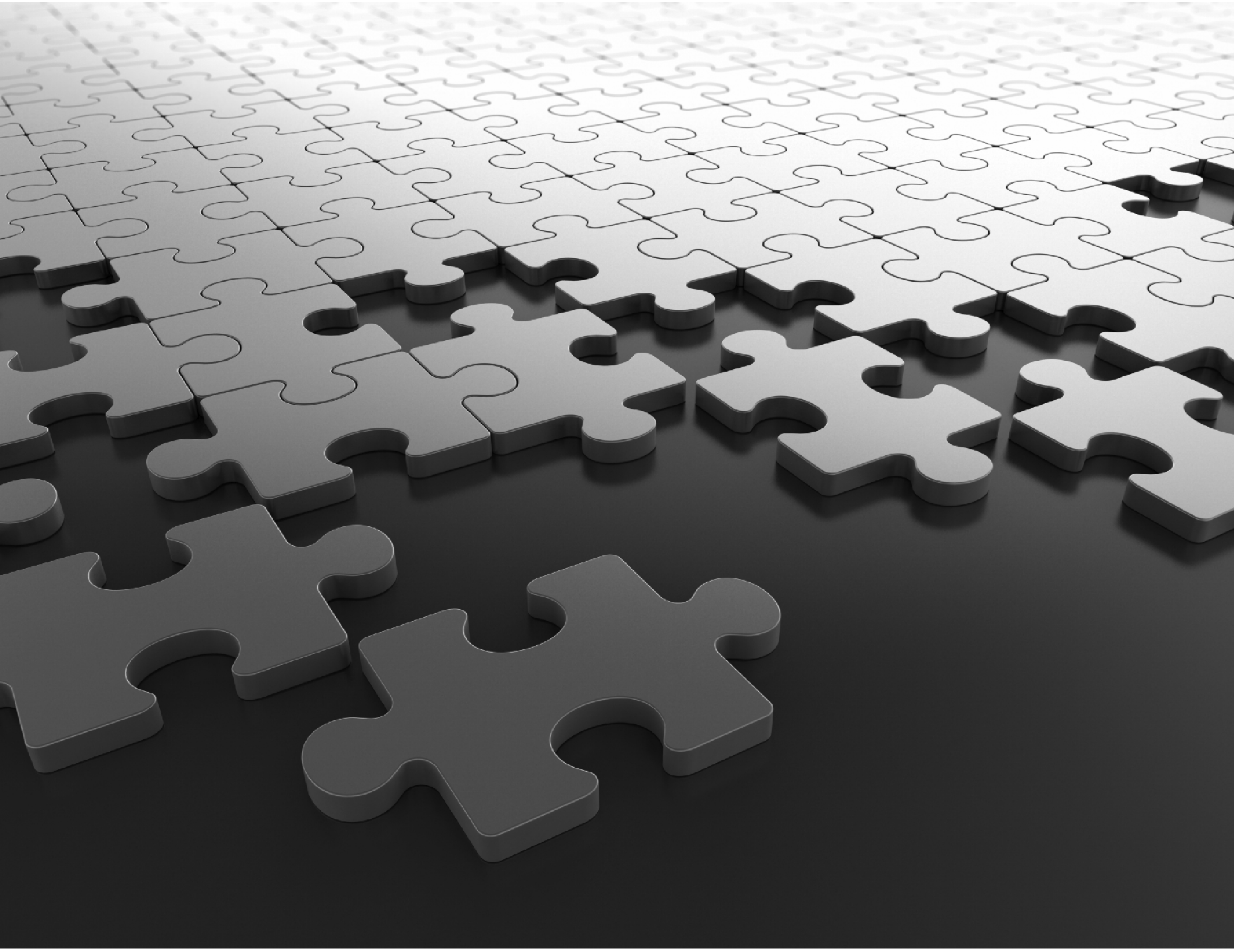
SARVAM UMG
System Manual



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SARVAM UMG

The Universal Media Gateway

System Manual

Documentation Disclaimer

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

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Contents

CHAPTER 1

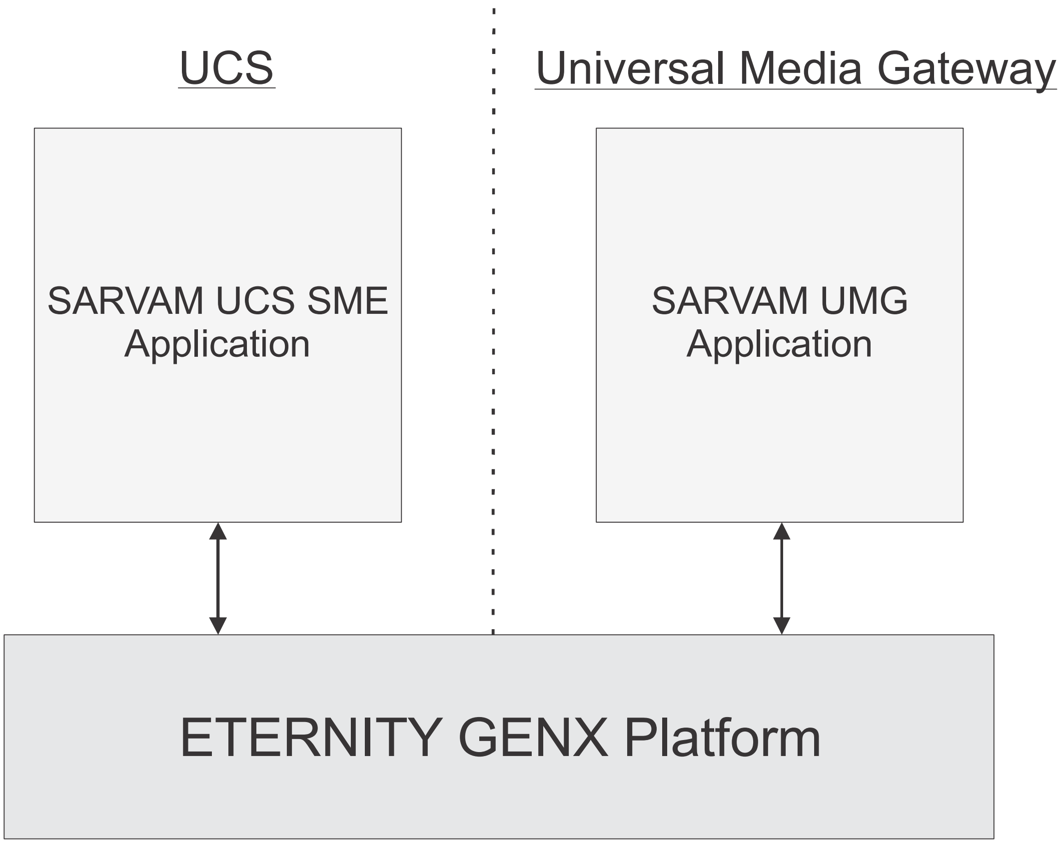
Overview

About the Product

The ETERNITY GENX is the common platform for SARVAM UCS and SARVAM UMG Application. The ETERNITY GENX Platform refers to an entity that includes the entire assembly of cards and the hardware enclosure.

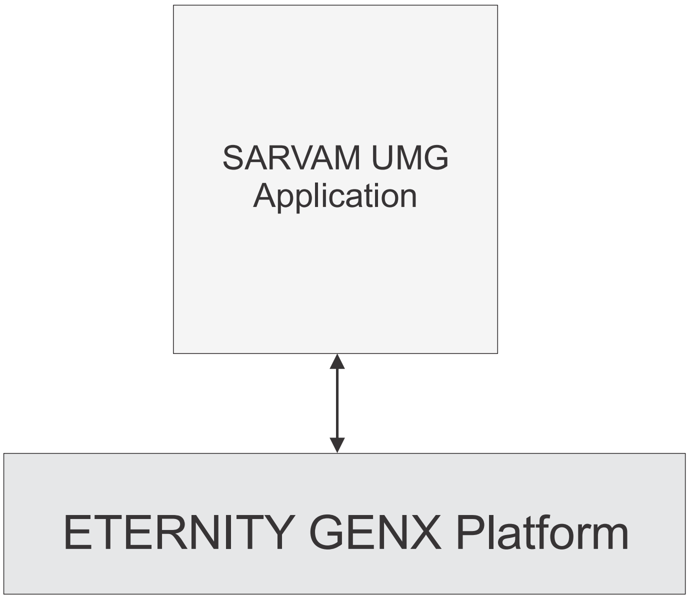
You can use the ETERNITY GENX as the Unified Communication Server or the Universal Media Gateway depending upon the Application License you purchase.

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



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

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



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

CHAPTER 2 **Introduction**

Welcome

Thank you for choosing our product SARVAM UMG! We hope you will make optimum use of this intelligent, feature-packed multi-port — FXO, FXS, Mobile, BRI, T1E1, SIP — Universal Media Gateway. Please read this document carefully before installing your SARVAM UMG.

About this System Manual

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

You may also refer to the SARVAM UMG *Quick Start* for quick installation. To download the Quick Start, scan the QR Code printed on the Product Label/Packaging Label.

For instructions on using the features of the SARVAM UMG refer to the SARVAM UMG *User Card*. The documentation can be found at https://www.matrixtelesol.com/product-manuals.html

For product registration and warranty related details, please visit http://www.matrixcomsec.com/product-registration-form.html

Intended Audience

This System Manual is aimed primarily at **Network and System Engineers**, who will install, configure and maintain the SARVAM UMG.

System Engineers are persons who customize the system configuration to meet the requirements of the organization/users. It is assumed that they have some experience in installing and configuring the multi-port — FXO, FXS, Mobile, BRI, T1E1, SIP — Universal Media Gateway.

Parts of this document containing description of telephony features are aimed at **End Users**, who are the persons/ organizations who will actually use the SARVAM UMG.

Organization of this Document

This System Manual contains the following chapters:

- **Know Your SARVAM UMG** - describes the system and its design, application scenarios, the interfaces, and the hardware.

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

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

- **Configuring SARVAM UMG** - contains description of the different tools and options available to configure SARVAM UMG using the web-based programming tool, Jeeves. It provides detailed description to configure the various extension and trunk port types - FXS, FXO, BRI, Mobile, T1E1 and SIP.

- **Basic Settings:** Provides instructions for configuring the basic parameters of SARVAM UMG, which are sufficient to get the system into operation.

- **Advanced Settings:** Contains instructions for configuring the more advanced features and facilities of SARVAM UMG.

- **Features**: Describes the telephony features of SARVAM UMG and provides instructions for End Users to use these features.

- **Maintenance**: Provides instructions for back-up, generating reports and debugging.

- **Status:** Displays the status of the System, Network, SIP Trunk, Mobile Ports, BRI Ports, T1E1 Ports and FXO Ports.

How to Read this System Manual

This System Manual is organized in such a way that you will find all the information you need quickly and easily.

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.

Conventions used in this System Manual

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

Note: *It indicates something that requires your special attention or it reminds you of something you need to do when you are using the SARVAM UMG.*

Tip: *It indicates a helpful hint giving you an alternative way to operate the SARVAM UMG or carry out a procedure more efficiently.*

Caution: *It indicates an action or condition that is likely to result in malfunction or damage to the SARVAM UMG or your property.*

Warning: *It indicates a hazard or an action that will cause damage to the SARVAM UMG and /or cause bodily harm to the user.*

Terminology used in this System Manual

The word '**Gateway**', **'Universal Gateway'** and **'Universal Media Gateway'** are used interchangeably and synonymously to mean SARVAM UMG.

The word **'System'** refers to the ETERNITY GENX platform when used as the Universal Media Gateway.

Some of the terms specific to this document are defined below:

Term	Usage in the document
System Engineer (SE)	The person who installs, configures and maintains SARVAM UMG.
User	The person who uses SARVAM UMG.
Caller / Calling party	The person who make calls using SARVAM UMG.
Callee / Called party	The person to whom calls are made using SARVAM UMG.
Source / Originating Port	A port from which a call originates.
Destination / Terminating Port	A port on which a call terminates.

Using this System Manual, we hope, you will be able to install, operate and make optimum use of SARVAM UMG. However, if you encounter any technical problems, please contact your dealer/reseller or the Matrix Customer Care.

CHAPTER 3

Know your SARVAM UMG

Introduction

When ETERNITY GENX platform is used with SARVAM UMG application, it acts as a Universal Media Gateway with 12 universal slots. The system supports various Ports — FXS, FXO, BRI, T1E1, Mobile and SIP Trunks.

It is a unique convergence of innovative technology, intelligent software features and an effective solution for accessing internet-based telephone services. It offers a robust SIP Stack which gives a thorough interoperability and usability with various SIP trunk providers and IP-PBX.

The system provides an efficient and unrestricted simultaneous communication (incoming and outgoing); thus offering high performance and high reliability in a compact, modular design.

ETERNITY GENX Platform

The Matrix ETERNITY GENX platform offers you a valuable and flexible approach to run multiple applications on the same platform. The universal slot platform and the modular design of the cards allows you to start with the minimum required configuration and expand the system capacity later, by adding more cards to the universal slots. So, you can invest progressively in scaling up the system as the communication needs of your organization grow.

Depending upon the Application License you purchase, you can use the ETERNITY GENX platform as the Unified Communication Server or as the Universal Media Gateway.

Universal Connectivity

SARVAM UMG offers Universal Connectivity as it provides access to multiple networks — SLT, CO, ISDN BRI, ISDN PRI, Mobile and VoIP — on a single platform.

Interoperability

SARVAM UMG provides you the compatibility to work with other systems without any restricted access or implementation.

Flexibility and Scalability

SARVAM UMG is designed to support both Homogeneous and Hybrid Port cards in its universal slots. The Hybrid cards offer all possible communication interface options to match your communication requirement.

The system is also designed to provide very high level of flexibility and scalability to meet your future communication needs. You can start with the minimum required configuration and expand the system capacity later, by adding more cards to the universal slots as per your organization's requirements.

Cost Saving and Productivity Enhancing Features

Key features like CDR Buffer with a large capacity, that normally warrant additional investment in most other brands, are in-built into this system.

Intelligent feature like CLI based Routing ensure efficient call management and prompt response to callers. It handles the calls on all ports simultaneously, allowing full traffic on all ports.

System Key Features

- Access Codes
- Automatic Number Translation
- Call Progress Tones
- Class of Service
- CLI Based Routing
- Call Detail Record (CDR)
- Region Selection
- Destination Number Determination
- Destination Port Determination
- Digest Authentication on SIP
- DDI Based Routing
- Emergency Number Dialing
- Peer to Peer Calling
- SIM PIN
- System Debug
- Web-based configuration

Also refer "Technical Specifications - ETERNITY GENX" and "Technical Specifications - SARVAM UMG" for the Technical Specifications.

For a complete list of features along with their access codes, see "Features at Glance".

Easy Installation and Operation

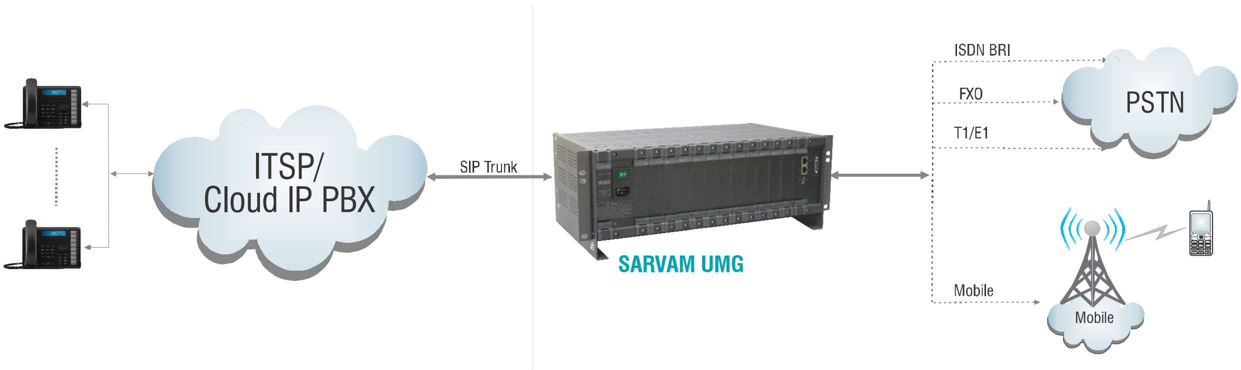
The universal slot architecture of ETERNITY GENX allows easy installation and removal of cards.

The built-in web server, Jeeves allows you to configure the system parameters and features using a Web browser.

Applications of SARVAM UMG

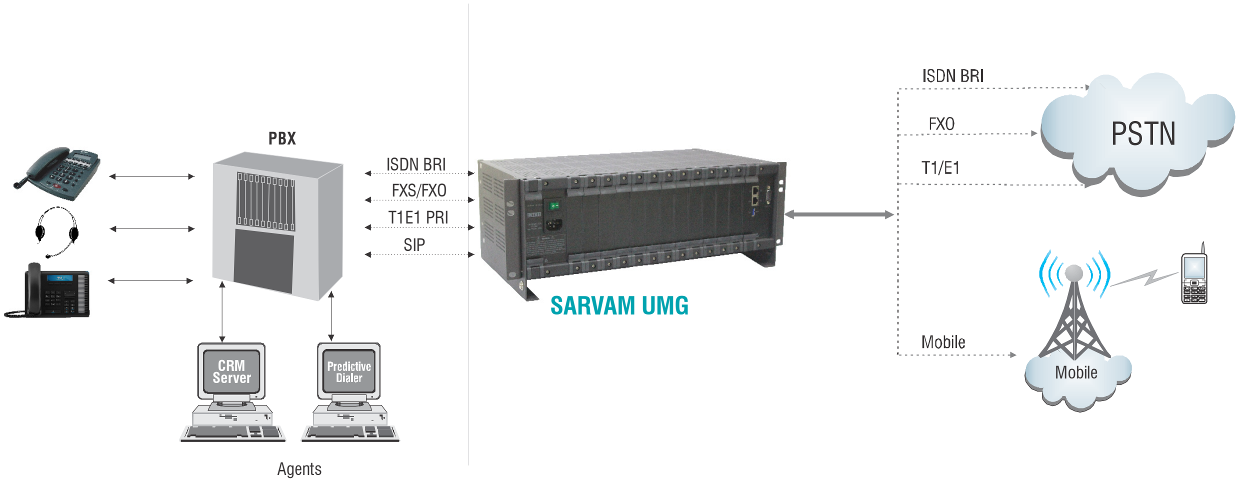
The Matrix SARVAM UMG can be deployed in small to large enterprises and institutions: manufacturing units, corporate offices, banking and financial institutions, software firms, shopping malls, hospitals, hotels-motels, call centers and in institutions.

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

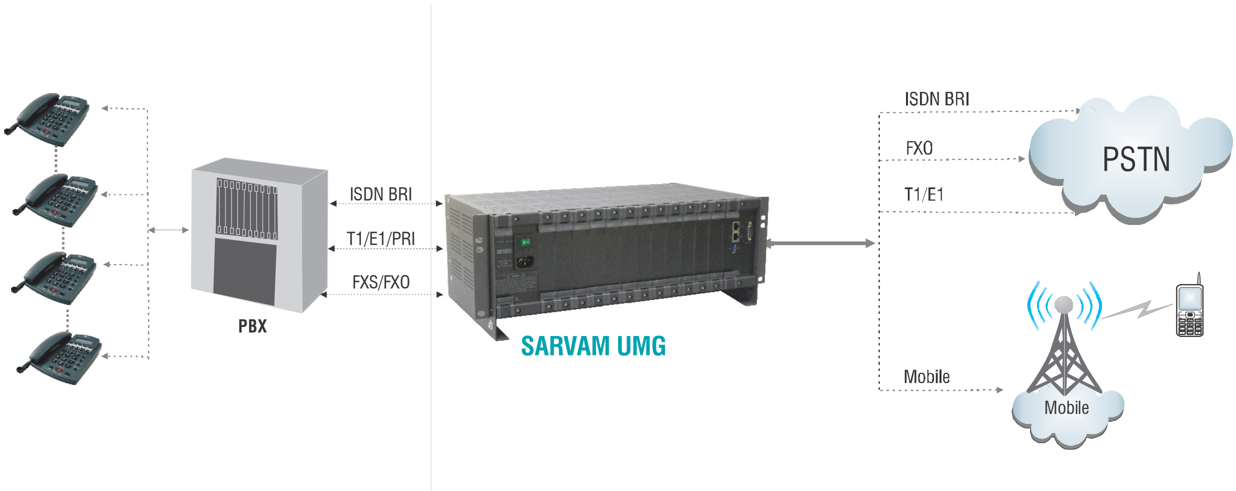


Universal Gateway for ITSP

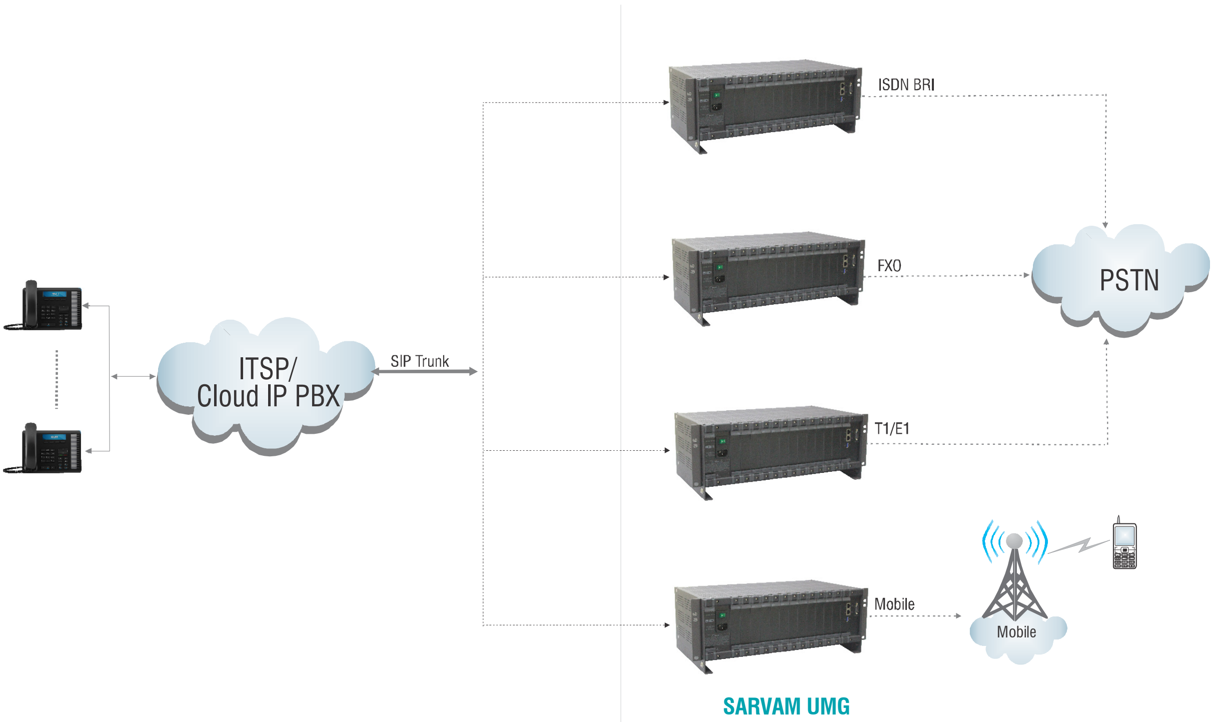
Universal Gateway for IP-PBX



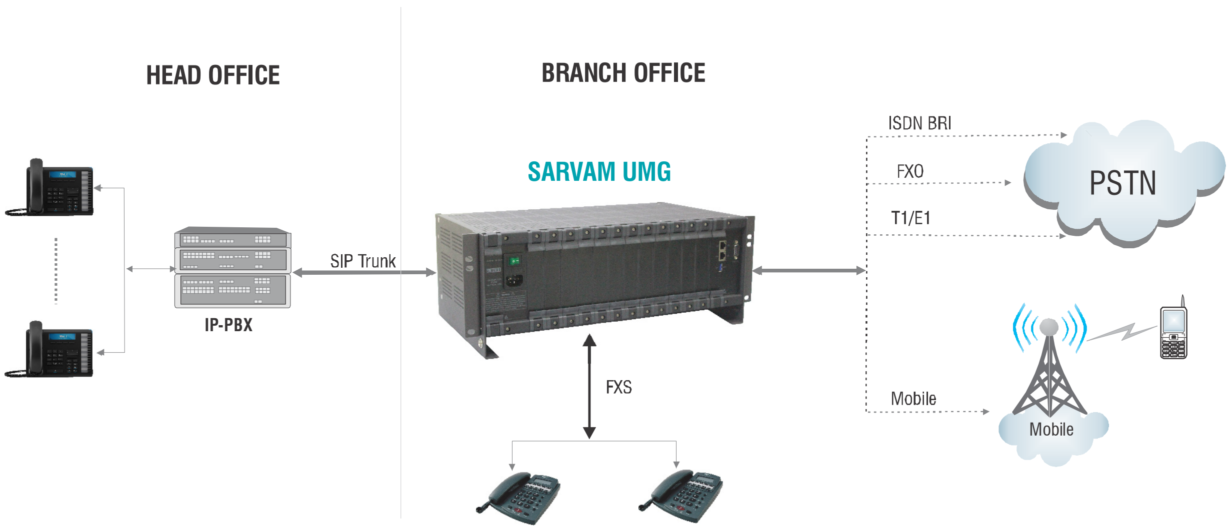
Universal Gateway for Call Centres



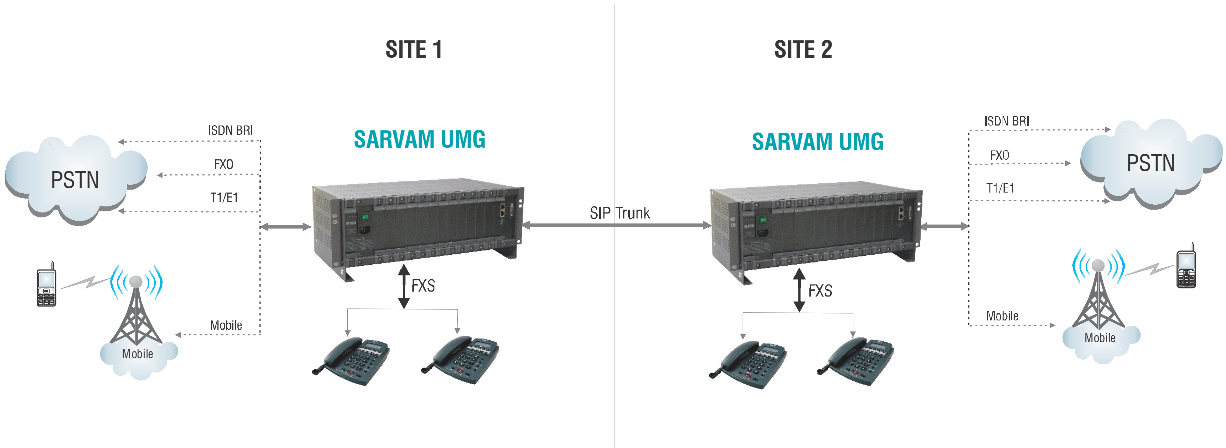
Universal Gateway for TDM PBX



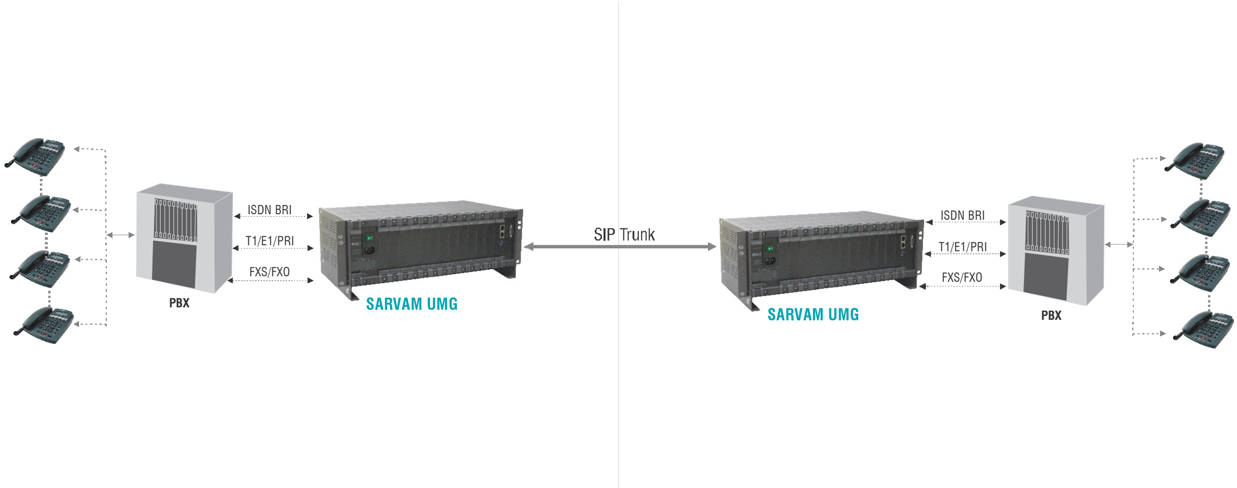
Universal Gateway for Calling Card Applications



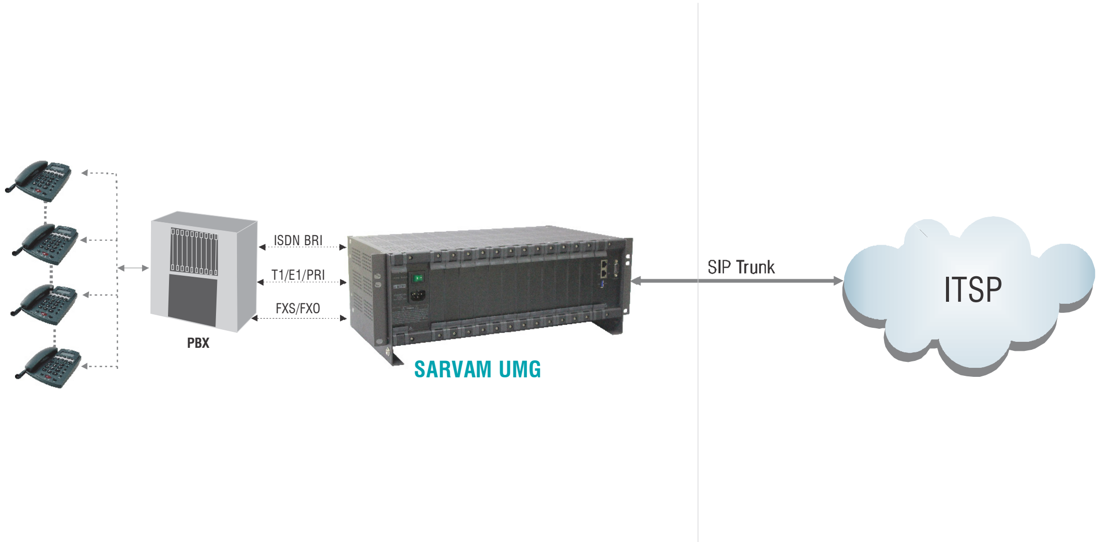
Universal Gateway for Site to Branch Connectivity



Universal Gateway for Site to Site Connectivity



Universal Gateway for TDM PBX Site-to-Site Connectivity

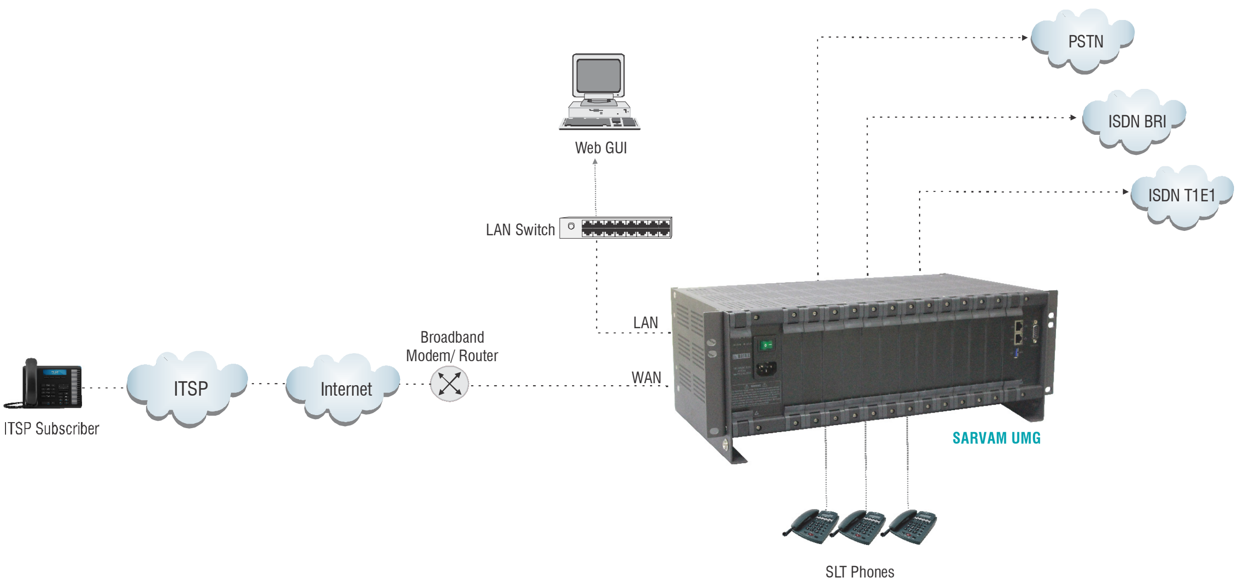


Universal Gateway for TDM PBX SIP Connectivity

SIP Trunk Utility

If connecting to the Public IP Network,

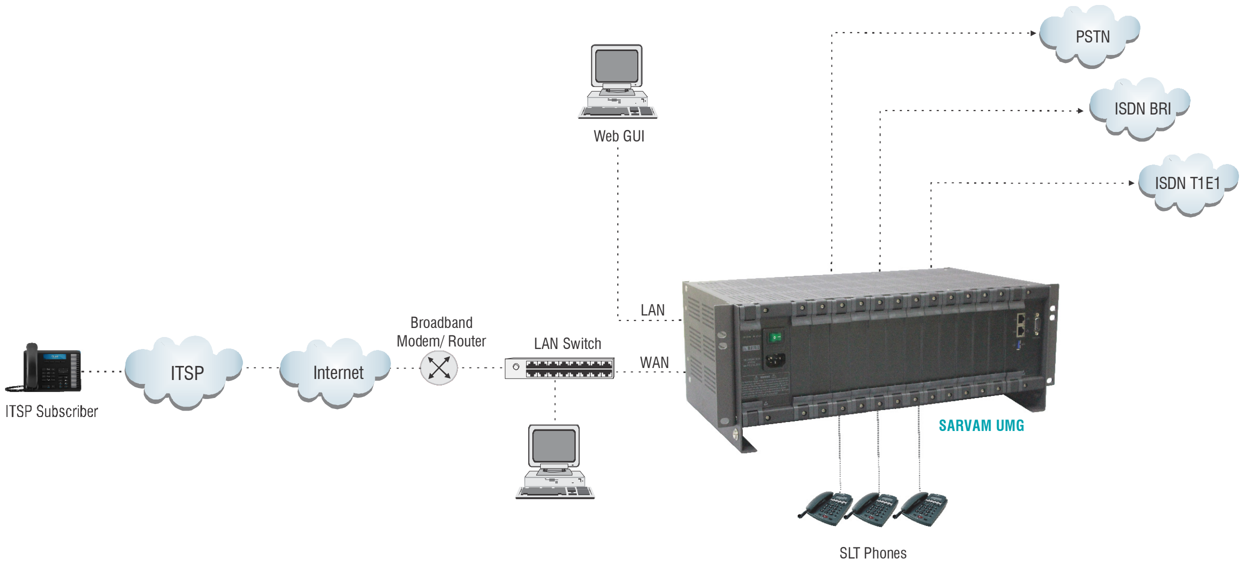
- Plug one end of the RJ45 Ethernet cable into the WAN Port of ETERNITY GENX platform when used with SARVAM UMG application and the other end into the Broadband Router/Modem.



Connecting SARVAM UMG to the Public IP Network

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

- Plug one end of the RJ45 Ethernet cable into the WAN Port of ETERNITY GENX platform when used with SARVAM UMG application and the other end into the LAN Switch/Hub.



Connecting SARVAM UMG to the Private IP Network

Hardware Overview

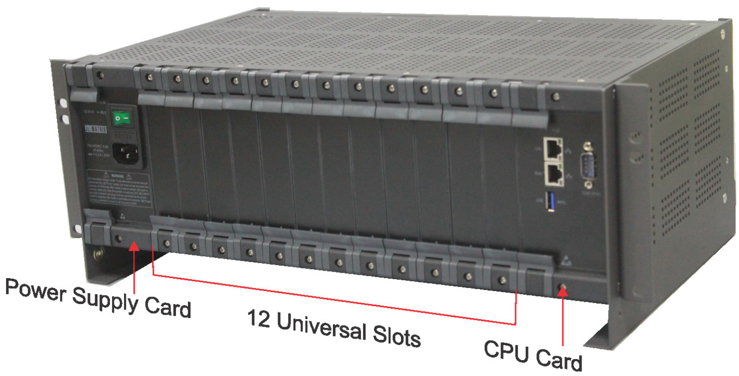
The Enclosure

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

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

Illustrated below is the design of the enclosure and the position of the slots of ETERNITY GENX.

ETERNITY GENX



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

The Cards

The ETERNITY GENX houses the following types of cards:

1. Power Supply Card - PSUNI 250W Card or PS48VDC 250W Card
2. CPU Card
3. SLT Card
4. CO Card
5. CO+SLT Card
6. BRI Card
7. T1E1PRI Card
8. GSM/3G/4G Card

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

The Interfaces

The SARVAM UMG supports the following interfaces for connecting to different telecom networks, standard telephones and other external devices.

The CO Interface

The CO Interface enables the SARVAM UMG to be connected to the POTS Network. The POTS Networks across the world support various standards and differ in features. For example, some networks support Caller ID Presentation using DTMF signaling, while some support Caller ID Presentation using FSK signaling; some networks offer 600 Ω Impedance, while others offer complex impedance.SARVAM UMG's versatile architecture allows it to be connected to such networks differing in their characteristics.

The CO Interface supports following features:

- Programmable AC Impedance - 600Ω, 900Ω and various complex impedances.
- Answer Supervision/Polarity Reversal
- Selectable Disconnect Supervision - Polarity Reversal, Open Loop Disconnect
- Selectable Caller ID Detection - 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 ISDN T1E1PRI Interface

The ISDN T1E1PRI Interface enables SARVAM UMG to be connected to digital networks over T1[1] and E1[2] carrier lines.

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

- PRI
- Robbed Bit Signaling (RBS)

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

- PRI
- Channel Associated Signaling (CAS)

The ISDN T1E1PRI Interface supports the following features:

- Terminal (TE) mode and Network (NT) Mode

1. *T1 PRI (T-Carrier) offers 23 Bearer Channels and one Signaling Channel (23B+D). It is used in North America, Japan and Korea.*
2. *E1 PRI (E-Carrier) offers 30 Bearer Channels and two Signaling Channels (30B+D). It is used in all countries, except North America, Japan and Korea.*

The ISDN BRI Interface

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

The ISDN BRI Interface has the following features:

- Signaling types - 2B+D Signaling
- Terminal (TE) mode
- Network (NT) mode

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

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

The Mobile Interface

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

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

The Mobile Interface supports the following features:

- GSM 2G, 3G and 4G LTE network support.
- Selectable GSM Frequency Bands for 2G - 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).
- Single Rooftop (RT) antenna for 'High Gain'

The SARVAM UMG Mobile Interface does not support GPRS features, Fax and Data services, and network supported services.

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 SARVAM UMG and extensions of other Systems that are networked with the SARVAM UMG.

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

The VoIP Interface supports adaptive jitter buffer for reducing delay.

The key features of the VoIP Interface are:

- Upto 250 SIP Trunks - for Proxy or Peer-to-Peer (non-Proxy).
- 128 Maximum Simultaneous Voice Calls (as per License).
- Selectable Network Assignment (Connection Type) - Static IP, DHCP, PPPoE.
- Selectable DNS - Automatic and Static.
- Dynamic DNS for mobile SIP devices.
- 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.
- Broad Voice Codec Selection -G.729, G.723, GSM FR, iLBC (30ms), iLBC (20ms), GSM EFR, G.711 (u-Law), G.711 (A-Law)
- Quality of Service - RTP DiffServe/ToS
- Voice Mail Subscription for SIP Trunks.

SIP Trunks

The SARVAM UMG application supports a maximum of 250 SIP Trunks, allowing you to subscribe to as many as 250 different Internet Telephony Service Providers (ITSP).

You can connect SARVAM UMG application to the IP network, which may be Public Internet.

The Single Line Telephone Interface

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

The SLT Interface has the following features:

- Selectable Caller ID Presentation - DTMF, FSK
- 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.

Computer

You can connect SARVAM UMG to a standalone computer or to a LAN Switch over the Ethernet Port of the SARVAM UMG.

PC connectivity is required to:

- access the web-based configuration tool Jeeves.
- capture and download Call Detail Records (CDR)
- capture and download System Activity Log, System Fault Log and System DSP Log.

CHAPTER 4

Before You Start

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

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

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

Before you begin to install and set up the ETERNITY GENX Platform, make sure you have the following items:

- SARVAM UMG Application License
- A Main Distribution Frame (MDF)
- A suitable location to install the Main Distribution Frame and the ETERNITY GENX platform.
- Cables for trunk lines and extensions.
- The Cards of ETERNITY GENX.
- One or more Single Line Telephone for testing.
- Power supply.
- One or more active Two-wire trunk lines for test calls.
- A modem for the ISDN T1E1PRI line.
- An NT1 termination device for the ISDN BRI line.
- Appropriate cables and connectors to set up and test the WAN interface of the ETERNITY GENX and the LAN connection.
- A standalone PC or a PC connected in a LAN that can PING the ETERNITY GENX.
- A SIM card to test mobile network connectivity.
- A SIP Account to test VoIP connectivity.

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

The Main Distribution Frame (MDF)

The MDF connects outside telephone lines coming from the local exchange to the CO lines of the ETERNITY GENX.

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

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

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

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

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

Location of the MDF

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

Selecting the Installation Site for ETERNITY GENX

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

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

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

Read "Protecting ETERNITY GENX Platform and Yourself".

Selecting Cables

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

Selecting Extension Telephones

Select appropriate telephone instruments to be connected as extension phones. You may connect any standard telephone instrument like rotary phone, Pulse/tone switchable push-button phone, Feature phone or Cordless phone. So, you can also use your existing telephone instruments.

Providing Power Supply Source

- The ETERNITY GENX work with input voltages 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.

Terminating Trunk and Extension Cables on the MDF

- Terminate the CO Trunk Line cables from the CO (public telephone exchange) into the 'Trunk Lines' side of the MDF using the punch tool for Krone modules.

- Terminate all the extension cables (connected to the wall sockets/outlets) into the 'Station Lines' side of the MDF using the punch tool for Krone modules.

- Label the trunk and extension line cables for easy identification and keep a record of the trunk and extension lines in an updatable format.

  Where multiple wiring cabinets/distribution frames are used, label each frame and reference its number on the corresponding outlet.

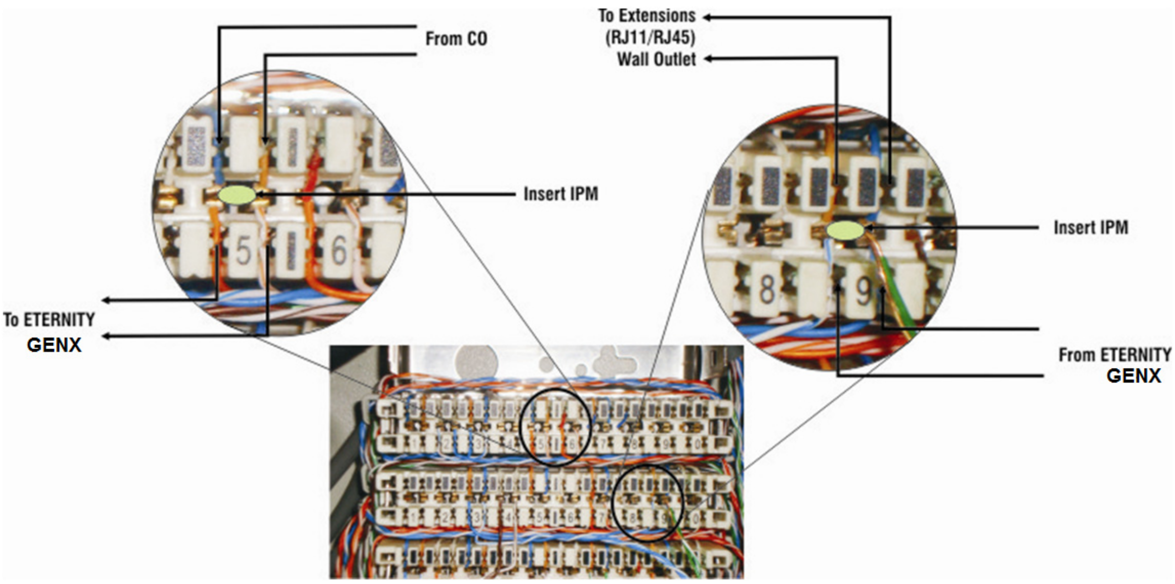
- Install Primary Protection modules with Gas Discharge Tubes (GDT) and fuses on entry points for all trunk lines. This is to protect the system from heavy voltages from trunk lines and overhead stations.

  The product warranty does not cover damages resulting from lack of primary protection on trunk lines.

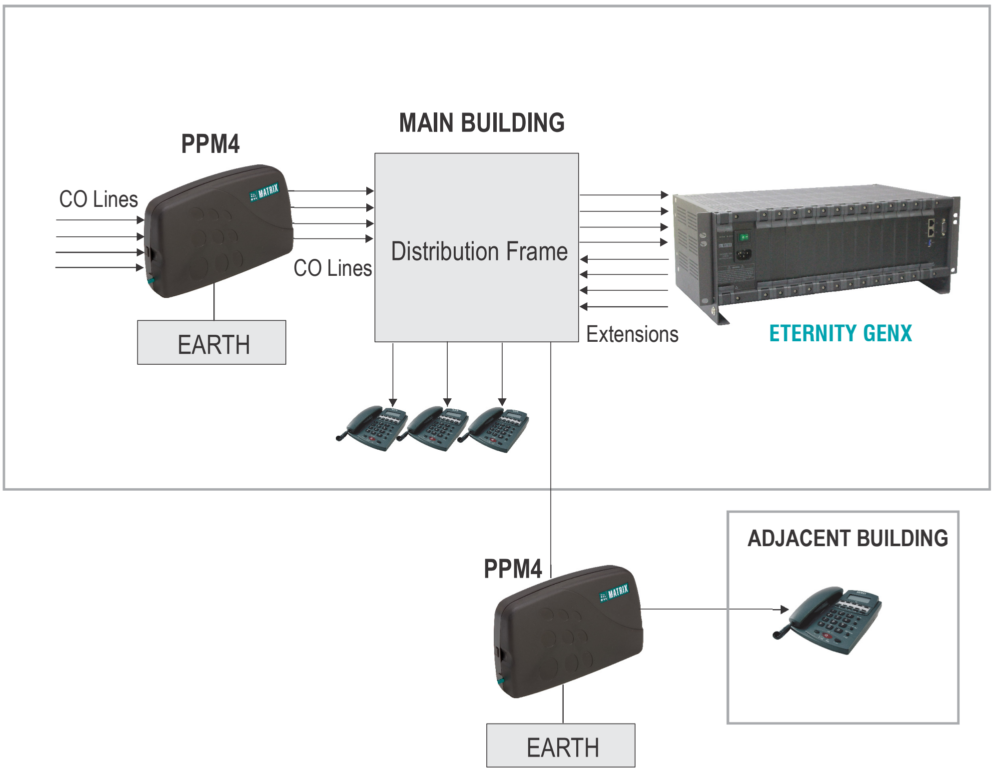
- It is recommended that you also install Primary Protection modules with GDT and fuses on all Extension lines, particularly off-premise extensions.

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

- A typical connection between the ETERNITY GENX and the MDF is illustrated in the figure below.



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



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

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

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

Input Protection Modules (IPM)

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

Primary Protection Module - PPM4

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

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

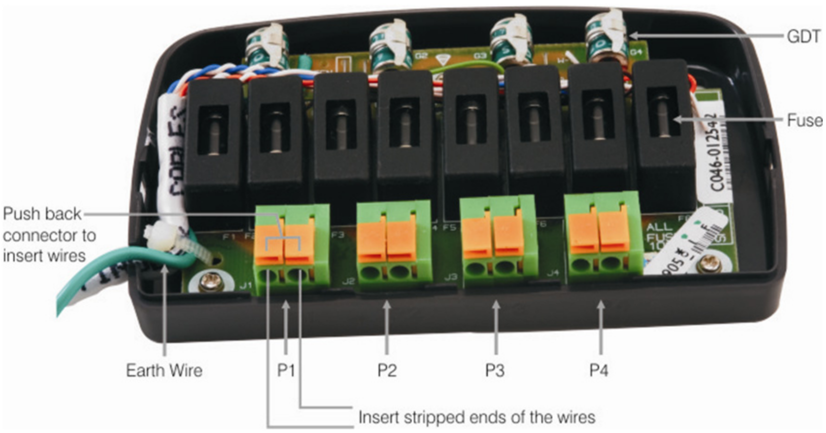
The Fuses in the PPM4 are an over current protection device. Whenever the current builds up beyond the permissible limit, (generally 100mA), the fuse opens to protect the circuit ahead.

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

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

Installing PPM4

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



1. Unpack the PPM and check the package contents.

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

.

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

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

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

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

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

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

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

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

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

    To remove the wires,

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

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

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

    Refer the following table for connection details of the wires:

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

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

15. Mount the PPM, if not done already.

Protecting ETERNITY GENX Platform and Yourself

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

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

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

Location

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

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

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

Power Supply

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

- The ETERNITY GENX does not work in isolation from the environment. Power is fed to the system for functioning of the system. Being an ETERNITY GENX platform with SARVAM UMG application, it the interfaces for various types of trunk lines and extensions. So there are chances of heavy voltages entering the system through these interfaces. Also, static charges could find their way through the system components.

Protect the system from heavy voltages from the mains

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

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

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

Protect the system from Lightning

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

Protect the system from static charges

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

Telecom Earth (Ground)

- The Earth (Ground) is the most important safety procedure to prevent electrical shocks and fires. It protects from lightning strikes, electrical transients, static discharges, electromagnetic interference and electrical hazards.
- Ensure that a proper electrical earth and a telecom earth are in place for the safety of people and the system. Telecom earth is a dedicated earth for the ETERNITY GENX/any other telecom equipment.

- Provide a separate Telecom Earth (Ground) to the system installation. Providing a separate earth to the telecom equipment eliminates the possibility of any back-voltage on the earth.
- Refer “How to Make the Telecom Earth” for instructions on making the perfect earth (ground).

Shock and Fire Hazard

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

Cleaning and maintenance

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

Service and Repair

- Unplug this product from the wall outlet and refer servicing to a qualified service person under the following conditions:
  - When the power supply cord or plug is damaged or frayed.
  - If liquid has been spilled into the product.

- If the product has been exposed to rain or water.
- If the product has been dropped or the cabinet has been damaged.
- If the product exhibits a distinct change in performance.

Battery

ETERNITY GENX contains a 3VDC/18mAh Manganese Lithium Coin Battery (ML 1220 - Rechargeable) of diameter 12.5mm and height 2.0mm. The Battery is located on the CPU Card. The Battery should be replaced only by authorised dealers of Matrix. End Users must not attempt to replace it.

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

Disposal

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

Warning for RF Safety

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

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

CHAPTER 5

Installing ETERNITY GENX

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

Firmware Version V1R3 and earlier

The Matrix ETERNITY GENX is shipped factory fitted with the Power supply card, the CPU Card in their respective fixed slots (refer the section “Know your SARVAM UMG”).

The cards - BRI, T1E1PRI, GSM, CO, SLT - are shipped separately as per the order placed by individual customers. These cards are installed in any of the universal slots.

If you upgrade the system firmware to V1R4 and later, the Expansion Slots license will be applicable for the universal slots. No universal slots will be functional by default. You must purchase the SARVAM EXP4 SME license to activate the universal slots as required.

After upgrading the system with V1R4, if required you can again downgrade the system to V1R3.In this, case all the universal slots will be functional.

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

Firmware Version V1R4 and later

The Matrix ETERNITY GENX is shipped factory fitted with the Power supply card, the CPU Card in their respective fixed slots (refer the section “Know your SARVAM UMG”).

The cards - BRI, T1E1PRI, GSM, CO, SLT - are shipped separately as per the order placed by individual customers. These cards are installed in any of the universal slots.

If you have upgraded the system firmware to V1R4 and later in the old ETERNITY GENX system, the Expansion Slots license will be applicable for the universal slots. No universal slots will be functional by default. You must purchase the SARVAM EXP4 SME license to activate the universal slots as required.

If you have purchased the new ETERNITY GENX system with the firmware V1R4, the Expansion Slots license will be applicable for the universal slots. The first four universal slots after the power supply card will be functional by default. If you require more functional universal slots, you must purchase the SARVAM EXP4 SME license.

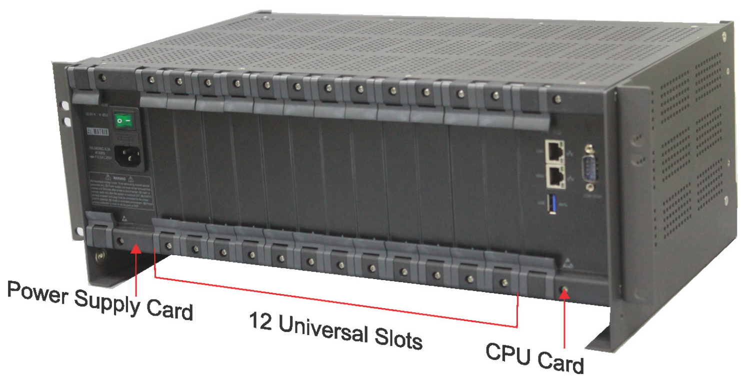
Each SARVAM EXP4 SME license will provide the activation for next four universal slots in sequence.

System purchased with V1R4 firmware must not be downgraded to earlier versions.

For details, see "Expansion Slots" under "License Management".

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

ETERNITY GENX



The extreme left slot is reserved for the Power Supply card and the extreme right slot is reserved for the CPU card.

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

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

Mounting the System

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

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

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

Connecting Input Power Supply

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

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

Inserting Cards

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

8. Select a free slot from the universal slots.

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

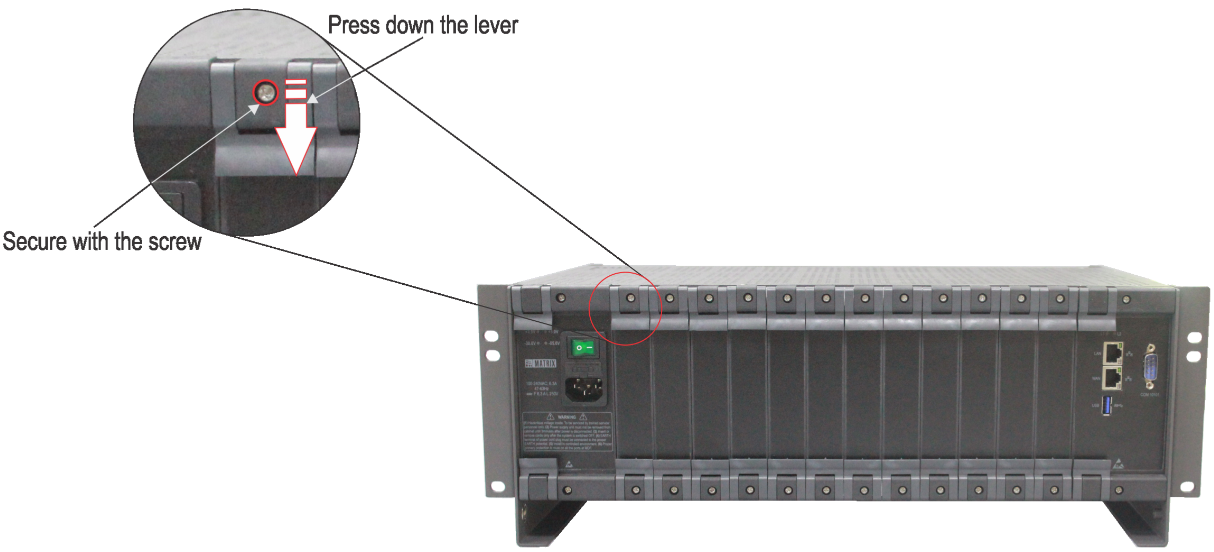
10. Hold the card with the connectors facing you. Do not grab the card from both ends.

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

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

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

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



14. Tighten the screws on either side of the bracket.

15. Following the above steps, install each card into the universal slots.

Detailed installing instructions are provided for each card - SLT, CO, BRI, T1E1, GSM - later in this section. Refer to them when installing each card type.

16. To remove a card:

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

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

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

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

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

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

The Power Supply Card

Two types of Power Supply Cards are supported by the Matrix ETERNITY GENX — ETERNITY GE Card PSUNI (250 W) and ETERNITY GE Card PS48V (250W) and ETERNITY GE CARD PSBB.

- **ETERNITY GE Card PSUNI 250W** with 100-240VAC, 47-63Hz Mains as Input AC Voltage Power Supply and output as 250W.

  This card is designed on the SMPS scheme. As this card does not have any provision for battery backup, it is recommended that a UPS be connected to keep the system powered during outages.

  This card has four Green LEDs, a Mains Switch, and a Socket assembly for connecting the mains cord.

- **ETERNITY GE Card PS48VDC 250W** with 48VDC as Input DC Power Supply Voltage and output as 250W. A Float cum Boost Charger (FCBC) is required to feed 48VDC power to the card. The FCBC works on input AC mains.

  The card has four Green LEDs (for four different voltages), one Red LED (for input reverse indication, an MCB Switch, a power ON/OFF Switch, and a 3-way termination block for connecting the power cord.

  Both, the PSUNI card and the PS48V Card provide DC output voltages as: +3.5V, +5.0V, -30V and -85V. These are indicated by LEDs.

- **ETERNITY GE CARD PS Battery Backup (PSBB)** with 100-240VAC, 47-63Hz Mains as Input AC Voltage Power Supply.

  This card is designed on the SMPS scheme and has a provision for Battery Backup. The card is supplied with a Battery cable to connect a 24VDC external battery. When the system is functioning using the AC power supply the battery will also be charged by the card. This card has no LEDs.

Installing the Power Supply Card

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

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

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

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

AC Power Supply Card must be removed from the platform only three minutes after the power supply is switched off.

Make sure you do not place the Power Supply Card on any conductive surface.

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

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

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

5. If installing the PSUNI card, connect the three-pin power cord into the socket of the PSUNI card and plug in the cord into the mains supply.

   You may connect the PSUNI Card to a UPS to keep the system live during power outages.

   Select a UPS considering the typical power consumption of ETERNITY GENX. See “Power Consumption Table”.

6. If installing the **Eternity GE Card PS48V (250W)**, connect the Float cum Boost Charger (FCBC) or AC-DC Power Supply with 48VDC Output. Terminate the power cord from the FCBC output into the 3-way termination block on the PS48V card.

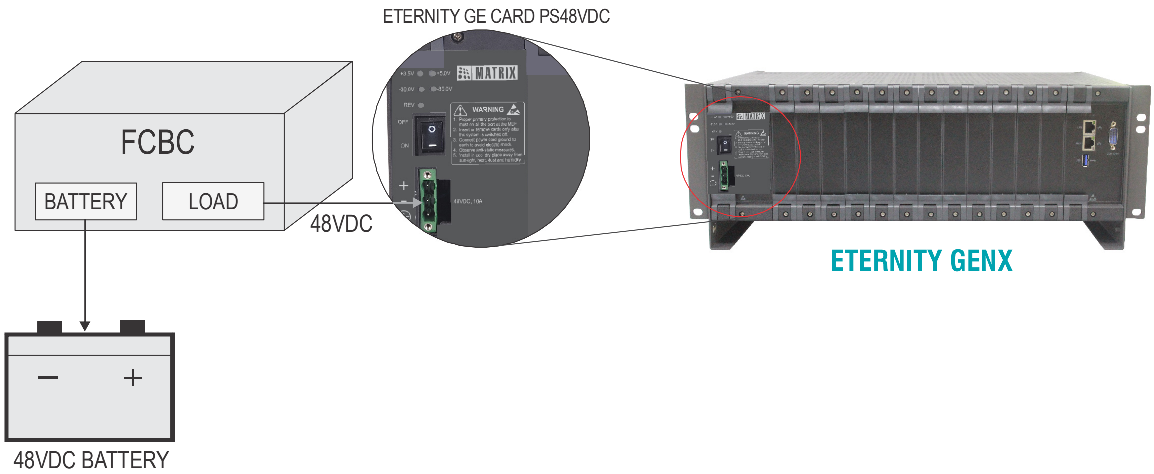
   Polarity is critical. Ensure that the wires are connected with the correct polarity. Follow the standard color codes used by FCBC manufacturers:

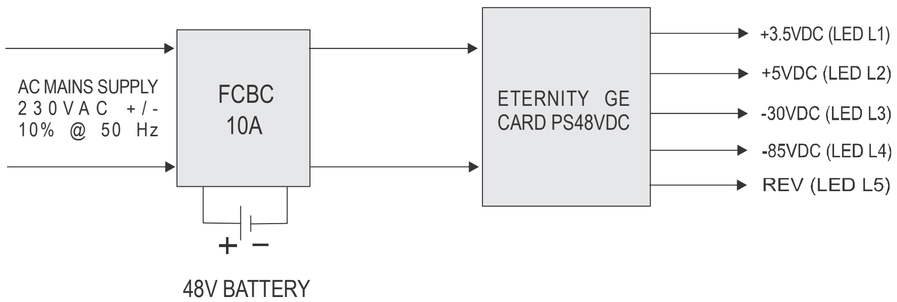
Color	Signal	Symbol
Red	+48VDC	+
Black	GND	-
Green	Earth	⏚

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

Connecting an FCBC to the ETERNITY GENX

As PS48VDC Card works with FCBC, the input voltage working range has hysterisis.The power supply turns ON at Voltage greater than 45VDC and it remains ON till 40VDC during battery discharge condition. Moreover, the input voltage for the card must be from 40VDC to 60VDC.





7. Connect the Battery Backup[3]. Battery backup time depends upon the total load. The total load is the sum of system's load and load of active extensions.

8. If installing **Eternity GE CARD PSBB**, **c**onnect the Battery Backup. Terminate the power cord from the Battery Backup output into the 3-way termination block on the PS Battery Backup card. The Battery cable supplied with the card has a connector on one end and the other end is free. Insert the cable connector into the connector on the card as shown below. At the other free end connect the Battery.

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

Color	Signal
Red	+24VDC
Black	GND

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

Color	Signal
Green	Earth

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

The following table displays the maximum number of off-hook ports supported

Condition	**Number of off-hook ports**
Continuous run condition	55 off-hook ports
Temporary overload	85 off-hook ports

9. Switch on power supply, after completing all other installation tasks.

Power Consumption Table

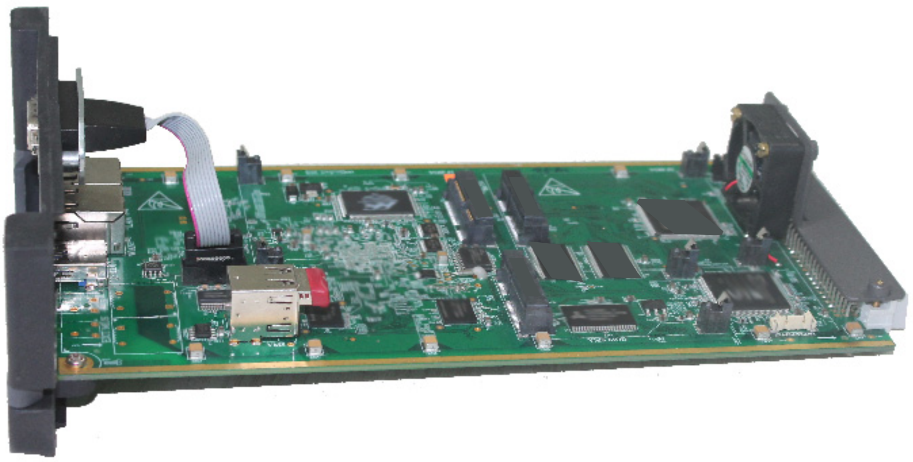
Platform	Number of Ports	Power Supply	Battery Backup[a]
			50% Load
ETERNITY GENX	240	PS48V DC[b]	7 hrs
		PSBB[c]	4 hrs

a. This is applicable to only FXS Ports. The above figures may vary depending upon the system configuration and number of SLT ports in off-hook condition.
b. This is with respect to 48V | 28Ah Battery.
c. This is with respect to 24V | 28Ah Battery.

The Power Supply unit can prove hazardous due to high voltage power it carries. Make sure you get it serviced by a trained service person only.

The CPU Card

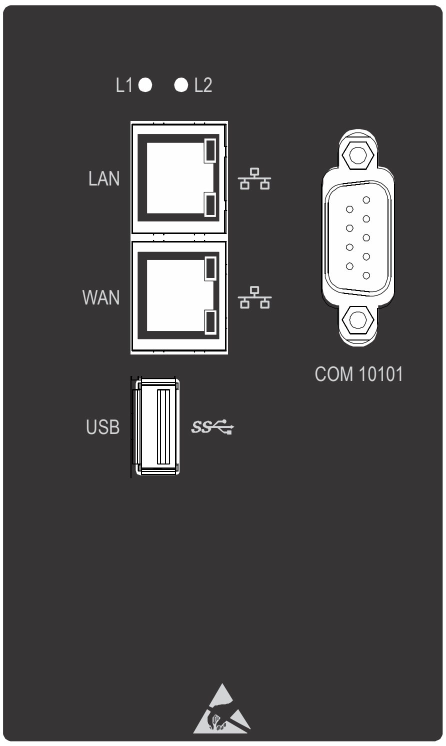
The ETERNITY GENX CPU Card hosts the SARVAM UMG Application. It supports two NX DBM VOCODER64 modules. The module NX DBM VOCODER64 is optional and can be purchased separately.



This card manages the entire system and controls all other slave cards (SLT, CO, CO+SLT, BRI, T1E1, GSM etc.). All configuration and programming information is stored in this card. It also carries out the VoIP functionality.

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

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



Ports and Connectors:

Port	Connector	Description
LAN	RJ45	Used for connecting the Ethernet cable into LAN Port to connect to a PC or a LAN Switch.
WAN	RJ45	Used for connecting the Ethernet cable into WAN Port to connect to a Broadband Router/Modem.
USB	-	For future use
COM	DB-9	For future use

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

LAN Interface

The LAN Port is provided to:

- connect a PC or a LAN Switch. This port is used for operating the web-based programming software, Jeeves.
- capture "System Activity Log", "System Fault Log" and "System Debug".

WAN Interface

The WAN Port is provided to:

- connect a LAN Switch/Router/Modem.
- connect the CPU Card to the public network over a Router/Modem.
- register the SIP Trunks with the ITSP.
- capture "System Activity Log", "System Fault Log" and "System Debug".

VoIP Interface

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

VOCODER Channels

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

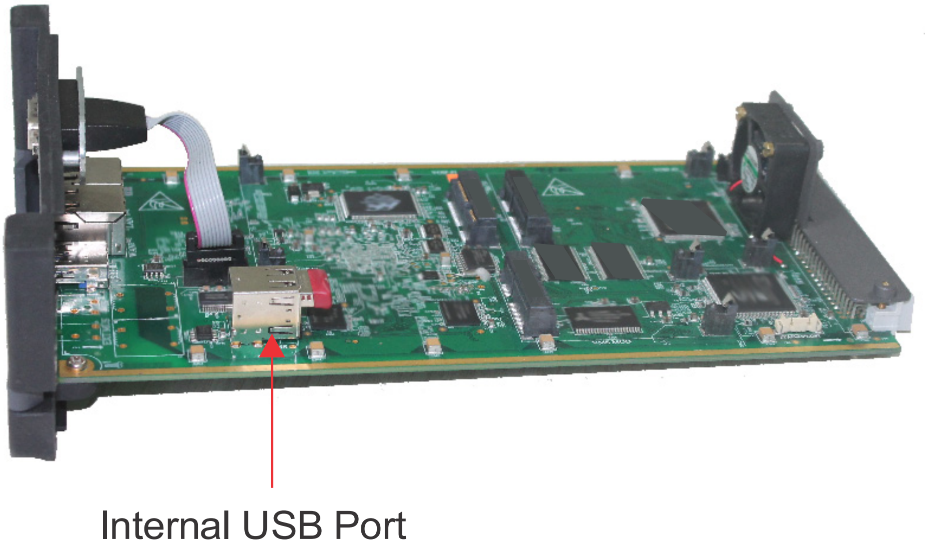
If you require more than 64 Vocoder channels, you can install another NX DBM VOCODER64 Module.

4. *During the Demo period, the number of Vocoder channels supported will be equal to the total number of channels available in the Vocoder module/s installed in the System. If the Demo period is paused or gets expired, then the number of Vocoder channels supported would be as per the license you purchase.*

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

Internal USB Port

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



The pendrive supports FAT32 file format. It contains the SARVAM UMG Application.

Do not remove the pendrive.

When you select the SARVAM UMG SME Application, the system fetches the application from the pendrive.

External USB Port (Device Port) 3.0

This is provided for future use.

COM Port

This is provided for future use.

LED

The CPU card has two dual color (Green and Red) LEDs.

- LED 1 - L1 works as a Heart Bit of CPU Card. In Normal Condition, L1 will be turned ON Green for 1 sec and OFF for 1 sec.
- LED 2 - L2 indicates the Layer Application status. In Normal condition, L2 will be turned ON Orange and will blink very fast.
- Both L1 and L2 also indicate Application status.

Case 1: L1 is steady GREEN/OFF and LED2 is OFF. This means the Application is hung and there is some problem in the Application code.

Case 2: LED1 is steady GREEN/OFF and LED2 is GREEN/RED/ORANGE. This means the Layer is hung and there is some problem in the Layer code.

Jumpers

The position and function of the Jumpers on the CPU Card are:

Jumper Number	Position	Function
J1	AB	Default SE Password.
	BC (default)	Normal.

Installing the VOCODER Module

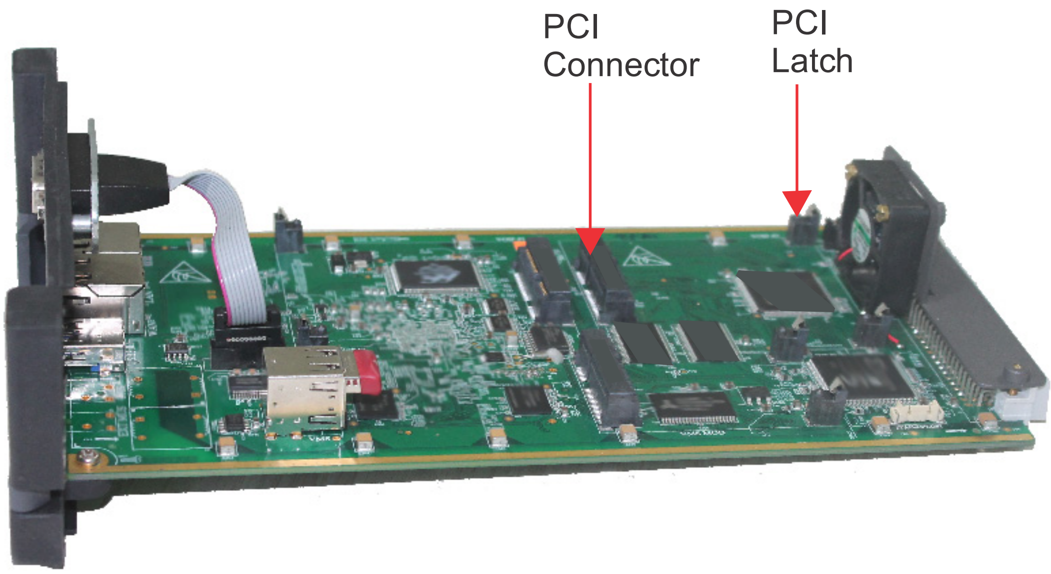
To install,

- Unpack the NX DBM VOCODER64 Module.

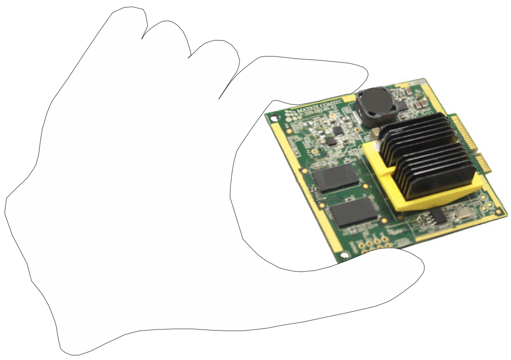
- If the CPU Card is already installed, switch off power supply, unplug the power cord. Remove the screws securing the card. Lift the levers on the mounting bracket to release the card. As the card emerges from the slot, ease it out of the slot.

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

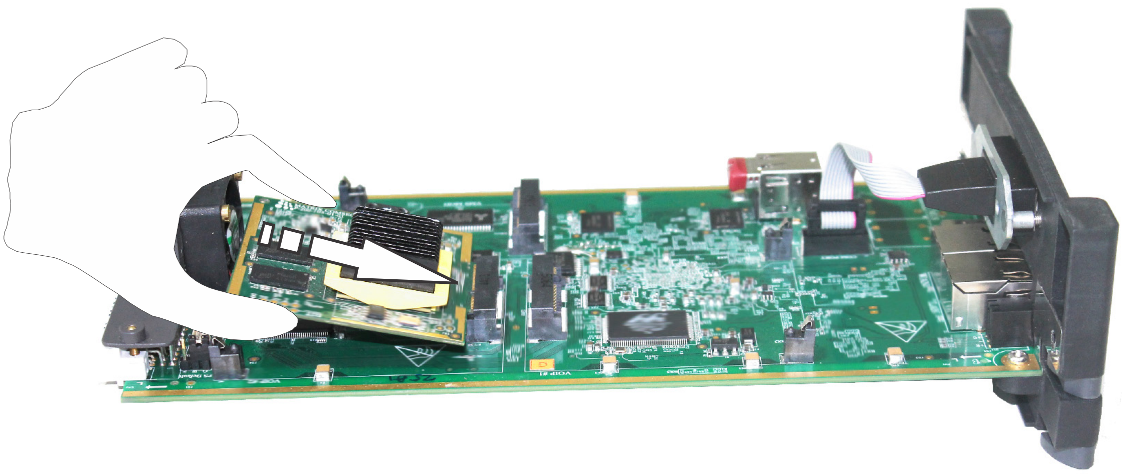
- The NX DBM VOCODER64 Module is to be mounted on the CPU board.



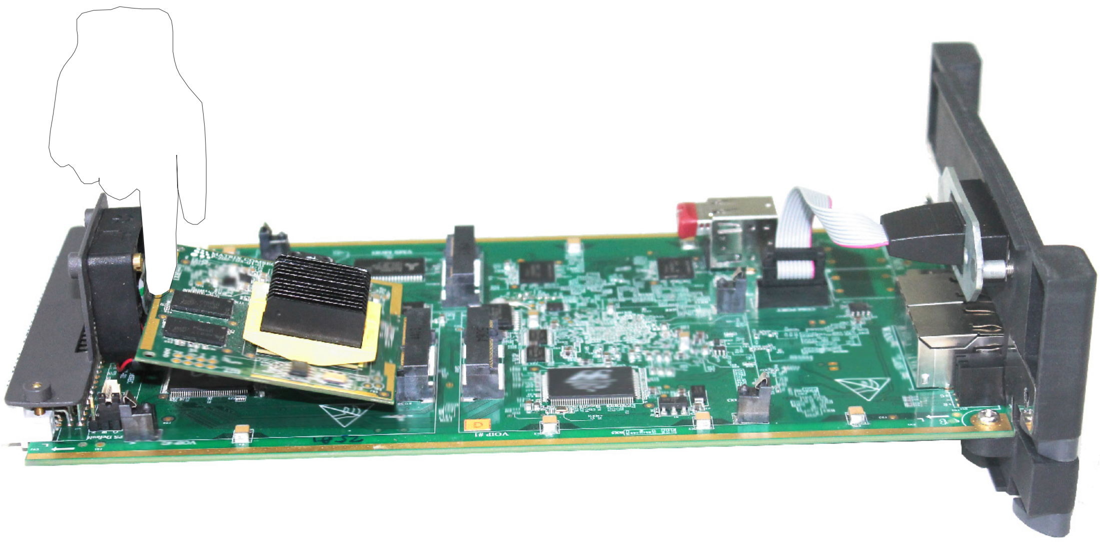
- Locate the PCI Connector and PCI Latch on the mainboard.



- Carefully hold the NX DBM VOCODER64 Module from the edges. Make sure you do not touch the PCB area.



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

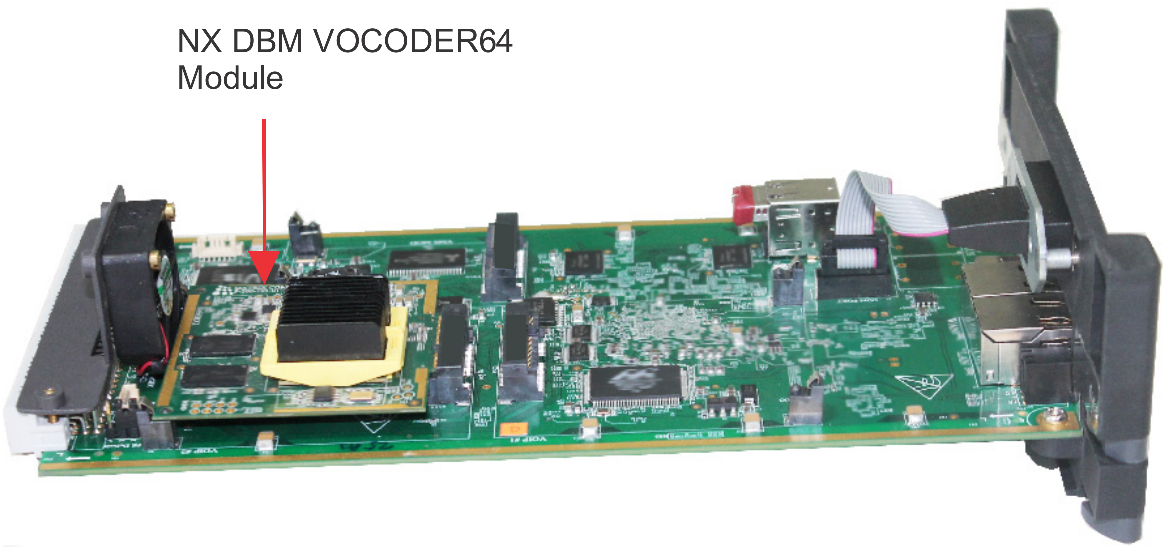


- Press the Module with a finger to fix the latches perfectly into the mounting holes. Make sure you do not touch the PCB area of the module except the yellow line provided for grounding at the front end of the module.

  Do not apply excessive pressure.

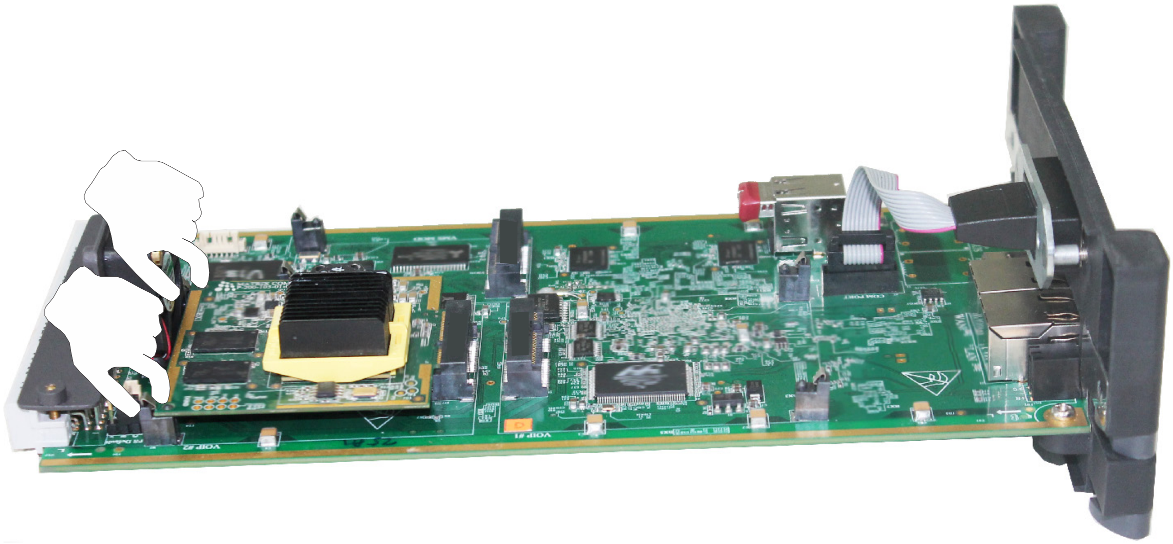
Removing the VOCODER Module

- Locate the NX DBM VOCODER64 Module you want to remove from the CPU Card.

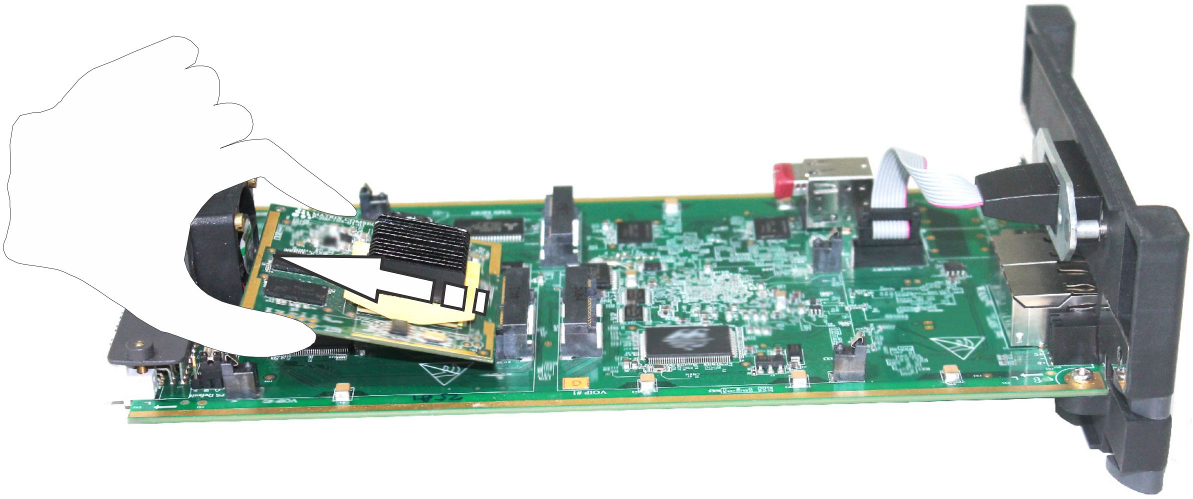


- Press both the latches together.

Make sure you support the base of the latches from behind with your thumbs.



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



The CO Card

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

The CO Card is available in the following configurations:

CO Cards for ETERNITY GENX

Card Name	Configuration and Application
ETERNITY GE Card CO16	16-port card to connect 16 Two-wire Trunk lines from the CO network
ETERNITY GE Card CO8	8-port card to connect 8 Two-wire Trunk lines from the CO network
ETERNITY GE Card CO4+SLT16	Combination card, with 4 CO ports to connect 4 CO analog trunk lines and 16 SLT ports to connect 16 Single Line Telephones

Connectors

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

LED

The CO16 and CO8 Cards have two LEDs. The LED 2 indicates the health of the card during the Reset Cycle.

LED Pattern of CO Card

LED 2 (L2)

PORT Status	LED Color	LED Cadence
Commands from Application to CO Port.	GREEN	Toggle[a]
Events to Application from CO Port.	RED	

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

Installing the CO Card

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

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

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

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

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

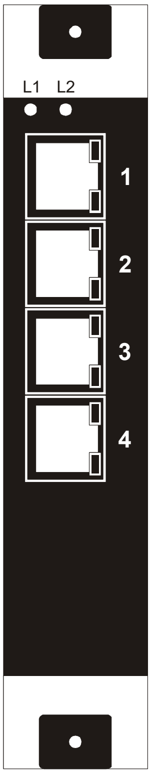
5. Press down the lever on the card mounting brackets to secure the card in its slot. Fix the mounting bracket in place with the two screws provided.

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

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

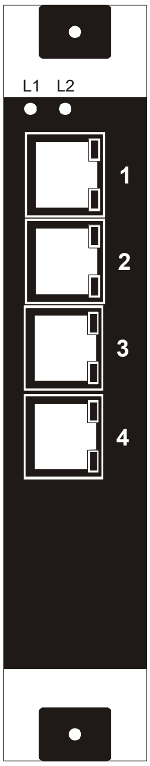
   Refer the illustrations below for the pinout details of the connectors on each card.

ETERNITY GE Card CO16



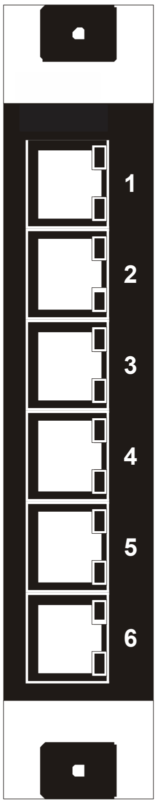
Connector	Color	Connection	H/w Port Offset
RJ45-1	Blue - (Blue & White)	CO	01
	Orange - (Orange & White)	CO	02
	Green - (Green & White)	CO	03
	Brown - (Brown & White)	CO	04
RJ45-2	Blue - (Blue & White)	CO	05
	Orange - (Orange & White)	CO	06
	Green - (Green & White)	CO	07
	Brown - (Brown & White)	CO	08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

ETERNITY GE Card CO8



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

ETERNITY GE Card CO4+SLT16



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

7. Plug in the RJ45 end of the Trunk Card cables into the respective connectors. Refer to the connector diagrams illustrated above for each CO Card type.

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

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

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

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

The BRI Card

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

The BRI Card is available in the following configuration:

BRI Card of ETERNITY GENX

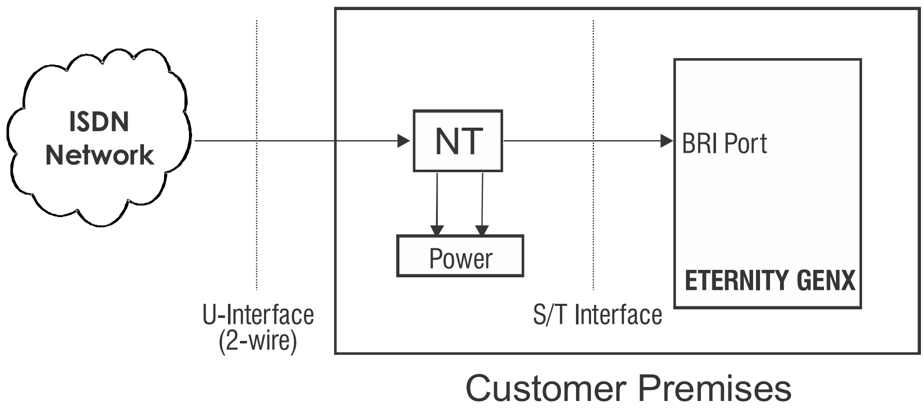
Card Name	Configuration and Application
ETERNITY GE Card BRI4	4-Port card to connect 4 ISDN BRI Lines or ISDN Compatible Devices

Connectors

The BRI card has 4 RJ45 Connectors. A separate cable is supplied for each connector.

ISDN BRI - Installation Scenarios

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



Where,

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

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

TE and NT Modes

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

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

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

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

  OR

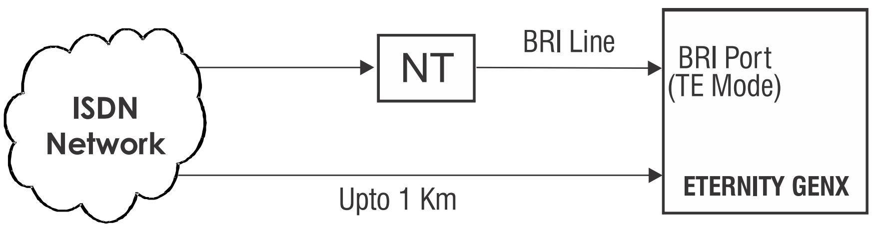
- the BRI port of the other ISDN System in the NT mode and the BRI Port of the ETERNITY GENX in the TE mode.

Also refer the topics “BRI Port - Network” and “BRI Port - Terminal” to know more.

Types of BRI Configuration

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

Point-to-Point Configuration



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

Point-to-Multipoint Configuration

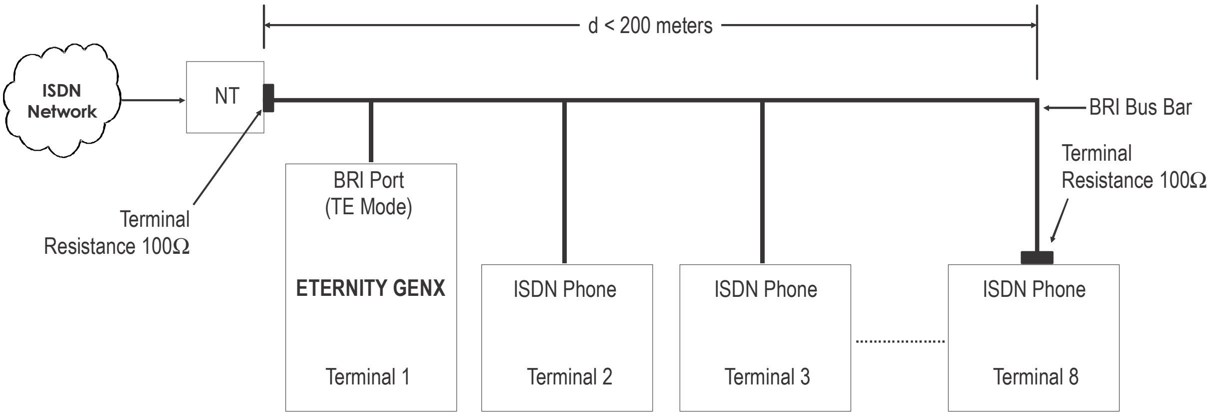
A maximum of 8 ISDN equipment can be connected on a single BRI Bus line in a Point-to-Multipoint configuration.

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

a. Short Passive Bus Configuration

b. Extended Passive Bus Configuration

Short Passive Bus Configuration

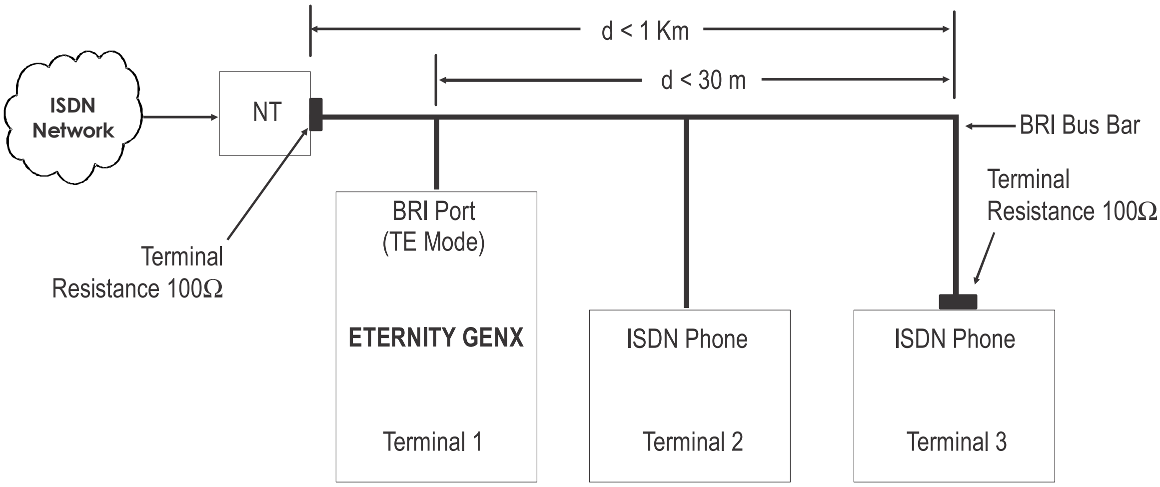


Where,
TE = Terminal Equipment or ISDN device (End user device)
NT = Network Termination provided by the ISDN Service Provider
d = Distance from NT to the last TE equipment.

In a Short Passive Bus Configuration,

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

Extended Passive Bus Configuration



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

In an Extended Passive Bus Configuration,

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

Installing the BRI Card

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

Setting Orientation Type of BRI Port

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

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

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

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

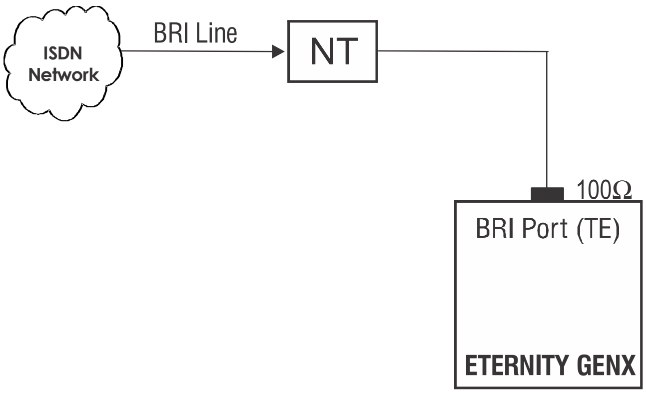
   To set Orientation Type of the BRI Port, you must access the Web-based configuration tool, Jeeves.

   Under **Basic Settings**, click the **BRI Port** link, click the desired port number and under General Parameters set the **Orientation Type**.

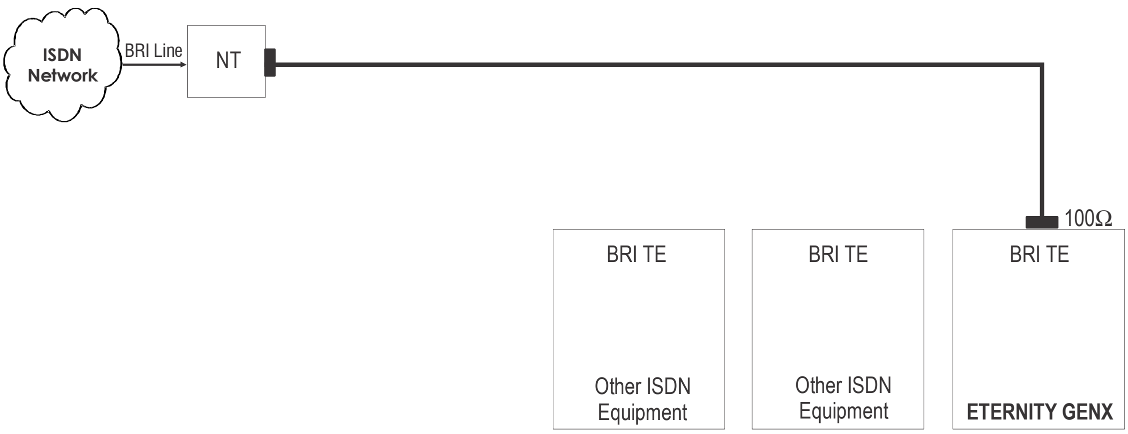
Inserting Termination Resistance

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

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



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



   In a Point-to-Multipoint configuration, 100Ω termination can be provided on either of the following:

- Last TE equipment
- Last point of the bus bar where the last TE equipment is connected.

- When BRI port is configured in the NT mode.

- If the S0 bus itself supports Terminating resistors, Termination Resistance need not be inserted.
  - *BRI Port is configured as TE and connected in a Point-to-Point Configuration as illustrated above.*
  - *BRI Port is configured as NT.*
- *Termination need not be inserted if the BRI port of ETERNITY GENX (configured in TE mode) is connected as any terminal other than the last terminal on the S0 bus (in a Multi-point configuration).*

Termination in TE Equipment (BRI Port)

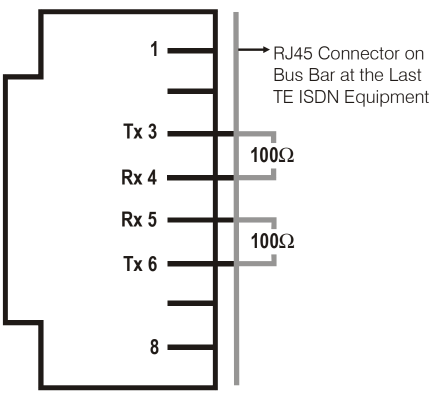
6. To set the 100Ω termination on the BRI port set the Jumpers on the BRI module of each port as given in the table below.

Function	**Module 1 (M1)**				**Module 2 (M2)**			
	BRI Port 1		BRI Port 2		BRI Port 3		BRI Port 4	
	Jumper Position		Jumper Position		Jumper Position		Jumper Position	
	J6	J8	J7	J9	J6	J8	J7	J9
To insert 100Ω termination	AB	AB	AB	AB	AB	AB	AB	AB
To remove 100Ω termination	BC	BC	BC	BC	BC	BC	BC	BC

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

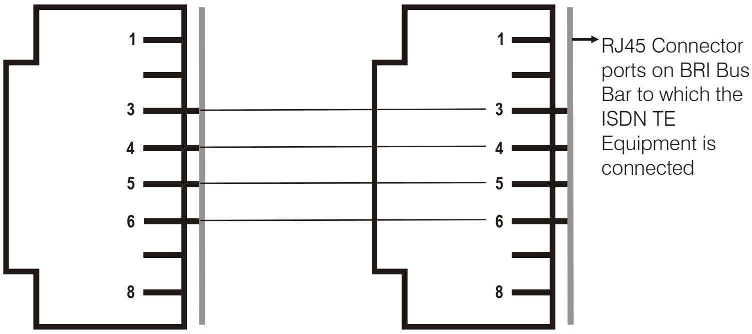
Termination in the Bus Bar

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



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

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



- The above figure shows the connection details of two ports on the BRI Bus Bar. Similarly, you can connect 8 ports on the Bus Bar, keeping in mind the Termination Resister for the NT and the Last TE on the Bus bar.

- Pin number 3, 4, 5 and 6 of the RJ45 connector are used for connectivity.

- Pin number 3 and 6 are used for Transmit (Tx) and pin number 4 and 5 are used for Receive (Rx) from the ISDN TE side.

- Pin number 3 and 6 are used for Receive (Rx) and pin number 4 and 5 are used for Transmit (Tx) from the NT side.

Feeding Power to the Terminal

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

   To do this,

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

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

Function	**Module 1 (M1)**				**Module 2 (M2)**			
	BRI Port 1		BRI Port 2		BRI Port 3		BRI Port 4	
	Jumper Position		Jumper Position		Jumper Position		Jumper Position	
	J4	J5	J2	J3	J4	J5	J2	J3
To feed power on Tx and Rx wires (Phantom Power)	AB	AB	AB	AB	AB	AB	AB	AB

Function	Module 1 (M1)				Module 2 (M2)			
	BRI Port 1		BRI Port 2		BRI Port 3		BRI Port 4	
	Jumper Position		Jumper Position		Jumper Position		Jumper Position	
	J4	J5	J2	J3	J4	J5	J2	J3
To feed power on separate pair of wires	BC	BC	BC	BC	BC	BC	BC	BC

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

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

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

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

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

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

Pin Number	Pin Details
4	Tx
5	Rx

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

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

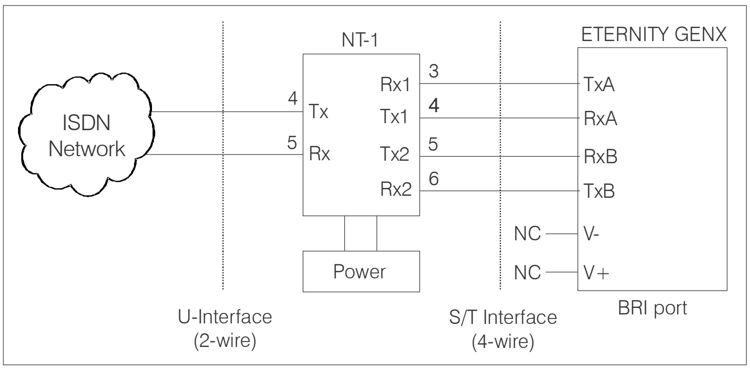
Pinout and Cable Details of BRI4 Port in TE Mode

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

Pinout and Cable Details of BRI4 Port in NT Mode

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

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



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

LED Pattern of the BRI Card

LED 2 (L2)

PORT Status	LED Color	LED Cadence
Commands from Application to BRI Port.	GREEN	Toggle[a] at each command
Events to Application from BRI Port.	RED	Toggle[b] at each event

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

b. Same as above note.

The Mobile Card

The Mobile Card interfaces the ETERNITY GENX with 2G/3G/4G networks. It routes calls made and received over mobile networks, like a mobile handset.

The Mobile Cards are available in 2G, 3G and 4G variants.

The Mobile Card does not support GPRS features, Fax and Data services, network supported services except CLIR and USSD.

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

The Mobile card is available in the following configurations:

The Mobile Card for ETERNITY GENX

Card Name	Configuration and Application
ETERNITY GE Card GSM4	4-port card to connect to 4 GSM networks (4 SIM Cards can be installed).To know more, refer to "ETERNITY GE Card GSM4/ GSM4 3G without SIM Hot-swap". For Hardware Design V2R2, CPLD V2R2 and PCB V2R1 This version onwards SIM Hot Swap is supported, that is the SIM card can be removed and inserted in the SIM Slots without turning off the system. To know more, refer to "ETERNITY GE Card GSM4/ GSM4 3G/ GSM4 4G with SIM Hot-swap".
ETERNITY GE Card GSM4 3G	4-port card to connect to 4 GSM networks with 3G support (4 SIM Cards can be installed). To know more, refer to "ETERNITY GE Card GSM4/ GSM4 3G without SIM Hot-swap". For Hardware Design V2R2, CPLD V2R2 and PCB V2R1 This version onwards SIM Hot Swap is supported, that is the SIM card can be removed and inserted in the SIM Slots without turning off the system. To know more, refer to "ETERNITY GE Card GSM4/ GSM4 3G/ GSM4 4G with SIM Hot-swap".
ETERNITY GE CARD GSM4 4G	4-port card to connect to 4 GSM networks with 4G support (4 SIM Cards can be installed). To know more, refer to "ETERNITY GE Card GSM4/ GSM4 3G/ GSM4 4G with SIM Hot-swap".

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

SIM cards from different service providers can be used.

Antenna

ETERNITY GE Card GSM4 has a single antenna for the four ports. A splitter connects all the four ports on the card into a single antenna. An antenna cable is also provided, giving you the flexibility to move the antenna to another position (in case of weak signal).

Personal Identification Number (PIN)

The SIM cards can be protected from unauthorized use by programming a Personal Identification Number (PIN) on the SIM. If the wrong SIM PIN is entered thrice in a row, by a user, the SIM Card suspects the user and asks for the Personal Unlock Keyword (PUK).

Installing the Mobile Card

To be able to connect ETERNITY GENX to 2G/3G/4G networks, you must have one of the above Mobile card types installed in the system.

1. To install the Mobile Card,

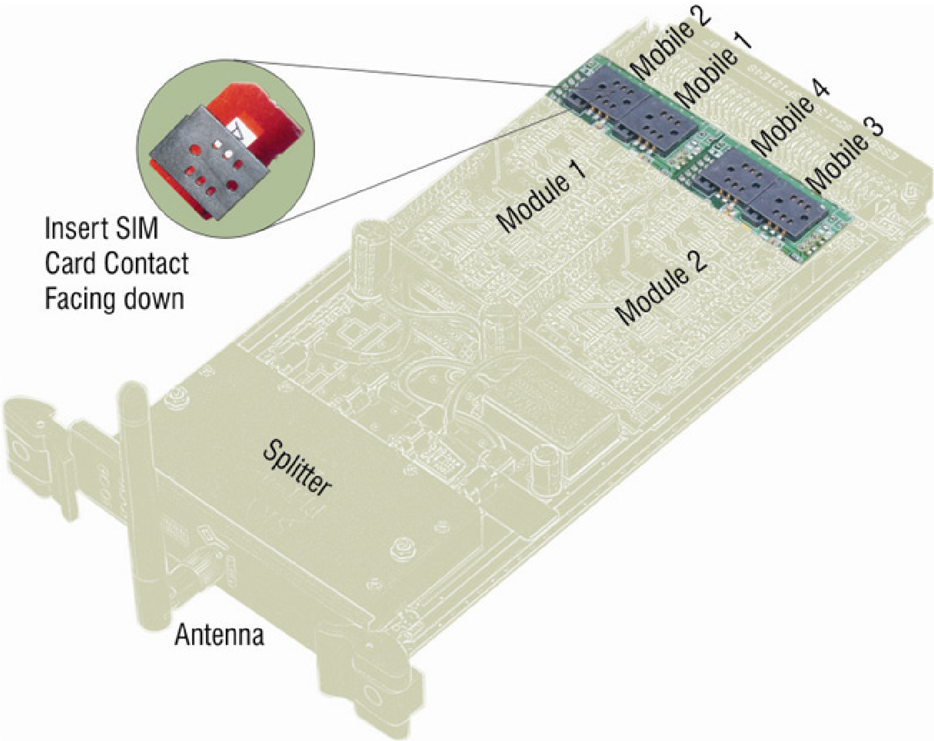
    - If using a 2G/3G/4G card, get the SIM Card from the 2G/3G/4G service provider of your choice ready. Use SIM PIN protection, if required.

Disable Call Waiting in the SIM, else it may result in call disconnection.

2. Make sure that the ETERNITY GENX is installed at a location where sufficient network coverage is available. The power supply should be turned off, and you must be wearing an electrostatic discharge preventive wrist strap and a have a grounding mat, before you begin handling the card.

3. Unpack the Mobile card and verify the package contents.

ETERNITY GE Card GSM4/ GSM4 3G without SIM Hot-swap



Enabling PIN Protection on SIM

To protect the SIM card from unauthorized use, you need to provide a Personal Identification Number (PIN) on the SIM.

4. To do so,
   - insert the SIM into a mobile handset first.
   - enable PIN Protection from the mobile handset.
   - assign the desired SIM PIN value.
   - make sure you configure this value in SIM PIN for the Mobile Port using Jeeves. See “Mobile Port” for detailed instructions.
   - remove the SIM from the mobile handset.

If you do not want to use PIN protection, insert the SIM in the mobile handset and disable PIN protection. Remove the SIM Card from the mobile handset.

5. Insert the SIM card (PIN changed to the desired value), with its connector side down into the SIM holder on the Mobile card. You can insert multiple SIM cards of the same GSM service provider or of different service providers.

6. Insert the Mobile card into the guide rails of the universal slot you have selected for this card. Make sure that the card is inserted deep enough to make perfect contact with the connectors on the backplane. Now, press down the levers on the card mount bracket to secure the card in its slot.

7. Connect the antenna provided with the card on the splitter connector on the front panel of the card. You may also use the antenna cable to place the antenna at another position.

8. Repeat Steps 1-7 to insert another Mobile card.

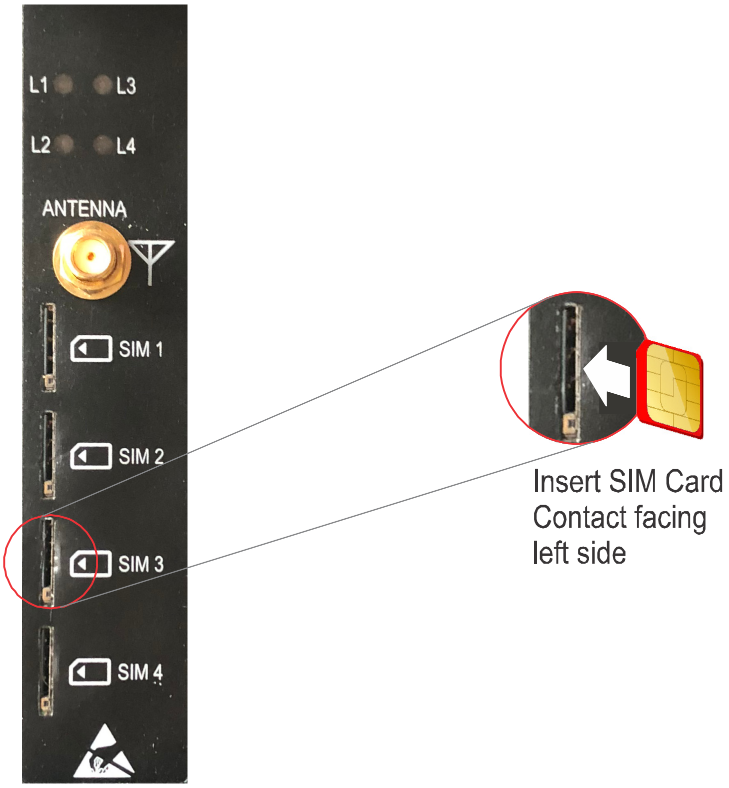
9. If you have completed all installations tasks, power the system.

10. Wait for the system to register with the Mobile network. By default, the Mobile ports are set to select and register with the Mobile networks automatically. Now, observe the LED Patterns of the Mobile Ports.

- *At every power up of the system, it takes about 3 minutes for the Mobile ports to get registered with the network. Once registration with the GSM network is completed, the mobile port can be used.*

- *Each time the Mobile Port sends a request, such as Registration Request, the system waits for the duration of the Network Response Timer. This Timer signifies the time for which the Mobile Port waits for a response from the Mobile network. It is fixed as 150 seconds for all Mobile ports.*

ETERNITY GE Card GSM4/ GSM4 3G/ GSM4 4G with SIM Hot-swap



Enabling PIN Protection on SIM

To protect the SIM card from unauthorized use, you need to provide a Personal Identification Number (PIN) on the SIM.

4. To do so,
   - insert the SIM into a mobile handset first.
   - enable PIN Protection from the mobile handset.
   - assign the desired SIM PIN value.
   - make sure you configure this value in SIM PIN for the Mobile Port using Jeeves. See “Mobile Port” for detailed instructions.
   - remove the SIM from the mobile handset.

If you do not want to use PIN protection, insert the SIM in the mobile handset and disable PIN protection. Remove the SIM Card from the mobile handset.

5. Insert the SIM card with its contact side facing left into the SIM slot located on the fascia of ETERNITY GE Card.

6. Push the SIM backwards into the slot until you hear a click and the SIM is locked in place.

7. To unlock the SIM, push the protruded portion of the SIM backwards again and release it.

The Mobile cards with SIM Hot - swap are designed keeping in mind the Standard Nano SIM size. In case, you face any issues due to the SIM size, contact your respective Service Provider for assistance.

8. Repeat the same steps to insert another SIM Card. You can insert multiple SIM cards of the same GSM service provider or of different service providers.

9. Insert the Mobile card into the guide rails of the universal slot you have selected for this card. Make sure that the card is inserted deep enough to make perfect contact with the connectors on the backplane. Now, press down the levers on the card mount bracket to secure the card in its slot.

10. Connect the antenna provided with the card on the splitter connector on the front panel of the card. You may also use the antenna cable to place the antenna at another position.

11. If you have completed all installations tasks, power the system.

12. Wait for the system to register with the Mobile network. By default, the Mobile ports are set to select and register with the Mobile networks automatically. Now, observe the LED Patterns of the Mobile Ports.

- *At every power up of the system, it takes about 3 minutes for the Mobile ports to get registered with the network. Once registration with the GSM network is completed, the mobile port can be used.*

- *Each time the Mobile Port sends a request, such as Registration Request, the system waits for the duration of the Network Response Timer. This Timer signifies the time for which the Mobile Port waits for a response from the Mobile network. It is fixed as 150 seconds for all Mobile ports.*

LED Pattern of Mobile Ports

ETERNITY GE GSM4 Card has 4 tri-colour LEDs. The number of LEDs on the GSM card corresponds with the number of mobile ports on the card.

After the Reset cycle is complete, during normal functioning, the LEDs color and cadence is described in the table below for various events on the Mobile port:

Event	Color	Cadence in msec (1 cadence is of 3000 msec)
Port idle	-	LED OFF
Port Active (All States other than Ring and Speech)	Red	Continuous ON
Ring Event	Green	400ms ON-200ms OFF400ms ON-200ms OFF
Speech	Green	Continuous ON
GSM initialization	Orange	200ms ON-200ms OFF-200ms ON-200ms OFF-200ms ON-200ms OFF-200ms ON-200ms OFF-200ms ON-1200ms OFF (5 blinks)
PUK required	Orange	200ms ON-200ms OFF-200ms ON-200ms OFF-200ms ON-200ms OFF-200ms ON-1600ms OFF-
SIM PIN faulty	Orange	200ms ON-200ms OFF-200ms ON-200ms OFF-200ms ON-2000ms OFF (3 blinks)
SIM Absent	Orange	200ms ON-200ms OFF-200ms ON-2400ms OFF (2 blinks)
Network Link Down (Absence of GSM Network)	Orange	200ms ON-2800ms OFF

The T1E1PRI Card

The ETERNITY GE Card T1E1PRI Single provides the interface to connect ETERNITY GENX to ISDN PRI Network.

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

- PRI
- Robbed Bit Signaling (RBS)

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

- PRI
- Channel Associated Signaling (CAS)

The T1E1PRI Card is available in the following configuration:

T1E1PRI Card for ETERNITY GENX

Card Name	Configuration and Application
ETERNITY GE Card T1E1PRI Single	1-Port card to connect 1 ISDN T1/E1 PRI Line or ISDN Compatible Device

Connectors

The T1E1PRI card has an RJ45 Connector. A cable with RJ45 plugs on both ends is supplied for the connector.

LED

The ETERNITY GE Card T1E1PRI Single has 2 LEDs - L1 and L2 - for indicating the port states.

Installing the T1E1PRI Card

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

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

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

Setting Line Termination Resistor

4. To set the Line Termination Resistor for the PRI Port for T1 or E1 Connectivity, you must change the position of the jumper J5. Refer the table below.

Jumper	Position	Meaning
J5	BC	To set termination resistance of 120Ω for E1 connectivity
J5	AB	To set termination resistance of 100Ω for T1 connectivity

- By default J5 is set to BC position to provide 120Ω termination resistance for E1 connectivity.
- To set 100Ω termination resistance for T1 connectivity, set jumper J5 to AB position.

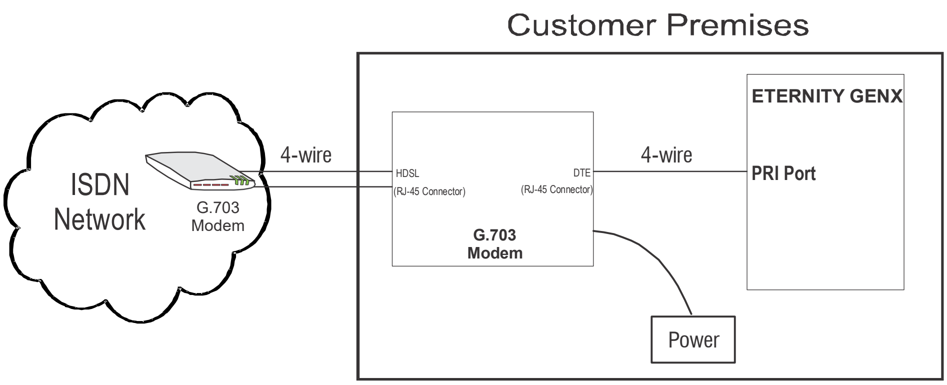
5. Insert the T1E1PRI Card into the guide rails of the free slot you selected for the card. Make sure that the connectors on the card make perfect contact with those of the slot on the backplane motherboard.

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

Connecting ISDN T1/E1 PRI Lines

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

   The diagram below illustrates this.



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

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

9. Refer the following pin details for connecting the Network Termination Unit with the ETERNITY GENX.

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

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

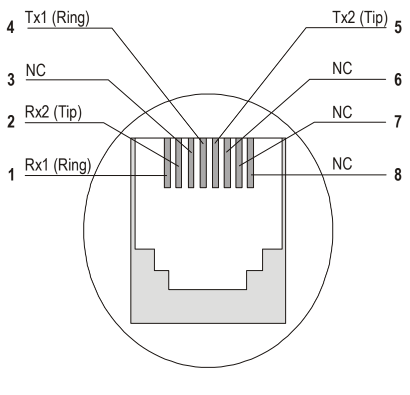
Pin details of DTE Interface of G.703 Modem. (HDSL Network Interface Unit)

Pin Number	Pin Details
1	TX1 (Tip)
2	TX2 (Ring)
3	Not used
4	RX1 (Ring)
5	RX2 (Tip)
6	Not used
7	Not used
8	Not used

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

Pin details of ETERNITY GENX T1E1PR1 Port

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



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

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

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

LED Patterns

The ETERNITY GE Card T1E1PRI has 2 LEDs: L1 and L2. Given below are the LED Patterns defined for each port state in different signaling types supported by the ETERNITY GENX.

1. Port Active Mode

Signaling Type: E1-PRI

LED1 Pattern:

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

LED2 Pattern:

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

Signaling Type: E1-CAS

LED1 Pattern:

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

LED2 Pattern:

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

Signaling Type: T1-RBS or T1-PRI

LED1 Pattern:

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

LED2 Pattern:

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

2. *Port Disable Mode*

LED1 Pattern:

Port Status	Color	Cadence
Port Disable	RED	Continuous ON

LED2 Pattern:

Port Status	Color	Cadence
Port Disabled	OFF	OFF

The Single Line Telephone Card

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

The SLT Card is available in the following configurations:

SLT Cards for ETERNITY GENX

Card Name	Configuration and Application
ETERNITY GE Card SLT20	20-port card to connect 20 Single Line Telephones
ETERNITY GE Card SLT16	16-port card to connect 16 Single Line Telephones
ETERNITY GE Card SLT8	8-port card to connect 8 Single Line Telephones
ETERNITY GE Card CO4+SLT16	Combination card with 4 ports to connect 4 Two-wire Trunk lines, and 16 ports to connect 16 Single Line Telephones

The maximum number of SLT ports supported are 240. However, the maximum number of simultaneous off-hook SLT ports supported are 120.

Connectors

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

LED

The Cards SLT8 and SLT16 have 2 LEDs, while SLT20 has no LED. The LED 2 indicates the health of the card during the Reset Cycle.

LED Pattern of SLT Card

LED 2 (L2)

PORT Status	LED Color	LED Cadence
Commands from Application to SLT Port.	GREEN	Toggle[a] at each command
Events to Application from SLT Port.	RED	Toggle[b] at each event

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

b. *Same as above note.*

Installing Single Line Telephones

To be able to connect Single Line Telephones as Extensions to your ETERNITY GENX, you must install at least one of the above mentioned SLT cards in the System.

1. Decide the number of SLT extensions required and arrange for as many telephone instruments.
   You may use any standard telephone instrument like a rotary phone, a pulse-tone switchable push-button phone, a feature phone or a cordless phone.

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

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

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

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

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

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

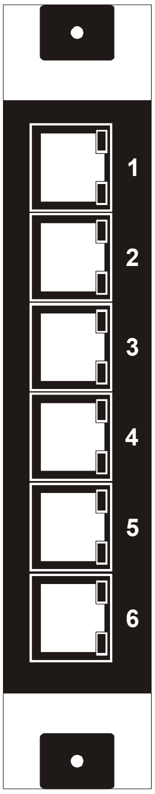
If you are installing more than one SLT card, you can install the second card in any other free slot. It is not necessary to install the second/third card in the subsequent slots.

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

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

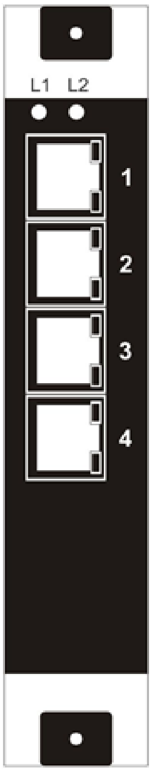
   Refer the illustrations below for pin out details of each connector.

ETERNITY GE Card SLT20



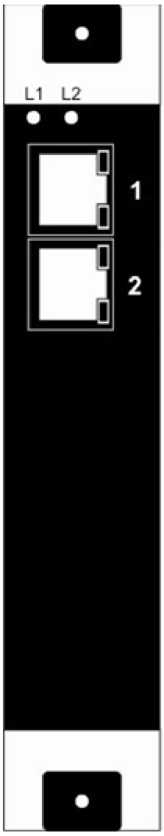
Connector	Color	Connection	H/w Port Offset
RJ45-1	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	01 02 03 04
RJ45-2	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	05 06 07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

ETERNITY GE Card SLT16



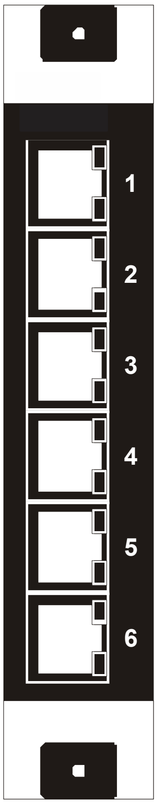
Connector	Color	Connection	H/w Port Offset
RJ45-1	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	01 02 03 04
RJ45-2	Blue - (Blue & White) Orange - (Orange & White) Green - (Green & White) Brown - (Brown & White)	SLT SLT SLT SLT	05 06 07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

ETERNITY GE Card SLT8



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

ETERNITY GE Card CO4+SLT16



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

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

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

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

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

Connecting SLT instruments

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

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

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

Starting Up ETERNITY GENX

Power ON

1. If you have completed all the installation tasks, switch on power supply.
   - For PSUNI card installed in the system, connect the three-pin plug of the power cord from the ETERNITY GENX into the AC outlet, and switch on power supply.
   - For PS48V card installed in the system, keep the MCB Switch ON and power the FCBC.
2. Observe the Reset Cycle.

Reset Cycle

- Reset Cycle (Power-ON Self Test) takes about 2 minutes to finish.
- All the LEDs and the cards of the System are turned on.

Interpreting LEDs

The functioning of the LEDs of the system and the various cards and their meaning are summarized at the end of the installation instructions for each Card Type.

Refer to the LED Patterns described for each Card Type to verify if the system is operating properly and locate faults, where they occur.

CHAPTER 6

Configuring ETERNITY GENX

ETERNITY GENX Platform provides a Graphic User Interface (GUI), Jeeves, the proprietary web-based configuration software of Matrix. The built-in web server Jeeves allows you to select the Application you want to run on the ETERNITY GENX platform.

Cards will work only after you select the application you want to run on the ETERNITY GENX Platform.

The accessibility to the Web-based GUI is secured by a password. This password cannot be used to configure the system using commands.

To be able to access Jeeves,

- the LAN/WAN Port of ETERNITY GENX 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.

If the computer for accessing Jeeves is connected in a LAN Switch and the WAN Port of SARVAM UMG is connected behind a NAT router, make sure that both the LAN and WAN connections are in different Subnet Masks.

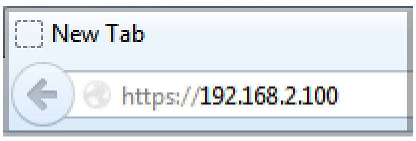
To login,

- Open the browser (Internet Explorer/Mozilla Firefox) on the PC (Standalone or LAN PC) to which the ETERNITY GENX is connected.
- Make sure the IP Address of the computer and the LAN Port of ETERNITY GENX do not conflict, and that both are in the same Subnet.

  The default IP Address of the LAN Port is: **192.168.2.100**

  The default Subnet Mask of the LAN Port is: **255.255.255.000**

  Change the Subnet of the computer, if necessary.
- In the address bar of the browser, enter https://192.168.2.100.

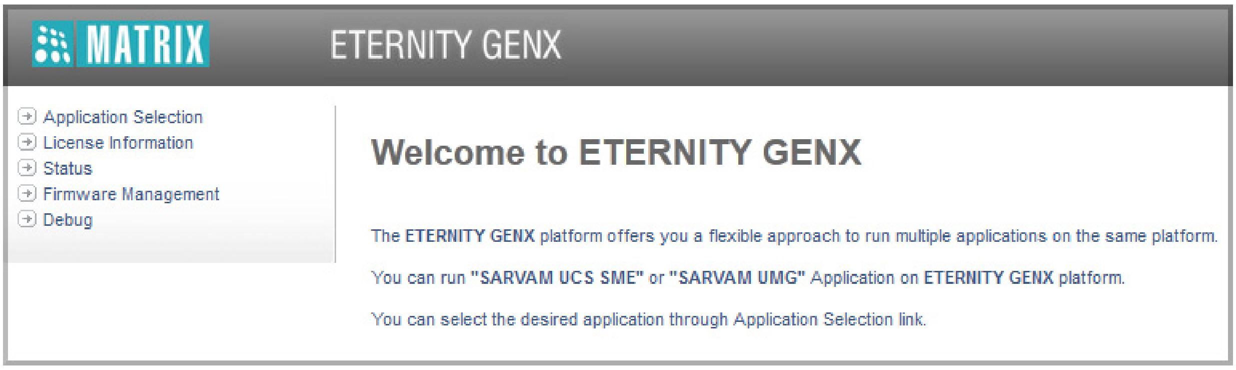


*If you enter the IP Address **192.168.2.100** directly, you will be redirected to the HTTPS protocol for secure access. Click the https://192.168.2.100 link on the page.*

- The **Login** page will open.
- In **Login Password**, enter ***1234***, the default Password[5].

- Click the **Login** button.
- On successful login, the **Home** page of Jeeves opens.

The left navigation bar displays the links — **Application Selection, License Information, Status, Firmware Management** and **Debug**.



Application Selection enables you to select the application you wish to run on the ETERNITY GENX platform. You may select — SARVAM UCS SME or SARVAM UMG.

License Information displays the License key along with the License details of the applications.

Status displays the system details and the status of all the ports.

---

5. *Enter the default password only for the first time. If you login again after switching from the respective application, enter the new updated password.*

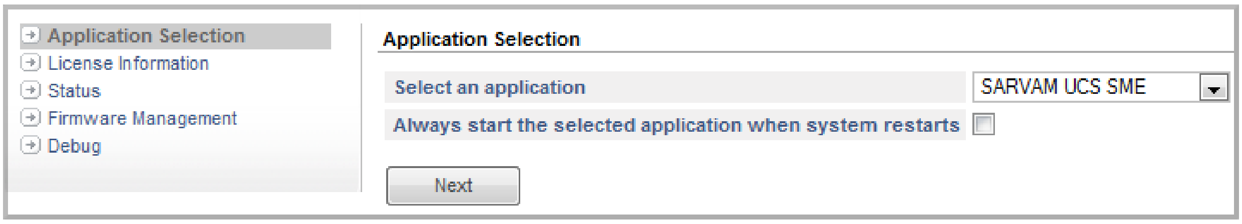
Firmware Management enables you to manage the upgradation of the system software with a click of a button.

Debug allows you to enable and configure the debug settings.

Application Selection

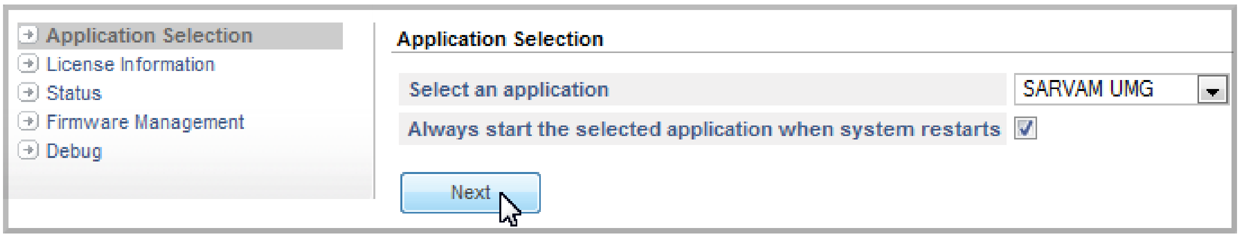
Through **Application Selection**, you can select the application you wish to run on the ETERNITY GENX Platform.

- In **Select an Application**, you may select the required application — SARVAM UCS SME or SARVAM UMG. Default: SARVAM UCS SME.



- Select the **Always start the selected application when system restarts** check box to enable. The selected application will start whenever the system restarts.

  Keep the check box disabled only if you want to select the application to be run on the ETERNITY GENX platform everytime the system restarts. Default: Disabled.



- Now, click on the **Next** button to proceed to the SARVAM UMG Configuration.

When you click the Next button, you will be redirected to the SARVAM UMG Application. To return back to this page, see "Switch Application" in "System Detail".

Once you are redirected to the SARVAM UMG Application,

- If you have purchased the SARVAM UMG SME License, you must activate it. For detailed instructions, refer "How to activate your License".

- If you have not purchased the license and you wish to use the features on trial basis, you can use the Demo Provision. Demo Provision enables you to use the SARVAM UMG application, free of cost for a period of 60 days.

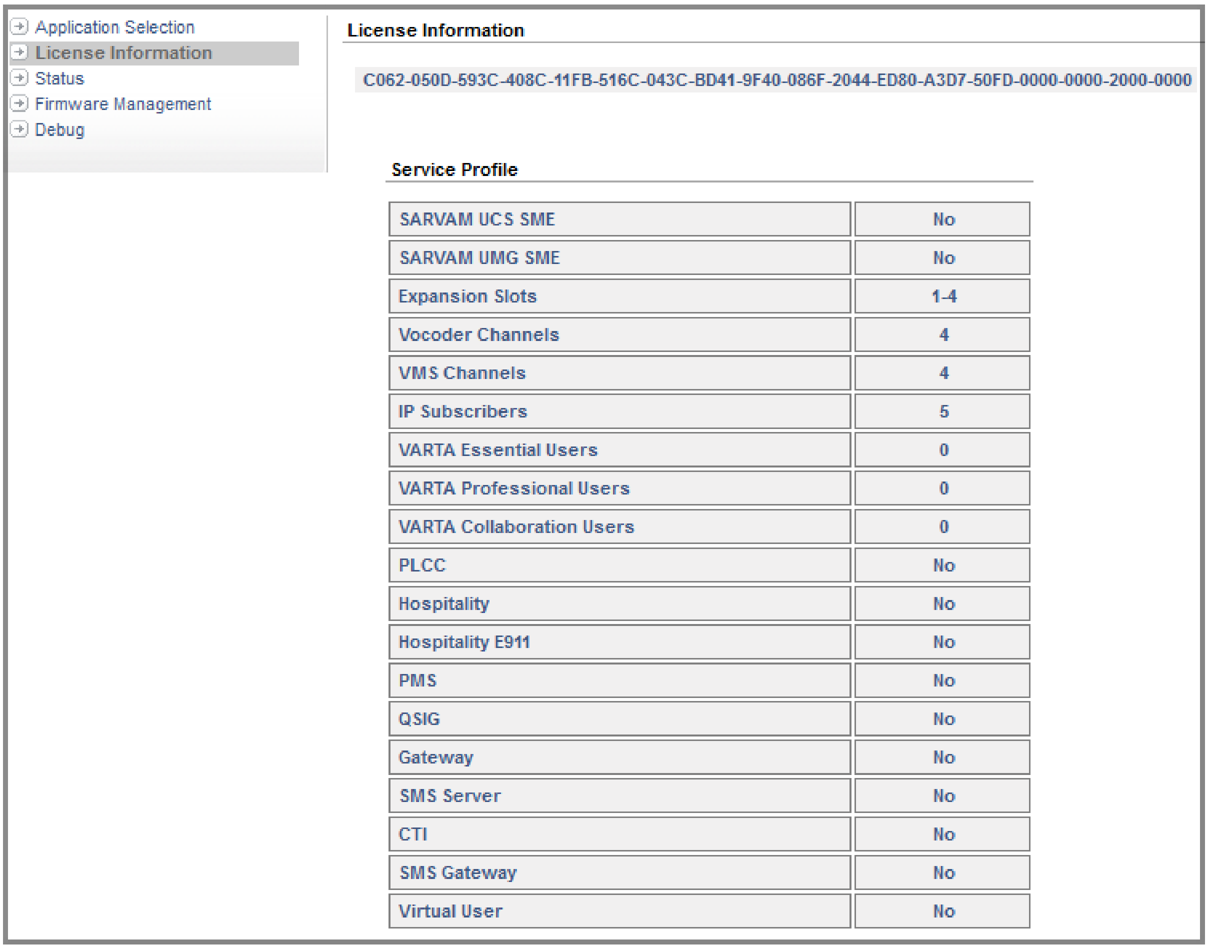
  During the Demo Provision you can access and use all the features and functionalities[6] supported by the application. For detailed instructions, refer "Demo Provision".

6. *The number of Vocoder channels that will be supported in demo period will be equal to the total number of channels available in the Vocoder module/s installed in the System.*

- If you do not have the license for the SARVAM UMG Application and you do not start the Demo Period, the system will disconnect all the connected calls[7] from any port after 60 seconds.

License Information

License Information displays the License key along with the License details of the applications. Whenever you want to activate license for any software application, you can check the current license key from here.



SARVAM UCS SME	No
SARVAM UMG SME	No
Expansion Slots	1-4
Vocoder Channels	4
VMS Channels	4
IP Subscribers	5
VARTA Essential Users	0
VARTA Professional Users	0
VARTA Collaboration Users	0
PLCC	No
Hospitality	No
Hospitality E911	No
PMS	No
QSIG	No
Gateway	No
SMS Server	No
CTI	No
SMS Gateway	No
Virtual User	No

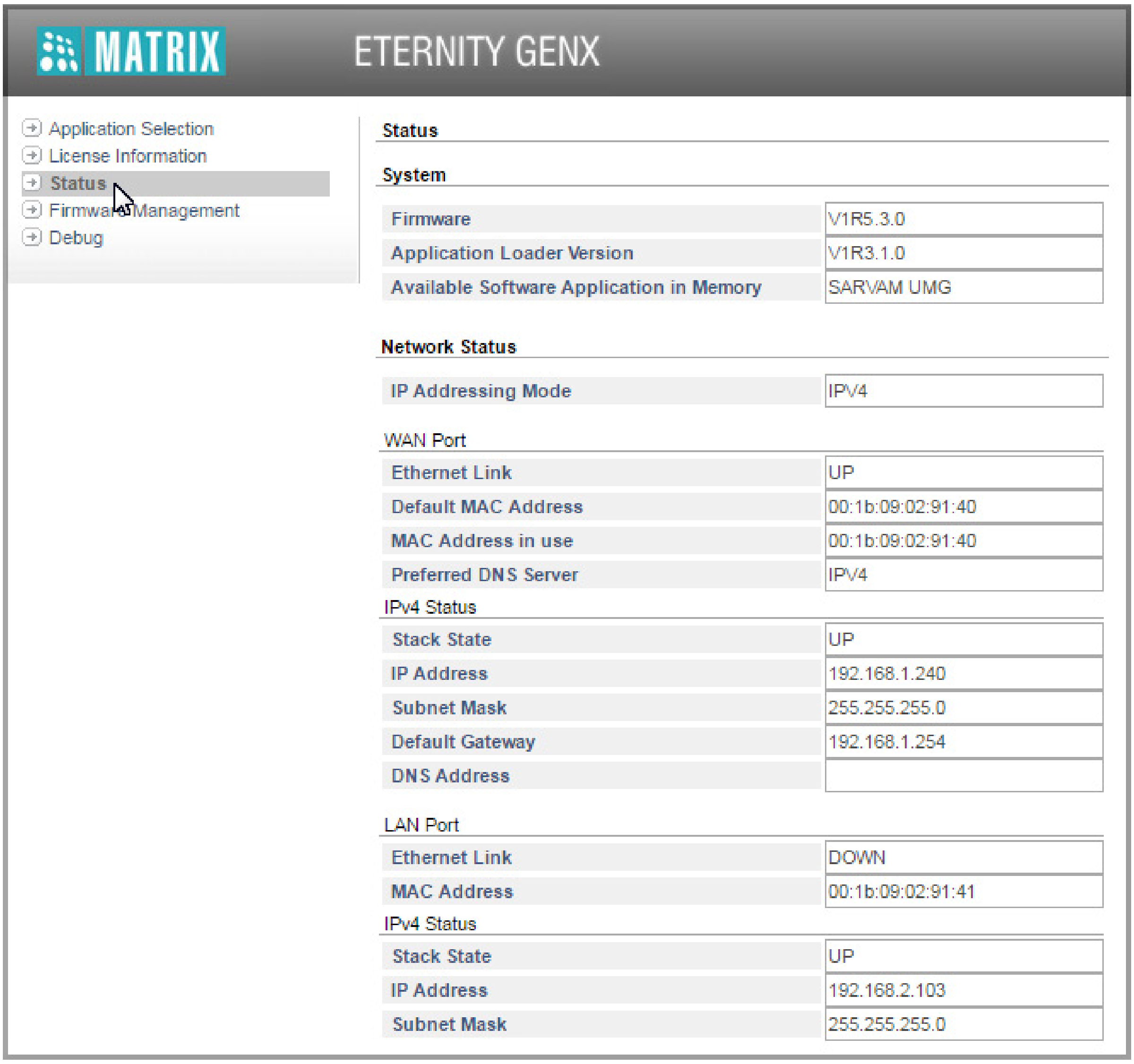
To activate the license, refer topic “How to activate your License”.

To know more regarding the licenses, refer topic “License Management”.

7. *Connected calls means where speech is connected between the calling party port and the called party port even if the called party port is not matured.*

Status

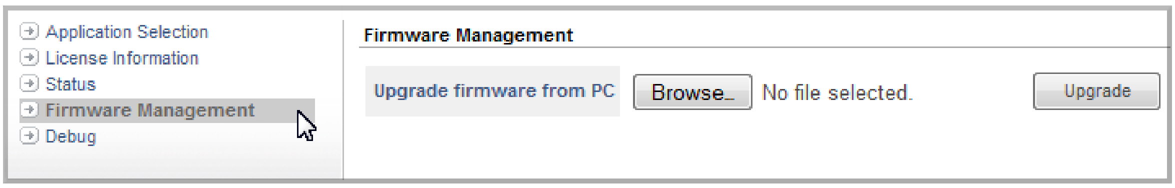
Status displays the system details, that is, Product name and the Available software application, Network status, WAN Port and LAN Port status.



Firmware Management

To upgrade firmware of ETERNITY GENX,

- In **Upgrade Firmware from PC**, click the **Browse** button to reach the location on the local disk where the firmware files are stored in your PC. Make sure that the file is a zip file with **.zip** extension.



- After selecting the required firmware zip file from the PC, click the **Upgrade** button.

The system starts the upgradation process. After successful upgradation and validating the file, the system restarts with the upgraded firmware.

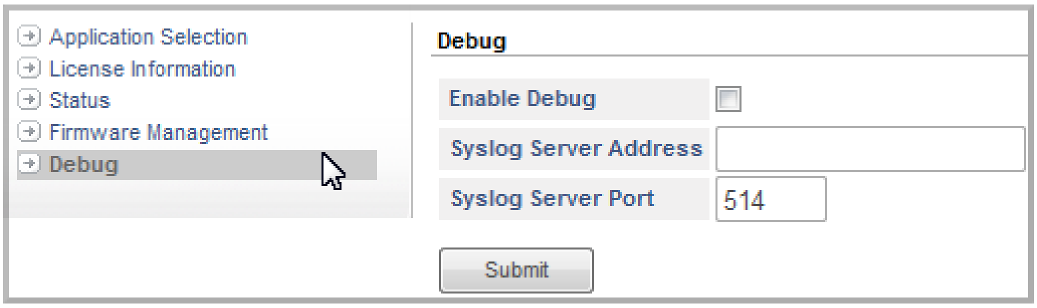
If you select a file other than zip file, an error message is displayed when you click the Upgrade button.

Debug

ETERNITY GENX supports Syslog Client for debugging. Debug messages are sent to the remote Syslog Server.

- Select the **Enable Debug** check box to enable system debug. Default: Disabled.

  You will be able to configure the Debug Settings only after you enable this check box.



- In **Syslog Server IP Address**, enter the remote Syslog Server IP Address. Default: Blank.

  In **Syslog Server Port**, enter the port number. The range of the server port is 514, 1024 to 65535. Default: 514.

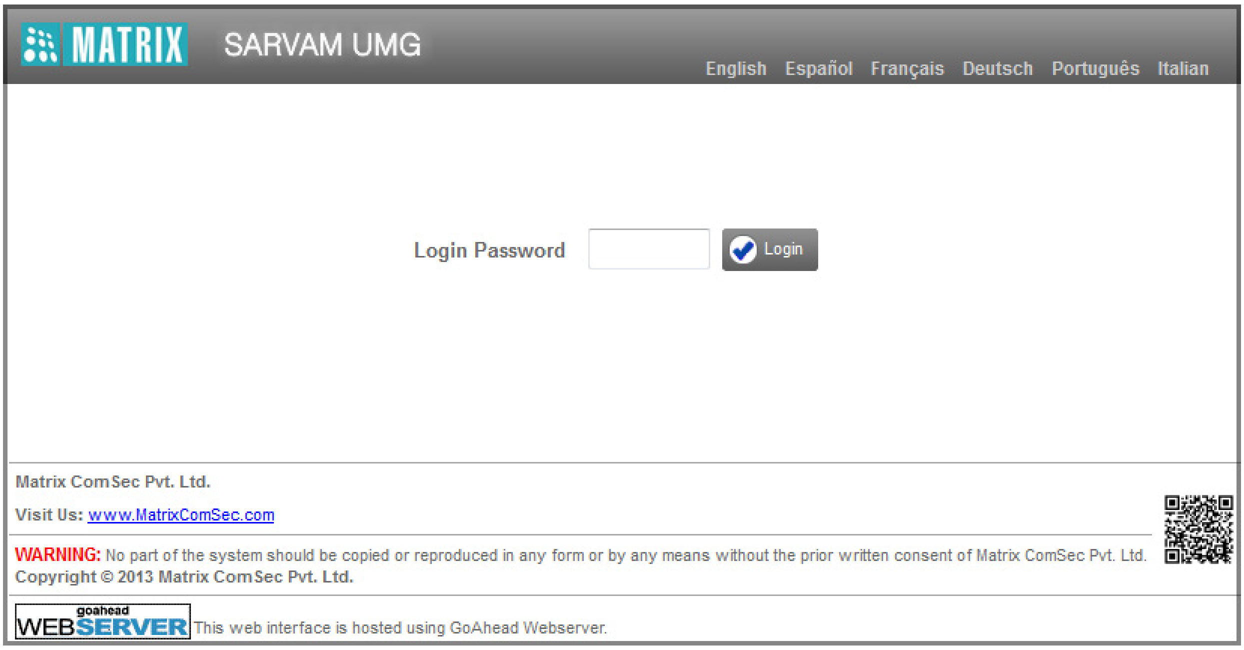
CHAPTER 7

Configuring SARVAM UMG

SARVAM UMG Application provides an embedded web server with a Graphic User Interface (GUI), *Jeeves*, for configuration.

To access SARVAM UMG Jeeves,

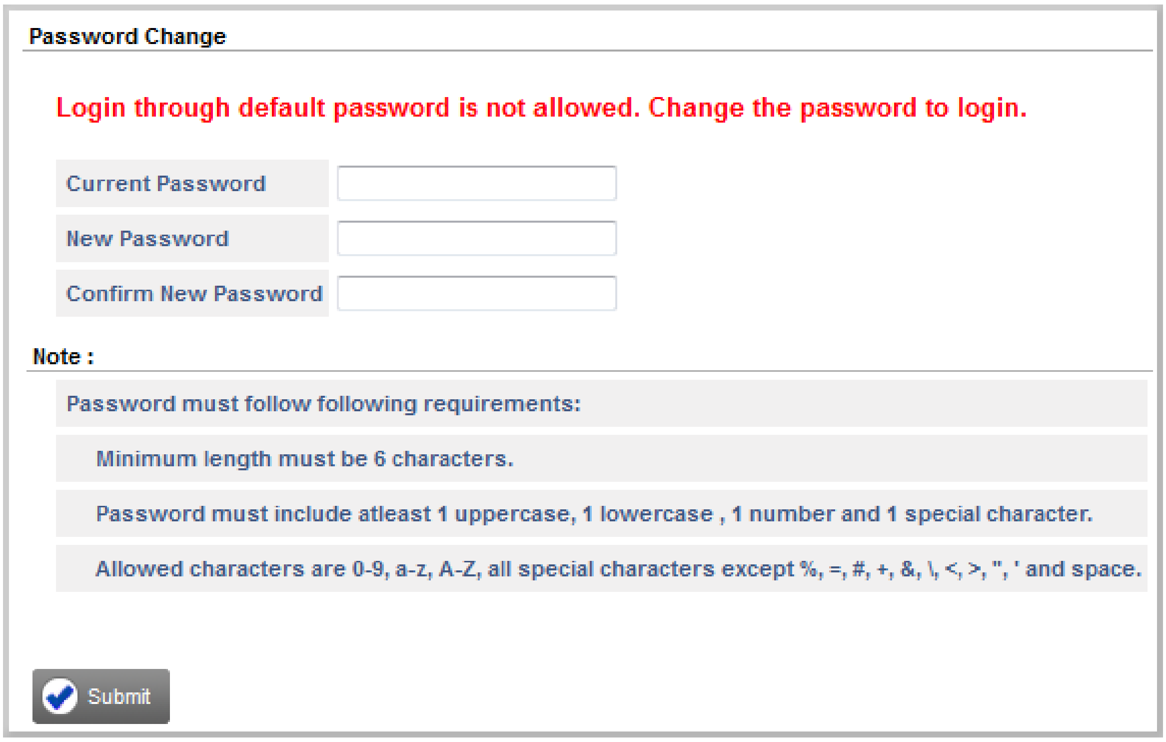
- In **Login Password**, enter ***1234***, the default Password.



- Click the **Login** button.

Before you start configuring the system, if you wish to view or download the SARVAM UMG Quick Start/ SARVAM UMG User Card or other product related documents, you can click or scan the QR Code present on the login page of Jeeves.

- You are prompted to change the default password.



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

- Enter the **New Password**. All ASCII characters (except Percentage **%**, Hash **#,** Equal to **=,** Plus **+**, And **&**, Backslash ****, Less than **<,** Greater than **>,** Apostrophe **'** **,** Double Quote **"** and **Space**) and digits 0 to 9 are allowed.

  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.

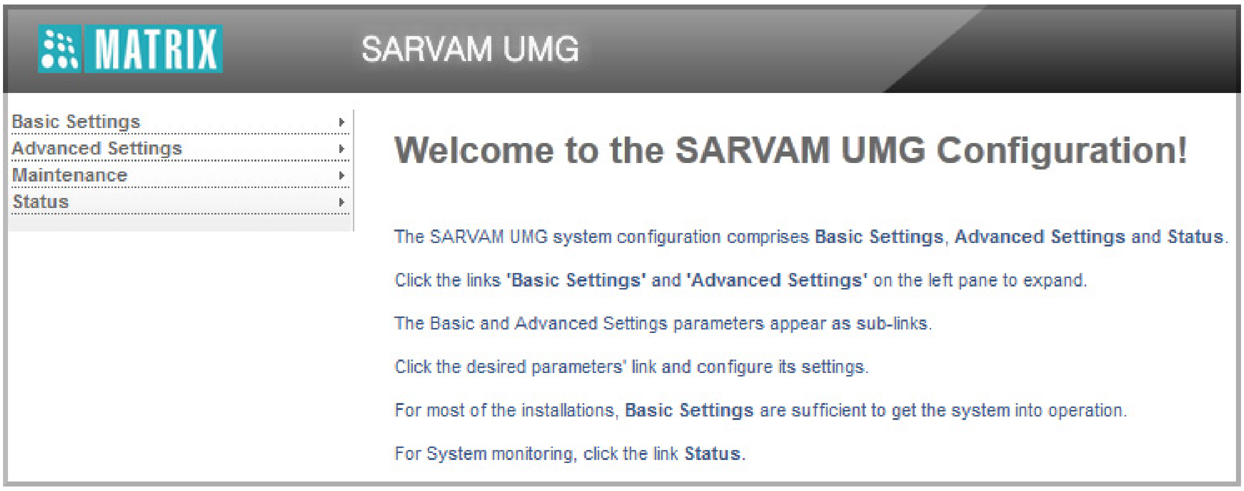
- Click **Submit.** You will be re-directed to the Login page again.

- In **Login Password**, enter the new password.

As this password is meant for restricting access to the SE mode, we strongly recommend you to:
- *Keep the password secret.*
- *Select a complex password that cannot be easily guessed.*
- *Change the password regularly. See* *"Login Password" for instructions.*

On successful login, the **Home** page of Jeeves opens.

The left navigation bar displays the links **Basic Settings**, **Advanced Settings, Maintenance** and **Status.**



Basic Settings break down the complexities of configuration and are sufficient to get your system into operation.

Advanced Settings enable you to configure the advanced features and facilities of SARVAM UMG.

Maintenance allows you to carry out system maintenance and monitoring activities like uploading/ upgrading firmware and configuration, system debug, system restart.

Status allows you to view the system details and the status of all the ports.

If you have not activated the license for SARVAM UMG Application, you may do so through the online process. For details, refer “How to activate your License”.

If the SARVAM UMG SME License is not activated and Demo Period is not started, the system will disconnect all the connected calls[8] from any port after 60 seconds.

You may now configure the Basic Settings of SARVAM UMG.

8. *Connected calls means where speech is connected between the calling party port and the called party port even if the called party port is not matured.*

CHAPTER 8

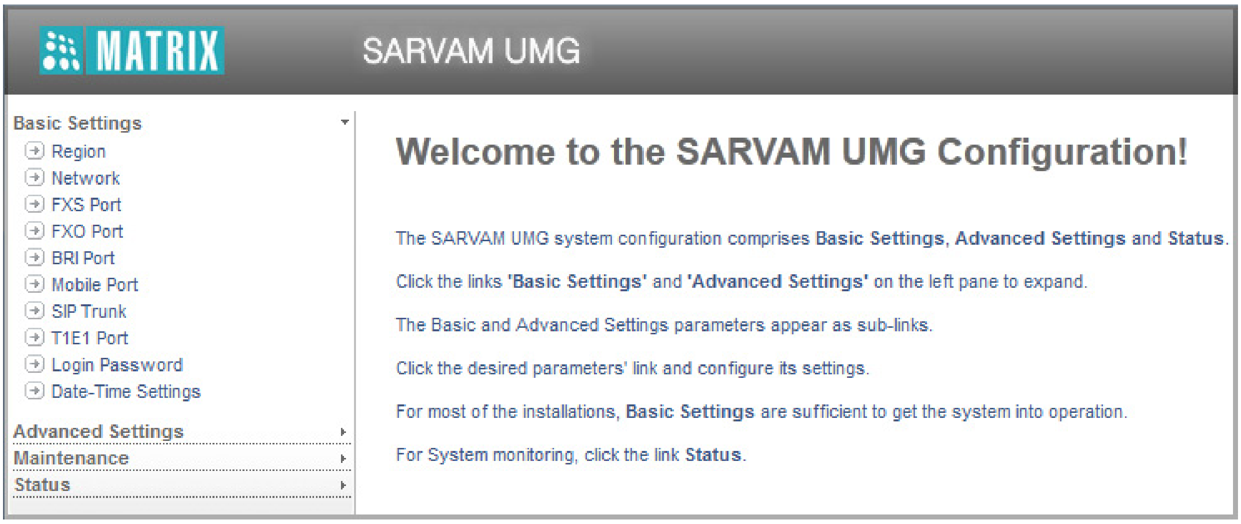
Basic Settings

The Basic Settings lets you configure the basic parameters of SARVAM UMG. Configuring these, you will be able to operate the system efficiently.

To configure Basic Settings,

- Click the **Basic Settings** link to expand.

  The links to the different basic parameters appear on the left navigation bar.



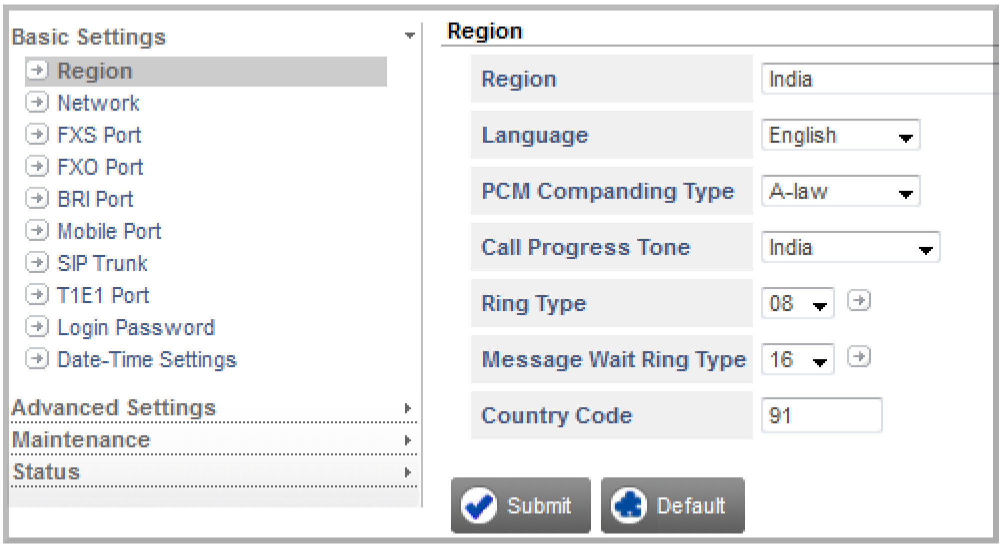
- Click the sublink of the required parameter: **Region**, **Network, FXS Port**, **FXO Port**, **BRI Port, Mobile Port**, **SIP Trunk**, **T1E1 Port, Login Password** and **Date -Time Settings**.
- The selected parameter page opens.
  - Click **Expand** to expand a link and display all parameters under the link.
  - Click **Collapse** to collapse a link and hide all parameters under the link.
  - Click **Settings** to configure / edit the settings of a parameter further.
  - Click the **Submit** button to save changes made on the page.
  - Click the **Default** button to assign factory set values to all the parameters on the page.

- Click the **Add** button to add a new record.
- Click the **Delete** button to delete a record.
- Click the **Close** button to exit a window.
- Click **Logout** to end the login session and exit Jeeves. You will return to the login page of Jeeves.

Region

To configure Region and other region specific parameters,

- Click the **Region** link.



Region

- In **Region**, select the name of the country where SARVAM UMG is installed. Default: India.

  When you change Region, an alert message will appear on the screen **"Changing Region shall assign default values to all parameters of the system. Do you want to continue?"** Click OK. All country specific parameters will be assigned default values. See "Default Region Table" in the Appendix for country specific default values.

Language

- In **Language**, select the language in which you want the pages of the Jeeves to be presented.

  SARVAM UMG supports various languages — English, Italian, Spanish, French, German, and Portuguese. Default: English.

  After selecting the Language, you need to login again. The Jeeves will now appear in the language you selected.

You can also select a Language of your choice on the login page of Jeeves. The language you select will be applied for the current session only.

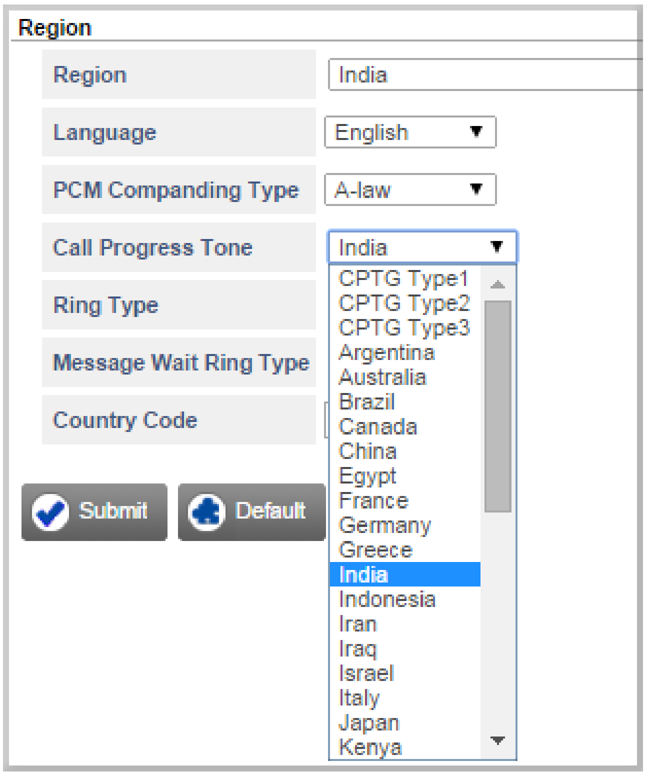
PCM Companding Type

- SARVAM UMG automatically sets the **PCM Companding Type** according to the Region you select. You can change the PCM Companding Type—A-law or µ–law as per your requirement. Default: A-law (for India).

Call Progress Tones

- Select **Call Progress Tone**. SARVAM UMG supports country specific Call Progress Tone Generation (CPTG) to simulate the same tones of the local PSTN to which it is connected. The Call Progress Tones supported by SARVAM UMG for different countries is presented in the *Appendix*. For details, see "Call Progress Tones".

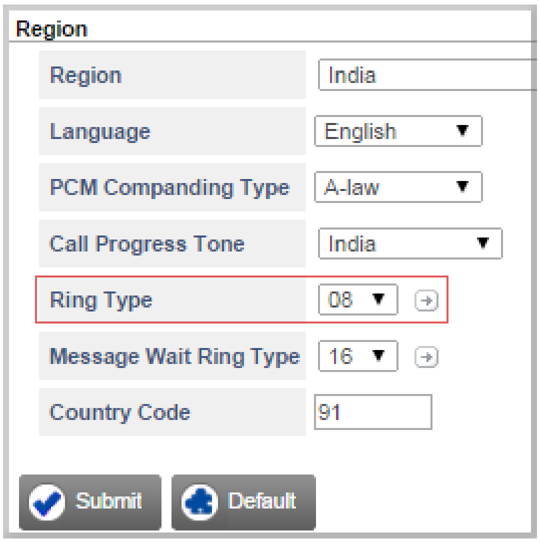
  - To match the call progress tone of the country where SARVAM UMG is installed, select the Country accordingly. Default: India



Ring Type

- SARVAM UMG automatically sets the **Ring Type** according to the Region you select. You can change the Ring Type as per your requirement. Default: 08 (India).

- To configure the Ring Type, click **Settings**.



The **Ring Type** table opens.

Ring Type

Ring Type	Ring Cadence						Supported Country
	ON Time 1 (msec)	OFF Time 1 (msec)	ON Time 2 (msec)	OFF Time 2 (msec)	ON Time 3 (msec)	OFF Time 3 (msec)	
1	Infinite						
2	750	750	0	0	0	0	
3	500	1500	0	0	0	0	
4	750	2250	0	0	0	0	
5	1500	500	0	0	0	0	
6	1000	4000	0	0	0	0	Brazil, Greece, Italy, Netherland, Switzerland, Finland, Germany
7	2000	4000	0	0	0	0	Egypt, USA, Canada, Namibia
8	400	200	400	2000	0	0	Australia, India, Singapore, South Africa, UK, Ireland, Malaysia
9	400	200	400	200	400	2000	
10	1000	2000	0	0	0	0	Japan
11	1000	3000	0	0	0	0	China, korea, Russia, Belgium, Taiwan
12	1000	5000	0	0	0	0	Portugal, Sweden
13	1500	3000	0	0	0	0	Spain
14	1500	3500	0	0	0	0	France
15	2000	3000	0	0	0	0	Israel, New Zealand, Poland, Thailand, UAE, Czechia, Norway, Hongkong, Austria, Hungary, Slovakia
16	3500	5500	790	1100	0	0	

Close

The table presents you with the number of **Ring Types**, **1** to **16**, supported by the system, the **Ring Cadence** of each Ring Type, and the countries where each Ring Type is supported.

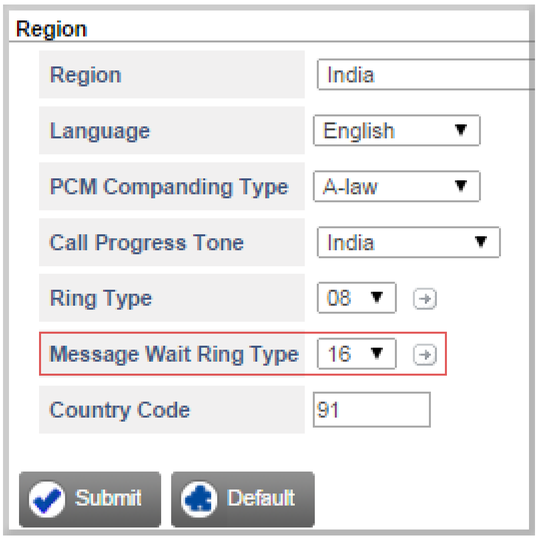
- Note the Ring Type number that you wish to assign.
- Close the window to return to the **Region** page.
- In **Ring Type**, select the desired Ring Type number.

Message Wait Ring Type

- This parameter is related to the “Message Wait Indication on SIP Trunks” feature. When you select *Message Wait Notification* type as *Ring* on the FXS Port, a Short, Fast ring is played to indicate the arrival of a new message.

  If required, you may change the **Message Wait Ring Type.** Default: 16 for all Regions.

  - To configure the Message Wait Ring Type, click **Settings**.



The **Ring Type** table opens. SARVAM UMG supports 16 different Ring Types.

Ring Type

Ring Type	Ring Cadence						Supported Country
	ON Time 1 (msec)	OFF Time 1 (msec)	ON Time 2 (msec)	OFF Time 2 (msec)	ON Time 3 (msec)	OFF Time 3 (msec)	
1	Infinite						
2	750	750	0	0	0	0	
3	500	1500	0	0	0	0	
4	750	2250	0	0	0	0	
5	1500	500	0	0	0	0	
6	1000	4000	0	0	0	0	Brazil, Greece, Italy, Netherland, Switzerland, Finland, Germany
7	2000	4000	0	0	0	0	Egypt, USA, Canada, Namibia
8	400	200	400	2000	0	0	Australia, India, Singapore, South Africa, UK, Ireland, Malaysia
9	400	200	400	200	400	2000	
10	1000	2000	0	0	0	0	Japan
11	1000	3000	0	0	0	0	China, korea, Russia, Belgium, Taiwan
12	1000	5000	0	0	0	0	Portugal, Sweden
13	1500	3000	0	0	0	0	Spain
14	1500	3500	0	0	0	0	France
15	2000	3000	0	0	0	0	Israel, New Zealand, Poland, Thailand, UAE, Czechia, Norway, Hongkong, Austria, Hungary, Slovakia
16	3500	5500	790	1100	0	0	

Close

- Note the Ring Type number that you wish to assign.
- Close the window to return to the **Region** page.
- In **Message Wait Ring Type**, select the Ring Type number of your choice.

Country Code

- SARVAM UMG automatically sets the **Country Code** according to the Region you select. You can change the Country Code as per your requirement. Default: 91 (India).

  If you have kept **Remove Country Code from CLI received** check box enabled in the *System Parameters*, the system will remove the Country Code configured here from the CLI received on the source port.

- Click the **Submit** button to save.

Network

SARVAM UMG may be installed typically, in a Public IP Network or in a Private network, behind a NAT Router.

When SARVAM UMG is installed in a Public IP Network,

- the WAN Port of SARVAM UMG is connected to a Broadband Router/Modem.
- Public IP is assigned to the WAN Port.

When SARVAM UMG is installed in a Private Network, behind a NAT Router,

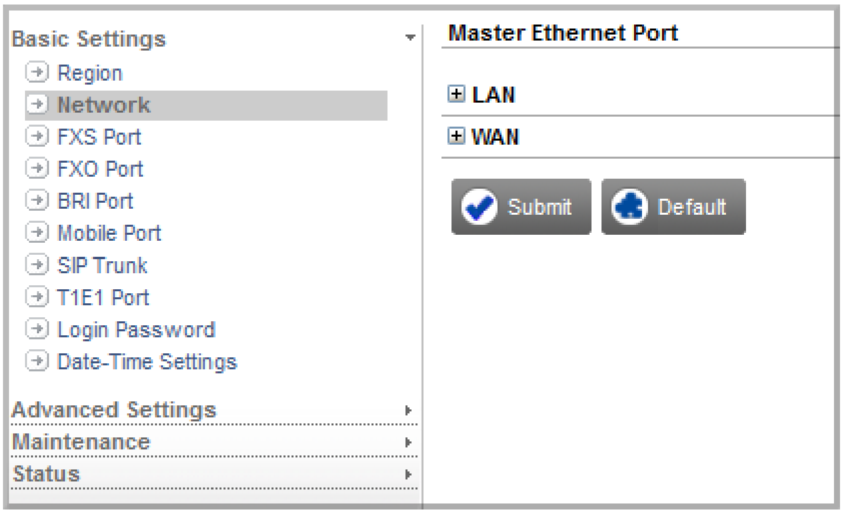
- the WAN Port of SARVAM UMG is connected to the LAN Switch/Hub.
- Private IP is assigned to the WAN Port.

Depending on your installation scenario, configure the Network Port Parameters.

To configure Network parameters,

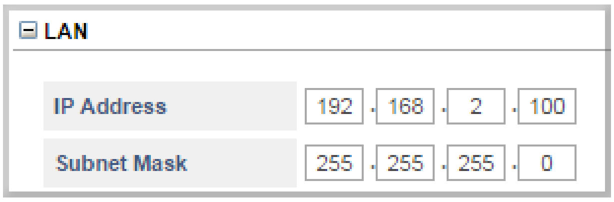
- Click the **Basic Settings** link to expand.

- Click the **Network** link. The Network Parameters page opens.



LAN

- Click **LAN** to expand.

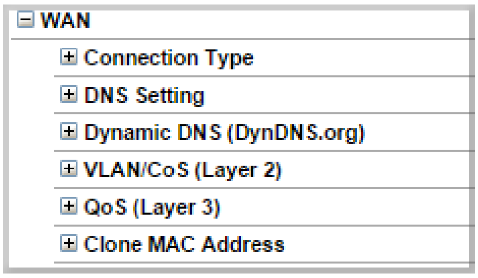


- In **IP Address**, the current IP Address of the LAN Port is displayed. Default: **192.168.2.100**
- In **Subnet Mask**, the current Subnet Mask of the LAN Port is displayed. Default: **255.255.255.000**
  If required, you may change the LAN Port IP Address and Subnet Mask.

When your SARVAM UMG is installed in a Private Network, make sure the LAN Port and the WAN Port are connected in different subnets.

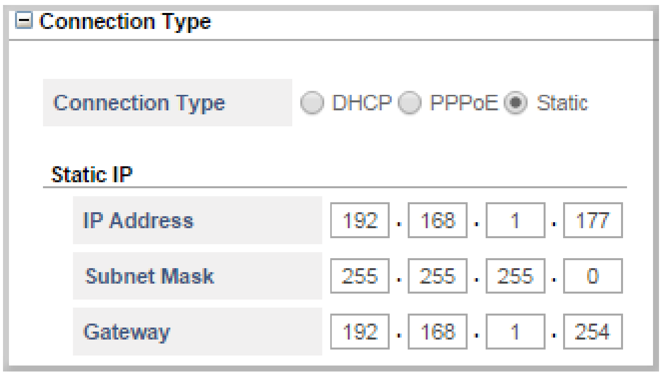
WAN

- Click **WAN** to expand.



Connection Type

- Click **Connection Type** to expand.



- Select the network connection type, that is, the IP Addressing Scheme used by your network to assign the IP address to the WAN Port: Static, DHCP, PPPoE. Default: Static.
  - Select **Static**, if your network uses Static IP addressing and configure the following parameters.

    - In **IP Address**, enter the IP Address you obtained from your Network Administrator for the WAN Port of SARVAM UMG. Make sure that the IP Address does not conflict with that of any other device on the LAN. Default: 192.168.1.100

    - In **Subnet Mask**, enter the Subnet Mask you obtained from your Network Administrator for the WAN Port. Default: 255.255.255.0

    - In **Gateway**, enter the IP Address of the Router's LAN Interface as the Default Gateway IP Address. Default: 192.168.1.254

  - Select **DHCP,** if your network uses DHCP Addressing. Whenever SARVAM UMG is restarted, the DHCP server will dynamically assign an IP Address, Subnet Mask and Gateway Address to the WAN Port. You have to configure the Domain Name Server (DNS) Address only, if not already provided by your Internet Service Provider.

  - Select **PPPoE,** if your network uses PPPoE addressing. The PPPoE server will automatically assign an IP Address, Subnet Mask and Gateway Address to the WAN Port of SARVAM UMG. You need to configure the following parameters provided by your Internet Service Provider:

    - In **PPPoE User ID,** enter the User Name provided by the Internet Service Provider. The User ID can be a maximum of 64 characters.

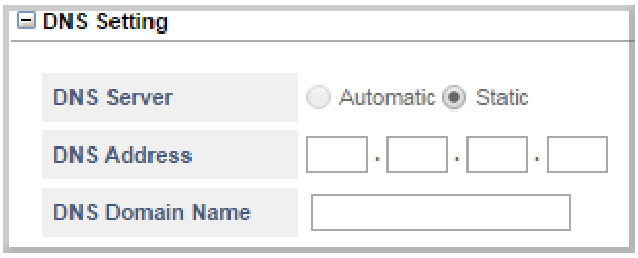
    - In **PPPoE Password,** enter the User Password provided by the Internet Service Provider. The password can be a maximum of 64 characters.

    - In **PPPoE Service Name,** enter the Service Name, if provided by your Internet Service Provider. The Service Name may be a maximum of 64 characters. If Service Name is not provided, leave this blank.

DNS Setting

Configure the Domain Name Server (DNS) settings as provided by your Internet Service Provider. You may consult your Network Administrator.

- Click **DNS Setting** to expand.



- Select **DNS Server** as **Automatic** or **Static** 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.

- In **DNS Address,** enter the DNS Server Address. In **DNS Domain Name**, enter the DNS Domain Name if provided to you by your Network Administrator.

- Select **Automatic** if:
  - your network uses DHCP or PPPoE IP Addressing.
  - the DHCP/ PPPoE server of your network assigns the DNS Address automatically.

Dynamic DNS (DynDNS.org)

Dynamic DNS (DDNS) is a service that maps internet domain names to IP addresses. DDNS Service Provider provides the host name/domain name to the internet devices and also embeds DDNS client in the internet device. By doing so, whenever a new IP Address is assigned to the internet host, the DDNS client running in the internet host updates its new IP address in the Dynamic DNS server.

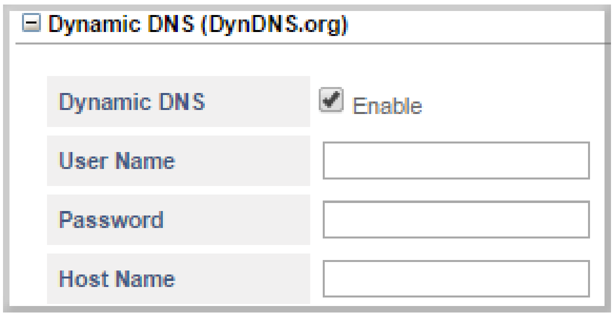
When the WAN Port of SARVAM UMG is assigned a dynamic IP, its new IP Address needs to be updated regularly with the various devices or networks which utilise the WAN Port settings to function. Dynamic DNS resolves this by mapping a domain name to the WAN Port IP Address, which SARVAM UMG can update in the Dynamic DNS Server.

Once the IP Address of the system is updated in the DNS server, any caller on the IP network can reach the system by dialing the host name/domain of the system.

SARVAM UMG supports Dynamic DNS Server client of the Service Provider Dynamic DNS.org. To use this service, you must first register with DynDNS.org and then do the following:

- Click **Dynamic DNS (DynDNS.org)** to expand.

- Select the **Dynamic DNS Enable** check box.



- Enter the **User Name** you created on DynDNS.org. The name can be a maximum of 40 characters.

- Enter the **Password** you created for the User Name on DynDNS.org. The password can be a maximum of 24 characters.

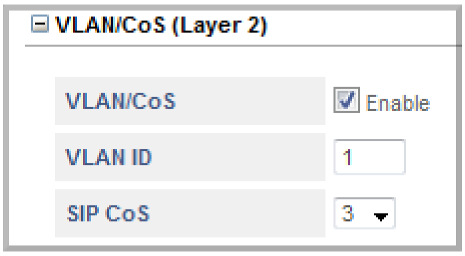
- Enter the **Host Name** you created on the DynDNS.org here. The Host Name can be a maximum of 40 characters.

VLAN/CoS

If SARVAM UMG is connected in a VLAN, configure the **VLAN/CoS**. This parameter enables the SARVAM UMG 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[9].

- Click **VLAN/CoS (Layer 2)** to expand.

- Select the **VLAN/CoS** check box to enable VLAN ID tagging on all packets generated by the system. Default: Disabled.

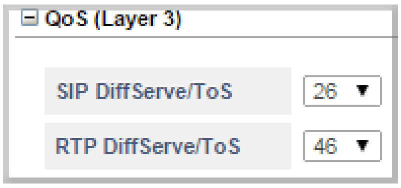


- Enter the **VLAN ID** that you have assigned to the VLAN in which the SARVAM UMG is connected. Valid range is 0 to 4094. Default: 1.

- For **SIP CoS**, define the CoS (priority) bits which will be added in all SIP packets. Valid range is 0 to 7. Default: 3

QoS (Layer 3)

- Click **QoS (Layer 3)** to expand.



- SARVAM UMG will send all SIP messages using SIP QoS setting, enter the **SIP DiffServe/ ToS** as per your requirement. Valid range is 00 to 63. Default: 26.

- SARVAM UMG will send all the RTP packets with RTP QoS setting, enter the **RTP DiffServe/ ToS** as per your requirement. Valid range is 00 to 63. Default: 46.

Clone MAC Address

- Click **Clone MAC Address** to expand.

- If you want to clone the MAC address, select the **Clone MAC Address** check box.

In **MAC Address (Cloned)**, enter the desired MAC address you want to clone in hexadecimal format, e.g. 00:50:c2:55:b0:10.

9. *The IEEE 802.1P standard allows Layer2 switches to prioritize the traffic, thus providing Quality of Service (QoS), i.e. 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.*

Restoring Default LAN IP Address

To restore the Default LAN IP Address by changing the Jumper **(J1)** settings on the CPU Card,

- Switch off the power supply.
- Remove the CPU Card from the slot.
- Locate Jumper **(J1)** on the PCB of the CPU Card.
- Change the position of the Jumper from ***BC*** to ***AB*** (that is, from 'Normal' to 'Reset Login Password')
- Reinsert the CPU Card into the slot.
- Switch ON the system and wait for the system to initialize.
- Switch OFF the system and remove the CPU Card from the slot.
- Change the Jumper position from ***AB*** to the original position ***BC***.
- Insert the CPU Card back into the slot.
- Switch ON the system.

