night Mode' key**: When this check box is enabled, you can switch to Day Mode (Working Hrs) or Night Mode (Non-Working Hrs) on pressing the DSS Key. For more details, see “Day Night Mode”. Default: disabled.

- **Replace '+' from CLI:** The GSM network presents the calling party number with prefix '+' to the called party. SARVAM UCS allows you to remove '+' and replace it with an appropriate number string as required. To do this, you must enable **Replace '+' from CLI** by selecting this check box.

  You may also configure the number string with which ‘+’ is to be replaced in the CLI in the **Replace '+' from CLI with the number string** field. Enter the number string with which you want to replace '+' received as prefix of calling party number**.**

  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 (i.e. left blank), the CLI number would be presented as 919327237228.

- **Disconnect Built-In Auto Attendant Call, when Dialed Number is busy:** When this check box is enabled, the Built-In Auto Attendant call will be disconnected if the landing extension is busy. The Built-In Auto Attendant call will not be routed to the Operator. Default: disabled.

- **Disconnect Built-In Auto Attendant call, when Dialed Number is not responding:** When this check box is enabled the system disconnects the Built-In Auto Attendant call if there is no reply from the landing destination extension. The Built-In Auto Attendant call will not be routed to the Operator. Default: disabled.

- **Disconnect Built-In Auto Attendant call, when caller does not Dial any digit:** When this check box is enabled, the system will disconnect the Built-In Auto Attendant call if the caller fails to dial a digit within the First Digit Wait Timer. The Built-In Auto Attendant 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 the Extension creating 3 party conference, disconnects during Conference**.

  - 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.

- **Play Beep when Conference/Dial-in Conference begins:** This is a common check box for two features of SARVAM UCS: “Conference-Multiparty” and “Conference Dial-In”. When this check box is enabled,

  - the system plays beeps to the other participants in a Dial-In Conference when a new participant joins in (i.e. dials in to an on-going Dial-In Conference)

- the system plays beeps to the other participants connected in a Multi-Party Conference, when a new participant is included.

If you disable this check box, no warning Beep will be played indicating the addition of a new participant, to the existing participants in a Dial-In or Multi-party conference.

- **Play Beep when Raid/Call Taping/Conversation Recording Starts:** This is a common check box for the features "Raid", "Call Taping" and "Conversation Recording". When this check box 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 check box is disabled, no indication will be given to the opposite party when the call is being taped/conversation is being recorded.

- **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 check box 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.

- **Ignore call forward set by member extension, when call is routed on Routing/Dept. Group:** If you want the system to ignore the call forward set by member extensions in a Routing Group/Department Group and place calls on them, select this check box. Default: Disabled.

- **Call Proceeding Tone for Multistage Dialing:** This check box is used in "Multi-Stage Dialing" where you need to configure Pause and Wait for Answer in the Substitute Number string 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 using the Multi-Stage, 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 the substitute 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).

- **Companding Algorithm:** 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. Select the companding Algorithm according to the Regulatory Requirement of the country where SARVAM UCS is installed.

- **Language of SE and SA Web Interface:** 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. If you want to change the default language set on Region selection, select the required language from the combo box. The language selected on this page will be applicable when you login the next session of SARVAM UCS.

- **Display Presence Status during Call on Extended IP Phone:** If you want to allow Extended IP extension users to view the presence status of the extension they are calling, you must enable this check box. Default: Disabled.

- **Apply RCOC only if the caller calls back on the same trunk from which the call was made:** Enable this 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.

- **Stuttered Dial Tone When DND is set**: If you want SARVAM UCS to play a Stuttered Dial Tone on the extension when DND is set, you must enable this check box. Default: Disabled.

- **Detect Possible toll bypass attempt by Extn. during IC Call from CO Line & Drop the Call**: 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 outgoing call as a fraudulent call. To drop this call, enable this check box. Default: Disabled.

- **Play beeps when Assistant present in 3-Party Conference**: Select this check box, if you want SARVAM UCS to play beeps during the conference to indicate the presence of the Operator. For details, see “Conference-3 Party”.

- **Play beeps when Assistant present in Multiparty Conference**: Select this check box, if you want SARVAM UCS to play beeps during the conference to indicate the presence of the Operator. For details, see “Conference-Multiparty” and “Conference Dial-In”.

- **Play beeps when Assistant leaves the Conference**: Select this check box, if you want SARVAM UCS to play beeps to the conference participants when the Operator leaves the conference.

- **Call Proceeding Tone for 1st caller of a SIP Extension**: 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 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.

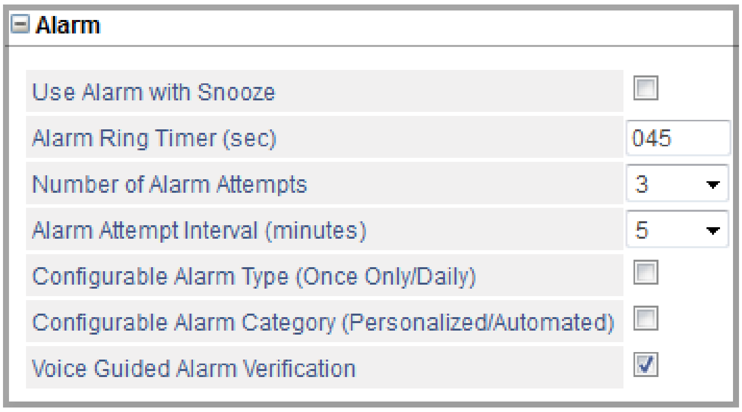
- **Silent**: If this option is selected, the extension user will get Silence (no tone), while the call is being routed.

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.

Alarm

- Click **Alarm** to expand.



- Configure the following Alarm and Reminder parameters:
- **Use Alarm with Snooze:** When you use Snooze functionality with an Alarm, the system does not consider an alarm call as answered until the extension user has dialed the acknowledgement code ‘0’ after going Off-Hook. By default, Snooze is disabled.
- **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:** 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

Feature	Ring Type	Ring Text
Internal Call	Short, very slow	internal
Trunk Call	Double	external
Auto Call Back	Short, slow	acb
Auto Redial	Long, very slow	autord
Alarm	Long, fast	selfalram
Emergency	Long, fast	emergency
Operator Alarm	Long, fast	operatoralarm
Message Wait	Short, fast	msgwait

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

Enable Incoming CLI Modification	
Country Code	
Area Code	
International Prefix	
National Prefix	
Area Code required to make Local calls?	Yes
Prefix Area Code	

This parameter 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. See "Incoming CLI Modification" to know more.

If you receive CLI in dialable format, there is no need to use this feature. In such case, keep the check box disabled. You do not need to configure 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 "CO Trunks", "Mobile Trunks" and "SIP 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 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:** This is the Area Code of the place where 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.

- **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. For example, '00' can be set as the prefix for dialing International numbers.

- **National Prefix:** These are digits required as Prefix for dialing long distance, National (within the country) numbers. The prefix may be upto 5 digits, with numbers from 00000 to 99999. For example, '0' can be 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:

  - ***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.

DND Text Message

- Click **DND Text Message** to expand.

DND Text Message

Message No.	Message Text
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

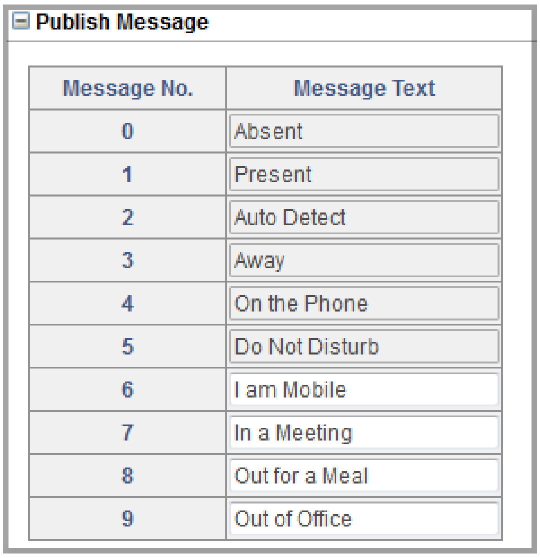
SARVAM UCS allows you to configure different DND Text Messages, which extension users can select when setting DND on their extensions. The DND text message they select is displayed to the calling extensions. See "Do Not Disturb (DND)" to know more.

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 Presence, 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.

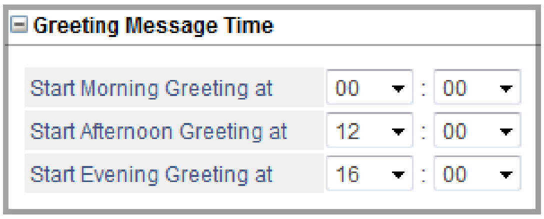
Message No.	Message Text
0	Absent
1	Present
2	Auto Detect
3	Away
4	On the Phone

Message No.	Message Text
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.

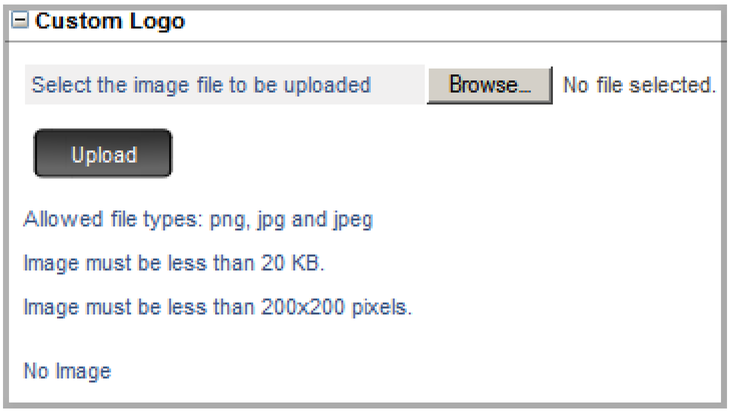
  The time must be in HH:MM format. The valid range for Hours (HH) is 00 to 23 and for Minutes (MM) is 00 to 59. By default the time in **Start Morning Greeting at** is set to 00:00, **Start Afternoon Greeting at** is set to 12:00 and **Start Evening Greeting at** is set to 16:00.

  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 and iPhone — SARVAM UCS supports uploading of any customized logo. This logo shall appear on the Home screen of VARTA ADR100 and VARTA AMP100.

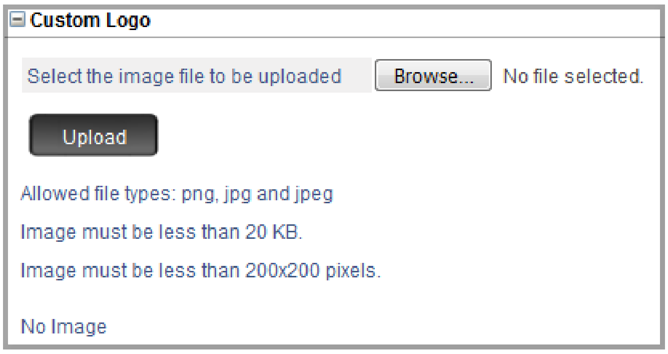
Make sure the image you want to upload fulfills the following requirement:
- The image must be a JPEG, JPG or PNG file
- Maximum Dimension of the image : 200 x 200px
- Maximum File Size of the image : 20KB

To upload the file,

- Click **Browse** to **Select the image file to be uploaded** from the location on the local disk.

- Click **Upload.**

  A preview of the image you uploaded appears.



- If you wish to remove the image, click **Remove.**

If you upload another image directly without removing the previous image, the system will automatically overwrite the image.

- Click **Submit.**

System Security

Access to 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. For details, see “System Engineer (SE) Password” and “System Administrator (SA) Password”.

You can access the FTP Server of the Extended IP Phones. This access is provided for maintenance and it is secured by means of a Password. For details, see “FTP Access for Extended IP Phones”.

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 confidential.
- Select a complex password that cannot be easily guessed.
- Change the password regularly.
- Not to 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,

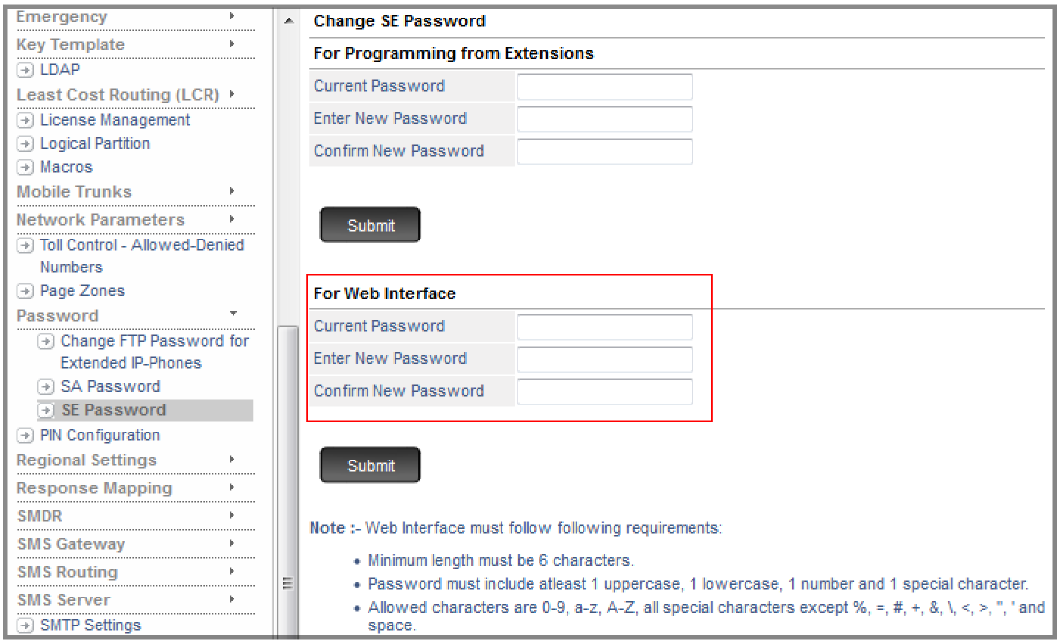
- the password will be valid for 90 days only and you will not be able to login with the existing password after 90 days. You will be prompted to change the password.
- if you enter a wrong password for five times consecutively within 10 minutes, the system will block the source IP Address for 10 minutes. This activity will be logged in the “System Activity Log”.

- *The SE password is stored in the CPU. If you forget the SE password, the only way to restore the default SE password is to change the Jumper settings of the CPU module.*
- *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 for Accessing Jeeves

- Login as System Engineer.
- Under **Advanced Settings**, click **Password**.

- Click **SE Password**.



- Under **For Web Interface**,
  - Enter **Current Password**.
  - **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**.

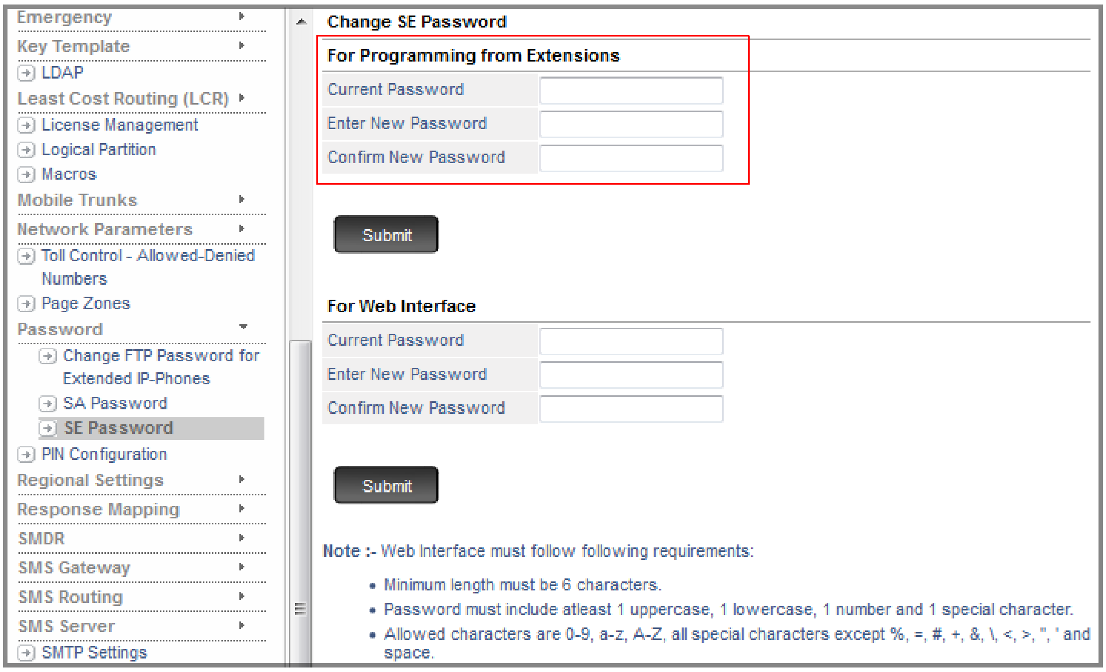
For Programming from Extensions

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. It should be changed and kept confidential.

Changing the SE Password for programming from extensions

- Login as System Engineer.
- Under **Advanced Settings**, click **Password**.

- Click **SE 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**.

Forgot the SE Password?

If you have already changed the default SE Password (1234) 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 Settings on the CPU module. To do this,

- Remove the top cover of the enclosure.
- Locate and change the position of the Jumper **J7** from ***BC*** to ***AB***.
- Replace the cover of the enclosure.
- Switch ON the system and wait for 15 seconds.
- Switch OFF the system and remove the enclosure cover.
- Change the Jumper position from ***AB*** to the original position ***BC***.
- Replace the enclosure cover.
- 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 again change the SE password using Jeeves as described above.

You can configure only the basic network and debug parameters from your telephone using the default SE Password, 1234. To know the detailed list of parameters, which you can configure using the default SE Password, see *"Basic SE Commands".*

If you do not restore the jumper position 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.

When you change the default settings of the jumper (J7) on the CPU module, the parameter— Allow Web Server Access From—will also be set to default.

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 *"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 confidential.
- Select a complex password that cannot be easily guessed.
- Change the password regularly.
- Not to use the **"Remember Password**" property of your Web Browser.

The SA 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.

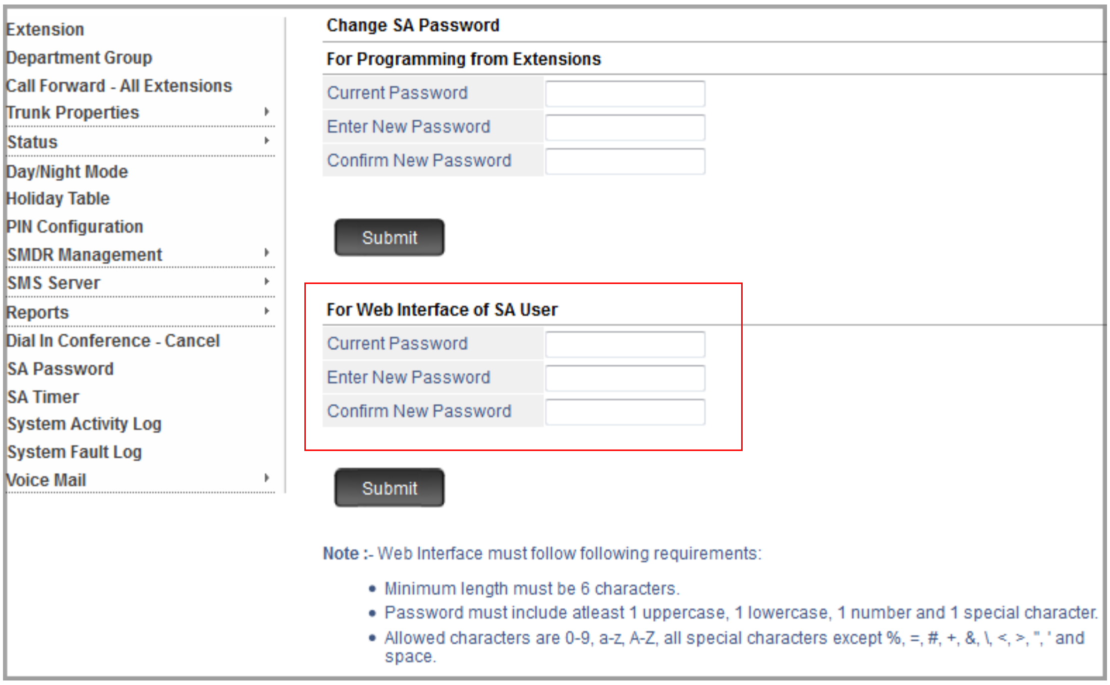
To provide additional security,

- the password will be valid for 90 days 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".

Changing SA Password for Accessing Jeeves

- Login as System Administrator.

- Click **SA Password**.



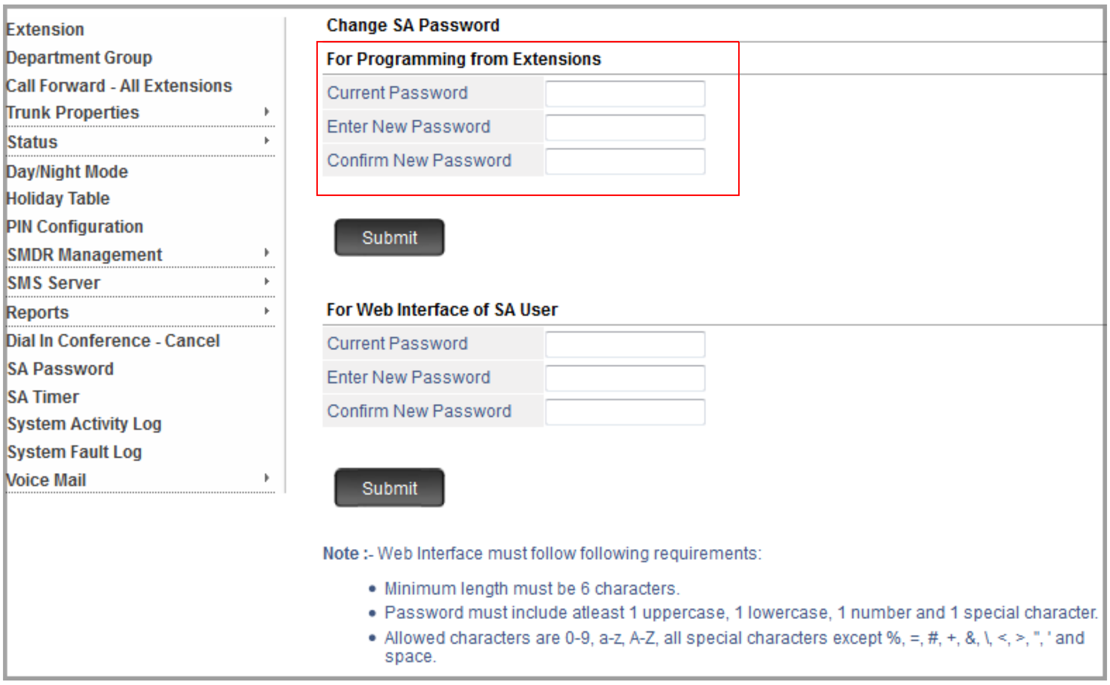
- Under **For Web Interface of SA User**,
    - Enter **Current Password.**
    - **Enter New Password**. All ASCII characters (except Percentage **%**, Hash **#,** Equal to **=,** Plus **+**, And **&**, Backslash **\,** Less than **<,** Greater than **>,** Apostrophe **'** , Double Quote **"** and **Space**) 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**.

For Programming from Extensions

Changing SA Password for Programming from Extensions

- Login 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**.

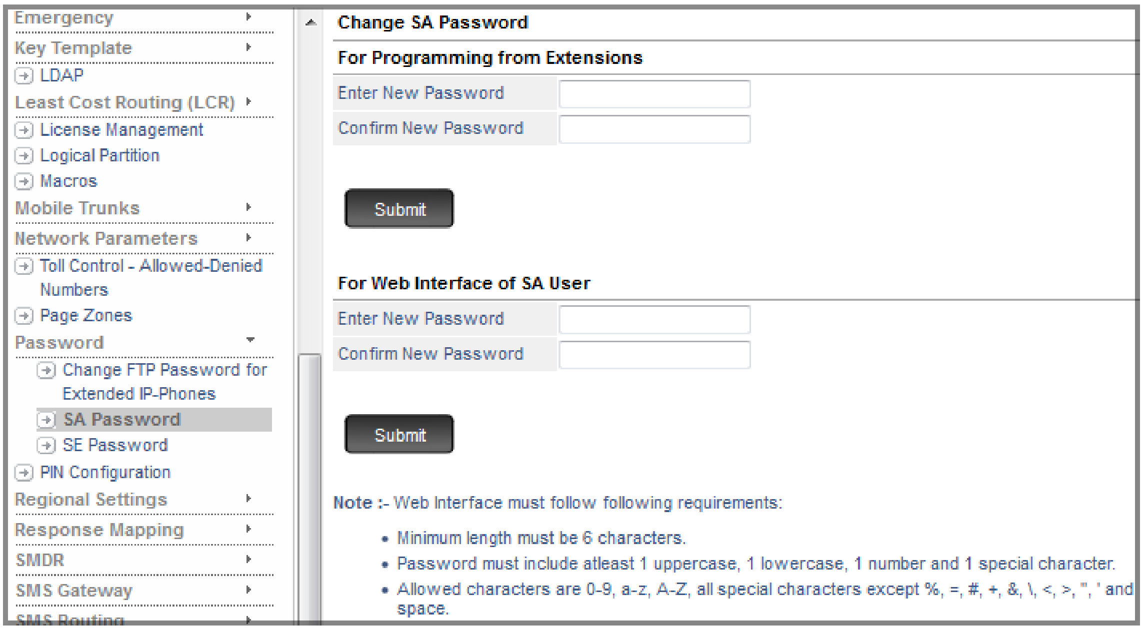
Forgot the SA Password?

In case the System Administrator has forgotten the password, a new password can be issued by the System Engineer only.

To issue a new SA Password from Jeeves,

- Login as System Engineer.
- Under **Advanced Settings**, click **Password**.

- Click **SA Password**.



- To change/issue a new password **For Web Interface of SA 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 at least 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 change/issue a new password **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. The default SE Password is 1234.

  - **Re-enter New Password** to confirm.

  - Click **Submit**.

User Password

Extension Users can secure their respective 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.

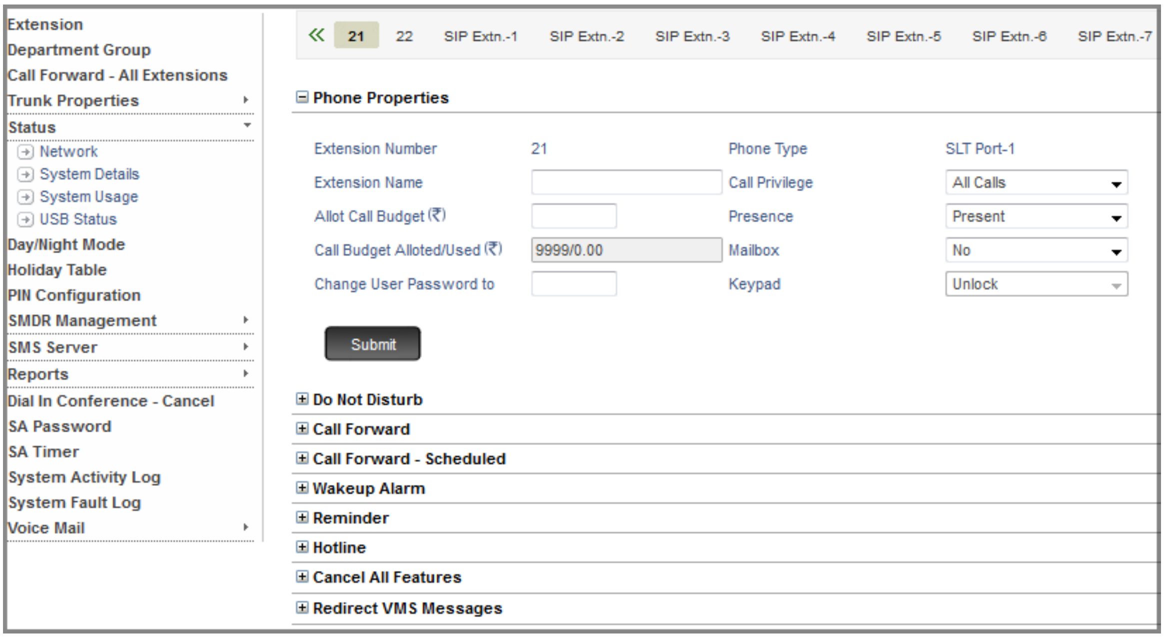
Refer the topic “User Password” to know more.

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.

To issue a new User password using Jeeves,

- Login as System Administrator.
- Click **Extension**.
- Now, click the desired Extension number on the top bar. To scroll to the desired extension type and number, click **>>**.
- The page of the desired extension will appear.
- Click **Phone Properties** to expand.



- Enter the new User Password in **Change User Password to**. The User Password may be a combination of 4 digits. Valid digits: 0 to 9.
- Click **Submit**.

To issue a new User password using a telephone,

- Enter SA mode from a SLT.
- Dial ***1072-012-Extension Number-New User Password***
- Exit SA mode.