The LAN IP Address will be restored to default, **192.168.2.100**

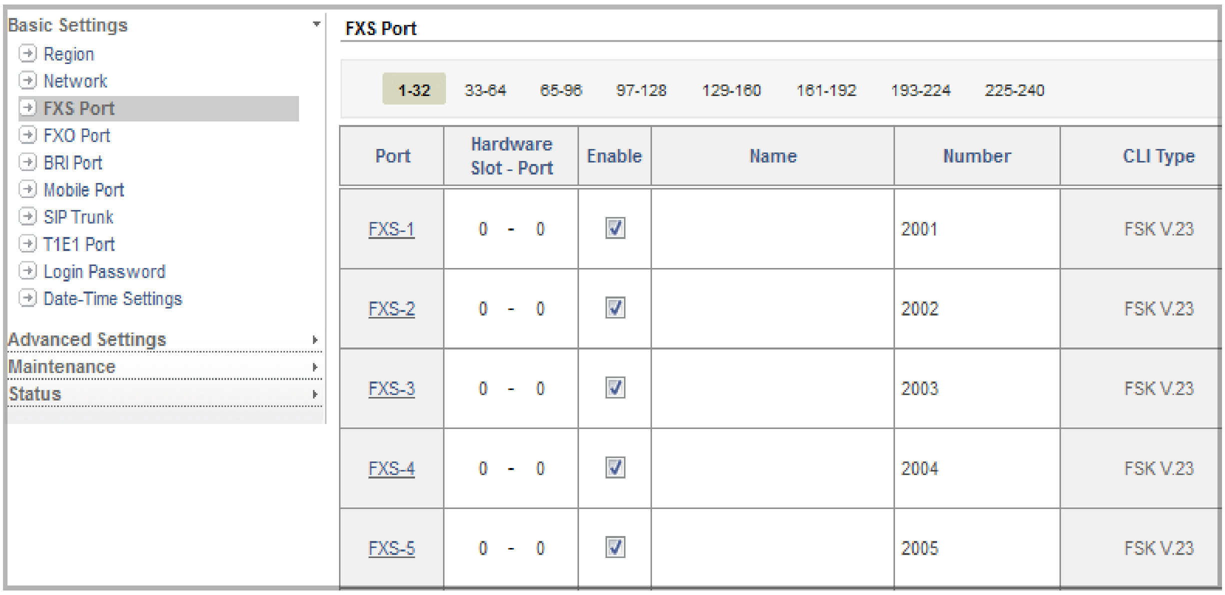
*When you restore the default LAN IP Address (***192.168.2.100***) by changing the Jumper position, a few other parameters will also be set to default. See* *“Restoring Default Settings by changing the Jumper Position”* *for details.*

FXS Port

SARVAM UMG supports FXS Ports to which you can connect a Gateway or any standard telephone instrument.

To configure the parameters of the FXS Port,

- Click the **Basic Settings** link to expand.
- Click the **FXS Port** link.



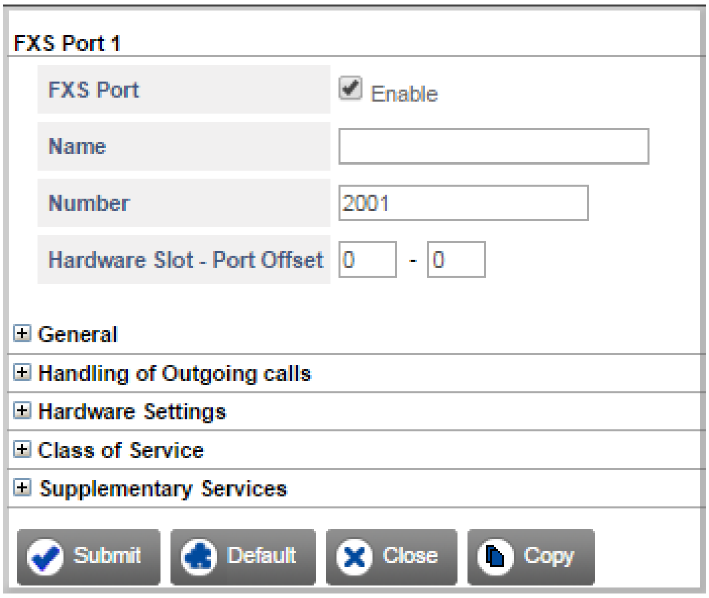
The FXS Port page displays the following parameters:

- **Port:** It displays the FXS Port numbers. Click on the desired FXS Port number to configure the Port Parameters.
- **Hardware Slot-Port:** The SARVAM UMG can automatically detect and assign the Hardware Slot and Port numbers to the FXS software ports. However, if required you may change the Hardware Slot and Port assigned to the FXS software port. In this case, enter the desired Hardware Slot and Port number.
- **Enable:** Keep the **FXS Ports** enabled. Clear the FXS Port **Enable** check box, only if you do not want to use the respective port. Default: Enabled.
- **Name:** Assign a Name to the FXS Port for identification. The Name can be a maximum of 24 characters.
- **Number:** Assign a Number to the FXS Port for identification. The Number can be a maximum of 16 digits.
- **CLI Type:** It displays the type of CLI you select — DTMF, FSK V2.3, FSK Bellcore.
- **Outgoing Call Routing:** It displays the Outgoing Call Routing Method you select.

To configure the FXS Port parameters,

- Click **FXS-1**.

The **FXS Port 1** window opens.



- Keep the **FXS Port** check box enabled.

  Clear the **FXS Port Enable** check box, only when you do not want to use this FXS Port. Default: Enabled.

- You may assign a **Name** to the FXS Port. When a call is made through this port, the name you assigned will appear on the phone of the called party, if the phone supports CLI display.

  The name you assign may consist of a maximum of 24 characters. Default: Blank.

- You can assign a **Number** to the FXS Port. When a call is made through this port, the number you assigned will appear on the phone of the called party, if the phone supports CLI display.

  The number you assign to the FXS Port can have a maximum of 16 characters. Valid characters are 0 to 9, *, # and +. Default: 2001.

- Configure the **Hardware Slot - Port Offset**, if required. 'Slot' is the number of the universal slot in which the SLT Card is inserted. 'Port' is the number of the SLT (FXS) hardware port on which the telephone instrument is connected.

  The SARVAM UMG can automatically detect and assign the hardware slot and port numbers to the SLT software ports.

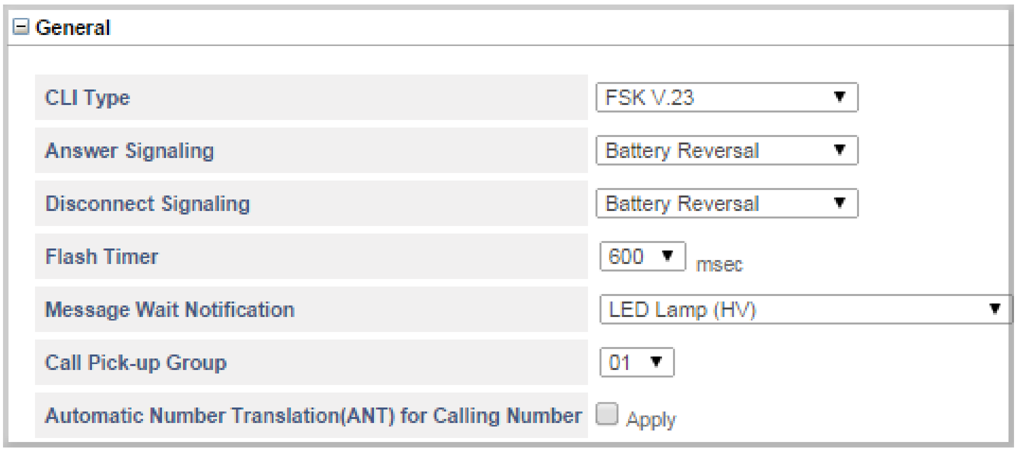
  For example: if you have inserted the SLT8 Card in Slot number 02 of SARVAM UMG, the system will assign the hardware slot 02 and port numbers 01-08 to the SLT Software Ports from 001 to 008 respectively.

  However, if required, you may change the Hardware Slot and Port assigned to the FXS software port. In this case, enter the desired Hardware Slot and Port number.

  If you want to de-assign the Hardware Slot and Port, enter '00' in both fields. By default, Hardware Slot-Port is 00–00.

General

- Click **General** to expand.



- Select the appropriate **CLI Type**, according to the CLI Type supported by the telephone instrument/ Gateway connected to the FXS Port.

  SARVAM UMG supports three signaling protocols for CLI on the FXS Port — DTMF, FSK V.23 and FSK BellCore. Default: FSK V.23

- Select the appropriate **Answer Signaling** Type on the FXS Port.

  Answer Signaling is a signal generated on the FXS Port to indicate that the called party has answered (call maturity).

  - Select **None,** if no answer signaling is to be generated on the FXS Port.
  - Select **Battery Reversal**, if answer signaling is to be generated in the form of Battery Reversal on the FXS Port.

    Default: Battery Reversal

- Select the appropriate **Disconnect Signaling** Type on the FXS Port.

  A Disconnect Signal is the signal generated on the FXS port to indicate that the called party has disconnected the call.

  - Select **None**, if no signaling is to be generated on the FXS Port for call disconnection.
  - Select **Battery Reversal**, if call disconnection is to be signaled in the form of Battery Reversal.
  - Select **Open Loop Disconnect**, if call disconnection is to be signaled in the form of Open Loop Disconnect signal. If you select this option, you must configure the Open Loop Disconnect Timer.
    - Set the duration of **Open Loop Disconnect Timer** as per your requirement. Valid range is 001 to 999 msec. Default: 500 msec.

    Default: Battery Reversal.

- Set the duration of the **Flash Timer**. This is the time for which Flash will be detected on the FXS Port. SARVAM UMG uses this event to activate various features — Call Hold, Call Transfer, etc. Valid range is 100 to 900 msec. Default: 600 msec.

- If you have subscribed to *Message Wait Indication* for the voicemail service from your ITSP, and have selected this FXS Port as the destination for receiving Message Wait Indication[10], you may select the desired type of **Message Wait Notification** from the following options.

  - Select **Stuttered Dial Tone**, if you want new message indication in the form of a stuttered dial tone, whenever the user picks up the phone connected to the FXS Port.

  - Select **LED Lamp (HV),** if the phone connected to the FXS Port is equipped with a 'Message Wait' lamp and you want new messages to be indicated on this LED lamp using High Voltage.

  - Select **Ring**, if you want the arrival of a new message to be indicated by the *Message Wait Ring* (a Short, Fast ring).

    You can select a different Ring Type to indicate message wait. For instructions, see "Message Wait Ring Type".

    You can also set the duration for which the ring is to be played (Ring Timer), the number of times the ring is to be played (Ring Count) and the interval between rings (Ring Interval). For instructions, see "Message Wait" in "System Parameters".

  - Select **LED Lamp (FSK)**, if the phone connected to the FXS Port is equipped with a 'Message Wait' lamp and you want new messages to be indicated on this LED lamp using FSK CLI.

  - Select **Stuttered Dial Tone + LED Lamp (HV),** if you want new message indication on the LED Lamp using High Voltage as well as in the form of a stuttered dial tone, when the user picks up the phone connected to the FXS Port.

  - Select **Stuttered Dial Tone + LED Lamp (FSK)**, if you want new message indication on the LED Lamp using FSK CLI as well as in the form of a stuttered dial tone, when the user picks up the phone connected to the FXS Port.

  Default: LED Lamp (HV)

  Whenever a new message arrives in the Mailbox of the SIP Trunk, SARVAM UMG gives notification to this (destination) FXS Port according to the type of *Message Wait Notification* you select.

  To know more about this feature and the Notification options, see "Message Wait Indication on SIP Trunks".

- If you want to allow the Call Pick-up feature on this FXS Port, assign the FXS Port to a **Call Pick-up Group**[11].

  Call Pick-up feature allows the FXS Port user to answer calls that are ringing on any FXS Port in the same Call Pick-up Group, by dialing an access code. To know more about this feature, see "Call Pick-up". By default, all the FXS Ports are assigned to group 1.

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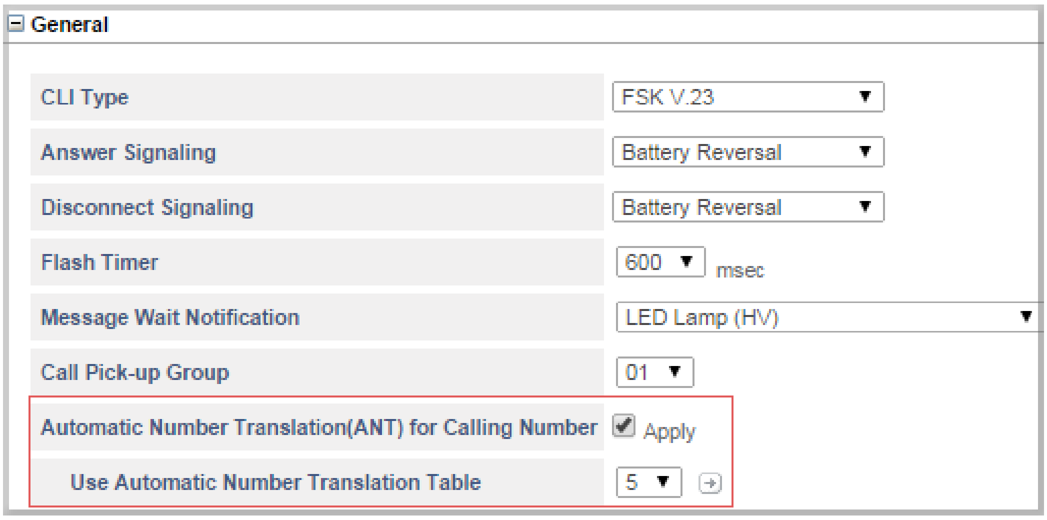
10. *You have selected the number of this FXS Port for the **Send Message Notification on** parameter, under **MWI Parameters** you configured on the SIP Trunk.*
11. *The number of Call Pick-up Groups will vary according to the number of FXS Ports present in the system.*

Make sure that the Call Pick-up feature is enabled in the *Class of Service* of all the FXS Ports in the group. See “Class of Service”.

'0' is used to de-assign an FXS Port from a Call Pick-Up Group.

- You can apply Automatic Number Translation (ANT) logic on CLI number when call is to be placed on the FXS Port.

  - To apply ANT logic on the Calling Numbers, select the **Automatic Number Translation (ANT) for Calling Number** check box. Default: Disabled.

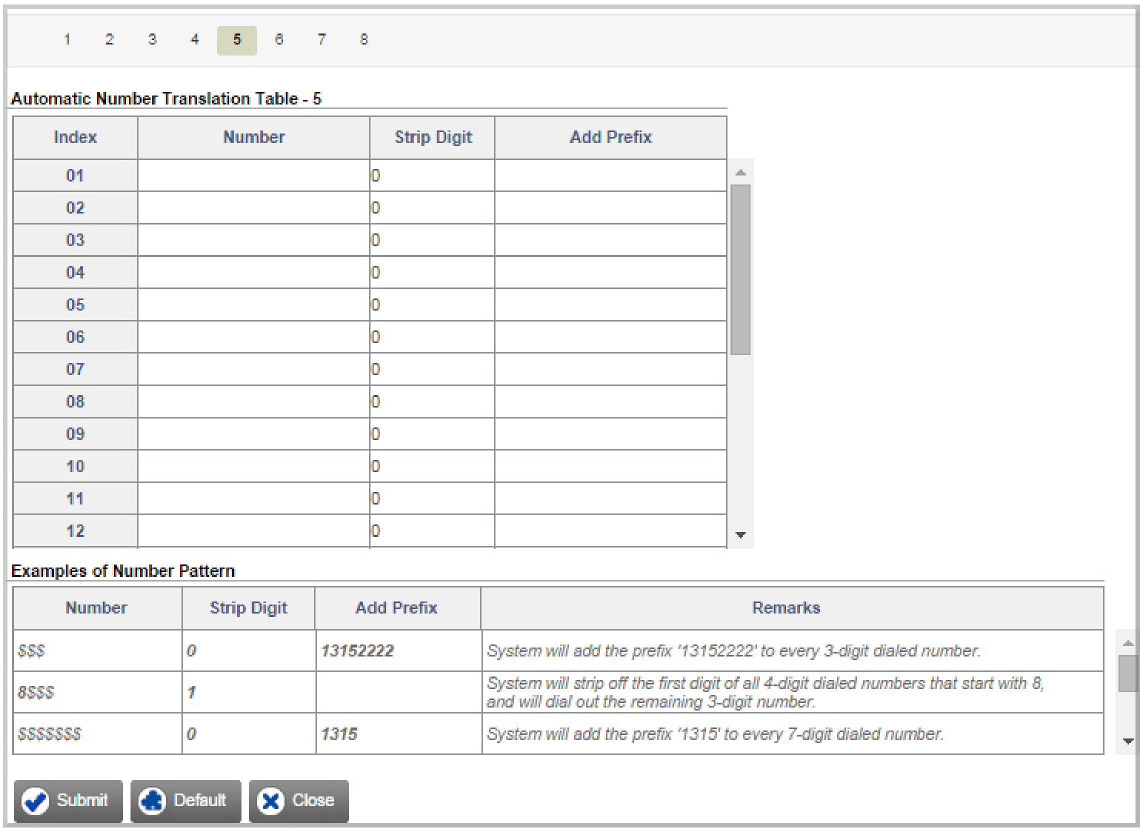


  - In **Use Automatic Number Translation Table**, select the ANT Table number you have configured for the Calling Numbers. Default: Table 5.

    If you have not configured the Automatic Number Translation Table,

    - Click **Settings** .

The **Automatic Number Translation Table** window opens.



Index	Number	Strip Digit	Add Prefix
01		0	
02		0	
03		0	
04		0	
05		0	
06		0	
07		0	
08		0	
09		0	
10		0	
11		0	
12		0	

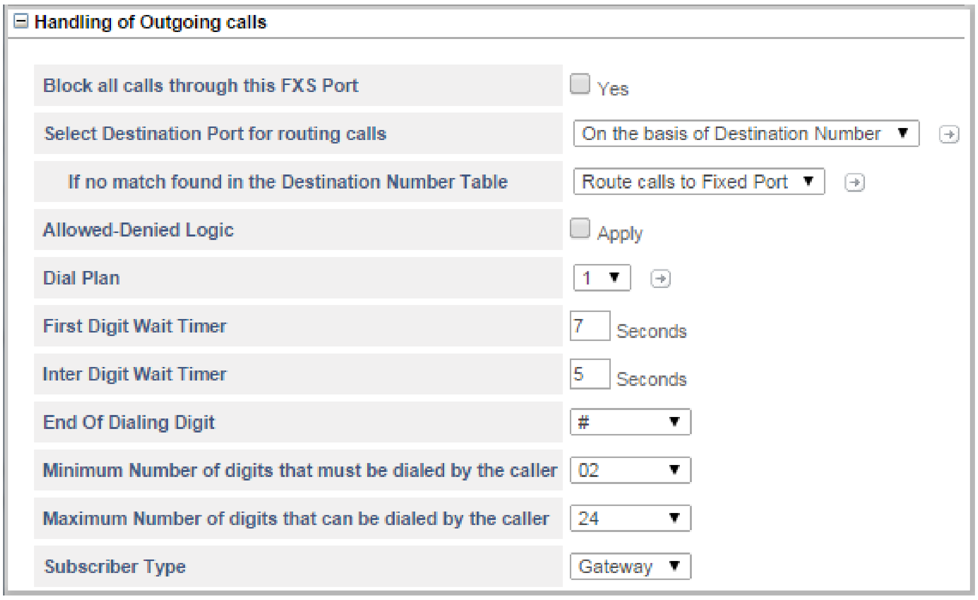
Examples of Number Pattern

Number	Strip Digit	Add Prefix	Remarks
$$$	*0*	*13152222*	*System will add the prefix '13152222' to every 3-digit dialed number.*
8$$$	*1*		*System will strip off the first digit of all 4-digit dialed numbers that start with 8, and will dial out the remaining 3-digit number.*
$$$$$$$	*0*	*1315*	*System will add the prefix '1315' to every 7-digit dialed number.*

- You may configure the default Automatic Number Translation Table or any other Table. See “Automatic Number Translation (ANT)” to configure the ANT Table.
- Click **Submit** to save the ANT Table and close the window.
- Return to ANT parameter and assign the ANT Table you configured.
- Click **Submit**.

Handling of Outgoing Calls

- Click **Handling of Outgoing Calls** to expand.



- If you do not want to route calls through this port, select the **Block all calls through this FXS Port** check box. Default: Disabled.

Destination Port Determination

- In **Select Destination Port for routing calls**, select the method to be used for determining the Destination Port for routing calls from the FXS Port. You may select any one of these options:
  - Fixed
  - on the basis of Destination Number

  Default: on the basis of Destination Number

  Read further for instructions on selecting and configuring each of these destination port determination methods.

If the destination number to be dialed out is an IP Address, SARVAM UMG will not check the Destination Port Determination Method. Instead, it will route the call using the SIP Trunk / Group programmed for IP Dialing. (To know more, see the feature description "IP Dialing").

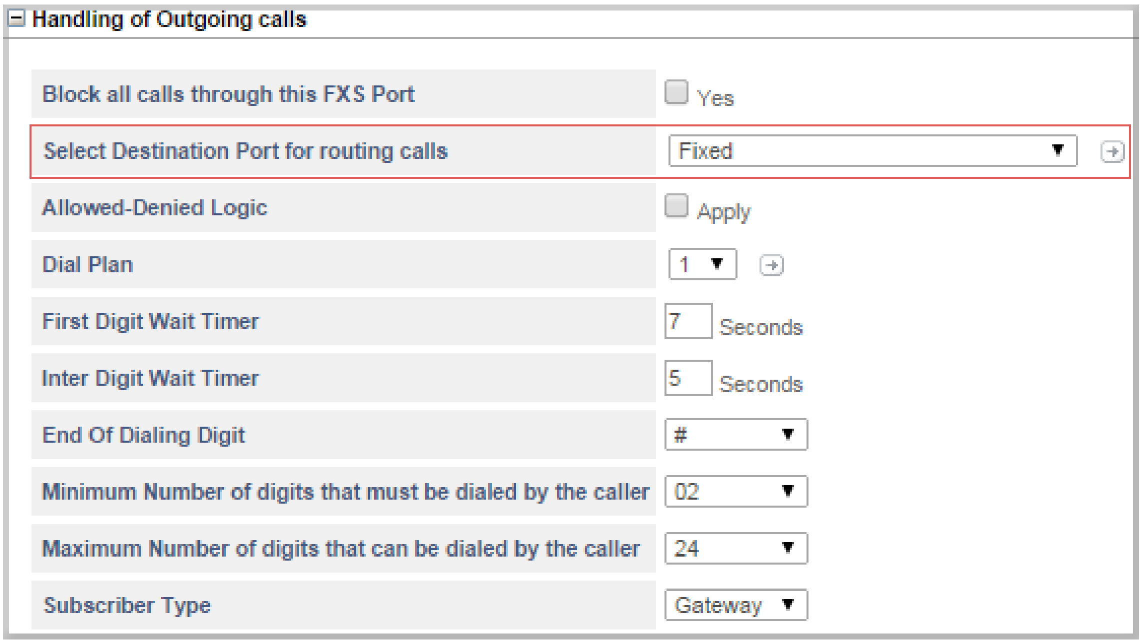
Fixed

In this method, outgoing calls made from the FXS Port are routed to a Fixed Destination Port, irrespective of the number dialed from the FXS Port.

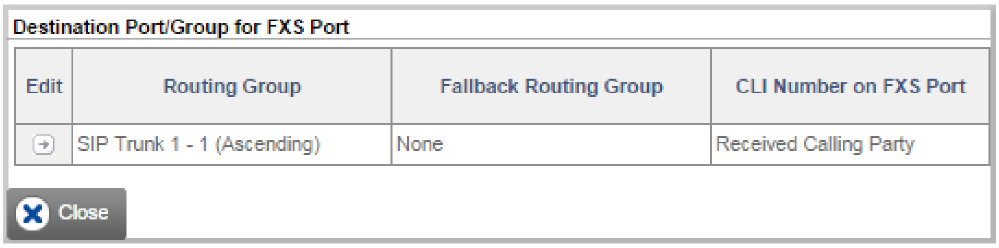
To apply this method, do the following:

- In **Select Destination Port for routing calls**, select **Fixed** option.

- Click **Settings** .



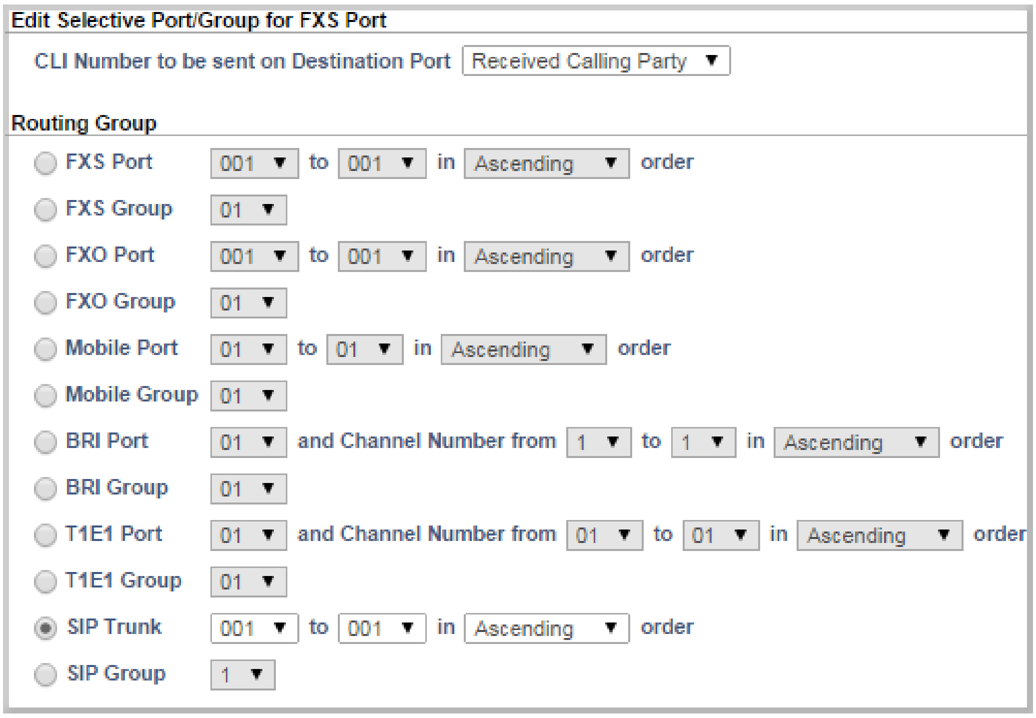
The **Destination Port/Group for FXS Port** window opens.



The default **Routing Group** and **Fallback Routing Groups** appear.

- If you wish to change the default Routing Group options, click **Edit** .

The **Edit Selective Port/Group for FXS Port** window opens.

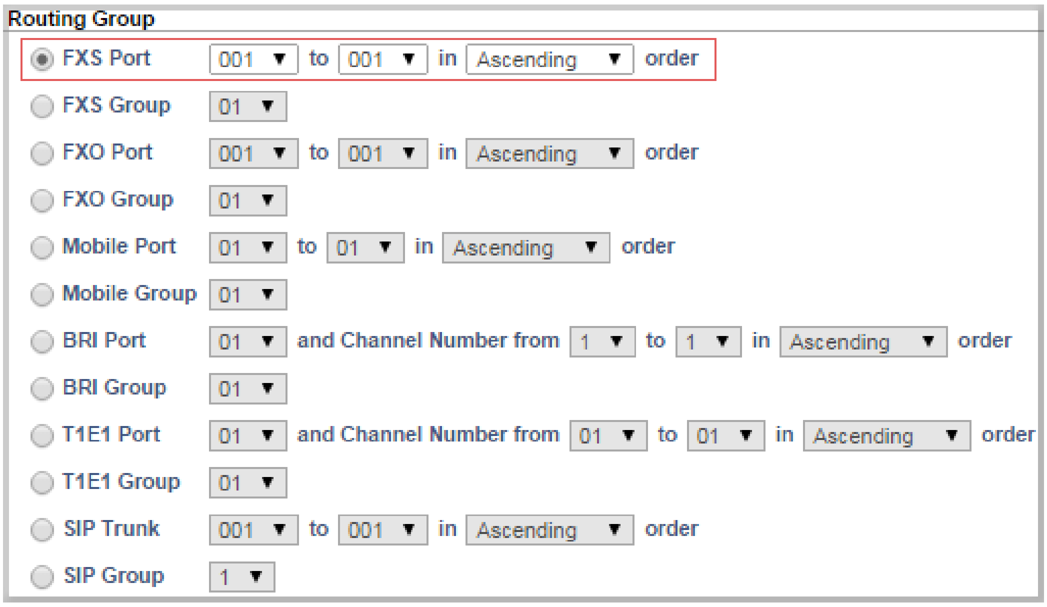


- Select the **CLI Number to be sent on Destination Port**. You can select Received Calling Party or Received Called Party. Default: Received Calling Party.

CLI to be sent on Destination Port is applicable when FXS Port/Group is determined as the Destination Port.

- Create the **Routing Group**.
  - To create a group of *sequential* **FXS Ports** as members,

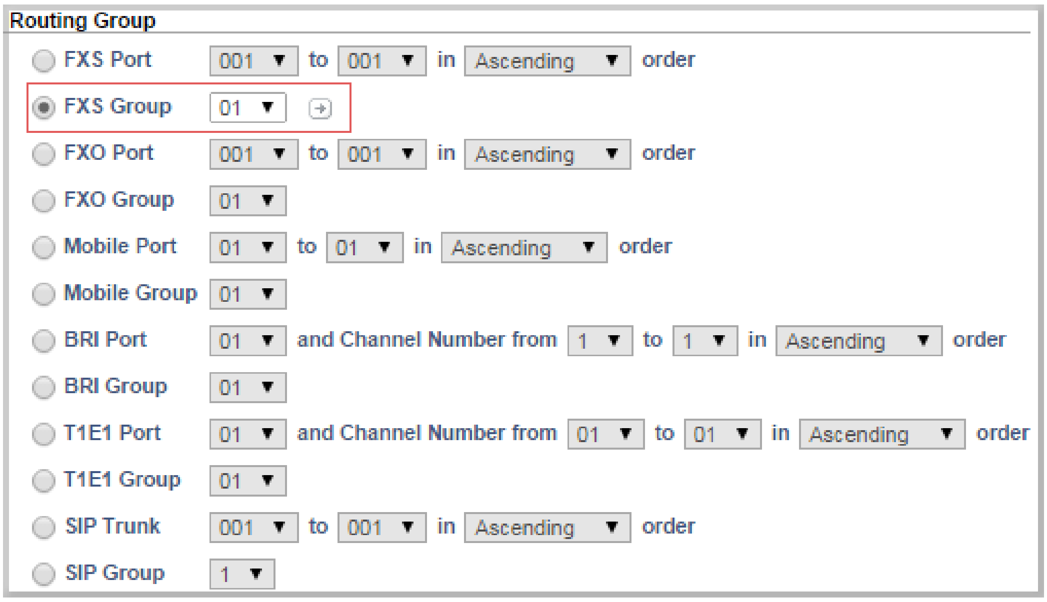
- Select the desired **FXS Port** numbers as members. Default:1.



- In **in - order**, select the order in which the system should hunt for a free member FXS Port to route the call.

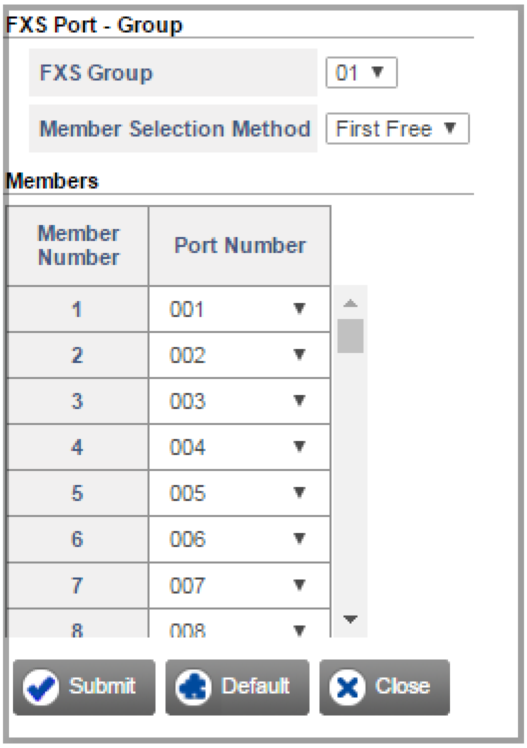
  Select **Ascending** to start hunting from the first to the last member FXS Port. Select **Descending** to start hunting from the last to the first member FXS Port. Default: Ascending.

- To create a group of *not-sequential* **FXS Ports** as members,
  - Select a **FXS Group**.



  - Select **FXS Group** number. Default:1.
  - Click **Settings**.

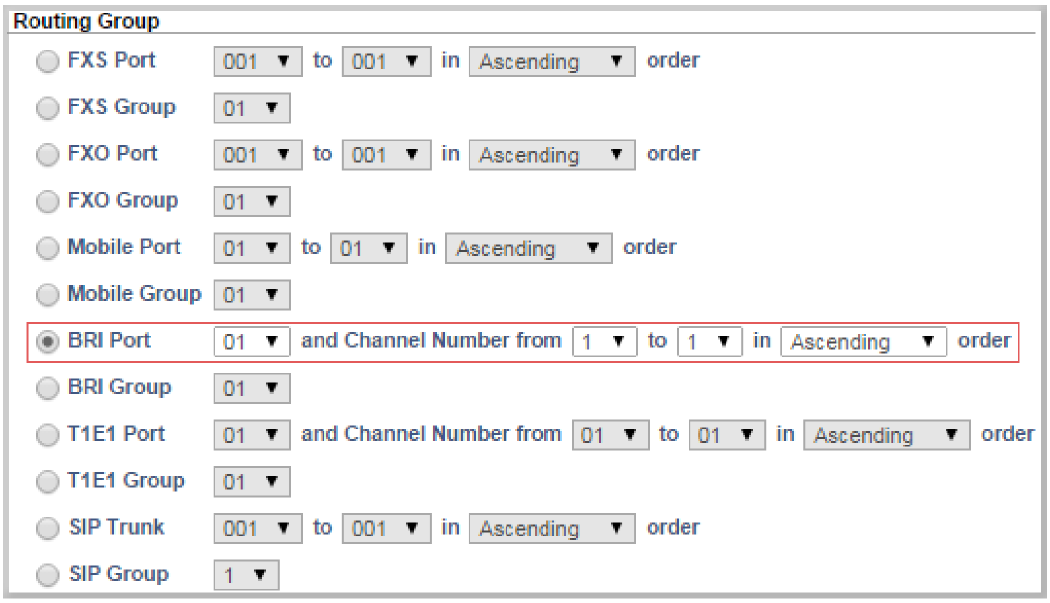
The **FXS Port - Groups** window opens.



- Create the FXS Group. For detailed instructions on creating groups, see the topic "Group" under *Advanced Settings*.

- Similarly, you can create a group of *sequential* and *not-sequential* FXO Ports, Mobile Ports and SIP Trunks.

- To create a routing group of *sequential* **BRI Channels** as members,

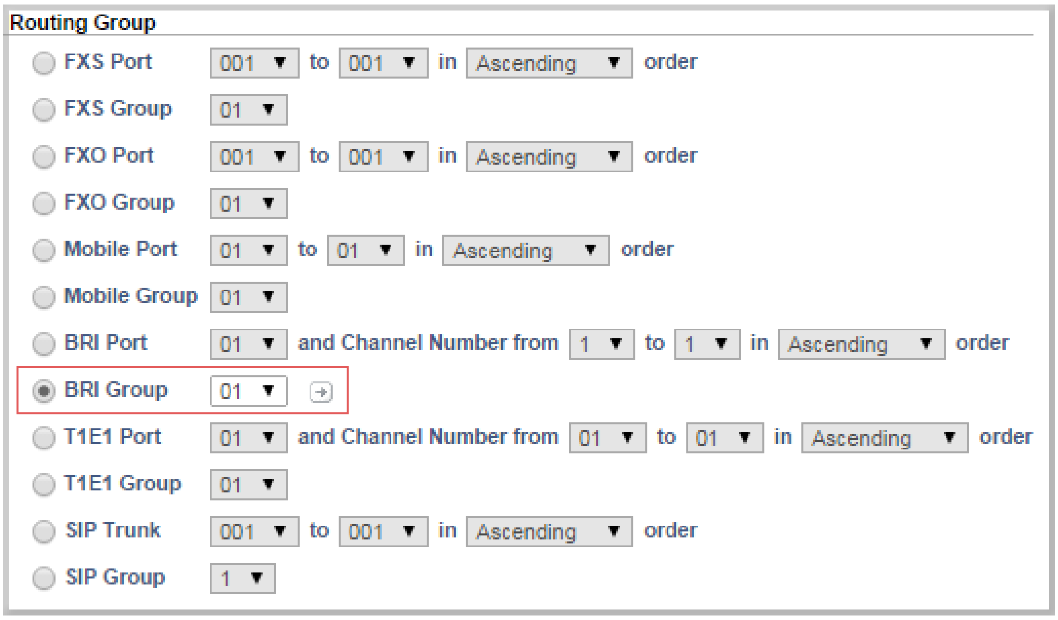


- Select the **BRI Port** Number. Default: 1.

- In Channel Number **From - to**, select the **Start Channel Number** and the **End Channel Number**, respectively.
- In **in - order**, select the order in which the system should hunt for a free member Channel to route the call.

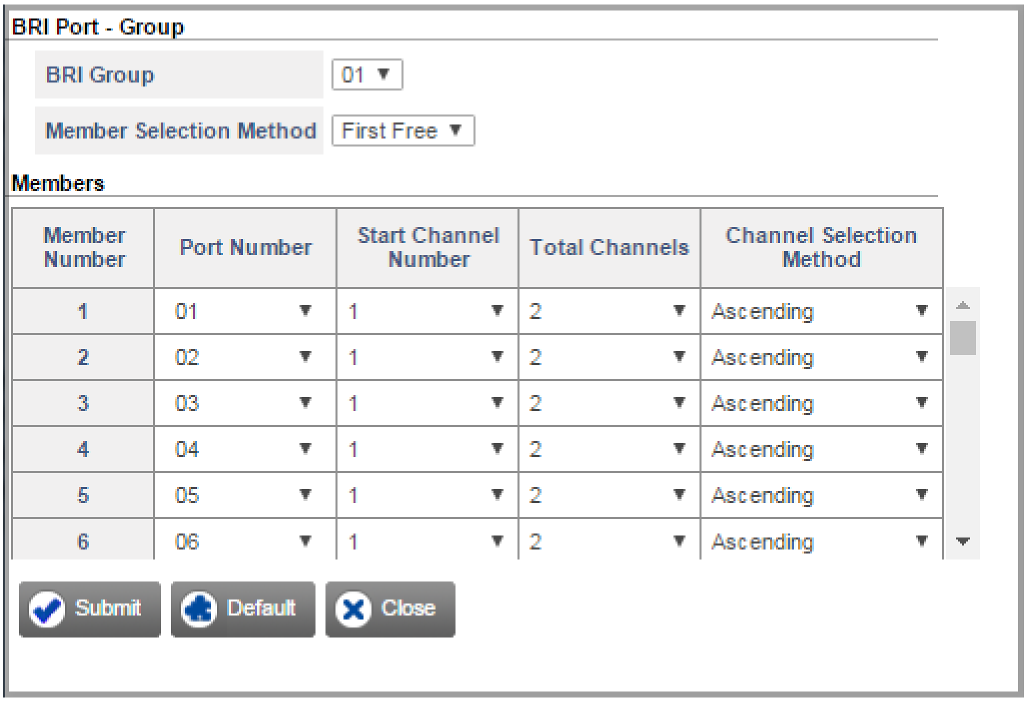
  Select **Ascending** to start hunting from the first to the last member channel. Select **Descending** to start hunting from the last to the first member channel. Default: Ascending.

- To create a group of *not-sequential* **BRI Channels** as members,
  - Select **BRI Group**.



  - Select a **BRI Group** number. Default:1.
  - Click **Settings** .

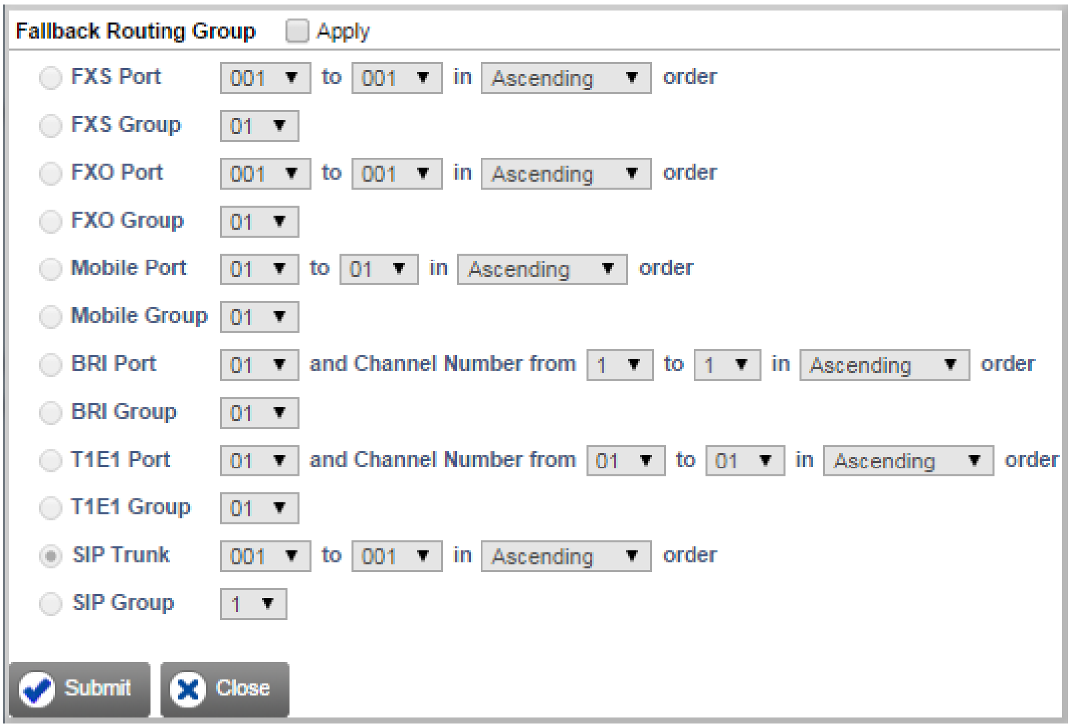
The **BRI Port - Groups** window opens.



- Create the BRI Group. For detailed instructions on creating groups, see the topic "Group" under *Advanced Settings.*

- Similarly, you can create a group of *sequential* and *not-sequential* T1E1 Ports.

- You may create the **Fallback Routing Group**.



- To do this,

- Select the **Apply** check box.
- Follow the same instructions provided earlier for creating *sequential* and *not-sequential* groups of FXS Ports, FXO Ports, Mobile Ports, BRI Ports, T1E1 Ports and SIP Trunks.

- Click **Submit** to save changes. The **Edit** window closes.
- The entry you edited appears in the **Destination Port/Group for FXS Port** window.
- Close the **Destination Port/Group for FXS Port** window to return to the Handling of Calls window.

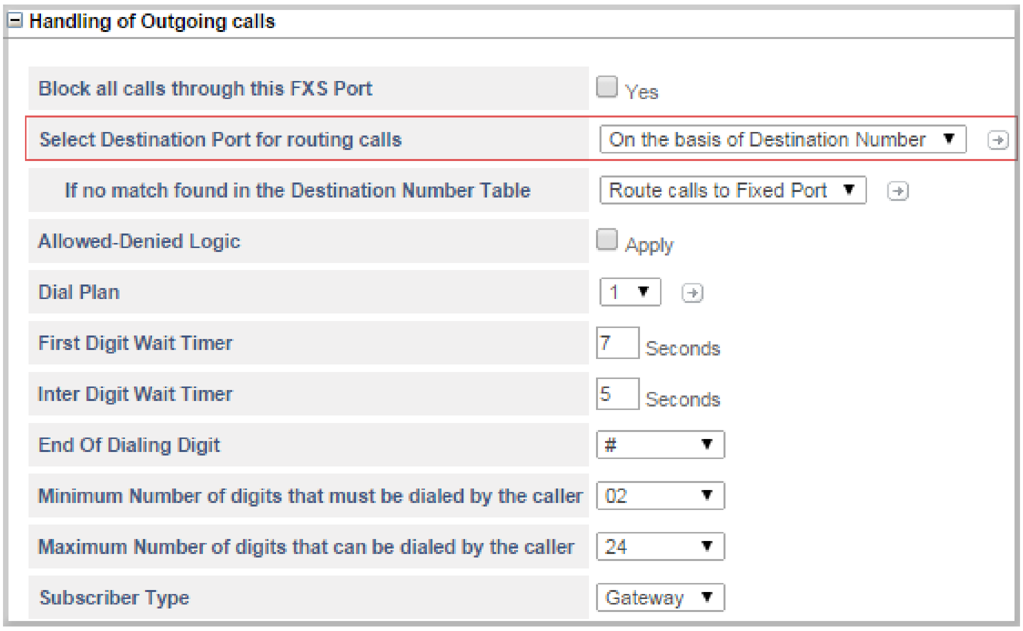
On the basis of Destination Number

In this method, outgoing calls made from the FXS Port are routed to the destination port on the basis of the destination number (called party number) dialed by the caller.

You must configure the called party numbers in the **Destination Number Based** Table. SARVAM UMG will match the called party number dialed by the caller with the entries of this table. If a match is found for the number in the table, the call is routed to the destination.

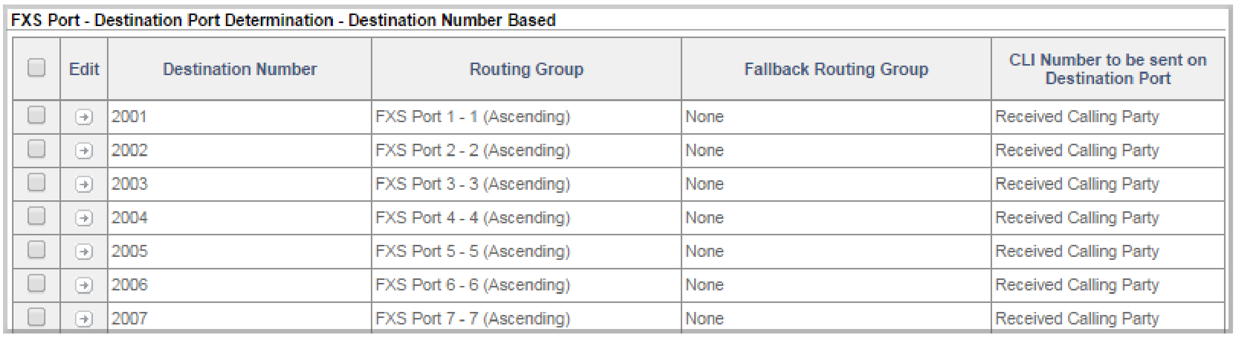
To apply this method, do the following:

- In **Select Destination Port for routing calls**, select **On the basis of Destination Number** option.



- Click **Settings** .

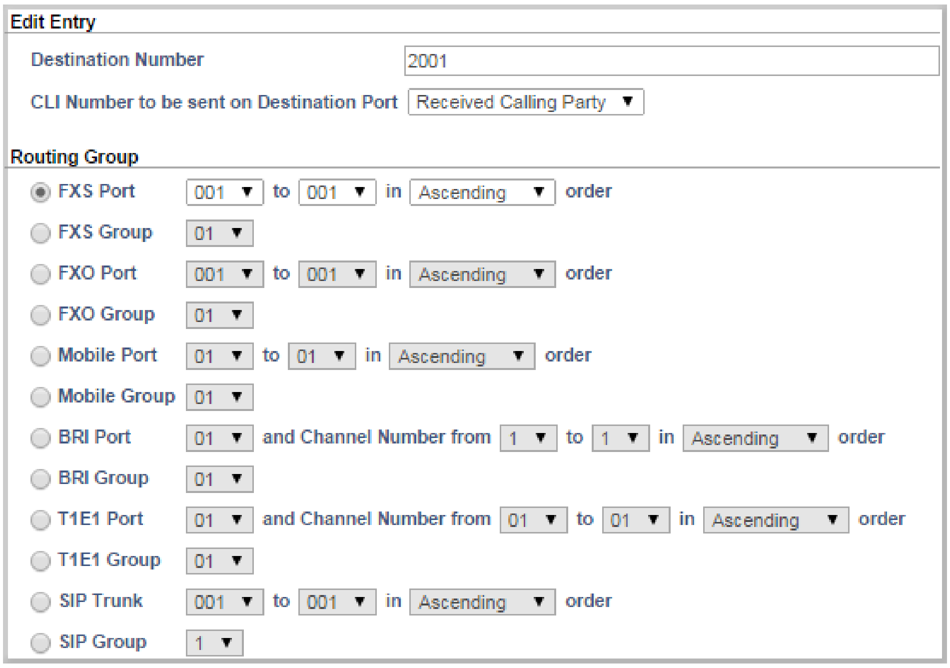
The **FXS Port - Destination Port Determination - Destination Number Based** table window opens.



FXS Port - Destination Port Determination - Destination Number Based

	Edit	Destination Number	Routing Group	Fallback Routing Group	CLI Number to be sent on Destination Port
		2001	FXS Port 1 - 1 (Ascending)	None	Received Calling Party
		2002	FXS Port 2 - 2 (Ascending)	None	Received Calling Party
		2003	FXS Port 3 - 3 (Ascending)	None	Received Calling Party
		2004	FXS Port 4 - 4 (Ascending)	None	Received Calling Party
		2005	FXS Port 5 - 5 (Ascending)	None	Received Calling Party
		2006	FXS Port 6 - 6 (Ascending)	None	Received Calling Party
		2007	FXS Port 7 - 7 (Ascending)	None	Received Calling Party

- The table displays 120 default entries, which you can edit as per your requirement.
- Click **Edit** to edit the first entry in the table.
- The **Edit Entry** window opens.



- In **Destination Number**, enter the number you expect the callers to dial. You may enter upto 64 characters (Digits + "Wildcard Characters") in this field. Valid characters are 0 to 9, *, #, X, T, Comma [,], Hyphen [-], Caret [^]. Default: Blank.

Wildcard Characters

SARVAM UMG 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.

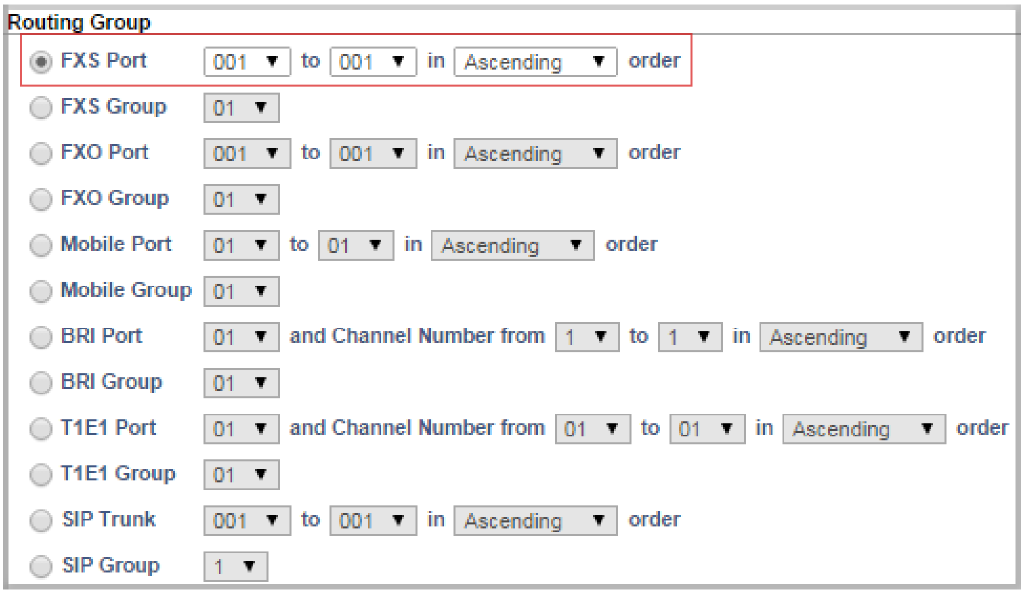
+	+ (plus) can be configured as a first character of the Destination Number string in the *SIP Trunk-Destination Port Determination-Destination Number Based* table only.
[-]	Hyphen within the bracket, defines a range. Only digits 0-9 are allowed within a bracket.
[,]	Comma within a bracket is used as 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.

- Select the **CLI Number to be sent on Destination Port**. You can select Received Calling Party or Received Called Party. Default: Received Calling Party.

CLI to be sent on Destination Port is applicable when FXS Port/Group is determined as the Destination Port.

- Create the **Routing Group**.

  - To create a group of *sequential* **FXS Ports** as members,



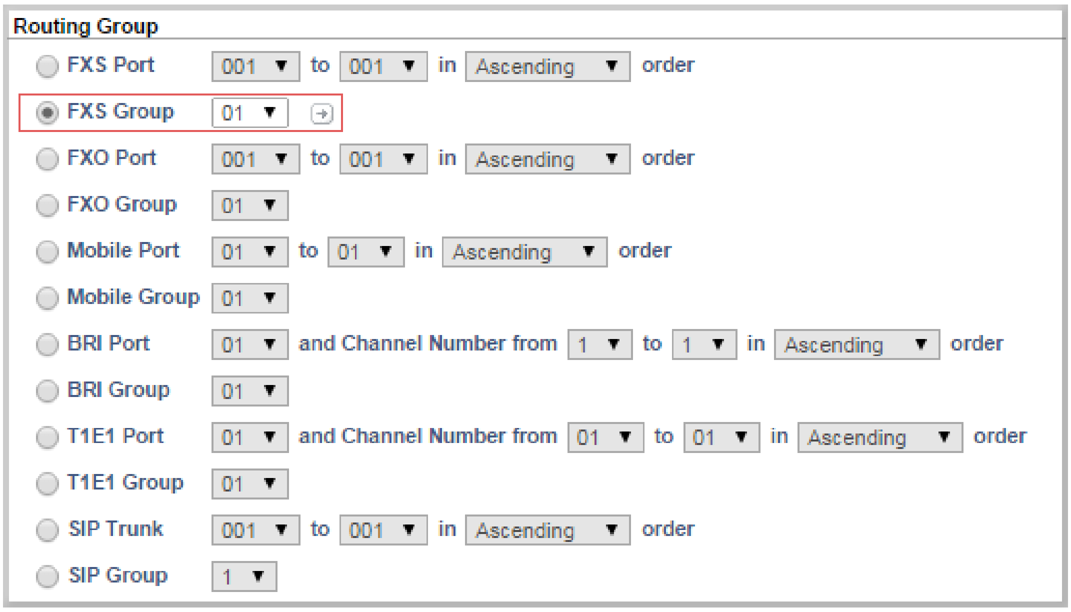
  - Select the desired **FXS Port** numbers as members. Default:1.

  - In **in - order**, select the order in which the system should hunt for a free member FXS Port to route the call.

    Select **Ascending** to start hunting from the first to the last member FXS Port. Select **Descending** to start hunting from the last to the first member FXS Port. Default: Ascending.

  - To create a group of *not-sequential* **FXS Ports** as members,

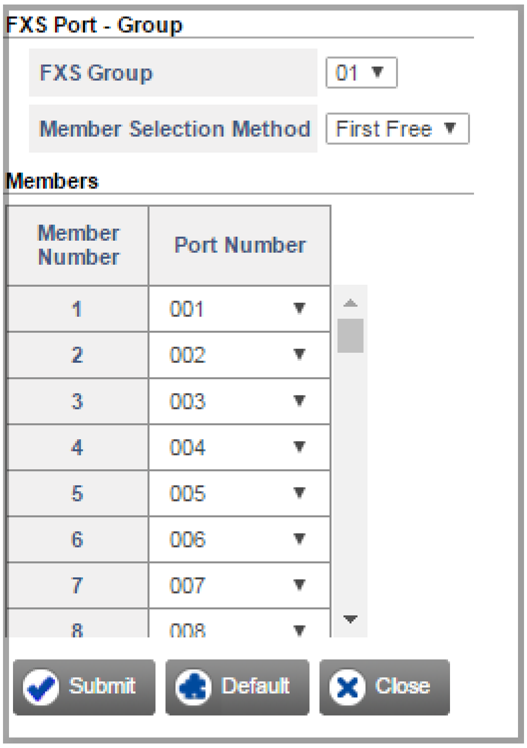
- Select a **FXS Group**.



- Select **FXS Group** number. Default:1.

- Click **Settings** .

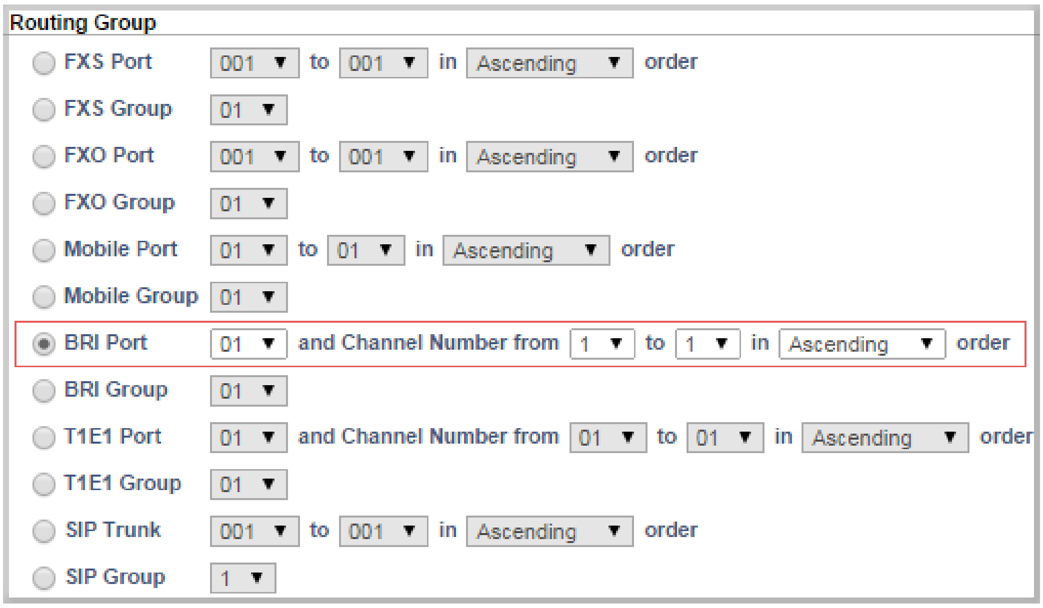
  The **FXS Port - Groups** window opens.



- Create the FXS Group. For detailed instructions on creating groups, see the topic "Group" under *Advanced Settings*.

- Similarly, you can create a group of *sequential* and *not-sequential* FXO Ports, Mobile Ports and SIP Trunks.

- To create a routing group of *sequential* **BRI Channels** as members,

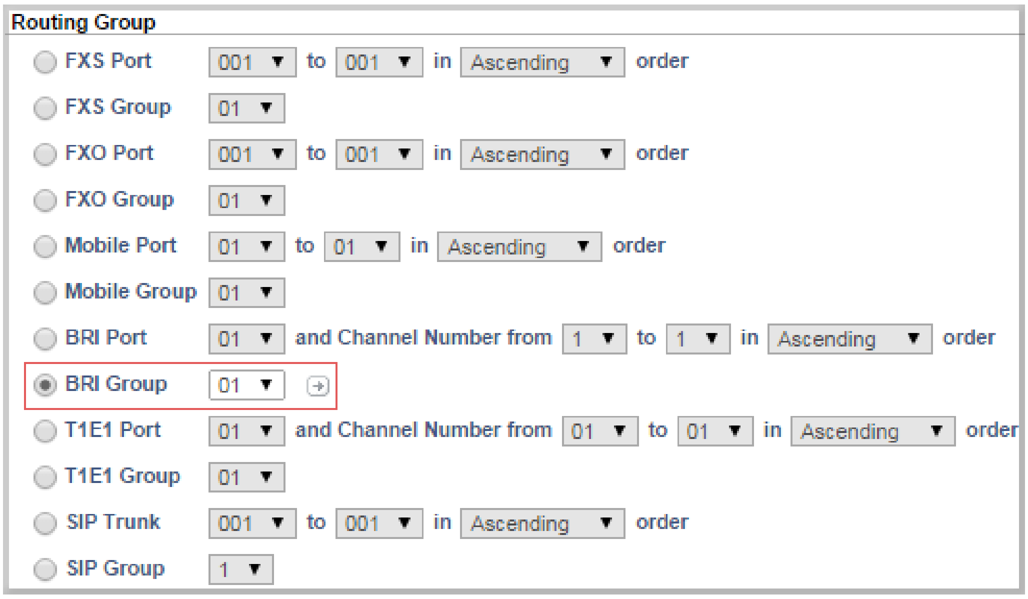


  - Select the **BRI Port** Number. Default: 1.
  - In Channel Number **From - to**, select the **Start Channel Number** and the **End Channel Number**, respectively.
  - In **in - order**, select the order in which the system should hunt for a free member Channel to route the call.

    Select **Ascending** to start hunting from the first to the last member channel. Select **Descending** to start hunting from the last to the first member channel. Default: Ascending.

- To create a group of *not-sequential* **BRI Channels** as members,

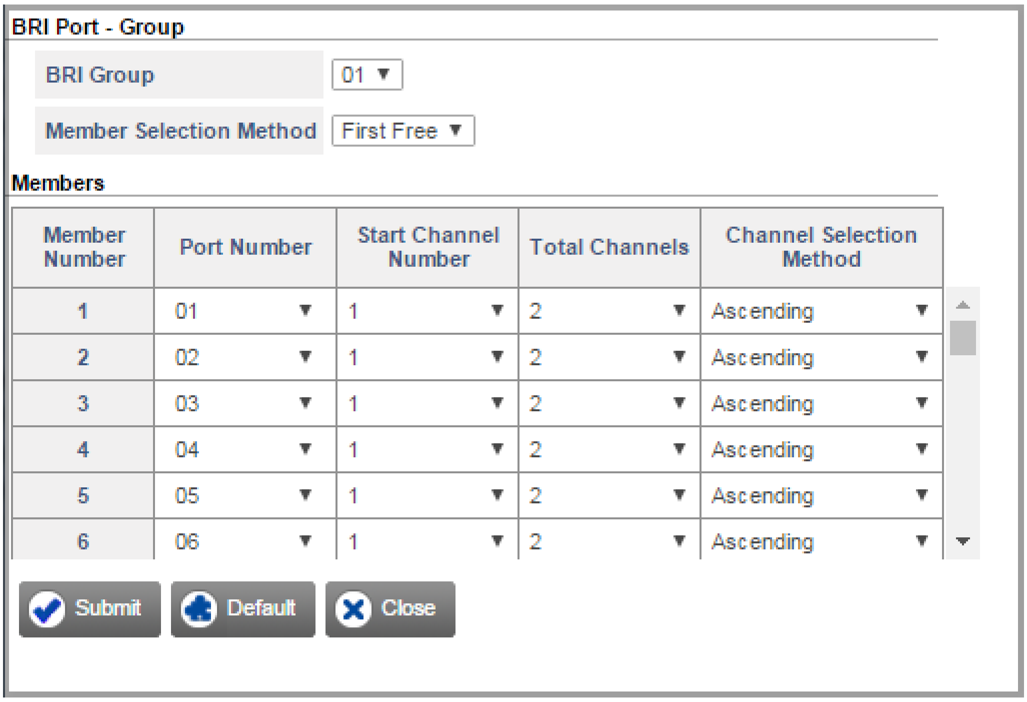
- Select **BRI Group**.



- Select a **BRI Group** number. Default:1.

- Click **Settings** .

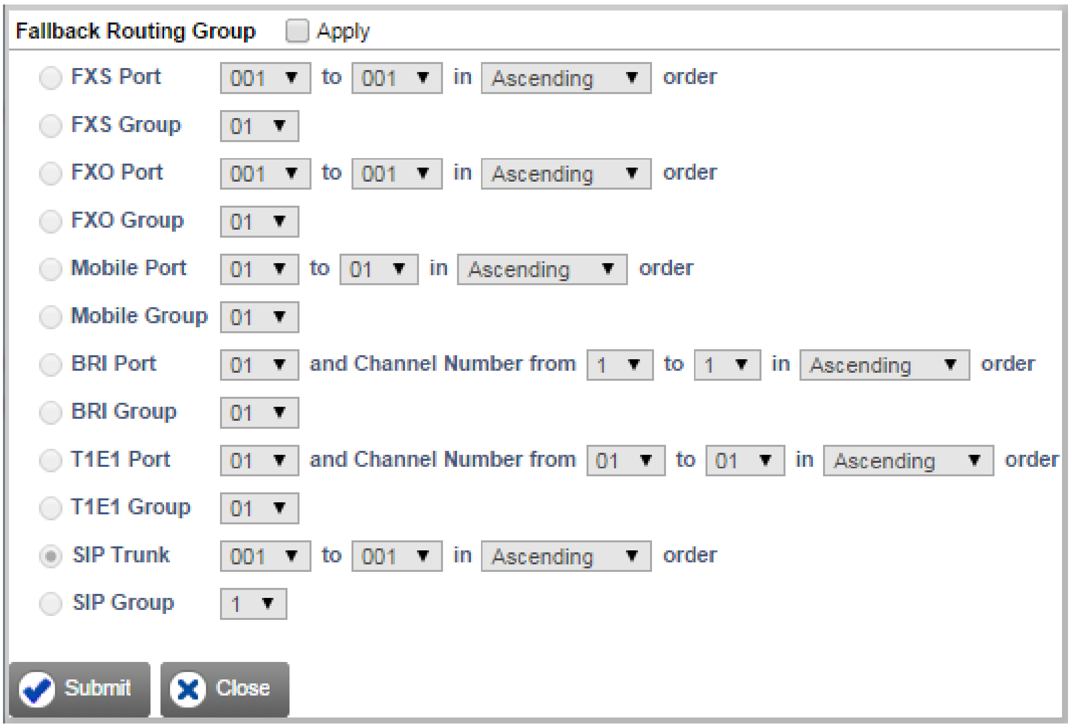
  The **BRI Port - Groups** window opens.



- Create the BRI Group. For detailed instructions on creating groups, see the topic “Group” under *Advanced Settings.*

- Similarly, you can create a group of *sequential* and *not-sequential* T1E1 Ports.

- You may create the **Fallback Routing Group**.

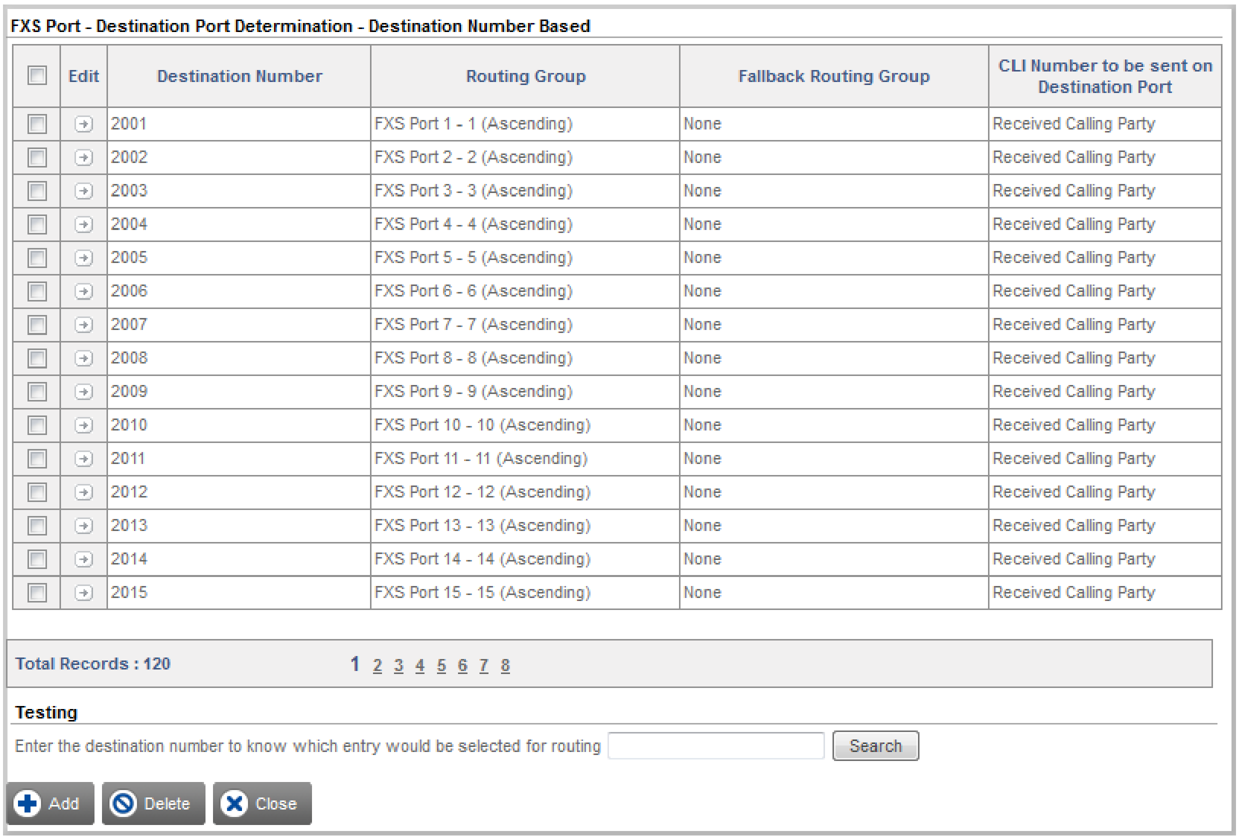


- To do this,
  - Select the **Apply** check box.
  - Follow the same instructions provided earlier for creating *sequential* and *not-sequential* groups of FXS Ports, FXO Ports, Mobile Ports, BRI Ports, T1E1 Ports and SIP Trunks.
- Click **Submit** to save changes. The **Edit Entry** window closes.
- The changes appear in the **FXS Port - Destination Port Determination - Destination Number Based** table.
- Follow the same steps as above to edit another entry to this table.
- To delete an entry, select the check box and click the **Delete** button.

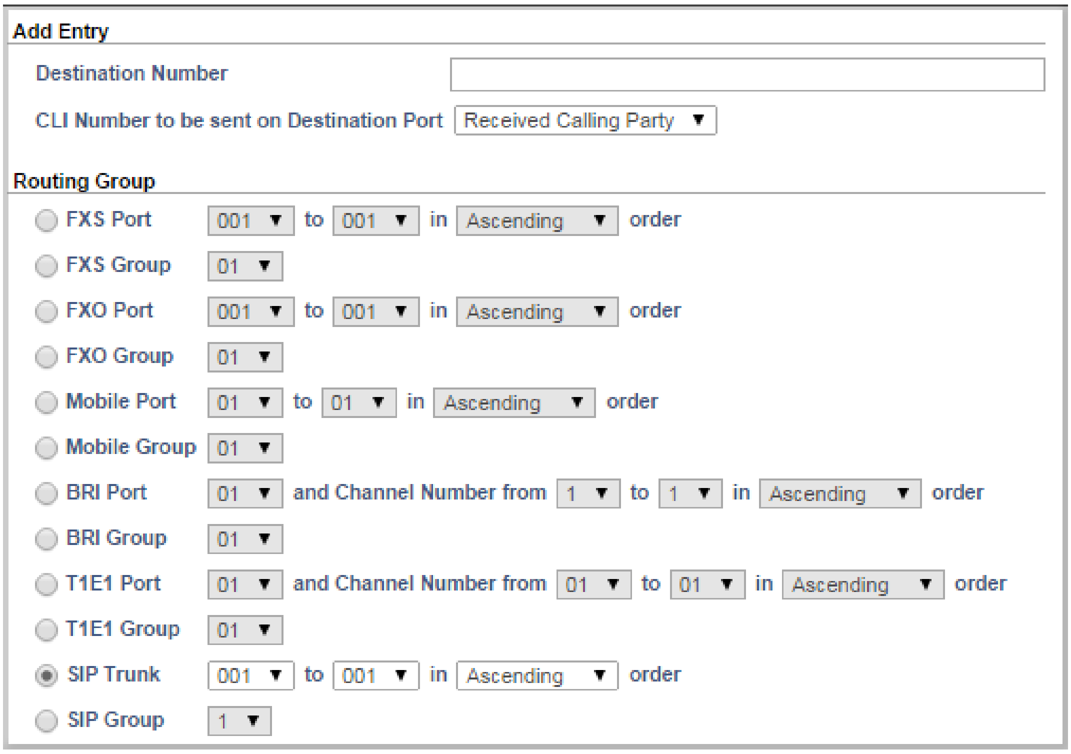
If there are multiple entries in the Destination Number Based table, to search a particular entry in the table, under Testing enter the desired number to know which entry would be selected for routing.

- The table can have a maximum of 300 entries. In the case of T1E1 and BRI Ports, the table can have a maximum of 250 entries.

- To add a new entry, click **Add**.



- The **Add Entry** window opens.



- Create the **Routing Group** and **Fallback Routing Group** as per your requirement.

- Click **Submit** and close the window.

- Close the window if you have finished adding/editing entries.

  You can also configure the **Destination Number Based** Table from Advanced Settings. For instructions, see "Destination Port Determination" under Advanced Settings.

- Select a method for routing outgoing calls, if no match is found in the Destination Number Table.

  In **If no match found in the Destination Number Table**, select the desired option for routing the call. You may select — Route calls to Fixed Port or Disconnect Call. Default: Route calls to Fixed Port.

  If you select *Route calls to Fixed Port*, click Settings to configure the Destination Port/Group for routing the call. For instructions, see "Fixed".

Allowed - Denied Logic

You can apply the Allowed-Denied logic on the FXS Port (source port) if you want to allow or restrict the dialing of particular numbers. You can use this feature for Toll Control.

The Allowed-Denied Number Logic makes use of two Number lists:

- **Allowed Numbers List:** This is the list of numbers that can be dialed out from the FXS Port.

- **Denied Numbers List:** This list contains the numbers that are to be restricted from being dialed out from the FXS Port.

When Allowed-Denied Logic is enabled on a source port, for each number dialed from the port, SARVAM UMG uses the best-match-found logic to compare the dialed number with the Allowed Number list and the Denied Number list.

The number is allowed to be dialed, if it:
- matches with both lists.
- matches with Allowed Number list, but not with the Denied Number list.
- matches with neither the Allowed List nor the Denied List.

The number is denied, if it matches with the Denied Number list, but not with the Allowed Number list.

The system does not apply the Allowed-Denied Logic:
- When dialed number string matches with any Access Code.
- When dialed number string matches with any Emergency Number.

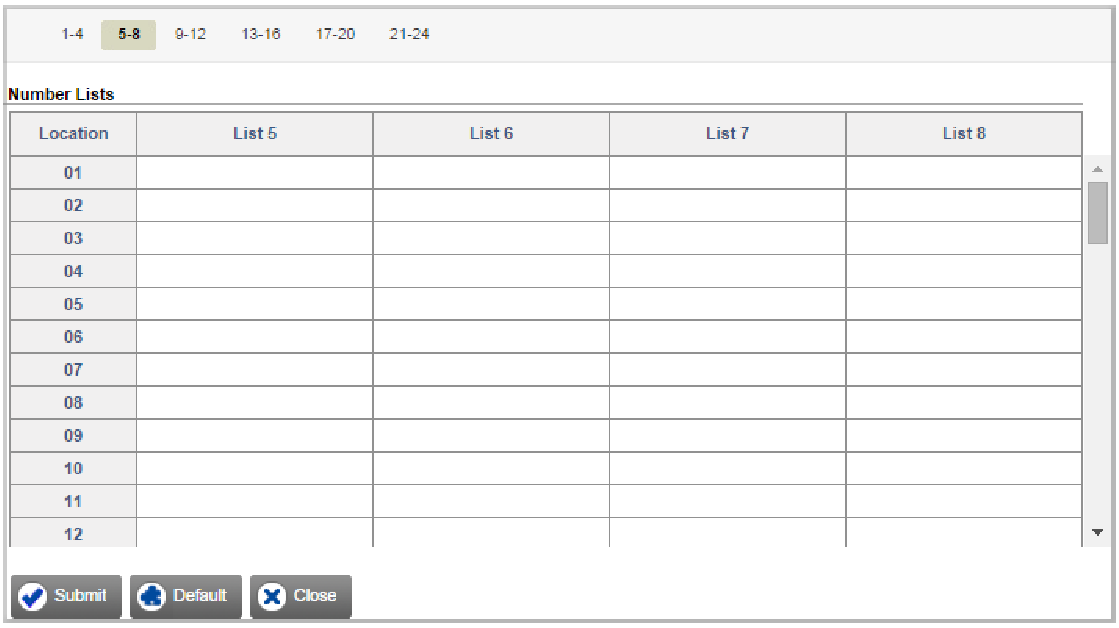
To apply Allowed - Denied Logic on the FXS Port,

- Select the **Allowed - Denied Logic** check box.

- In the **Allowed Number List**, select the list number you have configured with numbers you want to allow to be dialed out from the FXS Port. Default: 05.

  If you have not configured the Allowed Number List,

  - Click **Settings** .
  - The Number Lists window opens.



  - You may configure the default Allowed Number List 5 or any other list. See “Number Lists” to configure the allowed numbers.
  - Click **Submit** to save the Allowed Number List and close the window.

- In the **Denied Number List**, select the list number you have configured with numbers you want to restrict to be dialed out from the FXS Port. Default: 06.

  If you have not configured the Denied Number List,

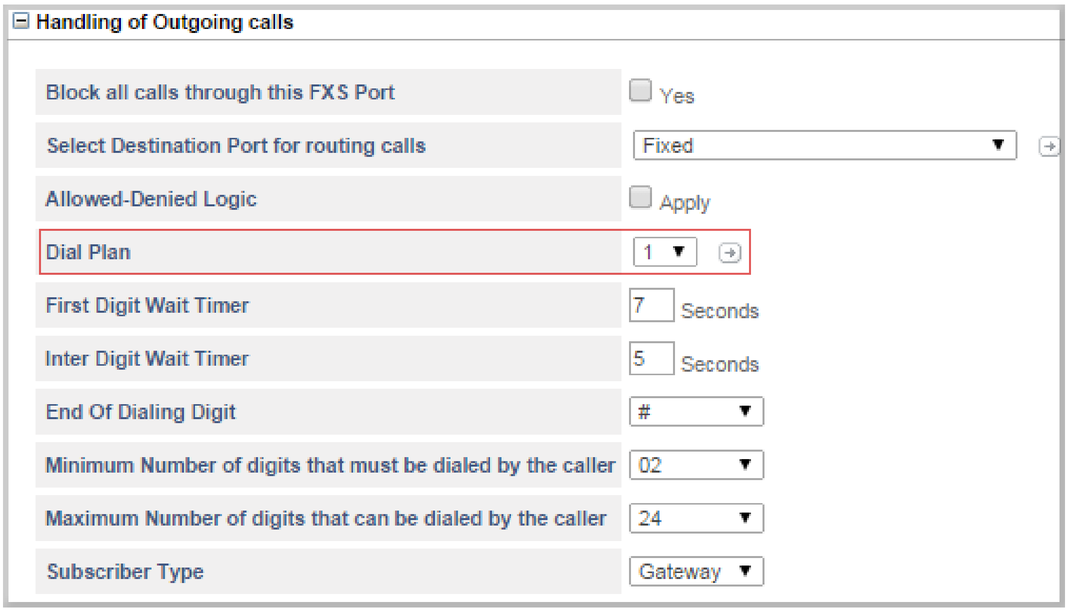
  - Click **Settings** . The Number Lists window opens.
  - You may configure the default Denied Number List 6 or any other list. See “Number Lists” to configure the restrict numbers.
  - Click **Submit** to save the Denied Number List and close the window.

Dial Plan

SARVAM UMG supports 8 Dial Plans with total 64 entries in each table. The Dial Plan contains a series of digits and/or wildcard characters.

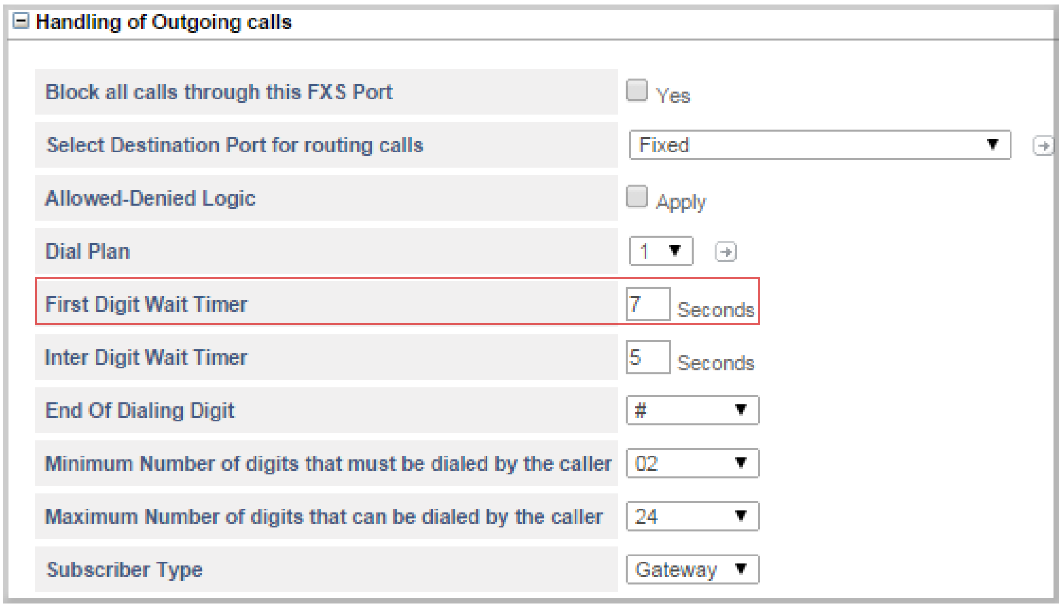
When a user dials a number, it is compared with the Destination Number configured in the Dial Plan. If a match is found, the system routes the call immediately without waiting for End of Dialing and if a match is not found, the system will wait for the End of Dialing and then route the call as per the Destination Port Selection method configured.

- To configure the **Dial Plan Table**, click **Settings** .



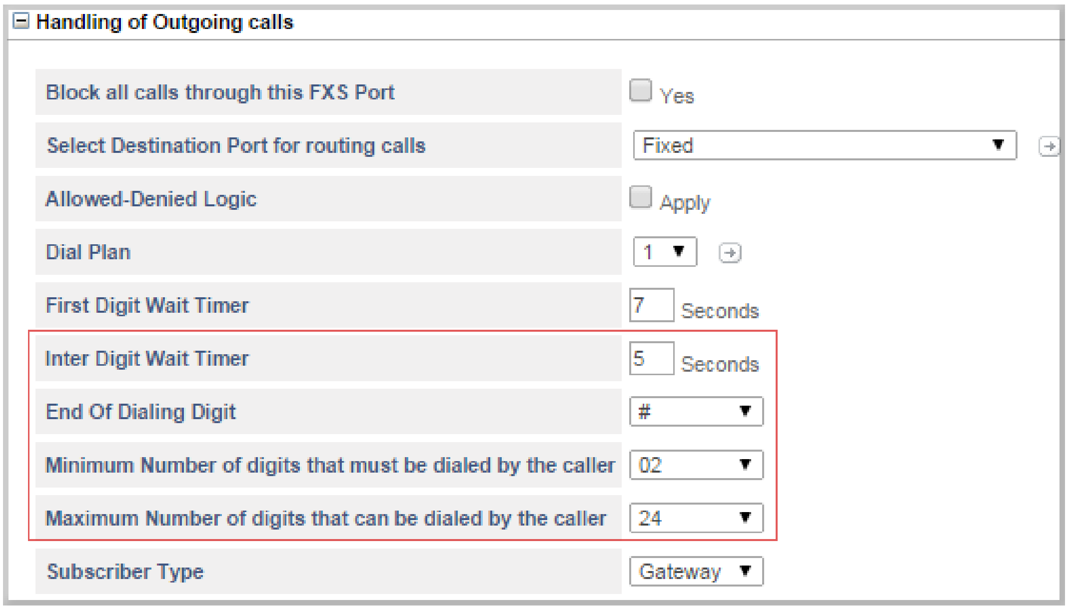
For detailed instructions, see "Dial Plan".

First Digit Wait Timer



- Set the duration of the **First Digit Wait Timer**. This is the time in seconds for the which the system will wait for the user to dial the destination number. Valid range is 01 to 99 seconds. Default: 7 seconds.

End-of-Dialing



- Set the duration of the **Inter Digit Wait Timer**. This is the time for which you want the system to wait while receiving the digits dialed by the user, to consider it as end-of-dialing. You may change this timer, if required. Valid range is 01 to 99 seconds. Default: 5 seconds.

- As **End of Dialing Digit**, select whether the system should consider # or * as termination digit to detect end of dialing. Default: #

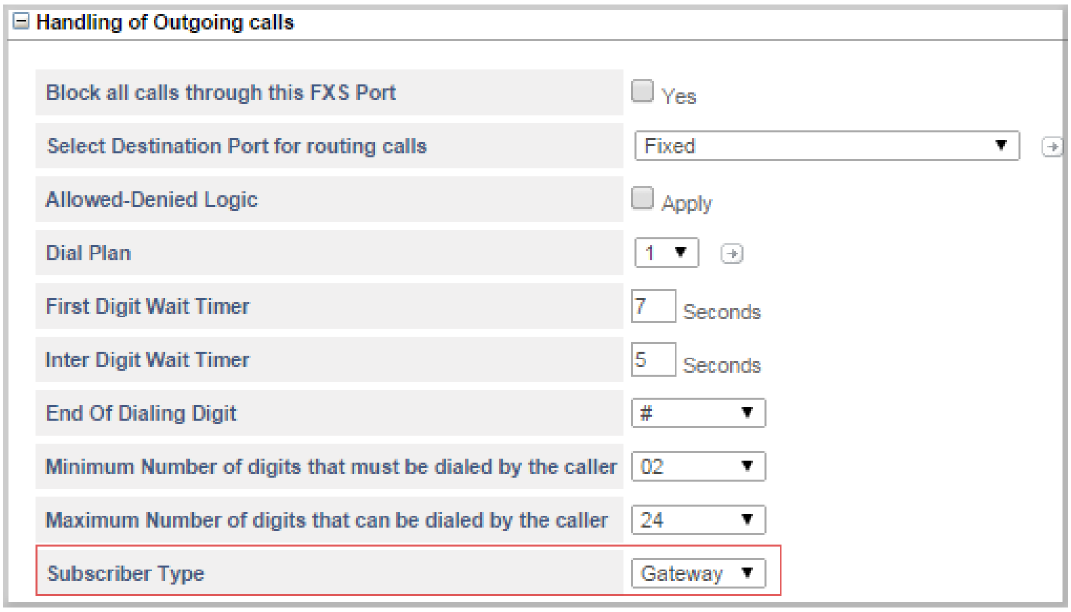
- In **Minimum number of digits that can be dialed by the caller**, select the minimum number of digits to be dialed by the user for the system to consider it as a valid number. Valid range is 01 to 24 digits. Default: 2 digits.

- In **Maximum number of digits that can be dialed by the caller**, select the maximum number of digits to be dialed by the user for the system to consider it as end-of-dialing. Valid range is 01 to 24 digits. Default: 24 digits.

  When the user dials a number, the system will match it with the above end-of-dialing indications and accept the one that matches first.

Subscriber Type

- Select the **Subscriber Type** for SARVAM UMG.



When SARVAM UMG is interfaced with a Service Provider — ITSP, the Matrix ETERNITY IP-PBX, or any other System— you can access the supplementary features supported by the Service Provider as well as the features of the SARVAM UMG. These features can be accessed by dialing flash.

By selecting the Subscriber Type, you may choose to access the features of the service provider, or to primarily access the features of SARVAM UMG.

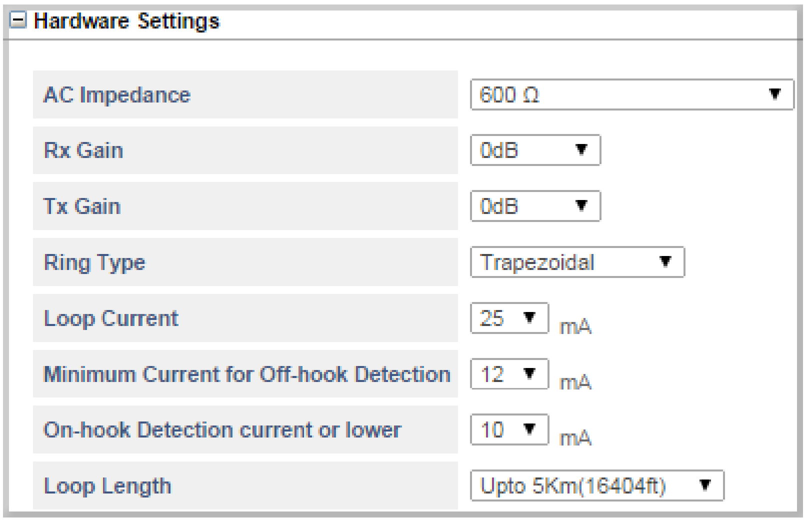
- Select **Network,** if you want to use the supplementary services supported by the Gateway. When you set SARVAM UMG in the Network mode, you can access the service provider features by dialing flash. You will not be able to access the local features of SARVAM UMG.

- Select **Gateway,** if you want to use primarily the supplementary features of SARVAM UMG. In the Gateway mode, you will also be able to access the supplementary services of the service provider which require dialing of Flash. To know more, see “Supplementary Services of Service Provider”.

Default: Gateway.

Hardware Settings

- Click **Hardware Settings** to expand.



- In **AC Impedance**, select the appropriate impedance according to the AC Impedance supported by the device connected to the FXS Port of SARVAM UMG. The device may be a phone or a PBX.

  You may select — 600 Ω, 900 Ω, 350 Ω + (1000 Ω || 0.21 µF), 220 Ω + (820 Ω || 120 nF) or 270 Ω + (750 Ω || 150 nF). Default: 600 Ω.

- **Rx Gain** enables you to adjust the volume level of the speech received from the remote party. Select the Rx Gain accordingly. Default: 0dB.

- **Tx Gain** enables you to adjust the volume level of the speech transmitted to the remote party. Select the Tx Gain accordingly. Default: 0dB.

- Select the **Ring Type** to be generated on the FXS Port. You can select — Low Sinusoidal, Low Trapezoidal, Sinusoidal or Trapezoidal. Default: Trapezoidal.

- Select the **Loop Current** according to the *Loop Length* you select. You can select — 25 mA, 30 mA, 35 mA or 40 mA. Default: 25 mA.

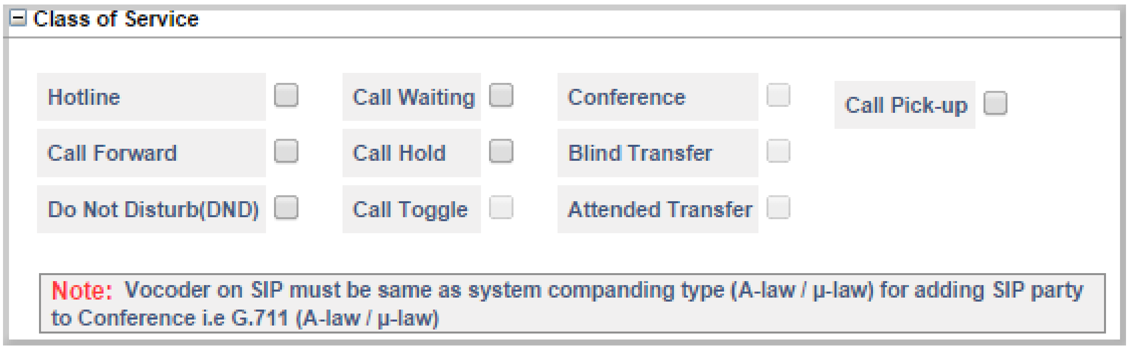
- Select the **Minimum Current for Off-hook Detection (mA)** as per your requirement. You can select—10mA, 12mA, 14mA or 16mA. Default: 12 mA.

- Select the **On-hook Detection current or lower** as per your requirement. You can select — 10mA, 12mA, 14mA or 16mA. Default: 10 mA.

- Select the **Loop Length** — **Upto 5 Km (16404 ft)** or **Above 5 Km (16404 ft)** — depending on SARVAM UMG's installation scenario. The Loop Length is the distance between the SARVAM UMG and the telephone instrument connected to the FXS Port. Default: Upto 5 Km (16404 ft).

Class of Service

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



- Select the features of SARVAM UMG that you want to allow in **Class of Service**[12] (CoS) of the FXS Port.

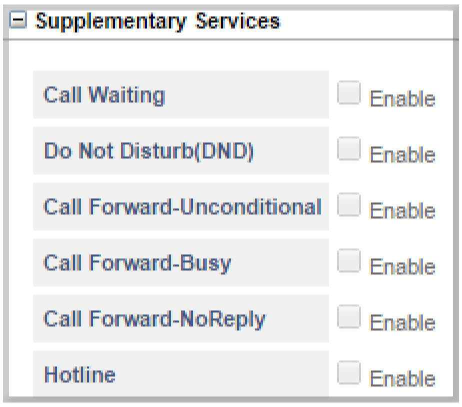
  By default all the features are denied.

  To allow a feature, select the respective check box.

  To deny a feature, clear the respective check box.

Supplementary Services

- Click **Supplementary Services** to expand.



SARVAM UMG offers the supplementary features — Call Waiting, Do Not Disturb (DND), Call Forward - Unconditional, Call Forward - Busy, Call Forward - No Reply and Hotline.

- Select the **Enable** check box of the features you want to use.
- By default all the features are disabled.

To use any Supplementary feature, first make sure that you have enabled it the Class of Service.

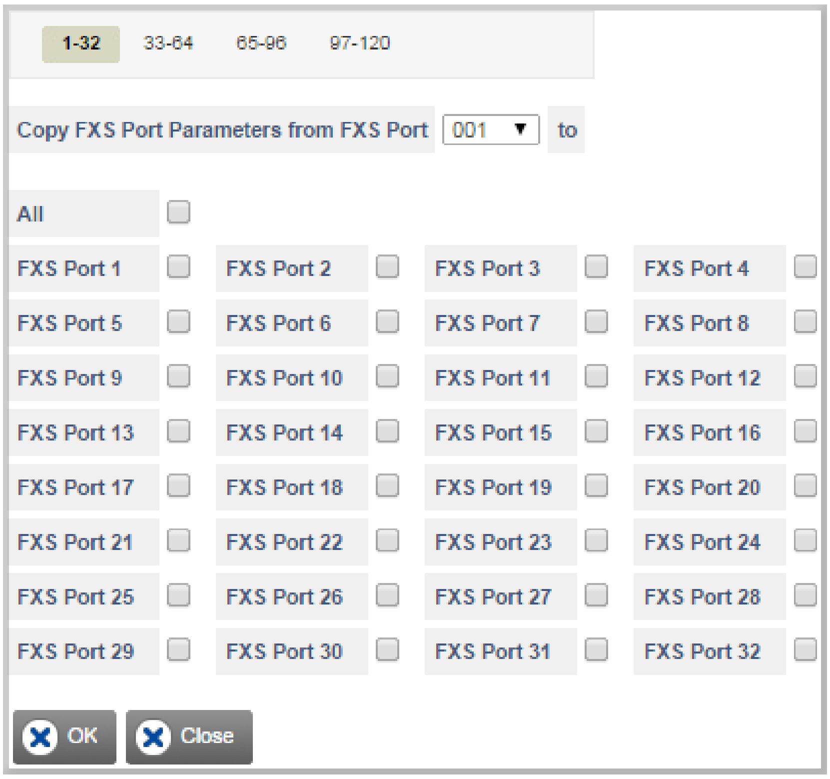
- If you have completed the configuration of FXS-1, click **Submit** to save settings.

12. *Class of Service (CoS) defines the set of features of SARVAM UMG that the phone connected to the FXS Port is to be allowed access to.*

- To configure the next FXS Port, click the FXS Port number tab and follow the same instructions as given earlier.

Copy Port Parameters

- You can also copy the settings of a FXS Port to another FXS Port using the **Copy** button. To do this,
  - Click the **Copy** button. The **Copy FXS Port Parameters** window opens.



  - In the **Copy FXS Port Parameters from FXS Port** box, select the number of the port you want to copy settings *From*. Select the check box of the respective port numbers you want to copy the settings *To*.
  - If you want to copy the settings *To* all the ports, select the **All** check box.
  - Click the **OK** button.
- Once you have copied the settings, you can again edit the specific parameters of the FXS Port you copied the settings to.

FXO Port

SARVAM UMG supports FXO Ports, which may be either

- interfaced with the analog trunk from CO and used to route incoming calls to FXS Ports, BRI Ports, T1E1 Ports, SIP Trunks and Mobile Ports.
  - or -
- interfaced with FXS Port of the System.

To configure the parameters of the FXO Port,

- Click the **Basic Settings** link to expand.
- Click the **FXO Port** link.

The FXO Port page displays the following parameters:

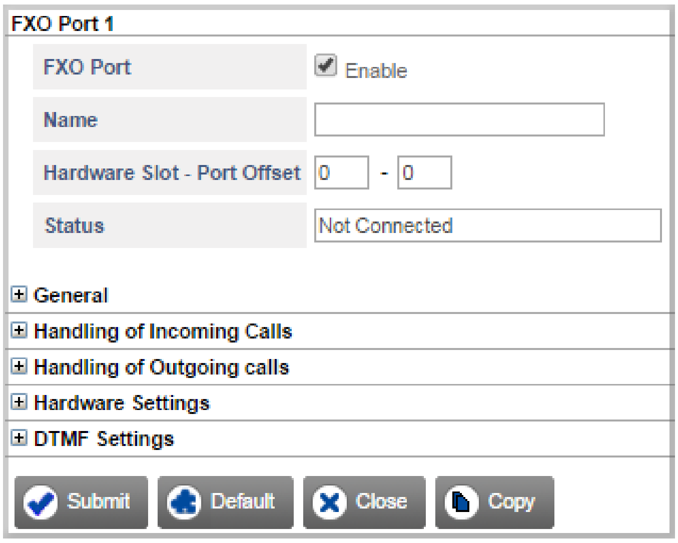
- **Port:** It displays the FXO Port numbers. Click on the desired FXO Port number to configure the Port Parameters.
- **Hardware Slot-Port:** The SARVAM UMG can automatically detect and assign the Hardware Slot and Port numbers to the FXO software ports. However, if required you may change the Hardware Slot and Port assigned to the FXO software port. In this case, enter the desired Hardware Slot and Port number.
- **Enable:** Keep the **FXO Ports** enabled. Clear the FXO Port **Enable** check box, only if you do not want to use the respective port. Default: Enabled.
- **Name:** Assign a Name to the FXO Port for identification. The Name can be a maximum of 24 characters.
- **Status:** This displays the status of connectivity.
- **CLI Type:** It displays the type of CLI you select — DTMF, FSK V2.3, FSK Bellcore.

- **Incoming Call Routing:** It displays the Incoming Call Routing Method you select.

To configure the FXO Port parameters,

- Click **FXO-1**.

  The **FXO Port 1** window opens.



- Keep the **FXO Port** check box enabled.

  Clear the **FXO Port Enable** check box only when you do not want to use this FXO Port. Default: Enabled.

- You can assign a **Name** to the FXO Port, which will be displayed to the called party, if the called party telephone instrument supports CLI display. Default: Blank

  The name you assign may consist of a maximum of 24 characters.Default: Blank.

- SARVAM UMG will assign the **Hardware Slot - Port Offset** automatically, when any card is inserted in the system.

  Hardware slot is the number of the universal slot of SARVAM UMG in which the CO Card is inserted. Range of slot number is 1-12. Port is the number of CO (FXO) hardware port on the card to which the CO line is connected.

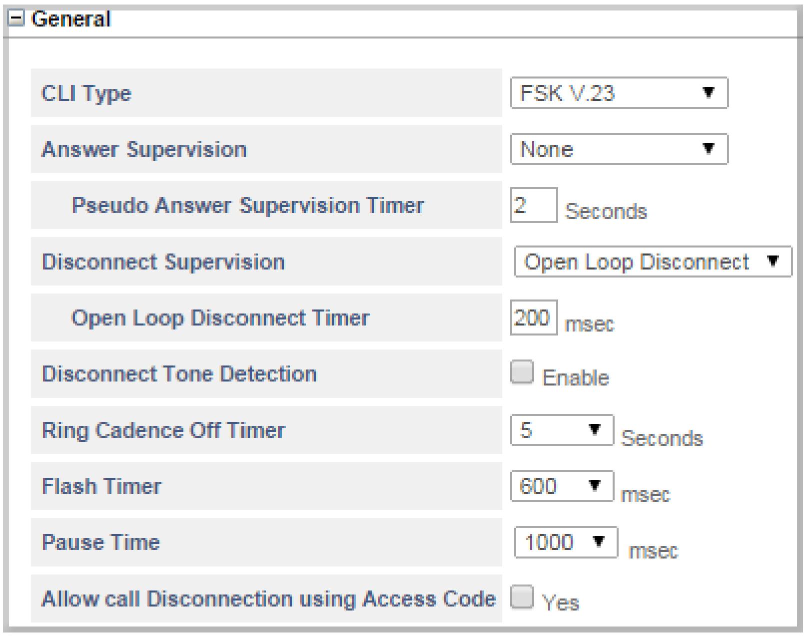
  However, if required, you may change the Hardware Slot and Port assigned to the FXO software port. In this case, enter the desired Hardware Slot and Port number.

  If you want to de-assign the Hardware Slot and Port, enter '00' in both fields. By default, Hardware Slot-Port is 00–00.

- **Status** displays the status of the connectivity of the FXO Port, that is, ‘Not Connected’ or ‘Connected’.

General

- Click **General** to expand.



- Select the **CLI Type —** DTMF-India, DTMF-ETSI, DTMF-Denmark, DTMF-Brazil, DTMF-Any, FSK V.23 or FSK Bellcore — according to the CLI type supported by your CO network. Default: FSK V.23

- SARVAM UMG uses **Answer Supervision** signaling to indicate that the call made through the FXO Port has been answered by the remote party.

  Select the appropriate **Answer Supervision** — None or Battery Reversal — supported by your CO network.

  - Select **None,** when no signaling is available from the CO network. The call will be considered as matured on the expiry of the *Pseudo Answer Supervision Timer*.

    - Set the duration of the **Pseudo Answer Supervision Timer** as per your requirement.Valid range is 01 to 99 seconds. Default: 2 seconds.

  - If you select **Battery Reversal,** SARVAM UMG will consider the call as matured only if reversal of polarity is detected on the FXO Port. Pseudo Answer Supervision Timer will not be applicable in this case.

    Default: None.

When an outgoing call is made from the FXO Port and Answer Supervision is set to Battery Reversal, the system will disconnect the call if it does not mature within 120 seconds.

- **Disconnect Supervision** is the signal given by the CO network to detect far end disconnection.

  Whenever a call (incoming or outgoing) on the FXO Port is disconnected by the remote party, the CO network will send Disconnect signal. SARVAM UMG will detect this signal and release the FXO Port. Select the appropriate **Disconnect Supervision** — None, Battery Reversal or Open Loop Disconnect — according to the type supported by your CO network.

- Select **None** when no signaling is available from the CO network.
- Select **Battery Reversal** when call disconnection is signaled in the form of reversed polarity.

  When the call is disconnected by the remote user, the reversed polarity is signalled and the FXO Port is released (free). The caller will get an error tone.

- Select **Open Loop Disconnect** when call disconnection is signaled in the form of an Open Loop signal. The system will check Open Loop Disconnect signal for the time configured in the *Open Loop Disconnect Timer*.

  Only if, the Open Loop signal is detected continuously for the time configured in *Open Loop Disconnect Timer,* it will be considered as a valid Open Loop signal for releasing the port.

  If you select this option, you must configure the Open Loop Disconnect Timer.

  - Set the duration of the **Open Loop Disconnect Timer** as per your requirement. Valid range is 001 to 999 msec. Default: 200 msec.

Default: Open Loop Disconnect.

- **Disconnect Tone Detection** is used by the system to release the FXO Port, when the remote party goes On-hook or disconnects the call. The tone detection is applicable for both incoming and outgoing calls from the FXO Port.

  Select the **Disconnect Tone Detection** check box, if you want the system to detect Call Disconnect Tone sent by the CO network on the FXO Port.

  - Select the **Disconnect Tone Type** that you want the system to use as the Call Disconnect Tone on the FXO Port. Default: Disconnect Tone 1

    If required, you can customize the frequencies and cadences of the Disconnect Tone as per your requirement. For instructions, see “Disconnect Tone”.

- Set the duration of the **Ring Cadence-OFF Timer**, to set the OFF time for Ring cadence. During the incoming call on FXO Port, if the CO gives ring in which the Ring OFF period is quite long, the system will consider that the ring has been stopped, and will stop ringing the FXS port, even though the incoming call is still present.

  To get accurate indication, the system supports Ring Cadence OFF timer on FXO Port so that ring can continue even for incoming calls with long Ring OFF period. Valid range is 1 to 9 seconds. Default: 5 seconds.

- Set the duration of the **Flash Timer**. This is the time for which Flash will be generated on the FXO Port. SARVAM UMG uses this event to activate various features — Call Hold, Call Transfer, etc. Default: 600 msec.

- Set the duration of the **Pause Timer** to add delay while a call is being made from the FXO Port. After the FXO Port goes Off-hook, SARVAM UMG adds some delay before dialing out the number. During this time, no digit is dialed by the system on the FXO Port. This is used when the exchange takes some time to detect that the FXO Port is Off-hook. Default: 500 msec.

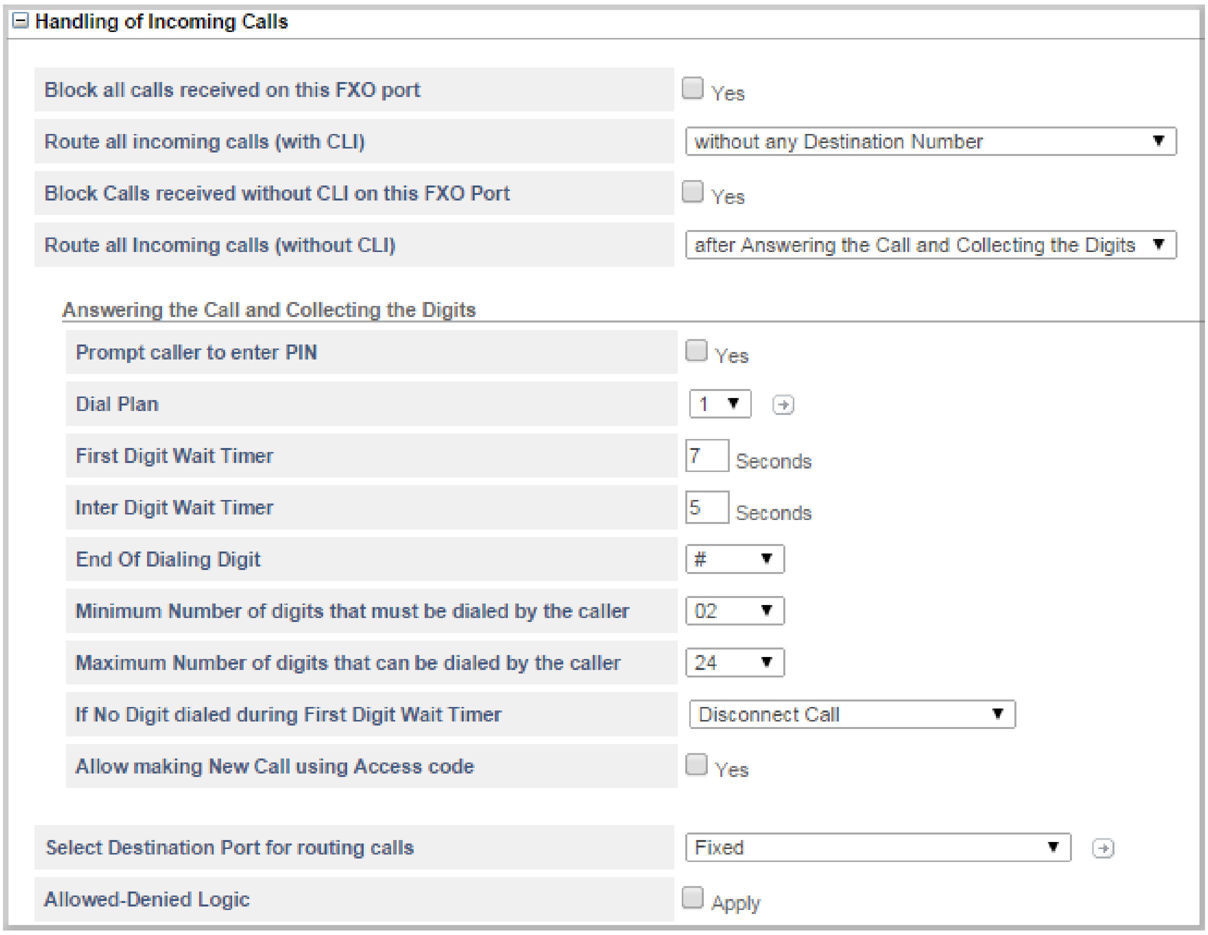
This timer will also be used while applying ANT logic, if you have configured Pause (^) in the Add Prefix column of ANT table. See “Automatic Number Translation (ANT)” for more details.

- Select the **Allow Call Disconnection using Access code** check box, if you want to enable the feature Disconnect Call using Access Code on the FXO Port. See “Disconnecting a Call using Access Code”.

- Click **Submit** to save the settings.

Handling of Incoming Calls

- Click **Handling of Incoming Calls** to expand.



- If you do not want to route calls received on this FXO Port, select the **Block all calls received on this FXO Port** check box. Default: Disabled.

Destination Number Determination

Select the desired destination number determination method for routing incoming calls *with* and *without* CLI.

- To **Route all Incoming calls (with CLI)**, you may select from any of the following methods.
  - without any Destination Number
  - to the Fixed Destination Number
  - on the basis of Calling Party Number
  - after Answering the Call and Collecting the Digits

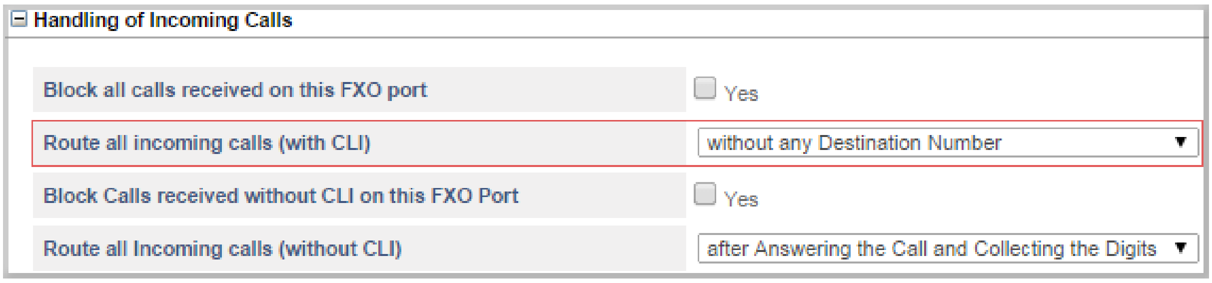
  Default: without any Destination Number

If the destination number to be dialed out is an IP Address, SARVAM UMG will not check the Destination Port Determination Method. Instead, it will route the call using the SIP Trunk / Group programmed for IP Dialing. (To know more, see the feature description "IP Dialing").

Route Calls without any Destination Number

In this method, all calls received on the FXO Port are directly routed to the destination port, regardless of the Destination Number.

- To apply this method, in **Route all incoming calls (with CLI)**, select **without any Destination Number**.

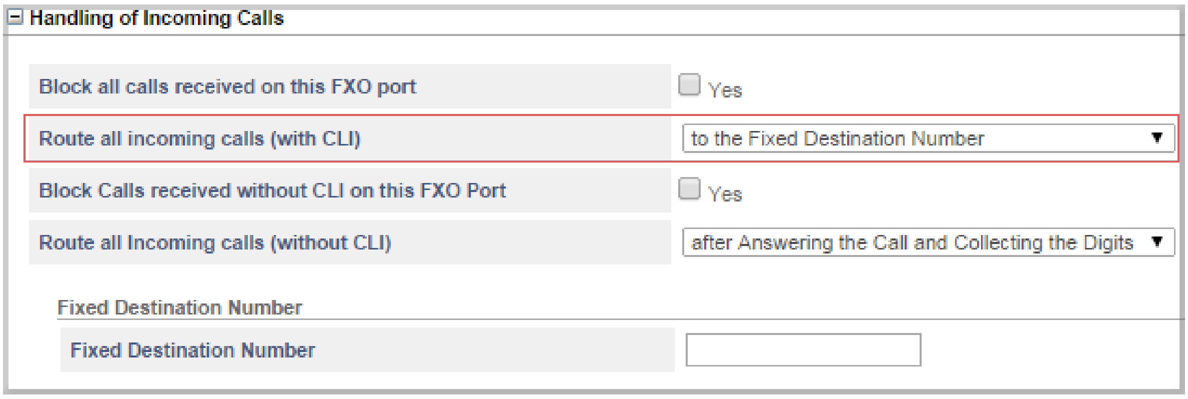


Route to the Fixed Destination Number

In this method, calls received on the FXO Port are routed to a fixed destination number, which you must configure for this port.

To apply this method, do the following:

- In **Route all incoming calls (with CLI)**, select **to the Fixed Destination Number**.



- In **Fixed Destination Number**, enter the desired destination number.

  The Destination Number may consist of a maximum of 24 digits. Valid digits are 0 to 9, *, # and (.) dot. Default: Blank.

- Click **Submit** to save the changes.

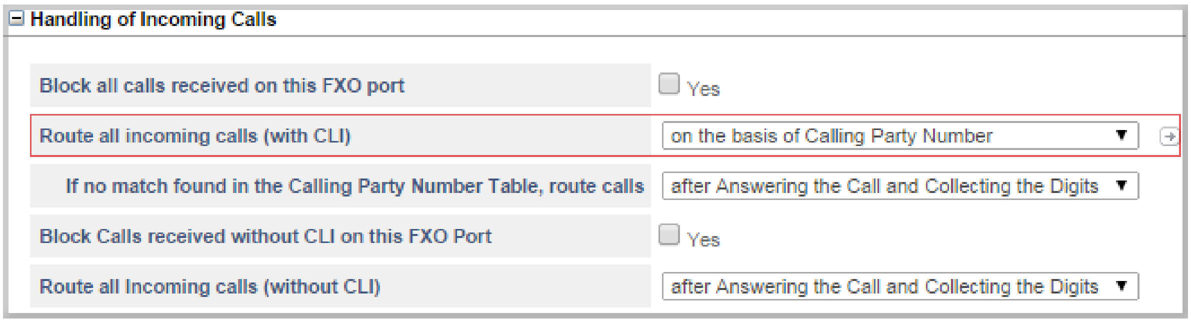
Route on the basis of Calling Party Number

In this method, a call received on the FXO Port is routed to a specific destination number, as per the calling party's number. For this, the calling party numbers and their corresponding destination numbers must be configured in the Calling Party Number Based Table.

Whenever there is an incoming call on the FXO Port, SARVAM UMG will match the Calling Party Number with the entries of the Calling Party Number Based Table. If a match is found, the call will be routed to the destination number configured for that Calling Party Number.

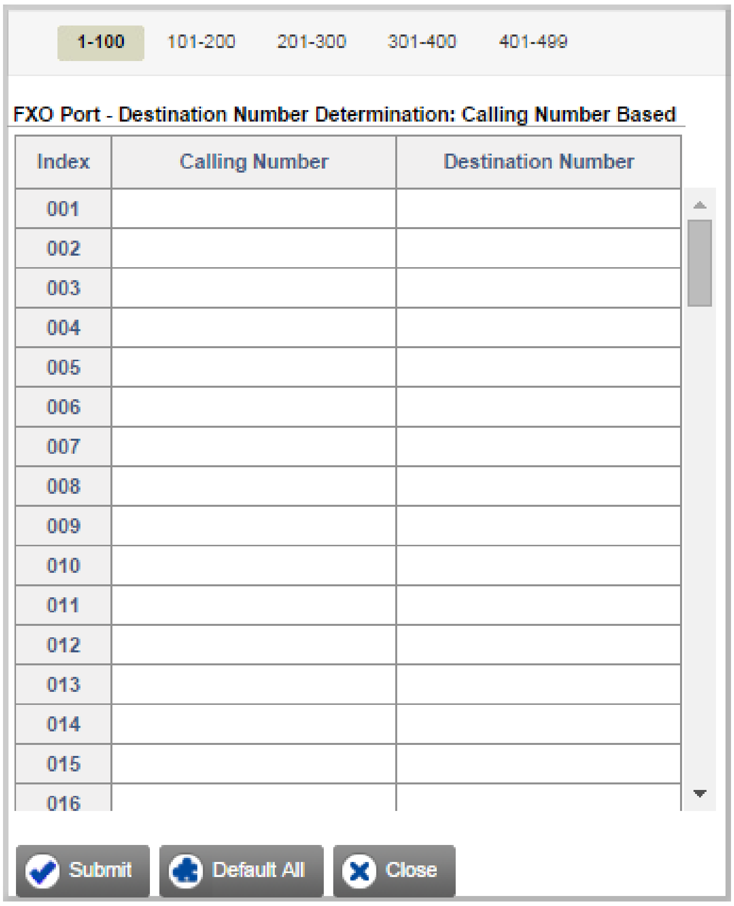
To apply this method, do the following:

- In **Route all incoming calls (with CLI)**, select **on the basis of Calling Party Number**.



- Click **Settings** .

The **FXO Port - Destination Number Determination: Calling Number Based** Table opens.



- Configure the **Calling Number Based** table for the FXO Port. You can enter upto 499 Calling Party Numbers and their corresponding Destination Numbers in this table.

- In **Calling Number**, enter the calling party numbers. The Calling numbers may consist of a maximum of 24 characters. Default: Blank.

- For each calling party number, enter a corresponding destination number in **Destination Number**. Destination numbers may consist of a maximum of 24 characters. Digits 0 to 9, *, # and (.) dot are allowed. Default: Blank.

- Click **Submit** to save your entries. Close the window to return to the FXO Port window.

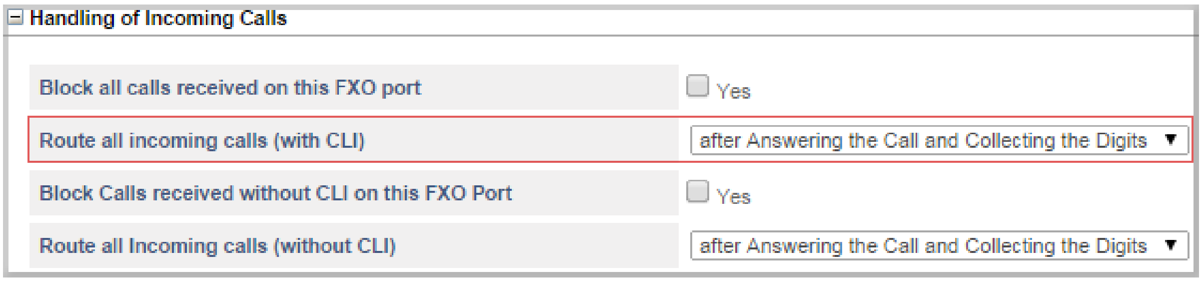
  You can also configure the **Calling Number Based** Table from *Advanced Settings* link. For instructions, see “Destination Number Determination” under *Advanced Settings*.

- For incoming calls with Calling Party Numbers that do not match with the Calling Party Number Table, you may select the destination number determination method.

  In the **If no match found in the Calling Party Number Table, route calls** box, you may select either **to the Fixed Destination Number** or **after Answering the Call and Collecting the Digits**. Default: after Answering the Call and Collecting the Digits.

Route After Answering the Call and Collecting the Digits

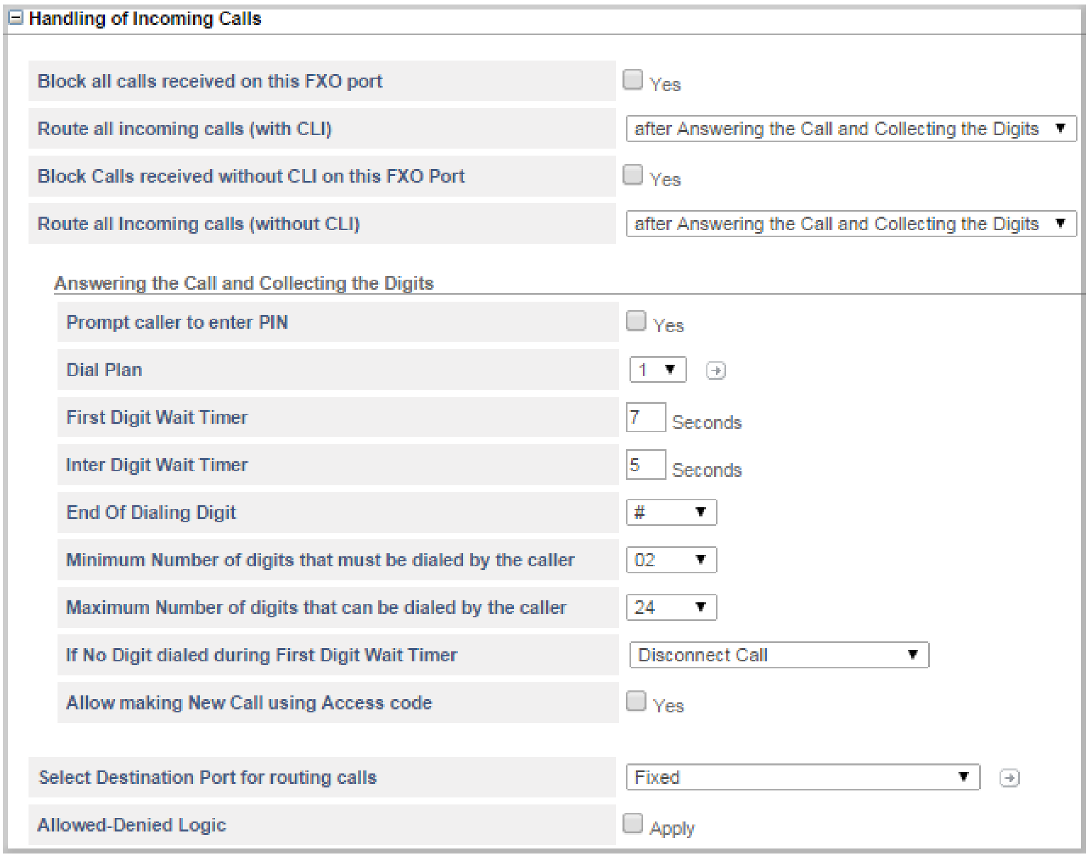
In this method, the system answers the incoming call on the FXO Port and plays dial tone to the caller, allowing the caller to dial the desired number. The number dialed by the caller is considered as the destination number.



To apply this method, do the following:

- In **Route all incoming calls (with CLI)**, select **after Answering the Call and Collecting the Digits**.

  The related parameters appear under **Answering the Call and Collecting the Digits**.



- If you want to enable PIN Authentication on the FXO Port, select the **Prompt caller to enter PIN** check box.

  If you enable this check box, you must also configure the PIN Authentication Table. To know more about this feature and for detail instructions, see “PIN Authentication” under *Advanced Settings*.

    - SARVAM UMG supports 8 Dial Plans with total 64 entries in each table. When a user dials a number, it is compared with the Destination Number configured in the Dial Plan. If a match is found, the system routes the call immediately without waiting for End of Dialing and if a match is not found, the system will wait for the End of Dialing and then route the call as per the Destination Port Selection method configured.

      Select the **Dial Plan** table number you configured for this port. If you have not configured the Dial Plan table you may do so now,

        - Click **Settings** . The Dial Plan Table opens.

        - Configure the numbers in the table. For detailed instructions, see “Dial Plan”.

    - Set the duration of the **First Digit Wait Timer**. This is the duration for which you want the system to wait for the caller to dial the destination number after the dial tone. Valid range is 01 to 99 seconds. Default: 7 seconds.

    - Set the duration of the **Inter Digit Wait Timer**. This is the duration for which you want the system to wait while receiving the digits dialed by the caller to consider it as End of Dialing. You may change this timer, if required. Valid range is 01 to 99 seconds. Default: 05 seconds.

    - As **End of Dialing Digit**, select whether the system should consider **#** or ***** as termination digit to detect end of dialing. Default: #

    - In **Minimum number of digits that can be dialed by the caller**, select the minimum number of digits to be dialed by the user for the system to consider it as a valid number**.** Valid range is 01 to 24 digits. Default: 2 digits.

    - In **Maximum number of digits that can be dialed by the Caller**, select the maximum number of digits to be dialed by the user for the system to consider it as end-of-dialing**.** Valid range is 01 to 24 digits. Default: 24 digits.

      When the caller dials a number, the system will match it with the above end-of-dialing indications and accept the one that matches first.

- If the caller fails to dial the number during the First Digit Wait Timer, you can either have the system disconnect the call or route the call to a fixed destination number.

  In **If No Digit dialed during First Digit Wait Timer**, you can select — **Disconnect the Call** or **Use Fixed Destination Number**. Default: Disconnect Call.

    - If you select **Use Fixed Destination Number**, enter the desired destination number in **Fixed Destination Number.** The Destination number may consist of a maximum of 24 digits. Valid digits are 0 to 9, *, # and (.) dot. Default: Blank.

- Select the **Allow making New Call using Access Code** check box, if you want to enable the feature Making New Call using Access Code on the FXO Port. See “Making a New Call using Access Code”.

- Click **Submit** to save settings.

- If you do not want to route calls without CLI through this port, select the **Block Calls received without CLI on this FXO Port** check box.

- To **Route all Incoming calls (without CLI),** you may select from any of the following methods:
  - to the Fixed Destination Number, see "Route to the Fixed Destination Number".
  - after Answering the Call and Collecting the Digits, see "Route After Answering the Call and Collecting the Digits".
    Default: after Answering the Call and Collecting the Digits.

Destination Port Determination

Select the Destination Port for routing calls for the FXO Port. You may select from any of the following options:

- Fixed
- on the basis of Destination Number
- on the basis of Calling Party Number
  Default: Fixed

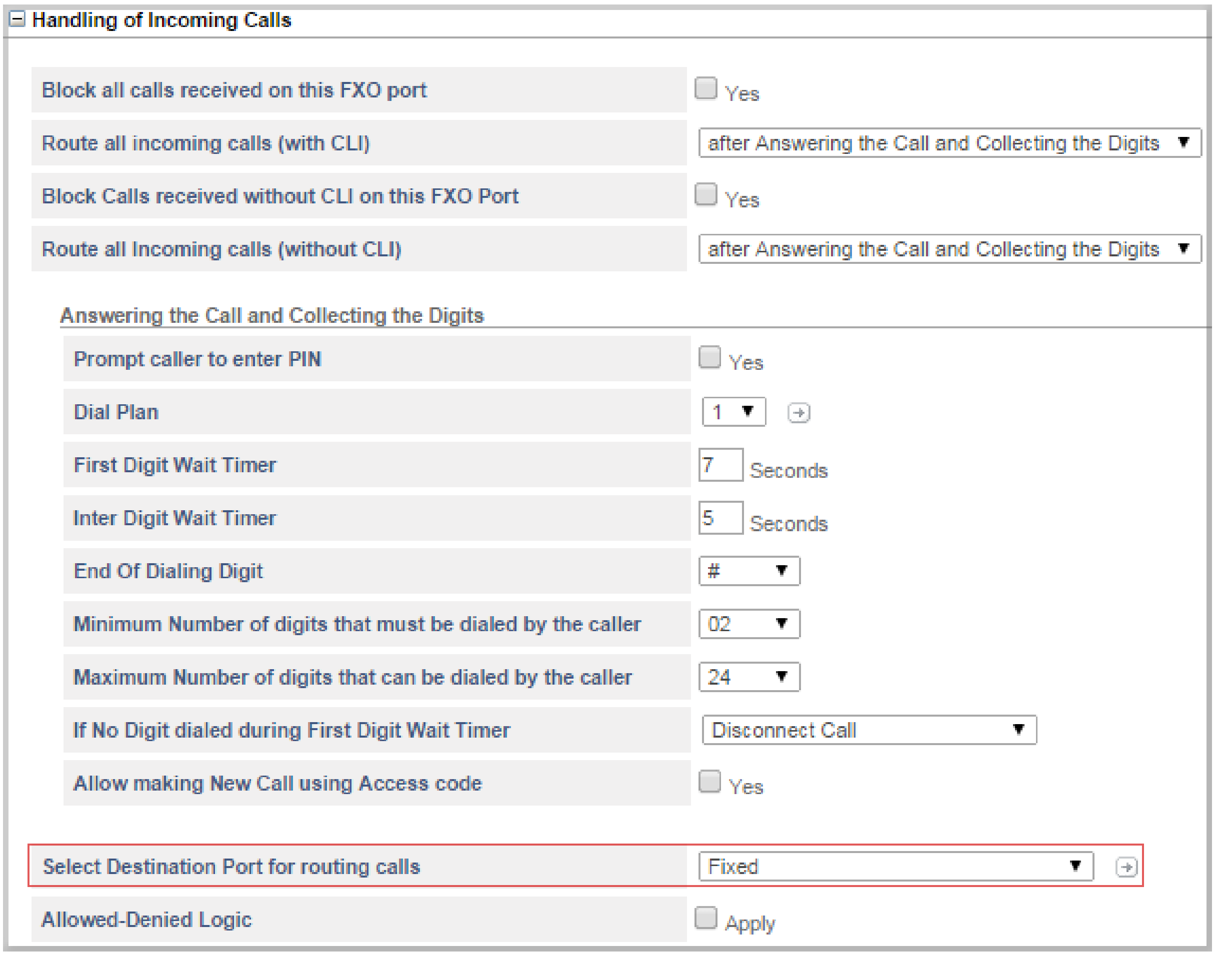
If the destination number to be dialed out is an IP Address, SARVAM UMG will not check the Destination Port Determination Method. Instead, it will route the call using the SIP Trunk / Group programmed for IP Dialing. (See "IP Dialing" to know more).

Fixed

In this method, calls received on the FXO Port are routed to a Fixed Destination Port, irrespective of the number dialed on the FXO Port.

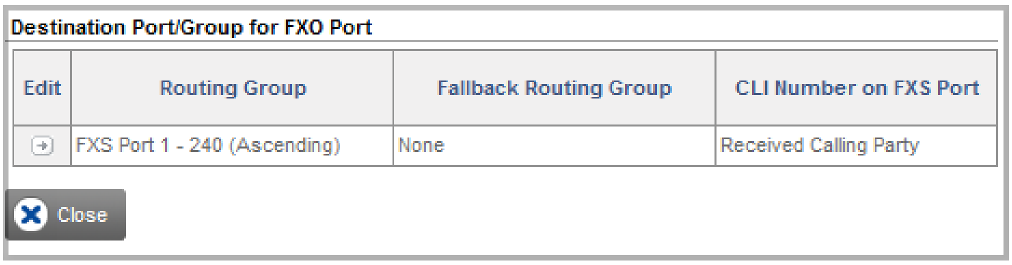
To apply this method, do the following:

- In **Select Destination Port for routing calls**, select **Fixed** option.



- Click **Settings**.

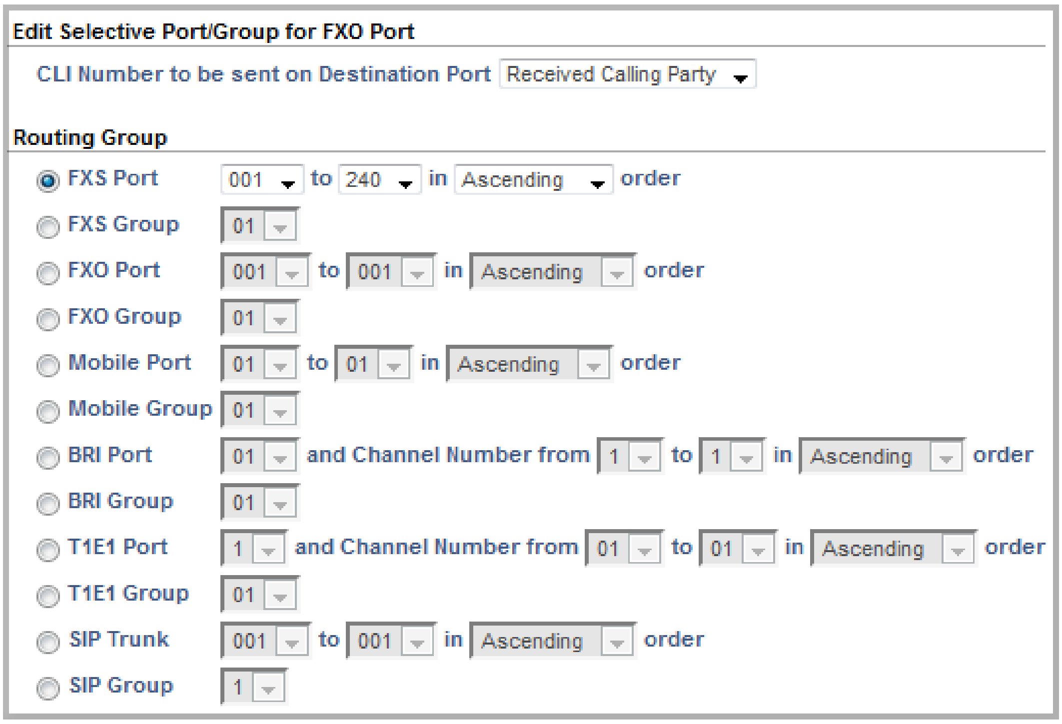
The **Destination Port/Group for FXO Port** window opens.



The default **Routing Group** and **Fallback Routing Groups** appear.

- If you wish to change the default Routing Group options, click **Edit**.

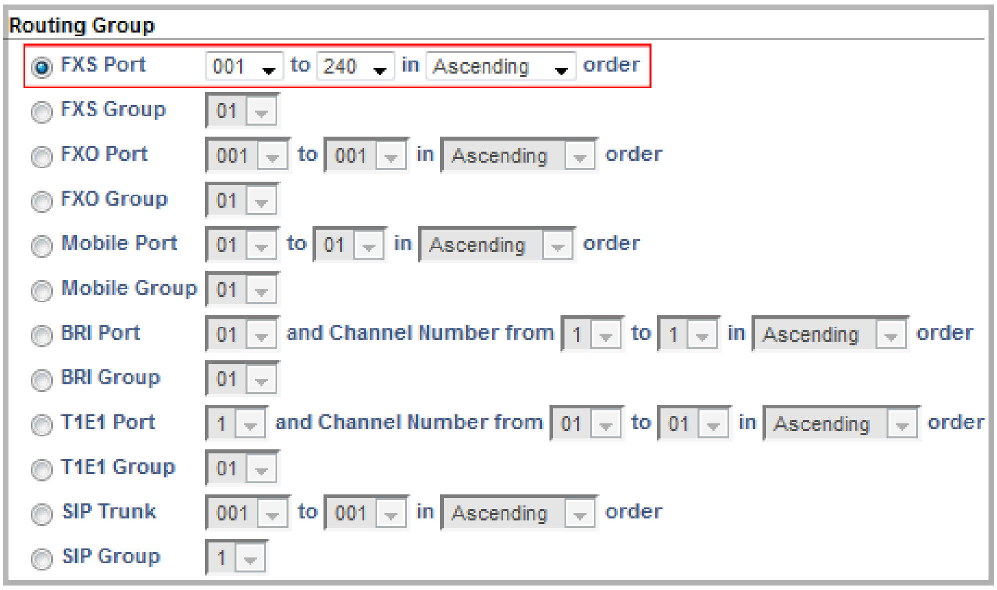
The **Edit Selective Port/Group for FXO Port** window opens.



- Select the **CLI Number to be sent on Destination Port**. You can select Received Calling Party or Received Called Party. Default: Received Calling Party.

CLI to be sent on Destination Port is applicable when FXS Port/Group is determined as the Destination Port.

- Create the **Routing Group**.
  - To create a group of *sequential* **FXS Ports** as members,



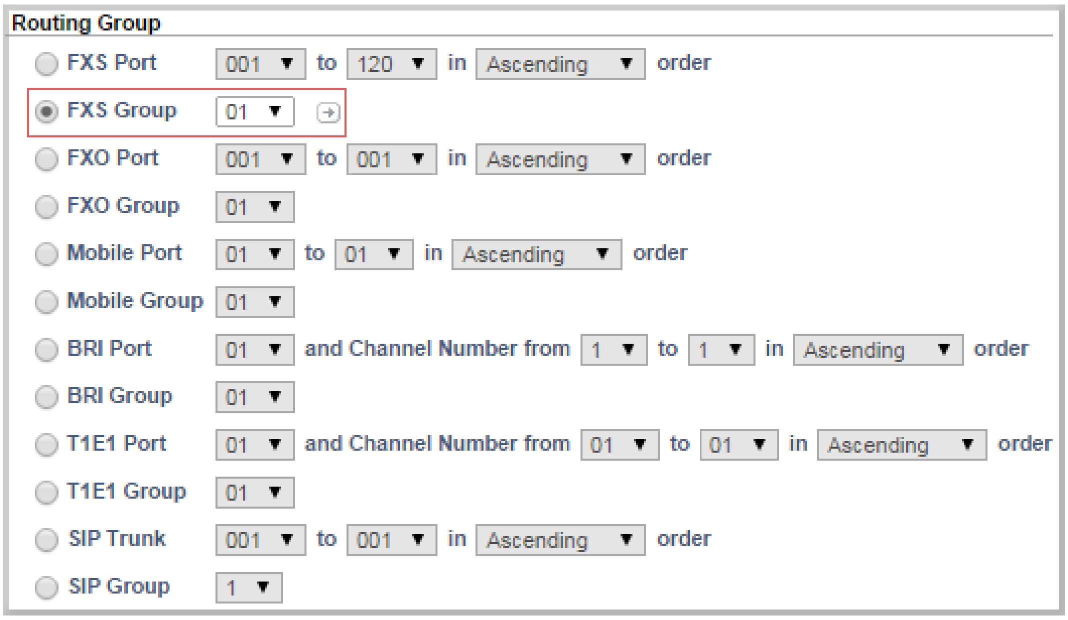
    - Select the desired **FXS Port** numbers as members. Default:1.

- In **in - order**, select the order in which the system should hunt for a free member FXS Port to route the call.

  Select **Ascending** to start hunting from the first to the last member FXS Port. Select **Descending** to start hunting from the last to the first member FXS Port. Default: Ascending.

- To create a group of *not-sequential* **FXS Ports** as members,

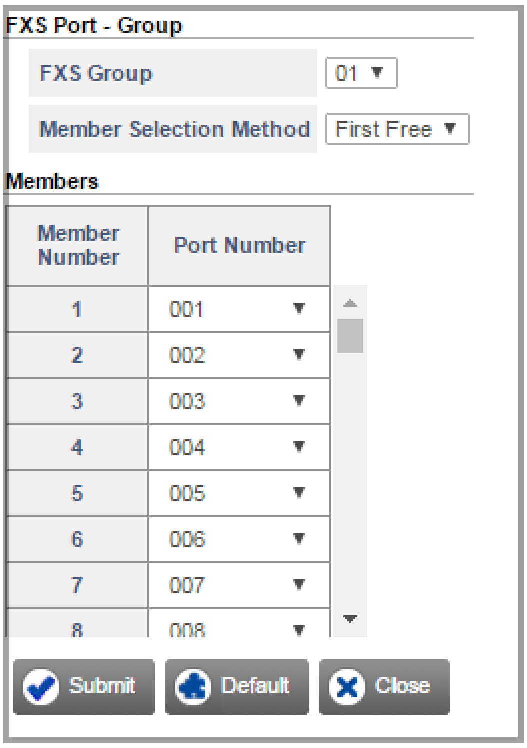
  - Select a **FXS Group**.



  - Select **FXS Group** number. Default:1.

  - Click **Settings** .

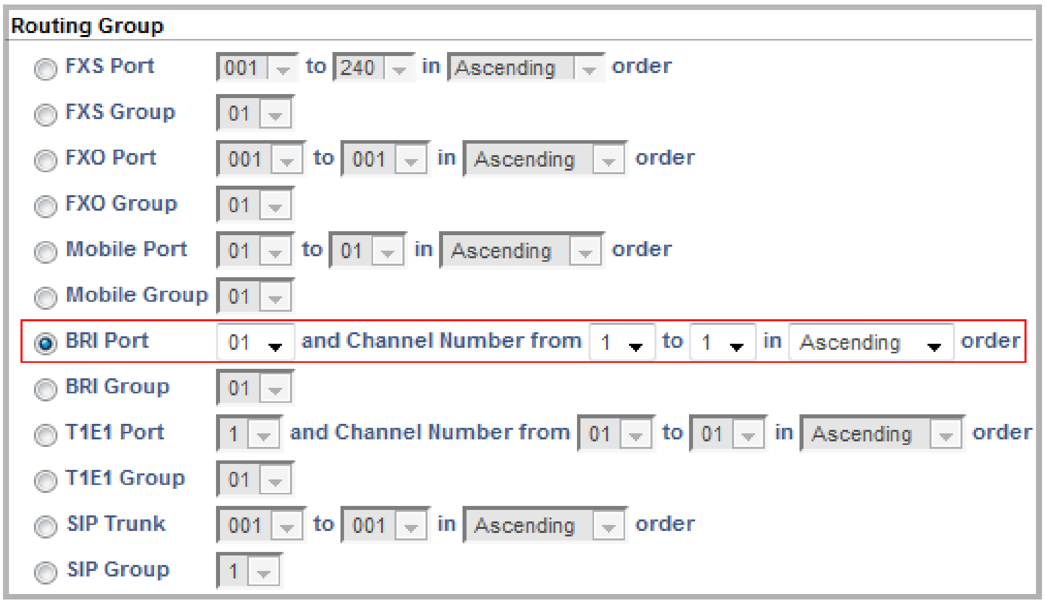
- The **FXS Port - Groups** window opens.



- Create the FXS Group. For detailed instructions on creating groups, see the topic "Group" under *Advanced Settings*.

- Similarly, you can create a group of *sequential* and *not-sequential* FXO Ports, Mobile Ports and SIP Trunks.

- To create a routing group of *sequential* **BRI Channels** as members,

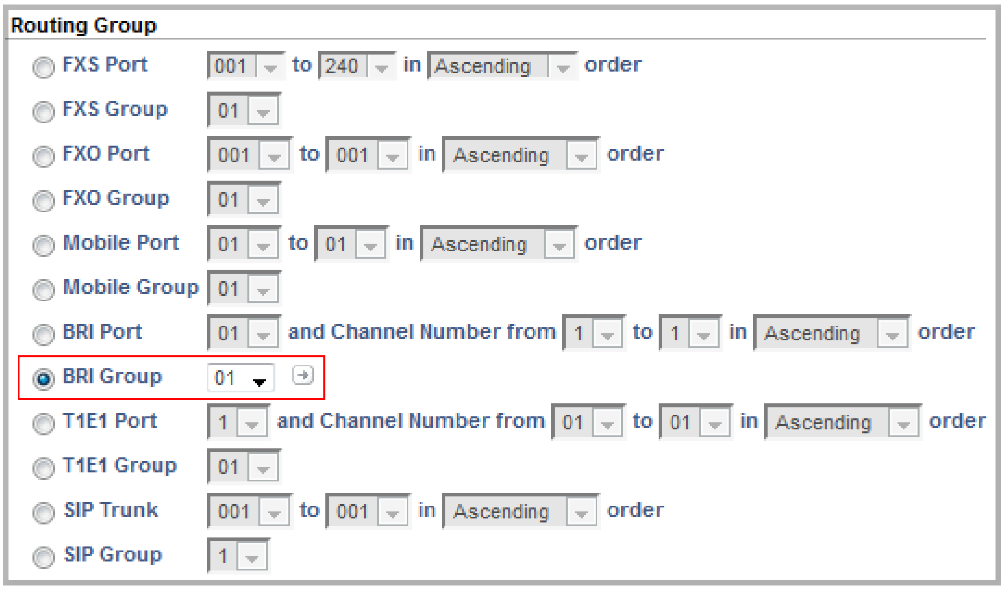


- Select the **BRI Port** Number. Default: 1.

- In Channel Number **From - to**, select the **Start Channel Number** and the **End Channel Number**, respectively.
- In **in - order**, select the order in which the system should hunt for a free member Channel to route the call.

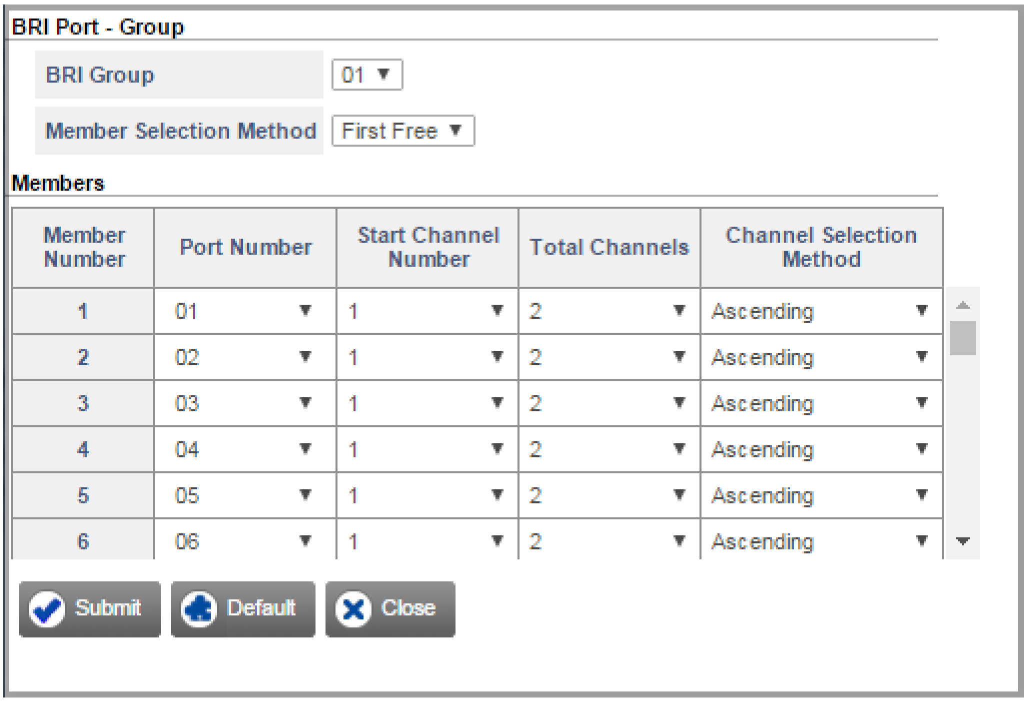
  Select **Ascending** to start hunting from the first to the last member channel. Select **Descending** to start hunting from the last to the first member channel. Default: Ascending.

- To create a group of *not-sequential* **BRI Channels** as members,
  - Select **BRI Group**.



  - Select a **BRI Group** number. Default:1.

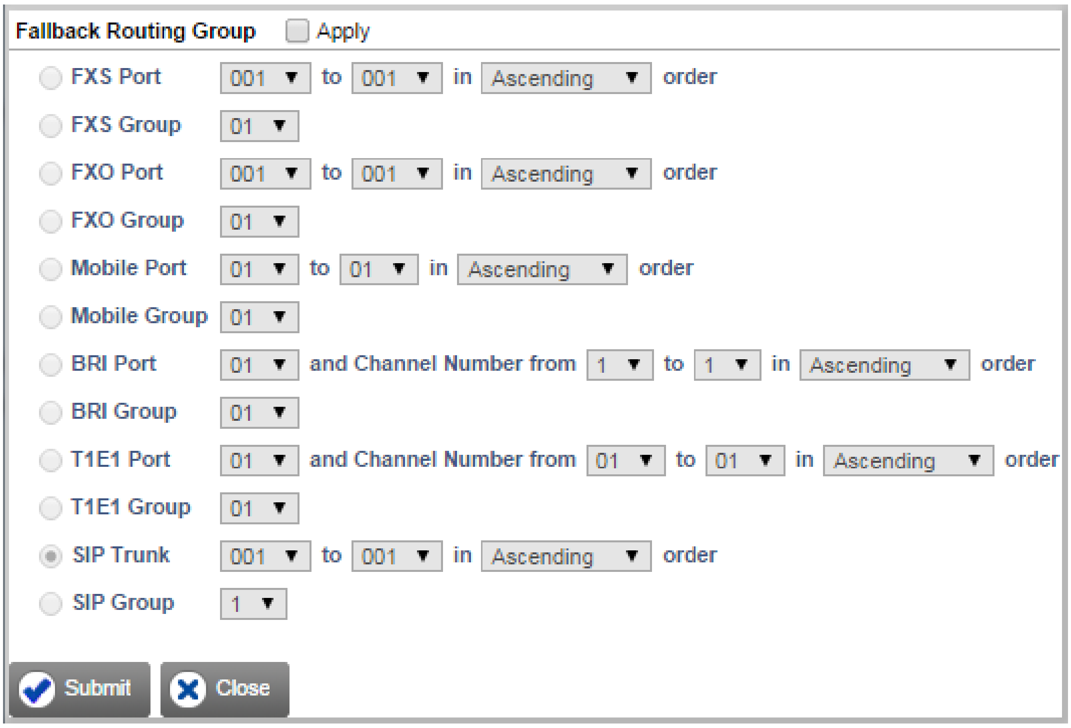
- Click **Settings** . The **BRI Port - Groups** window opens.



- Create the BRI Group. For detailed instructions on creating groups, see the topic “Group” under *Advanced Settings.*

- Similarly, you can create a group of *sequential* and *not-sequential* T1E1 Ports.

- You may create the **Fallback Routing Group**.



- To do this,
    - Select the **Apply** check box.
    - Follow the same instructions provided earlier for creating *sequential* and *not-sequential* groups of FXS Ports, FXO Ports, Mobile Ports, BRI Ports, T1E1 Ports and SIP Trunks.
- Click **Submit** to save changes. The **Edit** window closes.
- The entry you edited appears in the **Destination Port/Group for FXO Port** window.
- Close the **Destination Port/Group for FXO Port** window to return to the Handling of Calls window.

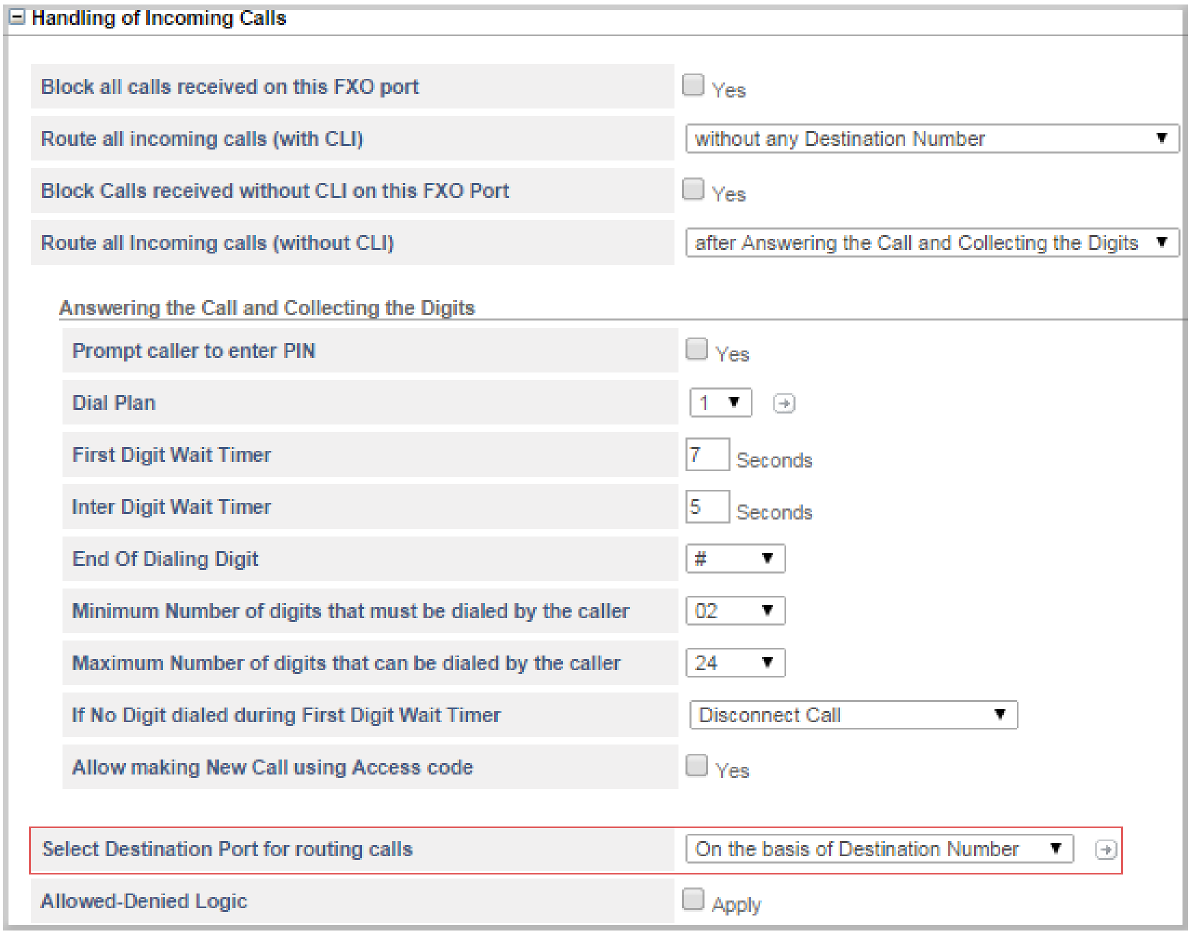
On the basis of Destination Number

In this method, incoming calls on the source port are routed to the destination port on the basis of the destination number (called party number) dialed by the caller.

You must configure the called party numbers in the **Destination Number Based** Table. SARVAM UMG will match the called party number dialed by the caller with the entries of this table. If a match is found for the number in the table, the call is routed to the destination.

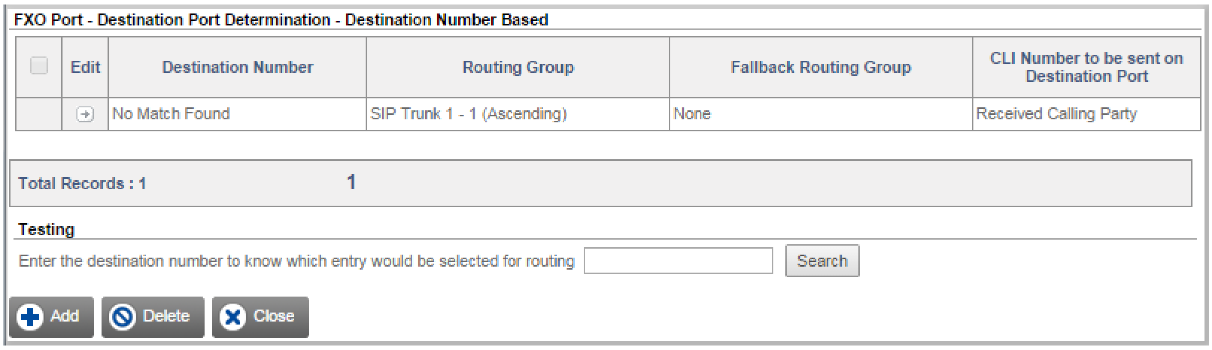
To apply this method, do the following:

- In **Select Destination Port for routing calls**, select **On the basis of Destination Number** option.

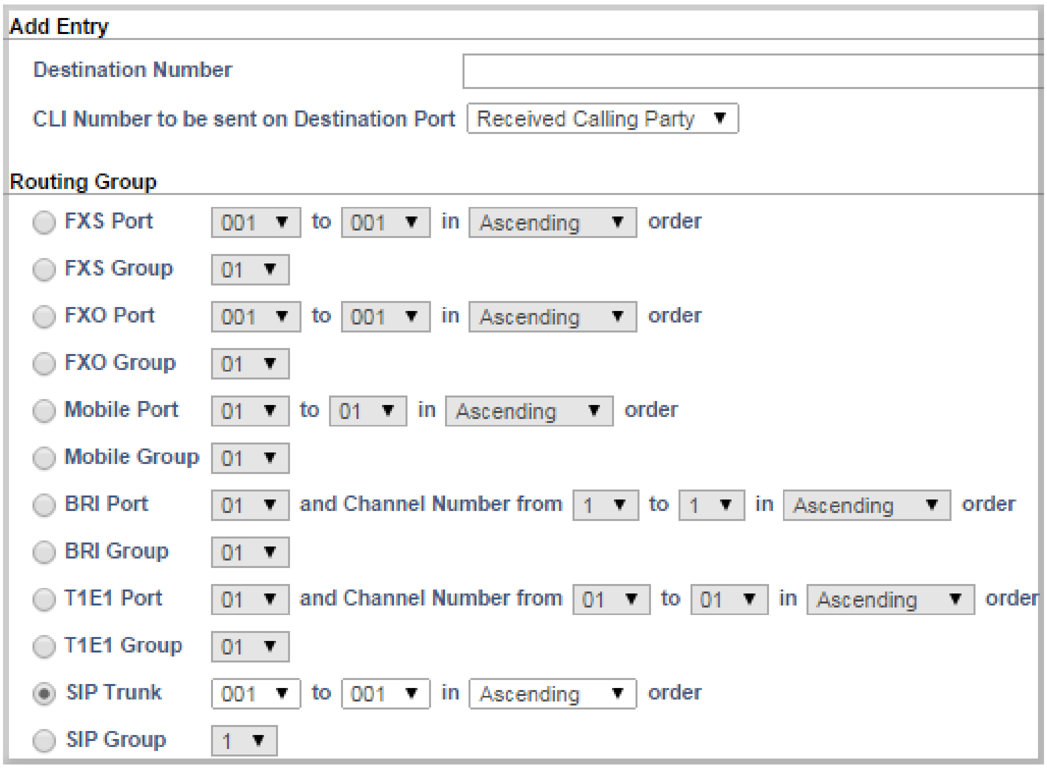


- Click **Settings** .

The **FXO Port - Destination Port Determination - Destination Number Based** table window opens.



- To add a new entry, click **Add**. The **Add Entry** window opens. You can add upto 250 entries.



- In **Destination Number**, enter the number you expect the callers to dial. You may enter upto 64 characters (Digits + "Wildcard Characters") in this field. Valid characters are 0 to 9, *, #, X, T, Comma [,], Hyphen [-], Caret [^]. Default: Blank.

Wildcard Characters

SARVAM UMG supports following characters.

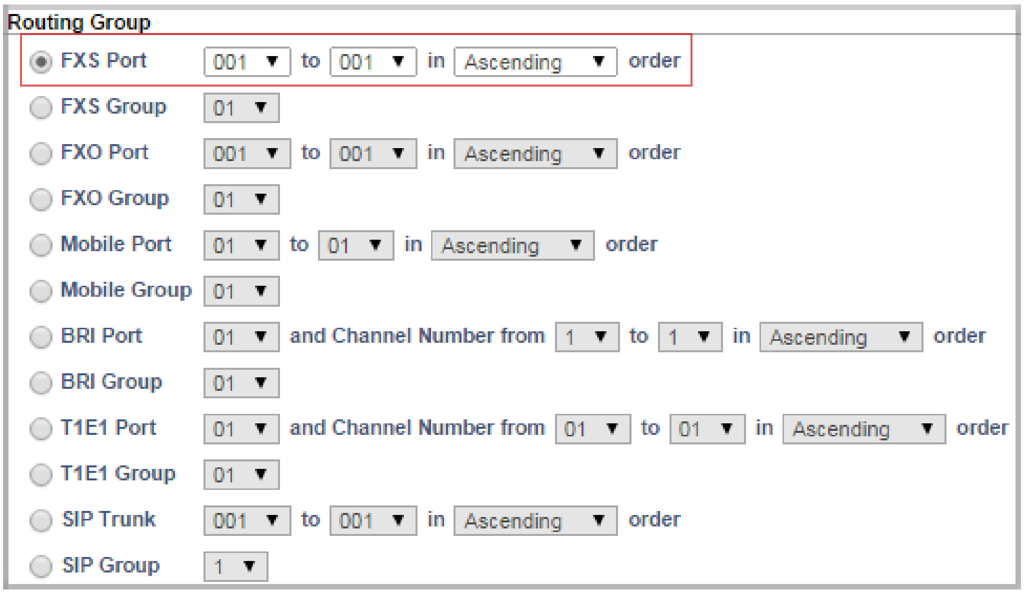
Character	Description
X (letter X)	X represents any single digit from 0 to 9.
#	When # is configured in a number string, it will not be considered as End of Dialing.

*	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.

- Select the **CLI Number to be sent on Destination Port**. You can select Received Calling Party or Received Called Party. Default: Received Calling Party.

CLI to be sent on Destination Port is applicable when FXS Port/Group is determined as the Destination Port.

- Create the **Routing Group**.
    - To create a group of *sequential* **FXS Ports** as members,

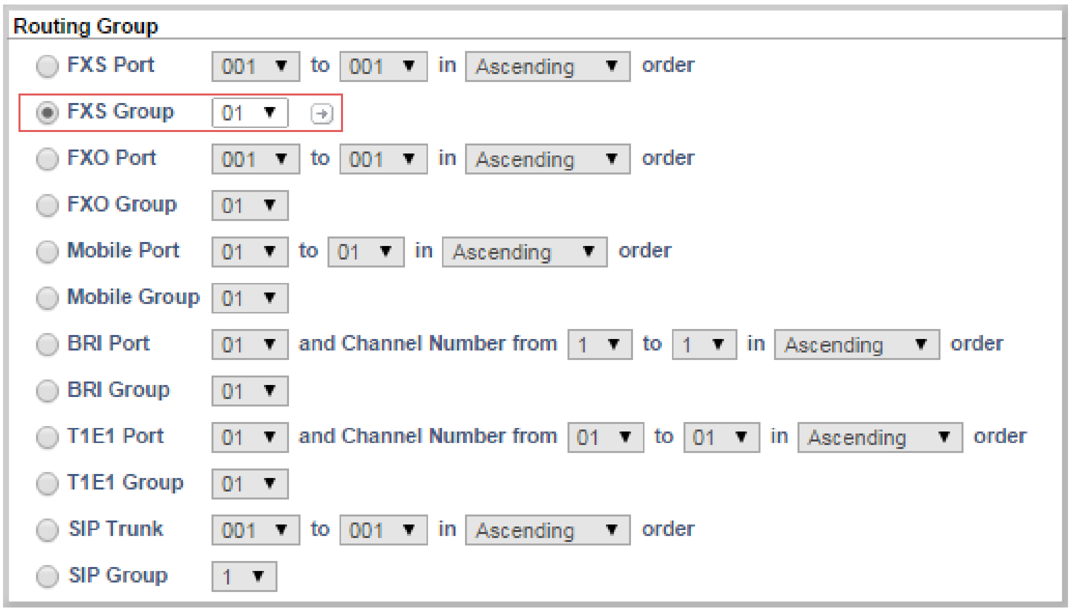


        - Select the desired **FXS Port** numbers as members. Default:1.
        - In **in - order**, select the order in which the system should hunt for a free member FXS Port to route the call.

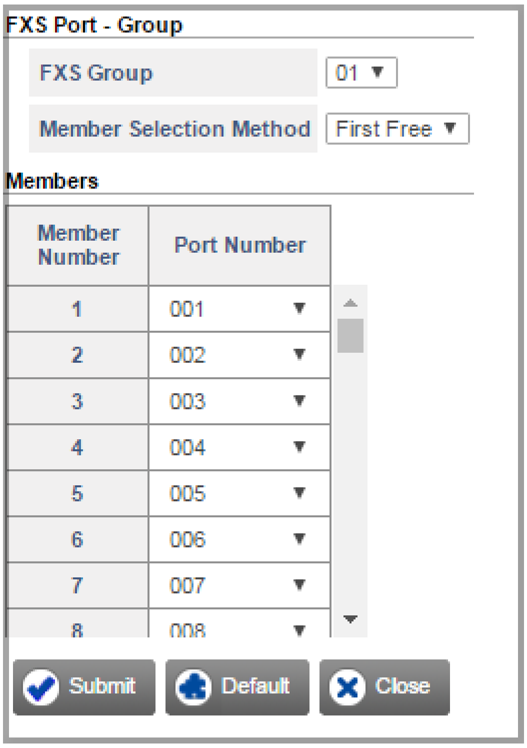
          Select **Ascending** to start hunting from the first to the last member FXS Port. Select **Descending** to start hunting from the last to the first member FXS Port. Default: Ascending.

    - To create a group of *not-sequential* **FXS Ports** as members,

- Select a **FXS Group**.

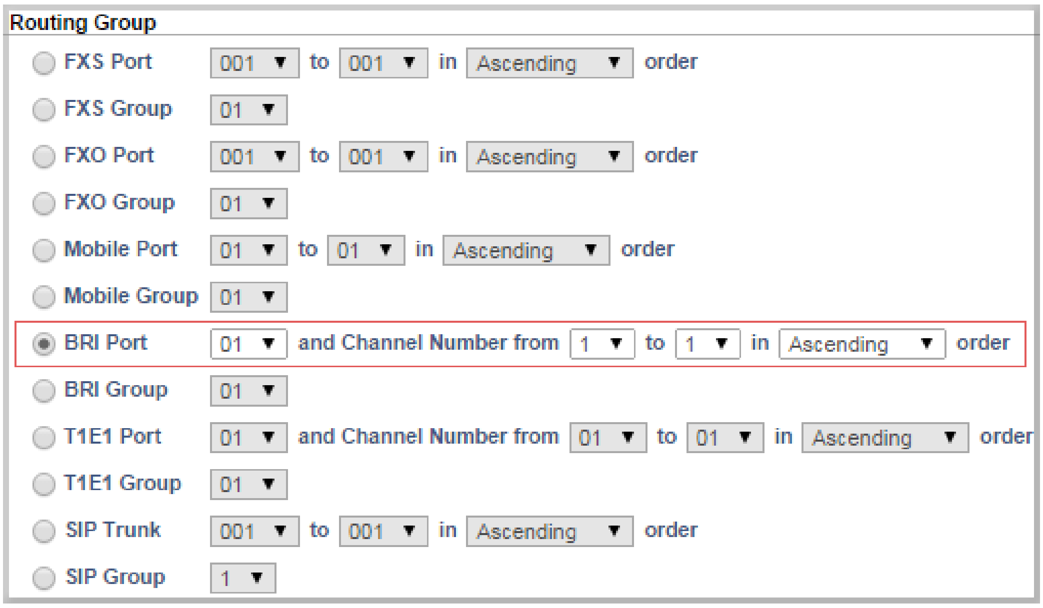


- Select **FXS Group** number. Default:1.
- Click **Settings** .
- The **FXS Port - Groups** window opens.



- Create the FXS Group. For detailed instructions on creating groups, see the topic “Group” under *Advanced Settings*.

- Similarly, you can create a group of *sequential* and *not-sequential* FXO Ports, Mobile Ports and SIP Trunks.
- To create a routing group of *sequential* **BRI Channels** as members,

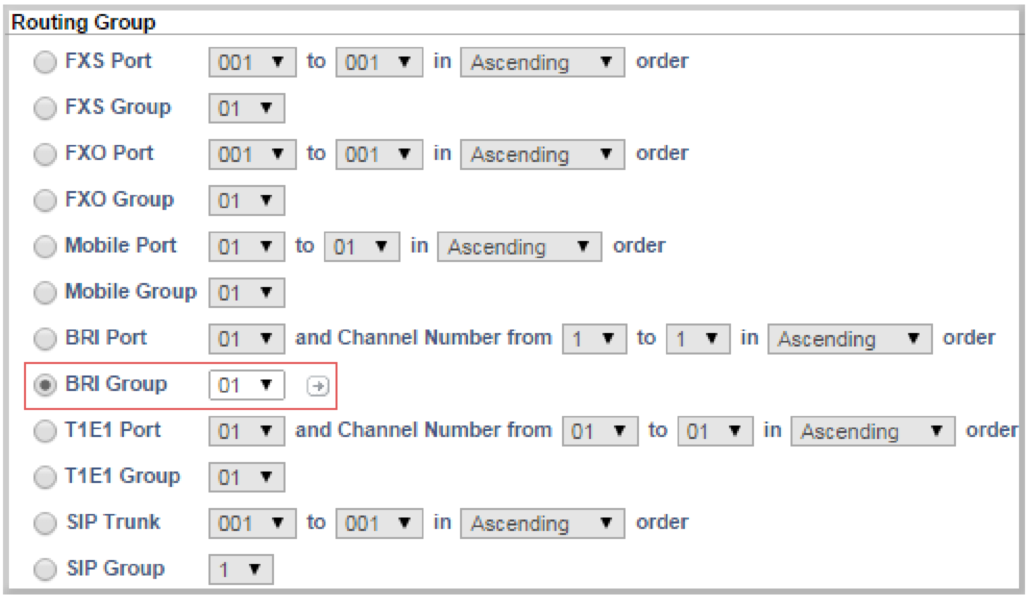


  - Select the **BRI Port** Number. Default: 1.
  - In Channel Number **From - to**, select the **Start Channel Number** and the **End Channel Number**, respectively.
  - In **in - order**, select the order in which the system should hunt for a free member Channel to route the call.

    Select **Ascending** to start hunting from the first to the last member channel. Select **Descending** to start hunting from the last to the first member channel. Default: Ascending.

- To create a group of *not-sequential* **BRI Channels** as members,

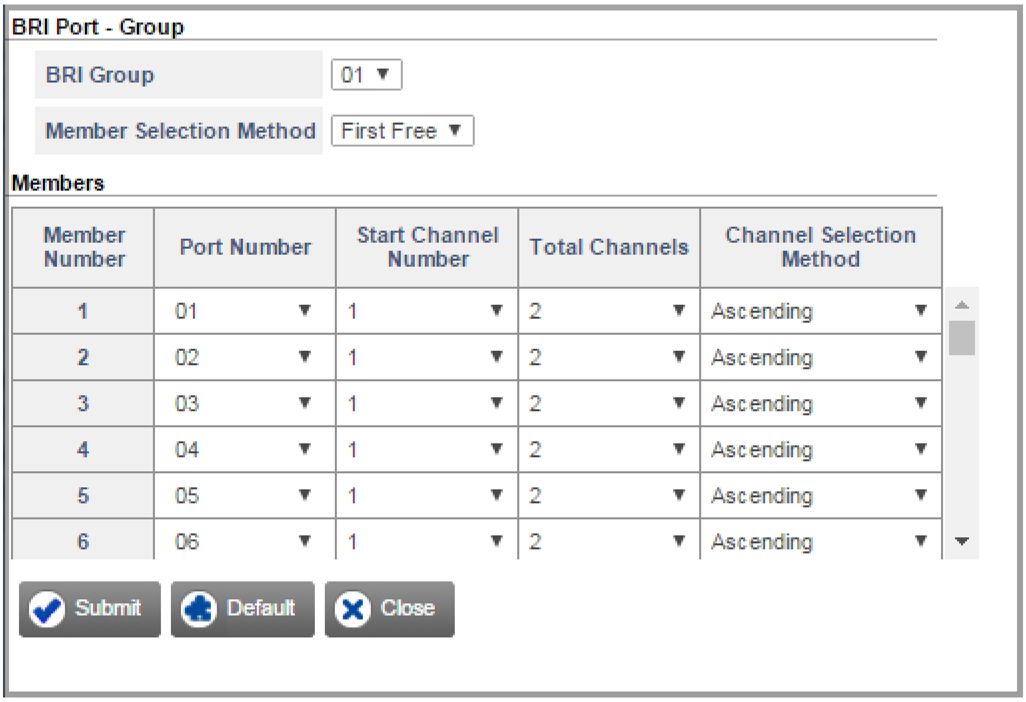
- Select **BRI Group**.



- Select a **BRI Group** number. Default:1.

- Click **Settings** .

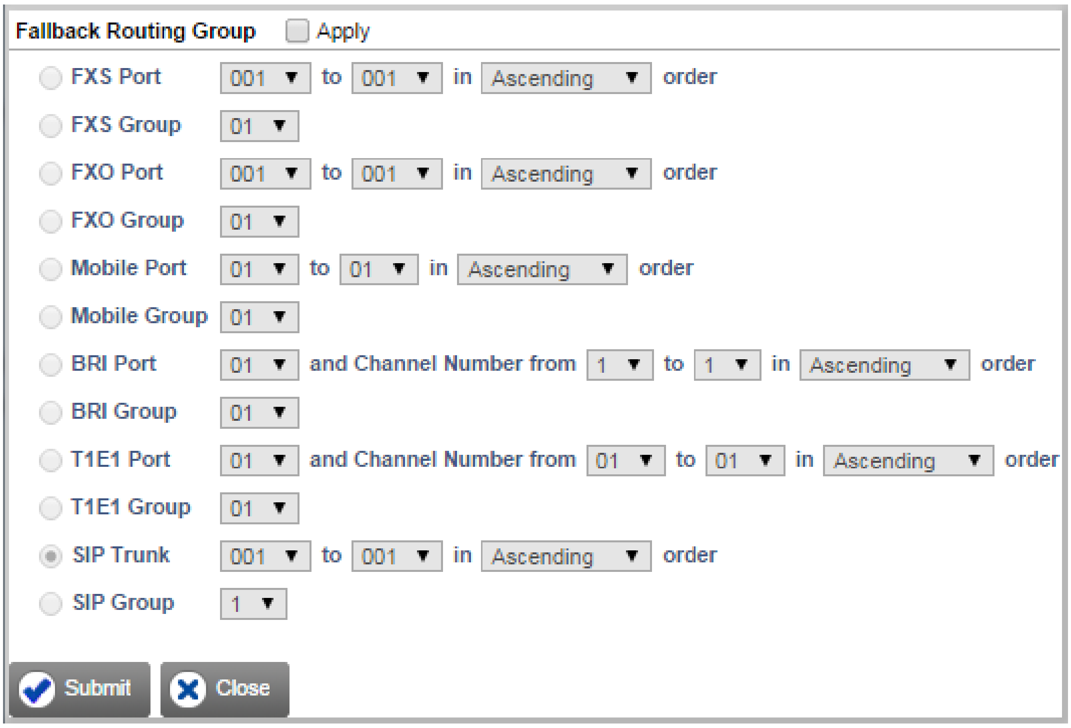
- The **BRI Port - Groups** window opens.



- Create the BRI Group. For detailed instructions on creating groups, see the topic “Group” under *Advanced Settings.*

- Similarly, you can create a group of *sequential* and *not-sequential* T1E1 Ports.

- You may create the **Fallback Routing Group**.



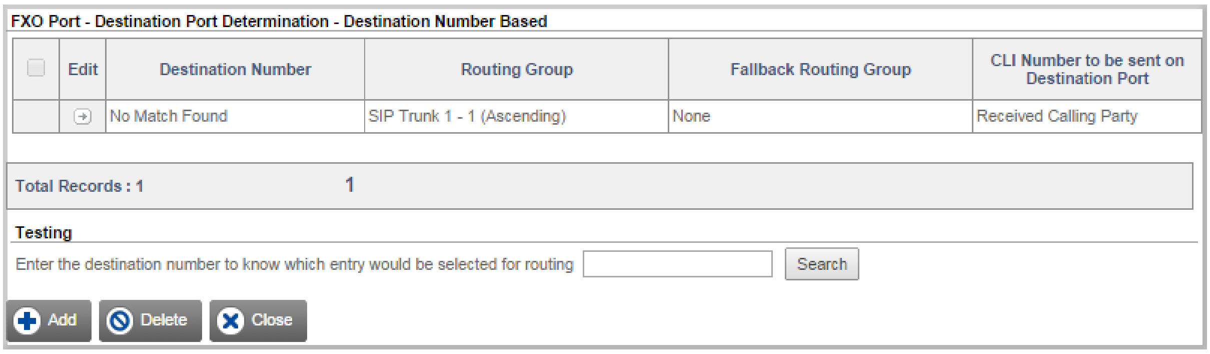
- To do this,
  - Select the **Apply** check box.
  - Follow the same instructions provided earlier for creating *sequential* and *not-sequential* groups of FXS Ports, FXO Ports, Mobile Ports, BRI Ports, T1E1 Ports and SIP Trunks.
- Click **Submit** to save changes. The **Add Entry** window closes.
- The entry you added appears in the **FXO Port - Destination Port Determination - Destination Number Based** table.
- Follow the same steps as above to add another entry to this table.
- To delete an entry, select the check box and click the **Delete** button.