Additional Security to Extension Users

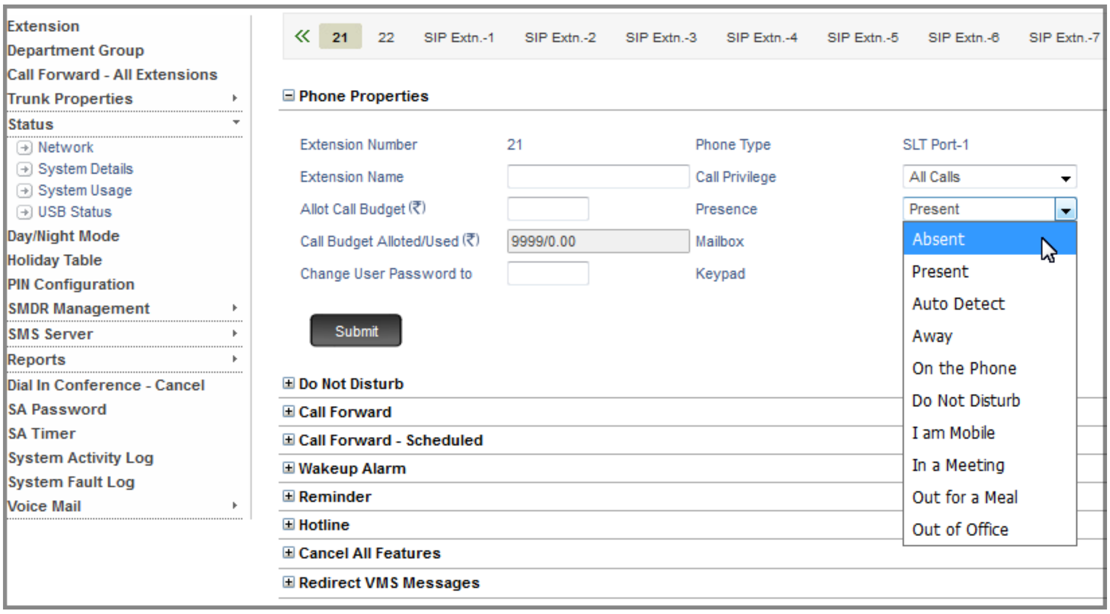
It is also possible to secure unattended extensions. This may be necessary when extension users forget to lock their extensions or are away from their desks. In such situations, the System Administrator can protect the extension from unauthorized access and use by:

- locking the Keypad of the extension phone. This is possible only on Extended IP Phones.
- setting the User Status for the extension as “Absent”; possible on 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 SIP extensions and set SIP and SLT extension users as ‘Absent’ using Jeeves and by dialing commands from a telephone.

To secure an extension from Jeeves,

- Login as System Administrator.
- Click **Extension**.
- Now, click the desired Extension number on the top bar. To scroll to the desired extension type and number, click **>>**.
- The page of the desired extension will appear.
- Click **Phone Properties** to expand.



- Set the user status for the parameter **Presence** by selecting the option **Absent** from the combo box. When you want to change user status again to present, select **Present**.
- Select the option **Lock** for the parameter **Keypad**. This parameter will be configurable only if the selected extension is a Extended IP Phone. 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**.

To secure an extension using a telephone,

- Enter SA mode from a SLT.
- Dial ***1072-013-Extension Number-1*** to lock keypad of a SIP extension.
- Dial ***1072-013-Extension Number-0*** to unlock keypad of a SIP 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”.*
- *Extended IP Phone extension users can also lock the keypad of their phones from the Phone Menu. Refer “Extended IP Phone/VARTA UC Client - Operation” for instructions on navigating the phone menu.*
- *It is also possible to lock/unlock the Extended IP Phone 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.

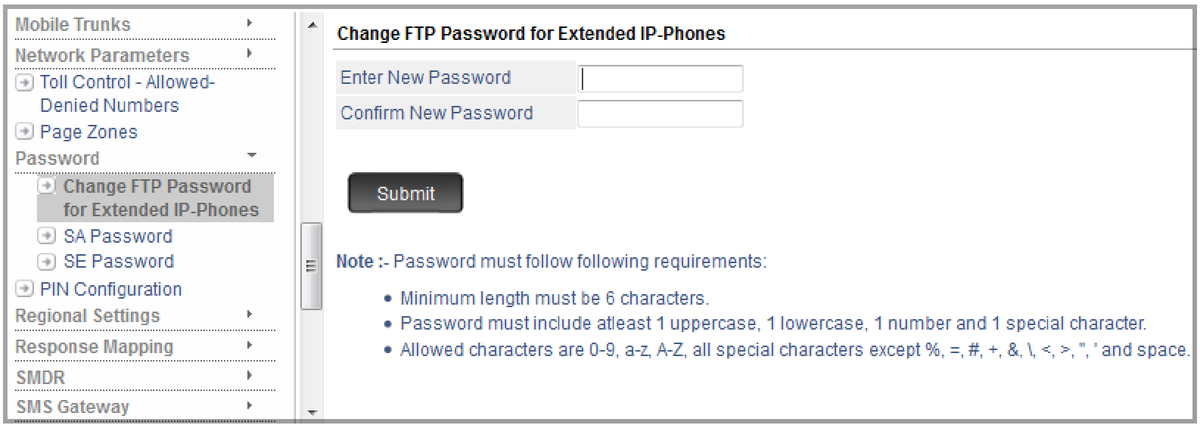
- *When you set the System to default, the FTP Password will not be set to default.*
- *If you use jumpers to restore the default SE Password, the FTP Password will not be set to default.*

Changing the FTP Password using Jeeves

- Login as System Engineer.

- Under **Advanced Settings**, click **Password**.

• Click **Change FTP Password for Extended IP-Phones.**



- **Enter New Password**.
- In **Confirm New Password**, re-enter the new password to confirm.
- Click **Submit**.

System Timers and Counts

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 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

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.	2 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

Name	Description	Range	Default
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

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	5 seconds
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.	0 to 255	10 seconds
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.	1 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.	1 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.	1 to 255	5 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
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 tries
Message Notification Ring Timer (sec)	The time for which the extension on which the Message Wait Notification Call is made will ring.	1 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
Call Park Timer (min)	The time after which the call comes back to the extension that has parked the call.	1 to 255	2 minutes
Call Park Release Timer (min)	The time after which the parked call gets disconnected.	1 to 255	3 minutes
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 Indication type for the extension.	0 to 255	10 attempts

Name	Description	Range	Default
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 Indication 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 Indication 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
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 Extended IP Phone which put it on hold.	1 to 999	120 seconds
Exclusive Hold Retrieval Timer (min)	This is the time for which a call put on Exclusive Hold remains connected in the Extended IP Phone. If the call put on Exclusive Hold is not retrieved within this timer, the call is returned to the 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

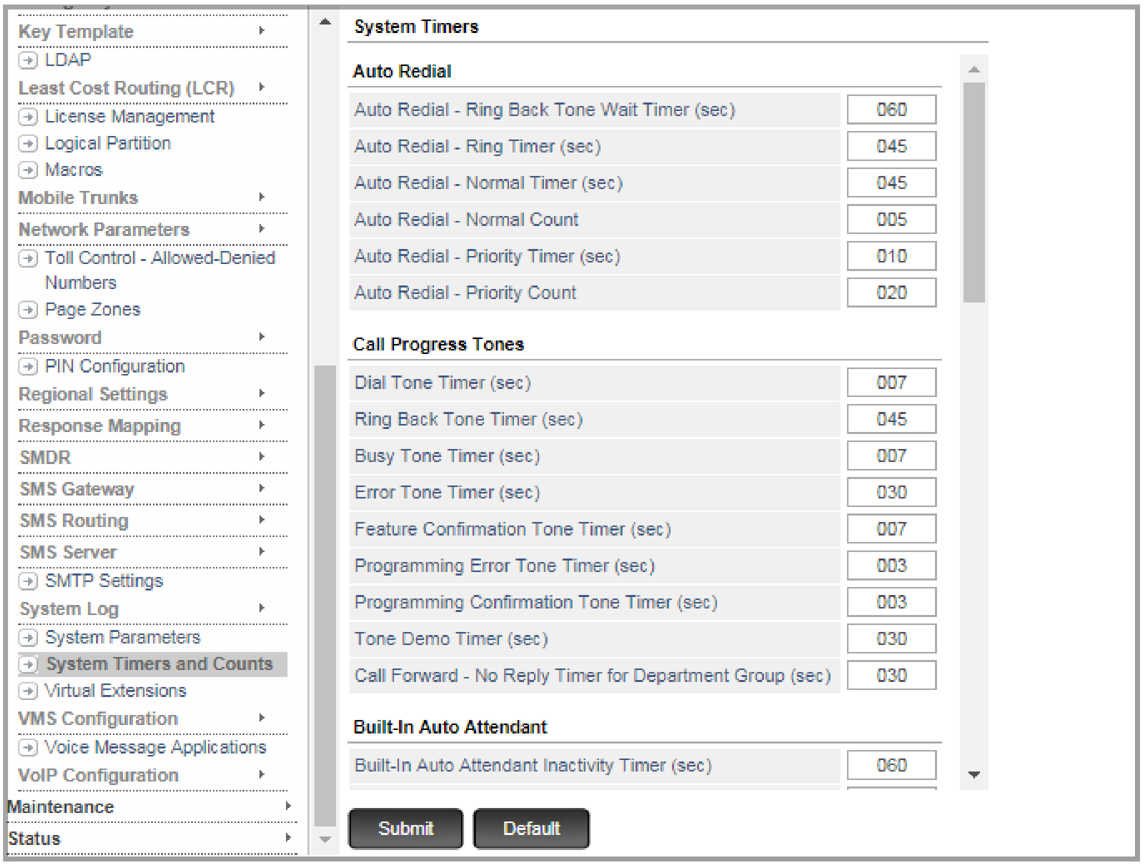
Name	Description	Range	Default
Emergency Reporting Call - Ring Timer (min)	The time for which the Operators extension rings, to inform the operator the number of the extension that dialed out an emergency number.	1 to 255	10 minutes
Held Call Disconnection Timer (min)*	The time after which the system will disconnect the held call.	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 minutes
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 SPARSH VP330, SPARSH VP210 and Matrix VARTA Mobile UC Clients.*

How to configure

To configure Timers and Counts,

- Login as System Engineer.
- Under **Advanced Settings**, click **System Timers and Counts**.



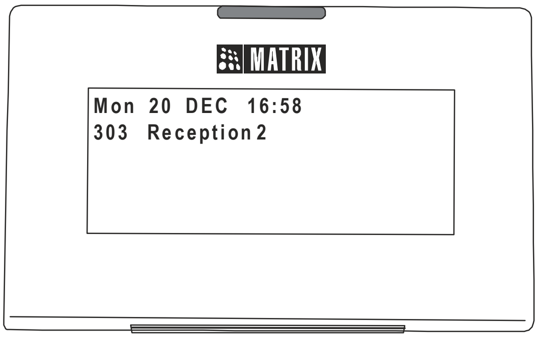
- Change the value of the Timer or Count by entering the desired duration or count in the respective fields.
- Click **Submit**.

Time Zone Display

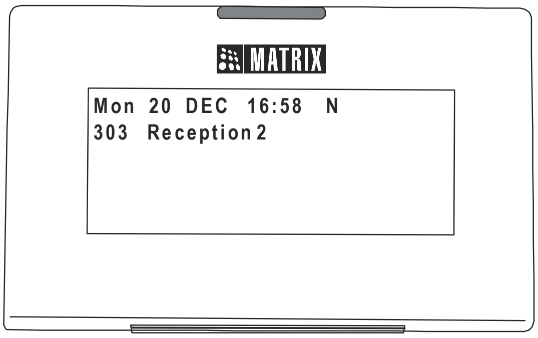
The current time zone—Working Hours or Non-working Hours—is displayed on the LCD of the Extended IP Phone[164].

During Non-working hours the letter ‘N’ is displayed on the LCD of the Extended IP Phone in the idle state, and during Break hours, the letter ‘B’ is displayed.

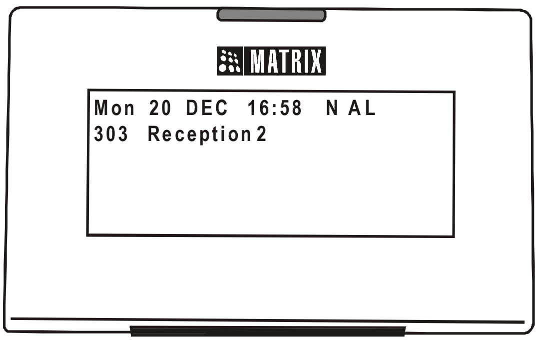
During working hours, in the idle state, the phone display will look like this:



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:



164. *This feature is explained with reference to SPARSH VP248. The display varies with Phone Type.*

Toll Control

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 of Call Privilege allows the extension users to call certain areas and restricts it from calling others. The extension users can also be restricted from dialing of specific telephone numbers.

SARVAM UCS supports the following types of Call Privileges:

- **No Calls:** Dialing of all external numbers is restricted. Only internal (extension-to-extension) calls are allowed.

  If you have enabled Global Directory part1, Part 2 and 3 in the Class of Service of the extension, the numbers configured in these will also be allowed to be dialed out.

  Therefore, if you want to restrict the extension user to dial external numbers, you must deny Global Directory Part-1, Part 2 and Part 3 in the Class of Service of that extension.
- **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 list of *Local* numbers.
- **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 list of *Regional* numbers.
- **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 list of *National* numbers.
- **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 list of *International* Numbers.
- **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 (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* number 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 four 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 whose allotted budget is consumed. 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 (i.e. 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 the **Day** (Working hours) and the **Night/Break** (Non-working hours). For instance, you may define 'International Calls' as Call Privilege for the Day and 'No Calls' as Call Privilege for the Night/Break.

  By default, Call Privilege ‘No Calls’ is selected for the Day and Night/Break.

- **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.

SARVAM UCS offers you the flexibility to redefine the Call Privilege for each of the above Toll Control Levels according to the requirements of the extension users.

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 of Call Privilege detected	Logic system will check
Local calls	Local Number List configured.
Regional calls	Regional Number List configured.
National calls	National Number List configured.
International calls	International Number List configured.
Limited Calls	Limited Calls Number List (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 each digit of the dialed number string with the number strings programmed as Allowed and Denied numbers in the Lists of the Local/Regional/National/Limited Calls 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, SIP.

For Toll Control to work, you need to configure the lists of Local Numbers, Regional Numbers, National Numbers, International Numbers and Limited Calls, 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.

- On one column of each list, write down the numbers you want to permit as Allowed Numbers. On 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 Numbers	Denied Numbers
1		
2		
:		
:		
999		

List of Regional Numbers for Call Privilege - Regional Calls

Sr. No.	Allowed Numbers	Denied Numbers
1		
2		
:		
:		
999		

List of Regional Numbers for Call Privilege - International Calls

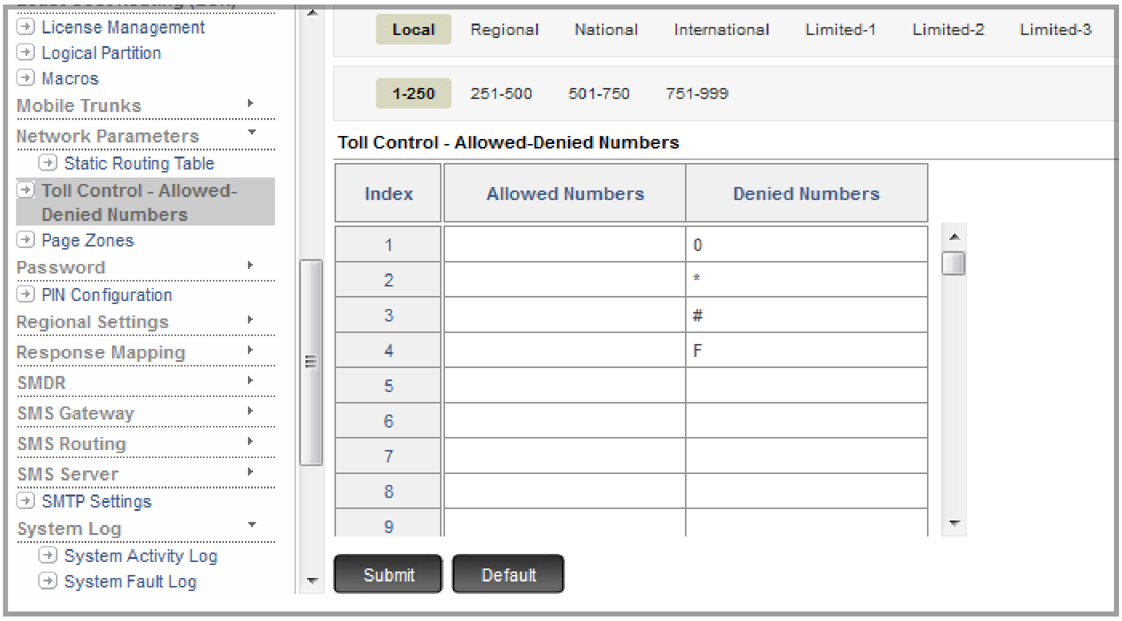
Sr. No.	Allowed List	Denied List
1		
2		
:		
:		
999		

List of Limited numbers Call Privilege - Limited Calls 1

Sr. No.	Allowed Numbers	Denied Numbers
1		
2		
:		
:		
999		

Toll Control - Allowed Denied Numbers

- Login as System Engineer.
- Under **Advanced Settings**, click **Toll Control - Allowed Denied Numbers**.



Local Calls

- Click the **Local** tab.
- Define **Local Calls** number strings. You may enter up to 999 local numbers.
- In the column **Allowed Numbers**, enter the number strings to be allowed to be dialed out by the system.

  You may enter only the first digit of the number string, or a part of the string, or the complete number string.

  Each number string you enter must not exceed 16 characters.
- In the column **Denied Numbers**, enter the number strings which you want to restrict from being dialed out.
- Click **Submit.**
- Enter as many local numbers as you want by clicking the Index number links 001-250, 251-500, 500-750, 751-999.

  Click **Submit.**

Regional Calls

- Click the **Regional** tab.
- Enter the **Allowed Numbers** and **Denied Numbers** in their respective columns.
- Click **Submit.**
- Enter as many Regional numbers as you want by clicking the links 001-250, 251-500, 500-750, 751-999.

- Click **Submit**.

National Calls

- Click the **National** tab.
- Enter the **Allowed Numbers** and **Denied Numbers** in their respective columns.
- Click **Submit**.
- Enter as many National numbers as you want by clicking the Index number links 001-250, 251-500, 500-750, 751-999.
- Click **Submit**.

International Calls

- Click the **International** tab.
- Enter the **Allowed Numbers** and **Denied Numbers** in their respective columns.
- Click **Submit**.
- Enter as many International numbers as you want by clicking the Index number links 001-250, 251-500, 500-750, 751-999.
- Click **Submit**.

Limited Calls

The Call Privilege Type 'Limited Calls' allows the dialing of specific telephone numbers, which may be local, regional, national or international numbers.

- Click **Limited 1** tab.
- Enter the **Allowed Numbers** and **Denied Numbers** in their respective columns.
- Click **Submit** to save.
- Enter as many Regional, National numbers as you want, by clicking the Index number links 001-250, 251-500, 500-750, 751-999.

  Click **Submit**.
- To configure another Limited Number List, click the tab **Limited 2**, **Limited 3** and follow the same instructions as above.

Assign Toll Control to extension users

Configure Toll Control levels for each extension port of SARVAM UCS. If you have not already assigned a Toll Control level to the extension ports at the time of configuring them, you may do so now.

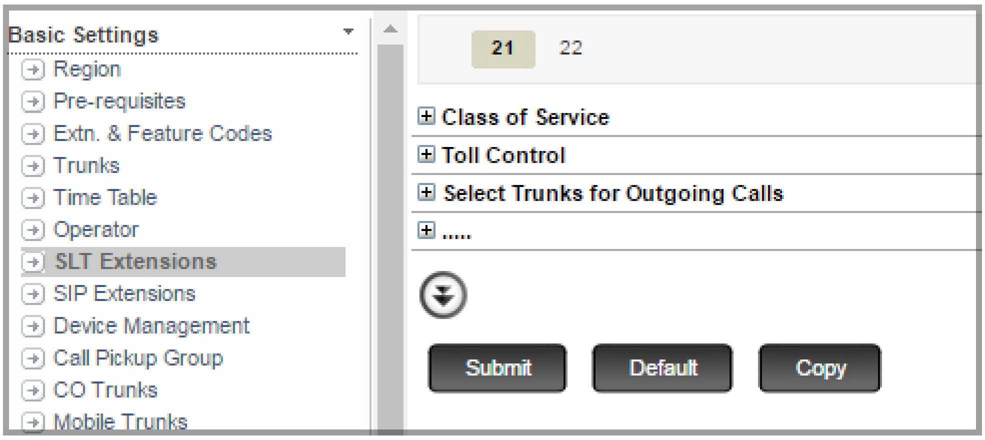
- Login as System Engineer.

- Click **Basic Settings**.

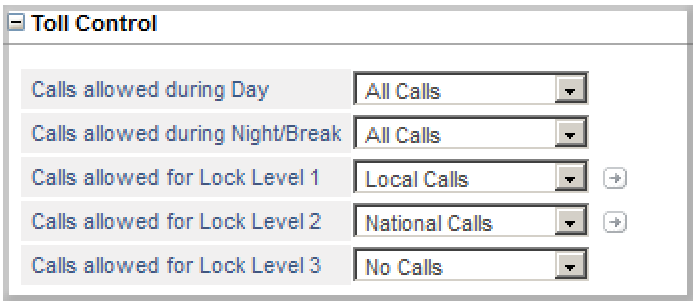
- Click the link of the extension type, it may be:
  - **"SLT Extensions"**
  - **"SIP Extensions"**

  The page of the selected extension type opens.

  For example, if you select SLT Extensions, the SLT Extensions page opens.



- Click **Toll Control** to expand.



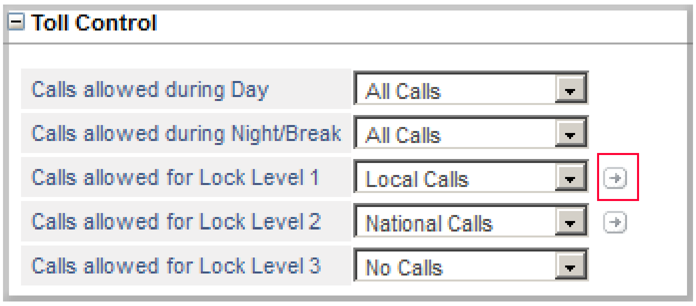
- Click **More** to expand all Toll Control parameters.

- In the **Calls allowed during Day** box, select the call privilege type for the Day time:
  - No Calls
  - Local Calls
  - Regional Calls
  - National Calls
  - International Calls
  - All Calls
  - Limited Calls 1
  - Limited Calls 2
  - Limited Calls 3

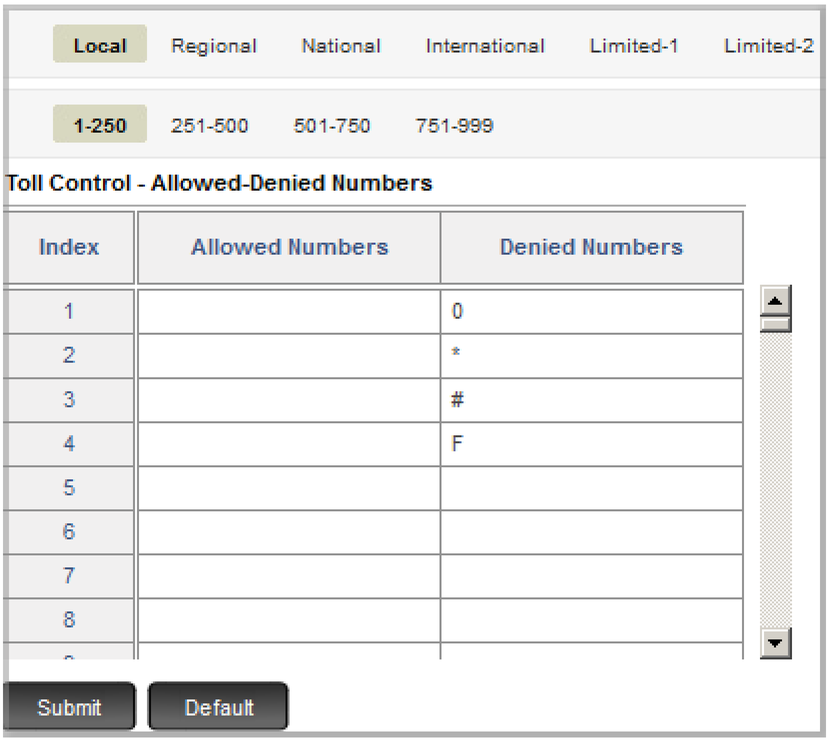
- *The Call Privileges 'No Calls' and 'All Calls' do not require any Allowed Denied Numbers to be configured.*
- *The Call Privilege Type 'Limited Calls' allows the dialing of specific telephone numbers only*.

- For the type of call privilege you selected, the number list configured in the “Toll Control - Allowed Denied Numbers” page will be automatically assigned.

If you have not configured number lists already, you may do so now:



- Click the **Settings** icon. The number list of the call privilege type you selected opens in a window.



For instance, if you selected **National Calls** as **Calls allowed during Day**, when you click the **Settings** icon, the list for National numbers will open.

- In the **Allowed Numbers** column, enter the long distance numbers within the country that are to be permitted.

  You may enter only the first digit of the number string, or a part of the string, or the complete number string.

  Each number string you enter must not exceed 16 characters.

In the **Denied Numbers** column, enter the long distance numbers that are to be restricted. The number strings may be only the first digit of the string, a prefix, or the complete string, not exceeding 16 characters.

- Enter as many numbers as you want by clicking the Index number links 001-250, 251-500, 500-750, 751-999.
- Click **Submit** at the bottom of the page to save the entries.
- Close the window.
- In the **Calls allowed during Night/Break** box, select the call privilege type for the Night/Break time.
- To configure the number list for the selected call privilege type, click the **Settings** icon. The number list of the call privilege type you selected opens in a window.

  Follow the same steps as described above to configure this table.
- Follow the same steps to select the call privilege type and configure the number list for
  - **Calls allowed for Lock Level 1**
  - **Calls allowed for Lock Level 2**
  - **Calls allowed for Lock Level 3**

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.

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 the caller.

Trunk Auto Answer is useful in call centers, railway inquiry, 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 destination extension/s. If not answered by any extension, the call is answered by the system.

How it works

Trunk Auto Answer is useful when *Operator* or *Extension/s* or *Department Groups* is selected as the landing destination for incoming calls for the Day, Break and Night.

So, you can enable Trunk Auto Answer on a CO, Mobile or SIP trunk, only if you have selected Operator or Extension/s as the destination for incoming calls on that trunk.

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—Operator or Extensions—selected for the time of the day.

  The system starts the *Built-In Auto Attendant Inactivity Timer* (default: 60 seconds). The Trunk Auto Answer 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 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 Trunk Auto Answer 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 call 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 (Operator or Extensions) 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 Extensions and waits for the duration of the *Delayed Trunk Auto Answer Timer* (configurable; default:10 seconds) for any of the extensions 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 plays Ring Back Tone to the caller for the duration of this timer.