If there are multiple entries in the Destination Number Based table, to search a particular entry in the table, under Testing enter the desired number in the ***Enter the destination number to know which entry would be selected for routing*** *search box.*

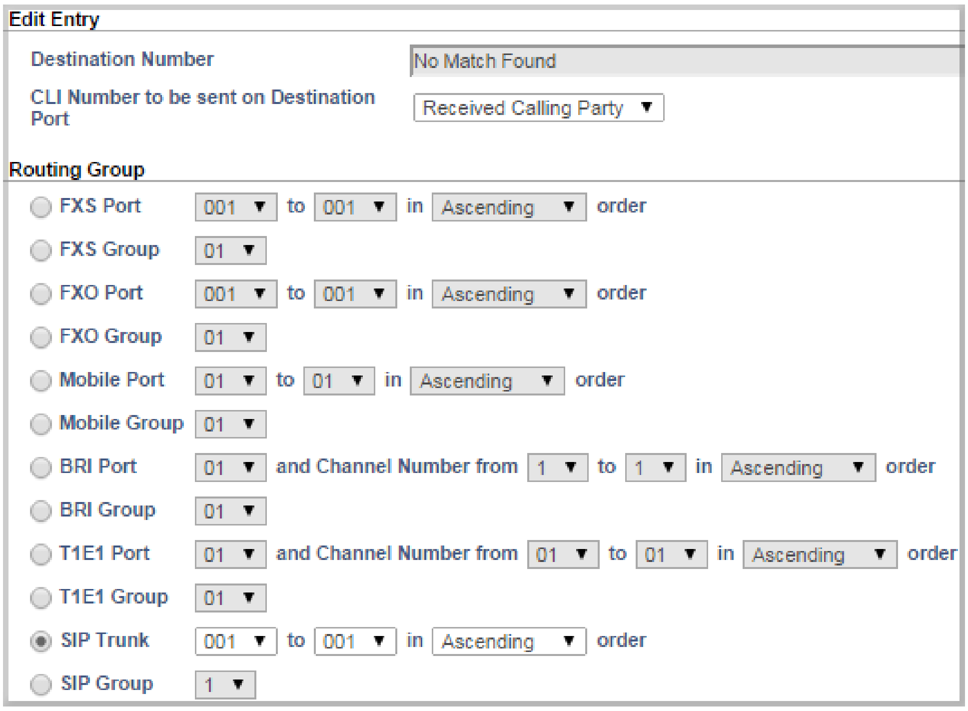
- By default, SIP Trunk 1-1 (Ascending) is assigned as the Routing Group, for routing calls from numbers that do not match with any of the destination numbers you configured (No Match Found).

  To change the default Routing Group and to create the Fallback Routing Group for the No Match Found numbers entry,

- For the No Match Found entry in the table, click **Edit** .



- The **Edit Entry** window opens.



- Create the **Routing Group** and **Fallback Routing Group** as per your requirement.

- Click **Submit** and close the window.

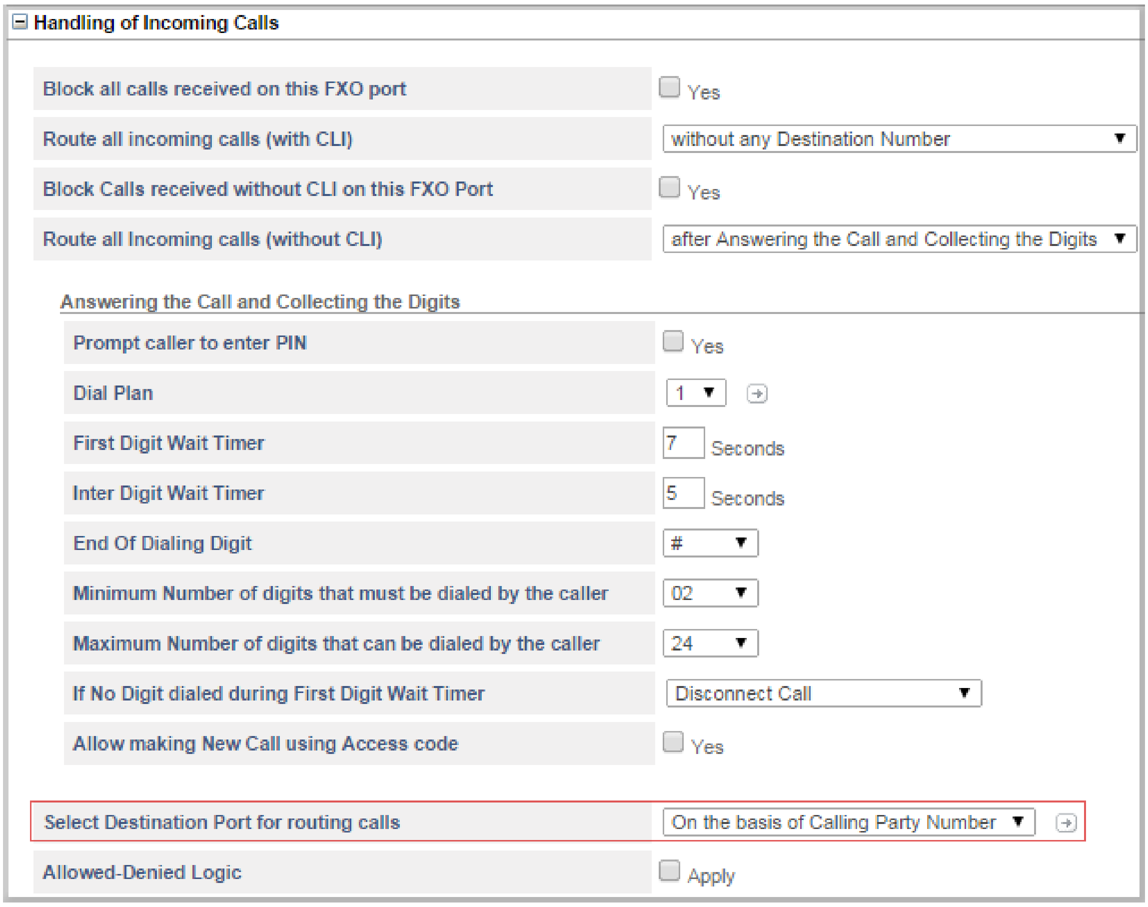
- Close the window if you have finished adding/editing entries.

You can also configure the **Destination Number Based** Table from *Advanced Settings*. For instructions, see “Destination Port Determination” under *Advanced Settings.*

On the basis of Calling Party Number

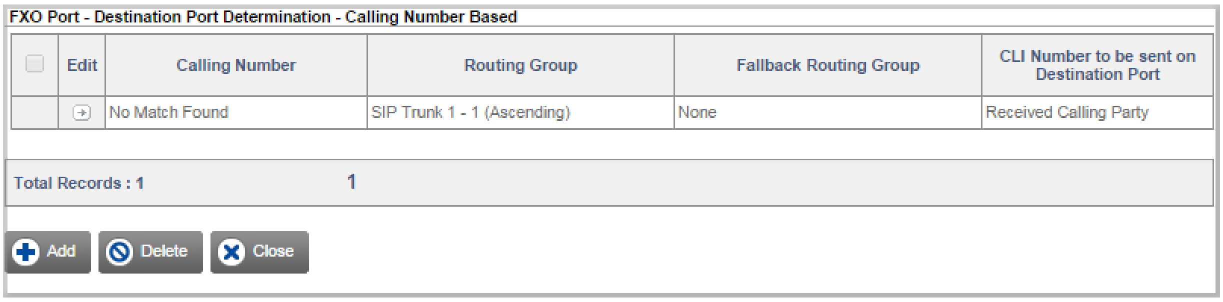
In this method, incoming calls on the FXO Port are routed to a specific port as per the calling party's number. To apply this method, do the following:

- In **Select Destination Port for routing calls**, select **On the basis of Calling Party Number** option.

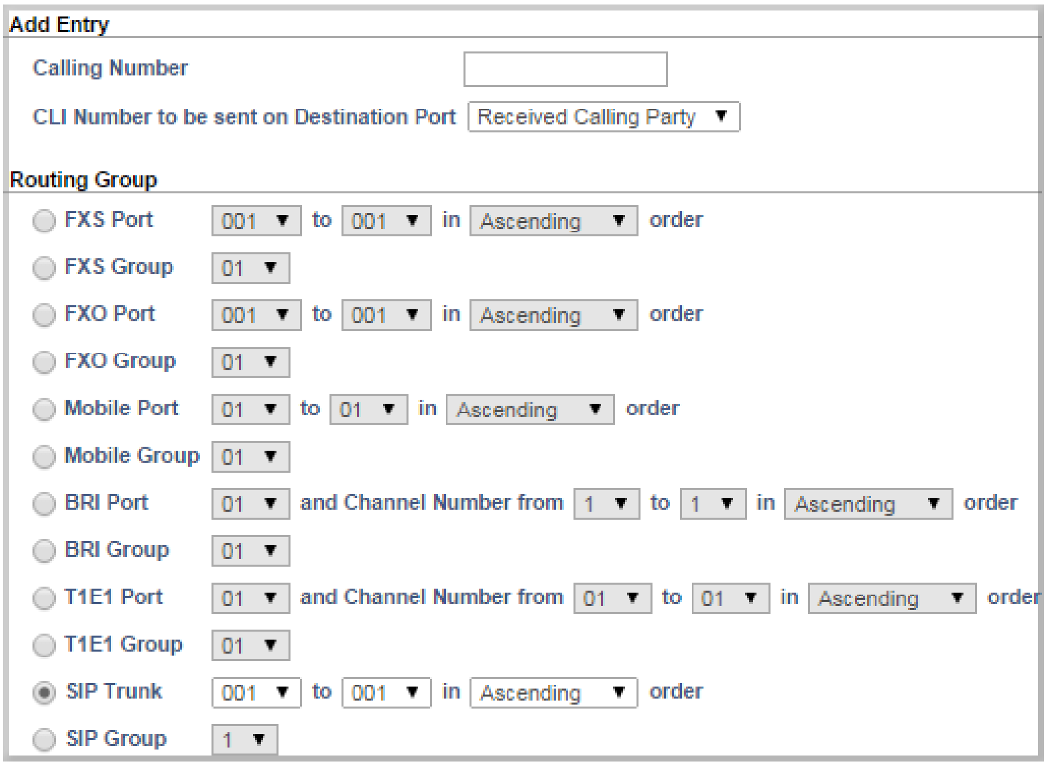


- Click **Settings** .

The **FXO Port - Destination Port Determination - Calling Number Based** table window opens.



- To add a new entry, click **Add**. The **Add Entry** window opens. You can add upto 499 entries.



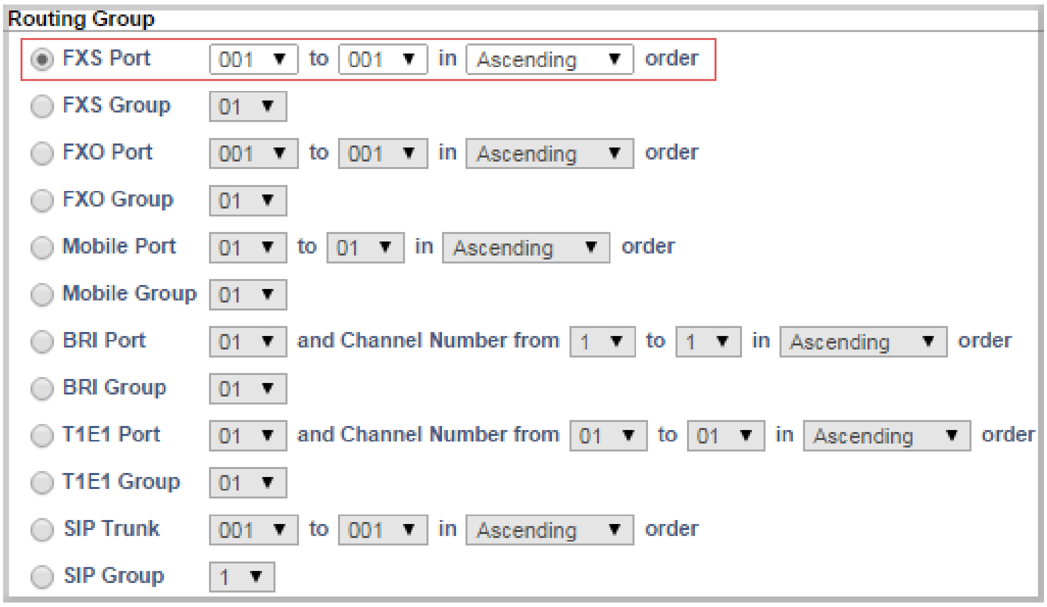
- In **Calling Number**, enter the number (max. 24 characters) from which you expect calls to be received. Valid digits are 0 to 9, *, #, (dot). Default: Blank.

- Select the **CLI Number to be sent on Destination Port**. You can select Received Calling Party or Received Called Party. Default: Received Calling Party.

CLI to be sent on Destination Port is applicable when FXS Port/Group is determined as the Destination Port.

- Create the **Routing Group**.

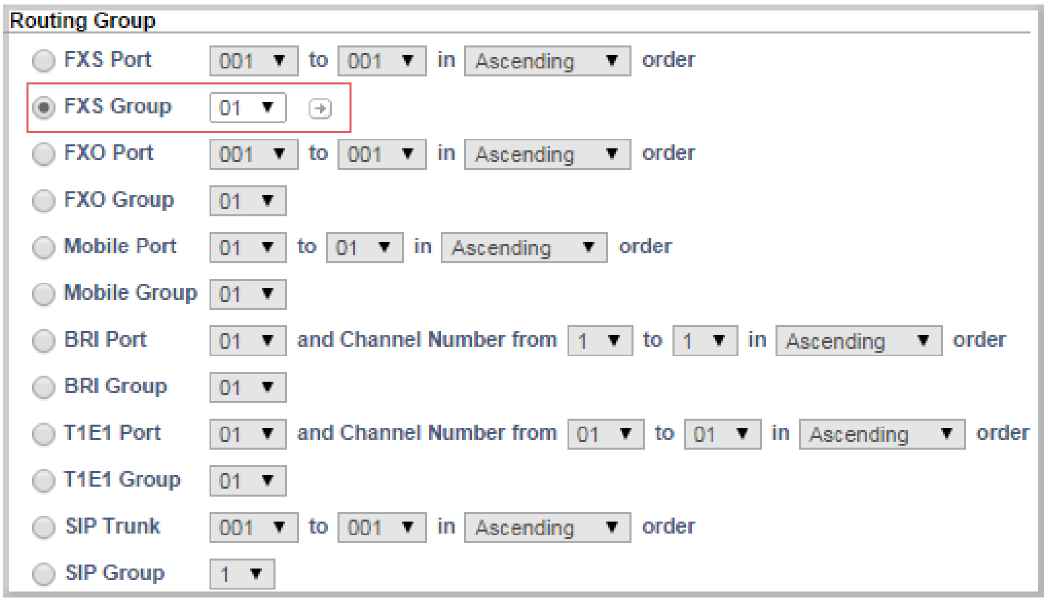
- To create a group of *sequential* **FXS Ports** as members,



  - Select the desired **FXS Port** numbers as members. Default:1.
  - In **in - order**, select the order in which the system should hunt for a free member FXS Port to route the call.

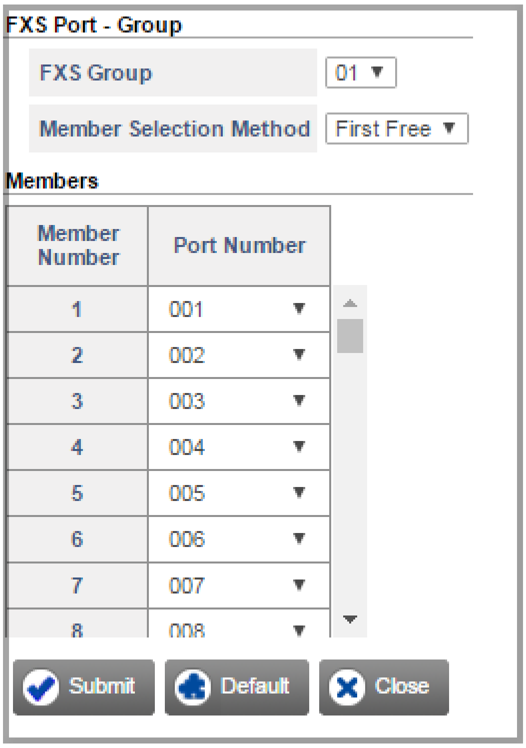
    Select **Ascending** to start hunting from the first to the last member FXS Port. Select **Descending** to start hunting from the last to the first member FXS Port. Default: Ascending.

- To create a group of *not-sequential* **FXS Ports** as members,
  - Select a **FXS Group**.



  - Select **FXS Group** number. Default:1.

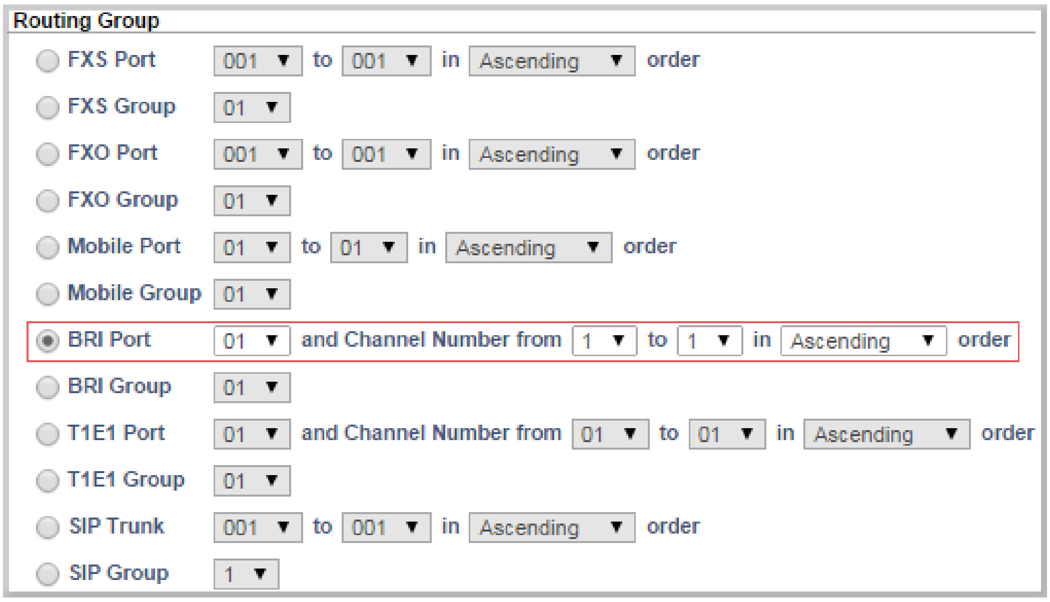
- Click **Settings** .
- The **FXS Port - Groups** window opens.



- Create the FXS Group. For detailed instructions on creating groups, see the topic “Group” under *Advanced Settings*.

- Similarly, you can create a group of *sequential* and *not-sequential* FXO Ports, Mobile Ports and SIP Trunks.

- To create a routing group of *sequential* **BRI Channels** as members,



- Select the **BRI Port** Number. Default: 1.

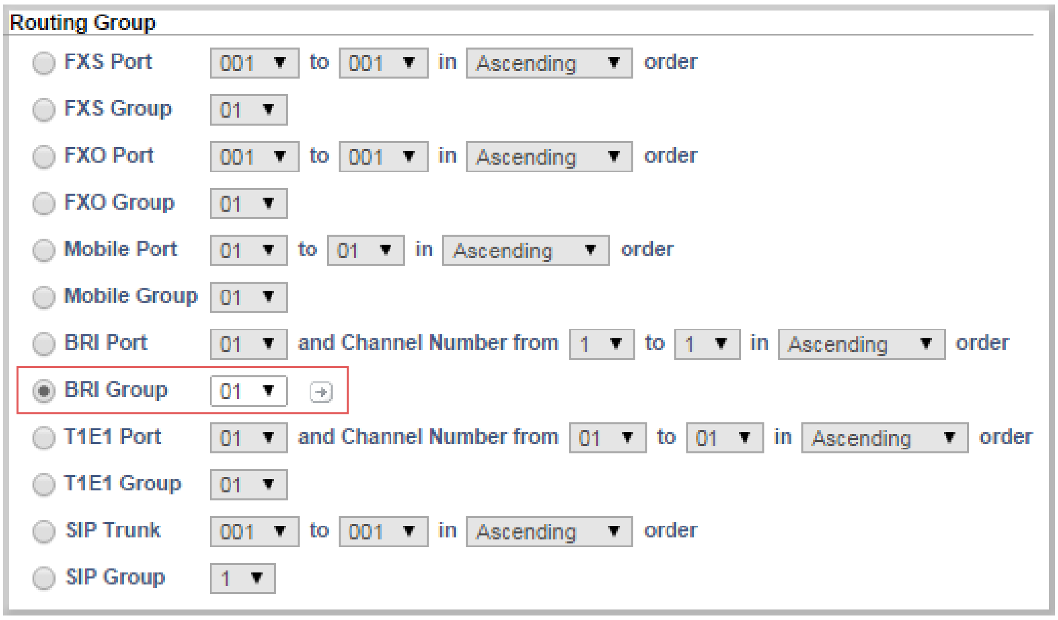
- In Channel Number **From - to**, select the **Start Channel Number** and the **End Channel Number**, respectively.

- In **in - order**, select the order in which the system should hunt for a free member Channel to route the call.

  Select **Ascending** to start hunting from the first to the last member channel. Select **Descending** to start hunting from the last to the first member channel. Default: Ascending.

- To create a group of *not-sequential* **BRI Channels** as members,

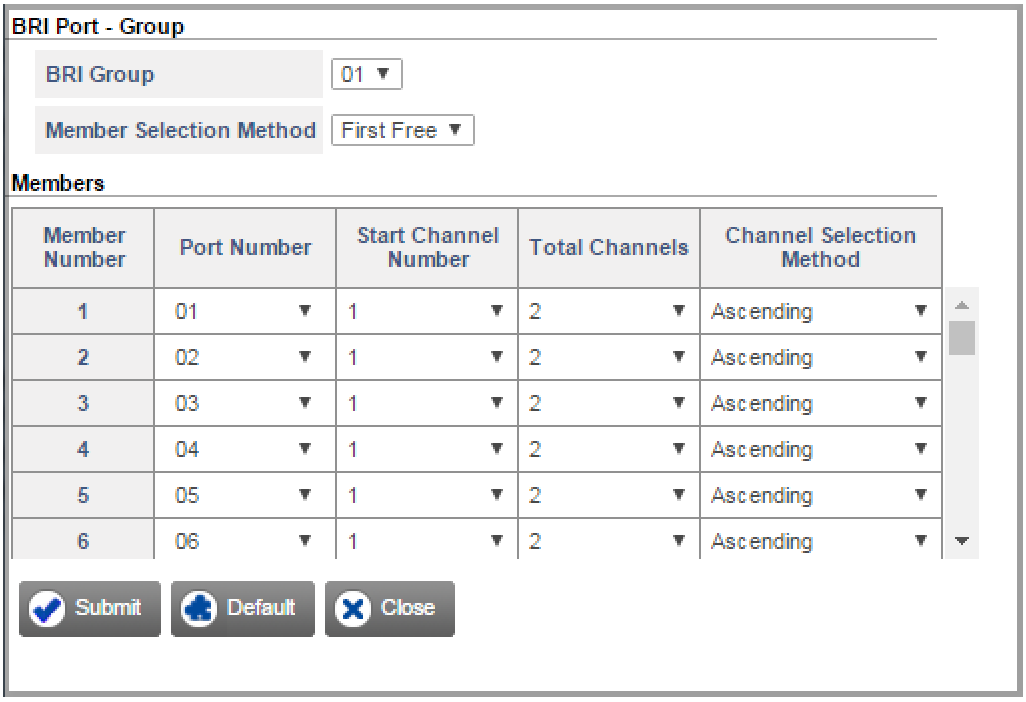
  - Select **BRI Group**.



  - Select a **BRI Group** number. Default:1.

  - Click **Settings** .

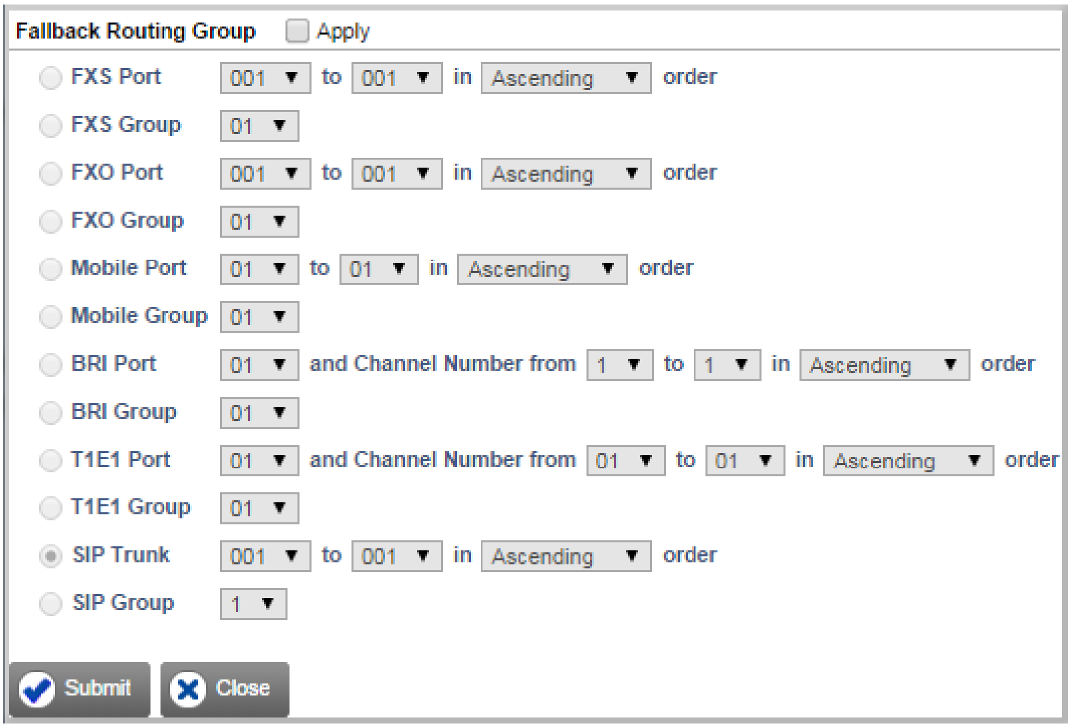
- The **BRI Port - Groups** window opens.



- Create the BRI Group. For detailed instructions on creating groups, see the topic "Group" under *Advanced Settings.*

- Similarly, you can create a group of *sequential* and *not-sequential* T1E1 Ports.

- You may create the **Fallback Routing Group**.



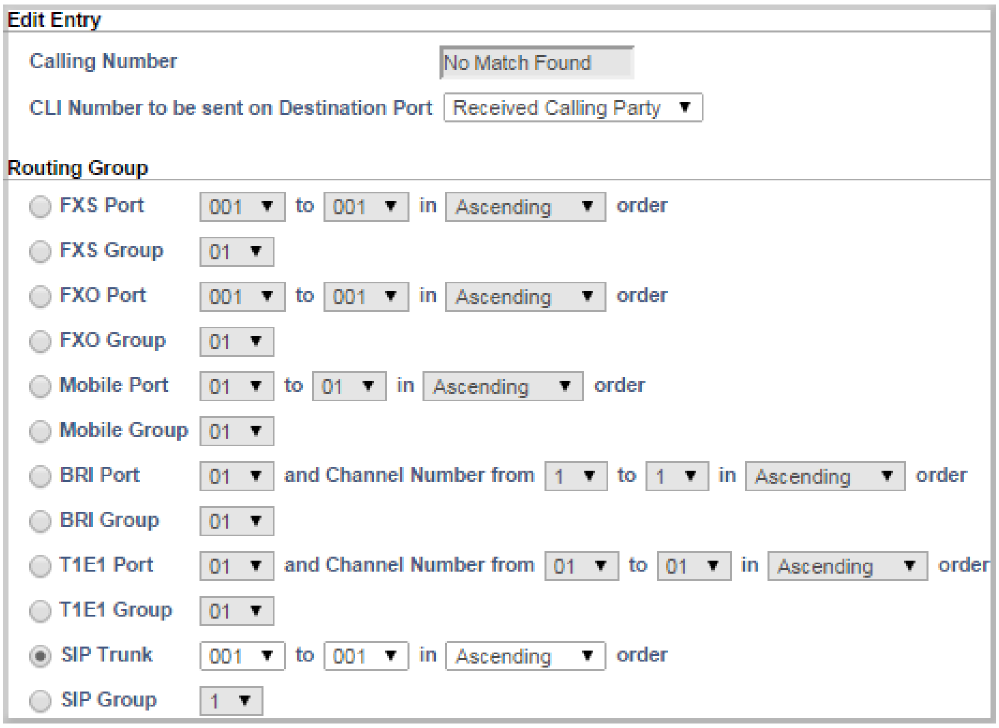
- To do this,

- Select the **Apply** check box.
- Follow the same instructions provided earlier for creating *sequential* and *not-sequential* groups of FXS Ports, FXO Ports, Mobile Ports, BRI Ports, T1E1 Ports and SIP Trunks.

- Click **Submit** to save changes. The **Add Entry** window closes.
- The entry you added appears in the **FXO Port - Destination Port Determination - Calling Number Based** table.
- Follow the same steps as above to add another entry to this table.
- To delete an entry, select the check box and click the **Delete** button.
- By default, SIP Trunk 1-1 (Ascending) is assigned as the Routing Group, for routing calls from numbers that do not match with any of the destination numbers you configured (No Match Found).

  To change the default Routing Group and to create the Fallback Routing Group for the No Match Found numbers entry,

  - For the No Match Found entry in the table, click **Edit** .



  - The **Edit Entry** window opens.
  - Create the **Routing Group** and **Fallback Routing Group** as per your requirement.
  - Click **Submit** and close the window.

- Close the window if you have finished adding/editing entries.

You can also configure the **Calling Number Based** Table from *Advanced Settings*. For instructions, see "Destination Port Determination" under *Advanced Settings.*

Allowed - Denied Logic

You can apply the Allowed-Denied logic on the FXO Port (source port) if you want to allow or restrict the dialing of particular numbers. You can use this feature for Toll Control.

The Allowed-Denied Number Logic makes use of two Number lists:

- **Allowed Numbers List:** This is the list of numbers that can be dialed by the caller on the FXO Port.

- **Denied Numbers List:** This list contains the numbers that are to be restricted from being dialed by the caller on the FXO Port.

When Allowed-Denied Logic is enabled on a source port, for each number dialed from the port, SARVAM UMG uses the best-match-found logic to compare the dialed number with the Allowed Number list and the Denied Number list.

The number is allowed to be dialed, if it:
- matches with both lists.
- matches with Allowed Number list, but not with the Denied Number list.
- matches with neither the Allowed List nor the Denied List.

The number is denied, if it matches with the Denied Number list, but not with the Allowed Number list.

The system does not apply the Allowed-Denied Logic:
- When dialed number string matches with any Access Code.
- When dialed number string matches with any Emergency Number.
- When any one of the following is selected to Route all Incoming Calls (with CLI):
  - on the basis of Calling Party Number
  - to a Fixed Destination Number

To apply Allowed - Denied Logic on the FXO Port,

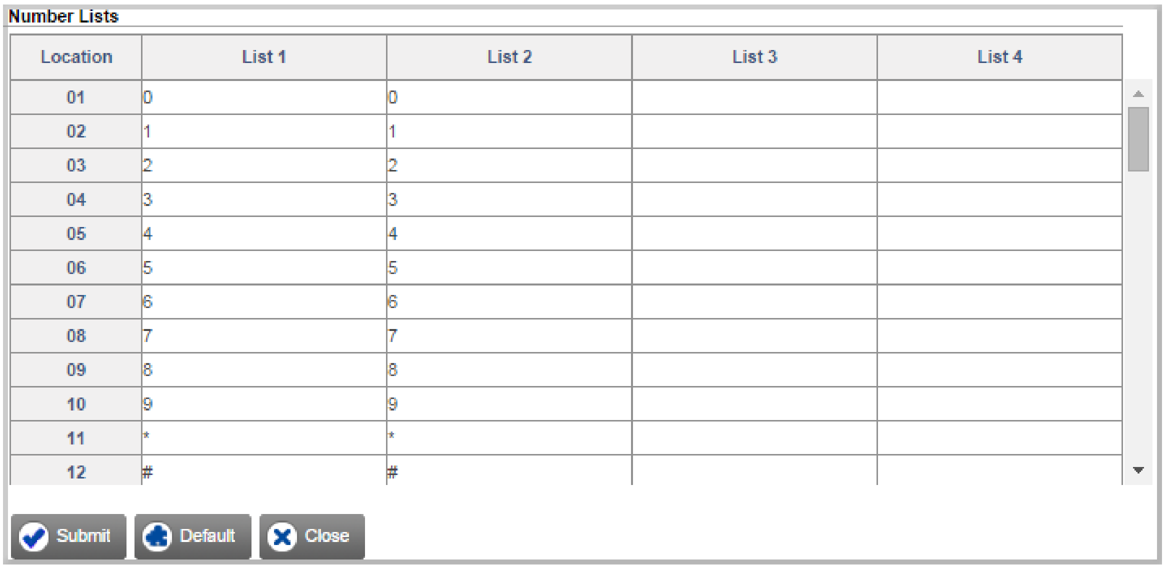
- Select the **Allowed - Denied Logic** check box.

  - In the **Allowed Number List**, select the list number you have configured with numbers you want to allow to be dialed out from the FXO Port. Default: 01

    If you have not configured the Allowed Number List,

    - Click **Settings** .

- The Number Lists window opens.



- You may configure the default Allowed Number List or any other list. See "Number Lists" to configure the allowed numbers.
- Click **Submit** to save the Allowed Number List and close the window.

- In the **Denied Number List**, select the list number you have configured with numbers you want to restrict to be dialed out from the FXO Port. Default: 02.

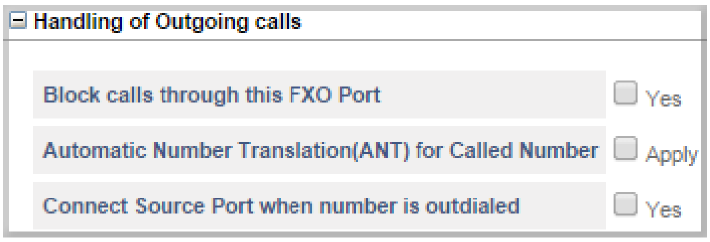
  If you have not configured the Denied Number List,

  - Click **Settings** . The Number Lists window opens.
  - You may configure the default Denied Number List or any other list. See "Number Lists" to configure the restricted numbers.
  - Click **Submit** to save the Denied Number List and close the window.

Handling of Outgoing Calls

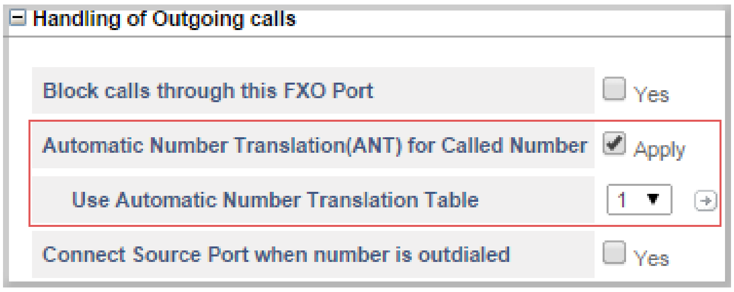
When a FXO Port is determined as the destination port, the numbers dialed from this port constitute outgoing calls.

- Click **Handling of Outgoing Calls** to expand.



- If you do not want to route outgoing calls through this port, select the **Block calls through this FXO Port Yes** check box.

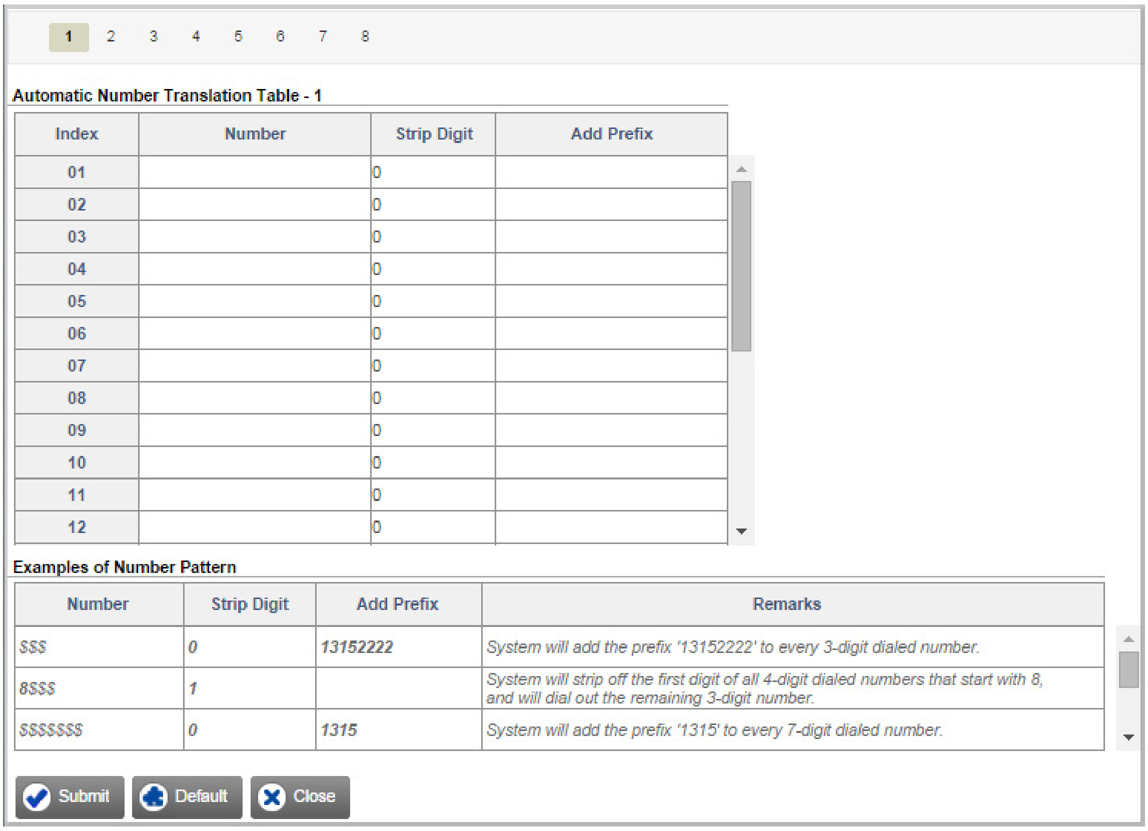
- You can apply **Automatic Number Translation logic** on outgoing calls made from the FXO Port.
    - To apply ANT logic on the Called Numbers, select the **Automatic Number Translation (ANT) for Called Number** check box. Default: Disabled.



- In **Use Automatic Number Translation Table**, select the ANT Table number you have configured for the FXO Port. Default: Table 1.

  If you have not configured the Automatic Number Translation Table,

    - Click **Settings** .
    - The **Automatic Number Translation Table** window opens.



- You may configure the default Automatic Number Translation Table or any other Table. See "Automatic Number Translation (ANT)" to configure the ANT Table.
- Click **Submit** to save the ANT Table and close the window.

    - Return to ANT parameter and assign the ANT Table you configured.

  - Click **Submit** to apply List.

- Select the **Connect Source Port when number is outdialed** check box to enable. This will connect the Source Port with the Destination Port without waiting for the call on the Destination Port to mature. Default: Disabled.

  In all Destination Number Determination methods except *After Answering the Call and Collecting the Digits*, the Source Port gets connected to the Destination Port only after the call has matured, i.e. the called party has answered the call. Until the call matures, the caller hears only Ring Back Tone played by the network.

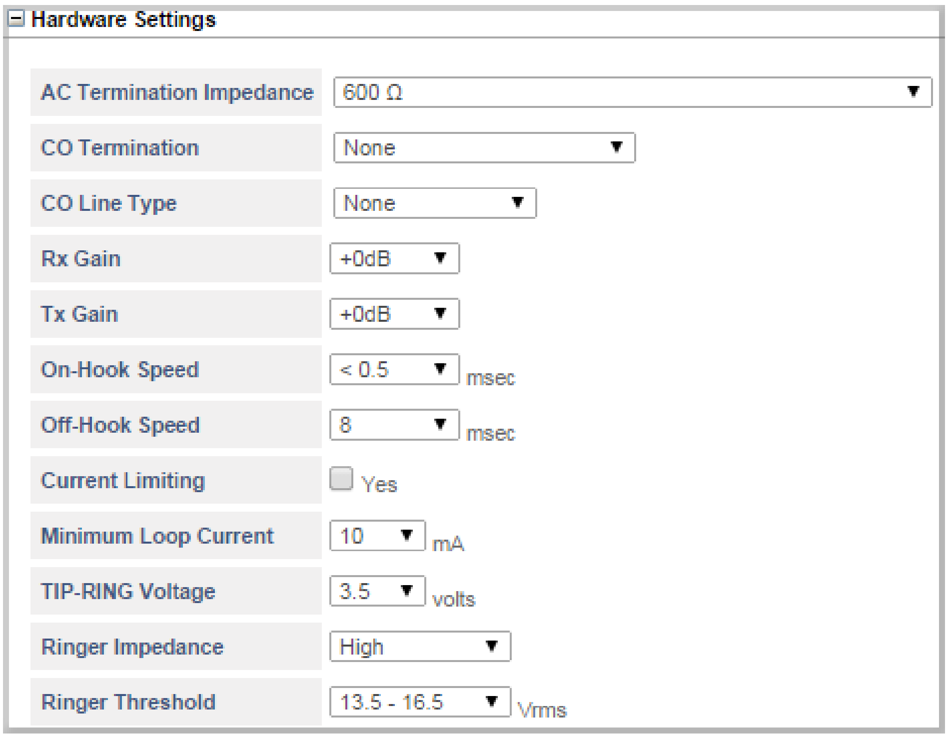
  By connecting the Source Port with the Destination Port immediately after the number is dialed, the caller can know the state of the call — called party is busy, not responding, not reachable or is rejecting the call.

- Click **Submit** to save.

*If you enable **Connect Source Port when number is outdialed**, you will not be able to provide the features "Making a New Call using Access Code" and "Disconnecting a Call using Access Code" to users.*

Hardware Settings

- Click **Hardware Settings** to expand.



- In **AC Termination Impedance**, select the appropriate Impedance of the FXO Port as per the AC Termination Impedance supported by your CO Network. Default: 600 Ω.

- In **CO Termination**, select the appropriate line impedance match. This would depend on the region where SARVAM UMG is deployed. Default: None.

This parameter allows you to increase near-end echo cancellation on the FXO Port. Near-end echo is primarily caused due to the mismatch between AC Termination Impedance (presented by FXO Port of SARVAM UMG to the line) and CO Termination (Impedance presented by the Central Office to the line) as well as 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, CO Termination and a Line Type that closely models the line connecting the FXO Port of SARVAM UMG to the Central Office.

- In **CO Line Type**, select a line type that closely models the line connecting SARVAM UMG to the Central Office. You may select a specific EIA line model (EIA-0 to EIA-7) or a specific wire gauge with it's length (2000 ft. 22/24/26awg). Default: None.

You are recommended to conduct the AC Impedance Test for the line connected to the FXO Port to determine the most appropriate values for the AC Termination Impedance, CO Termination and the CO Line Type. For more information see the topic “AC Impedance Test (FXO)”.

- **Rx Gain** enables you to adjust the volume level of the speech received from the remote party. Select the Rx Gain accordingly. Default: +0dB.

- **Tx Gain** enables you to adjust the volume level of the speech transmitted to the remote party. Select the Tx Gain accordingly. Default: +0dB.

- In **On-Hook Speed**, set the time period required for the line-side device (DAA) to go On-hook.

  It is the time duration between On-hook bit clearance until loop current equals zero. You can select — <0.5, 3 or 26. Default: <0.5 msec.

- In **OFF- Hook Speed**, set the time that would be required by the line transients to settle. Only after this time period, the transmission or reception can take place. You can select — 512, 128, 64 or 8. Default: 8 msec.

- If you want to limit the loop current, select the **Current Limiting** check box. The Loop Current will be limited to a maximum of 60mA. Default: Disabled.

- Set the **Minimum Loop Current** required by the line-side device (DAA) to operate. You can select — 10, 12, 14 or 16 — as per your requirement. Default: 10 mA.

- Set the **TIP-RING Voltage (Volts)** to adjust the TIP/Ring Voltage on the line side. You can select — 3.1, 3.2, 3.35 or 3.5. Default: 3.5 volts.

  In countries where low voltage is required, use lower TIP/RING voltage. Adjust the values of the TIP-RING Voltage to match your country requirements.

- Set the **Ringer Impedance** — High or Synthesized — for the FXO Port according to your country-specific requirement.

  High signifies 20 MΩ Ringer Impedance. This is the default Ringer Impedance provided on the line side by the DAA module of the FXO Port. The DAA Module can provide higher impedance when **Synthesized** impedance is selected.

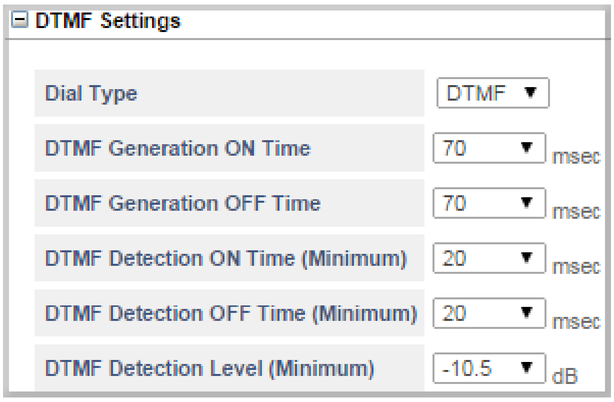
  Some countries like Poland, South Africa and Slovenia require higher ring impedance which is achieved by the DAA module, when Ringer Impedance is set to Synthesized impedance. Default: High.

- Set the **Ringer Threshold** to the desired value. This parameter defines the level below which the FXO Port would not validate the Ring signal and the level above which it would validate the Ring signal. Set Ringer Threshold — 13.5 - 16.5, 19.35 - 23.65 or 40.5 - 49.5. Default: 13.5 - 16.5 Vrms.

- Click **Submit** to save.

DTMF Settings

- Click **DTMF Settings** to expand.



- Select the appropriate **Dial Type** option as supported by your CO Network. You can select — Pulse or DTMF. Default: DTMF.

  - If you have selected **Dial Type** as **Pulse**, you must set the **Pulse Dial Ratio** as per your country. You can select — 40:60, 50:50 or 33:67. Default: 33:67.

- Select the appropriate **DTMF Generation ON Time** for the FXO Port. This is the time for which the DTMF tone is generated. Valid range is 50 to 200 msec. Default: 70 msec.

- Select the appropriate **DTMF Generation OFF Time** for the FXO Port. This is the time for which the system should wait before dialing the successive DTMF digits so that the CO Network can detect the dialed digits. Valid range is 50 to 200 msec. Default: 70 msec.

- Select the appropriate **DTMF Detection ON Time (Minimum)** for the FXO Port. This is the minimum time period for which the DTMF signal should be present in order to be detected. Valid range is 20 to 200 msec. Default: 20 msec.

- Select the appropriate **DTMF Detection OFF Time (Minimum)** for the FXO Port. This is the minimum time period between successive DTMF digits. Valid range is 20 to 200 msec. Default: 20 msec.

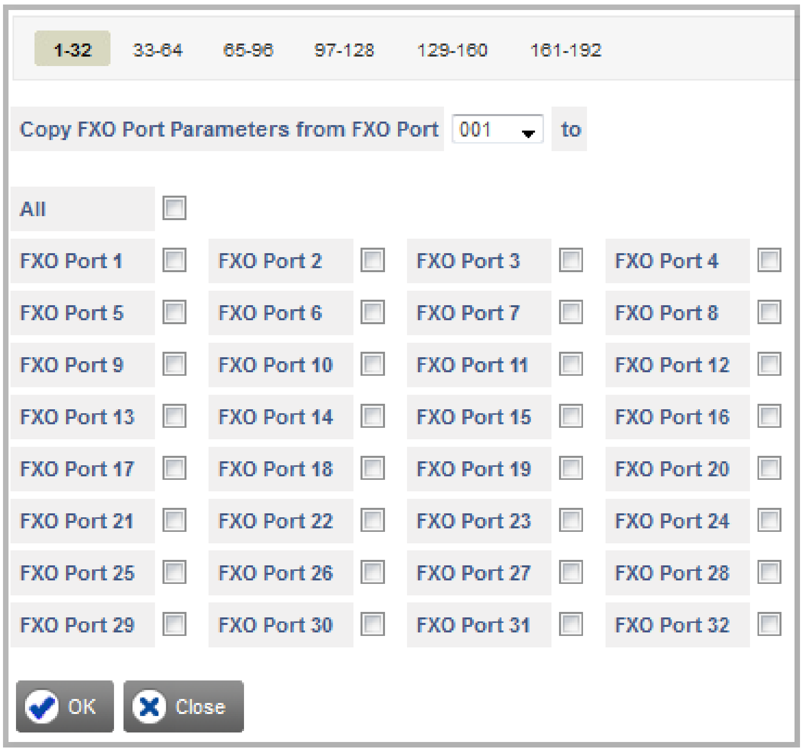
- Select the appropriate **DTMF Detection Level (Minimum)** for the FXO Port. This is the minimum level (dB) of the DTMF digit to be considered as valid. Default: -10.5 dB.

- To configure the next FXO Port, click the FXO Port number and follow the same instructions as given earlier.

Copy Port Parameters

- You can also copy the settings of a FXO Port to another FXO Port using the **Copy** button. To do this,

- Click the **Copy** button. The **Copy FXO Port Parameters** window opens.



- In the **Copy FXO Port Parameters from FXO Port** box, select the number of the port you want to copy settings *From*. Select the check box of the respective port numbers you want to copy the settings *To*.
- If you want to copy the settings *To* all the ports, select the **All** check box.
- Click the **OK** button.

- Once you have copied the settings, you can again edit the specific parameters of the FXO Port you copied the settings to.

BRI Port - Network

SARVAM UMG supports BRI Ports to which terminal equipments — ISDN Phone, Video Conferencing Unit or a Dialogic Card— can be connected. You must configure the **Orientation** of the BRI Port as **Network**.

- Click the **Basic Settings** link to expand.
- Click the **BRI Port** link.

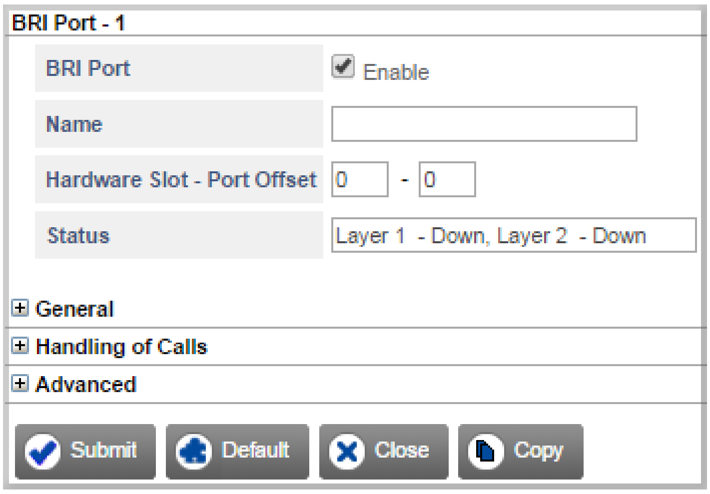
The BRI Port page displays the following parameters:

- **Port:** It displays the BRI Port numbers. Click on the desired BRI Port number to configure the Port Parameters.
- **Hardware Slot-Port:** The SARVAM UMG can automatically detect and assign the Hardware Slot and Port numbers to the BRI software ports. However, if required you may change the Hardware Slot and Port assigned to the BRI software port. In this case, enter the desired Hardware Slot and Port number.
- **Enable:** Keep the **BRI Ports** enabled. Clear the BRI Port **Enable** check box, only if you do not want to use the respective port. Default: Enabled.
- **Name:** Assign a Name to the BRI Port for identification. The Name can be a maximum of 24 characters.
- **Status:** This displays the status of Layer 1 and Layer 2, that is, Up or Down.
- **Interface Type:** It displays the type of interface you select — Point to Point or Point to Multipoint.
- **Orientation:** It displays the type of orientation you select — Network or Terminal.
- **Call Routing:** It displays the Call Routing Method you select.

To configure the **BRI Port** as **Network**,

- Click **BRI-1**.

The **BRI Port-1** window opens.



- Keep the **BRI Port** check box enabled.

  Clear the **BRI Port Enable** check box only when you do not want to use this BRI Port. Default: Enabled.

- You can assign a **Name** to the BRI Port, which will be displayed to the called party, if the called party's telephone instrument supports name display. Default: Blank

  The name you assign may consist of a maximum of 24 characters.Default: Blank.

- SARVAM UMG will assign the **Hardware Slot - Port Offset** automatically, when any card is inserted in the system.

  Hardware slot is the number of the universal slot of SARVAM UMG in which the BRI Card is inserted. Range of slot number is 1-12. Port is the number of BRI hardware port on the card to which the BRI line is connected.

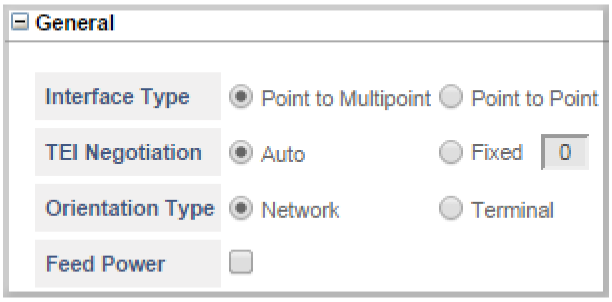
  However, if required, you may change the Hardware Slot and Port assigned to the BRI software port. In this case, enter the desired Hardware Slot and Port number.

  If you want to de-assign the Hardware Slot and Port, enter '00' in both fields. By default, Hardware Slot-Port is 00–00.

- **Status** displays the status of the BRI Port.

General Parameters

- Click **General** to expand.



- Select the **Interface Type** as **Point-to-Point** or **Point-to-Multipoint** according to your installation. Default: Point-to-Multipoint.

- Select **TEI Negotiation** for the BRI Port as **Auto** or **Fixed**. Default: Auto.

  TEI (Terminal Endpoint Identifier) Negotiation mode is used to integrate the SARVAM UMG with a specific ISDN-System. The TEI value programmed for BRI-NT Port should match the TEI value programmed in the Terminal Equipment connected to it.

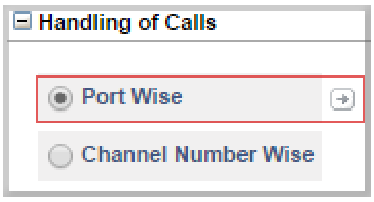
  - If you set TEI Negotiation to **Fixed**, enter the value of the Fixed TEI Negotiation from 00 to 63. Default: 00.

- Keep the **Orientation Type** of the BRI Port as **Network**.

- Enable the **Feed Power** check box, if you want the system to feed power to the terminal equipments connected to the BRI Port. Default: Disabled.

Handling of Calls

- Click **Handling of Calls** to expand.



- Select the method to route calls from the BRI Port. SARVAM UMG provides two options for **Handling of Calls**: Port Wise and Channel Number Wise.

  - **Port Wise**: Select this method to apply the call routing method for the entire port, that is, for all the channels and all called party numbers received.

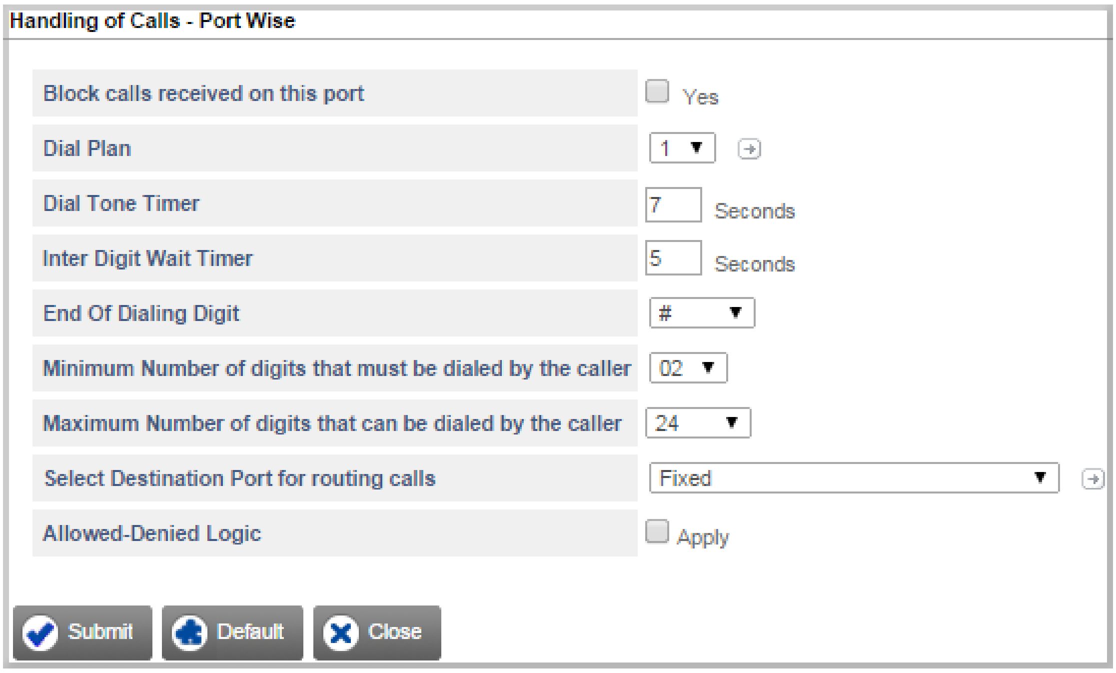
  - **Channel Number Wise**: Select this method to apply a different call routing method for each channel.

- For the call routing option you select, click **Settings** and configure the parameters.

Port Wise

To configure Handling of Calls Port wise,

- Select the **Port Wise** check box.
- Click **Settings** .
- The **Handling of Calls - Port wise** window opens.



- Keep the **Block calls received on this port** check box disabled.

  Select this check box only if you do not want to route calls through this port.

- SARVAM UMG supports 8 Dial Plans with total 64 entries in each table. When a user dials a number, it is compared with the Destination Number configured in the Dial Plan. If a match is found, the system routes the call immediately without waiting for End of Dialing and if a match is not found, the system will wait for the End of Dialing and then route the call as per the Destination Port Selection method configured.

  Select the **Dial Plan** table number you configured for this port. If you have not configured the Dial Plan table you may do so now,

  - Click **Settings** . The Dial Plan Table opens.
  - Configure the numbers in the table. For detailed instructions, see "Dial Plan".

- Set the duration of the **Dial Tone Timer** as per your requirement. This is the time for which SARVAM UMG will play Dial Tone to the caller. On expiry of this timer, the system plays error tone to the caller. Valid range is 01 to 99 seconds. Default: 7 seconds.

- Set the duration of the **Inter Digit Wait Timer**. This is the duration for which you want the system to wait while receiving the digits dialed by the caller to consider it as End of Dialing. You may change this timer, if required. Valid range is 01 to 99 seconds. Default: 05 seconds.

- As **End of Dialing Digit**, select whether the system should consider **#** or ***** as termination digit to detect end of dialing. Default: #

- In **Minimum number of digits that can be dialed by the caller**, select the minimum number of digits to be dialed by the user for the system to consider it as a valid number. Valid range is 01 to 24 digits. Default: 2 digits.

- In **Maximum number of digits that can be dialed by the caller**, define the maximum number of digits to be dialed by the user for the system to consider it as End of Dialing. Valid range is 01 to 24 digits. Default: 24 digits.

Destination Port Determination

For the port/channel number, select the Destination Port for routing calls from the following options:

- Fixed
- On the basis of Destination Number
- On the basis of Calling Party Number

Default: Fixed.

Read the description and follow the instructions for each of these destination port selection methods given below.

If the destination number to be dialed out is an IP Address, SARVAM UMG will not check the Destination Port Determination Method. Instead, it will route the call using the SIP Trunk / Group programmed for IP Dialing. (To know more, see the feature description “IP Dialing”).

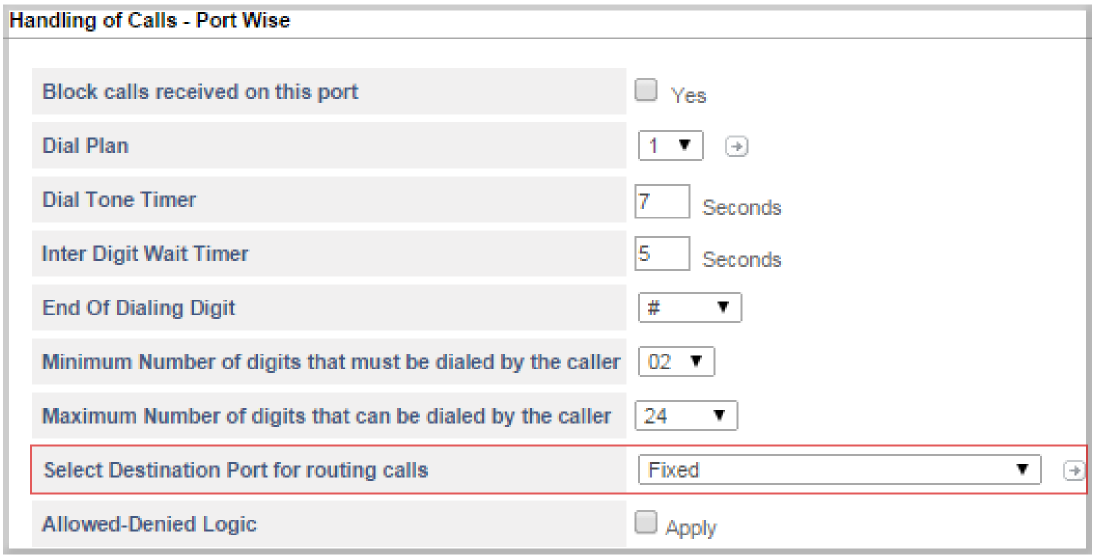
Fixed

In this method, calls received on the BRI Port are routed to a Fixed Destination Port, irrespective of the number dialed on the BRI Port.

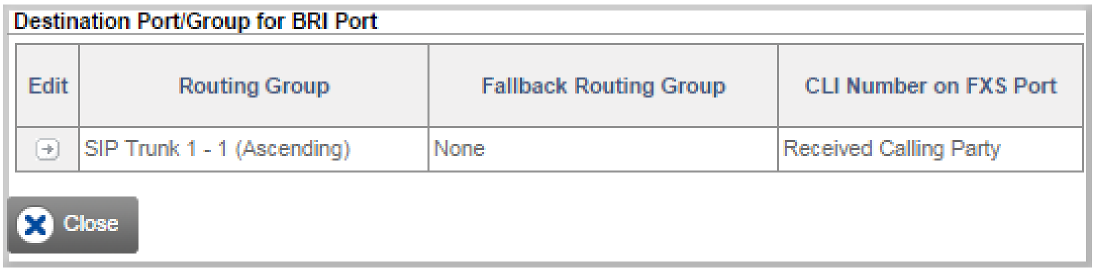
To apply this method, do the following:

- In **Select Destination Port for routing calls**, select **Fixed** option.

- Click **Settings** .



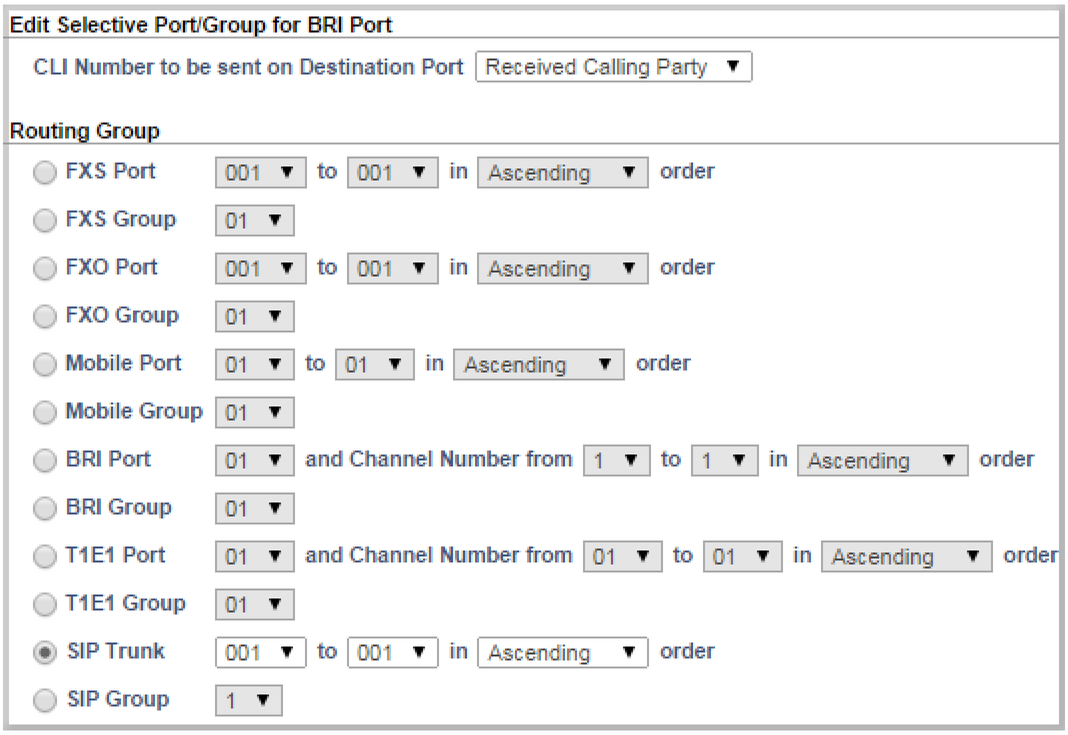
The **Destination Port/Group for BRI Port** window opens.



The default **Routing Group** and **Fallback Routing Groups** appear.

- If you wish to change the default Routing Group options, click **Edit** .

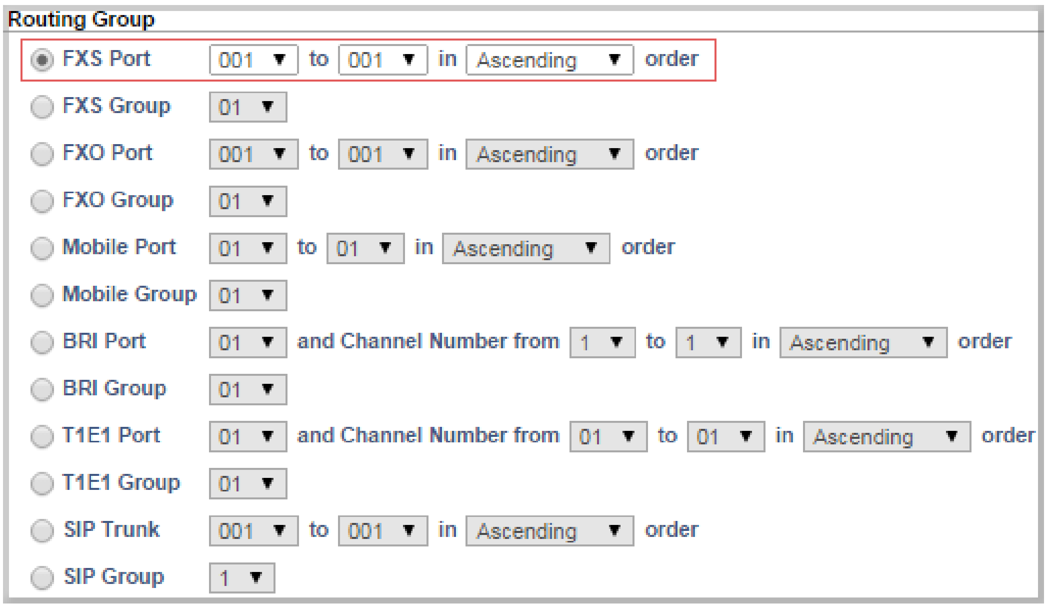
The **Edit Selective Port/Group for BRI Port** window opens.



- Select the **CLI Number to be sent on Destination Port**. You can select Received Calling Party or Received Called Party. Default: Received Calling Party.

CLI to be sent on Destination Port is applicable when FXS Port/Group is determined as the Destination Port.

- Create the **Routing Group**.
    - To create a group of *sequential* **FXS Ports** as members,



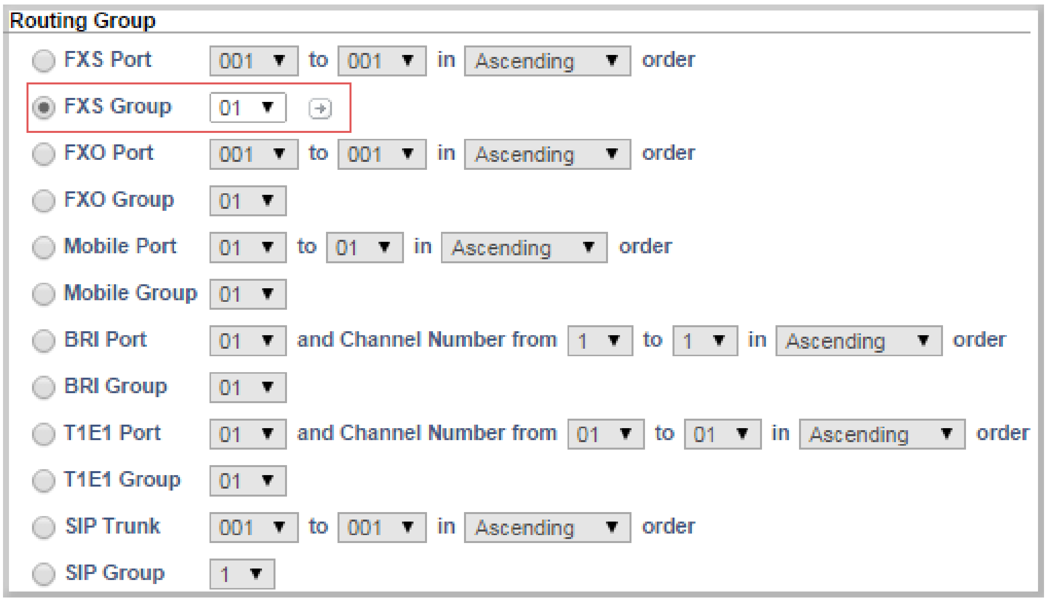
- Select the desired **FXS Port** numbers as members. Default:1.

- In **in - order**, select the order in which the system should hunt for a free member FXS Port to route the call.

  Select **Ascending** to start hunting from the first to the last member FXS Port. Select **Descending** to start hunting from the last to the first member FXS Port. Default: Ascending.

- To create a group of *not-sequential* **FXS Ports** as members,

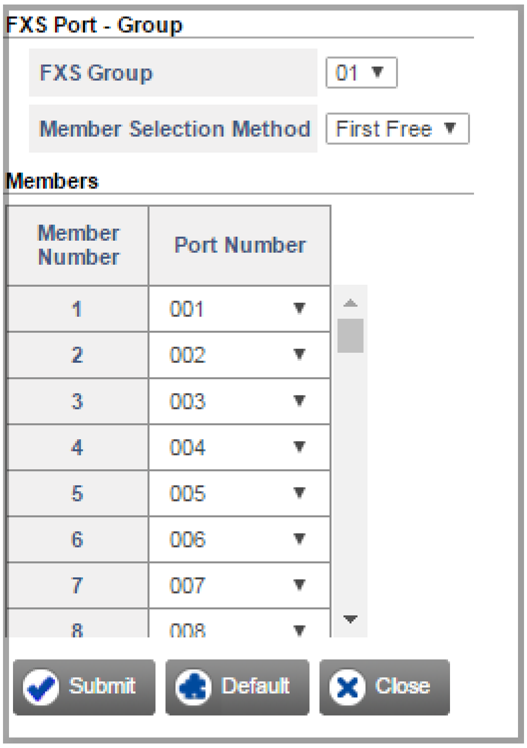
  - Select a **FXS Group**.



  - Select **FXS Group** number. Default:1.

  - Click **Settings** .

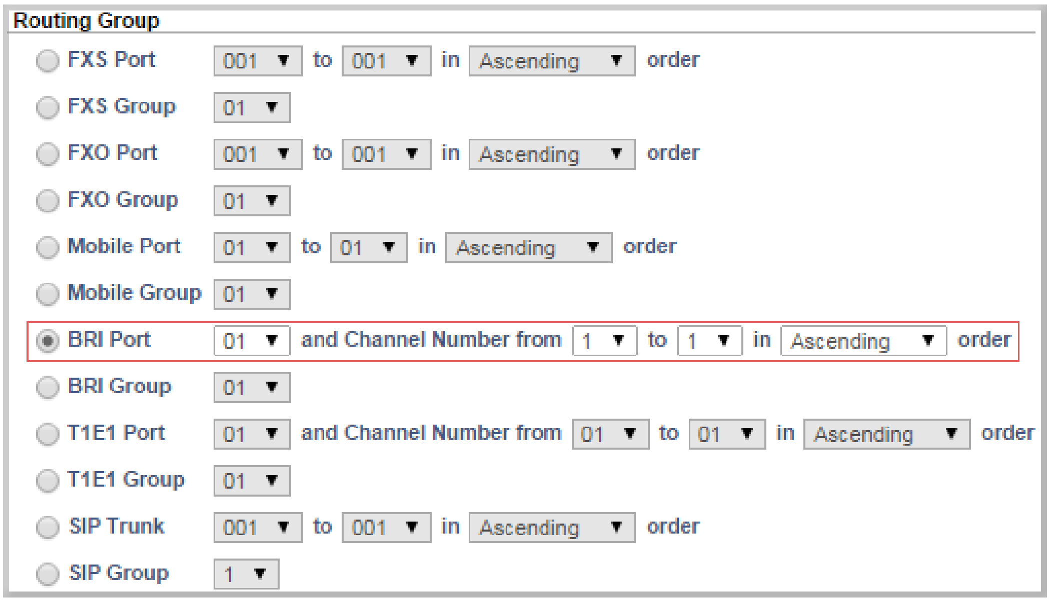
- The **FXS Port - Groups** window opens.



- Create the FXS Group. For detailed instructions on creating groups, see the topic “Group” under *Advanced Settings*.

- Similarly, you can create a group of *sequential* and *not-sequential* FXO Ports, Mobile Ports and SIP Trunks.

- To create a routing group of *sequential* **BRI Channels** as members,



- Select the **BRI Port** Number. Default: 1.

- In Channel Number **From - to**, select the **Start Channel Number** and the **End Channel Number**, respectively.

- In **in - order**, select the order in which the system should hunt for a free member Channel to route the call.
  Select **Ascending** to start hunting from the first to the last member channel. Select **Descending** to start hunting from the last to the first member channel. Default: Ascending.

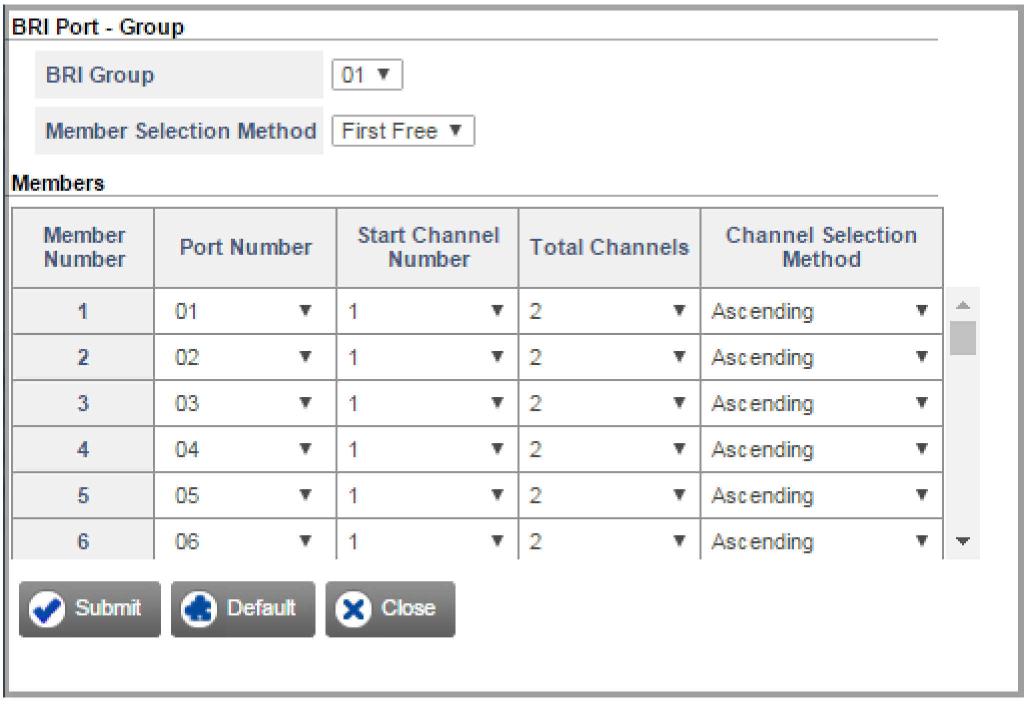
- To create a group of *not-sequential* **BRI Channels** as members,

  - Select **BRI Group**.

  - Select a **BRI Group** number. Default:1.

  - Click **Settings** .

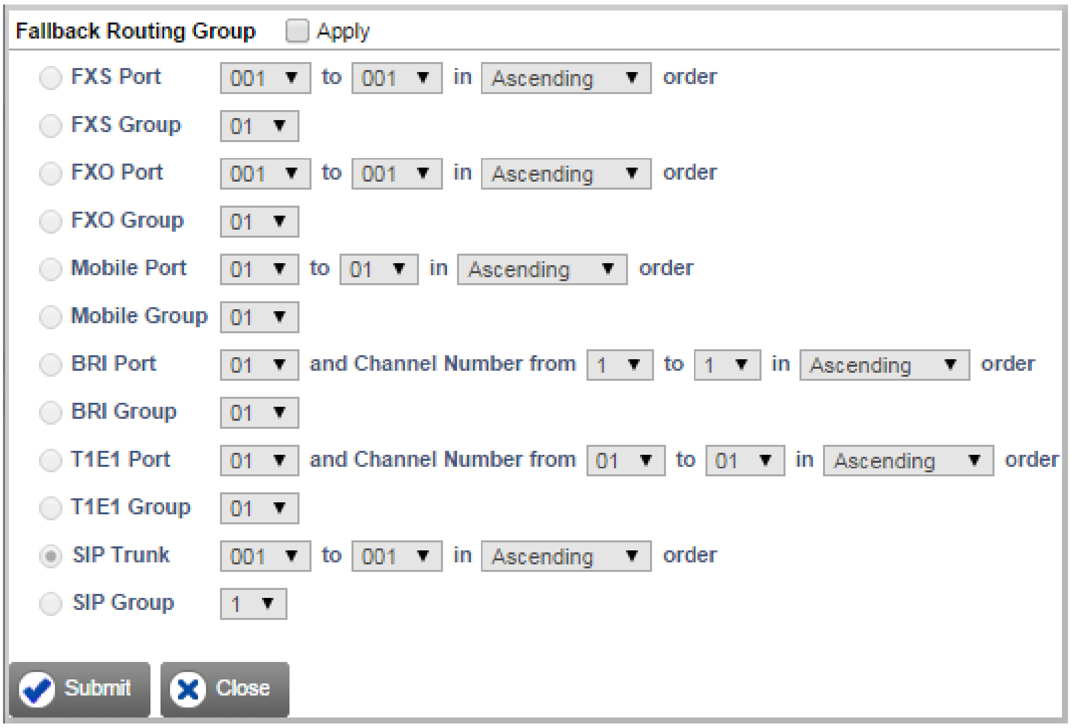
- The **BRI Port - Groups** window opens.



- Create the BRI Group. For detailed instructions on creating groups, see the topic "Group" under *Advanced Settings.*

- Similarly, you can create a group of *sequential* and *not-sequential* T1E1 Ports.

- You may create the **Fallback Routing Group**.



- To do this,
  - Select the **Apply** check box.
  - Follow the same instructions provided earlier for creating *sequential* and *not-sequential* groups of FXS Ports, FXO Ports, Mobile Ports, BRI Ports, T1E1 Ports and SIP Trunks.
- Click **Submit** to save changes. The **Edit** window closes.
- The entry you edited appears in the **Destination Port/Group for BRI Port** window.
- Close the **Destination Port/Group for BRI Port** window to return to the Handling of Calls window.

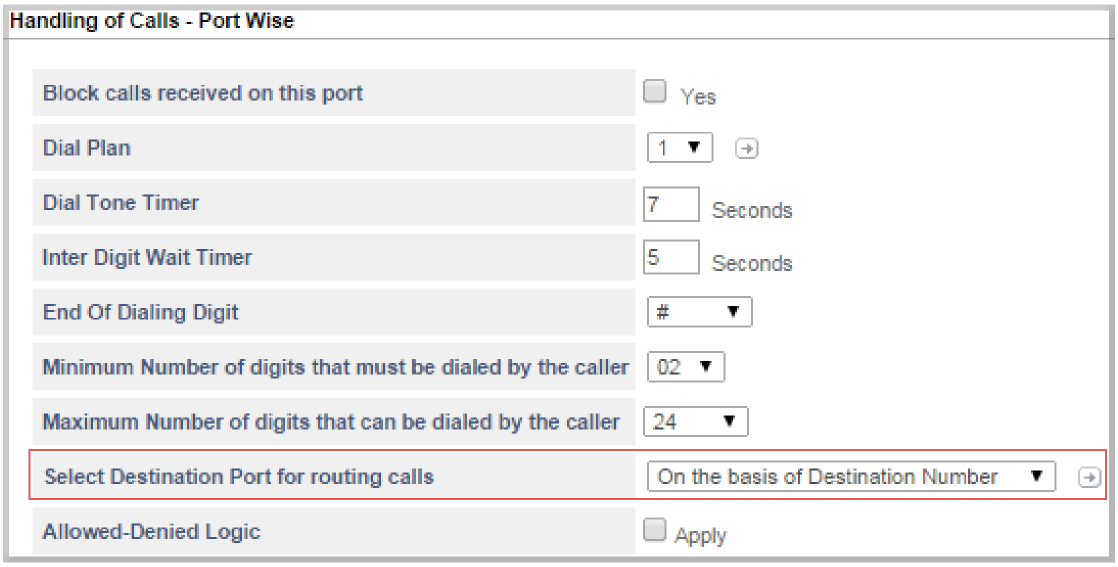
On the basis of Destination Number

In this method, incoming calls on the source port are routed to the destination port on the basis of the destination number (called party number) dialed by the caller.

You must configure the called party numbers in the **Destination Number Based** Table. SARVAM UMG will match the called party number dialed by the caller with the entries of this table. If a match is found for the number in the table, the call is routed to the destination.

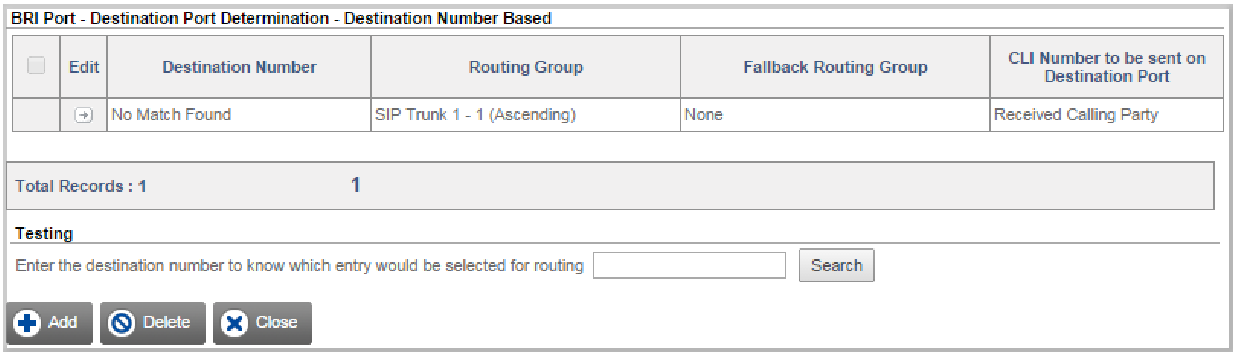
To apply this method, do the following:

- In **Select Destination Port for routing calls**, select **On the basis of Destination Number** option.

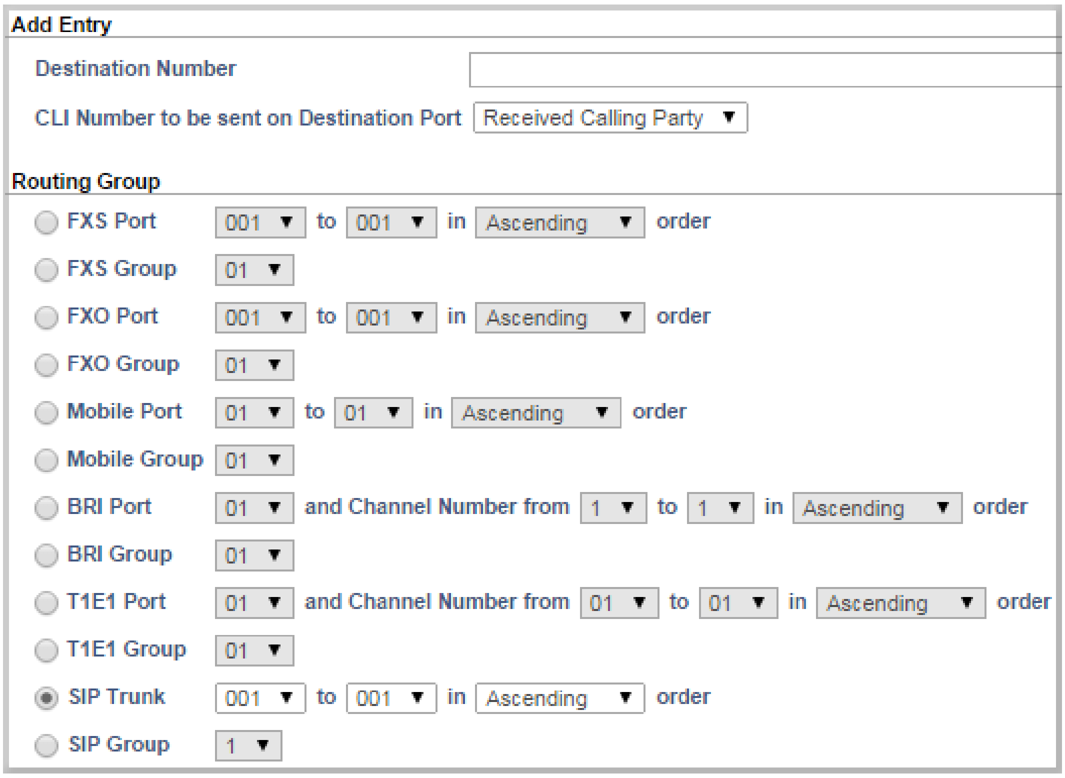


- Click **Settings** .

The **BRI Port - Destination Port Determination - Destination Number Based** table window opens.



- To add a new entry, click **Add**. The **Add Entry** window opens. You can add upto 1000 entries.



- In **Destination Number**, enter the number you expect the callers to dial. You may enter upto 64 characters (Digits + "Wildcard Characters") in this field. Valid characters: 0 to 9, *, #, X, T, Comma [,], Hyphen [-], Caret [^]. Default: Blank.

Wildcard Characters

SARVAM UMG 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.

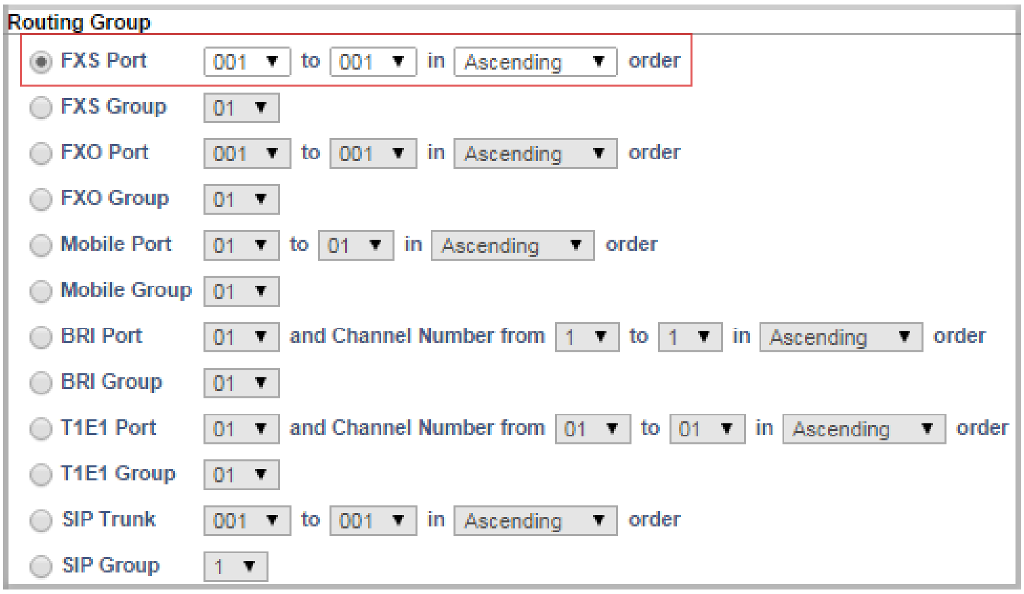
+	+ (plus) can be configured as a first character of the Destination Number string in the *SIP Trunk-Destination Port Determination-Destination Number Based* table only.
[-]	Hyphen within the bracket, defines a range. Only digits 0-9 are allowed within a bracket.
[,]	Comma within a bracket is used as 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.

- Select the **CLI Number to be sent on Destination Port**. You can select Received Calling Party or Received Called Party. Default: Received Calling Party.

CLI to be sent on Destination Port is applicable when FXS Port/Group is determined as the Destination Port.

- Create the **Routing Group**.

  - To create a group of *sequential* **FXS Ports** as members,



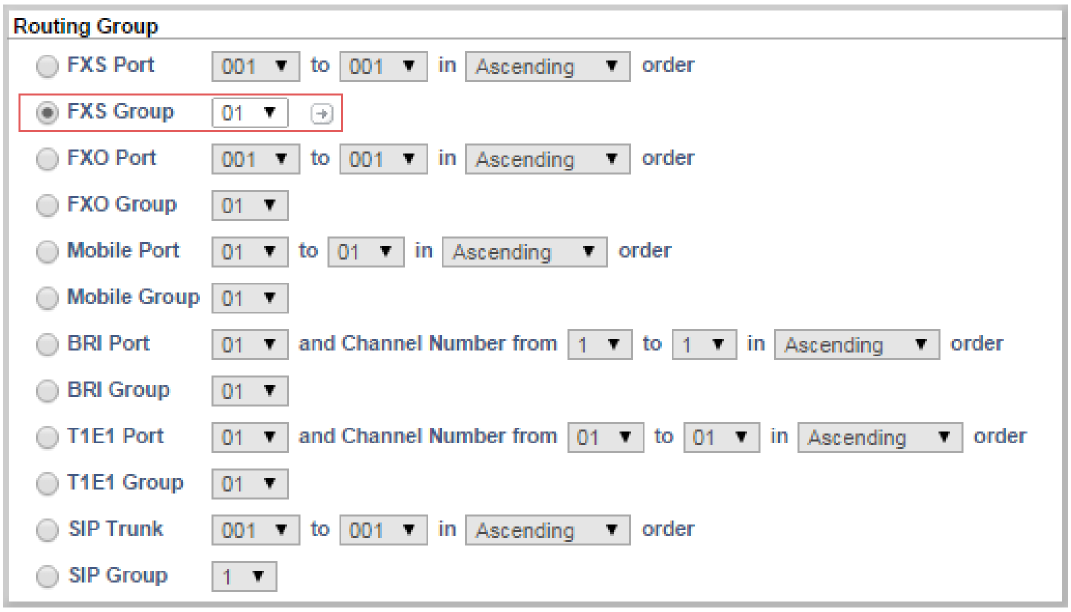
    - Select the desired **FXS Port** numbers as members. Default:1.

    - In **in - order**, select the order in which the system should hunt for a free member FXS Port to route the call.

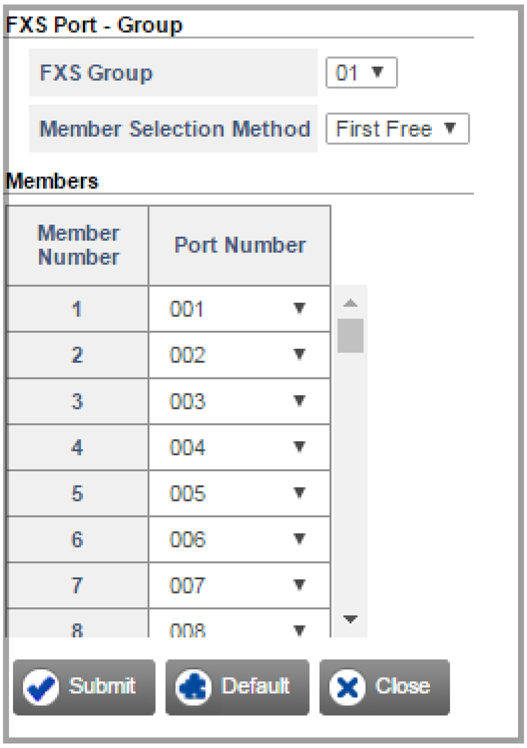
      Select **Ascending** to start hunting from the first to the last member FXS Port. Select **Descending** to start hunting from the last to the first member FXS Port. Default: Ascending.

  - To create a group of *not-sequential* **FXS Ports** as members,

- Select a **FXS Group**.



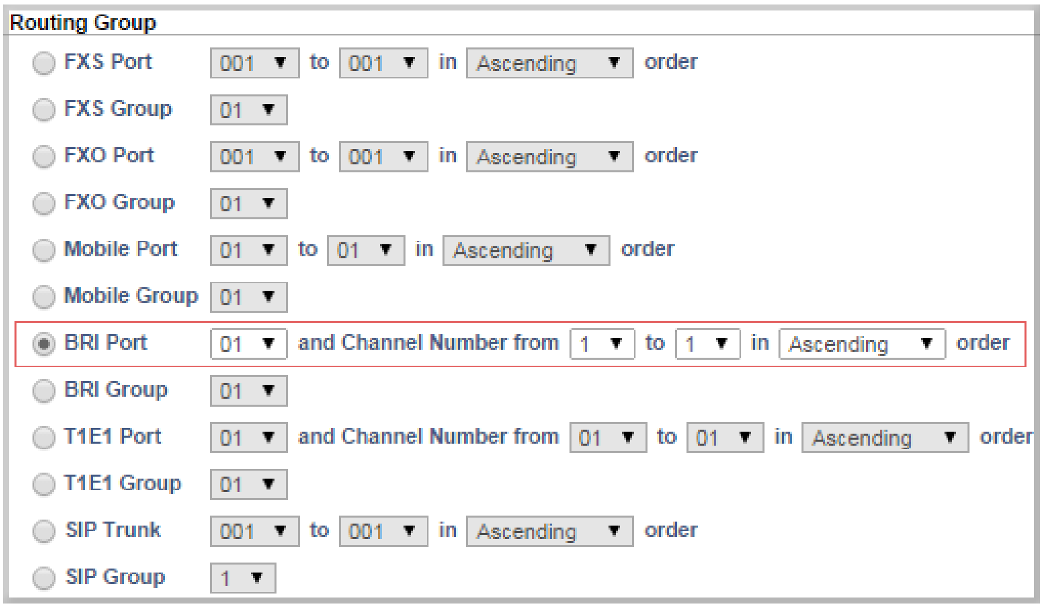
- Select **FXS Group** number. Default:1.
- Click **Settings** .
- The **FXS Port - Groups** window opens.



- Create the FXS Group. For detailed instructions on creating groups, see the topic “Group” under *Advanced Settings*.

- Similarly, you can create a group of *sequential* and *not-sequential* FXO Ports, Mobile Ports and SIP Trunks.

- To create a routing group of *sequential* **BRI Channels** as members,



  - Select the **BRI Port** Number. Default: 1.

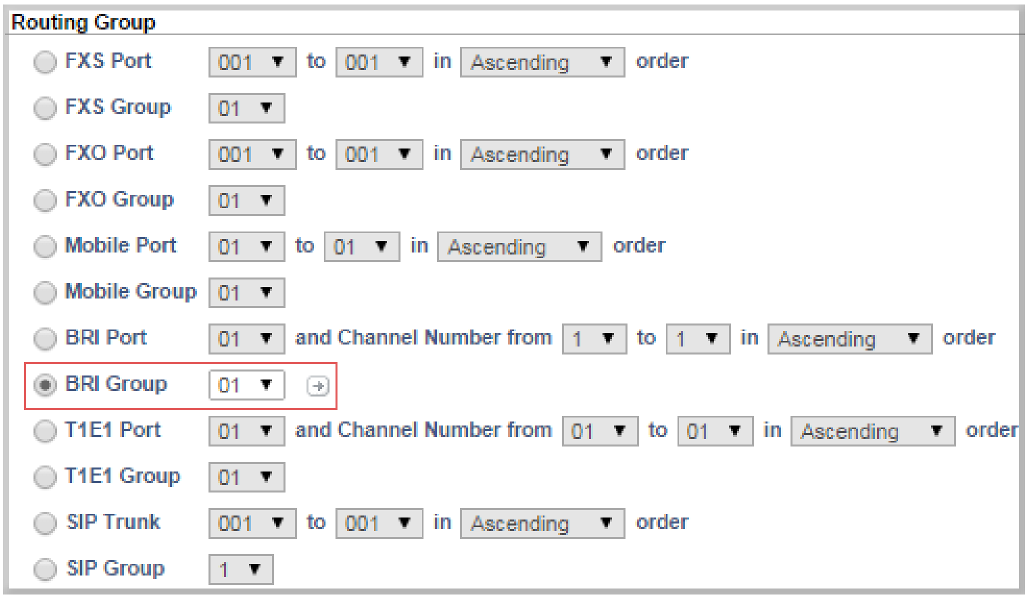
  - In Channel Number **From - to**, select the **Start Channel Number** and the **End Channel Number**, respectively.

  - In **in - order**, select the order in which the system should hunt for a free member Channel to route the call.

    Select **Ascending** to start hunting from the first to the last member channel. Select **Descending** to start hunting from the last to the first member channel. Default: Ascending.

- To create a group of *not-sequential* **BRI Channels** as members,

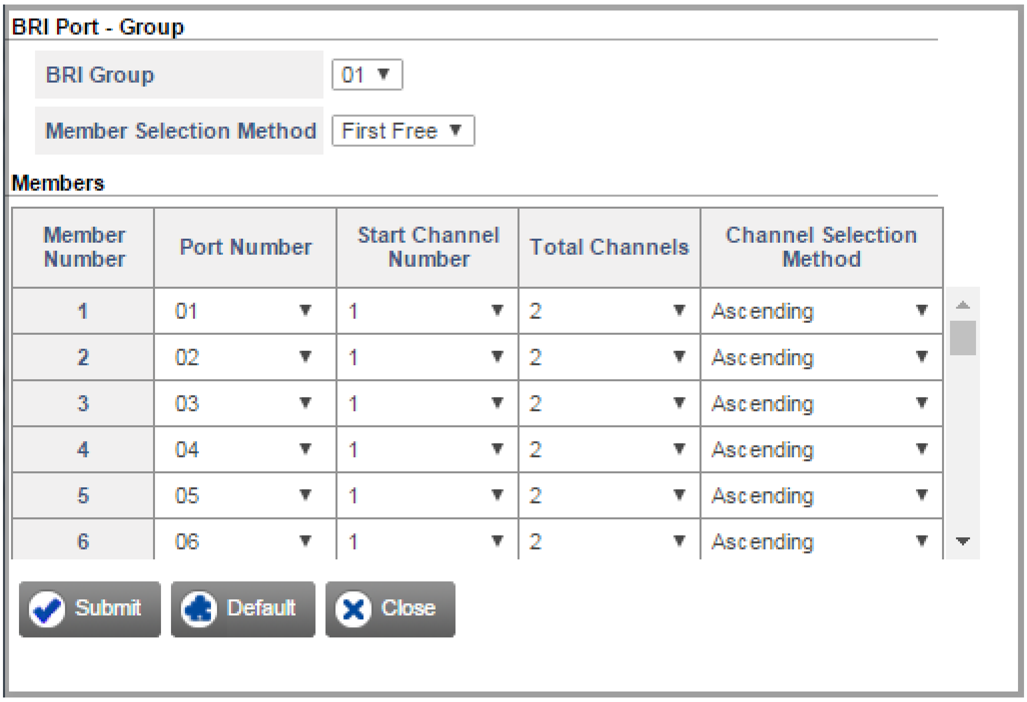
- Select **BRI Group**.



- Select a **BRI Group** number. Default:1.

- Click **Settings** .

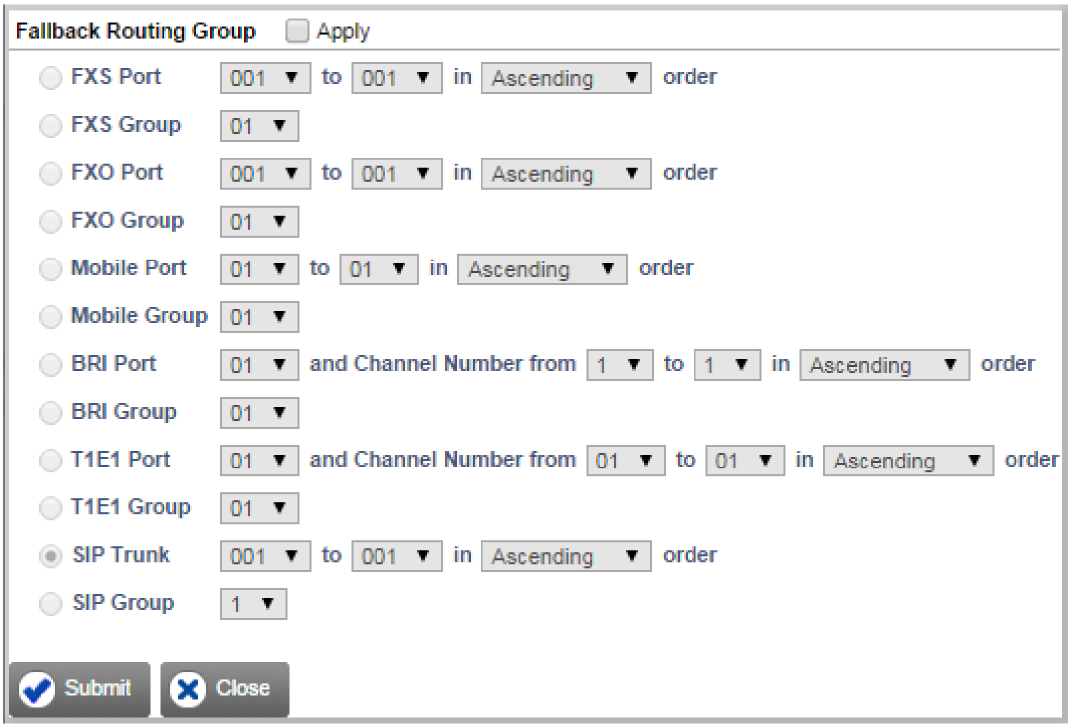
- The **BRI Port - Groups** window opens.



- Create the BRI Group. For detailed instructions on creating groups, see the topic "Group" under *Advanced Settings.*

- Similarly, you can create a group of *sequential* and *not-sequential* T1E1 Ports.

- You may create the **Fallback Routing Group**.



- To do this,
    - Select the **Apply** check box.
    - Follow the same instructions provided earlier for creating *sequential* and *not-sequential* groups of FXS Ports, FXO Ports, Mobile Ports, BRI Ports, T1E1 Ports and SIP Trunks.
- Click **Submit** to save changes. The **Add Entry** window closes.
- The entry you added appears in the **BRI Port - Destination Port Determination - Destination Number Based** table.
- Follow the same steps as above to add another entry to this table.
- To delete an entry, select the check box and click the **Delete** button.

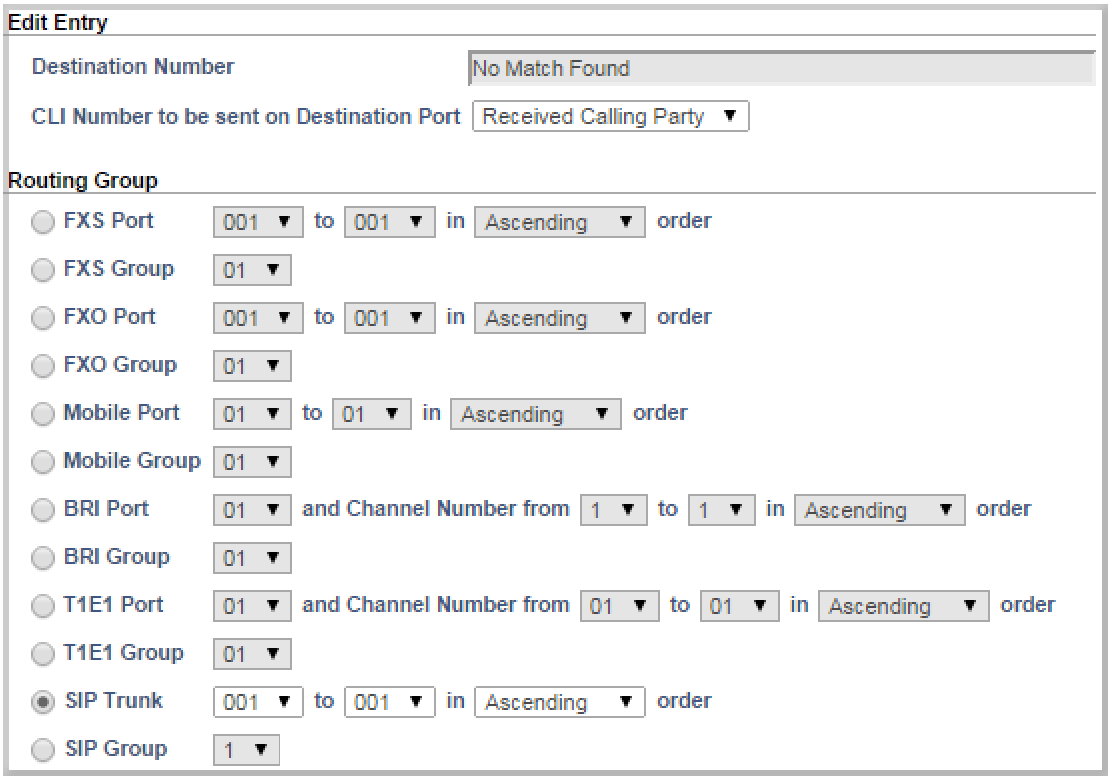
If there are multiple entries in the Destination Number Based table, to search a particular entry in the table, under Testing enter the desired number in the ***Enter the destination number to know which entry would be selected for routing*** *search box.*

- By default, SIP Trunk 1-1 (Ascending) is assigned as the Routing Group, for routing calls from numbers that do not match with any of the destination numbers you configured (No Match Found).

  To change the default Routing Group and to create the Fallback Routing Group for the No Match Found numbers entry,

    - For the No Match Found entry in the table, click **Edit** .

- The **Edit Entry** window opens.



- Create the **Routing Group** and **Fallback Routing Group** as per your requirement.

- Click **Submit** and close the window.

- Close the window if you have finished adding/editing entries.

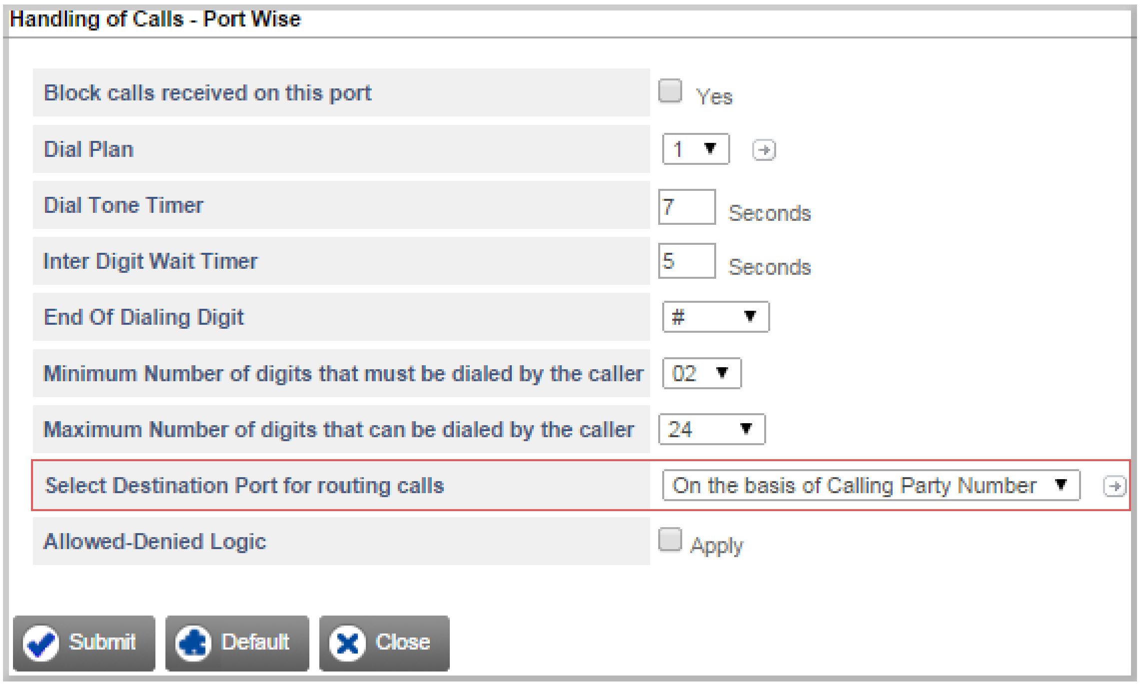
  You can also configure the **Destination Number Based** Table from *Advanced Settings*. For instructions, see “Destination Port Determination” under *Advanced Settings.*

On the basis of Calling Party Number

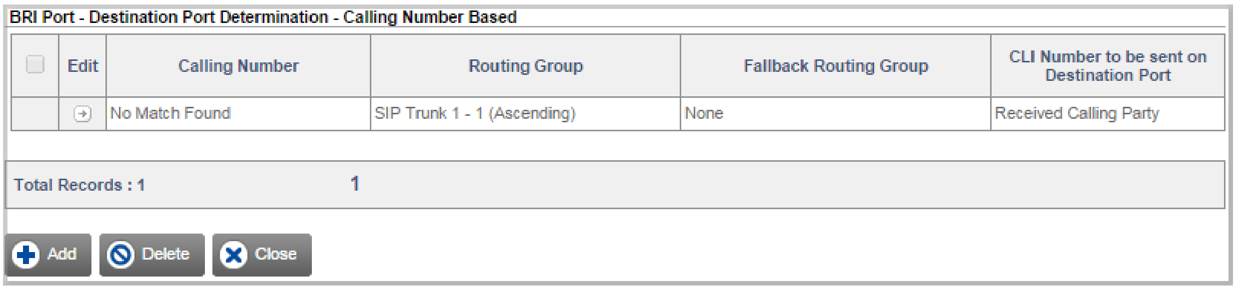
In this method, incoming calls on the BRI Port are routed to a specific port as per the calling party's number.
To apply this method, do the following:

- In **Select Destination Port for routing calls**, select **On the basis of Calling Party Number** option.

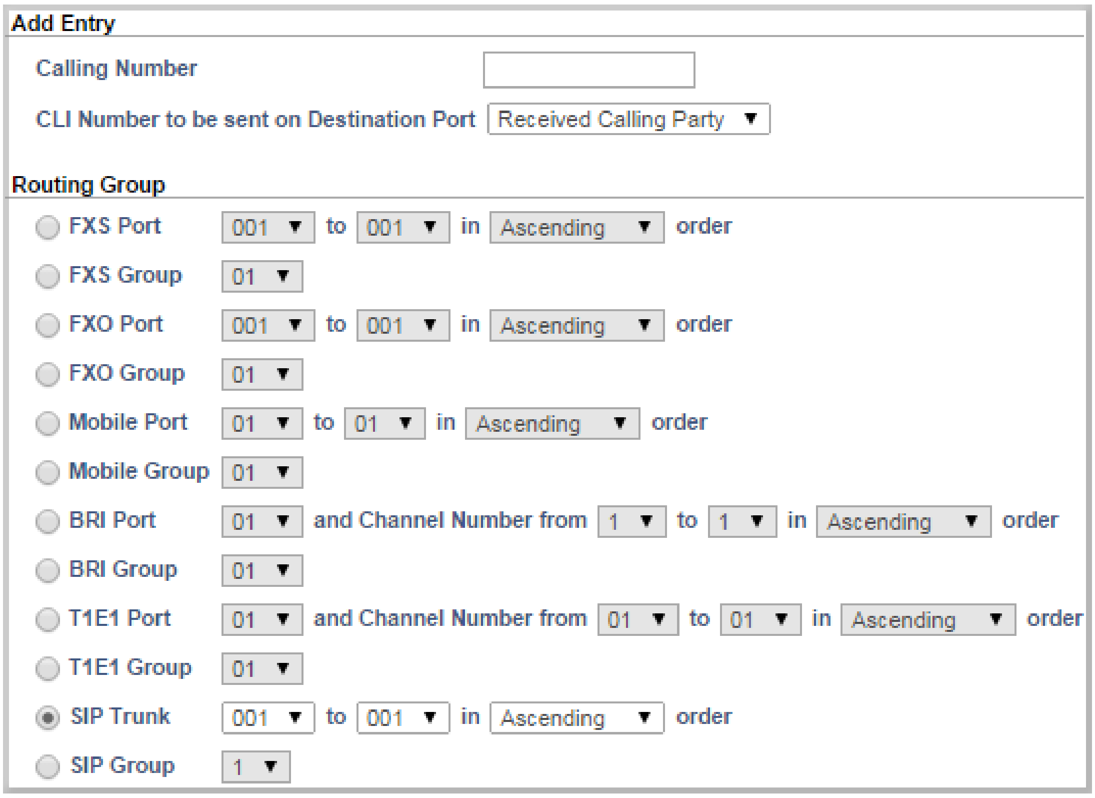
- Click **Settings** .



The **BRI Port - Destination Port Determination - Calling Number Based** table window opens.



- To add a new entry, click **Add**. The **Add Entry** window opens. You can add upto 499 entries.



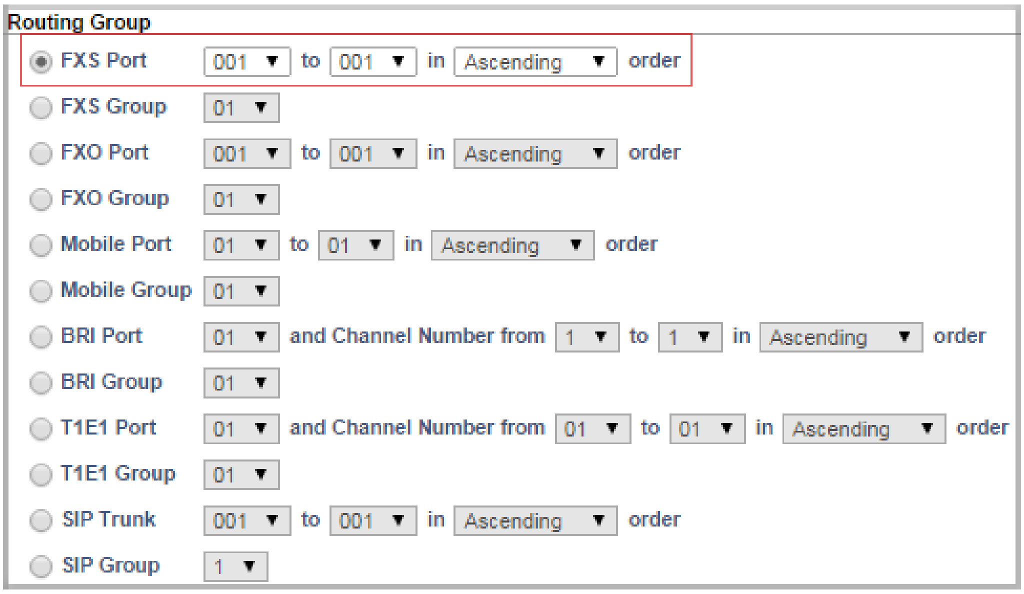
- In **Calling Number**, enter the number (max. 24 characters) from which you expect calls to be received. Valid digits are: 0 to 9, *, #, (dot). Default: Blank.

- Select the **CLI Number to be sent on Destination Port**. You can select Received Calling Party or Received Called Party. Default: Received Calling Party.

CLI to be sent on Destination Port is applicable when FXS Port/Group is determined as the Destination Port.

- Create the **Routing Group**.

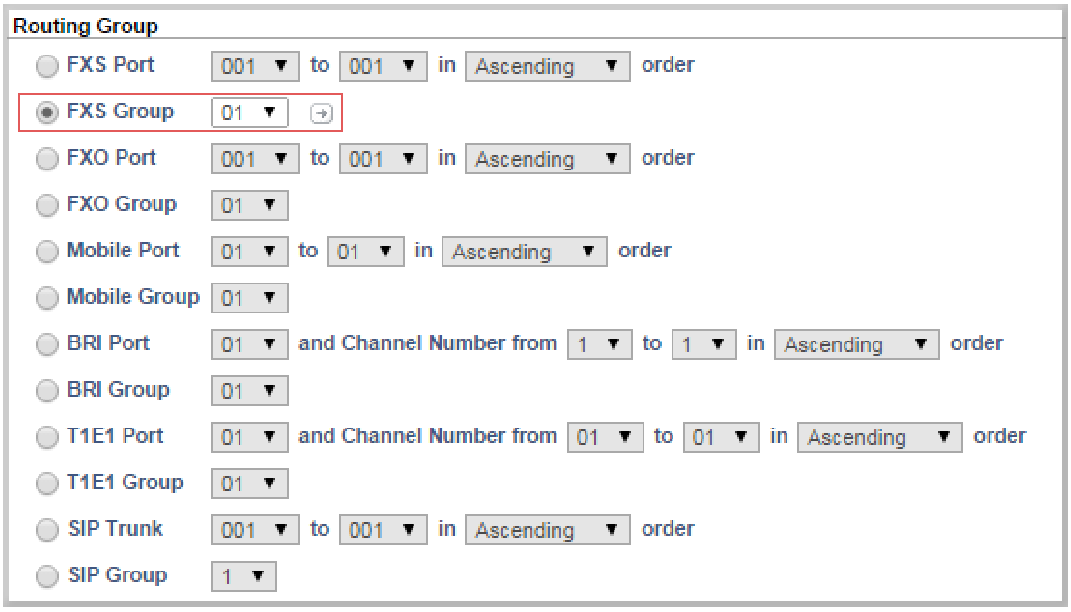
- To create a group of *sequential* **FXS Ports** as members,



  - Select the desired **FXS Port** numbers as members. Default:1.
  - In **in - order**, select the order in which the system should hunt for a free member FXS Port to route the call.

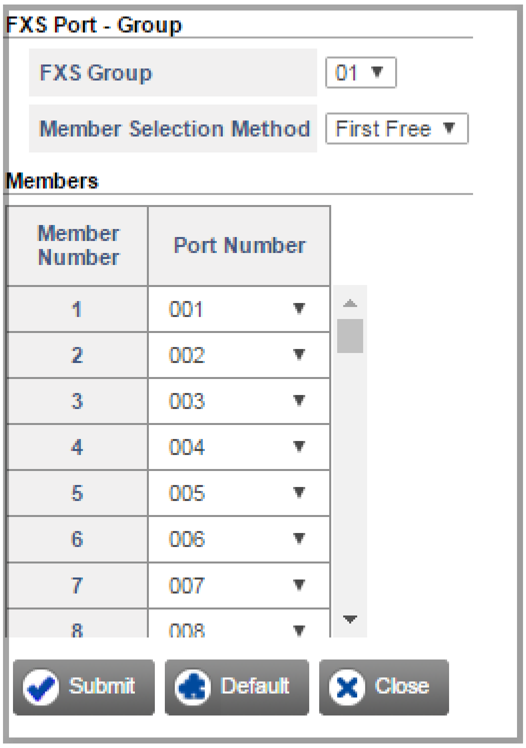
    Select **Ascending** to start hunting from the first to the last member FXS Port. Select **Descending** to start hunting from the last to the first member FXS Port. Default: Ascending.

- To create a group of *not-sequential* **FXS Ports** as members,
  - Select a **FXS Group**.



  - Select **FXS Group** number. Default: 1.

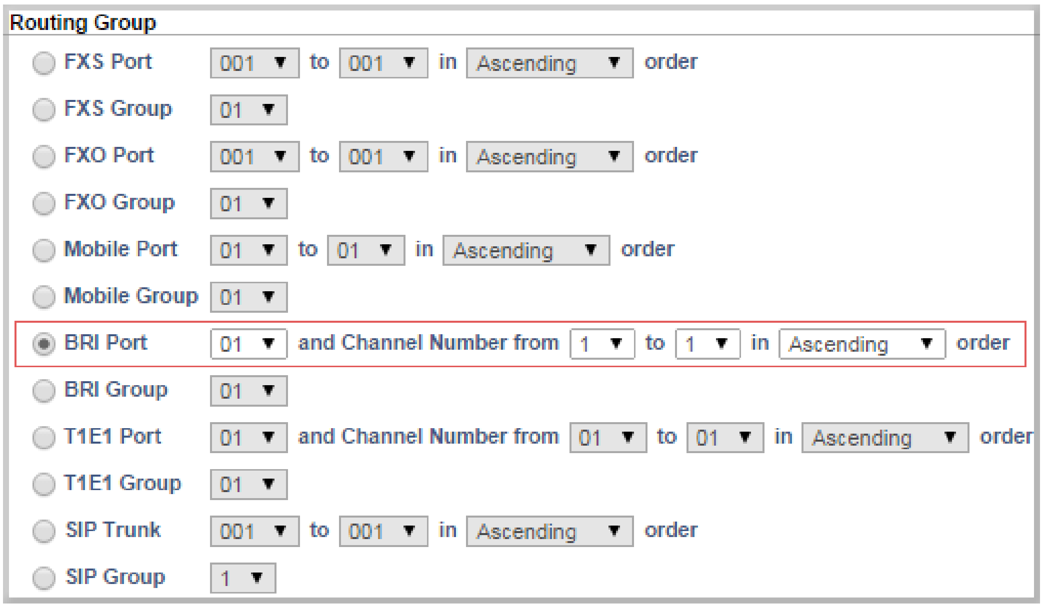
- Click **Settings** .
- The **FXS Port - Groups** window opens.



- Create the FXS Group. For detailed instructions on creating groups, see the topic “Group” under *Advanced Settings*.

- Similarly, you can create a group of *sequential* and *not-sequential* FXO Ports, Mobile Ports and SIP Trunks.

- To create a routing group of *sequential* **BRI Channels** as members,

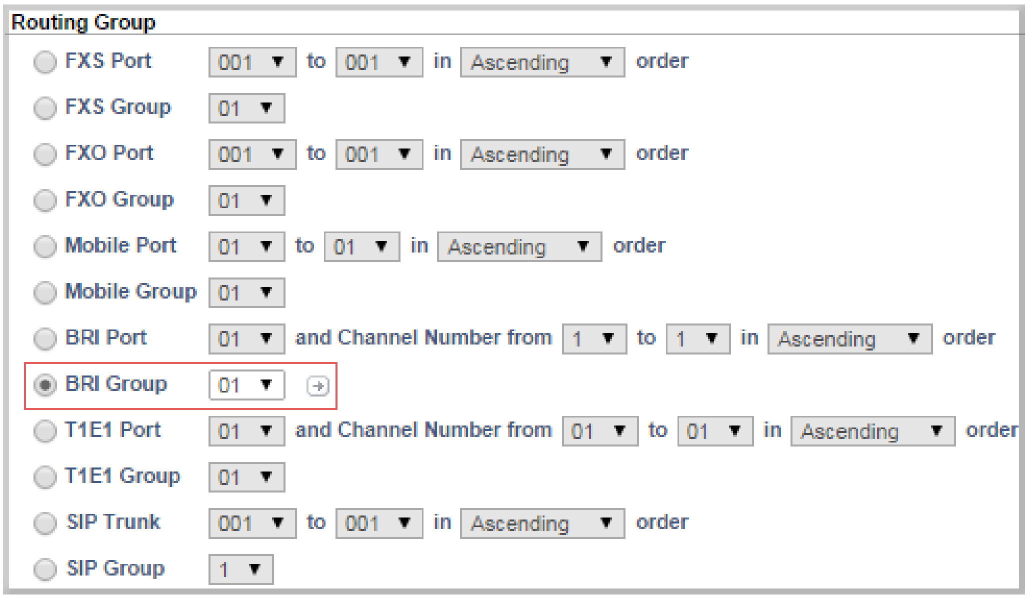


  - Select the **BRI Port** Number. Default: 1.
  - In Channel Number **From - to**, select the **Start Channel Number** and the **End Channel Number**, respectively.
  - In **in - order**, select the order in which the system should hunt for a free member Channel to route the call.

    Select **Ascending** to start hunting from the first to the last member channel. Select **Descending** to start hunting from the last to the first member channel. Default: Ascending.

- To create a group of *not-sequential* **BRI Channels** as members,

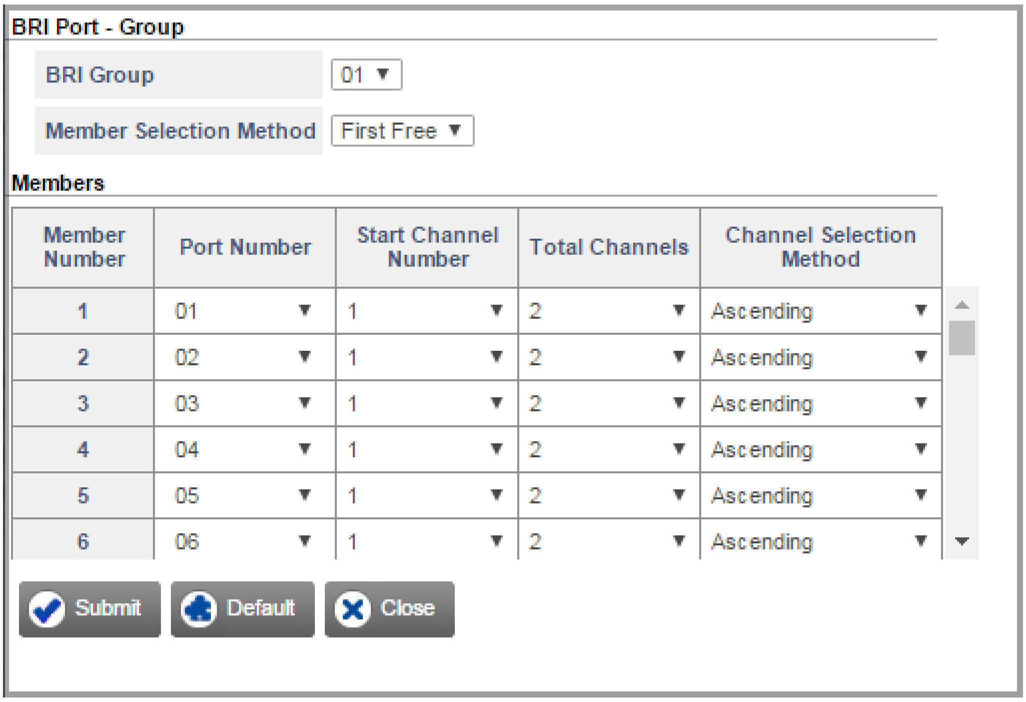
- Select **BRI Group**.



- Select a **BRI Group** number. Default:1.

- Click **Settings** .

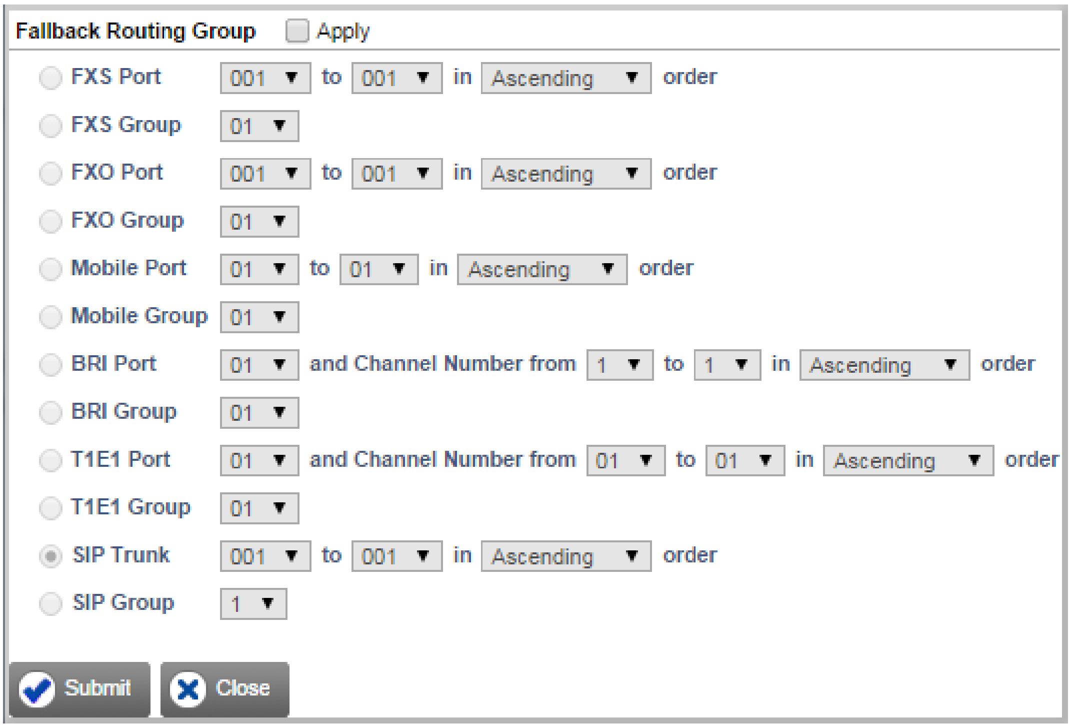
- The **BRI Port - Groups** window opens.



- Create the BRI Group. For detailed instructions on creating groups, see the topic “Group” under *Advanced Settings.*

- Similarly, you can create a group of *sequential* and *not-sequential* T1E1 Ports.

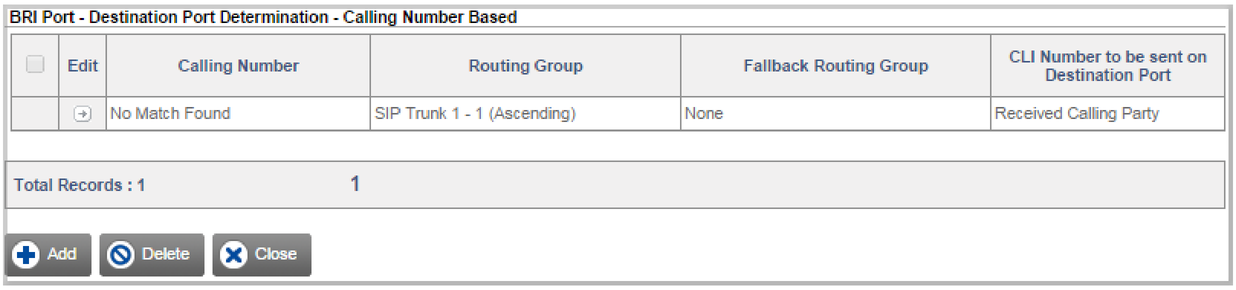
- You may create the **Fallback Routing Group**.



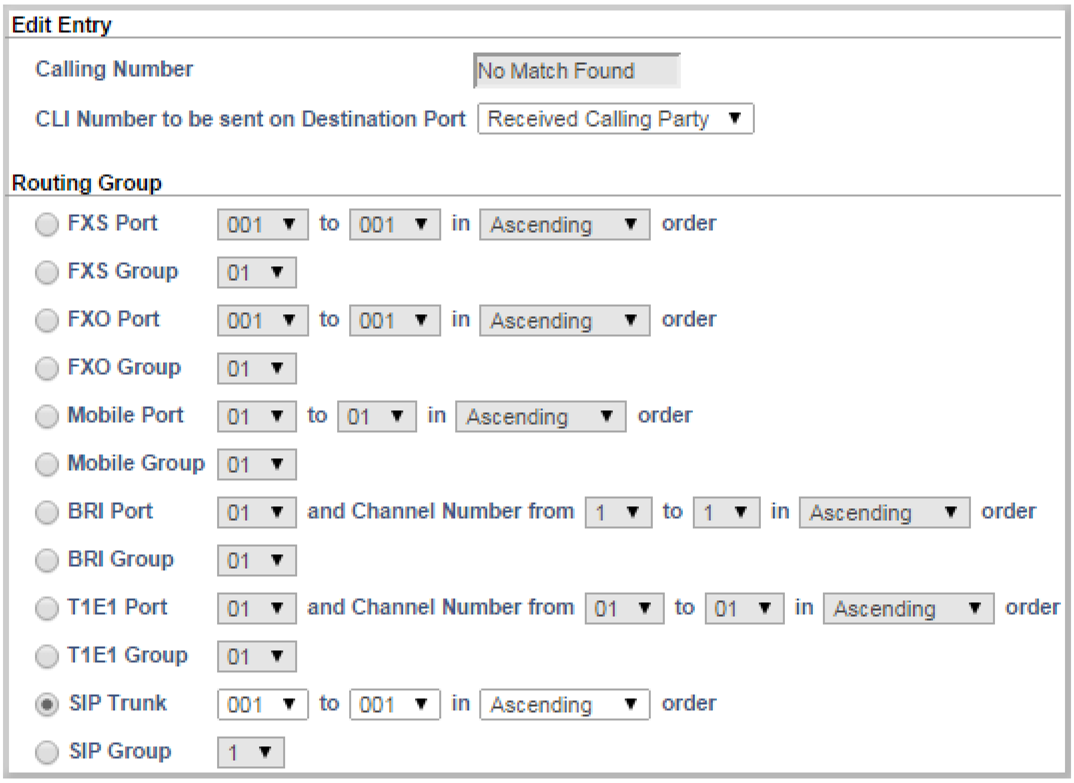
- To do this,
  - Select the **Apply** check box.
  - Follow the same instructions provided earlier for creating *sequential* and *not-sequential* groups of FXS Ports, FXO Ports, Mobile Ports, BRI Ports, T1E1 Ports and SIP Trunks.
- Click **Submit** to save changes. The **Add Entry** window closes.
- The entry you added appears in the **BRI Port - Destination Port Determination - Calling Number Based** table.
- Follow the same steps as above to add another entry to this table.
- To delete an entry, select the check box and click the **Delete** button.
- By default, SIP Trunk 1-1 (Ascending) is assigned as the Routing Group, for routing calls from numbers that do not match with any of the destination numbers you configured (No Match Found).

  To change the default Routing Group and to create the Fallback Routing Group for the No Match Found numbers entry,

- For the No Match Found entry in the table, click **Edit** .



- The **Edit Entry** window opens.



- Create the **Routing Group** and **Fallback Routing Group** as per your requirement.

- Click **Submit** and close the window.

- Close the window if you have finished adding/editing entries.

You can also configure the **Calling Number Based** Table from *Advanced Settings*. For instructions, see “Destination Port Determination” under *Advanced Settings.*

Allowed - Denied Logic

You can apply the Allowed-Denied logic on the BRI Port (source port) if you want to allow or restrict the dialing of particular numbers. You can use this feature for Toll Control.

The Allowed-Denied Number Logic makes use of two Number lists:

- **Allowed Numbers List:** This is the list of numbers that can be dialed out from the BRI Port.

- **Denied Numbers List:** This list contains the numbers that are to be restricted from being dialed out from the BRI Port.

When Allowed-Denied Logic is enabled on a source port, for each number dialed from the port, SARVAM UMG uses the best-match-found logic to compare the dialed number with the Allowed Number list and the Denied Number list.

The number is allowed to be dialed, if it:
- matches with both lists.
- matches with Allowed Number list, but not with the Denied Number list.
- matches with neither the Allowed List nor the Denied List.

The number is denied, if it matches with the Denied Number list, but not with the Allowed Number list.

The system does not apply the Allowed-Denied Logic:
- When dialed number string matches with any Access Code.
- When dialed number string matches with any Emergency Number.

To apply Allowed - Denied Logic on the BRI Port,

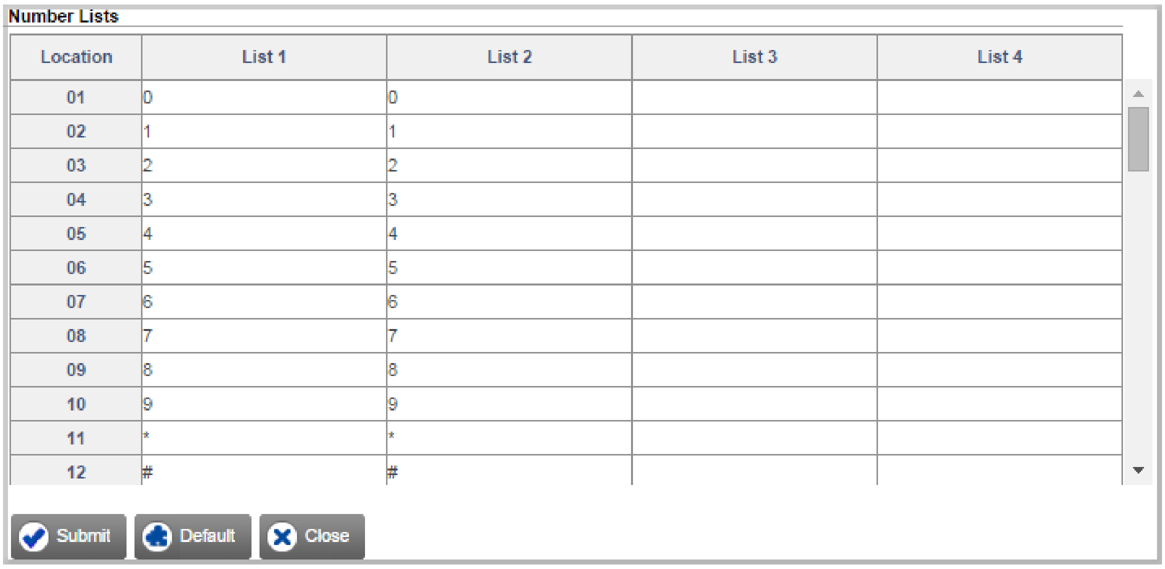
- Select the **Allowed - Denied Logic** check box.

  - In the **Allowed Number List**, select the list number you have configured with numbers you want to allow to be dialed out from the BRI Port. Default: 01

    If you have not configured the Allowed Number List,

    - Click **Settings**.

- The Number Lists window opens.



Location	List 1	List 2	List 3	List 4
01	0	0		
02	1	1		
03	2	2		
04	3	3		
05	4	4		
06	5	5		
07	6	6		
08	7	7		
09	8	8		
10	9	9		
11	*	*		
12	#	#		

- You may configure the default Allowed Number List 1 or any other list. See “Number Lists” to configure the allowed numbers.
- Click **Submit** to save the Allowed Number List and close the window.

- In the **Denied Number List**, select the list number you have configured with numbers you want to restrict to be dialed out from the BRI Port. Default: 02

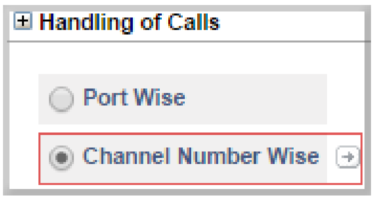
  If you have not configured the Denied Number List,

  - Click **Settings**. The Number Lists window opens.
  - You may configure the default Denied Number List 2 or any other list. See “Number Lists” to configure the restrict numbers.
  - Click **Submit** to save the Denied Number List and close the window.

Channel Number Wise

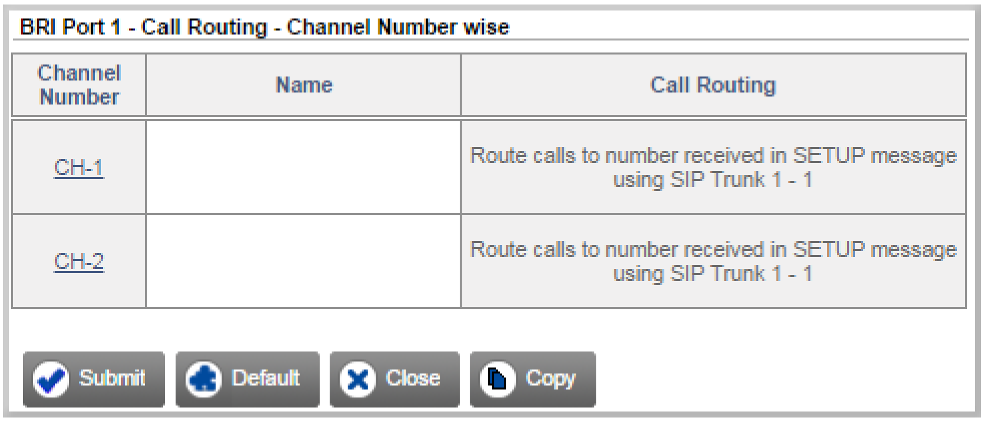
To configure Handling of Calls for each channel,

- Click the **Channel Number Wise** check box.



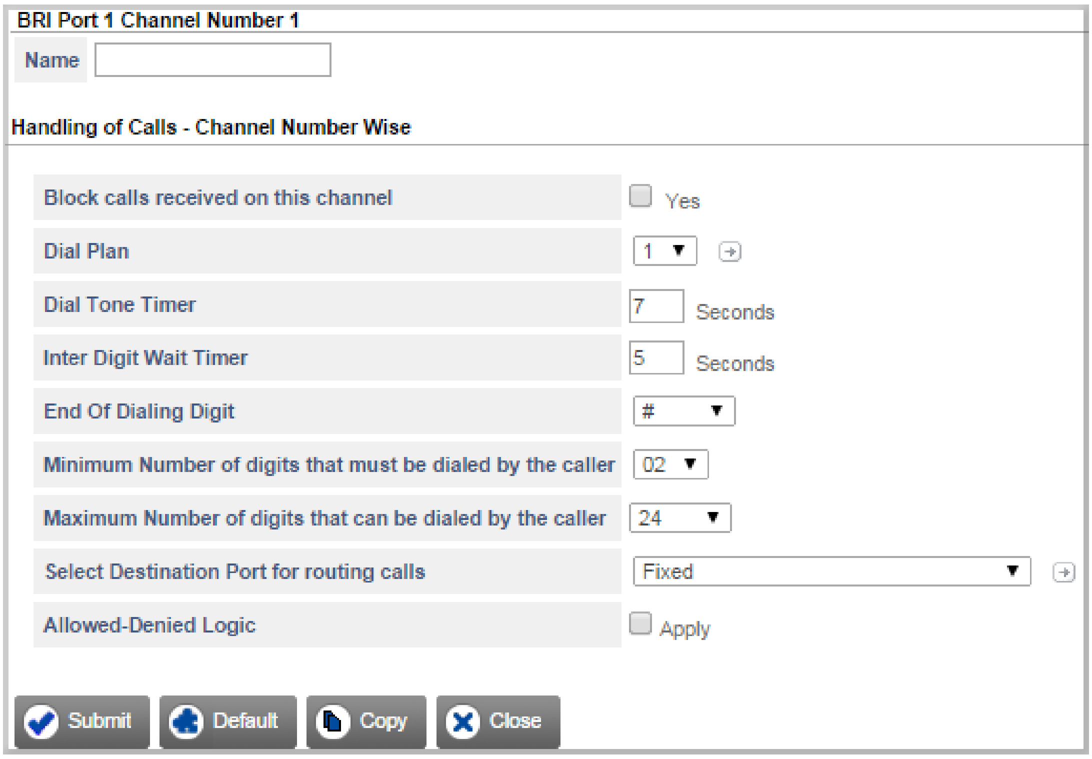
- Click **Settings**.

- The **BRI Port 1 - Call Routing - Channel Number Wise** window opens.



- To configure Channel 1 parameters, click **CH-1.**

- The **BRI Port 1 Channel Number 1** window opens.



- Assign a **Name** to the channel for identification.

- Under **Handling of Calls - Channel Number Wise** configure the following parameters.

  - Keep the **Block calls received on this channel** check box disabled.

    Select this check box only if you do not want to route calls received on this channel.

  - SARVAM UMG supports 8 Dial Plans with total 64 entries in each table. When a user dials a number, it is compared with the Destination Number configured in the Dial Plan. If a match is found, the system routes the call immediately without waiting for End of Dialing and if a match is not found, the system will wait for the End of Dialing and then route the call as per the Destination Port Selection method configured.

Select the **Dial Plan** table number you configured for this port. If you have not configured the Dial Plan table you may do so now,

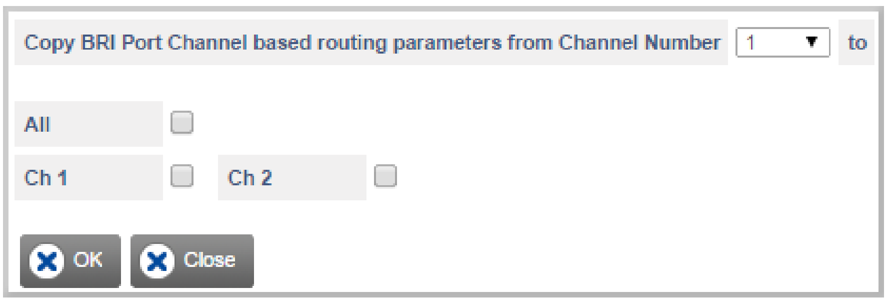
- Click **Settings** the Dial Plan Table opens.
- Configure the numbers in the table. For detailed instructions, see “Dial Plan”.

- Set the duration of the **Dial Tone Timer** as per your requirement. This is the time for which SARVAM UMG will play Dial Tone to the caller. Default: 7 seconds. On expiry of this timer, the system plays error tone to the caller.
- Set the duration of the **Inter Digit Wait Timer**. Define the number of seconds the system should wait while receiving the digits dialed by the caller to consider it as End of Dialing. The range of this timer is 01 to 99 seconds. Default: 5 seconds.
- As **End of Dialing Digit,** select whether the system should consider # or * as termination digit to detect end of dialing. Default: #
- In **Minimum number of digits that can be dialed by the caller**, select the minimum number of digits to be dialed by the user for the system to consider it as a valid number**.** Valid range is 01 to 24 digits. Default: 2 digits.
- In **Maximum number of digits that can be dialed by the caller**, define the maximum number of digits to be dialed by the user for the system to consider it as End of Dialing**.** Valid range is 01 to 24 digits. Default: 24 digits.
- **Select the Destination Port for routing calls**, see “Destination Port Determination”.
- To apply the **Allowed - Denied Logic**, see “Allowed - Denied Logic”.

- Click **Submit** to save the settings.

Copy Channel Based Routing Parameters

- You can also copy the settings of a BRI Channel to another BRI Channel using the **Copy** button. To do this,
  - Click the **Copy** button. The **Copy BRI Port Channel based routing parameters from Channel Number** window opens.

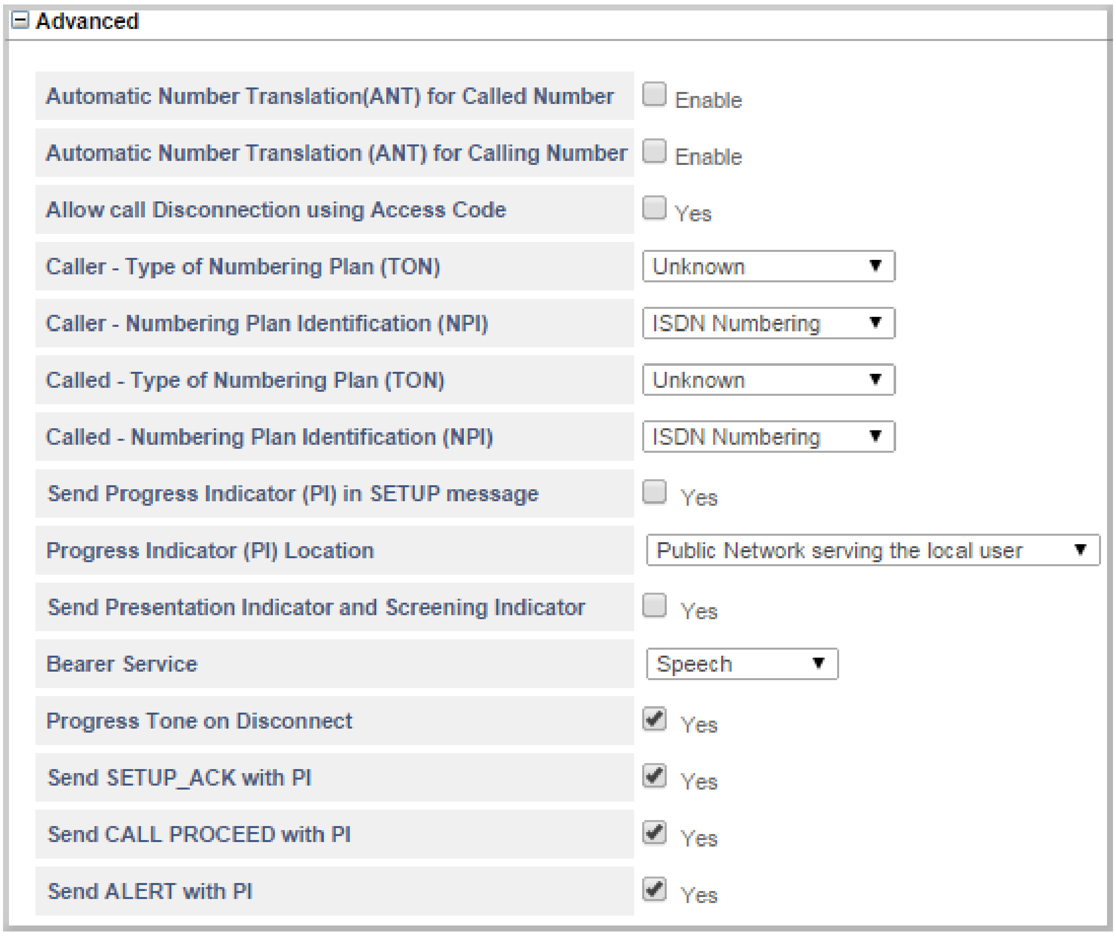


- In the **Copy BRI Port Channel based routing parameters from Channel Number** box, select the number of the channel you want to copy settings *From*. Select the check boxes of the desired channel numbers you want to copy the settings *To*.
- If you want to copy the settings *To* all the channels, select the **All** check box.
- Click the **OK** button.
- Once you have copied the settings, you can again edit the specific parameters of the BRI Channel you copied the settings to.

- Close the **BRI Port 1 Channel Number 1** window.
- To configure Channel 2, click **CH-2** on the **BRI Port 1 - Call Routing - Channel Number Wise** window and follow the same instructions as given above.
- Close the **BRI Port 1 - Call Routing - Channel Number Wise** window.

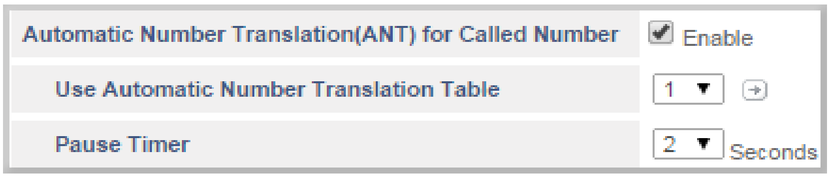
Advanced

- Click **Advanced** to expand.



- You can apply **Automatic Number Translation logic** on outgoing calls made from the BRI Port.

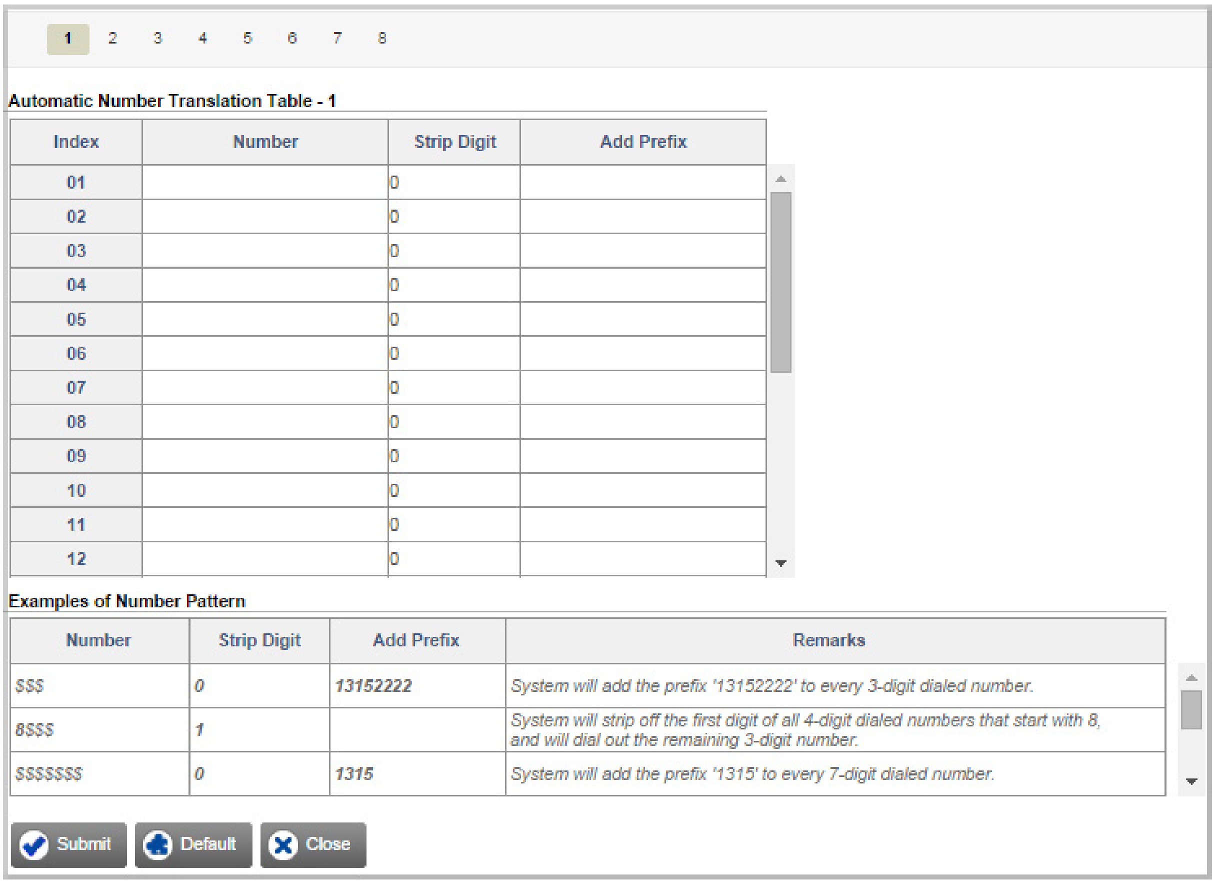
- To apply ANT logic on the Called Numbers, select the **Automatic Number Translation (ANT) for Called Number** check box. Default: Disabled.



  - In **Use Automatic Number Translation Table**, select the ANT Table number you have configured for the Called Numbers. Default: Table 1.

    If you have not configured the Automatic Number Translation Table,

    - Click **Settings** .
    - The **Automatic Number Translation Table** window opens.



    - You may configure the default Automatic Number Translation Table or any other Table. See "Automatic Number Translation (ANT)" to configure the ANT Table.
    - Click **Submit** to save the ANT Table and close the window.
    - Return to ANT parameter and assign the ANT Table you configured.
    - Click **Submit**.

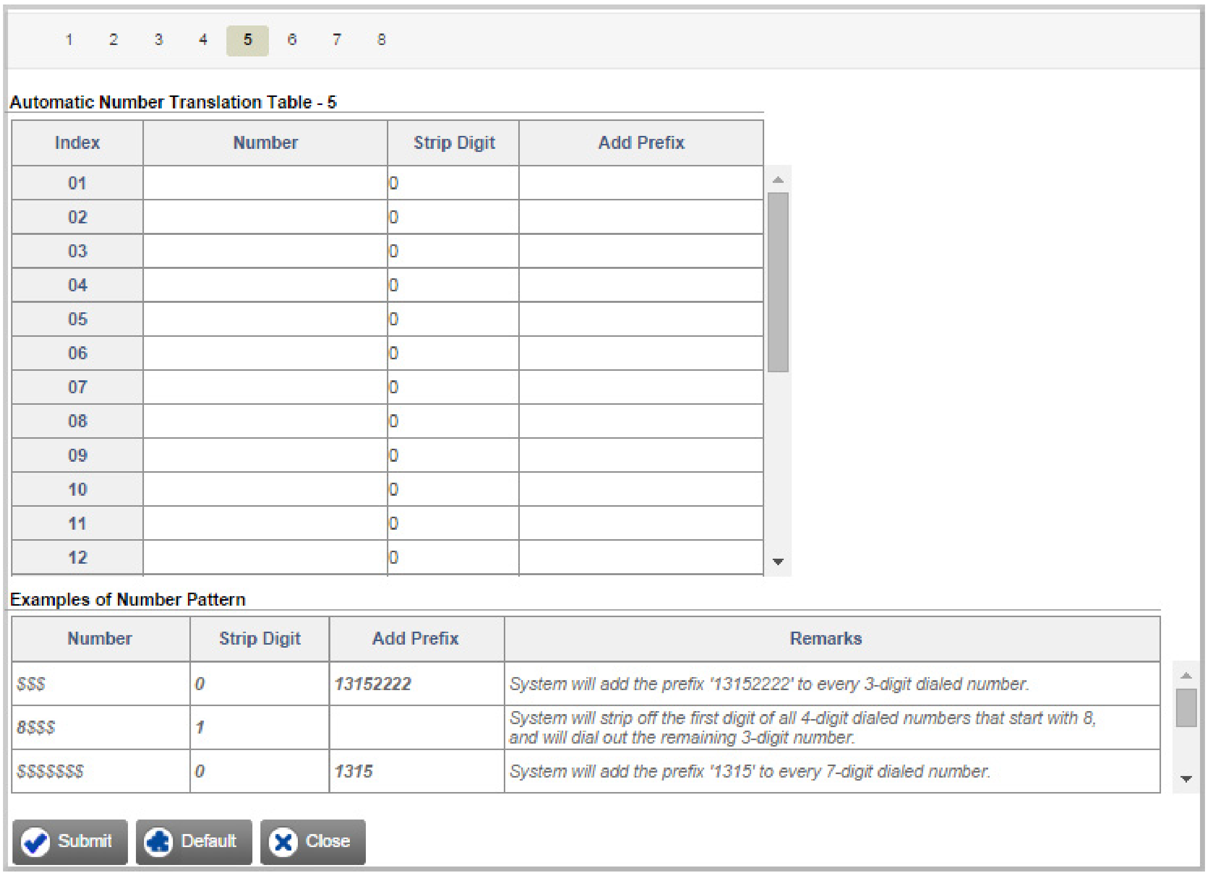
- Set the duration of the **Pause Timer**, if you have configured ^ (Pause) in the Add Prefix column of the ANT Table. Valid range is 1 to 9 seconds. Default: 2 seconds.

- To apply ANT logic on the Calling Numbers, click the **Automatic Number Translation (ANT) for Calling Number** check box. Default: Disabled.

- In **Use Automatic Number Translation Table**, select the ANT Table number you have configured for the Calling Numbers. Default: Table 5.

  If you have not configured the Automatic Number Translation Table,

  - Click **Settings**.
  - The **Automatic Number Translation Table** window opens.



Index	Number	Strip Digit	Add Prefix
01		0	
02		0	
03		0	
04		0	
05		0	
06		0	
07		0	
08		0	
09		0	
10		0	
11		0	
12		0	

Number	Strip Digit	Add Prefix	Remarks
$$$	0	13152222	System will add the prefix '13152222' to every 3-digit dialed number.
8$$$	1		System will strip off the first digit of all 4-digit dialed numbers that start with 8, and will dial out the remaining 3-digit number.
$$$$$$$	0	1315	System will add the prefix '1315' to every 7-digit dialed number.

  - You may configure the default **Automatic Number Translation Table - 5** or any other Table. See "Automatic Number Translation (ANT)" to configure the ANT Table.
  - Click **Submit** to save the ANT Table and close the window.
  - Return to ANT parameter and assign the ANT Table you configured.

- To enable the feature Disconnect Call using Access Code on the BRI Port, select the **Allow Call Disconnection using Access code** check box. To know more about this feature, see “Disconnecting a Call using Access Code”.

- Select the required option for sending the **Caller-Type of Numbering Plan (TON) —** Unknown, International, National, Network Specific, Subscriber, Abbreviated or Reserved. Default: Unknown.

- Select the required option for sending the **Caller-Numbering Plan Identification (NPI)** — Unknown, ISDN Numbering, Data Numbering, Telex Numbering, National Numbering, Private or Reserved. Default: ISDN Numbering.

- Select the required option for sending the **Called-Type of Numbering Plan (TON)** — Unknown, International, National, Network Specific, Subscriber, Abbreviated or Reserved. Default: Unknown.

- Select the required option for sending the **Called-Numbering Plan Identification (NPI) —**Unknown, ISDN Numbering, Data Numbering, Telex Numbering, National Numbering, Private or Reserved. Default: ISDN Numbering.

- Select the **Send Progress Indicator (PI) in SETUP message** check box if you want the progress indicator value to be sent in SETUP message. Default: Disabled.

- Set the **Progress Indicator (PI) value in SETUP message** to the desired value. You can select — 1 or 3.

  Progress indicator 1 indicates that the call is not end-to-end ISDN and further call progress information may be available in-band.

  Progress indicator 3 indicates that the origination address is non-ISDN. This value will be included in the Setup Message to indicate whether the calling party is an ISDN device or not. Default: 1.

- Select the **Progress Indicator (PI) Location** for the SETUP message.The location is a progress indicator information element that indicates from where the message is coming. Default: Public Network serving the local user.

- Select the **Send Presentation Indicator and Screening Indicator** check box, if you want the system to display the presentation and screening information to the remote end. Default: Disabled.

  - Select the required **Presentation Indicator**. This allows remote end to know whether CLI Number should be displayed to user or not. You can select — Presentation Allowed, Presentation Restricted or Received from Source Port. Default: Received from Source Port.

  - Select the required **Screening Indicator**. This indicates whether the information is provided by the user or the network along with the screening details — not screened, verified and passed or verified and failed. Default: User-provided, not screened.

- Select the **Bearer Service** supported by your service provider. This will be sent in the SETUP Message. You can select — Speech, 3.1 KHz Audio. Default: Speech

- Select the **Progress Tone on Disconnect** check box, if you want the system to play the progress tone on the port when the call is released by the remote end or disconnected by the system. Default: Enabled.

- Select the **Send SETUP_ACK with PI** check box, if you want the system to send PI (Progress indicator) element in Setup Ack message. Default: Enabled.

- Select the **Send CALL PROCEED with PI** check box, if you want the system to send PI (Progress indicator) in Proceed message. Default: Enabled.

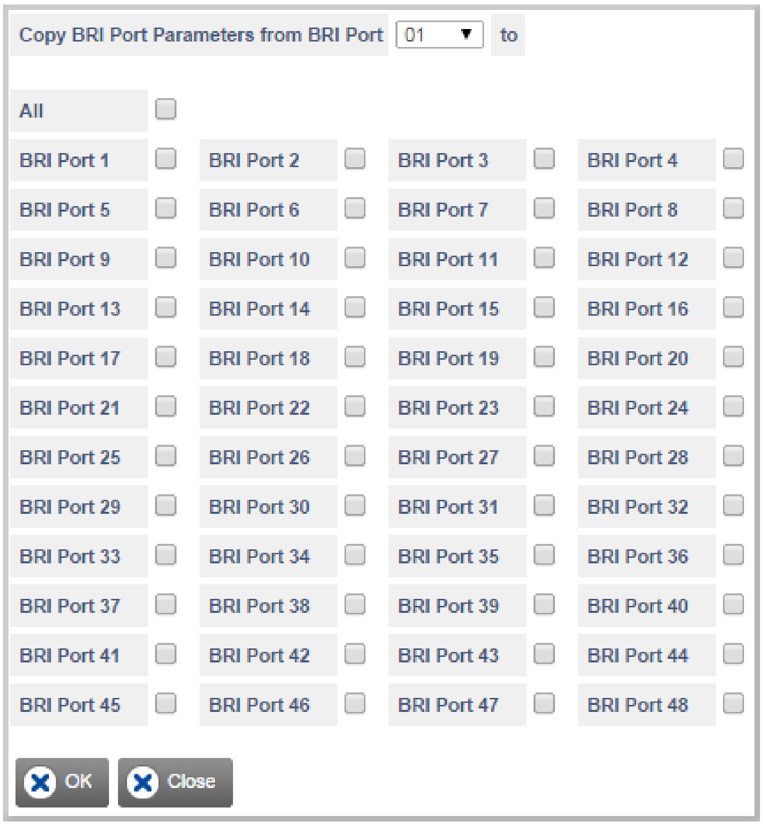
- Select the **Send ALERT with PI** check box, if you want the system to send PI (Progress indicator) in Alert message. Default: Enabled.

To configure the next BRI Port click the BRI Port number tab and follow the same instructions as given above or you can copy the parameters from another port.

Copy BRI Port Parameters

- You can also copy the settings of a BRI Port to another BRI Port using the **Copy** button. To do this,

  - Click the **Copy** button. The **Copy BRI Port Parameters** window opens.



  - In the **Copy BRI Port Parameters from BRI Port** box, select the number of the port you want to copy settings *From*. Select the check box of the respective port numbers you want to copy the settings *To*.

  - If you want to copy the settings *To* all the ports, select the **All** check box.

  - Click the **OK** button.

  - Once you have copied the settings, you can again edit the specific parameters of the BRI Port you copied the settings to.

BRI Port - Terminal

If you have connected the BRI line from your ISDN Service Provider directly to BRI Port of the SARVAM UMG over the NT1 device or connected the BRI Port to an ISDN System (NT), you must configure the **Orientation Type** of the port as **Terminal**.

- Click the **Basic Settings** link to expand.
- Click the **BRI Port** link.

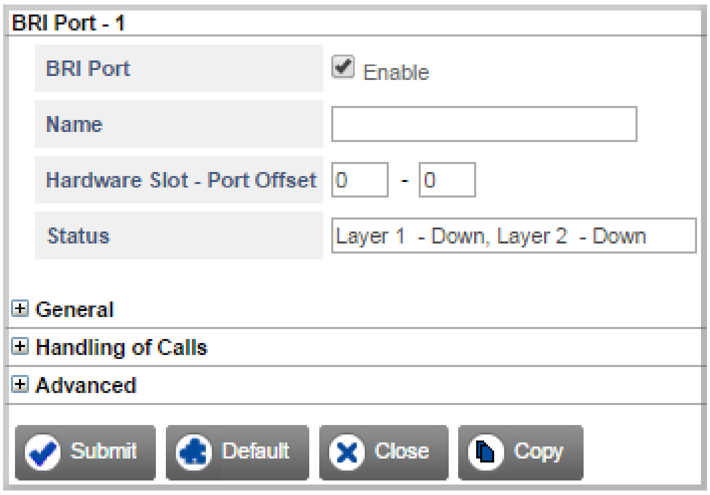
The BRI Port page displays the following parameters:

- **Port:** It displays the BRI Port numbers.Click on the desired BRI Port number to configure the Port Parameters.
- **Hardware Slot-Port:** The SARVAM UMG can automatically detect and assign the Hardware Slot and Port numbers to the BRI software ports. However, if required you may change the Hardware Slot and Port assigned to the BRI software port. In this case, enter the desired Hardware Slot and Port number.
- **Enable:** Keep the **BRI Ports** enabled. Clear the BRI Port **Enable** check box, only if you do not want to use the respective port. Default: Enabled.
- **Name:** Assign a Name to the BRI Port for identification. The Name can be a maximum of 24 characters.
- **Status:** This displays the status of Layer 1 and Layer 2, that is, Up or Down.
- **Interface Type:** It displays the type of interface you select — Point to Point or Point to Multipoint.
- **Orientation:** It displays the type of orientation you select — Network or Terminal.
- **Call Routing:** It displays the Call Routing Method you select.

To configure BRI Port as **Terminal**,

- Click **BRI-1**.

  The **BRI Port-1** window opens.



- Keep the **BRI Port** check box enabled.

  Clear the **BRI Port Enable** check box only when you do not want to use this BRI Port. Default: Enabled.

- You can assign a **Name** to the BRI Port, which will be displayed to the called party, if the called party telephone instrument supports name display. Default: Blank

  The name you assign may consist of a maximum of 24 characters.Default: Blank.

- SARVAM UMG will assign the **Hardware Slot - Port Offset** automatically, when any card is inserted in the system.

  Hardware slot is the number of the universal slot of SARVAM UMG in which the BRI Card is inserted. Range of slot number is 1-12. Port is the number of BRI hardware port on the card to which the BRI line is connected.

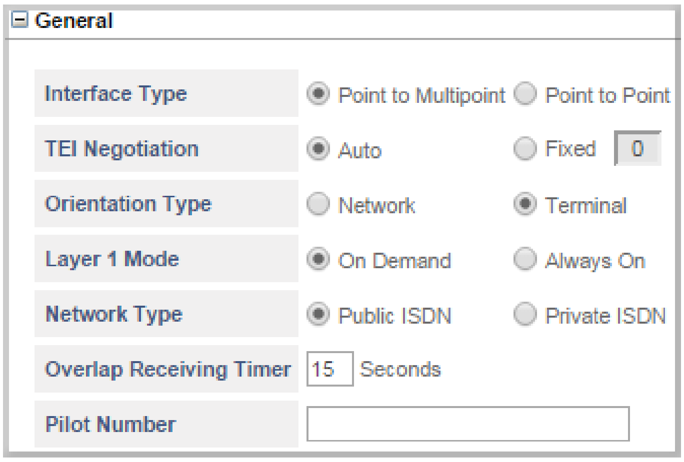
  However, if required, you may change the Hardware Slot and Port assigned to the BRI software port. In this case, enter the desired Hardware Slot and Port number.

  If you want to de-assign the Hardware Slot and Port, enter '00' in both fields. By default, Hardware Slot-Port is 00–00.

- **Status** displays the status of the BRI Port.

General Parameters

- Click **General** to expand.



- Select the **Interface Type** as **Point-to-Point** or **Point-to-Multipoint** according to your installation. Default: Point-to-Multipoint.

- Select **TEI Negotiation** for the BRI Port as **Auto** or **Fixed**. Default: Auto.

  TEI (Terminal Endpoint Identifier) Negotiation mode is used to integrate the SARVAM UMG with a specific ISDN-System. The TEI value programmed for BRI-NT Port should match the TEI value programmed in the Terminal Equipment connected to it.

  - If you set TEI Negotiation to **Fixed**, enter the value of the Fixed TEI Negotiation from 00 to 63. Default: 00.

- Select the **Orientation Type** of the BRI Port as **Terminal**.

- Select the required **Layer 1 Mode** as **On Demand** or **Always On** as supported by your Service Provider. Default: On Demand.

- Select the **Network Type** according to your installation. Default: Public ISDN.

  - Select **Public ISDN**, if the BRI Port is directly connected to the ISDN network.

  - Select **Private ISDN**, if the BRI Port is connected to the NT Port of an ISDN System.

  The calling party number sent while routing the call through the BRI Port will depend on the Network Type, whether Public Network or Private.

- Set the duration (in seconds) of the **Overlap Receiving Timer**. On expiry of this timer the system will process the incoming call as per the Incoming Call Routing logic configured.

  The Overlap Receiving Timer is applied only while receiving the called party number information in the overlap receiving mode. The Timer is not relevant for the port in overlap sending mode.

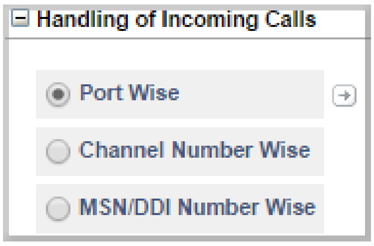
  The range of Overlap Receiving Timer is from 01 to 99 seconds. Default: 15 seconds.

- Enter the **Pilot Number.** This number will be sent as the calling party number when the call is routed using the BRI Port and when Reverse DDI logic is not applied on the port.

  You need to ask your service provider for the Pilot Number if you have selected **Public ISDN** as the **Network Type** for the BRI Port. The length of the Pilot Number can be of maximum 24 digits and the valid digits are 0-9, #,*. Default: Blank.

Handling of Incoming Calls

Click **Handling of Incoming Calls** to expand.



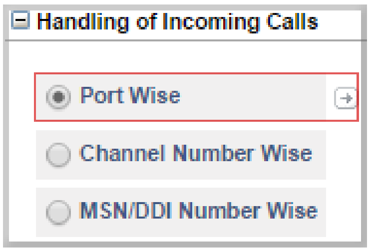
Select the method to route the incoming calls from the BRI Port. SARVAM UMG provides three options for **Handling of Incoming Calls:** Port wise, Channel Number wise and MSN/DDINumber wise. Default: Port wise.

- **Port Wise:** Select this method to apply the call routing method for the entire port.
- **Channel Number Wise:** Select this method to apply a different call routing method for each channel. You can configure a different incoming call routing option for each channel.
- **MSN/DDI Number Wise:** Select this method to apply a different call routing method for each MSN number given by the Service Provider for the BRI Line. SARVAM UMG allows you to configure upto 4 MSN Numbers.

Port Wise

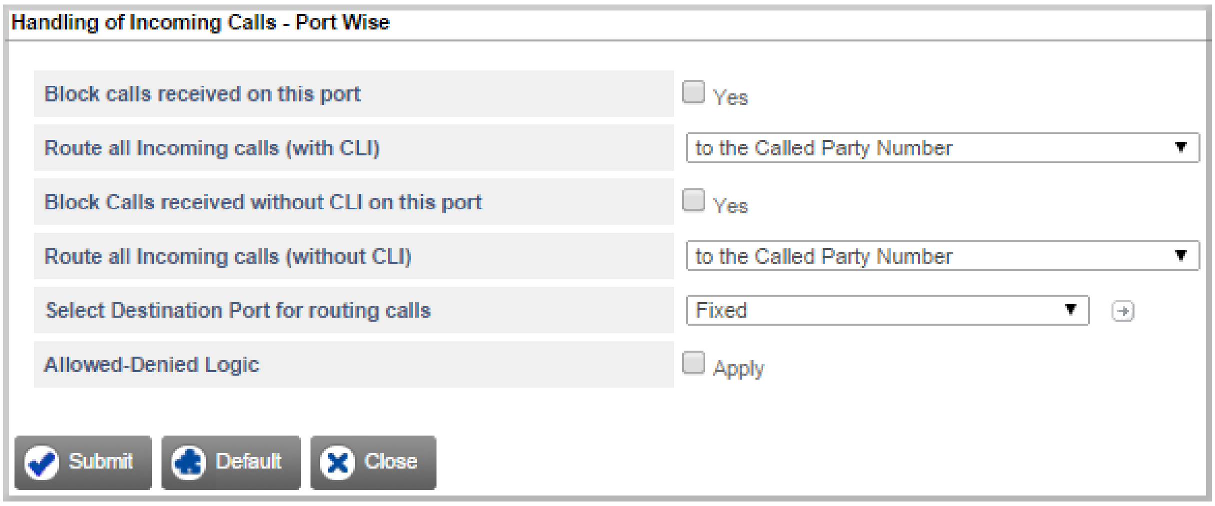
To configure Handling of Incoming Calls Port Wise,

- Select the **Port Wise** check box.



- Click **Settings** .

- The **Handling of Incoming Calls - Port Wise** window opens.



- Keep the **Block calls received on this port** check box disabled.

  Select this check box only if you do not want to route calls received on this port.