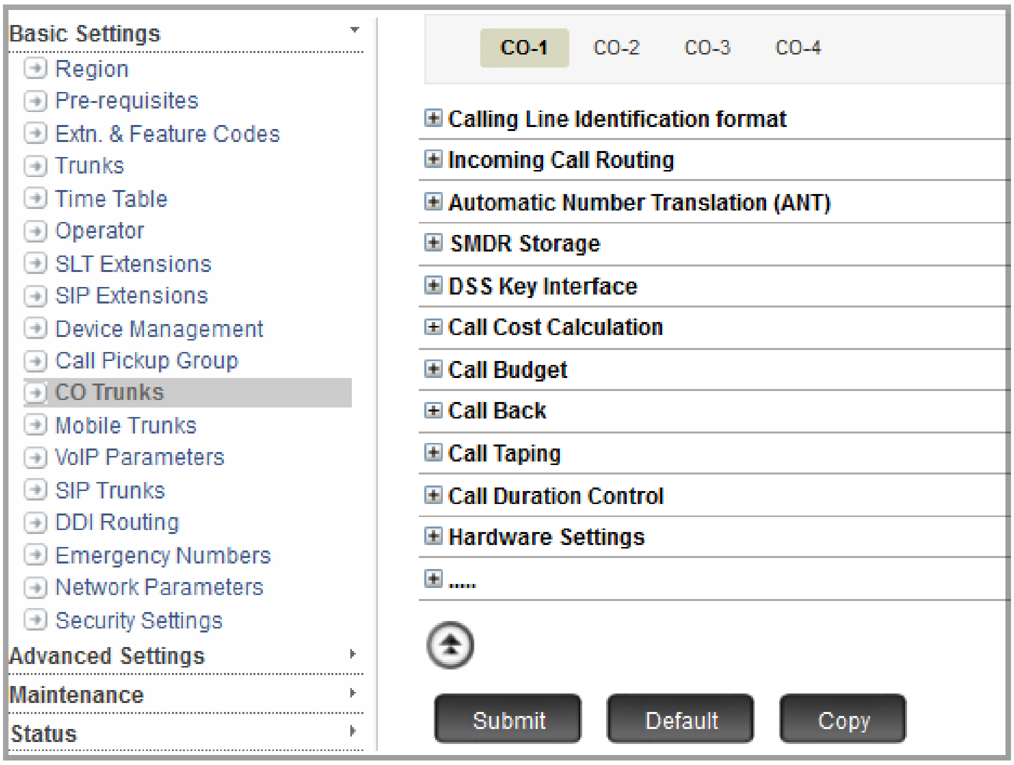
- If any of the landing destinations answers the call before the expiry of the Built-In Auto Attendant Inactivity Timer, the system stops the Built-In Auto Attendant Inactivity Timer and the Ring Back Tone message, and connects the caller to the extension that answered the call.
- If none of the landing 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

To configure Trunk auto Answer,

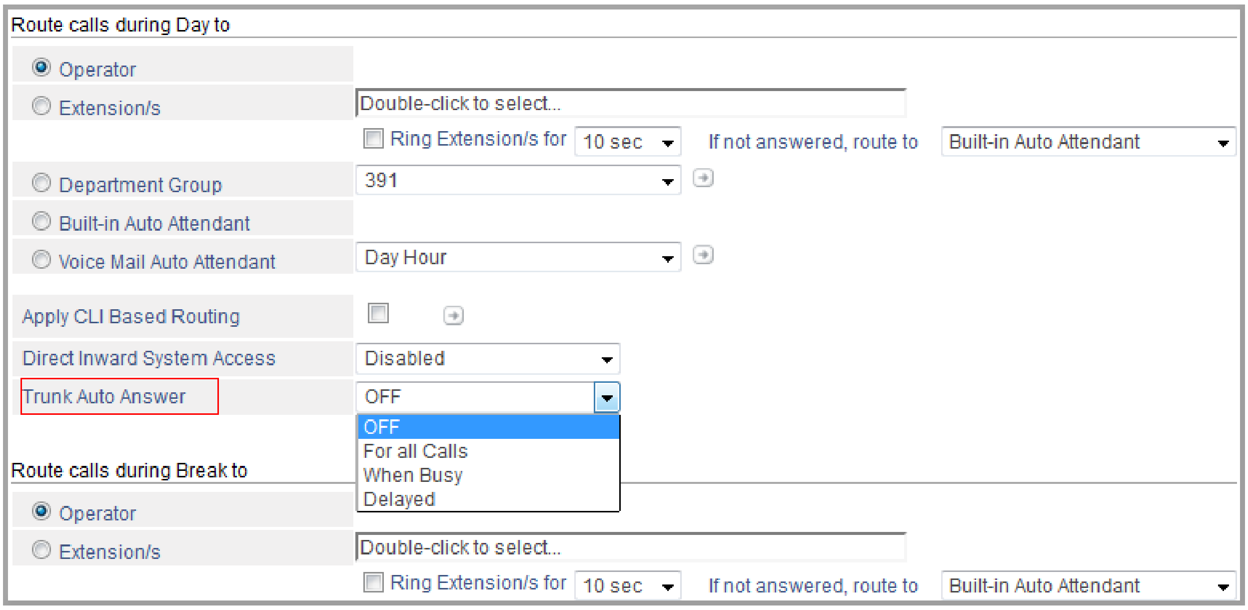
- Enable Trunk Auto Answer on the desired trunk—CO, Mobile, SIP.
  - Under **Basic Settings**, click the desired trunk port type.
  - Select the desired Trunk port number.



  - Click **Incoming Call Routing** to expand.
  - If **Operator** or **Extensions** or **Department Group** is selected as the destination to **Route calls during Day to, Route calls during Break to** and **Route calls During Night to**, configure Trunk Auto Answer.

    It is not mandatory to configure Trunk Auto Answer for all the time zones. You may configure for any time zone, as per your requirement.

- Select the Type of Trunk Auto Answer for the Day, for Break and for the Night as: **For all Calls** or **When Busy** or **Delayed**. You may select the same or different options for the Day, Break and Night, as per your requirement.



- Select **Trunk Auto Answer Greeting message**, **Trunk Auto Answer Ring Back Tone Message** and **Trunk Auto Answer Busy Bye Message** for the Day, Break and Night.

  You may select different Greeting, Ring Back Tone and Busy Bye Message for each time zone.

  If you do not want to play messages for **Trunk Auto Answer Ring Back Tone**, you may select the options: Music-on-Hold or Do not Play as per your requirement.

- 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 “Voice Message Applications” for instructions on recording and assigning voice modules to greeting messages.

- Configure the Trunk Auto Answer related **Timers**, if required. The following Timers are of relevance to the Trunk Auto Answer Feature:
  - Built-In Auto Attendant Inactivity Timer (default: 60 seconds)
  - Ring Back Tone Timer (default: 45 seconds)
  - Busy Tone Timer (default: 7 seconds)

  You may change the duration of these timers as per your requirement. See “System Timers and Counts”.

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.

Trunk Call Waiting

The Trunk Call Waiting feature gives indication to the extension user about another 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.

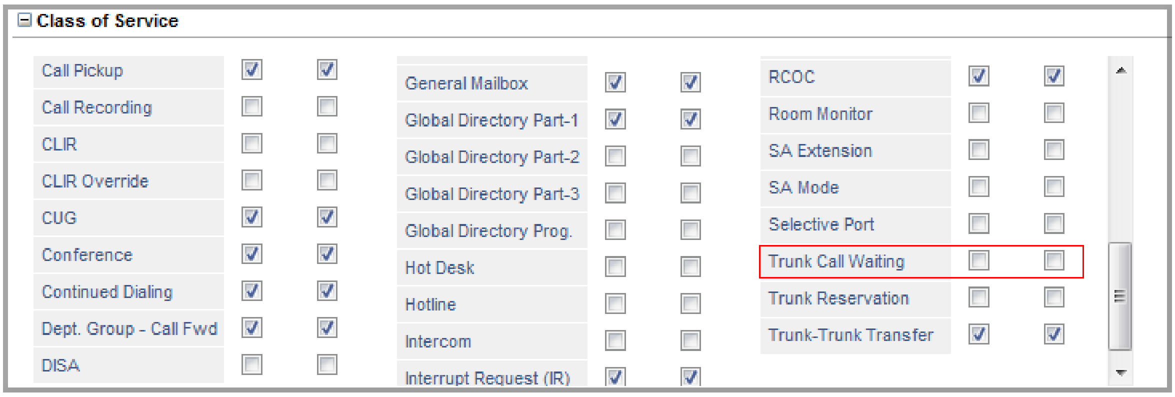
How it works

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 HOLD 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. By default, Trunk Call Waiting is disabled in the Class of Service of all Extension types: SLT, SIP. You may enable this feature on extensions which you want to provide Trunk Call Waiting indication.



For instructions, see “Class of Service (CoS)”, and the description for configuring the different Extension types under “Basic Settings”.

Keep this feature disabled, if you want to provide Privacy from Trunk Call intrusion beeps to an extension type. See “Privacy” to know more.

Trunk Reservation

This feature enables any extension user to reserve a trunk for exclusive use, for a specific time period.

Trunk Reservation can be requested from a SLT or an Extended IP Phone.

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.

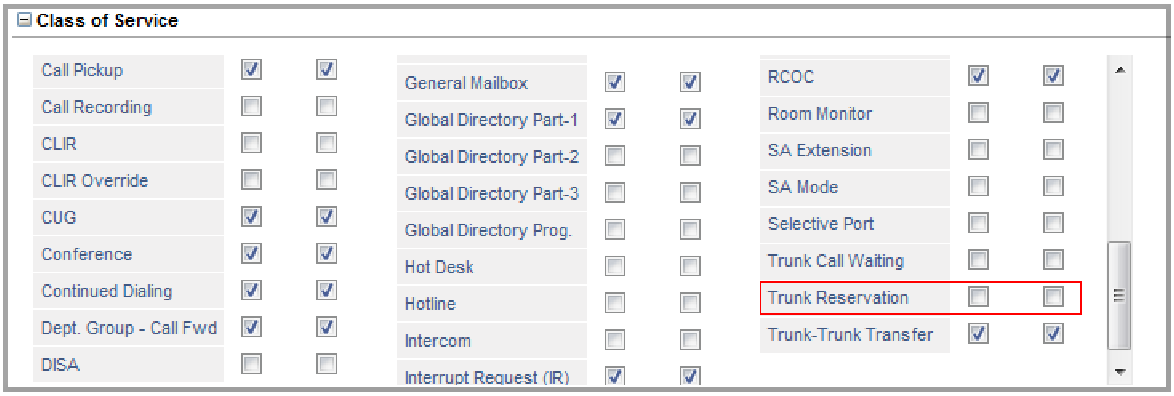
Extension user A is a Sales Executive. To complete the sales target, A needs to make long-distance calls to customers. Since there is full traffic on all the four trunks throughout the day, and these trunks are constantly busy, A would need a dedicated trunk line to save time and complete the target.

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:
  release the trunk manually, by canceling Trunk Reservation.
  OR
  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 (CoS)" for the Day and Night/Break time, as required.



See "Basic Settings" for instructions on configuring Class of Service of different extension port types. You may increase or decrease the duration of the Trunk Reservation Timer. See "System Timers and Counts" for instructions.

How to use

For 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 DSS Key assigned to Trunk Reservation.
  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

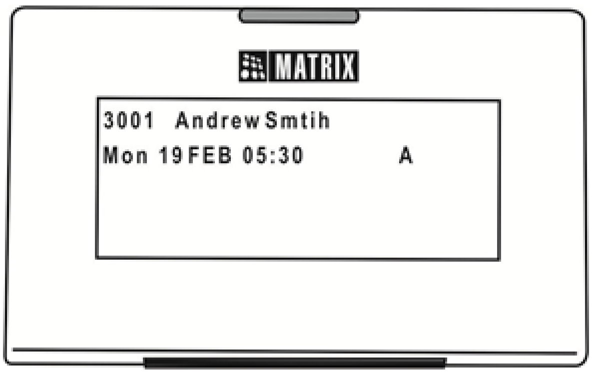
Extension users may sometimes want to leave their desks, and expecting to return soon, they may not have forwarded their calls or set Do Not Disturb on their extensions. In such cases, incoming calls will continue to land on the extension and go unanswered. The callers have no way of knowing that the 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 Extended IP Phone 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 Extended IP Phone user calls the extension which has set 'User Absent', the text message 'User Absent' will appear on the caller's phone display.

When a 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, refer to "Additional Security to Extension Users" under the topic *System Security.*

User Password

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 can 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
- User Absent/Present
- Hot Desk
- Phone Settings of the Extended IP Phone
- Mailbox of Voice Mail

The extension user must change the default password for all the above listed features except: Phone Settings, Mailbox of Voicemail. 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.*
- *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 Extended IP Phone Users

- Press ‘Enter’ key to enter Extended IP Phone 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.
- Dial New User Password.
- You get confirmation tone.
- Replace handset.

Video Call

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.

SARVAM UCS supports video calling for both SIP Trunks and SIP extensions.

How it Works

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 conform to the same type of video codecs[165] 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 conformed 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 VoIP channels[166] of SARVAM UCS are consumed. During the entire video call, those VoIP channels are reserved and used till either of the parties disconnects the call.

SARVAM UCS support a maximum of eight VoIP channels. If the RTP Mode is selected as Transcoding, the maximum number of simultaneous video calls supported will be 4. However, if you select RTP Mode as the RTP Relay or Direct RTP, the number of simultaneous video calls supported will be 8.

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 converted 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.

165. *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.*
166. *The Channels will be reserved only for Transcoding Mode.*

When the below mentioned features are accessed the call will always be an Audio call:

- When you 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 Auto Call Back Return call, Auto Redial Return call, Alarm call, Handover call etc.

- *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

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.

SARVAM UCS supports 20 Virtual Extensions.

How it works

The shared telephone instrument is called the Master Extension. A Master Extension can be a SLT or Matrix Extended IP Phone.

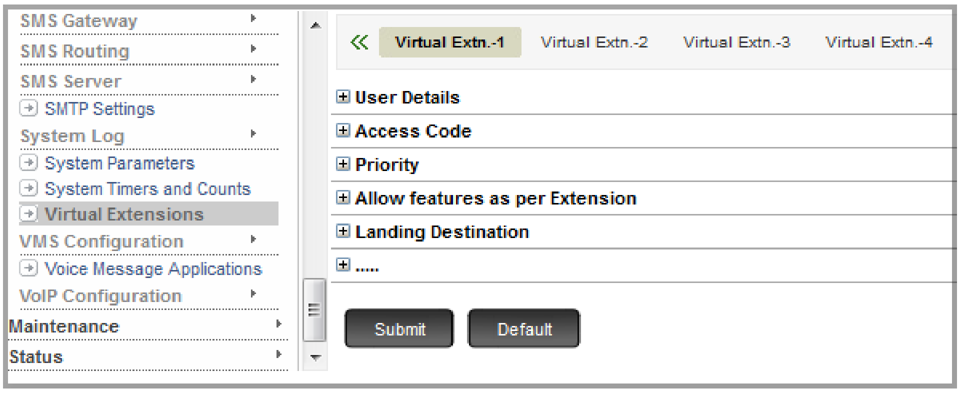
- Virtual Extensions are assigned to the Master Extension. A Master Extension can have multiple Virtual Extensions, but a Virtual Extension can have only one Master Extension.
- The Virtual Extension functions as any SLT or SIP extension of SARVAM UCS. It can be assigned all features and facilities, like Class of Service, Toll Control, Call Forward, Voice Mail, just like any other extension of SARVAM UCS.
- 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 automatically on completion of call* or *Walk out on user request.*

How to configure

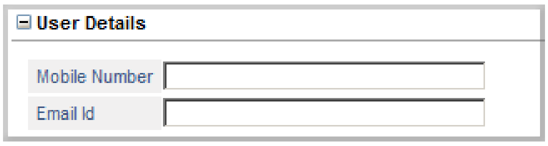
To configure Virtual Extensions,

- Login as System Engineer.
- Under **Advanced Settings**, click **Virtual Extensions**.

The Virtual Extensions page opens.



- Configure Virtual Extension 1.
- Click **User Details** to expand.



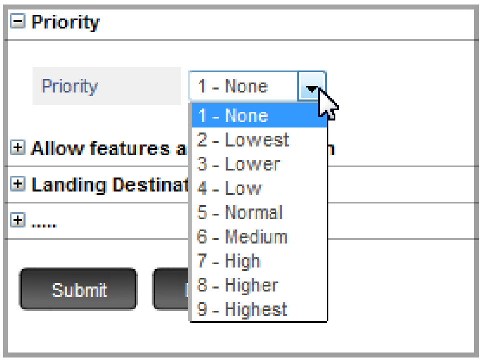
  - Enter the **Mobile Number** of the extension user. The Number can be a maximum of 16 digits. Default: Blank.
  - Enter the **Email ID** of the extension user. The Email ID can be a maximum of 64 characters. Default: Blank.
  - Click **Submit**.

    These parameters are used by the SMS Server application. To know more about this feature, see “SMS Server”.

- Click **Access Code** to expand.

  - In **Access Code**, assign a number to the Extension.
  - In **Name**, assign a name to the Extension for identification.

- Click **Priority** to expand.

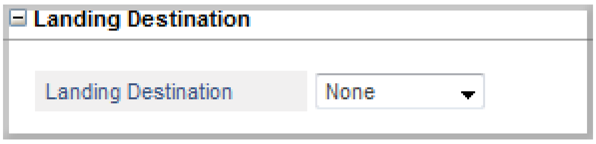


  - Select a Priority level from 1 to 9 which you want to assign to the Extension.

- Click **Allow Features as per Extension** to expand.

  - Select the extension whose features and facilities you want to assign to this Virtual Extension from the list.

- Click **Landing Destination** to expand.



  - Select the Master Extension to which you want to assign this Virtual Extension. Incoming calls for the Virtual Extension will land on this Master Extension, referred to as Landing Destination.

- Click the **More...** link to expand

  - Select the **SMS/Email Group Type**, you want to assign to the extension user. Default: Blank. See "SMS/Email Group" under "SMS Server" to know more.

- Click **Submit**.

To configure another Virtual Extension, click the **Virtual Ext.** <number> tab. Follow the same instructions as above to configure the extension.

How to use

For making outgoing calls, users of Virtual Extensions must use “Walk-In Class of Service”.

Voice Help

The Voice Help feature of SARVAM UCS allows you to record and play short voice messages to provide quick help/ guidance to the extension users. You can use Voice Help to announce important phone numbers or access codes of frequently used features, and similar functions.

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 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

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 “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 for which they are to be used.

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 requirement. 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 four 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 can be played according to the time zone, i.e. working hours, break hours and non-working hours.

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 Day:** "Welcome to Cotton Software**"**.
- **Built-In Auto Attendant - Welcome Greeting message for Break:** "Welcome to Cotton Software. This is lunch time. Please call after 2.00 pm".
- **Built-In Auto Attendant - Welcome Greeting Message for Night:** "Welcome to Cotton Software. We are closed for the day. Please call later".

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 alert them when they dial a wrong or invalid number or inform them when the dialed extension is busy or when there is no response from the dialed extension.

For example,

- **Built-In Auto Attendant - Dial Prompt:** "Please dial the extension number."
- **Built-In Auto Attendant - Destination Ringing Message:** "The number you dialed is ringing."
- **Built-In Auto Attendant - Wrong Dial Message:** "Sorry you have dialed an invalid number."
- **Built-In Auto Attendant - Destination Busy Message: "**The extension you have dialed is busy."
- **Built-In Auto Attendant - Destination No Reply Message: "**The extension you dialed is not responding."
- **Built-In Auto Attendant - No Dial Message:** "Sorry you have not dialed any number**"**.
- **Built-In Auto Attendant - Conference Number**[167]**:** "Please dial the Conference Number.**"**
- **Built-In Auto Attendant - Conference Password:** "Please dial the Conference Password.**"**
- **Built-In Auto Attendant - Call Transfer to Operator:** "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 voice message is played to serve the alarm to the extension on which Alarm is set. With the Voice Message for alarms, extension users can be greeted when they answer the alarm call.

  For example, "Good Morning. This is your wake up call."

You are recommended to record the message "Please press 0 to acknowledge." in the 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.

167. *Refer the description of the feature "Conference Dial-In".*

- **Voice Help:** A voice message can be recorded for providing quick help or guidance to extension users. You can use Voice Help to announce important phone numbers or access codes of frequently used features, and similar functions.

  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. 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 of the Error Tone, such as: *"Please check the number you have dialed".*

Voice message for Confirmation Tone

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 configured for the Call Privilege type set for the Toll Control assigned to the extension, a voice message can be played: *"Please check the number you have dialed."* or *“This facility is not available to 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 the extension.

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

ETERNITY 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 Built-In Auto Attendant Greetings Message on a Built-In Auto Attendant enabled trunk. On an Auto-Answer enabled trunk these messages are played before the Trunk Auto Answer greeting message. 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

Eternity allows extension users to make outgoing calls that can be routed through different networks. When outgoing calls are made using the Mobile or SIP, 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 play a voice message (Outgoing Call-Busy message) to the callers.

PIN Dialing

You can record and play a voice message instead of beeps to prompt the user to dial PIN while using the PIN Dialing feature. See "PIN Dialing" for more details.

Recording Voice Messages

You can record Voice Messages on voice modules from a 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 SLT.

Record the Voice Message, dial:

- ***2502-Voice Module Number***
  Where,
  Voice Module is from 01 to 16.

- If your phone is an Extended IP Phone, 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 as per the module size you wish to assign to the module.
- 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.

Make sure the naming convention of the voice message file should be in the correct format. The allowed characters for the filename are A to Z, a to z, 0 to 9, - and _.

How to configure

By default, Voice Module Number 01 to 13 are assigned to certain applications like Music on Hold, Built-In Auto Attendant and Trunk Auto Answer Greeting and Built-In Auto Attendant Guidance Messages. Each voice module also contains a pre-recorded voice message (in WAV format) for the particular application to which the module is assigned. See the table for details.

Voice Module Number	Voice Message Application	Voice Message
01	Music-On-Hold	
02	Morning Greeting (Built-In Auto Attendant)	Good Morning!
03	Afternoon Greeting (Built-In Auto Attendant)	Good Afternoon!
04	Evening Greeting (Built-In Auto Attendant)	Good Evening!
05	Built-In Auto Attendant Welcome Greeting for Day Time (Working Hours)	Welcome!
06	Built-In Auto Attendant Welcome Greeting for Night/Break 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.
14	Not assigned to any application	Blank
15	Not assigned to any application	Blank
16	Not assigned to any application	Blank

If these default Voice Modules and Voice Messages serve your purpose, you may use them. No further configuration is required, as these modules are loaded in the system by default.

However, if you want to use custom messages, you must first record the messages on voice modules.

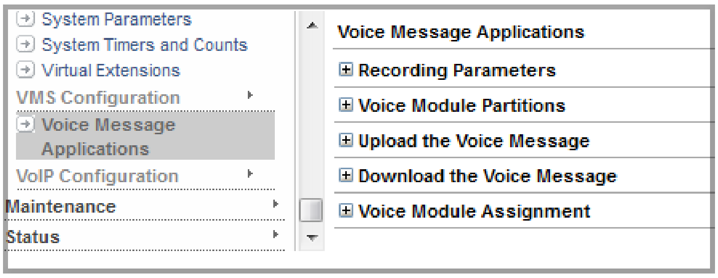
For example, if you want to use a different voice message for the Built-In Auto Attendant Welcome Greeting for Night Time, record a suitable message of your choice on Voice Module number 06.

If you want to use a voice message for DND Notification, you may record a suitable message on a blank module and assign this module to the DND Notification application.

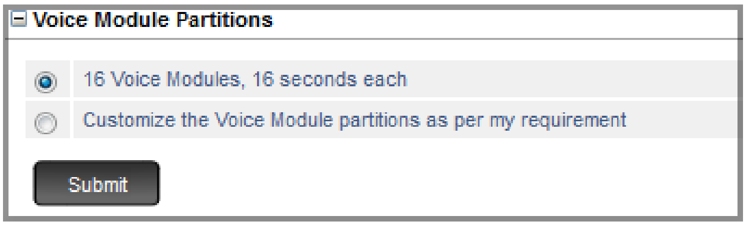
You can reassign the default Voice Modules to other Voice Message Applications as required. The same Voice Module can also be assigned to multiple Voice Message Applications.

Configuring Voice Messages Application parameters using Jeeves

- Login as System Engineer.
- Under **Advanced Settings**, click **Voice Message Applications**.



- 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. If you upload a customized MOH and then delete the same, the default MOH will be played by the system. This default MOH is stored in the system and cannot be deleted.*

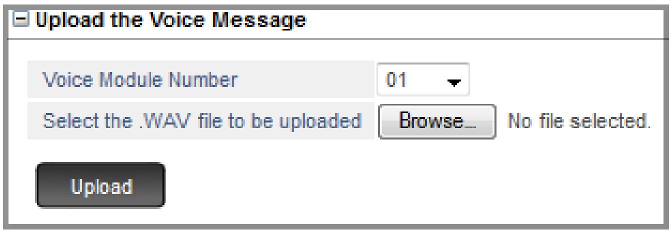
- *Only the first or the second Module Number can be assigned the Module Size as 240 seconds.*

- *When the modules sizes are changed, the system will automatically disable/enable the remaining module sizes.*

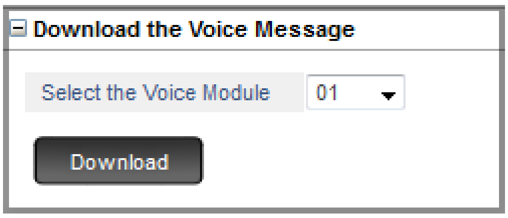
- *The messages from the disabled/resized Module Number will be deleted. In future, if you wish to use these Modules Numbers, you need to upload the messages in these modules again.*

- *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.**
- 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

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!
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.

- *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.*

- Click **Submit.**

Walk-In Class of Service

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' in to any extension port—SLT 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.

You can allow extension users either of the above types of Walk-In by configuring the **Walk-In/Out** mode in their extension port parameters, as:

- *Walk out automatically on completion of call*
  Or
- *Walk out on user request*

Walk-In Class of Service is protected with the User Password.To be able to use this feature, extension users must first change the default User Password **1111**.

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 SIP Phone with the number 301, with long distance calling facility (toll control: International calls). Extension user B has a SLT with the number 201, without long distance calling (toll control: local calls).

Here,

- 301 is the **Source Extension**, whose CoS and Toll Control and other features / facilities are to be used from another extension 201 by performing Walk-In.
- 201 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 201 by dialing:
  - the feature code for 'Walk-In Class of Service'.
  - A's extension number, 301.
  - 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 of the **Source Extension** 301 on the **Destination Extension** 201.

- Extension user A can make external and internal call from extension B.

- If Extension 301 has **Walk out automatically on completion of call** selected as the **Walk-In/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 201.

- If Extension 301 has **Walk out on user request** selected as the **Walk-In/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, if:

  - another extension user walks into extension 201
  - there is an incoming call on 201.

- *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 201 using Walk-In will be calculated and charged to extension 301 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 201, using Walk-In, will be recorded in the Station Message Detail Record of extension 301 only.*

Walking into another extension to access a feature

- Extension user A with number 301 has Call Forward in the Class of Service.

- Extension user B with number 201 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 (301)
    - A's User Password[168]

- On successful Walk-In, the system applies the Class of Service, Toll Control and other features / facilities of the Source Extension 301 on the Destination Extension 201.

- From extension 201, extension user A sets Call Forward for 301, to an external number.

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168. 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 301 and cancel Call Forward from 301, or can cancel Call Forward from 201, if user A is still walked-in on 201.

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 201, until user A has walked out from 201.*

Incoming calls on the Destination Extension 201 will work according to the setting of the Destination Extension only, whereas outgoing calls on the Destination Extension will work according to the settings of the Source Extension 301.

The following set of features of the Destination Extension will remain unaffected:
- *Key map*
- *Language*
- *Priority*
- *Call Pick-Up group*
- *Personal Directory*

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. It does not require any other configuration except selection of the Walk-In/Out mode for the extension in its port parameters.

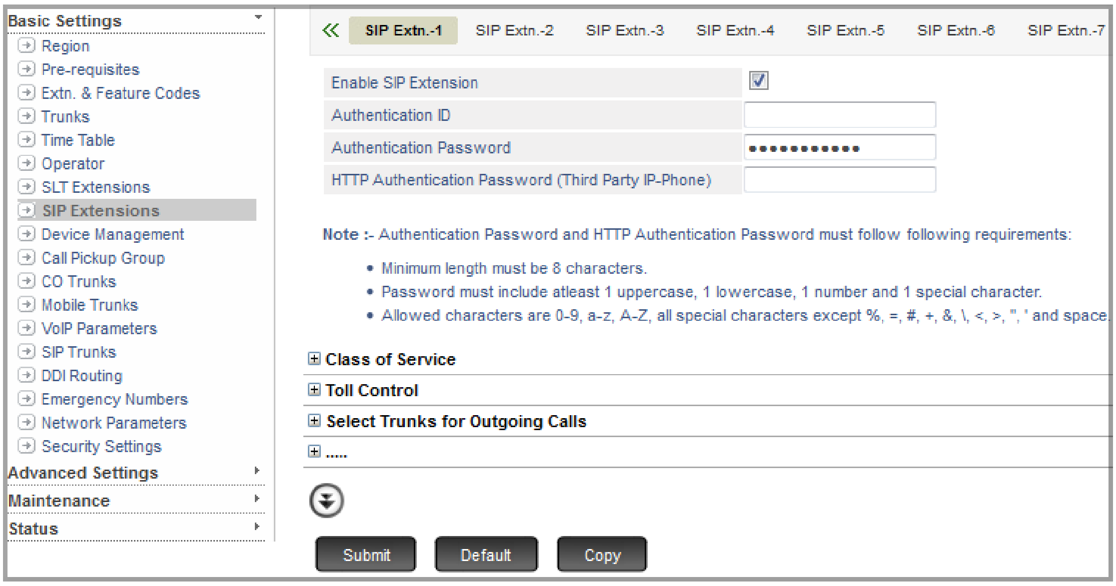
If you want to allow different walk-out modes to different extensions, then decide which of the extensions are to be allowed '*Walk out automatically on completion of call*' and which are to be allowed '*Walk out on user request*'.