Destination Number Determination

Select the desired destination number determination method for routing incoming calls *with* and *without* CLI.

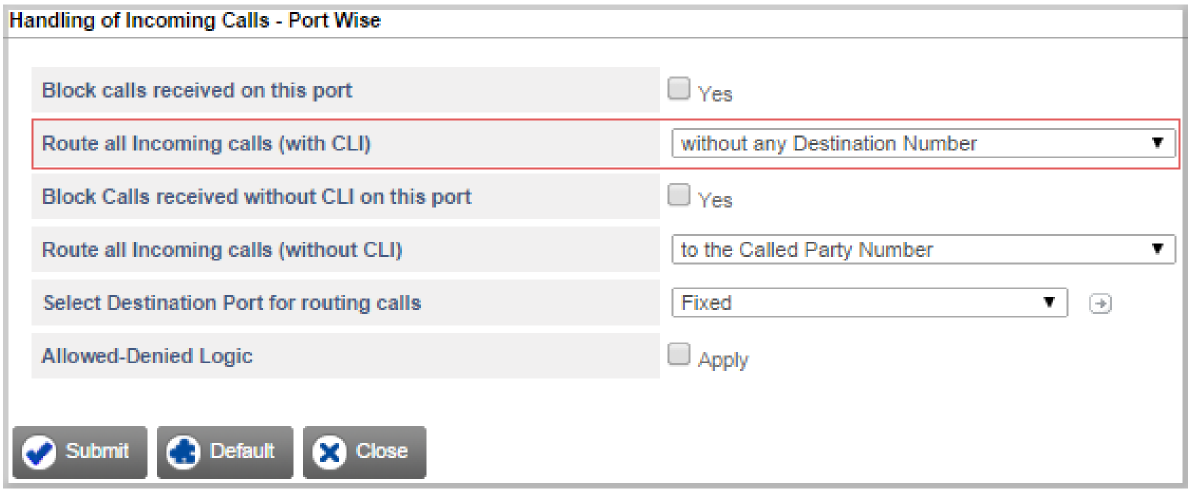
- To **Route all Incoming calls (with CLI)**, you may select from any of the following methods:
  - without any Destination Number
  - to a Fixed Destination Number
  - on the basis of Calling Party Number
  - on the basis of DDI Number
  - to the Called Party Number
  - after Answering the Call and Collecting the Digits
    Default: to the Called Party Number

  Read further for instructions on selecting and configuring each of these destination number determination methods.

If the destination number to be dialed out is an IP Address, SARVAM UMG will not check the Destination Port Determination Method. Instead, it will route the call using the SIP Trunk / Group programmed for IP Dialing. (See "IP Dialing" to know more).

Route Calls without any Destination Number

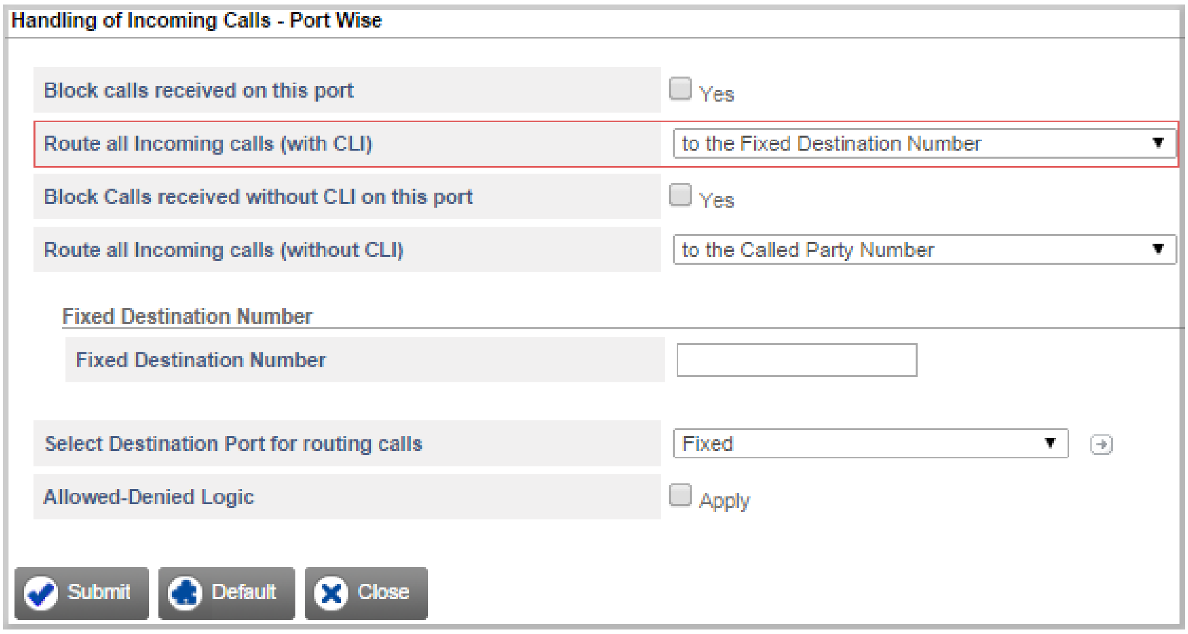
In this method, all calls received on the BRI Port are directly routed to the destination port, irrespective of the Destination Number.



- To apply this method, in the **Route all incoming calls (with CLI)**, select **without any Destination Number**.

Route to a Fixed Destination Number

In this method, calls received on the BRI Port are routed to a fixed destination number, which is configured for the BRI Port.



To apply this method, do the following:

- In **Route all Incoming calls (with CLI)**, select **to the Fixed Destination Number**.

- In the **Fixed Destination Number** box that appears, enter the desired destination number. The Destination Number may consist of a maximum of 24 digits. Valid digits are 0 to 9, *, # and. (dot/period). Default: Blank.

- Click **Submit** to save your settings.

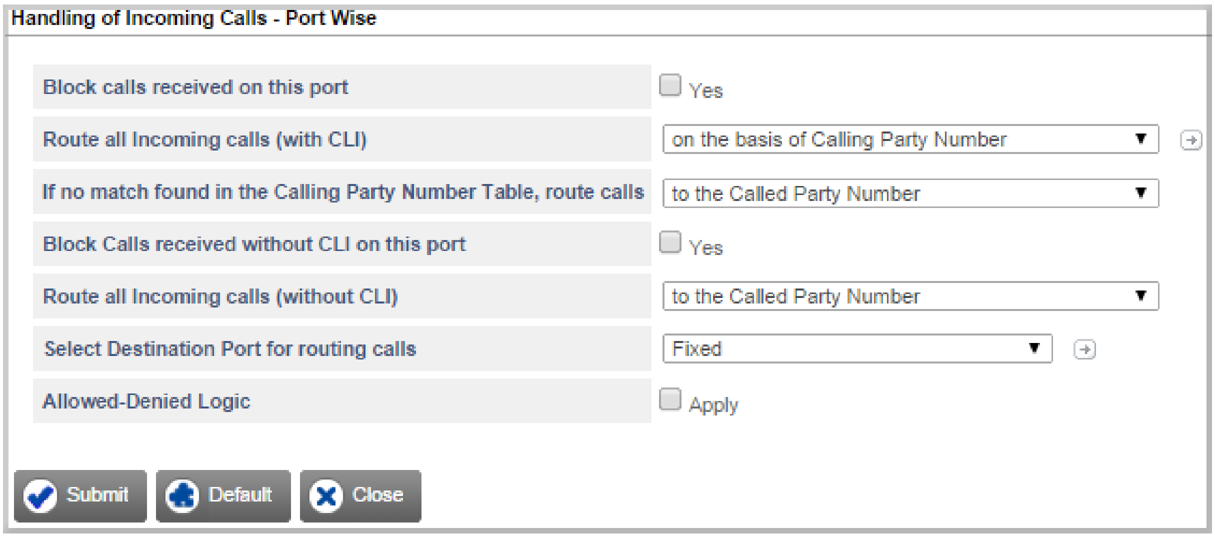
Route on the basis of Calling Party Number

In this method, a call received on the BRI Port is routed to a specific number, as per the calling party's number. You must configure the calling party numbers in the *Calling Party Number Based* Table.

When there is an incoming call on the BRI Port, SARVAM UMG will match the Calling Party Number with the entries of the Calling Party Number Based Table. If a match is found, the call is routed to the destination number configured for that Calling Party Number.

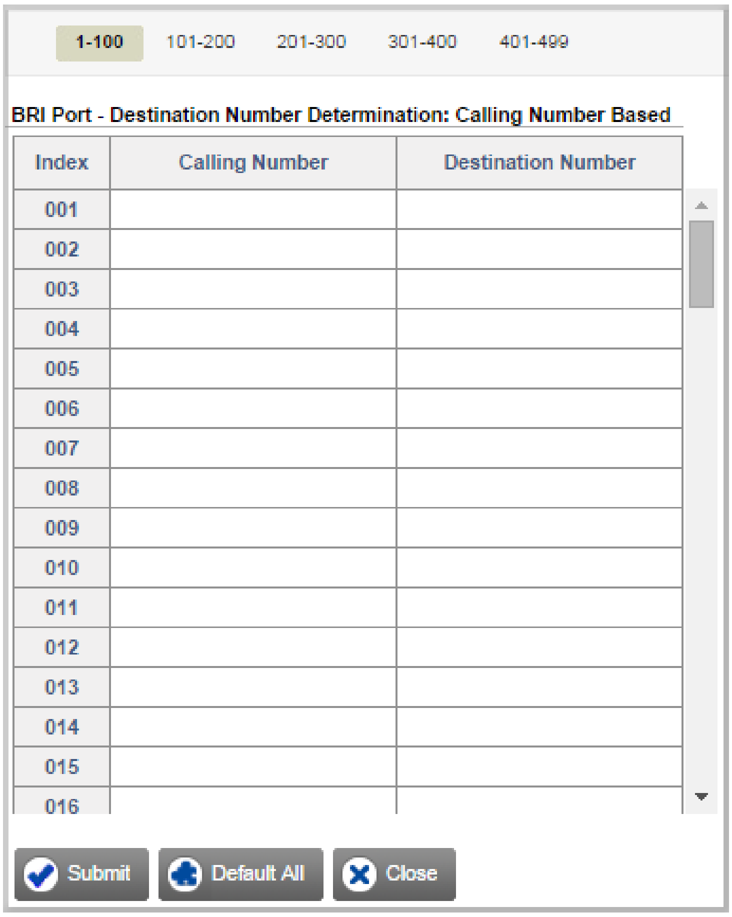
To apply this method, do the following:

- In **Route all Incoming calls (with CLI)**, select **on the basis of Calling Party Number**.



- Click **Settings** .

- The **BRI Port - Destination Number Determination: Calling Number Based** Table window opens.



- In **Calling Number**, enter the calling party numbers. The Calling numbers may consist of a maximum of 24 characters. Default: Blank.

- For each calling party number, enter a corresponding destination number in **Destination Number**. Destination numbers may consist of a maximum of 24 characters. Digits 0 to 9, *, # and (.) dot are allowed. Default: Blank.

- Click **Submit** to save your entries. Close the window to return to the **Handling of Incoming Calls - Port Wise** window.

  You can also configure the **Calling Number Based** table from *Advanced Settings.* For instructions, see "Destination Number Determination" under *Advanced Settings*.

- Select a method for routing incoming calls with CLI that *do not match* with any entries in the Calling Party Number Based Table.

  In the **If no match found in the Calling Party Number Table, route calls** box, select the desired method from the following options for processing the call:
  - to a Fixed Destination Number
  - on the basis of DDI Number
  - to the Called Party Number
  - after Answering the Call and Collecting the Digits

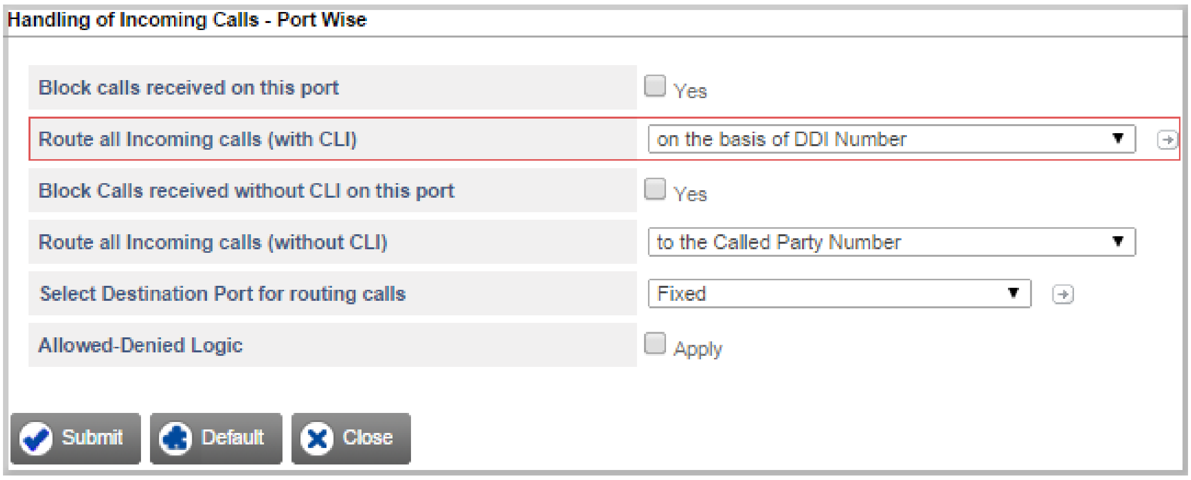
  Default: to the Called Party Number.

Route on the basis of DDI Number

In this method, incoming calls on the BRI Port are routed to specific numbers as per the DDI number received in the SETUP message on the BRI Port.

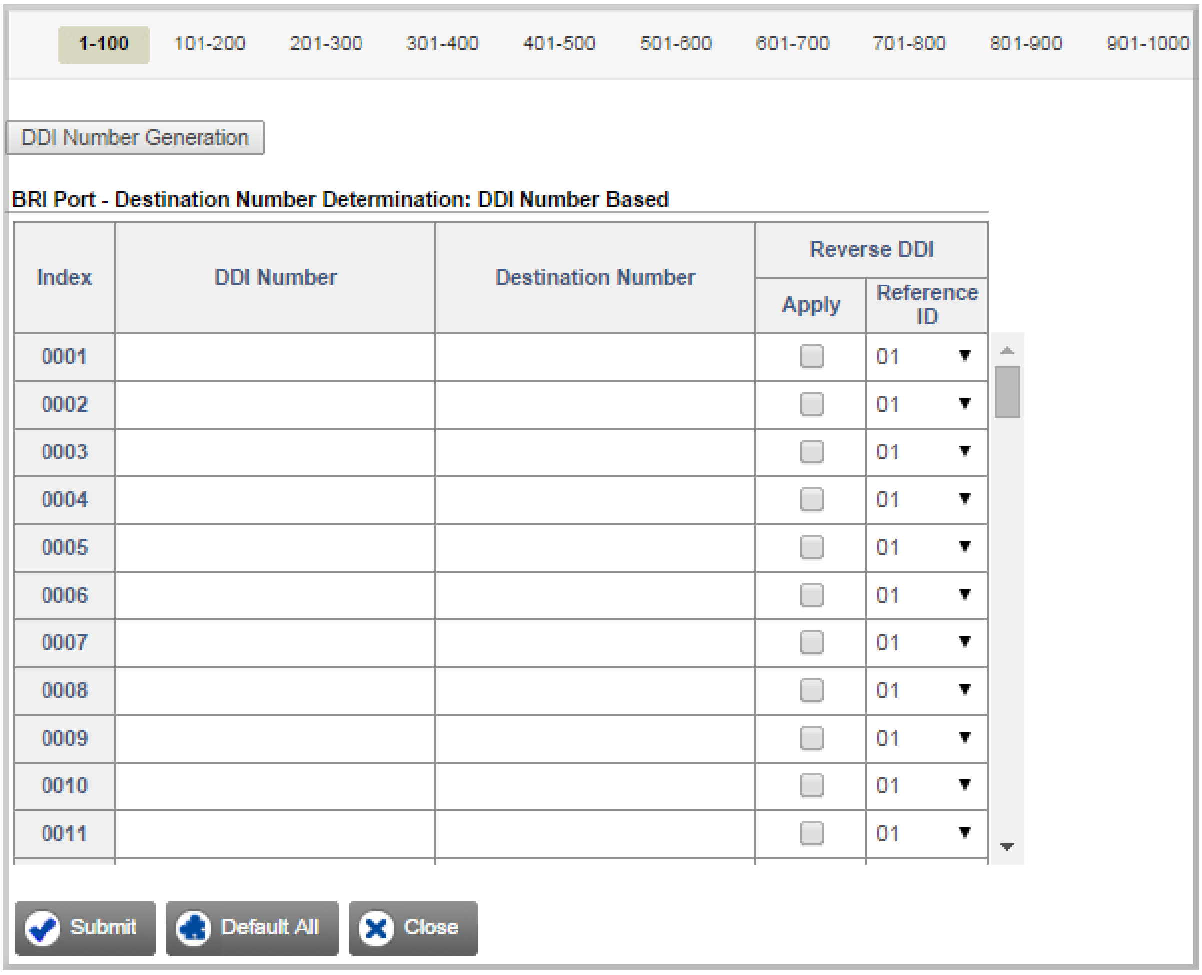
To apply this method, do the following:

- In **Route all Incoming calls (with CLI)**, select **on the basis of DDI Number**.



- Click **Settings** .

The **BRI Port - Destination Number Determination: DDI Number Based** Table opens.



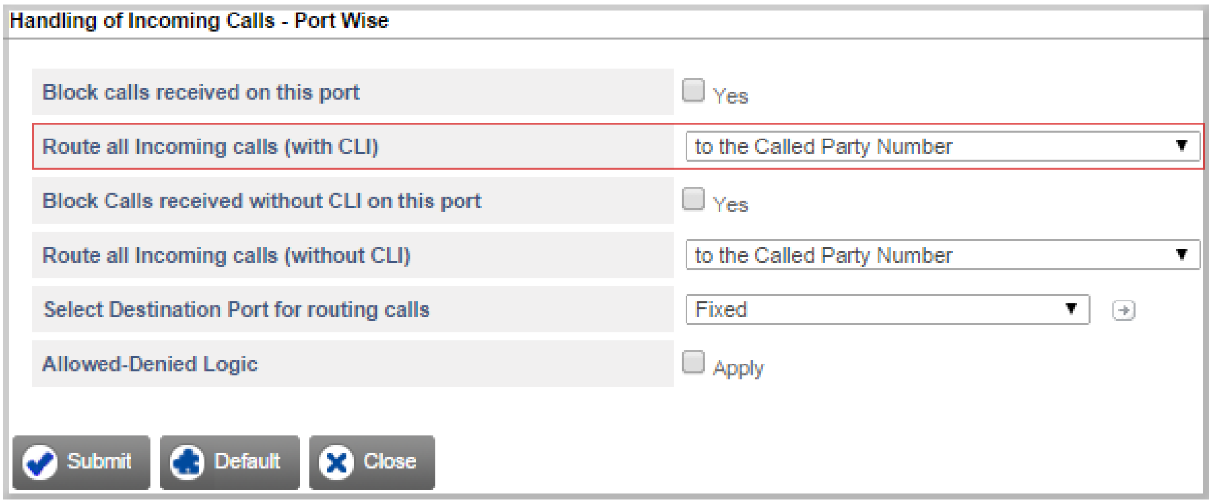
- In **DDI Number**, enter the DDI Numbers allotted by your service provider.
- For each DDI Number, enter the corresponding destination number in **Destination Number**.
- To apply **Reverse DDI** for each number, select the check boxes under **Apply** and select the **Reference ID** for the number. Default: Apply Reverse DDI is disabled and Reference ID is 1.
- Click **Submit** to save and close the window to return to the Handling of Incoming Calls - Port Wise window.

  You can also configure the **DDI Number Based** Table from *Advanced Settings.* For instructions, see "Destination Number Determination" under *Advanced Settings*.

Route to the Called Party Number

In this method, a call received on the BRI Port is routed to a specific number depending upon the called party number received in the SETUP Message on the BRI Port.

• To apply this method, in **Route all incoming calls (with CLI)**, select **to the Called Party Number**.

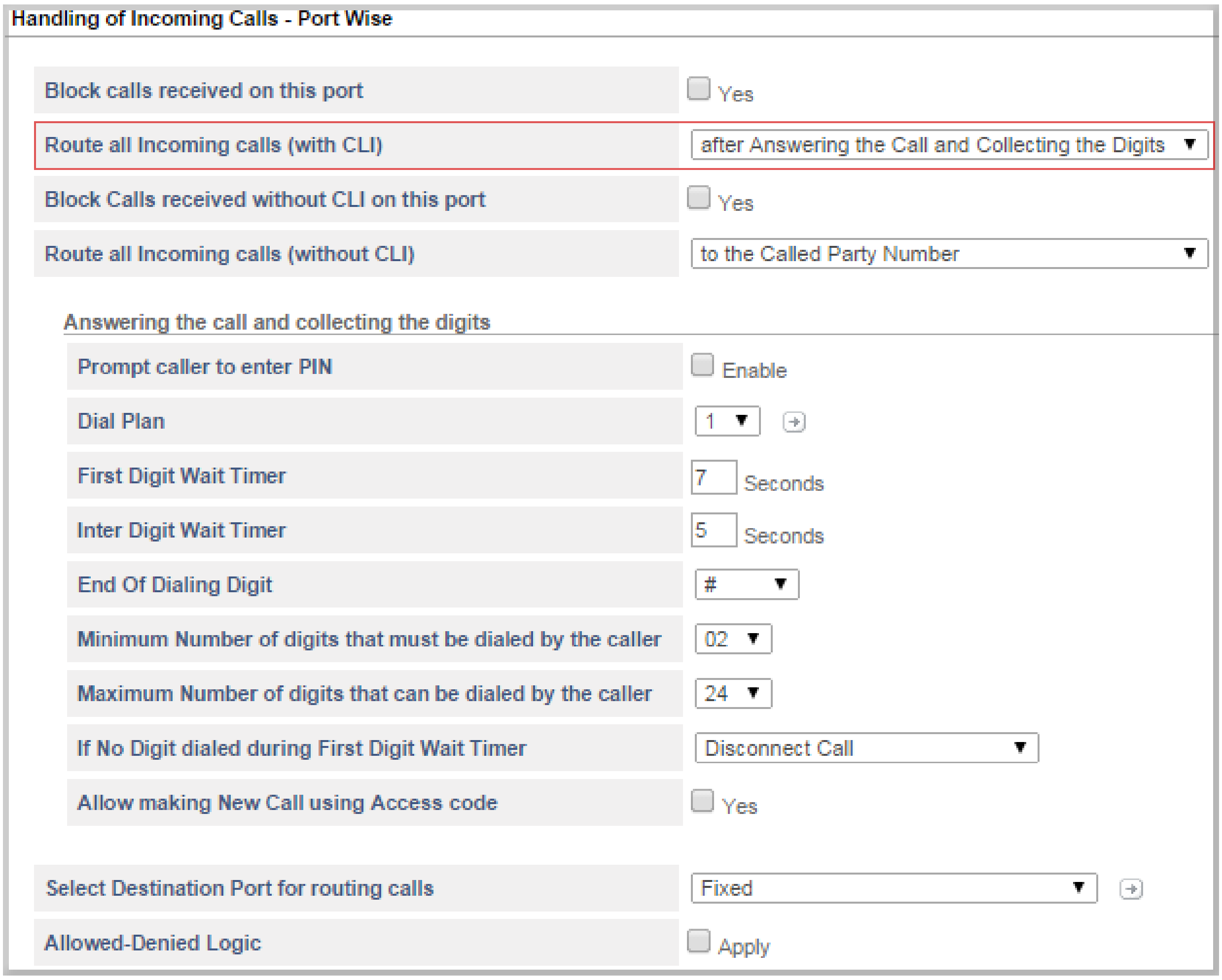


Route after Answering the Call and Collecting the Digits

In this method, the incoming call is answered and dial tone is played to the caller, allowing the caller to dial the desired number. The number dialed by the caller is considered as the destination number.

To apply this method, do the following:

- In **Route all Incoming calls (with CLI)**, select **after Answering the Call and Collecting the Digits**.



The related parameters of this method appear under **Answering the call and collecting the digits**.

- If you want to enable PIN Authentication on the BRI Port, select the **Prompt caller to enter PIN** check box.

  If you enable this check box, you must also configure the PIN Authentication Table. To know more about this feature and for detail instructions, see “PIN Authentication” under *Advanced Settings*.

- SARVAM UMG supports 8 Dial Plans with total 64 entries in each table. When a user dials a number, it is compared with the Destination Number configured in the Dial Plan. If a match is found, the system routes the call immediately without waiting for End of Dialing and if a match is not found, the system will wait for the End of Dialing and then route the call as per the Destination Port Selection method configured.

  Select the **Dial Plan** table number you configured for this port. If you have not configured the Dial Plan table you may do so now,

  - Click **Settings** the Dial Plan Table opens.
  - Configure the numbers in the table. For detailed instructions, see “Dial Plan”.

  - Set the duration of the **First Digit Wait Timer.** This is the duration for which you want the system to wait for the caller to dial the destination number after the dial tone. Valid range is 01 to 99 seconds. Default: 7 seconds.

  - You may configure the following options as End of Dialing indication:

    - Set the duration of the **Inter Digit Wait Timer**. This is the duration for which you want the system to wait while receiving the digits dialed by the caller to consider it as End of Dialing. You may change this timer, if required. Valid range is 01 to 99 seconds. Default: 05 seconds.

    - In **End of Dialing Digit,** select # or * as termination digit the system should consider to detect end of dialing. Default: #

    - In **Minimum number of digits that can be dialed by the caller**, select the minimum number of digits to be dialed by the user for the system to consider it as a valid number**.** Valid range is 01 to 24 digits. Default: 2 digits.

    - In **Maximum Number of digits that can be dialled by the caller**, select the maximum number of digits to be dialed by the user for the system to consider it as End of Dialing**.** Valid range is 01 to 24 digits. Default: 24 digits.

      When the caller dials a number, the system will match it with the above End of Dialing indications and accept the one that matches first.

  - If the caller fails to dial the number during the First Digit Wait Timer, you can either have the system disconnect the call or route the call to a fixed destination number.

    In the **If No Digit dialed during First Digit Wait Timer** box, select the desired option: **Disconnect the Call** or **Use Fixed Destination Number**. Default: Disconnect Call.

    - If you selected **Use Fixed Destination Number**, enter the desired destination number in the **Fixed Destination Number** field**.** The Destination number may consist of a maximum of 24 digits. Valid digits are 0 to 9, *, # and . (dot/period). Default: Blank.

  - *The First Digit Wait Timer is loaded as soon as the system answers the call.*

  - *When you dial the first digit, the First Digit Wait Timer is stopped and the system loads the Inter Digit Wait Timer.*

  - *SARVAM UMG reloads the Inter Digit Wait Timer:*
    - *each time you dial a new digit till the termination digit is detected.*
    - *when you have dialed the maximum number of digits configured as End of Dialing.*

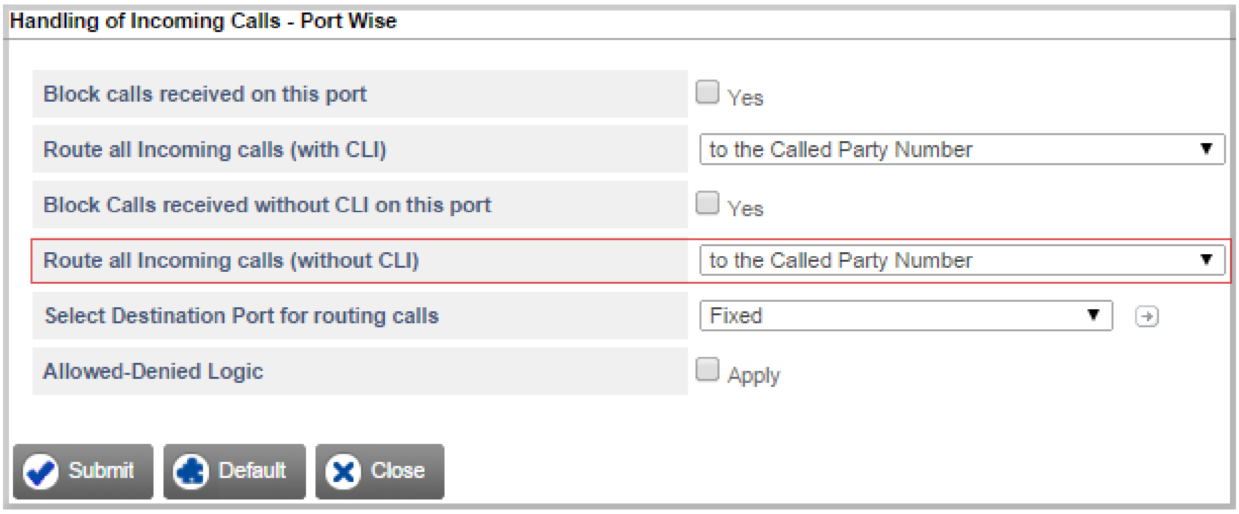
  - If you want to enable the feature Making New Call using Access Code on the BRI Port, select the **Allow making New Call using Access Code** check box. For further details, see “Making a New Call using Access Code”.

  - Click **Submit** to save settings.

- If you do not want to route the incoming calls received without CLI, through this BRI Port, select **Block Calls received without CLI on this Port** check box.

- To **Route all Incoming calls (without CLI),** you may select from any of the following methods:
    - to a Fixed Destination Number, see "Route to a Fixed Destination Number".
    - on the basis of DDI Number, see "Route on the basis of DDI Number".
    - to the Called Party Number, see "Route to the Called Party Number".
    - after Answering the Call and Collecting the Digits, see "Route after Answering the Call and Collecting the Digits".

    Default: to the Called Party Number.



Destination Port Determination

For the port/channel/MSN number, select the Destination Port for routing calls from the following options:

- Fixed
- On the basis of Destination Number
- On the basis of Calling Party Number

  Default: Fixed.

  Read the description and follow the instructions for each of these destination port selection methods given below.

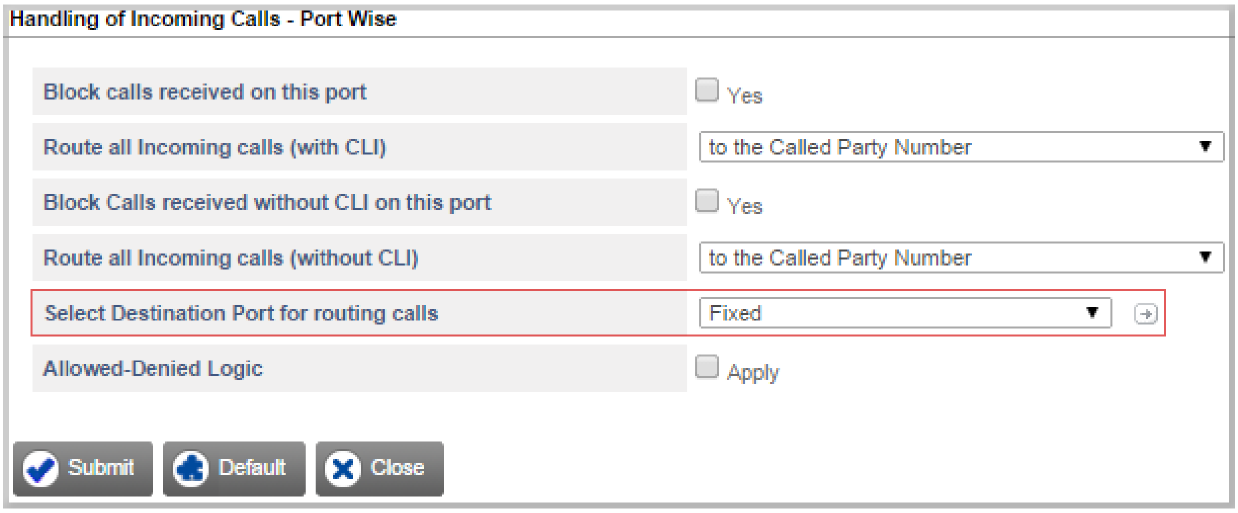
If the destination number to be dialed out is an IP Address, SARVAM UMG will not check the Destination Port Determination Method. Instead, it will route the call using the SIP Trunk / Group programmed for IP Dialing. (To know more, see the feature description "IP Dialing").

Fixed

In this method, calls received on the BRI Port are routed to a Fixed Destination Port, irrespective of the number dialed on the BRI Port.

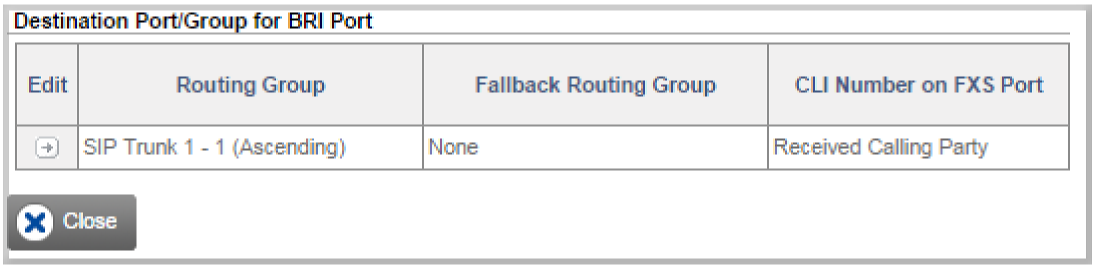
To apply this method, do the following:

- In **Select Destination Port for routing calls**, select **Fixed** option.



- Click **Settings** .

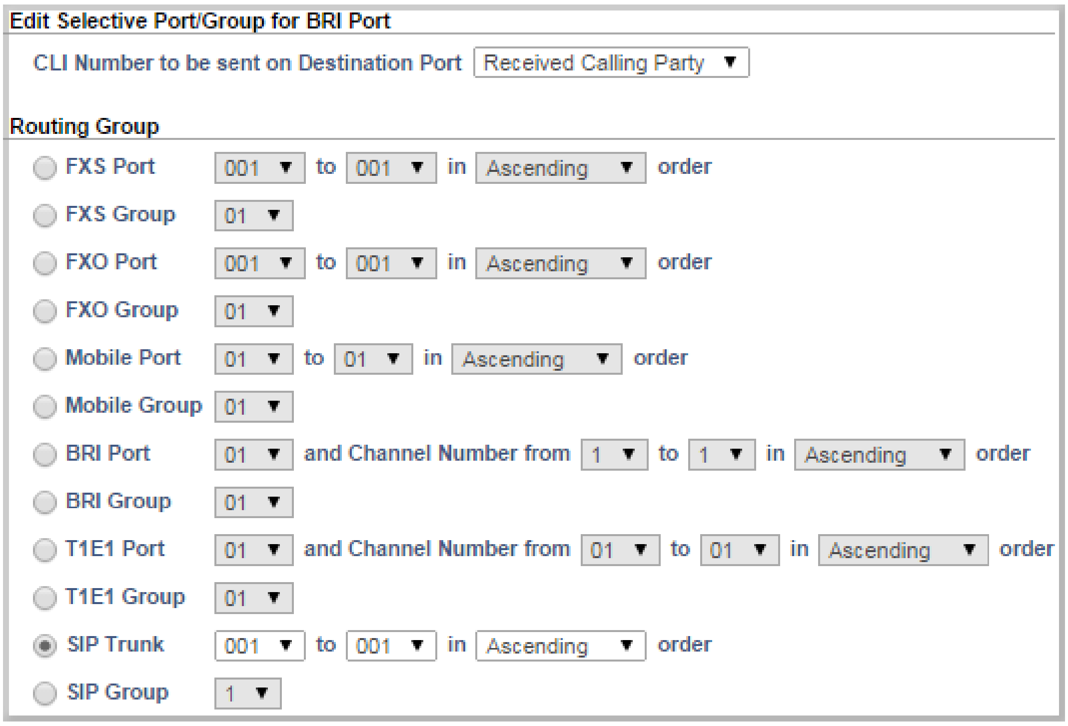
  The **Destination Port/Group for BRI Port** window opens.



  The default **Routing Group** and **Fallback Routing Groups** appear.

- If you wish to change the default Routing Group options, click **Edit** .

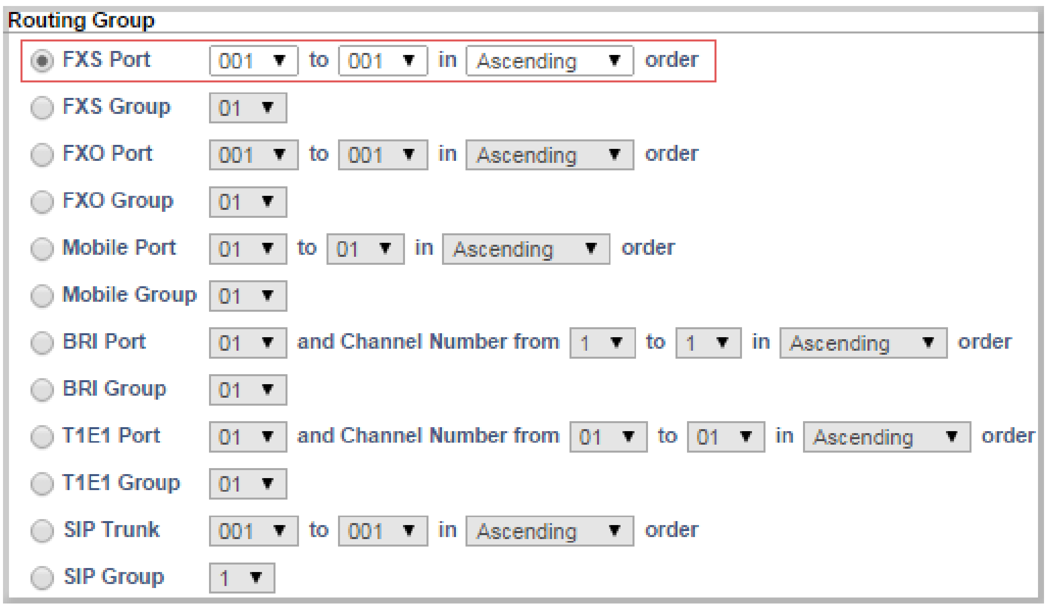
The **Edit Selective Port/Group for BRI Port** window opens.



- Select the **CLI Number to be sent on Destination Port**. You can select Received Calling Party or Received Called Party. Default: Received Calling Party.

CLI to be sent on Destination Port is applicable when FXS Port/Group is determined as the Destination Port.

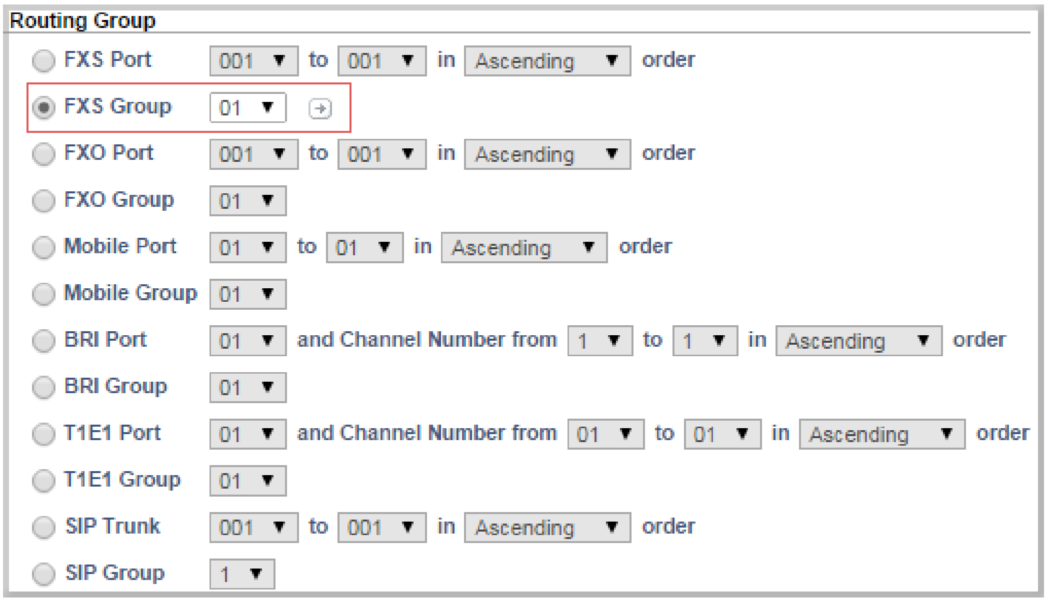
- Create the **Routing Group**.
    - To create a group of *sequential* **FXS Ports** as members,



   - Select the desired **FXS Port** numbers as members. Default:1.
   - In **in - order**, select the order in which the system should hunt for a free member FXS Port to route the call.

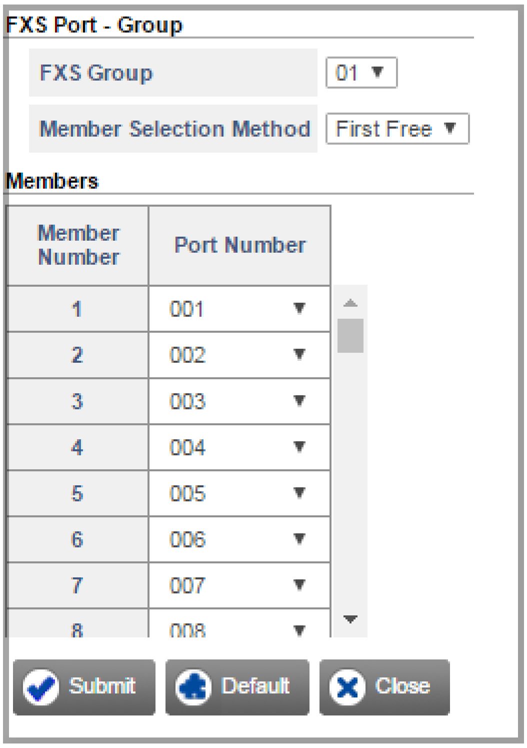
     Select **Ascending** to start hunting from the first to the last member FXS Port. Select **Descending** to start hunting from the last to the first member FXS Port. Default: Ascending.

- To create a group of *not-sequential* **FXS Ports** as members,
   - Select a **FXS Group**.



   - Select **FXS Group** number. Default:1.
   - Click **Settings** .

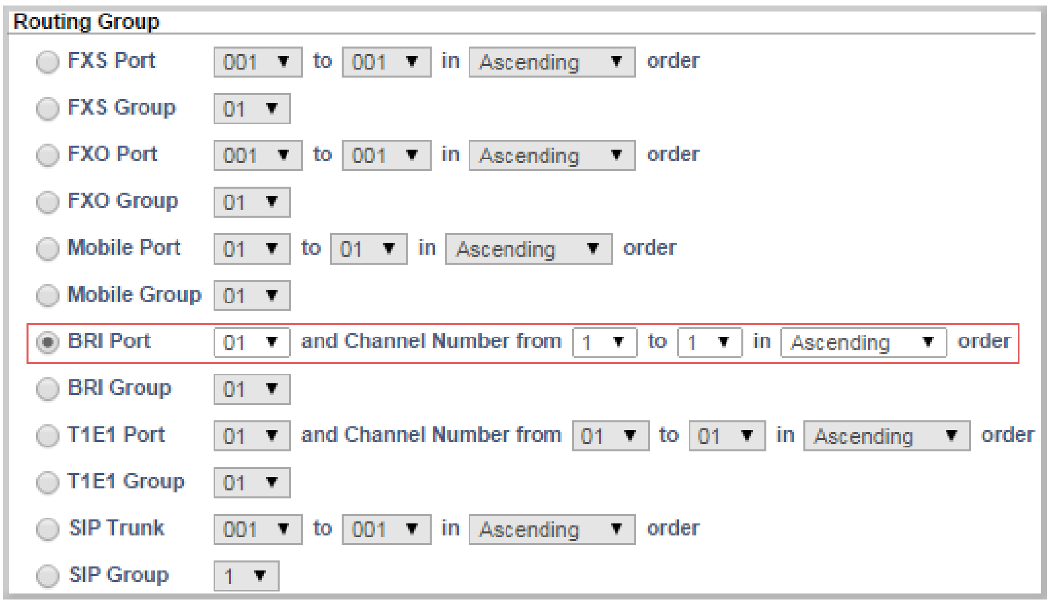
- The **FXS Port - Groups** window opens.



  - Create the FXS Group. For detailed instructions on creating groups, see the topic “Group” under *Advanced Settings*.

- Similarly, you can create a group of *sequential* and *not-sequential* FXO Ports, Mobile Ports and SIP Trunks.

- To create a routing group of *sequential* **BRI Channels** as members,



  - Select the **BRI Port** Number. Default: 1.

- In Channel Number **From - to**, select the **Start Channel Number** and the **End Channel Number**, respectively.

- In **in - order**, select the order in which the system should hunt for a free member Channel to route the call.
  Select **Ascending** to start hunting from the first to the last member channel. Select **Descending** to start hunting from the last to the first member channel. Default: Ascending.

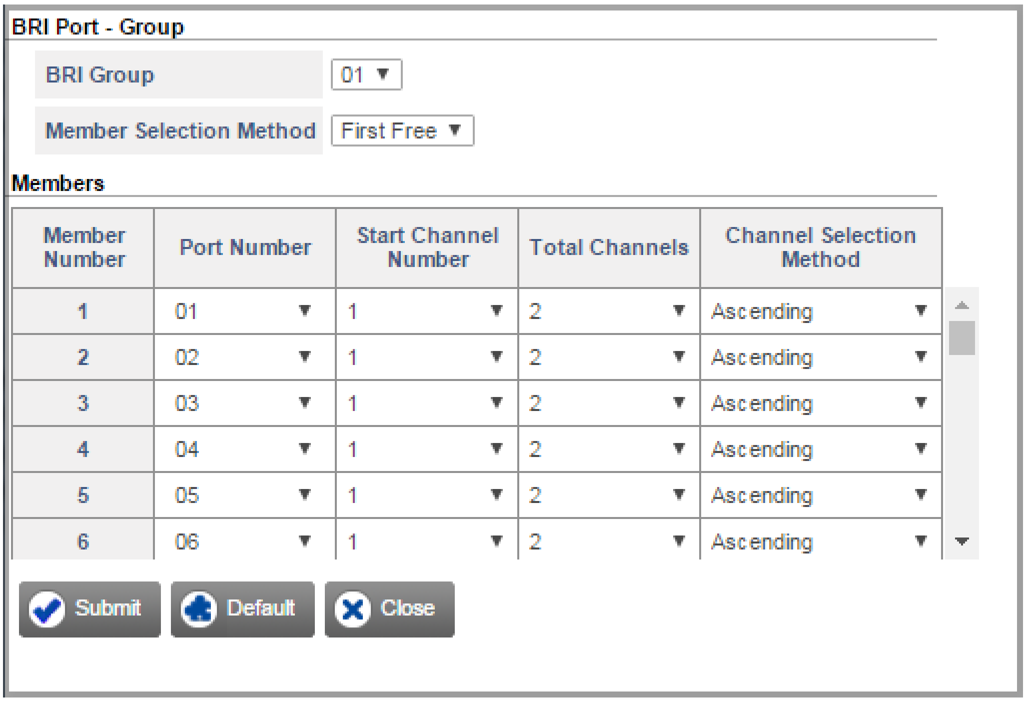
- To create a group of *not-sequential* **BRI Channels** as members,

  - Select **BRI Group**.

  - Select a **BRI Group** number. Default:1.

  - Click **Settings** .

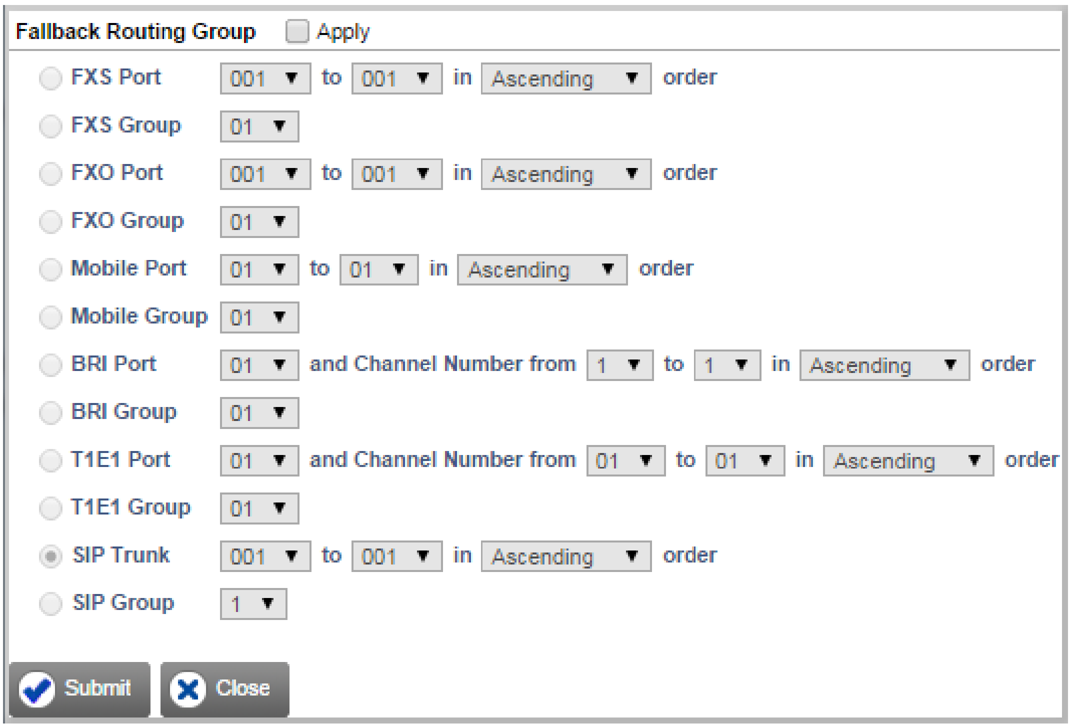
- The **BRI Port - Groups** window opens.



- Create the BRI Group. For detailed instructions on creating groups, see the topic "Group" under *Advanced Settings.*

- Similarly, you can create a group of *sequential* and *not-sequential* T1E1 Ports.

- You may create the **Fallback Routing Group**.



- To do this,
  - Select the **Apply** check box.
  - Follow the same instructions provided earlier for creating *sequential* and *not-sequential* groups of FXS Ports, FXO Ports, Mobile Ports, BRI Ports, T1E1 Ports and SIP Trunks.
- Click **Submit** to save changes. The **Edit** window closes.
- The entry you edited appears in the **Destination Port/Group for BRI Port** window.
- Close the **Destination Port/Group for BRI Port** window to return to the Handling of Calls window.

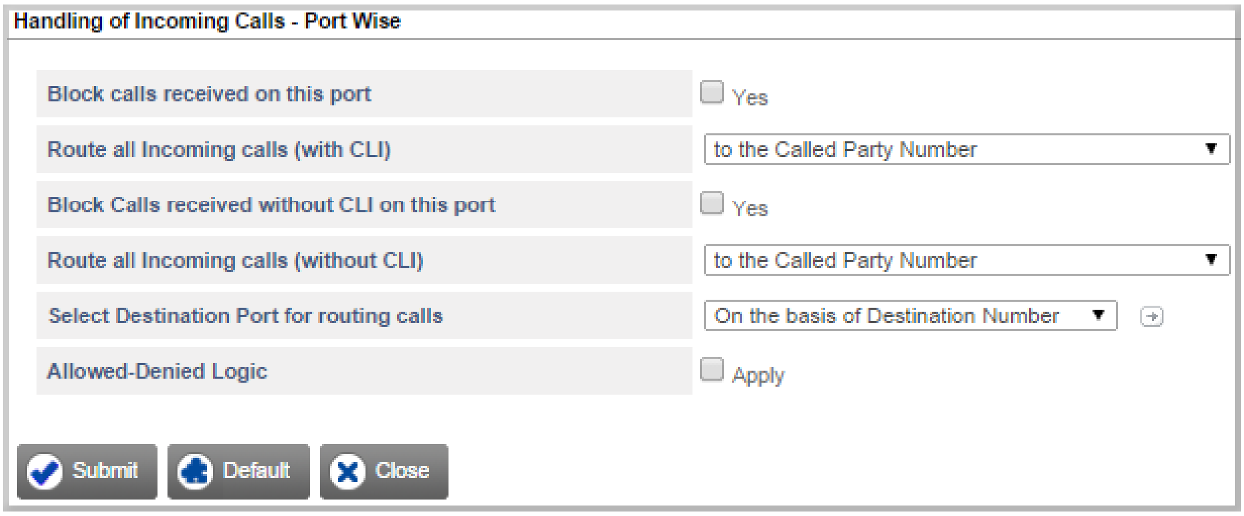
On the basis of Destination Number

In this method, incoming calls on the source port are routed to the destination port on the basis of the destination number (called party number) dialed by the caller.

You must configure the called party numbers in the **Destination Number Based** Table. SARVAM UMG will match the called party number dialed by the caller with the entries of this table. If a match is found for the number in the table, the call is routed to the destination.

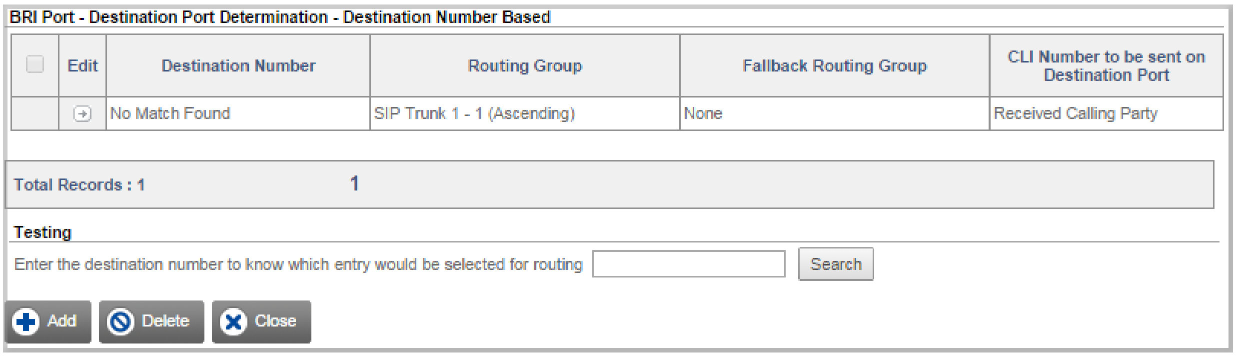
To apply this method, do the following:

- In **Select Destination Port for routing calls**, select **On the basis of Destination Number** option.

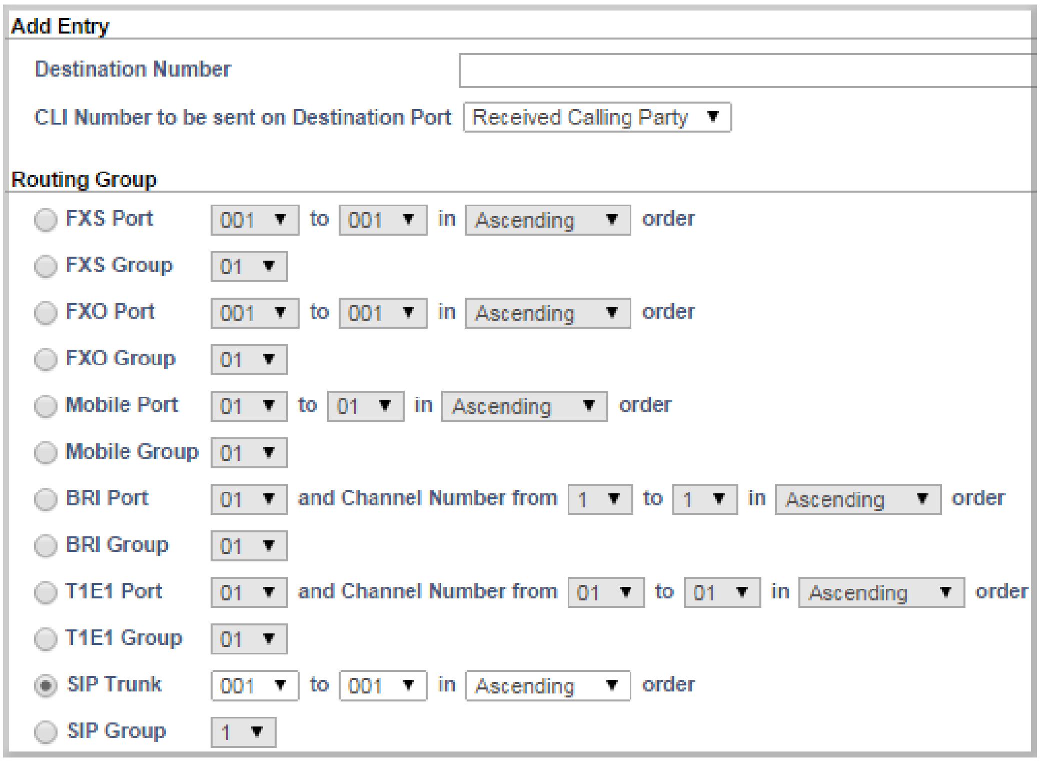


- Click **Settings** .

The **BRI Port - Destination Port Determination - Destination Number Based** table window opens.



- To add a new entry, click **Add**. The **Add Entry** window opens. You can add upto 1000 entries.



- In **Destination Number**, enter the number you expect the callers to dial. You may enter upto 64 characters (Digits + "Wildcard Characters") in this field. Valid characters are 0 to 9, *, #, X, T, Comma [,], Hyphen [-], Caret [^]. Default: Blank.

Wildcard Characters

SARVAM UMG 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.

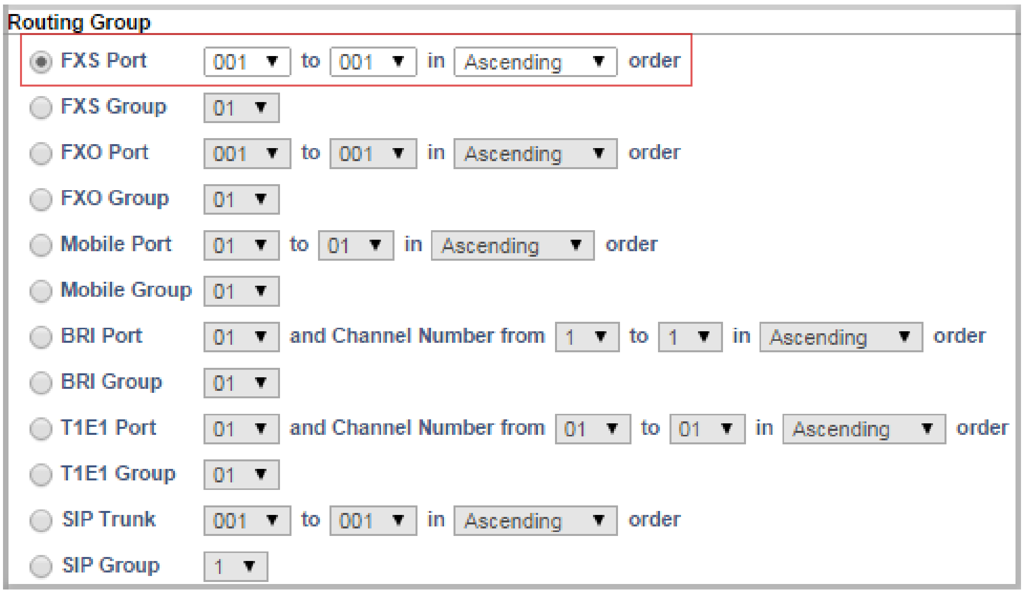
+	+ (plus) can be configured as a first character of the Destination Number string in the *SIP Trunk-Destination Port Determination-Destination Number Based* table only.
[-]	Hyphen within the bracket, defines a range. Only digits 0-9 are allowed within a bracket.
[,]	Comma within a bracket is used as 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.

- Select the **CLI Number to be sent on Destination Port**. You can select Received Calling Party or Received Called Party. Default: Received Calling Party.

CLI to be sent on Destination Port is applicable when FXS Port/Group is determined as the Destination Port.

- Create the **Routing Group**.

  - To create a group of *sequential* **FXS Ports** as members,



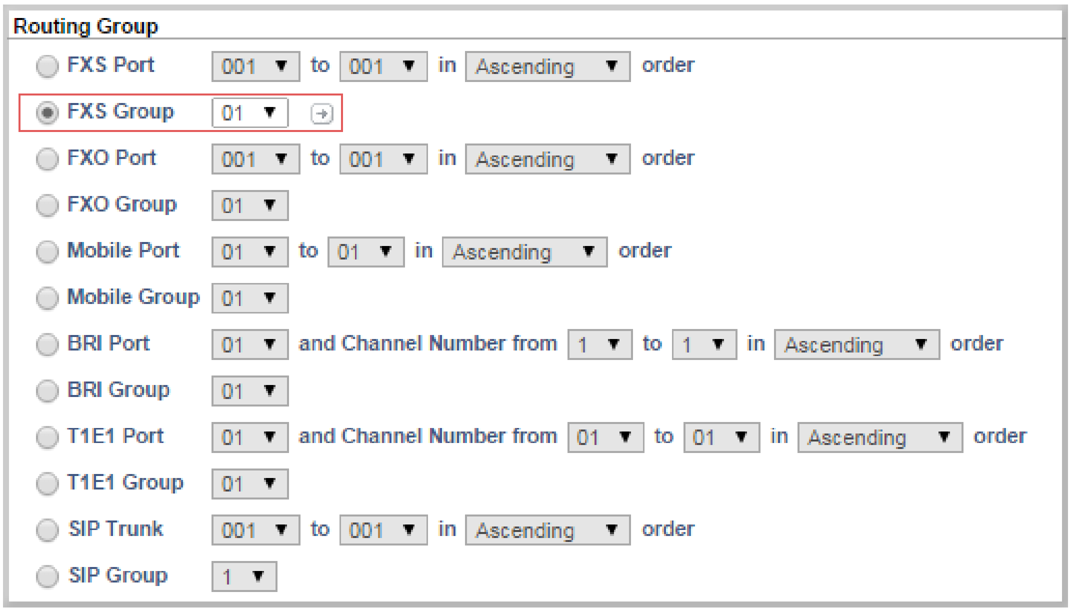
    - Select the desired **FXS Port** numbers as members. Default:1.

    - In **in - order**, select the order in which the system should hunt for a free member FXS Port to route the call.

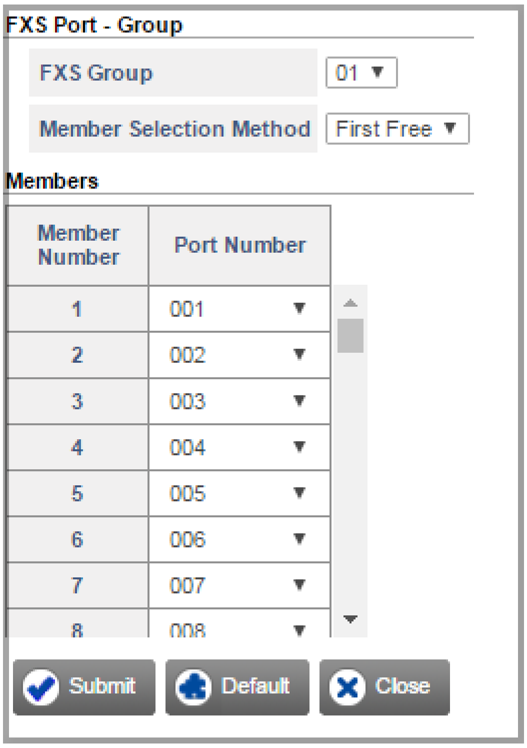
      Select **Ascending** to start hunting from the first to the last member FXS Port. Select **Descending** to start hunting from the last to the first member FXS Port. Default: Ascending.

  - To create a group of *not-sequential* **FXS Ports** as members,

- Select a **FXS Group**.



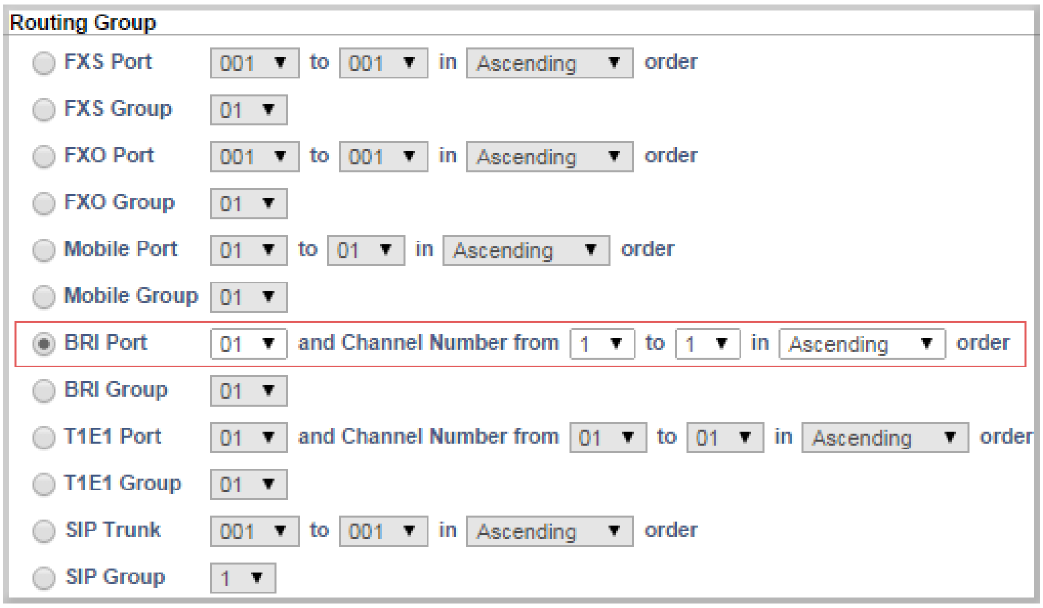
- Select **FXS Group** number. Default:1.
- Click **Settings** .
- The **FXS Port - Groups** window opens.



- Create the FXS Group. For detailed instructions on creating groups, see the topic “Group” under *Advanced Settings*.

- Similarly, you can create a group of *sequential* and *not-sequential* FXO Ports, Mobile Ports and SIP Trunks.

- To create a routing group of *sequential* **BRI Channels** as members,



  - Select the **BRI Port** Number. Default: 1.

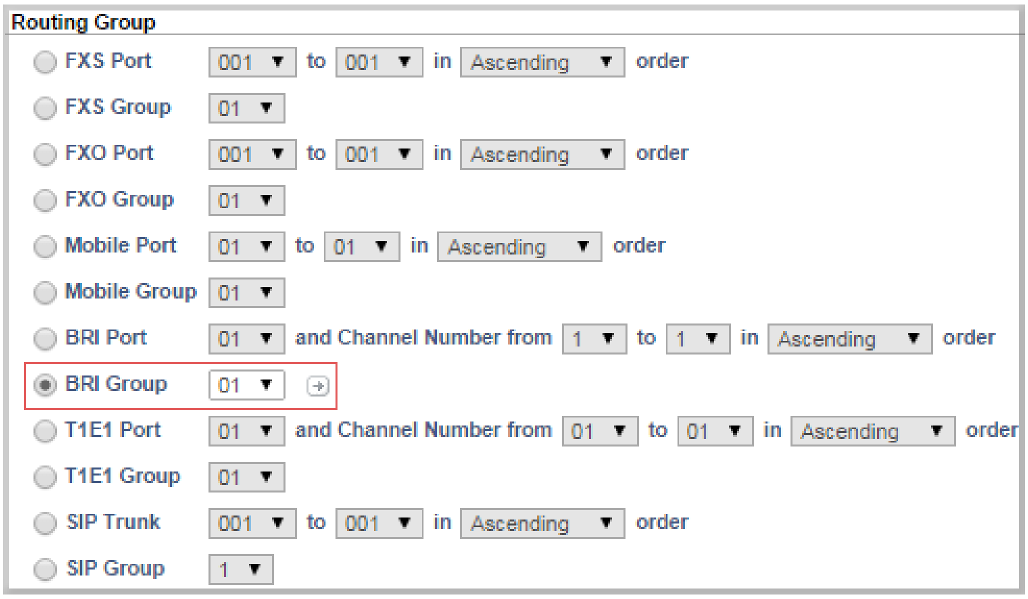
  - In Channel Number **From - to**, select the **Start Channel Number** and the **End Channel Number**, respectively.

  - In **in - order**, select the order in which the system should hunt for a free member Channel to route the call.

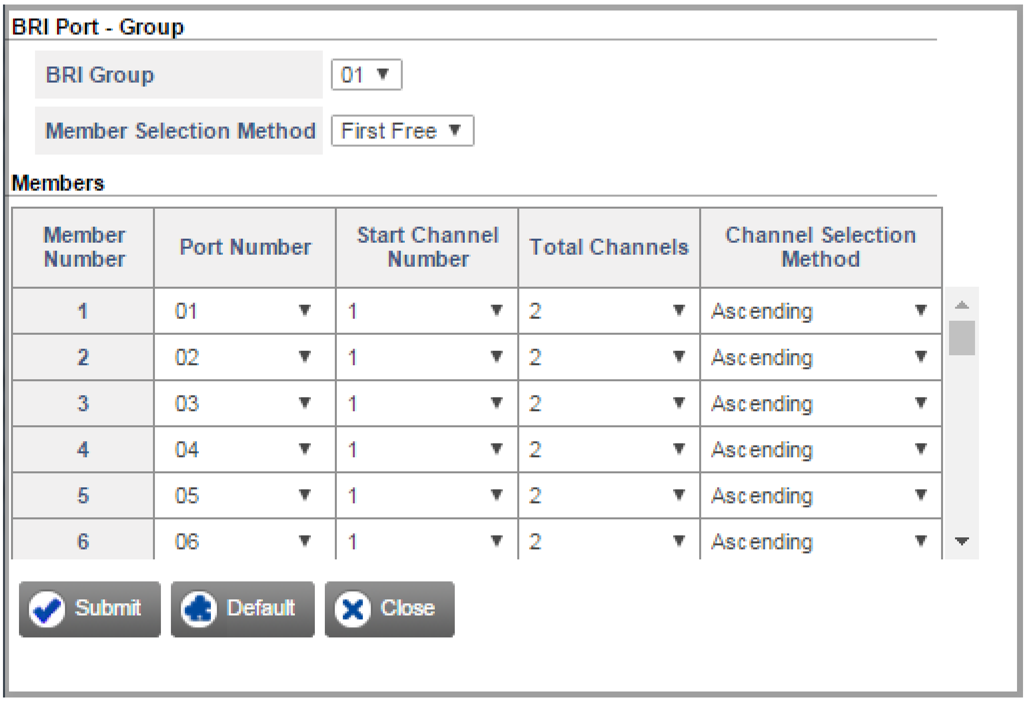
    Select **Ascending** to start hunting from the first to the last member channel. Select **Descending** to start hunting from the last to the first member channel. Default: Ascending.

- To create a group of *not-sequential* **BRI Channels** as members,

- Select **BRI Group**.



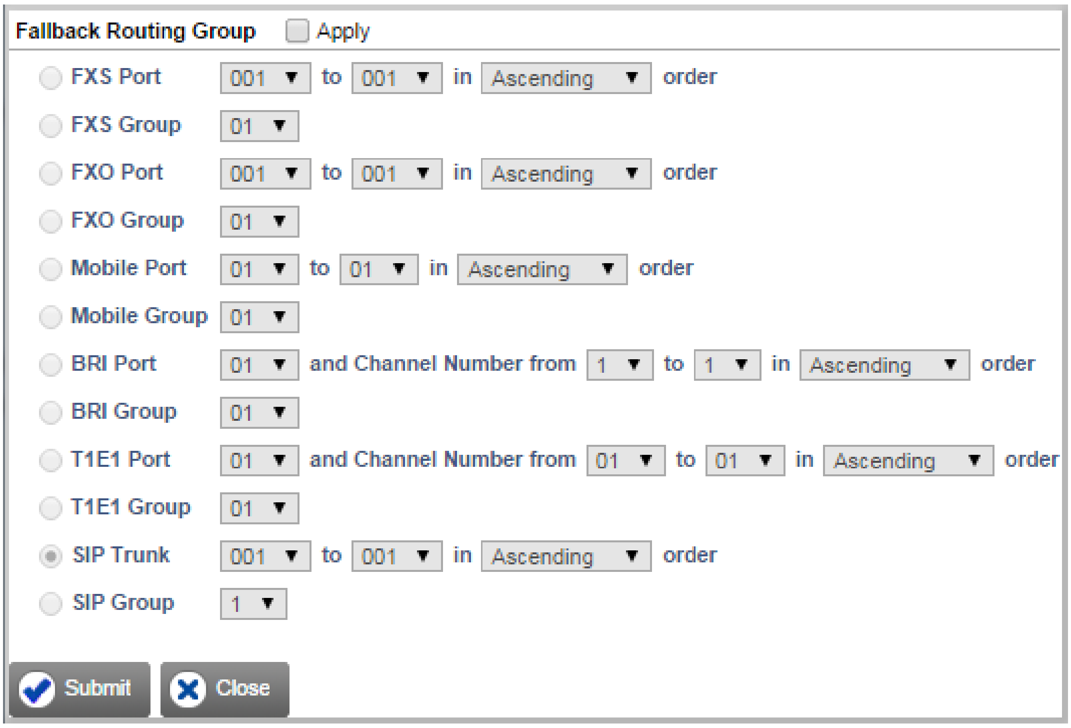
- Select a **BRI Group** number. Default:1.
- Click **Settings** .
- The **BRI Port - Groups** window opens.



- Create the BRI Group. For detailed instructions on creating groups, see the topic "Group" under *Advanced Settings.*

- Similarly, you can create a group of *sequential* and *not-sequential* T1E1 Ports.

- You may create the **Fallback Routing Group**.



- To do this,
  - Select the **Apply** check box.
  - Follow the same instructions provided earlier for creating *sequential* and *not-sequential* groups of FXS Ports, FXO Ports, Mobile Ports, BRI Ports, T1E1 Ports and SIP Trunks.
- Click **Submit** to save changes. The **Add Entry** window closes.
- The entry you added appears in the **BRI Port - Destination Port Determination - Destination Number Based** table.
- Follow the same steps as above to add another entry to this table.
- To delete an entry, select the check box and click the **Delete** button.

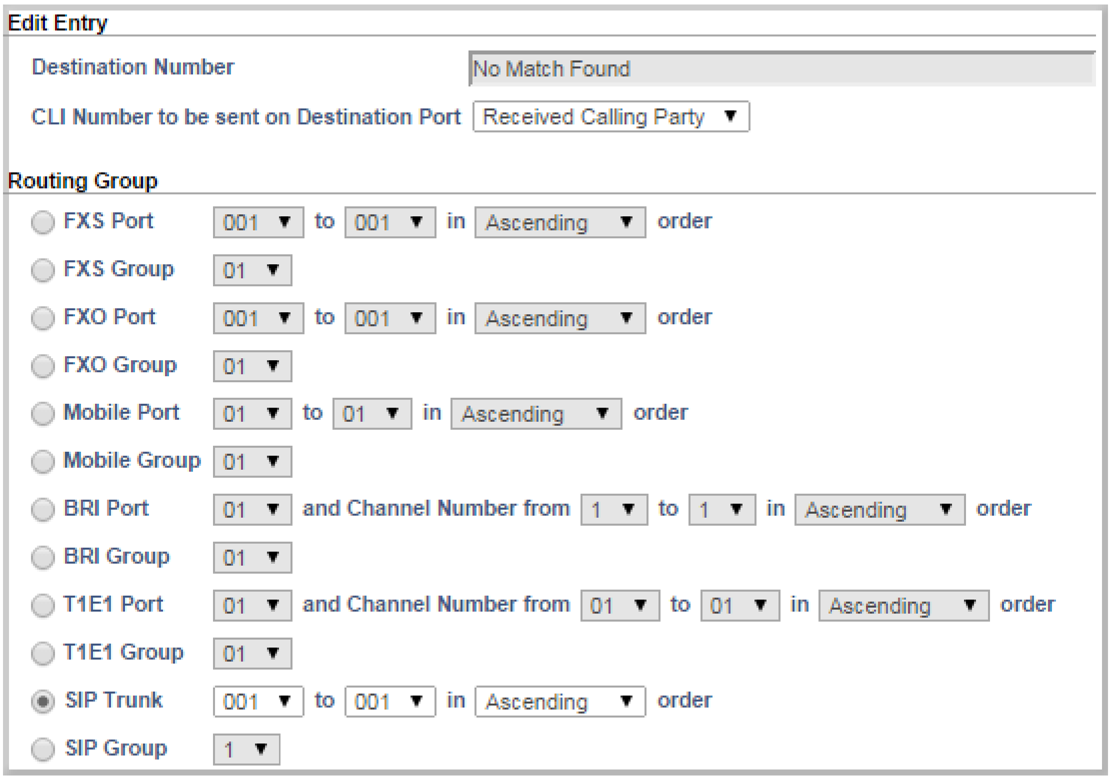
*If there are multiple entries in the Destination Number Based table, to search a particular entry in the table, under Testing enter the desired number in the **Enter the destination number to know which entry would be selected for routing** search box.*

- By default, SIP Trunk 1-1 (Ascending) is assigned as the Routing Group, for routing calls from numbers that do not match with any of the destination numbers you configured (No Match Found).

  To change the default Routing Group and to create the Fallback Routing Group for the No Match Found numbers entry,

  - For the No Match Found entry in the table, click **Edit** .

- The **Edit Entry** window opens.



- Create the **Routing Group** and **Fallback Routing Group** as per your requirement.

- Click **Submit** and close the window.

- Close the window if you have finished adding/editing entries.

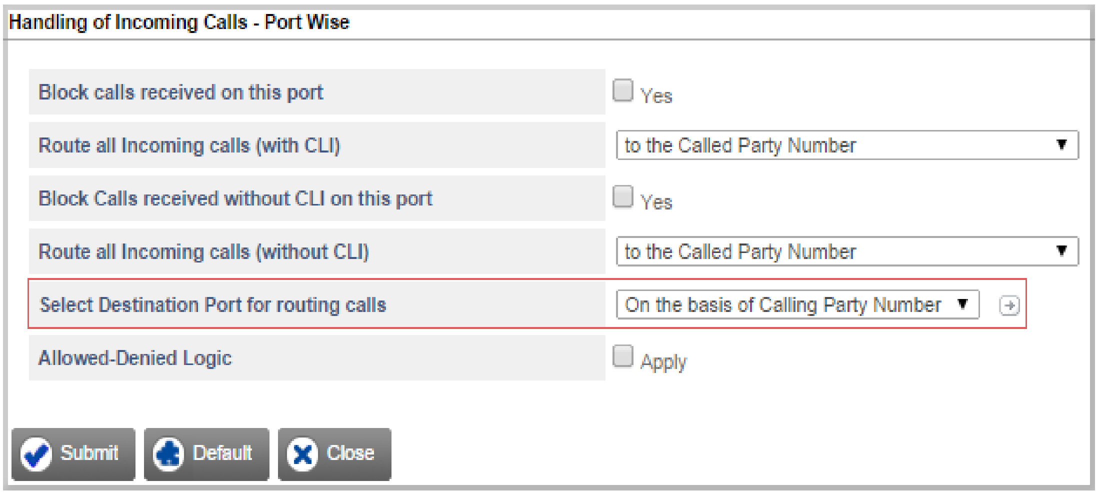
  You can also configure the **Destination Number Based** Table from *Advanced Settings*. For instructions, see "Destination Port Determination" under *Advanced Settings*.

On the basis of Calling Party Number

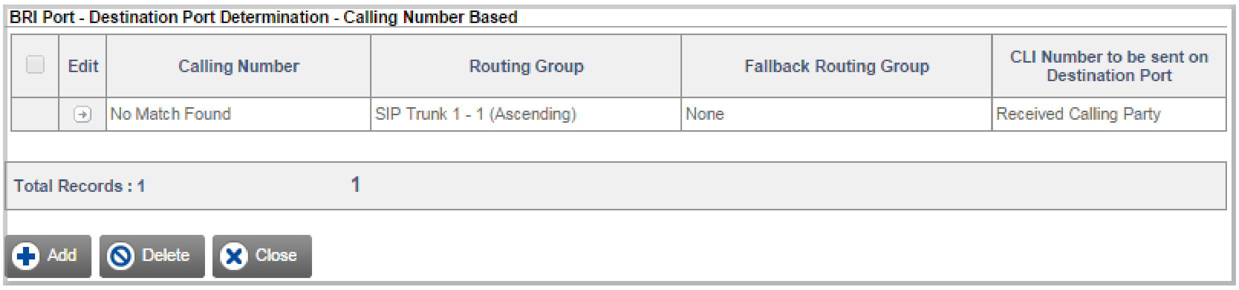
In this method, incoming calls on the BRI Port are routed to a specific port as per the calling party's number. To apply this method, do the following:

- In **Select Destination Port for routing calls**, select **On the basis of Calling Party Number** option.

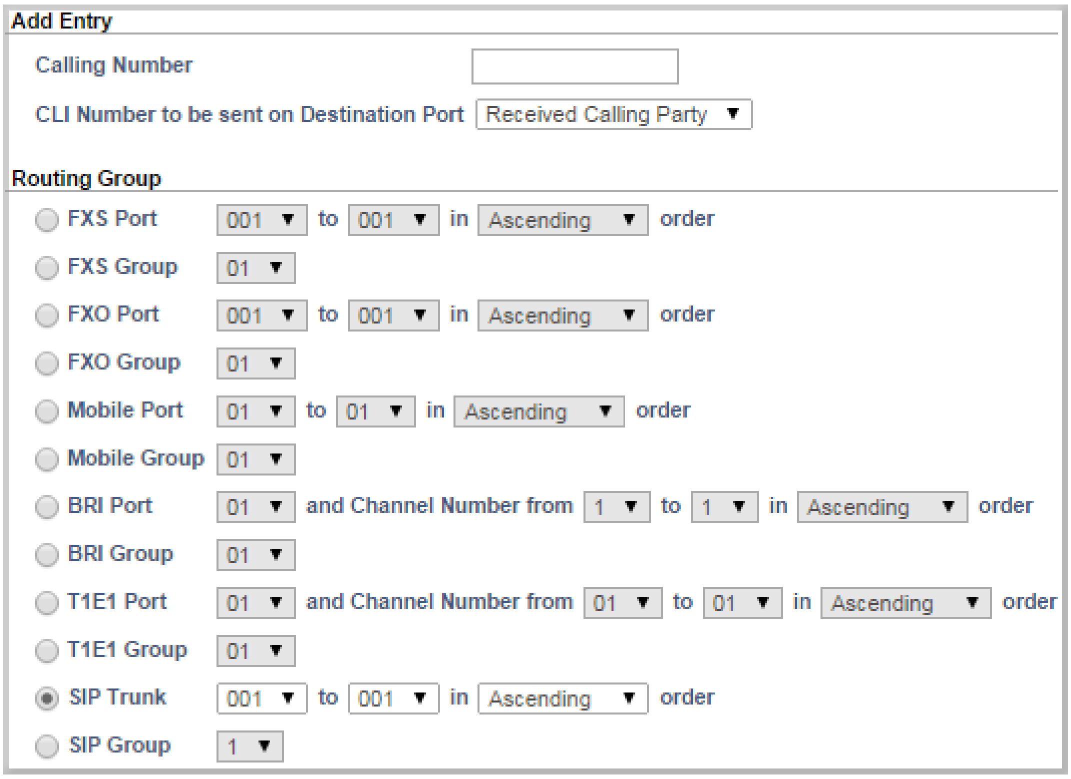
- Click **Settings** .



The **BRI Port - Destination Port Determination - Calling Number Based** table window opens.



- To add a new entry, click **Add**. The **Add Entry** window opens. You can add upto 499 entries.



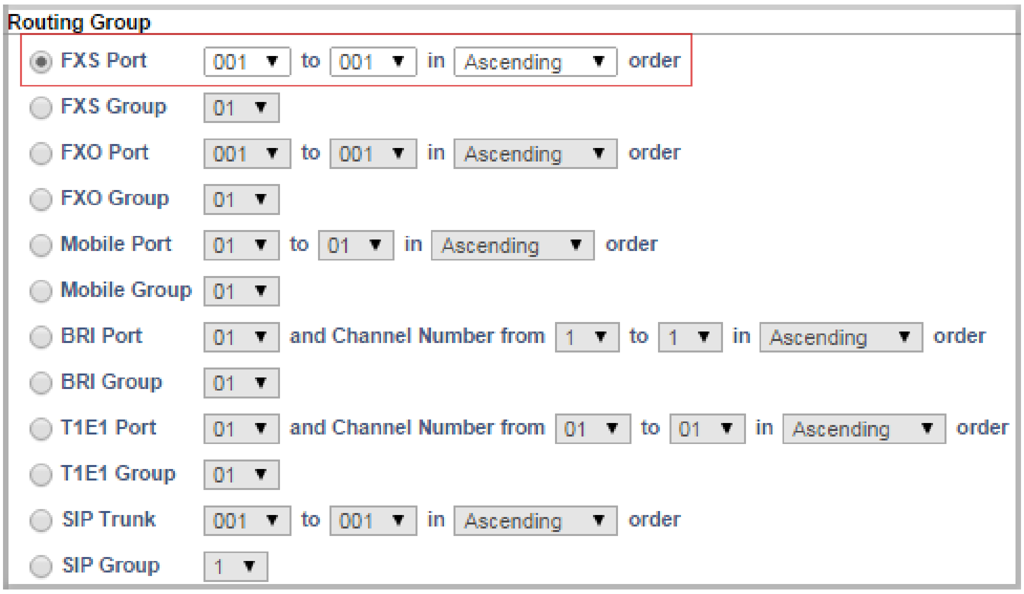
- In **Calling Number**, enter the number (max. 24 characters) from which you expect calls to be received. Valid digits are 0 to 9, *, #, (dot). Default: Blank.

- Select the **CLI Number to be sent on Destination Port**. You can select Received Calling Party or Received Called Party. Default: Received Calling Party.

CLI to be sent on Destination Port is applicable when FXS Port/Group is determined as the Destination Port.

- Create the **Routing Group**.

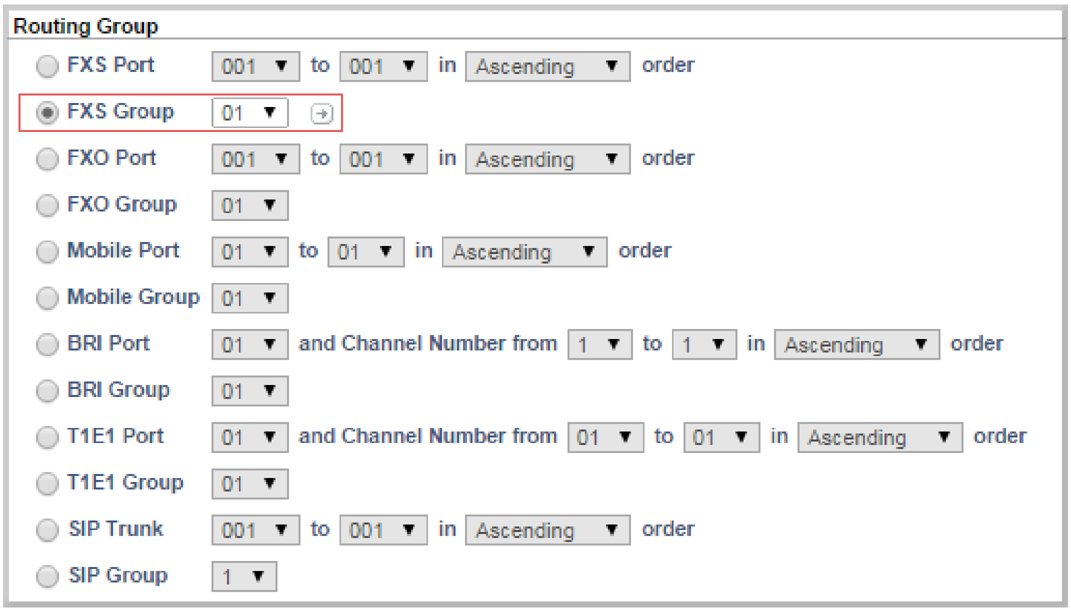
- To create a group of *sequential* **FXS Ports** as members,



  - Select the desired **FXS Port** numbers as members. Default:1.
  - In **in - order**, select the order in which the system should hunt for a free member FXS Port to route the call.

    Select **Ascending** to start hunting from the first to the last member FXS Port. Select **Descending** to start hunting from the last to the first member FXS Port. Default: Ascending.

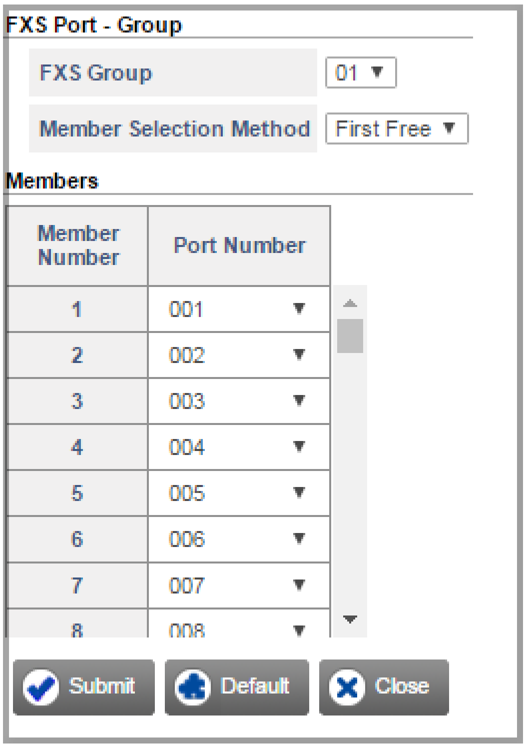
- To create a group of *not-sequential* **FXS Ports** as members,
  - Select a **FXS Group**.



  - Select **FXS Group** number. Default: 1.

- Click **Settings** .

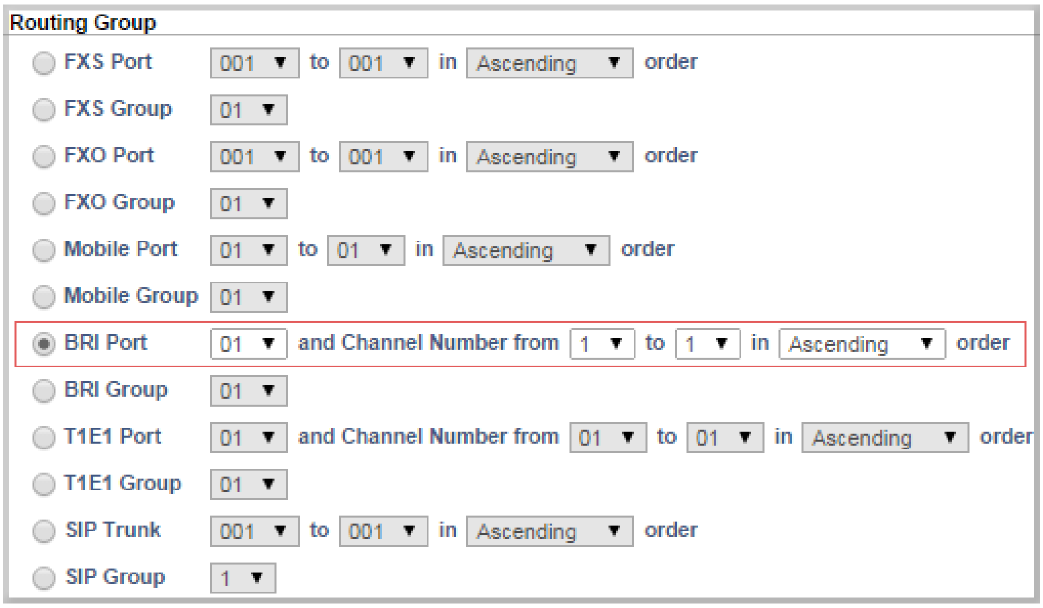
- The **FXS Port - Groups** window opens.



- Create the FXS Group. For detailed instructions on creating groups, see the topic “Group” under *Advanced Settings*.

- Similarly, you can create a group of *sequential* and *not-sequential* FXO Ports, Mobile Ports and SIP Trunks.

- To create a routing group of *sequential* **BRI Channels** as members,

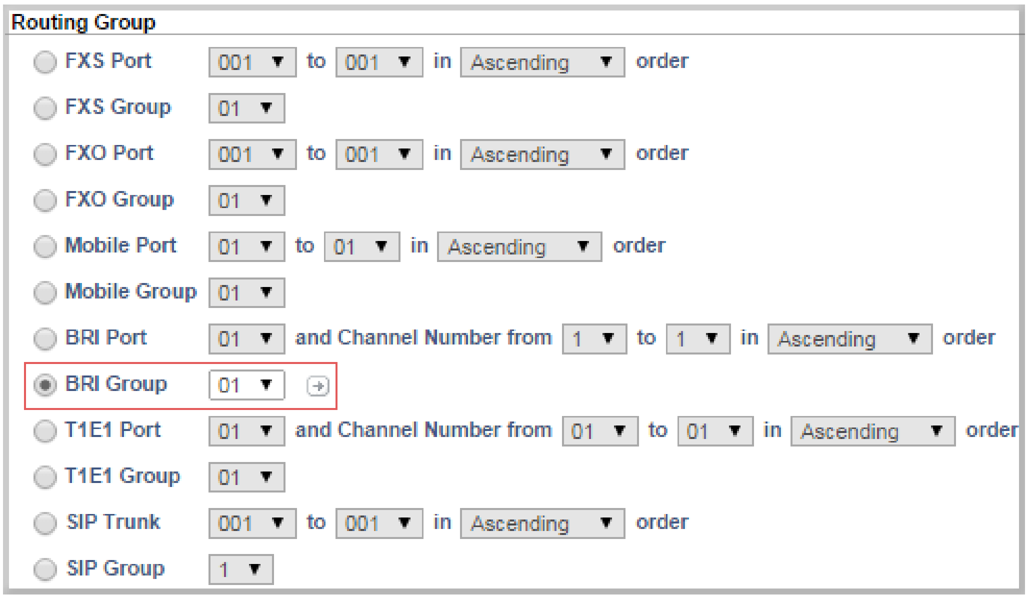


  - Select the **BRI Port** Number. Default: 1.
  - In Channel Number **From - to**, select the **Start Channel Number** and the **End Channel Number**, respectively.
  - In **in - order**, select the order in which the system should hunt for a free member Channel to route the call.

    Select **Ascending** to start hunting from the first to the last member channel. Select **Descending** to start hunting from the last to the first member channel. Default: Ascending.

- To create a group of *not-sequential* **BRI Channels** as members,

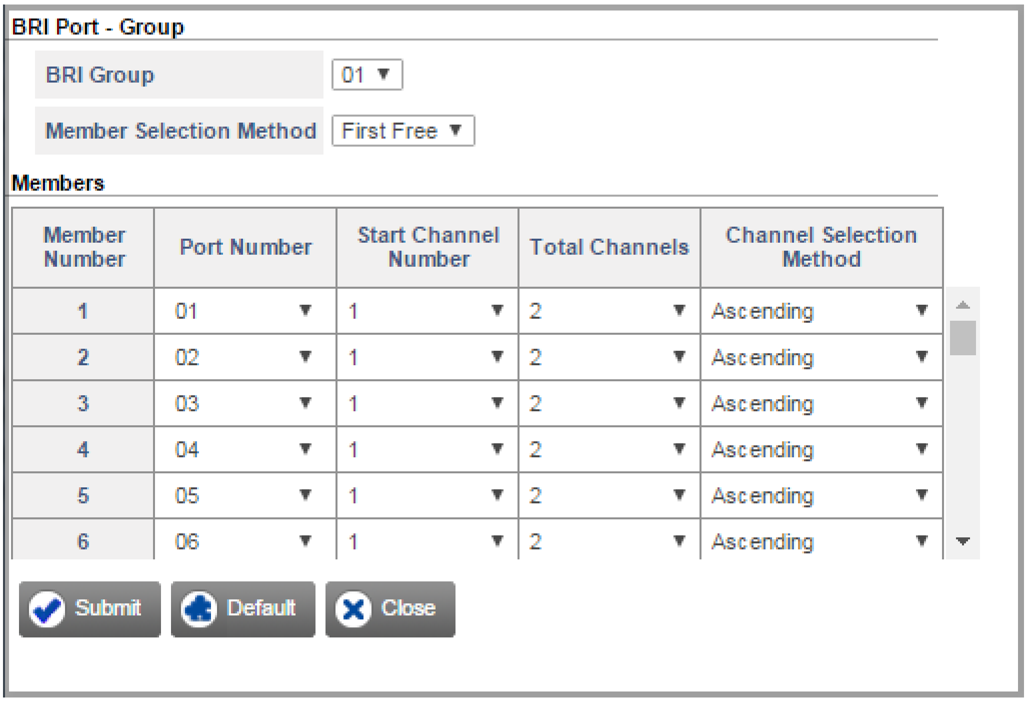
- Select **BRI Group**.



- Select a **BRI Group** number. Default:1.

- Click **Settings** .

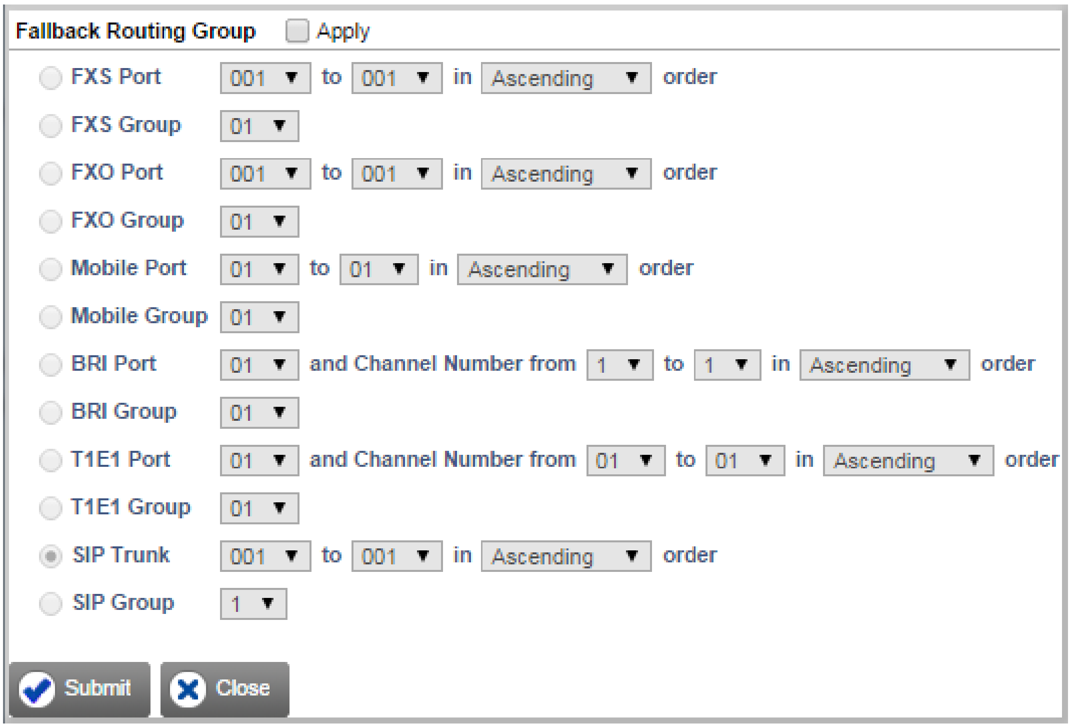
- The **BRI Port - Groups** window opens.



- Create the BRI Group. For detailed instructions on creating groups, see the topic "Group" under *Advanced Settings.*

- Similarly, you can create a group of *sequential* and *not-sequential* T1E1 Ports.

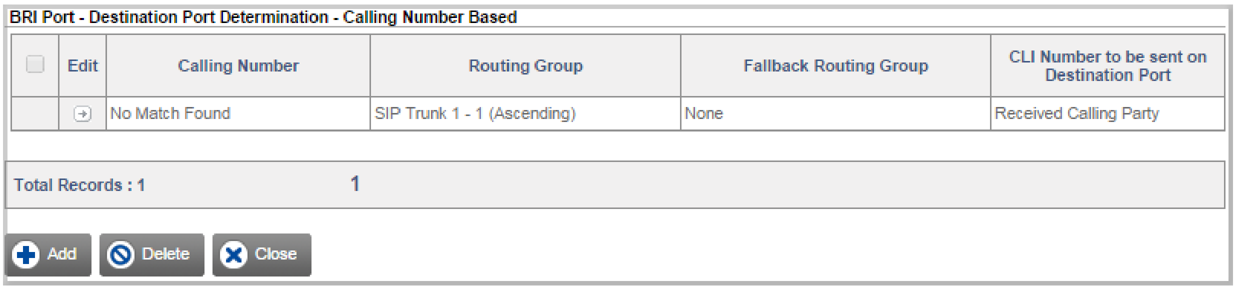
- You may create the **Fallback Routing Group**.



- To do this,
  - Select the **Apply** check box.
  - Follow the same instructions provided earlier for creating *sequential* and *not-sequential* groups of FXS Ports, FXO Ports, Mobile Ports, BRI Ports, T1E1 Ports and SIP Trunks.
- Click **Submit** to save changes. The **Add Entry** window closes.
- The entry you added appears in the **BRI Port - Destination Port Determination - Calling Number Based** table.
- Follow the same steps as above to add another entry to this table.
- To delete an entry, select the check box and click the **Delete** button.
- By default, SIP Trunk 1-1 (Ascending) is assigned as the Routing Group, for routing calls from numbers that do not match with any of the destination numbers you configured (No Match Found).

  To change the default Routing Group and to create the Fallback Routing Group for the No Match Found numbers entry,

- For the No Match Found entry in the table, click **Edit** .

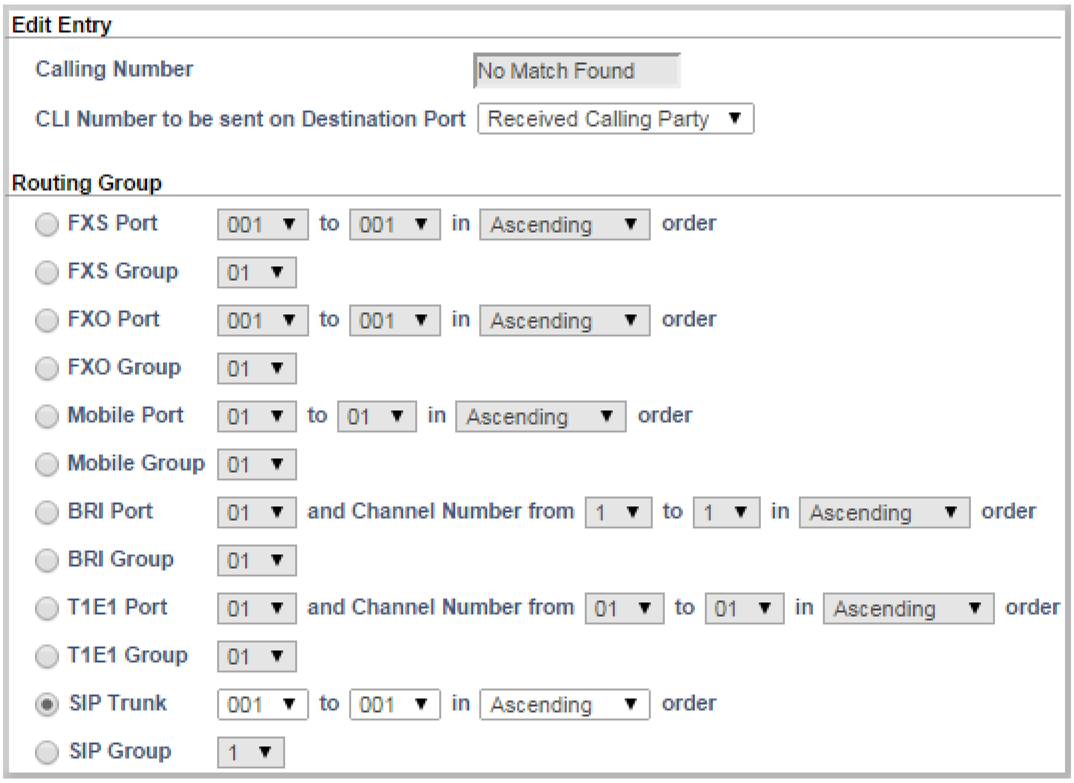


BRI Port - Destination Port Determination - Calling Number Based

	Edit	Calling Number	Routing Group	Fallback Routing Group	CLI Number to be sent on Destination Port
		No Match Found	SIP Trunk 1 - 1 (Ascending)	None	Received Calling Party

Total Records : 1 1

- The **Edit Entry** window opens.



- Create the **Routing Group** and **Fallback Routing Group** as per your requirement.

- Click **Submit** and close the window.

- Close the window if you have finished adding/editing entries.

  You can also configure the **Calling Number Based** Table from *Advanced Settings*. For instructions, see "Destination Port Determination" under *Advanced Settings.*

Allowed - Denied Logic

You can apply the Allowed-Denied logic on the BRI Port (source port) if you want to allow or restrict the dialing of particular numbers. You can use this feature for Toll Control.

The Allowed-Denied Number Logic makes use of two Number lists:

- **Allowed Numbers List:** This is the list of numbers that can be dialed out from the BRI Port.
- **Denied Numbers List:** This list contains the numbers that are to be restricted from being dialed out from the BRI Port.

When Allowed-Denied Logic is enabled on a source port, for each number dialed from the port, SARVAM UMG uses the best-match-found logic to compare the dialed number with the Allowed Number list and the Denied Number list.

The number is allowed to be dialed, if it:

- matches with both lists.
- matches with Allowed Number list, but not with the Denied Number list.
- matches with neither the Allowed List nor the Denied List.

The number is denied, if it matches with the Denied Number list, but not with the Allowed Number list.

The system does not apply the Allowed-Denied Logic:

- When dialed number string matches with any Access Code.
- When dialed number string matches with any Emergency Number.
- When any one of the following is selected to Route all Incoming Calls (with CLI):
    - on the basis of Calling Party Number
    - to a Fixed Destination Number
    - on the basis of DDI Number

To apply Allowed - Denied Logic on the BRI Port,

- Select the **Allowed - Denied Logic** check box.

- In the **Allowed Number List**, select the list number you have configured with numbers you want to allow to be dialed out from the BRI Port. Default: 01

    If you have not configured the Allowed Number List,

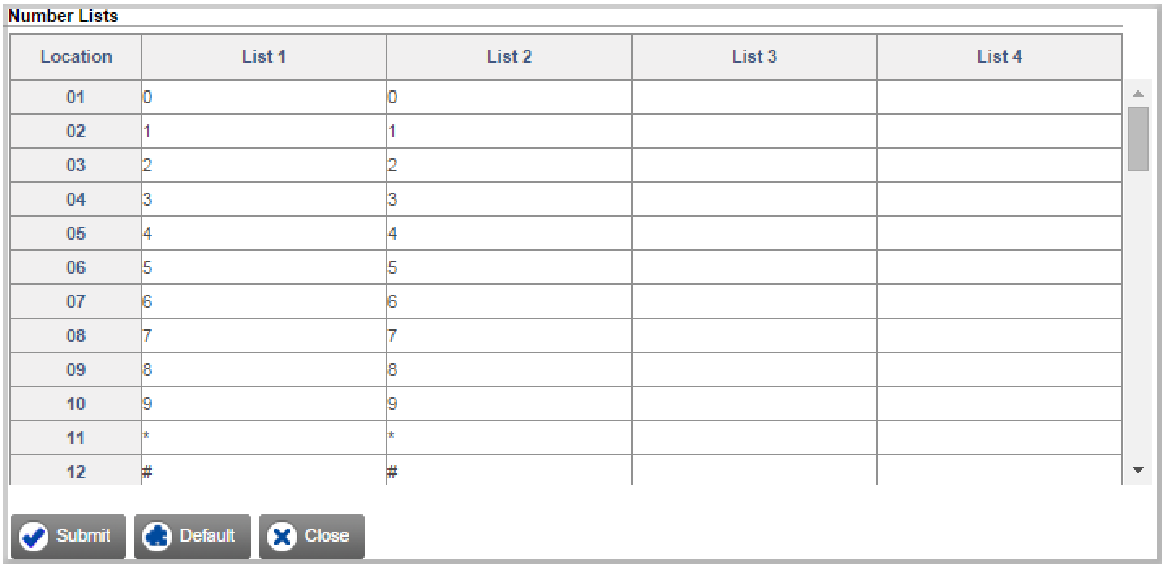
    - Click **Settings** . The Number Lists window opens.
    - You may configure the default Allowed Number List 1 or any other list. See “Number Lists” to configure the allowed numbers.
    - Click **Submit** to save the Allowed Number List and close the window.

- In the **Denied Number List**, select the list number you have configured with numbers you want to restrict to be dialed out from the BRI Port. Default: 02

If you have not configured the Denied Number List,

- Click **Settings** .

- The Number Lists window opens.



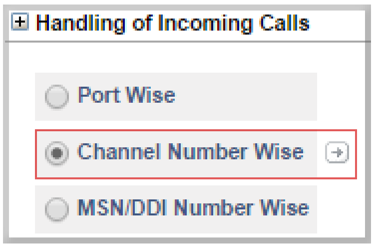
- You may configure the default Denied Number List 2 or any other list. See “Number Lists” to configure the restrict numbers.

- Click **Submit** to save the Denied Number List and close the window.

Channel Number Wise

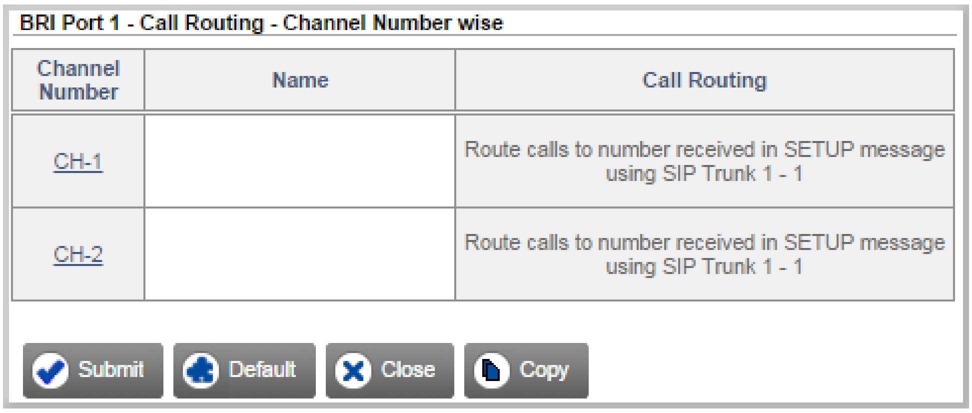
To configure Handling of Incoming Calls for each channel,

- Select the **Channel Number Wise** check box.

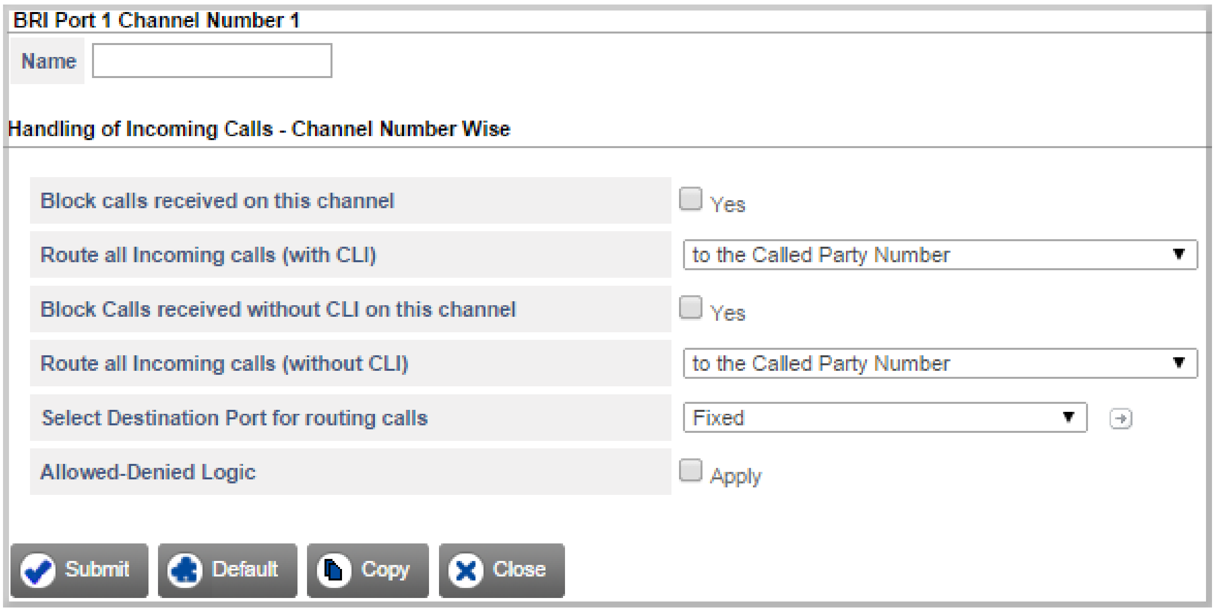


- Click **Settings** .

- The **BRI Port 1 - Call Routing - Channel Number Wise** window opens.



- Click the respective channel number — **CH-1** or **CH-2** — to configure the parameters**.**

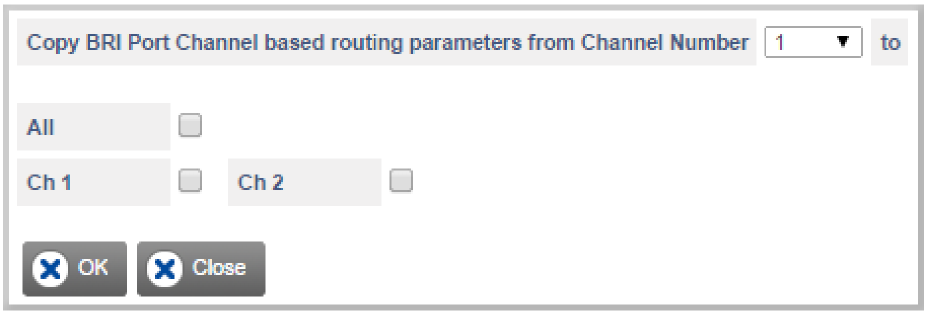


Configure the routing parameters for each channel.

- Block calls received on this channel.
- Route all Incoming Calls (with CLI), see “Handling of Incoming Calls”.
- Block Calls received without CLI on this channel.
- Route all Incoming Calls (without CLI), see “Handling of Incoming Calls”.
- Select the destination port for routing calls, see “Destination Port Determination”.
- Allowed-Denied Logic, see “Allowed - Denied Logic”.
- Handling of Outgoing Calls, see “Handling of Outgoing Calls”.

Copy Channel Based Routing Parameters

- You can also copy the settings of a BRI Channel to another BRI Channel using the **Copy** button. To do this,
    - Click the **Copy** button. The **Copy BRI Port Channel based routing parameters from Channel Number** window opens.

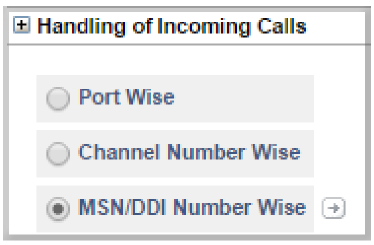


    - In the **Copy BRI Port Channel based routing parameters from Channel Number** box, select the number of the channel you want to copy settings *From*. Select the check boxes of the desired channel numbers you want to copy the settings *To*.
    - If you want to copy the settings *To* all the channels, select the **All** check box.
    - Click the **OK** button.
    - Once you have copied the settings, you can again edit the specific parameters of the BRI Channel you copied the settings to.
- Close the **BRI Port 1 Channel Number 1** window.
- To configure Channel 2, click **CH-2** on the **BRI Port 1 - Call Routing - Channel Number Wise** window and follow the same instructions as given above.
- Close the **BRI Port 1 - Call Routing - Channel Number Wise** window.

MSN/DDI Number Wise

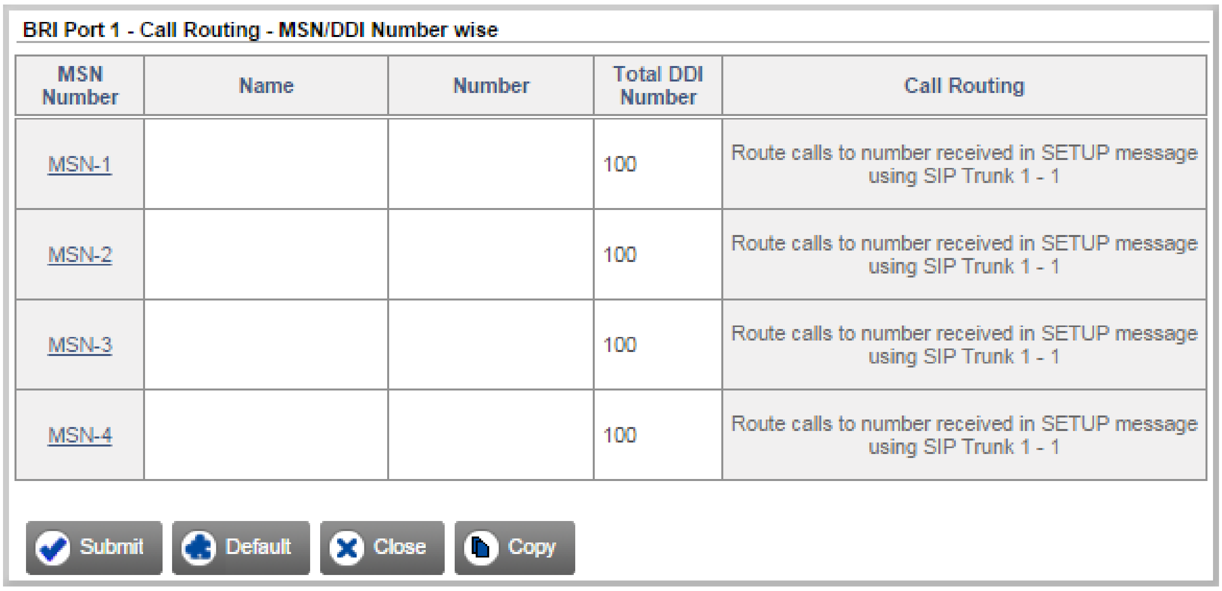
To configure Handling of Incoming Calls for each MSN Number,

- Select the **MSN/DDI Number Wise** check box.



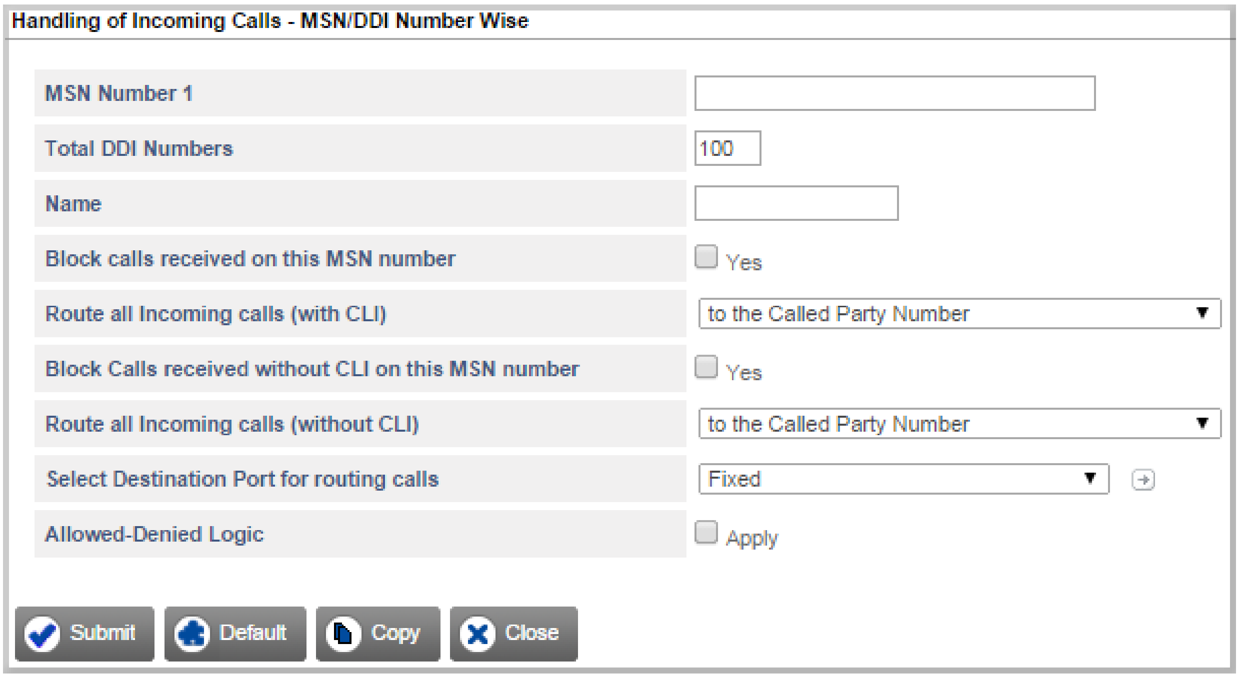
- Click **Settings** .

- The **BRI Port 1 - Call Routing - MSN/DDI Number Wise** window opens.


BRI Port 1 - Call Routing - MSN/DDI Number wise

MSN Number	Name	Number	Total DDI Number	Call Routing
MSN-1			100	Route calls to number received in SETUP message using SIP Trunk 1 - 1
MSN-2			100	Route calls to number received in SETUP message using SIP Trunk 1 - 1
MSN-3			100	Route calls to number received in SETUP message using SIP Trunk 1 - 1
MSN-4			100	Route calls to number received in SETUP message using SIP Trunk 1 - 1

- Click the respective MSN number — **MSN-1**, **MSN-2, MSN-3, MSN-4** — to configure the parameters**.**



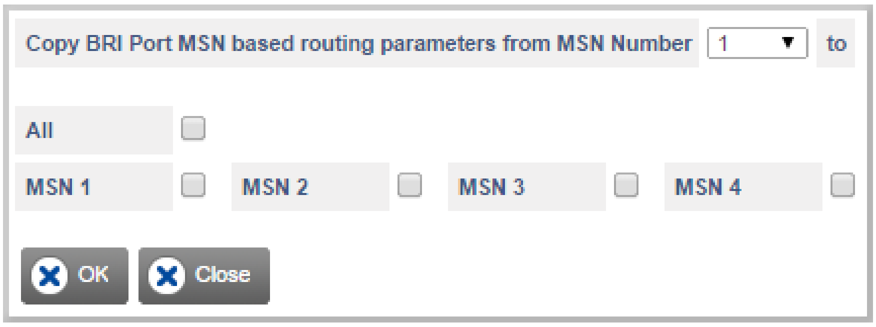
Configure the routing parameters for each MSN number:

- **MSN Number 1:** Enter the first **MSN Number** (max. 24 digits) provided by your Service Provider. Valid digits are 0-9, # and *. Default: Blank.
- **Total DDI Number:** Specify **Total DDI Numbers** provided by your Service Provider. Valid range is 1 to 9999. Default: 0100.
- **Name:** Assign a Name for identification.

- Block calls received on this MSN Number.
- Route all Incoming Calls (with CLI), see "Handling of Incoming Calls".
- Block Calls received without CLI on this MSN number.
- Route all Incoming Calls (without CLI), see "Handling of Incoming Calls".
- Select the destination port for routing calls, see "Destination Port Determination".
- Allowed-Denied Logic, see "Allowed - Denied Logic".
- Handling of Outgoing Calls, see "Handling of Outgoing Calls".

Copy MSN Based Routing Parameters

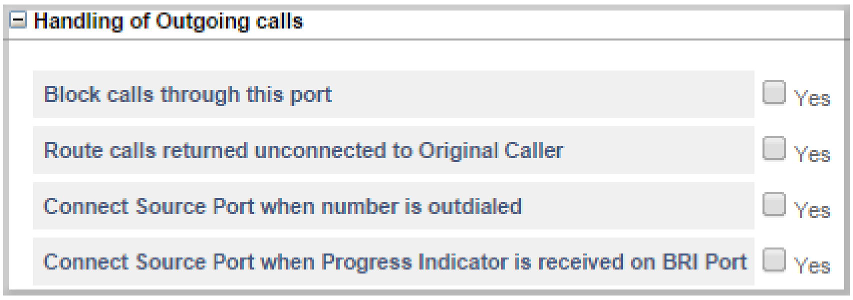
- You can also copy the settings of a MSN Number to another MSN Number using the **Copy** button. To do this,
  - Click the **Copy** button. The **Copy BRI Port MSN based routing parameters from MSN Number** window opens.



  - In the **Copy BRI Port MSN based routing parameters from MSN Number** box, select the MSN Number you want to copy settings *From*. Select the check boxes of the desired MSN Numbers you want to copy the settings *To*.
  - If you want to copy the settings *To* all the MSN Numbers, select the **All** check box.
  - Click the **OK** button.
  - Once you have copied the settings, you can again edit the specific parameters of the MSN Number you copied the settings to.
- Close the **BRI Port 1 MSN Number 1** window.
- To configure MSN Number 2, click **MSN-2** on the **BRI Port 1 - Call Routing - MSN/DDI Number Wise** window and follow the same instructions as given above.
- Close the **BRI Port 1 - Call Routing - MSN/DDI Number Wise** window.

Handling of Outgoing Calls

Click **Handling of Outgoing Calls** to expand.



When BRI Port is determined as the destination port, numbers dialed from this port constitute outgoing calls.

- If you do not want to route outgoing calls though this BRI Port, select the **Block calls through this port** check box.

- Enable **Route calls returned unconnected to Original Caller**, if you want SARVAM UMG to route outgoing calls made from this port that return unconnected back to the original caller.

  If you enable this feature, when an outgoing call is made using this port, and the Called Party is found busy or does not respond, SARVAM UMG stores the number of the calling party, the number of the called party and this port (through which the outgoing call was made). A record of each such call is stored for the duration of the Unconnected Calls Record Delete Timer (configurable; default: 999 minutes).

  If the called party returns the call before the expiry of this Timer, SARVAM UMG checks whether *Apply RCOC only if the caller calls back on the same trunk from which the call was made* is enabled or not, and accordingly places the incoming call to the original calling party. To change the duration of this timer, delete records of such calls and enable/disable the *Apply RCOC only if the caller calls back on the same trunk from which the call was made* check box, see "System Parameters".

- To connect the Source Port with the Destination Port without waiting for the call on the Destination Port to mature, enable the **Connect Source Port when number is outdialed** check box. Default: Disabled.

  In all Destination Number Determination methods except *After Answering the Call and Collecting the Digits*, the Source Port gets connected to the Destination Port only after the call has matured, that is, the called party has answered the call. Until the call matures, the caller hears only Ring Back Tone played by the network.

  By connecting the Source Port with the Destination Port immediately after the number is dialed, the caller can know the state of the call; if the called party is busy, not responding, not reachable or is rejecting the call.

*If you enable **Connect Source Port when number is outdialed**, you will not be able to provide the features "Making a New Call using Access Code" and "Disconnecting a Call using Access Code" to users.*

- To connect the Source Port with the Destination Port without waiting for the call on the Destination Port to mature, enable the **Connect Source Port when number is outdialed** check box. Default: Disabled.

  In all Destination Number Determination methods except *After Answering the Call and Collecting the Digits*, the Source Port gets connected to the Destination Port only after the call has matured, that is, the called party has answered the call. Until the call matures, the caller hears only Ring Back Tone played by the network.

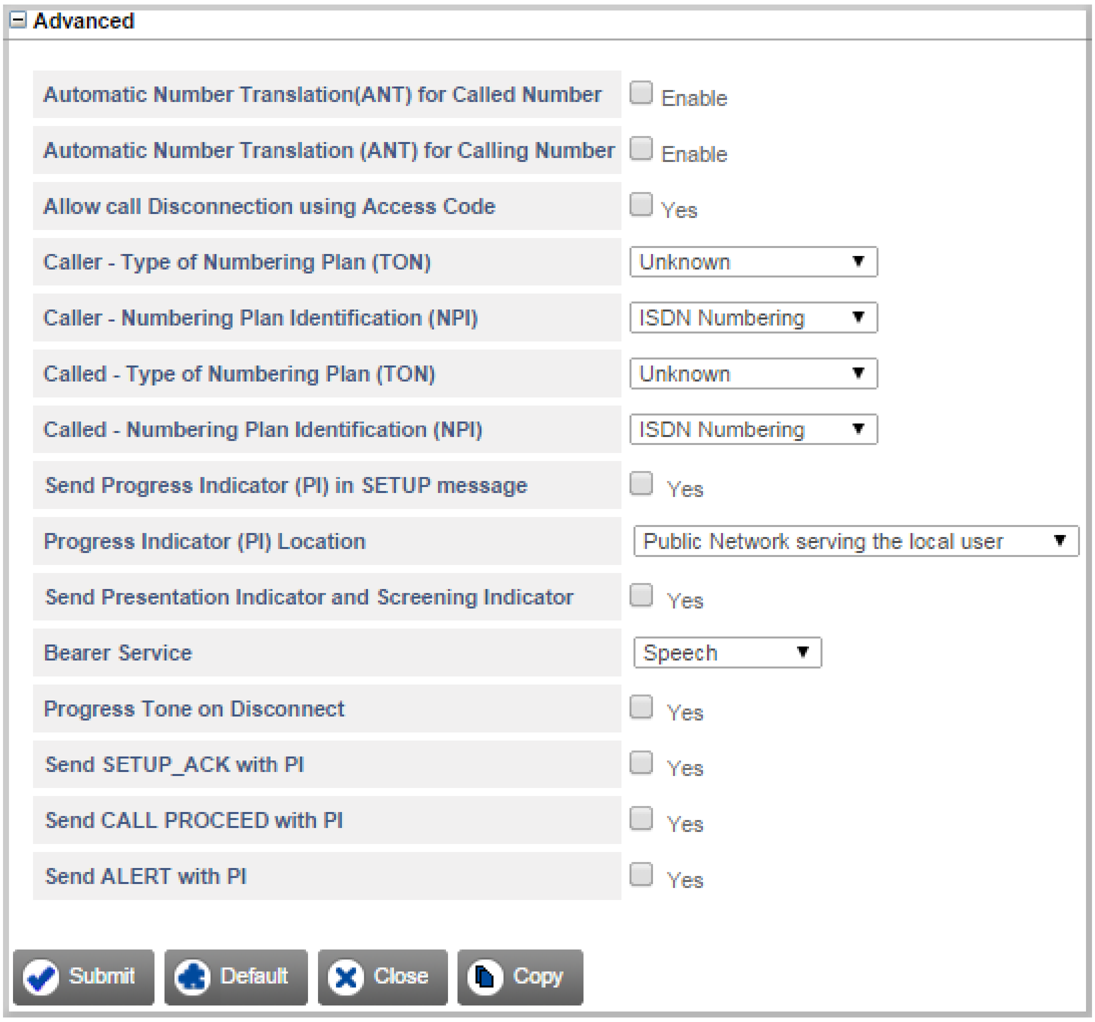
By connecting the Source Port with the Destination Port immediately after the number is dialed, the caller can know the state of the call; if the called party is busy, not responding, not reachable or is rejecting the call.

*If you enable **Connect Source Port when number is outdialed**, you will not be able to provide the features "Making a New Call using Access Code" and "Disconnecting a Call using Access Code" to users.*

- To connect to the Source port only when a Progress Indicator is received in either the call proceeding or an alerting message, enable the **Connect Source Port when Progress Indicator is received on BRI Port** check box. Default: Disabled.
- Click **Submit** to save.

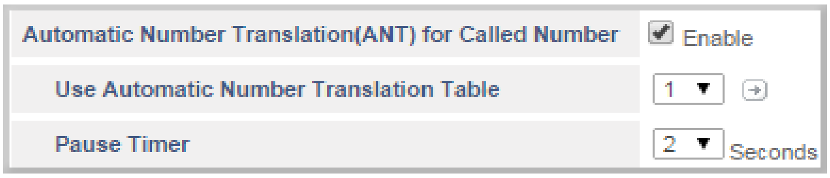
Advanced

- Click **Advanced** to expand.



- You can apply **Automatic Number Translation logic** on outgoing calls made from the BRI Port.

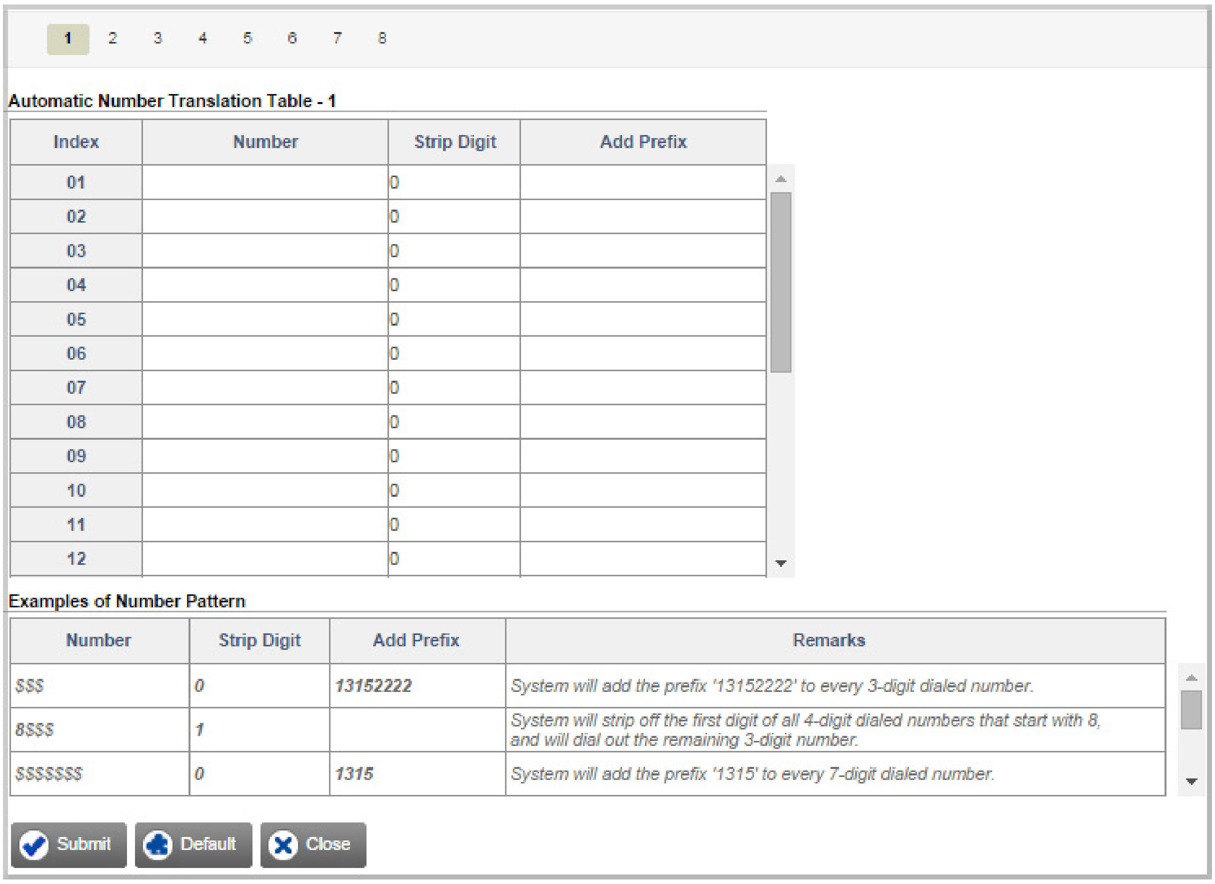
- To apply ANT logic on the Called Numbers, select the **Automatic Number Translation (ANT) for Called Number** check box. Default: Disabled.



- In **Use Automatic Number Translation Table**, select the ANT Table number you have configured for the Called Numbers. Default: Table 1.

  If you have not configured the Automatic Number Translation Table,

  - Click **Settings** .
  - The **Automatic Number Translation Table** window opens.



1 2 3 4 5 6 7 8

Automatic Number Translation Table - 1

Index	Number	Strip Digit	Add Prefix
01		0	
02		0	
03		0	
04		0	
05		0	
06		0	
07		0	
08		0	
09		0	
10		0	
11		0	
12		0	

Examples of Number Pattern

Number	Strip Digit	Add Prefix	Remarks
$$$	0	13152222	System will add the prefix '13152222' to every 3-digit dialed number.
8$$$	1		System will strip off the first digit of all 4-digit dialed numbers that start with 8, and will dial out the remaining 3-digit number.
$$$$$$$	0	1315	System will add the prefix '1315' to every 7-digit dialed number.

Submit Default Close

  - You may configure the default Automatic Number Translation Table or any other Table. See "Automatic Number Translation (ANT)" to configure the ANT Table.
  - Click **Submit** to save the ANT Table and close the window.
  - Return to ANT parameter and assign the ANT Table you configured.
  - Click **Submit**.

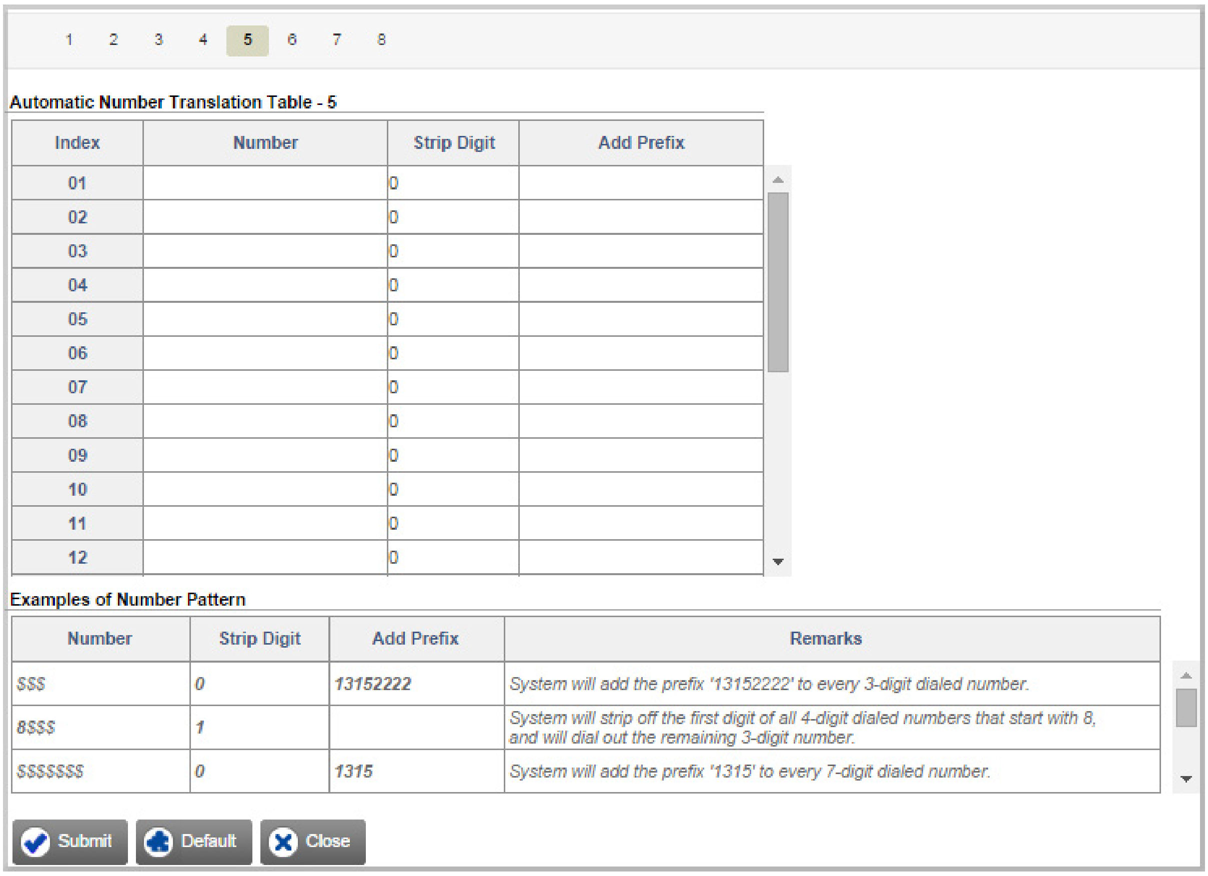
- Set the duration of the **Pause Timer**, if you have configured ^ (Pause) in the Add Prefix column of the ANT Table. Valid range is 1 to 9 seconds. Default: 2 seconds.

- To apply ANT logic on the Calling Numbers, select the **Automatic Number Translation (ANT) for Calling Number** check box. Default: Disabled.

- In **Use Automatic Number Translation Table**, select the ANT Table number you have configured for the Calling Numbers. Default: Table 5.

  If you have not configured the Automatic Number Translation Table,

  - Click **Settings**.
  - The **Automatic Number Translation Table** window opens.



Index	Number	Strip Digit	Add Prefix
01		0	
02		0	
03		0	
04		0	
05		0	
06		0	
07		0	
08		0	
09		0	
10		0	
11		0	
12		0	

Number	Strip Digit	Add Prefix	Remarks
$$$	0	13152222	System will add the prefix '13152222' to every 3-digit dialed number.
8$$$	1		System will strip off the first digit of all 4-digit dialed numbers that start with 8, and will dial out the remaining 3-digit number.
$$$$$$$	0	1315	System will add the prefix '1315' to every 7-digit dialed number.

  - You may configure the default **Automatic Number Translation Table - 5** or any other Table. See "Automatic Number Translation (ANT)" to configure the ANT Table.
  - Click **Submit** to save the ANT Table and close the window.
  - Return to ANT parameter and assign the ANT Table you configured.

- To enable the feature Disconnect Call using Access Code on the BRI Port, select the **Allow Call Disconnection using Access code** check box. To know more about this feature, see "Disconnecting a Call using Access Code".

- Select the required option for sending the **Caller- Type of Numbering Plan (TON) —** Unknown, International, National, Network Specific, Subscriber, Abbreviated or Reserved. Default: Unknown.

- Select the required option for sending the **Caller- Numbering Plan Identification (NPI)** — Unknown, ISDN Numbering, Data Numbering, Telex Numbering, National Numbering, Private or Reserved. Default: ISDN Numbering.

- Select the required option for sending the **Called-Type of Numbering Plan (TON)** — Unknown, International, National, Network Specific, Subscriber, Abbreviated or Reserved. Default: Unknown.

- Select the required option for sending the **Called-Numbering Plan Identification (NPI) —**Unknown, ISDN Numbering, Data Numbering, Telex Numbering, National Numbering, Private or Reserved. Default: ISDN Numbering.

- Select the **Send Progress Indicator (PI) in SETUP message** check box if you want the progress indicator value to be sent in SETUP message. Default: Disabled.

- Set the **Progress Indicator (PI) value in SETUP message** to the desired value. You can select — 1 or 3.

  Progress indicator 1 indicates that the call is not end-to-end ISDN and further call progress information may be available in-band.

  Progress indicator 3 indicates that the origination address is non-ISDN. This value will be included in the Setup Message to indicate whether the calling party is an ISDN device or not. Default: 1.

- Select the **Progress Indicator (PI) Location** for the SETUP message.The location is a progress indicator information element that indicates from where the message is coming. Default: Public Network serving the local user.

- Select the **Send Presentation Indicator and Screening Indicator** check box, if you want the system to display the presentation and screening information to the remote end. Default: Disabled.

  - Select the required **Presentation Indicator**. This allows remote end to know whether CLI Number should be displayed to user or not. You can select — Presentation Allowed, Presentation Restricted or Received from Source Port. Default: Received from Source Port.

  - Select the required **Screening Indicator**. This indicates whether the information is provided by the user or the network along with the screening details — not screened, verified and passed or verified and failed. Default: User-provided, not screened.

- Select the **Bearer Service** supported by your service provider. This will be sent in the SETUP Message. You can select — Speech, 3.1 KHz Audio. Default: Speech

- Select the **Progress Tone on Disconnect** check box, if you want the system to play the progress tone on the port when the call is released by the remote end or disconnected by the system. Default: Enabled.

- Select the **Send SETUP_ACK with PI** check box, if you want the system to send PI (Progress indicator) element in Setup Ack message. Default: Enabled.

- Select the **Send CALL PROCEED with PI** check box, if you want the system to send PI (Progress indicator) in Proceed message. Default: Enabled.

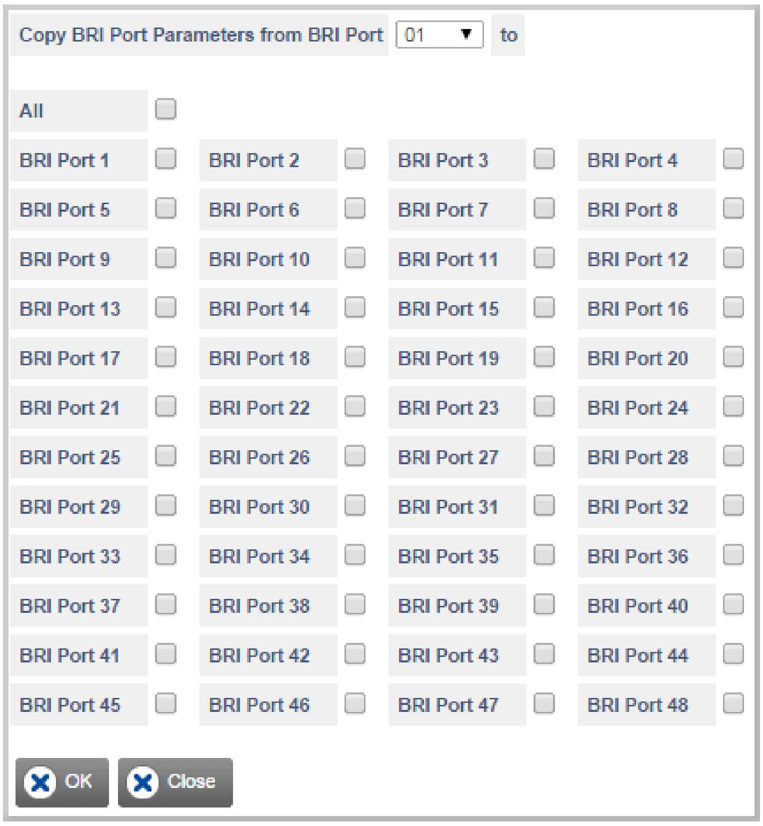
- Select the **Send ALERT with PI** check box, if you want the system to send PI (Progress indicator) in Alert message. Default: Enabled.

To configure the next BRI Port, click the BRI Port number tab and follow the same instructions as given above or you can copy the parameters from another port.

Copy BRI Port Parameters

- You can also copy the settings of a BRI Port to another BRI Port using the **Copy** button. To do this,

  - Click the **Copy** button. The **Copy BRI Port Parameters** window opens.



  - In the **Copy BRI Port Parameters from BRI Port** box, select the number of the port you want to copy settings *From*. Select the check box of the respective port numbers you want to copy the settings *To*.

  - If you want to copy the settings *To* all the ports, select the **All** check box.

  - Click the **OK** button.

  - Once you have copied the settings, you can again edit the specific parameters of the BRI Port you copied the settings to.

Mobile Port

The system supports a maximum of 48 Mobile ports. Before you begin configuring the Mobile ports, ensure that the Mobile Card has been installed correctly. The system supports 2G, 3G as well as 4G LTE network thereby providing higher bandwidth, greater connection speed and better underlying technology for VoIP calls. Access to these networks is achieved by mounting any of these — 2G, 3G or 4G — GSM modules on the Mobile card.

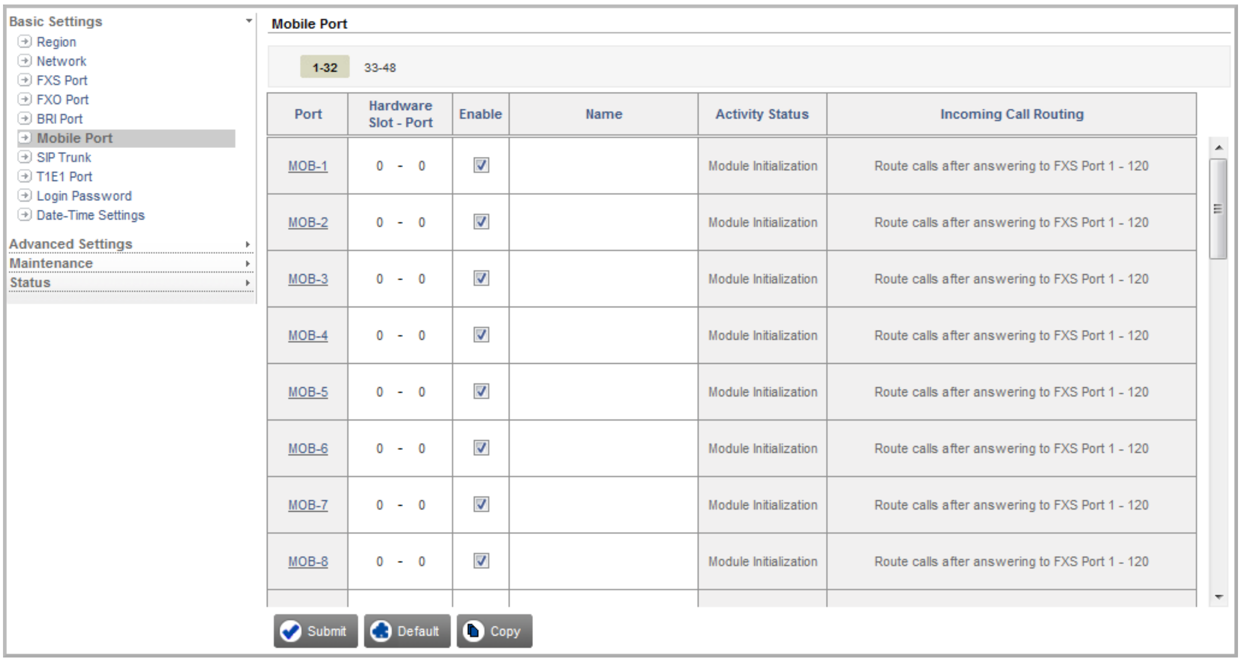
Configuration of Mobile Port involves:

1. Customizing the Mobile Port Parameters
2. Network Selection
3. Configuring VoLTE parameters. For details refer to “VoLTE Configuration”.
4. Uploading MBN files, For details refer to “MBN File Upload”.
5. Viewing Mobile Port Status

To configure a Mobile Port,

- Click the **Basic Settings** link to expand.

- Click the **Mobile Port** link.



The Mobile Port page displays the following parameters:

- **Port:** It displays the Mobile Port numbers. Click on the desired Mobile Port number to configure the Port Parameters.

- **Hardware Slot-Port:** The SARVAM UMG can automatically detect and assign the Hardware Slot and Port numbers to the Mobile software ports. However, if required you may change the Hardware Slot and Port assigned to the Mobile software port. In this case, enter the desired Hardware Slot and Port number.

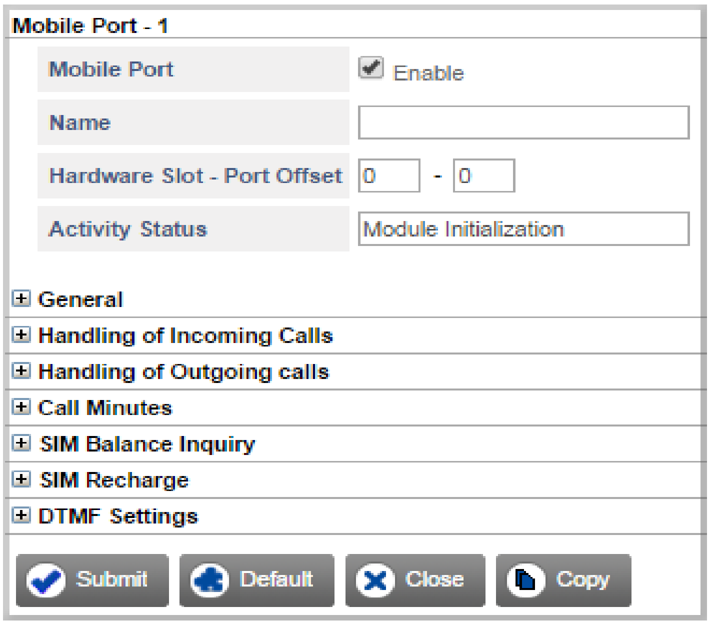
- **Enable:** Keep the **Mobile Ports** enabled. Clear the Mobile Port **Enable** check box, only if you do not want to use the respective port. Default: Enabled.

- **Name:** Assign a Name to the Mobile Port for identification. The Name can be a maximum of 24 characters.
- **Activity Status:** This displays the port activity status — Module Initialization, SIM PUK Required, SIM PIN Wrong, SIM Absent, SIM Present, Network Absent, Network Present.
- **Incoming Call Routing:** It displays the Incoming Call Routing Method you select.

To configure the **Mobile Port**,

- Click **MOB-1**.

  The **Mobile Port-1** window opens.



- Keep the **Mobile Ports** enabled.

  Clear this check box, only if you do not want to use the respective port. Default: Enabled.

- You can assign a **Name** to each Mobile Port. The name will be displayed as CLIP during incoming calls. The Name can be of maximum 24 characters. Default: Blank.

- SARVAM UMG will assign the **Hardware Slot - Port Offset** automatically, when any card is inserted in the system.

  Hardware Slot is the number of the universal slot of SARVAM UMG in which the Mobile Card is inserted. Range of slot number is 1-12. 'Port' is the number of the Mobile port in which you have inserted the SIM card.

  For example: if you have inserted the Mobile4 Card in Slot number 02 of SARVAM UMG, the system will assign the hardware slot 02 and port numbers 01-04 to the Mobile Software Ports from 01 to 04 respectively.

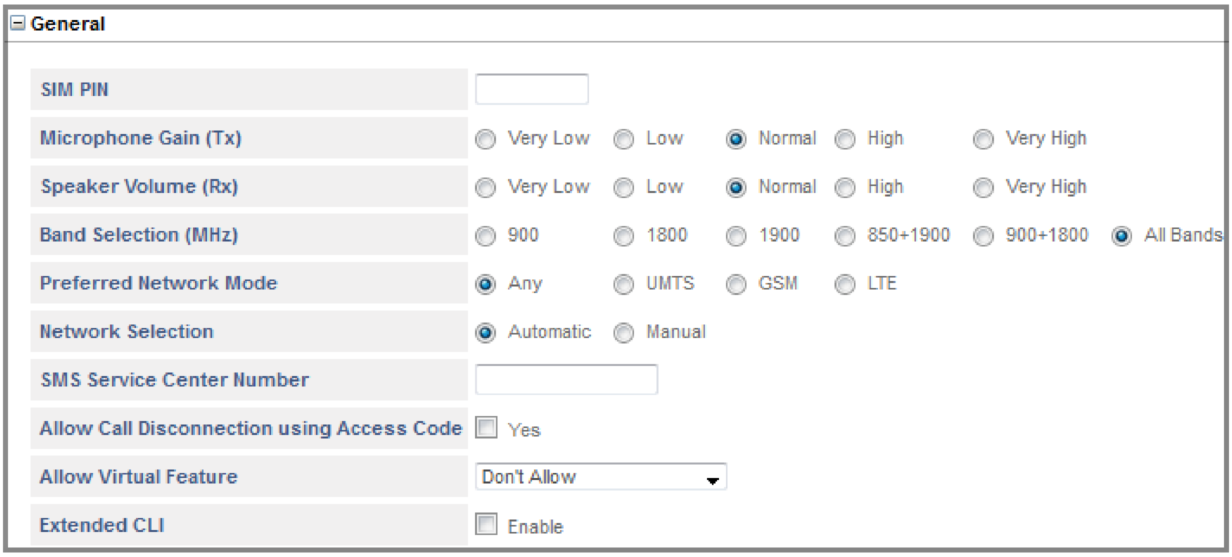
  However, if required, you may change the Hardware Slot and Port assigned to the Mobile software port. In this case, enter the desired Hardware Slot and Port number.

If you want to de-assign the Hardware Slot and Port, enter '00' in both fields. By default, Hardware Slot-Port is 00–00.

- **Activity Status** displays the port activity status — Module Initialization, SIM PUK Required, SIM PIN Wrong, SIM Absent, SIM Present, Network Absent, Network Present.

General

- Click **General** to expand.



- If you have enabled SIM PIN protection on the SIM Card, in **SIM PIN**, enter the same SIM PIN value (4 to 8 digits). Default: Blank.

- *If the wrong SIM PIN is entered thrice in a row, by a user, the SIM will be locked and will ask for the Personal Unlock Keyword (PUK).*

- *SIM PIN is not set to default or does not change, if SARVAM UMG is set to default or when you upgrade/downgrade firmware.*

- You can adjust the **Microphone Gain (Tx)** Gain of the Mobile Port to improve the audibility of the transmitting speech from SARVAM UMG. Select the desired Tx Gain — **Very Low, Low, Normal**, **High** and **Very High**. Default: Normal.

- You can adjust the **Speaker Volume (Rx)** of the Mobile Port to improve the audibility of incoming speech. Select the desired Rx Gain — **Very Low**, **Low**, **Normal**, **High** or **Very High**. Default: Normal.

- The Frequency **Band Selection (MHz)** supported by the GSM networks varies from country to country.

  You can select the Frequency Band used by the GSM Service Provider(s) in the country where SARVAM UMG is installed. The supported bands are — 900,1800,1900,850+1900, 900+1800 and All bands. Default: All Bands.

  Select 850 + 1900 GSM frequency band for countries which support both 850 and 1900 MHz frequencies for GSM network.

*If you select **LTE** as the **Preferred Network Mode**, **Band Selection (MHz)** will not be applicable.*

When you change the Frequency Band, the change will take effect after the next system restart or the next Mobile Port restart.

- If your SARVAM UMG has a 4G Mobile Port, the SIM Card of this port will get registered with either GSM (2G) or UMTS (3G) or LTE (4G) network, whichever is available. You can select the Network with which the SIM should be registered by setting the **Preferred Network Mode**.

  If the SIM installed in the Mobile Port supports GSM, UMTS and LTE services, and you wish the SIM to get registered with only one of these networks, you can restrict the other network by setting the Preferred Network Mode — Any, UMTS, GSM or LTE.

  - **Any:** If you select this option, the SIM will automatically register with the available network, that is, UMTS (3G) or GSM (2G) or LTE (4G). For example if you have a 3G SIM and 3G network is not available it will automatically fall-back to 2G network.
  - **UMTS:** Select this when you want the SIM to get registered with UMTS (3G) network only.
  - **GSM:** Select this when you want the SIM to get registered with GSM (2G) network only.
  - **LTE:** Select this when you want the SIM to get registered with LTE (4G) network only.

    Default: Any.

If you select the Preferred Network Mode as LTE only, calling functionality will be possible only if the Service provider supports calling through VoLTE.

  If your Mobile Port supports GSM only, do not change the default value of this parameter.

- Select the desired **Network Selection** mode — **Automatic** or **Manual**. Default: Automatic.
  - Select **Automatic** when you want the Mobile Port to automatically locate and register with the Network that supports the SIM card. At each power ON, the SIM in the Mobile Port will automatically register with the Network.
  - Select **Manual** when you want the Mobile Port to select the network operator according to the priority set by you. You also need to configure the list of network operators as per your preference.

    To apply **Manual** Network Selection,

    - Select **Manual** and click **Settings** .

        - The **Manual Network List Priority** window opens.

        - In the Priority levels, **Priority1** to **Priority 9**, enter the Network Operator Codes with which you want the SIM to register, as per your preference.

          The Network Operator Code may consist of a minimum of 5 digits and a maximum of 8 digits. Default: Blank.

    - Click **Submit** and then close the **Manual Network List Priority** window.

- Enter the **SMS Service Center Number**, as provided to you by your service provider. This number is used by the system while sending the SMS notifications.

- Select the **Allow Call Disconnection using Access code** check box to enable the Disconnect Call using Access Code feature on the Mobile Port. For further details, see "Disconnecting a Call using Access Code".

- Select the desired option in **Allow Virtual Feature** — Don't Allow, Allow All, As per Virtual User Table to determine the access of Virtual feature to the user during an ongoing call.

    - If you select **Don't Allow**, then the system will not allow access to Virtual User feature. However, you can access the feature access code for making or disconnecting a call on this mobile port.

    - If you select **Allow All**, then the system will allow access of Virtual feature. However, you cannot access the feature access code for making or disconnecting a call on this mobile port.

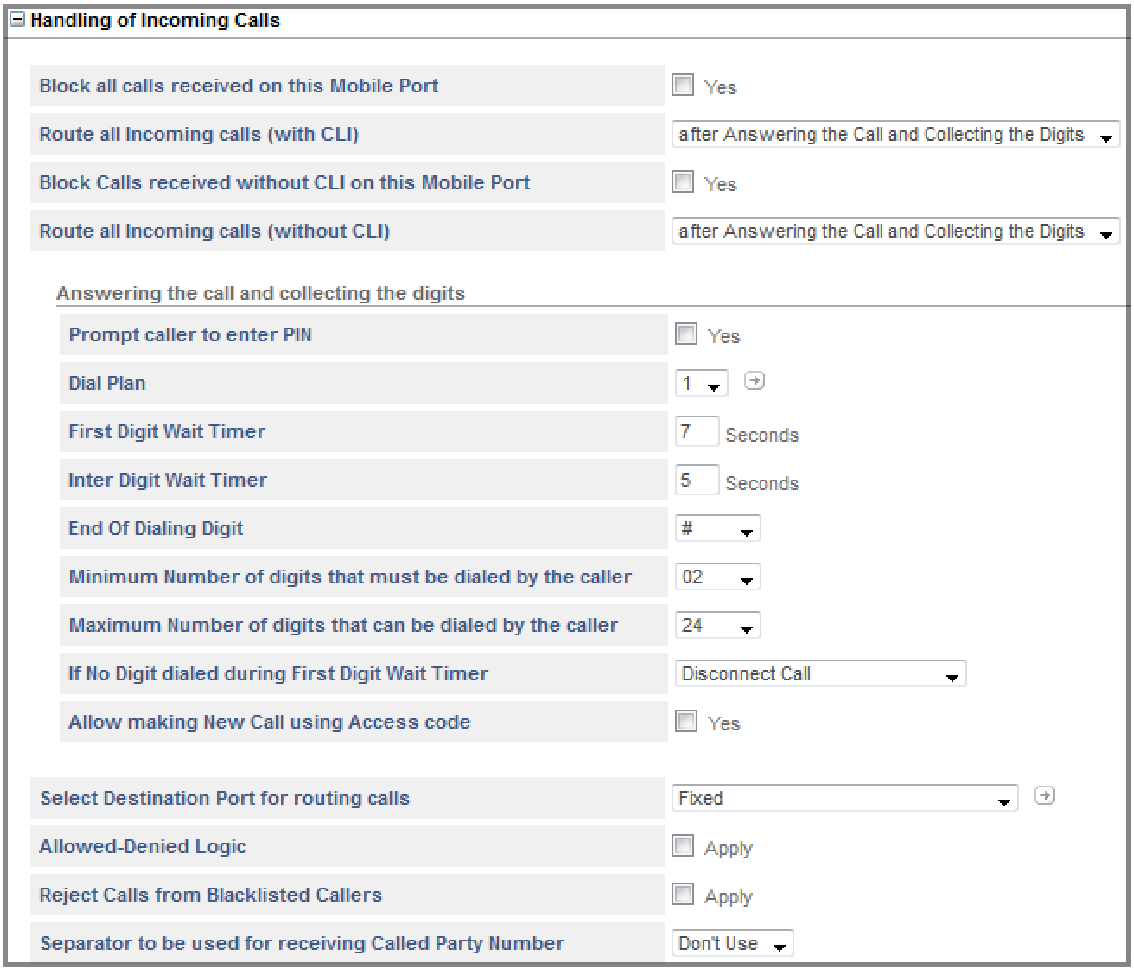
    - If you select **As per Virtual User Table,** then system will allow access of Virtual feature only to those Virtual Users as configured in the Virtual User Table.

- Select the **Extended CLI** check box only when a wrong CLI is being displayed. When this parameter is enabled, the current caller's CLI will be displayed. By default, it is disabled.

- Click **Submit** to save.

Handling of Incoming Calls

- Click **Handling of Incoming Calls** to expand.



Handling of Incoming Calls

Block all calls received on this Mobile Port	Yes
Route all Incoming calls (with CLI)	after Answering the Call and Collecting the Digits
Block Calls received without CLI on this Mobile Port	Yes
Route all Incoming calls (without CLI)	after Answering the Call and Collecting the Digits
Answering the call and collecting the digits	
Prompt caller to enter PIN	Yes
Dial Plan	1
First Digit Wait Timer	7 Seconds
Inter Digit Wait Timer	5 Seconds
End Of Dialing Digit	#
Minimum Number of digits that must be dialed by the caller	02
Maximum Number of digits that can be dialed by the caller	24
If No Digit dialed during First Digit Wait Timer	Disconnect Call
Allow making New Call using Access code	Yes
Select Destination Port for routing calls	Fixed
Allowed-Denied Logic	Apply
Reject Calls from Blacklisted Callers	Apply
Separator to be used for receiving Called Party Number	Don't Use

- If you do not want to route calls received on this Mobile Port, select **Block all calls received on this Mobile Port** check box. Default: Disabled.

Destination Number Determination

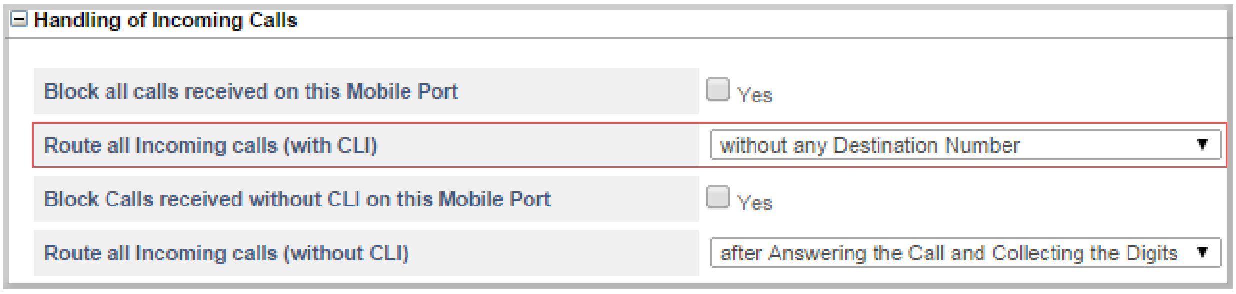
Select the desired destination number determination method for routing incoming calls *with* and *without* CLI.

- To **Route all Incoming calls (with CLI)**, you may select from any of the following methods.
  - without any Destination Number
  - to the Fixed Destination Number
  - on the basis of Calling Party Number
  - after Answering the Call and Collecting the Digits
    Default: without any Destination Number

If the destination number to be dialed out is an IP Address, SARVAM UMG will not check the Destination Port Determination Method. Instead, it will route the call using the SIP Trunk / Group programmed for IP Dialing. (See "IP Dialing" to know more).

Route Calls without any Destination Number

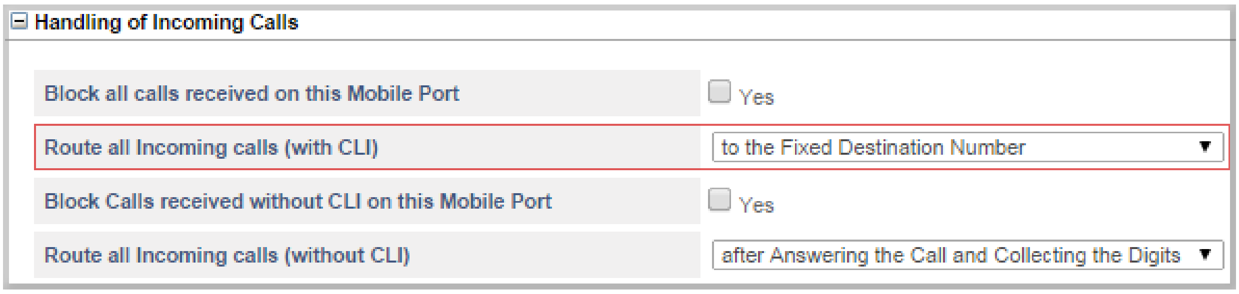
In this method, all calls received on the Mobile Port are directly routed to the destination port, regardless of the Destination Number.



- To apply this method, in **Route all incoming calls (with CLI)**, select **without any Destination Number.**

Route to the Fixed Destination Number

In this method, calls received on the Mobile Port are routed to a fixed destination number, which you must configure for this port.



To apply this method, do the following:

- In **Route all incoming calls (with CLI)**, select **to the Fixed Destination Number**.
- In **Fixed Destination Number**, enter the desired destination number.

  The Destination Number may consist of a maximum of 24 digits. Valid digits are 0 to 9, *, # and (.) dot.
  Default: Blank.
- Click **Submit** to save the changes.

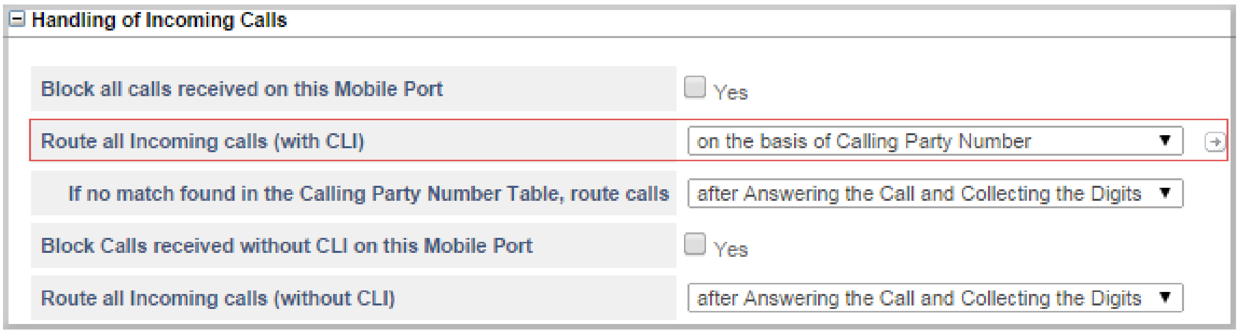
Route on the basis of Calling Party Number

In this method, a call received on the Mobile Port is routed to a specific destination number, as per the calling party's number. For this, the calling party numbers and their corresponding destination numbers must be configured in the Calling Party Number Based Table.
Whenever there is an incoming call on the Mobile Port, SARVAM UMG will match the Calling Party Number with the entries of the Calling Party Number Based Table. If a match is found, the call will be routed to the destination number configured for that Calling Party Number.

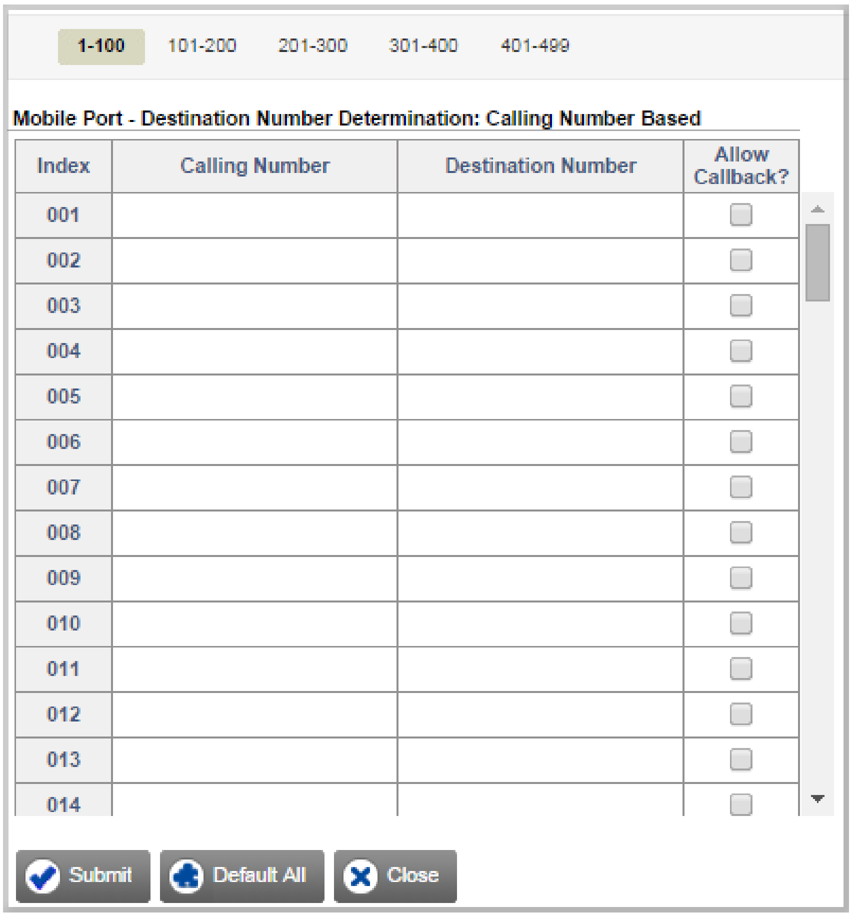
To apply this method, do the following:

- In **Route all incoming calls (with CLI)**, select **on the basis of Calling Party Number.**

- Click **Settings** .



The **Mobile Port Destination Number Determination: Calling Number Based** Table opens.



- Configure the **Calling Number Based** table for the Mobile Port. You can enter upto 499 Calling Party Numbers and their corresponding Destination Numbers in this table.
- In **Calling Number**, enter the calling party numbers. The Calling numbers may consist of a maximum of 24 characters. Default: Blank.
- For each calling party number, enter a corresponding destination number in **Destination Number**. Destination numbers may consist of a maximum of 24 characters. Digits 0 to 9, *, # and (.) dot are allowed. Default: Blank.

- Click **Submit** to save your entries. Close the window to return to the Mobile Port window.

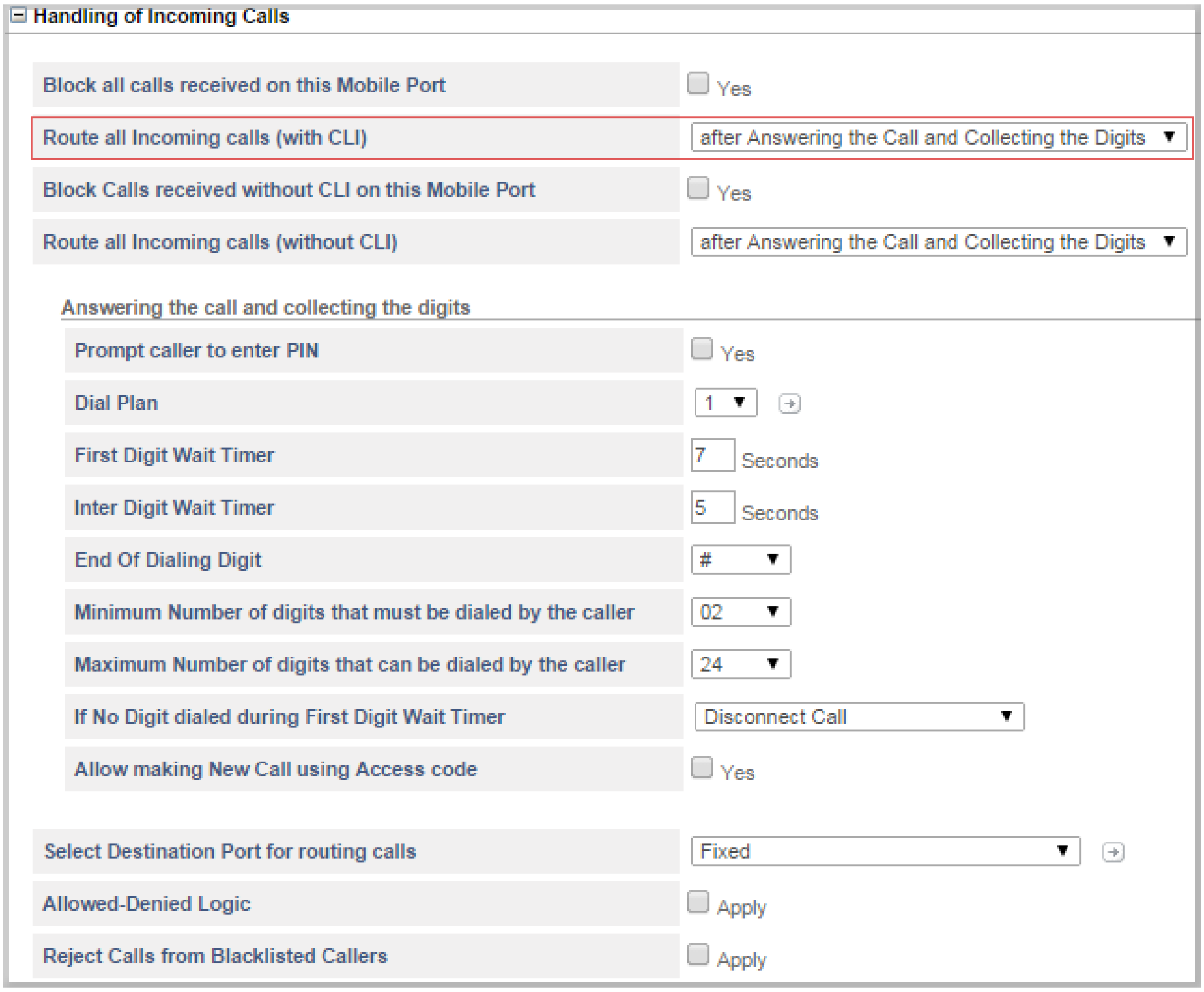
  You can also configure the **Calling Number Based** Table from *Advanced Settings* link. For instructions, see "Destination Number Determination" under *Advanced Settings*.

- For incoming calls with Calling Party Numbers that do not match with the Calling Party Number Table, you may select the destination number determination method.

  In the **If no match found in the Calling Party Number Table, route calls** box, you may select either **to the Fixed Destination Number** or **after Answering the Call and Collecting the Digits**. Default: after Answering the Call and Collecting the Digits.

Route After Answering the Call and Collecting the Digits

In this method, the system answers the incoming call on the Mobile Port and plays dial tone to the caller, allowing the caller to dial the desired number. The number dialed by the caller is considered as the destination number.



To apply this method, do the following:

- In **Route all incoming calls (with CLI)**, select **after Answering the Call and Collecting the Digits.**

  Under **Answering the Call and Collecting the Digits**,

- If you want to enable PIN Authentication on the Mobile Port, select the **Prompt caller to enter PIN** check box.

  If you enable this check box, you must also configure the PIN Authentication Table. To know more about this feature and for detail instructions, see "PIN Authentication" under *Advanced Settings*.

- SARVAM UMG supports 8 Dial Plans with total 64 entries in each table. When a user dials a number, it is compared with the Destination Number configured in the Dial Plan. If a match is found, the system routes the call immediately without waiting for End of Dialing and if a match is not found, the system will wait for the End of Dialing and then route the call as per the Destination Port Selection method configured.

  Select the **Dial Plan** table number you configured for this port. If you have not configured the Dial Plan table you may do so now,

  - Click **Settings** the Dial Plan Table opens.
  - Configure the numbers in the table. For detailed instructions, see "Dial Plan".

- Set the duration of the **First Digit Wait Timer (FDWT)**. This is the duration for which you want the system to wait for the caller to dial the destination number after the dial tone. Valid range is 01 to 99 seconds. Default: 7 seconds.

- Set the duration of the **Inter Digit Wait Timer**. This is the duration for which you want the system to wait while receiving the digits dialed by the caller to consider it as End of Dialing. You may change this timer, if required. Valid range is 01 to 99 seconds. Default: 05 seconds.

- As **End of Dialing Digit**, select whether the system should consider **#** or ***** as termination digit to detect end of dialing. Default: #

- In **Minimum number of digits that can be dialed by the caller**, select the minimum number of digits to be dialed by the user for the system to consider it as a valid number**.** Valid range is 01 to 24 digits. Default: 2 digits.

- In **Maximum number of digits that can be dialed by the Caller**, select the maximum number of digits to be dialed by the user for the system to consider it as end-of-dialing**.** Valid range is 01 to 24 digits. Default: 24 digits.

  When the caller dials a number, the system will match it with the above end-of-dialing indications and accept the one that matches first.

- If the caller fails to dial the number during the First Digit Wait Timer, you can either have the system disconnect the call or route the call to a fixed destination number.

  In **If No Digit dialed during First Digit Wait Timer (FDWT)**, you can select — **Disconnect the Call** or **Use Fixed Destination Number**. Default: Disconnect Call.

- If you select **Use Fixed Destination Number**, enter the desired destination number in **Fixed Destination Number.** The Destination number may consist of a maximum of 24 digits. Valid digits are 0 to 9, *, # and (.) dot. Default: Blank.

- Select the **Allow making New Call using Access Code** check box, if you want to enable the feature Making New Call using Access Code on the FXO Port. See "Making a New Call using Access Code".

- Click **Submit** to save settings.
- If you do not want to route calls without CLI through this port, select the **Block Calls received without CLI on this Mobile Port** check box.
- To **Route all Incoming calls (without CLI),** you may select from any of the following methods:
  - to the Fixed Destination Number, see “Route to the Fixed Destination Number”.
  - after Answering the Call and Collecting the Digits, see “Route After Answering the Call and Collecting the Digits”.

    Default: after Answering the Call and Collecting the Digits.

Destination Port Determination

Select the Destination Port for routing calls for the Mobile Port. You may select from any of the following options:

- Fixed
- On the basis of Destination Number
- On the basis of Calling Party Number

  Default: Fixed.

  Read the description and follow the instructions for each of these destination port selection methods given below.

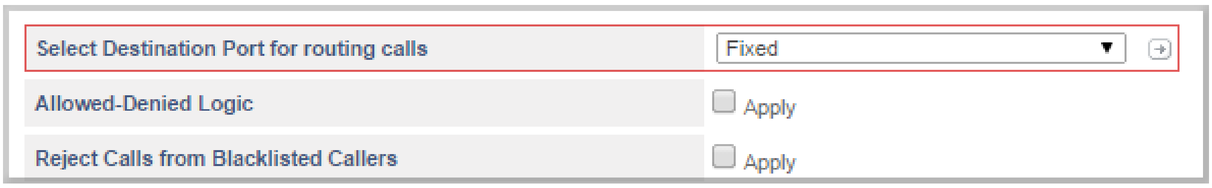
If the destination number to be dialed out is an IP Address, SARVAM UMG will not check the Destination Port Determination Method. Instead, it will route the call using the SIP Trunk / Group programmed for IP Dialing. (To know more, see the feature description “IP Dialing”).

Fixed

In this method, calls received on the Mobile Port are routed to a Fixed Destination Port, irrespective of the number dialed on the Mobile Port.

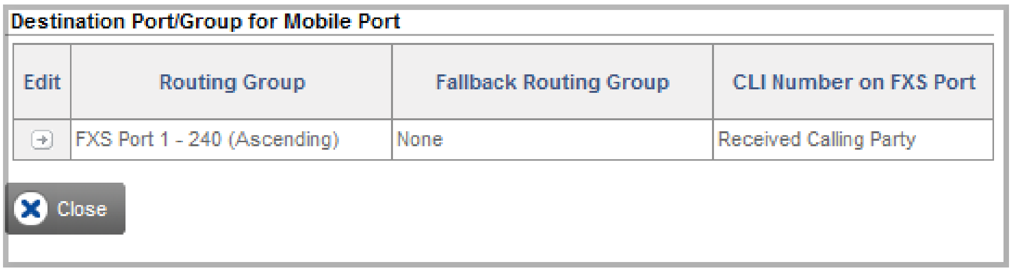
To apply this method, do the following:

- In **Select Destination Port for routing calls**, select **Fixed** option.



- Click **Settings** .

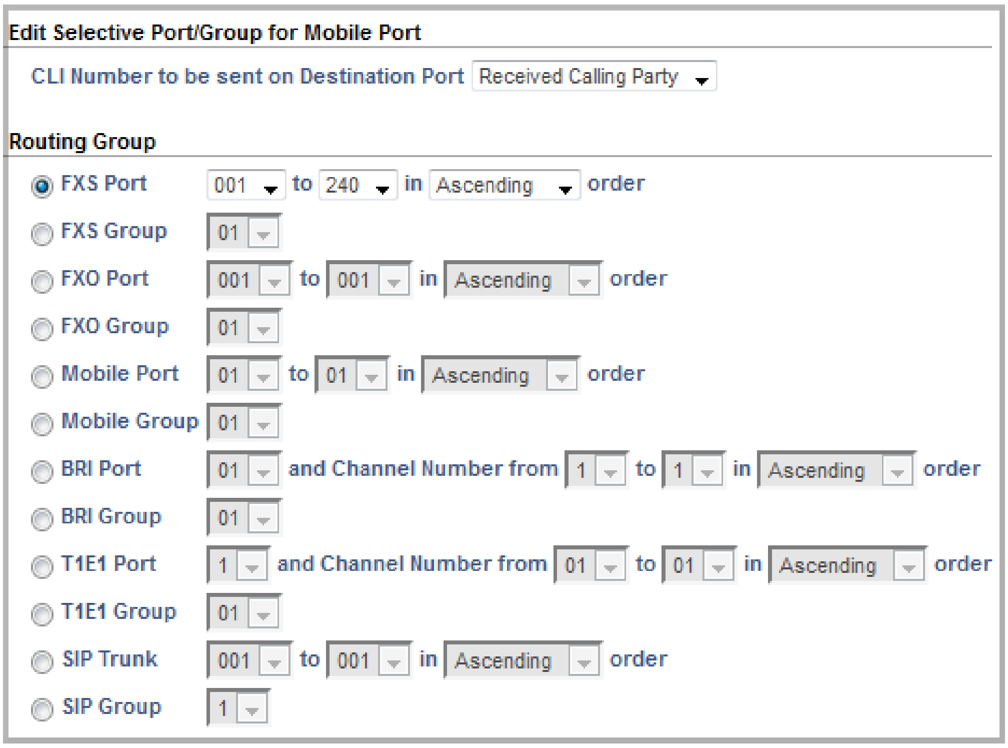
The **Destination Port/Group for Mobile Port** window opens.



The default **Routing Group** and **Fallback Routing Groups** appear.

- If you wish to change the default Routing Group options, click **Edit**.

The **Edit Selective Port/Group for Mobile Port** window opens.

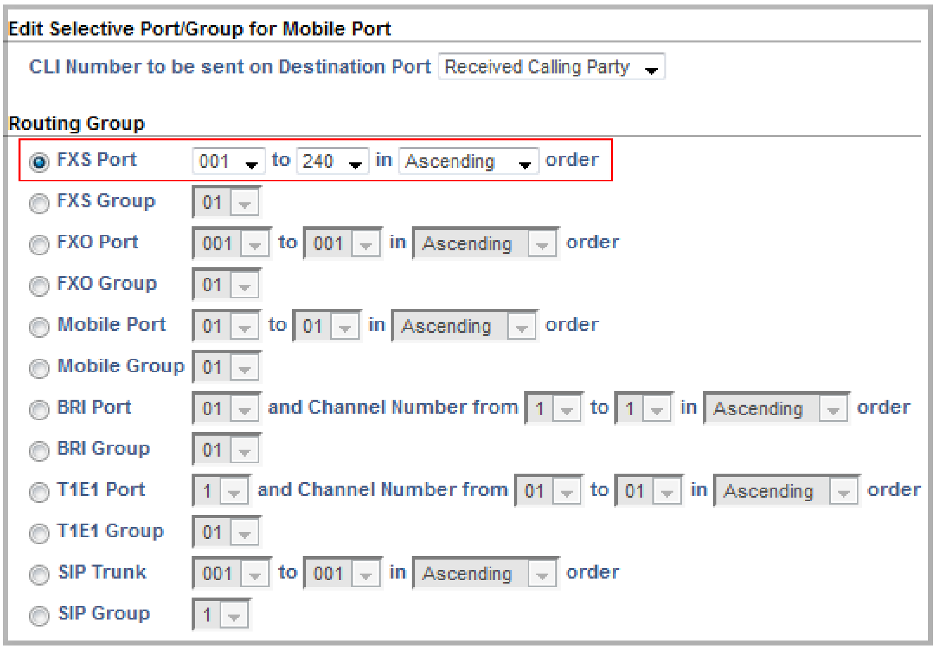


- Select the **CLI Number to be sent on Destination Port**. You can select Received Calling Party or Received Called Party. Default: Received Calling Party.

CLI to be sent on Destination Port is applicable when FXS Port/Group is determined as the Destination Port.

- Create the **Routing Group**.

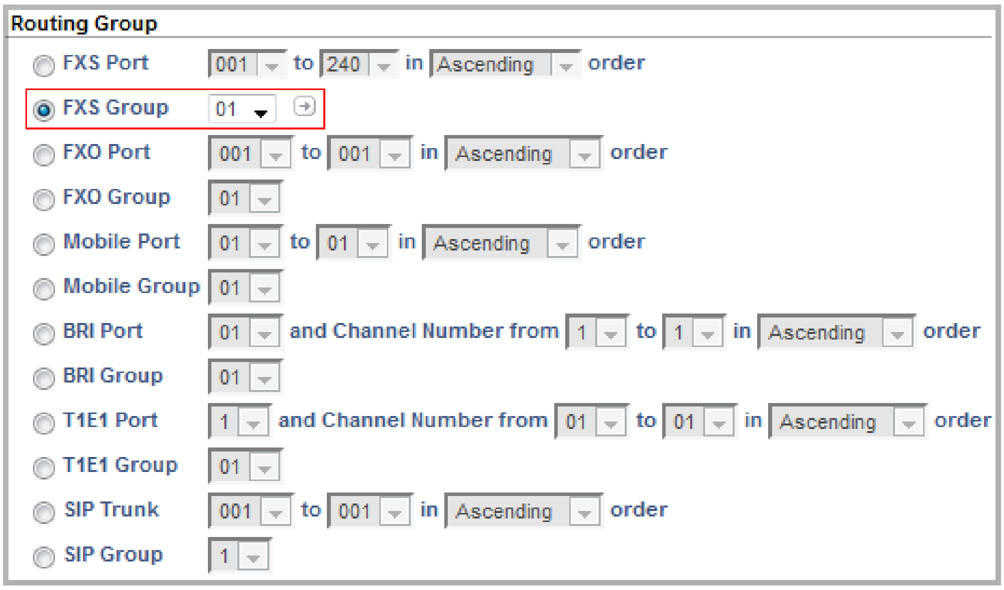
- To create a group of *sequential* **FXS Ports** as members,



  - Select the desired **FXS Port** numbers as members. Default: 1.
  - In **in - order**, select the order in which the system should hunt for a free member FXS Port to route the call.

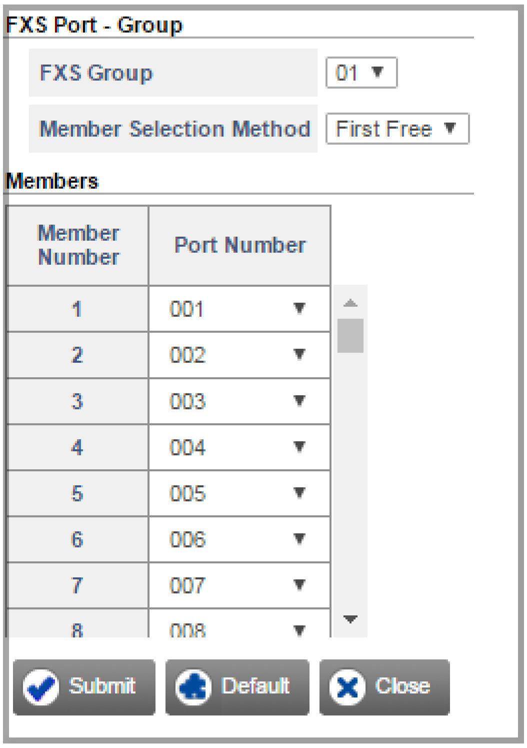
    Select **Ascending** to start hunting from the first to the last member FXS Port. Select **Descending** to start hunting from the last to the first member FXS Port. Default: Ascending.

- To create a group of *not-sequential* **FXS Ports** as members,
  - Select a **FXS Group**.



  - Select **FXS Group** number. Default: 1.

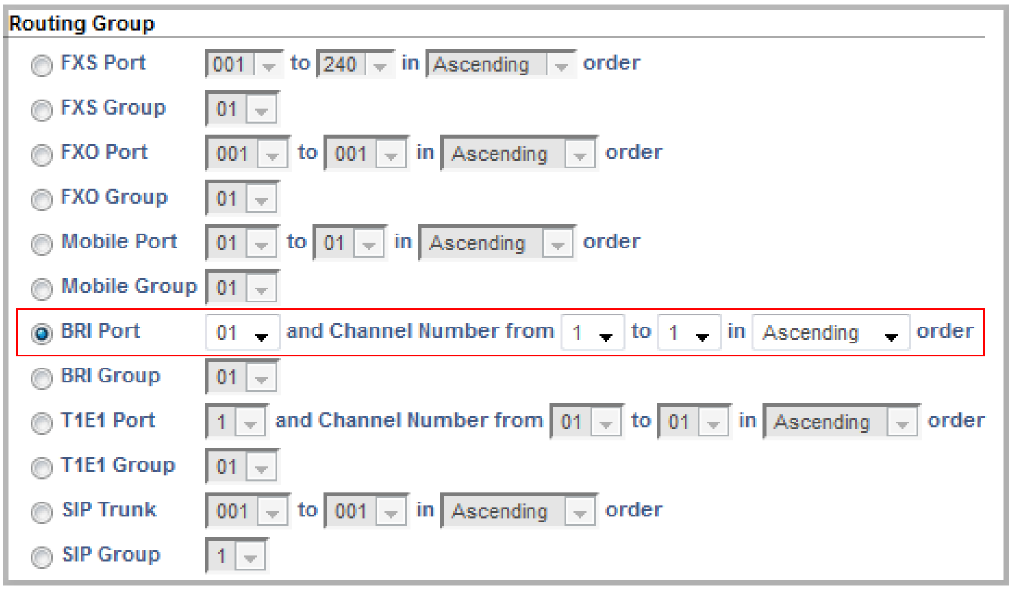
- Click **Settings** .
- The **FXS Port - Groups** window opens.



- Create the FXS Group. For detailed instructions on creating groups, see the topic “Group” under *Advanced Settings*.

- Similarly, you can create a group of *sequential* and *not-sequential* FXO Ports, Mobile Ports and SIP Trunks.

- To create a routing group of *sequential* **BRI Channels** as members,



- Select the **BRI Port** Number. Default: 1.

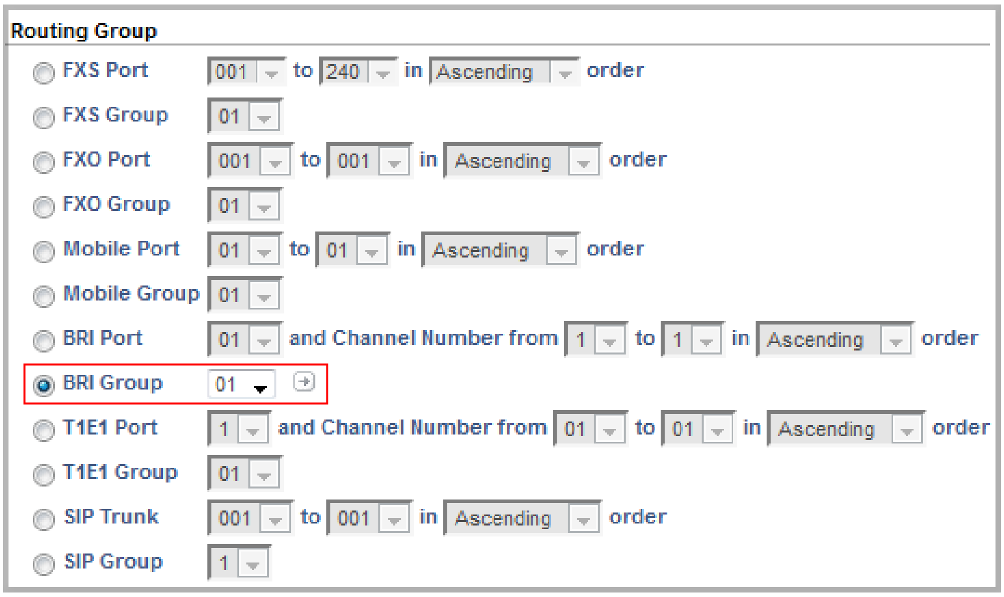
- In Channel Number **From - to**, select the **Start Channel Number** and the **End Channel Number**, respectively.

- In **in - order**, select the order in which the system should hunt for a free member Channel to route the call.

  Select **Ascending** to start hunting from the first to the last member channel. Select **Descending** to start hunting from the last to the first member channel. Default: Ascending.

- To create a group of *not-sequential* **BRI Channels** as members,

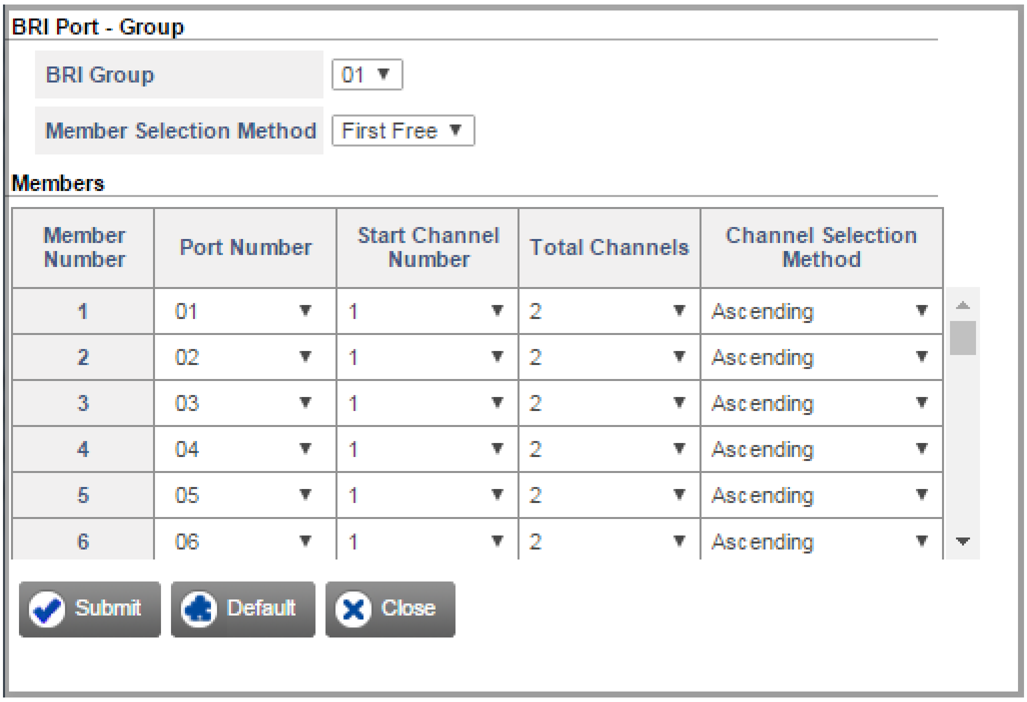
  - Select **BRI Group**.



  - Select a **BRI Group** number. Default: 1.

  - Click **Settings**.

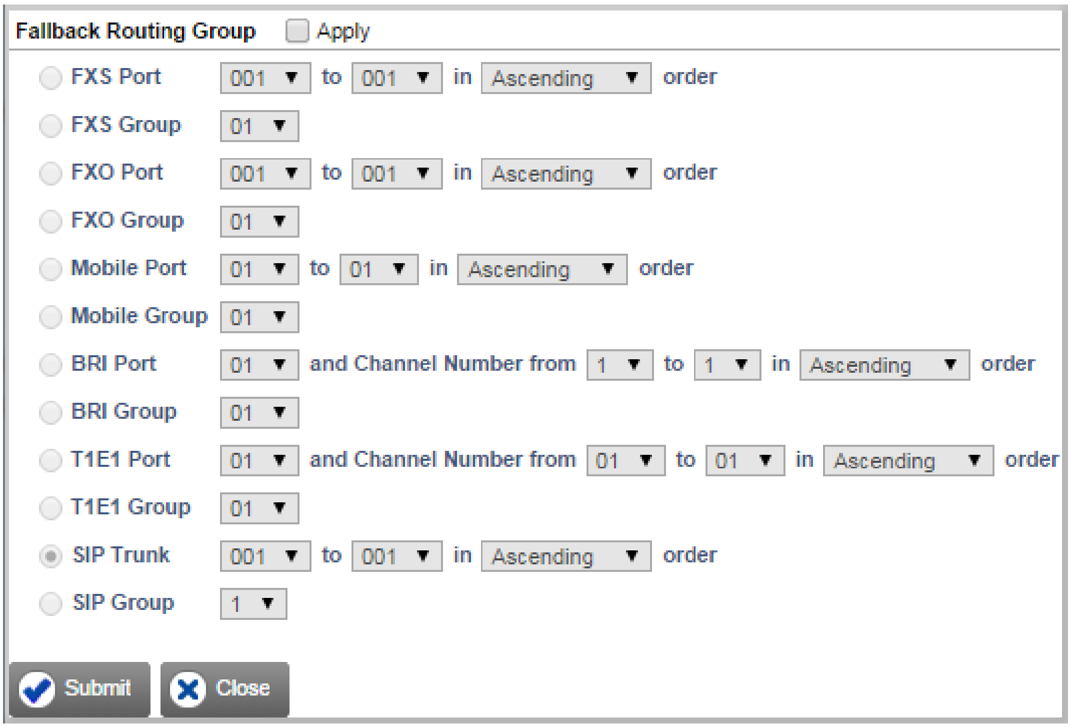
- The **BRI Port - Groups** window opens.



- Create the BRI Group. For detailed instructions on creating groups, see the topic “Group” under *Advanced Settings.*

- Similarly, you can create a group of *sequential* and *not-sequential* T1E1 Ports.

- You may create the **Fallback Routing Group**.



- To do this,
    - Select the **Apply** check box.
    - Follow the same instructions provided earlier for creating *sequential* and *not-sequential* groups of FXS Ports, FXO Ports, Mobile Ports, BRI Ports, T1E1 Ports and SIP Trunks.
- Click **Submit** to save changes. The **Edit** window closes.
- The entry you edited appears in the **Destination Port/Group for Mobile Port** window.
- Close the **Destination Port/Group for Mobile Port** window to return to the Handling of Calls window.

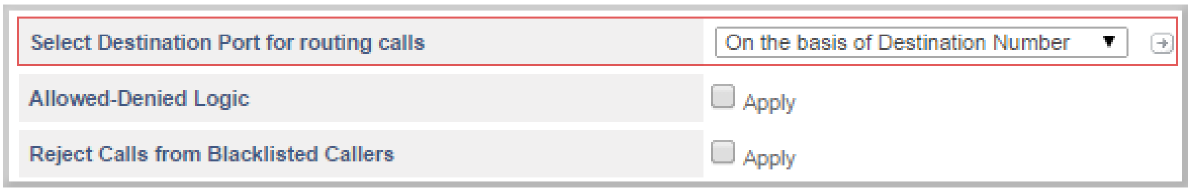
On the basis of Destination Number

In this method, incoming calls on the source port are routed to the destination port on the basis of the destination number (called party number) dialed by the caller.

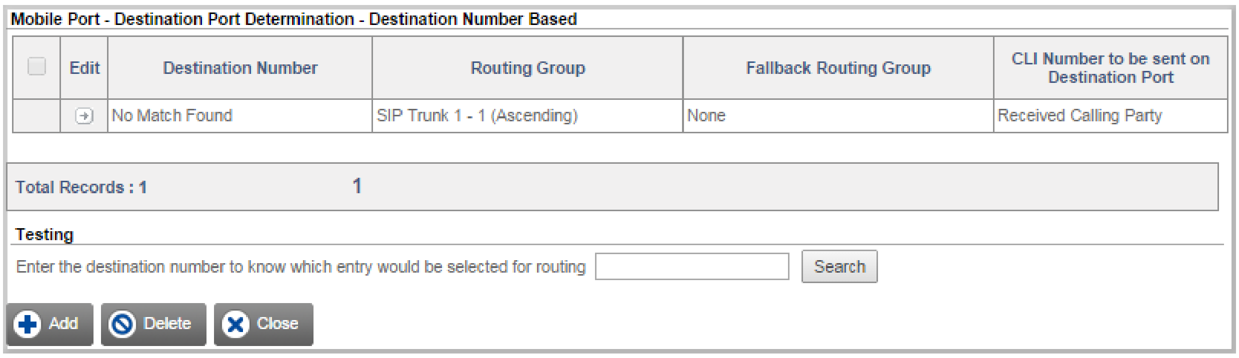
You must configure the called party numbers in the **Destination Number Based** Table. SARVAM UMG will match the called party number dialed by the caller with the entries of this table. If a match is found for the number in the table, the call is routed to the destination.

To apply this method, do the following:

- In **Select Destination Port for routing calls**, select **On the basis of Destination Number** option.
- Click **Settings** .



The **Mobile Port - Destination Port Determination - Destination Number Based** table window opens.


Mobile Port - Destination Port Determination - Destination Number Based

	Edit	Destination Number	Routing Group	Fallback Routing Group	CLI Number to be sent on Destination Port
		No Match Found	SIP Trunk 1 - 1 (Ascending)	None	Received Calling Party

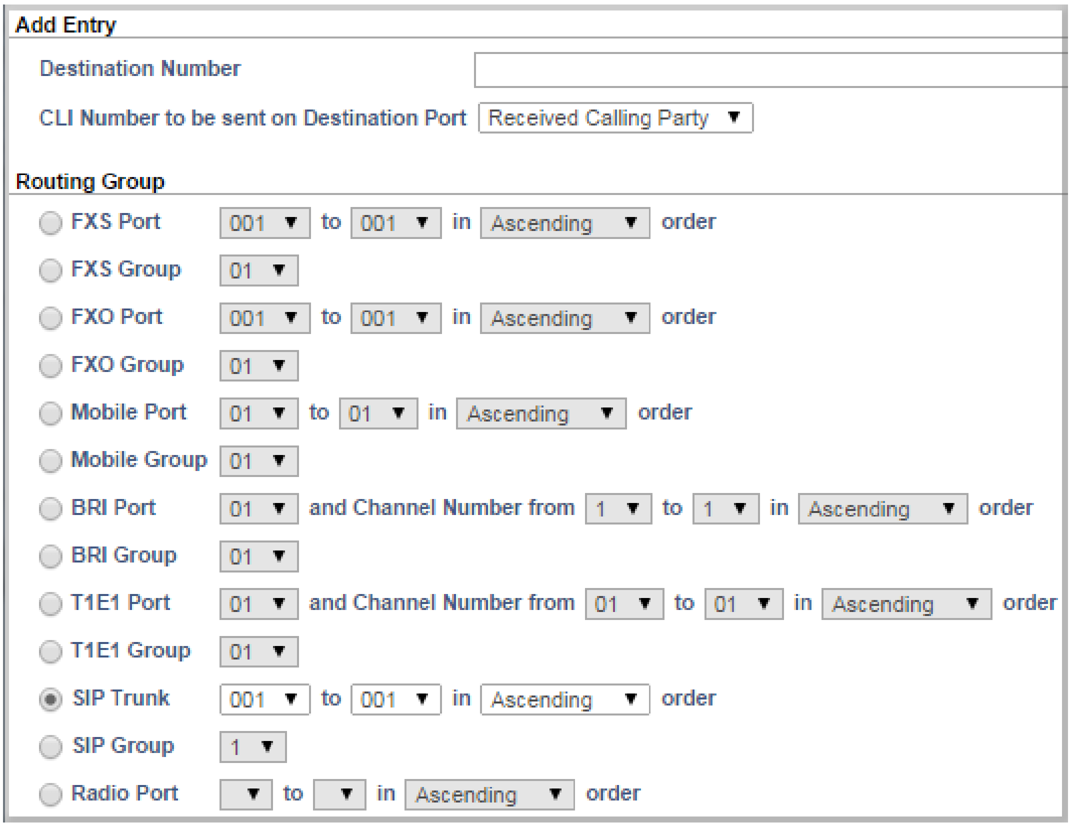
Total Records : 1 1

Testing

Enter the destination number to know which entry would be selected for routing Search

Add Delete Close

- To add a new entry, click **Add**. The **Add Entry** window opens. You can add upto 250 entries.



- In **Destination Number**, enter the number you expect the callers to dial. You may enter upto 64 characters (Digits + "Wildcard Characters") in this field. Valid characters are 0 to 9, *, #, X, T, Comma [,], Hyphen [-], Caret [^]. Default: Blank.

Wildcard Characters

SARVAM UMG supports following characters.

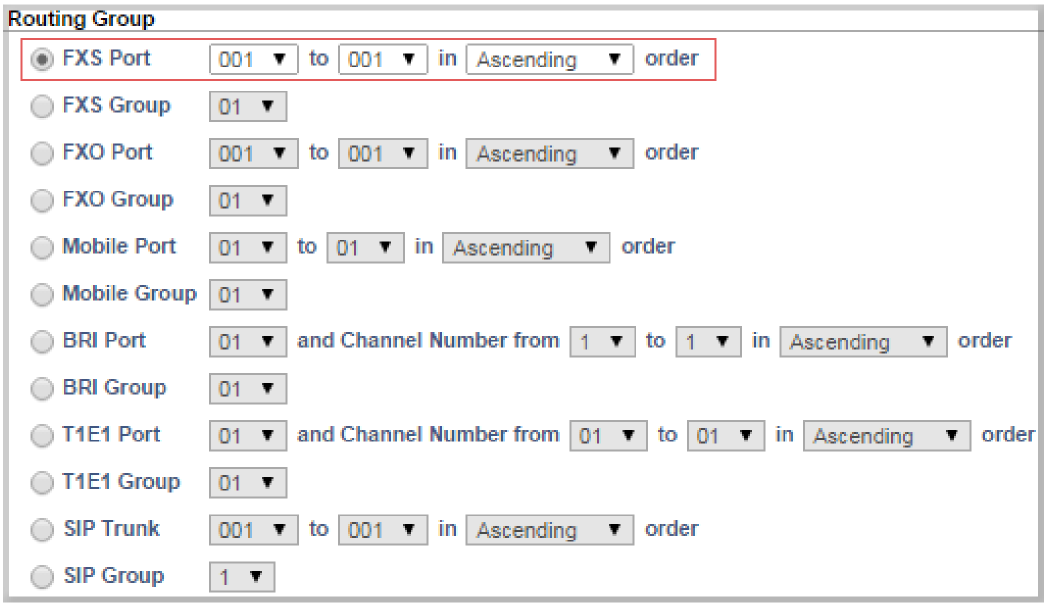
Character	Description
X (letter X)	X represents any single digit from 0 to 9.
#	When # is configured in a number string, it will not be considered as End of Dialing.
*****	When * is configured in a number string, it will not be considered as End of Dialing.
+	+ (plus) can be configured as a first character 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.

- Select the **CLI Number to be sent on Destination Port**. You can select Received Calling Party or Received Called Party. Default: Received Calling Party.

CLI to be sent on Destination Port is applicable when FXS Port/Group is determined as the Destination Port.

- Create the **Routing Group**.

  - To create a group of *sequential* **FXS Ports** as members,



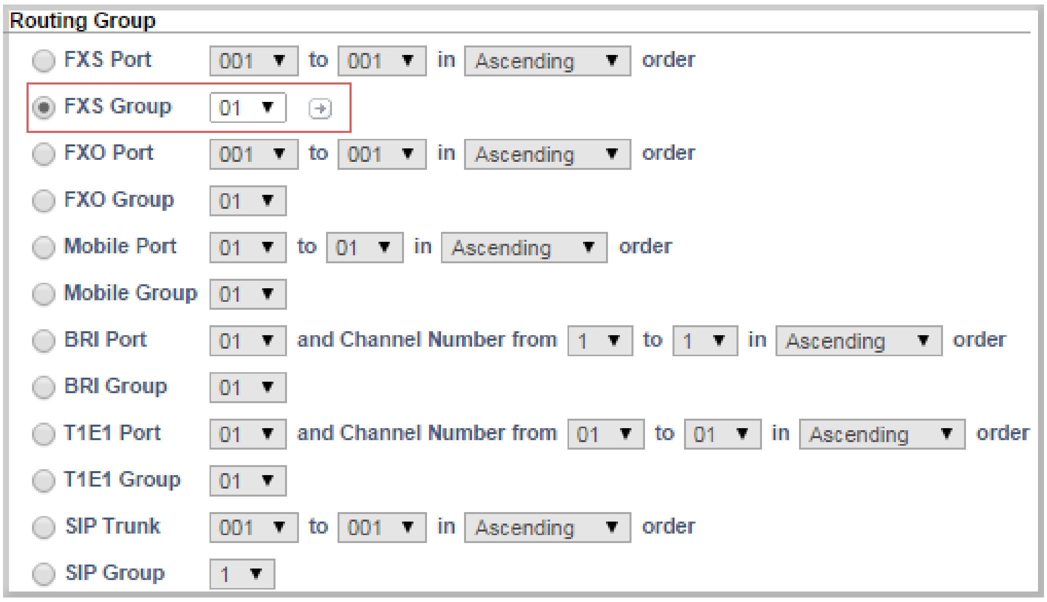
    - Select the desired **FXS Port** numbers as members. Default: 1.

    - In **in - order**, select the order in which the system should hunt for a free member FXS Port to route the call.

      Select **Ascending** to start hunting from the first to the last member FXS Port. Select **Descending** to start hunting from the last to the first member FXS Port. Default: Ascending.

  - To create a group of *not-sequential* **FXS Ports** as members,

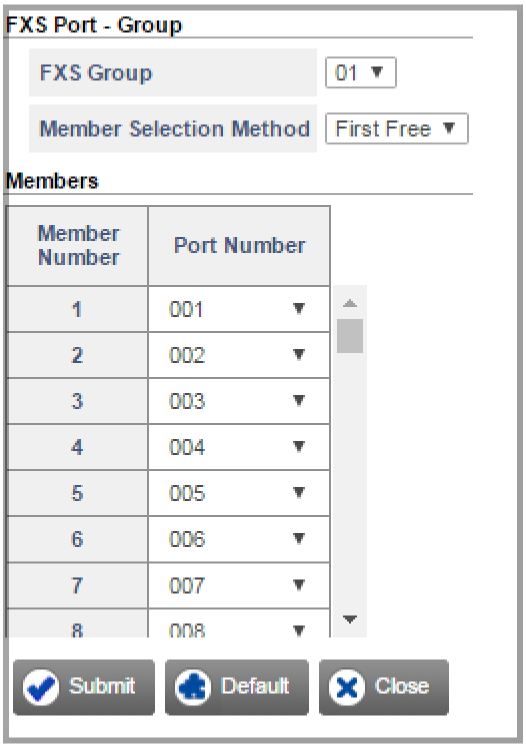
- Select a **FXS Group**.



- Select **FXS Group** number. Default: 1.

- Click **Settings** .

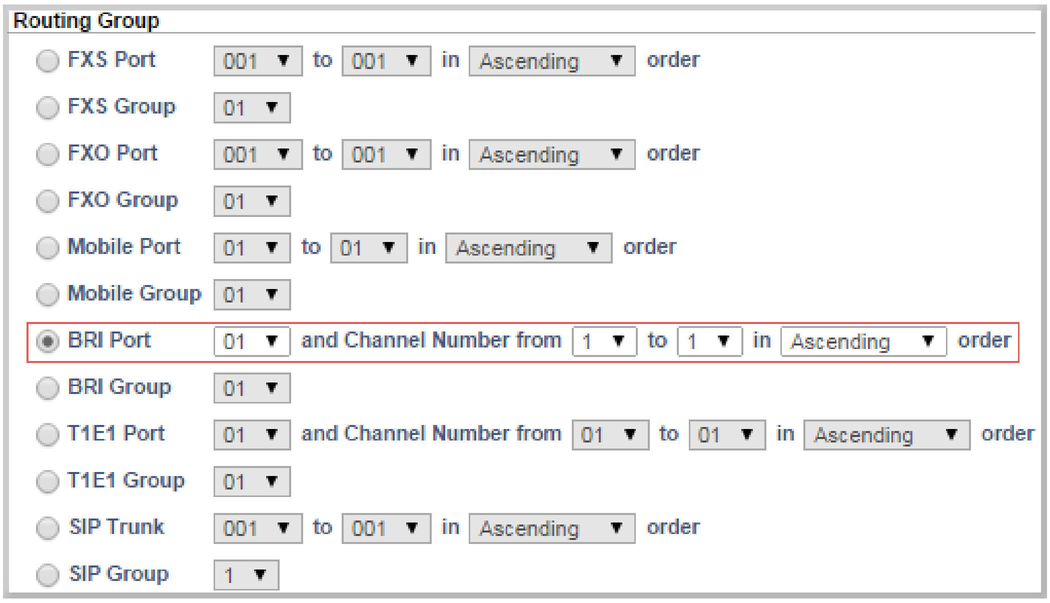
- The **FXS Port - Groups** window opens.



- Create the FXS Group. For detailed instructions on creating groups, see the topic “Group” under *Advanced Settings*.

- Similarly, you can create a group of *sequential* and *not-sequential* FXO Ports, Mobile Ports and SIP Trunks.

- To create a routing group of *sequential* **BRI Channels** as members,



  - Select the **BRI Port** Number. Default: 1.

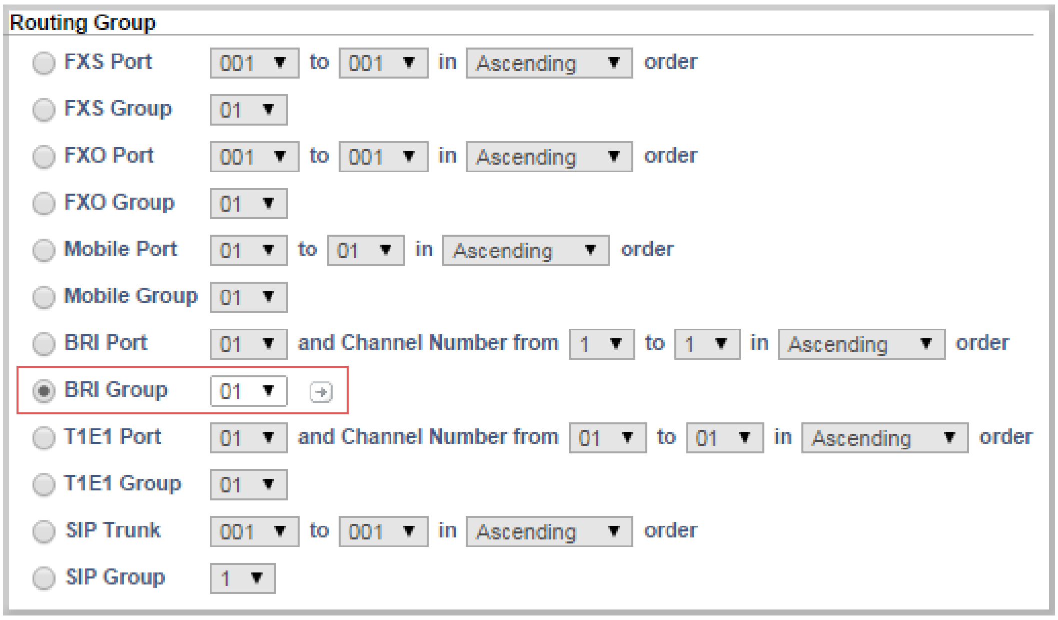
  - In Channel Number **From - to**, select the **Start Channel Number** and the **End Channel Number** respectively.

  - In **in - order**, select the order in which the system should hunt for a free member Channel to route the call.

    Select **Ascending** to start hunting from the first to the last member channel. Select **Descending** to start hunting from the last to the first member channel. Default: Ascending.

- To create a group of *not-sequential* **BRI Channels** as members,

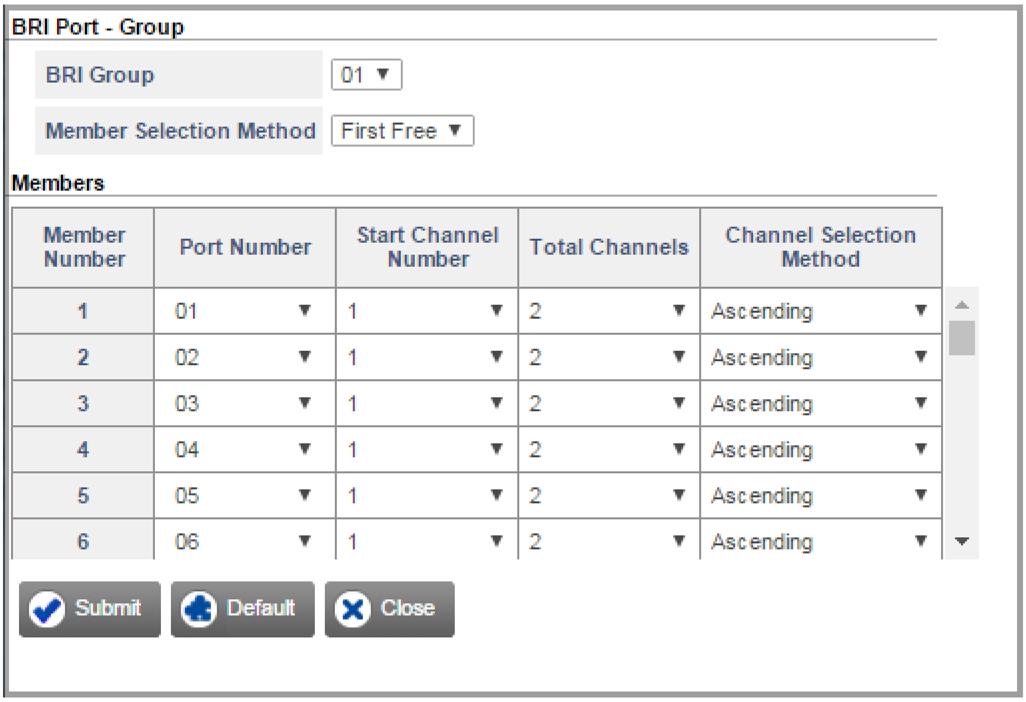
- Select **BRI Group**.



- Select a **BRI Group** number. Default: 1.

- Click **Settings** .

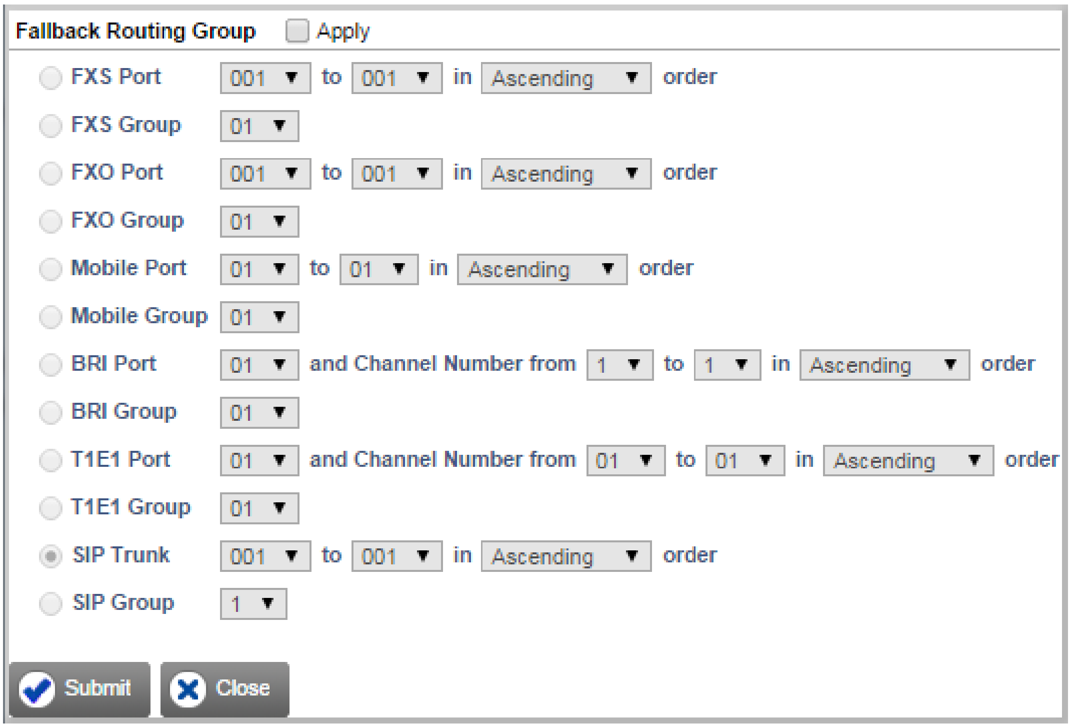
- The **BRI Port - Groups** window opens.



- Create the BRI Group. For detailed instructions on creating groups, see the topic "Group" under *Advanced Settings.*

- Similarly, you can create a group of *sequential* and *not-sequential* T1E1 Ports.

- You may create the **Fallback Routing Group**.



- To do this,
    - Select the **Apply** check box.
    - Follow the same instructions provided earlier for creating *sequential* and *not-sequential* groups of FXS Ports, FXO Ports, Mobile Ports, BRI Ports, T1E1 Ports and SIP Trunks.
- Click **Submit** to save changes. The **Add Entry** window closes.
- The entry you added appears in the **Mobile Port - Destination Port Determination - Destination Number Based** table.
- Follow the same steps as above to add another entry to this table.
- To delete an entry, select the check box and click the **Delete** button.

If there are multiple entries in the Destination Number Based table, to search a particular entry in the table, under Testing enter the desired number in the ***Enter the destination number to know which entry would be selected for routing*** *search box.*

- By default, SIP Trunk 1-1 (Ascending) is assigned as the Routing Group, for routing calls from numbers that do not match with any of the destination numbers you configured (No Match Found).

  To change the default Routing Group and to create the Fallback Routing Group for the No Match Found numbers entry,

    - For the No Match Found entry in the table, click **Edit**.

- The **Edit Entry** window opens.
- Create the **Routing Group** and **Fallback Routing Group** as per your requirement.
- Click **Submit** and close the window.

- Close the window if you have finished adding/editing entries.

  You can also configure the **Destination Number Based** Table from *Advanced Settings*. For instructions, see “Destination Port Determination” under *Advanced Settings.*

On the basis of Calling Party Number

In this method, incoming calls on the Mobile Port are routed to a specific port as per the calling party's number.

To apply this method, do the following:

- In **Select Destination Port for routing calls**, select **On the basis of Calling Party Number** option.

- Click **Settings** .

Select Destination Port for routing calls	On the basis of Calling Party Number ▼
Allowed-Denied Logic	Apply
Reject Calls from Blacklisted Callers	Apply

The **Mobile Port - Destination Port Determination - Calling Number Based** table window opens.

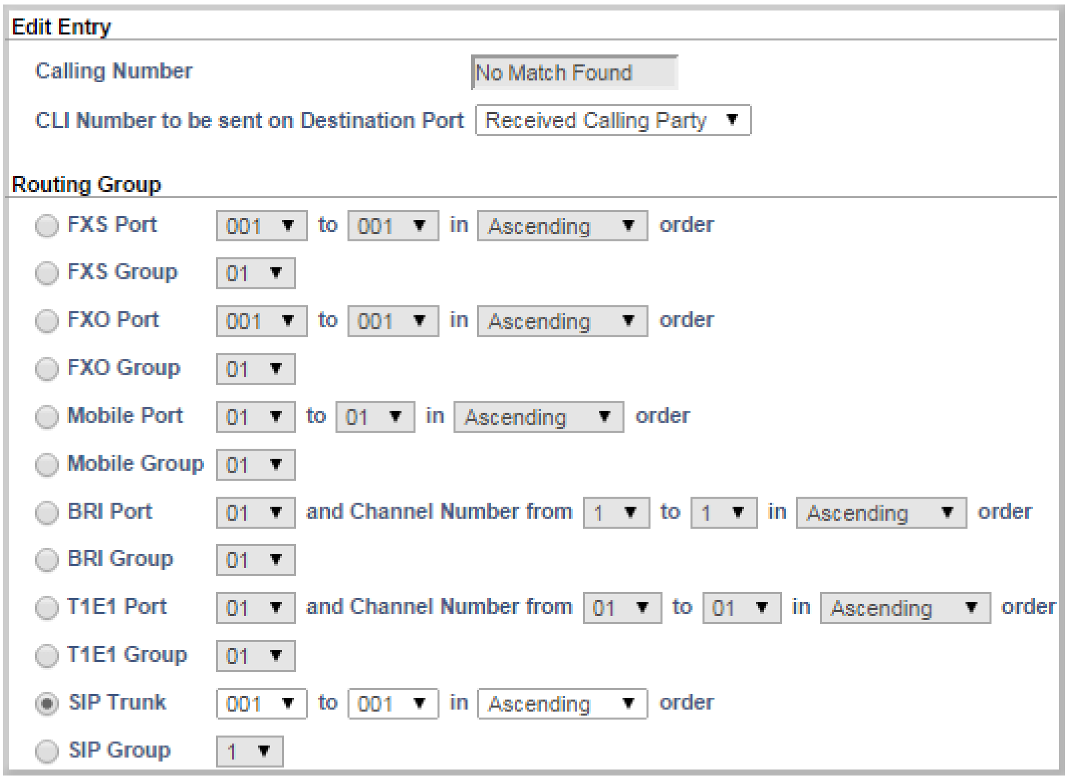
Mobile Port - Destination Port Determination - Calling Number Based

	Edit	Calling Number	Routing Group	Fallback Routing Group	CLI Number to be sent on Destination Port
		No Match Found	SIP Trunk 1 - 1 (Ascending)	None	Received Calling Party

Total Records : 1 1

Add Delete Close

- To add a new entry, click **Add**. The **Add Entry** window opens. You can add upto 499 entries.



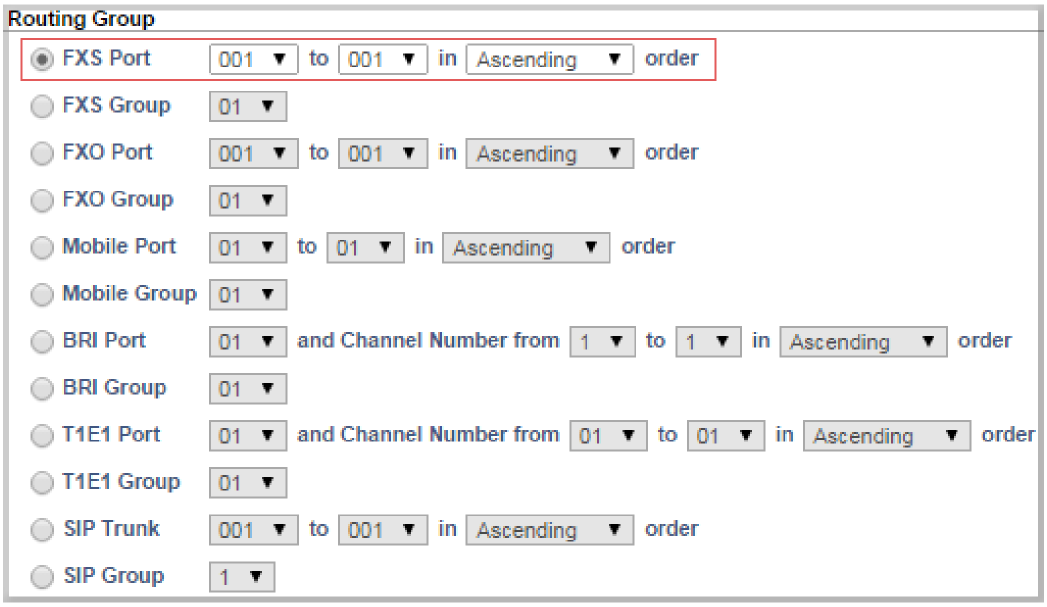
- In **Calling Number**, enter the number (max. 24 characters) from which you expect calls to be received. Valid digits are 0 to 9, *, #, +. Default: Blank.

- Select the **CLI Number to be sent on Destination Port**. You can select Received Calling Party or Received Called Party. Default: Received Calling Party.

CLI to be sent on Destination Port is applicable when FXS Port/Group is determined as the Destination Port.

- Create the **Routing Group**.

- To create a group of *sequential* **FXS Ports** as members,



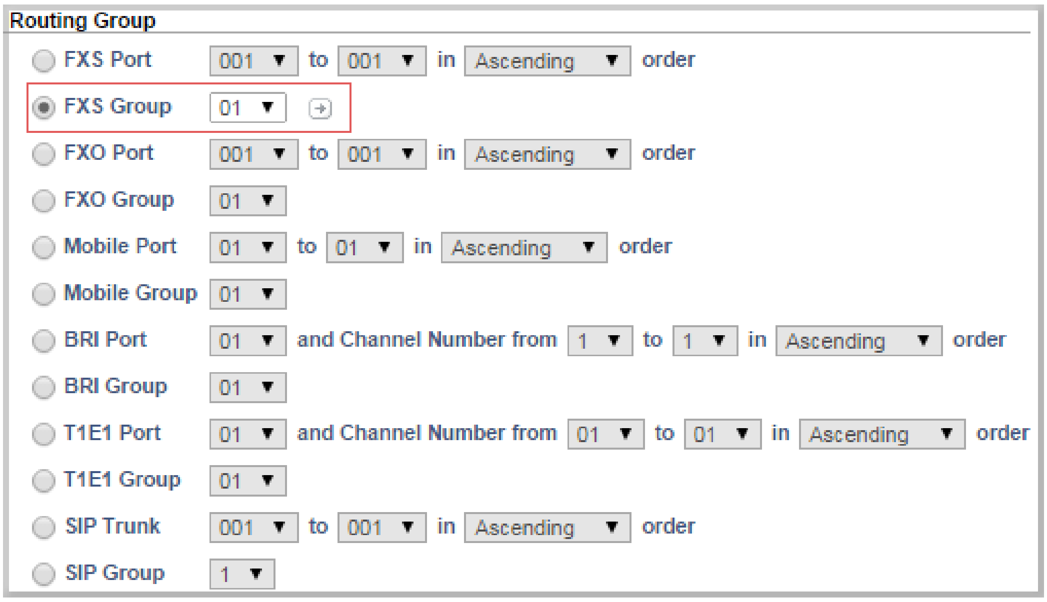
  - Select the desired **FXS Port** numbers as members. Default: 1.

  - In **in - order**, select the order in which the system should hunt for a free member FXS Port to route the call.

    Select **Ascending** to start hunting from the first to the last member FXS Port. Select **Descending** to start hunting from the last to the first member FXS Port. Default: Ascending.

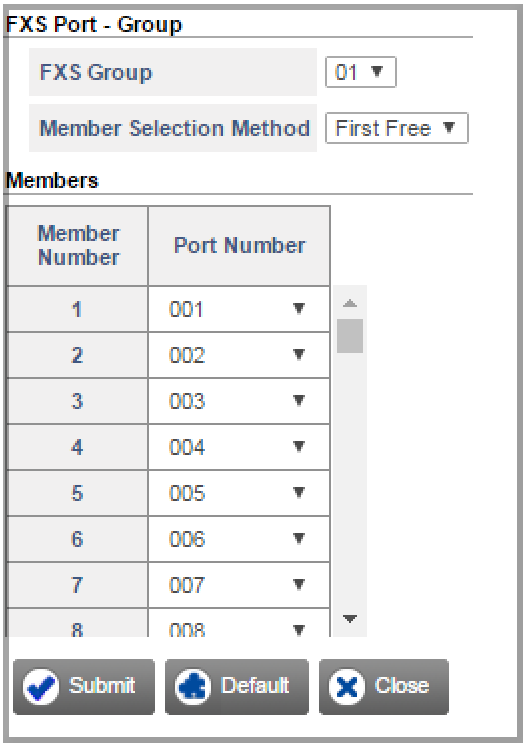
- To create a group of *not-sequential* **FXS Ports** as members,

  - Select a **FXS Group**.



  - Select **FXS Group** number. Default: 1.

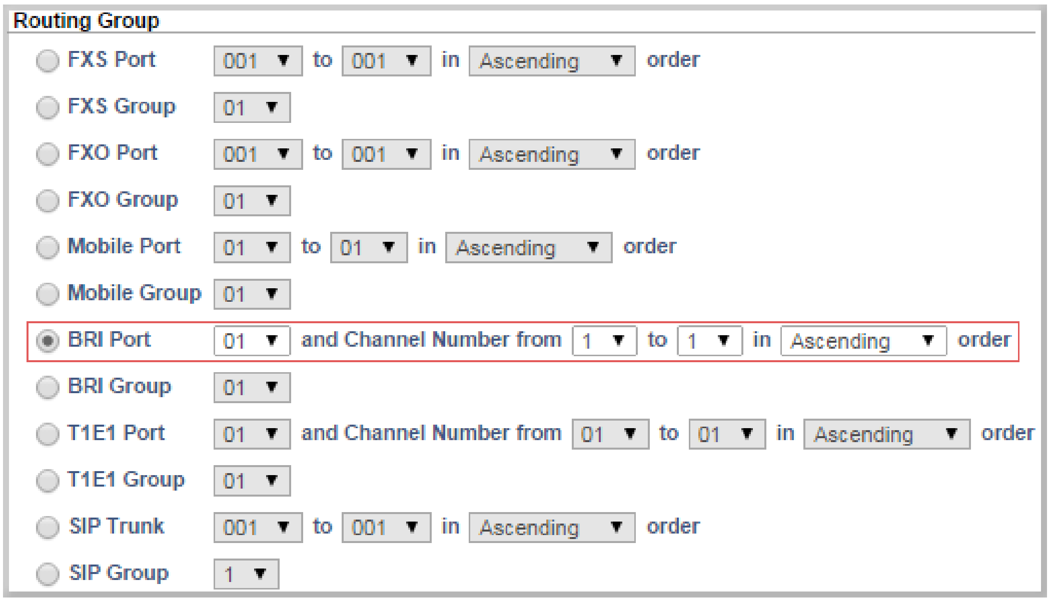
- Click **Settings** .
- The **FXS Port - Groups** window opens.



- Create the FXS Group. For detailed instructions on creating groups, see the topic "Group" under *Advanced Settings*.

- Similarly, you can create a group of *sequential* and *not-sequential* FXO Ports, Mobile Ports and SIP Trunks.

- To create a routing group of *sequential* **BRI Channels** as members,



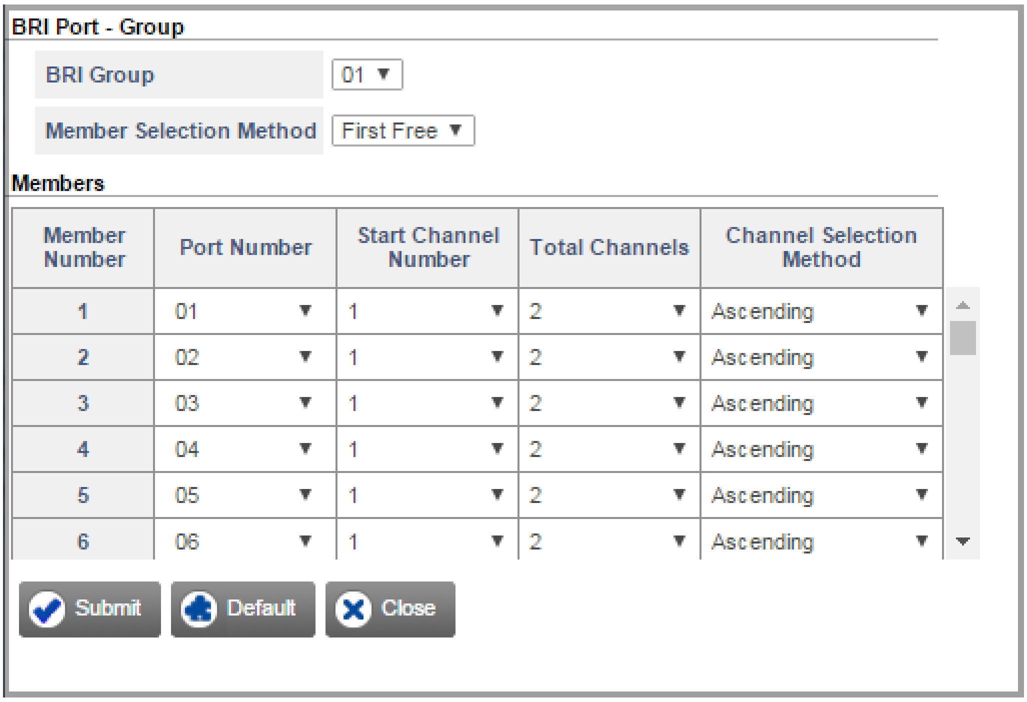
- Select the **BRI Port** Number. Default: 1.
- In Channel Number **From - to**, select the **Start Channel Number** and the **End Channel Number**, respectively.
- In **in - order**, select the order in which the system should hunt for a free member Channel to route the call.

  Select **Ascending** to start hunting from the first to the last member channel. Select **Descending** to start hunting from the last to the first member channel. Default: Ascending.

- To create a group of *not-sequential* **BRI Channels** as members,
  - Select **BRI Group**.

  - Select a **BRI Group** number. Default:1.
  - Click **Settings** .

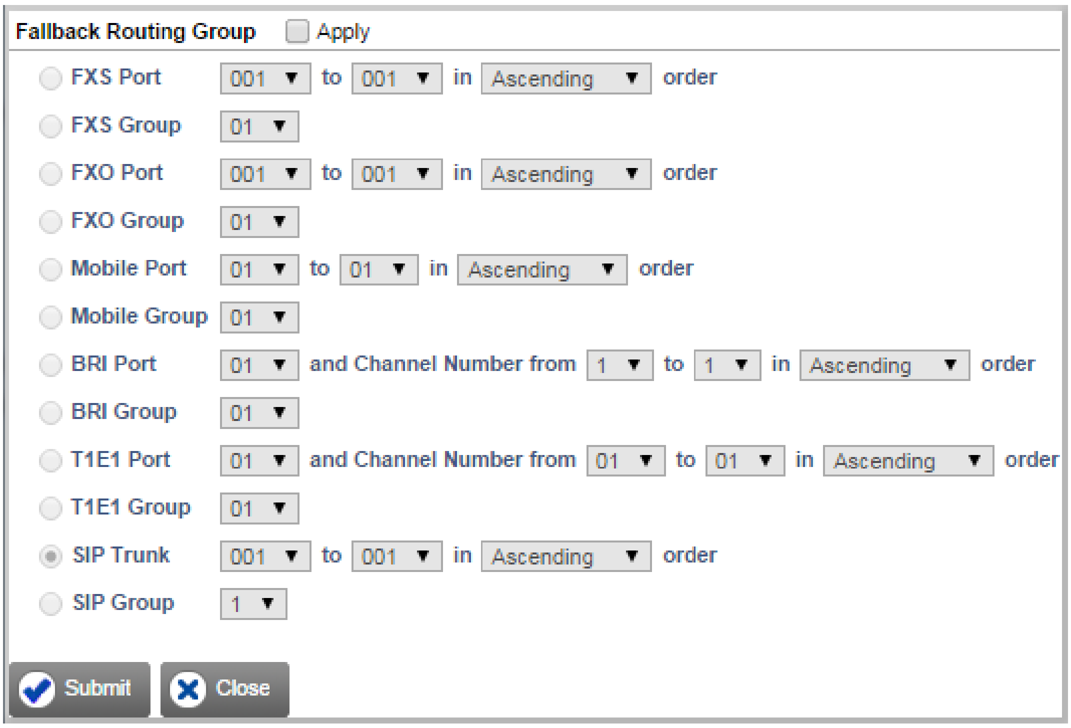
- The **BRI Port - Groups** window opens.



- Create the BRI Group. For detailed instructions on creating groups, see the topic "Group" under *Advanced Settings.*

- Similarly, you can create a group of *sequential* and *not-sequential* T1E1 Ports.

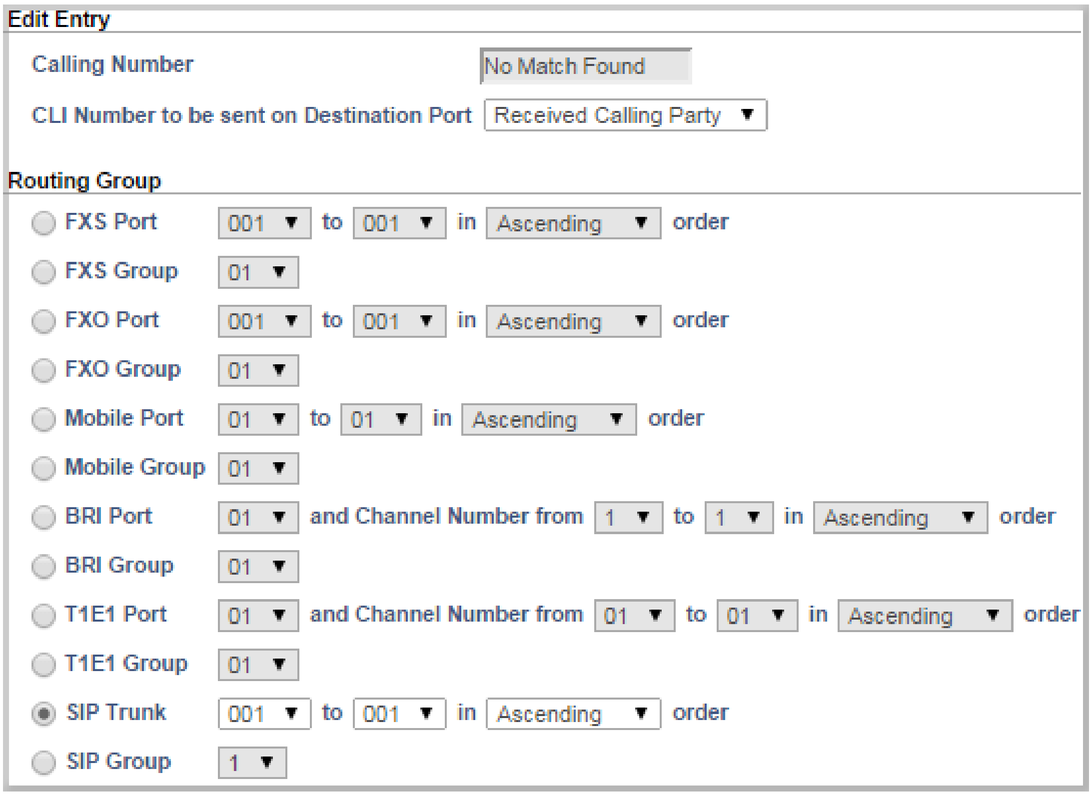
- You may create the **Fallback Routing Group**.



- To do this,
    - Select the **Apply** check box.
    - Follow the same instructions provided earlier for creating *sequential* and *not-sequential* groups of FXS Ports, FXO Ports, Mobile Ports, BRI Ports, T1E1 Ports and SIP Trunks.
- Click **Submit** to save changes. The **Add Entry** window closes.
- The entry you added appears in the **Mobile Port - Destination Port Determination - Calling Number Based** table.
- Follow the same steps as above to add another entry to this table.
- To delete an entry, select the check box and click the **Delete** button.
- By default, SIP Trunk 1-1 (Ascending) is assigned as the Routing Group, for routing calls from numbers that do not match with any of the destination numbers you configured (No Match Found).

  To change the default Routing Group and to create the Fallback Routing Group for the No Match Found numbers entry,

    - For the No Match Found entry in the table, click **Edit**.



    - The **Edit Entry** window opens.
    - Create the **Routing Group** and **Fallback Routing Group** as per your requirement.
    - Click **Submit** and close the window.

- Close the window if you have finished adding/editing entries.

  You can also configure the **Calling Number Based** Table from *Advanced Settings*. For instructions, see "Destination Port Determination" under *Advanced Settings.*

Allowed - Denied Logic

You can apply the Allowed-Denied logic on the Mobile Port (source port) if you want to allow or restrict the dialing of particular numbers. You can use this feature for Toll Control.

The Allowed-Denied Number Logic makes use of two Number lists:

- **Allowed Numbers List:** This is the list of numbers that can be dialed by the caller.

- **Denied Numbers List:** This list contains the numbers that are to be restricted from being dialed by the caller on the Mobile Port.

When Allowed-Denied Logic is enabled on a source port, for each number dialed from the port, SARVAM UMG uses the best-match-found logic to compare the dialed number with the Allowed Number list and the Denied Number list.

The number is allowed to be dialed, if it:
- matches with both lists.
- matches with Allowed Number list, but not with the Denied Number list.
- matches with neither the Allowed List nor the Denied List.

The number is denied, if it matches with the Denied Number list, but not with the Allowed Number list.

The system does not apply the Allowed-Denied Logic:
- When dialed number string matches with any Access Code.
- When dialed number string matches with any Emergency Number.
- When any one of the following is selected to Route all Incoming Calls (with CLI):
  - on the basis of Calling Party Number
  - to a Fixed Destination Number

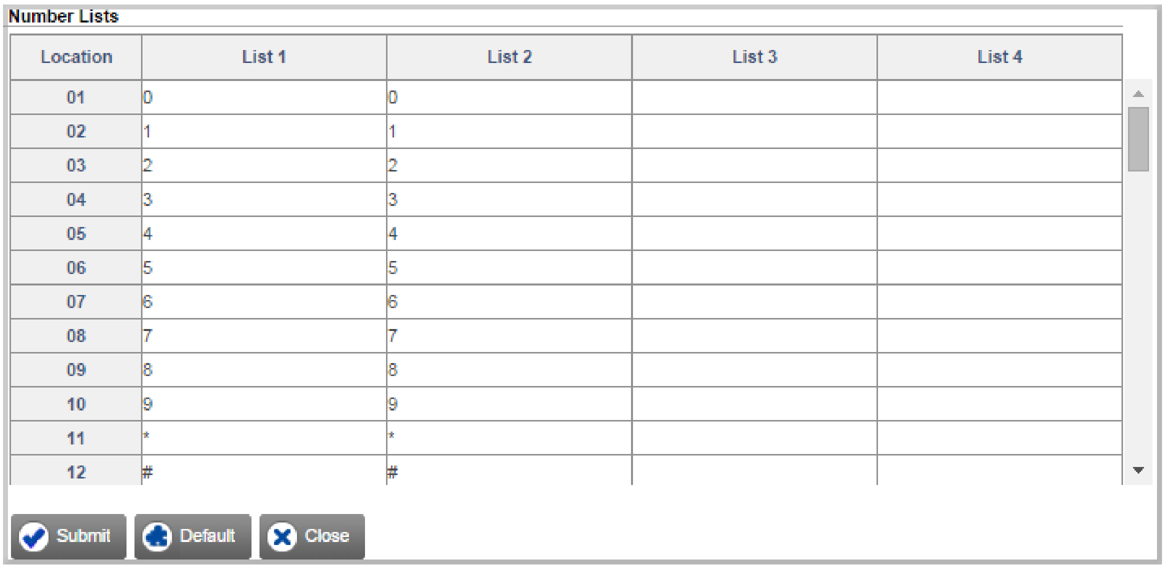
To apply Allowed - Denied Logic on the Mobile Port,

- Select the **Allowed - Denied Logic** check box.

  - In the **Allowed Number List**, select the list number you have configured with numbers you want to allow to be dialed out from the Mobile Port. Default: 01

    If you have not configured the Allowed Number List,

- Click **Settings** . The Number Lists window opens.



Number Lists

Location	List 1	List 2	List 3	List 4
01	0	0		
02	1	1		
03	2	2		
04	3	3		
05	4	4		
06	5	5		
07	6	6		
08	7	7		
09	8	8		
10	9	9		
11	*	*		
12	#	#		

Submit Default Close

- You may configure the default Allowed Number List or any other list. See "Number Lists" to configure the allowed numbers.

- Click **Submit** to save the Allowed Number List and close the window.

- In the **Denied Number List**, select the list number you have configured with numbers you want to restrict to be dialed out from the Mobile Port. Default: 02.

  If you have not configured the Denied Number List,

  - Click **Settings** . The Number Lists window opens.

  - You may configure the default Denied Number List or any other list. See "Number Lists" to configure the restricted numbers.

  - Click **Submit** to save the Denied Number List and close the window.

Black Listed Callers

With the Black Listed Callers feature you can block incoming calls from specific numbers on the Mobile Port. Thus all incoming calls from the numbers you have 'blacklisted' will be automatically rejected by SARVAM UMG.

To apply Black Listed Callers on Mobile Port,

- Select the **Reject Calls from Blacklisted Callers** check box.

- In the **Blacklisted Callers Number List**, select the list number you have configured with the numbers of unwanted callers. Default:16

  If you have not configured the Blacklisted Callers Number List,

  - Click **Settings** . The Number Lists window opens.

- You may configure the default Blacklisted Callers Number List or any other list. See “Number Lists” to configure the numbers of unwanted callers.

Separator for Called Party Number

With the Separator feature, the system can differentiate the Calling and the Called Number when there is an incoming call on the Mobile port of SARVAM UMG.

The CLIP format received from the Service Provider supporting the Separator Feature will be displayed as *Calling Number(Separator)Called Number.*

When the separator provided by the service provider matches with the separator configured in the system, the incoming call will be routed to the destination port using Called Number received or a dial tone will be provided to the caller if Called Number matches the configured number in *Destination Number List for Manual Dial*.

If the separator is not received or does not match, then calls will be routed according to the incoming call routing set for the port, that is, the Destination Number Determination and Destination Port Determination methods configured.

To apply the Separator feature on the Mobile Port,

- Select the desired option in **Separator to be used for receiving Called Party Number.** You can select — ‘#’ , ‘*’ or Don’t Use. Default: Don’t Use.
    - When you select ‘#’ or ‘*’, **Destination Number List for Manual Dial** is displayed. Select the list number you have configured with numbers for which you want the system to provide the caller with a dial tone. Default: None. When the dial tone is played to the callers, s/he can manually dial the destination number and call will be routed according to the configuration in “Route After Answering the Call and Collecting the Digits”.

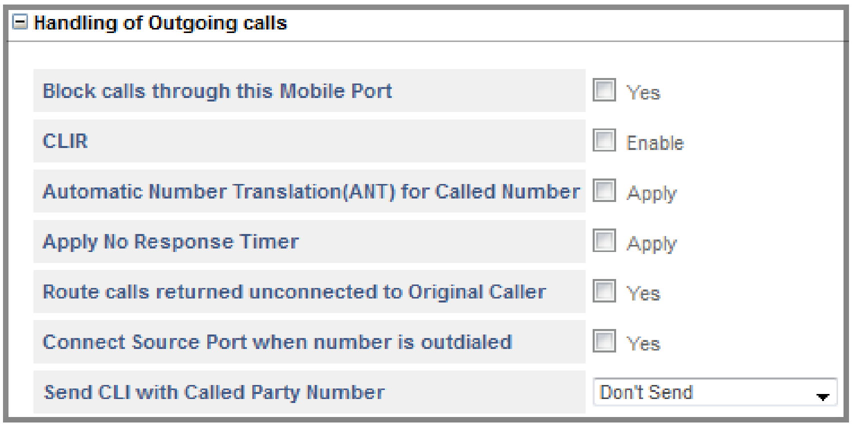
      If you have not configured the Destination Number List for Manual Dial,

    - Click **Settings** . The Number Lists window opens.
    - You may configure any list. See “Number Lists” to configure the numbers for Manual Dial.
- Click **Submit** and close the window.

Handling of Outgoing Calls

Click **Handling of Outgoing Calls** to expand.

When Mobile Port is determined as the destination port, numbers dialed from this port constitute outgoing calls.

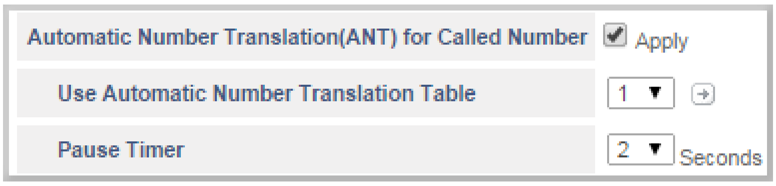


- If you do not want to route outgoing calls though this Mobile Port, select the **Block calls through this Mobile Port** check box. Default: Disabled.

- By default, the CLI of the Mobile Port is sent to the called party when outgoing calls are made using the Mobile Port. If you do not want to send CLI, enable the **CLIR** check box. Default: Disabled.

- You can apply **Automatic Number Translation logic** on outgoing calls made from the Mobile Port.

  - To apply ANT logic on the Called Numbers, select the **Automatic Number Translation (ANT) for Called Number** check box. Default: Disabled.

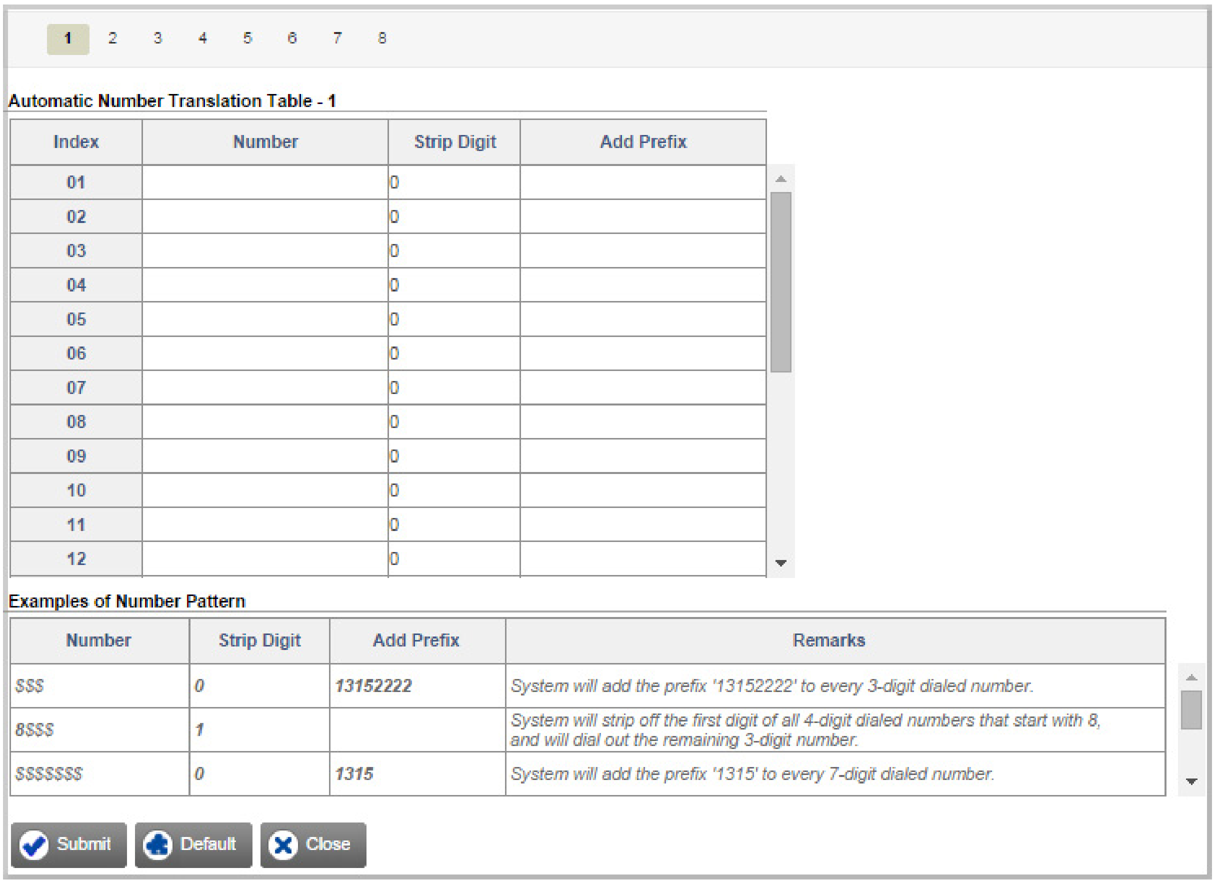


  - In **Use Automatic Number Translation Table**, select the ANT Table number you have configured for the Mobile Port. Default: Table 1.

    If you have not configured the Automatic Number Translation Table,

    - Click **Settings** .

- The **Automatic Number Translation Table** window opens.



- You may configure the default Automatic Number Translation Table or any other Table. See "Automatic Number Translation (ANT)" to configure the ANT Table.

- Click **Submit** to save the ANT Table and close the window.

- Return to ANT parameter and assign the ANT Table you configured.

- Set the duration of the **Pause Timer**, if you have configured ^ (Pause) in the Add Prefix column of the ANT Table. Valid range is 1 to 9 seconds. Default: 2 seconds.

- Select the **Apply No Response Timer** check box. The system will route the call through a Fallback Routing Group or Port, if a response other than—Alert, Connect, Busy, No Reply, Disconnect with cause as Busy or No Reply—is received from the network within the specified time period. Default: Disabled.

  - Set the duration of the **No Response Timer**, if you have enabled the *Apply No Response Timer* option. It is the time for which SARVAM UMG will wait for the valid response from the network for any request. If no valid response is received before the expiry of this timer, SARVAM UMG will fallback to alternate Routing Group or Port for further processing of the call. Valid range is 01 to 99 seconds. Default: 10 seconds.

To apply Fallback logic on the Mobile Port, make sure you have enabled Fallback Routing Group under "Destination Port Determination".

- Enable **Route calls returned unconnected to Original Caller**, if you want SARVAM UMG to route outgoing calls made from this port that return unconnected back to the original caller.

If you enable this feature, when an outgoing call is made using this port, and the Called Party is found busy or does not respond, SARVAM UMG stores the number of the calling party, the number of the called party and this port (through which the outgoing call was made). A record of each such call is stored for the duration of the Unconnected Calls Record Delete Timer (configurable; default: 999 minutes).

If the called party returns the call before the expiry of this Timer, SARVAM UMG checks whether *Apply RCOC only if the caller calls back on the same trunk from which the call was made* is enabled or not, and accordingly places the incoming call to the original calling party. To change the duration of this timer, delete records of such calls and enable/disable the *Apply RCOC only if the caller calls back on the same trunk from which the call was made* check box, see "System Parameters".

- To connect the Source Port with the Destination Port without waiting for the call on the Destination Port to mature, enable the **Connect Source Port when number is outdialed** check box. Default: Disabled.

  In all Destination Number Determination methods except *After Answering the Call and Collecting the Digits*, the Source Port gets connected to the Destination Port only after the call has matured, that is, the called party has answered the call. Until the call matures, the caller hears only Ring Back Tone played by the network.

  By connecting the Source Port with the Destination Port immediately after the number is dialed, the caller can know the state of the call; if the called party is busy, not responding, not reachable or is rejecting the call.

- To send the separator and Calling Number in CLI while placing an outgoing call through a Mobile Port, in **Send CLI with Called Party Number**, select **—** Don't Send, Send with '#' Separator, Send with '*' Separator. Default: Don't Send.

  - When you select *Send with '#' Separator* or *Send with '*' Separator* then select the **Maximum digits of Called Party Number for sending CLI.** Default: 5. If the Called Number is greater than 5 digits then system will not append the separator and Calling Number with the Called Number.

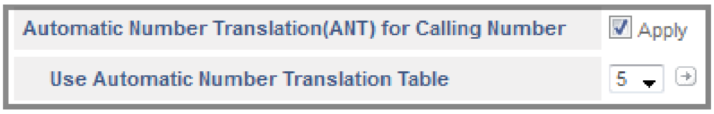
    For example 1: Called number-9898, Calling number-4545 then the system will dial out 9898#4545.

    For example 2: Called number 9898989898, Calling number-4545, then the system will dial out only 9898989898 because the Called number digits are greater than the *Maximum digits of Called party for sending CLI.*

When you have enabled this feature, the CLI format will be dialed out as Called Number(Separator)Calling Number.

  - You can apply **Automatic Number Translation** logic on outgoing calls made from the Mobile Port.

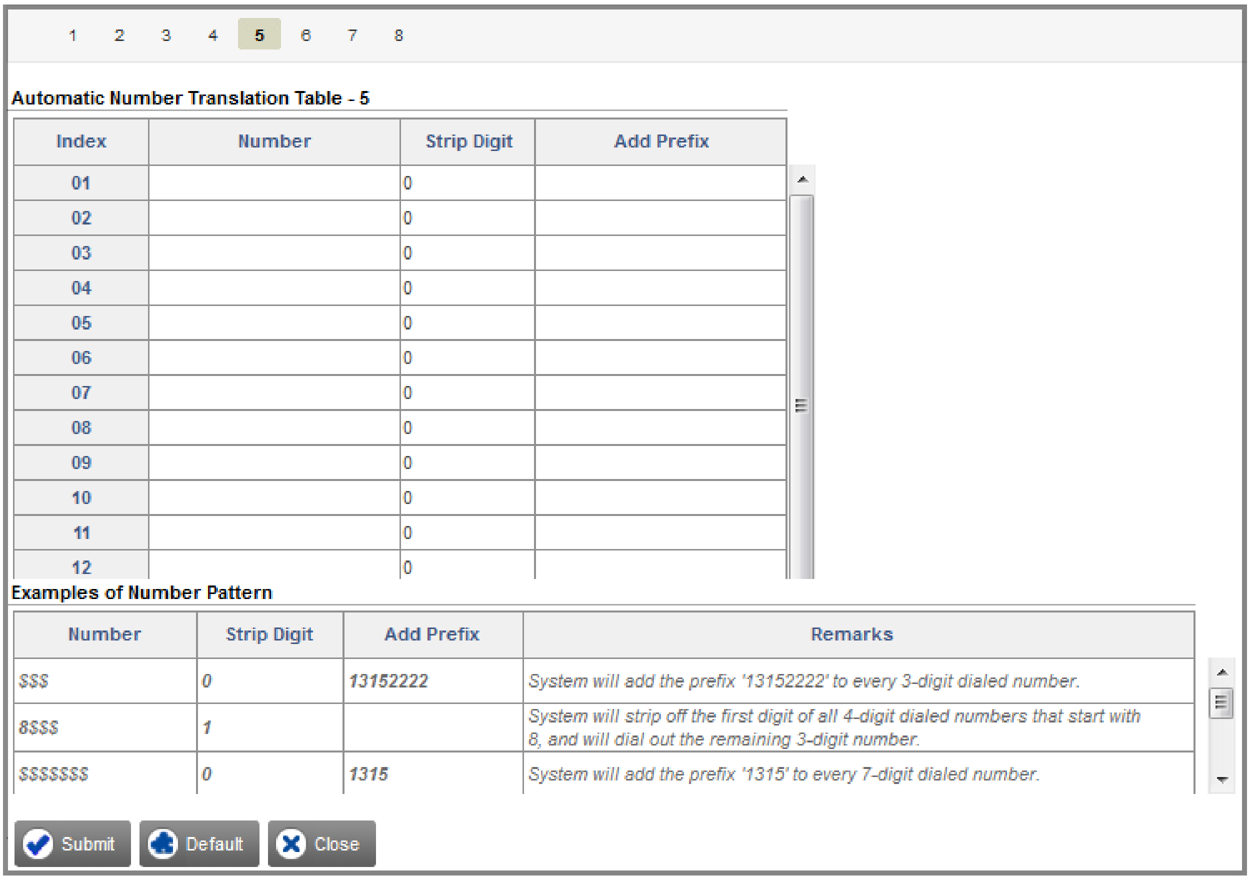
- To apply ANT logic on the Calling Numbers, select the **Automatic Number Translation (ANT) for Calling Number** check box. Default: Disabled.



- In Use Automatic Number Translation Table, select the ANT Table number you have configured for the Mobile Port. Default: Table 5.

  If you have not configured the Automatic Number Translation Table,

- Click **Settings** . The **Automatic Number Translation Table** window opens.



- You may configure the default Automatic Number Translation Table or any other Table. See “Automatic Number Translation (ANT)” to configure the ANT Table.

- Click **Submit** to save the ANT Table and close the window.

- Return to ANT parameter and assign the ANT Table you configured.

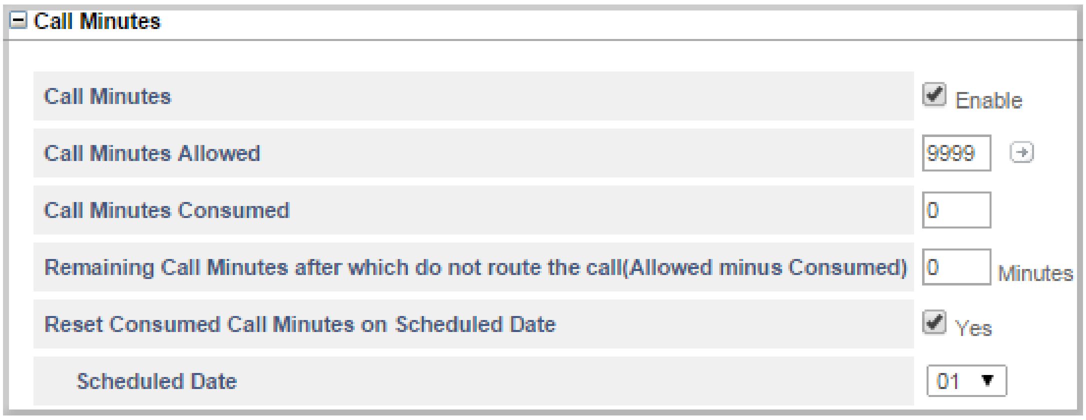
- Click **Submit** to save.

*If you enable **Connect Source Port when number is outdialed**, you will not be able to provide the features “Making a New Call using Access Code” and “Disconnecting a Call using Access Code” to users.*

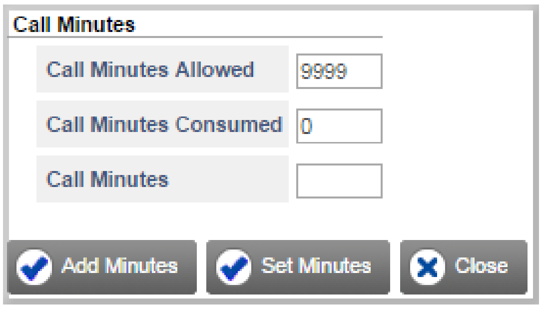
Call Minutes

Mobile Service Providers offer different tariff schemes to their subscribers. For example, mobile service providers in India offer first 500 Minutes free, CUG calling, first 500 minutes calling at 30 paise. SARVAM UMG allows you to take advantage of these tariff schemes by configuring Call Minutes on the Mobile Port.

- Click **Call Minutes** to expand.
- Select the **Call Minutes** check box to apply this feature.



- The **Call Minutes Allowed** displays the minutes allotted to the Mobile Port for making outgoing calls. Valid range is 0000 to 9999. Default: 9999.
- To change the value of Call Minutes Allowed,
  - Click **Settings** .



  - The **Call Minutes** window opens.
    - In **Call Minutes**, enter the desired value.
    - Click the **Set Minutes** button. The value entered appears in the **Call Minutes Allowed**.
  - If you want to add minutes to the existing value,
    - In **Call Minutes**, enter the desired value you want to add.
    - Click the **Add Minutes** button. The value you entered gets added to the existing minutes and appears in the **Call Minutes Allowed**.

- Close the window to return to the main page.
- The **Call Minutes Allowed** displays the Call Minutes set/added by you.
- The minutes consumed are displayed in **Call Minutes Consumed**.
- You may block the outgoing calls from the Mobile Port after certain call minutes have been consumed. To do so, configure the **Remaining Call Minutes after which do not route the call (Allowed minus Consumed).** The system will block the outgoing calls made from this port when the call minutes you configured are remaining. Valid range is 000 to 999. Default: 000.
- Select the **Reset Consumed Call Minutes on Scheduled Date** check box, if you want the system to automatically reset the value of Call Minutes on a scheduled date. Also select the desired **Schedule Date**.

SIM Balance and Recharge

SARVAM UMG supports Balance Inquiry and Recharging of the SIM Card installed in its Mobile Ports[13].

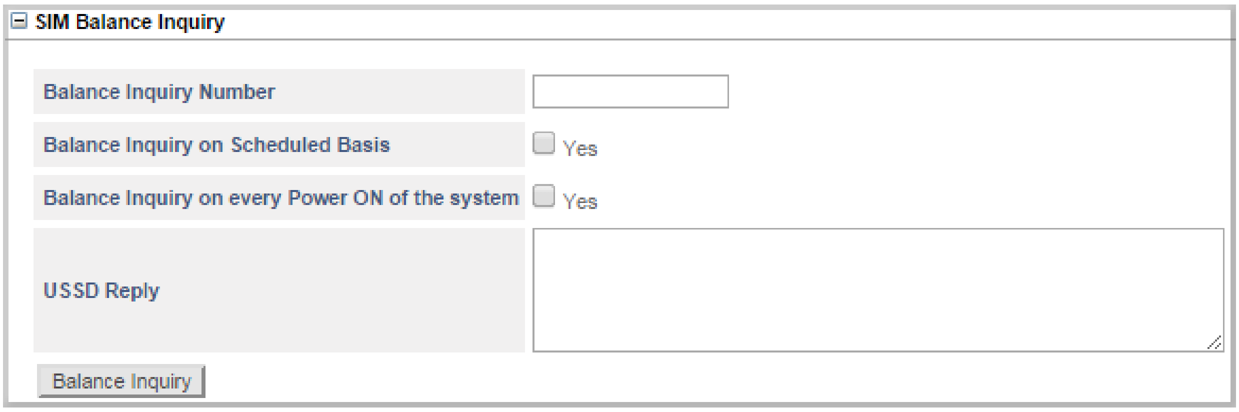
To be able to use this feature, first collect the following information from your Network Operator:

- **Balance Inquiry Number:** This is the number provided by the Network Operator to the subscribers to check Balance. Different Network Operators have different numbers. For example, the Balance Inquiry number of Vodafone is ***141#.***
- **Recharging Service Number:** This is the number provided by the Network Operators to their subscribers for Recharging Service. Different Network Operators have different numbers for Recharging Service. For example, the Recharging Service Number of Vodafone is ***140***.

SIM Balance Inquiry

To check the SIM Balance using Jeeves,

- Click **SIM Balance Inquiry** to expand.

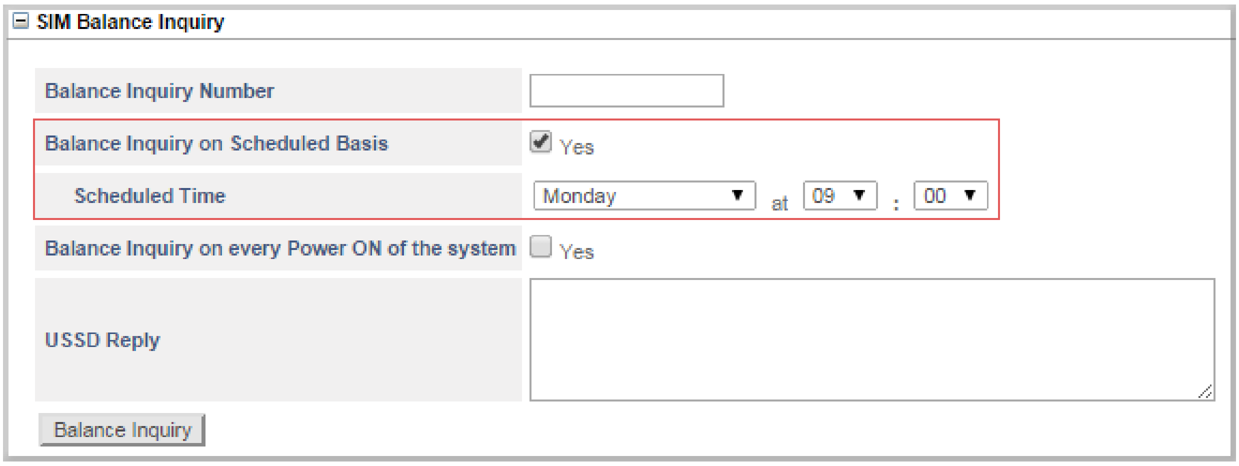


- In **Balance Inquiry Number**, enter the number provided to you by the Network Operator to check Balance. The number can be a maximum of 16 digits. The digits allowed are 0-9, * and #.
- Click **Submit.**

13. *SARVAM UMG 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.*

- To run the query, click the **Balance Inquiry** button.

- If you want the system to check the SIM Balance at fixed intervals, select the **Balance Inquiry on Scheduled Basis** check box and configure the following:



  - In **Schedule Time - at**, enter the day and the time when you want the system to make the Balance query.
- Select the **Balance Inquiry on every Power ON of the system** check box, if you want the system to make the SIM Balance query at every Power ON.

- The response received from the mobile network (including possible error messages) will be displayed under **USSD Reply**.

SIM Recharge

To Recharge the SIM,

- Click **SIM Recharge** to expand.

- In **Recharge Number**, enter the number provided to you by the Network Operator to recharge the SIM. The number can be a maximum of 8 digits. The digits allowed are 0-9, * and #.

- Click **Submit.**

- Click the **Recharge** button.

- A new window opens. In **Enter Recharge PIN Number**, enter the number printed on the Recharge Voucher.

- Click **OK**.

- The response received from the mobile network (including possible error messages) will be displayed under **USSD-Reply**.

BCCH Locking

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 UMG 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.

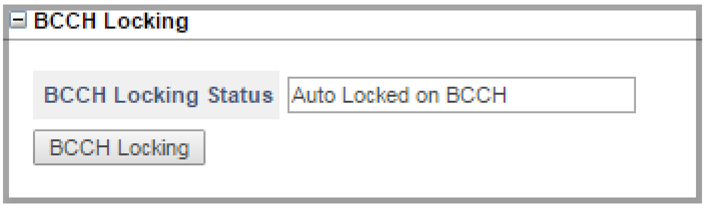
SARVAM UMG enables you to lock the Mobile Port to a particular cell or channel or BTS (Base Transceiver Station) for various reasons such as:
- better network availability
- minimum call drop due to bad signal/ network failure, etc.

- *To use this feature, make sure GSM or LTE is selected as the **Preferred Network Mode** of the Mobile Port.*

- *BCCH Locking will not be supported if Quectel UC20 (3G module) is installed in your system.*

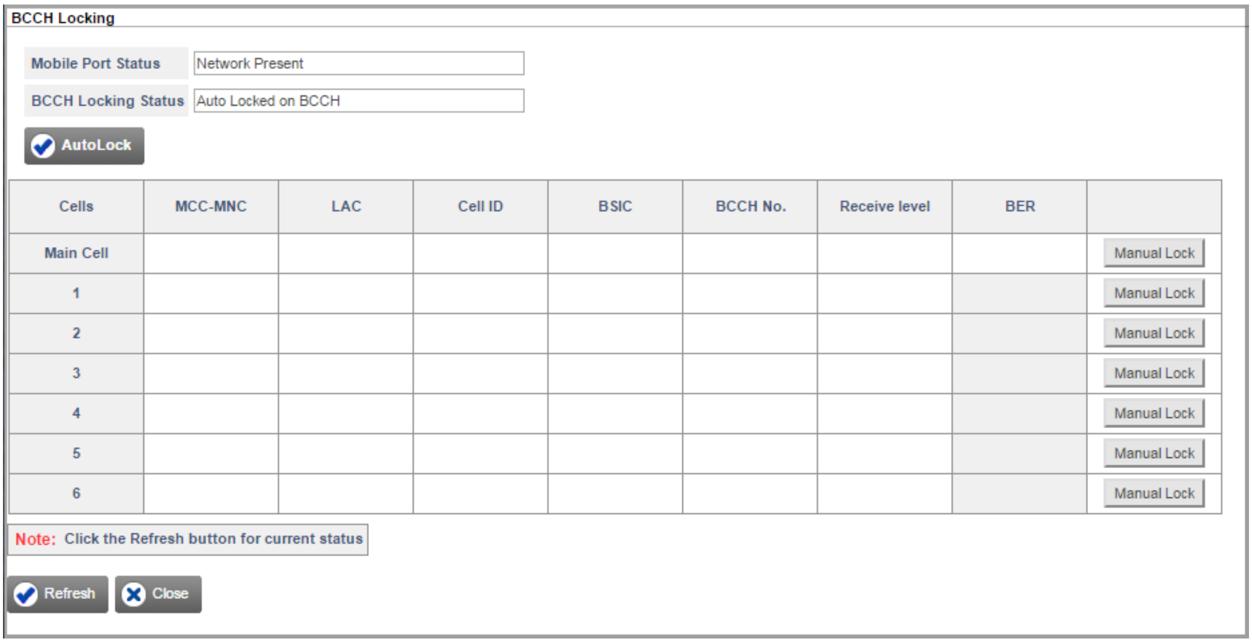
To select the BCCH Locking Mode for the Mobile Port,

- Click **BCCH Locking** to expand.



- By default, the port is Auto Locked with the Main Cell which has the highest signal strength.

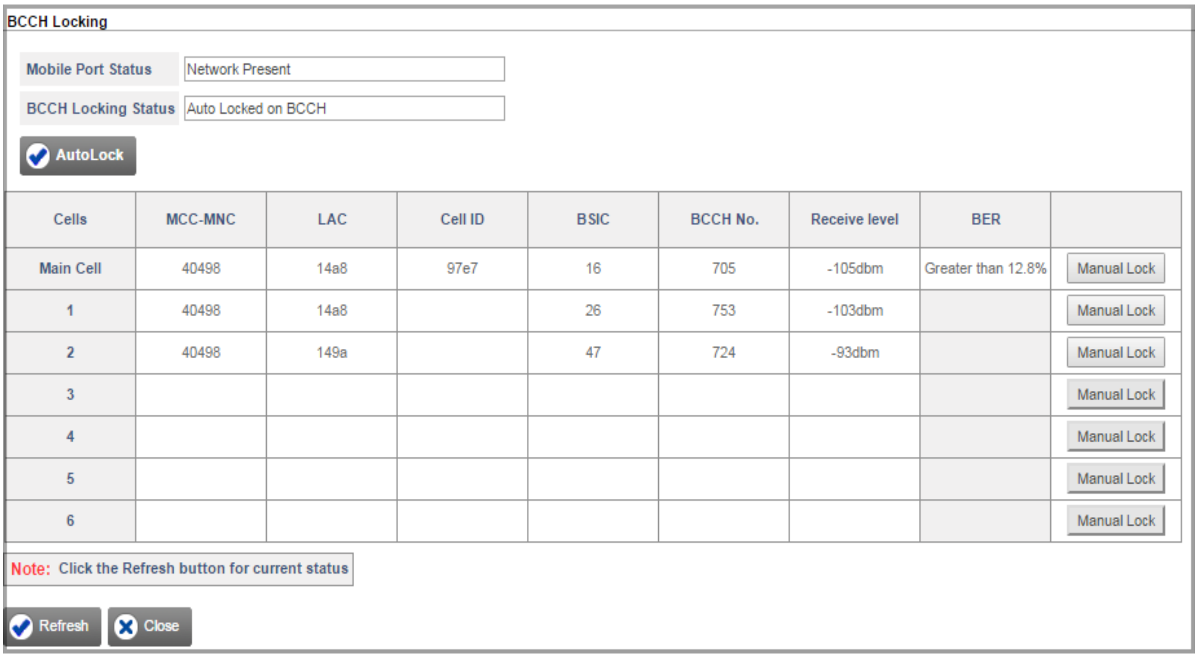
- To check the signal strength and/or change the locking mode, click **BCCH Locking**. The BCCH Locking page opens.



Cells	MCC-MNC	LAC	Cell ID	BSIC	BCCH No.	Receive level	BER	
Main Cell								Manual Lock
1								Manual Lock
2								Manual Lock
3								Manual Lock
4								Manual Lock
5								Manual Lock
6								Manual Lock

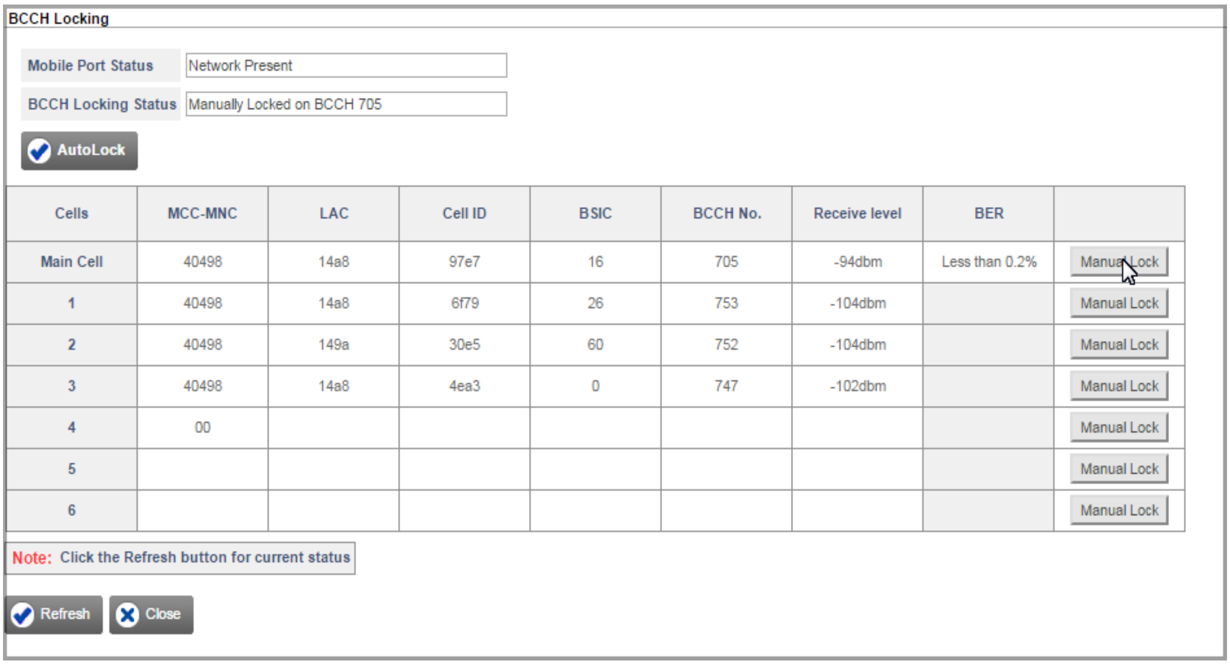
- The **Mobile Port Status** and **BCCH Locking Status** is displayed.

- Click **Refresh**. The system fetches the details of the various cells.



Cells	MCC-MNC	LAC	Cell ID	BSIC	BCCH No.	Receive level	BER	
Main Cell	40498	14a8	97e7	16	705	-105dbm	Greater than 12.8%	Manual Lock
1	40498	14a8		26	753	-103dbm		Manual Lock
2	40498	149a		47	724	-93dbm		Manual Lock
3								Manual Lock
4								Manual Lock
5								Manual Lock
6								Manual Lock

- To lock the port with the desired cell, click **Manual Lock**.



BCCH Locking

Mobile Port Status: Network Present

BCCH Locking Status: Manually Locked on BCCH 705

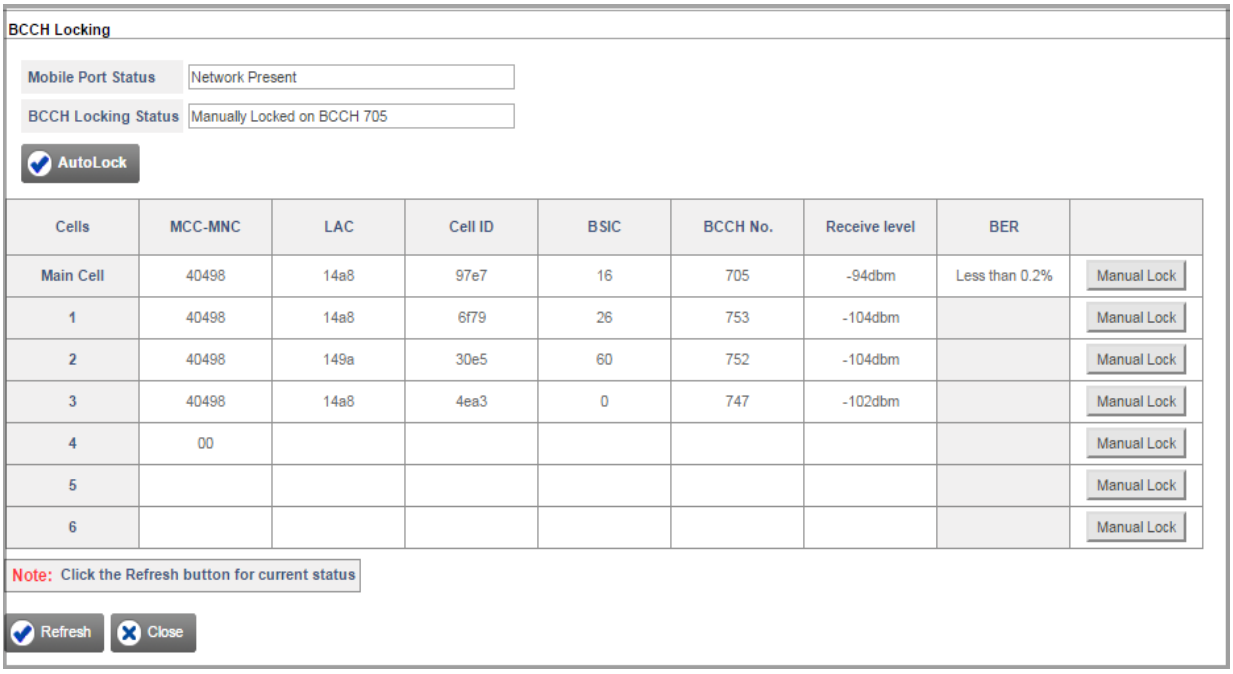
AutoLock

Cells	MCC-MNC	LAC	Cell ID	BSIC	BCCH No.	Receive level	BER	
Main Cell	40498	14a8	97e7	16	705	-94dbm	Less than 0.2%	Manual Lock
1	40498	14a8	6f79	26	753	-104dbm		Manual Lock
2	40498	149a	30e5	60	752	-104dbm		Manual Lock
3	40498	14a8	4ea3	0	747	-102dbm		Manual Lock
4	00							Manual Lock
5								Manual Lock
6								Manual Lock

Note: Click the Refresh button for current status

Refresh Close

- Once the port is manually locked with the desired cell, the BCCH Locking Status is updated.



BCCH Locking

Mobile Port Status: Network Present

BCCH Locking Status: Manually Locked on BCCH 705

AutoLock

Cells	MCC-MNC	LAC	Cell ID	BSIC	BCCH No.	Receive level	BER	
Main Cell	40498	14a8	97e7	16	705	-94dbm	Less than 0.2%	Manual Lock
1	40498	14a8	6f79	26	753	-104dbm		Manual Lock
2	40498	149a	30e5	60	752	-104dbm		Manual Lock
3	40498	14a8	4ea3	0	747	-102dbm		Manual Lock
4	00							Manual Lock
5								Manual Lock
6								Manual Lock

Note: Click the Refresh button for current status

Refresh Close

Now, if you wish to change the cell with which the port is manually locked,

- You must click **Auto Lock**. The page displays the details of the various cells.
- Click **Manual Lock** of the desired cell.

If you want the port to be locked automatically with the cell that has the higher signal strength, click **Auto Lock**.

VoLTE Configuration

VoLTE Configuration will be visible only when 4G SIM is inserted.

Voice over Long Term Evolution (VoLTE) is a standard that is used for high-definition voice calling service. VoLTE offers faster connection speed and advance technology for VoIP calls with better audio clarity and added security.

SARVAMUMG gives you the facility to access 4G LTE network depending upon the GSM module mounted on the Mobile card. This is made possible through an MBN file which is required for voice calling over LTE. Every mobile service provider has a unique MBN file that is already preloaded in the 4G module. The system also provides additional facility to manually upload MBN file of newly emerging mobile service providers in the market that offer VoLTE services.

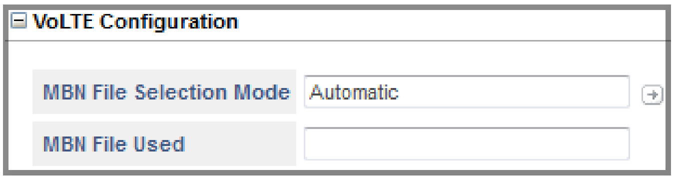
The 4G module of SARVAM UMG offers backward compatibility, thereby enabling the devices to automatically get registered with 3G/2G network when not within the coverage area of 4G LTE network. To know more, refer "The Mobile Card for ETERNITY GENX".

To access 4G LTE network ensure you have performed the following:

- Make sure the Mobile card with 4G module is installed in the system.
- Inserted the Nano SIM cards of the desired service providers.
- Configured the VoLTE parameters.
- Uploaded MBN files. For details refer to "MBN File Upload".
- Configured the Mobile Ports.

To configure LTE (4G) parameters for the Mobile Port,

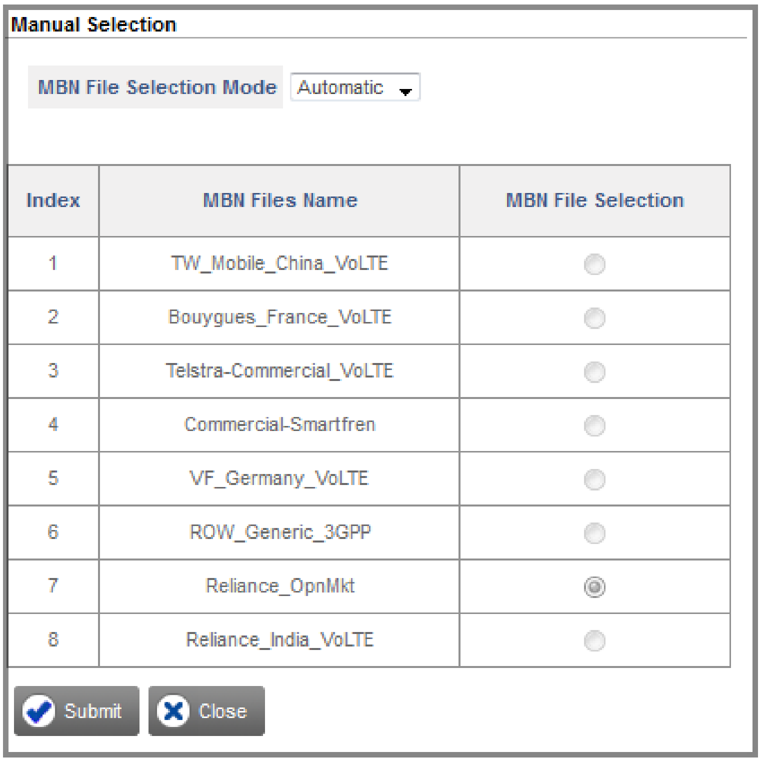
- Click **VoLTE Configuration** to expand.



- **MBN File Selection Mode** displays the selected mode.

- **MBN File Used:** The module stores various MBN files. This displays the name of currently used MBN file for the particular Mobile Port.

- Click **Settings** , to change the selection mode or select the desired MBN files from the list of available MBN files, The **Manual Selection** table opens. It displays all the MBN files supported by the module.



Manual Selection

MBN File Selection Mode: Automatic

Index	MBN Files Name	MBN File Selection
1	TW_Mobile_China_VoLTE	○
2	Bouygues_France_VoLTE	○
3	Telstra-Commercial_VoLTE	○
4	Commercial-Smartfren	○
5	VF_Germany_VoLTE	○
6	ROW_Generic_3GPP	○
7	Reliance_OpnMkt	◉
8	Reliance_India_VoLTE	○

Submit Close

- 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 in the module and the same are displayed in the table.
- When you enable **Automatic** mode for selecting the MBN file, the system automatically selects the MBN file of the mobile service provider depending upon the SIM card inserted into the slot.

  The table is un-editable and displays the selected MBN file.
- When you enable **Manual** mode for selecting the MBN file, then make sure you select the MBN file from the list of available MBN files.

  The table displays all the MBN files supported by the module. Click on the corresponding radio button to select the desired MBN file.

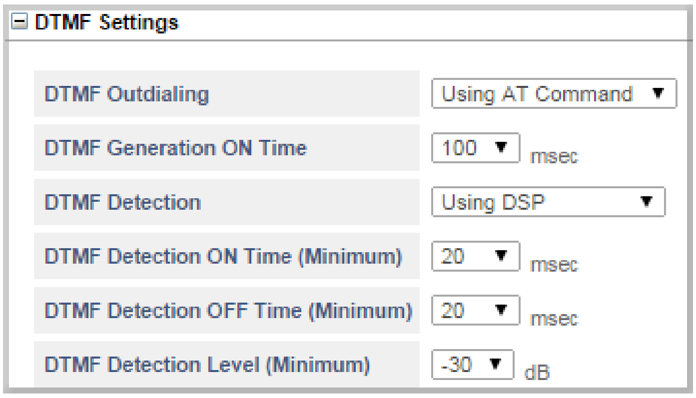
  You can also upload MBN file of a newly emerged VoLTE supporting service provider in the market. To do so, refer “MBN File Upload”. After the file is uploaded it appears in the list.
- Click **Submit** to save your settings.

MBN File Selection Mode and Manual Selection table will be disabled when the 4G SIM is registered with 2G/3G network.

DTMF Settings

To configure the DTMF Settings for the Mobile Port,

- Click **DTMF Settings** to expand.



- Select the desired **DTMF Outdialing** option as **Inband** or **Using AT Command**. Default: Using AT Command.

- Select the appropriate **DTMF Generation ON Time** for the Mobile Port. This parameter determines the time for which the DTMF digit will remain ON, while being out dialed by the system. You may select 100 msec or 200 msec. Default: 100 msec.

- Select the appropriate **DTMF Detection** option. You may select **Using DSP** or **Using GSM Engine**. Default: Using DSP.

  If you have selected DTMF Detection option as *Using GSM Engine*, configure the following:

  - Select the appropriate **DTMF Detection Minimum ON Duration**. Valid range is 20 to 100 msec. Default: 30 msec.

  If you have selected DTMF Detection option as *Using DSP*, configure the following:

  - Select the appropriate **DTMF Detection ON Time (Minimum)** for the Mobile Port. This is the minimum time period for which the DTMF signal should be present in order to be detected. Valid range is 20 to 200 msec. Default: 34 msec.

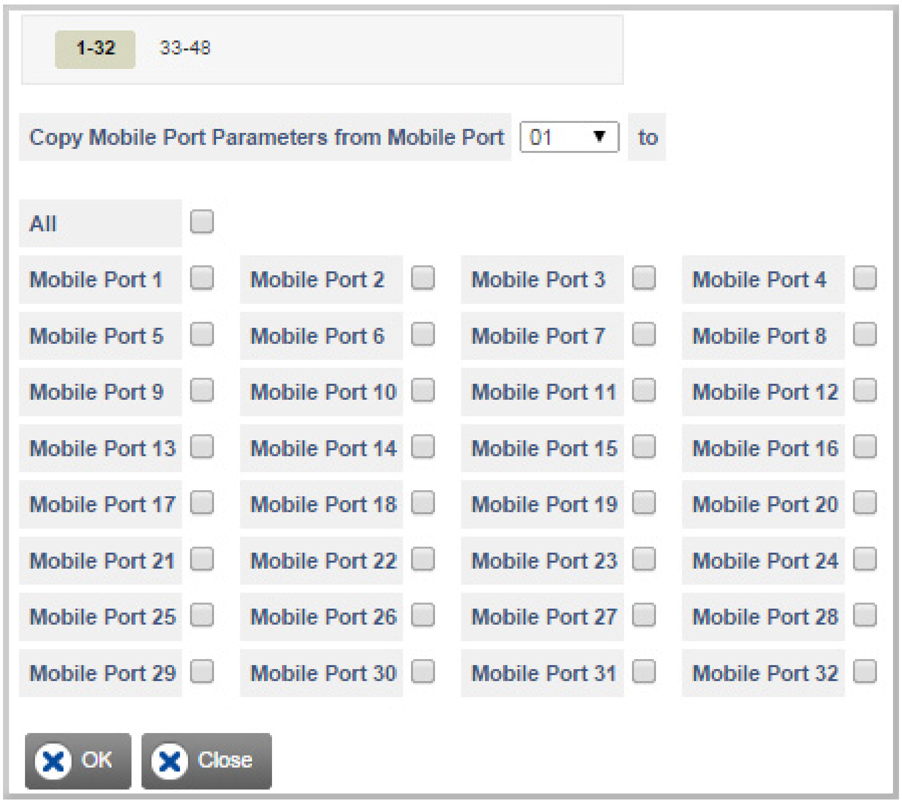
  - Select the appropriate **DTMF Detection OFF Time (Minimum)** for the Mobile Port. This is the minimum time period between successive DTMF digits. Valid range is 20 to 200 msec. Default: 68 msec.

  - Select the appropriate **DTMF Detection Level (Minimum)** for the Mobile Port. This is the minimum level (dB) of the DTMF digit to be considered as valid. Default: -10.5 dB.

Copy Port Parameters

- You can also copy the settings of a Mobile Port to another Mobile Port using the **Copy** button. To do this,

- Click the **Copy** button. The **Copy Mobile Port Parameters** window opens.



- In the **Copy Mobile Port Parameters from Mobile Port** box, select the number of the port you want to copy settings *From*. Select the check box of the respective port numbers you want to copy the settings *To*.

- If you want to copy the settings *To* all the ports, select the **All** check box.

- Click the **OK** button.

  Once you have copied the settings, you can again edit the specific parameters of the Mobile Port you copied the settings to.

SIP Trunk

SARVAM UMG supports 250 SIP Trunks. You can register all SIP Trunks with the same ITSP or with different ITSPs. These SIP Trunks may be configured as Proxy or Peer-to-Peer (non-proxy).

- Click the **Basic Settings** link to expand.
- Click the **SIP Trunk** link.

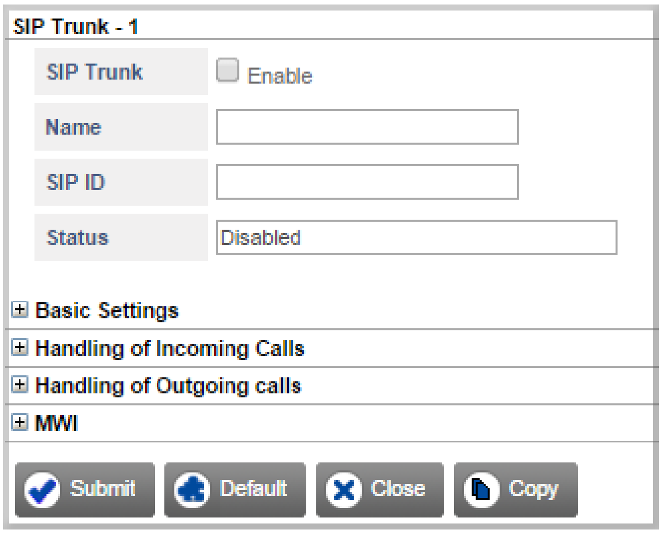
The SIP Trunk page displays the following parameters:

- **Trunk:** It displays the SIP Trunk numbers. Click on the desired SIP Trunk number to configure the SIP Trunk Parameters.
- **Enable:** Click the check box to enable the SIP Trunk.
- **Name:** Assign a Name to the SIP Trunk for identification. The Name can be a maximum of 24 characters.
- **Status:** This displays the status of SIP Trunk.
- **SIP ID:** This displays the SIP ID assigned to the SIP Trunk.
- **SIP Registration:** Keep the **SIP Registration** enabled. Clear this check box only if you do not want to enable SIP Registration for the respective SIP Trunk. Default: Enabled.
- **SIP Network Profile:** It displays the Network Profile you select for the SIP Trunk. To configure the Network Profile, click on **Network Profile**.
- **Incoming Call Routing:** It displays the Incoming Call Routing Method selected for the SIP Trunk.

To configure the **SIP Trunk** parameters,

- Click **SIP-1**.

  The **SIP Trunk-1** window opens.



- Select the **SIP Trunk** check box to enable the SIP Trunk. You may disable the SIP Trunk, if you do not want to route calls through this Trunk. Default: Disabled.

- You can assign a **Name** to the SIP Trunk for identification. Default: Blank.

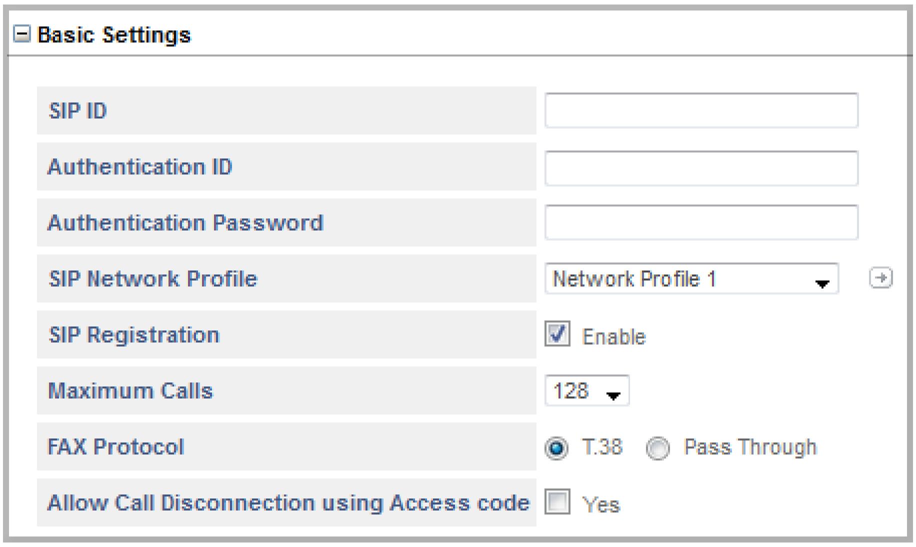
  Assign the name of the ITSP with which the trunk is registered, or any other name of your choice. The name will appear on the display of the remote party's phone, when a call is made through this SIP Trunk.

- In **SIP ID**, the SIP ID that you assign under *Basic Settings* is displayed.

- **Status** displays the status of the SIP Trunk.

Basic Settings

Click **Basic Settings** to expand.



- In **SIP ID**, enter the SIP ID provided by your ITSP. For example, if the SIP URI provided by the ITSP is 12345@abc.com, enter 12345 in this field. Default: Blank.

  The SIP ID is the number 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.

- Enter the **Authentication ID** (User ID) provided by your ITSP. Default: Blank.

- Enter the **Authentication Password** provided by your ITSP. Default: Blank.

- In **SIP Network Profile**, you can either select the default **Network Profile 1** or **Add New Network Profile** option.

  - Click **Settings** to configure the parameters of the selected Network Profile.

    For detailed instructions, see "SIP Network Profile".

    You can also configure the SIP Network Profile from *Advanced Settings*. For instructions, see "SIP Network Profile" under Advanced Settings.

- Keep the **SIP Registration** check box enabled.

  SARVAM UMG will send the REGISTER message to Registrar Proxy or Outbound Proxy as applicable.

  Clear the check box, only if you want to disable registration. Default: Enabled.

- In **Maximum Calls**, select the number of simultaneous calls you want to allow on this SIP Trunk.

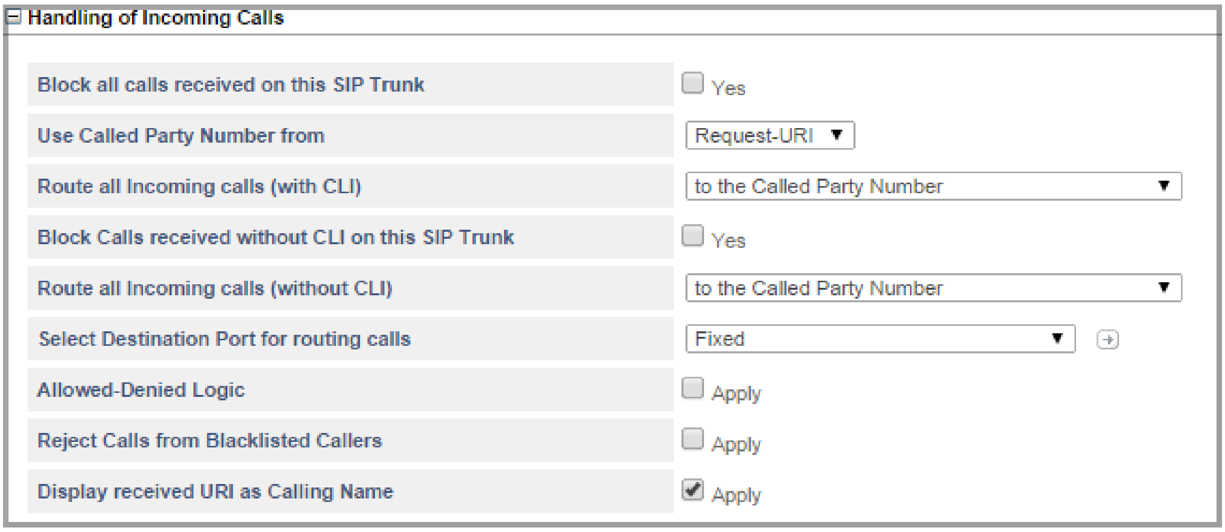
The maximum number of simultaneous SIP calls depend upon the number of Vocoder channels[14] supported.

- Select the desired **Fax Protocol**, to send and receive the Fax over IP:
  - **T.38:** If you select this option, the device you are sending the fax to, must also support this protocol.
  - **Pass Through:** Select this option, if you need to send fax over G.711. The device you are sending fax to must also use G.711.

    Default: T.38.
- Select the **Allow Call Disconnection using Access code** check box to enable. Default: Disabled. To know more about the feature, see "Disconnecting a Call using Access Code".

Handling of Incoming Calls

Click **Handling of Incoming Calls** to expand.



- Keep the **Block all calls received on this SIP Trunk** check box disabled.

  Select this check box only if you do not want to route calls received on this SIP Trunk.
- By default, SARVAM UMG identifies the Called Party Number for routing the incoming call on the SIP Trunk further, by the number received in the **Request-URI** of the INVITE message.

  If you want the system to identify the Called Party Number from the 'To Field' of the INVITE message, in the **Use Called Party Number From** parameter, select the **To Field** option.

Destination Number Determination

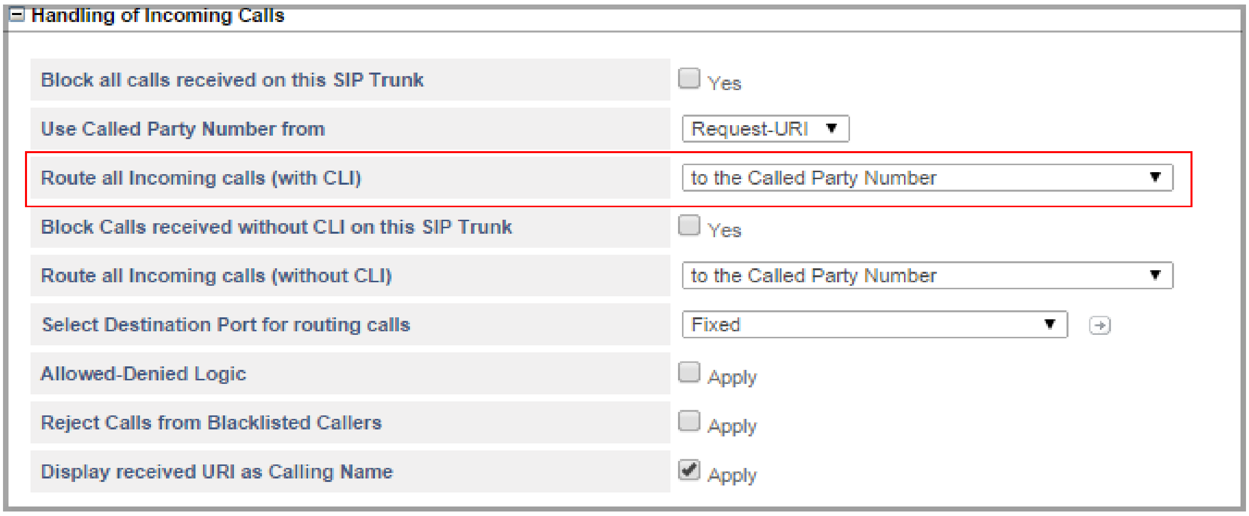
Select the desired destination number determination method for routing incoming calls *with* and *without* CLI.

- To **Route all Incoming calls (with CLI)**, you may select from any of the following methods:

14. *During the Demo period, the number of Vocoder channels that will be supported will be equal to the total number of channels available in the Vocoder module/s installed in the System. If the Demo period is paused or gets expired, then the number of supported Vocoder channels will be as per the license you purchase.*

- without any Destination Number
- to a Fixed Destination Number
- on the basis of Calling Party Number
- on the basis of DDI Number
- to the Called Party Number
- after Answering the Call and Collecting the Digits

  Default: to the Called Party Number

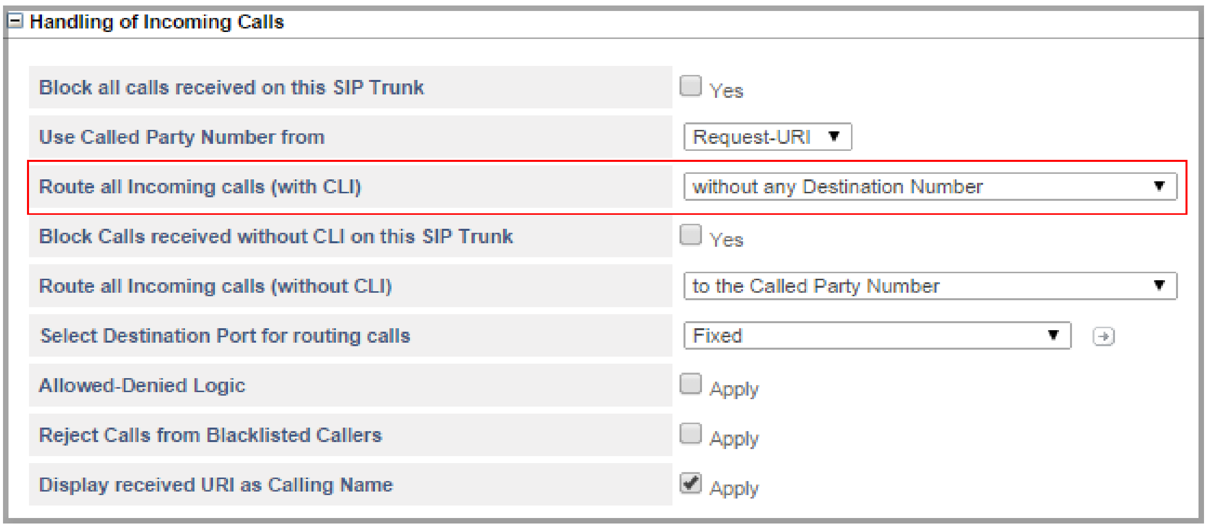


Read further for instructions on selecting and configuring each of these destination number determination methods.

If the destination number to be dialed out is an IP Address, SARVAM UMG will not check the Destination Port Determination Method. Instead, it will route the call using the SIP Trunk / Group programmed for IP Dialing. (See "IP Dialing" to know more).

Route Calls without any Destination Number

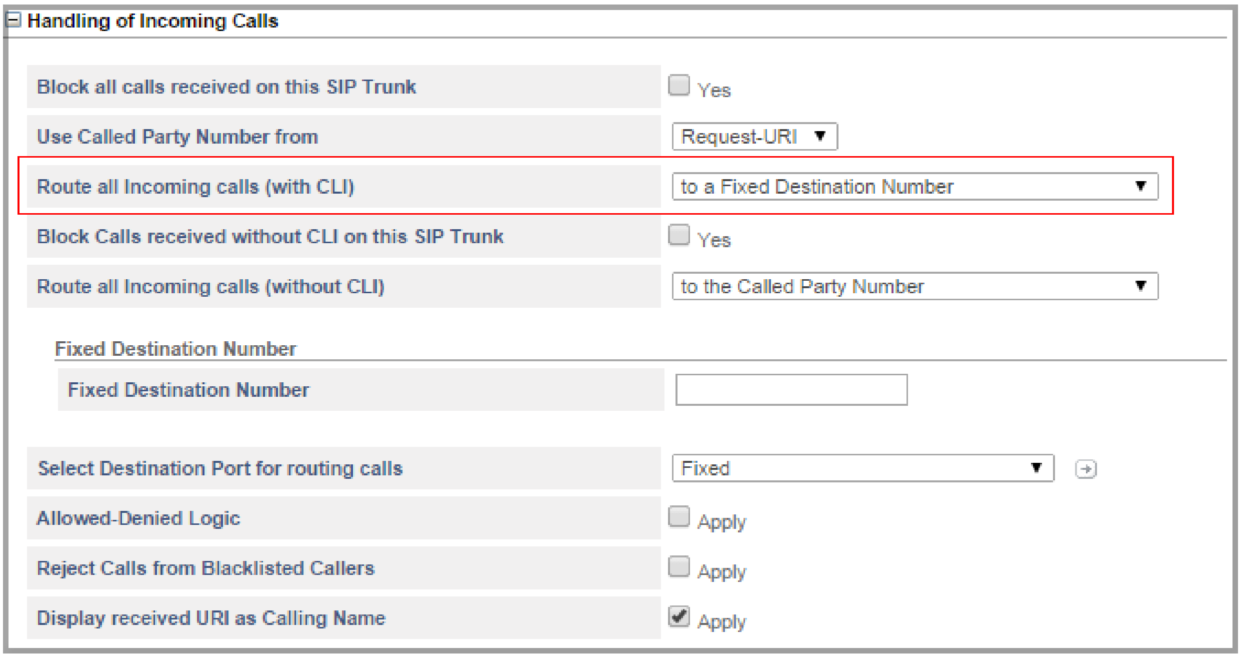
In this method, all calls received on the SIP Trunk are directly routed to the destination port, irrespective of the Destination Number.



- To apply this method, in **Route all incoming calls (with CLI)**, select **without any Destination Number**.

Route to a Fixed Destination Number

In this method, calls received on the SIP Trunk are routed to a fixed destination number, which is configured for the SIP Trunk.



To apply this method, do the following:

- In **Route all Incoming calls (with CLI)**, select **to the Fixed Destination Number**.
- In the **Fixed Destination Number** box that appears, enter the desired destination number. The Destination Number may consist of a maximum of 24 digits. Valid digits are 0 to 9, *, # and . (dot/period). Default: Blank.
- Click **Submit** to save your settings.

Route on the basis of Calling Party Number

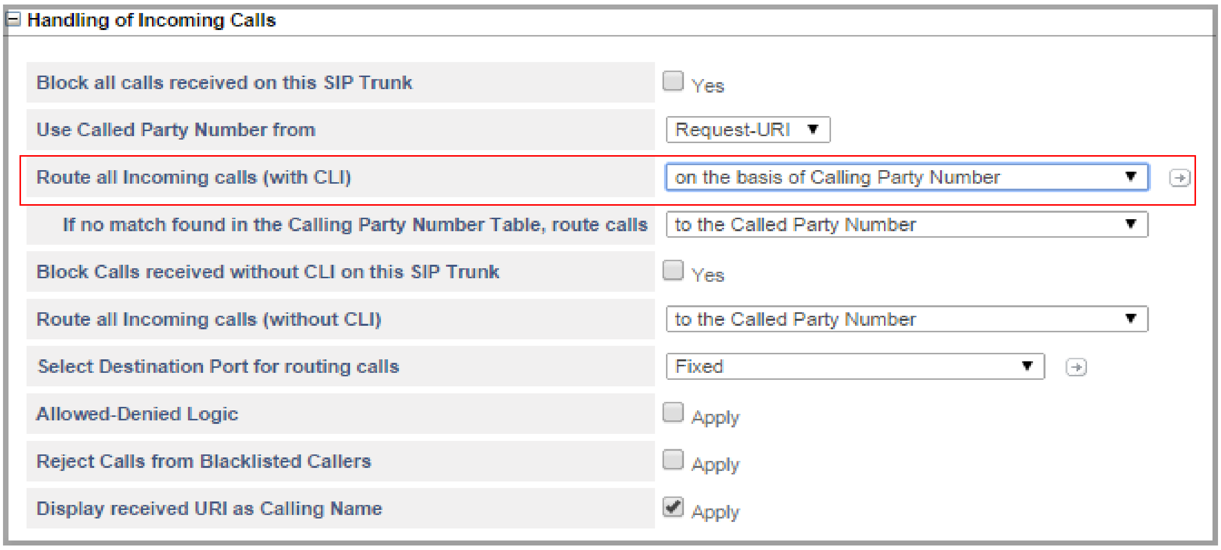
In this method, a call received on the SIP Trunk is routed to a specific number, as per the calling party's number. You must configure the calling party numbers in the *Calling Party Number Based* Table.

When there is an incoming call on the SIP Trunk, SARVAM UMG will match the Calling Party Number with the entries of the Calling Party Number Based Table. If a match is found, the call is routed to the destination number configured for that Calling Party Number.

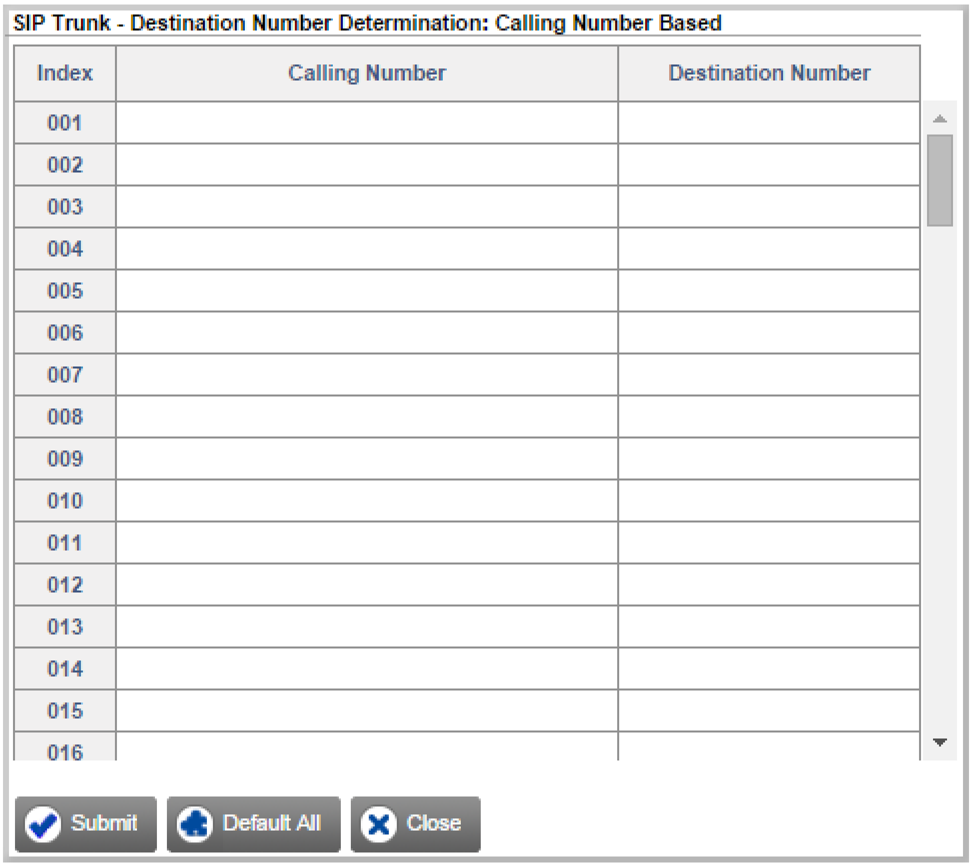
To apply this method, do the following:

- In **Route all Incoming calls (with CLI)**, select **on the basis of Calling Party Number**.

- Click **Settings** .



- The **SIP Trunk- Destination Number Determination: Calling Number Based** Table window opens.



- In **Calling Number**, enter the calling party numbers. The Calling numbers may consist of a maximum of 24 characters. Default: Blank.

- For each calling party number, enter a corresponding destination number in **Destination Number**. Destination numbers may consist of a maximum of 24 characters. Digits 0 to 9, *, # and (.) dot are allowed. Default: Blank.

- Click **Submit** to save your entries. Close the window to return to the main page.

  You can also configure the **Calling Number Based** table from *Advanced Settings.* For instructions, see “Destination Number Determination” under *Advanced Settings*.

- Select a method for routing incoming calls with CLI that *do not match* with any entries in the Calling Party Number Based Table.

  In **If no match found in the Calling Party Number Table, route calls**, select the desired method from the following options for processing the call:
  - to a Fixed Destination Number
  - to the Called Party Number
  - on the basis of DDI Number
  - after Answering the Call and Collecting the Digits

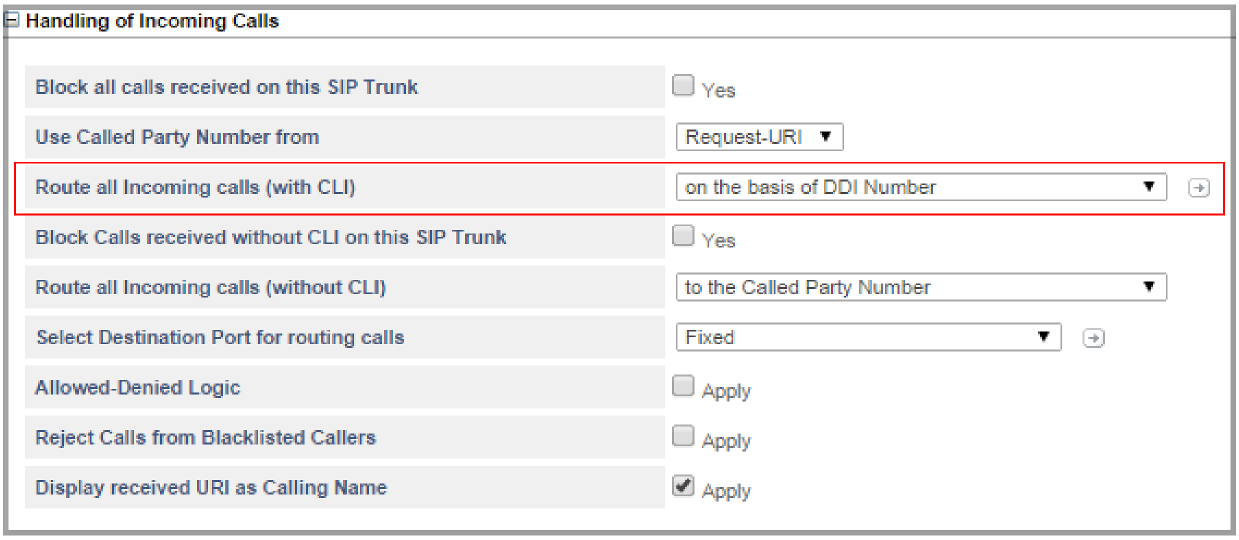
  Default: to the Called Party Number.

Route on the basis of DDI Number

In this method, incoming calls on the SIP Trunk are routed to specific numbers as per the DDI number received in the SETUP message on the SIP Trunk.

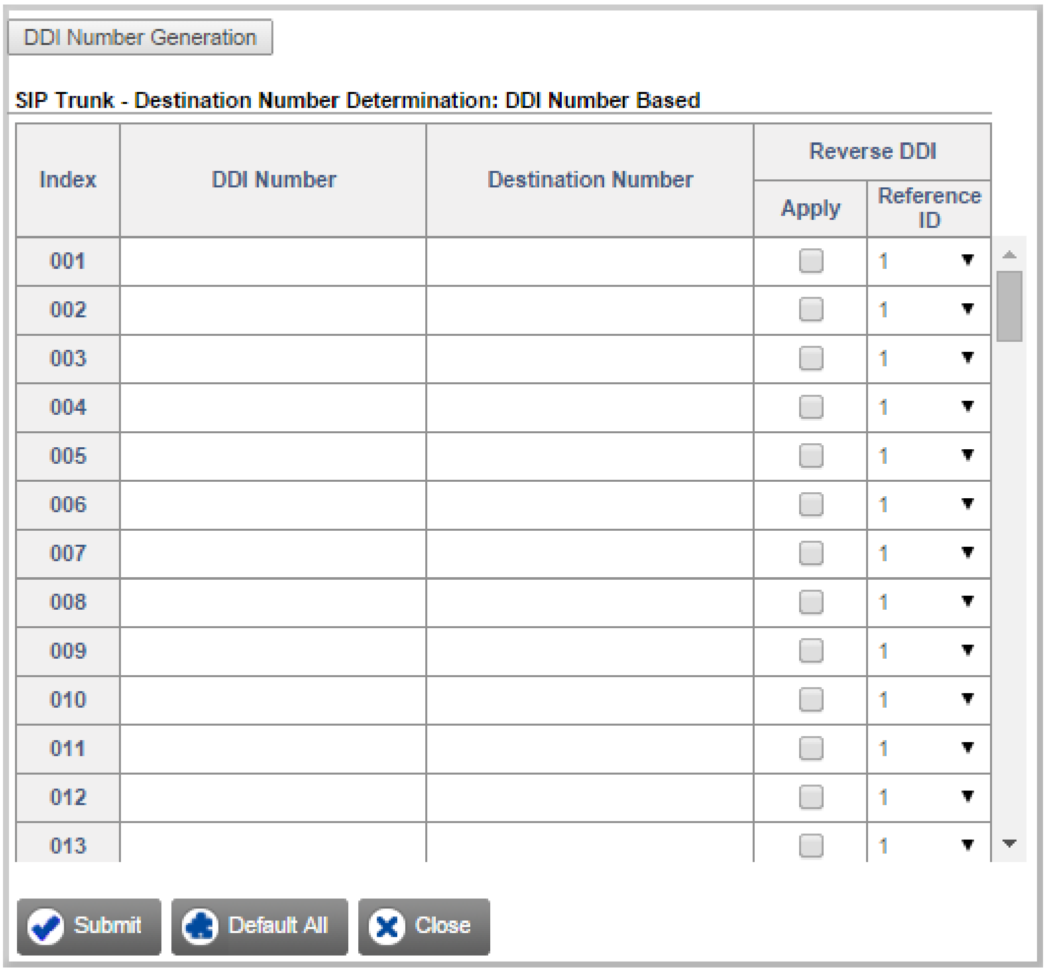
To apply this method, do the following:

- In **Route all Incoming calls (with CLI)**, select **on the basis of DDI Number**.



- Click **Settings** .

The **SIP Trunk - Destination Number Determination: DDI Number Based** Table opens.



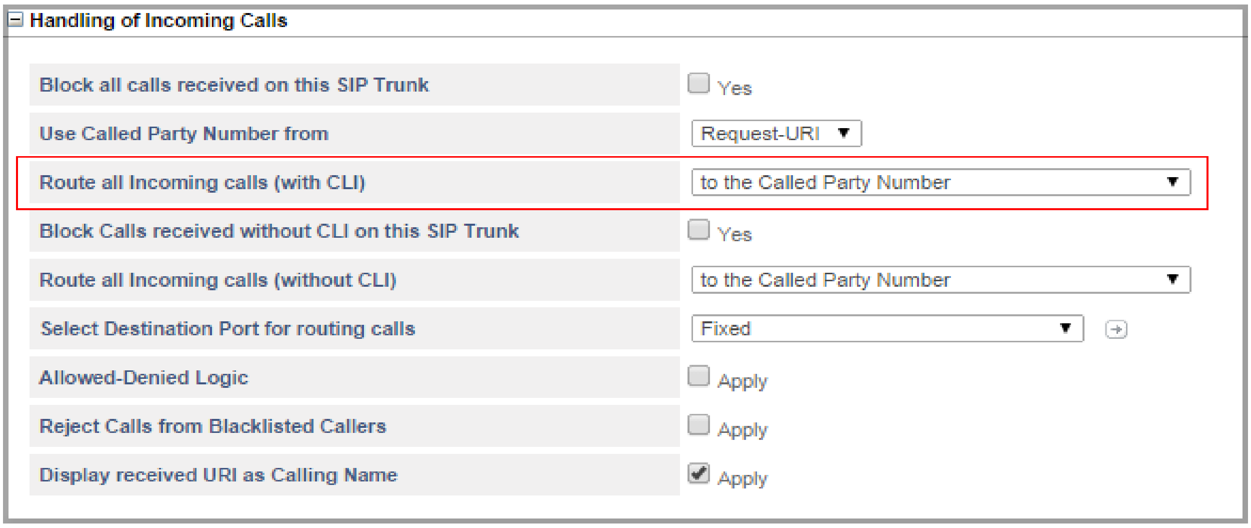
- In **DDI Number**, enter the DDI Numbers allotted by your service provider.
- For each DDI Number, enter the corresponding destination number in **Destination Number**.
- To apply **Reverse DDI** for each number, select the check boxes under **Apply** and select the **Reference ID** for the number. Default: Apply Reverse DDI is disabled and Reference ID is 1.
- Click **Submit** to save and close the window to return to the main page.

  You can also configure the **DDI Number Based** Table from *Advanced Settings.* For instructions, see "Destination Number Determination" under *Advanced Settings*.

Route to the Called Party Number

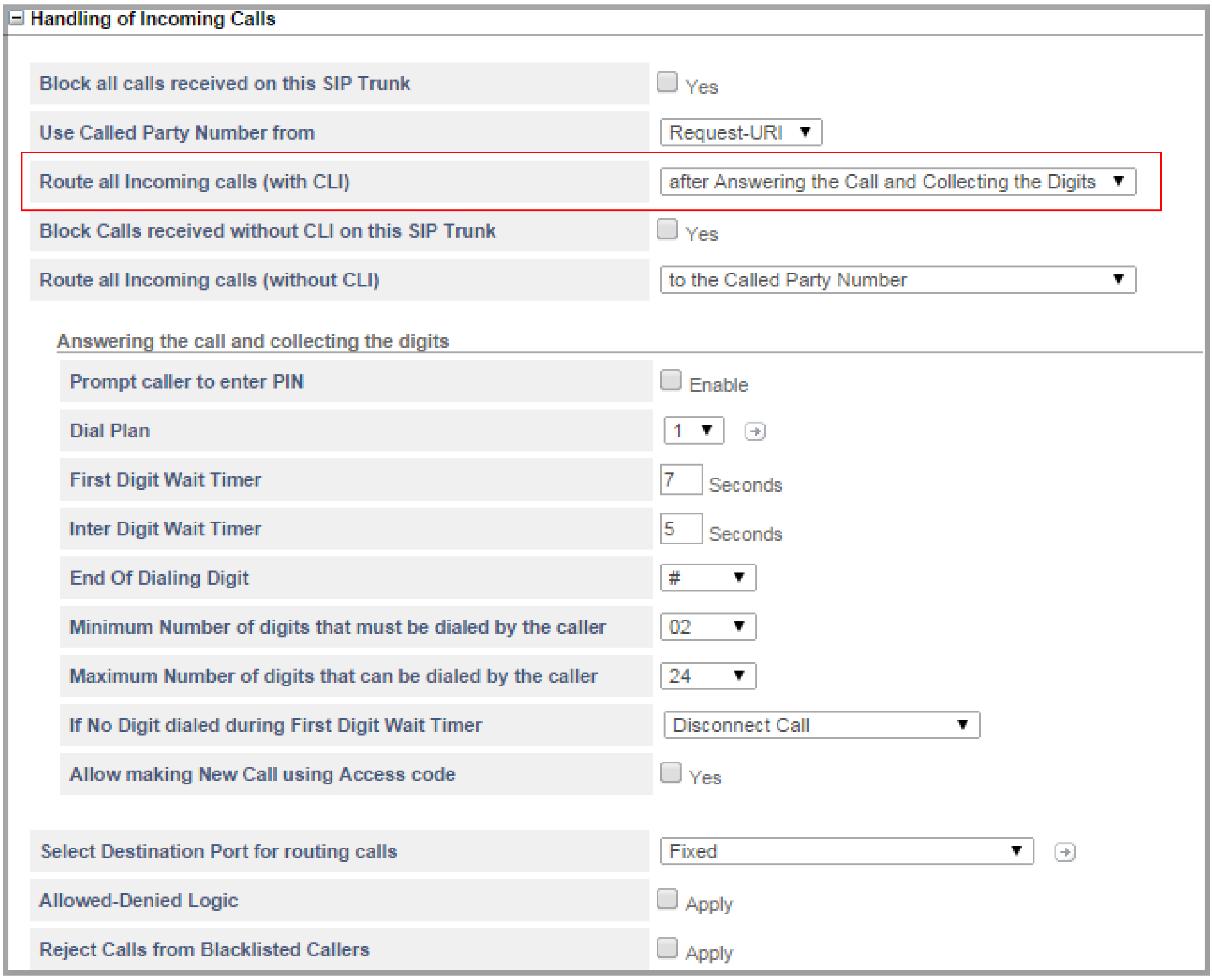
In this method, a call received on the SIP Trunk is routed to a specific number depending upon the called party number received in the SETUP Message on the SIP Trunk.

- To apply this method, in **Route all incoming calls (with CLI)**, select **to the Called Party Number**.



Route after Answering the Call and Collecting the Digits

In this method, the incoming call is answered and dial tone is played to the caller, allowing the caller to dial the desired number. The number dialed by the caller is considered as the destination number.



To apply this method, do the following:

- In **Route all Incoming calls (with CLI)**, select **after Answering the Call and Collecting the Digits**.

  The related parameters of this method appear under **Answering the call and collecting the digits**.

  - If you want to enable PIN Authentication on the SIP Trunk, select the **Prompt caller to enter PIN** check box.

    If you enable this check box, you must also configure the PIN Authentication Table. To know more about this feature and for detailed instructions, see “PIN Authentication” under *Advanced Settings*.

  - SARVAM UMG supports 8 Dial Plans with total 64 entries in each table. When a user dials a number, it is compared with the Destination Number configured in the Dial Plan. If a match is found, the system routes the call immediately without waiting for End of Dialing and if a match is not found, the system will wait for the End of Dialing and then route the call as per the Destination Port Selection method configured.

Select the **Dial Plan** table number you configured for this port. If you have not configured the Dial Plan table you may do so now,

  - Click **Settings** the Dial Plan Table opens.
  - Configure the numbers in the table. For detailed instructions, see "Dial Plan".

- Set the duration of the **First Digit Wait Timer**. This is the duration for which you want the system to wait for the caller to dial the destination number after the dial tone. Valid range is 01 to 99 seconds. Default: 7 seconds
- You may configure the following options as End of Dialing indication:
  - Set the duration of the **Inter Digit Wait Timer**. This is the duration for which you want the system to wait while receiving the digits dialed by the caller to consider it as End of Dialing. You may change this timer, if required. Valid range is 01 to 99 seconds. Default: 05 seconds.
  - In **End of Dialing Digit,** select # or * as termination digit the system should consider to detect end of dialing. Default: #
  - In **Minimum number of digits that can be dialed by the caller**, select the minimum number of digits to be dialed by the user for the system to consider it as a valid number**.** Valid range is 01 to 24 digits. Default: 2 digits.
  - In **Maximum Number of digits that can be dialed by the caller**, select the maximum number of digits to be dialed by the user for the system to consider it as End of Dialing**.** Valid range is 01 to 24 digits. Default: 24 digits.

    When the caller dials a number, the system will match it with the above End of Dialing indications and accept the one that matches first.
- If the caller fails to dial the number during the First Digit Wait Timer, you can either have the system disconnect the call or route the call to a fixed destination number.

  In **If No Digit dialed during First Digit Wait Timer**, select the desired option: **Disconnect the Call** or **Use Fixed Destination Number**. Default: Disconnect Call.
  - If you selected **Use Fixed Destination Number**, enter the desired destination number in the **Fixed Destination Number** field. The Destination number may consist of a maximum of 24 digits. Valid digits are 0 to 9, *, # and . (dot/period). Default: Blank.

- *The First Digit Wait Timer is loaded as soon as the system answers the call.*
- *When you dial the first digit, the First Digit Wait Timer is stopped and the system loads the Inter Digit Wait Timer.*
- *SARVAM UMG reloads the Inter Digit Wait Timer:*
  - *each time you dial a new digit till the termination digit is detected.*
  - *each time you dial a new digit till the entry is not matched in Dial Plan.*
  - *when you have dialed the maximum number of digits configured as End of Dialing.*

- If you want to enable the feature Making New Call using Access Code on the SIP Trunk, select the **Allow making New Call using Access Code** check box. For further details, see "Making a New Call using Access Code".

- Click **Submit** to save settings.

- If you do not want to route the incoming calls received without CLI, through this SIP Trunk, select **Block Calls received without CLI on this SIP Trunk** check box.

- To **Route all Incoming calls (without CLI),** you may select from any of the following methods:

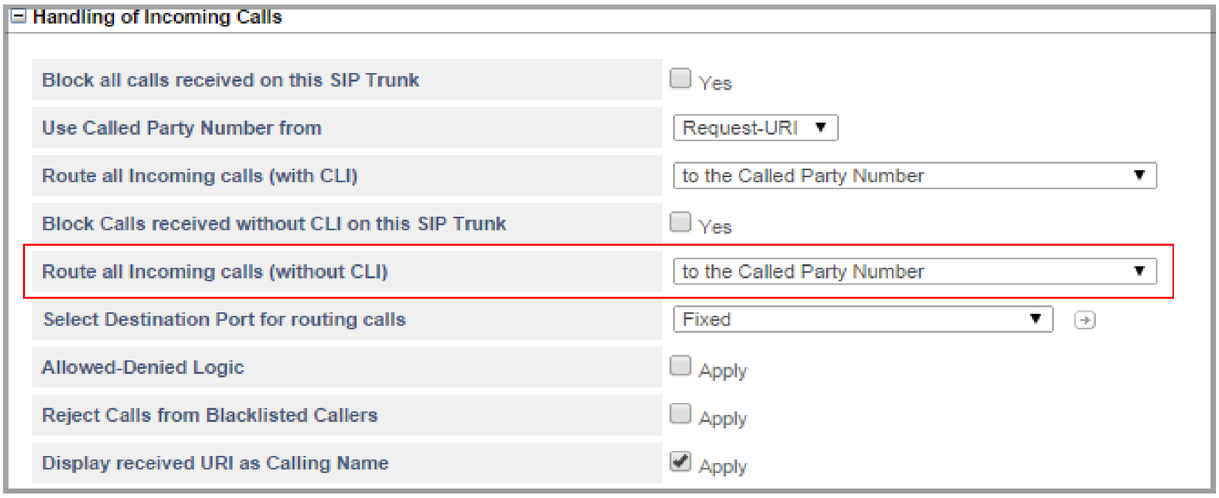
  - to a Fixed Destination Number, see "Route to a Fixed Destination Number".

  - on the basis of DDI Number, see "Route on the basis of DDI Number".

  - to the Called Party Number, see "Route to the Called Party Number".

  - after Answering the Call and Collecting the Digits, see "Route after Answering the Call and Collecting the Digits".

    Default: to the Called Party Number.



Destination Port Determination

For the SIP Trunk, select the Destination Port for routing calls from the following options:

- Fixed
- On the basis of Destination Number
- On the basis of Calling Party Number

  Default: Fixed.

  Read the description and follow the instructions for each of these destination port selection methods given below.

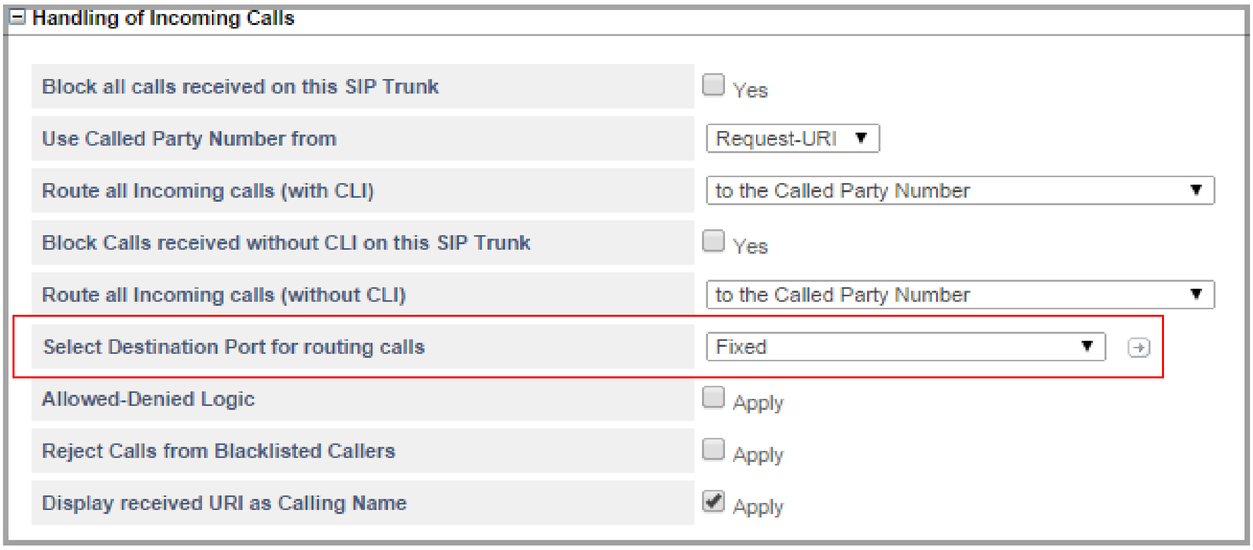
If the destination number to be dialed out is an IP Address, SARVAM UMG will not check the Destination Port Determination Method. Instead, it will route the call using the SIP Trunk / Group programmed for IP Dialing. (To know more, see the feature description "IP Dialing").

Fixed

In this method, calls received on the SIP Trunk are routed to a Fixed Destination Port, irrespective of the number dialed on the SIP Trunk.

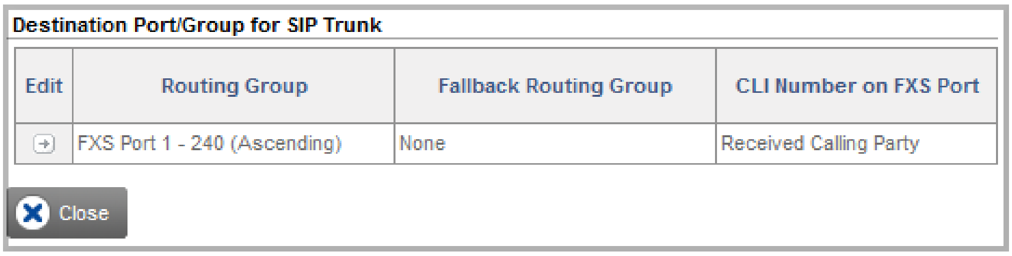
To apply this method, do the following:

- In **Select Destination Port for routing calls**, select **Fixed** option.



- Click **Settings** .

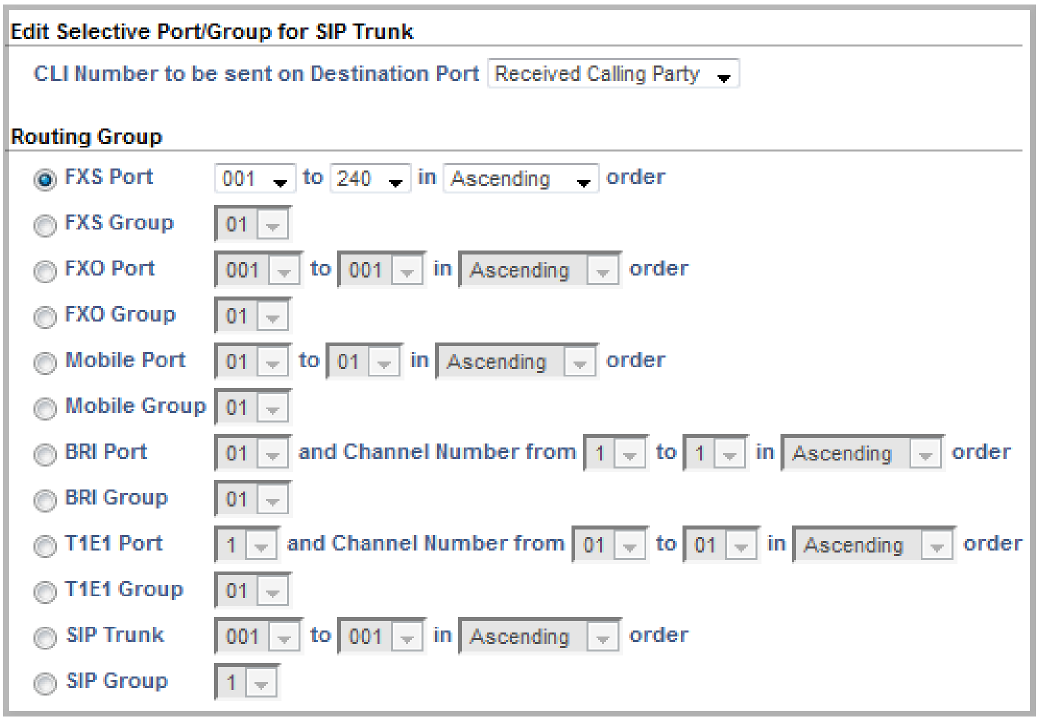
  The **Destination Port/Group for SIP Trunk** window opens.



  The default **Routing Group** and **Fallback Routing Groups** appear.

- If you wish to change the default Routing Group options, click **Edit** .

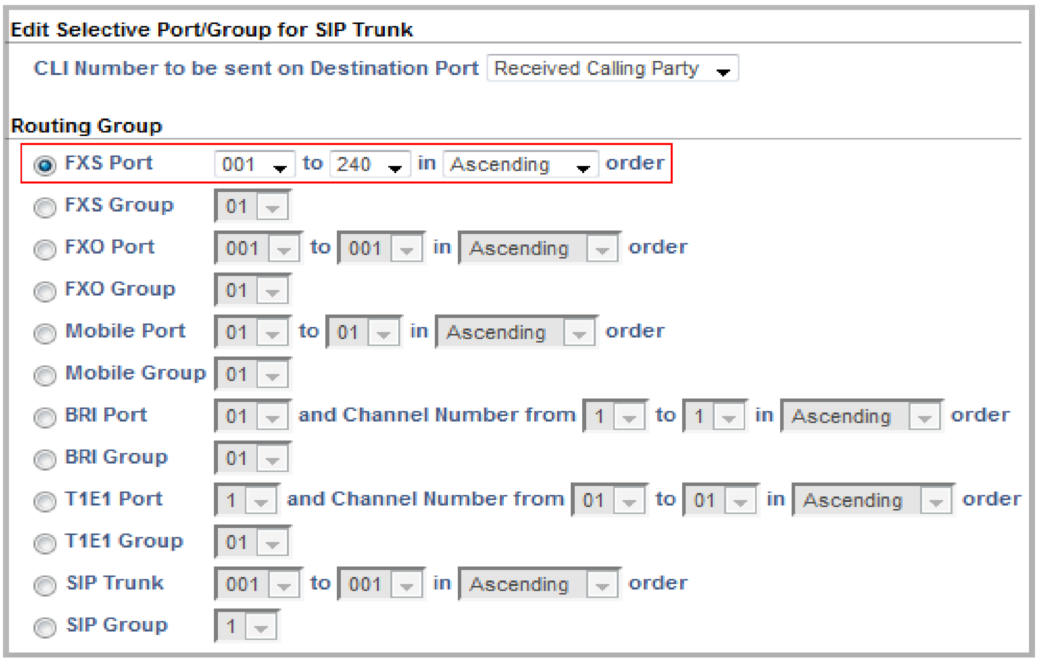
The **Edit Selective Port/Group for SIP Trunk** window opens.



- Select the **CLI Number to be sent on Destination Port**. You can select Received Calling Party or Received Called Party. Default: Received Calling Party.

CLI to be sent on Destination Port is applicable when FXS Port/Group is determined as the Destination Port.