To configure the Walk-In/Out mode for extensions,

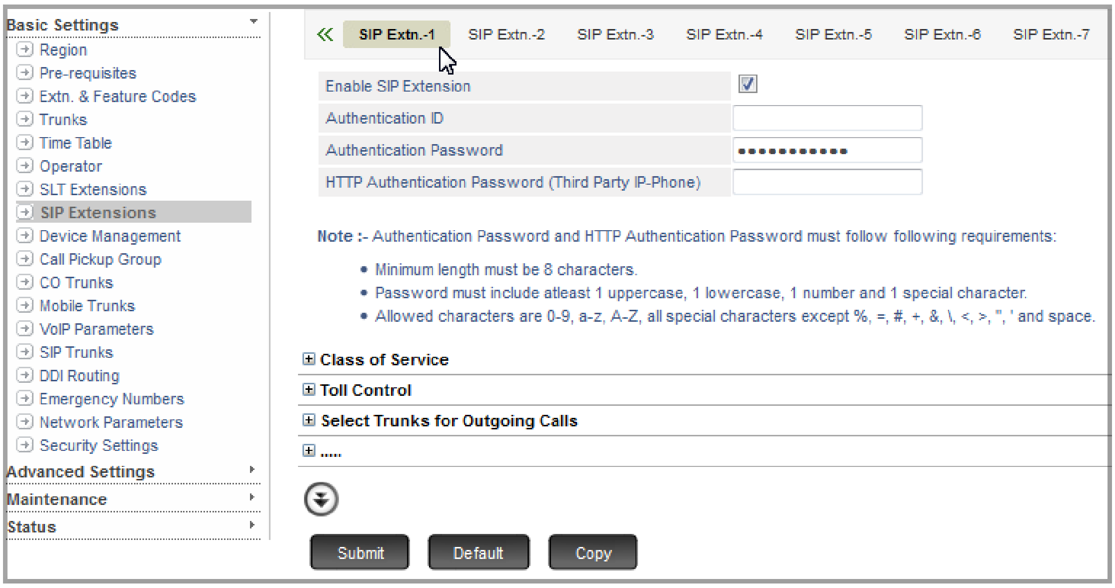
- Login as System Engineer.

- Under **Basic Settings**, click the desired extension type — **SLT Extension** or **SIP Extension** as per your requirement.

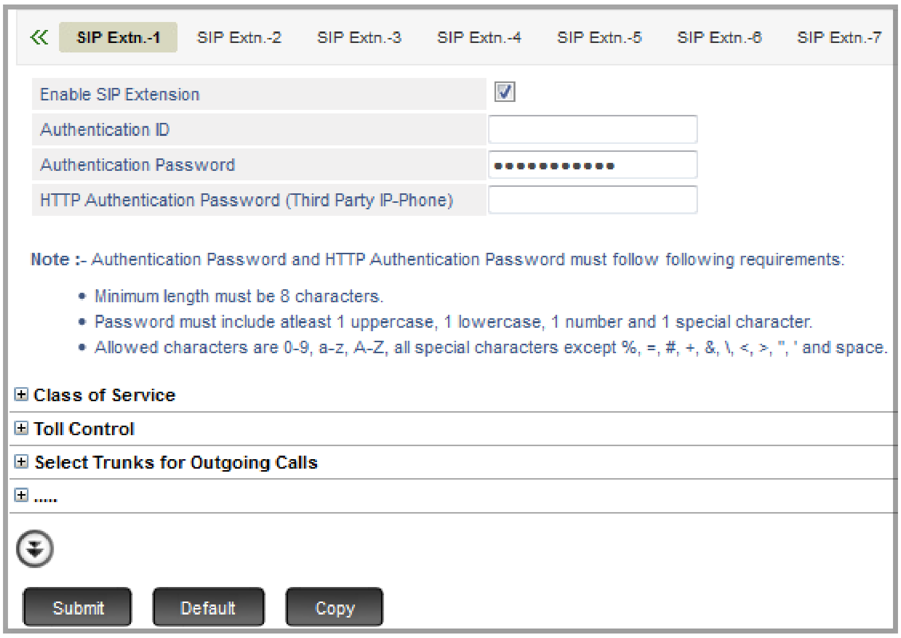
The page of the selected extension type opens.



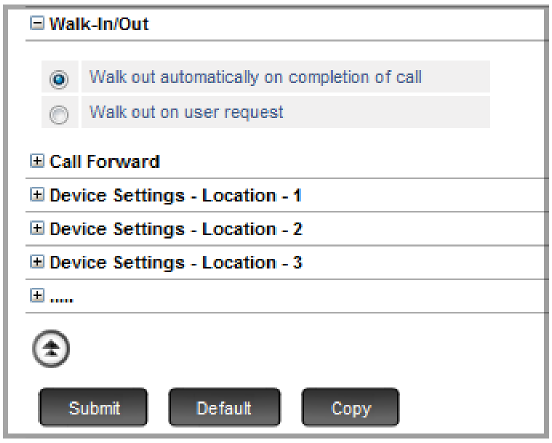
- To select an extension number on the page, select the **Extension Number** tab.



- Click **More** to expand parameter options for the selected Extension number.



- Click **Walk-In/Out** to expand.



- Select the walk-out mode, as required:
  - **Walk out automatically on completion of call**
  - **Walk out on user request**
- Click **Submit**.

How to use

For 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 has set ‘Walk out on user request’ as Walk-Out mode. If the Source Extension has set ‘Walk out automatically on completion of call’ 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 'Walk out automatically on completion of 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 10

Voice Mail Features

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".

If you have CDMA Mobile SIM is installed in your system, it is recommended to avoid using Voice Mail features as the DTMF Detection might not work efficiently.

To access your Mailbox from your own extension:

- Dial 390.
- 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 3801/3802/3803.
- 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 390 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.

Accessing General Mailbox

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 mailbox 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 mailbox 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 by 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 **New Message Delivery Option in Mailbox Full Condition** option in the **Voicemail Settings** of the desired extensions.

  For the option **New Message Delivery Option in Mailbox Full Condition,** select **Deliver to General Mailbox**.

- Enable the General Mailbox feature in the "Class of Service (CoS)" of the extensions.

- If required, assign a Password for accessing the General Mailbox in the "General Mailbox Settings".

For instructions on configuring these parameters see "SIP Extensions" under "Basic Settings".

How to use

For 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.

Alarms and Reminders

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

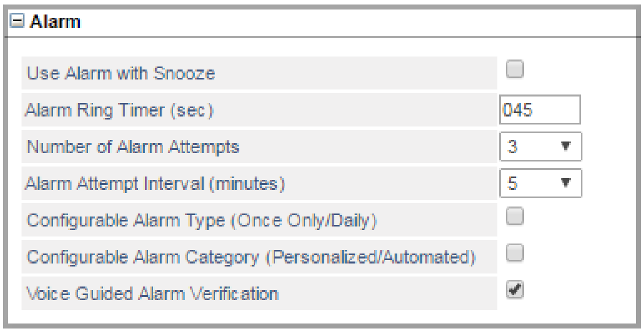
The VMS allows you to enable/disable the **Alarm Verification** for alarms and reminders, allowing extension users who want to use alarms and reminders to confirm:

- 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.

To configure Alarm Verification,

- Login as System Engineer.
- Under **Advanced Settings**, click **System Parameters**.
- On the System Parameters page, click **Alarm** to expand.



- Select the **Voice Guided Alarm Verification** check box to enable Alarm Verification. Default: Enabled.
- Click **Submit.**

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[169]. 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 …." followed by the prompt: To Confirm[170], Press 1, To Re-enter, Press 2.".

169.The Date and time format depends on the Region/Country selected for the system.
170. This option will not be played if Alarm Verification is disabled in the System Parameters.

- 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 cancelled." 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."
- 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[171]. 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."

171.The date format in the prompt will be MM DD YYYY, if you selected USA as the Region/Country for your system.

- 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 cancelled." 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."

  - 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[172], 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."

172.This option will not be played, if Alarm Verification is disabled in the System parameters.

- Dial # to cancel Reminder → VMS responds with: "Your Reminder is cancelled." 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."

Broadcast Message

Broadcasting Message allows you to send the same message to all extension users having voicemail, 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".
- 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.

Call Transfer Types

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”.

Dial by Extension Number

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 at the home position.

How to use

External Callers

- 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 — Day Hour, Break Hour, Night Hour. In the above prompt the greeting is as per Day hour.

- 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.

Extension Users

When you call the VMS by dialing **3801/3802/3803**,

- 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 ‘#’.

- 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.

Dial By Name

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.

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

External Callers

- 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 — Day Hour, Break Hour, Night Hour. In the above prompt the greeting is as per Day hour.

- The caller dial 6 → VMS prompts: "Please enter first three letters of the name."
- If multiple matches are found the VMS prompts: "More than on 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: "Sorry 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.

Extension Users

When you call the VMS by dialing **3801/3802/3803**,

- 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 6 → VMS prompts: "Please enter first three letters of the name."
- Dial valid digits → VMS prompts: "To confirm press 1, to cancel press 2."

- Dial 1 → VMS transfers the call as per the transfer type of the selected extension. Talk.
    - If the caller dials invalid digits, the VMS prompts: "Sorry no match found." followed by the prompt: "Please enter first three letters of the name."

Extension users can use "Dial By Name" also to reach another extension.

Email Based Notification

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".

Forwarding Messages

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

When you call the VMS by dialing **390**,

- 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.

Join Conference Dial-In using VMAA

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.
- 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

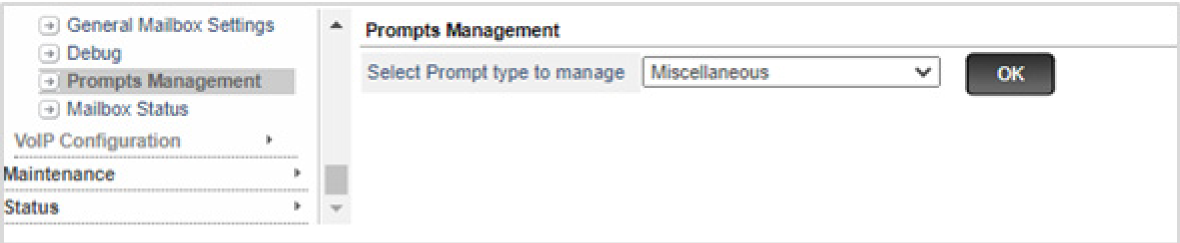
- For instructions to enable Voice Mail Auto Attendant on the desired trunks, see "CO Trunks", "Mobile Trunks" and "SIP Trunks" under *Basic Settings*.
- To customize the Conference prompts as per your requirement, see "Recording Voice Messages".

- 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".

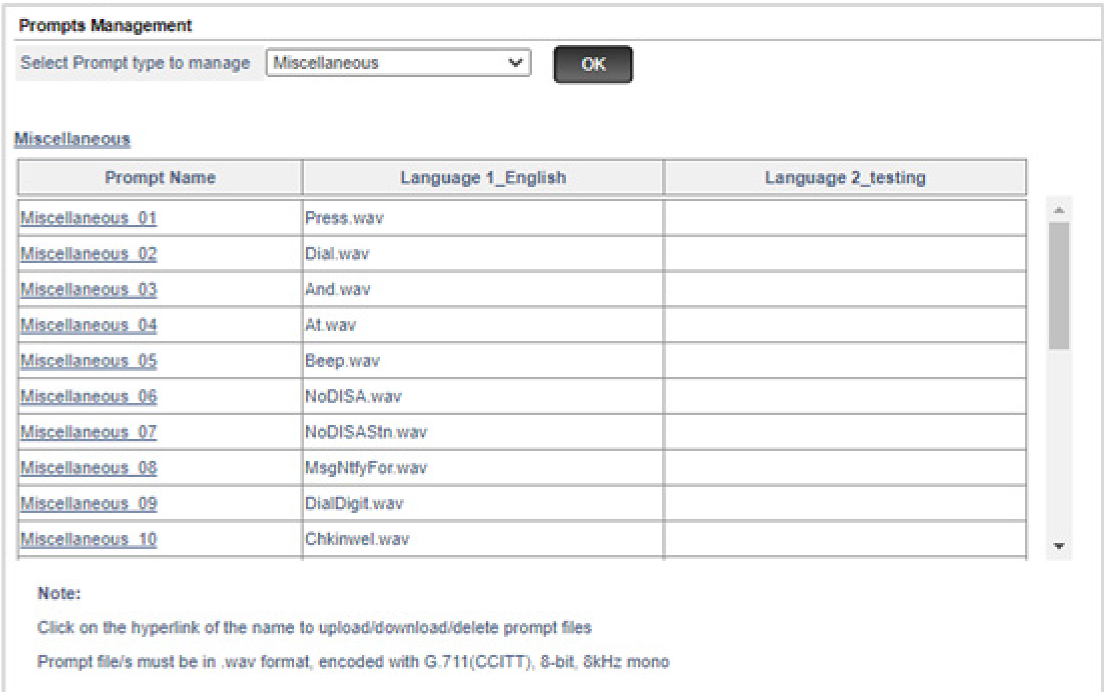
Dial-In Prompt Upload

After upgrading the system, to upload the VMS Dial-In Prompt, follow the steps mentioned below:

- Under **Configuration**, click **VMS Configuration**.
- Click **Prompts Management.**

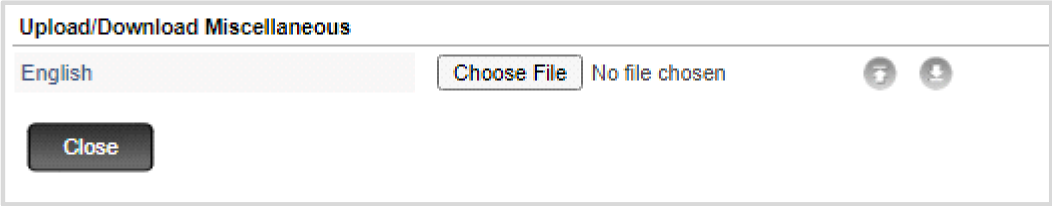


- In **Select Prompt type to manage** select **Miscellaneous**.

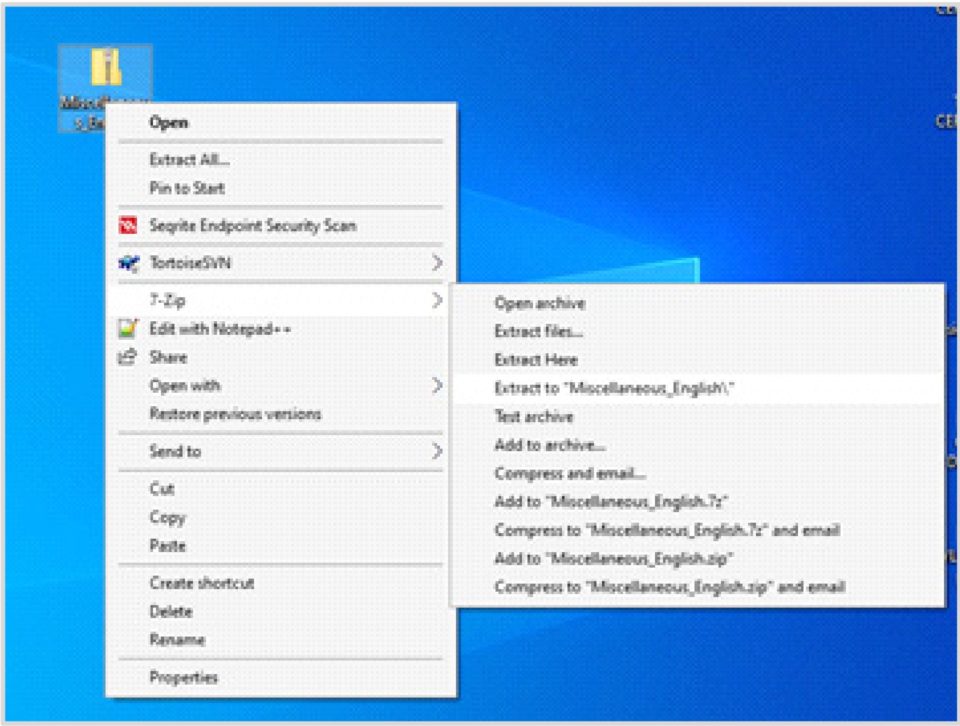


- Click **OK**.
- Click on **Miscellaneous**

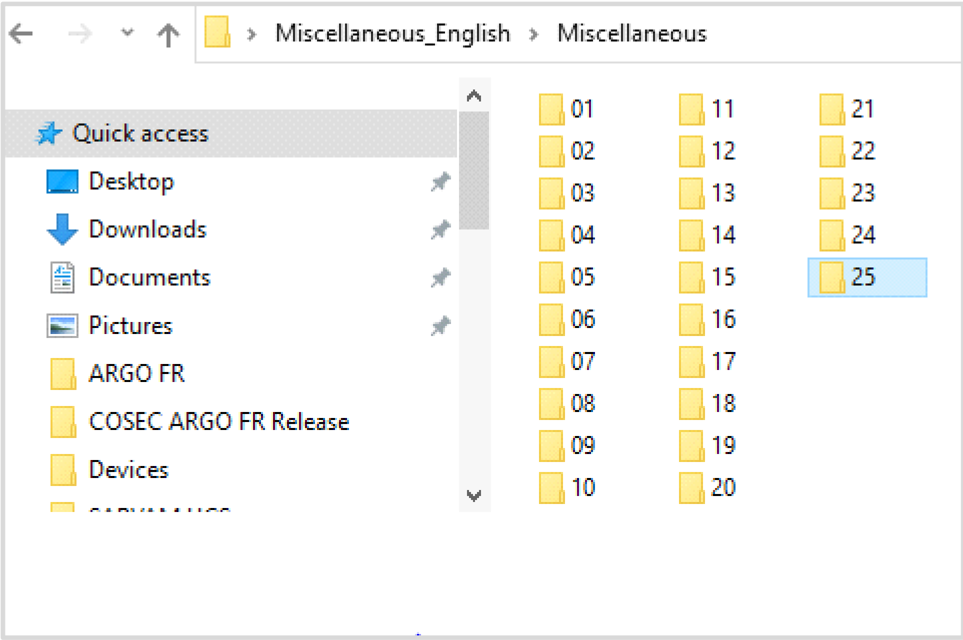
- To download all the prompts of, click .



- The Miscellaneous_English zip folder is downloaded. Extract the folder files.



- Now click the extracted folder and create a new folder 25 in it.



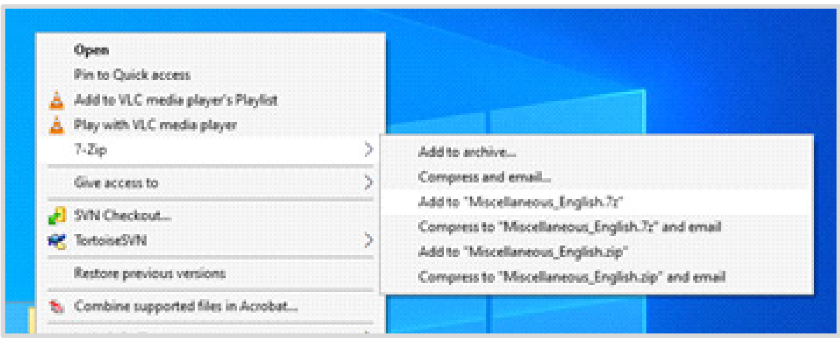
- Click to open Folder number 01, it has two files. Make sure you delete one file which you do not require. This folder must not have two files.

- Copy the new prompt in folder 25 from the USB package (path:Voicemail\database\prompts\language_1\Miscellaneous\25)

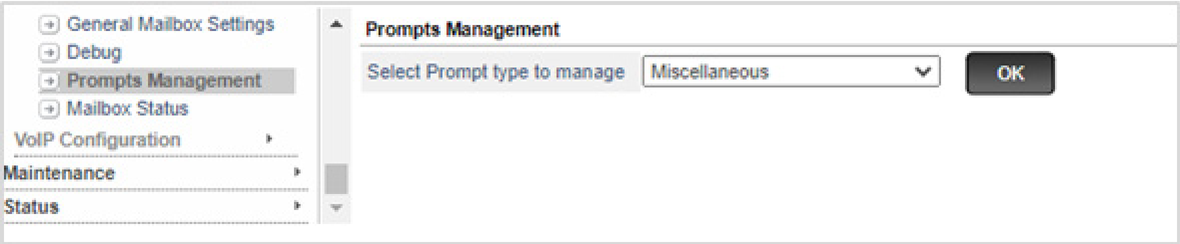
Contact the Dealer/Distributor or Technical Support for the USB package.

- Go back to the extracted Miscellaneous folder and open the folder 25. Paste the prompt here.

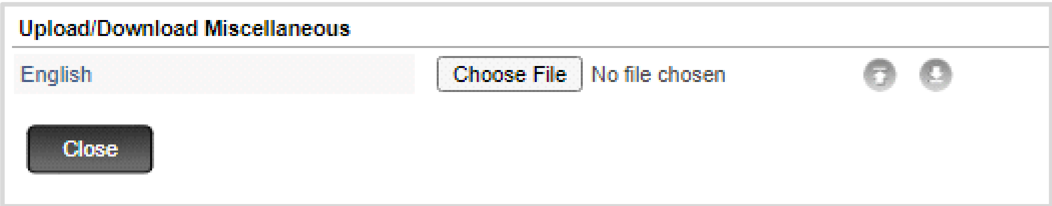
- Zip the Miscellaneous folder.



- 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.

Leaving a Message

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 — Day Hour, Break Hour, Night Hour. In the above prompt the greeting is as per Day hour.

- 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".

Listening to Messages

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 extension 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 **390**,

- 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 '#'.

Mailbox Settings

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 **390**,
- 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".

Message Notification

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.

Message Verification

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

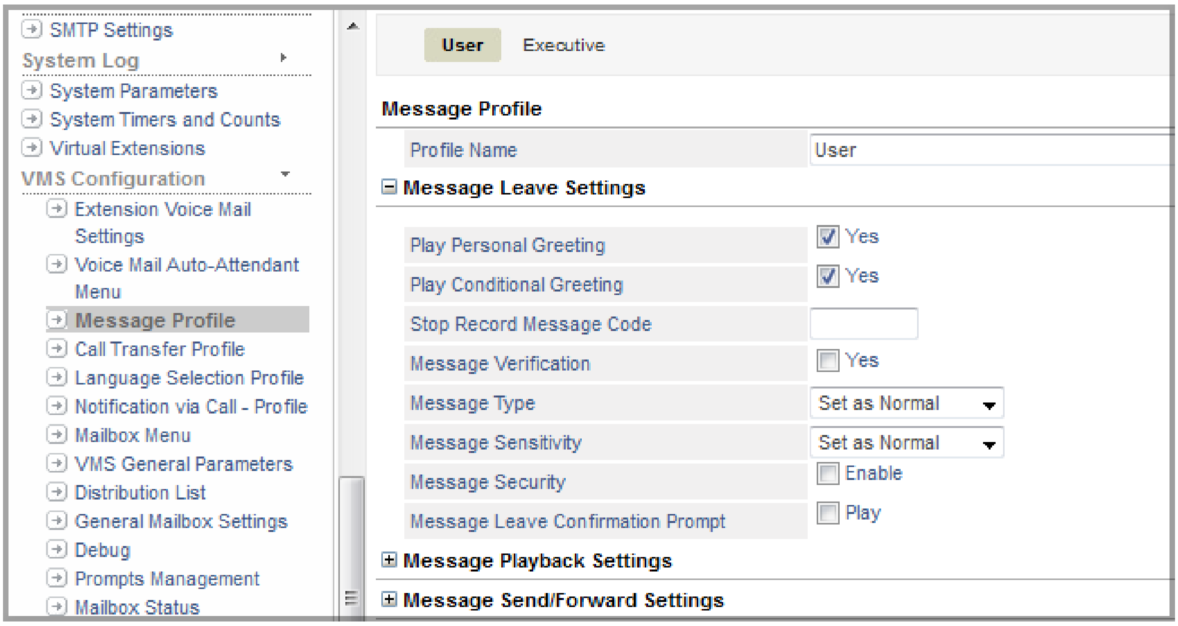
- For Message Verification to work, it must be enabled in the VMS.
- With Message Verification enabled, each time a caller or an extension user records a message, the VMS offers to the caller/extension user the option to verify the recorded message and re-record the message, if they want.
- When the 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,

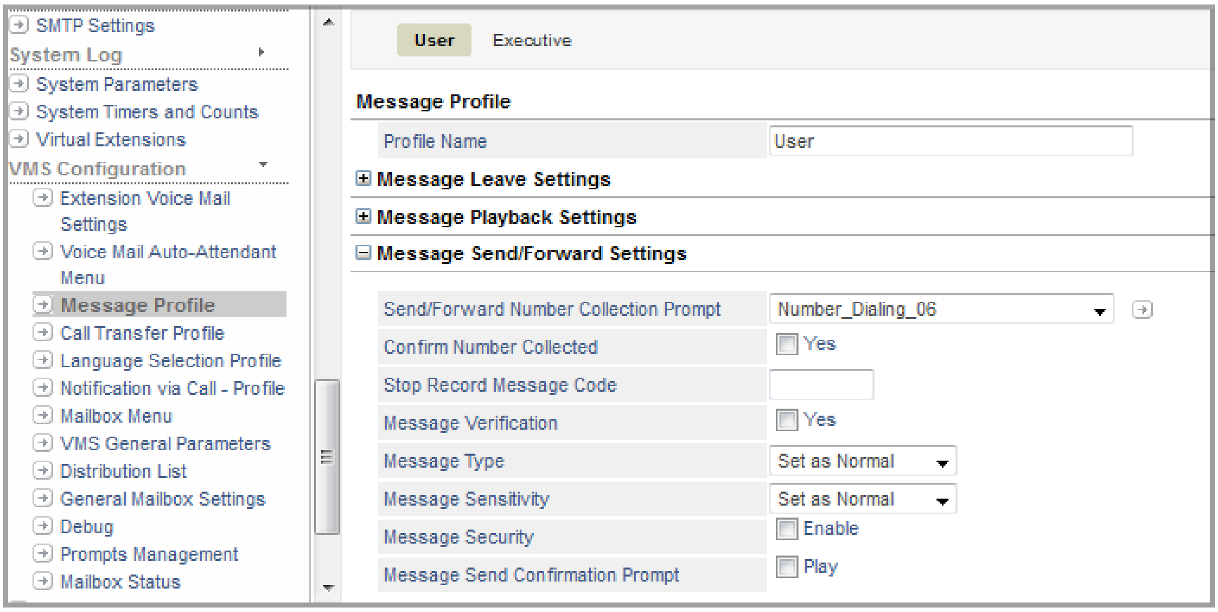
- Login as System Engineer.
- Click **Advanced Settings**, click **VMS Configuration.**
- Click **Message Profile**.
- To allow external callers to check the message they have recorded in the mailbox of an extension user or extension user to check the message they have recorded as a reply to any message.

- Click **Message Leave Settings** to expand.



- Select the **Message Verification** check box, to enable.
- Click **Submit**.

- To 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**.

Message Wait Notification via Call

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 Voice Mail 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 “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”.

Recording Personal Greetings

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 **390**,

- 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

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 **390**,

- 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 '#' ".

Redirecting Messages

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

To Redirect Messages using phone,

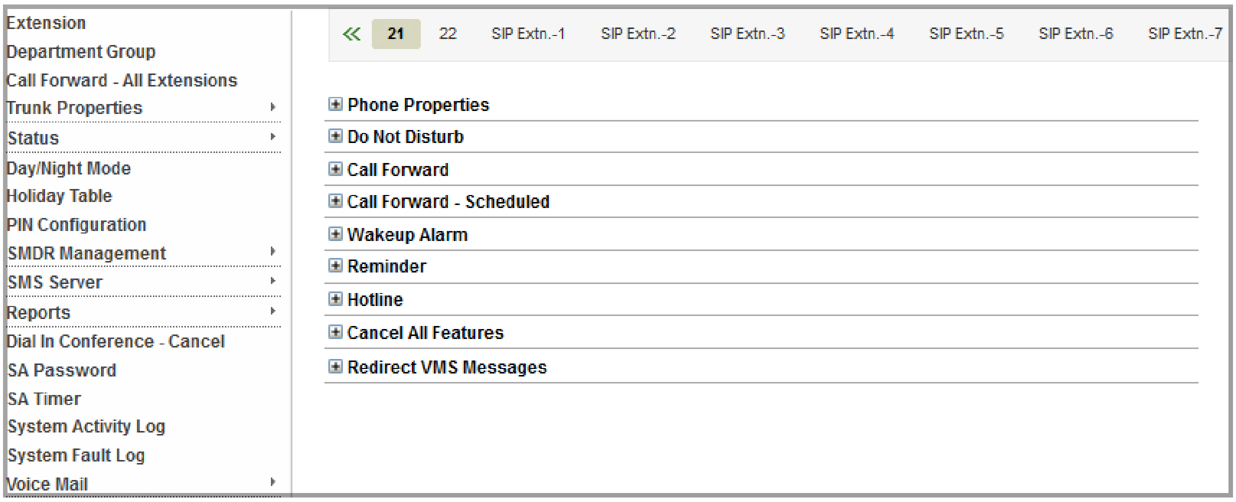
- Call the VMS by dialing **390**.
- The VMS takes you to your mailbox.
- VMS responds with: "You have <n> new messages" followed by the prompt: "Enter your mailbox password".
- Enter your mailbox password → VMS prompts: "You have n new/no new messages."
- VMS prompts: "To listen to new messages press '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

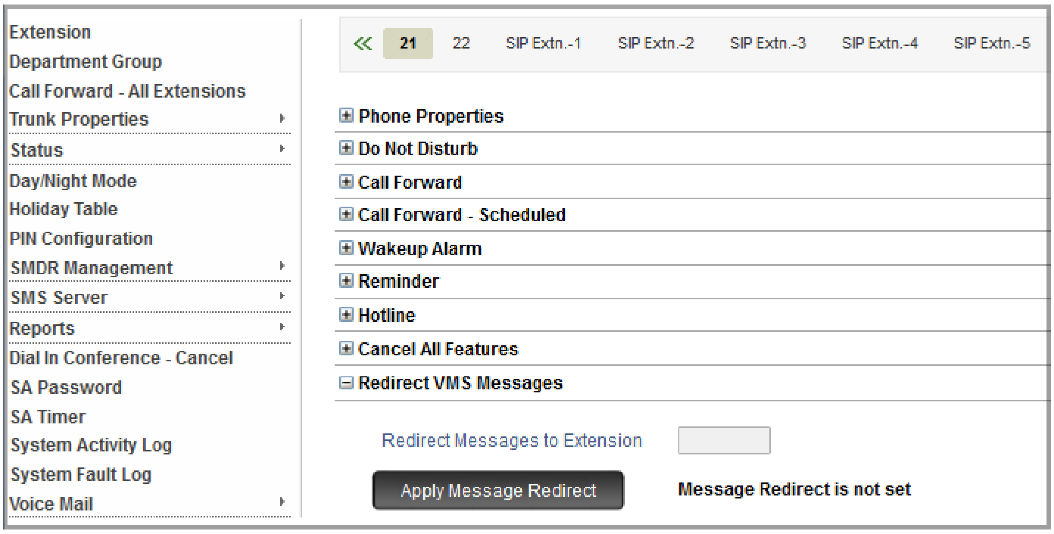
- Login as System Administrator.

- Click **Extension**.



- Click the desired Extension tab.

- Click **Redirect VMS Messages** to expand.



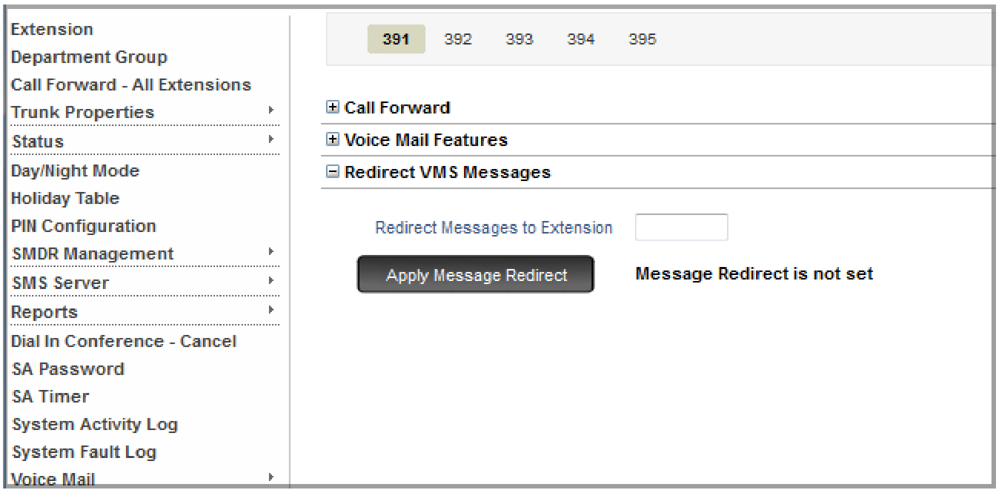
- In **Redirect Messages to Extension**, enter the extension number on which you want the messages to be redirected.

- Click the **Apply Message Redirection** button.

For Department Group

- Login 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 SLT.

  To redirect Messages, dial:
  - ***1072-314-Source Station-Code-Destination Station***
    Where,
    Source Station is the Access Code of any SLT, SIP Extension, Department Group.
    Destination Station is the Access Code of any SLT, SIP Extension, Department Group or General Mailbox.

    Code is
    0 to Cancel Message Redirection
    1 is to Set Message Redirection

- Exit SA mode.

Sending Messages

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 **390**,

- 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.

VMS DISA Login

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.
- you must select **Voice Mail Auto Attendant** as the **Auto Attendant** option on the desired trunks: CO, Mobile, SIP.
- you must enable DISA in the “Class of Service (CoS)” of the extension on to which the caller is allowed to log into using (PIN Authentication).
- you must change the default **User Password** (1111) of the extension on to which 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, Day hours/ Break hours/ Night 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).
- 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 “CO Trunks”, “Mobile Trunks”, and “SIP Trunks” under *Basic Settings*.
- For instructions to enable DISA in the “Class of Service (CoS)” of the extensions which you want to allow callers to access using DISA, see SLT/SIP extension under “Basic Settings”.
- 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”.
- 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”.

CHAPTER 11 **Maintenance**

Certificate Management

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 "VoIP Parameters".
- Select **Transport Mode**[173] 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.

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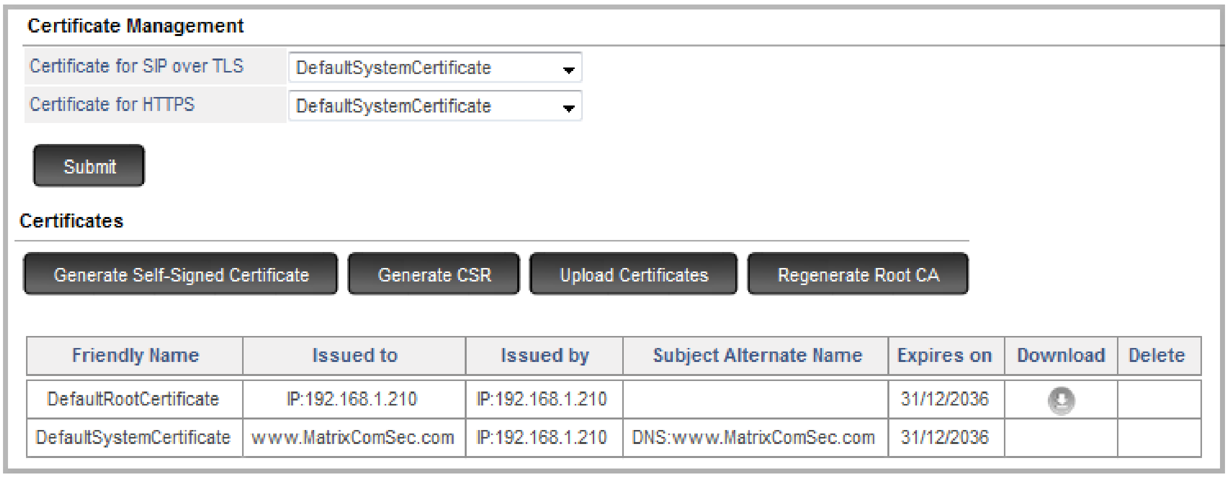
173. Not applicable for SPARSH VP248 and Standard SIP Phones.

- 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**.
- 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

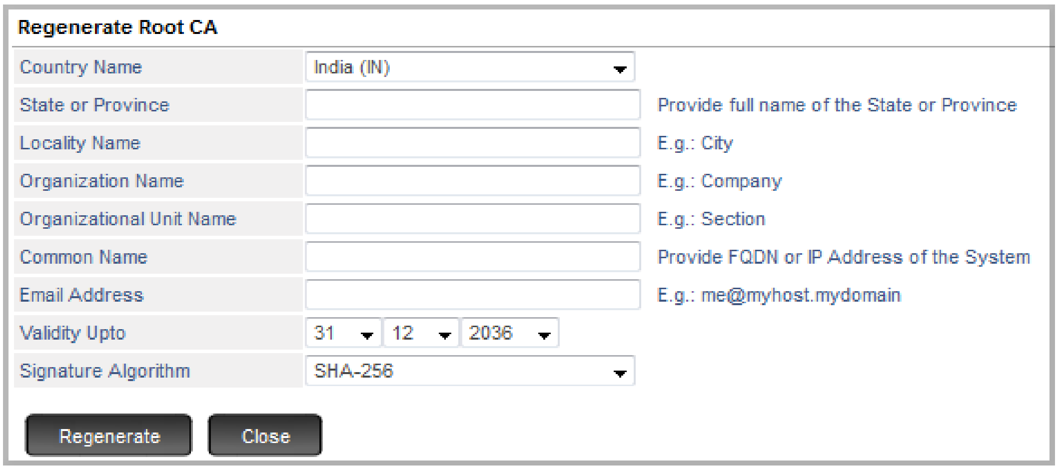
Regenerate Root CA

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

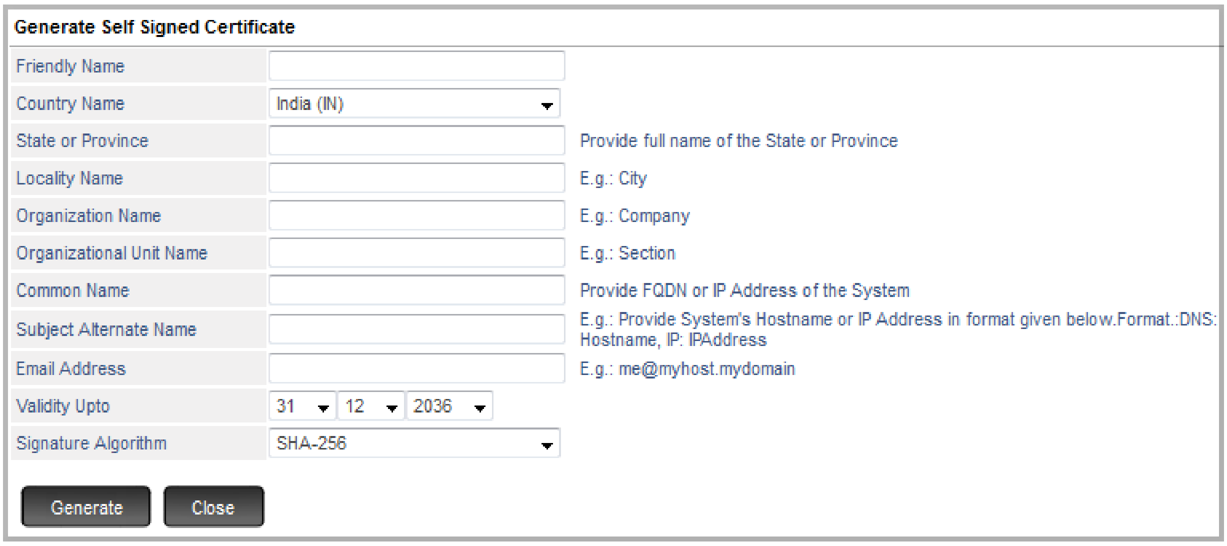
Depending on your installation scenario, you can generate a Self-Signed System Certificate using the details of your own organization.

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.

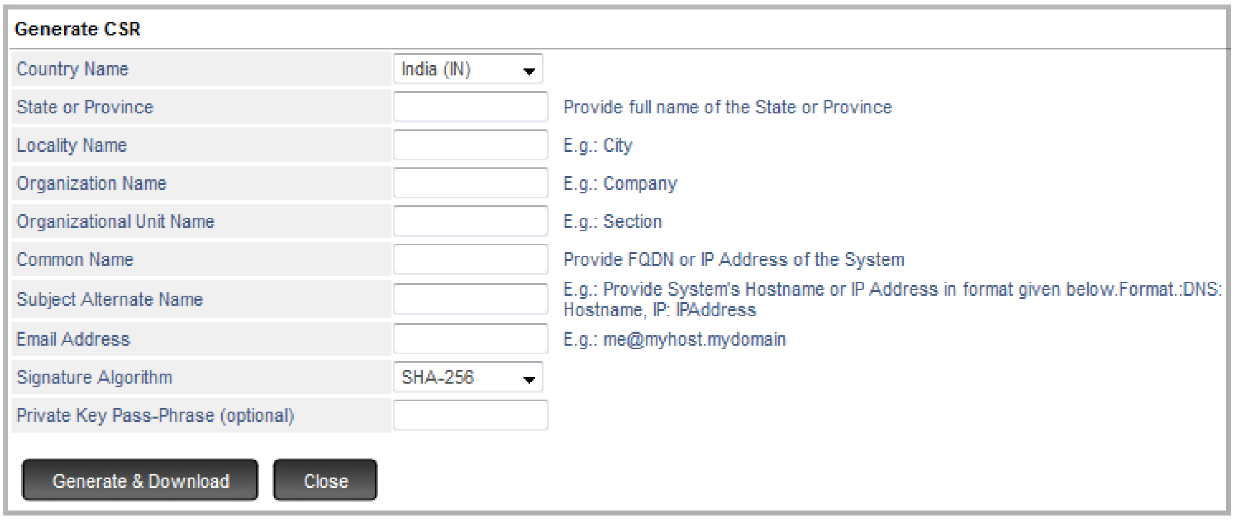
  A message appears to confirm that the Certificates have been generated successfully.
- Click **OK** to close the window**.** You will be redirected to the Certificate Management page.

  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.



Generate CSR

Country Name	India (IN)	
State or Province		Provide full name of the State or Province
Locality Name		E.g.: City
Organization Name		E.g.: Company
Organizational Unit Name		E.g.: Section
Common Name		Provide FQDN or IP Address of the System
Subject Alternate Name		E.g.: Provide System's Hostname or IP Address in format given below.Format.:DNS: Hostname, IP: IPAddress
Email Address		E.g.: me@myhost.mydomain
Signature Algorithm	SHA-256	
Private Key Pass-Phrase (optional)		

Generate & Download | Close

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.*

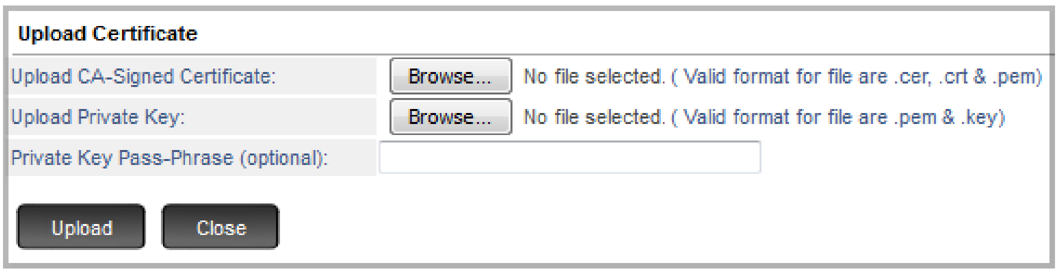
Upload Certificates

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.

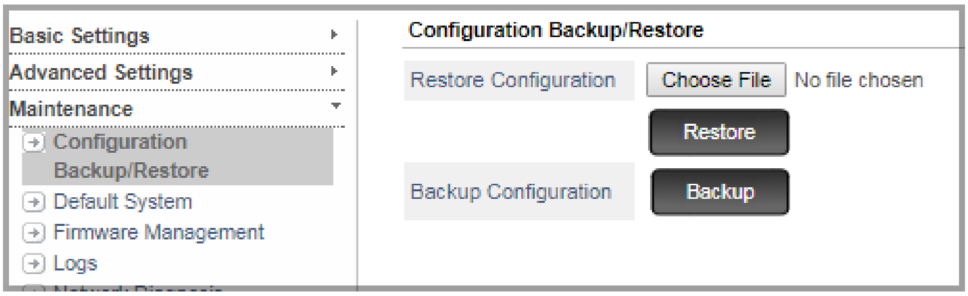
  The CA-Signed System Certificate you uploaded will appear in the **Certificates** list.

  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

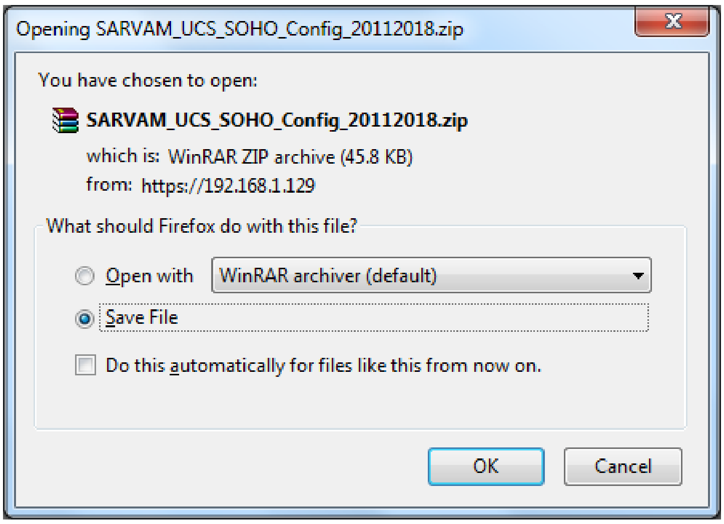
- To restore configuration files from any hard drive to the system, **Restore Configuration** option is provided.
- Click on the **Browse** button to reach the location on the local disk where the configuration files are stored in PC. Make sure that the file is a zip file.
- After selecting the required 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_SOHO_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.

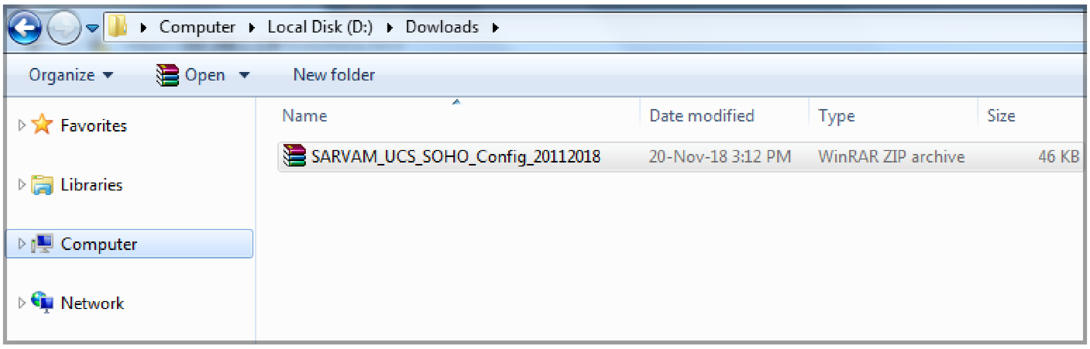
- *The above window display depends upon the browser you are using. Check the* ***Download Settings*** *of your browser and set the Download path accordingly.*

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.

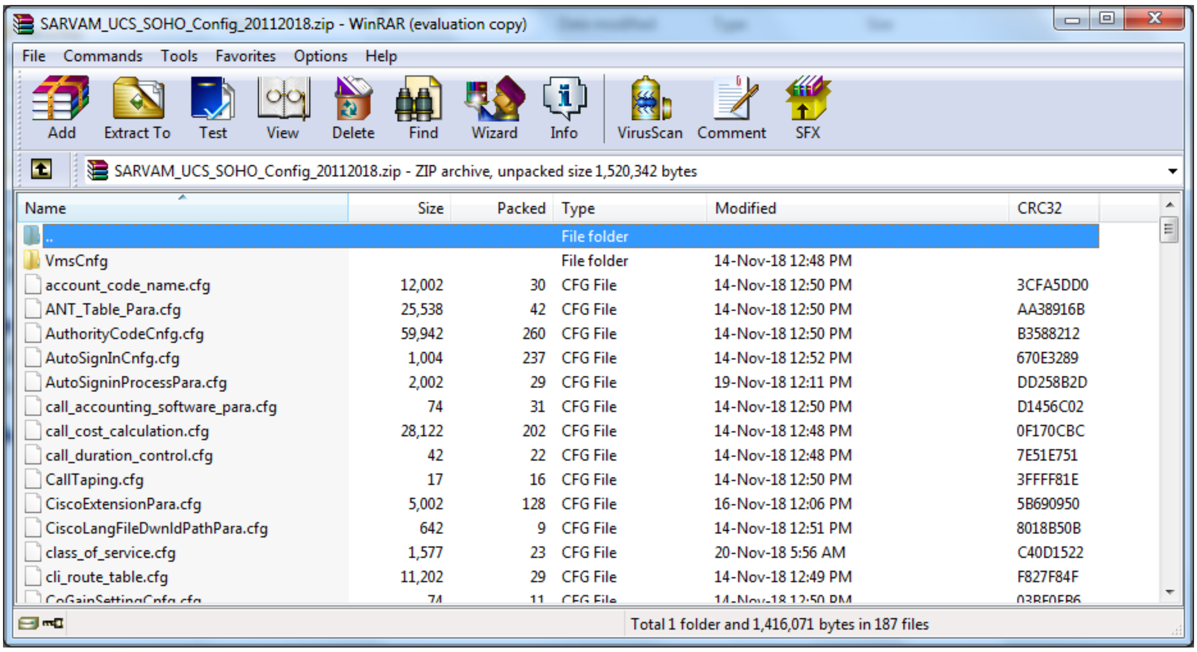
- *If you are using Mozilla Firefox (version 3.5.1 recommended), before you save the configuration files, set the* ***Downloads*** *option of your browser as* ***Always ask me where to save files****.*

- 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_SOHO_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

SARVAM UCS is supplied with preset values for system and feature settings, 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".

How it works

The default settings are to be loaded or restored in the following situations:

1. ***Installing 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.

Country Code	Country Name	Default Time Zone	Default DST Mode	Default DST Schedule Type	CPTG[a]	Default 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	
006	Australia (Note2) (Adelaide)	GMT+09:30	Scheduled	2	05	English		9	0	

Country Code	Country Name	Default Time Zone	Default DST Mode	Default DST Schedule Type	CPTG[a]	Default Language	Distinctive Ring[b]	Opr	TAC	Abbr. Dialing
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	9	
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	Portugues e		0	9	
020	Brazil (Brasilia, Rio de Janeiro, Sao Paulo)	GMT-03:00	Scheduled	4	06	Portugues e		0	9	
021	Brazil (Manaus)	GMT-04:00			06	Portugues e		0	9	
022	Brazil (Acre)	GMT-05:00			06	Portugues e		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	
042	Fiji	GMT+12:00				English				
043	Finland	GMT+02:00	Scheduled	8		English		9	0	

Country Code	Country Name	Default Time Zone	Default DST Mode	Default DST Schedule Type	CPTG[a]	Default Language	Distinctive Ring[b]	Opr	TAC	Abbr. Dialing
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				
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 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	9	
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 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. ***Malfunctioning of the System***.

   When there is a system malfunction, possibly caused by a configuration error that you are unable to diagnose, you may restore default settings.

Restoring Default Settings

Default settings can be loaded or restored from the SE mode (software default).

To be able to do this, you must have the **SE password**.

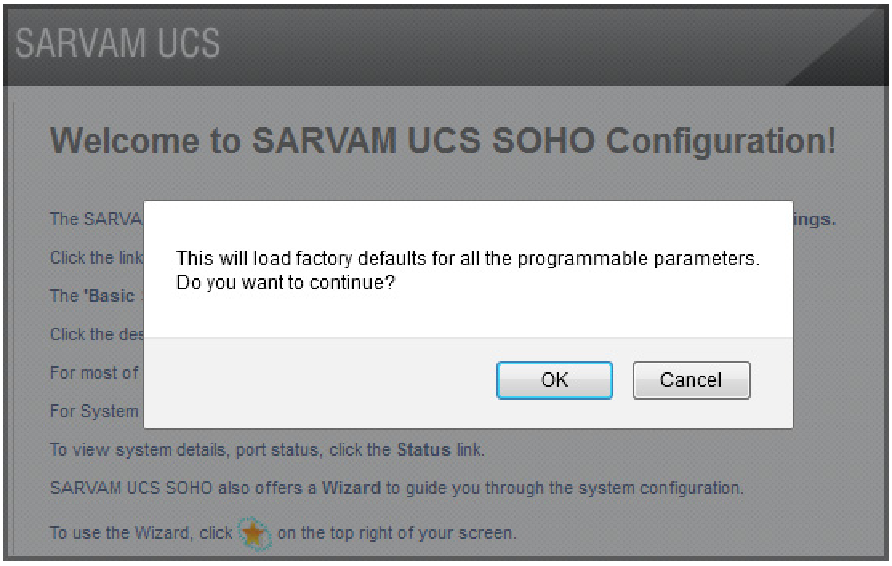
All parameters will be assigned default values except the following, when the system is set to default:

- SE Password
- SA Password

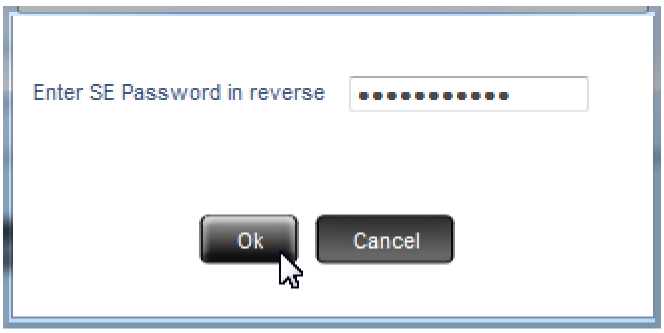
- Date Format
- CPTG Type
- Voice Module
- System Activity Log
- System Fault Log
- SMDR
- SIM PIN of Mobile
- Network Parameters

Restoring Default Settings using Jeeves

- Login as System Engineer.
- Under **Maintenance**, click **Default the System**.



- 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. E.g.: if it is 1234, enter 4321 and click **OK**.
- The system will restart and load the default settings.

- You are returned to the Login page of Jeeves.
- You may now enter SE mode again.
- The names configured for the Extensions will disappear and the default extension numbers assigned to the extensions will appear.

Without the SE Password, you cannot restore default values using software. If you forget the SE Password, you must first restore default SE password by changing the Jumper Settings on the CPU module. See *"System Security"* *for more details.*

Modified default parameter values for Firmwares later than V1R6.7

Parameters	Old Default Value	New Default Value	New Default Value's impact on the functionality
Extensions - SLT and SIP			
Class of Service			
Closed User Group (CUG)	Enabled	Disabled	Extension Users will not be able to access this feature by default.
Global Directory Part-1	Enabled	Disabled	Extension Users will not be able to access this feature by default.
Trunk-Trunk Transfer	Enabled	Disabled	Extension Users will not be able to access this feature by default.
Toll Control			
Calls allowed during Day	All Calls	No Calls	Extension Users will not be able to make external call.
Calls allowed during Night/Break	All Calls	No Calls	Extension Users will not be able to make external call.
Calls allowed for Lock Level 1	Local	No Calls	Extension Users will not be able to make external call.
Calls allowed for Lock Level 2	National	No Calls	Extension Users will not be able to make external call.
Calls allowed for Lock Level 3	No Calls	No Calls	Extension Users will not be able to make external call.
More			
Call Duration Control	Apply as per CDC profile 1 (the Apply CDC for calls matching with numbers is blank)	Apply as per CDC profile 1 (the Apply CDC for calls matching with numbers is configured)	CDC will be applied on All Extension for external calls.
CO Trunks			
Call Budget	Disabled	Enabled	Call Budget will be applied on the Trunk. It is set as 300 minutes.
SIP Trunk and Mobile Ports			
Call Budget	Disabled	Enabled	Call Budget will be applied on the Trunk. It is set as 300 minutes.

SIP Trunk as Peer-to-Peer and Treat Incoming Calls as Station			
Class of Service			
Global Directory Part-1	Enabled	Disabled	Extension Users will not be able to access this feature by default.
Toll Control			
Calls allowed during Day	All Calls	No Calls	Extension Users will not be able to make external calls.
Calls allowed during Night/Break	All Calls	No Calls	Extension Users will not be able to make external calls
More			
Call Duration Control	Apply as per CDC profile 1 (the Apply CDC for calls matching with numbers is blank)	Apply as per CDC profile 1 (the Apply CDC for calls matching with numbers is configured)	CDC will be applied on All Extensions for external calls
Logical Partitioning	CO, Mobile, VoIP	CO, Mobile, SIP, SIP Extension	Users will not be able to make SIP Trunk calls from SIP Extensions. Logical Partition Table will be set to default for all regions respectively.

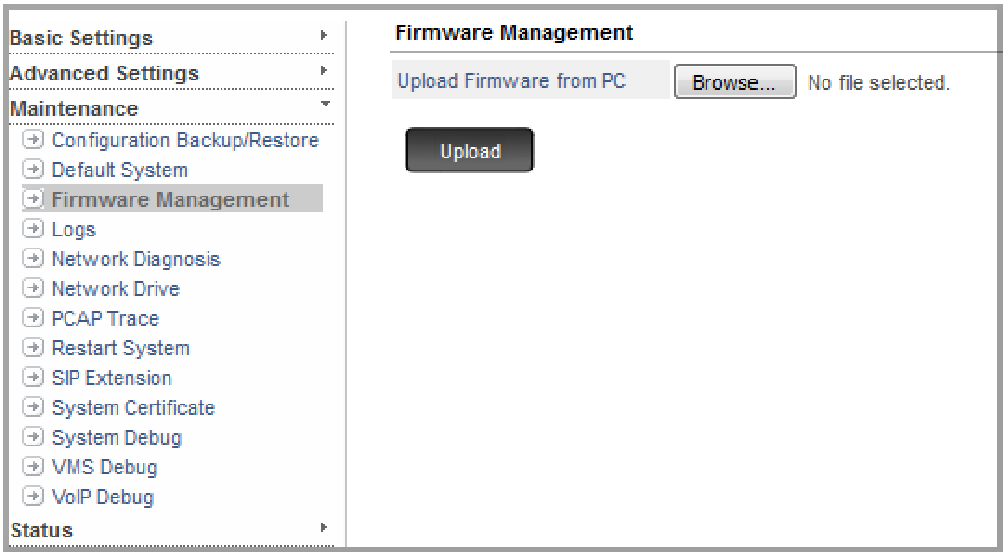
Firmware Management

SARVAM UCS enables you to manage the upgradation of the system firmware from a personal Computer. You can upgrade the system firmware stored on your local computer 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 Version later than V1R6.7" and "Modified default parameter values for Firmwares later than V1R6.7".