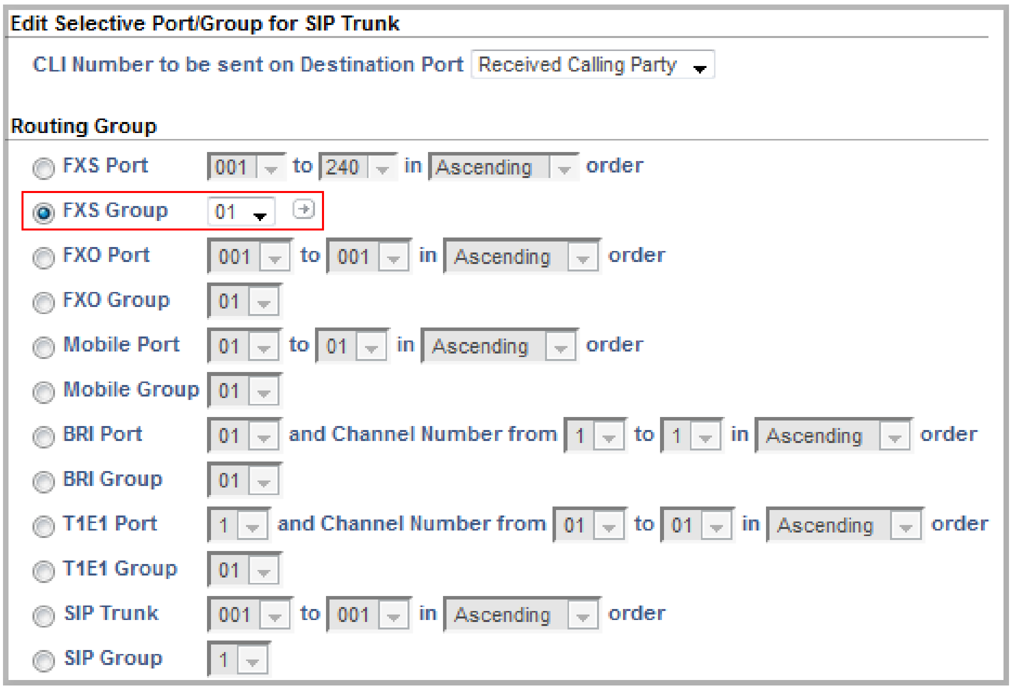
- Create the **Routing Group**.
    - To create a group of *sequential* **FXS Ports** as members,



- Select the desired **FXS Port** numbers as members. Default: 1.
- In **in - order**, select the order in which the system should hunt for a free member FXS Port to route the call.

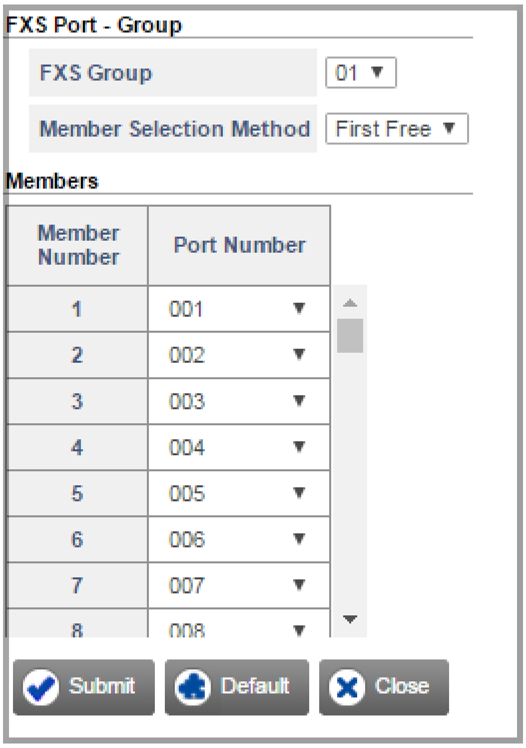
  Select **Ascending** to start hunting from the first to the last member FXS Port. Select **Descending** to start hunting from the last to the first member FXS Port. Default: Ascending.

- To create a group of *not-sequential* **FXS Ports** as members,
  - Select a **FXS Group**.



  - Select **FXS Group** number. Default: 1.
  - Click **Settings** .

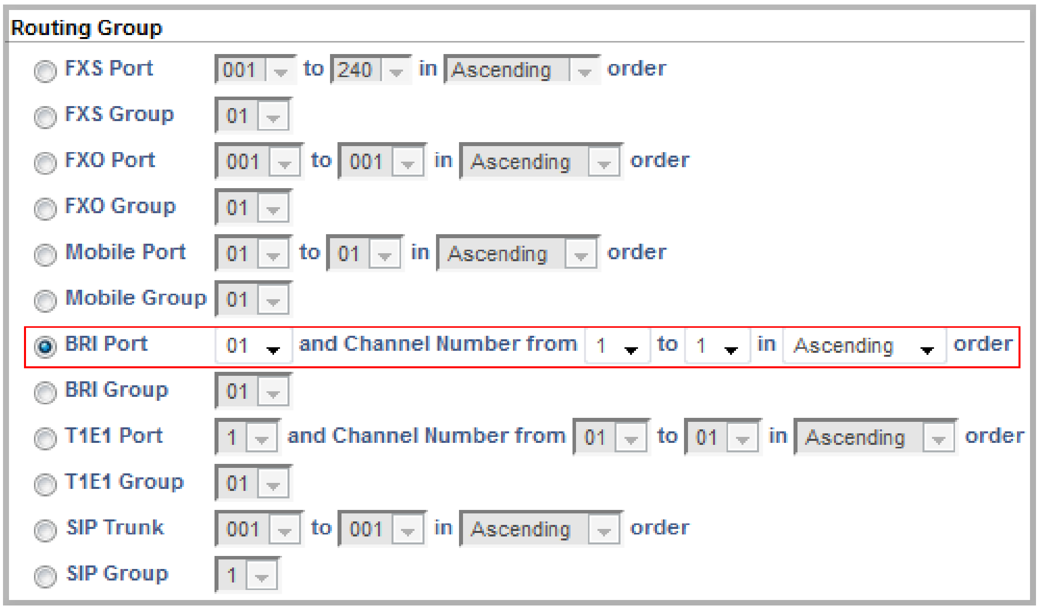
- The **FXS Port - Groups** window opens.



  - Create the FXS Group. For detailed instructions on creating groups, see the topic “Group” under *Advanced Settings*.

- Similarly, you can create a group of *sequential* and *not-sequential* FXO Ports, Mobile Ports and SIP Trunks.

- To create a routing group of *sequential* **BRI Channels** as members,



  - Select the **BRI Port** Number. Default: 1.

- In Channel Number **From - to**, select the **Start Channel Number** and the **End Channel Number**, respectively.

- In **in - order**, select the order in which the system should hunt for a free member Channel to route the call.

  Select **Ascending** to start hunting from the first to the last member channel. Select **Descending** to start hunting from the last to the first member channel. Default: Ascending.

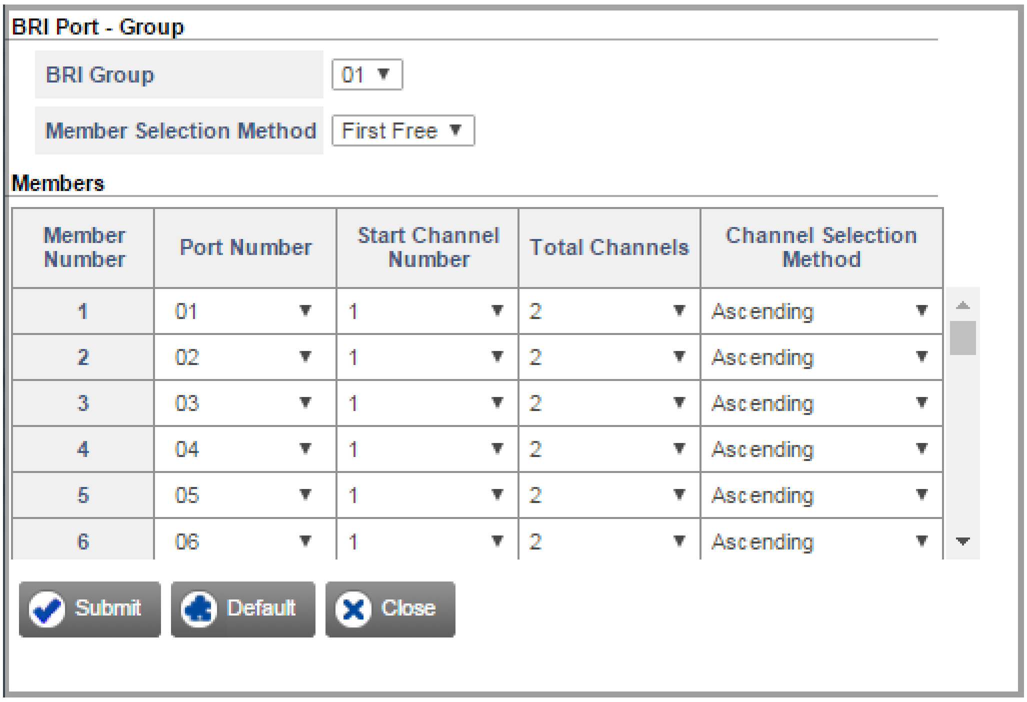
- To create a group of *not-sequential* **BRI Channels** as members,

  - Select **BRI Group**.

  - Select a **BRI Group** number. Default: 1.

  - Click **Settings**.

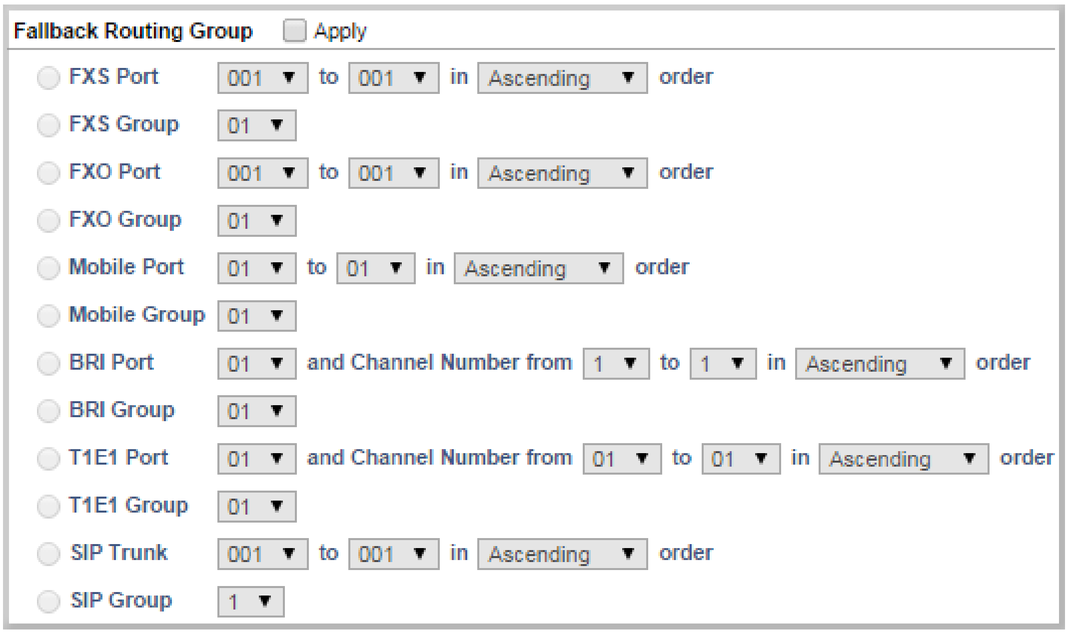
- The **BRI Port - Groups** window opens.



  - Create the BRI Group. For detailed instructions on creating groups, see the topic "Group" under *Advanced Settings.*

- Similarly, you can create a group of *sequential* and *not-sequential* T1E1 Ports.

- You may create the **Fallback Routing Group**.



- To do this,

  - Select the **Apply** check box.

- Follow the same instructions provided earlier for creating *sequential* and *not-sequential* groups of FXS Ports, FXO Ports, Mobile Ports, BRI Ports, T1E1 Ports and SIP Trunks.

- Click **Submit** to save changes. The **Edit** window closes.
- The entry you edited appears in the **Destination Port/Group for SIP Trunk** window.
- Close the **Destination Port/Group for SIP Trunk** window to return to the main page.

On the basis of Destination Number

In this method, incoming calls on the source port are routed to the destination port on the basis of the destination number (called party number) dialed by the caller.

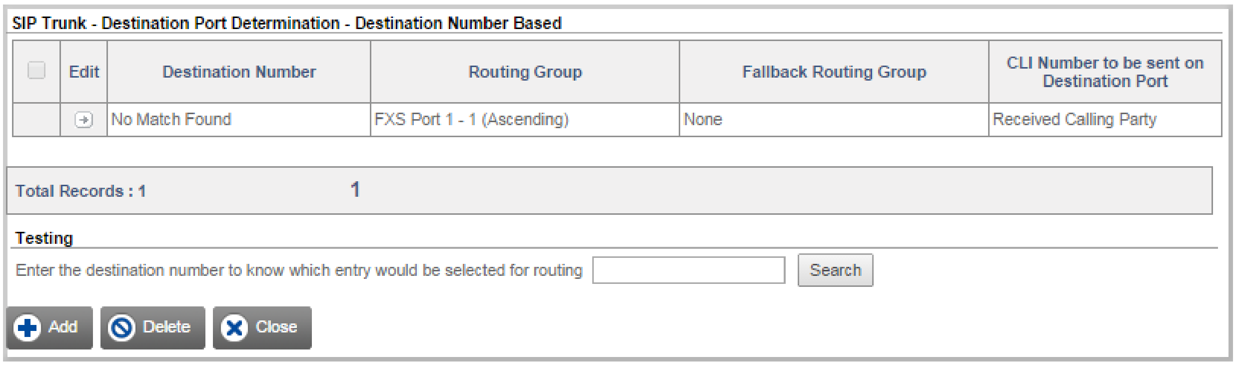
You must configure the called party numbers in the **Destination Number Based** Table. SARVAM UMG will match the called party number dialed by the caller with the entries of this table. If a match is found for the number in the table, the call is routed to the destination.

To apply this method, do the following:

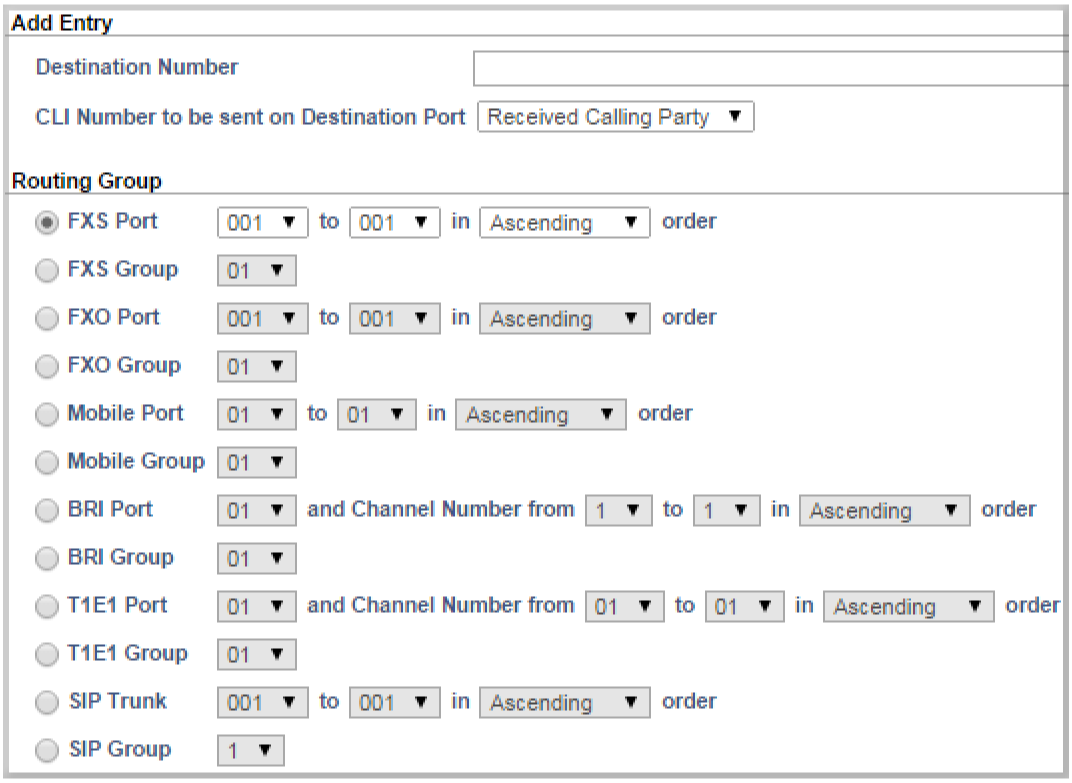
- In **Select Destination Port for routing calls**, select **On the basis of Destination Number** option.

- Click **Settings** .

The **SIP Trunk - Destination Port Determination - Destination Number Based** table window opens.



- To add a new entry, click **Add**. The **Add Entry** window opens. You can add upto 250 entries.



- In **Destination Number**, enter the number you expect the callers to dial. You may enter upto 64 characters (Digits + "Wildcard Characters"). Valid characters are 0 to 9, *, #, X, T, Comma [,], Hyphen [-], Caret [^]. Default: Blank.

Wildcard Characters

SARVAM UMG supports following characters.

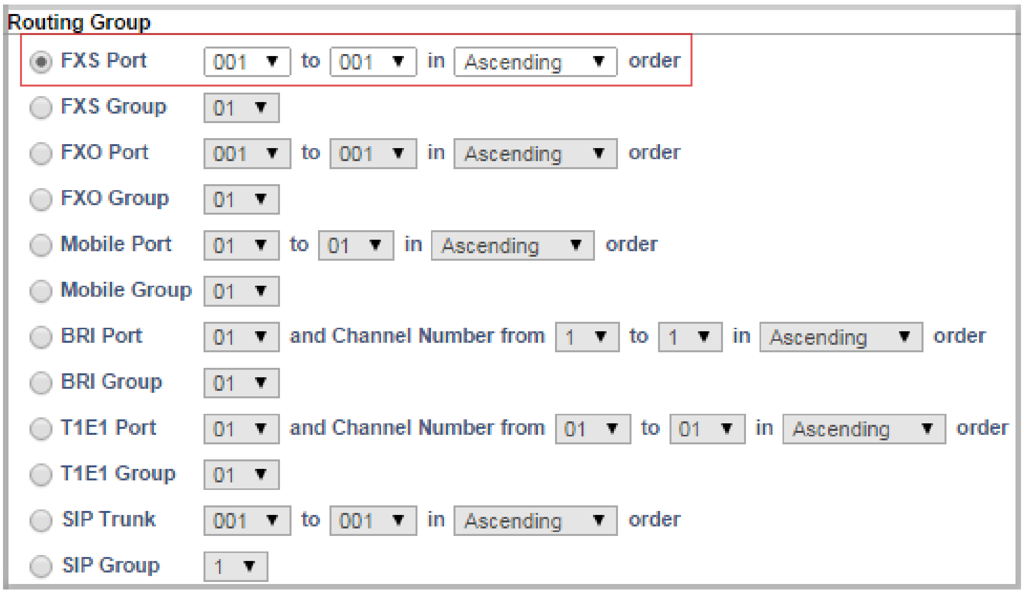
Character	Description
X (letter X)	X represents any single digit from 0 to 9.
#	When # is configured in a number string, it will not be considered as End of Dialing.

*****	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.

- Select the **CLI Number to be sent on Destination Port**. You can select Received Calling Party or Received Called Party. Default: Received Calling Party.

CLI to be sent on Destination Port is applicable when FXS Port/Group is determined as the Destination Port.

- Create the **Routing Group**.
  - To create a group of *sequential* **FXS Ports** as members,

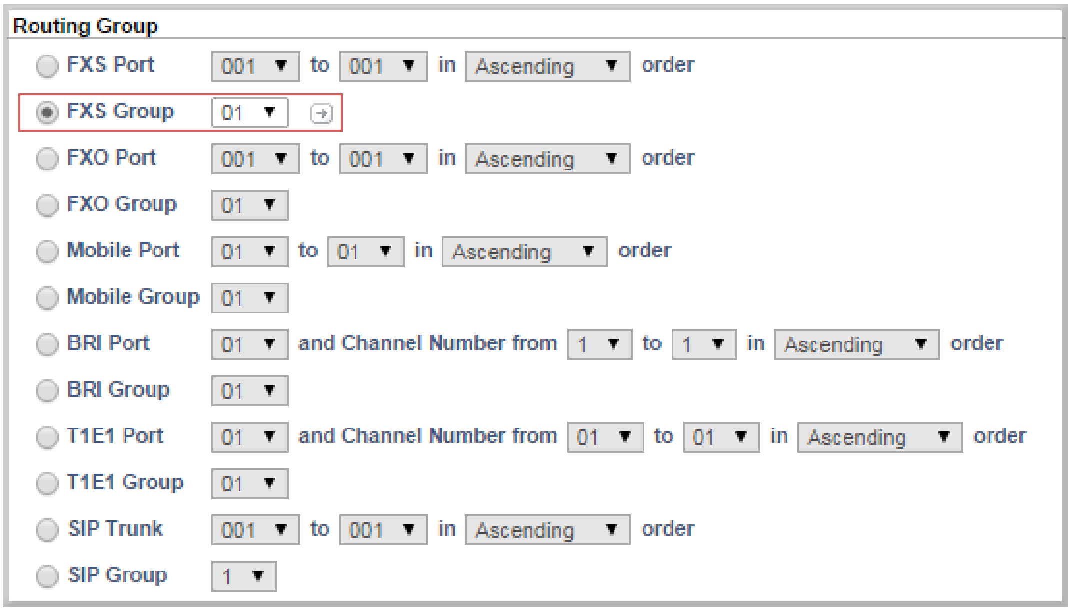


    - Select the desired **FXS Port** numbers as members. Default: 1.
    - In **in - order**, select the order in which the system should hunt for a free member FXS Port to route the call.

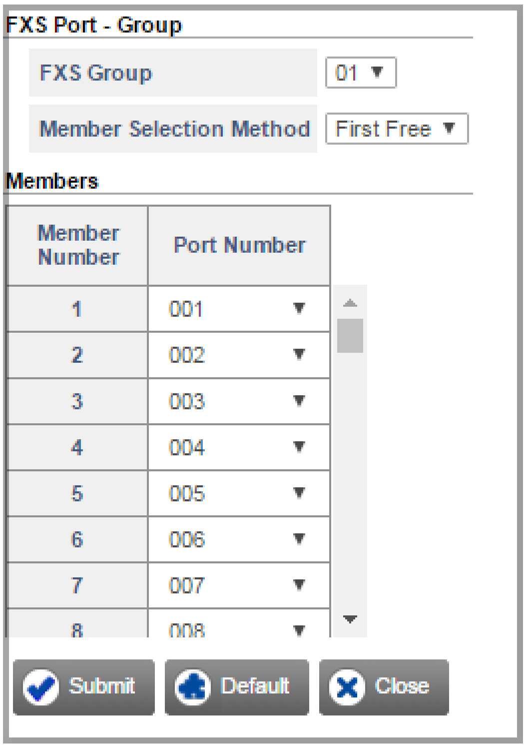
      Select **Ascending** to start hunting from the first to the last member FXS Port. Select **Descending** to start hunting from the last to the first member FXS Port. Default: Ascending.

  - To create a group of *not-sequential* **FXS Ports** as members,

- Select a **FXS Group**.



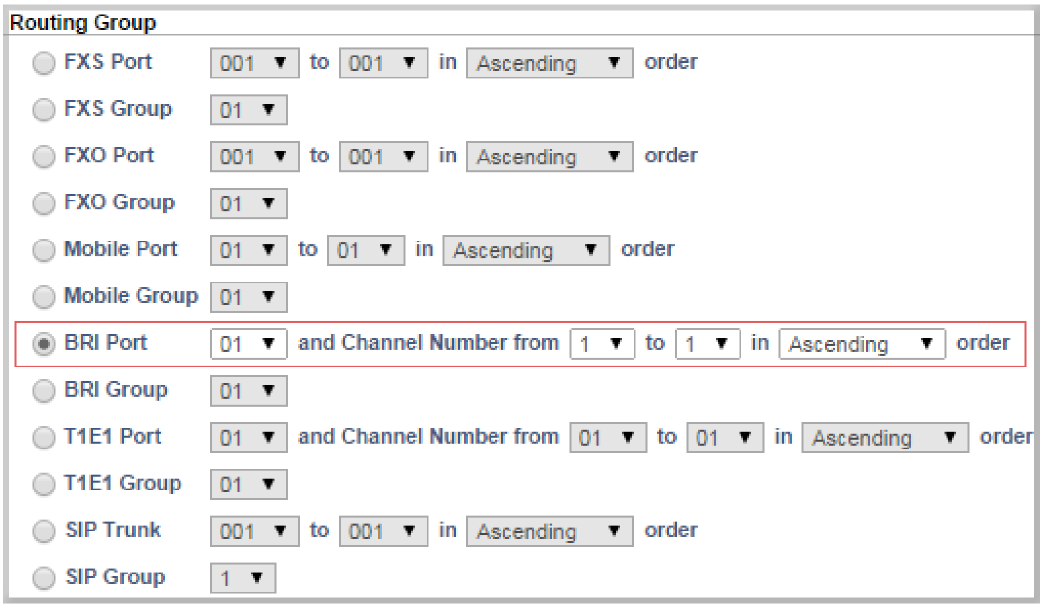
- Select **FXS Group** number. Default: 1.
- Click **Settings** .
- The **FXS Port - Groups** window opens.



- Create the FXS Group. For detailed instructions on creating groups, see the topic “Group” under *Advanced Settings*.

- Similarly, you can create a group of *sequential* and *not-sequential* FXO Ports, Mobile Ports and SIP Trunks.

- To create a routing group of *sequential* **BRI Channels** as members,



  - Select the **BRI Port** Number. Default: 1.

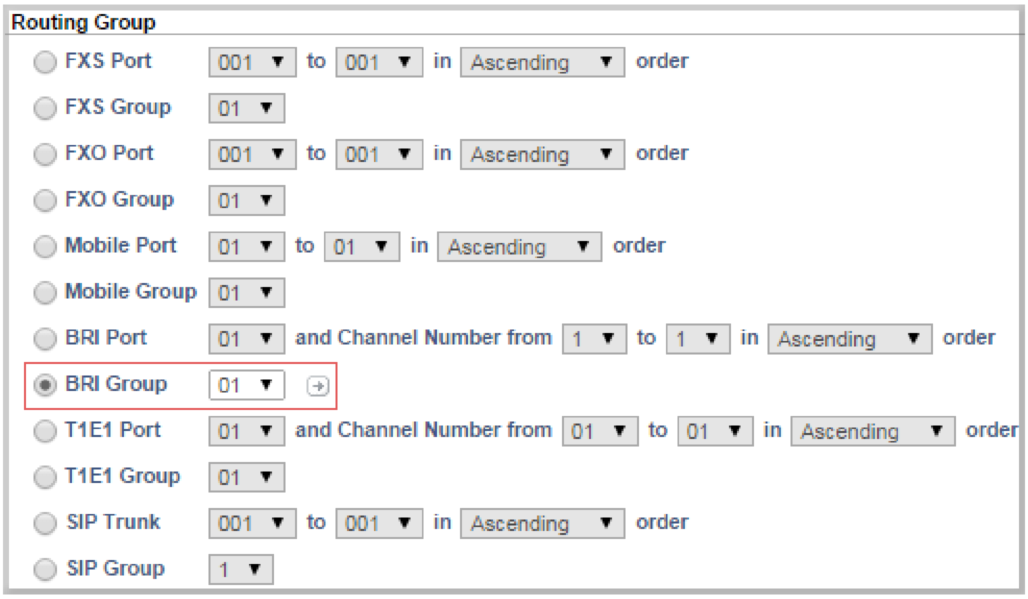
  - In Channel Number **From - to**, select the **Start Channel Number** and the **End Channel Number**, respectively.

  - In **in - order**, select the order in which the system should hunt for a free member Channel to route the call.

    Select **Ascending** to start hunting from the first to the last member channel. Select **Descending** to start hunting from the last to the first member channel. Default: Ascending.

- To create a group of *not-sequential* **BRI Channels** as members,

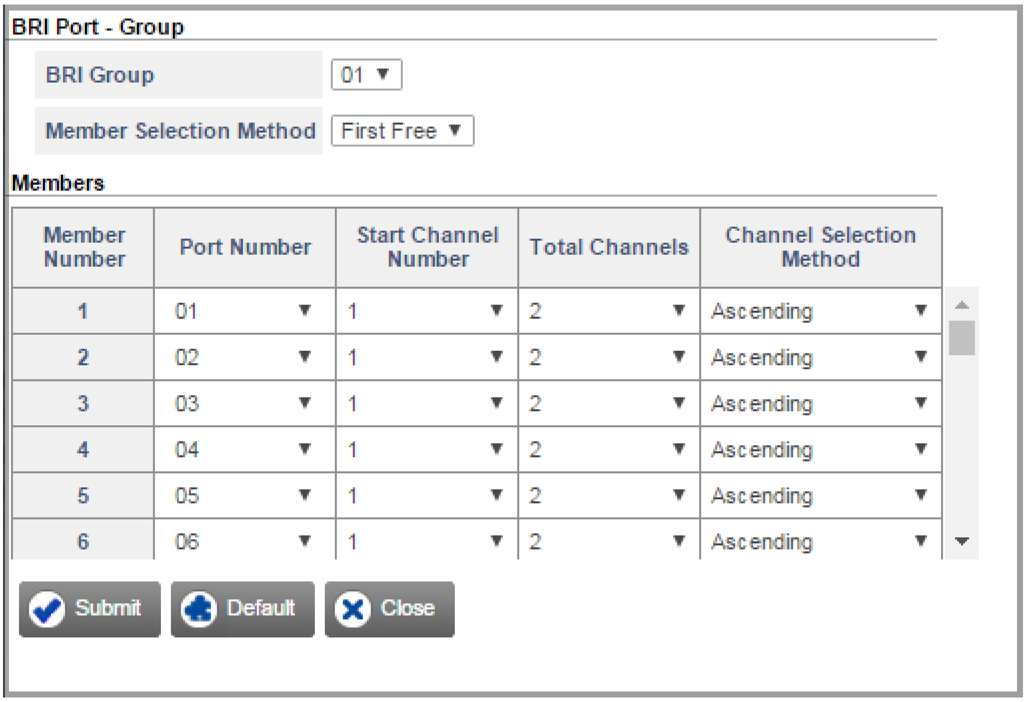
- Select **BRI Group**.



- Select a **BRI Group** number. Default: 1.

- Click **Settings** .

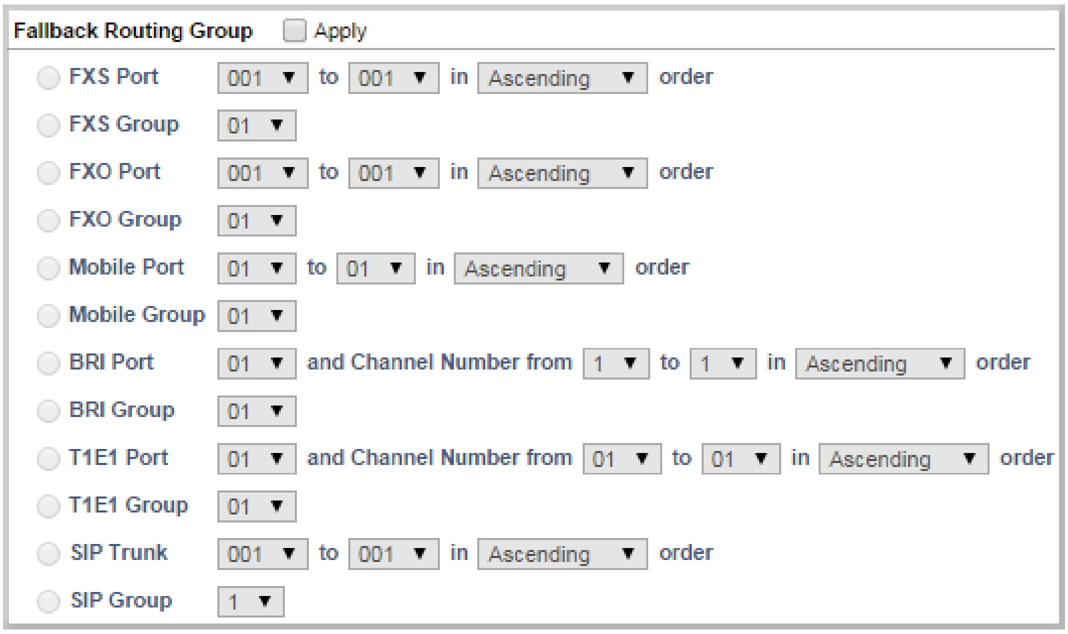
- The **BRI Port - Groups** window opens.



- Create the BRI Group. For detailed instructions on creating groups, see the topic “Group” under *Advanced Settings.*

- Similarly, you can create a group of *sequential* and *not-sequential* T1E1 Ports.

- You may create the **Fallback Routing Group**.



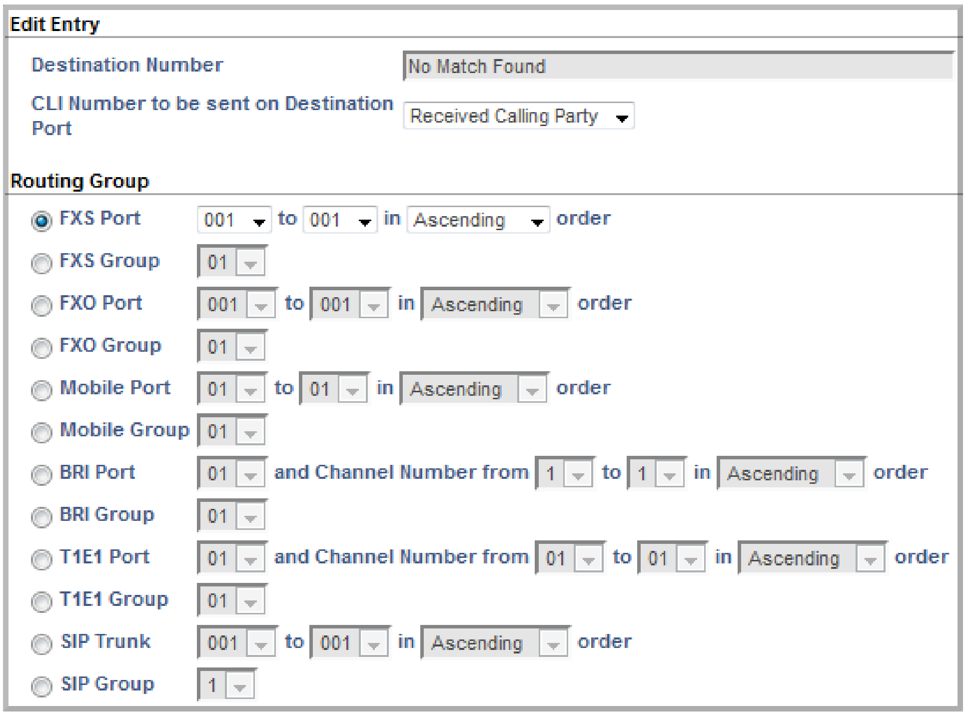
- To do this,
  - Select the **Apply** check box.
  - Follow the same instructions provided earlier for creating *sequential* and *not-sequential* groups of FXS Ports, FXO Ports, Mobile Ports, BRI Ports, T1E1 Ports and SIP Trunks.
- Click **Submit** to save changes. The **Add Entry** window closes.
- The entry you added appears in the **SIP Trunk - Destination Port Determination - Destination Number Based** table.
- Follow the same steps as above to add another entry to this table.
- To delete an entry, select the check box and click the **Delete** button.

If there are multiple entries in the Destination Number Based table, to search a particular entry in the table, under Testing enter the desired number to know which entry would be selected for routing search box.

- By default, FXS Port 1 - 1 (Ascending) is assigned as the Routing Group, for routing calls from numbers that do not match with any of the destination numbers you configured (No Match Found).

  To change the default Routing Group and to create the Fallback Routing Group for the No Match Found numbers entry,

- For the No Match Found entry in the table, click **Edit**.



- The **Edit Entry** window opens.
- Create the **Routing Group** and **Fallback Routing Group** as per your requirement.
- Click **Submit** and close the window.

- Close the window if you have finished adding/editing entries.

  You can also configure the **Destination Number Based** Table from *Advanced Settings*. For instructions, see “Destination Port Determination” under *Advanced Settings.*

On the basis of Calling Party Number

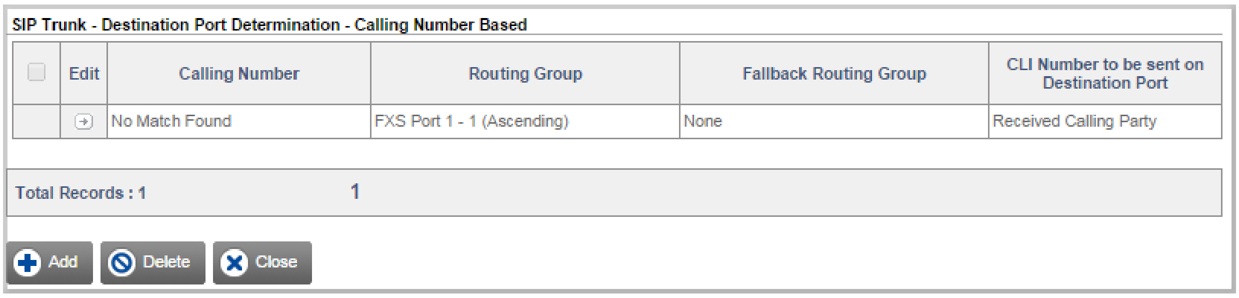
In this method, incoming calls on the SIP Trunk are routed to a specific port as per the calling party's number.

To apply this method, do the following:

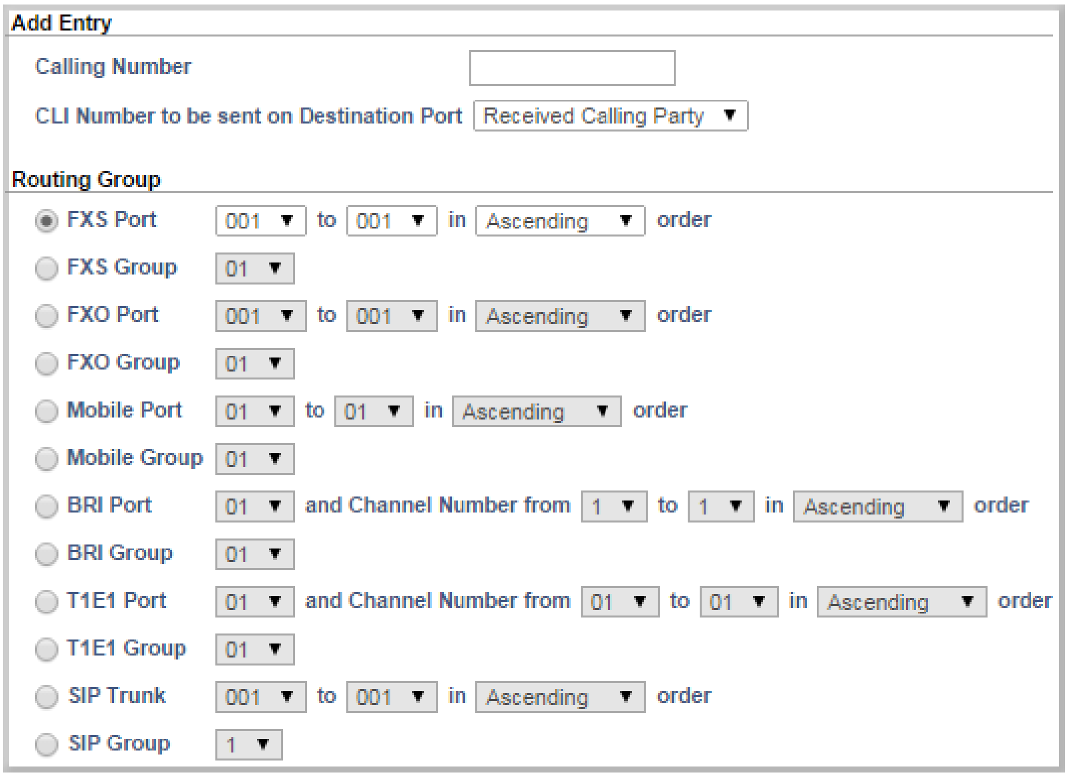
- In **Select Destination Port for routing calls**, select **On the basis of Calling Party Number** option.

- Click **Settings**.

The **SIP Trunk - Destination Port Determination - Calling Number Based** table window opens.



- To add a new entry, click **Add**. The **Add Entry** window opens. You can add upto 499 entries.



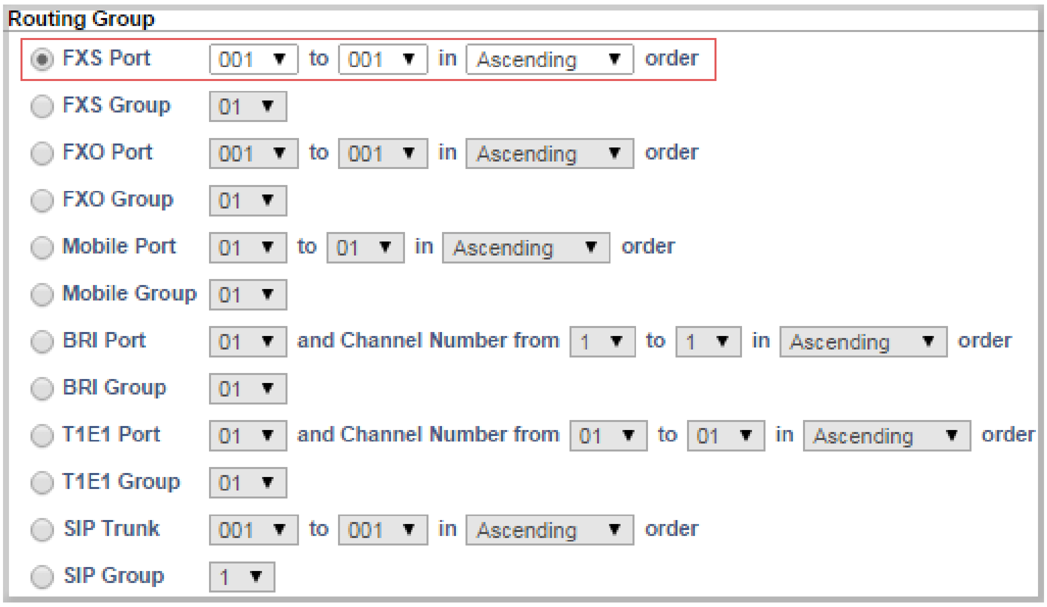
- In **Calling Number**, enter the number (max. 24 characters) from which you expect calls to be received. Valid digits are 0 to 9, *, #, (dot). Default: Blank.

- Select the **CLI Number to be sent on Destination Port**. You can select Received Calling Party or Received Called Party. Default: Received Calling Party.

CLI to be sent on Destination Port is applicable when FXS Port/Group is determined as the Destination Port.

- Create the **Routing Group**.

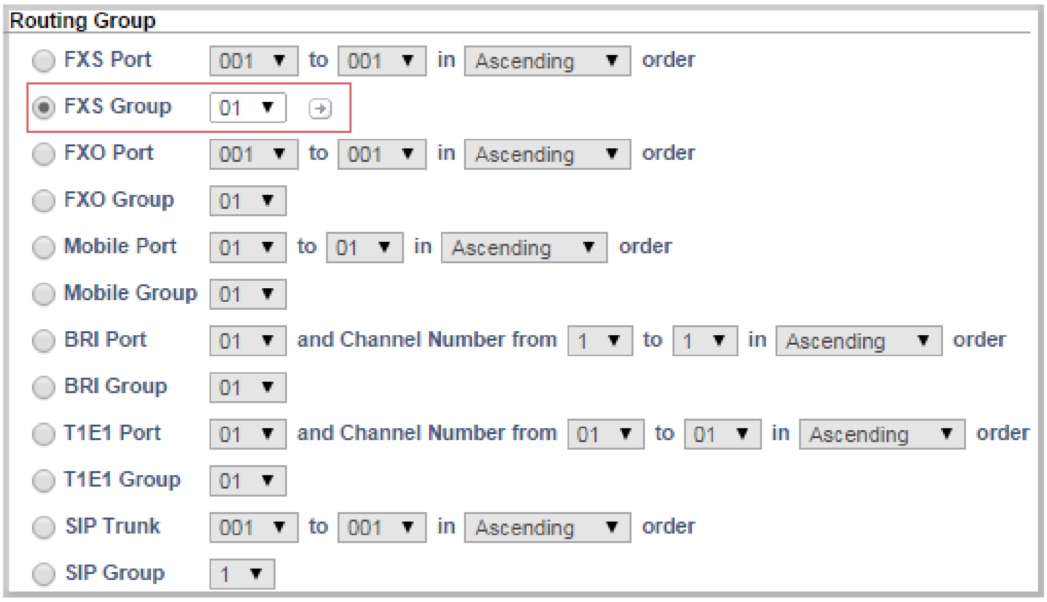
- To create a group of *sequential* **FXS Ports** as members,



  - Select the desired **FXS Port** numbers as members. Default: 1.
  - In **in - order**, select the order in which the system should hunt for a free member FXS Port to route the call.

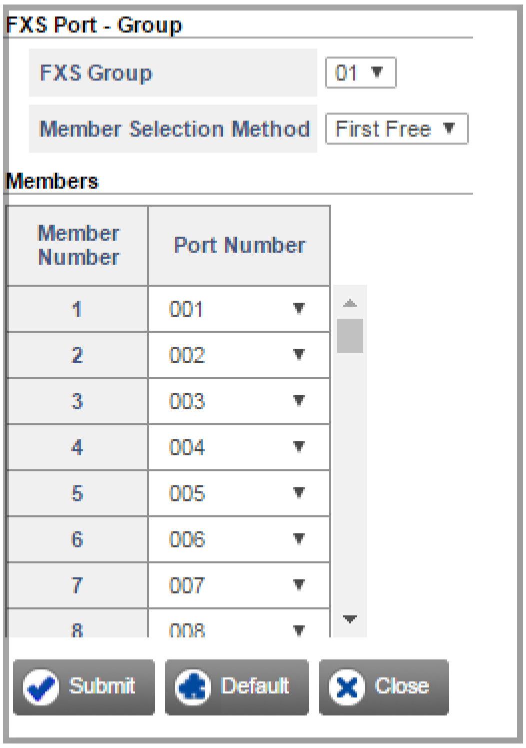
    Select **Ascending** to start hunting from the first to the last member FXS Port. Select **Descending** to start hunting from the last to the first member FXS Port. Default: Ascending.

- To create a group of *not-sequential* **FXS Ports** as members,
  - Select a **FXS Group**.



  - Select **FXS Group** number. Default: 1.

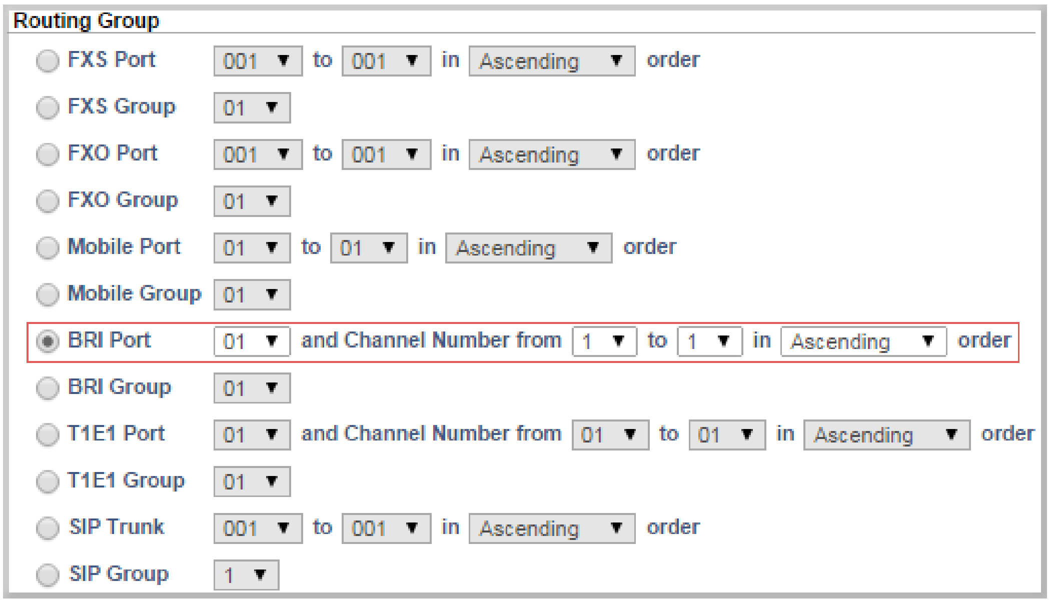
- Click **Settings** .
- The **FXS Port - Groups** window opens.



- Create the FXS Group. For detailed instructions on creating groups, see the topic “Group” under *Advanced Settings*.

- Similarly, you can create a group of *sequential* and *not-sequential* FXO Ports, Mobile Ports and SIP Trunks.

- To create a routing group of *sequential* **BRI Channels** as members,

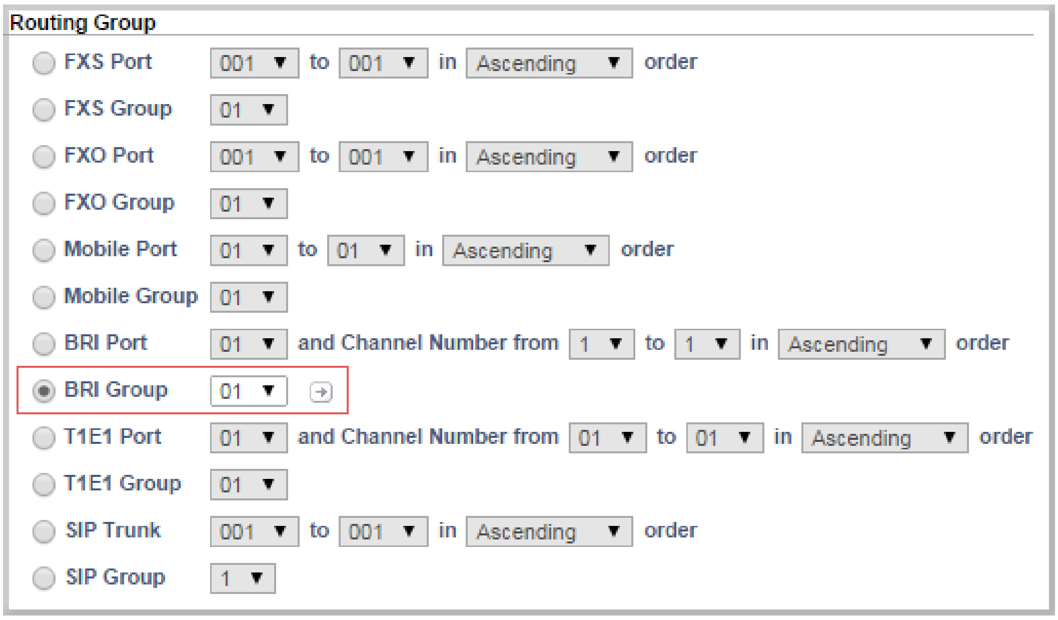


  - Select the **BRI Port** Number. Default: 1.
  - In Channel Number **From - to**, select the **Start Channel Number** and the **End Channel Number**, respectively.
  - In **in - order**, select the order in which the system should hunt for a free member Channel to route the call.

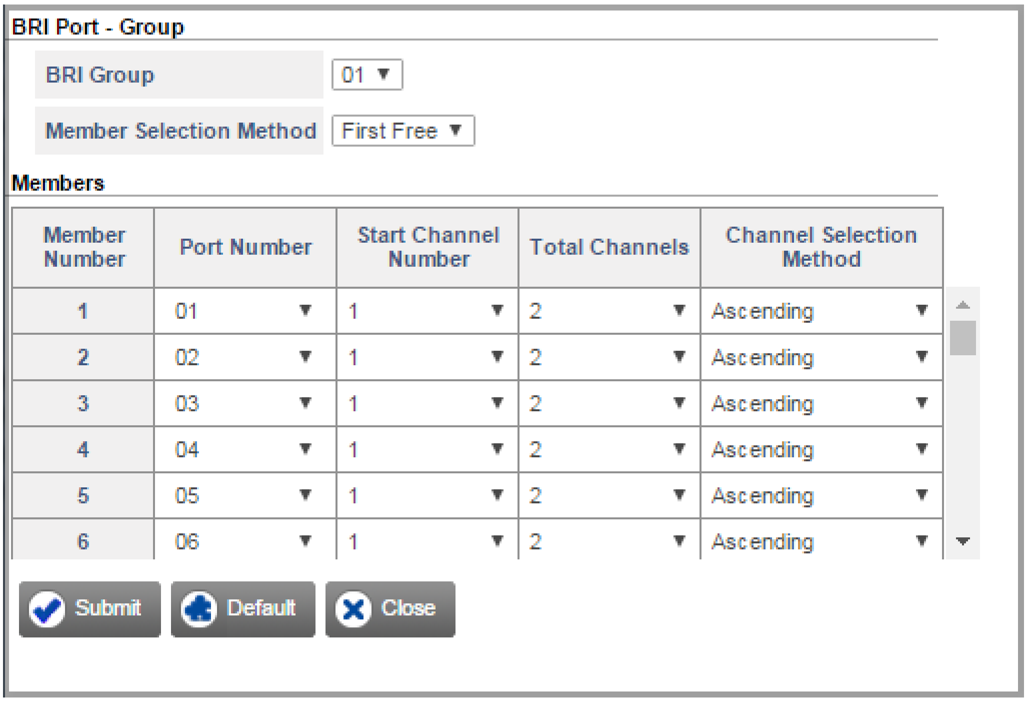
    Select **Ascending** to start hunting from the first to the last member channel. Select **Descending** to start hunting from the last to the first member channel. Default: Ascending.

- To create a group of *not-sequential* **BRI Channels** as members,

- Select **BRI Group**.



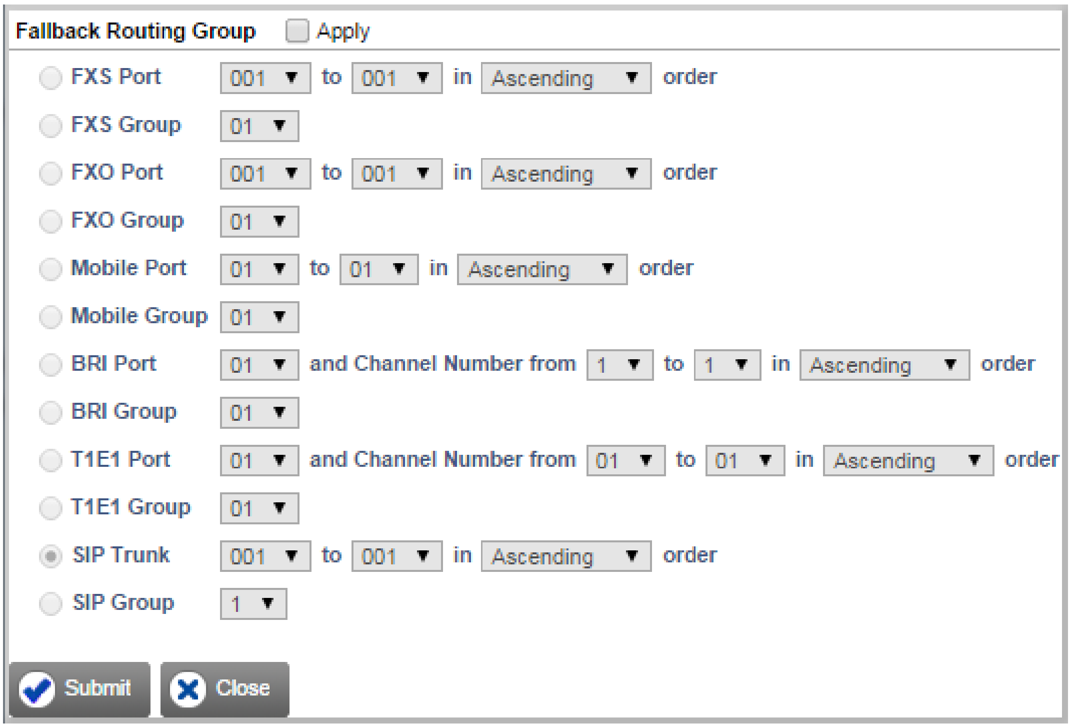
- Select a **BRI Group** number. Default: 1.
- Click **Settings** .
- The **BRI Port - Groups** window opens.



- Create the BRI Group. For detailed instructions on creating groups, see the topic “Group” under *Advanced Settings.*

- Similarly, you can create a group of *sequential* and *not-sequential* T1E1 Ports.

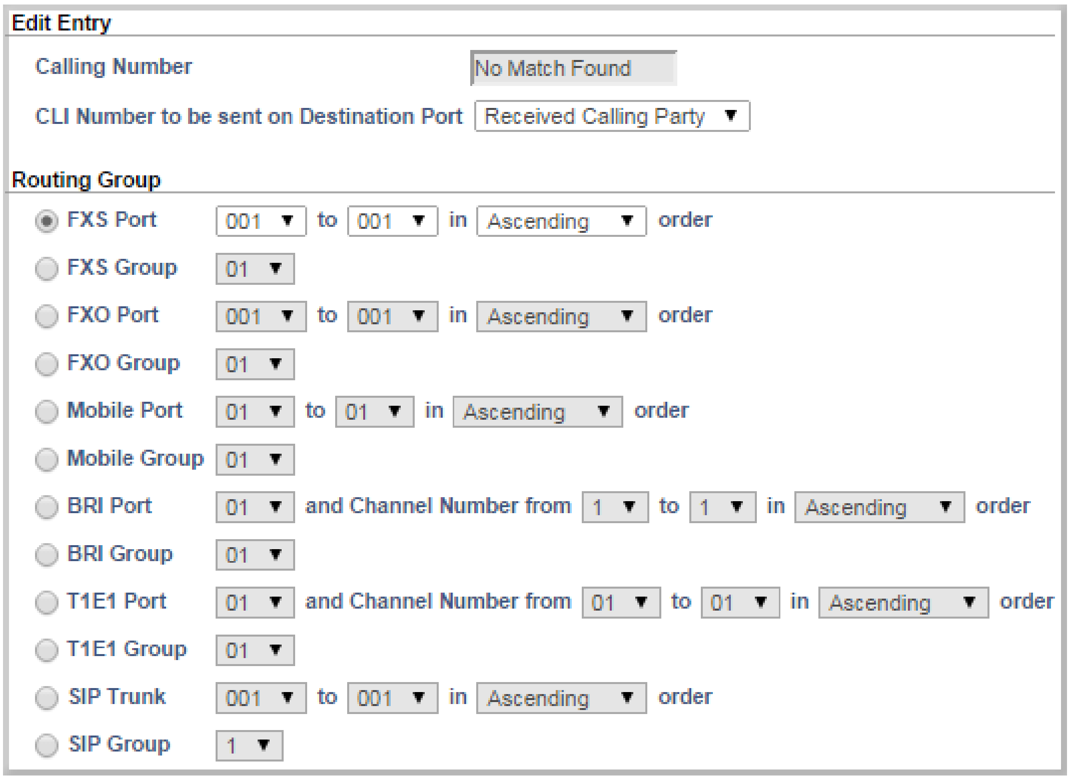
- You may create the **Fallback Routing Group**.



- To do this,
  - Select the **Apply** check box.
  - Follow the same instructions provided earlier for creating *sequential* and *not-sequential* groups of FXS Ports, FXO Ports, Mobile Ports, BRI Ports, T1E1 Ports and SIP Trunks.
- Click **Submit** to save changes. The **Add Entry** window closes.
- The entry you added appears in the **SIP Trunk - Destination Port Determination - Calling Number Based** table.
- Follow the same steps as above to add another entry to this table.
- To delete an entry, select the check box and click the **Delete** button.
- By default, FXS Port 1 - 1 (Ascending) is assigned as the Routing Group, for routing calls from numbers that do not match with any of the destination numbers you configured (No Match Found).

  To change the default Routing Group and to create the Fallback Routing Group for the No Match Found numbers entry,

- For the No Match Found entry in the table, click **Edit**.



- The **Edit Entry** window opens.
- Create the **Routing Group** and **Fallback Routing Group** as per your requirement.
- Click **Submit** and close the window.

- Close the window if you have finished adding/editing entries.

  You can also configure the **Calling Number Based** Table from *Advanced Settings*. For instructions, see "Destination Port Determination" under *Advanced Settings.*

Allowed - Denied Logic

You can apply the Allowed-Denied logic on the SIP Trunk (source port) if you want to allow or restrict the dialing of particular numbers. You can use this feature for Toll Control.

The Allowed-Denied Number Logic makes use of two Number lists:

- **Allowed Numbers List:** This is the list of numbers that can be dialed out from the SIP Trunk.
- **Denied Numbers List:** This list contains the numbers that are to be restricted from being dialed out from the SIP Trunk.

When Allowed-Denied Logic is enabled on a source port, for each number dialed from the port, SARVAM UMG uses the best-match-found logic to compare the dialed number with the Allowed Number list and the Denied Number list.

The number is allowed to be dialed, if it:

- matches with both lists.
- matches with Allowed Number list, but not with the Denied Number list.
- matches with neither the Allowed List nor the Denied List.

The number is denied, if it matches with the Denied Number list, but not with the Allowed Number list.

The system does not apply the Allowed-Denied Logic:

- When dialed number string matches with any Access Code.
- When dialed number string matches with any Emergency Number.
- When any one of the following is selected to Route all Incoming Calls (with CLI):
  - on the basis of Calling Party Number
  - to a Fixed Destination Number
  - on the basis of DDI Number

To apply Allowed - Denied Logic on the SIP Trunk,

- Select the **Allowed - Denied Logic** check box.

- In the **Allowed Number List**, select the list number you have configured with numbers you want to allow to be dialed out from the SIP Trunk. Default: 07

  If you have not configured the Allowed Number List,

  - Click **Settings** . The Number Lists window opens.
  - You may configure the default Allowed Number List or any other list. See "Number Lists" to configure the allowed numbers.
  - Click **Submit** to save the Allowed Number List and close the window.

- In the **Denied Number List**, select the list number you have configured with numbers you want to restrict to be dialed out from the SIP Trunk. Default: 08

  If you have not configured the Denied Number List,

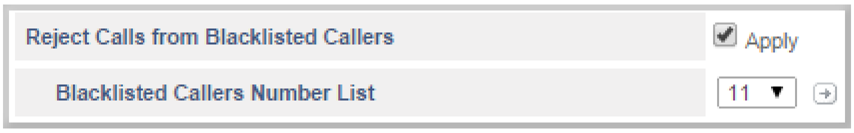
  - Click **Settings** . The Number Lists window opens.
  - You may configure the default Denied Number List or any other list. See "Number Lists" to configure the restricted numbers.
  - Click **Submit** to save the Denied Number List and close the window.

Black Listed Callers

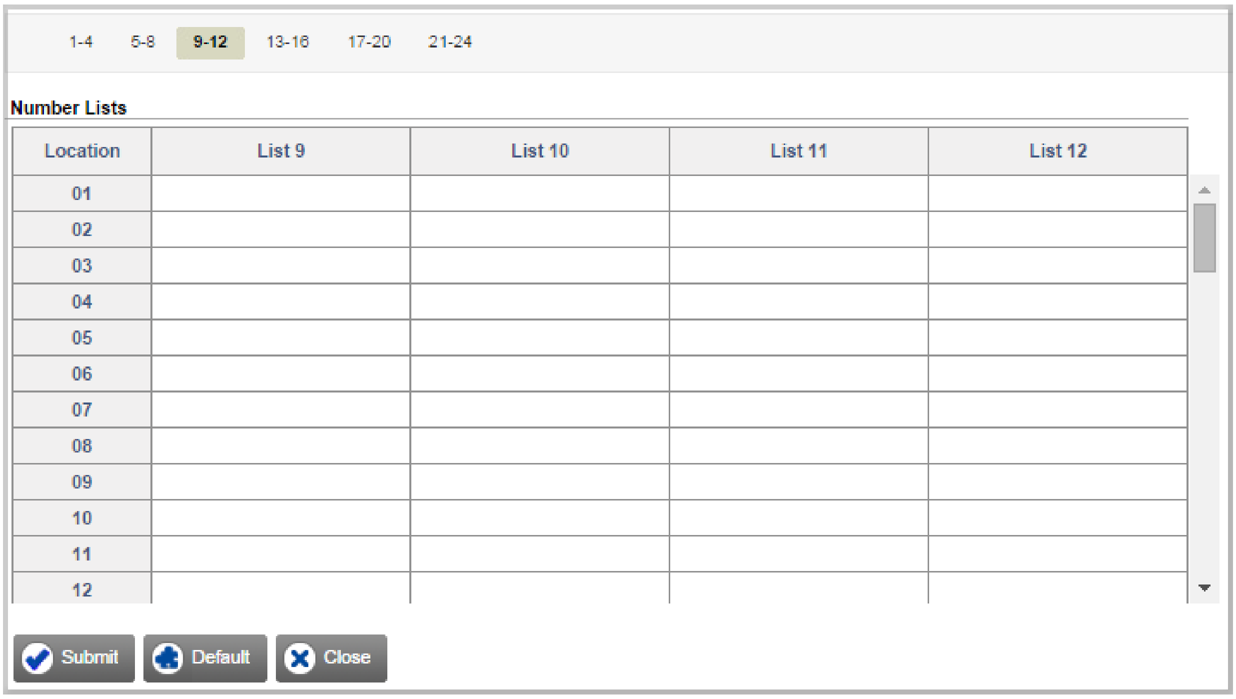
With the Black Listed Callers feature you can block incoming calls from specific addresses/numbers on SIP Trunks. Thus all incoming calls from the numbers you have 'blacklisted' will be automatically rejected by SARVAM UMG.

To apply Black Listed Callers on SIP Trunk,

- Select the **Reject Calls from Blacklisted Callers** check box.



- Configure the **Black Listed Callers** table.
- To do this,
  - Click **Settings** .
  - The Number List window opens.



  - By default, Number List 11 is assigned as Black Listed Callers List.
  - Enter the numbers of unwanted callers in this list.
- Click **Submit** to save the entries and close the window to return to the main page.

Display Received URI as Calling Name

- Keep the **Display Received URI as Calling Name** check box enabled.

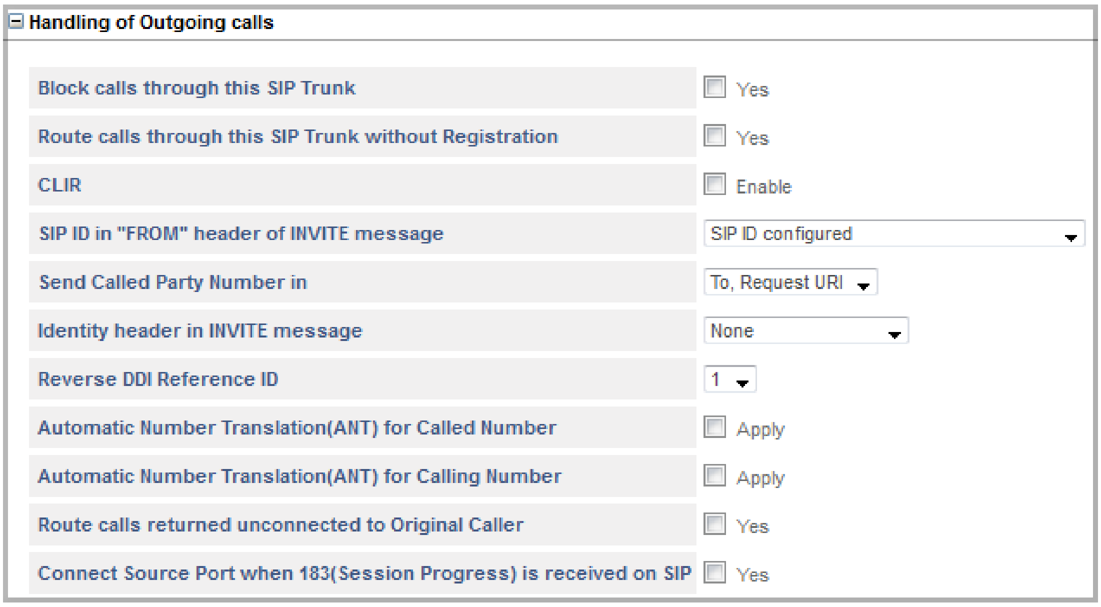
When Name is received in the "FROM" header for incoming call on the SIP Trunk, SARVAM UMG will display name and received URI as calling name. When Name is not received, SARVAM UMG will display only the received URI as calling name.

You may disable this check box, if you do not want the system to display the received URI as calling name on the SIP Trunk.

- Click **Submit**.

Handling of Outgoing Calls

When a SIP Trunk is determined as the destination port, numbers dialed from this port constitute outgoing calls.



- Click **Handling of Outgoing calls** to expand.
- If you do not want to route outgoing calls through this SIP Trunk, select the **Block calls through this SIP Trunk** check box.
- To allow the users to make outgoing calls irrespective of whether the SIP Trunk has been successfully registered with the proxy or not, select the **Route Calls through this SIP Trunk without Registration** check box.

  By default, the system does not allow outgoing calls to be made if the status of the SIP Trunk is 'not registered'.
- By default, the CLI of the SIP Trunk is sent to the called party when outgoing calls are made using the SIP Trunk. If you do not want to send CLI, enable the **CLIR** check box. Default: Disabled.

- SARVAM UMG supports flexible options for sending **SIP ID in "FROM" header of INVITE message** during an outgoing call. You may select the desired option — SIP ID configured, Caller ID received on Source Port, Caller ID after applying Reverse DDI logic, Fixed Number. Default: SIP ID configured.
  - If you select *Caller ID after applying Reverse DDI logic*, SARVAM UMG allows you to configure the desired option for **If no match found using Reverse DDI logic** — SIP ID configured, Caller ID received on Source Port, Fixed Number. Default: SIP ID configured.
  - If you select *Fixed Number* option for **SIP ID in "FROM" header of INVITE message** or **If no match found using Reverse DDI logic**, you must configure the **Fixed Number**. The Fixed Number can be a maximum of 24 characters. Characters 0-9, +, * # and dot(.) are allowed. Default: Blank.
- SARVAM UMG 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.
- SARVAM UMG also offers flexible options for sending **Identity header in INVITE message** during an outgoing call. You may select the desired option — None, P-Preferred Identity, P-Asserted Identity— according to the Identity header supported by your service provider. Default: None.
  - SARVAM UMG supports flexible options for sending **SIP ID in Identity header of INVITE message** during an outgoing call. You may select the desired option — Send SIP ID configured, Send Caller ID received on Source Port, Send Caller ID after applying Reverse DDI logic, Send Fixed Number. Default: Send SIP ID configured.

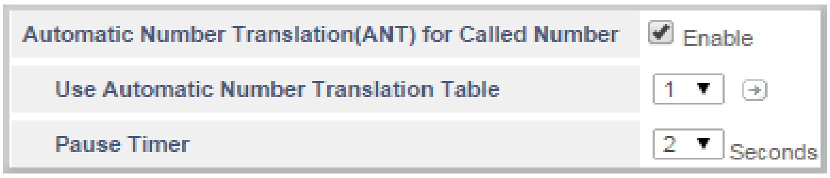
SIP ID in Identity header of INVITE message** can be configured only when you have selected either P-Preferred Identity or P-Asserted Identity as **Identity header in INVITE message.

  - If you select *Send Caller ID after applying Reverse DDI logic*, SARVAM UMG allows you to configure the desired option for **If no match found using Reverse DDI logic** — Send SIP ID configured, Send Caller ID received on Source Port, Send Fixed Number. Default: SIP ID configured.
  - If you select *Send Fixed Number* as an option for **SIP ID in Identity header of INVITE message** or **If no match found using Reverse DDI logic**, you must configure the **Fixed Number**. The Fixed Number can be a maximum of 24 digits. Characters 0-9, +, *, # and dot(.) are allowed. Default: Blank.

*If you have enabled **CLIR** and **SIP ID in Identity header in INVITE message** is configured, then SARVAM UMG will add **Privacy = ID** header in the INVITE message during an outgoing call from the SIP Trunk.*

- Select **Reverse DDI Reference ID**, if you have selected either/ both of the following:
    - *Caller ID after applying Reverse DDI logic* option as the **SIP ID in "FROM" header of INVITE message**.
    - *Send Caller ID after applying Reverse DDI logic* option as the **SIP ID in Identity header of INVITE message.**

      SARVAM UMG will compare the Reference ID configured on the SIP Trunk with the one configured in the SIP Trunk - Destination Number Determination: DDI Number Based Table. If a match is found, SARVAM UMG will send the corresponding DDI Number to the Called Party.
- You can apply **Automatic Number Translation logic** on outgoing calls made from the SIP Trunk.
    - To apply ANT logic on the Called Numbers, select the **Automatic Number Translation (ANT) for Called Number** check box. Default: Disabled.

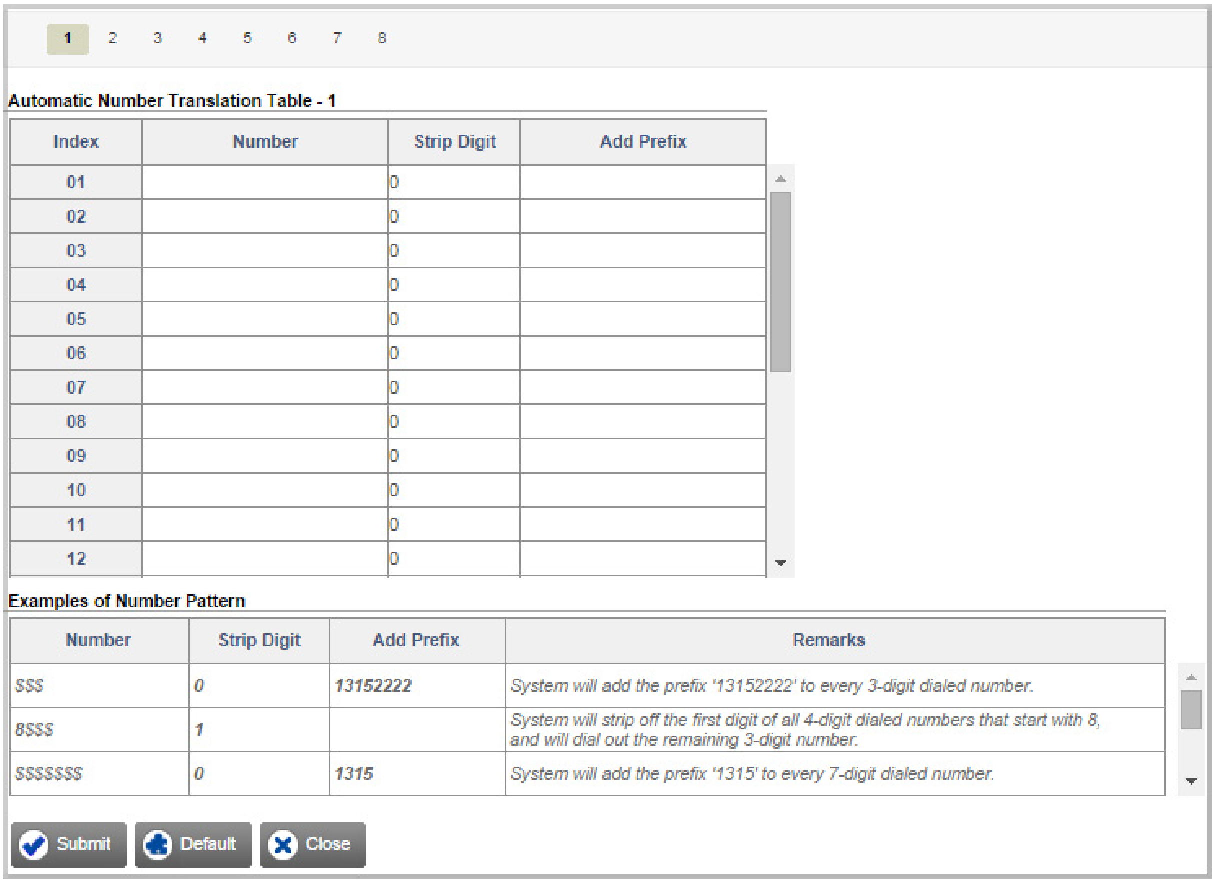


    - In **Use Automatic Number Translation Table**, select the ANT Table number you have configured for the Called Numbers. Default: Table 1.

      If you have not configured the Automatic Number Translation Table,

        - Click **Settings** .

- The Automatic Number Translation Table window opens.



- You may configure the default Automatic Number Translation Table or any other Table. See "Automatic Number Translation (ANT)" to configure the ANT Table.

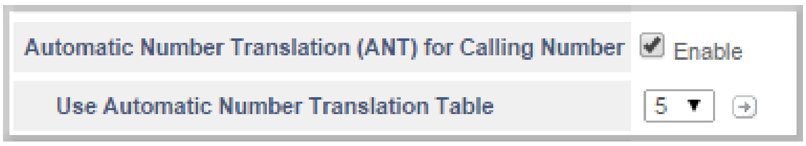
- Click **Submit** to save the ANT Table and close the window.

- Return to ANT parameter and assign the ANT Table you configured.

- Click **Submit**.

- Set the duration of the **Pause Timer**, if you have configured ^ (Pause) in the Add Prefix column of the ANT Table. Valid range is 1 to 9 seconds. Default: 2 seconds.

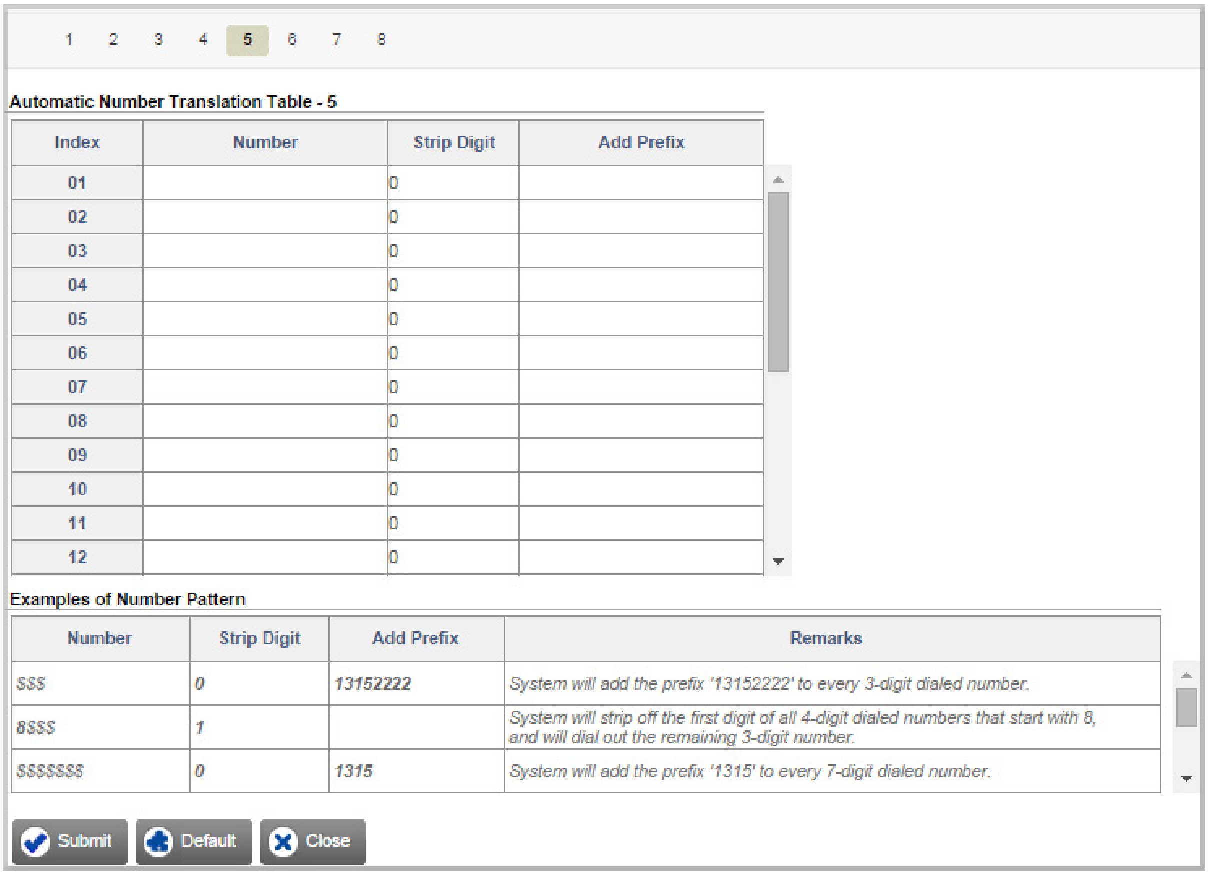
- To apply ANT logic on the Calling Numbers, select the **Automatic Number Translation (ANT) for Calling Number** check box. Default: Disabled.



- In **Use Automatic Number Translation Table**, select the ANT Table number you have configured for the Calling Numbers. Default: Table 5.

  If you have not configured the Automatic Number Translation Table,

- Click **Settings** .
- The Automatic Number Translation Table window opens.



1 2 3 4 5 6 7 8

Automatic Number Translation Table - 5

Index	Number	Strip Digit	Add Prefix
01		0	
02		0	
03		0	
04		0	
05		0	
06		0	
07		0	
08		0	
09		0	
10		0	
11		0	
12		0	

Examples of Number Pattern

Number	Strip Digit	Add Prefix	Remarks
$$$	*0*	*13152222*	*System will add the prefix '13152222' to every 3-digit dialed number.*
8$$$	*1*		*System will strip off the first digit of all 4-digit dialed numbers that start with 8, and will dial out the remaining 3-digit number.*
$$$$$$$	*0*	*1315*	*System will add the prefix '1315' to every 7-digit dialed number.*

Submit Default Close

- You may configure the default Automatic Number Translation Table 5 or any other Table. See "Automatic Number Translation (ANT)" to configure the ANT Table.
- Click **Submit** to save the ANT Table and close the window.
- Return to ANT parameter and assign the ANT Table you configured.
- Click **Submit** to apply List.

- Select the **Route calls returned unconnected to Original Caller** check box, if you want SARVAM UMG to route outgoing calls made from this Trunk that return unconnected back to the original caller.

  If you enable this feature, when an outgoing call is made using this Trunk, and the Called Party is found busy or does not respond, SARVAM UMG stores the number of the calling party, the number of the called party and this trunk (through which the outgoing call was made). A record of each such call is stored for the duration of the Unconnected Calls Record Delete Timer (configurable; default: 999 minutes).

  If the called party returns the call before the expiry of this Timer, SARVAM UMG checks whether *Apply RCOC only if the caller calls back on the same trunk from which the call was made* is enabled or not, and accordingly places the incoming call to the original calling party. To change the duration of this timer, delete records of such calls and enable/disable the *Apply RCOC only if the caller calls back on the same trunk from which the call was made* check box, see "System Parameters".

- To connect the Source Port with the Destination Port without waiting for the call on the Destination Port to mature, select the **Connect Source Port when 183(Session Progress) is received on SIP** check box to enable. Default: Disabled.

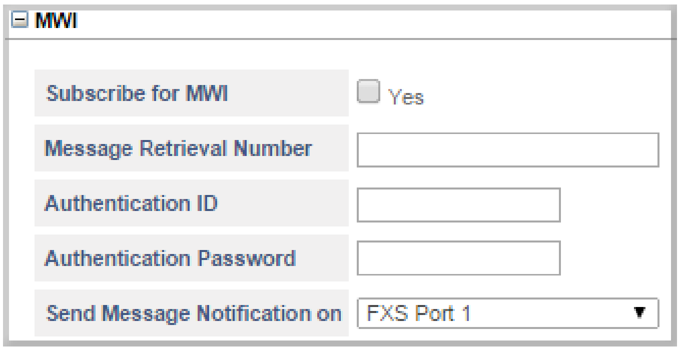
  In all Destination Number Determination methods except *After Answering the Call and Collecting the Digits*, the Source Port gets connected to the Destination Port only after the call has matured, that is, the called party has answered the call. Until the call matures, the caller hears only Ring Back Tone played by the network.

  By connecting the Source Port with the Destination Port immediately after the number is dialed, the caller can know the state of the call; if the called party is busy, not responding, not reachable or is rejecting the call.

If you enable **Connect Source Port when 183(Session Progress) is received on SIP**, *you will not be able to provide the features "Making a New Call using Access Code" and "Disconnecting a Call using Access Code" to users.*

- Click **Submit** to save the changes.

Message Wait Indication (MWI)



Message Wait Indication parameters are applicable only when SIP Trunk is configured as Proxy.

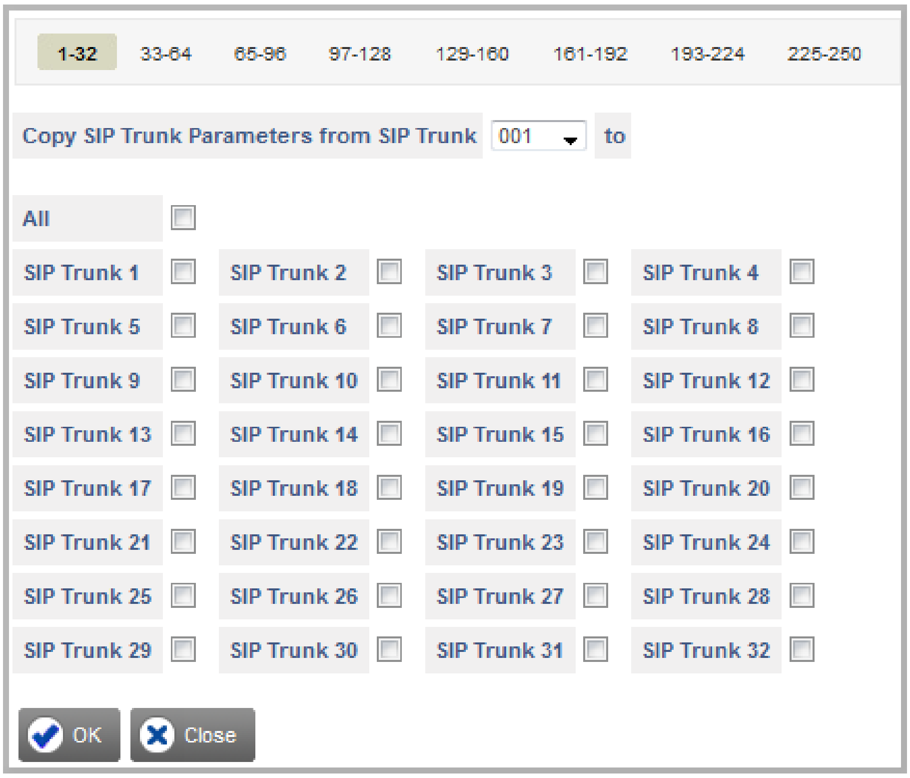
- If you have subscribed for Message Wait Indication on the SIP Trunk for the voicemail service from your ITSP, click **MWI** and configure the following parameters:
  - Select the **Subscribe for MWI** check box. Default: Disabled.
  - In **Message Retrieval Number**, enter the number provided to you by your ITSP. This number is used for retrieval of voicemail on the SIP Trunk. The Message Retrieval Number may consist of a maximum of 24 characters. Valid range is 0 to 9, * and # are allowed. Default: Blank.
  - Enter the **Authentication ID** (User ID) provided by your ITSP. Default: Blank.
  - Enter the **Authentication Password** provided by your ITSP. Default: Blank.
  - In **Send Message Notification on**, select the FXS Port on which Message Wait Indication is to be sent whenever there is a new message on the SIP Trunk. Default: FXS Port 1.

To know more about MWI, see “Message Wait Indication on SIP Trunks”.

- If you have completed the configuration of SIP Trunk 1, click **Submit** to save settings.
- Close the window.
- To configure the next SIP Trunk, click the desired SIP Trunk number and follow the same instructions as given above.

Copy SIP Trunk Parameters

- You can also copy the settings of a SIP Trunk to another SIP Trunk using the **Copy** button. To do this,
  - Click the **Copy** button. The **Copy SIP Trunk Parameters** window opens.



  - In the **Copy SIP Trunk Parameters from SIP Trunk** box, select the number of the trunk you want to copy settings *From*. Select the check box of the respective trunk numbers you want to copy the settings *To*.
  - If you want to copy the settings *To* all the trunks, select the **All** check box.
  - Click the **OK** button.
  - Once you have copied the settings, you can again edit the specific parameters of the SIP Trunk you copied the settings to.

T1 Port

SARVAM UMG supports the T1/E1 Ports to which you can connect the T1 or E1 line.

- Click the **Basic Settings** link to expand.

- Click the **T1E1 Port** link.

Port	Hardware Slot - Port	Enable	Name	Status	Line Signaling	Orientation	Call Routing
T1E1-1	0 - 0	☑		Layer 1 - Down Layer 2 - Down	E1 - PRI ETSI NET5	Terminal	Route calls to number received in SETUP message using SIP Trunk 1 - 1
T1E1-2	0 - 0	☑		Layer 1 - Down Layer 2 - Down	E1 - PRI ETSI NET5	Terminal	Route calls to number received in SETUP message using SIP Trunk 1 - 1
T1E1-3	0 - 0	☑		Layer 1 - Down Layer 2 - Down	E1 - PRI ETSI NET5	Terminal	Route calls to number received in SETUP message using SIP Trunk 1 - 1
T1E1-4	0 - 0	☑		Layer 1 - Down Layer 2 - Down	E1 - PRI ETSI NET5	Terminal	Route calls to number received in SETUP message using SIP Trunk 1 - 1
T1E1-5	0 - 0	☑		Layer 1 - Down Layer 2 - Down	E1 - PRI ETSI NET5	Terminal	Route calls to number received in SETUP message using SIP Trunk 1 - 1
T1E1-6	0 - 0	☑		Layer 1 - Down Layer 2 - Down	E1 - PRI ETSI NET5	Terminal	Route calls to number received in SETUP message using SIP Trunk 1 - 1
T1E1-7	0 - 0	☑		Layer 1 - Down Layer 2 - Down	E1 - PRI ETSI NET5	Terminal	Route calls to number received in SETUP message using SIP Trunk 1 - 1
T1E1-8	0 - 0	☑		Layer 1 - Down Layer 2 - Down	E1 - PRI ETSI NET5	Terminal	Route calls to number received in SETUP message using SIP Trunk 1 - 1

The T1E1 Port page displays the following parameters:

- **Port:** It displays the T1E1 Port numbers. Click on the desired T1E1 Port number to configure the Port Parameters.

- **Hardware Slot-Port:** The SARVAM UMG can automatically detect and assign the Hardware Slot and Port numbers to the T1E1 software ports. However, if required you may change the Hardware Slot and Port assigned to the T1E1 software port. In this case, enter the desired Hardware Slot and Port number.

- **Enable:** Keep the **T1E1 Ports** enabled. Clear the T1E1 Port **Enable** check box, only if you do not want to use the respective port. Default: Enabled.

- **Name:** Assign a Name to the T1E1 Port for identification. The Name can be a maximum of 24 characters.

- **Status:** This displays the status of Layer 1 and Layer 2, that is, Up or Down.

- **Line Signaling:** It displays the Carrier Type, Signaling Type and the ISDN Switch Variant you select.

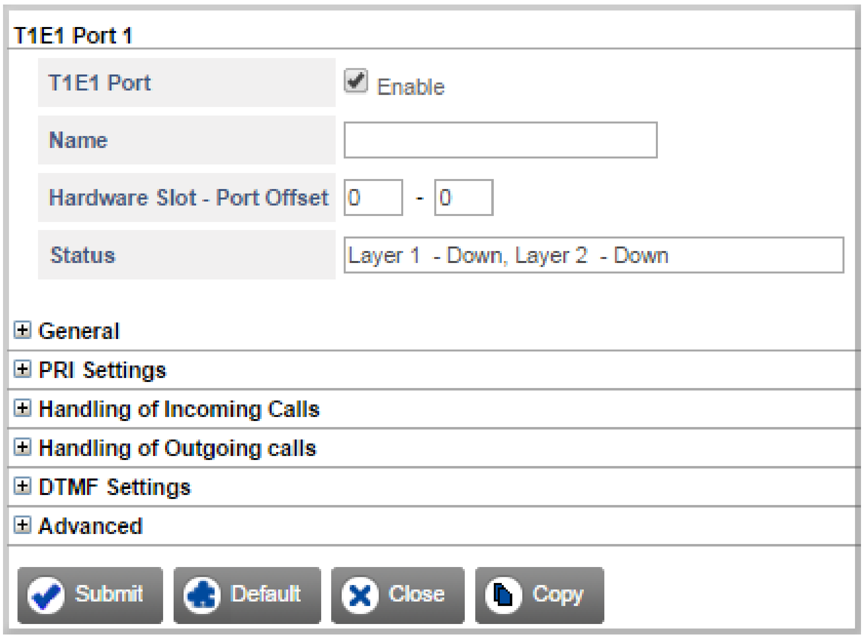
- **Orientation:** It displays the type of orientation you select — Network or Terminal.

- **Call Routing:** It displays the Call Routing Method you select.

To configure the **T1E1 Port**,

- Click **T1E1-1**.

The **T1E1 Port-1** window opens.



- Keep the **T1E1 Port** check box enabled.

  Clear the **T1E1 Port Enable** check box only when you do not want to use this T1E1 Port. Default: Enabled.

- You can assign a **Name** to the T1E1 Port, which will be displayed to the called party, if the called party telephone instrument supports name display.

  The name you assign may consist of a maximum of 24 characters. Default: Blank.

- SARVAM UMG will assign the **Hardware Slot - Port Offset** automatically, when any card is inserted in the system.

  Hardware slot is the number of the universal slot of SARVAM UMG in which the T1E1 Card is inserted. Range of slot number is 1-12. Port is the number of T1E1 hardware port on the card to which the T1E1 line is connected.

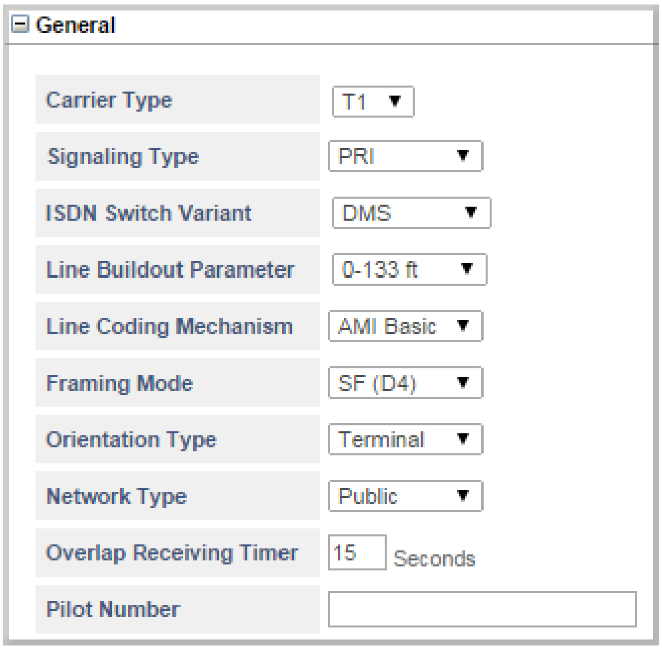
  However, if required, you may change the Hardware Slot and Port assigned to the T1E1 software port. In this case, enter the desired Hardware Slot and Port number.

  If you want to de-assign the Hardware Slot and Port, enter '00' in both fields. By default, Hardware Slot-Port is 00–00.

- **Status** displays the status of the T1E1 Port.

General

- Click **General** to expand.
- Select **T1** as the **Carrier Type**. Default: E1.



- Select **Signaling Type**. The Signaling Type signifies the type of signaling to be used on the T1 line.

  SARVAM UMG supports — PRI and RBS signaling for T1 line. Default: PRI.

  - If you select **PRI,** you must configure the PRI parameters. For instructions, see “PRI Settings”.
  - If you select **RBS,** you must configure the RBS parameters. For instructions, see “RBS Settings”.

- ISDN supports a variety of service provider switches. These switches are designed using ISDN standard protocol. The type of switch you select determines various factors — the number of ISDN devices that could be handled, the B-Channel that would support voice, video, data, etc. Each country uses their own specific type of ISDN switch.

  In **ISDN Switch Variant,** select — DMS, US NI2 or ATT 5ESS — as the ISDN Switch Variant. Default: DMS. This parameter is applicable only if you select PRI as the Signaling Type.

- Select the T1 **Line Buildout Parameter** for T1E1 Port. You may select — 0-133 ft., 133-266 ft., 266-399 ft., 399-533 ft., 533-655 ft., -7.5 dB,-16 dB, -22.5 dB. Default: 0-133 ft.

- Line Coding is a mechanism to code the digital data into electrical pulses for the purpose of transmission over the communication channel.

  Select the **Line Coding Mechanism —** AMI Basic or B8ZS. Default: AMI Basic.

- **Framing** is a formatting resource that splits the digital data into time slots of 8 bits each. Each time slot is treated as single transmission unit. These frames enable the receiver to interpret the data.

  Select the **Framing Mode** as per your requirement. You can select SF (D4) or ESF. Default: SF (D4)

- **SF (D4)** refers to the Superframe with 12 concatenated frames.
- **ESF** refers to the Extended Superframe with 24 concatenated frames. This type of framing mode also has cyclic redundancy checking, a maintenance channel and is used for Common Channel Signaling (CCS).

- Select the **Orientation Type** for the port as **Terminal** or **Network**, according to your installation scenario. Default: Terminal.

  If you select *Terminal* as Orientation Type, select the **Network Type** — Public or Private — to specify whether the T1 line is from a **Public** Network (telephone exchange) or from a **Private** Network (to the NT port of a System). Default: Public.

  - For **Terminal** as the Orientation Type, configure — "Handling of Incoming Calls" and "Handling of Outgoing Calls".
  - For **Network** as the Orientation Type, configure "Handling of Calls".

- Enter the **Pilot Number** provided by your service provider for the T1 line connected to the T1/E1 Port. Pilot Number is necessary for sending the calling party number when the call is routed using T1/E1 Port and Reverse DDI logic is not applied. Valid digits are 0 to 9, #, *. Default: Blank.

Handling of Incoming Calls

Click **Handling of Incoming Calls** to expand.

Select the method to route the incoming calls from the T1E1 Port.

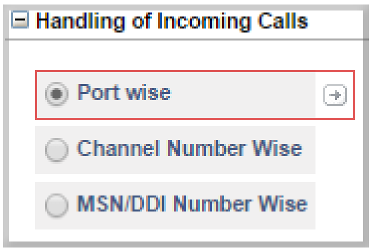
SARVAM UMG provides three options for **Handling of Incoming Calls —** Port Wise, Channel Number Wise and MSN/DDI Number Wise. Default: Port Wise.

- **Port Wise:** Select this method to apply the call routing method for the entire port.
- **Channel Number Wise:** Select this method to apply a different call routing method for each channel. You can configure a different incoming call routing option for each channel.
- **MSN/DDI Number Wise:** Select this method to apply a different call routing method for each MSN number given by the Service Provider for the T1 Line. SARVAM UMG allows you to configure upto 8 MSN Numbers.

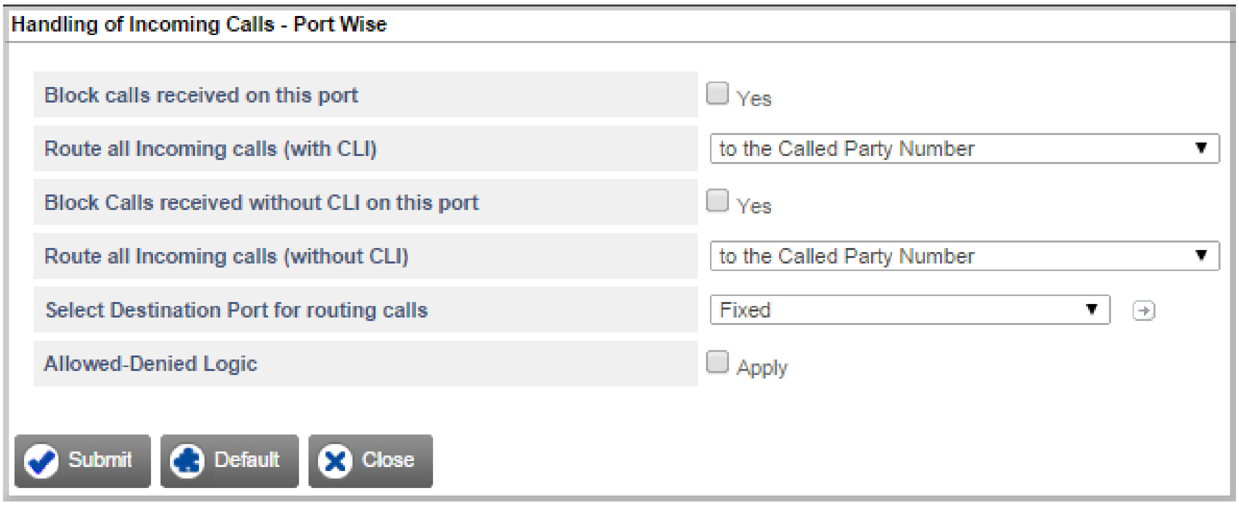
Port Wise

To configure Handling of Incoming Calls Port Wise,

- Select the **Port Wise** check box.



- Click **Settings** .
- The **Handling of Incoming Calls - Port Wise** window opens.



- Keep the **Block calls received on this port** check box disabled.

  Select this check box only if you do not want to route calls received on this port.

Destination Number Determination

Select the desired destination number determination method for routing incoming calls *with* and *without* CLI.

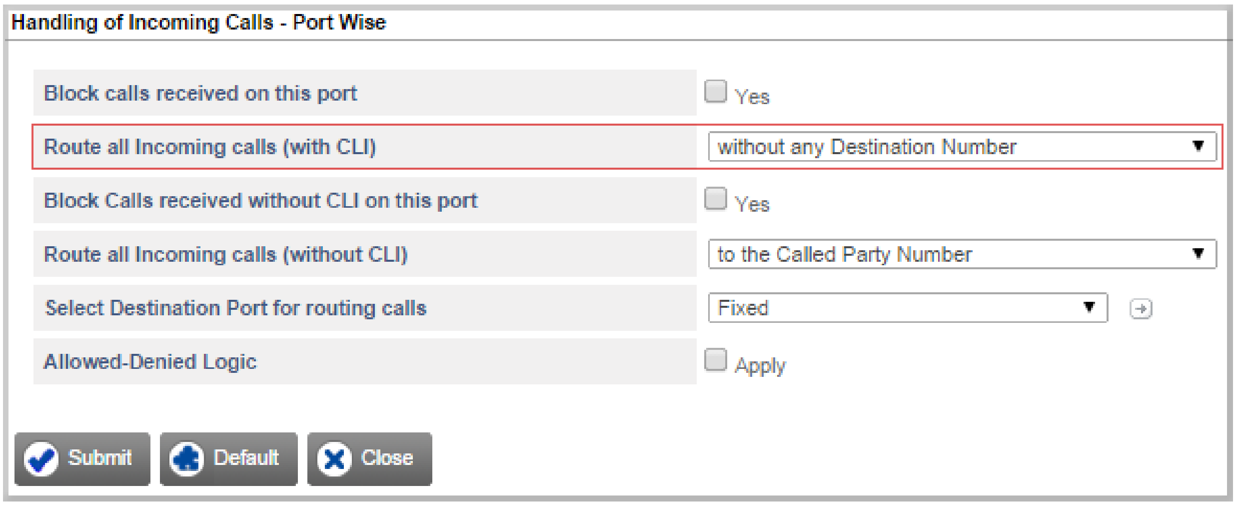
- To **Route all Incoming calls (with CLI)**, you may select from any of the following methods:
  - without any Destination Number
  - to a Fixed Destination Number
  - on the basis of Calling Party Number
  - on the basis of DDI Number
  - to the Called Party Number
  - after Answering the Call and Collecting the Digits
    Default: to the Called Party Number

  Read further for instructions on selecting and configuring each of these destination number determination methods.

If the destination number to be dialed out is an IP Address, SARVAM UMG will not check the Destination Port Determination Method. Instead, it will route the call using the SIP Trunk / Group programmed for IP Dialing. (See "IP Dialing" to know more).

Route Calls without any Destination Number

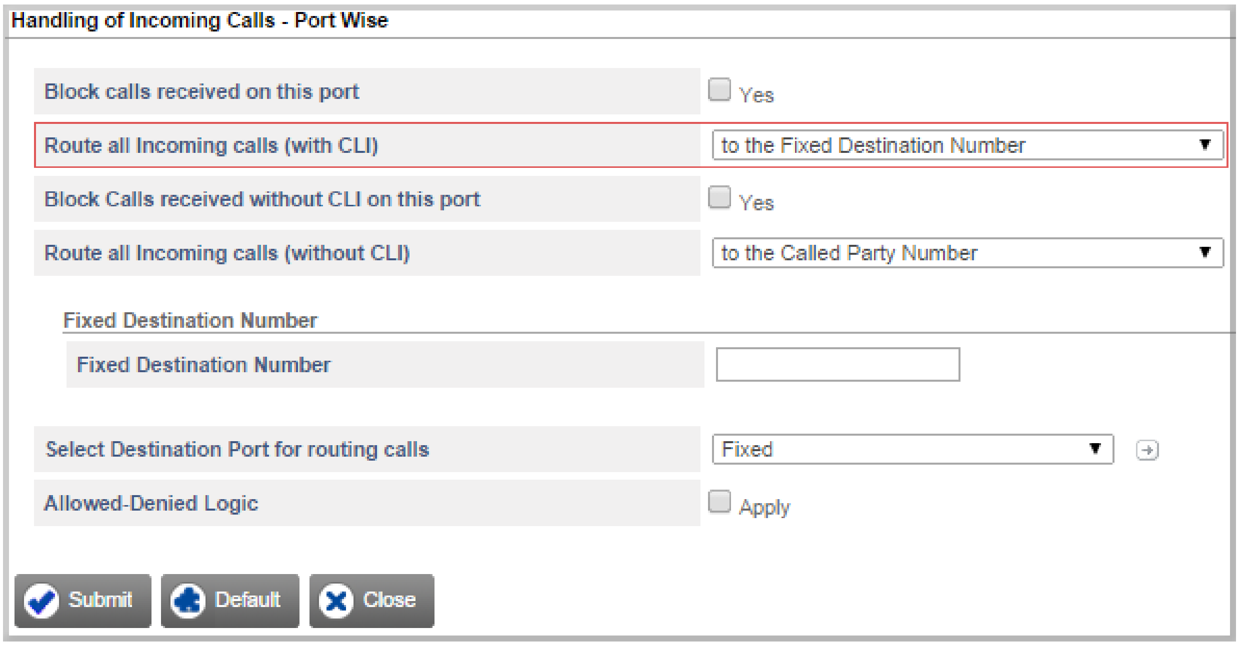
In this method, all calls received on the T1E1 Port are directly routed to the destination port, irrespective of the Destination Number.



- To apply this method, in **Route all incoming calls (with CLI)**, select **without any Destination Number**.

Route to a Fixed Destination Number

In this method, calls received on the T1E1 Port are routed to a fixed destination number, which is configured for the T1E1 Port.



To apply this method, do the following:

- In **Route all Incoming calls (with CLI)**, select **to the Fixed Destination Number**.
- In the **Fixed Destination Number** box that appears, enter the desired destination number. The Destination Number may consist of a maximum of 24 digits. Valid digits are 0 to 9, *, # and. (dot/period). Default: Blank.
- Click **Submit** to save your settings.

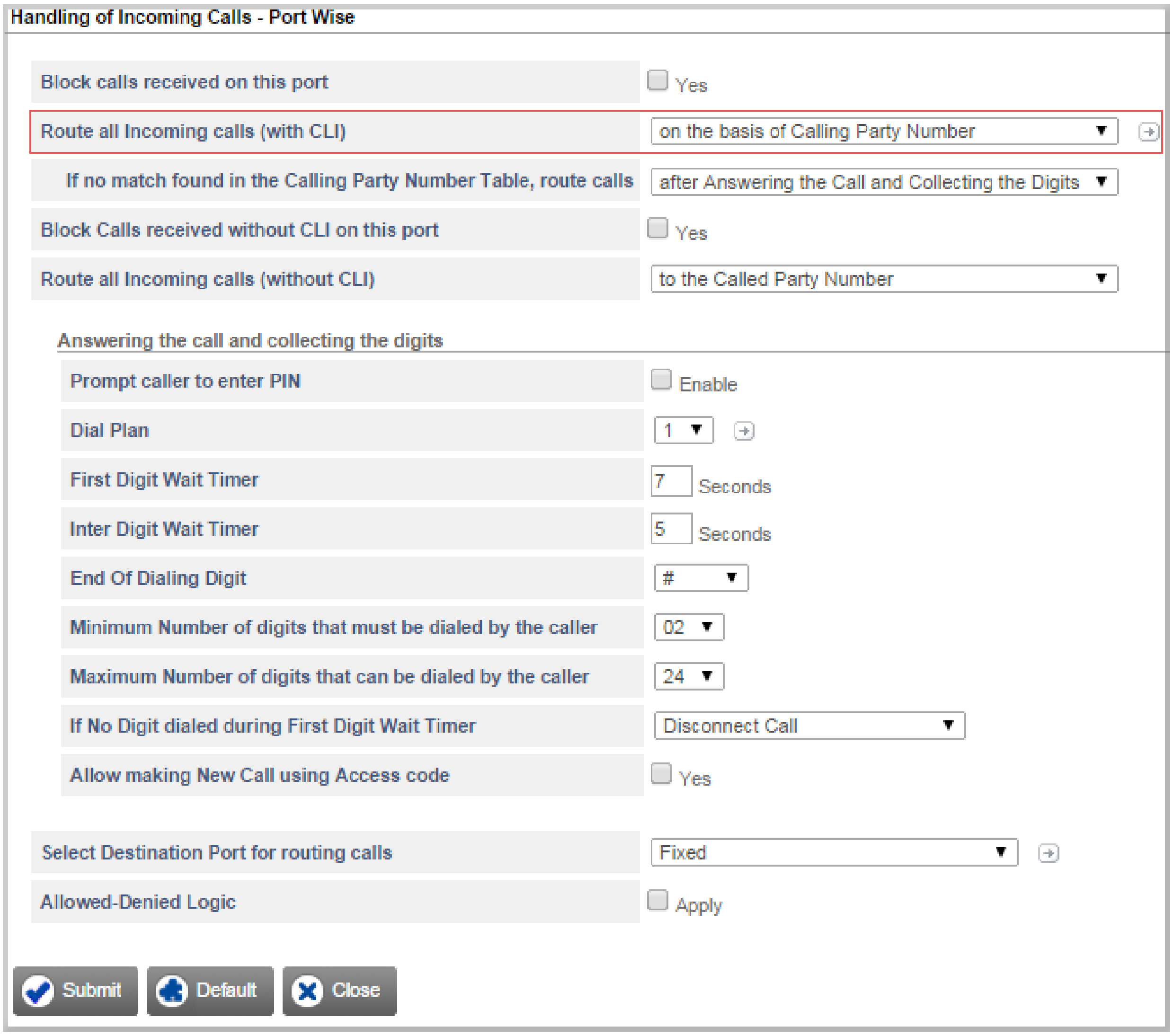
Route on the basis of Calling Party Number

In this method, a call received on the T1E1 Port is routed to a specific number, as per the calling party's number. You must configure the calling party numbers in the *Calling Party Number Based* Table.

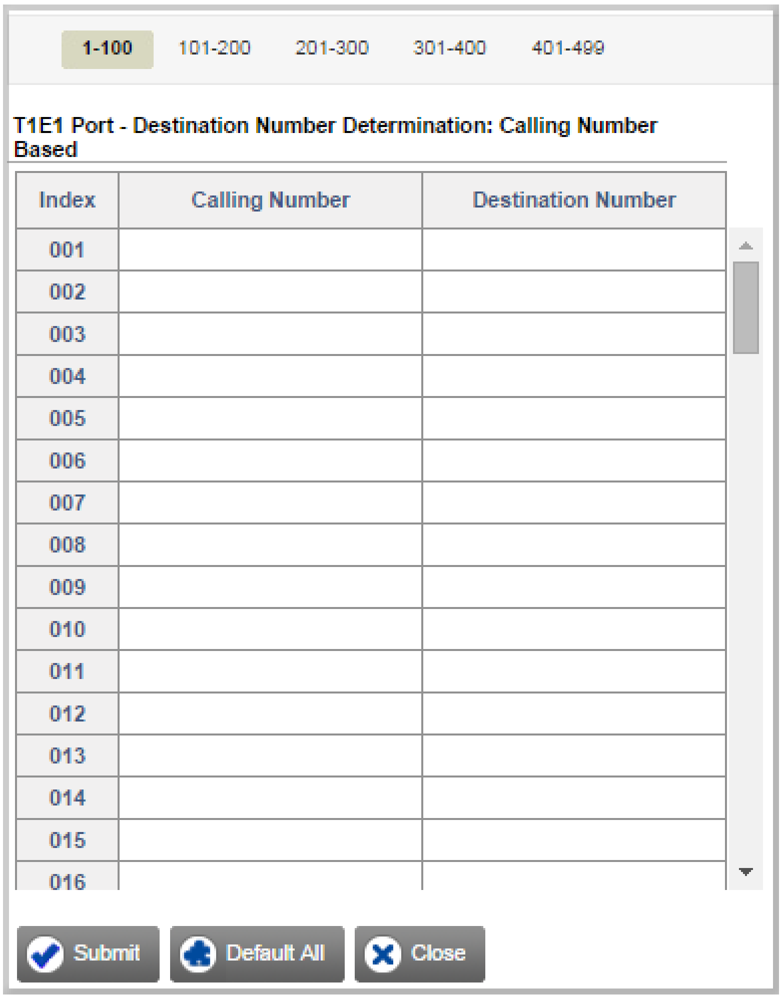
When there is an incoming call on the T1E1 Port, SARVAM UMG will match the Calling Party Number with the entries of the Calling Party Number Based Table. If a match is found, the call is routed to the destination number configured for that Calling Party Number.

To apply this method, do the following:

- In **Route all Incoming calls (with CLI)**, select **on the basis of Calling Party Number**.



- Click **Settings** .
    - The **T1E1 Port - Destination Number Determination: Calling Number Based** Table window opens.



- In **Calling Number**, enter the calling party numbers. The Calling numbers may consist of a maximum of 24 characters. Default: Blank.
- For each calling party number, enter a corresponding destination number in **Destination Number**. Destination numbers may consist of a maximum of 24 characters. Digits 0 to 9, *, # and (.) dot are allowed. Default: Blank.

- Click **Submit** to save your entries. Close the window to return to the **Handling of Incoming Calls - Port Wise** window.

  You can also configure the **Calling Number Based** table from *Advanced Settings.* For instructions, see "Destination Number Determination" under *Advanced Settings*.

- Select a method for routing incoming calls with CLI that *do not match* with any entries in the Calling Party Number Based Table.

In the **If no match found in the Calling Party Number Table, route calls** box, select the desired method from the following options for processing the call:

- to a Fixed Destination Number
- on the basis of DDI Number
- to the Called Party Number
- after Answering the Call and Collecting the Digits

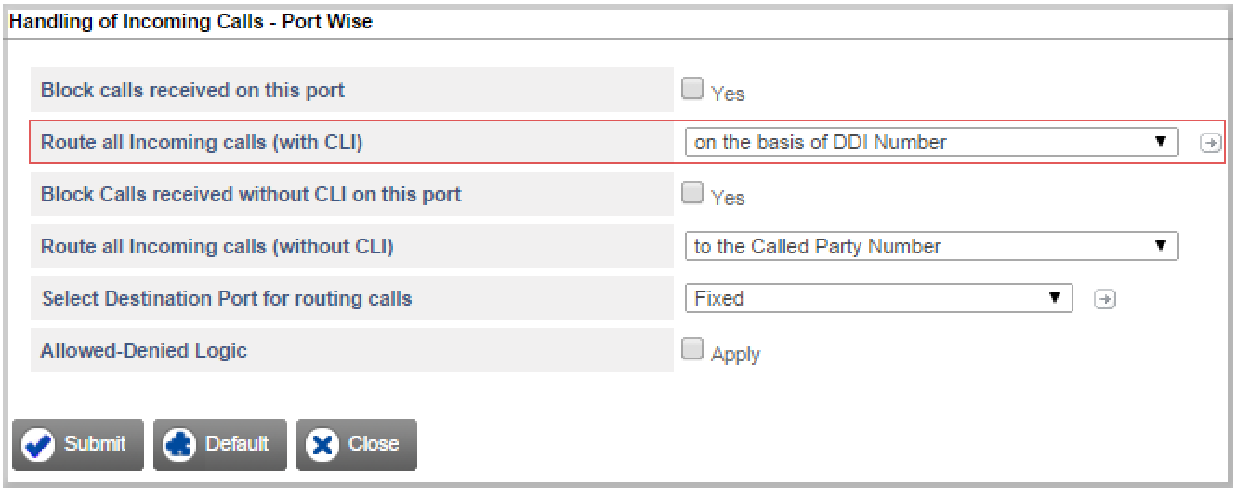
Default: to the Called Party Number.

Route on the basis of DDI Number

In this method, incoming calls on the T1E1 Port are routed to specific numbers as per the DDI number received in the SETUP message on the T1E1 Port.

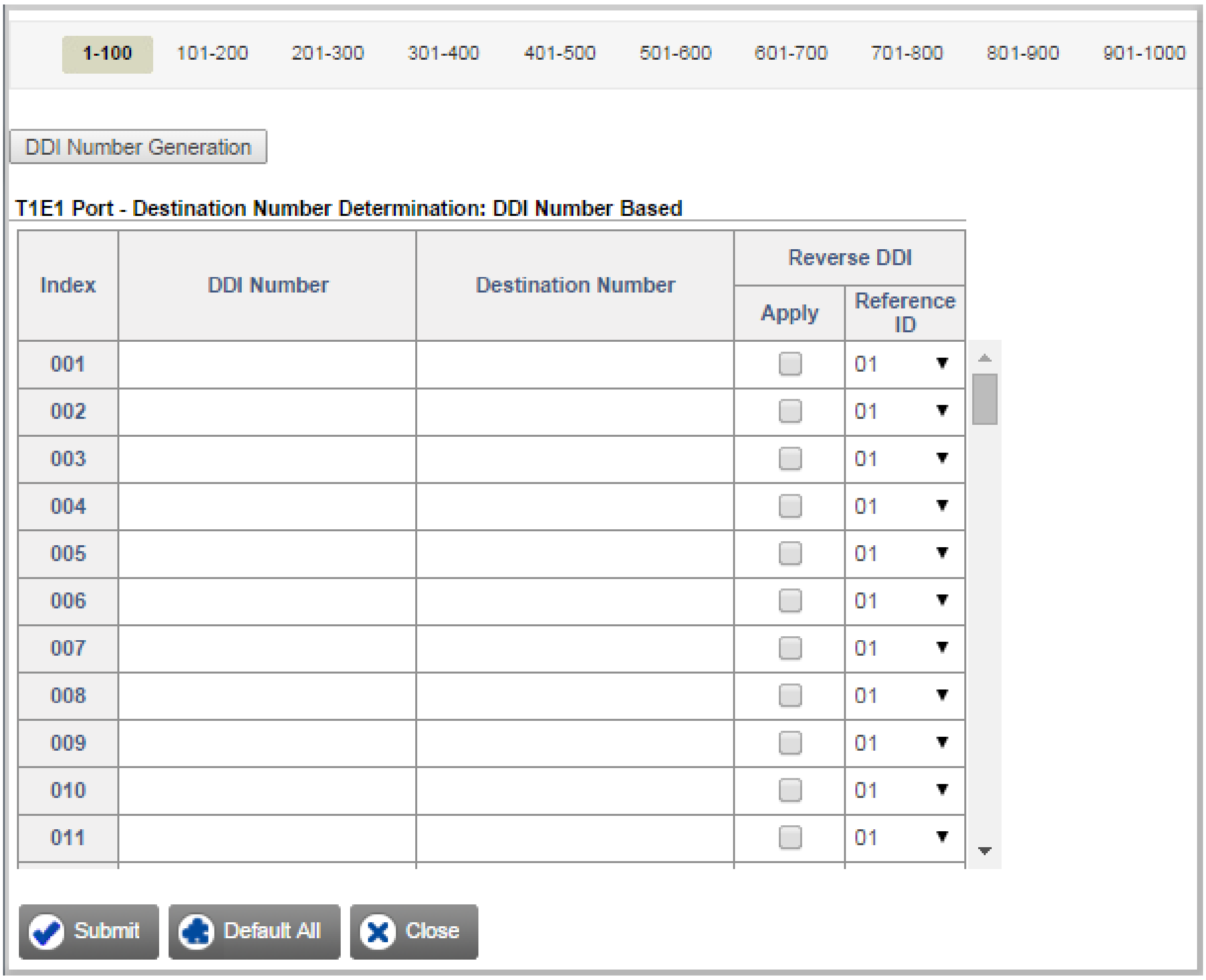
To apply this method, do the following:

- In **Route all Incoming calls (with CLI)**, select **on the basis of DDI Number**.



- Click **Settings** .

The **T1E1 Port - Destination Number Determination: DDI Number Based** Table opens.



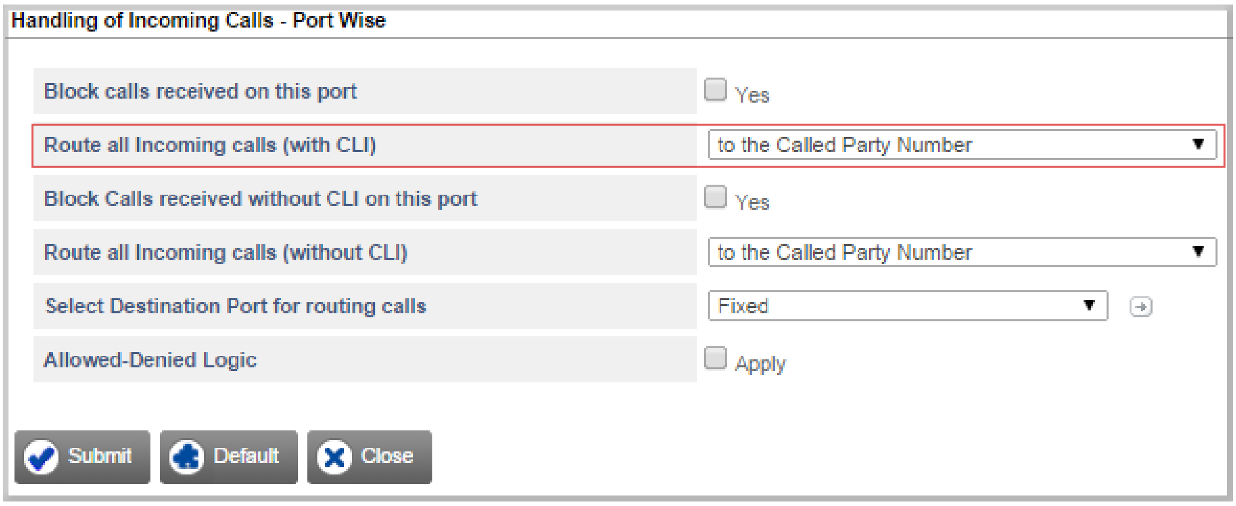
- In **DDI Number**, enter the DDI Numbers allotted by your service provider.
- For each DDI Number, enter the corresponding destination number in **Destination Number**.
- To apply **Reverse DDI** for each number, select the check boxes under **Apply** and select the **Reference ID** for the number. Default: Apply Reverse DDI is disabled and Reference ID is 1.
- Click **Submit** to save and close the window to return to the **Handling of Incoming Calls - Port Wise** window.

  You can also configure the **DDI Number Based** Table from *Advanced Settings.* For instructions, see “Destination Number Determination” under *Advanced Settings*.

Route to the Called Party Number

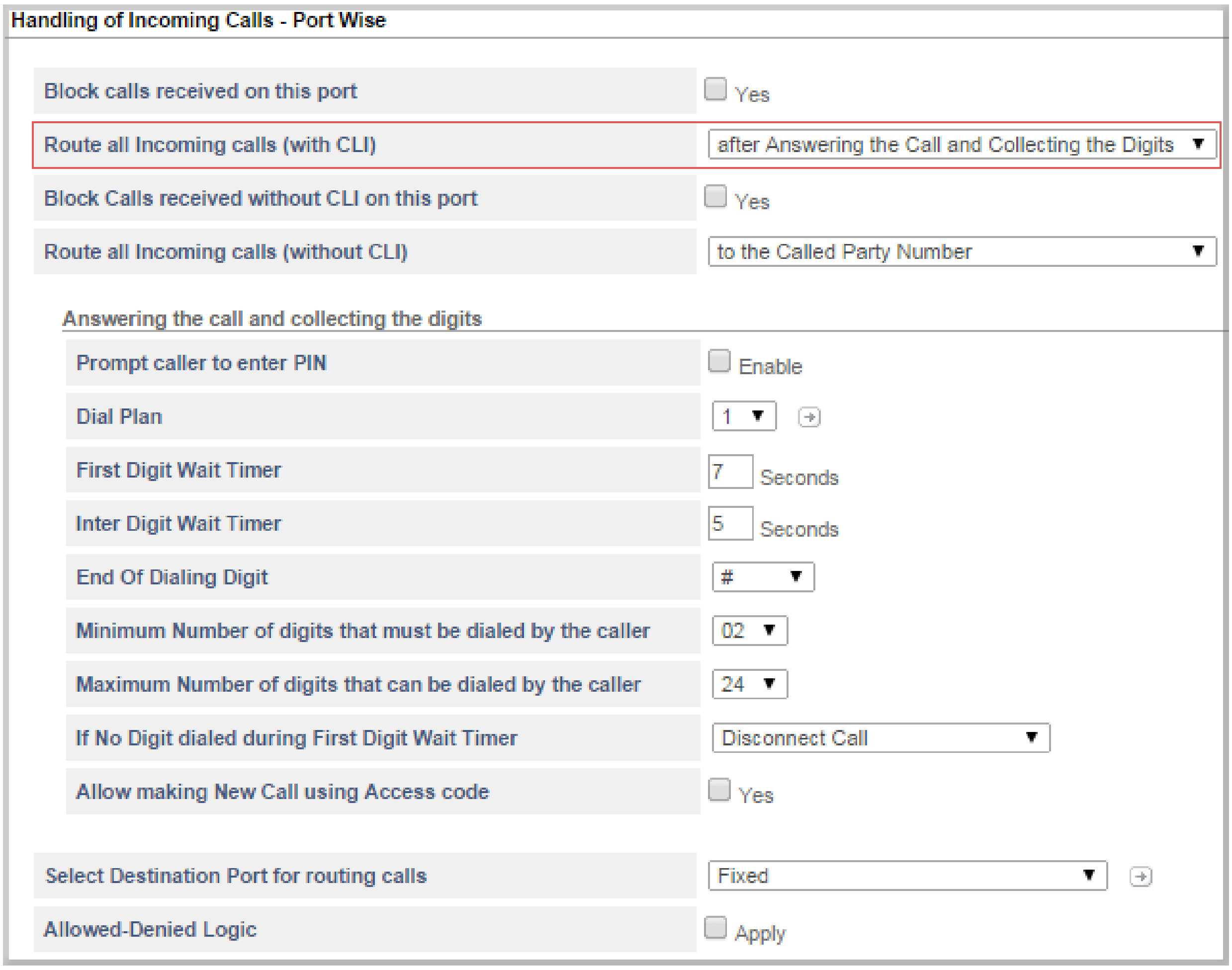
In this method, a call received on the T1E1 Port is routed to a specific number depending upon the called party number received in the SETUP Message on the T1E1 Port.

- To apply this method, in **Route all incoming calls (with CLI)**, select **to the Called Party Number**.



Route after Answering the Call and Collecting the Digits

In this method, the incoming call is answered and dial tone is played to the caller, allowing the caller to dial the desired number. The number dialed by the caller is considered as the destination number.



To apply this method, do the following:

- In **Route all Incoming calls (with CLI)**, select **after Answering the Call and Collecting the Digits**.

  The related parameters of this method appear under **Answering the call and collecting the digits**.

  - If you want to enable PIN Authentication on the T1E1 Port, select the **Prompt caller to enter PIN** check box.

    If you enable this check box, you must also configure the PIN Authentication Table. To know more about this feature and for detail instructions, see "PIN Authentication" under *Advanced Settings*.

  - SARVAM UMG supports 8 Dial Plans with total 64 entries in each table. When a user dials a number, it is compared with the Destination Number configured in the Dial Plan. If a match is found, the system routes the call immediately without waiting for End of Dialing and if a match is not found, the system will wait for the End of Dialing and then route the call as per the Destination Port Selection method configured.

    Select the **Dial Plan** table number you configured for this port. If you have not configured the Dial Plan table you may do so now,

    - Click **Settings** the Dial Plan Table opens.

    - Configure the numbers in the table. For detailed instructions, see "Dial Plan".

  - Set the duration of the **First Digit Wait Timer**. This is the duration for which you want the system to wait for the caller to dial the destination number after the dial tone. Valid range is 01 to 99 seconds. Default: 7 seconds

  - You may configure the following options as End of Dialing indication:

    - Set the duration of the **Inter Digit Wait Timer**. This is the duration for which you want the system to wait while receiving the digits dialed by the caller to consider it as End of Dialing. You may change this timer, if required. Valid range is 01 to 99 seconds. Default: 05 seconds.

    - In **End of Dialing Digit,** select # or * as termination digit the system should consider to detect end of dialing. Default: #

    - In **Minimum number of digits that can be dialed by the caller**, select the minimum number of digits to be dialed by the user for the system to consider it as a valid number**.** Valid range is 01 to 24 digits. Default: 2 digits.

    - In **Maximum Number of digits that can be dialled by the caller**, select the maximum number of digits to be dialed by the user for the system to consider it as End of Dialing**.** Valid range is 01 to 24 digits. Default: 24 digits.

      When the caller dials a number, the system will match it with the above End of Dialing indications and accept the one that matches first.

  - If the caller fails to dial the number during the First Digit Wait Timer, you can either have the system disconnect the call or route the call to a fixed destination number.

In the **If No Digit dialed during First Digit Wait Timer** box, select the desired option: **Disconnect the Call** or **Use Fixed Destination Number**. Default: Disconnect Call.

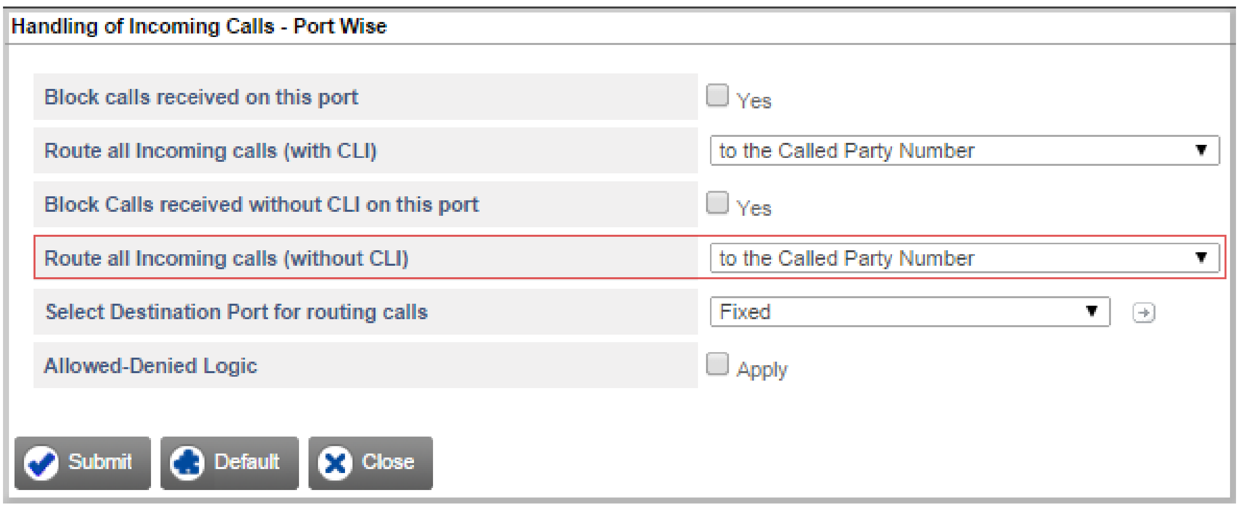
- If you selected **Use Fixed Destination Number**, enter the desired destination number in the **Fixed Destination Number** field. The Destination number may consist of a maximum of 24 digits. Valid digits are 0 to 9, *, # and . (dot/period). Default: Blank.

- *The First Digit Wait Timer is loaded as soon as the system answers the call.*
- *When you dial the first digit, the First Digit Wait Timer is stopped and the system loads the Inter Digit Wait Timer.*
- *SARVAM UMG reloads the Inter Digit Wait Timer:*
  - *each time you dial a new digit till the termination digit is detected.*
  - *when you have dialed the maximum number of digits configured as End of Dialing.*

- If you want to enable the feature Making New Call using Access Code on the T1E1 Port, select the **Allow making New Call using Access Code** check box. For further details, see “Making a New Call using Access Code”.
- Click **Submit** to save settings.

- If you do not want to route the incoming calls received without CLI, through this T1E1 Port, select **Block Calls received without CLI on this Port** check box.
- To **Route all Incoming calls (without CLI),** you may select from any of the following methods:
  - to a Fixed Destination Number, see “Route to a Fixed Destination Number”.
  - on the basis of DDI Number, see “Route on the basis of DDI Number”.
  - to the Called Party Number, see “Route to the Called Party Number”.
  - after Answering the Call and Collecting the Digits, see “Route after Answering the Call and Collecting the Digits”.

  Default: to the Called Party Number.



Destination Port Determination

For the port/channel/MSN number, select the Destination Port for routing calls from the following options:

- Fixed
- On the basis of Destination Number
- On the basis of Calling Party Number

  Default: Fixed.

  Read the description and follow the instructions for each of these destination port selection methods given below.

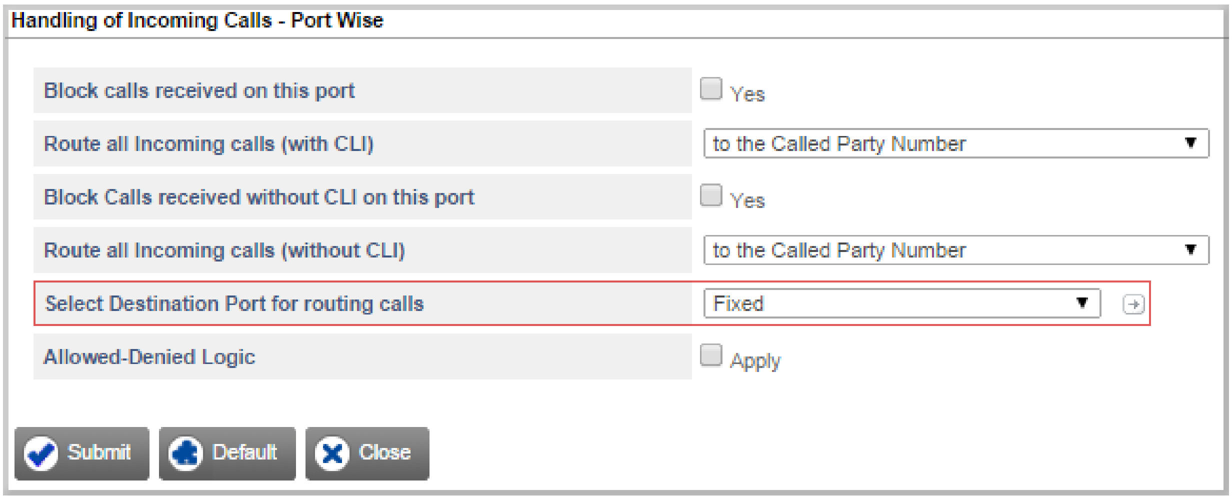
If the destination number to be dialed out is an IP Address, SARVAM UMG will not check the Destination Port Determination Method. Instead, it will route the call using the SIP Trunk / Group programmed for IP Dialing. (To know more, see the feature description "IP Dialing").

Fixed

In this method, calls received on the T1E1 Port are routed to a Fixed Destination Port, irrespective of the number dialed on the T1E1 Port.

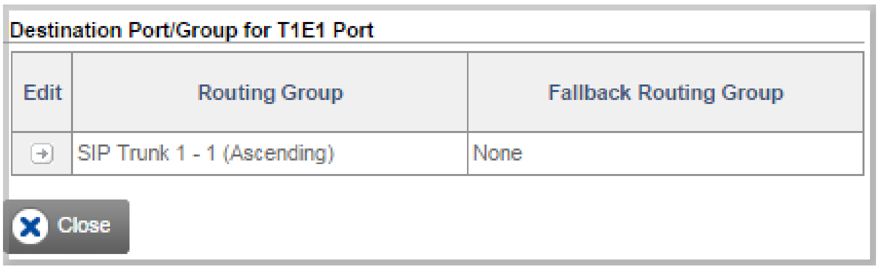
To apply this method, do the following:

- In **Select Destination Port for routing calls**, select **Fixed** option.



- Click **Settings** .

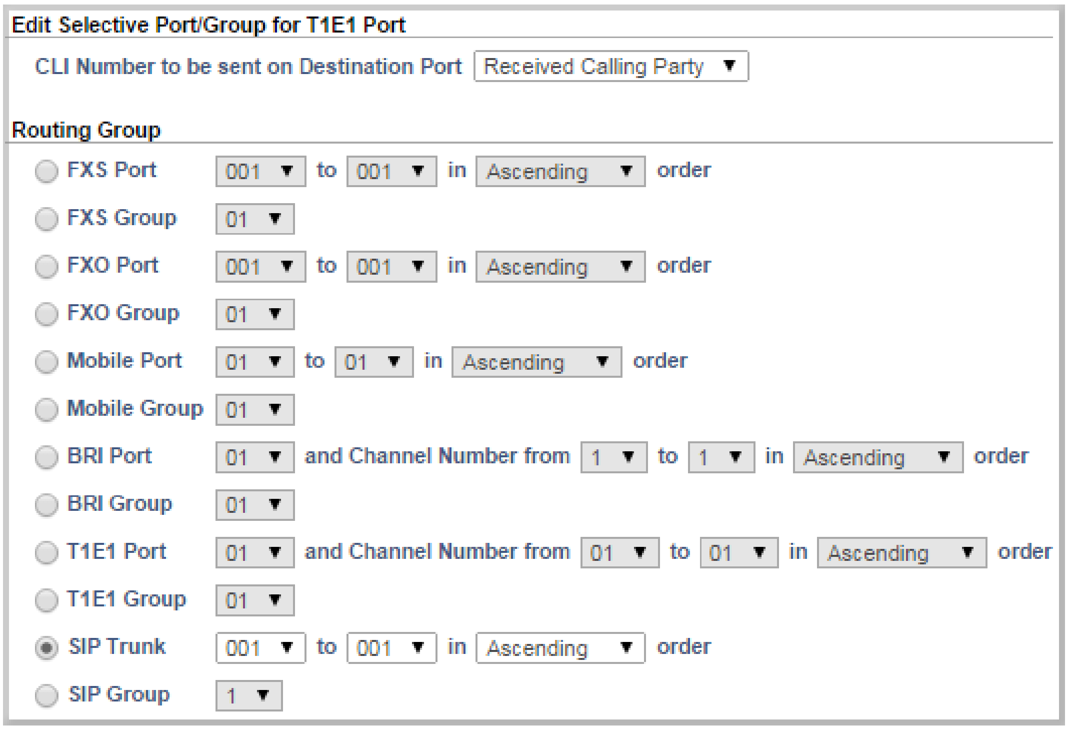
  The **Destination Port/Group for T1E1 Port** window opens.



The default **Routing Group** and **Fallback Routing Groups** appear.

- If you wish to change the default Routing Group options, click **Edit** .

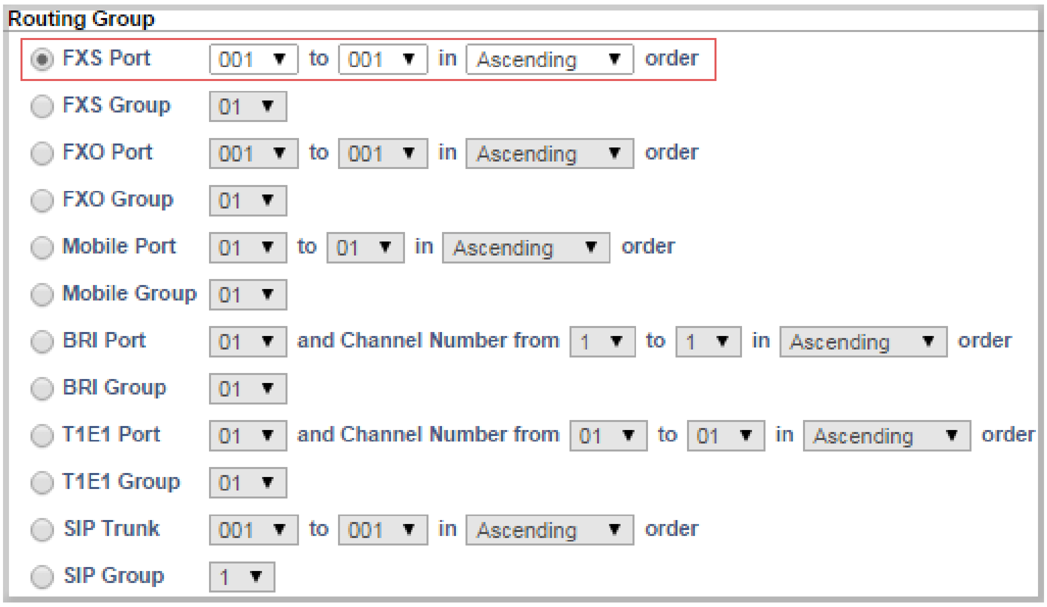
  The **Edit Selective Port/Group for T1E1 Port** window opens.



- Select the **CLI Number to be sent on Destination Port.** You can select Received Calling Party or Received Called Party. Default: Received Calling Party.

CLI to be sent on Destination Port is applicable when FXS Port/Group is determined as the Destination Port.

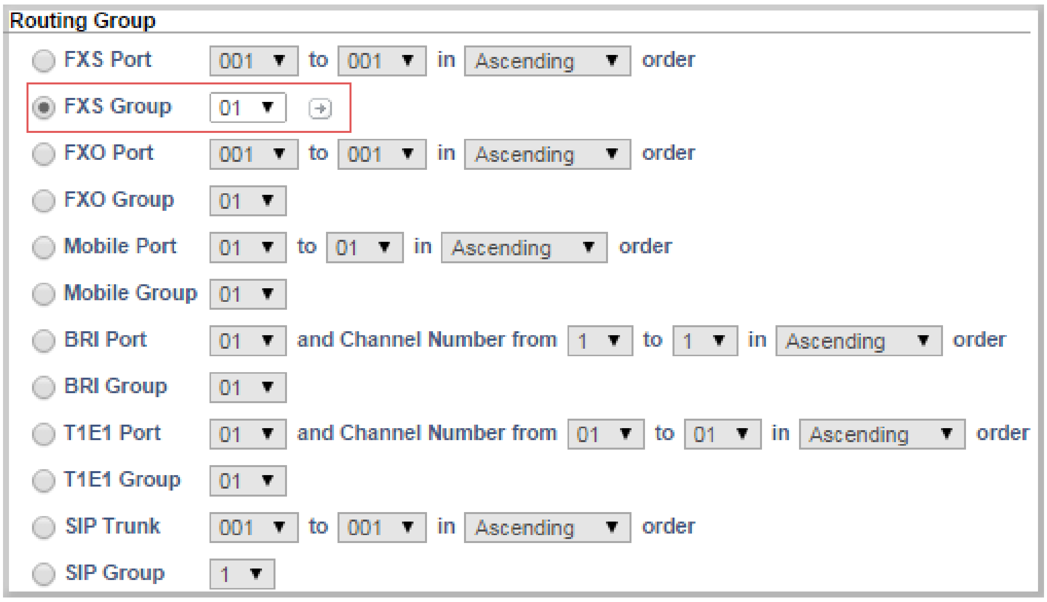
- Create the **Routing Group**.
  - To create a group of *sequential* **FXS Ports** as members,



- Select the desired **FXS Port** numbers as members. Default: 1.
- In **in - order**, select the order in which the system should hunt for a free member FXS Port to route the call.

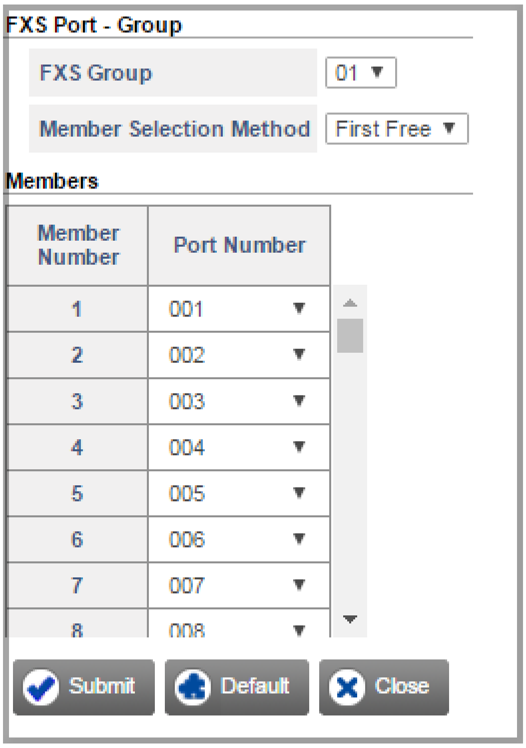
  Select **Ascending** to start hunting from the first to the last member FXS Port. Select **Descending** to start hunting from the last to the first member FXS Port. Default: Ascending.

- To create a group of *not-sequential* **FXS Ports** as members,
  - Select a **FXS Group**.



  - Select **FXS Group** number. Default: 1.
  - Click **Settings** .

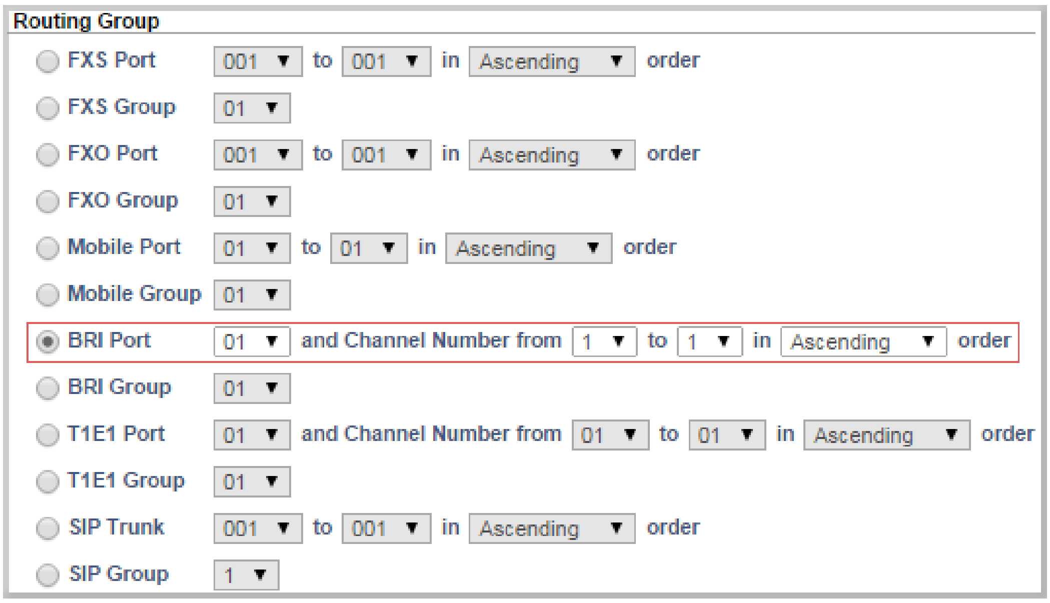
- The **FXS Port - Groups** window opens.



- Create the FXS Group. For detailed instructions on creating groups, see the topic "Group" under *Advanced Settings*.

- Similarly, you can create a group of *sequential* and *not-sequential* FXO Ports, Mobile Ports and SIP Trunks.

- To create a routing group of *sequential* **BRI Channels** as members,

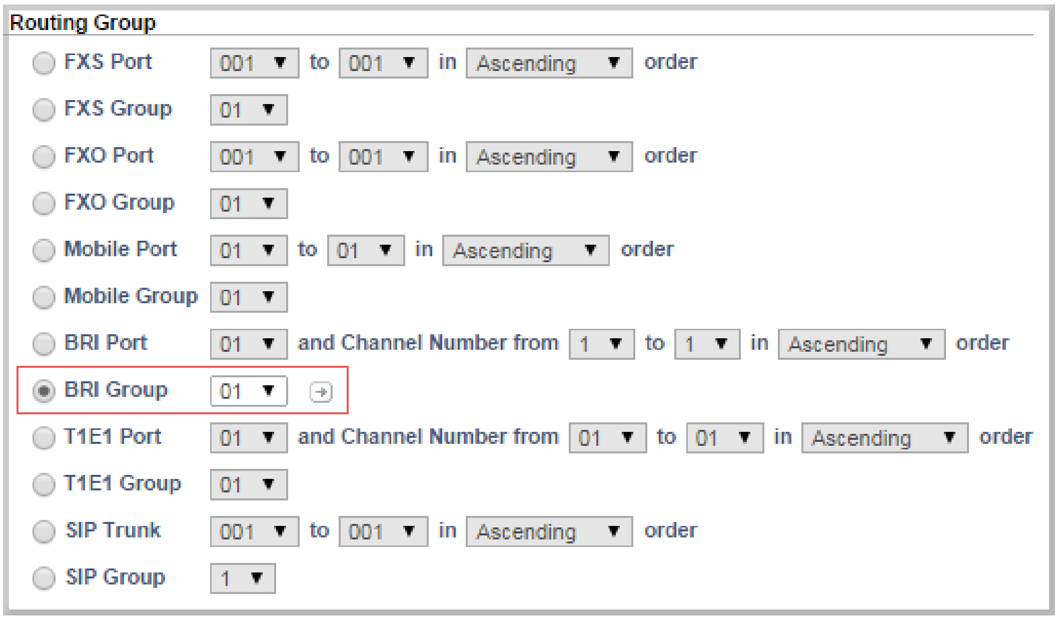


- Select the **BRI Port** Number. Default: 1.

- In Channel Number **From - to**, select the **Start Channel Number** and the **End Channel Number**, respectively.
- In **in - order**, select the order in which the system should hunt for a free member Channel to route the call.

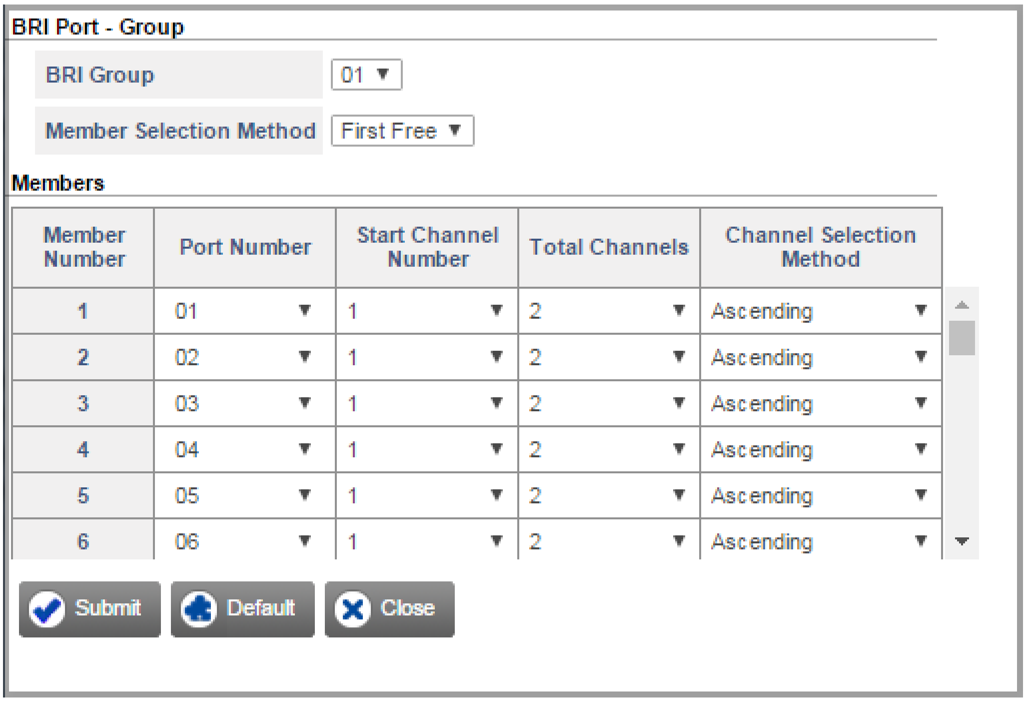
  Select **Ascending** to start hunting from the first to the last member channel. Select **Descending** to start hunting from the last to the first member channel. Default: Ascending.

- To create a group of *not-sequential* **BRI Channels** as members,
  - Select **BRI Group**.



  - Select a **BRI Group** number. Default: 1.
  - Click **Settings** .

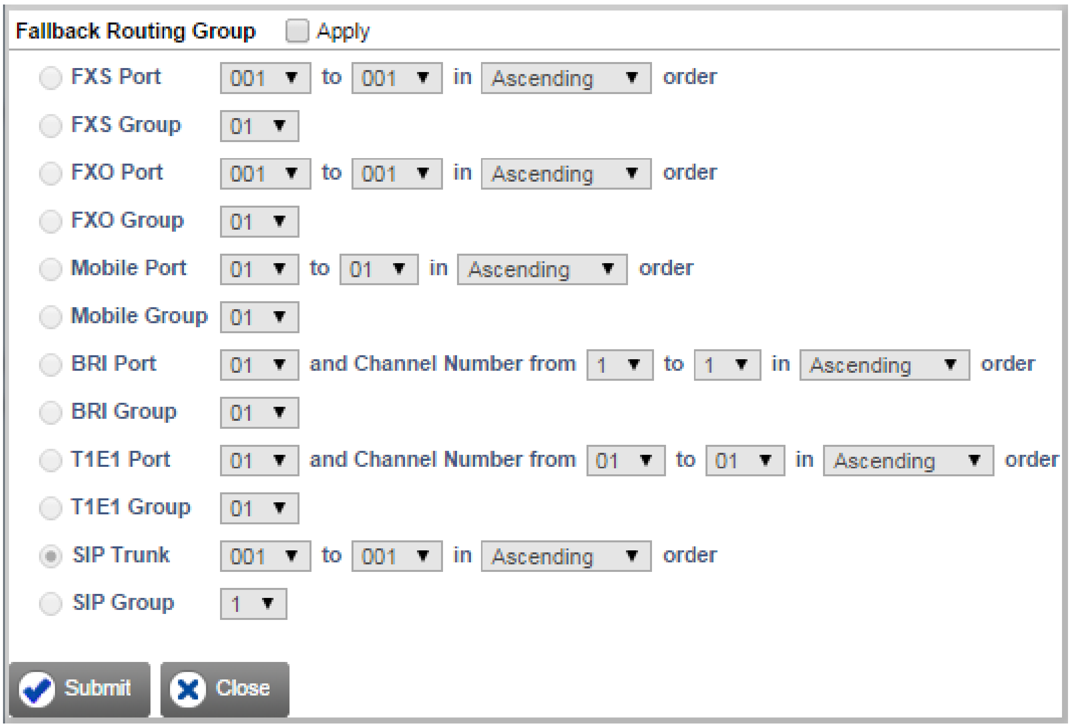
- The **BRI Port - Groups** window opens.



  - Create the BRI Group. For detailed instructions on creating groups, see the topic "Group" under *Advanced Settings.*

 - Similarly, you can create a group of *sequential* and *not-sequential* T1E1 Ports.

- You may create the **Fallback Routing Group**.



- To do this,
  - Select the **Apply** check box.
  - Follow the same instructions provided earlier for creating *sequential* and *not-sequential* groups of FXS Ports, FXO Ports, Mobile Ports, BRI Ports, T1E1 Ports and SIP Trunks.
- Click **Submit** to save changes. The **Edit** window closes.
- The entry you edited appears in the **Destination Port/Group for T1E1 Port** window.
- Close the **Destination Port/Group for T1E1 Port** window to return to the **Handling of Calls** window.

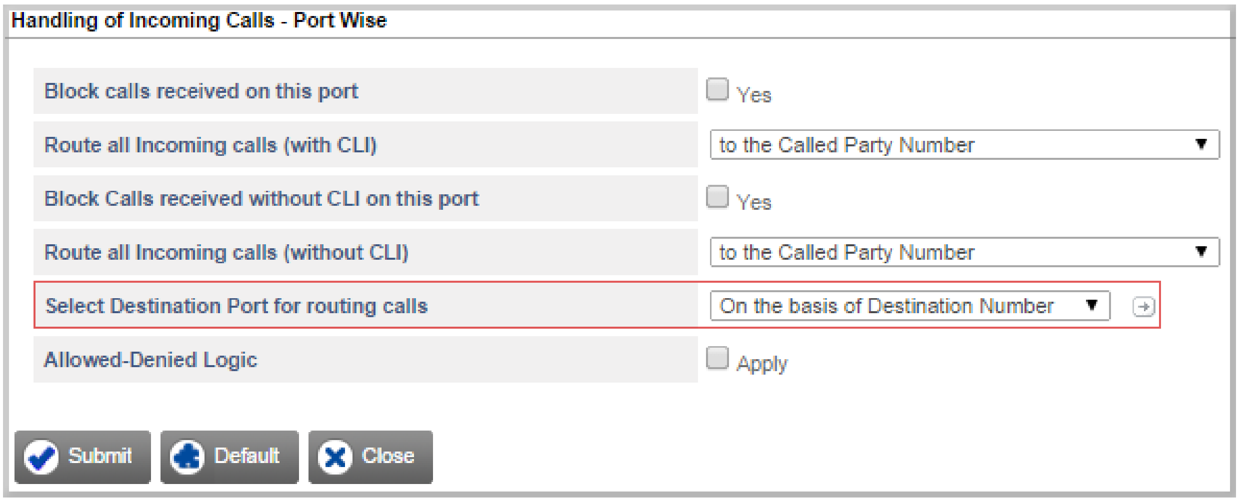
On the basis of Destination Number

In this method, incoming calls on the source port are routed to the destination port on the basis of the destination number (called party number) dialed by the caller.

You must configure the called party numbers in the **Destination Number Based** Table. SARVAM UMG will match the called party number dialed by the caller with the entries of this table. If a match is found for the number in the table, the call is routed to the destination.

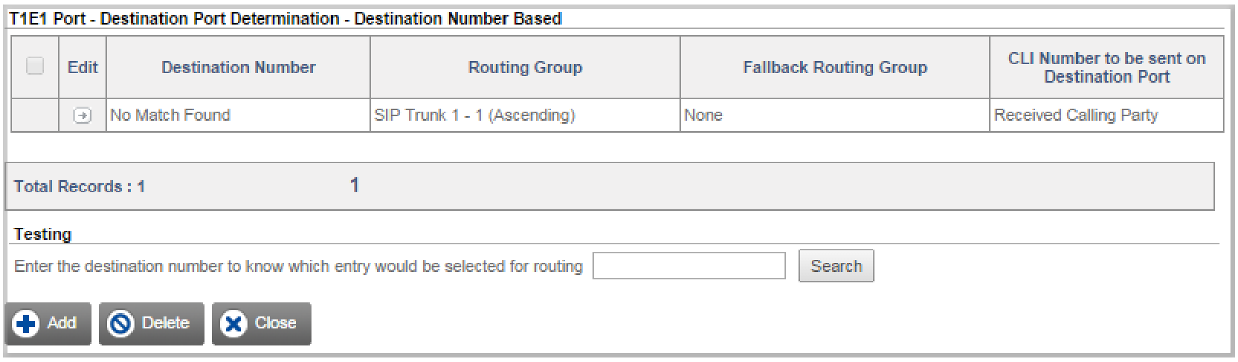
To apply this method, do the following:

- In **Select Destination Port for routing calls**, select **On the basis of Destination Number** option.

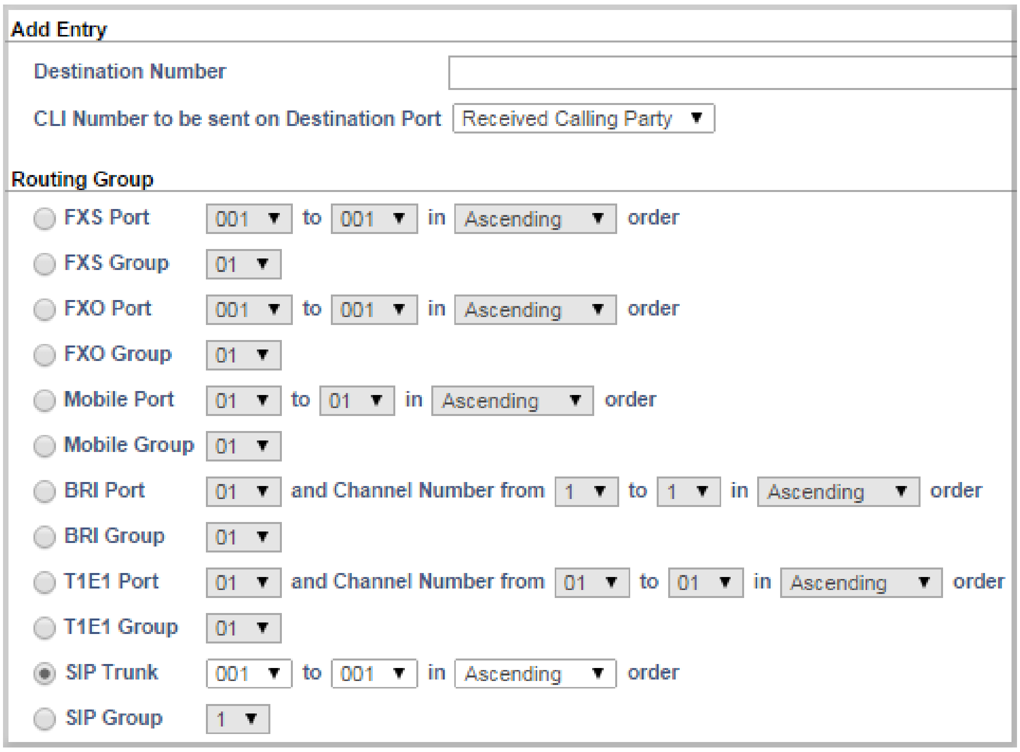


- Click **Settings** .

The **T1E1 Port - Destination Port Determination - Destination Number Based** table window opens.



- To add a new entry, click **Add**. The **Add Entry** window opens. You can add upto 1000 entries.



- In **Destination Number**, enter the number you expect the callers to dial. You may enter upto 64 characters (Digits + "Wildcard Characters") in this field. Valid characters are 0 to 9, *, #, X, T, Comma [,], Hyphen [-], Caret [^]. Default: Blank.

Wildcard Characters

SARVAM UMG supports following characters.

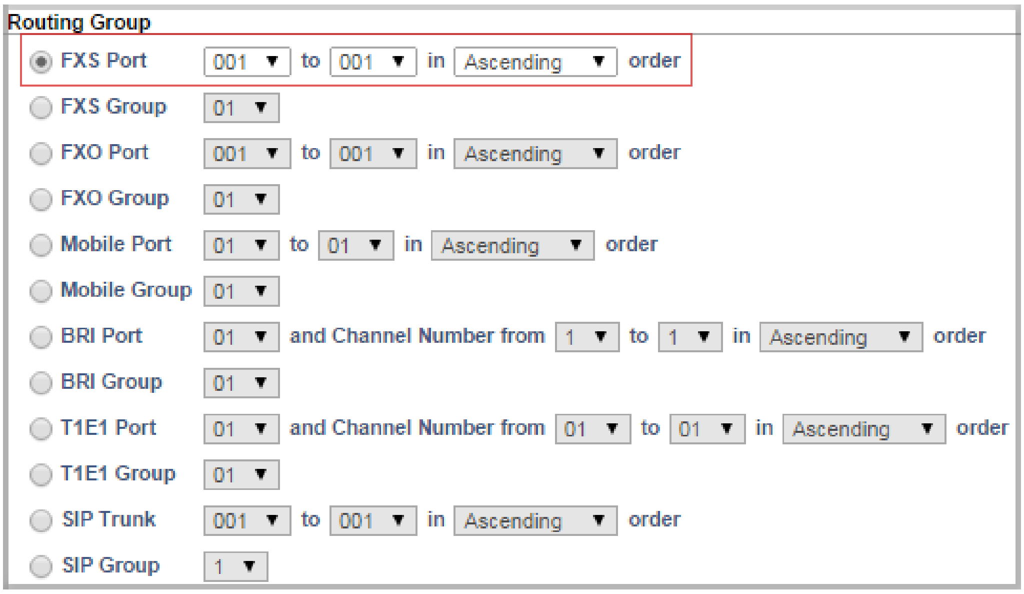
Character	Description
X (letter X)	X represents any single digit from 0 to 9.
#	When # is configured in a number string, it will not be considered as End of Dialing.
*****	When * is configured in a number string, it will not be considered as End of Dialing.

+	+ (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.

- Select the **CLI Number to be sent on Destination Port**. You can select Received Calling Party or Received Called Party. Default: Received Calling Party.

CLI to be sent on Destination Port is applicable when FXS Port/Group is determined as the Destination Port.

- Create the **Routing Group**.
    - To create a group of *sequential* **FXS Ports** as members,

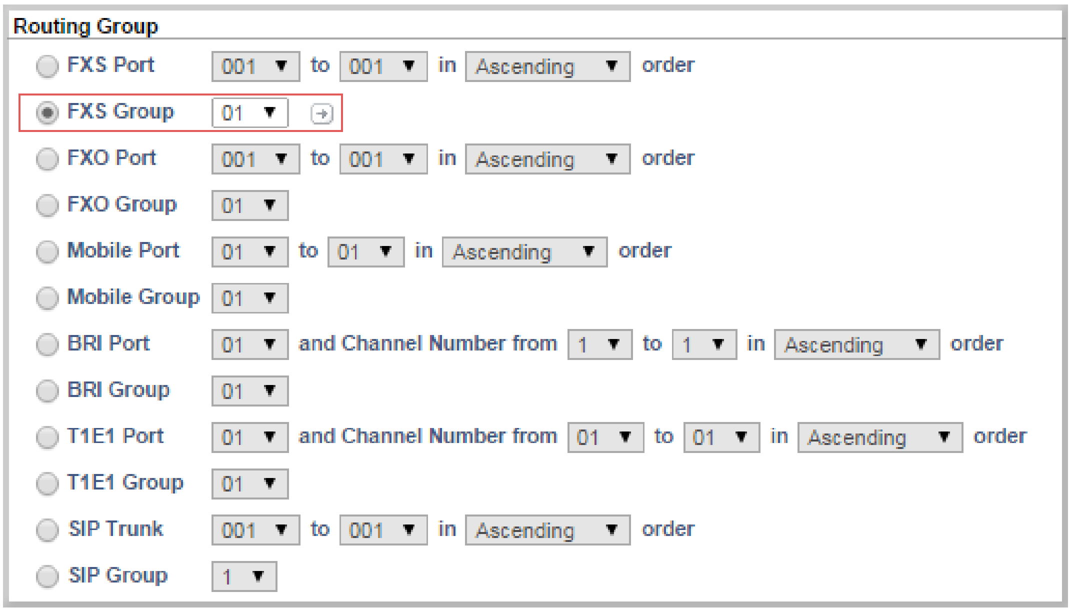


- Select the desired **FXS Port** numbers as members. Default: 1.
- In **in - order**, select the order in which the system should hunt for a free member FXS Port to route the call.

  Select **Ascending** to start hunting from the first to the last member FXS Port. Select **Descending** to start hunting from the last to the first member FXS Port. Default: Ascending.

- To create a group of *not-sequential* **FXS Ports** as members,

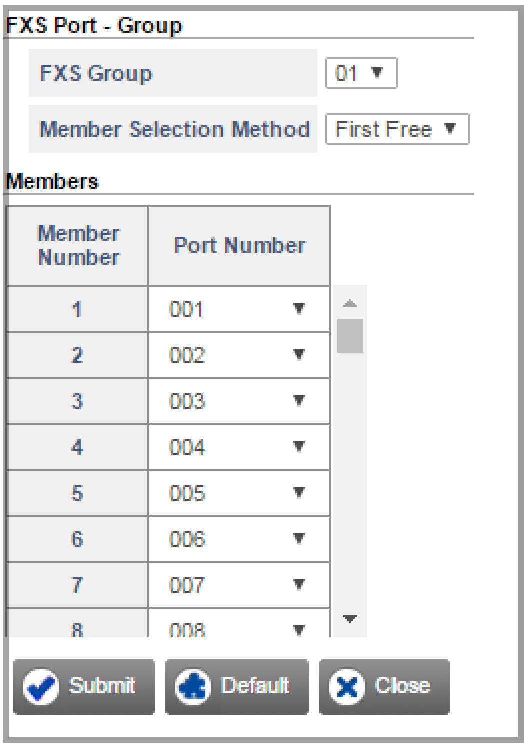
- Select a **FXS Group**.



- Select **FXS Group** number. Default: 1.

- Click **Settings** .

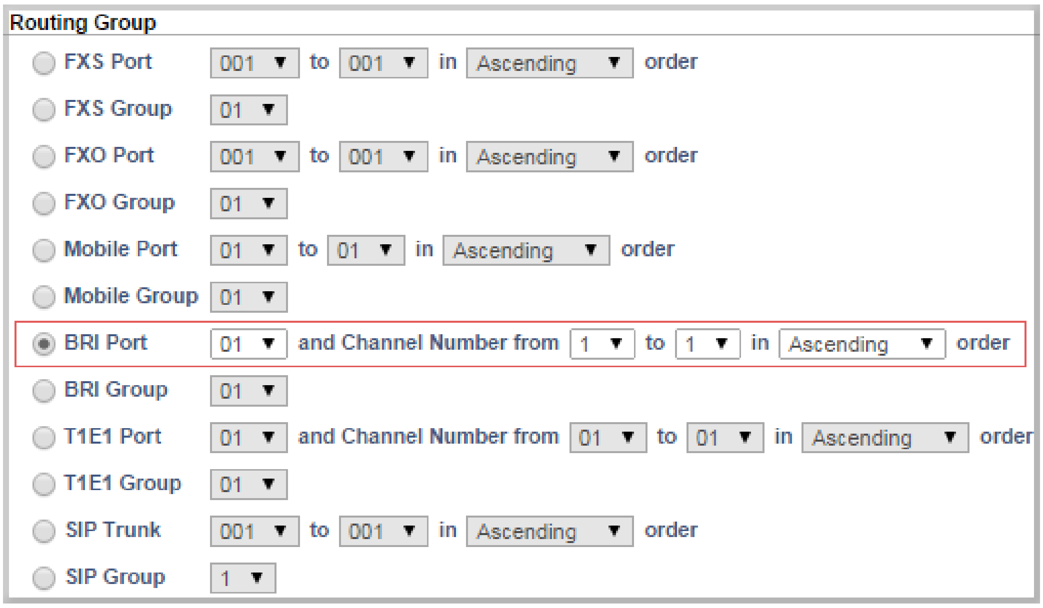
- The **FXS Port - Groups** window opens.



- Create the FXS Group. For detailed instructions on creating groups, see the topic "Group" under *Advanced Settings*.

- Similarly, you can create a group of *sequential* and *not-sequential* FXO Ports, Mobile Ports and SIP Trunks.

- To create a routing group of *sequential* **BRI Channels** as members,



  - Select the **BRI Port** Number. Default: 1.

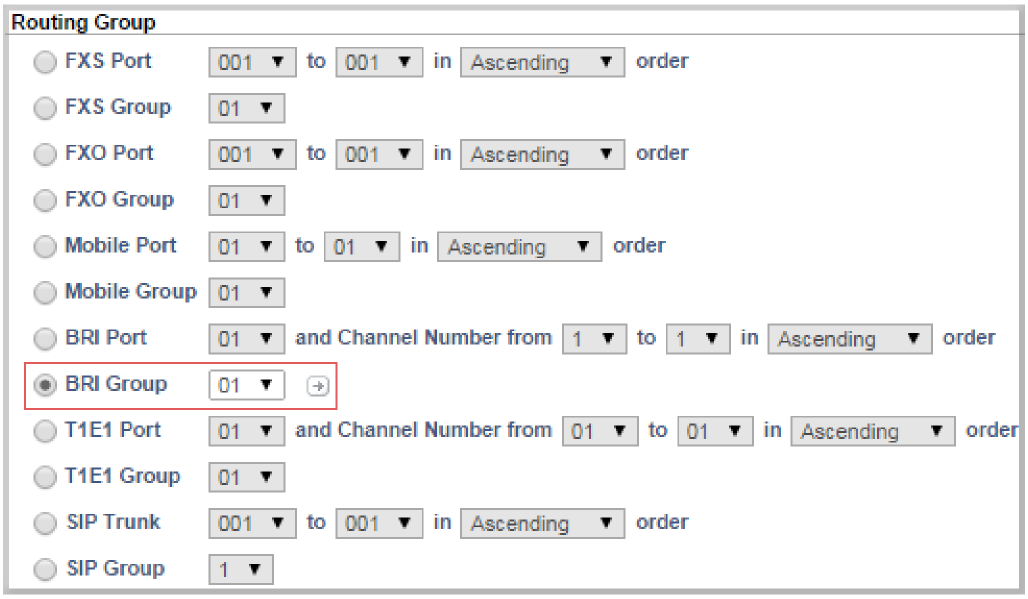
  - In Channel Number **From - to**, select the **Start Channel Number** and the **End Channel Number**, respectively.

  - In **in - order**, select the order in which the system should hunt for a free member Channel to route the call.

    Select **Ascending** to start hunting from the first to the last member channel. Select **Descending** to start hunting from the last to the first member channel. Default: Ascending.

- To create a group of *not-sequential* **BRI Channels** as members,

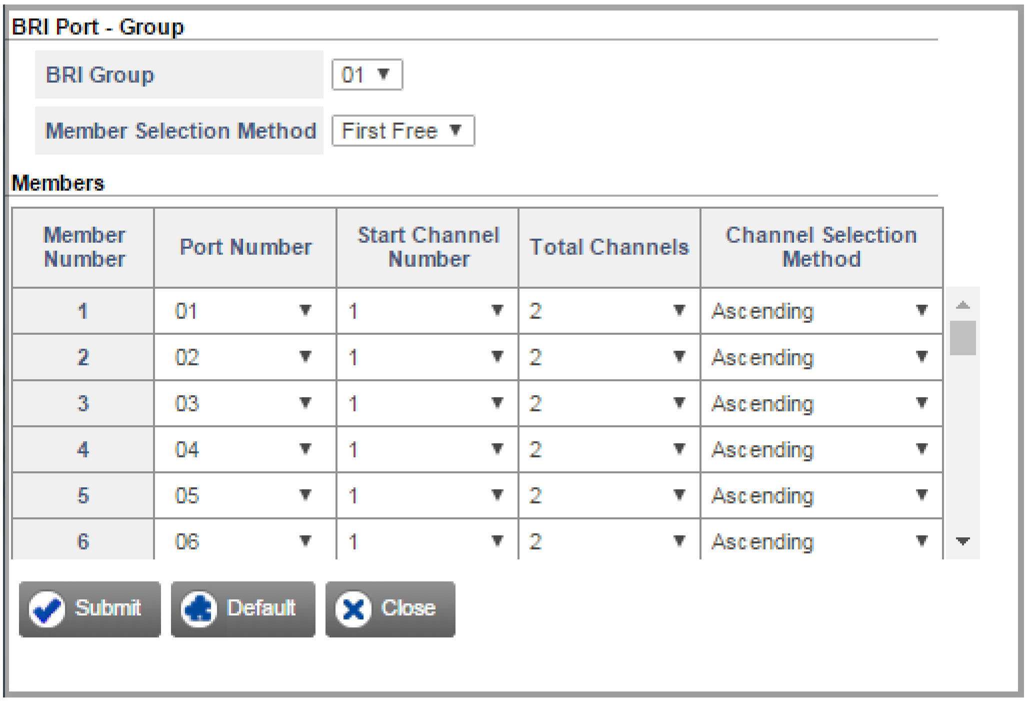
- Select **BRI Group**.



- Select a **BRI Group** number. Default: 1.

- Click **Settings** .

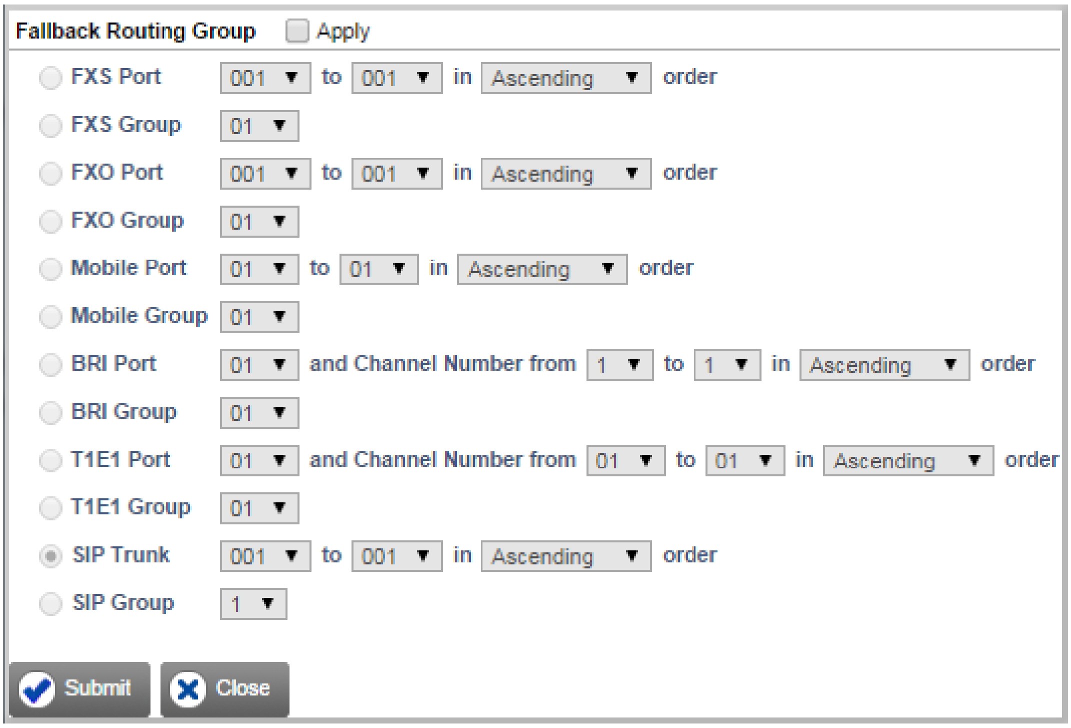
- The **BRI Port - Groups** window opens.



- Create the BRI Group. For detailed instructions on creating groups, see the topic "Group" under *Advanced Settings.*

- Similarly, you can create a group of *sequential* and *not-sequential* T1E1 Ports.

- You may create the **Fallback Routing Group**.



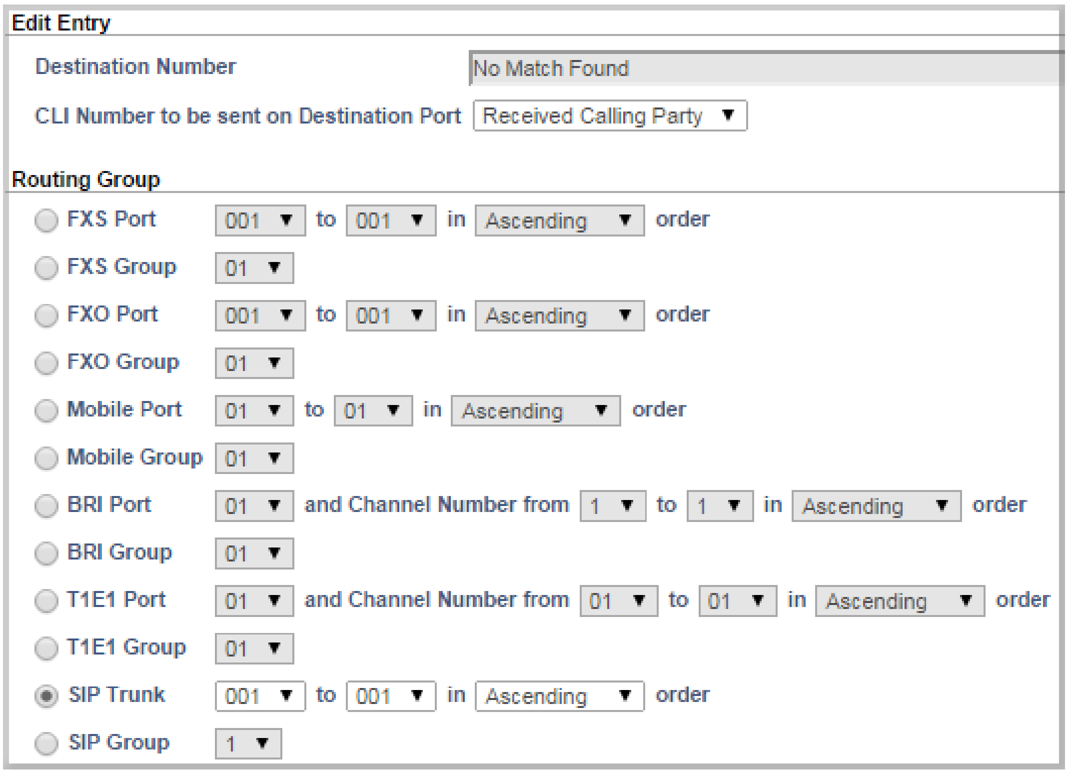
- To do this,
  - Select the **Apply** check box.
  - Follow the same instructions provided earlier for creating *sequential* and *not-sequential* groups of FXS Ports, FXO Ports, Mobile Ports, BRI Ports, T1E1 Ports and SIP Trunks.
- Click **Submit** to save changes. The **Add Entry** window closes.
- The entry you added appears in the **T1E1 Port - Destination Port Determination - Destination Number Based** table.
- Follow the same steps as above to add another entry to this table.
- To delete an entry, select the check box and click the **Delete** button.

If there are multiple entries in the Destination Number Based table, to search a particular entry in the table, under Testing enter the desired number to know which entry would be selected for routing.

- By default, SIP Trunk 1-1 (Ascending) is assigned as the Routing Group, for routing calls from numbers that do not match with any of the destination numbers you configured (No Match Found).

  To change the default Routing Group and to create the Fallback Routing Group for the No Match Found numbers entry,

- For the No Match Found entry in the table, click **Edit** .



- The **Edit Entry** window opens.
- Create the **Routing Group** and **Fallback Routing Group** as per your requirement.
- Click **Submit** and close the window.

- Close the window if you have finished adding/editing entries.

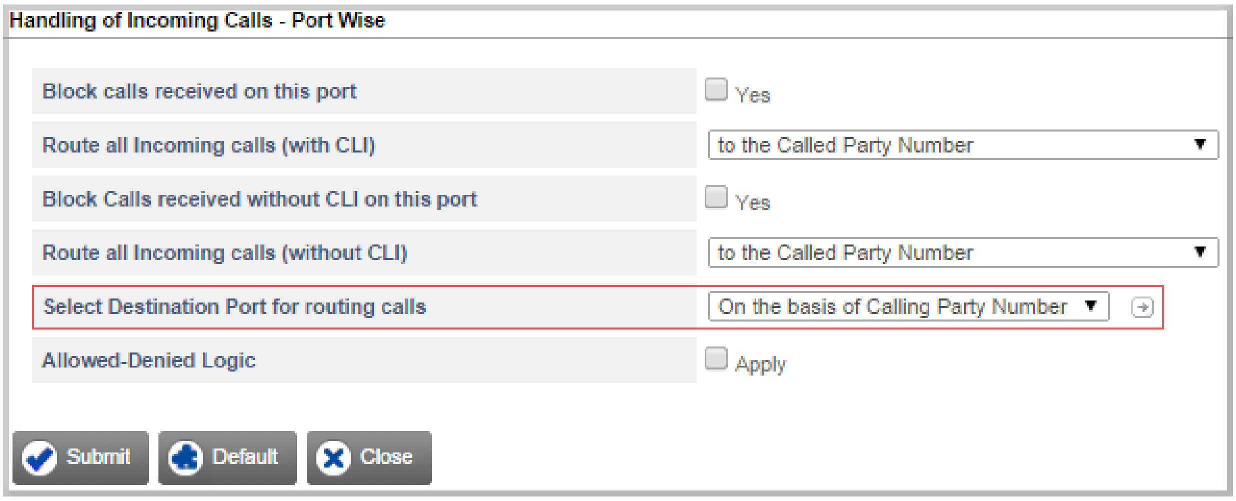
  You can also configure the **Destination Number Based** Table from *Advanced Settings*. For instructions, see "Destination Port Determination" under *Advanced Settings.*

On the basis of Calling Party Number

In this method, incoming calls on the T1E1 Port are routed to a specific port as per the calling party's number.

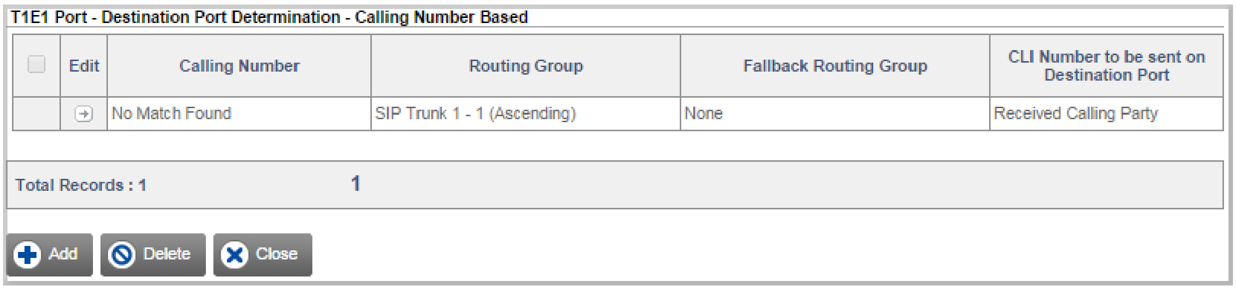
To apply this method, do the following:

- In **Select Destination Port for routing calls**, select **On the basis of Calling Party Number** option.



- Click **Settings** .

The **T1E1 Port - Destination Port Determination - Calling Number Based** table window opens.

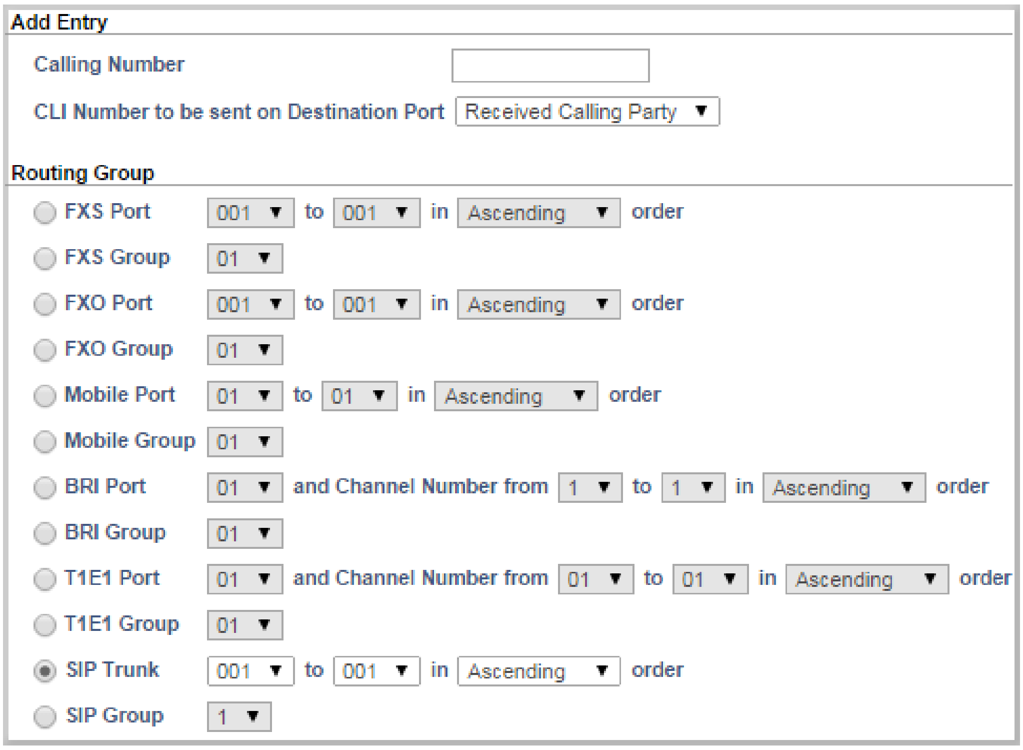

T1E1 Port - Destination Port Determination - Calling Number Based

	Edit	Calling Number	Routing Group	Fallback Routing Group	CLI Number to be sent on Destination Port
		No Match Found	SIP Trunk 1 - 1 (Ascending)	None	Received Calling Party

Total Records : 1 1

Add Delete Close

- To add a new entry, click **Add**. The **Add Entry** window opens. You can add upto 499 entries.



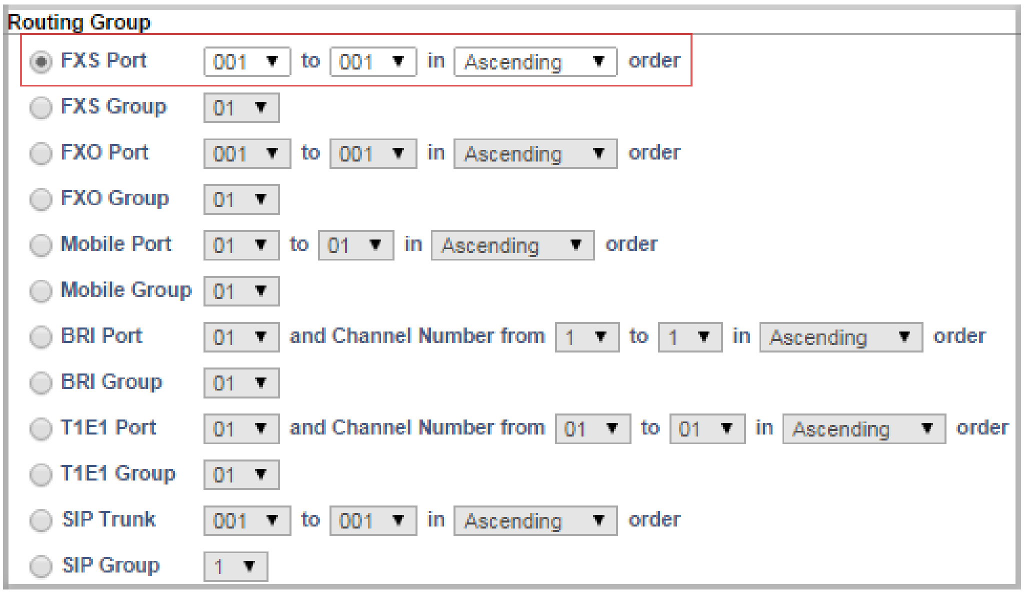
- In **Calling Number**, enter the number (max. 24 characters) from which you expect calls to be received. Valid digits are 0 to 9, *, #, (dot). Default: Blank.

- Select the **CLI Number to be sent on Destination Port**. You can select Received Calling Party or Received Called Party. Default: Received Calling Party.

CLI to be sent on Destination Port is applicable when FXS Port/Group is determined as the Destination Port.

- Create the **Routing Group**.

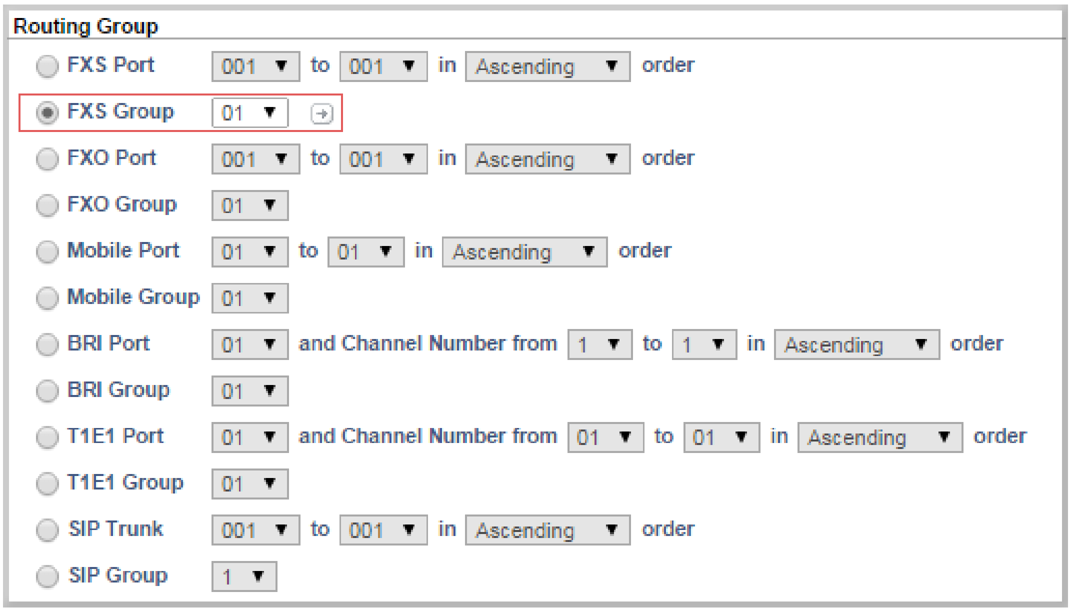
- To create a group of *sequential* **FXS Ports** as members,



  - Select the desired **FXS Port** numbers as members. Default: 1.
  - In **in - order**, select the order in which the system should hunt for a free member FXS Port to route the call.

    Select **Ascending** to start hunting from the first to the last member FXS Port. Select **Descending** to start hunting from the last to the first member FXS Port. Default: Ascending.

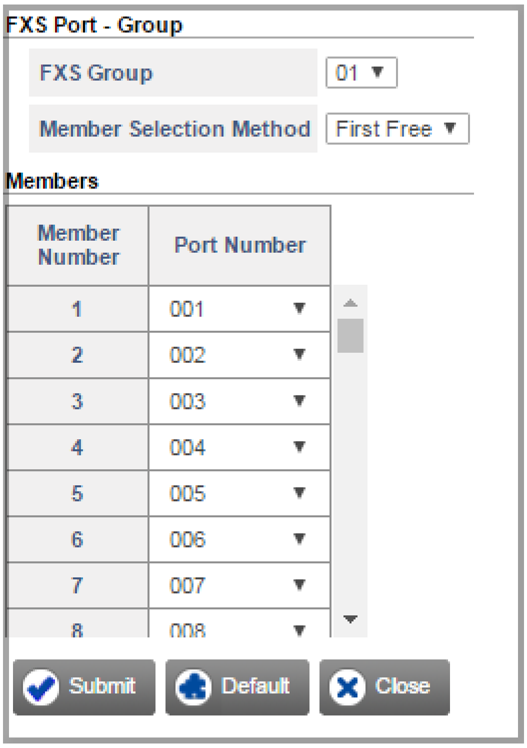
- To create a group of *not-sequential* **FXS Ports** as members,
  - Select a **FXS Group**.



  - Select **FXS Group** number. Default:1.

- Click **Settings** .

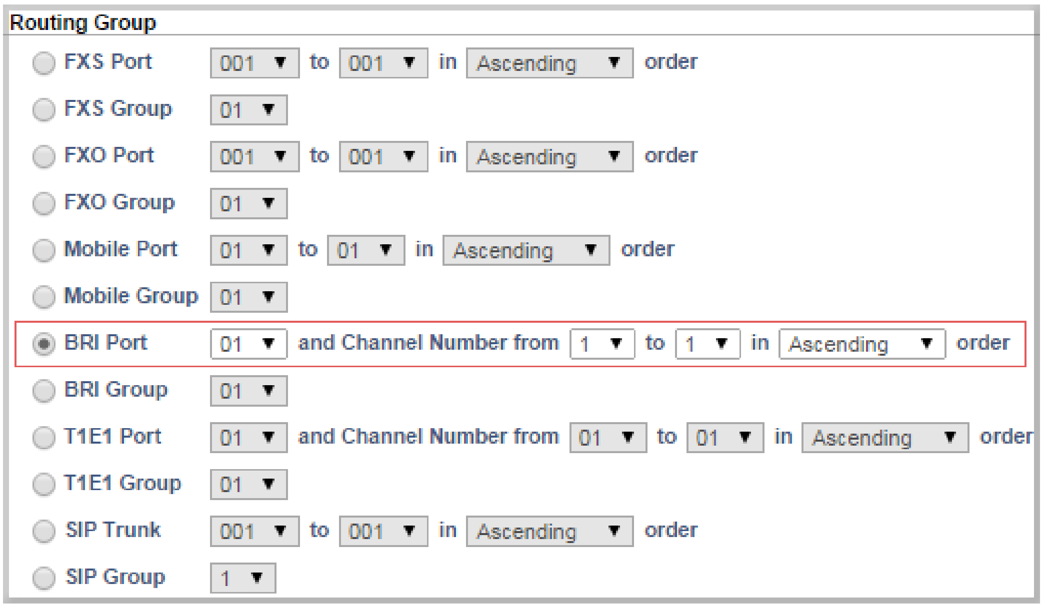
- The **FXS Port - Groups** window opens.



- Create the FXS Group. For detailed instructions on creating groups, see the topic “Group” under *Advanced Settings*.

- Similarly, you can create a group of *sequential* and *not-sequential* FXO Ports, Mobile Ports and SIP Trunks.

- To create a routing group of *sequential* **BRI Channels** as members,

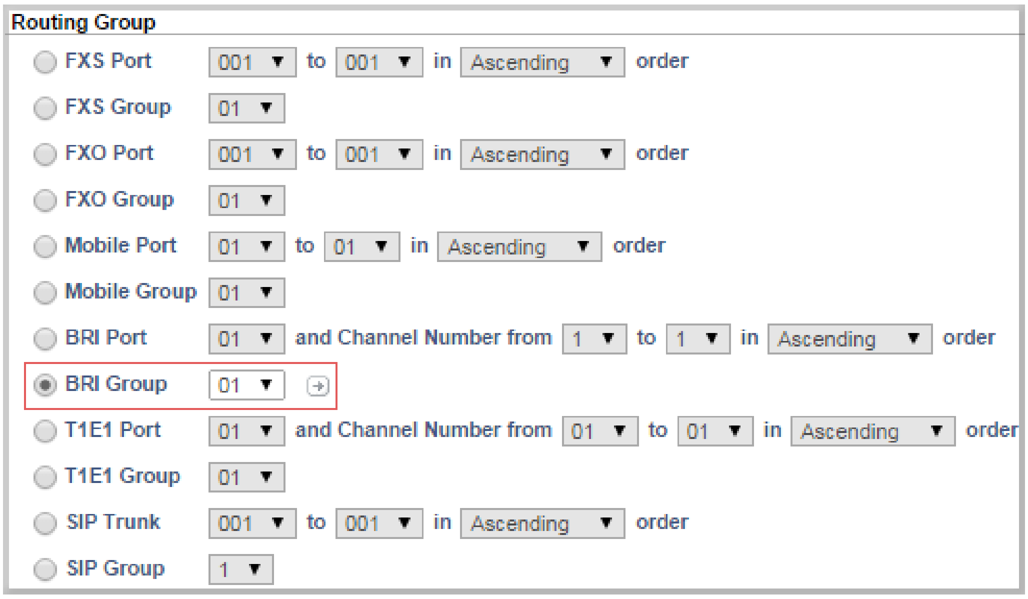


  - Select the **BRI Port** Number. Default: 1.
  - In Channel Number **From - to**, select the **Start Channel Number** and the **End Channel Number**, respectively.
  - In **in - order**, select the order in which the system should hunt for a free member Channel to route the call.

    Select **Ascending** to start hunting from the first to the last member channel. Select **Descending** to start hunting from the last to the first member channel. Default: Ascending.

- To create a group of *not-sequential* **BRI Channels** as members,

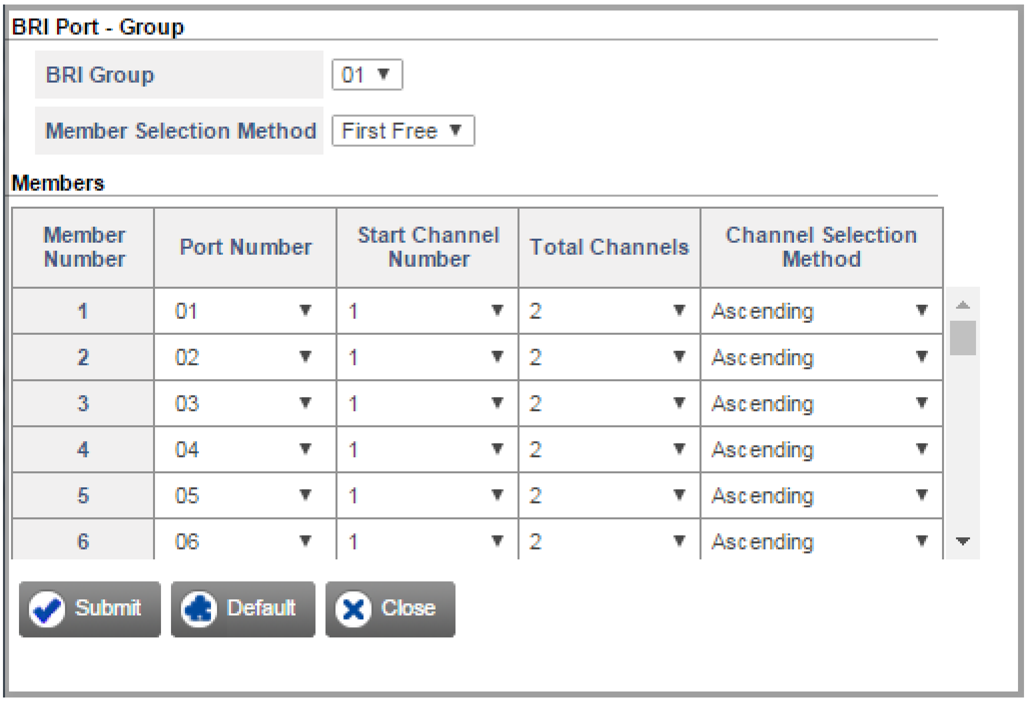
- Select **BRI Group**.



- Select a **BRI Group** number. Default: 1.

- Click **Settings** .

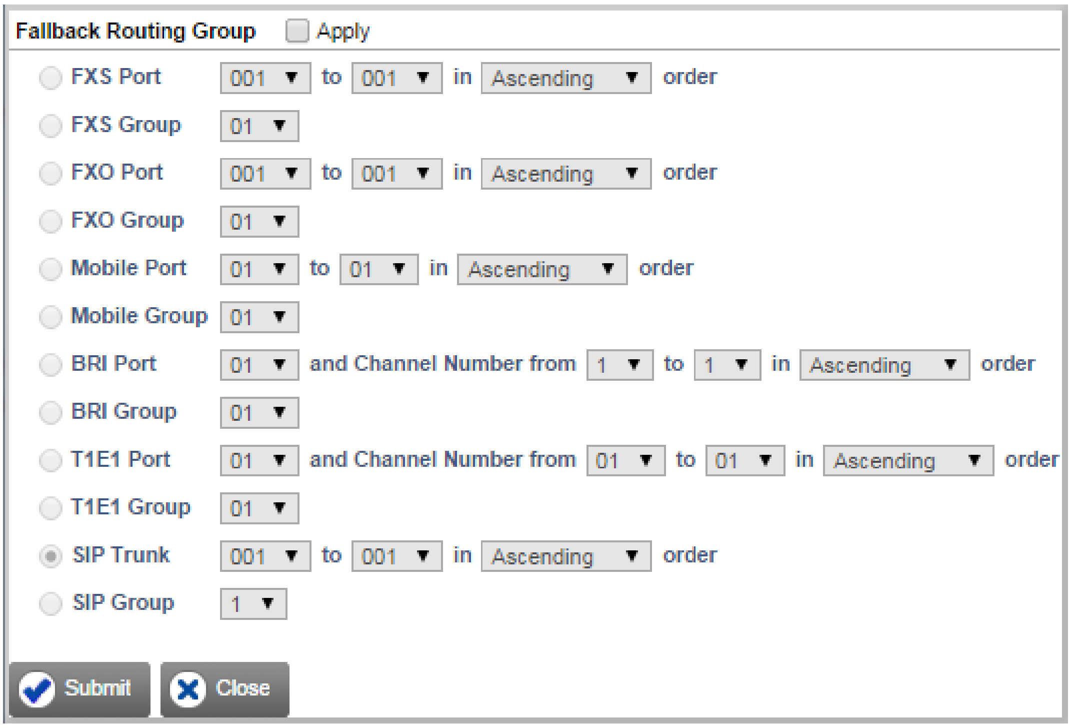
- The **BRI Port - Groups** window opens.



- Create the BRI Group. For detailed instructions on creating groups, see the topic “Group” under *Advanced Settings.*

- Similarly, you can create a group of *sequential* and *not-sequential* T1E1 Ports.

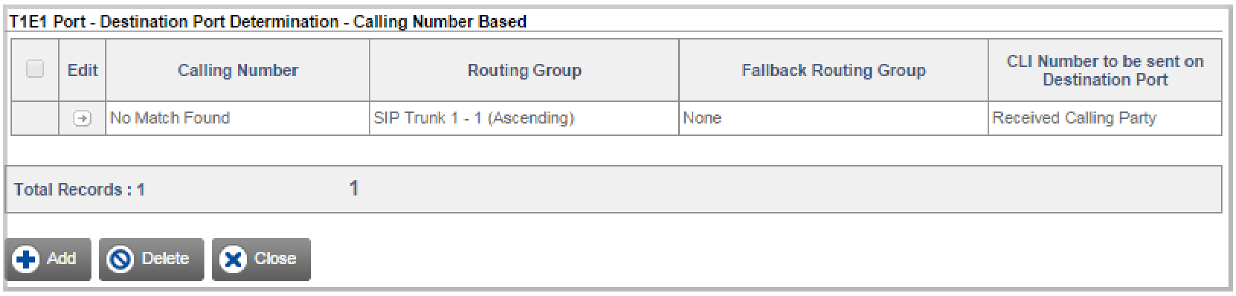
- You may create the **Fallback Routing Group**.



- To do this,
    - Select the **Apply** check box.
    - Follow the same instructions provided earlier for creating *sequential* and *not-sequential* groups of FXS Ports, FXO Ports, Mobile Ports, BRI Ports, T1E1 Ports and SIP Trunks.
- Click **Submit** to save changes. The **Add Entry** window closes.
- The entry you added appears in the **T1E1 Port - Destination Port Determination - Calling Number Based** table.
- Follow the same steps as above to add another entry to this table.
- To delete an entry, select the check box and click the **Delete** button.
- By default, SIP Trunk 1-1 (Ascending) is assigned as the Routing Group, for routing calls from numbers that do not match with any of the destination numbers you configured (No Match Found).

  To change the default Routing Group and to create the Fallback Routing Group for the No Match Found numbers entry,

- For the No Match Found entry in the table, click **Edit** .



T1E1 Port - Destination Port Determination - Calling Number Based

	Edit	Calling Number	Routing Group	Fallback Routing Group	CLI Number to be sent on Destination Port
		No Match Found	SIP Trunk 1 - 1 (Ascending)	None	Received Calling Party

Total Records : 1 1

Add Delete Close

- The **Edit Entry** window opens.

- Create the **Routing Group** and **Fallback Routing Group** as per your requirement.

- Click **Submit** and close the window.

- Close the window if you have finished adding/editing entries.

  You can also configure the **Calling Number Based** Table from *Advanced Settings*. For instructions, see "Destination Port Determination" under *Advanced Settings.*

Allowed - Denied Logic

You can apply the Allowed-Denied logic on the T1E1 Port (source port) if you want to allow or restrict the dialing of particular numbers. You can use this feature for Toll Control.

The Allowed-Denied Number Logic makes use of two Number lists:

- **Allowed Numbers List:** This is the list of numbers that can be dialed out from the T1E1 Port.

- **Denied Numbers List:** This list contains the numbers that are to be restricted from being dialed out from the T1E1 Port.

When Allowed-Denied Logic is enabled on a source port, for each number dialed from the port, SARVAM UMG uses the best-match-found logic to compare the dialed number with the Allowed Number list and the Denied Number list.

The number is allowed to be dialed, if it:
- matches with both lists.
- matches with Allowed Number list, but not with the Denied Number list.
- matches with neither the Allowed List nor the Denied List.

The number is denied, if it matches with the Denied Number list, but not with the Allowed Number list.

The system does not apply the Allowed-Denied Logic:
- When dialed number string matches with any Access Code.
- When dialed number string matches with any Emergency Number.
- When any one of the following is selected to Route all Incoming Calls (with CLI):
  - on the basis of Calling Party Number
  - to a Fixed Destination Number
  - on the basis of DDI Number

To apply Allowed - Denied Logic on the T1E1 Port,

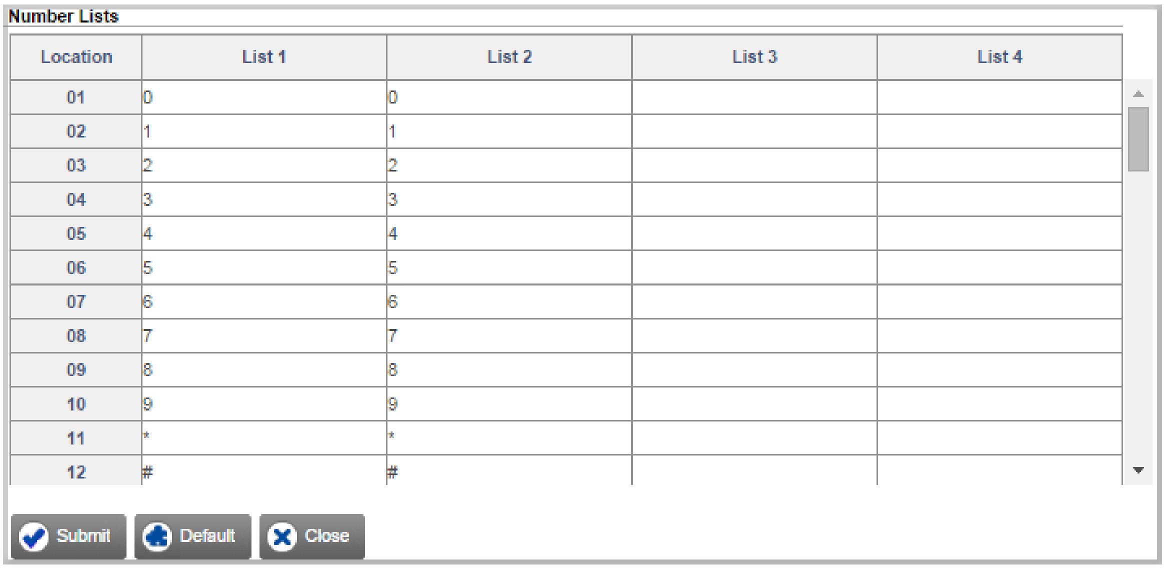
- Select the **Allowed - Denied Logic** check box.

- In the **Allowed Number List**, select the list number you have configured with numbers you want to allow to be dialed out from the T1E1 Port. Default: 01

  If you have not configured the Allowed Number List,

  - Click **Settings** .

  - The **Number Lists** window opens.



Number Lists

Location	List 1	List 2	List 3	List 4
01	0	0		
02	1	1		
03	2	2		
04	3	3		
05	4	4		
06	5	5		
07	6	6		
08	7	7		
09	8	8		
10	9	9		
11	*	*		
12	#	#		

  - You may configure the default Allowed Number List 1 or any other list. See “Number Lists” to configure the allowed numbers.

  - Click **Submit** to save the Allowed Number List and close the window.

- In the **Denied Number List**, select the list number you have configured with numbers you want to restrict to be dialed out from the T1E1 Port. Default: 02

  If you have not configured the Denied Number List,

  - Click **Settings** . **The Number Lists** window opens.

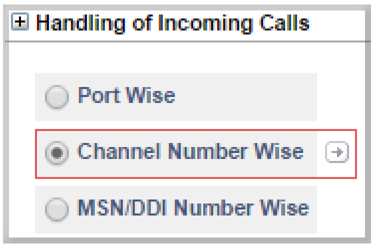
  - You may configure the default Denied Number List 2 or any other list. See “Number Lists” to configure the restrict numbers.

- Click **Submit** to save the Denied Number List and close the window.

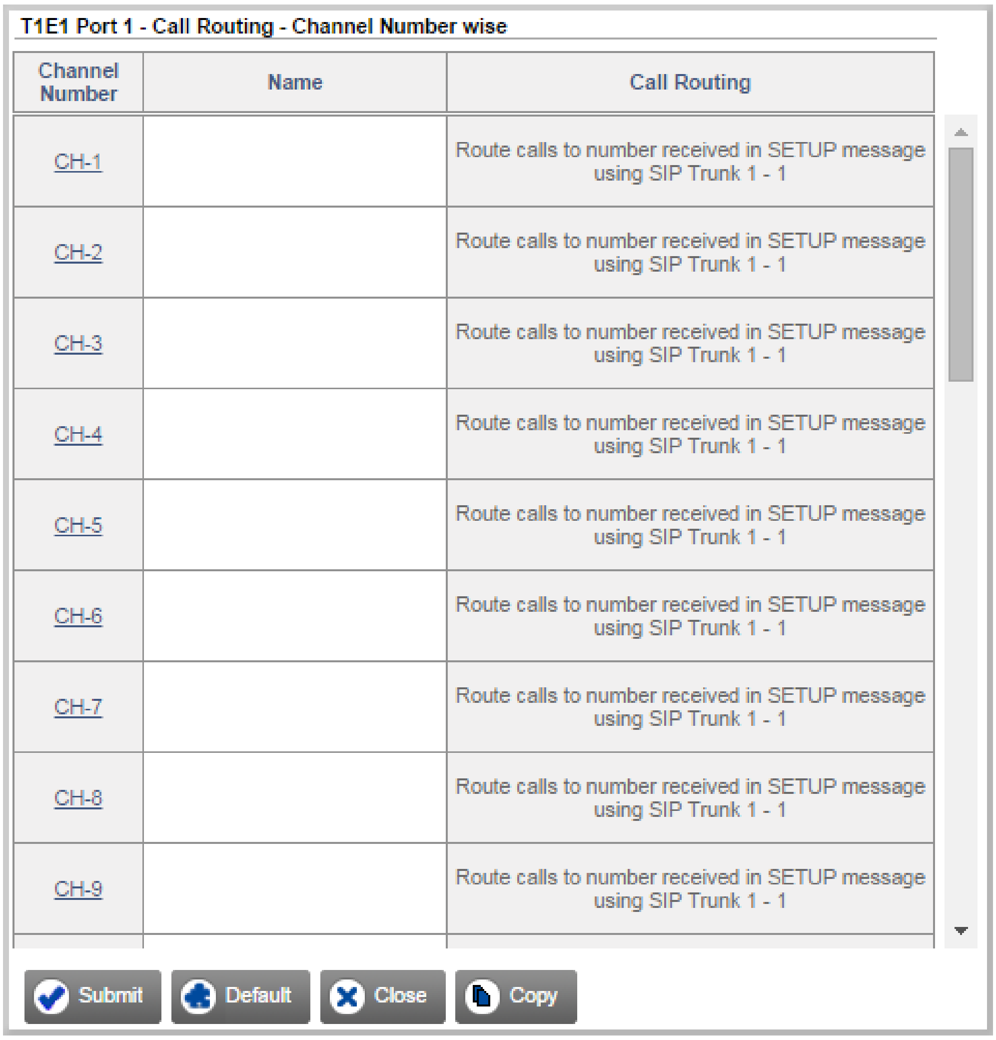
Channel Number Wise

To configure Handling of Incoming Calls for each channel,

- Select the **Channel Number Wise** check box.

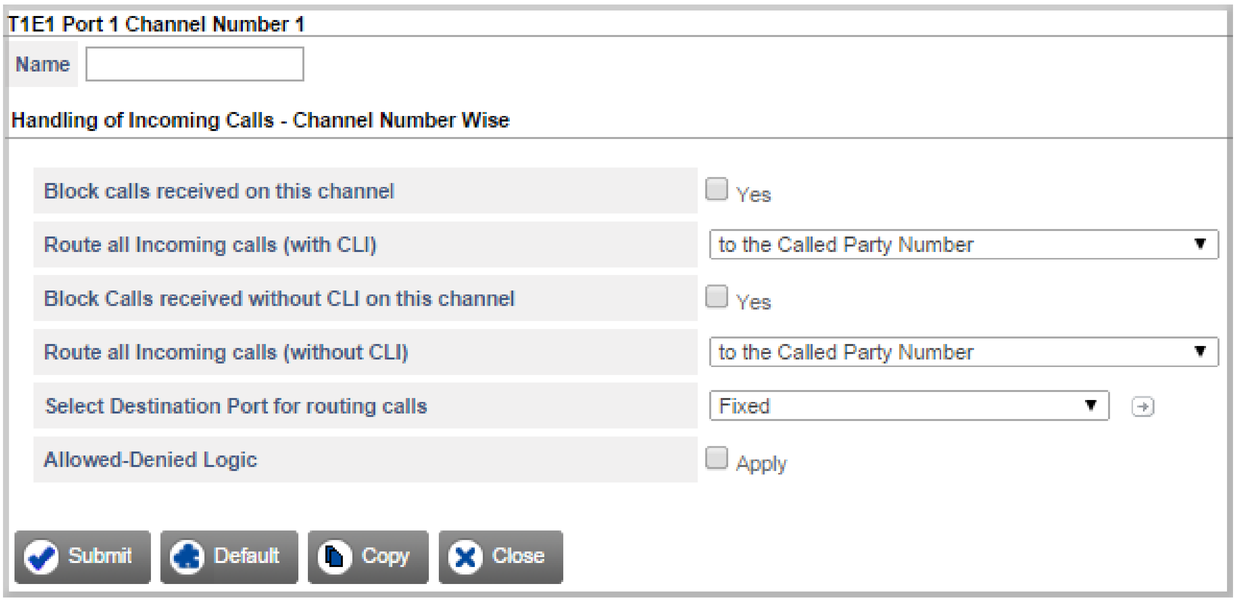


- Click **Settings** .
- The **T1E1 Port 1 - Call Routing - Channel Number Wise** window opens.


T1E1 Port 1 - Call Routing - Channel Number wise

Channel Number	Name	Call Routing
CH-1		Route calls to number received in SETUP message using SIP Trunk 1 - 1
CH-2		Route calls to number received in SETUP message using SIP Trunk 1 - 1
CH-3		Route calls to number received in SETUP message using SIP Trunk 1 - 1
CH-4		Route calls to number received in SETUP message using SIP Trunk 1 - 1
CH-5		Route calls to number received in SETUP message using SIP Trunk 1 - 1
CH-6		Route calls to number received in SETUP message using SIP Trunk 1 - 1
CH-7		Route calls to number received in SETUP message using SIP Trunk 1 - 1
CH-8		Route calls to number received in SETUP message using SIP Trunk 1 - 1
CH-9		Route calls to number received in SETUP message using SIP Trunk 1 - 1

- Click the respective channel number to configure the parameters.



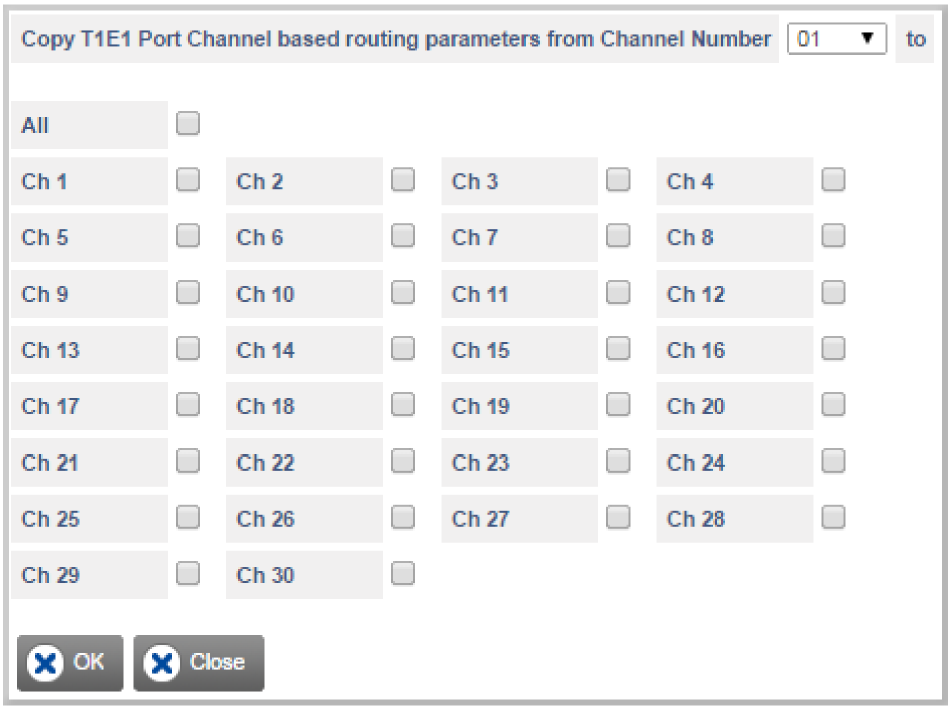
Configure the routing parameters for each channel.

- Block calls received on this channel.
- Route all Incoming Calls (with CLI), see "Handling of Incoming Calls".
- Block Calls received without CLI on this channel.
- Route all Incoming Calls (without CLI), see "Handling of Incoming Calls".
- Select the destination port for routing calls, see "Destination Port Determination".
- Allowed-Denied Logic, see "Allowed - Denied Logic".
- Handling of Outgoing Calls, see "Handling of Outgoing Calls".

Copy Channel Based Routing Parameters

- You can also copy the settings of a T1E1 Channel to another T1E1 Channel using the **Copy** button. To do this,

- Click the **Copy** button. The **Copy T1E1 Port Channel based routing parameters from Channel Number** window opens.



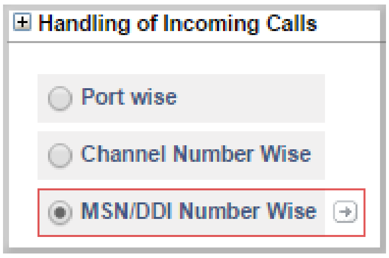
- In the **Copy T1E1 Port Channel based routing parameters from Channel Number** box, select the number of the channel you want to copy settings *From*. Select the check boxes of the desired channel numbers you want to copy the settings *To*.
- If you want to copy the settings *To* all the channels, select the **All** check box.
- Click the **OK** button.
- Once you have copied the settings, you can again edit the specific parameters of the T1E1 Channel you copied the settings to.

- Close the **T1E1 Port 1 Channel Number 1** window.
- To configure any Channel, click the respective channel number on the **T1E1 Port 1 - Call Routing - Channel Number Wise** window and follow the same instructions as given above.
- Close the **T1E1 Port 1 - Call Routing - Channel Number Wise** window.

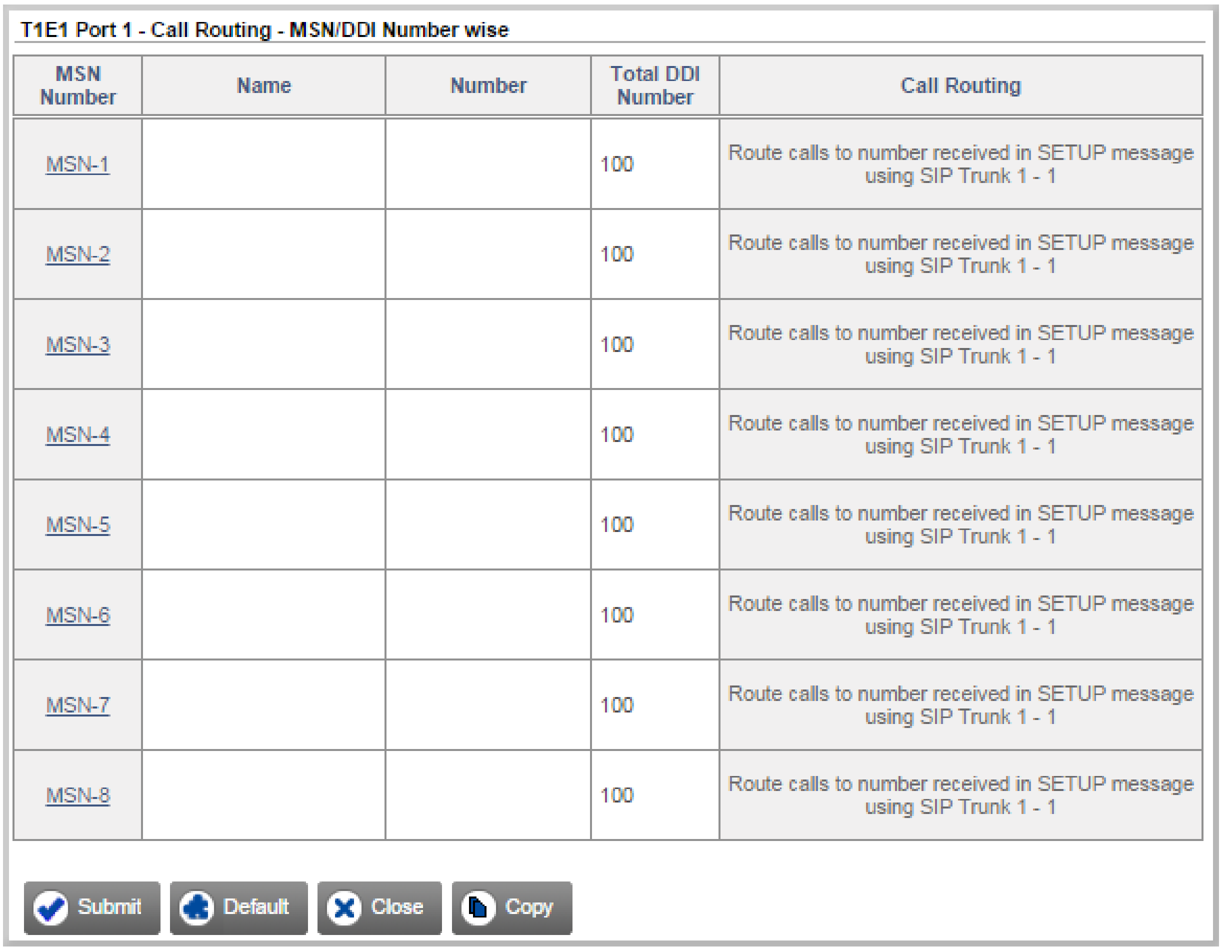
MSN/DDI Number Wise

To configure Handling of Incoming Calls for each MSN Number,

- Select the **MSN/DDI Number Wise** check box.



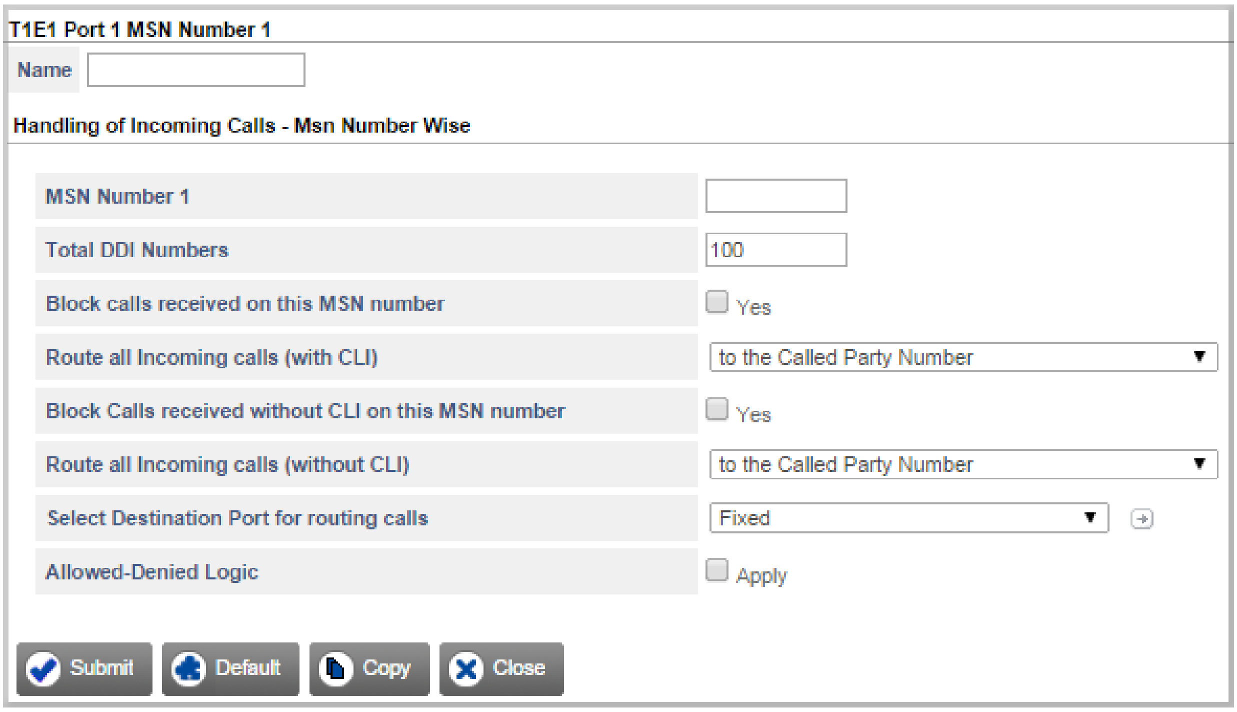
- Click **Settings** .
- The **T1E1 Port 1 - Call Routing - MSN/DDI Number Wise** window opens.


T1E1 Port 1 - Call Routing - MSN/DDI Number wise

MSN Number	Name	Number	Total DDI Number	Call Routing
MSN-1			100	Route calls to number received in SETUP message using SIP Trunk 1 - 1
MSN-2			100	Route calls to number received in SETUP message using SIP Trunk 1 - 1
MSN-3			100	Route calls to number received in SETUP message using SIP Trunk 1 - 1
MSN-4			100	Route calls to number received in SETUP message using SIP Trunk 1 - 1
MSN-5			100	Route calls to number received in SETUP message using SIP Trunk 1 - 1
MSN-6			100	Route calls to number received in SETUP message using SIP Trunk 1 - 1
MSN-7			100	Route calls to number received in SETUP message using SIP Trunk 1 - 1
MSN-8			100	Route calls to number received in SETUP message using SIP Trunk 1 - 1

Submit Default Close Copy

- Click the respective MSN number to configure the parameters.



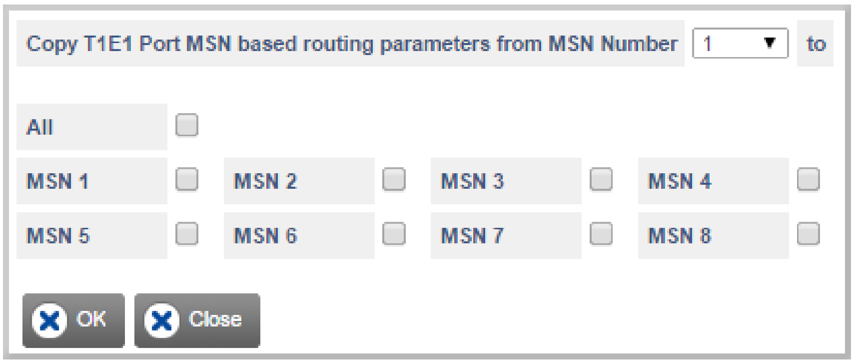
Configure the routing parameters for each MSN number:

- **MSN Number 1:** Enter the first **MSN Number** (max. 24 digits) provided by your Service Provider. Valid digits are 0-9, # and *. Default: Blank.
- **Total DDI Number:** Specify **Total DDI Numbers** provided by your Service Provider. Valid range is 1 to 9999. Default: 0100.
- **Name:** Assign a Name for identification.
- Block calls received on this MSN Number.
- Route all Incoming Calls (with CLI), see “Handling of Incoming Calls”.
- Block Calls received without CLI on this MSN number.
- Route all Incoming Calls (without CLI), see “Handling of Incoming Calls”.
- Select the destination port for routing calls, see “Destination Port Determination”.
- Allowed-Denied Logic, see “Allowed - Denied Logic”.
- Handling of Outgoing Calls, see “Handling of Outgoing Calls”.

Copy MSN Based Routing Parameters

- You can also copy the settings of a MSN Number to another MSN Number using the **Copy** button. To do this,

- Click the **Copy** button. The **Copy T1E1 Port MSN based routing parameters from MSN Number** window opens.



- In the **Copy T1E1 Port MSN based routing parameters from MSN Number** box, select the MSN Number you want to copy settings *From*. Select the check boxes of the desired MSN Numbers you want to copy the settings *To*.

- If you want to copy the settings *To* all the MSN Numbers, select the **All** check box.

- Click the **OK** button.

- Once you have copied the settings, you can again edit the specific parameters of the MSN Number you copied the settings to.

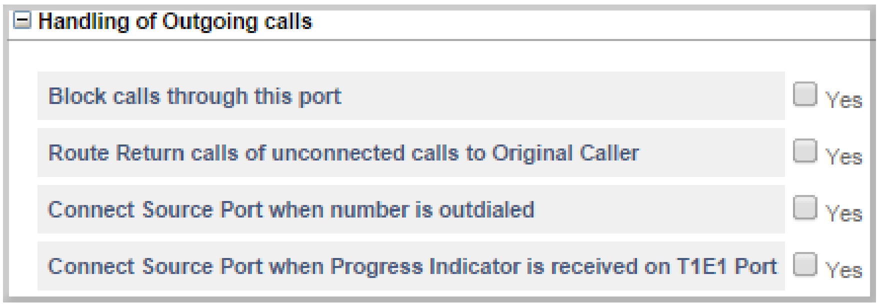
- Close the **T1E1 Port 1 MSN Number 1** window.

- To configure any MSN Number, click the respective MSN Number on the **T1E1 Port 1 - Call Routing - MSN/DDI Number Wise** window and follow the same instructions as given above.

- Close the **T1E1 Port 1 - Call Routing - MSN/DDI Number Wise** window.

Handling of Outgoing Calls

Click **Handling of Outgoing Calls** to expand.



When T1/E1 Port is determined as the destination port, numbers dialed from this port constitute outgoing calls.

For outgoing calls from T1/E1 Port, you can apply the features Automatic Number Translation (ANT) and Route Calls Returned Unconnected to Original Caller.

- Select the **Block calls through this port** check box, if you do not want to route outgoing calls though this port.

- Enable **Route Return calls of unconnected calls to Original Caller** check box, if you want SARVAM UMG to route outgoing calls made from this port that return unconnected back to the original caller. Default: Disabled.

  If you enable this feature, when an outgoing call is made using this port, and the Called Party is found busy or does not respond, SARVAM UMG stores the number of the calling party, the number of the called party and this port (through which the outgoing call was made). A record of each such call is stored for the duration of the Unconnected Calls Record Delete Timer (configurable; default: 999 minutes).

  If the called party returns the call before the expiry of this Timer, SARVAM UMG checks whether *Apply RCOC only if the caller calls back on the same trunk from which the call was made* is enabled or not, and accordingly places the incoming call to the original calling party. To change the duration of this timer, delete records of such calls and enable/disable the *Apply RCOC only if the caller calls back on the same trunk from which the call was made* check box, see "System Parameters".

- To connect the Source Port with the Destination Port without waiting for the call on the Destination Port to mature, enable the **Connect Source Port when number is outdialed** check box. Default: Disabled.

  In all Destination Number Determination methods except *After Answering the Call and Collecting the Digits*, the Source Port gets connected to the Destination Port only after the call has matured, that is, the called party has answered the call. Until the call matures, the caller hears only Ring Back Tone played by the network.

  By connecting the Source Port with the Destination Port immediately after the number is dialed, the caller can know the state of the call; if the called party is busy, not responding, not reachable or is rejecting the call.

- Enable **Connect Source Port when Progress Indicator is received on T1E1 Port** check box, to connect Source Port with the Destination Port as soon as Progress Indicator is received on T1E1 Port without waiting for the call on the Destination Port to mature. Default: Disabled.

If you enable Connect Source Port when Progress Indicator is received on T1E1 Port, you will not be able to provide the features "Making a New Call using Access Code" and "Disconnecting a Call using Access Code" to users.

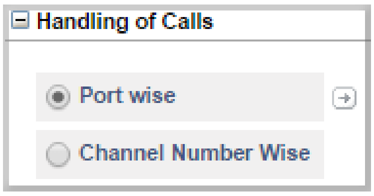
- Click **Submit** to save settings.

Handling of Calls

Click **Handling of Calls** to expand.

If you have selected **Network** as Orientation Type,

- Select the method to route the incoming calls from the **T1E1 Port**.

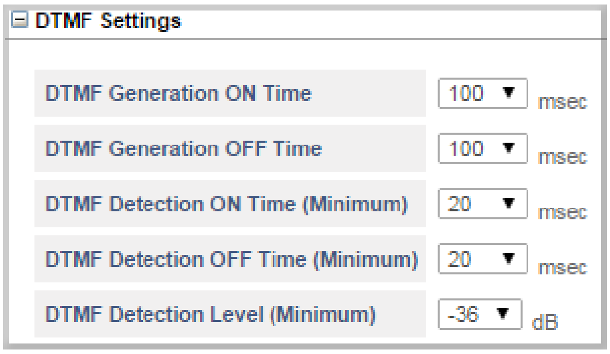


SARVAM UMG provides two options for **Handling of Calls** when the Port is set in Network mode.

- **Port Wise:** Select this method to apply the call routing method for the entire port. See "Port Wise".
- **Channel Number Wise:** Select this method to apply a different call routing method for each of the 30 channels. See "Channel Number Wise".

DTMF Settings

- Click **DTMF Settings** to expand.



- Select the appropriate **DTMF Generation ON Time** for the T1E1 Port. This is the time for which the DTMF digit which is to be outdialed remains ON. Valid range is 50 to 250 msec. Default: 100 msec.

- Select the appropriate **DTMF Generation OFF Time** for the T1E1 Port. This is the time for which system should wait before dialing the successive DTMF digits so that the T1E1 network can detect the dialed digits. Valid range is 50 to 250 msec. Default: 100 msec.

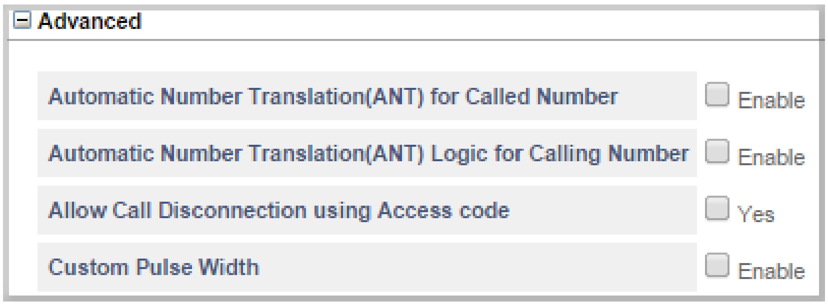
- Select the appropriate **DTMF Detection ON Time (Minimum)** for the T1E1 Port. This is the minimum time period for which the DTMF signal should be present in order to be detected. Valid range is 10 to 200 msec. Default: 20 msec.

- Select the appropriate **DTMF Detection OFF Time (Minimum)** for the T1E1 Port. This is the minimum time period between successive DTMF digits. Valid range is 10 to 200 msec. Default: 20 msec.

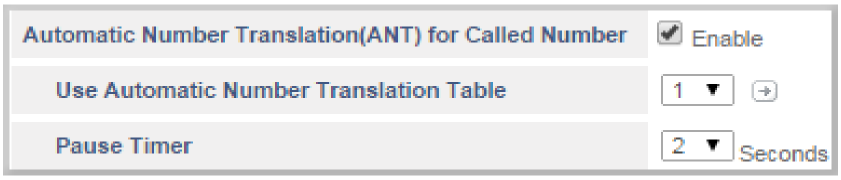
- Select the appropriate **DTMF Detection Level (Minimum)** for the T1E1 Port. This is the minimum level (dB) of the DTMF digit to be considered as valid. Default: -36 dB.

Advanced

- Click **Advanced** to expand.



- You can apply **Automatic Number Translation logic** on outgoing calls made from the T1E1 Port.
- To apply ANT logic on the Called Numbers, select the **Automatic Number Translation (ANT) for Called Number** check box. Default: Disabled.

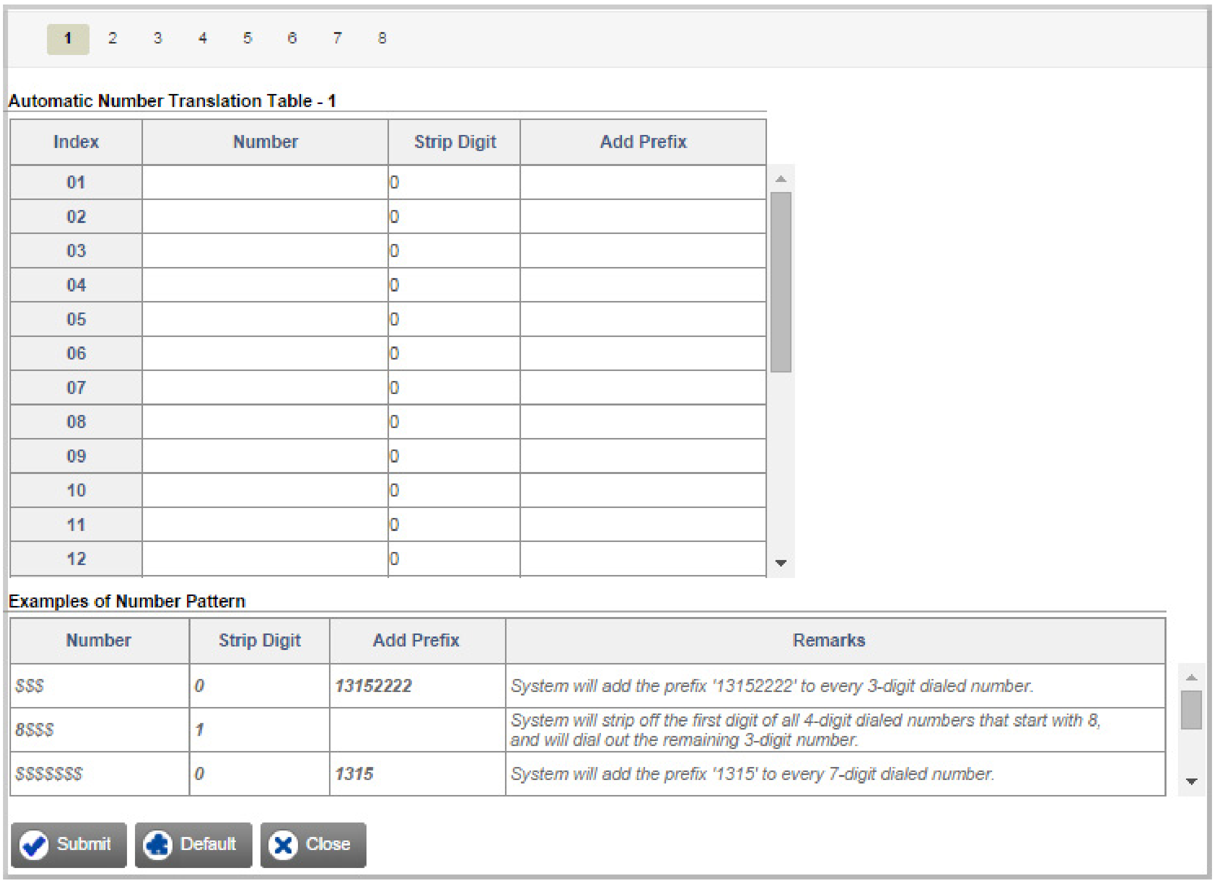


  - In **Use Automatic Number Translation Table**, select the ANT Table number you have configured for the Called Numbers. Default: Table 1.

    If you have not configured the Automatic Number Translation Table,

    - Click **Settings**.

- The **Automatic Number Translation Table** window opens.



1 2 3 4 5 6 7 8

Automatic Number Translation Table - 1

Index	Number	Strip Digit	Add Prefix
01		0	
02		0	
03		0	
04		0	
05		0	
06		0	
07		0	
08		0	
09		0	
10		0	
11		0	
12		0	

Examples of Number Pattern

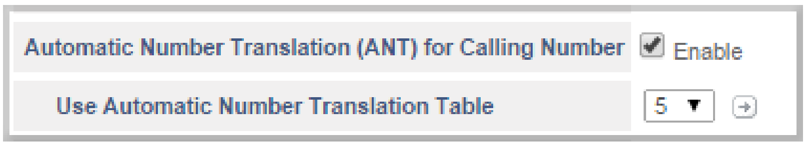
Number	Strip Digit	Add Prefix	Remarks
$$$	0	13152222	System will add the prefix '13152222' to every 3-digit dialed number.
8$$$	1		System will strip off the first digit of all 4-digit dialed numbers that start with 8, and will dial out the remaining 3-digit number.
$$$$$$$	0	1315	System will add the prefix '1315' to every 7-digit dialed number.

Submit Default Close

- You may configure the default Automatic Number Translation Table or any other Table. See "Automatic Number Translation (ANT)" to configure the ANT Table.
- Click **Submit** to save the ANT Table and close the window.
- Return to ANT parameter and assign the ANT Table you configured.
- Click **Submit**.

- Set the duration of the **Pause Timer**, if you have configured ^ (Pause) in the Add Prefix column of the ANT Table. Valid range is 1 to 9 seconds. Default: 2 seconds.

- To apply ANT logic on the Calling Numbers, click the **Automatic Number Translation (ANT) for Calling Number** check box. Default: Disabled.

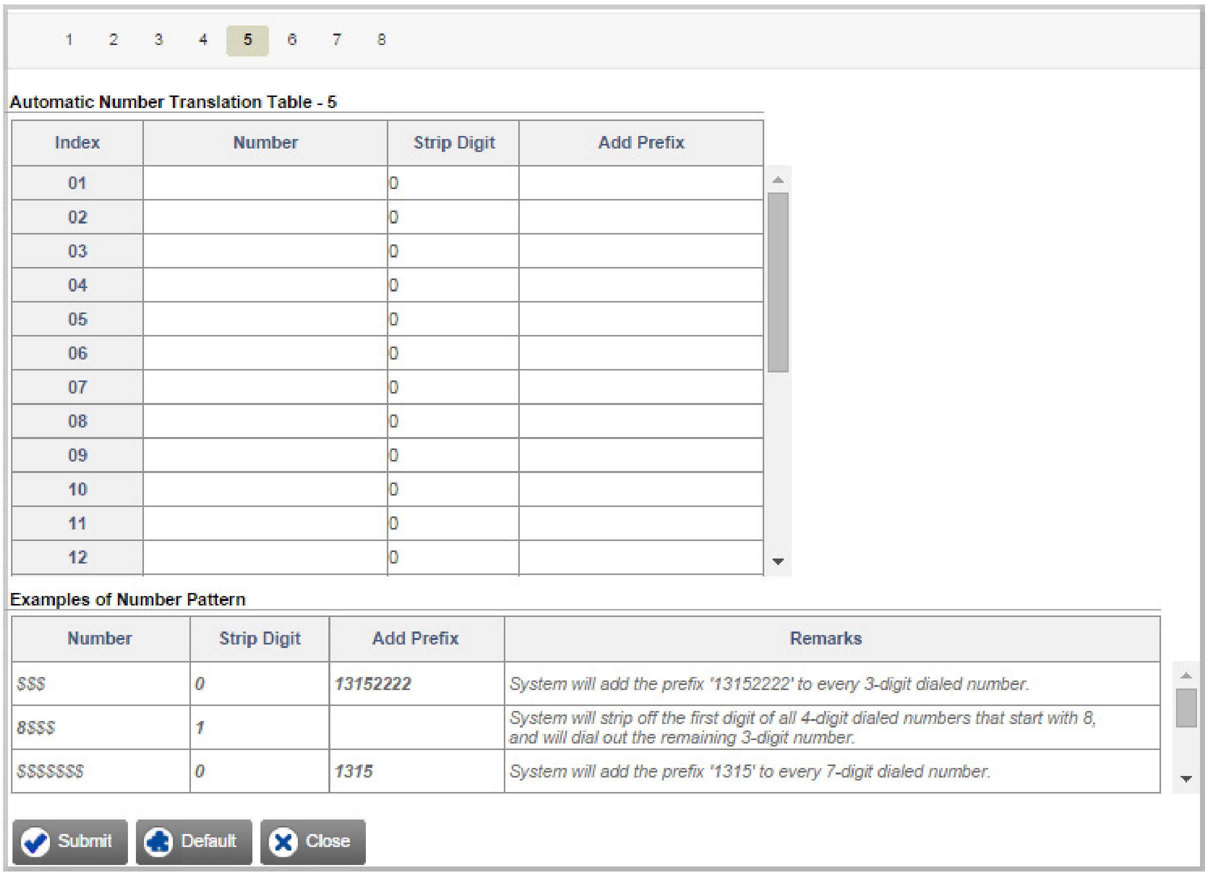


- In the **Use Automatic Number Translation Table**, select the ANT Table number you have configured for the Calling Numbers. Default: Table 5.

  If you have not configured the Automatic Number Translation Table,

- Click **Settings** .

- The **Automatic Number Translation Table** window opens.



Index	Number	Strip Digit	Add Prefix
01		0	
02		0	
03		0	
04		0	
05		0	
06		0	
07		0	
08		0	
09		0	
10		0	
11		0	
12		0	

Number	Strip Digit	Add Prefix	Remarks
$$$	0	13152222	System will add the prefix '13152222' to every 3-digit dialed number.
8$$$	1		System will strip off the first digit of all 4-digit dialed numbers that start with 8, and will dial out the remaining 3-digit number.
$$$$$$$	0	1315	System will add the prefix '1315' to every 7-digit dialed number.

- You may configure the default **Automatic Number Translation Table - 5** or any other Table. See “Automatic Number Translation (ANT)” to configure the ANT Table.

- Click **Submit** to save the ANT Table and close the window.

- Return to ANT parameter and assign the ANT Table you configured.

- To enable the feature Disconnect Call using Access Code on the T1E1 Port, select the **Allow Call Disconnection using Access code** check box. To know more about this feature, see “Disconnecting a Call using Access Code”.

- To customize the pulse width option and set the pulse shapes configure the **Custom Pulse** parameters. SARVAM UMG generates pulse shapes which match the country standard, where it is installed. However, if the standard pulse shape does not match, SARVAM UMG enables you to customize the pulse width to match your exact requirements.

  To use customize pulse width option and set the pulse shape in 1 to 4 phases, keep the T1/E1 Custom Pulse Width (CPW) flag enabled.

  - In **Custom Pulse Width Word 1,** set the pulse width for setting pulse shape in the 1st phase. Valid range is 0 to 127. Default: 63.

- In **Custom Pulse Width Word 2,** set the pulse width for setting pulse shape in the 2nd phase.Valid range is 001 to 127. Default: 58.

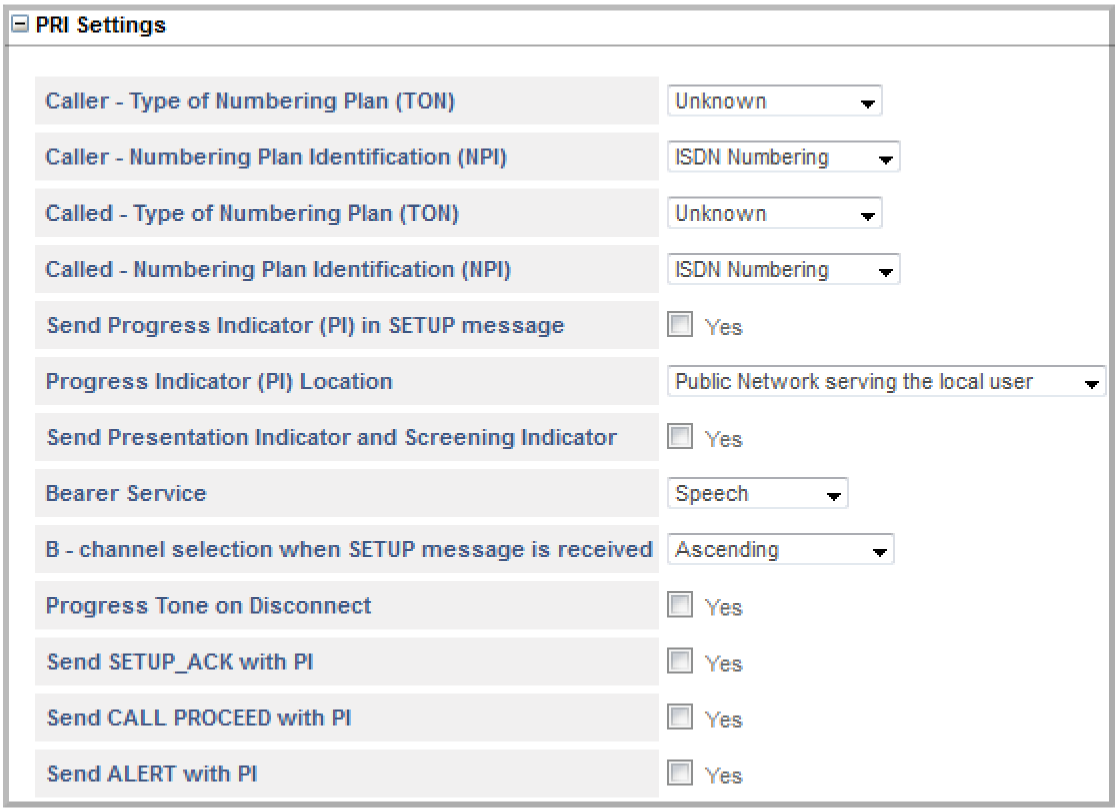
- In **Custom Pulse Width Word 3,** set the pulse width for setting pulse shape in the 3rd phase. Valid range is 001 to 127. Default: 76.

- In **Custom Pulse Width Word 4,** set the pulse width for setting pulse shape in the 4th phase. Valid range is 001 to 127. Default: 0.

PRI Settings

If you have selected **PRI** as **Signaling Type**, configure the PRI parameters.

- Click **PRI Settings** to expand.



- Set the duration of the **Overlap Receiving Timer**. This timer is relevant while receiving the called party number information in overlap receiving mode. Valid range is 01 to 99 seconds. Default: 5 seconds.

- Select the required option for sending the **Caller-Type of Numbering Plan (TON) —** Unknown, International, National, Network Specific, Subscriber, Abbreviated or Reserved. Default: Unknown.

- Select the required option for sending the **Caller-Numbering Plan Identification (NPI)** — Unknown, ISDN Numbering, Data Numbering, Telex Numbering, National Numbering, Private or Reserved. Default: ISDN Numbering.

- Select the required option for sending the **Called-Type of Numbering Plan (TON)** — Unknown, International, National, Network Specific, Subscriber, Abbreviated or Reserved. Default: Unknown.

- Select the required option for sending the **Called-Numbering Plan Identification (NPI) —** Unknown, ISDN Numbering, Data Numbering, Telex Numbering, National Numbering, Private or Reserved. Default: ISDN Numbering.

- Select the **Send Progress Indicator (PI) in SETUP message** check box if you want the progress indicator value to be sent in SETUP message. Default: Disabled.

- Set the **Progress Indicator (PI) value in SETUP message** to the desired value. You can select — 1 or 3.

  Progress indicator 1 indicates that the call is not end-to-end ISDN and further call progress information may be available in-band.

  Progress indicator 3 indicates that the origination address is non-ISDN.
  This value will be included in the Setup Message to indicate whether the calling party is an ISDN device or not. Default: 1.

- Select the **Progress Indicator (PI) Location** for the SETUP message.The location is a progress indicator information element that indicates from where the message is coming. Default: Public Network serving the local user.

- Select the **Send Presentation Indicator and Screening Indicator** check box, if you want the system to display the presentation and screening information to the remote end. Default: Disabled.

  - Select the required **Presentation Indicator**. This allows remote end to know whether CLI Number should be displayed to user or not. You can select — Presentation Allowed, Presentation Restricted or Received from Source Port. Default: Received from Source Port.

  - Select the required **Screening Indicator**. This indicates whether the information is provided by the user or the network along with the screening details — not screened, verified and passed or verified and failed. Default: User-provided, not screened.

- Select the **Bearer Service** supported by your service provider. This will be sent in the SETUP Message. You can select — Speech, 3.1 KHz Audio. Default: Speech

- Select the **B - channel selection when SETUP message is received**[15] as per your requirement.

- Select the **Progress Tone on Disconnect** check box, If you want the system to play the progress tone on the T1E1 Port when call is released by the remote end or released by the system. Default: Disabled.

- Select the **Send SETUP_ACK with PI** check box, if you want the system to send PI (Progress indicator) element in Setup Ack message. Default: Disabled.

- Select the **Send CALL PROCEED with PI** check box, if you want the system to send PI (Progress indicator) in Proceed message. Default: Disabled.

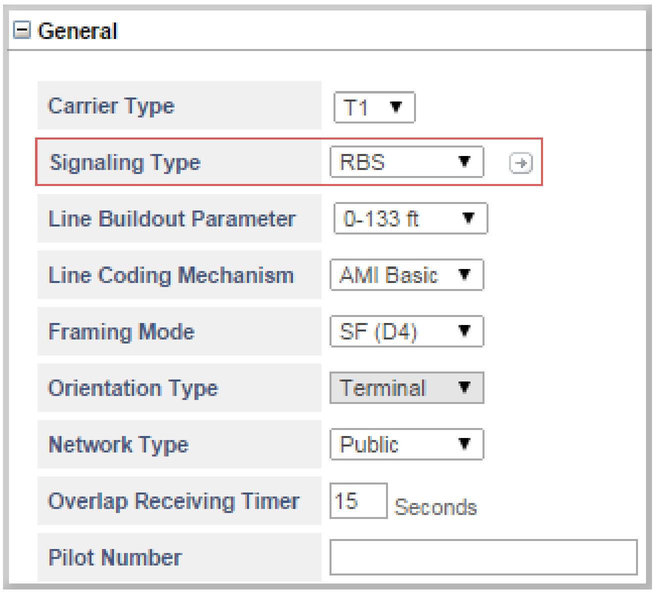
- Select the **Send ALERT with PI** check box, if you want the system to send PI (Progress indicator) in Alert message. Default: Disabled.

- Click **Submit** to save changes.

15. This parameter is not applicable in this version, it is meant for future use.

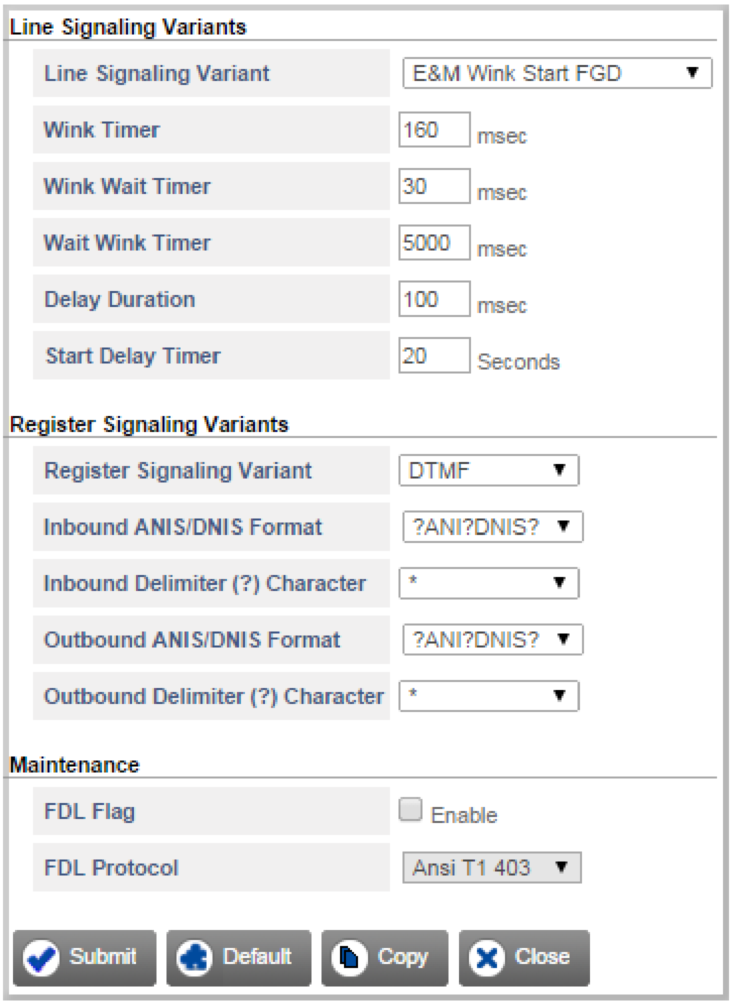
RBS Settings

If you have selected **RBS** as **Signaling Type**, configure the RBS parameters.



- Click **Settings** .

- RBS parameters window opens.



Line Signaling Variants

- Select the **Line Signaling Variant**. You can select — FXS Loop Start, FXO Loop Start, FXS Ground Start, FXO Ground Start, E&M Immediate Dial/Start, E&M Wink Start or E&M Wink Start FGD. Default: E&M Wink Start FGD.

- Set the duration of the **Wink Timer**. This signifies the momentary Off-Hook condition to acknowledge end of an outgoing call. Valid range is 0001 to 9999 msec. Default: 160 msec.

- Set the duration of the **Wink Wait Timer**. This signifies the maximum time the system should wait before sending a wink start signal after an incoming seizure is detected. Valid range is 0001 to 9999 msec. Default: 30 msec.

Ensure that this timer is greater than the Wink Wait Timer of the other end.

- Set the duration of the **Wait Wink Timer**. This signifies the time for which SARVAM UMG will wait for receiving the DNIS after sending the outgoing seizure signal. Valid range is 0001 to 9999 msec. Default: 5000 msec.

Make sure that this timer is greater than the Wait Wink Timer of the other end.

- Set the duration of the **Delay Duration**. This signifies the time after which the DNIS information is to be sent while making an outgoing call. Valid range is 0001 to 9999 msec. Default: 100 msec.

- Set the duration of the **Start Delay Timer**. This signifies the time for which SARVAM UMG waits for receiving DNIS from the network. This timer is loaded on receiving the Off-hook (I/C Seizure) on the receive channel (while receiving an incoming call). Valid range is 0001 to 9999 msec. Default: 20 msec.

Register Signaling Variant

- Select the **Register Signaling Variant** for T1/E1 Ports. You can select — DTMF, Decadic, MFC R2 or MFC R1. Default: DTMF.

- Select the **Inbound ANI/DNIS Format** for T1/E1 Ports. You can select — ANI, DNIS, ?ANI?, ?DNIS?, ?ANI?DNIS? or ?DNIS?ANI?. Default: ?ANI?DNIS?.

- Enter **Inbound Delimiter (?) Character** as per your requirement. Characters supported in this field are 0-9, #, *, A, B, C and D. Default: *

- Select the **Outbound ANI/DNIS Format** for T1/E1 Port. You can select — ANI, DNIS, ?ANI?, ?DNIS?, ?ANI?DNIS? or ?DNIS?ANI?. Default: ?ANI?DNIS?.

- Enter **Outbound Delimiter (?) Character** as per your requirement. Characters supported in this field are 0-9, #, *, A, B, C and D. Default: *

Maintenance

- FDL is used for communicating general maintenance information for transmitting user defined information within the T1 link. General maintenance information is in the form of Performance Message Report which is generated by SARVAM UMG. Depending upon the FDL Protocol, the Performance Message Report is sent every second, or sent on request.

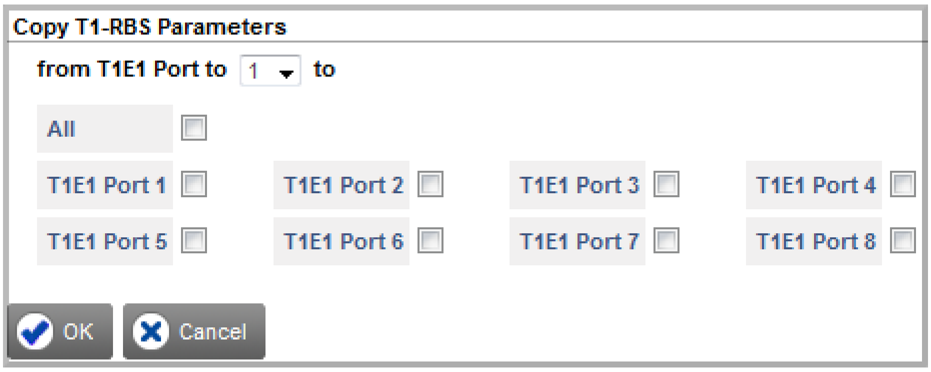
- Select the **FDL Flag** check box to enable, if the Network (Public or Private) to which SARVAM UMG is connected supports FDL. Default: Disabled.

- After enabling the FDL Flag, select the **FDL Protocol** — ANSI T1 403 or AT&T 54016 — for reporting the performance monitoring. Default: ANSI T1 403.

Copy T1-RBS Parameters

- You can also copy the settings of a T1-RBS from one T1E1 port to the another using the **Copy** button. To do this,

- Click the **Copy** button. The **Copy T1-RBS Parameters** window opens.

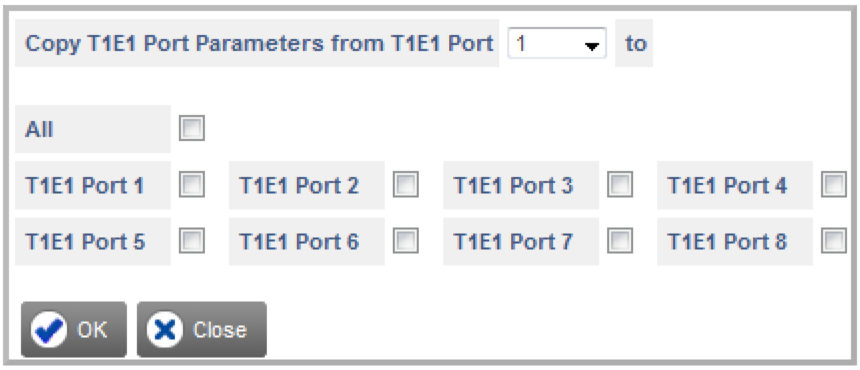


- In the **from T1E1 Port to** box, select the number of the T1-E1 Port you want to copy settings *From*. Select the check boxes of the desired port numbers you want to copy the settings *To*.
- If you want to copy the settings *To* all the T1E1 Ports, select the **All** check box.
- Click the **OK** button.
- Once you have copied the settings, you can again edit the specific parameters of T1-RBS you copied the settings to.

- Close the **Copy T1-RBS Parameters** window.
- Click **Submit** to save changes.
- Close the window to return to the main page.

Copy T1E1 Port Parameters

- You can also copy the settings of a T1E1 Port to another T1E1 Port using the **Copy** button. To do this,
    - Click the **Copy** button. The **Copy T1E1 Port Parameters** window opens.



- In the **Copy T1E1 Port Parameters from T1E1 Port** box, select the number of the port you want to copy settings *From*. Select the check box of the respective port numbers you want to copy the settings *To*.
- If you want to copy the settings *To* all the ports, select the **All** check box.

- Click the **OK** button.
- Once you have copied the settings, you can again edit the specific parameters of the **T1E1** Port you copied the settings to.

E1 Port

SARVAM UMG supports the T1/E1 Ports to which you can connect the T1 or E1 line.

- Click the **Basic Settings** link to expand.

- Click the **T1E1 Port** link.

The T1E1 Port page displays the following parameters:

- **Port:** It displays the T1E1 Port numbers. Click on the desired T1E1 Port number to configure the Port Parameters.

- **Hardware Slot-Port:** The SARVAM UMG can automatically detect and assign the Hardware Slot and Port numbers to the T1E1 software ports. However, if required you may change the Hardware Slot and Port assigned to the T1E1 software port. In this case, enter the desired Hardware Slot and Port number.

- **Enable:** Keep the **T1E1 Ports** enabled. Clear the T1E1 Port **Enable** check box, only if you do not want to use the respective port. Default: Enabled.

- **Name:** Assign a Name to the T1E1 Port for identification. The Name can be a maximum of 24 characters.

- **Status:** This displays the status of Layer 1 and Layer 2, that is, Up or Down.

- **Line Signaling:** It displays the Carrier Type, Signaling Type and the ISDN Switch Variant you select.

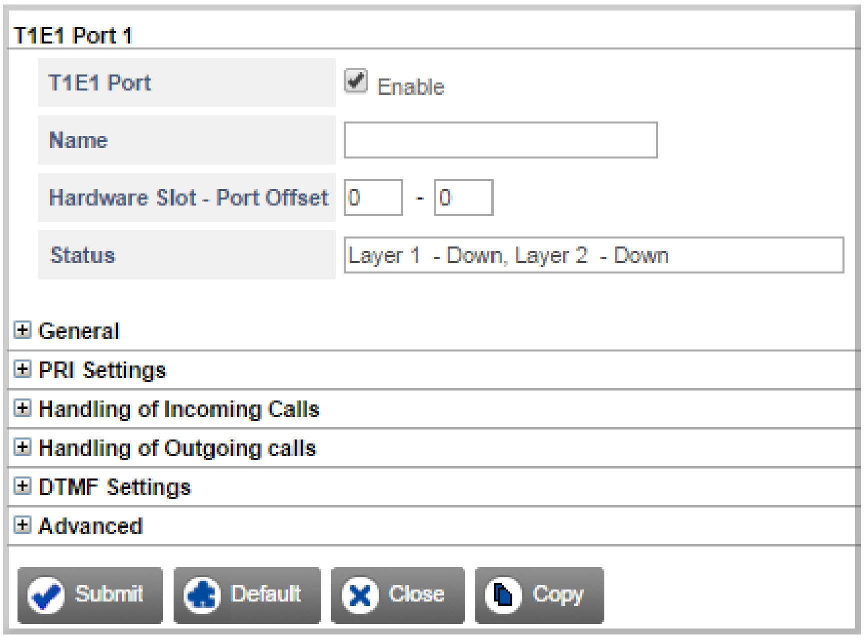
- **Orientation:** It displays the type of orientation you select — Network or Terminal.

- **Call Routing:** It displays the Call Routing Method you select.

To configure the **T1E1 Port**,

- Click **T1E1-1**.

The **T1E1 Port 1** window opens.



- Keep the **T1E1 Port** check box enabled.

  Clear the **T1E1 Port Enable** check box only when you do not want to use this T1E1 Port. Default: Enabled.

- You can assign a **Name** to the T1E1 Port, which will be displayed to the called party, if the called party telephone instrument supports CLI display.

  The name you assign may consist of a maximum of 24 characters. Default: Blank.

- SARVAM UMG will assign the **Hardware Slot - Port Offset** automatically, when any card is inserted in the system.

  Hardware slot is the number of the universal slot of SARVAM UMG in which the T1E1 Card is inserted. Range of slot number is 1-12. Port is the number of T1E1 hardware port on the card to which the T1E1 line is connected.

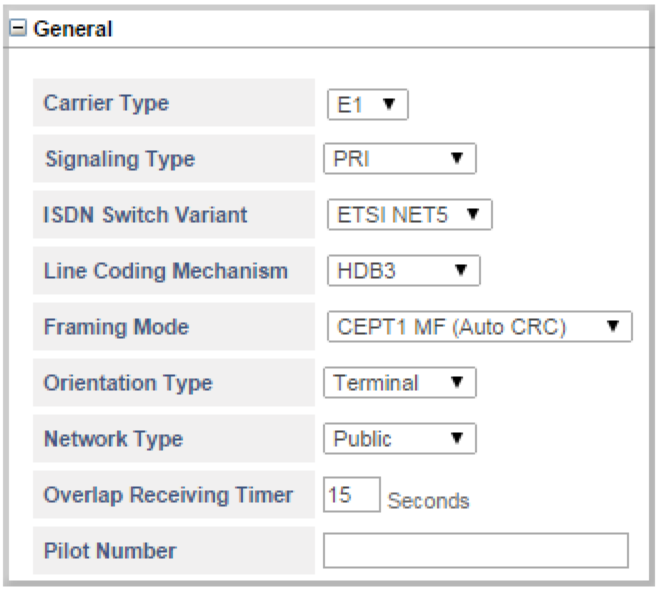
  However, if required, you may change the Hardware Slot and Port assigned to the T1E1 software port. In this case, enter the desired Hardware Slot and Port number.

  If you want to de-assign the Hardware Slot and Port, enter '00' in both fields. By default, Hardware Slot-Port is 0 – 0.

- **Status** displays the status of the T1E1 Port.

General

- Click **General** to expand.



- Select **E1** as the **Carrier Type**. Default: E1.
- Select **Signaling Type**. The Signaling Type signifies the type of signaling to be used on the E1 line.

  SARVAM UMG supports — PRI and CAS signaling for E1 line. Default: PRI.

  - If you select **PRI,** you must configure the PRI parameters. For instructions, see “PRI Settings”.
  - If you select **CAS,** you must configure the CAS parameters. For instructions, see “CAS Settings”.
- ISDN supports a variety of service provider switches. These switches are designed using ISDN standard protocol. The type of switch you select determines various factors — the number of ISDN devices that could be handled, the B-Channel that would support voice, video, data, etc. Each country uses their own specific type of ISDN switch.

  The system supports only ETSI NET5 as the **ISDN Switch Variant.** This parameter is applicable only if you select PRI as the Signaling Type.
- Line Coding is a mechanism to code the digital data into electrical pulses for the purpose of transmission over the communication channel.

  Select the **Line Coding Mechanism —** AMI Basic or HDB3. Default: AMI Basic.
- **Framing** is a formatting resource that splits the digital data into time slots of 8 bits each. Each time slot is treated as single transmission unit. These frames enable the receiver to interpret the data.

  Select the **Framing Mode** as per your requirement. You can select **—** CEPT1 MF (No CRC), CEPT1 MF (Forced CRC) or CEPT1 MF (Auto CRC) Framing Modes. Default: CEPT1 MF (Auto CRC).
- Select the **Orientation Type** for the port as **Terminal** or **Network**, according to your installation scenario. Default: Terminal.

If you have selected *Terminal* as Orientation Type, select the **Network Type** — Public or Private — to specify whether the T1 line is from a **Public** Network (telephone exchange) or from a **Private** Network (to the NT port of a System). Default: Public.

  - For **Terminal** as the Orientation Type, configure — “Handling of Incoming Calls” and “Handling of Outgoing Calls”.

  - For **Network** as the Orientation Type, configure “Handling of Calls”.

- Enter the **Pilot Number** provided by your service provider for the E1 line connected to the T1/E1 Port. Pilot Number is necessary for sending the calling party number when the call is routed using T1/E1 Port and Reverse DDI logic is not applied. Valid digits are 0 to 9, #, *. Default: Blank.

Handling of Incoming Calls

Click **Handling of Incoming Calls** to expand.

Select the method to route the incoming calls from the T1E1 Port.

SARVAM UMG provides three options for **Handling of Incoming Calls —** Port Wise, Channel Number Wise and MSN/DDI Number Wise. Default: Port Wise.

- **Port Wise:** Select this method to apply the call routing method for the entire port.

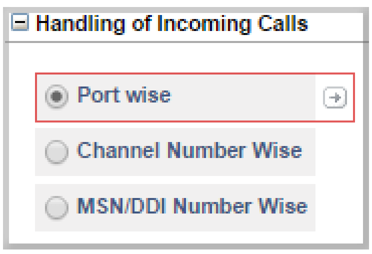
- **Channel Number Wise:** Select this method to apply a different call routing method for each channel. You can configure a different incoming call routing option for each channel.

- **MSN/DDI Number Wise:** Select this method to apply a different call routing method for each MSN number given by the Service Provider for the E1 Line. SARVAM UMG allows you to configure upto 8 MSN Numbers.

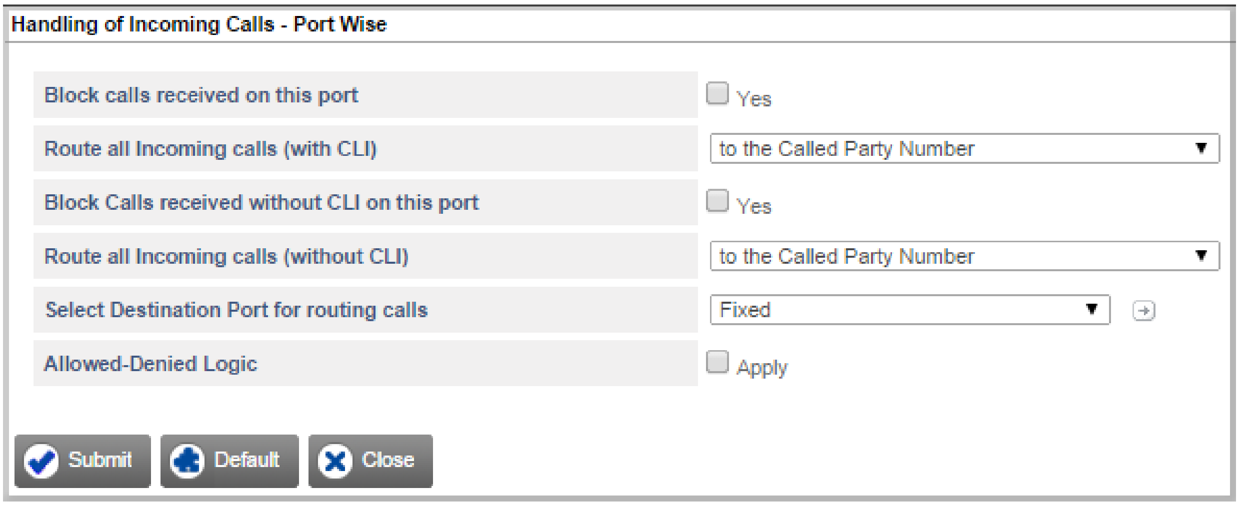
Port Wise

To configure Handling of Incoming Calls Port Wise,

- Select the **Port Wise** check box.



- Click **Settings** .
- The **Handling of Incoming Calls - Port Wise** window opens.



- Keep the **Block calls received on this port** check box disabled.

  Select this check box only if you do not want to route calls received on this port.

Destination Number Determination

Select the desired destination number determination method for routing incoming calls *with* and *without* CLI.

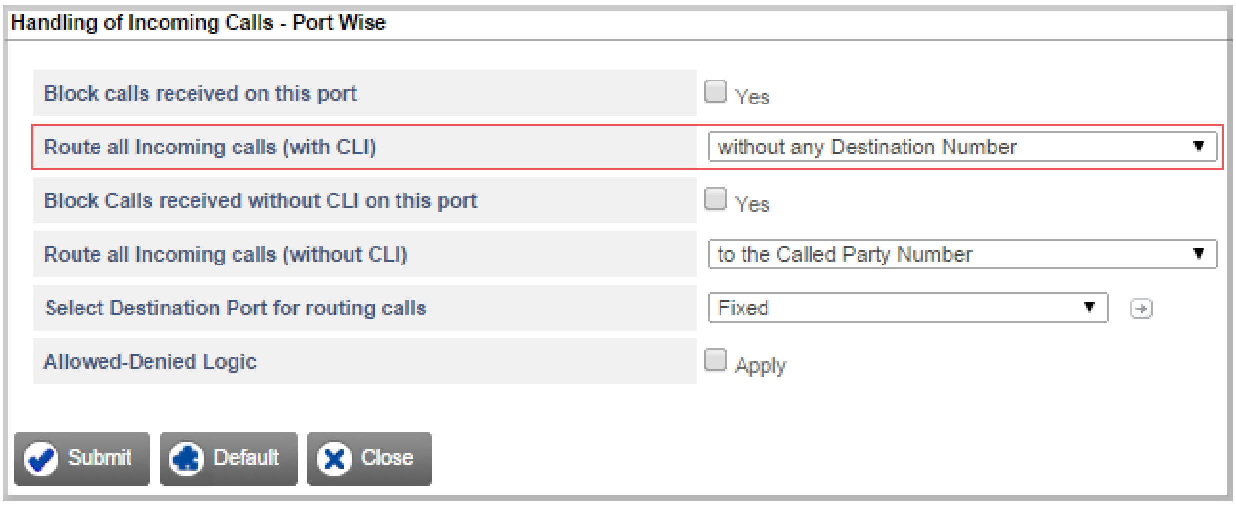
- To **Route all Incoming calls (with CLI)**, you may select from any of the following methods:
  - without any Destination Number
  - to a Fixed Destination Number
  - on the basis of Calling Party Number
  - on the basis of DDI Number
  - to the Called Party Number
  - after Answering the Call and Collecting the Digits
    Default: to the Called Party Number

  Read further for instructions on selecting and configuring each of these destination number determination methods.

If the destination number to be dialed out is an IP Address, SARVAM UMG will not check the Destination Port Determination Method. Instead, it will route the call using the SIP Trunk / Group programmed for IP Dialing. (See "IP Dialing" to know more).

Route Calls without any Destination Number

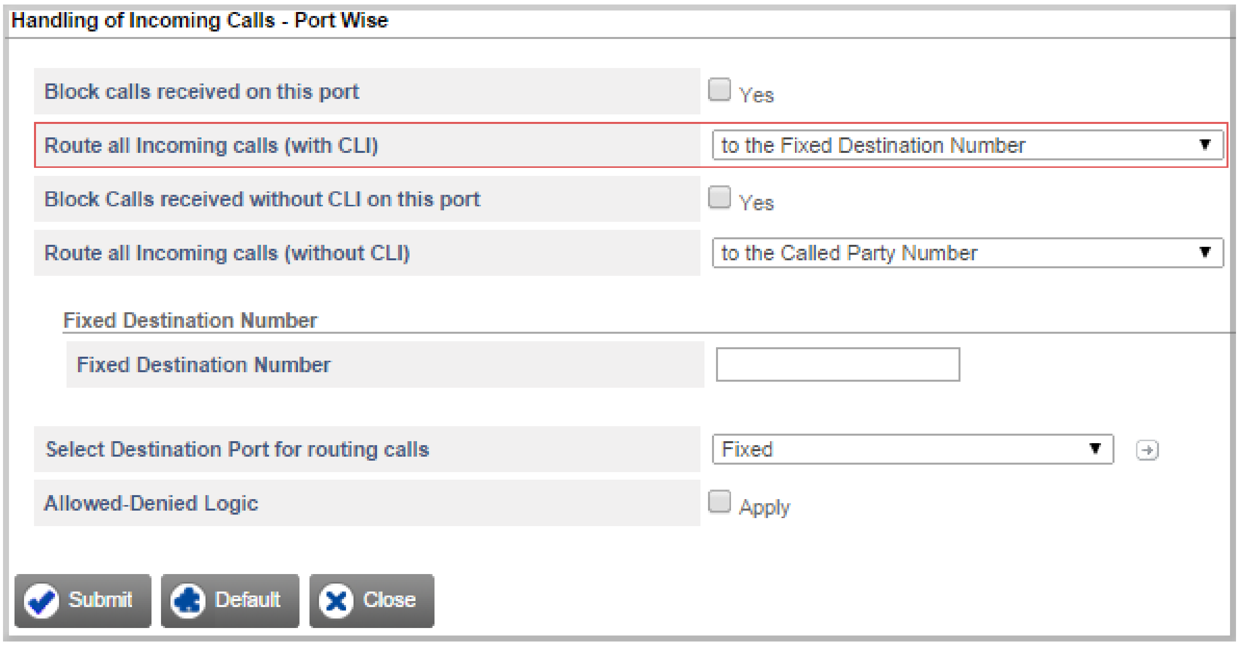
In this method, all calls received on the T1E1 Port are directly routed to the destination port, irrespective of the Destination Number.



- To apply this method, in **Route all incoming calls (with CLI)**, select **without any Destination Number**.

Route to a Fixed Destination Number

In this method, calls received on the T1E1 Port are routed to a fixed destination number, which is configured for the T1E1 Port.



To apply this method, do the following:

- In **Route all Incoming calls (with CLI)**, select **to the Fixed Destination Number**.
- In the **Fixed Destination Number** box that appears, enter the desired destination number. The Destination Number may consist of a maximum of 24 digits. Valid digits are 0 to 9, *, # and. (dot/period). Default: Blank.

- Click **Submit** to save your settings.

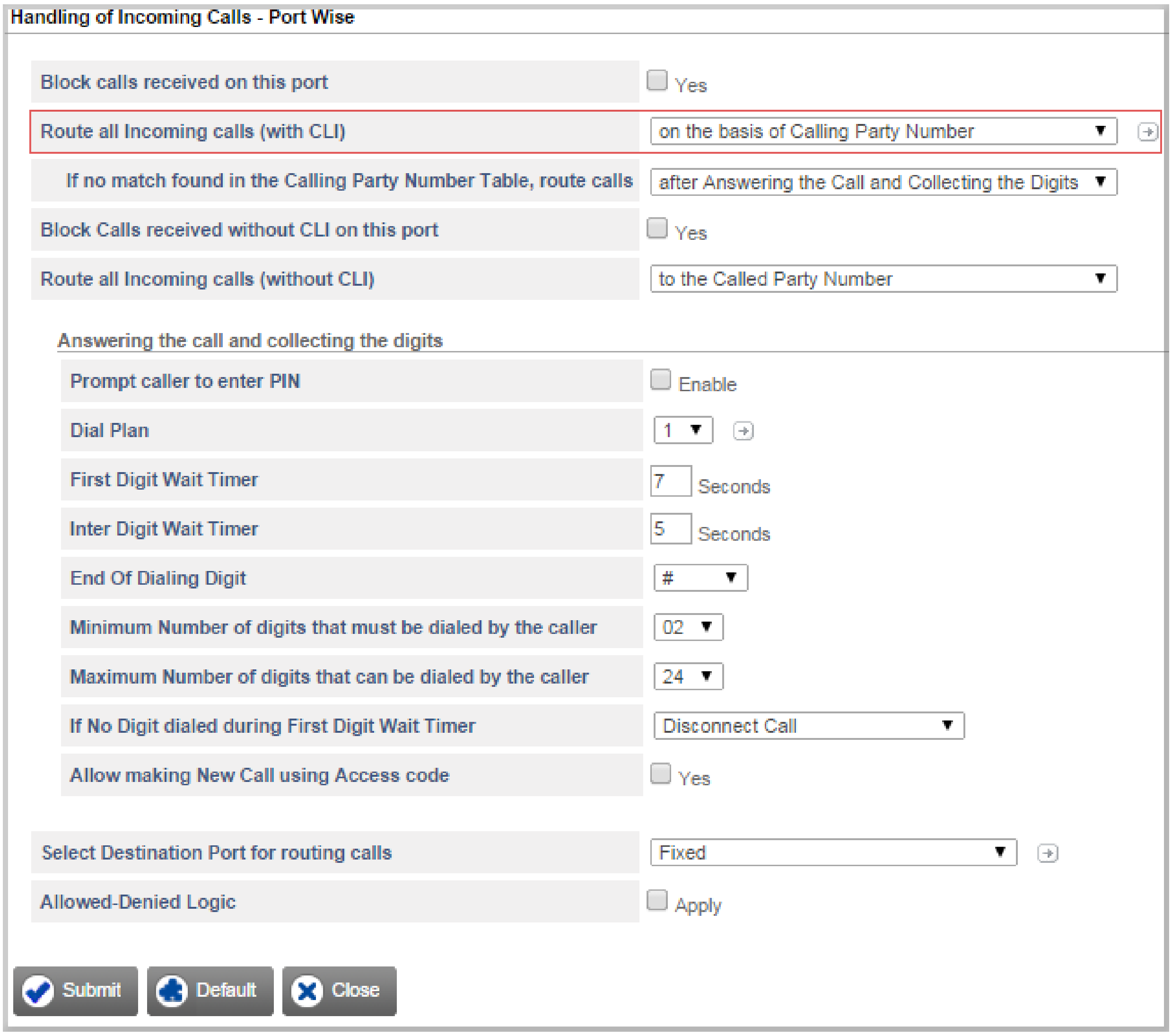
Route on the basis of Calling Party Number

In this method, a call received on the T1E1 Port is routed to a specific number, as per the calling party's number. You must configure the calling party numbers in the *Calling Party Number Based* Table.

When there is an incoming call on the T1E1 Port, SARVAM UMG will match the Calling Party Number with the entries of the Calling Party Number Based Table. If a match is found, the call is routed to the destination number configured for that Calling Party Number.

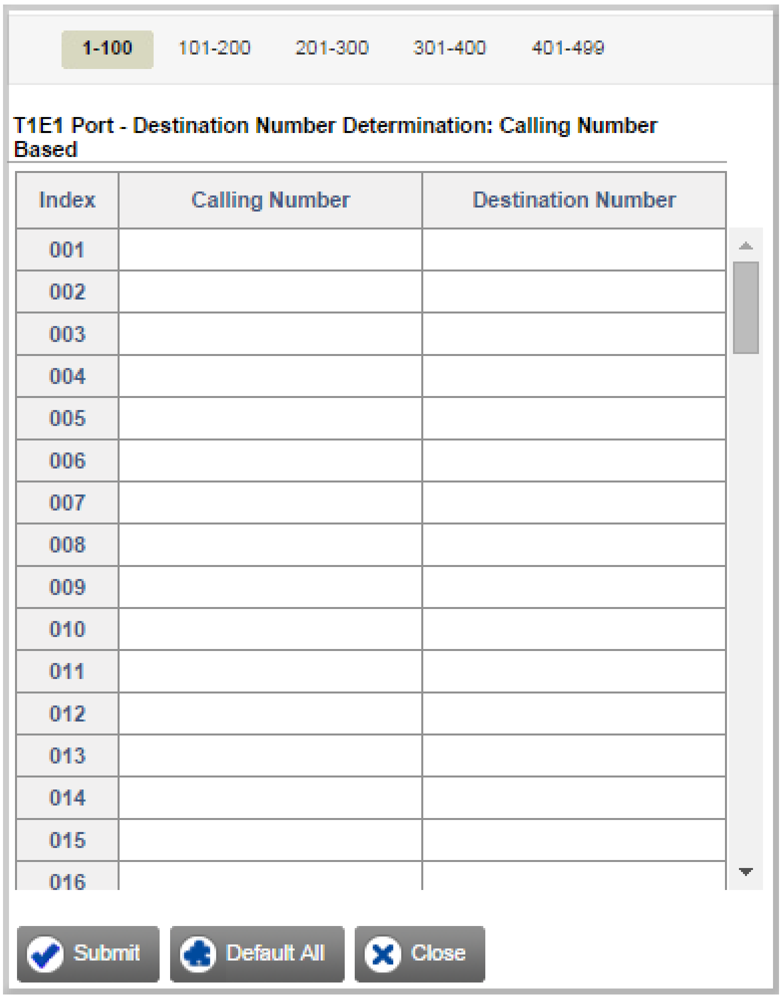
To apply this method, do the following:

- In **Route all Incoming calls (with CLI)**, select **on the basis of Calling Party Number**.



- Click **Settings** .

- The **T1E1 Port - Destination Number Determination: Calling Number Based** Table window opens.



  - In **Calling Number**, enter the calling party numbers. The Calling numbers may consist of a maximum of 24 characters. Default: Blank.

  - For each calling party number, enter a corresponding destination number in **Destination Number**. Destination numbers may consist of a maximum of 24 characters. Digits 0 to 9, *, # and (.) dot are allowed. Default: Blank.

- Click **Submit** to save your entries. Close the window to return to the **Handling of Incoming Calls - Port Wise** window.

  You can also configure the **Calling Number Based** table from *Advanced Settings.* For instructions, see “Destination Number Determination” under *Advanced Settings*.

- Select a method for routing incoming calls with CLI that *do not match* with any entries in the Calling Party Number Based Table.

  In the **If no match found in the Calling Party Number Table, route calls** box, select the desired method from the following options for processing the call:
  - to a Fixed Destination Number
  - on the basis of DDI Number
  - to the Called Party Number
  - after Answering the Call and Collecting the Digits

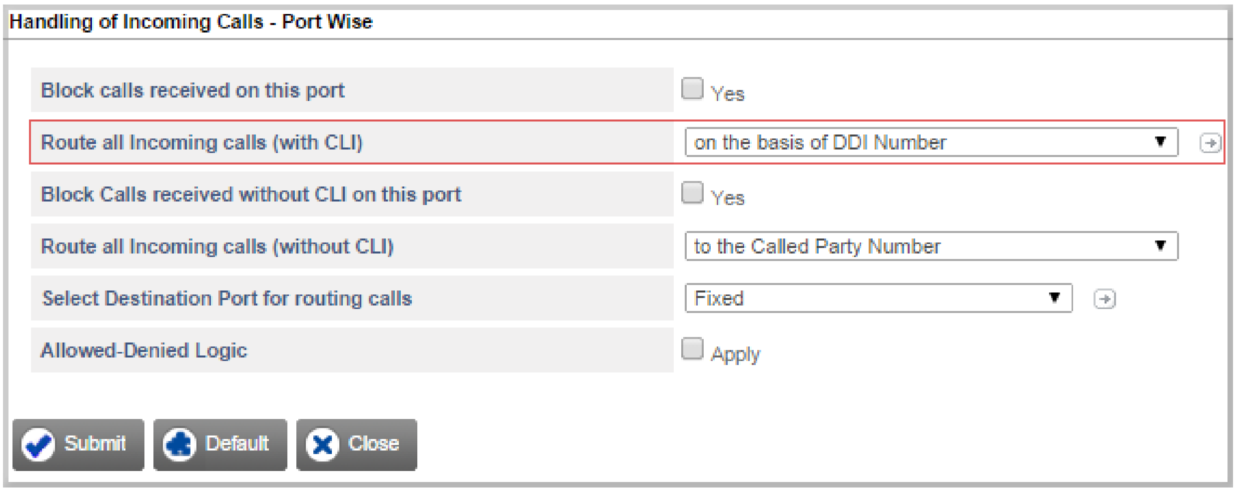
  Default: to the Called Party Number.

Route on the basis of DDI Number

In this method, incoming calls on the T1E1 Port are routed to specific numbers as per the DDI number received in the SETUP message on the T1E1 Port.

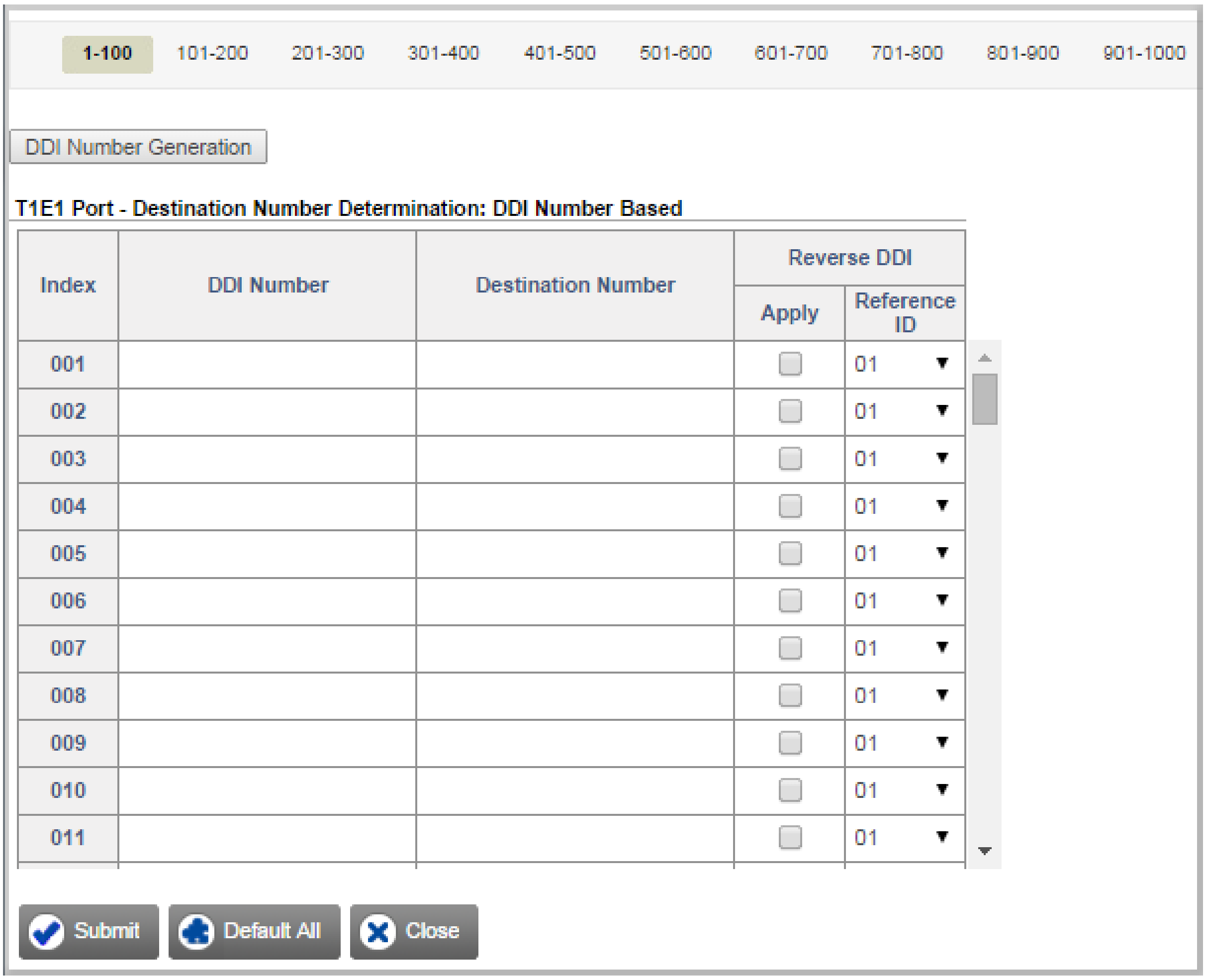
To apply this method, do the following:

- In **Route all Incoming calls (with CLI)**, select **on the basis of DDI Number**.



- Click **Settings** .

The **T1E1 Port - Destination Number Determination: DDI Number Based** Table opens.



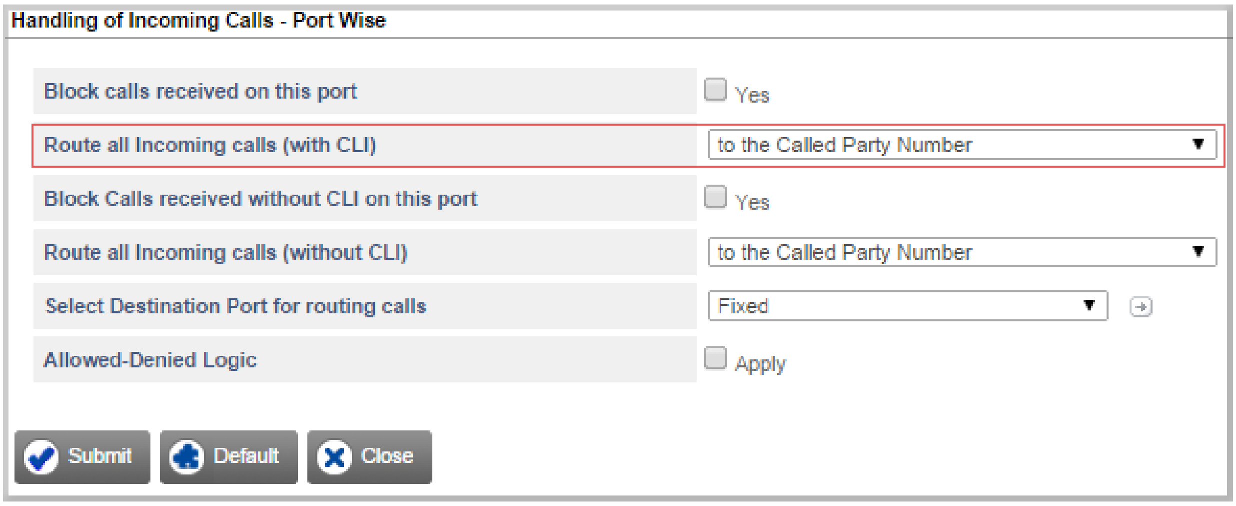
- In **DDI Number**, enter the DDI Numbers allotted by your service provider.
- For each DDI Number, enter the corresponding destination number in **Destination Number**.
- To apply **Reverse DDI** for each number, select the check boxes under **Apply** and select the **Reference ID** for the number. Default: Apply Reverse DDI is disabled and Reference ID is 1.
- Click **Submit** to save and close the window to return to the **Handling of Incoming Calls - Port Wise** window.

  You can also configure the **DDI Number Based** Table from *Advanced Settings.* For instructions, see "Destination Number Determination" under *Advanced Settings*.

Route to the Called Party Number

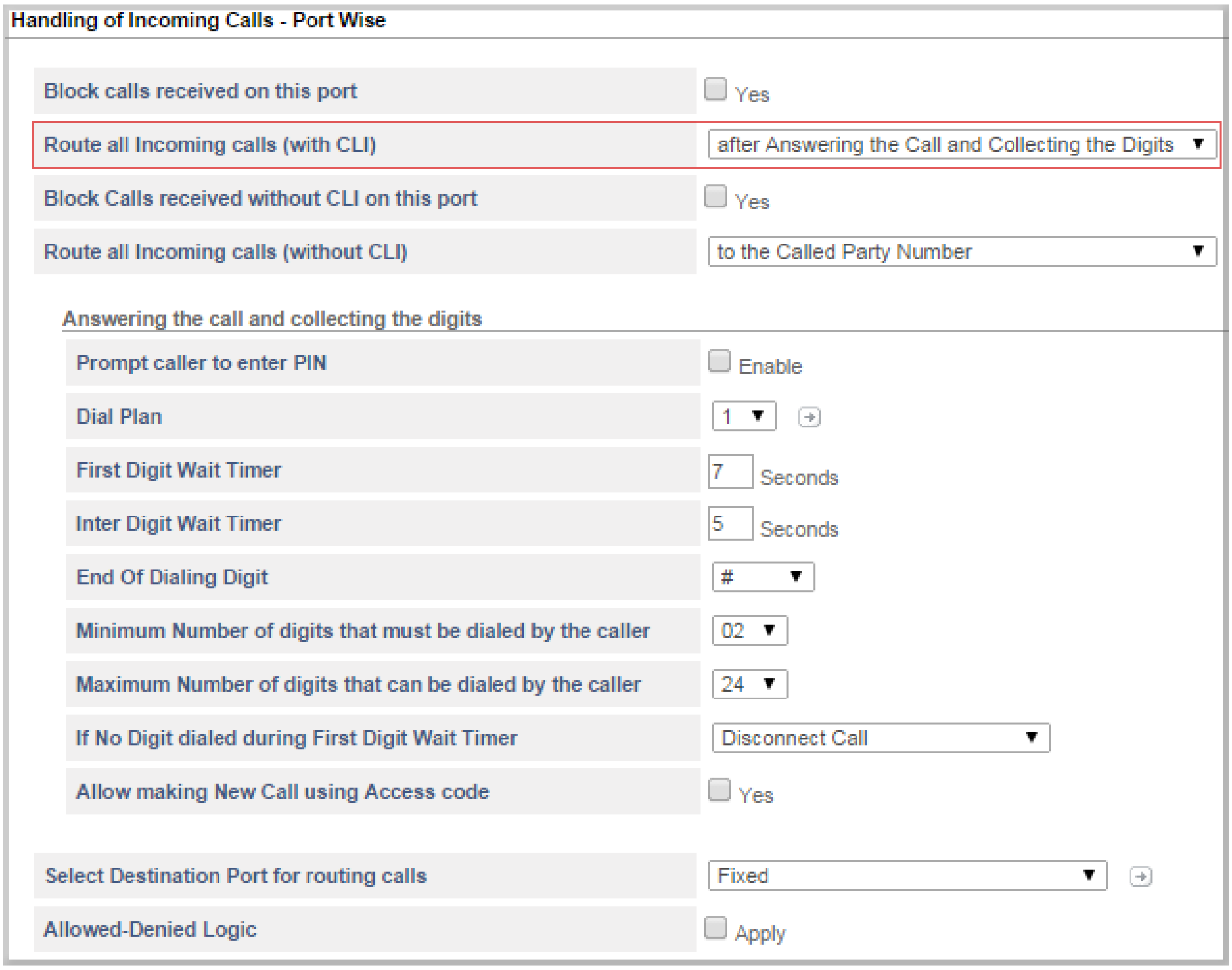
In this method, a call received on the T1E1 Port is routed to a specific number depending upon the called party number received in the SETUP Message on the T1E1 Port.

- To apply this method, in **Route all incoming calls (with CLI)**, select **to the Called Party Number**.



Route after Answering the Call and Collecting the Digits

In this method, the incoming call is answered and dial tone is played to the caller, allowing the caller to dial the desired number. The number dialed by the caller is considered as the destination number.



To apply this method, do the following:

- In **Route all Incoming calls (with CLI)**, select **after Answering the Call and Collecting the Digits**.

  The related parameters of this method appear under **Answering the call and collecting the digits**.

  - If you want to enable PIN Authentication on the T1E1 Port, select the **Prompt caller to enter PIN** check box.

    If you enable this check box, you must also configure the PIN Authentication Table. To know more about this feature and for detail instructions, see “PIN Authentication” under *Advanced Settings*.

  - SARVAM UMG supports 8 Dial Plans with total 64 entries in each table. When a user dials a number, it is compared with the Destination Number configured in the Dial Plan. If a match is found, the system routes the call immediately without waiting for End of Dialing and if a match is not found, the system will wait for the End of Dialing and then route the call as per the Destination Port Selection method configured.

    Select the **Dial Plan** table number you configured for this port. If you have not configured the Dial Plan table you may do so now,

    - Click **Settings** . The Dial Plan Table opens.

    - Configure the numbers in the table. For detailed instructions, see “Dial Plan”.

  - Set the duration of the **First Digit Wait Timer**. This is the duration for which you want the system to wait for the caller to dial the destination number after the dial tone. Valid range is 01 to 99 seconds. Default: 7 seconds

  - You may configure the following options as End of Dialing indication:

    - Set the duration of the **Inter Digit Wait Timer**. This is the duration for which you want the system to wait while receiving the digits dialed by the caller to consider it as End of Dialing. You may change this timer, if required. Valid range is 01 to 99 seconds. Default: 05 seconds.

    - In **End of Dialing Digit,** select # or * as termination digit the system should consider to detect end of dialing. Default: #

    - In **Minimum number of digits that can be dialed by the caller**, select the minimum number of digits to be dialed by the user for the system to consider it as a valid number. Valid range is 01 to 24 digits. Default: 2 digits.

    - In **Maximum Number of digits that can be dialled by the caller**, select the maximum number of digits to be dialed by the user for the system to consider it as End of Dialing. Valid range is 01 to 24 digits. Default: 24 digits.

      When the caller dials a number, the system will match it with the above End of Dialing indications and accept the one that matches first.

  - If the caller fails to dial the number during the First Digit Wait Timer, you can either have the system disconnect the call or route the call to a fixed destination number.

In the **If No Digit dialed during First Digit Wait Timer** box, select the desired option: **Disconnect the Call** or **Use Fixed Destination Number**. Default: Disconnect Call.

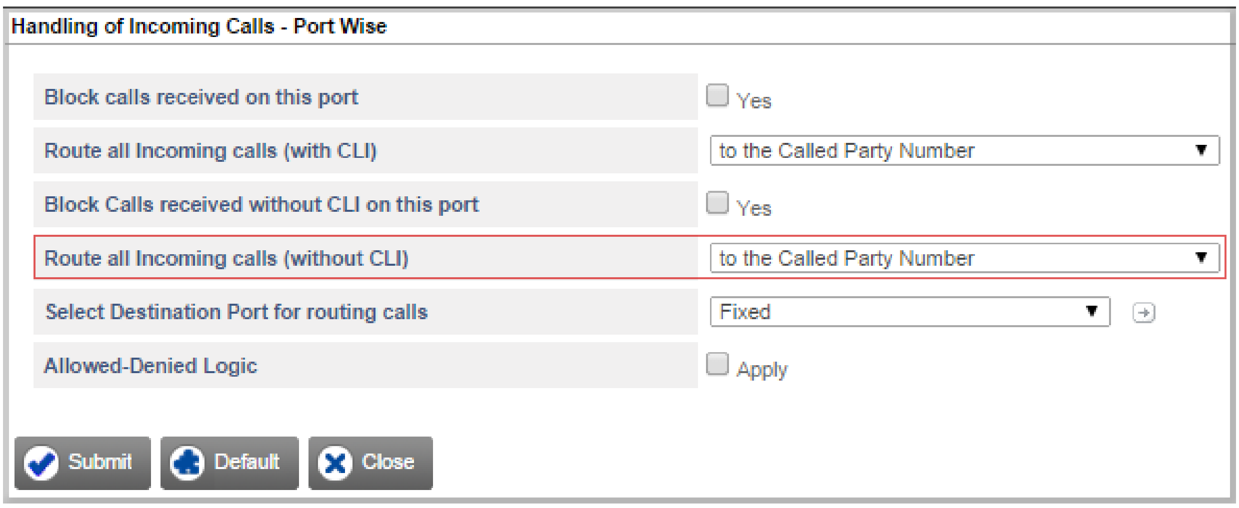
- If you selected **Use Fixed Destination Number**, enter the desired destination number in the **Fixed Destination Number** field. The Destination number may consist of a maximum of 24 digits. Valid digits are 0 to 9, *, # and . (dot/period). Default: Blank.

- *The First Digit Wait Timer is loaded as soon as the system answers the call.*
- *When you dial the first digit, the First Digit Wait Timer is stopped and the system loads the Inter Digit Wait Timer.*
- *SARVAM UMG reloads the Inter Digit Wait Timer:*
  - *each time you dial a new digit till the termination digit is detected.*
  - *when you have dialed the maximum number of digits configured as End of Dialing.*

- If you want to enable the feature Making New Call using Access Code on the T1E1 Port, select the **Allow making New Call using Access Code** check box. For further details, see "Making a New Call using Access Code".
- Click **Submit** to save settings.

- If you do not want to route the incoming calls received without CLI, through this T1E1 Port, select **Block Calls received without CLI on this Port** check box.
- To **Route all Incoming calls (without CLI),** you may select from any of the following methods:
  - to a Fixed Destination Number, see "Route to a Fixed Destination Number".
  - on the basis of DDI Number, see "Route on the basis of DDI Number".
  - to the Called Party Number, see "Route to the Called Party Number".
  - after Answering the Call and Collecting the Digits, see "Route after Answering the Call and Collecting the Digits".

  Default: to the Called Party Number.



Destination Port Determination

For the port/channel/MSN number, select the Destination Port for routing calls from the following options:

- Fixed
- On the basis of Destination Number
- On the basis of Calling Party Number

  Default: Fixed.

  Read the description and follow the instructions for each of these destination port selection methods given below.

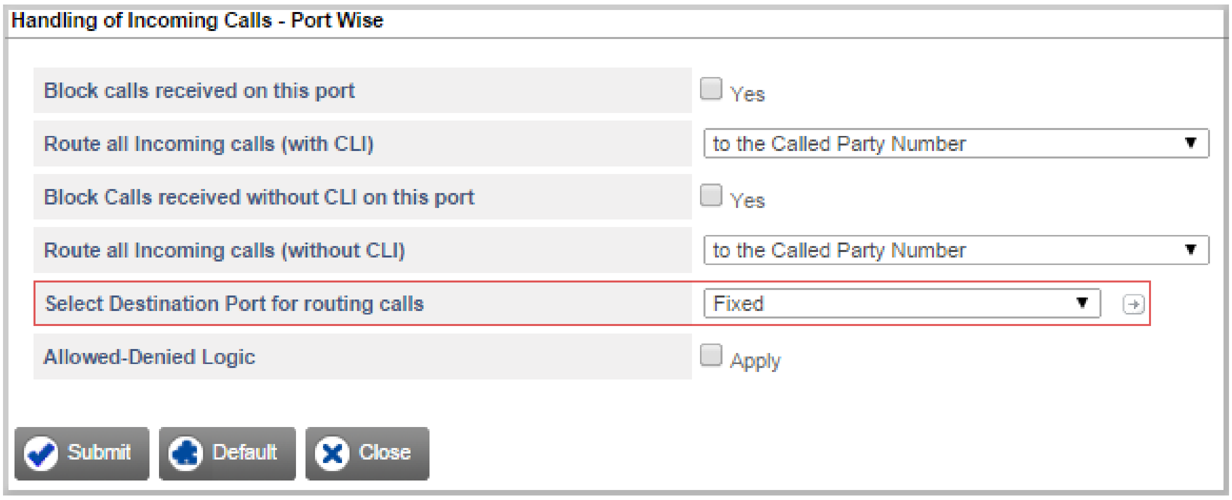
If the destination number to be dialed out is an IP Address, SARVAM UMG will not check the Destination Port Determination Method. Instead, it will route the call using the SIP Trunk / Group programmed for IP Dialing. (See “IP Dialing” to know more).

Fixed

In this method, calls received on the T1E1 Port are routed to a Fixed Destination Port, irrespective of the number dialed on the T1E1 Port.

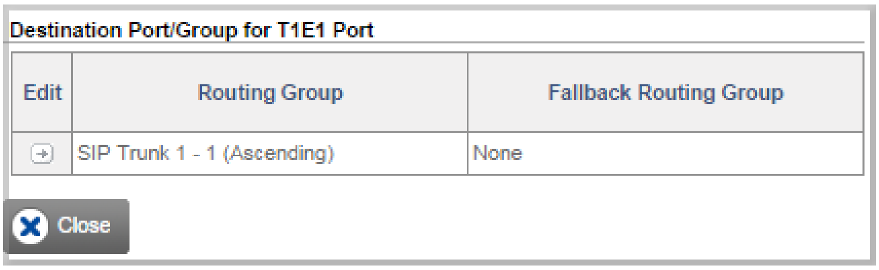
To apply this method, do the following:

- In **Select Destination Port for routing calls**, select **Fixed** option.



- Click **Settings** .

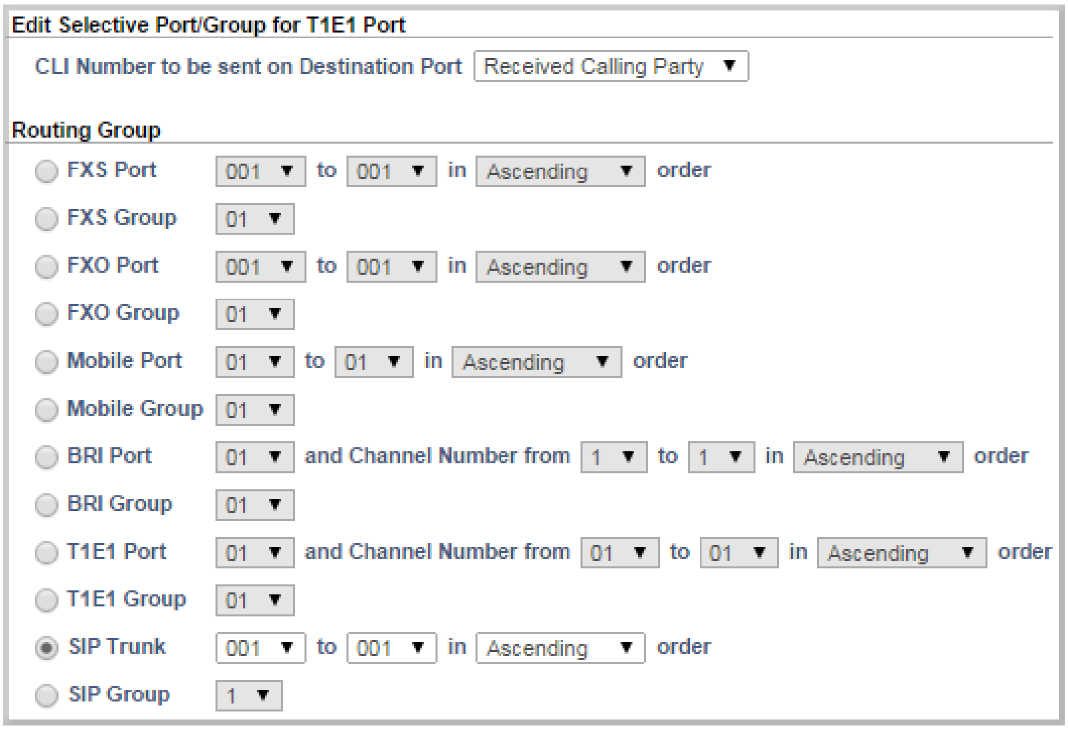
  The **Destination Port/Group for T1E1 Port** window opens.



The default **Routing Group** and **Fallback Routing Groups** appear.

- If you wish to change the default Routing Group options, click **Edit** .

  The **Edit Selective Port/Group for T1E1 Port** window opens.

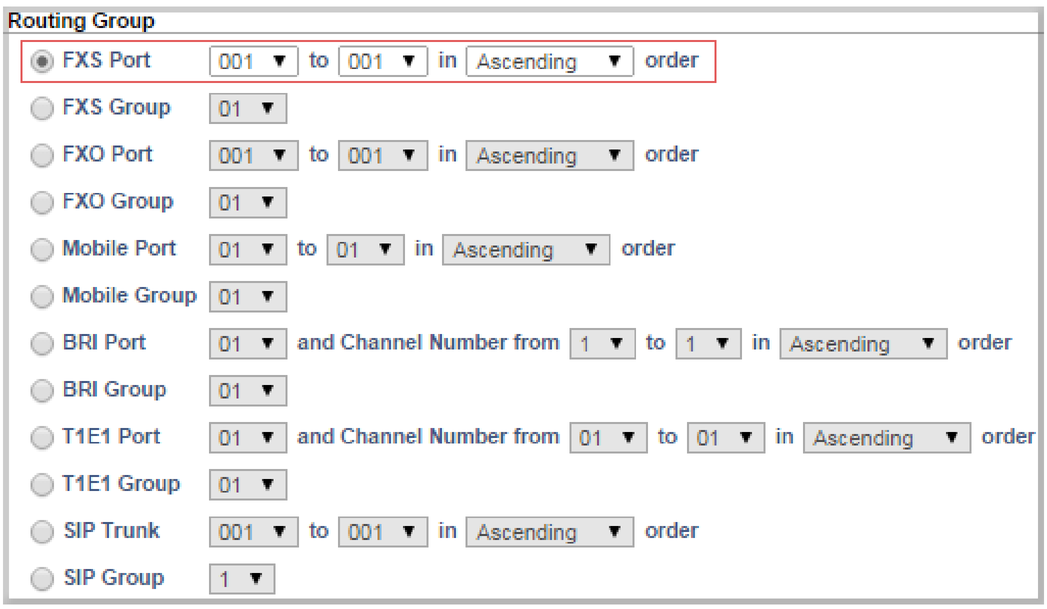


- Select the **CLI Number to be sent on Destination Port**. You can select Received Calling Party or Received Called Party. Default: Received Calling Party.

CLI to be sent on Destination Port is applicable when FXS Port/Group is determined as the Destination Port.

- Create the **Routing Group**.

  - To create a group of *sequential* **FXS Ports** as members,



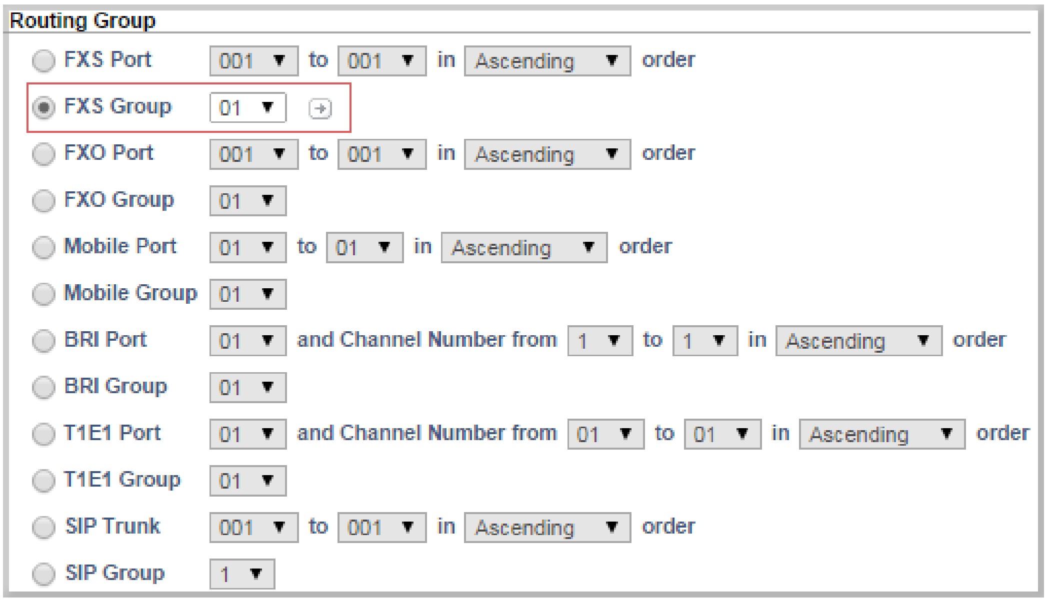
- Select the desired **FXS Port** numbers as members. Default: 1.

- In **in - order**, select the order in which the system should hunt for a free member FXS Port to route the call.

  Select **Ascending** to start hunting from the first to the last member FXS Port. Select **Descending** to start hunting from the last to the first member FXS Port. Default: Ascending.

- To create a group of *not-sequential* **FXS Ports** as members,

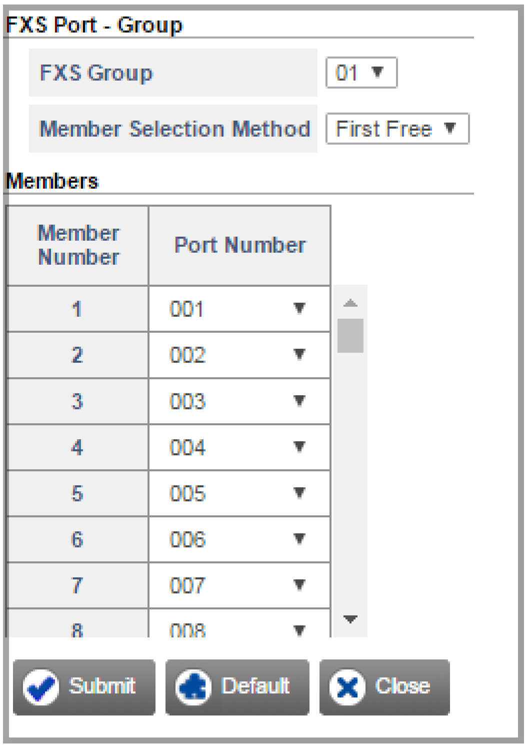
  - Select a **FXS Group**.



  - Select **FXS Group** number. Default: 1.

  - Click **Settings** .

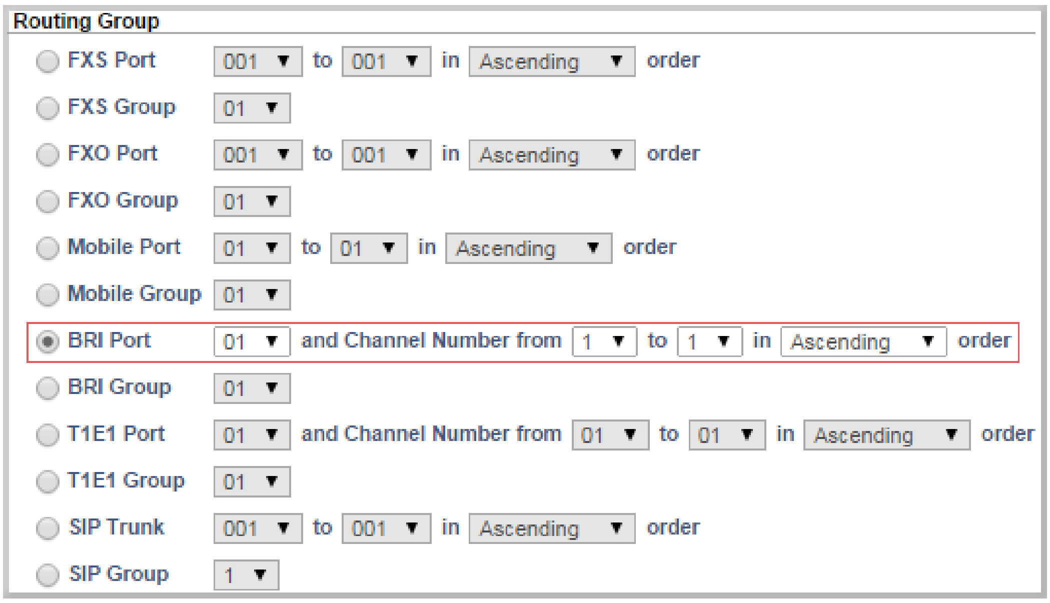
- The **FXS Port - Groups** window opens.



- Create the FXS Group. For detailed instructions on creating groups, see the topic “Group” under *Advanced Settings*.

- Similarly, you can create a group of *sequential* and *not-sequential* FXO Ports, Mobile Ports and SIP Trunks.

- To create a routing group of *sequential* **BRI Channels** as members,

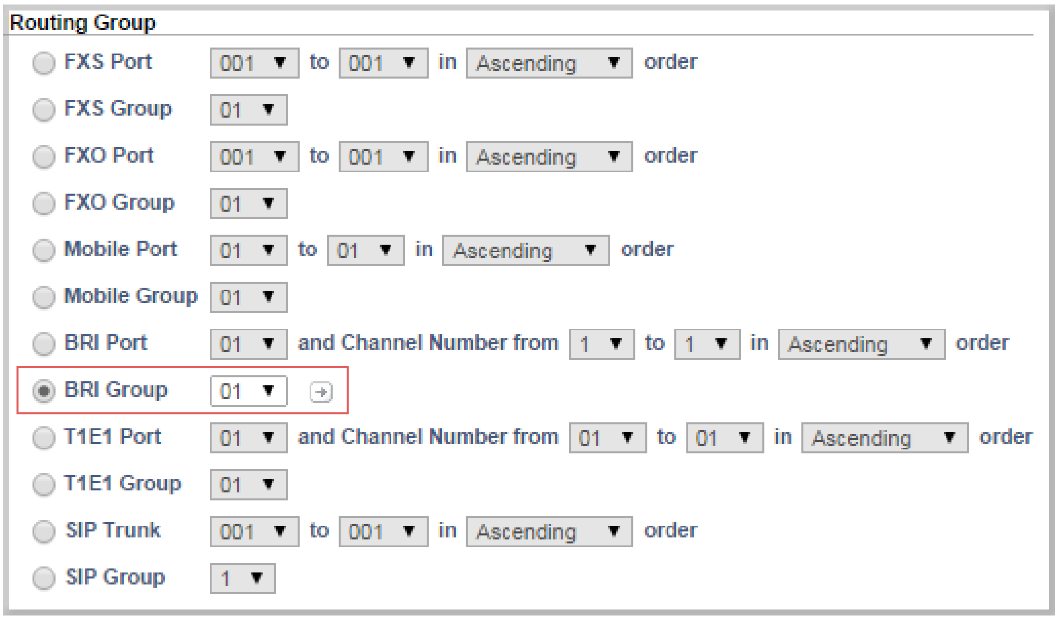


- Select the **BRI Port** Number. Default: 1.

- In Channel Number **From - to**, select the **Start Channel Number** and the **End Channel Number**, respectively.
- In **in - order**, select the order in which the system should hunt for a free member Channel to route the call.

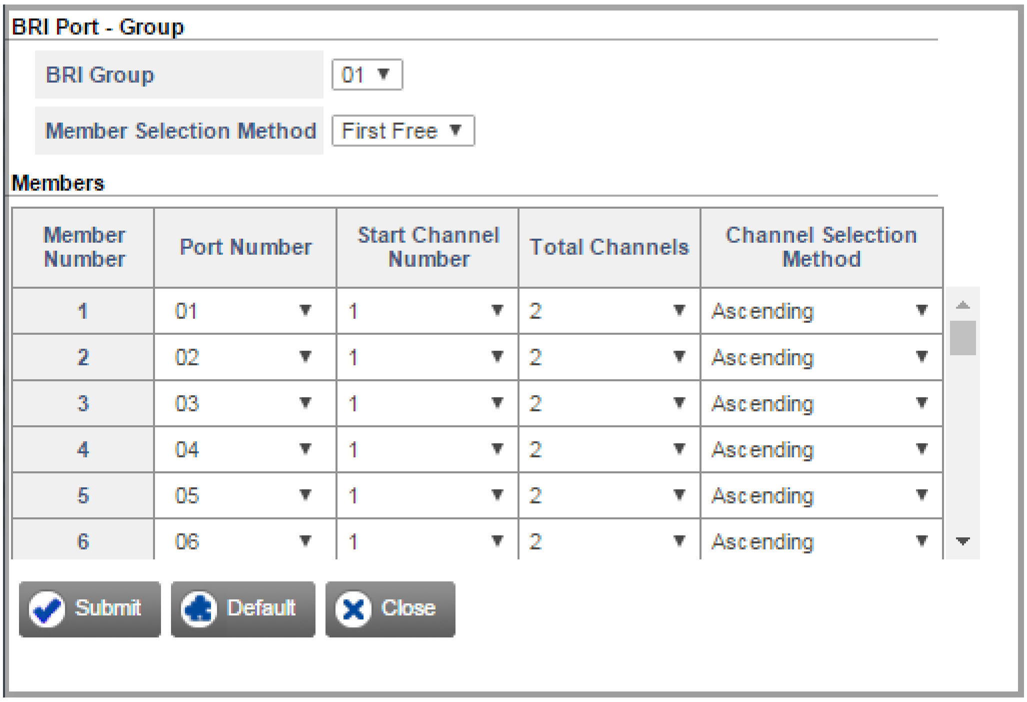
  Select **Ascending** to start hunting from the first to the last member channel. Select **Descending** to start hunting from the last to the first member channel. Default: Ascending.

- To create a group of *not-sequential* **BRI Channels** as members,
  - Select **BRI Group**.



  - Select a **BRI Group** number. Default: 1.
  - Click **Settings**.

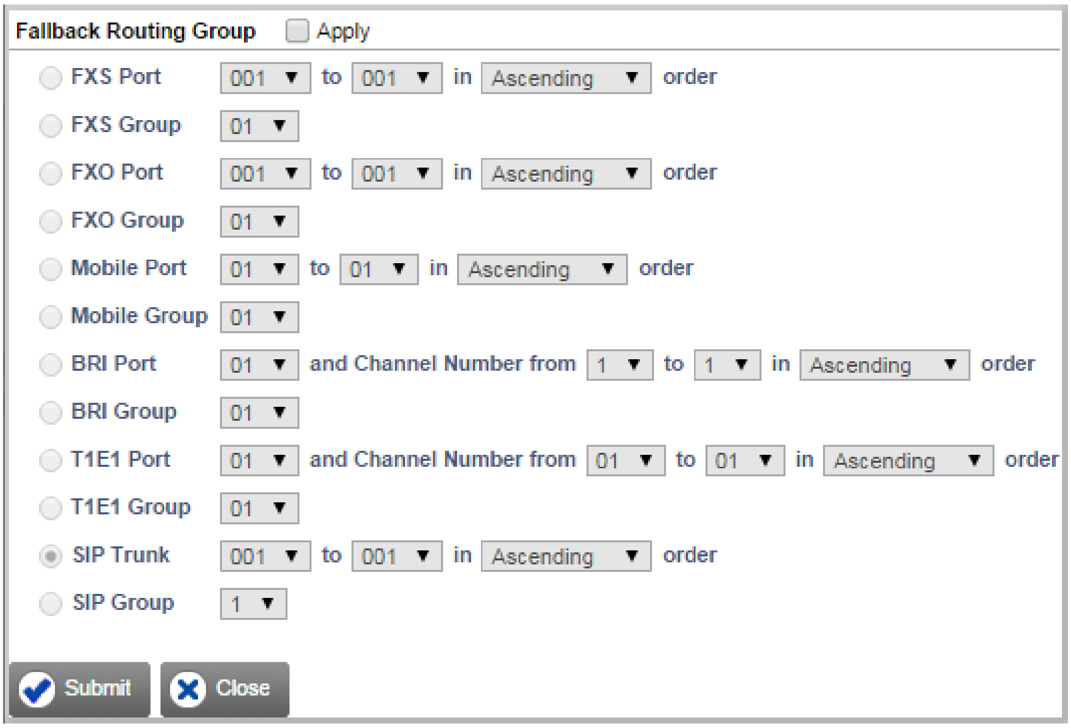
- The **BRI Port - Groups** window opens.



- Create the BRI Group. For detailed instructions on creating groups, see the topic "Group" under *Advanced Settings.*

- Similarly, you can create a group of *sequential* and *not-sequential* T1E1 Ports.

- You may create the **Fallback Routing Group**.



- To do this,
  - Select the **Apply** check box.
  - Follow the same instructions provided earlier for creating *sequential* and *not-sequential* groups of FXS Ports, FXO Ports, Mobile Ports, BRI Ports, T1E1 Ports and SIP Trunks.
- Click **Submit** to save changes. The **Edit** window closes.
- The entry you edited appears in the **Destination Port/Group for T1E1 Port** window.
- Close the **Destination Port/Group for T1E1 Port** window to return to the **Handling of Calls** window.

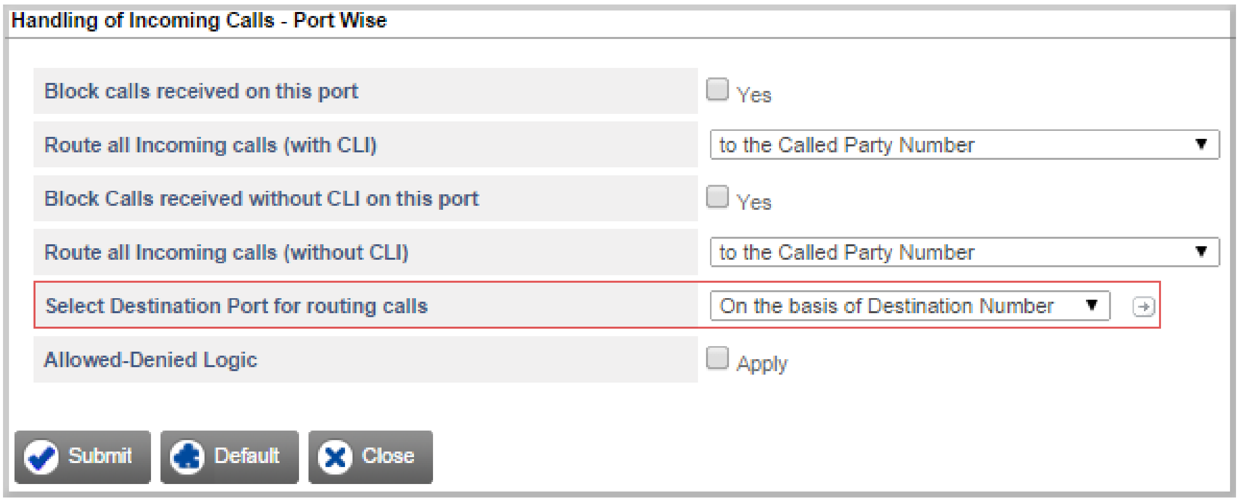
On the basis of Destination Number

In this method, incoming calls on the source port are routed to the destination port on the basis of the destination number (called party number) dialed by the caller.

You must configure the called party numbers in the **Destination Number Based** Table. SARVAM UMG will match the called party number dialed by the caller with the entries of this table. If a match is found for the number in the table, the call is routed to the destination.

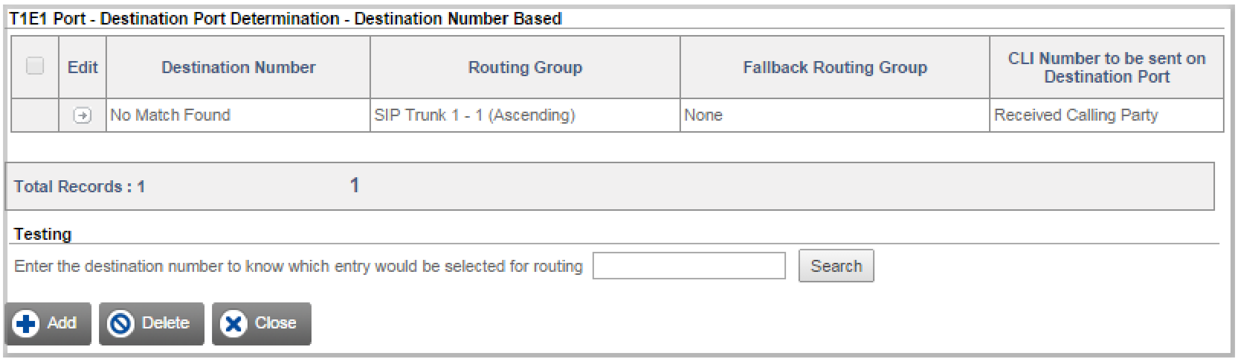
To apply this method, do the following:

- In **Select Destination Port for routing calls**, select **On the basis of Destination Number** option.

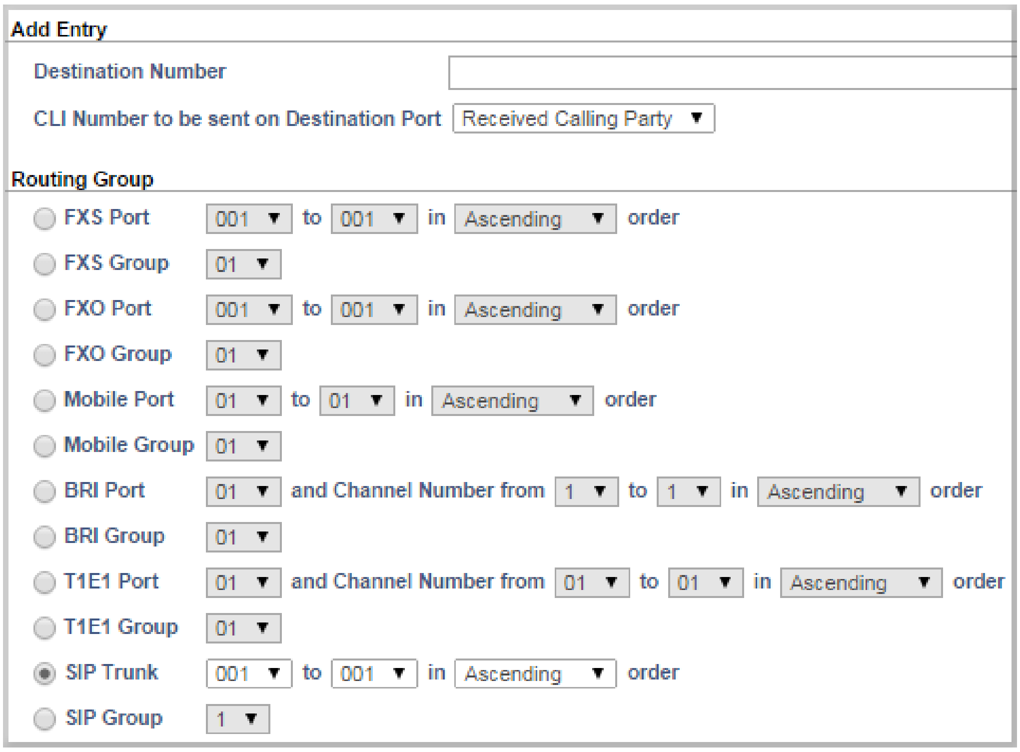


- Click **Settings** .

The **T1E1 Port - Destination Port Determination - Destination Number Based** table window opens.



- To add a new entry, click **Add**. The **Add Entry** window opens. You can add upto 1000 entries.



- In **Destination Number**, enter the number you expect the callers to dial. You may enter upto 64 characters (Digits + "Wildcard Characters") in this field. Valid characters are 0 to 9, *, #, X, T, Comma [,], Hyphen [-], Caret [^]. Default: Blank.

Wildcard Characters

SARVAM UMG supports following characters.

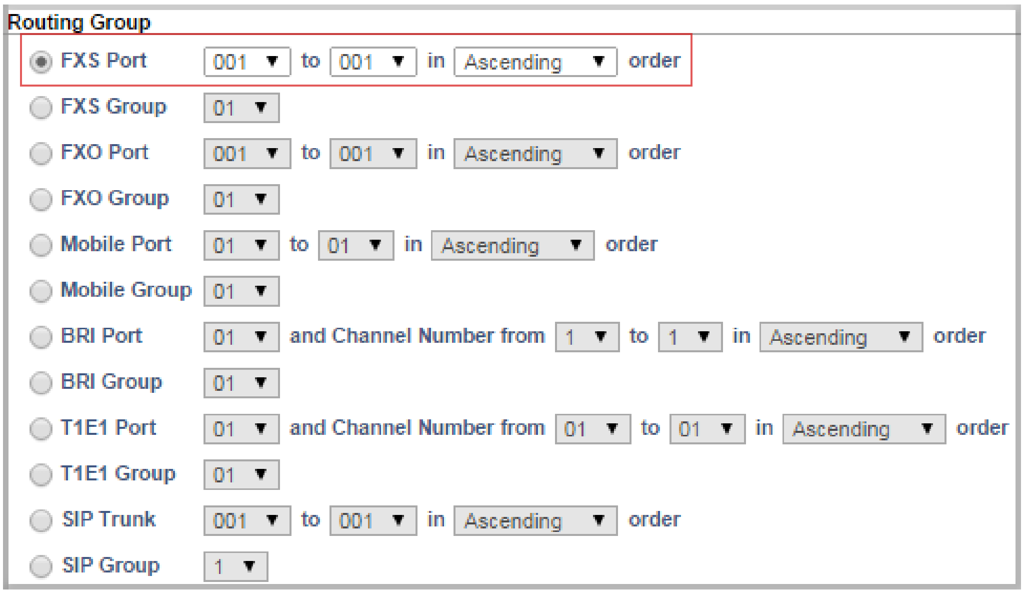
Character	Description
X (letter X)	X represents any single digit from 0 to 9.
#	When # is configured in a number string, it will not be considered as End of Dialing.
*****	When * is configured in a number string, it will not be considered as End of Dialing.

+	+ (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.

- Select the **CLI Number to be sent on Destination Port**. You can select Received Calling Party or Received Called Party. Default: Received Calling Party.

CLI to be sent on Destination Port is applicable when FXS Port/Group is determined as the Destination Port.

- Create the **Routing Group**.
    - To create a group of *sequential* **FXS Ports** as members,

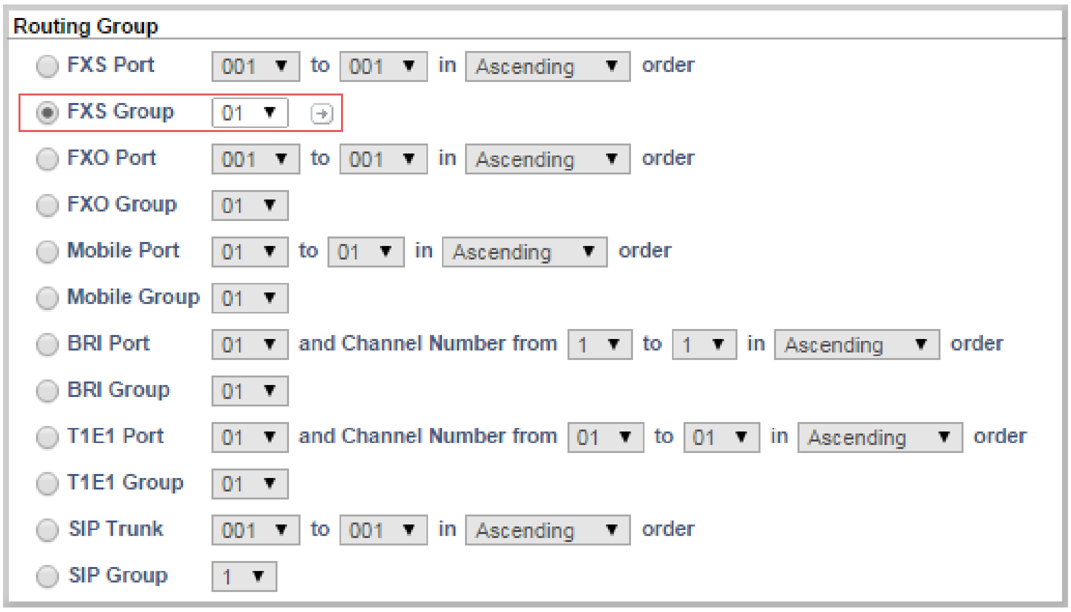


- Select the desired **FXS Port** numbers as members. Default: 1.
- In **in - order**, select the order in which the system should hunt for a free member FXS Port to route the call.

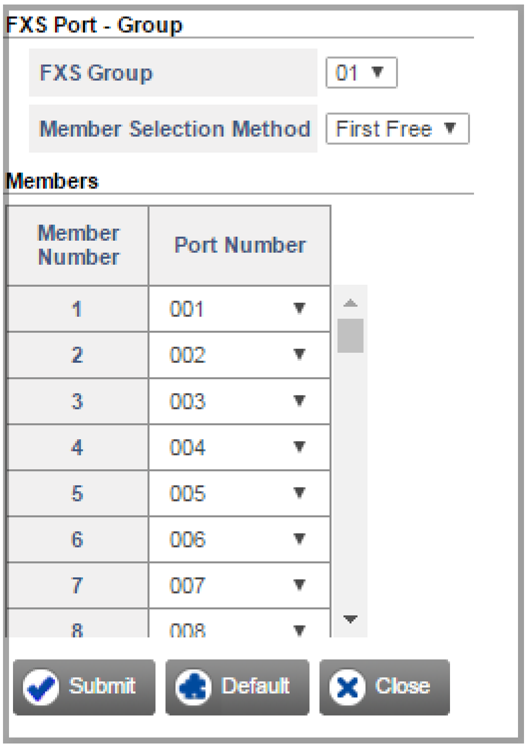
  Select **Ascending** to start hunting from the first to the last member FXS Port. Select **Descending** to start hunting from the last to the first member FXS Port. Default: Ascending.

- To create a group of *not-sequential* **FXS Ports** as members,

- Select a **FXS Group**.



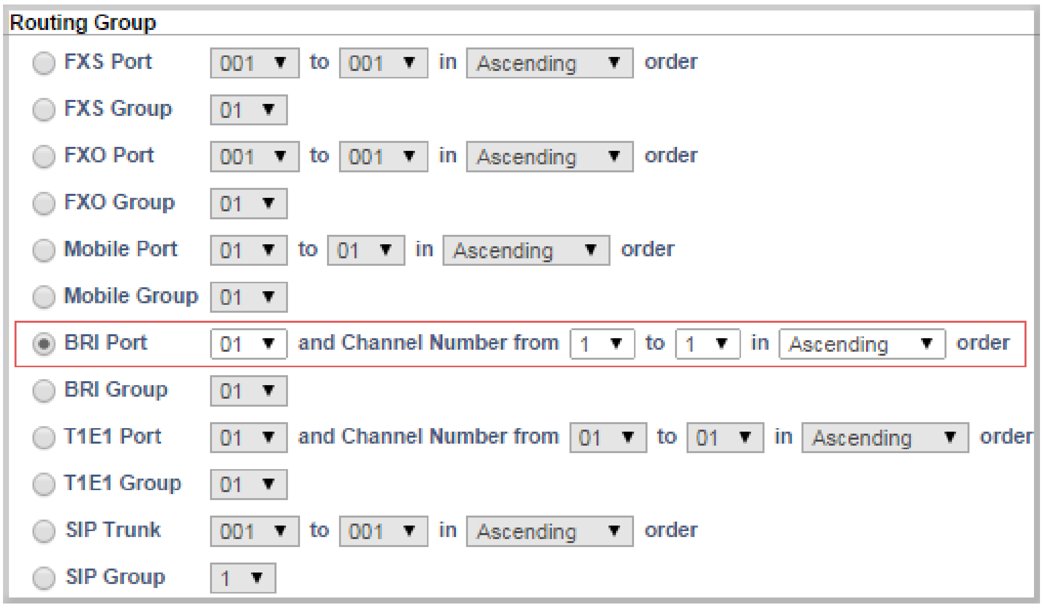
- Select **FXS Group** number. Default: 1.
- Click **Settings** .
- The **FXS Port - Groups** window opens.



- Create the FXS Group. For detailed instructions on creating groups, see the topic “Group” under *Advanced Settings*.

- Similarly, you can create a group of *sequential* and *not-sequential* FXO Ports, Mobile Ports and SIP Trunks.

- To create a routing group of *sequential* **BRI Channels** as members,



  - Select the **BRI Port** Number. Default: 1.

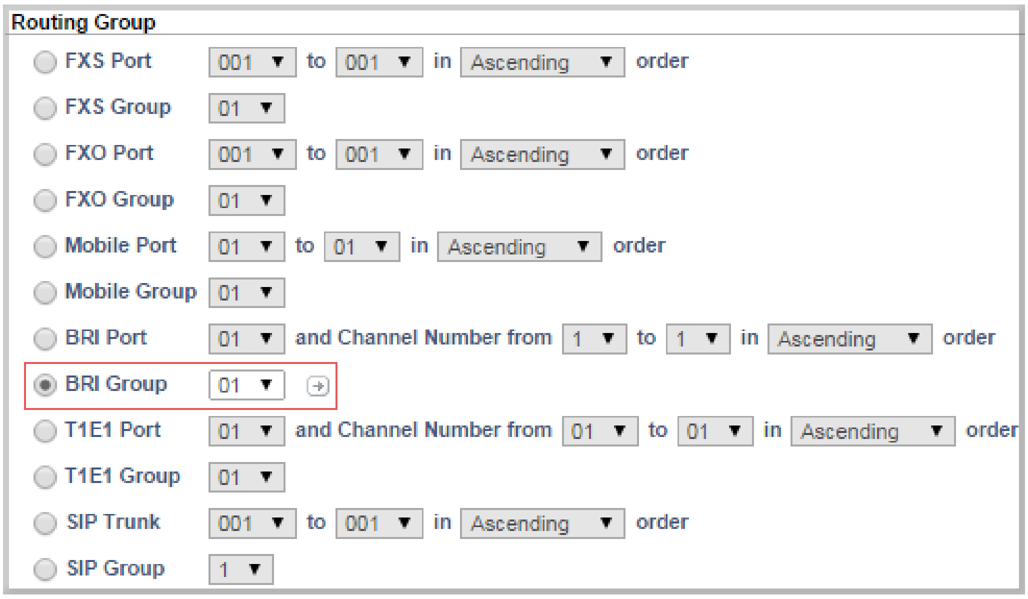
  - In Channel Number **From - to**, select the **Start Channel Number** and the **End Channel Number**, respectively.

  - In **in - order**, select the order in which the system should hunt for a free member Channel to route the call.

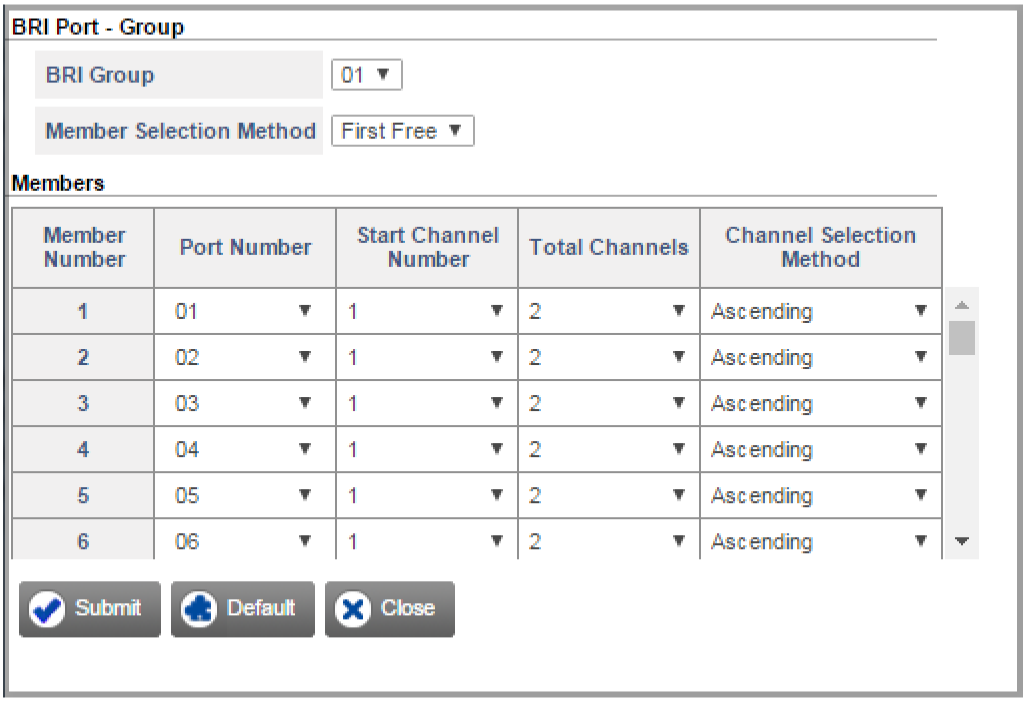
    Select **Ascending** to start hunting from the first to the last member channel. Select **Descending** to start hunting from the last to the first member channel. Default: Ascending.