How to configure

- Login as System Engineer.
- Under **Maintenance**, click **Firmware Management**.



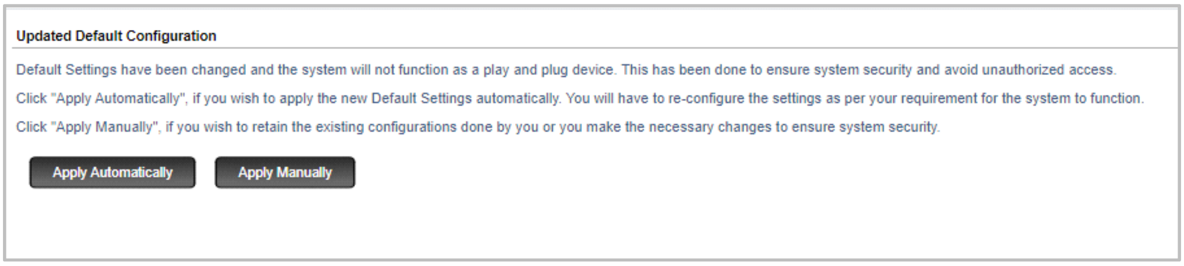
To upgrade the firmware,

- **Upload Firmware from PC**: Click the Browse button to select the firmware file from the local disk on the computer. Make sure that it is a zip file**.**
- After selecting the required firmware file from the PC, click the **Upload** 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.

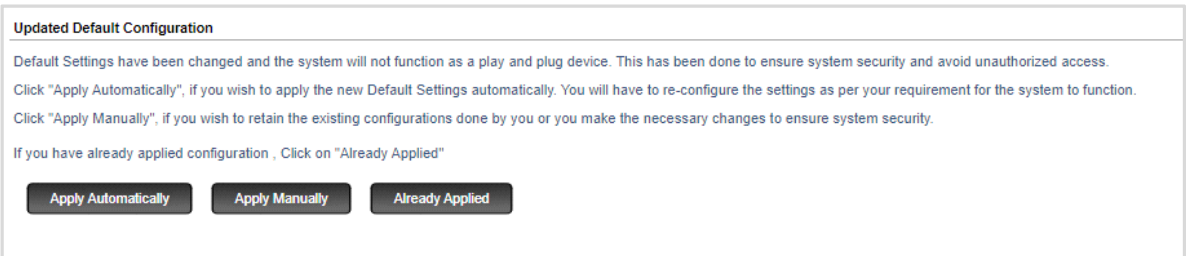
- *If you select a file other than .zip file, an error message is displayed when you click on the Upload button.*
- *For upgradation, make sure the internal USB is connected.*
- *The Firmware upgrade package contains — System Firmware as well as the Extended Firmware.*

After updating Firmware Version later than V1R6.7

- If you select, Restart, the system restarts and the new firmware is uploaded
- 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

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**.

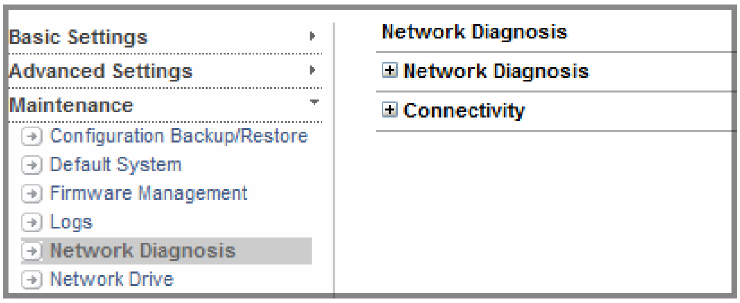
- Click on the desired file to downloaded.

Network Diagnosis

SARVAM UCS provides you an option to check the Internet/Ethernet connectivity using Ping and Traceroute as the diagnostic tools. You can allows check the connectivity between the system and the APNS Server.

How to configure

- Login as System Engineer.
- Under **Maintenance**, click **Network Diagnosis**.



- Click **Network Diagnosis** to expand.
- In **Diagnostic Utility**, select the diagnostic tool — **Ping** or **Traceroute** — to check the Internet/Ethernet 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: 0032 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.

  Similarly, you can also check the connectivity between the system and the APNS/FCM Server. To know more, refer to “APNS Connectivity” and “FCM Connectivity”.

Network Drive

The Network Drive 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.

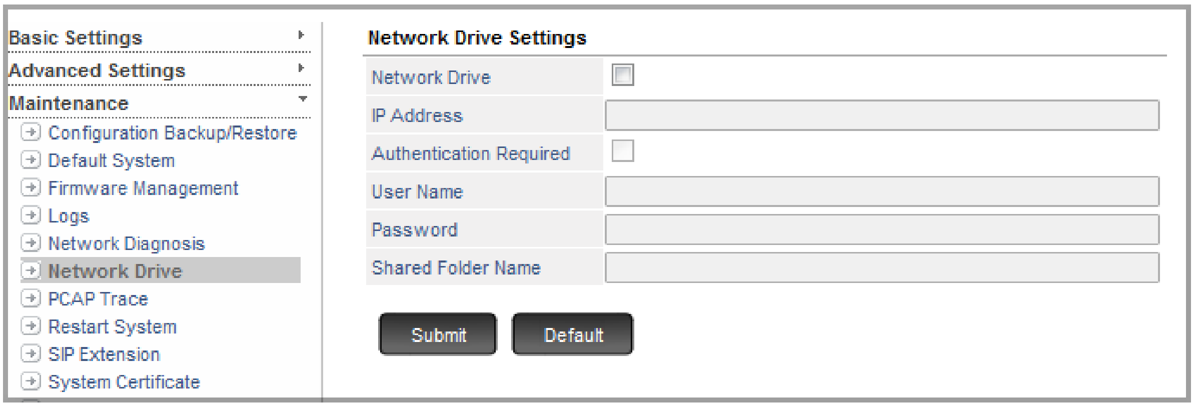
Backup on Network Drive provides you the facility to retrieve valuable information in case internal USB is full.

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

- Login 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

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.

SARVAMUCS supports PCAP Trace for the Ethernet Port 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

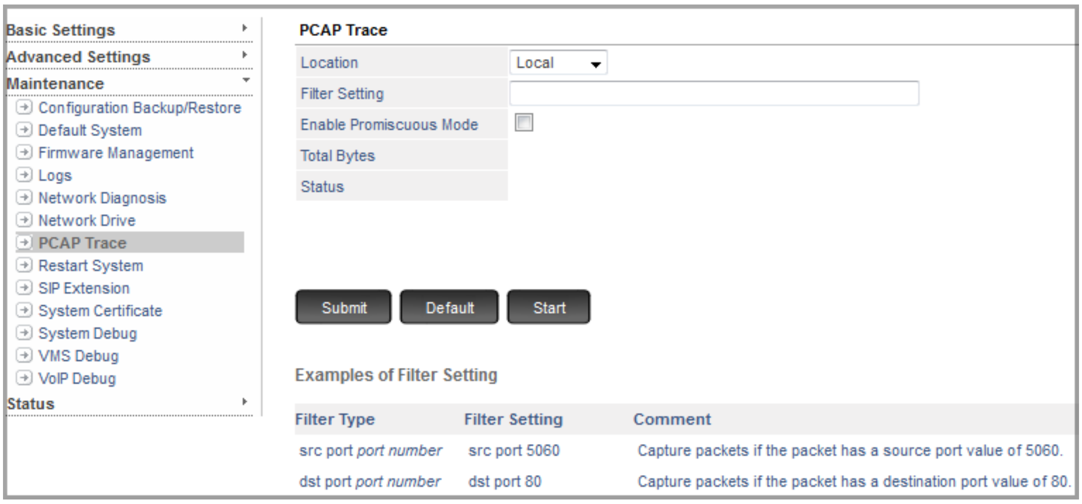
Using PCAP Trace of SARVAM UCS

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,

- Login as System Engineer.

- Under **Maintenance**, click **PCAP Trace**.



- **Location**: Select the location — Local or Remote — on which you want to store the PCAP data.

- **Filter Setting**: Decide the type of packets to be captured and set the Filter accordingly. The Filter 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 to the following examples to know how to set the Filters.

  - To capture packets which are transmitted from the system i.e. from IP address 192.168.1.191:
    - Filter Settings = src 192.168.1.191

  - To capture packets which are received for the system i.e. to IP address 92.168.1.191:
    - Filter Settings = dst 192.168.1.191

  - To capture only packets which are received by the system having 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 i.e. from IP address 192.168.1.191 and port number 161:
    - Filter Settings = src 192.168.1.191 and port 161

  Examples of Filter Setting Syntax:

Filter Type	Filter Setting	Logic
src port number	src port 5060	Capture packets if the packet has a source port value of 5060.
dst port number	dst port 80	Capture packets if the packet has a destination port value of 80.
port number	port 5060	Capture packets if the packet has either source or destination port value of 5060.
src host ip address	src host 192.168.1.176	Capture packets if the source IP address is 192.168.1.176

Filter Type	Filter Setting	Logic
dst host ip address	dst host 192.168.1.176	Capture packets if the destination IP address is 192.168.1.176.
host ip address	host 192.168.1.176	Capture packets if either source or destination IP address is 192.168.1.176

It is not mandatory to set Filters. When the Filter Settings field is left blank, the system will capture all packets.

- **Enable Promiscuous Mode**: Select the Enable Promiscuous Mode check box, if required.

  When you enable Promiscuous mode, SARVAM UCS will capture all network traffic. However, this will work only in a non-switched environment.

  When Promiscuous Mode check box is disabled, the system will capture only traffic that is directly related to it. Only traffic to, from or routed through SARVAM will be picked up by the PCAP Trace.

- Click **Submit.**

'Filter Setting' 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.
- **Status:** It displays the Number of Packets and bytes captured as per the filter setting.

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 **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 trace files 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:** By default, WAN is selected.

- **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.

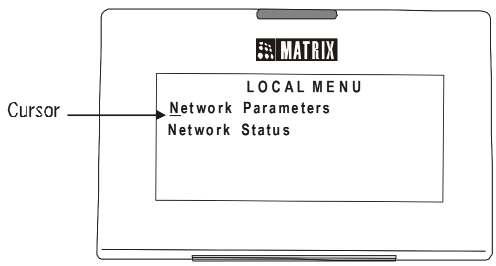
To be able to use PCAP Trace for a Matrix Extended IP Phone, there must be a DSS Key for accessing the Local Menu on the phone. If you have not already configured the DSS Key for Local Menu, you may do so now. For instructions on configuring DSS keys of the Extended IP Phone, see "Configuring SIP Extension Settings as per the Device Type" under *SIP Extensions*.

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.

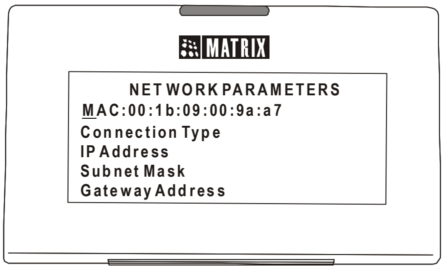
- The Local Menu appears on your phone display.



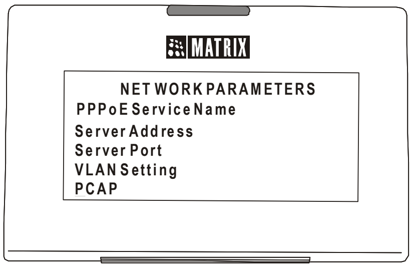
- The cursor appears under 'Network Parameters'.

- Press Enter key to select 'Network Parameters'.

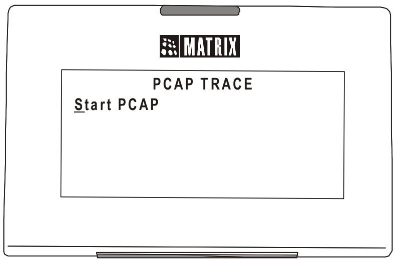
- The Network Parameters submenu appears.



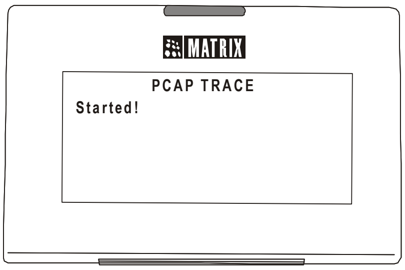
- Scroll with the DOWN navigation key to PCAP.



- Press Enter key.
- ‘Start PCAP’ appears on your phone display.



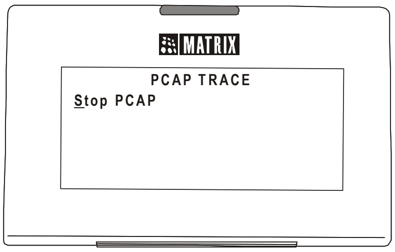
- Press Enter key. You get a confirmation message Started!



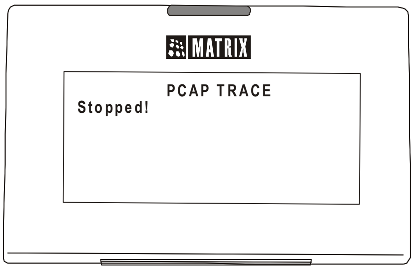
To stop PCAP,

- 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.

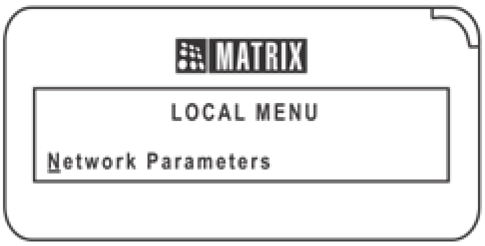
Using PCAP Trace for Matrix SPARSH VP310 Extended IP Phone

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,

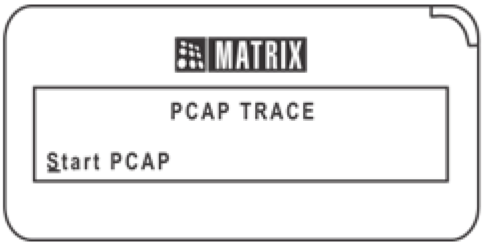
- When the phone is in idle state, press the DOWN key V to access the Network Settings.
- The cursor appears under ‘Network Parameters’.



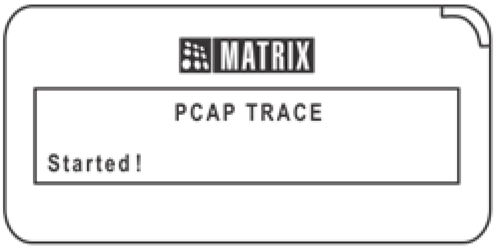
- Press Enter key to select ‘Network Parameters’.
- The Network Parameters submenu appears.

- Scroll with the DOWN navigation key to PCAP.

- Press Enter key.
- ‘Start PCAP’ appears on your phone display.



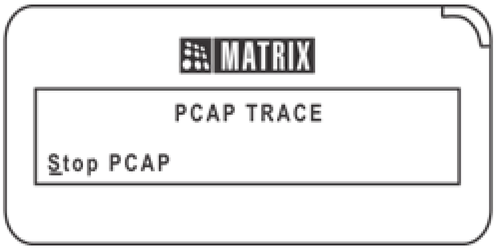
- Press Enter key. You get a confirmation message ‘Started!’



To stop PCAP,

- Press the DOWN key V 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 current User Password set for the phone.
- Click **Log on**.
- On successful login, the FTP window will open. You will see the different folders in this window.
- Click the **ramdisk** folder.
- In the **ramdisk** folder, right click the **trace.pcap** file 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.

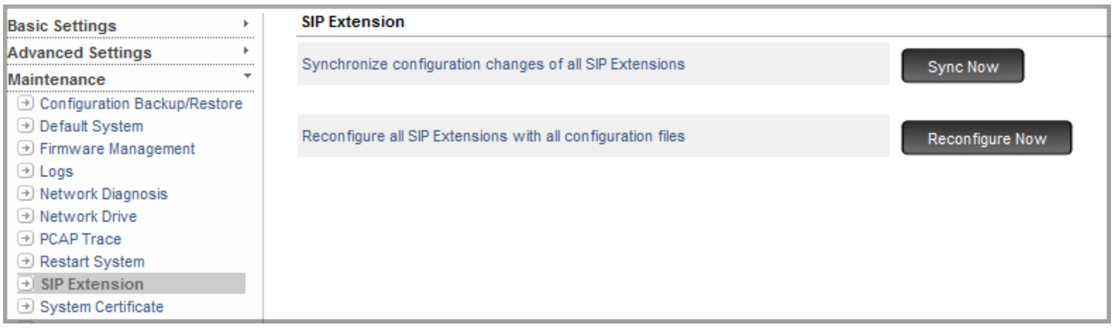
- *You may also use FireFTP, if you are using Mozilla Fire Fox. Make sure your browser has the* ***FireFTP*** *Add-on installed.*
- *To download the Trace file for SPARSH VP330, SPARSH VP210 and SPARSH VP510, refer to their respective User Guides.*

SIP Extension

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

- Login as System Engineer.
- Under **Maintenance,** click **SIP Extension**.



- Click **Sync Now** to synchronize configuration changes of all SIP Extensions, that is, all Extended Clients, SPARSH VP110, SPARSH VP710 and Third Party IP-Phones.

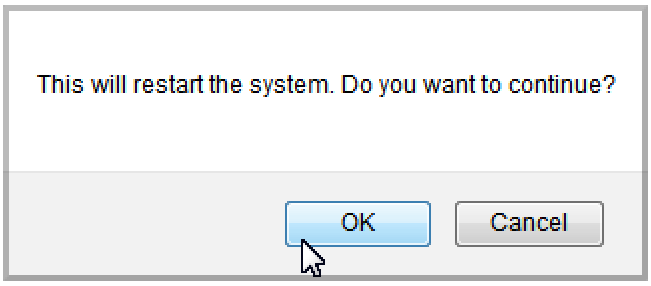
  Only those configuration files in which there are changes will be synchronized.

- Click **Reconfigure Now** to reconfigure all SIP Extensions with all configuration files.

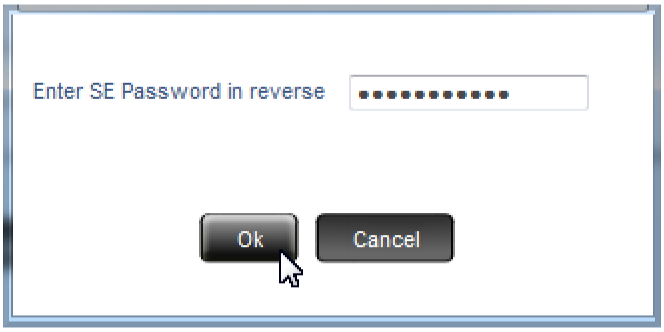
Restart System

To restart the System,

- Login as System Engineer.
- Under **Maintenance**, click **Restart System**.



- You will get an alert message, “This will restart the system. Do you want to continue?”.
- Click **OK**.
- Enter Reverse SE Password.



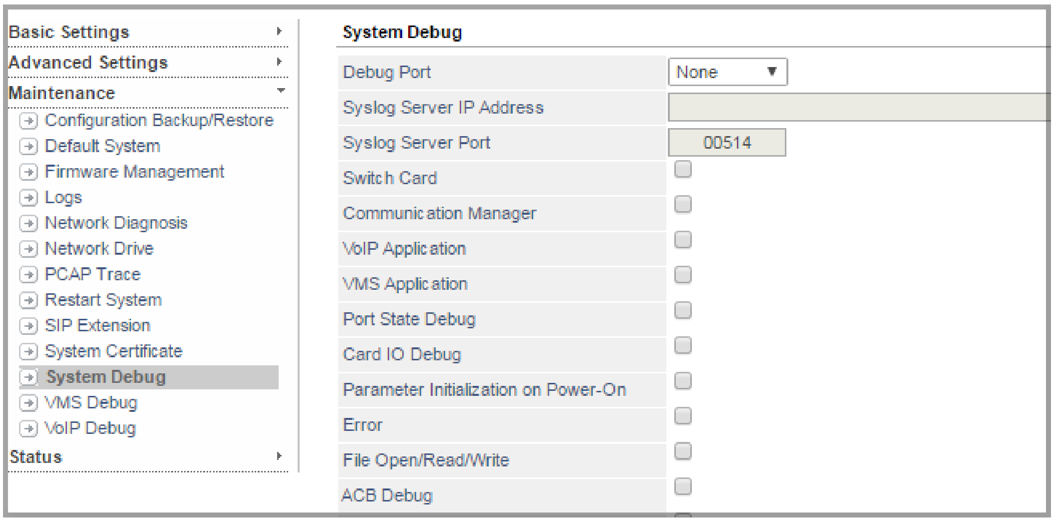
- The SE password you enter must be the current password. For Example: if it is 1234, enter 4321 and click OK.
- The system will restart.

System Debug

You can monitor the state of ports and IO operations for troubleshooting and identifying faults and errors using System Debug. SARVAM UCS supports Syslog Client for debugging. Debug messages are sent to the remote Syslog Server.

How to configure

- Login as System Engineer.
- Under **Maintenance**, click **System Debug**.



Debug Settings

- Select the **Debug Port** to which you want to send the debug. Default: Ethernet.
- In **Syslog Server IP Address and Port**, enter the desired server address. and server port number. Default: port number 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
  - Parameter Initialization on Power-On
  - Error
  - File Open/Read/Write
  - ACB Debug
  - Maturity/AOC Debug
  - Auto-upgradation/RCOC Debug
  - WebJeeves

- SMS Server Main Application
- APNS
- Auto Sign-In
- Xml Feature
- Email Process

ARM Debug

- In **Syslog Server IP Address and Port**, enter the server address and 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 the **Syslog Server IP Address and Port**, enter the server address and 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 Parameters

- 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

- Select the **Apply** check box to enable the port debug.

- Select the **Port Type** you want to debug:
  - SLT
  - CO
  - Mobile
  - SIP Trunk

- In **Start Port**, enter the number of the selected Port Type from which the system should start debugging.

- In **End Port**, enter the number of the selected Port Type after which the system should end debugging.

  If you want to debug a range of ports, configure the starting and ending port numbers in the **Start Port** and **End Port**.

To debug a single port, define the Start and End Port numbers as the same.

- In **System Debug**, select **Enable** to start Port debug.
- Click **Submit.**

I/O Debug

- Select the **Apply** check box to enable the I/O debug.
- Select the **Port Type** for which you want to debug I/O communication:
  - CO-SLT
  - Mobile
- In **Start Port**, enter the number of the selected Port Type from which the system should start debugging.
- In **End Port**, enter the number of the selected Port Type after which the system should end debugging.

  If you want to debug a range of ports, configure the starting and ending port numbers in the **Start Port** and **End Port**.

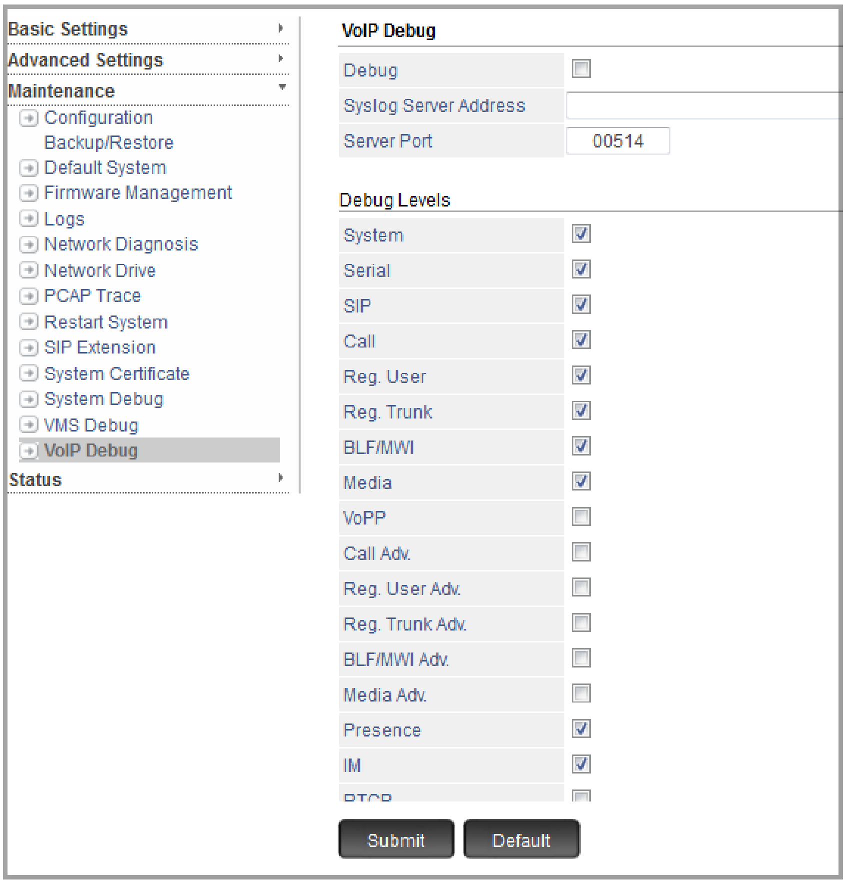
  To debug a single port, define the Start and End Port numbers as the same.
- In **System Debug**, select **Enable** to start I/O debug.
- Click **Submit.**

VoIP Debug

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 SARVAM UCS which supports multiple debug levels.

How to configure

- Login as System Engineer.
- Under **Maintenance**, click **VoIP Debug**.



- Select the **Debug** check box to enable.
  - Configure **Syslog Server Address** and **Server 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 desired **Debug Levels** 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.

CHAPTER 12

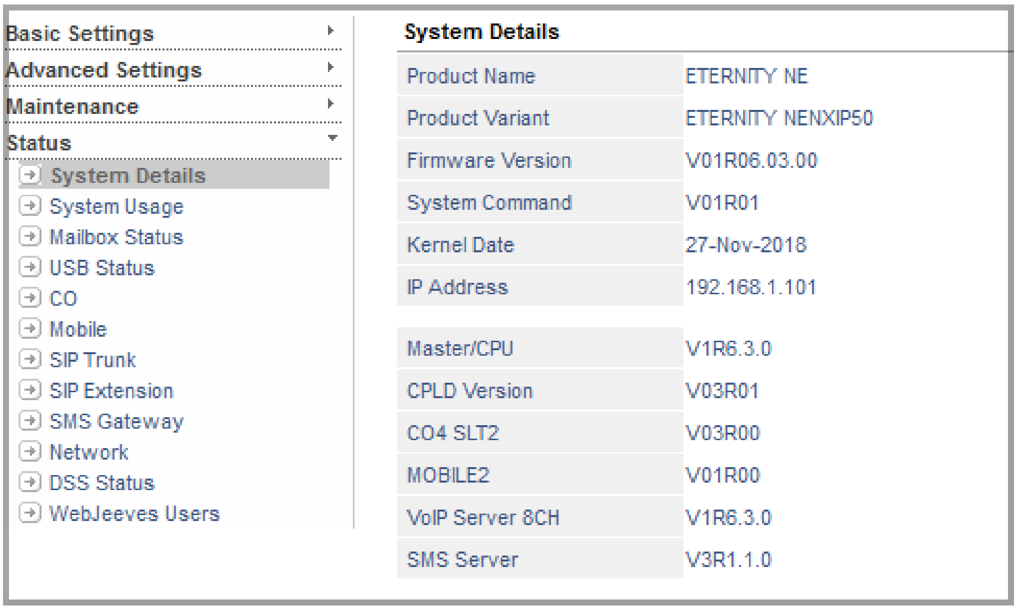
System Status

System Details

System Details displays the current Firmware version-revision and other important details related to the system.

To view the System Details,

- Login as System Engineer.
- Under **Status**, click **System Details**.

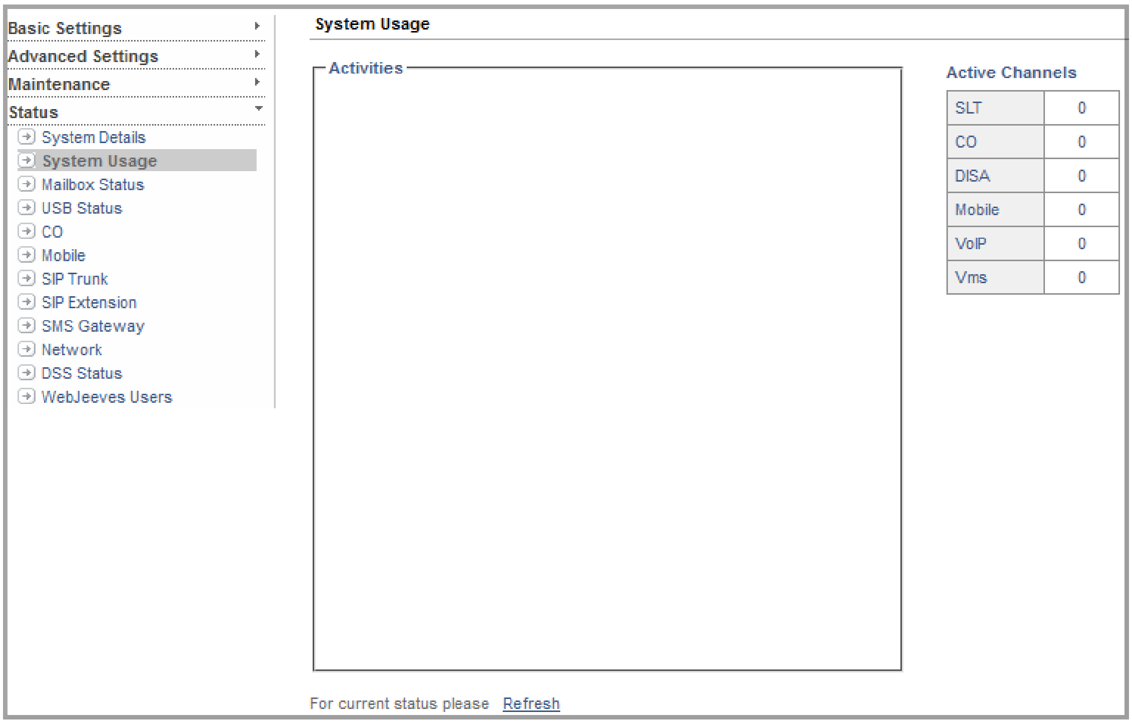


System Usage

System Usage displays the status of the calls along with the ports and trunks involved.

To view the System Usage,

- Login as System Engineer.
- Under **Status**, click **System Usage**.



System Performance

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 WAN port, FLASH 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.

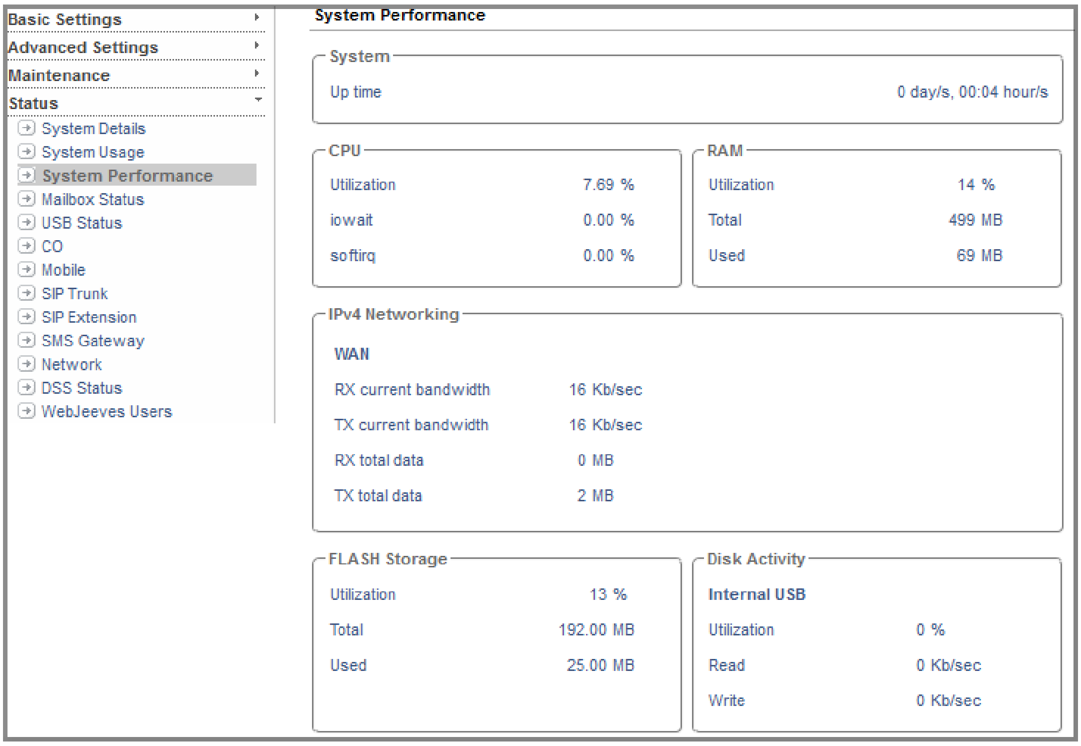
When the CPU and/or Memory (RAM) usage exceeds 80%, it is considered as high usage and the system sends notifications through Email, if configured. These events are also logged in the System Activity Log.

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 **System Performance**.

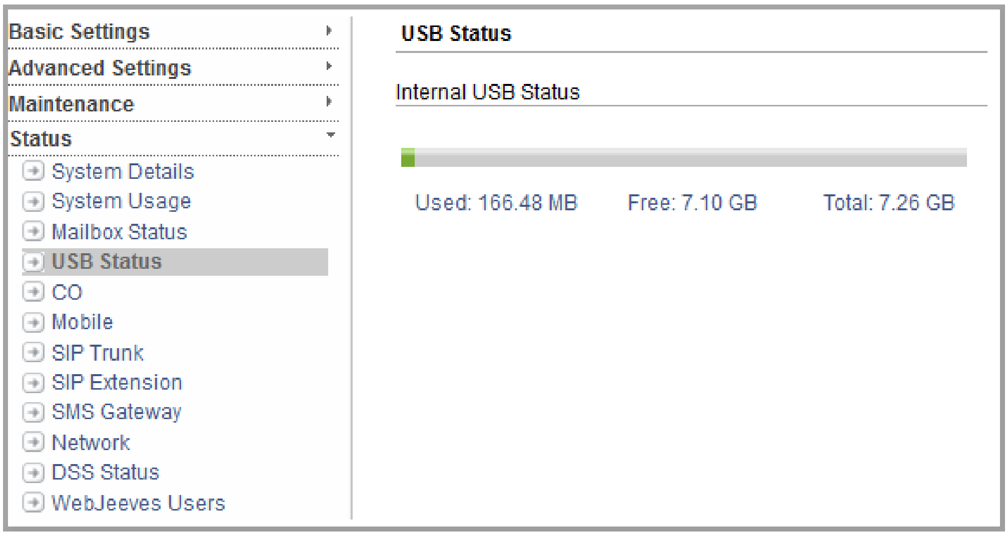


USB Status

The USB Status page displays the status of the internal USB (factory fitted) connected with the system. It also displays the memory status (used, free, total memory space) of the connected USB.

To view the USB Status,

- Login as System Engineer.
- Under **Status**, click **USB Status**.

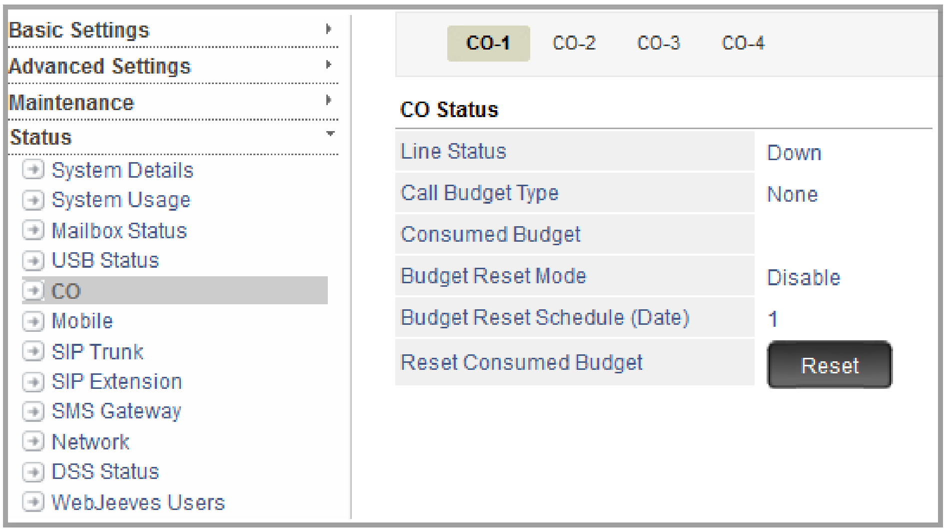


CO Status

The CO Status Page displays the status of the CO trunks connected to the system. It also displays various details related to the CO trunks.

To view CO Trunk Status,

- Login as System Engineer.
- Under **Status**, click **CO**.



For each CO Trunk, the following details will be displayed:

- CO Port Number/ Name
- Line Status
- Call Budget Type
- Allotted Budget 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)

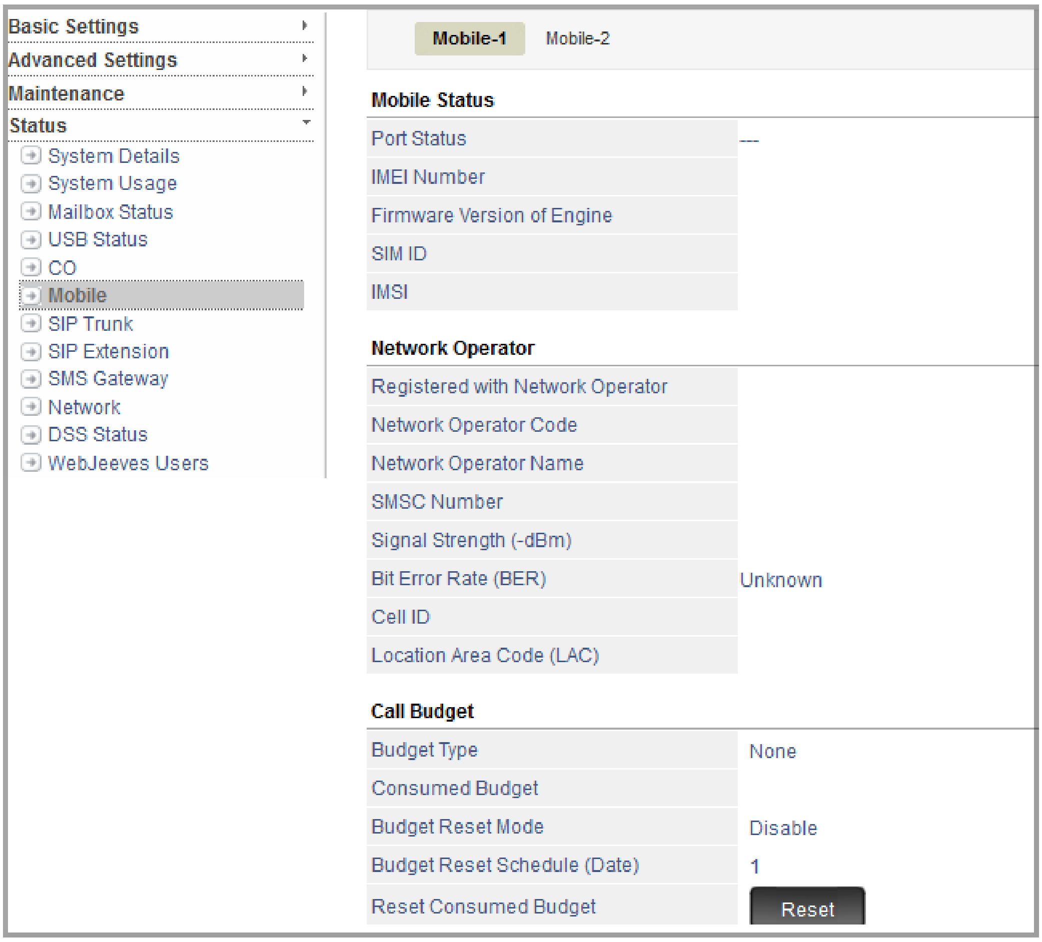
Similarly, to view the status of other CO trunks, click the respective tab.

Mobile Status

The Mobile Status Page displays the status of the Mobile trunks connected to the system. It also displays various details related to the Mobile trunks.

To view Mobile Port Status,

- Login as System Engineer.
- Under **Status**, click **Mobile**.



For each Mobile Trunk, the following details will be displayed:

Mobile Status

- **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 Number:** This is the unique identification number of each Mobile port.
- **Firmware Version of Engine:** This shows the firmware version of the Mobile Engine.

- **SIM ID:** 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:** International Mobile Subscriber Identity (IMSI) is a unique number stored in the SIM Card.

Network Operator

- **Registered with Network Operator:** This shows the type of network with which the Mobile port is registered, whether GSM, GSM Compact, 3G or UMTS, 4G.
- **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.
- **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.
- **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).
- **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).

Call Budget

- **Budget Type:** This shows the Call Budget Type, i.e. whether Amount, Minutes or Number of Calls, set on the Mobile port.
- **Budget Amount:** This shows the sum allotted as Call Budget on the Mobile port.
- **Consumed Budget:** This shows the sum/number of minutes/number of calls of the allotted Call Budget that has been used up on the Mobile port.
- **Budget Reset Mode:** This shows the whether manual or scheduled reset of the consumed call budget is set on the Mobile port.
- **Budget Reset Schedule (Date):** This shows whether the consumed Call Budget on the Mobile port is to be reset Daily or on a particular date of a month.
- **Reset Consumed Budget** (Amount/Minutes/Calls)**:** This editable field allows the System Engineer to reset the consumed Call Budget Amount/Minutes/Calls at any time, manually.

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.

- **Budget Reset Mode**: This shows if you have enabled Scheduled Reset for the SMS budget.

- **SMS Budget Reset Schedule (Date):** This shows the date on which the SMS Budget will be reset.

- **Reset Consumed 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)**.

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 to *"Resetting SMS Budget".*

ASR/ACD

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- **Call Duration:** This is the total call duration of matured outgoing calls[174] 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[175] 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 reset 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 reset 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 reset ASR and ACD when the total number of calls reaches 999. When you reset ASR and ACD the number of call matured and the number of calls dialed is reset to 0.

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174. Matured calls are outgoing calls for which 'CONNECT' message was received from the network.
175. The total number of outgoing calls made includes the number of times the ATD has been sent from the Mobile port to the network.

ASR and ACD can be reset anytime, even when the total number of calls is less than 999.

When ACD is reset, only the 'Total Call Duration' maintained for the ACD calculation will be reset. The 'Total Call Duration' of the Call Budget, i.e. the consumed minutes maintained for the Call Budget on the mobile port will remain unaffected.

- To view the status of other mobile ports, click the respective Mobile port tab.

Resetting SMS Budget

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:

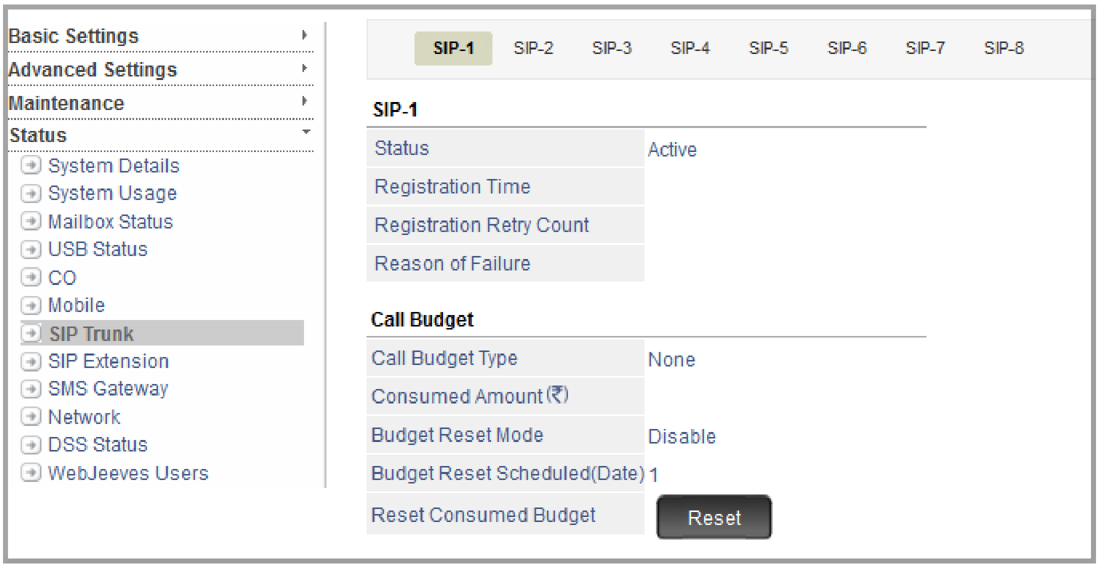
- Login as System Administrator.
- Under **Trunks Properties**, click **Mobile**.
- Scroll to **SMS Budget**.
    - **Consumed SMS (Total):** This shows the number of SMS of the allotted SMS Budget that has been used up on the Mobile port.
    - **Consumed SMS (Daily):** This shows the number of SMS of the allotted Daily SMS Budget that has been used up on the Mobile port.
    - **Budget Reset Mode:** 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 Budget:** Enable this checkbox to reset the consumed SMS Budget, manually. It will reset both **Consumed SMS (Total)** and **Consumed SMS (Daily)**.
- Click **Submit**.

SIP Trunk Status

The SIP Trunk Status Page displays the status of the SIP trunks connected to the system. It also displays various details related to the SIP trunks.

To view SIP Trunk Status,

- Login as System Engineer.
- Under **Status**, click **SIP Trunk**.



- For each SIP trunk (number), the following details will be displayed:
  - Status
  - Registration Time
  - Registration Retry Count
  - Reason for Failure
  - Call Budget Type
  - Allotted Budget Amount/Minutes/Calls
  - Consumed Amount/Minutes/Calls
  - Budget Reset Mode
  - Budget Reset Scheduled (Date)
  - Reset Consumed Budget (this is not a status indicator. It is for resetting the Consumed Call Budget manually)

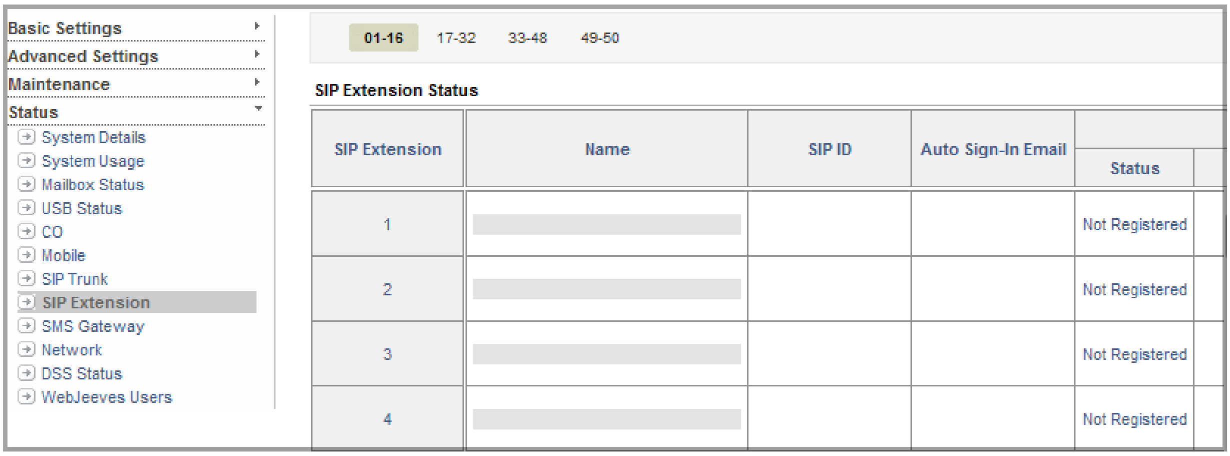
Similarly, to view the status of other SIP trunks, click the respective SIP Trunk tab.

SIP Extension Status

The SIP Extension Status Page displays the status of the SIP Extensions connected to the system. It also displays various details related to the SIP Extensions.

To view SIP Extension Status,

- Login as System Engineer.
- Under **Status**, click **SIP Extension**.



- The SIP Extension Status page will display the following for each SIP Extension,
  - SIP Extension number
  - SIP Extension name
  - 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 to “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.

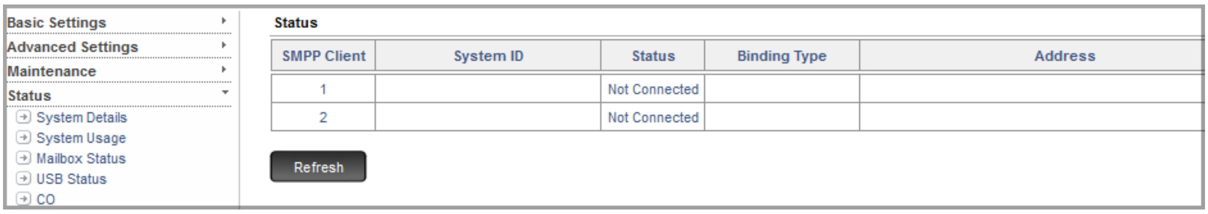
*You can also view the SIP Extension Status from the **Status** link. To view, click SIP Extension under Status.*

SMS Gateway

The SMS Gateway Status Page displays the status of the SMS Gateways.

To view the SMS Gateway Status,

- Login as System Engineer.
- Under **Status**, click **SMS Gateway**.



- The following parameters are displayed for each SMPP Client registered with SARVAM:
  - **System ID:** This is the ID received in the SMPP Client's connection request.
  - **Status:** This 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 displays the type of SMPP Client Binding, that is Transmitter, Receiver or Transceiver.
  - **Address:** This displays the SMPP Client's IP Address and Port.

Network Status

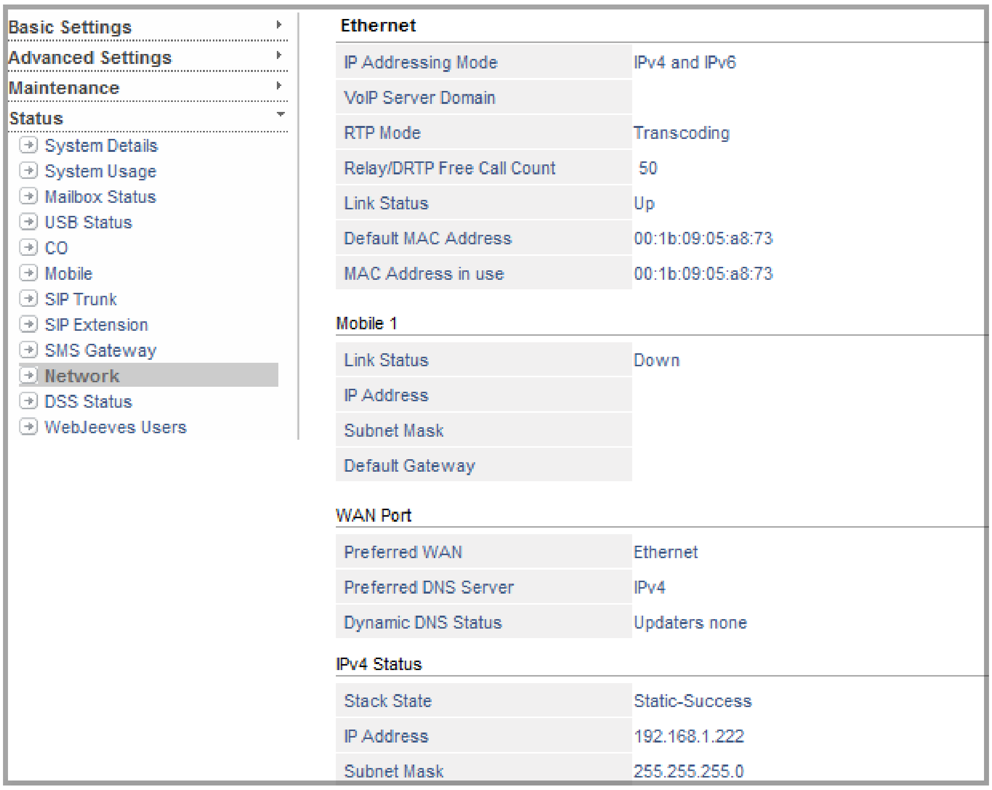
The Network Status Page displays the various network related details such as statuses of Ethernet Port, Mobile Port and WAN Port. It also displays the IPv6 and IPv4 related parameters along with other DHCP related details.

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 need to be re-initialized.

To re-initialize the WAN parameters, click the IPv4 Network Reinitialization or IPv6 Network Reinitialization button under WAN.

To view Network (Ethernet) port Status,

- Login as System Engineer.
- Under **Status**, click **Network**.

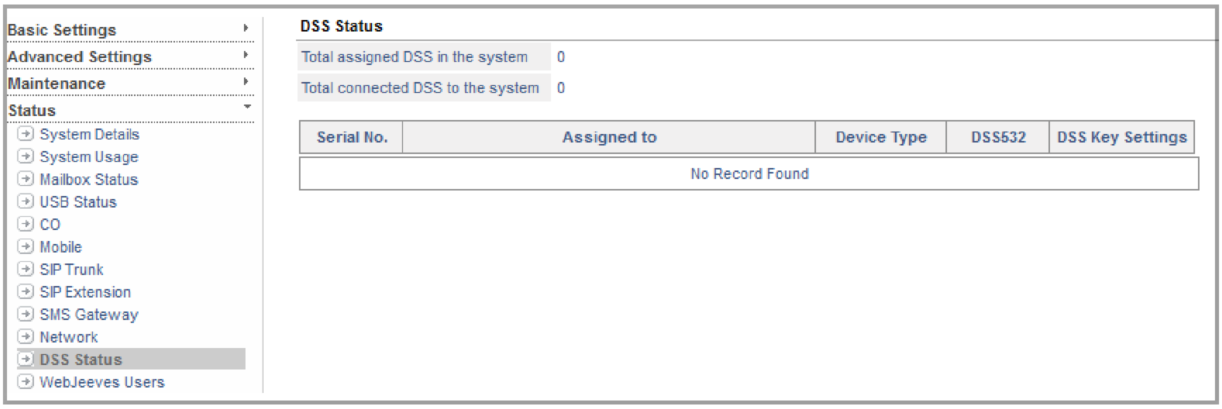


DSS Status

The DSS Status Page displays the status of the various DSS Consoles connected to the system.

To view the status of the DSS Console,

- Login as System Engineer.
- Under **Status**, click **DSS Status**.



The DSS Status Page will display the following,

- **Total assigned DSS in the system**
- **Total connected DSS to the system**

It also displays the following for each connected DSS Console,

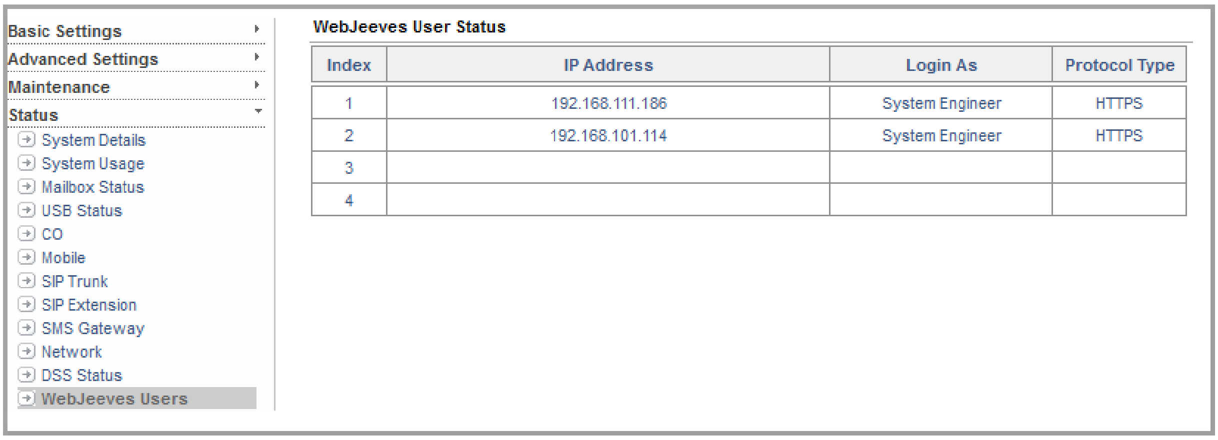
- **Serial Number**
- **Assigned to** displays the extension number to which the DSS Console is connected.
- **Device Type** displays the model of Extended IP Phone —SPARSH VP510— to which the DSS Console is connected.
- **DSS532** displays the status of the console, that is Assigned or Connected.
- **DSS Key Settings** provides the link to **Key Settings**. Click this link, if you wish to configure the DSS Keys.

WebJeeves Users

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 NE

TECHNOLOGY	
Type of Switching	PCM/TDM Digital Switching (100% Non-blocking)
Processor	32-bit RISC

SLT (ANALOG STATION)	
Signaling	Loop Start
Dialing	DTMF and Pulse (10/20PPS)
Off Hook AC Impedance	600/900/Complex
Off Hook Current	40mA Max
Loop Limit	1800 Ohms 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: -10dB to 6dB, Rx Gain:-10dB to 6dB
Ringing	Trapezoidal 60VRMS/25Hz and Sinusoidal 52VRMS/25Hz
REN	3
CLI Presentation	DTMF, FSK ITU-T V.23 and FSK Bellcore 202
Protection	Over Voltage Secondary Protection
Physical Connector	RJ11

CO (2-WIRE TRUNK)	
Signaling	Loop Start
Loop Limit	1200 Ohms
Off Hook AC Impedance	600/900/Complex
Pulse Dialing	10/20PPS
DTMF Dialing and Reception	ITU-T Q.23 & Q.24
Return Loss	>18dB

CO (2-WIRE TRUNK)	
Longitudinal Balance	>50dB
Transmission Level adjust	Tx Gain: -15dB to +10 dB, 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	RJ11

GSM	
Frequency Band (MHz)	Quad-Band: GSM850,EGSM900, DCS1800, PCS1900
Compliant	ETSI GSM Phase 2/2+
SIM Card	One SIM Per GSM Port
SIM Interface	1.8V, 3V
Transmission Power	Class 4 (2W) at GSM850 MHz and EGSM900 MHz Band Class 1 (1W) at DCS 1800 MHz and PCS1900 MHZ Band
RF Sensitivity	Better than -106dBm
Protocol	AT Command Interface
External Antenna	One Antenna per GSM Port, 1.8/3.0*dBi, 50? SMA (Male) Connector, Omni Directional with Cable of 3 Meters Length

UMTS (3G)	
Frequency Band (MHz)	Quad-Band: GSM850,EGSM900, DCS1800, PCS1900 UMTS A Module: Tri-Band: WCDMA 850/1900/2100 UMTS E Module: Tri-Band: WCDMA 900/1900/2100
Compliant	ETSI GSM Phase 2/2+
SIM Card	One SIM Per UMTS (3G) Port
SIM Interface	1.8V, 3V
Transmission Power	Class 4 (2W) at GSM850 MHz and EGSM900 MHz Band, Class 1 (1W) at DCS 1800 MHZ and PCS1900 MHz Band, Class 3 (0.25W) at WCDMA 850/1900/2100 MHz Band
RF Sensitivity	Better than -106dBm at GSM850, EGSM900, DCS1800 and PCS1900, Better than -108 dBm at WCDMA 850, Better than -108 dBm at WCDMA 1900/2100
Protocol	AT Command Interface
External Antenna	One Antenna per UMTS (3G)Port, 1.8/3.0*dBi, 50? SMA (Male) Connector, Omni Directional with Cable of 3 Meters Length

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)

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

VoIP	
VoIP Protocols	SIP v2, SIP over TCP, Symmetric RTP, RTCP, 100rel/PRACK
Network Protocol	IPv4,TCP,UDP, STUN,ARP,ICMP,PPP,DNS
SIP	Maximum 8 SIP Accounts Per System, Out Bound Proxy Support, Display Name, User Name, Password, URL, Proxy URL, Register Interval
Line Echo Cancellation	G.168 With 32/64/128ms Tail Length
Voice	Dynamic Jitter Buffer (Adaptive), Comfort Noise Generation and Voice Activity Detection
NAT	STUN and NAT Keep Alive
Voice CODECs	G.711 (A-law, µ-Law), G.723, G.729AB, GSM-FR and iLBC
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
Data Network	Ethernet (RJ45) Port, Auto MDIX (10/100 base T)

LED Indications	
Power	1 LED Single Color (GREEN)
System Status	1 LED Single Color (GREEN)

Power Supply	
Input	External Adapter - 24VDC, 2.5A
Power Consumption (Typical)	15W

Mechanical	
Dimension (WxHxD)	320.2 x 51.1 x 208.1 mm
Installation	Wall Mount, Table-Top, 19" Rack Mount
Unit Weight	1.23 kg
Shipping Weight	2.4 kg

Environment	
Operating Temperature	0°C to 45°C
Operating Humidity	5-95% RH, Non-Condensing
Storage Temperature	-20°C to 70°C
Storage Humidity	0-95% RH, Non-Condensing

Technical Specifications and Packing List of Extended IP Phones

SPARSH VP248

Packing List (Extended IP Phone/Open SIP Phone)

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.	Quick Installation Guide (printed copy)	1

Technical Specifications (Extended IP Phone)

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.

Technical Specifications (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

SPARSH VP310

Packing List (Extended IP Phone)

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.	Quick Installation Guide (printed copy)	1

Technical Specifications (Extended IP Phone)

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 VP510

Packing List (Extended IP Phone)

Sr. No.	Item	Quantity
1.	Phone body, Handset and Spring Cord	1
2.	Ethernet Cable	1
3.	Foot Stand	1
4.	Wall Mounting Template	1
5.	Quick Installation Guide (printed copy)	1

Technical Specifications (Extended IP Phone)

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
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

DSS532

Packing List

Sr. No.	Item	Quantity
1.	DSS532 Console Unit	1
2.	DSS Extender	1
3.	Foot Stand	1
4.	RJ11 Cable	1
5	Clamps (2 DSS-Phone Clamps and 2 DSS-DSS Clamps)	4
6.	DSS532 Quick Installation Guide (printed copy)	1

Technical Specifications

Terminals Supported	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

SPARSH VP330

Packing List (Extended IP Phone)

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.	Quick Installation Guide (printed copy)	1

Technical Specifications (Extended IP Phone)

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

SPARSH VP210 (Extended/Standard)

Sr. No.	Item	Quantity
1.	Phone, Handset and Spring Cord	1
2.	Ethernet Cable	1
3.	Foot Stand	1

SPARSH VP210 (Extended IP Phone)

LCD Display	128 x 64 Graphical LCD
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), 10/100/1000 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 IP Phone)

LCD Display	128 x 64 Graphical LCD
VoIP	
VoIP Protocols	SIP v2, SDP, RTP (RFC 2833), SRTP
Network Protocol	IPv4, 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.

SPARSH VP710

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

System Resources

SPECIFICATIONS	ETERNITY NE312	ETERNITY NE416	ETERNITY NE402
CO Lines	3	4	4
Single Line Telephones	12	16	2
VoIP Trunks*	8	8	8
SIP Extensions#	16	16	
GSM/UMTS(3G) Trunks*	2	2	2
Voice Mail System*	4 Channels	4 Channels	
Voice Modules (for Auto-attendant, Voice Tones, Voice Help)	16 Modules of 14 seconds each	16 Modules of 14 seconds each	
Ethernet (RJ45) Port	Auto MDIX (10/100 Base-T)		

** Optional and Field Pluggable Module*
 # The VoIP Module comes with built-in 8 IP Users, an additional of 8 IP users can be availed with IP8 Users License

ETERNITY Platforms

Product Name	Description
ETERNITY NE312	3 CO Ports, 2 GSM/UMTS(3G) Ports, 12 SLTs and up to 16 SIP Extensions
ETERNITY NE416	4 CO Ports, 2 GSM/UMTS(3G) Ports, 16 SLTs and up to 16 SIP Extensions
ETERNITY NE402	4 CO Ports, 2 GSM/UMTS(3G) Ports, 2 SLTs and up to 50 SIP Extensions

ETERNITY NE Optional Modules

ETERNITY NE GSM	Module to insert 1 SIM card for Voice Telephony over GSM Networks
ETERNITY NE UMTS A	Module to insert 1 SIM card for Voice Telephony over UMTS (3G) Networks

ETERNITY NE UMTS E	Module to insert 1 SIM card for Voice Telephony over UMTS (3G) Networks
ETERNITY NE IP SERVER	Module to provide VoIP telephony with 4 SIP Trunk Connectivity
ETERNITY NE VMS (License)	4 Channel Voice Mail to attend 4 Simultaneous Calls and Voice Mailboxes for Individual Extensions

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 or download the documentation of the following products by scanning the QR code printed on the Product Label/Packaging Label of the respective product.

ETERNITY NENXIP50

Sr. No.	Item	Quantity
1.	ETERNITY NENXIP50	1
2.	Power Adapter	1
3.	8 GB USB 2.0 Pen-Drive (Flash Drive)	1
4.	Power Cord as per country standard	1
5.	Wall Mounting Template	1
6.	Self tapping screws and screw grips for wall mounting	2
7.	Eternity NE VS (VoIP Server Module)	1

ETERNITY NENX312

Sr. No.	Item	Quantity
1.	ETERNITY NENX312	1
2.	Power Adapter	1
3.	8 GB USB 2.0 Pen-Drive (Flash Drive)	1
4.	Power Cord as per country standard	1
5.	Wall Mounting Template	1
6.	Self tapping screws and screw grips for wall mounting	2

ETERNITY NENX416

Sr. No.	Item	Quantity
1.	ETERNITY NENX416	1
2.	Power Adapter	1
3.	8 GB USB 2.0 Pen-Drive (Flash Drive)	1
4.	Power Cord as per country standard	1
5.	Wall Mounting Template	1
6.	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
5.	Self tapping screws and screw grips for wall mounting	2

SPARSH VP310

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

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

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

DSS532

Sr. No.	Item	Quantity
1.	DSS532 Console Unit	1
2.	DSS Extender	1
3.	Foot Stand	1
4.	RJ11 Cable	1
5.	Clamps (2 DSS-Phone Clamps and 2 DSS-DSS Clamps)	4

SPARSH VP110

Sr. No.	Item	Quantity
1.	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.	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

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	househould	
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 tested on IP Phones of different Brands

Features and Supportive Phones	
Feature	**Phones Supported**
Intercom	GrandStream GXP2020
	GrandStream GXV3140
	GrandStream GXP2120
	Yealink T28P
	Yealink T26P
	Yealink T22P
	Yealink T20P
	Snom 300
	Cisco SPA504G
	SPARSH VP110
	SPARSH VP710
Distinctive Ring	Snom 300
	Yealink T28P
	Yealink T26P
	Yealink T22P
	Yealink T20P
	Polycom VVX1500D
	SPARSH VP110
	SPARSH VP710
Last Caller Recall	Polycom VVX1500D
	Cisco SPA504G
	SPARSH VP110
	SPARSH VP710
Paging	Grandstream GXP2020
	Grandstream GXV3140
	GrandStream GXP2120
	Yealink T28P
	Yealink T26P
	Yealink T22P
	Yealink T20P
	Snom 300
	Cisco SPA504G
	SPARSH VP110
	SPARSH VP710
Conversation Recording	Yealink T28P/T26P/T22P/T20P
	Snom 300
	SPARSH VP110
	SPARSH VP710

Features and Supportive Phones	
Feature	**Phones Supported**
Call Park and Retrieve	Cisco SPA504G
	Polycom WX1500D
	Snom 300
	SPARSH VP110
	SPARSH VP710
Group Call Pickup and Selective Call Pickup	Cisco SPA504G
	Polycom WX1500D
	SPARSH VP110
	SPARSH VP710
SCA and Line Seize	Cisco 504G
	Polvcom WX1500D
	Snom 300
	Yealink T28P/T26P/TT22P/T20P
	Grandstream GXP2120
	SPARSH VP110
	SPARSH VP710
Resume Call Transfer	Yealink T28P and T26P
	SPARSH VP110
	SPARSH VP710
Semi-Attend Transfer	Grandstream GXP2020
	Grandstream GXP2120
	Yealink T28P/T26P/TT22P/T20P
	Polycom WX1500D
	SPARSH VP110
	SPARSH VP710
Busy Lamp Field	Snom 300
	Cisco SPA504G
	Yealink T28P/T26P/TT22P/T20P
	Grandstream GXP2020
	Grandstream GXP2120
	Polycom WX1500D
	SPARSH VP110
	SPARSH VP710
Support of Call Hold	Snom 300 (only when a Key is configured as Extension)
	SPARSH VP110
	SPARSH VP710

Features and Supportive Phones	
Feature	**Supportable Phone**
SARVAM UCS allows these extensions to be monitored using Busy Lamp Field	SLT SIP Extension
SARVAM UCS allows these trunks to be monitored using Busy Lamp Field	SIP CO Mobile
Support of Call Hold Indication in BLF	SLT (Consultation Hold) CO (Consultation Hold, Exclusive Hold, Global Hold) Mobile (Consultation Hold, Exclusive Hold, Global Hold) SIP Trunk (Consultation Hold, Exclusive Hold, Global Hold) SIP (Consultation Hold)

SARVAM UCS Features supported with RTP/Direct RTP

SARVAM UCS features that use DSP Channels

If RTP mode is set as RTP Relay or Direct RTP, system will use DSP channels for the following features:

- Call Taping
- DISA Call
- Voice Mail
- DID
- Trunk Auto Answer

SARVAM UCS features that use DSP 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 DSP 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

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 Extension	***131-Extension/Department Group/VMS***
Call Forward-All Calls to External Extension	***131-Trunk Access Code-Dest. Number-#****
Call Forward-If Busy	***132-Extension/Department Group/VMS***
Call Forward-If Busy-All Calls to External Number	***132-Trunk Access Code-Dest. Number-#****
Call Forward-If No Reply	***133-Extension/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-Extension/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 - When Not Registered

Call Forward-When Not Registered	****13***

Call Forward - Scheduled

Call Forward-Scheduled	***1175-Time Zone***

Call Hold

Put the caller on Hold	***Flash***
Retrieve the caller	***Flash***

Call Park

To Park a Call	***115-Orbit Number***
To Retrieve the Parked Call	***116-Orbit Number***

Call Pick Up

Call Pick Up-General	***4***
Call Pick Up-Selective	***12-Extension***

Call Toggle

Call Toggle (Toggle)	***Flash-1***

Call Transfer

Call Transfer to Extension	***Speech with Extension-Flash-Extension (Transfer Target)-OnHook***
Call Transfer to Extension (External Number)	***Speech with External Number-Flash-Extension (Transfer Target)-OnHook***
Call Transfer to Trunk (External)	***Speech with External Number -Flash-#-TAC-External Number (Transfer Target)-Flash-#***
Call Transfer to Trunk	***Speech with Extension -Flash-TAC-External Number-Go OnHook***
Blind Transfer to Mail Box (VMS)	***Flash-1078-Extension***

Calling Line Identification Restriction

Enable/disable CLIR	***103***

Cancel All Extension Features

Cancel all Extension features	***1051***

Conference 3-Party

Conference 3-Party	***Flash-*3***
Conference-Unsupervised	***Flash-#***

Conference Dial-In

Schedule a Conference	***194-Conference Number-Conference Password***
Initiate a Conference	***195-Conference Number-Conference Password***
To include a Extension in a midway of conference	***191***
Cancel a Conference	***196-Conference Number-Conference Password/ 190***

Conference Multiparty

Conference Multiparty	***Flash-191***
To exclude a Extension from mid of conference	***192***
Terminate Conference	***190***

Conversation Recording

Conversation Recording	***Flash-1095***

Department Call

Department Call	***Department Number (391-395)***

Direct Inward System Access

Enter DISA Mode	***1079-Extension 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***

Flashing on Trunks

Flashing on Trunks	***Flash-****

Follow Me

Set Call Follow Me	***135-Extension-User Password***
Cancel Call Follow Me	***130***

Forced Answer

Forced Answer	***5***

Forced Call Disconnection

Forced Call Disconnection	***#****

General Mailbox

To access General Mailbox	***1176-Follow Voice Mail prompts***

Hot Desking

Set Hot Desk	***1091-Your Extension Number-User Password***

Hotline

Set Hotline	***151-Extension***

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***

Intercom

Intercom	****5-Extension***

Last Caller Recall

Last Caller Recall	***1092***

Last Number Redial

Last Number Redial	***7***

Live Call Supervision

Set Live Call Supervision	***1098-Destination Extension***

Meet Me Paging

Meet Me Paging-Caller	***1093-Page Zone Number***
Meet Me Paging-Called Party	***1093-Extension Number of the Caller***

Message Wait

Message Wait Set/Cancel	***1076-Extension-Code***
Message Wait Retrieval	***1077***

Mute

Mute	***1052***

Operator

Call to Operator	***9***

Paging

Paging	***1074-Page Zone Number***

PIN Dialing

PIN Dialing	***2**

Presence

Publishing Presence by extension user	***104-Password-Index No.***
To view Presence Status	***1097-Extension Number***

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-Extension***

Selective Port Access

Selective Port Access	***69-Port Type-Port Number***

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***

Voice Mail

Voice Mail System	***390***

Walk-In Class Of Service

To Walk-In	***111-1-Your Extension Number-Your User Password***
To Walk-Out	***111-0***

SARVAM UCS Features supported in Terminals

Features	Supported In						
	VP330	VP210	VP248/310/510	Extended VP710	VARTA ADR100	VARTA AMP100	VARTA WIN200
Abbreviated Dialing	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Access Codes	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Account Codes	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Alarms	Yes	Yes	Yes	No	No	No	Yes
Auto Answer	Yes	Yes	Yes	Yes	No	No	No
Auto Call Back	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Auto Redial	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Barge-In	Yes	Yes	Yes	Yes	Yes	Yes	Yes
BLF for Trunks	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Call Chaining	Yes	Yes	Yes	No	No	No	Yes
Call Cost Display	No	No	Yes	No	No	No	No
Call Duration Display	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Call Forward	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Call Forward - Remote	No	No	Yes	No	No	No	No
Call Forward - Scheduled	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Call Forward - Not Registered	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Call Hold	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Call Park	Yes	Yes	Yes	No	No	No	Yes
Call Logs	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Call Pickup	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Call Toggle	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Call Transfer	Yes	Yes	Yes	Yes	Yes	Yes	Yes
CLIP	Yes	Yes	Yes	Yes	Yes	Yes	Yes
CLIR	Yes	Yes	Yes	No	No	No	Yes
Cancel All Station Features	No	No	Yes	No	No	No	No
CUG	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Conference 3-Party	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Conference Multiparty	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Conference Dial In	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Conversation Recording	Yes	Yes	Yes	Yes	Yes	Yes	Yes
COSEC Integration	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Daylight Saving Time (DST)	Yes	Yes	Yes	NA	NA	NA	NA
Department Call	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Distinctive Rings	Yes	Yes	Yes	No	No	No	Yes
Do Not Disturb (DND)	Yes	Yes	Yes	Yes	Yes	Yes	Yes
DSS Call Pick-Up	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Dynamic Lock	Yes	Yes	Yes	No	No	No	Yes
Emergency Conference	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Emergency Dialing	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Extended IP Phone/Mobile Softphone Client - Operation	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Flashing on Trunks (Continued Dialing)	Yes	Yes	Yes	No	No	No	Yes
Follow Me	Yes	Yes	Yes	No	No	No	No

Features	Supported In						
	VP330	VP210	VP248/310/510	Extended VP710	VARTA ADR100	VARTA AMP100	VARTA WIN200
Forced Answer	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Forced Call Disconnection	Yes	Yes	Yes	No	No	No	Yes
Handover (Manual) and Handoff	Yes	Yes	No	Yes	Yes	Yes	Yes
Handover (Automatic)	Yes	Yes	No	No	Yes	Yes	Yes
Hot Desking	No	No	No	No	No	No	No
Hotline	Yes	Yes	Yes	No	No	No	No
Intercom	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Interrupt Request (IR)	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Last Caller Recall	Yes	Yes	Yes	No	No	No	No
Last Number Redial	Yes	Yes	Yes	No	No	No	No
Live Call Supervision	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Macros	No	No	Yes	No	No	No	No
Meet Me Paging	Yes	Yes	Yes	No	No	No	No
Message Wait	Yes	Yes	Yes	No	No	No	Yes
Mute	Yes	Yes	Yes	Yes	Yes	Yes	Yes
OFF-Hook Alert	No	No	No	No	No	No	No
One Touch Transfer	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Paging	Yes	Yes	Yes	Yes	Yes	Yes	Yes
PIN Dialing	Yes	Yes	Yes	No	No	No	No
Presence	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Quick Dial	Yes	Yes	Yes	No	No	No	No
Raid	No	No	Yes	No	No	No	No
Reminder	Yes	Yes	Yes	No	No	No	Yes
Room Monitor	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Selective Port Access	Yes	Yes	Yes	No	No	No	No
Self Ring Test	No	No	Yes	No	No	No	No
System Activity Log Display	No	No	Yes	No	No	No	No
System Fault Log Display	No	No	Yes	No	No	No	No
Time Zone Display	No	No	Yes	No	No	No	No
Trunk Call Waiting	No	No	No	No	No	No	No
Trunk Reservation	Yes	Yes	Yes	No	No	No	Yes
User Absent/Present	Yes	Yes	Yes	Yes	Yes	Yes	Yes
User Password	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Video Call	No	No	No	Yes	Yes	Yes	Yes
Virtual Extension	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Virtual Extension	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Walk-In Class of Service	Yes	Yes	Yes	No	No	No	No
Voice Mail Features	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Others*							
IM	No	No	No	Yes	Yes	Yes	Yes
SMS	No	No	No	Yes	Yes	Yes	Yes
SMS	No	No	No	Yes	Yes	Yes	Yes
BLF for Extensions	Yes	No	No	Yes	Yes	Yes	Yes
Soft Keys	Yes	No	No	Yes	Yes	Yes	Yes
Contact Grouping	No	No	No	No	No	No	Yes
Favorites	Yes	No	No	Yes	Yes	Yes	Yes

* Refer to the respective User Guide for details

Acronyms

ACB	Auto Call Back
ANT	Automatic Number Translation
CAS	Call Accounting Software
CCC	Call Cost Calculation
CCWT	External Call Waiting Tone
CDC	Call Duration Control
CDR	Call Detail Record
CLIP	Calling Line Identification and Presentation
CLIR	Calling Line Identity Restriction
CO	Central Office
CoS	Class of Service
CPD	Call Progress Detection
CPTG	Call Progress Tone (Generation)
CUG	Closed User Group
DDI	Direct Dialing-In
DHCP	Dynamic Host Configuration Protocol
DISA	Direct Inward System Access
DND	Do Not Disturb
DNS	Domain Name System
DST	Daylight Saving Time
DTMF	Dual Tone Multi Frequency
FIFO	First In First Out
FM	Frequency Modulation
FSK	Frequency Shift Keying
FTP	File Transfer Protocol
GSM	Global System for Mobile
ICWT	Internal Call Waiting Tone
IMEI	International Mobile Equipment Identity
IP	Internet Protocol
IR	Interrupt Request
ISP	Internet Service Provider
ITU	International Telecommunication Union
LAN	Local Area Network

LCD	Liquid Crystal Display
LCR	Least Cost Routing
LED	Light Emitting Diode
LIFO	Last in First Out
MAC	Media Access Control Address
MCC	Mobile Country Code
MNC	Mobile Network Code
MOH	Music on Hold
MS	Mobile Station
PBX	Private Branch Exchange
PC	Personal Computer
PIN	Personal Identification Number
POTS	Plain Old Telephone Systems
PPDC	Pre PSTN Digit Count
PS	Power Supply
PSTN	Public Switched Telephone Network
PUK	Personal Unlock Key
RBT	Ring Back Tone
RF	Radio Frequency
RTC	Real Time Clock
SA	System Administrator
SAL	System Activity Log
SE	System Engineer
SFL	System Fault Log
SIP	Session Initiated Protocol
SIM	Subscriber Identity Module
SLT	2 wire Analog Station, Single Line Telephone
SMDR	Station Message Detail Recording
TAC	Trunk Access Code
TCP/IP	Transmission Control Protocol/Internet Protocol
CO	Two Wire Trunks, 2 wire Analog Trunk
UPS	Un-interrupted Power Supply
VMS	Voice Mail System
VMA	Voice Message Application
WAN	Wide Area Network

Basic SE Commands

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 Ethernet 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 Ethernet 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 Ethernet Port	***2150***
To view Subnet Mask of Ethernet Port	***2151***
To view Gateway Address of Ethernet	***2152***
To view Ethernet link Status	***2162***

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:

1. If the product is used other than under normal use and is not properly serviced and maintained by qualified technicians.
2. If the product is not maintained under proper environmental conditions.
3. If the product is subjected to abuse, damage, misuse, neglect, fire, power flow, acts of God, accident.
4. 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.
5. If the product is operated outside the product's specifications or used without designated protections.
6. 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

Federal Communications Commission Statement

Part 15: Class A Information

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 which case the user will be required to correct the interference at his own expense.

TEC Certificate:

सत्यमेव जयते

दूरसंचार विभाग,भारत सरकार

DEPARTMENT OF TELECOMMUNICATIONS, GOVERNMENT OF INDIA

दूरसंचार अभियांत्रिकी केंद्र

TELECOMMUNICATION ENGINEERING CENTRE

CERTIFICATE OF PROVISIONAL MANDATORY CONFORMANCE

No.: 672900034 **Dated:** 2019-10-21

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 NENX416

ESSENTIAL REQUIREMENT NO. :
TEC67291906

SOFTWARE VERSION:
V01R06

HARDWARE VERSION:
V3R4

VALID FROM:
2019-10-21

VALID TO:
2021-10-20

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: 2019.10.21 10:34:19 +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	672900034.1	ETERNITY NENX312
2	672900034.2	ETERNITY NENXIP50

List of Interfaces Tested

1. 2 Wire Trunk
2. Fast 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	Module present in Associated model-2
Model No.							ETERNITY NENX416	ETERNITY NENX312	ETERNITY NENXIP50
Power Supply	NA	V3R4	V01R06			Power supply to ETERNITY NE is applied from Adapter	Y	Y	Y
CPU Card	NA	V3R4	V01R06			CPU Module on base PCB	Y	Y	Y
Analog Trunks (CO)	NA	V3R4	V01R06			Small Office - Home Office PABX with 4 Analog Trunk Line (Co) - ETERNITY NENX416 3 Analog Trunk Line (Co) -	Y (4)	Y (3)	Y (4)

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	Module present in Associated model-2
						ETERNITY NENX312 4 Analog Trunk Line (Co) - ETERNITY NENXIP50			
Analog Extensions (SLT)	NA	V3R4	V01R06			Small Office - Home Office PABX with 16 Analog Extension (SLT) - ETERNITY NENX416 12 Analog Extension (SLT) - ETERNITY NENX312 2 Analog Extension	Y (16)	Y (12)	Y (2)

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	Module present in Associated model-2
						(SLT) - ETERNITY NENXIP50			
ETERNITY NE VS	3000000254	V1R5	V1R6			VoIP Server module	Y	Y	Y

CE Certificates:

Regulatory Information for Terminals

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 [176]

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.

176.Common for SPARSH VP248, SPARSH VP310, SPARSH VP330, SPARSH VP510, 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. Vadodara

CE

MATRIX COMSEC PVT. LTD.
Head Office: 394 GIDC, Makarpura, Vadodara-390 010, India. Ph: +91 265 2630555, Fax: +91 265 2636598, Email: Inquiry@MatrixComSec.com, **www.MatrixComSec.com**
Factory: 39-GIDC, Waghodia-391 760, Dist. Vadodara, India. Ph: +91 2668 262056/57, Fax: +91 2668 262631

SPARSH VP310

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.
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

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. Ganesh Jivani
Director
Date: 28.09.2017

CE

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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

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.
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

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.

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

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- The firmware of this product also includes some of the Open-Source software released under GNU General Public License (GPL) Version 2. Terms of this license is printed in full below.

- The source of the open source software used in this product is available on CD, upon written request from:

  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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```

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How to Apply These Terms to Your New Programs

If you develop a new program, and you want it to be of the greatest possible use to the public, the best way to achieve this is to make it free software which everyone can redistribute and change under these terms.

To do so, attach the following notices to the program. It is safest to attach them to the start of each source file to most effectively convey the exclusion of warranty; and each file should have at least the "copyright" line and a pointer to where the full notice is found.

```
<one line to give the program's name and a brief idea of what it does.>
Copyright (C) <year>  <name of author>

This program is free software; you can redistribute it and/or modify
it under the terms of the GNU General Public License as published by
the Free Software Foundation; either version 2 of the License, or
(at your option) any later version.

This program is distributed in the hope that it will be useful,
but WITHOUT ANY WARRANTY; without even the implied warranty of
```

```
    NERCHANTABILITY or FITNESS FOR A PARTICULAR PURPOSE.  See the
    GNU General Public License for more details.

    You should have received a copy of the GNU General Public License along
    with this program; if not, write to the Free Software Foundation, Inc.,
    51 Franklin Street, Fifth Floor, Boston, MA 02110-1301 USA.
```

Also add information on how to contact you by electronic and paper mail.

If the program is interactive, make it output a short notice like this when it starts in an interactive mode:

```
    Gnomovision version 69, Copyright (C) year name of author
    Gnomovision comes with ABSOLUTELY NO WARRANTY; for details type `show w'.
    This is free software, and you are welcome to redistribute it
    under certain conditions; type `show c' for details.
```

The hypothetical commands `show w' and `show c' should show the appropriate parts of the General Public License. Of course, the commands you use may be called something other than `show w' and `show c'; they could even be mouse-clicks or menu items--whatever suits your program.

You should also get your employer (if you work as a programmer) or your school, if any, to sign a "copyright disclaimer" for the program, if necessary. Here is a sample; alter the names:

```
 Yoyodyne, Inc., hereby disclaims all copyright interest in the program
`Gnomovision' (which makes passes at compilers) written by James Hacker.

  <signature of Ty Coon>, 1 April 1989
  Ty Coon, President of Vice
```

This General Public License does not permit incorporating your program into proprietary programs. If your program is a subroutine library, you may consider it more useful to permit linking proprietary applications with the library. If this is what you want to do, use the GNU Lesser General Public License instead of this License.

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