- To create a group of *not-sequential* **BRI Channels** as members,

- Select **BRI Group**.



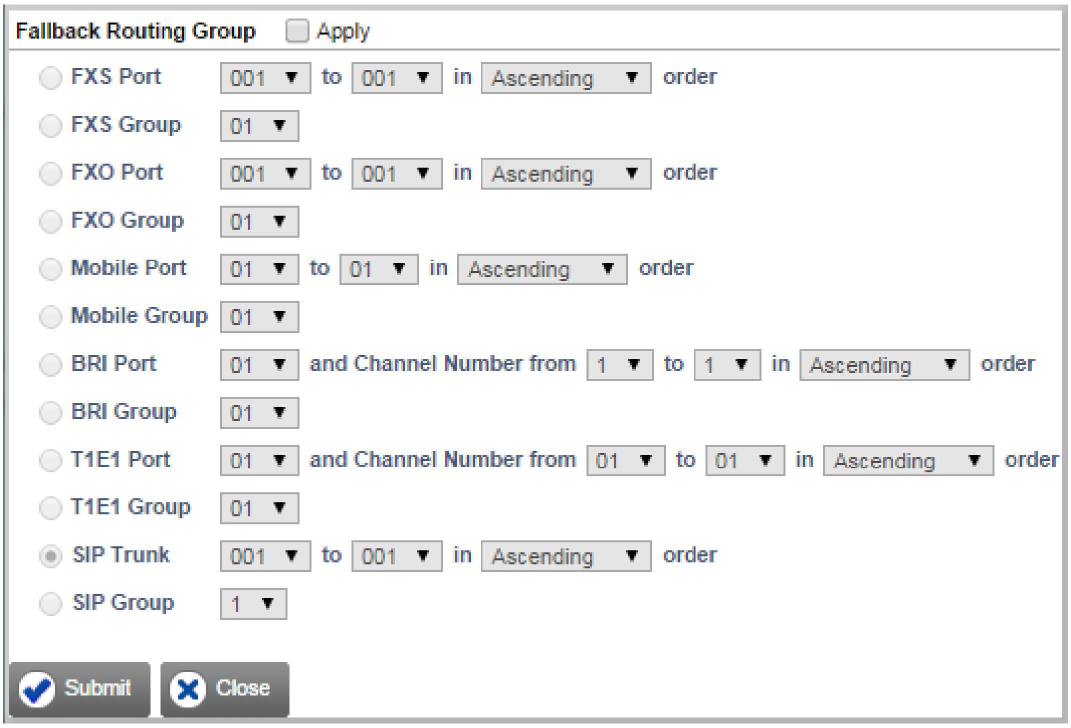
- Select a **BRI Group** number. Default: 1.
- Click **Settings** .
- The **BRI Port - Groups** window opens.



- Create the BRI Group. For detailed instructions on creating groups, see the topic “Group” under *Advanced Settings.*

- Similarly, you can create a group of *sequential* and *not-sequential* T1E1 Ports.

- You may create the **Fallback Routing Group**.



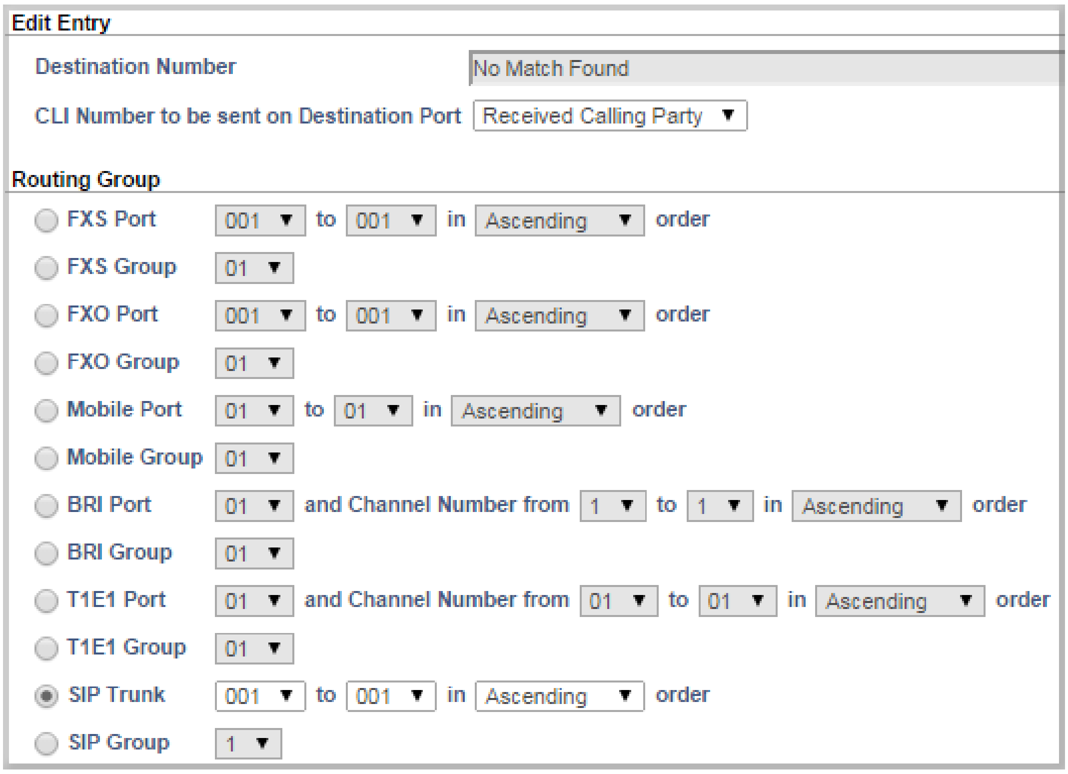
- To do this,
  - Select the **Apply** check box.
  - Follow the same instructions provided earlier for creating *sequential* and *not-sequential* groups of FXS Ports, FXO Ports, Mobile Ports, BRI Ports, T1E1 Ports and SIP Trunks.
- Click **Submit** to save changes. The **Add Entry** window closes.
- The entry you added appears in the **T1E1 Port - Destination Port Determination - Destination Number Based** table.
- Follow the same steps as above to add another entry to this table.
- To delete an entry, select the check box and click the **Delete** button.

*If there are multiple entries in the Destination Number Based table, to search a particular entry in the table, under Testing enter the desired number in the **Enter the destination number to know which entry would be selected for routing** search box.*

- By default, SIP Trunk 1-1 (Ascending) is assigned as the Routing Group, for routing calls from numbers that do not match with any of the destination numbers you configured (No Match Found).

  To change the default Routing Group and to create the Fallback Routing Group for the No Match Found numbers entry,

- For the No Match Found entry in the table, click **Edit**.



- The **Edit Entry** window opens.
- Create the **Routing Group** and **Fallback Routing Group** as per your requirement.
- Click **Submit** and close the window.

- Close the window if you have finished adding/editing entries.

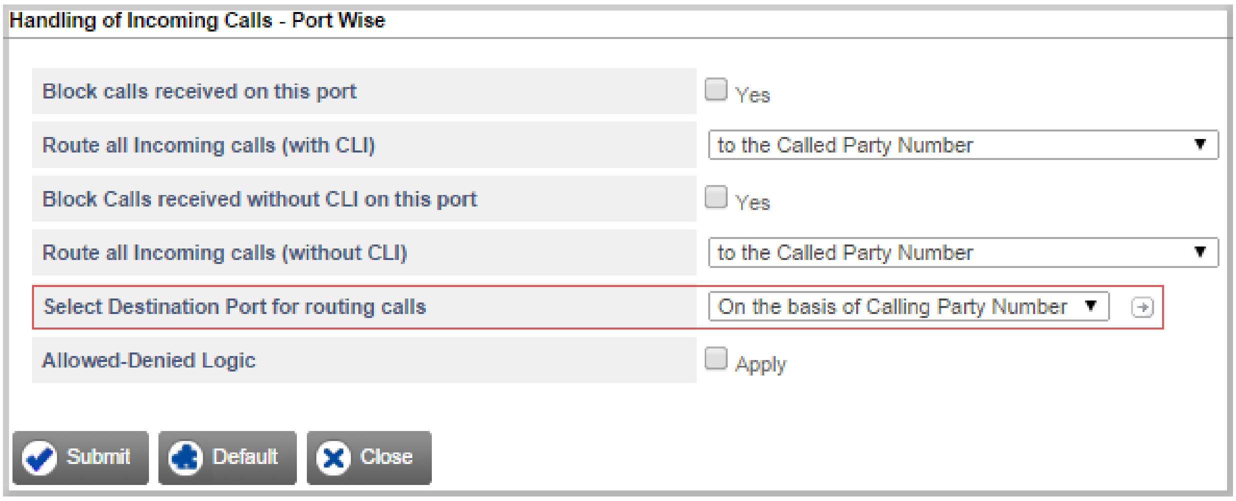
  You can also configure the **Destination Number Based** Table from *Advanced Settings*. For instructions, see "Destination Port Determination" under *Advanced Settings.*

On the basis of Calling Party Number

In this method, incoming calls on the T1E1 Port are routed to a specific port as per the calling party's number.

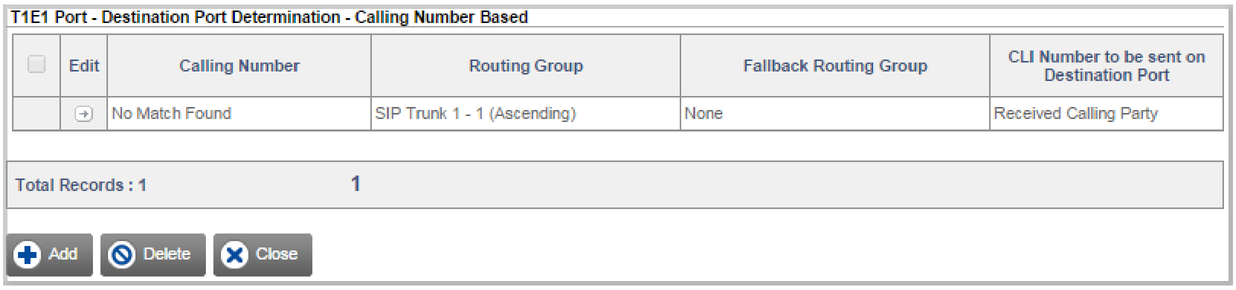
To apply this method, do the following:

- In **Select Destination Port for routing calls**, select **On the basis of Calling Party Number** option.

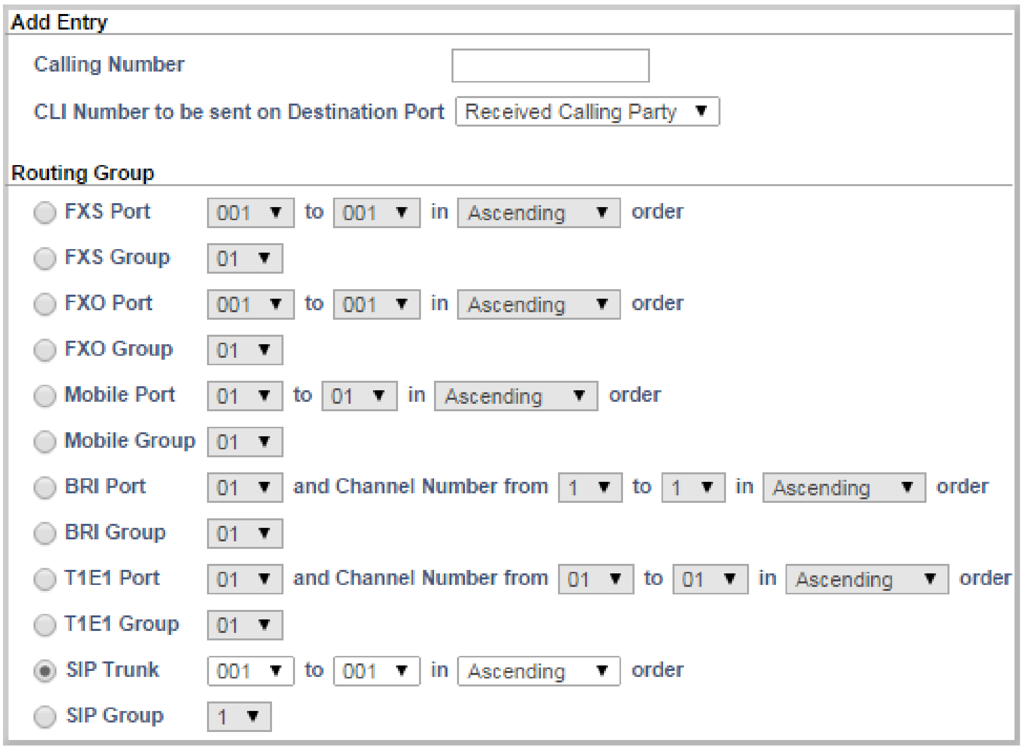


- Click **Settings** .

The **T1E1 Port - Destination Port Determination - Calling Number Based** table window opens.



- To add a new entry, click **Add**. The **Add Entry** window opens. You can add upto 499 entries.



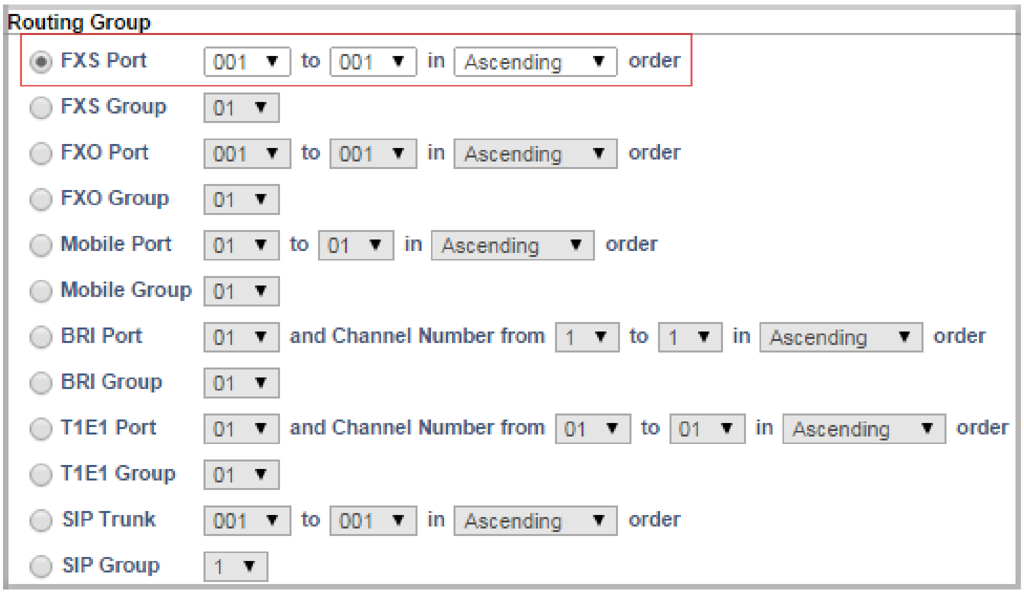
- In **Calling Number**, enter the number (max. 24 characters) from which you expect calls to be received. Valid digits are 0 to 9, *, #, (dot). Default: Blank.

- Select the **CLI Number to be sent on Destination Port**. You can select Received Calling Party or Received Called Party. Default: Received Calling Party.

CLI to be sent on Destination Port is applicable when FXS Port/Group is determined as the Destination Port.

- Create the **Routing Group**.

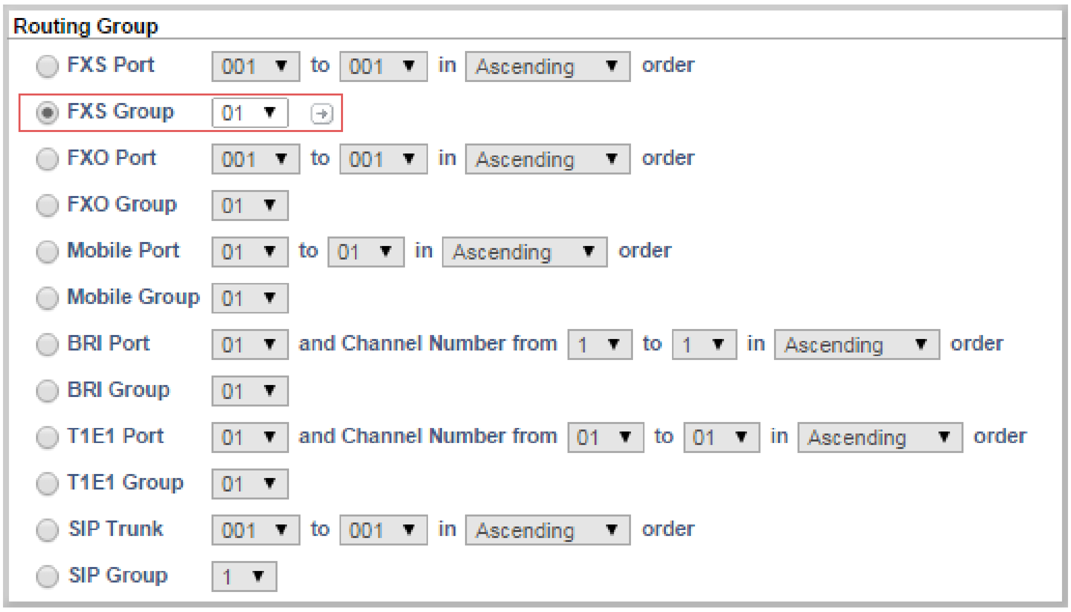
- To create a group of *sequential* **FXS Ports** as members,



  - Select the desired **FXS Port** numbers as members. Default: 1.
  - In **in - order**, select the order in which the system should hunt for a free member FXS Port to route the call.

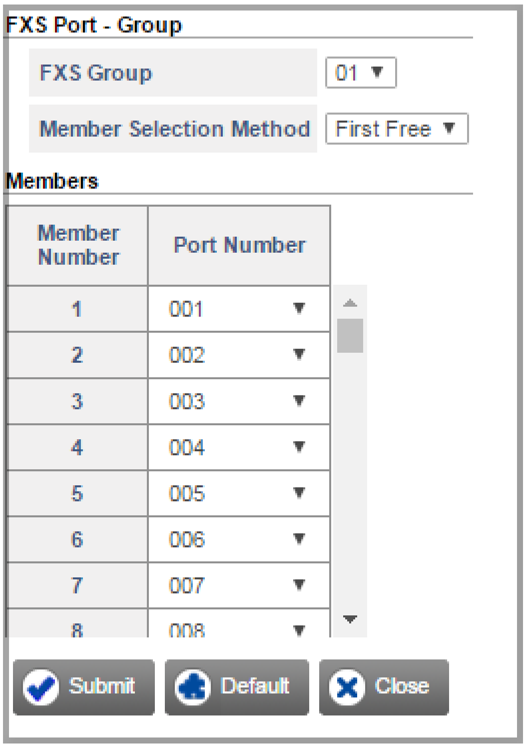
    Select **Ascending** to start hunting from the first to the last member FXS Port. Select **Descending** to start hunting from the last to the first member FXS Port. Default: Ascending.

- To create a group of *not-sequential* **FXS Ports** as members,
  - Select a **FXS Group**.



  - Select **FXS Group** number. Default: 1.

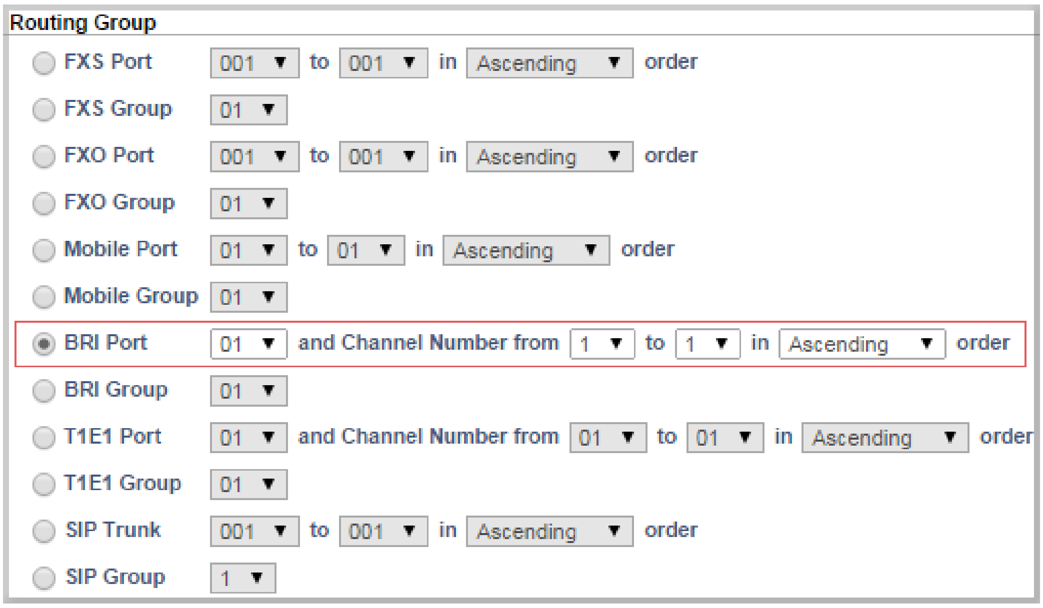
- Click **Settings** .
- The **FXS Port - Groups** window opens.



- Create the FXS Group. For detailed instructions on creating groups, see the topic “Group” under *Advanced Settings*.

- Similarly, you can create a group of *sequential* and *not-sequential* FXO Ports, Mobile Ports and SIP Trunks.

- To create a routing group of *sequential* **BRI Channels** as members,

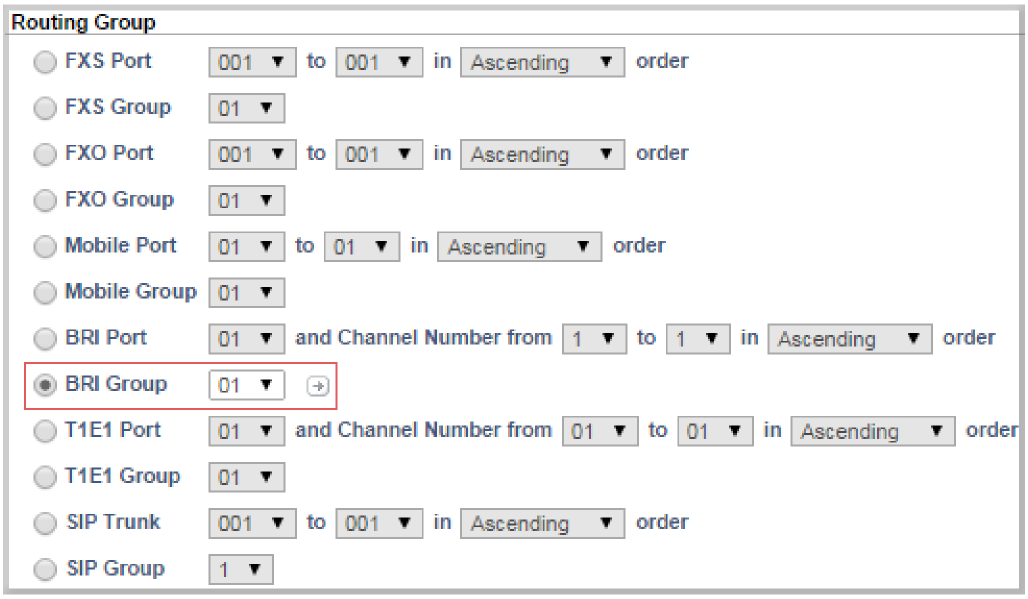


  - Select the **BRI Port** Number. Default: 1.
  - In Channel Number **From - to**, select the **Start Channel Number** and the **End Channel Number**, respectively.
  - In **in - order**, select the order in which the system should hunt for a free member Channel to route the call.

    Select **Ascending** to start hunting from the first to the last member channel. Select **Descending** to start hunting from the last to the first member channel. Default: Ascending.

- To create a group of *not-sequential* **BRI Channels** as members,

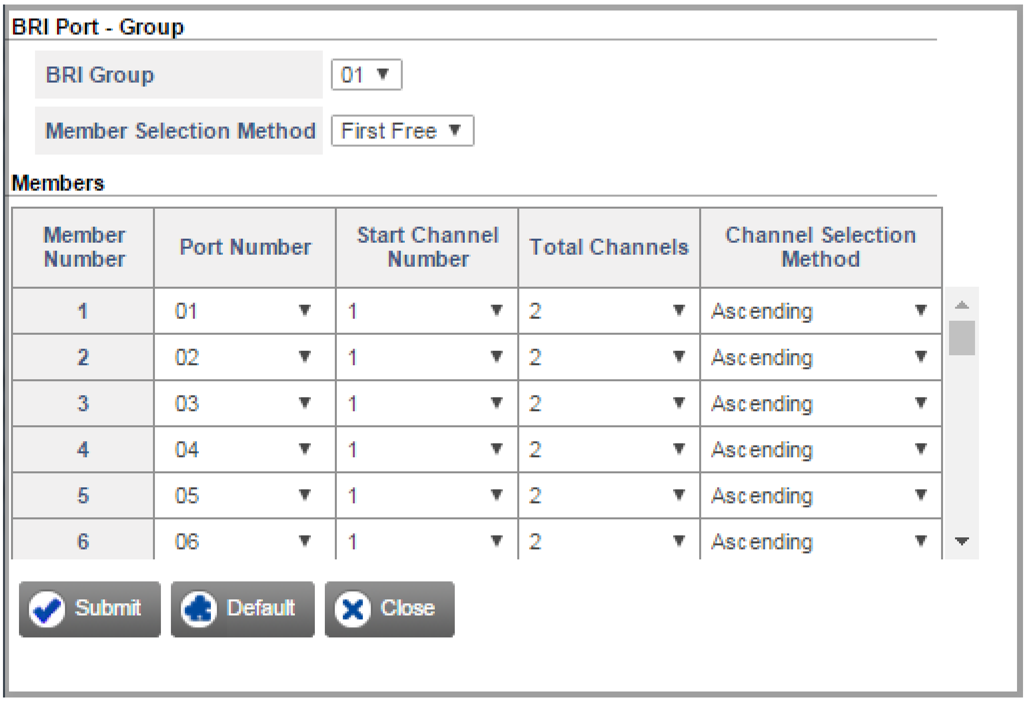
- Select **BRI Group**.



- Select a **BRI Group** number. Default: 1.

- Click **Settings** .

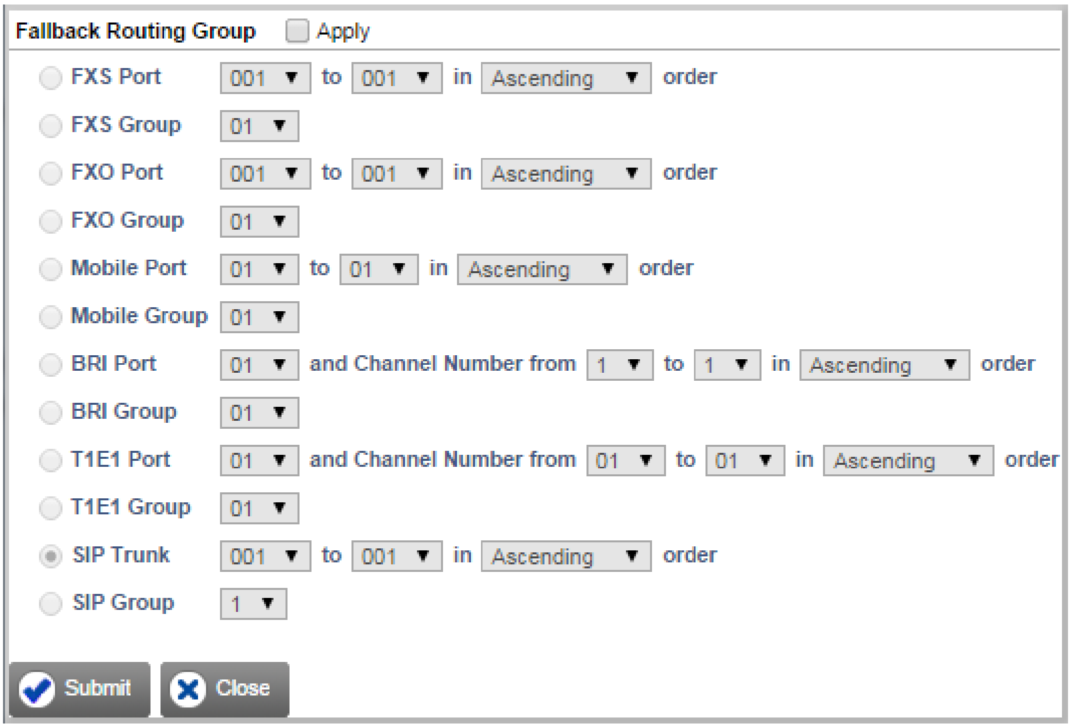
- The **BRI Port - Groups** window opens.



- Create the BRI Group. For detailed instructions on creating groups, see the topic “Group” under *Advanced Settings.*

- Similarly, you can create a group of *sequential* and *not-sequential* T1E1 Ports.

- You may create the **Fallback Routing Group**.



- To do this,
  - Select the **Apply** check box.
  - Follow the same instructions provided earlier for creating *sequential* and *not-sequential* groups of FXS Ports, FXO Ports, Mobile Ports, BRI Ports, T1E1 Ports and SIP Trunks.
- Click **Submit** to save changes. The **Add Entry** window closes.
- The entry you added appears in the **T1E1 Port - Destination Port Determination - Calling Number Based** table.
- Follow the same steps as above to add another entry to this table.
- To delete an entry, select the check box and click the **Delete** button.
- By default, SIP Trunk 1-1 (Ascending) is assigned as the Routing Group, for routing calls from numbers that do not match with any of the destination numbers you configured (No Match Found).

  To change the default Routing Group and to create the Fallback Routing Group for the No Match Found numbers entry,

- For the No Match Found entry in the table, click **Edit**.

T1E1 Port - Destination Port Determination - Calling Number Based

	Edit	Calling Number	Routing Group	Fallback Routing Group	CLI Number to be sent on Destination Port
		No Match Found	SIP Trunk 1 - 1 (Ascending)	None	Received Calling Party

Total Records : 1 1

Add Delete Close

- The **Edit Entry** window opens.
- Create the **Routing Group** and **Fallback Routing Group** as per your requirement.
- Click **Submit** and close the window.

- Close the window if you have finished adding/editing entries.

  You can also configure the **Calling Number Based** Table from *Advanced Settings*. For instructions, see "Destination Port Determination" under *Advanced Settings.*

Allowed - Denied Logic

You can apply the Allowed-Denied logic on the T1E1 Port (source port) if you want to allow or restrict the dialing of particular numbers. You can use this feature for Toll Control.

The Allowed-Denied Number Logic makes use of two Number lists:

- **Allowed Numbers List:** This is the list of numbers that can be dialed out from the T1E1 Port.
- **Denied Numbers List:** This list contains the numbers that are to be restricted from being dialed out from the T1E1 Port.

When Allowed-Denied Logic is enabled on a source port, for each number dialed from the port, SARVAM UMG uses the best-match-found logic to compare the dialed number with the Allowed Number list and the Denied Number list.

The number is allowed to be dialed, if it:
- matches with both lists.
- matches with Allowed Number list, but not with the Denied Number list.
- matches with neither the Allowed List nor the Denied List.

The number is denied, if it matches with the Denied Number list, but not with the Allowed Number list.

The system does not apply the Allowed-Denied Logic:
- When dialed number string matches with any Access Code.
- When dialed number string matches with any Emergency Number.
- When any one of the following is selected to Route all Incoming Calls (with CLI):
  - on the basis of Calling Party Number
  - to a Fixed Destination Number
  - on the basis of DDI Number

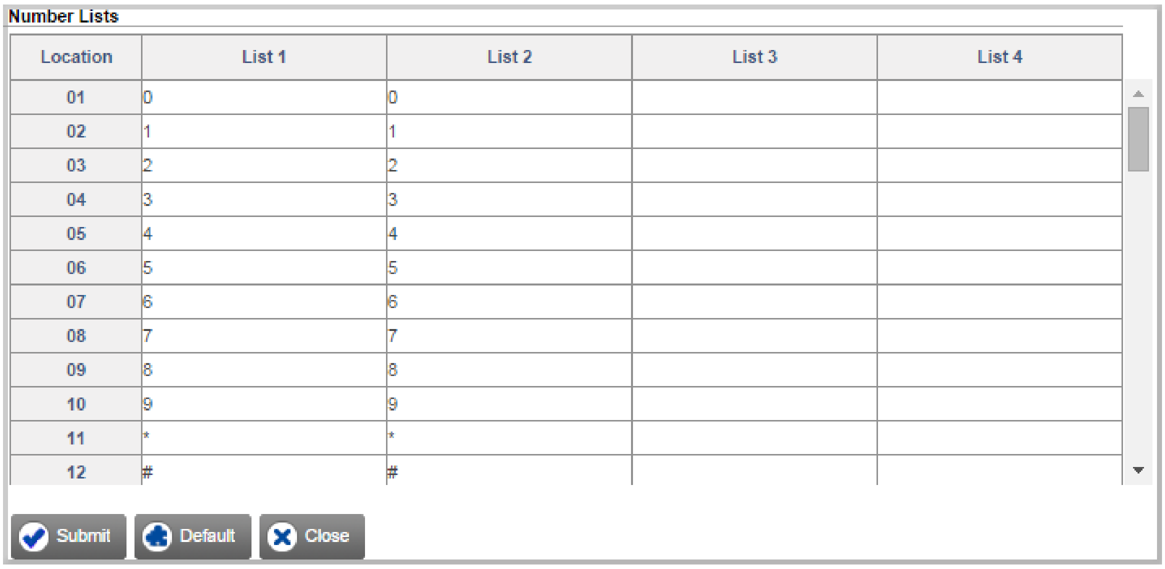
To apply Allowed - Denied Logic on the T1E1 Port,

- Select the **Allowed - Denied Logic** check box.

- In the **Allowed Number List**, select the list number you have configured with numbers you want to allow to be dialed out from the T1E1 Port. Default: 01

  If you have not configured the Allowed Number List,

  - Click **Settings** .
  - The **Number Lists** window opens.


Number Lists

Location	List 1	List 2	List 3	List 4
01	0	0		
02	1	1		
03	2	2		
04	3	3		
05	4	4		
06	5	5		
07	6	6		
08	7	7		
09	8	8		
10	9	9		
11	*	*		
12	#	#		

Submit Default Close

  - You may configure the default Allowed Number List 1 or any other list. See “Number Lists” to configure the allowed numbers.
  - Click **Submit** to save the Allowed Number List and close the window.

- In the **Denied Number List**, select the list number you have configured with numbers you want to restrict to be dialed out from the T1E1 Port. Default: 02

  If you have not configured the Denied Number List,

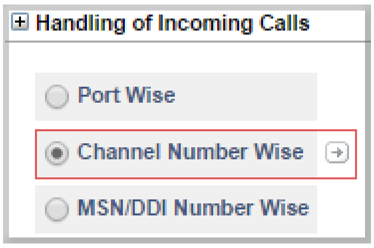
  - Click **Settings** . **The Number Lists** window opens.
  - You may configure the default Denied Number List 2 or any other list. See “Number Lists” to configure the restrict numbers.

- Click **Submit** to save the Denied Number List and close the window.

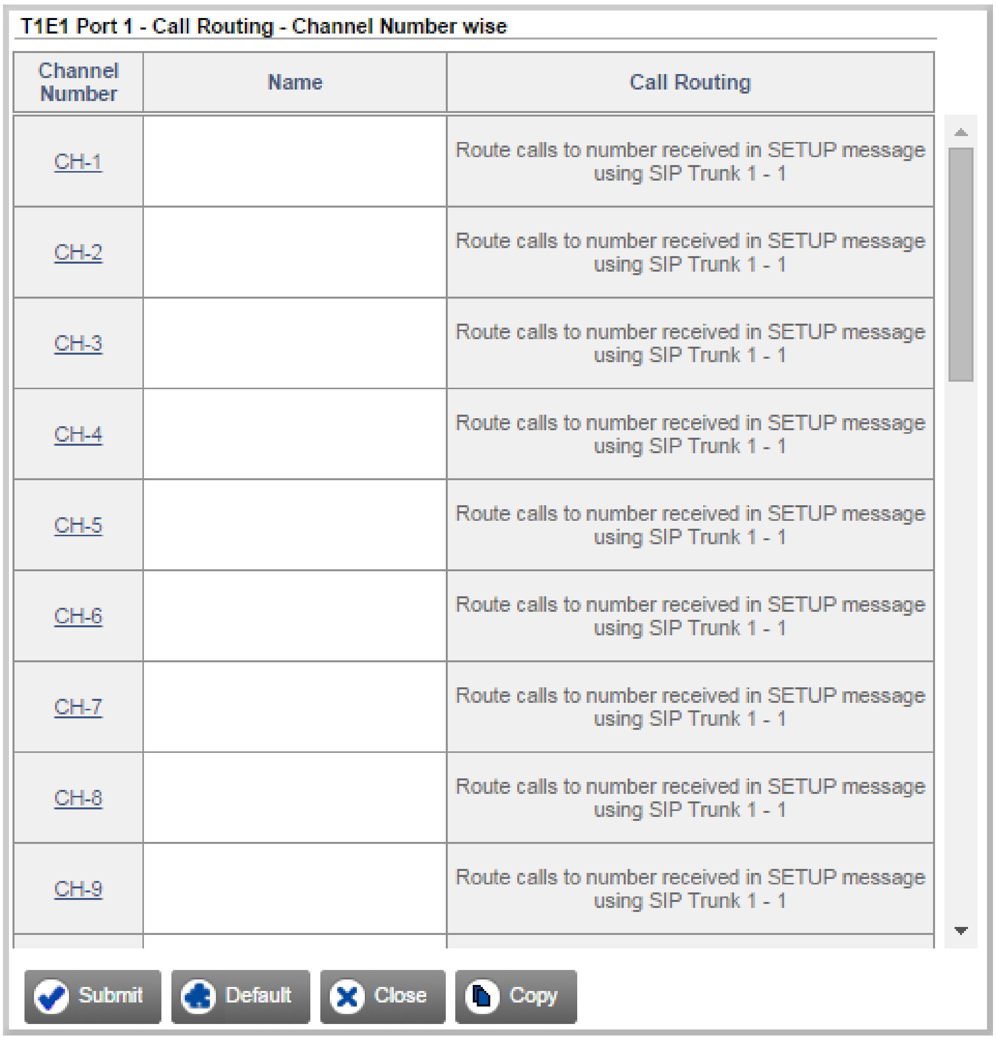
Channel Number Wise

To configure Handling of Incoming Calls for each channel,

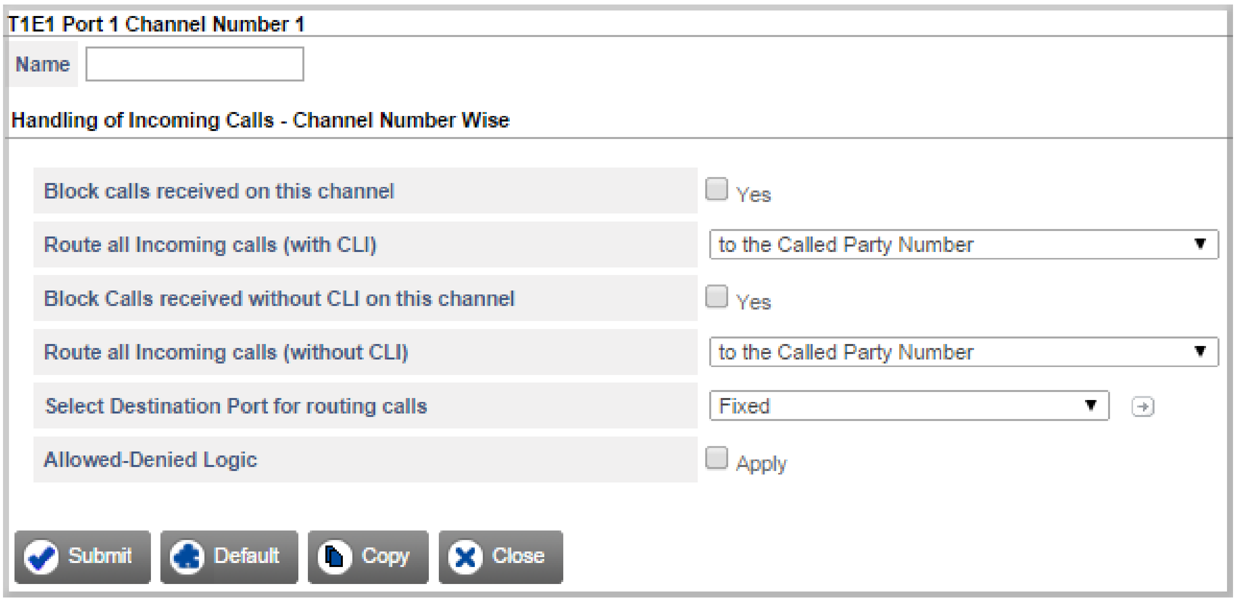
- Select the **Channel Number Wise** check box.



- Click **Settings** .
- The **T1E1 Port 1 - Call Routing - Channel Number Wise** window opens.



- Click the respective channel number to configure the parameters.



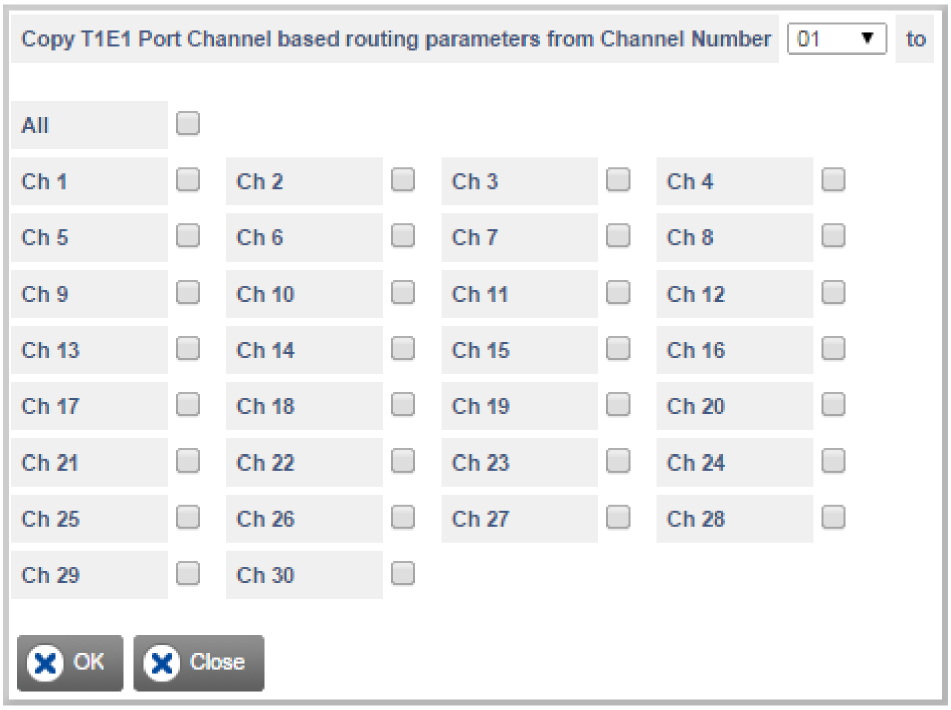
Configure the routing parameters for each channel.

- Block calls received on this channel.
- Route all Incoming Calls (with CLI), see “Handling of Incoming Calls”.
- Block Calls received without CLI on this channel.
- Route all Incoming Calls (without CLI), see “Handling of Incoming Calls”.
- Select the destination port for routing calls, see “Destination Port Determination”.
- Allowed-Denied Logic, see “Allowed - Denied Logic”.
- Handling of Outgoing Calls, see “Handling of Outgoing Calls”.

Copy Channel Based Routing Parameters

- You can also copy the settings of a T1E1 Channel to another T1E1 Channel using the **Copy** button. To do this,

- Click the **Copy** button. The **Copy T1E1 Port Channel based routing parameters from Channel Number** window opens.



- In the **Copy T1E1 Port Channel based routing parameters from Channel Number** box, select the number of the channel you want to copy settings *From*. Select the check boxes of the desired channel numbers you want to copy the settings *To*.

- If you want to copy the settings *To* all the channels, select the **All** check box.

- Click the **OK** button.

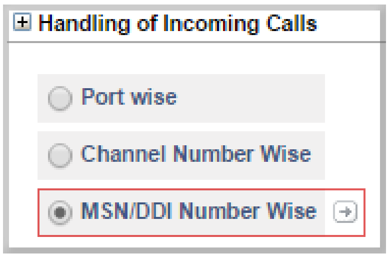
- Once you have copied the settings, you can again edit the specific parameters of the T1E1 Channel you copied the settings to.

- Close the **T1E1 Port 1 Channel Number 1** window.
- To configure any Channel, click the respective channel number on the **T1E1 Port 1 - Call Routing - Channel Number Wise** window and follow the same instructions as given above.
- Close the **T1E1 Port 1 - Call Routing - Channel Number Wise** window.

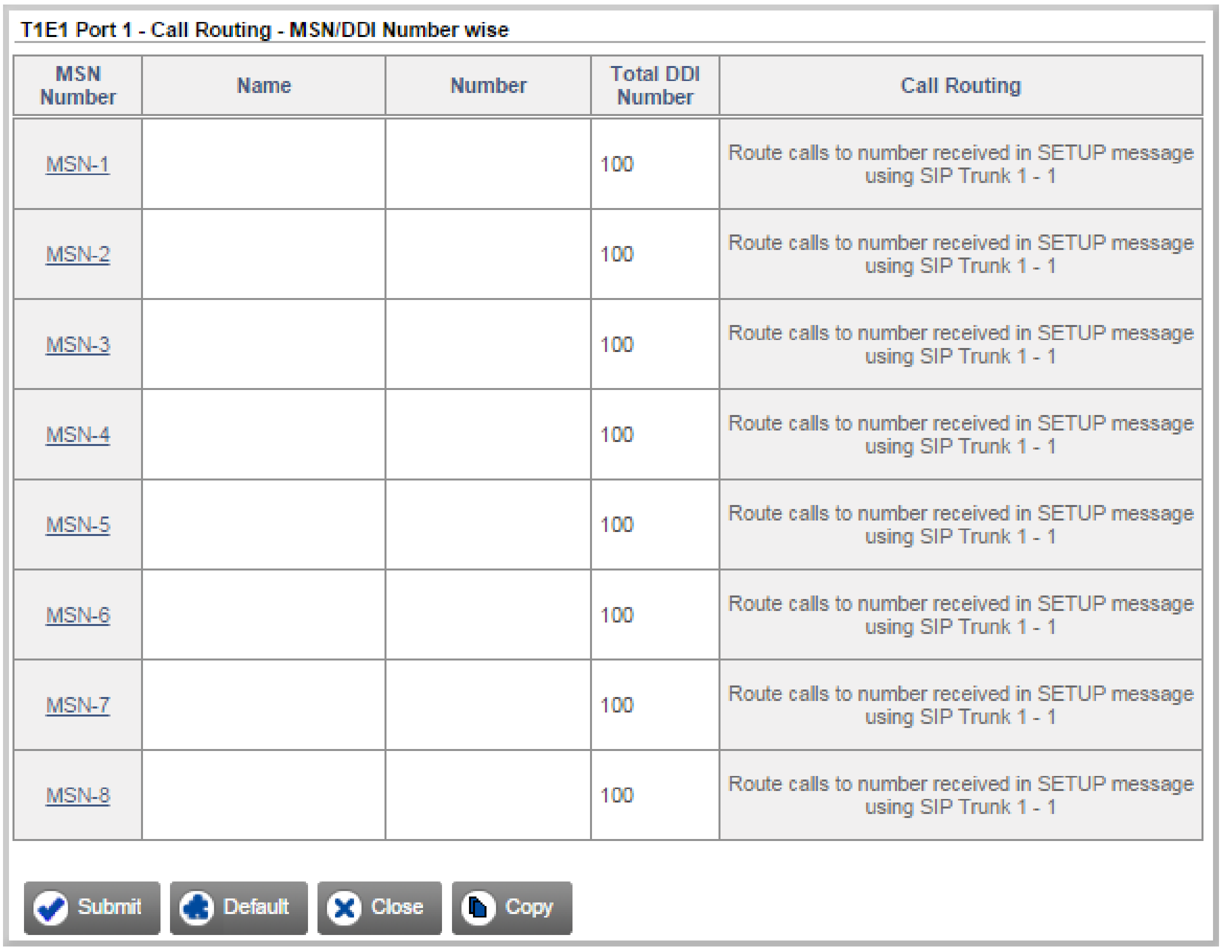
MSN/DDI Number Wise

To configure Handling of Incoming Calls for each MSN Number,

- Select the **MSN/DDI Number Wise** check box.



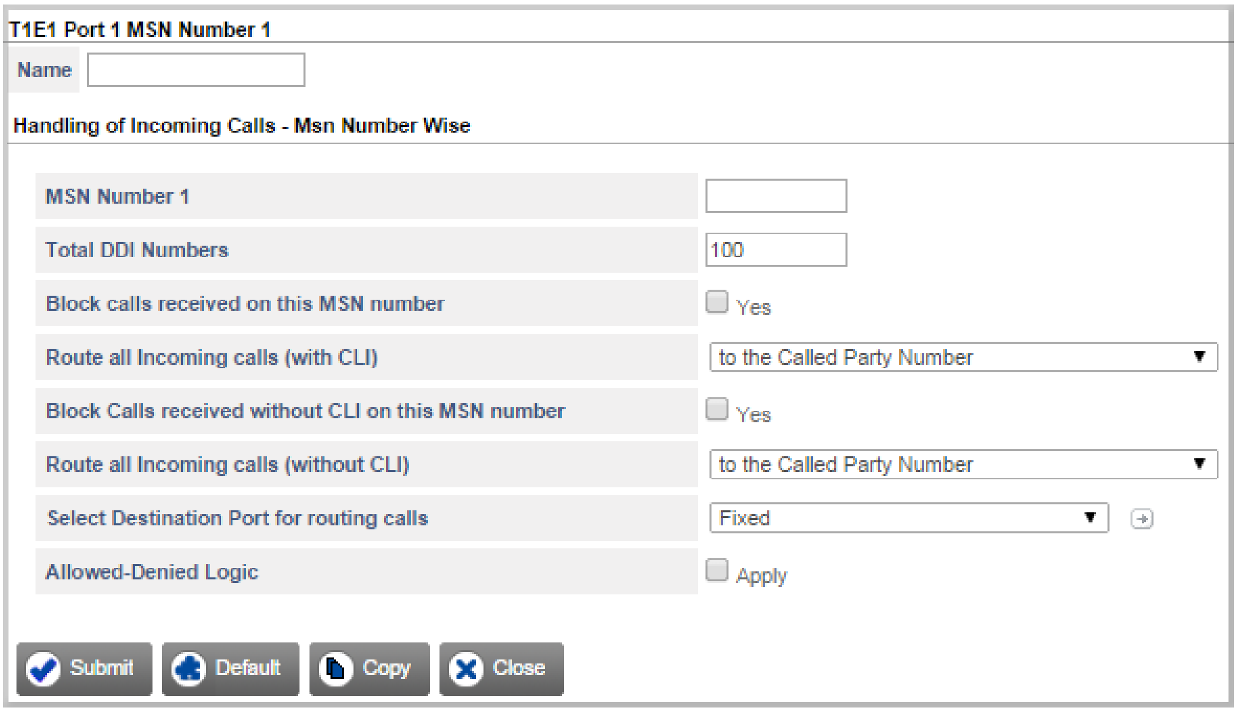
- Click **Settings** .
- The **T1E1 Port 1 - Call Routing - MSN/DDI Number Wise** window opens.



T1E1 Port 1 - Call Routing - MSN/DDI Number wise

MSN Number	Name	Number	Total DDI Number	Call Routing
MSN-1			100	Route calls to number received in SETUP message using SIP Trunk 1 - 1
MSN-2			100	Route calls to number received in SETUP message using SIP Trunk 1 - 1
MSN-3			100	Route calls to number received in SETUP message using SIP Trunk 1 - 1
MSN-4			100	Route calls to number received in SETUP message using SIP Trunk 1 - 1
MSN-5			100	Route calls to number received in SETUP message using SIP Trunk 1 - 1
MSN-6			100	Route calls to number received in SETUP message using SIP Trunk 1 - 1
MSN-7			100	Route calls to number received in SETUP message using SIP Trunk 1 - 1
MSN-8			100	Route calls to number received in SETUP message using SIP Trunk 1 - 1

- Click the respective MSN number to configure the parameters.



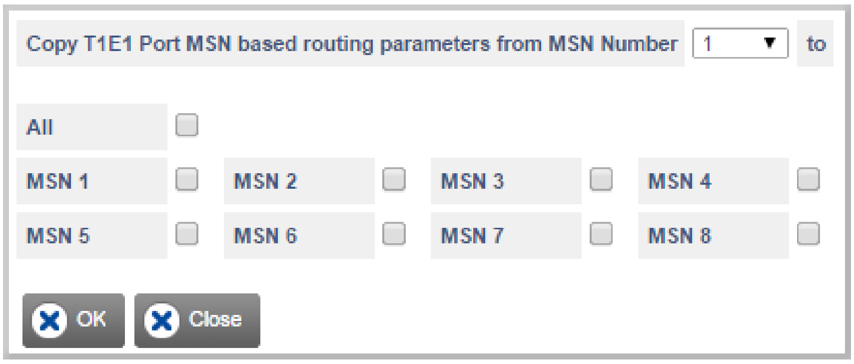
Configure the routing parameters for each MSN number:

- **MSN Number 1:** Enter the first **MSN Number** (max. 24 digits) provided by your Service Provider. Valid digits are 0-9, # and *. Default: Blank.
- **Total DDI Number:** Specify **Total DDI Numbers** provided by your Service Provider. Valid range is 1 to 9999. Default: 0100.
- **Name:** Assign a Name for identification.
- Block calls received on this MSN Number.
- Route all Incoming Calls (with CLI), see “Handling of Incoming Calls”.
- Block Calls received without CLI on this MSN number.
- Route all Incoming Calls (without CLI), see “Handling of Incoming Calls”.
- Select the destination port for routing calls, see “Destination Port Determination”.
- Allowed-Denied Logic, see “Allowed - Denied Logic”.
- Handling of Outgoing Calls, see “Handling of Outgoing Calls”.

Copy MSN Based Routing Parameters

- You can also copy the settings of a MSN Number to another MSN Number using the **Copy** button. To do this,

- Click the **Copy** button. The **Copy T1E1 Port MSN based routing parameters from MSN Number** window opens.

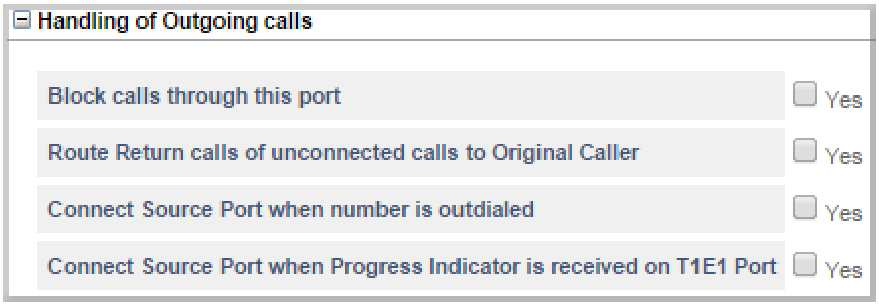


- In the **Copy T1E1 Port MSN based routing parameters from MSN Number** box, select the MSN Number you want to copy settings *From*. Select the check boxes of the desired MSN Numbers you want to copy the settings *To*.
- If you want to copy the settings *To* all the MSN Numbers, select the **All** check box.
- Click the **OK** button.
- Once you have copied the settings, you can again edit the specific parameters of the MSN Number you copied the settings to.

- Close the **T1E1 Port 1 MSN Number 1** window.
- To configure any MSN Number, click the respective MSN Number on the **T1E1 Port 1 - Call Routing - MSN/DDI Number Wise** window and follow the same instructions as given above.
- Close the **T1E1 Port 1 - Call Routing - MSN/DDI Number Wise** window.

Handling of Outgoing Calls

Click **Handling of Outgoing Calls** to expand.



When T1/E1 Port is determined as the destination port, numbers dialed from this port constitute outgoing calls.

For outgoing calls from T1/E1 Port, you can apply the features Automatic Number Translation (ANT) and Route Calls Returned Unconnected to Original Caller.

- Select the **Block calls through this port** check box, if you do not want to route outgoing calls though this port.

- Enable **Route Return calls of unconnected calls to Original Caller** check box, if you want SARVAM UMG to route outgoing calls made from this port that return unconnected back to the original caller. Default: Disabled.

  If you enable this feature, when an outgoing call is made using this port, and the Called Party is found busy or does not respond, SARVAM UMG stores the number of the calling party, the number of the called party and this port (through which the outgoing call was made). A record of each such call is stored for the duration of the Unconnected Calls Record Delete Timer (configurable; default: 999 minutes).

  If the called party returns the call before the expiry of this Timer, SARVAM UMG checks whether *Apply RCOC only if the caller calls back on the same trunk from which the call was made* is enabled or not, and accordingly places the incoming call to the original calling party. To change the duration of this timer, delete records of such calls and enable/disable the *Apply RCOC only if the caller calls back on the same trunk from which the call was made* check box, see "System Parameters".

- To connect the Source Port with the Destination Port without waiting for the call on the Destination Port to mature, enable the **Connect Source Port when number is outdialed** check box. Default: Disabled.

  In all Destination Number Determination methods except *After Answering the Call and Collecting the Digits*, the Source Port gets connected to the Destination Port only after the call has matured, that is, the called party has answered the call. Until the call matures, the caller hears only Ring Back Tone played by the network.

  By connecting the Source Port with the Destination Port immediately after the number is dialed, the caller can know the state of the call; if the called party is busy, not responding, not reachable or is rejecting the call.

- Enable **Connect Source Port when Progress Indicator is received on T1E1 Port** check box, to connect Source Port with the Destination Port as soon as Progress Indicator is received on T1E1 Port without waiting for the call on the Destination Port to mature. Default: Disabled.

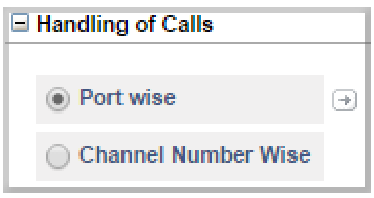
If you enable Connect Source Port when Progress Indicator is received on T1E1 Port, you will not be able to provide the features "Making a New Call using Access Code" and "Disconnecting a Call using Access Code" to users.

- Click **Submit** to save settings.

Handling of Calls

If you have selected **Network** as Orientation Type,

- Select the method to route the incoming calls from the **T1E1 Port**.



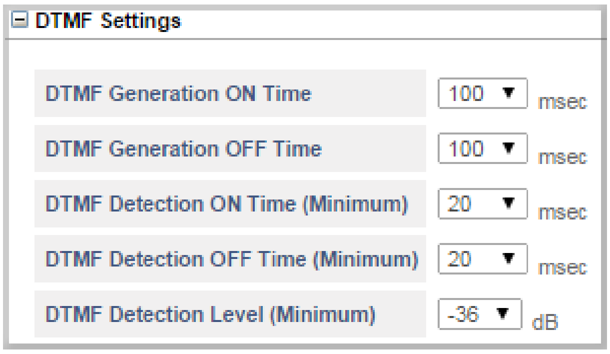
SARVAM UMG provides two options for **Handling of Calls** when the Port is set in Network mode.

- **Port Wise:** Select this method to apply the call routing method for the entire port. See “Port Wise”.

- **Channel Number Wise:** Select this method to apply a different call routing method for each of the 30 channels. See “Channel Number Wise”.

DTMF Settings

- Click **DTMF Settings** to expand.



- Select the appropriate **DTMF Generation ON Time** for the T1E1 Port. This is the time for which the DTMF digit which is to be outdialed remains ON. Valid range is 50 to 250 msec. Default: 100 msec.

- Select the appropriate **DTMF Generation OFF Time** for the T1E1 Port. This is the time for which system should wait before dialing the successive DTMF digits so that the T1E1 network can detect the dialed digits. Valid range is 50 to 250 msec. Default: 100 msec.

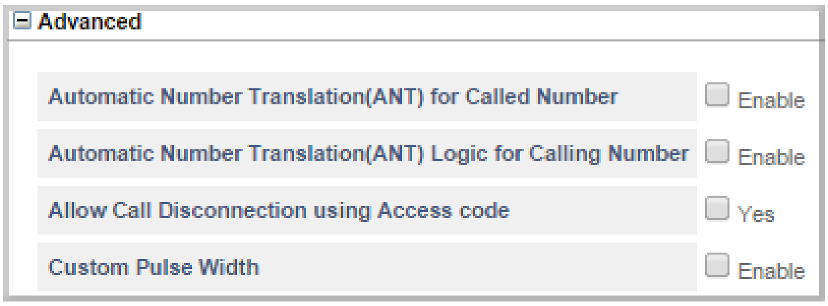
- Select the appropriate **DTMF Detection ON Time (Minimum)** for the T1E1 Port. This is the minimum time period for which the DTMF signal should be present in order to be detected. Valid range is 10 to 200 msec. Default: 20 msec.

- Select the appropriate **DTMF Detection OFF Time (Minimum)** for the T1E1 Port. This is the minimum time period between successive DTMF digits. Valid range is 10 to 200 msec. Default: 20 msec.

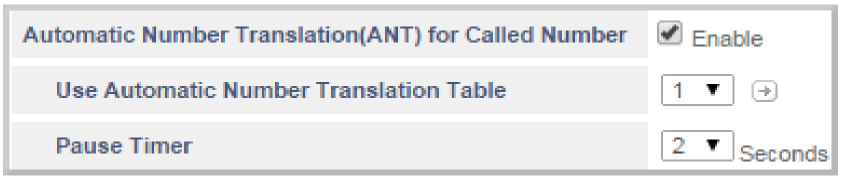
- Select the appropriate **DTMF Detection Level (Minimum)** for the T1E1 Port. This is the minimum level (dB) of the DTMF digit to be considered as valid. Default: -36 dB.

Advanced

- Click **Advanced**.



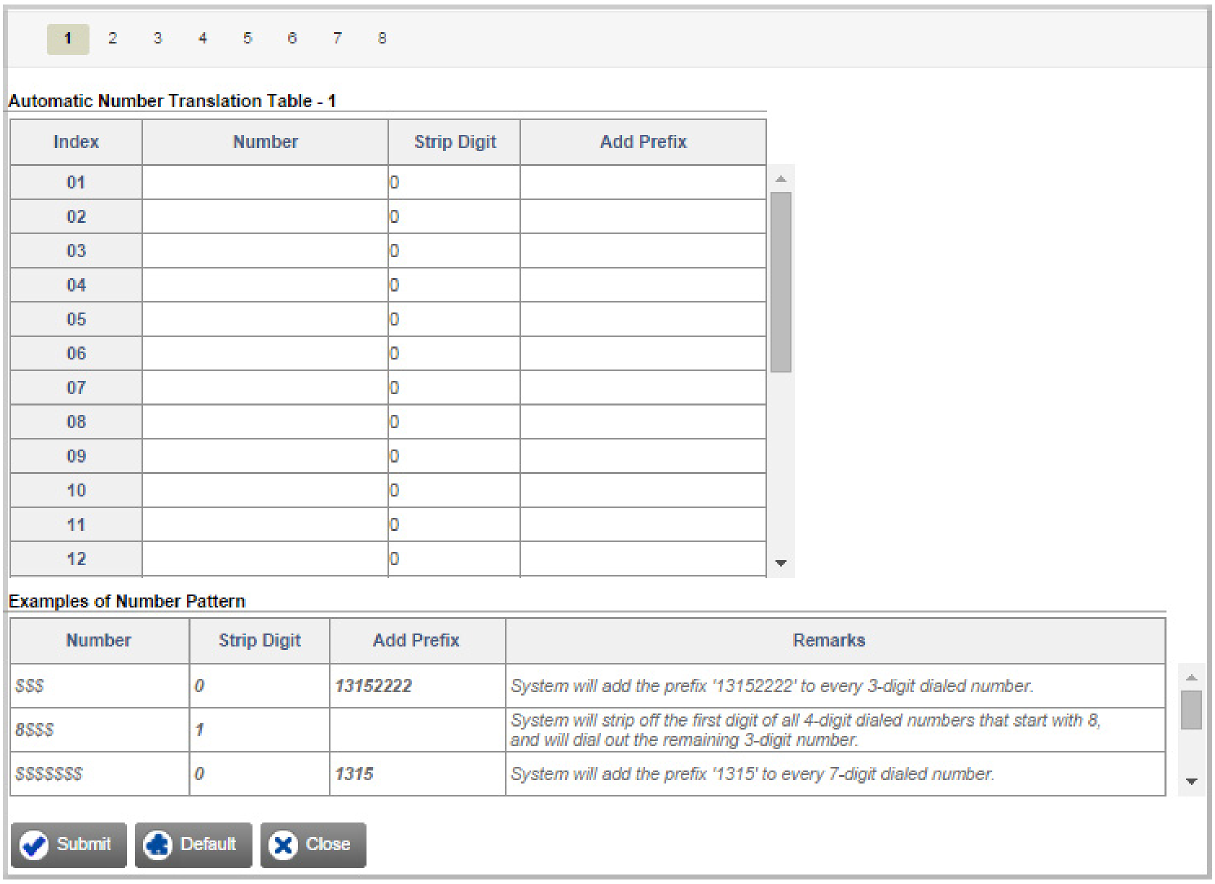
- You can apply **Automatic Number Translation logic** on outgoing calls made from the T1E1 Port.
- To apply ANT logic on the Called Numbers, select the **Automatic Number Translation (ANT) for Called Number** check box. Default: Disabled.



  - In **Use Automatic Number Translation Table**, select the ANT Table number you have configured for the Called Numbers. Default: Table 1.

    If you have not configured the Automatic Number Translation Table,

    - Click **Settings** .
    - The **Automatic Number Translation Table** window opens.



Automatic Number Translation Table - 1

Index	Number	Strip Digit	Add Prefix
01		0	
02		0	
03		0	
04		0	
05		0	
06		0	
07		0	
08		0	
09		0	
10		0	
11		0	
12		0	

Examples of Number Pattern

Number	Strip Digit	Add Prefix	Remarks
$$$	0	13152222	System will add the prefix '13152222' to every 3-digit dialed number.
8$$$	1		System will strip off the first digit of all 4-digit dialed numbers that start with 8, and will dial out the remaining 3-digit number.
$$$$$$$	0	1315	System will add the prefix '1315' to every 7-digit dialed number.

    - You may configure the default Automatic Number Translation Table or any other Table. See "Automatic Number Translation (ANT)" to configure the ANT Table.
    - Click **Submit** to save the ANT Table and close the window.
    - Return to ANT parameter and assign the ANT Table you configured.

- Click **Submit**.

- Set the duration of the **Pause Timer**, if you have configured ^ (Pause) in the Add Prefix column of the ANT Table. Valid range is 1 to 9 seconds. Default: 2 seconds.

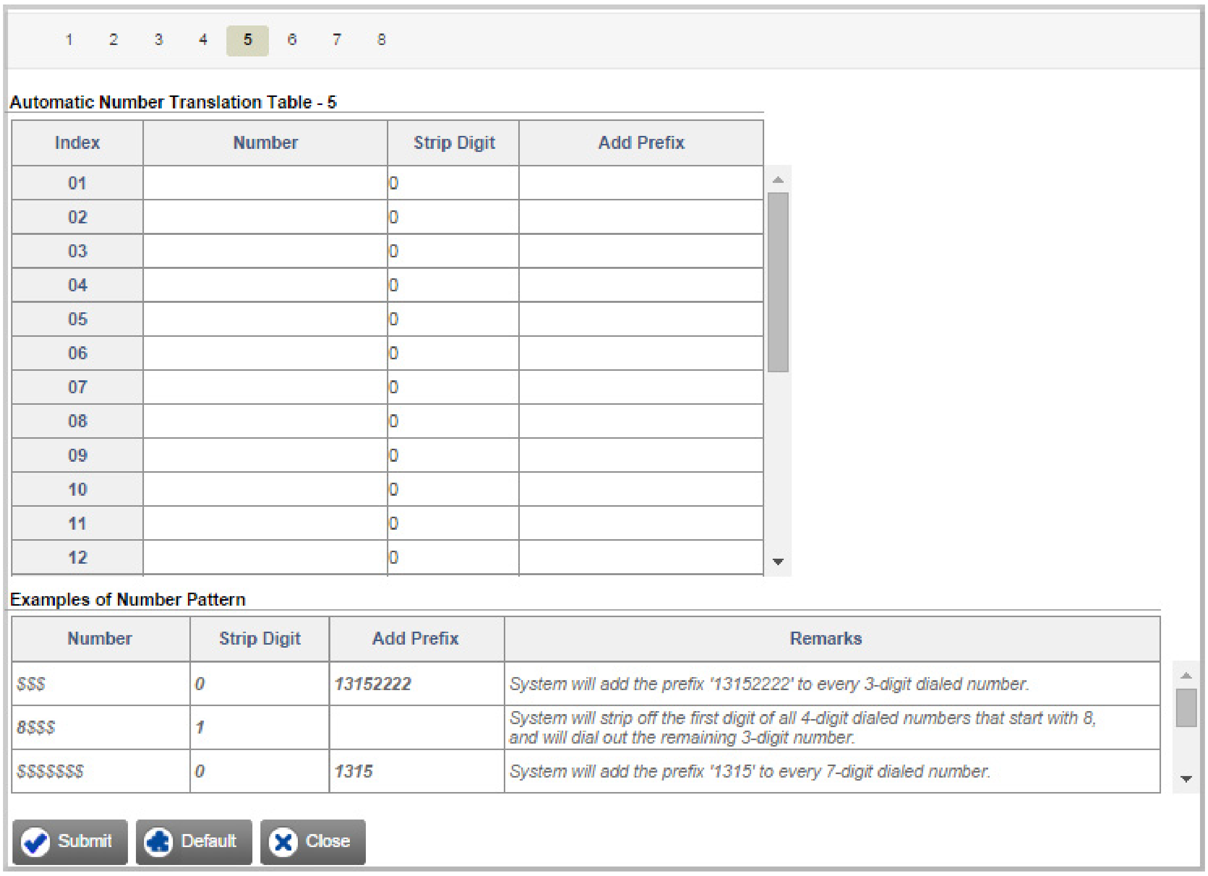
- To apply ANT logic on the Calling Numbers, click the **Automatic Number Translation (ANT) for Calling Number** check box. Default: Disabled.

- In the **Use Automatic Number Translation Table**, select the ANT Table number you have configured for the Calling Numbers. Default: Table 5.

  If you have not configured the Automatic Number Translation Table,

  - Click **Settings**.

  - The **Automatic Number Translation Table** window opens.



Index	Number	Strip Digit	Add Prefix
01		0	
02		0	
03		0	
04		0	
05		0	
06		0	
07		0	
08		0	
09		0	
10		0	
11		0	
12		0	

Number	Strip Digit	Add Prefix	Remarks
$$$	0	13152222	System will add the prefix '13152222' to every 3-digit dialed number.
8$$$	1		System will strip off the first digit of all 4-digit dialed numbers that start with 8, and will dial out the remaining 3-digit number.
$$$$$$$	0	1315	System will add the prefix '1315' to every 7-digit dialed number.

- You may configure the default Automatic Number Translation Table or any other Table. See "Automatic Number Translation (ANT)" to configure the ANT Table.

- Click **Submit** to save the ANT Table and close the window.

    - Return to ANT parameter and assign the ANT Table you configured.

- To enable the feature Disconnect Call using Access Code on the T1E1 Port, select the **Allow Call Disconnection using Access code** check box. To know more about this feature, see "Disconnecting a Call using Access Code".

- To customize the pulse width option and set the pulse shapes configure the **Custom Pulse** parameters. SARVAM UMG generates pulse shapes which match the country standard, where it is installed. However, if the standard pulse shape does not match, SARVAM UMG enables you to customize the pulse width to match your exact requirements.

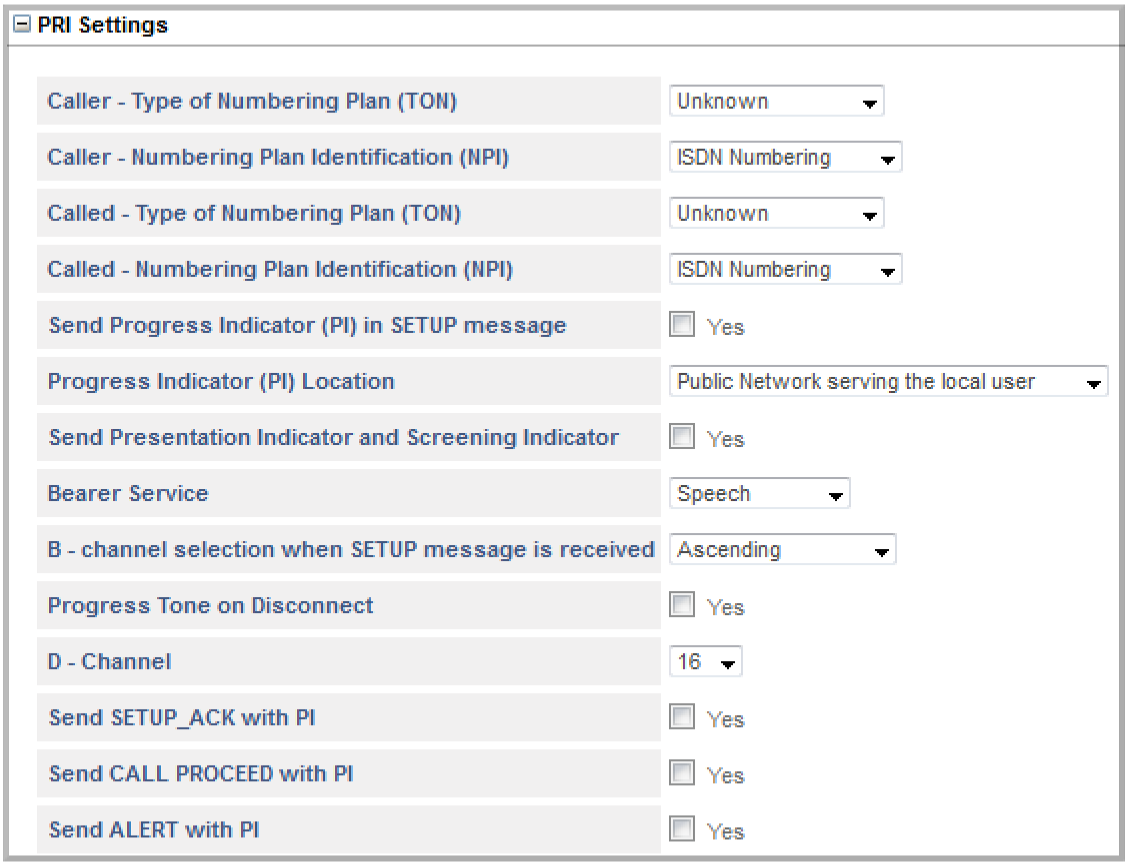
  To use customize pulse width option and set the pulse shape in 1 to 4 phases,

  - Select the **Custom Pulse Width** check box.
  - In **Custom Pulse Width Word 1,** set the pulse width for setting pulse shape in the 1st phase. Valid range is 0 to 127. Default: 63.
  - In **Custom Pulse Width Word 2,** set the pulse width for setting pulse shape in the 2nd phase.Valid range is 001 to 127. Default: 58.
  - In **Custom Pulse Width Word 3,** set the pulse width for setting pulse shape in the 3rd phase. Valid range is 001 to 127. Default: 76.
  - In **Custom Pulse Width Word 4,** set the pulse width for setting pulse shape in the 4th phase. Valid range is 001 to 127. Default: 0.

PRI Settings

If you have selected **PRI** as **Signaling Type**, configure the PRI parameters.

- Click **PRI Settings**.



- Set the duration of the **Overlap Receiving Timer**. This timer is relevant while receiving the called party number information in overlap receiving mode. Valid range is 01 to 99 seconds. Default: 5 seconds.

- Select the required option for sending the **Caller-Type of Numbering Plan (TON) —** Unknown, International, National, Network Specific, Subscriber, Abbreviated or Reserved. Default: Unknown.

- Select the required option for sending the **Caller-Numbering Plan Identification (NPI)** — Unknown, ISDN Numbering, Data Numbering, Telex Numbering, National Numbering, Private or Reserved. Default: ISDN Numbering.

- Select the required option for sending the **Called-Type of Numbering Plan (TON)** — Unknown, International, National, Network Specific, Subscriber, Abbreviated or Reserved. Default: Unknown.

- Select the required option for sending the **Called-Numbering Plan Identification (NPI) —**Unknown, ISDN Numbering, Data Numbering, Telex Numbering, National Numbering, Private or Reserved. Default: ISDN Numbering.

- Select the **Send Progress Indicator (PI) in SETUP message** check box if you want the progress indicator value to be sent in SETUP message. Default: Disabled.

- Set the **Progress Indicator (PI) value in SETUP message** to the desired value. You can select — 1 or 3.

  Progress indicator 1 indicates that the call is not end-to-end ISDN and further call progress information may be available in-band.
  Progress indicator 3 indicates that the origination address is non-ISDN.

  This value will be included in the Setup Message to indicate whether the calling party is an ISDN device or not. Default: 1.

- Select the **Progress Indicator (PI) Location** for the SETUP message.The location is a progress indicator information element that indicates from where the message is coming. Default: Public Network serving the local user.

- Select the **Send Presentation Indicator and Screening Indicator** check box, if you want the system to display the presentation and screening information to the remote end. Default: Disabled.

  - Select the required **Presentation Indicator**. This allows remote end to know whether CLI Number should be displayed to user or not. You can select — Presentation Allowed, Presentation Restricted or Received from Source Port. Default: Received from Source Port.

  - Select the required **Screening Indicator**. This indicates whether the information is provided by the user or the network along with the screening details — not screened, verified and passed or verified and failed. Default: User-provided, not screened.

- Select the **Bearer Service** supported by your service provider. This will be sent in the SETUP Message. You can select — Speech, 3.1 KHz Audio. Default: Speech

- Select the B - channel selection when SETUP message is received[16] as per your requirement.

- Select the **Progress Tone on Disconnect** check box, if you want the system to play the progress tone on the port when the call is released by the remote end or disconnected by the system. Default: Disabled.

- Select the **D - Channel** as per your requirement.

  You may select — channel 01 to channel 31 — for the signaling.

  Default: 16.

- Select the **Send SETUP_ACK with PI** check box, if you want the system to send PI (Progress indicator) element in Setup Ack message. Default: Disabled.

- Select the **Send CALL PROCEED with PI** check box, if you want the system to send PI (Progress indicator) in Proceed message. Default: Disabled.

- Select the **Send ALERT with PI** check box, if you want the system to send PI (Progress indicator) in Alert message. Default: Enabled.

- Click **Submit** to save changes.

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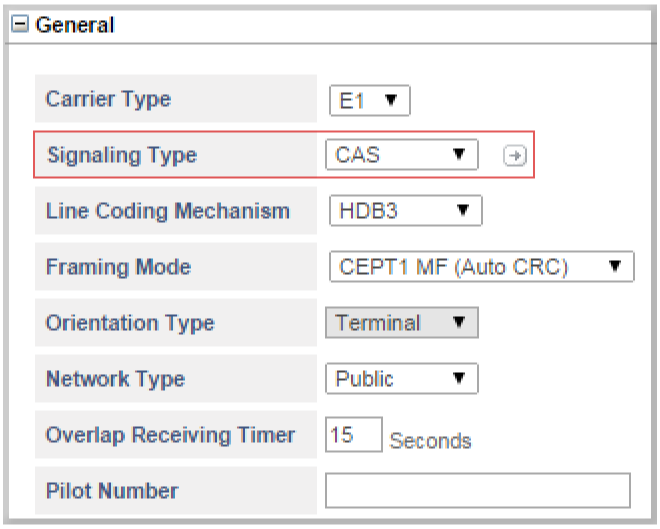
16. This parameter is not applicable in this version, it is meant for future use.

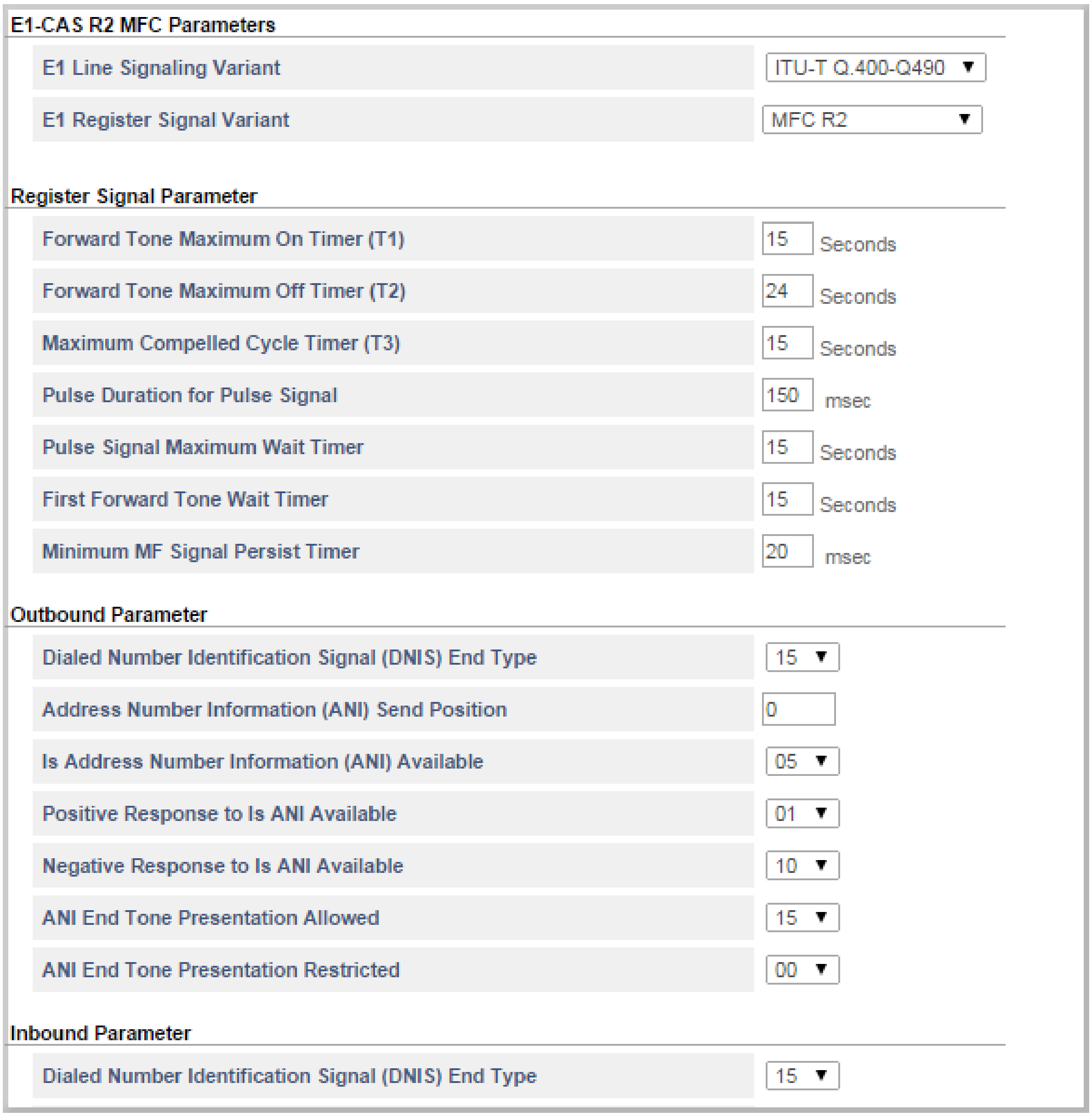
CAS Settings

If you have selected **CAS** as **Signaling Type**, configure the CAS parameters.



- Click **Settings** .

- The CAS parameters window opens.



- Configure the following parameters.

E1-CAS R2 MFC Parameters

- Select the appropriate **E1 Line Signaling Variant —** ITU-T Q.400-Q.490, Brazil. Default: ITU-T Q.400-Q.490.

  If you select Brazil, all the **CAS Parameters** will be assigned default values as per Brazil.

- Select the appropriate **E1 Register Signal Variant —** DTMF, MFC R2. Default: MFC R2.

  If you select DTMF, DNIS/ANI is transmitted in the corresponding speech channel using the DTMF signals as per ITU-T Q.23.

If you select MFC R2, DNIS/ANI is transmitted in the corresponding speech channel using the MFC R2 signals as per ITU-T Q.400-Q490.

Register Signal Parameters

- Set the duration of the **Forward Tone Maximum On Timer (T1)**. This specifies the maximum time for which the forward signal must remain ON from the outbound end. Valid range is 01 to 99 seconds. Default: 15 seconds.

- Set the duration of the **Forward Tone Maximum Off Timer (T2)**. This specifies the maximum time between two outgoing forward signals. Forward tone remains OFF during this time.Valid range is 01 to 99 seconds. Default: 24 seconds.

- Set the duration of the **Maximum Compelled Cycle Timer (T3)**. This specifies the maximum time within which one compelled signaling cycle shall end. Valid range is 01 to 99 seconds. Default: 15 seconds.

- Set the duration of the **Pulse Duration for Pulse Signal**. This specifies the pulse duration of the backward signals — A-3, A-4, A-6 and A-15 — that are pulsed to the outbound end. Pulse duration of these signals vary from country to country. Valid range is 001 to 999 msec. Default: 150 msec.

It is recommended that tolerance be fixed at +/- 25 msec.

- Set the duration of the **Pulsed Signal Maximum Wait Timer**. This specifies the time for which the outbound end waits for the pulsed signal. If the pulsed signal is not received during this time, the compelling signaling is said to be complete. Valid range is 01 to 99 seconds. Default: 15 seconds.

- Set the duration of the **First Forward Tone Wait Timer**. This specifies the time between receipt of line seizure signal and the first forward signal. Valid range is 08 to 24 seconds. Default: 15 seconds.

- Set the duration of the **Minimum MF Signal Persist Timer**. This specifies the minimum time for which the forward /backward signal shall be sustained on the line by the receiving end. Valid range is 001 to 255 msec. Default: 20 msec.

Outbound Parameters

- **Dialed Number Identification Signal (DNIS) End Type** specifies the DNIS End Type as per your requirement. This parameter is applicable only when DNIS length is set to 99 (i.e. variable). The outbound end indicates end of DNIS using a group I tone or using time out.

  Valid range is 00 to 15; where 00 indicates End of DNIS as time out. 01 to 15 indicates group I tone to declare End of DNIS. Default: 15.

- **Address Number Information (ANI) Send Position** specifies the number of DNIS digits after which address number information is to be sent. Address number information is usually sent on receiving the backward tone Send next digit or Send next ANI digit.

  If send next address number information tone is received then this parameter is not applicable. But if same tone is used by the inbound end to request the next ANI digit and the next DNIS digit, ANI is sent after the number of digits as set in this field. Valid range is 00 to 99. Default: 00.

- **Address Number Information (ANI) Available** signifies Group A tone (received from the inbound tone) that is to be interpreted as a question by the inbound end asking the outbound end whether the outbound end has ANI digits to be sent. Valid range is 01 to 15. Default: 05.

- **Positive Response to Is ANI Available** signifies the Group 1 tone that the outbound end will send to the inbound end as a response to *Is ANI Available tone from the inbound end*. The tone defined in this parameter indicates the Group 1 tone with which the Outbound end will respond to the inbound end to indicate that it has ANI digits to be sent. Valid range is 01 to 15. Default: 01.

- **Negative Response to Is ANI Available** signifies the Group 1 tone that the outbound end will send to the inbound end as a response to *Is ANI Available tone from the inbound end*. The tone defined in this parameter indicates the Group 1 tone with which the Outbound end will respond to the inbound end to indicate that it does not have ANI digits to be sent. Valid range is 01 to 15. Default: 10.

- **ANI End Tone Presentation Allowed** signifies the Group 1 tone used to signify end of ANI digits with Presentation Allowed. Valid range is 01 to 15. Default:15.

- **ANI End Tone Presentation Restricted** signifies the Group 1 tone used to signify end of ANI digits with Presentation Restricted. Valid range is 01 to 15. Default: 00.

Inbound Parameters

- **Dialed Number Identification Signal (DNIS) End Type** parameter is applicable only when the DNIS length is set to 99 (i.e. variable). The outbound end indicates end of DNIS using a group I tone or using time out.

  The range of DNIS End Type is from 00/ 01 to 15; where 00 indicates End of DNIS as time out, 01 to 15 indicates group I tone to declare End of DNIS. Default:15.

- **Dialed Number Identification Signal (DNIS) Digit Length** specifies the number of DNIS digits required by inbound end to indicate the Called party number during MFC R2 signaling. Valid range is 01 to 99. Default: 99.

- Enter the **Address Number Information (ANI) Request Position** as per your requirement. The inbound end may or may not request ANI digits. It may request ANI digits after receiving the first DNIS or after receiving second DNIS or even after receiving all the DNIS digits.

  The range of ANI Request Position is as follows:

ANI Request	Meaning
00	Never request ANI digits
01-98	Request ANI digits on receipt of these many DNIS digits
99	Request after receiving all the DNIS digits (complete DNIS)

  Default: 99.

- **Address Number Information (ANI) Length** specifies the number of ANI digits that would be expected by the inbound side as Calling Party Number during MFC R2 signaling. This parameter at the inbound side guides the inbound register to switch from requesting ANI digits back to requesting DID digits. Valid range is 00 to 99. Default: 99.

  - ANI Length = 00 signifies ANI is not sent by the Outbound end.
  - ANI Length = 99 signifies ANI Length is variable.

- **Ask Address Number Information (ANI)** specifies the backward group A tone used to ask the outbound end whether it has ANI digits to be sent. This parameter is also known as **Request ANI Category**. Valid range is 00/ 01 to 15. If no tone is sent by the inbound end, set this parameter to 00. For India, this parameter is set to 04. Default: 05.

- **Positive Response to Ask ANI** specifies that the Group 1 forward tone is to be received by the inbound end from the outbound which in turn indicates that outbound end has ANI digits to be sent. This parameter is also known as ANI category. Valid range is 01 to 15. Default: 01.

  For example, In India I-1 or I-10 is sent by the outbound end. In Kuwait, I-6 is sent. This parameter cannot be zero because; Is ANI Available request will be made by the inbound end only if the country supports this protocol.

- **Negative Response to Ask ANI** specifies the Group 1 forward tone to be received by the inbound end from the outbound which would indicate that outbound end has ANI digits to be sent. This parameter is also known as ANI category. Valid range is 01 to 15. Default: 10.

  For example, in India I-1 or I-10 is sent by the outbound end. In Kuwait, I-6 is sent. This parameter cannot be zero because; Is ANI Available request will be made by the inbound end only if the country supports this protocol.

- **ANI End Tone Presentation Allowed** specifies the Group I tone that the inbound end should expect from the outbound end to consider End of ANI digits with information that the Presentation of ANI by the outbound end is allowed. Valid range is 00/ 01 to 15. Default: 15. If no tone is sent, set this parameter to 00. For India use A-4, for China use A-1.

- **ANI End Tone Presentation Restricted** specifies the group I tone that the inbound end shall expect from the outbound end to consider End of ANI digits with an information that the Presentation of ANI by the outbound end is Restricted. Valid range is 00/ 01 to 15. Default: 00. If no tone is sent, set this parameter to 00. For India use A-4, for China use A-1.

- **Ask Calling Party Sub Category** specifies the group 1 tone that the inbound end shall expect from the outbound end to consider End of ANI digits with an information that the Presentation of ANI by the outbound end is Restricted. Select the check box to enable. Default: Disabled.

Forward Group II

- **Ordinary Subscriber** specifies the forward group II tone used to inform the inbound end that the calling party is an Ordinary Subscriber. This signal is sent in response to Calling Party Category signal Request from the inbound end. Valid range is 00/ 01 to 15. Default: 01. If this parameter is not applicable, assign 00.

- **Priority Subscriber** specifies the forward group II tone used to inform the inbound end that the calling party is a Priority Subscriber. This signal is sent in response to Calling Party Category signal Request from the inbound end. Valid range is 00/ 01 to 15. Default: 02. If this parameter is not applicable, assign 00.

- **Maintenance Equipment** specifies the forward group II tone used to inform the inbound end that the calling party is Maintenance equipment. Valid range is 00/ 01 to 15. Default: 03. If this parameter is not applicable, assign 00.

- **Operator** specifies the forward group II tone used to inform the inbound end that the calling party is Operator. Valid range is 00/ 01 to 15. Default: 05. If this parameter is not applicable, assign 00.

- **Pay Phone** specifies the forward group II tone used to inform the inbound end that the calling party is Pay Phone (Coin box). Valid range is 00/ 01 to 15. Default: 00. If this parameter is not applicable, assign 00.

- **Data Transmission** specifies the forward group II tone used to inform the inbound end that the call is a Data Call. Valid range is 00/ 01 to 15. Default: 06. If this parameter is not applicable, assign 00.

- **Interception Operator** specifies the forward group II tone used to inform the inbound end that the call is from Interception Operator. Valid range is 00/ 01 to 15. Default: 00. If this parameter is not applicable, assign 00.

Backward Group A

- **Send Next Digit (N+1) (DNIS)** specifies the backward group A tone used to request next digit. Be it ANI digit or DNIS digit. Valid range is 00/ 01 to 15. Default: 01. If you do not want to use any tone, assign 00.For India, use A-1 to signify Send DNIS Digit event.

- **Send Last But One Digit (N-1) (DNIS)** specifies the backward group A tone used to request last but one digit i.e. N-1 digit. Be it ANI digit or DNIS digit. Valid range is 00/ 01 to 15. Default: 02. If you do not want to use any tone, assign 00. For India, use A-9 to signify Send last but one digit event.

- **Send Last But Two Digits (N-2) (DNIS)** specifies the backward group A tone used to request last but two digits i.e. N-2 digit. Be it ANI digit or DNIS digit. Valid range is 00/ 01 to 15. Default: 07. If you do not want to use any tone, assign 00. For India, use A-7 to signify Send last but two digits event.

- **Send Last But Three Digits (N-3) (DNIS)** specifies the backward group A tone used to request last but three digits i.e. N-3 digit. Be it ANI digit or DNIS digit. Valid range is 00/ 01 to 15. Default: 08. If you do not want to use any tone, assign 00. For India use A-8 to signify Send last but three digits event.

- **Send Caller Party Category and ANI Digit** sends the calling party’s category requests transmission of a group II signal. Valid range is 00 to 15. Default: 05.

- **Address Completed, Change over of Group B** specifies the backward group A tone used to inform the inbound end that the incoming register at the inbound end needs no additional address digit and is about to go over to transmission of a group B signal conveying the status of equipment at the subscriber at the inbound end. Valid range is 00/ 01 to 15. Default: 03. If you do not want to use any tone, assign 00.

- **Send Calling Party Category and Change to Group C** specifies the backward group A tone used by the inbound end to request Calling Party Category from the outbound end. This tone also informs the outbound end to change to reception of Group C signal. Valid range is 00/ 01 to 15. Default: 00. If you do not want to use any tone, assign 00.

- **Congestion in the National Network** specifies the backward group A tone used to inform the congestion at the inbound end. Valid range is 00/ 01 to 15. Default: 04. If you do not want to use any tone, assign 00.

- **Send Calling Party Category** specifies the backward group A tone used to request calling party category. Valid range is 00/ 01 to 15. Default: 05. If you do not want to use any tone, assign 00.

  For India, use A-7 to signify 'Send calling party's category' event.

- **Address Completed, Charge, Set Speech Condition** specifies the backward group A tone used to inform the inbound end that the incoming register at the inbound end needs no additional address digit, but will not send Group B signals. Also charge the call on answer. Valid range is 00/ 01 to 15. Default: 06. If you do not want to use any tone, assign 00.

- **Repeat DNIS Digits from Beginning** specifies the backward group A tone used to inform the outbound end to send all the DNIS digits from the beginning. Valid range is 00/ 01 to 15. Default: 00. If you do not want to use any tone, assign 00.

- **Send Next ANI Digits** specifies the backward group A tone used to request next (first) ANI digit. Valid range is 00/ 01 to 15. Default: 05. If no such tone is sent, set this parameter to 00.

  A few countries use different tone to request next ANI digit and next DNIS digits. For example, India uses A-4, China uses A-1.

Backward Group B

- **Send Special Information Tone** specifies the backward group B tone used to inform the outbound end that the call cannot be made through because of reasons beyond those which are considered by the Protocol. Hence Special Information tone will be sent to the calling party. SARVAM UMG will send only the Group B signal and then disconnect the call. Valid range is 00/ 01 to 15. Default: 02. If you do not want to use any tone, assign 00.

- **Send Special Information Tone and Setup Speech Conditions** specifies the backward group B tone used to inform the outbound end that the call cannot be made through because of reasons beyond those which are considered by the Protocol. Hence, Special information tone will be sent to the calling party and request the outbound end to setup speech conditions.

  In this case, SARVAM UMG shall connect the calling party to the voice message of the system informing the caller that the call cannot be connected. Valid range is 00/ 01 to 15. Default: 02. If you do not want to use any tone, assign 00.

- **Subscriber Line Busy** specifies the backward group B tone used to inform the outbound end that the called subscriber is busy. Subscriber Line busy range is from 00/ 01 to 15. Default: 03. If you do not want to use any tone, assign 00.

- **Subscriber Line Free, Charge** specifies the backward group B tone used to inform the outbound end that the called subscriber is free and the call is to be charged on answer. Valid range is 00/ 01 to 15. Default: 06. If you do not want to use any tone, assign 00.

- **Subscriber Line Free, No Charge** specifies the backward group B tone used to inform the outbound end that the called subscriber is free, but the call is not to be charged on answer. This signal permits non-chargeable calls without the need for transferring **no charge** information by line signals. Valid range is 00/ 01 to 15. Default: 07. If you do not want to use any tone, assign 00.

- **Congestion** specifies the backward group A tone used to inform that congestion is encountered after changeover from Group-A to Group-B signals. Valid range is 00/ 01 to 15. Default: 04. If you do not want to use any tone, assign 00.

- **Unallocated Number** specifies the backward group B tone used to inform the outbound end that the number received is not in use. Valid range is 00/ 01 to 15. Default: 05. If you do not want to use any tone, assign 00.

- **Subscriber Line Out of Order** specifies the backward group B tone used to inform the outbound end that the called subscriber's line is out of order. Valid range is 00/ 01 to 15. Default: 08. If you do not want to use any tone, assign 00.

- **Call Rejected, No Indication** specifies the Group B backward tone used to inform the outbound end that the call is rejected but there is no indication of cause. Valid range is 00/ 01 to 15. Default: 00. If this parameter is not applicable, assign 00.

- **Alternative Answer Tone** specifies the Group B backward tone used to inform the outbound end that the call is accepted and the speech path is made through. Valid range is 00/ 01 to 15. Default: 00. If this parameter is not applicable, assign 00.

- **Changed Number** specifies the Group B backward tone used to inform the outbound end that the number dialed by the calling party is changed. However, this parameter is rarely used. Valid range is 00/ 01 to 15. Default: 00. If this parameter is not applicable, assign 00.

Backward Group C

- **Send Next ANI Digit** specifies the backward group C tone to request next (even first) ANI digit from the outbound end. Valid range is 00/ 01 to 15. Default: 00. If this parameter is not applicable, assign 00.

- **Request Transition Back to Group A and Restart from First DNIS (Group C)** specifies the backward group C tone to restart from the first DNIS and request transition to Group A. Valid range is 00/ 01 to 15. Default: 00. If this parameter is not applicable, assign 00.

- **Address Completed Change to Reception of Group B** specifies the backward group C tone used to signify Address completed, change to reception of Group B signal. Valid range is 00/ 01 to 15. Default: 00. If this parameter is not applicable, assign 00.

- **Congestion** specifies the backward group C tone used to signify Congestion. Valid range is 00/ 01 to 15. Default: 00. If this parameter is not applicable, assign 00.

- **Request Transition Back to Group A and Sent Next DNIS** specifies the backward group C tone used to signify request transition back to group A, and send next DNIS. Valid range is 00/ 01 to 15. Default: 00. If this parameter is not applicable, assign 00.

- **Request Transition Back to Group A and Repeat the Last DNIS** specifies the backward group C tone used to signify request transition back to group A, and repeat the last DNIS. Valid range is 00/ 01 to 15. Default: 00. If this parameter is not applicable, assign 00.

Line Signal Parameters

- By default **Line Signaling** check box is enabled. If you are unable to make outgoing calls, check with your Service Provider and disable this option.

- **C and D Bits** indicates the default values of C and D bits when the T1/E1 Port transmits line signals.

CD Bits	Meaning (Binary Value)
0	00 (C=0, D=0)
1	01
2	10
3	11

Default: 01 i.e. C=0 and D=1

The C and D bits received during an IC call should be ignored by the system.

- **Invert Bit A Flag** specifies whether A-bit is to be inverted before transmitting and on receiving. Select the check box to Invert Bit A. Default: Disabled (Do Not Invert Bit A).

- **Invert Bit B Flag** specifies whether B-bit is to be inverted before transmitting and on receiving. Select the check box to Invert Bit B. Default: Disabled (Do Not Invert Bit B1)

- **Invert Bit C Flag** specifies whether C-bit is to be inverted before transmitting and on receiving. Select the check box to Invert Bit C. Default: Disabled (Do Not Invert Bit C).

- **Invert Bit D Flag** specifies whether D-bit is to be inverted before transmitting and on receiving. Select the check box to enable, that is to Invert Bit D. Default: Disabled (Do Not Invert Bit D).

- **E1 Metering Bit** signifies the bit used by the network to signal metering pulses. You can select — None, Bit-A, Bit-B, Bit-C or Bit-D. Default: Bit-A.

- Set the duration of the **E1 Metering Pulse Minimum Timer**. This specifies the minimum time for which the metering bit is changed, to be recognized as a genuine metering pulse subject to E1 Metering Pulse Minimum timer. All Changes occurred for time less than this timer is ignored. Valid range is 20ms to 1000ms. Default: 150ms.

- **Clear Back Signal** is the signal used to signify that the called party has disconnected the line first. This is indicated in two ways: Release Guard (Ab =1) or Forced Release (Bb = 0). This parameter is country specific. Default: Release Guard.

- Set the duration of the **Release Timer**. This specifies the time for which the clear back signal should persist on the line to be recognized as a genuine clear back signal. This is also known as Clear Back timer. Valid range is 20 to 1000 msec. Default: 400 msec.

- Set the duration of the **Line Seizure Acknowledge Wait Timer**. This specifies the time for which the outbound end waits for seizure acknowledgement from the inbound end after sending the line seizure signal. On expiry of this timer, clear forward signal is sent by the outbound end. Alarm is to be generated. This timer is applicable only when acting as outbound end. Valid range is 0001 to 9999 msec. Default: 200msec.

- Set the duration of the **Release Guard Timer**. This specifies the time for which inbound register waits before declaring the channel idle (sending idle signal) when clear forward line signal is received from the outbound end. This timer is applicable for Forced Release signal. This timer is applicable only when acting as inbound end. This timer depends on the speed of switching and processing. Valid range is 0000 to 9999 msec. Default: 200msec.

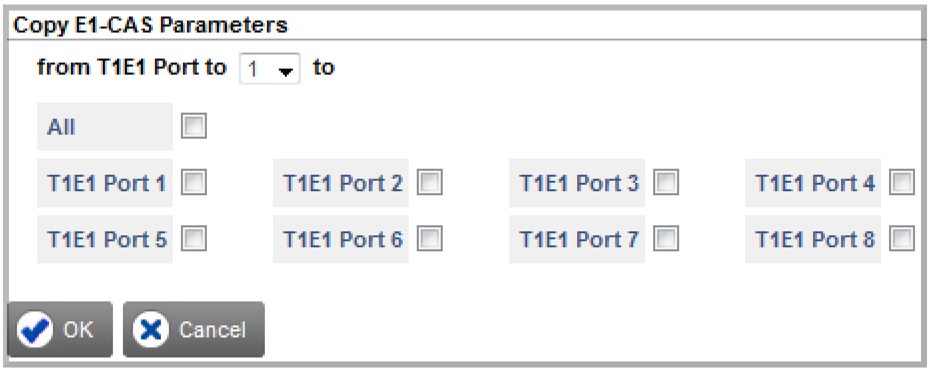
- Click **Submit** to save changes.

- Close the window to return to the main page.

Copy E1-CAS Parameters

- You can also copy the settings of a E1-CAS from one T1E1 port to the another using the **Copy** button. To do this,

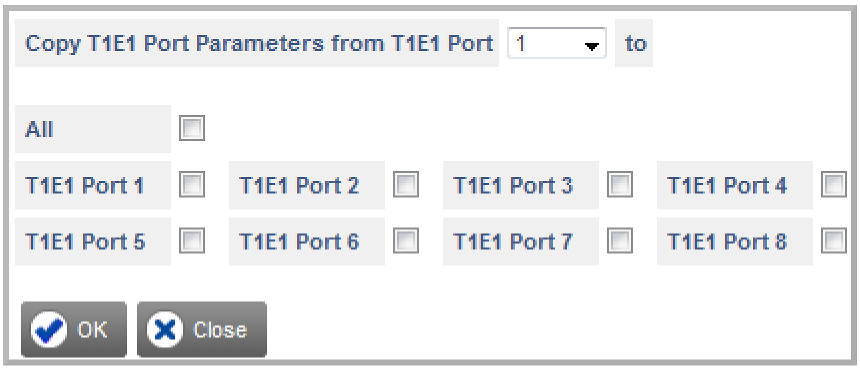
- Click the **Copy** button. The **Copy E1-CAS Parameters** window opens.



  - In the **from T1E1 Port to** box, select the number of the T1E1 Port you want to copy settings *From*. Select the check boxes of the desired port numbers you want to copy the settings *To*.
  - If you want to copy the settings *To* all the T1E1 Ports, select the **All** check box.
  - Click the **OK** button.
  - Once you have copied the settings, you can again edit the specific parameters of E1-CAS you copied the settings to.
- Close the **Copy E1-CAS Parameters** window.
- Click **Submit** to save changes.
- Close the window to return to the main page.

Copy T1E1 Port Parameters

- You can also copy the settings of a T1E1 Port to another T1E1 Port using the **Copy** button. To do this,
  - Click the **Copy** button. The **Copy T1E1 Port Parameters** window opens.



  - In the **Copy T1E1 Port Parameters from T1E1 Port** box, select the number of the port you want to copy settings *From*. Select the check box of the respective port numbers you want to copy the settings *To*.
  - If you want to copy the settings *To* all the ports, select the **All** check box.

- Click the **OK** button.
- Once you have copied the settings, you can again edit the specific parameters of the **T1E1** Port you copied the settings to.

Login Password

You can configure SARVAM UMG using Jeeves and by dialing commands from a telephone (only specific parameters).

Login Password for Jeeves

To configure the system, you must log into the Jeeves using the Jeeves Password.

The default Jeeves Password is 1234. When you login for the first time, you will be prompted to change the password.

The password must be as per the specifications given below:

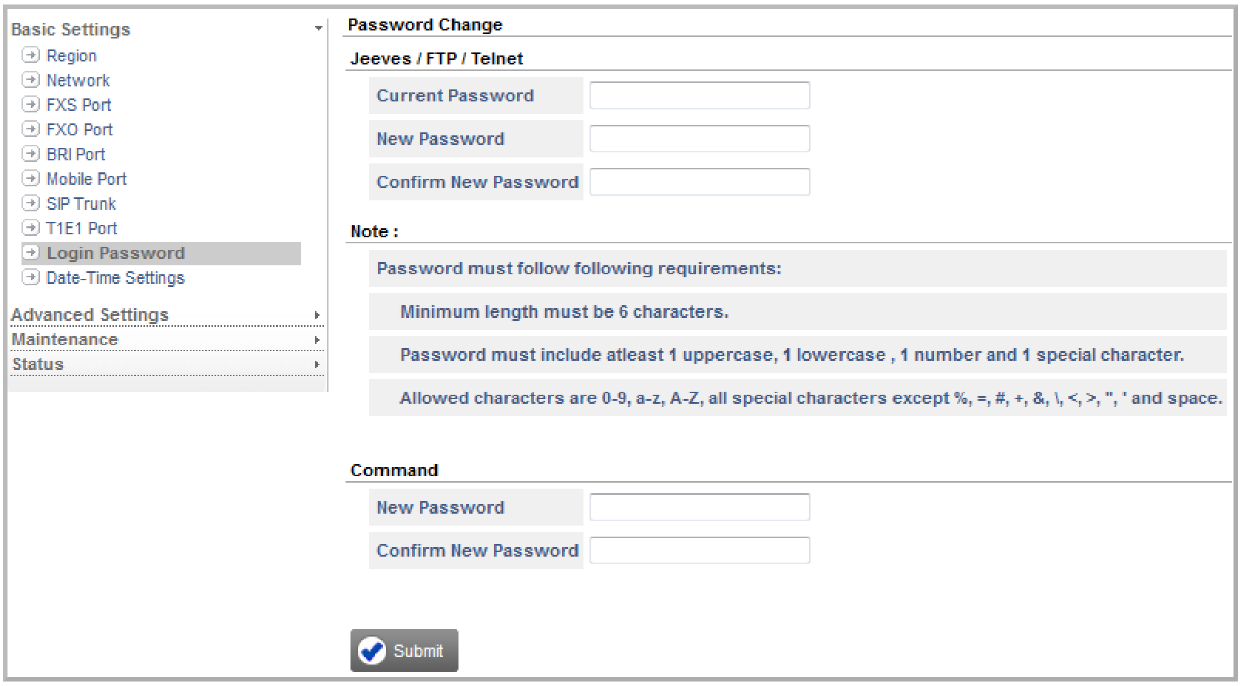
- It must not be less than 6 characters and can be of maximum 12 characters.
- Digits 0 to 9 and all ASCII characters are allowed, except Percentage **%**, Hash **#,** Equal to **=,** Plus **+**, And **&**, Backslash **\,** Less than **<,** Greater than **>,** Apostrophe **' ,** Double Quote **"** and **Space**.
- It must include atleast one upper-case, one lower-case, one number and one special character.

To provide additional security, if you enter a wrong password five times consecutively within 10 minutes, the system will block the source IP Address for 10 minutes. The notification (Warning) will be sent for this event to the SNMP Manager. See *"Simple Network Management Protocol (SNMP)"* *for more details.*

To change the Jeeves Password,

- Click the **Basic Settings** link to expand.

- Click the **Login Password** link.



Under **Jeeves/FTP/Telnet**,

- Enter **Current Password**.
- Enter **New Password**. Digits 0 to 9 and all ASCII characters are allowed, except Percentage **%**, Hash **#,** Equal to **=,** Plus **+**, And **&**, Backslash **\\,** Less than **<,** Greater than **>,** Apostrophe **'** , Double Quote **"** and **Space**.

  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** button to save your new password.

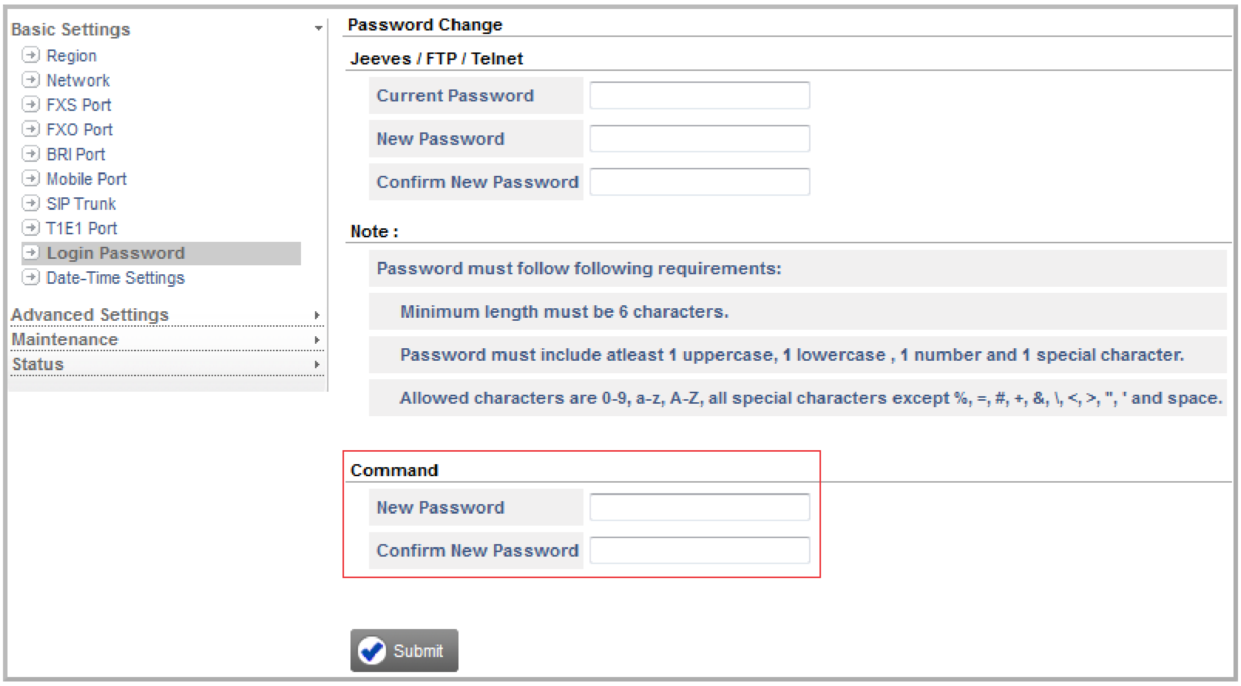
- *Password for Jeeves is case-sensitive.*
- *When you default the system, Jeeves Password will not be set to default.*

Login Password for System Commands

To configure system by dialing System Commands, you must enter the SE mode using the Command Password. This Password must not be less than 4 digits and can be of upto 8 digits. Digits 0-9 are allowed. The default Command Password is **1234.** You may change this Password using Jeeves.

To change the Command Password:

- Click the **Basic Settings** link to expand.
- Click the **Login Password** link.



Under **Command**,

- Enter **New Password**.
- In **Confirm New Password**, re-enter the new password to confirm.
- Click **Submit** to save.

When you default the system, Command Password will not be set to default.

Forgot the Login Password?

If you have already changed the default Jeeves/Command Password (1234) and are unable to recall or locate it, you can restore the default Jeeves/Command Password using the Jumper.

Restoring Default Login Password

To restore the Default Jeeves/Command Password by changing the Jumper **(J1)** Settings on the CPU Card,

- Switch off the power supply.
- Remove the CPU Card from the slot.
- Locate Jumper **(J1)** on the PCB of the CPU Card.
- Change the position of the Jumper from ***BC*** to ***AB*** (that is, from 'Normal' to 'Reset Login Password')
- Reinsert the CPU Card into the slot.
- Switch ON the system and wait for the system to initialize.
- Switch OFF the system and remove the CPU Card from the slot.
- Change the Jumper position from ***AB*** to the original position ***BC.***

- Insert the CPU Card back into the slot.
- Switch ON the system.

The Jeeves/Command Password will be restored to the default value, 1234.

When you change the jumper positions to restore default Jeeves/Command Password (1234),a few other parameters will also be set to default. See “Restoring Default Settings by changing the Jumper Position” for details.

Date-Time Settings

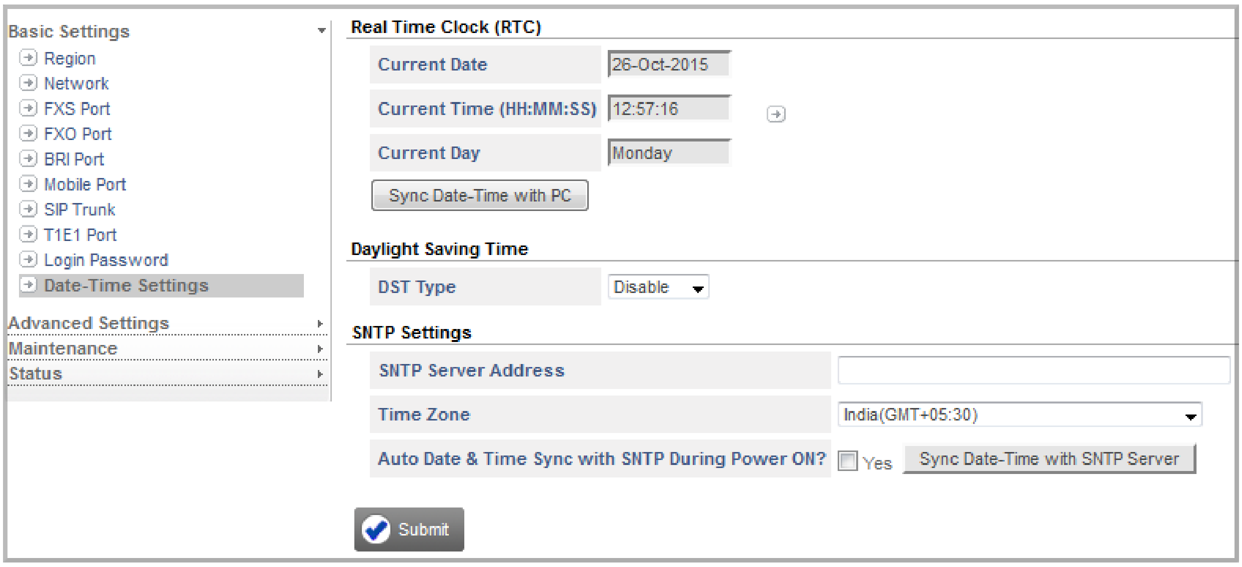
Real Time Clock

SARVAM UMG has a Real Time Clock (RTC) to store date and time. When you select the Region, the RTC parameters are set automatically.

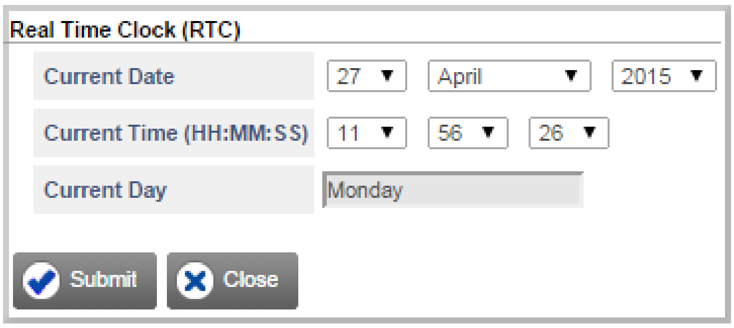
However, the RTC can drift over a long period. So, you may check and reset the RTC values at regular intervals to correct this drift.

To set the Real Time Clock,

- Click the **Basic Settings** link to expand.
- Click the **Date-Time Settings** link.
- The **Real Time Clock** parameters appear on your screen.



- Under **Real Time Clock (RTC)**, click **Settings** of the **Current Time (HH:MM:SS)**.
    - A new window opens.



- Set the **Current Date** in date-month-year format.

- Set the **Current Time** in hours-minutes-seconds format.

  The current day will be displayed automatically for the date and time you set.

- Close the window.

- Click **Submit** to save RTC settings.
- Click the **Sync Date-Time with PC** button, if you want to sync the system's date and time with that of your PC.

Daylight Saving Time

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[17] use it, though the start and end dates of DST vary by location and year.

SARVAM UMG supports Daylight Saving Time adjustment to enable you to set the Date and Time[18] of SARVAM UMG forward and backward according to the DST convention followed in your country.

You can set DST by: **Day and Month** or **Date and Month**.

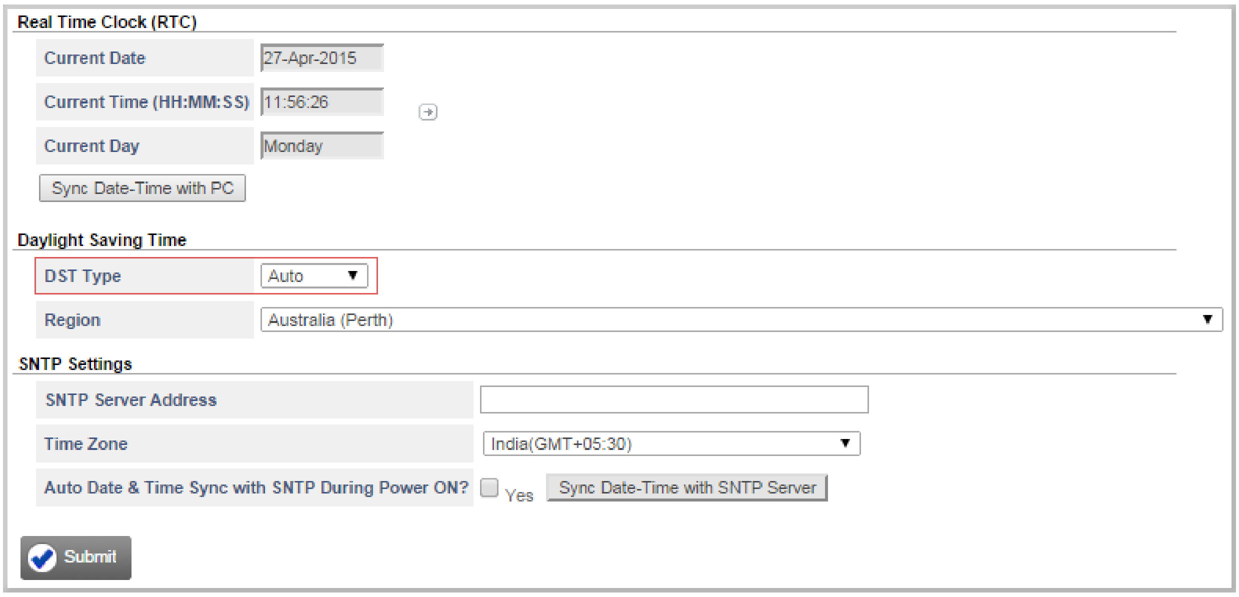
When SARVAM UMG is set to default, your DST settings will remain unchanged.

To configure DST,

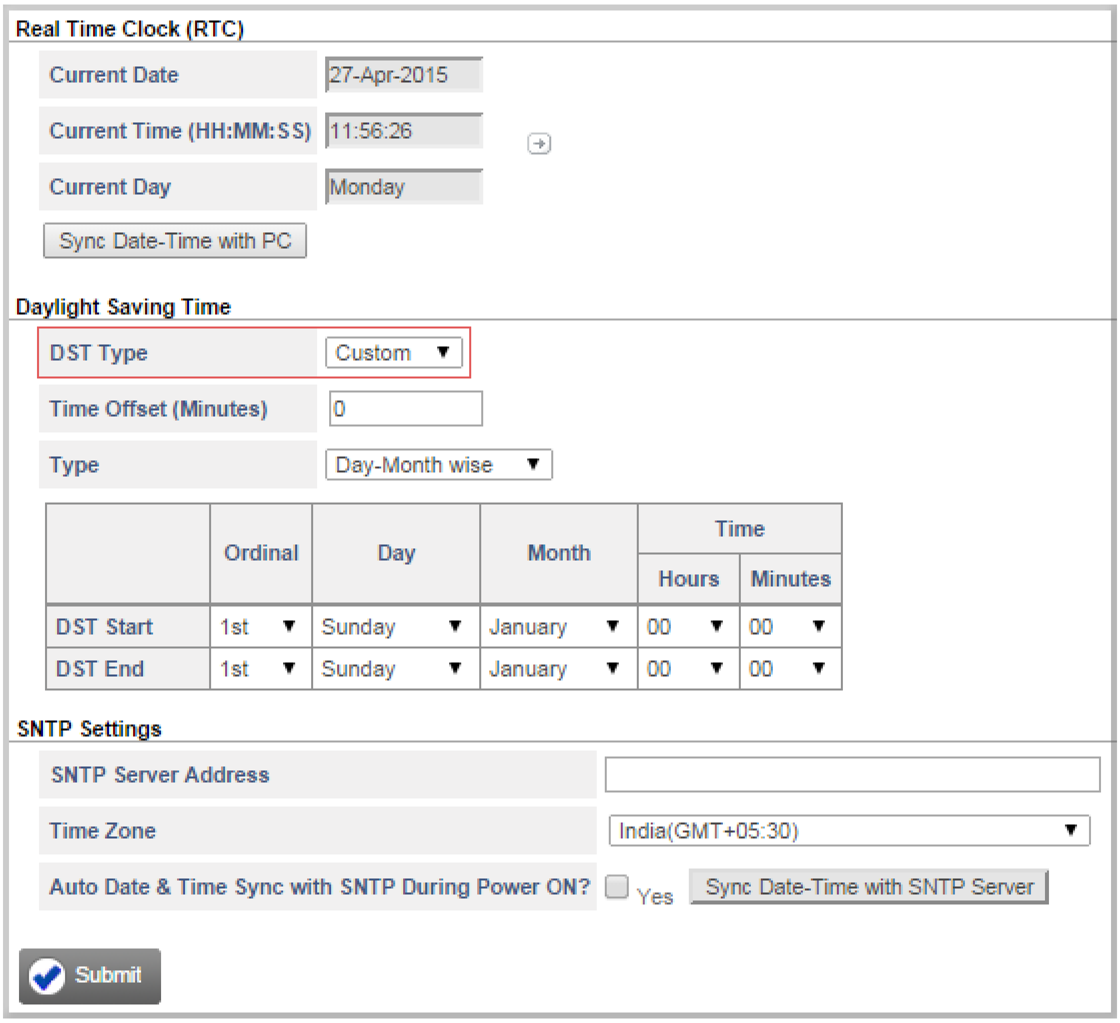
- Click the **Basic Settings** link to expand.
- Click the **Date and Time Settings** link.
- Go to **Daylight Saving Time** and do the following:
  - Select the **DST Type**. You may select **Auto** or **Custom**. If you do not want to apply DST select Disable. Default: Disabled.

17. *In most countries in Asia and Africa, and in certain countries of South America, DST is not observed.*
18. *SARVAM UMG sets its Date and Time according to the **Time Zone** you selected, and synchronizes the time according to the **SNTP Server** you selected. See* *"Region".*

- If you select **Auto**, you must select the **Region**. DST will be set automatically for the region you select.



- If you select **Custom**, you must configure the Time Offset and choose whether you want the DST to be applied by Day and Month or by Date and Month and define the DST Start and End time.



- In **Time Offset**, enter the time in minutes which the system should consider to forward the clock at the start of DST and to set the clock back when DST ends. Default: 60 minutes.

- Select the desired **Type** of DST as:

  - **Day-Month Wise**, if the DST in your country starts and ends on a particular day of the month. For example, if DST starts on the Second Sunday of March and ends on the First Sunday of October.

  –or–
  - **Date-Month Wise**, if the DST in your country starts and ends on a particular date of the month. For example, if DST starts on October 12 and ends on March 15.

  Default: Day-Month Wise.

- If you select **Day-Month Wise** option, you need to configure the Start and End time for DST.

  DST Start
  - Select the **Ordinal** day of the month when DST begins: 1st, 2nd, 3rd, 4th or 5th.
  - Select the **Day** of the month when DST begins: Sunday, Monday, Tuesday, Wednesday, Thursday, Friday, Saturday.
  - Select the **Month** when DST begins: January to December.
  - Set the **Time** when you want DST to begin in 24 hours format.

  Default: 1st Sunday March, Time 00 hours and 00 minutes.

  DST End
  - Select the **Ordinal** day of the month when DST ends: 1st, 2nd, 3rd, 4th or 5th.
  - Select the **Day** of the month when DST ends: Sunday, Monday, Tuesday, Wednesday, Thursday, Friday, Saturday.
  - Select the **Month** when DST ends: January to December.
  - Set the **Time** when you want DST to end in 24 hours format.

  Default: 1st Sunday September, Time 00 hours and 00 minutes.

When the DST of a particular country starts or ends on the Last Sunday or any other day, for instance, the last Tuesday, last Friday of the month, always set the Ordinal Number as '5th'.

- If you select **Date-Month Wise** option, configure the following parameters:

  DST Start
  - Select the **Month** when DST begins — January to December.
  - Select the **Date** on which DST begins — 1 to 31.
  - Set the **Time** when DST begins in 24 hours format.

  DST End
  - Select the **Month** when DST ends — January to December.
  - Select the **Date** on which DST ends — 1 to 31.
  - Set the **Time** when DST ends in 24 hours format.

- Click **Submit** to save your DST settings.

Example: If you are installing SARVAM UMG in a country in the European Union, as per the European Summer Time, the DST would start on the Last Sunday in March and end on the Last Sunday in October each year. Clocks are advanced by one hour at 01:00 hours GMT at the start of DST and set back by one hour at 01:00 hours GMT when DST ends. Let us take the example of setting DST, if SARVAM UMG were installed in Berlin, Germany. In the year 2011, the DST in Berlin starts on Sunday, 27 March at 02:00:00 hours and ends on Sunday 30 October at 03:00:00 hours. To set DST you must do the following:

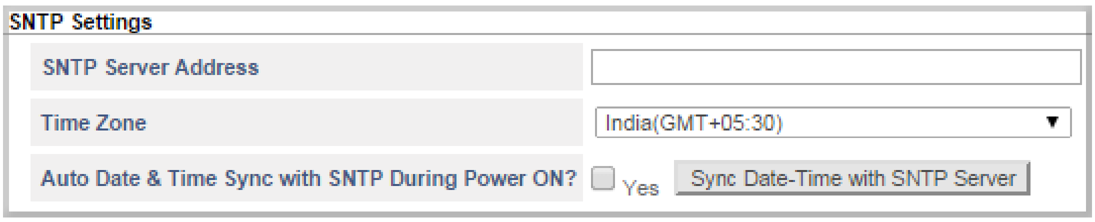
1. Select the **DST Type** as **Custom**.
2. Set the **Time Offset** as 60 minutes.
3. Select the option **Date-Month Wise** as **Type**[19].
4. Configure the **DST Start** as follows:
   - Select **March** as the **Month.**
   - Select **27th** as the **Date**.
   - Set **Time** to 01:59:59
5. Now, go to the option **DST End**, and configure as follows.
   - Select **October** as the **Month**.
   - Select **30th** as the **Date**.
   - Set **Time** to 02:59:59.
6. Click **Submit** to save DST settings.

On Sunday 27 March at 01:59:59 the SARVAM UMG will set the clock forward by 1 hour. On Sunday 30 October, SARVAM UMG sets the clock back by 1 hour at 02:59:59.

SNTP Settings

To use SNTP for synchronizing with the Real Time Clock,

- Under **Basic Settings**, click the **Date and Time Settings** link.
- Go to **SNTP Settings** on this page.



- In **SNTP Server Address**, enter the Time Server Address. The SNTP Server address can be of maximum 40 characters. Default: Blank.
- By default, the time zone for the country/region where SARVAM UMG is installed is automatically selected when you select ‘Region’. If required you may change the time zone by selecting the desired country/region from the **Time Zone** list. Default: India (GMT+05:30).
- If you want the system to synchronize date and time with the SNTP server automatically at Power On, select the **Auto Date and Time Sync with SNTP during Power ON?** check box.

19. You can also select Day-Month-wise as Type.

At every power ON, SARVAM UMG will synchronize its date and time with the Time Server address you have entered as SNTP Server Address.

By default, Auto Date and Time Sync with SNTP during Power ON is disabled.

- To synchronize date and time of SARVAM UMG with the SNTP server whenever required, click the **Sync Date and Time Server with SNTP** button.
- Click **Submit** to save the changes.

CHAPTER 9

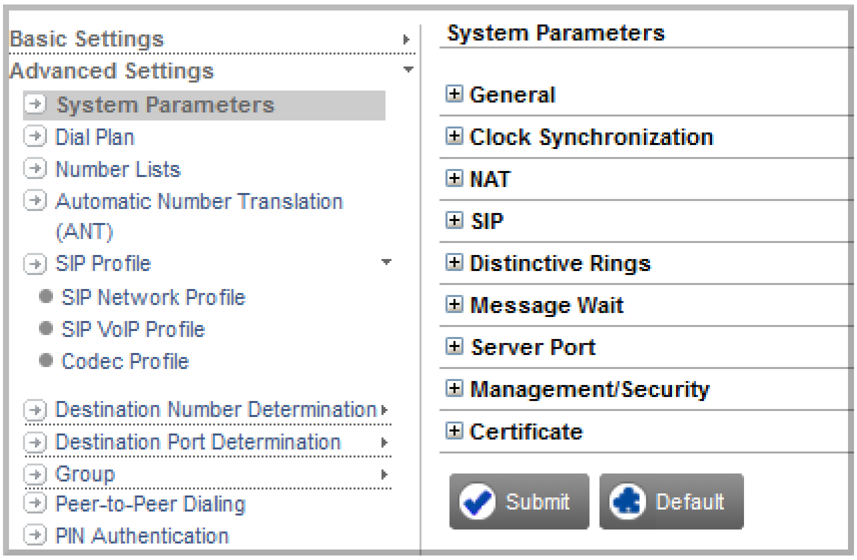
Advanced Settings

System Parameters

System Parameters are general parameters, related to features and facilities that are applied system-wide, such as System Name, NAT and SIP related parameters, Server Port, Certificates, DMZ etc.

To configure the System Parameters,

- Click the **Advanced Settings** link to expand.
- Click the **System Parameters** link.
- The **System Parameters** page opens.



General Parameters

- Click **General** to expand and configure the following.

- In **System Name,** enter a name that you wish to assign to SARVAM UMG. Naming SARVAM UMG will serve as an identifier when there are more than one SARVAM UMGs connected in the same LAN network.

  Valid range is 40 characters. Default: Blank.

- To use the IP Dialing feature, that is, to directly dial IP Addresses, in **SIP Trunk for IP Dialing** select a SIP Trunk or SIP Group for routing the call to the IP Address.

  The valid range for the SIP Trunk is 001 to 250 and for SIP Group is 1 to 9.

  When you assign a SIP Trunk, make sure you have enabled the SIP Trunk and configured its necessary parameters. For further details, see “SIP Trunk” under *Basic Settings.*

  When you assign a SIP Group, make sure you have configured the SIP Group. For further details, see “Group”.

  Default: SIP Group 1

  To know more about this feature, see “IP Dialing”.

- Select the **Play Routing Tone** check box, if you want the system to play routing tone while routing the call to the destination port. During an outgoing call, the routing tone indicates that the call is in progress. Default: Disabled.

- Select the **Call Release Timer** check box, if you want the system to release the ports involved in a call after a definite time period. Default: Enabled.

  This timer is loaded when a call gets matured and stops whenever a port involved in a call is released.

  - Set the duration of the **Release Timer**. Valid range is 001 to 999 minutes. Default: 999 minutes.

- Set the duration of the **Routing Group Busy Wait Timer**. This timer defines the duration for which you want SARVAM UMG to search for a free destination port in the Routing Group and the Fallback Routing Group in order to route and place the call. This timer is loaded when no destination port is free in both the Routing Group and the Fallback Routing Group. Valid range is 1 to 99 seconds. Default: 1 second.

- Set the duration of the **Transfer Notification Timer**. This timer defines the duration for which you want SARVAM UMG to wait for the notification regarding the status of the transferred call. It notifies whether the transfer target is busy, the call has been answered or the call has been disconnected etc.

  This timer is loaded when a user performs a transfer activity. The user (transferor) is notified of the status of the transfer activity within the time period you have set for this timer.

  Valid range is 1 to 999 seconds. Default: 60 seconds.

- Set the duration of the **Ring Timer**. This timer defines the duration for which you want SARVAM UMG to play a ring on the FXS Port to indicate the incoming call. This timer is loaded when the call is placed on the FXS Port, that is, either there is a ring event on the FXS Port or call waiting beeps are played when the FXS Port is busy. Valid range is 1 to 99 seconds. Default: 45 seconds.

- Set the duration of the **Error Tone Timer**. This timer defines the duration for which you want SARVAM UMG to will play the Error Tone. Valid range is 0 to 9 seconds. Default: 7 seconds.

- Set the duration of the **Error Tone Delay Timer**. This timer defines the duration for which you want SARVAM UMG to play the Error Tone whenever a call is disconnected during speech. Valid range is 00 to 99 seconds. Default: 0 seconds.

- Set the duration of the **Unconnected Calls Record Delete Timer**. This timer defines the duration for which you want SARVAM UMG to route those calls which have been returned unconnected from — BRI (Terminal) Port, T1E1 Port (T1 line), Mobile Port or SIP Trunk. These calls are routed back to the original callers. Default: 999 minutes.

  This timer will work only if you have enabled the **Route calls returned unconnected to Original Caller** check box under *Handling of Outgoing Calls* in the BRI (Terminal) Port, T1E1 Port (Terminal line), Mobile Port or the SIP Trunk.

  This timer is used whenever an outgoing call made using the port on which this feature is enabled is either found busy or unanswered. SARVAM UMG stores the numbers of the Calling Party, Called Party and the source port through which the outgoing calls were made.

The records of 200 such Unconnected Calls are stored using FIFO method. These records are deleted either on the expiry of this timer or when the incoming call from the called party is placed to the original calling party before the expiry of this timer.

You can also delete the records of unconnected calls without waiting for this timer to expire. To do this, click the **Clear Unconnected Call Records** button.

- Select the **Replace '+' from the CLI** check box, if you want the system to remove the prefix '+' from the CLI received. Default: Disabled.

    - In **Replace '+' from CLI with the number string**, enter the number string with which you want to replace '+' in the CLI received. Keep the number string blank to remove the '+' sign without adding any other prefix to it. Default: Blank.

      For example:
      The number string +919327237228 is received as CLI.
      If '00' is configured as the replacement string, the CLI number would be presented as 00919327237228.
      If no replacement string is configured, the CLI number would be presented as 919327237228.

- Select the **Remove Country Code from CLI received** check box, if you want the system to remove the country code from the CLI received on the source port before presenting it to the destination port. Default: Disabled.

  Make sure you have configured the "Country Code" under *Region* in the *Basic Settings*.

- Select **Display Configuration** as *Active,* if you want the system to display only those trunks and extension ports that are detected by it at Power-On for configuration.

  Select *Default,* if you want the system to display all the supported trunks and extension ports.

  Default: Default.

- Select the **Apply RCOC only if the caller calls back on the same trunk from which the call was made** check box, if you want SARVAM UMG to match the Trunk Port Parameters (Trunk Port Number and Type) of an incoming call with the entry in the RCOC table while applying RCOC logic on the "SIP Trunk", "BRI Port - Terminal", "E1 Port", "T1 Port" and "Mobile Port". Default: Disabled.

  If this check box is enabled, SARVAM UMG will match Trunk port parameters of the incoming call with the entry stored in RCOC table. If a match is found, it will route the incoming call to original caller.

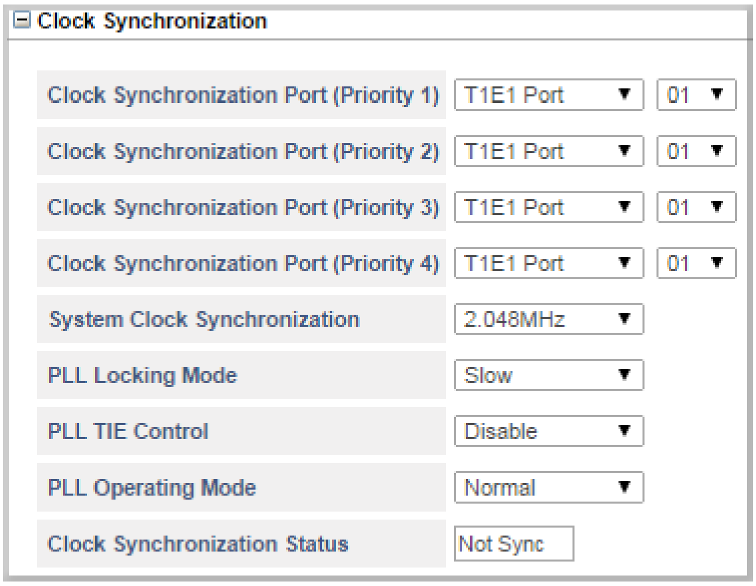
Clock Synchronization

When SARVAM UMG transmits or receives the data from the external lines, there must be proper synchronization between the transmitter and receiver. In case of improper synchronization, clock slips can occur. A clock slip can alter the data stream which could result in either the loss of data or the addition of unwanted noise in the data.

Clock Synchronization can be done in three ways— using the data clock, using the external clock (clock is sent by the network on a dedicated cable pair) or using the internal clock. SARVAM UMG does not support the external clock. When the SARVAM UMG is connected to the PSTN, then it is recommended to extract the clock from the incoming data whereas if the SARVAM UMG is used to form a part of a private network, you are recommended to use the internal clock. For example, if a private network is formed by connecting three SARVAM UMG, then one system should be programmed as master clock whereas other two should be programmed in the slave mode.

If two or more T1E1/BRI Ports are connected to the PSTN (or a Private Network) then in such case, clock will be extracted from the first T1E1/BRI Software port.

- Click **Clock Synchronization** to expand and configure the following.



- In **Clock Synchronization Port (Priority 1)**, select the desired Port — BRI, T1E1 or None — which you want as the first priority and also select the respective Port Number.

  Similarly, configure the **Clock Synchronization Port (Priority 2)**, **Clock Synchronization Port (Priority 3)** and **Clock Synchronization Port (Priority 4)**.

- In **System Clock Synchronization,** select the desired frequency for the synchronization. You can select — 8 KHz Derived, 8 KHz, 2.048 MHz or 1.54 MHz.

  Select System Clock Synchronization option as **8 KHz**, only for BRI Port.

  Select System Clock Synchronization option as **2.048 MHz**, only for T1E1 Port (E1 line).

  Select System Clock Synchronization option as **1.54 MHz**, only for T1E1 Port (T1 line).

  Default: 2.048 MHz for all the countries excluding USA. For USA, it is 1.54 MHz.

The system will restart when the frequency is changed.

- In **PLL Locking Mode**, select the speed required for clock synchronization. You can select — Fast or Slow. Default: Slow.

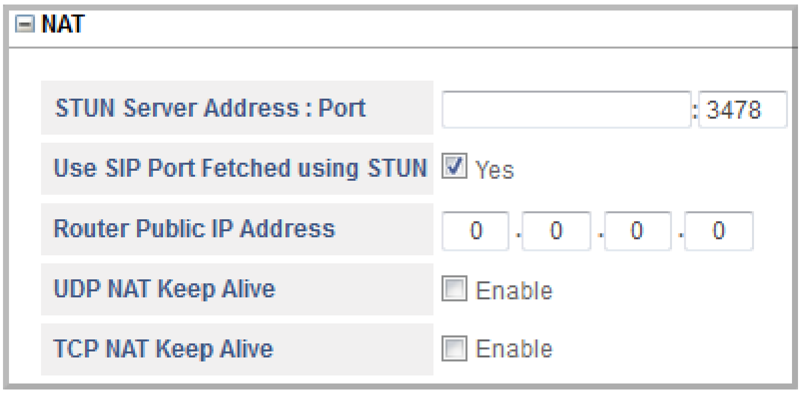
- In **PLL TIE Control**, you can select — Enable or Disable — as per your requirement. Default: Disabled.

- In **PLL Operating Mode**, you can select — Normal, Hold Over or Free Run — as the operating mode as per your requirement. Default: Normal

- In **Clock Synchronization Status**, the synchronization status of the clock is displayed.

NAT

- Click **NAT** to expand and configure the following.



- In **STUN Server Address: Port**, enter the STUN Server Address and the Listening Port of the STUN Server.

  STUN (Simple Traversal of UDP through NAT) server facilitates the traversing through most of the NATs, except the symmetric NATs. Configure this only when SARVAM UMG is located behind a NAT router which is not symmetric.

  The STUN Server Address can have maximum 40 characters.

  Valid range is 1024 to 65535. Default: 3478.

- Clear the **Use SIP Port Fetched using STUN** check box, if SARVAM UMG is located behind the NAT router and you have forwarded the SIP Listening Port of the SARVAM UMG in the router.

  Keep the **SIP Port fetched using STUN** check box enabled, if you have *not* forwarded the SIP Listening Port in the router.

*Make sure you have selected **NAT Type** as **STUN** in the SIP Trunk. See “SIP Network Profile”.*

- In **Router's Public IP Address**, enter the public IP address of the NAT router behind which the system is located. Default: Blank.

  Configure this only when the system is located behind the NAT router and a Static IP Address is assigned as the Public IP Address of the Router.

*Make sure you have selected **NAT Type** as **Router's IP Address** in the SIP Trunk. See “SIP Network Profile”.*

- Select the **UDP NAT Keep Alive** check box, if you want NAT Keep Alive messages to be sent to refresh the binding in the NAT router when SARVAM UMG is connected behind a NAT router and the SIP messages are transported over UDP. Default: Disabled.

  - Select the type of **Keep Alive Message** to be sent. You can select — REGISTER or NOTIFY as per your requirement. Default: NOTIFY.

    - In **Interval**, set the time period after which you want the system to send Keep Alive messages. This time period should be less than the NAT binding timer of the router. Valid range is 001 to 999 seconds. Default: 120 seconds.

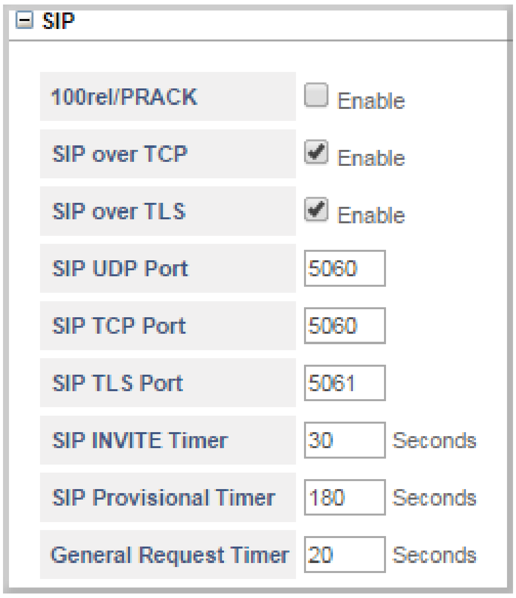
  - Select the **TCP NAT Keep Alive** check box, if you want NAT Keep Alive messages to be sent to refresh the binding in the NAT router when SARVAM UMG is connected behind a NAT router and the SIP messages are transported over TCP. Default: Disabled.

    - In **Interval**, set the time period after which you want the system to send Keep Alive messages. This time period should be less than the NAT binding timer of the router. Valid range is 001 to 999 seconds. Default: 120 seconds.

- Click **Submit** to save changes.

SIP

- Click **SIP** to expand and configure the following.



  - Select **100rel/PRACK** check box, if you want SARVAM UMG to use 100rel SIP trunk for reliable transmission of SIP provisional responses and PRACK for Provisional Acknowledgement. Default: Disabled.

  - Select **SIP Over TCP** check box, if you want SARVAM UMG to receive SIP messages over TCP. Default: Enabled.

    SARVAM UMG supports transporting of SIP messages over User Datagram Protocol (UDP) as well as Transfer Control Protocol (TCP) connection. Despite the advantages that SIP Over TCP offers, it is more common to use UDP to transport the SIP messages.

    Make sure that you have selected *TCP* as the *SIP Transport* option and have enabled the *TCP (Fallback to UDP)* check box. See *"Advanced"* in *"SIP Network Profile"*.

  - Select **SIP Over TLS** check box, if you want SARVAM UMG to receive SIP messages over TLS. Default: Enabled.

SARVAM UMG supports transporting of SIP messages over TLS. TLS protects SIP signaling against loss of integrity, confidentiality and replay.

Make sure that you have selected *TLS* as the *SIP Transport* option. See *"Advanced"* in *"SIP Network Profile"*.

- Configure the **SIP UDP Port.** This is the port on which SARVAM UMG listens for SIP messages transported over UDP. This port is also used as the source port for sending SIP messages to the remote peer. Valid range is 1031 to 65534. Default: 5060.

- Configure the **SIP TCP Port.** This is the port on which SARVAM UMG listens for SIP messages transported over TCP. This port is also used as the source port for sending SIP messages to the remote peer. Valid range is 1031 to 65534. Default: 5060.

- Configure the **SIP TLS Port.** This is the port on which the SARVAM UMG listens for SIP messages transported over TLS. This port is also used as the source port for sending SIP messages to the remote peer. Valid range is 1031 to 65534. Default: 5061.

- Set the duration of the **SIP INVITE Timer**. This timer defines the time period for which you want SARVAM UMG to wait for the response from the called party after sending the INVITE message. This timer starts after sending INVITE message to the called party and stops after receiving the provisional or the final response or when the call gets disconnected.

  On expiry of the timer, SARVAM UMG terminates the call process and gives an error tone to the user. Valid range is 10 to 200 seconds. Default: 30 seconds.

- Set the duration of the **SIP Provisional Timer**. This timer defines the time period for which you want SARVAM UMG to wait for the final response after receiving the provisional response from the called party. This timer starts when provisional response is received from the called party and stops after receiving the final response from the called party or when then call gets disconnected.

  On the expiry of the timer, SARVAM UMG terminates the call process and gives error tone to the user. Valid range is 10 to 200 seconds. Default: 180 seconds.

- Set the duration of the **General Request Timer**. This timer defines the time period for which you want SARVAM UMG to wait for the response to the transaction request. This timer starts when a transaction is initiated and stops after receiving the response to the request. On expiry of the timer, the SARVAM UMG clears the transaction. Valid range is 10 to 60 seconds. Default: 20 seconds.

- Click **Submit** to save changes.

If you have made any changes in the NAT or SIP Parameters, all the ongoing calls will be disconnected when you submit the page to save the changes.

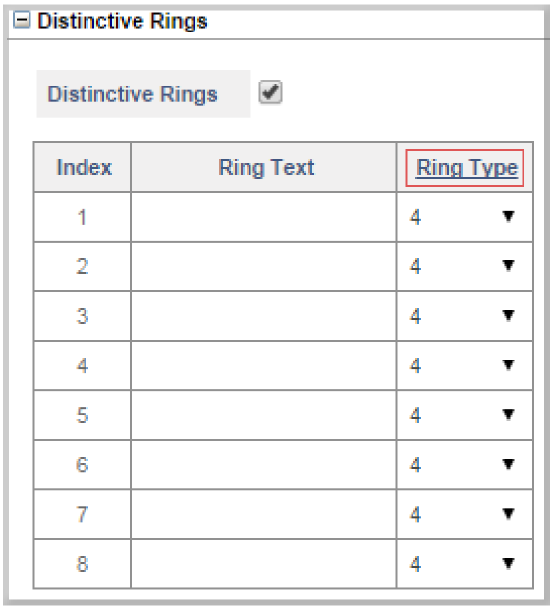
Distinctive Rings

Distinctive Rings are the ringing patterns used for distinguishing between different types of call events.

The distinctive ringing pattern is selected according to Alert-Info header that is included in INVITE message. For example: Alert-Info <Bellcore-dr2>, or Alert-Info <http://…/Bellcore-dr2>. 'dr2' defines that ringing pattern number 2 will be played on the FXS Port. If the Alert-Info header is missing, the default ring tone will be played on the FXS Port.

- Click **Distinctive Rings** to expand and configure the following.

  - Select the **Distinctive Rings** check box to enable.
  - In **Ring Text**, enter the Ring Text that you expect to receive in the Alert-Info header of the INVITE message during an incoming call. The Ring Text can be a maximum of 24 characters.
  - To assign the Ring Type,
    - Click on **Ring Type**.



- The **Ring Type** window opens.

Ring Type

Ring Type	Ring Cadence						Supported Country
	ON Time 1 (msec)	OFF Time 1 (msec)	ON Time 2 (msec)	OFF Time 2 (msec)	ON Time 3 (msec)	OFF Time 3 (msec)	
1	Infinite						
2	750	750	0	0	0	0	
3	500	1500	0	0	0	0	
4	750	2250	0	0	0	0	
5	1500	500	0	0	0	0	
6	1000	4000	0	0	0	0	Brazil, Greece, Italy, Netherland, Switzerland, Finland, Germany
7	2000	4000	0	0	0	0	Egypt, USA, Canada, Namibia
8	400	200	400	2000	0	0	Australia, India, Singapore, South Africa, UK, Ireland, Malaysia
9	400	200	400	200	400	2000	
10	1000	2000	0	0	0	0	Japan
11	1000	3000	0	0	0	0	China, korea, Russia, Belgium, Taiwan
12	1000	5000	0	0	0	0	Portugal, Sweden
13	1500	3000	0	0	0	0	Spain
14	1500	3500	0	0	0	0	France
15	2000	3000	0	0	0	0	Israel, New Zealand, Poland, Thailand, UAE, Czechia, Norway, Hongkong, Austria, Hungary, Slovakia
16	3500	5500	790	1100	0	0	

Close

The Ring Type numbers, Ring Cadence and the corresponding Supported Countries are displayed.

- Take a note of the Ring Type number you wish to configure for each Ring Text.
- Close the **Ring Type** window.

- Select the desired **Ring Type** number for each Ring Text.

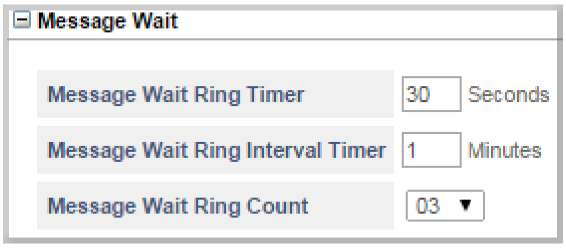
Keep the Distinctive Rings check box disabled, if you want SARVAM UMG to play country-specific Ring Type on the FXS Port. For details, see "Region".

- Click **Submit** to save changes.

Message Wait

This parameter is related to the "Message Wait Indication on SIP Trunks" feature. If you have selected *Message Wait Notification* type as *Ring* on the FXS Port, you may also configure the following parameters related to Message Wait Ring.

- Click **Message Wait** to expand and configure the following.



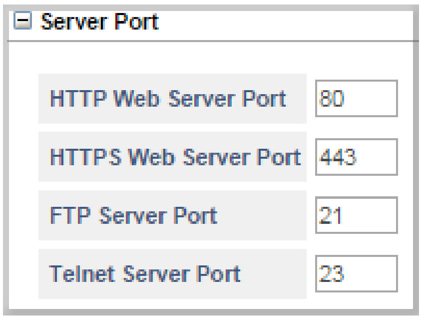
  - Set the duration of the **Message Wait Ring Timer.** This timer defines the time duration for which the Message Wait Ring will be played on the FXS Port for Message Wait Notification. Valid range is 01 to 60 seconds. Default: 30 seconds.

  - Set the duration of the **Message Wait Ring Interval Timer**. This timer defines the time duration after which SARVAM UMG will play the Message Wait Ring again on the FXS Port for the Message Wait Notification, if the previous ring was unanswered by the user. Valid range is 001 to 999 minutes. Default: 1 minute.

  - In **Message Wait Ring Count**, enter the number of times you want SARVAM UMG to play the Message Wait Ring on the FXS Port, until it is answered by the FXS Port user. Valid range is 1 to 10. Default: 3.

Server Port

- Click **Server Port** to expand and configure the following.



  - Configure the **HTTP Web Server Port.** SARVAM UMG has an embedded web server called *Jeeves*, for the system configuration. You can access *Jeeves* using HTTP. You can change it as per your requirement. Valid range is 80/ 1031 to 65535. Default: 80.

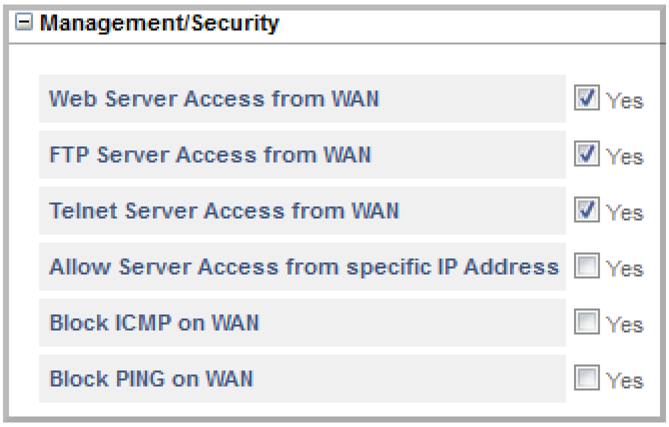
  - Configure the **HTTPS Web Server Port.** You can access *Jeeves* of SARVAM UMG using HTTPS. You can change it as per your requirement. Valid range is 443/ 1031 to 65535. Default: 443.

  - Configure the **FTP Server Port.** SARVAM UMG has an embedded FTP server for Software Upgrade. You can change it as per your requirement. Valid range is 21/ 1031 to 65535. Default: 21.

  - Configure the **Telnet Server Port.** You can access SARVAM UMG using Telnet. You can change it as per your requirement. Valid range is 23/ 1031 to 65535. Default: 23.

Management/Security

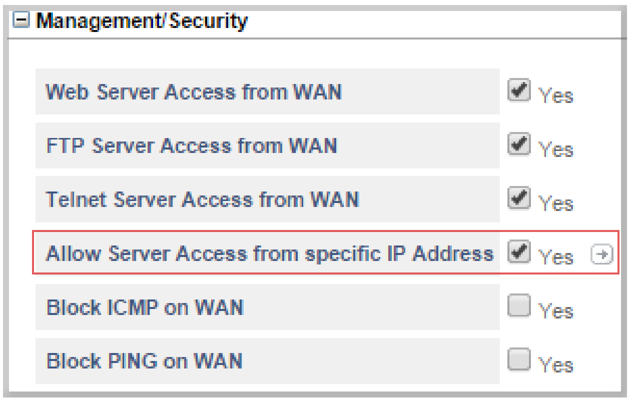
- Click **Management/Security** to expand and configure the following.



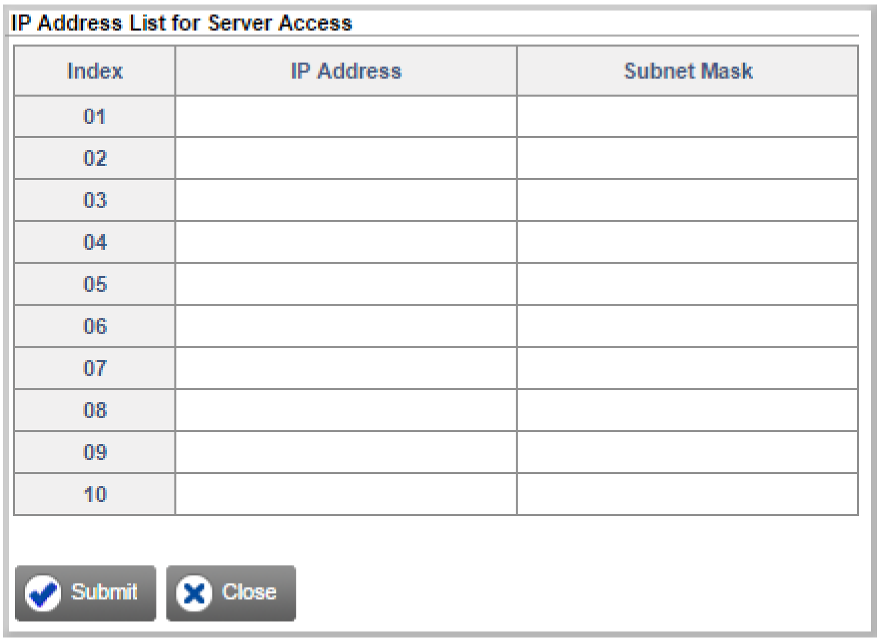
- Select the **Web Server Access from WAN** check box, if you want to allow users to access the system's Web Server (Jeeves) from the WAN Port. Default: Enabled.
- Select the **FTP Server Access from WAN** check box, if you want to allow users to access the system's FTP Server from the WAN Port. Default: Enabled.
- Select the **Telnet Server Access from WAN** check box, if you want to allow users to access the system using Telnet from the WAN Port. Default: Enabled.
- Select the **Allow Server access from specific IP Address** check box, if you want to allow access of the system to specific users only. Default: Disabled.

  If you enable this parameter, you must configure the IP Address table for Server Access.

  To configure the IP Address table, Click **Settings**.



The **IP Address List for Server Access** window opens. You can store 10 entries in this table.


IP Address List for Server Access

Index	IP Address	Subnet Mask
01		
02		
03		
04		
05		
06		
07		
08		
09		
10		

- Enter the IP Addresses and their respective Subnet Mask in the table.
- Click **Submit** and close the window.

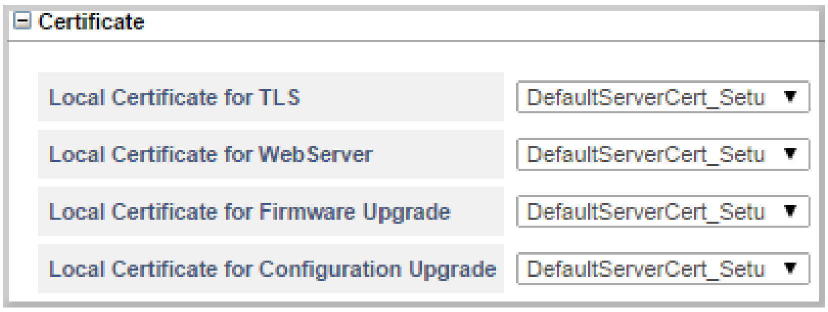
SARVAM UMG will allow system access only to those users whose IP Address matches with the one configured in the IP Address List for Server Access.

- Select the **Block ICMP on WAN** check box, if you want the system to discard the ICMP packets received on WAN. Default: Disabled.
- Select the **Block PING on WAN** check box, if you want the system to discard the PING request received on WAN. Default: Disabled.

Blocking of PING on WAN will prevent your network from being pinged or detected by other Internet users to acquire your IP Address.

Certificate

- Click **Certificate** to expand and select the certificate for each of the following.



- In **Local Certificate for TLS**, select the certificate to be used by the system for TLS.

- In **Local Certificate for Web Server**, select the certificate to be used by the system for accessing the Web Server.
- In **Local Certificate for Firmware Upgrade**, select the certificate to be used by the system for Firmware Upgrade.
- In **Local Certificate for Configuration Upgrade**, select the certificate to be used by the system for Configuration Upgrade.

To create as well as Upload/ Download Certificates, see “Certificate Manager”.

Dial Plan

SARVAM UMG supports 8 Dial Plans with total 64 entries in each table. The Dial Plan contains a series of digits and/or wildcard characters.

When a user dials a number, it is compared with the Destination Number configured in the Dial Plan. If a match is found, the system routes the call immediately without waiting for End of Dialing and if a match is not found, the system will wait for the End of Dialing and then route the call as per the Destination Port Selection method configured.

Dial Plan will be applied on the — FXO Port, Mobile Port, SIP Trunk, BRI Port (Terminal) and T1E1 Port (Terminal) — when,

- the Destination Number Selection method used for routing the call is **Answering the call and collecting the digits**.

  and

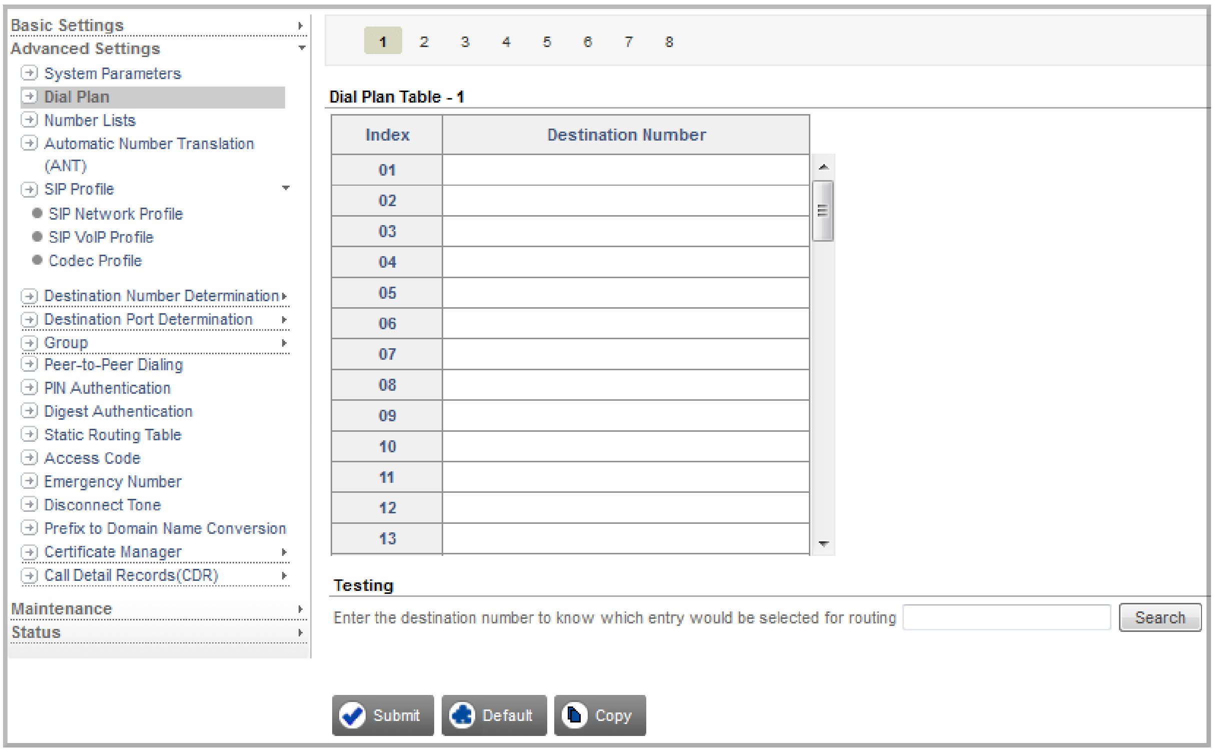
- the Destination Port Selection method is either **Fixed** or **Calling Number Based**.

Dial Plan will be applied on the — FXS Port, BRI Port (Network) and T1E1 Port (Network) — when,

- the Destination Port Selection method is either **Fixed** or **Calling Number Based**.

Configuring Dial Plan Table

- Click the **Advanced Settings** link.
- Click the **Dial Plan** link.



The Dial Plan Table allows you to configure upto 64 entries. Each entry is stored against an Index number.

For each entry,

- In **Destination Number**, enter the number you expect the callers to dial. You may enter upto 64 characters (Digits + "Wildcard Characters") in this field. Valid characters are 0 to 9, *, #, X, T, Comma [,], Hyphen [-], Caret [^]. Default: Blank.

- Click **Submit** to save.

If there are multiple entries in the Dial Plan table, to search a particular entry in the table, under Testing enter the desired number to know which entry would be selected for routing.

Wildcard Characters

SARVAM UMG 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 a Dial Plan can be configured.

Dial Plan Entry	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.

Number Lists

A Number List is a data structure that constitutes digit and character strings which must be configured for the system to support the features described in the following.

SARVAM UMG supports 24 Number Lists. Each Number List can contain upto 64 entries of a maximum of 24 characters each.

You need to configure Number Lists for the features described in the following. By default, each of these features is assigned particular Number Lists. You may retain the Number List assigned by default, or configure another Number List and assign this list to the feature.

Allowed - Denied Logic

You can apply the Allowed-Denied logic on a source port—FXS, FXO, SIP, BRI, Mobile, SIP and T1E1—if you want to allow or restrict the dialing of particular numbers. You can use this feature for Toll Control.

The Allowed-Denied logic makes use of two Number Lists:

- **Allowed Number List:** This is the list of numbers that can be dialed out from the source port.

- **Denied Number List:** This is the list of numbers that are to be restricted from being dialed out from the source port.

Both the lists must be first programmed separately for each port and then assigned to the respective port.

When Allowed-Denied Logic is enabled on a source port, for each number dialed from the port, SARVAM UMG uses the best-match-found logic to compare the dialed number with the Allowed Number List and the Denied Number List.

The number is allowed to be dialed, if the dialed number:
- matches with both lists.
- matches with Allowed Number List, but not with the Denied Number List.
- matches with neither the Allowed List nor the Denied List.

The number is denied, if it matches with the Denied Number List, but not with the Allowed Number List.

Allowed-Denied Number feature is not applicable in following cases:
- Destination number string matches with any Access Code.
- Destination number string matches with any Emergency Number.
- For Call Forward Number programmed.
- *Route all Incoming Calls (with CLI)* option selected is:
  - Fixed Destination Number
    - or -
  - on basis of Calling Party Number.

To apply this feature,

- you must configure the numbers you want to allow and restrict from being dialed out in the Allowed and Denied Number Lists.

By default, the following Number Lists are assigned for Allowed Denied Logic for each port type:

Port Type	Default Allowed Numbers List	Default Denied Numbers List
FXS Ports	List 05	List 06
FXO Ports	List 01	List 02
SIP Trunks	List 07	List 08
Mobile Ports	List 01	List 02
BRI Ports	List 01	List 02
T1E1 Ports	List 01	List 02

You may retain these lists or configure any other Number List from 01 to 24.

- enable **Allowed-Denied Logic** on the port type—FXS, FXO, SIP, Mobile, BRI, T1E1 — on which you want to apply this feature.
- configure the numbers you want to allow and the numbers you want to restrict in the default **Allowed Number List** and **Denied Number List** assigned to the port.

For instructions, see the following topics under *Basic Settings*:

"Handling of Incoming Calls"on "SIP Trunk"

"Handling of Incoming Calls" on "FXO Port"

"Handling of Incoming Calls" on "Mobile Port"

"Handling of Outgoing Calls" on "FXS Port"

"Handling of Calls" on "BRI Port - Network"

"Handling of Incoming Calls" on "BRI Port - Terminal"

"Handling of Incoming Calls" on "T1 Port"

"Handling of Incoming Calls" on "E1 Port"

If you do not want to use the default Number Lists assigned to the ports, you may select a different List Number and configure it. In this case, you must select the List Number you configured as the Allowed Number List/Denied Number List for the port.

Black Listed Callers

The Black Listed Callers feature enables you to block incoming calls from specific numbers and addresses on SIP Trunks and Mobile Ports. You can apply this feature on the Source Port only.

To use this feature,

- you must configure the numbers of unwanted callers in a Number List.

Make sure you have configured the full SIP URI (for example: 12345@abc.com) of the unwanted callers in the Blacklisted Callers Number List.

- enable the **Reject Calls from Blacklisted Callers** check box on the SIP Trunks and Mobile Ports on which you want to apply this feature.
- select the Number List you configured as **Black Listed Callers Number List**.

  For instructions, see the following topics under *Basic Settings*:

  - “Handling of Incoming Calls”on “SIP Trunk”
  - “Handling of Incoming Calls” on “Mobile Port”

Now, whenever there is an incoming call on the SIP Trunk or Mobile Port you have applied this feature, the SARVAM UMG will match the number with the Blacklisted Callers’ Number list you have assigned. If the number matches with any of the numbers you have blacklisted, the system will reject the call.

Make a list of numbers that you want to black list. Configure these numbers in a Number List. By default, Number List 16 is assigned as the Black Listed Callers List for the Mobile Ports and Number List 11 is assigned as the Black Listed Callers List for the SIP Trunks.

You may retain this list and configure all the numbers you want to black list in this list or you may configure different Number Lists for different ports and assign the lists to the ports.

Each number string in the List can have a maximum of 24 characters. If the callers' number exceeds 24 characters, the first 24 characters of the number will be checked. If the first 24 characters of the callers' number match perfectly with any of the numbers programmed in Blacklisted Callers List, the call will be rejected.

Call Detail Record Filters

SARVAM UMG enables you to generate reports of Call Detail Records using different filters. You can generate Call Detail Record report of calls made to specific numbers (Called Party Numbers) and calls received from specific numbers (Calling Party Numbers).

When you want to sort calls by Called Party and Calling Party Numbers, you must configure a Number list for each of these.

To generate Call Detail Records using Called Party and Calling Party Numbers as filters,

- make a list of Called Party Numbers and another list of Calling Party Numbers.
- configure a Number List with the Called Party Numbers and another Number List with the Calling Party Numbers.

  By default, Number list 01 is assigned for both Called Party and Calling Party numbers. You may retain this list and configure Called Party and Calling Party numbers in this list, or you may retain this for Called Party Numbers and configure another list number for Calling Party numbers. In this case you must assign the list you configured to the respective filter.

- assign the Called Party Number list you configured to the CDR filter **Called Party Number Matching with Number List**.
- assign the Calling Party Number list you configured to the CDR filter **Calling Party Number Matching with Number List**.

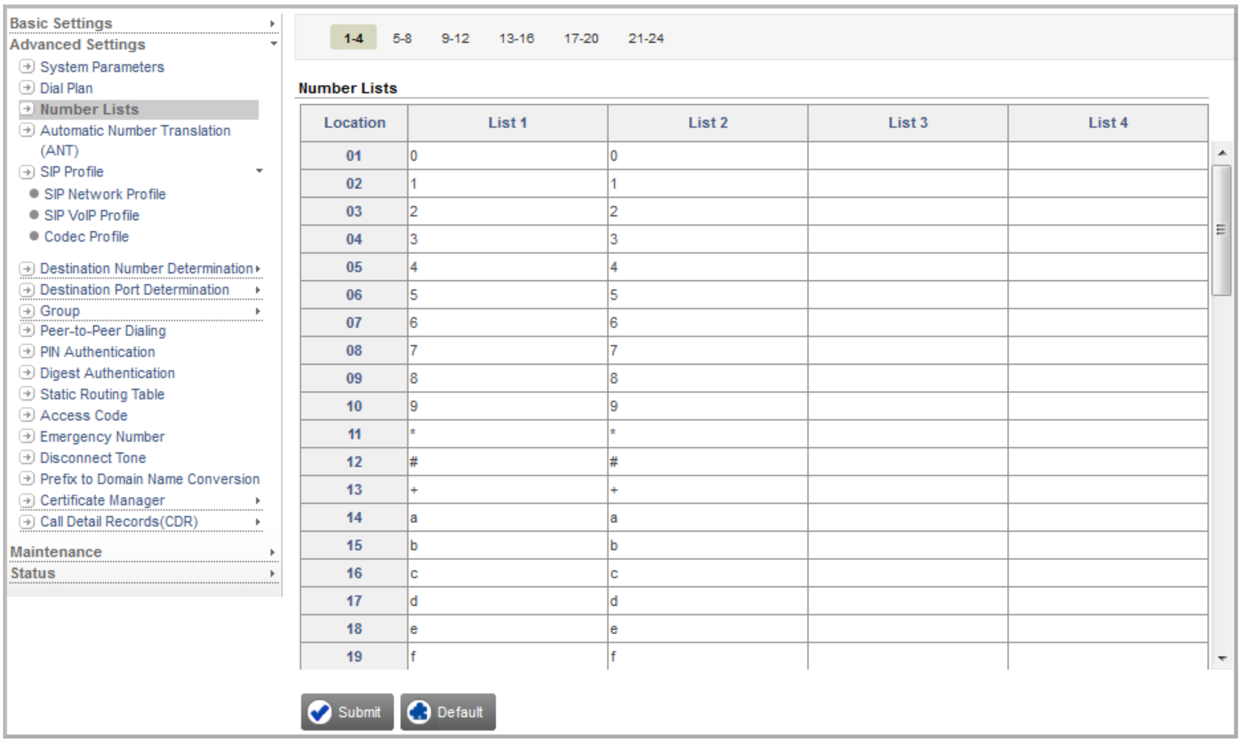
  For instructions, see "Call Detail Records (CDR)".

Configuring Number Lists

You must determine the purpose for which the list is required and accordingly prepare them.

To configure Number lists,

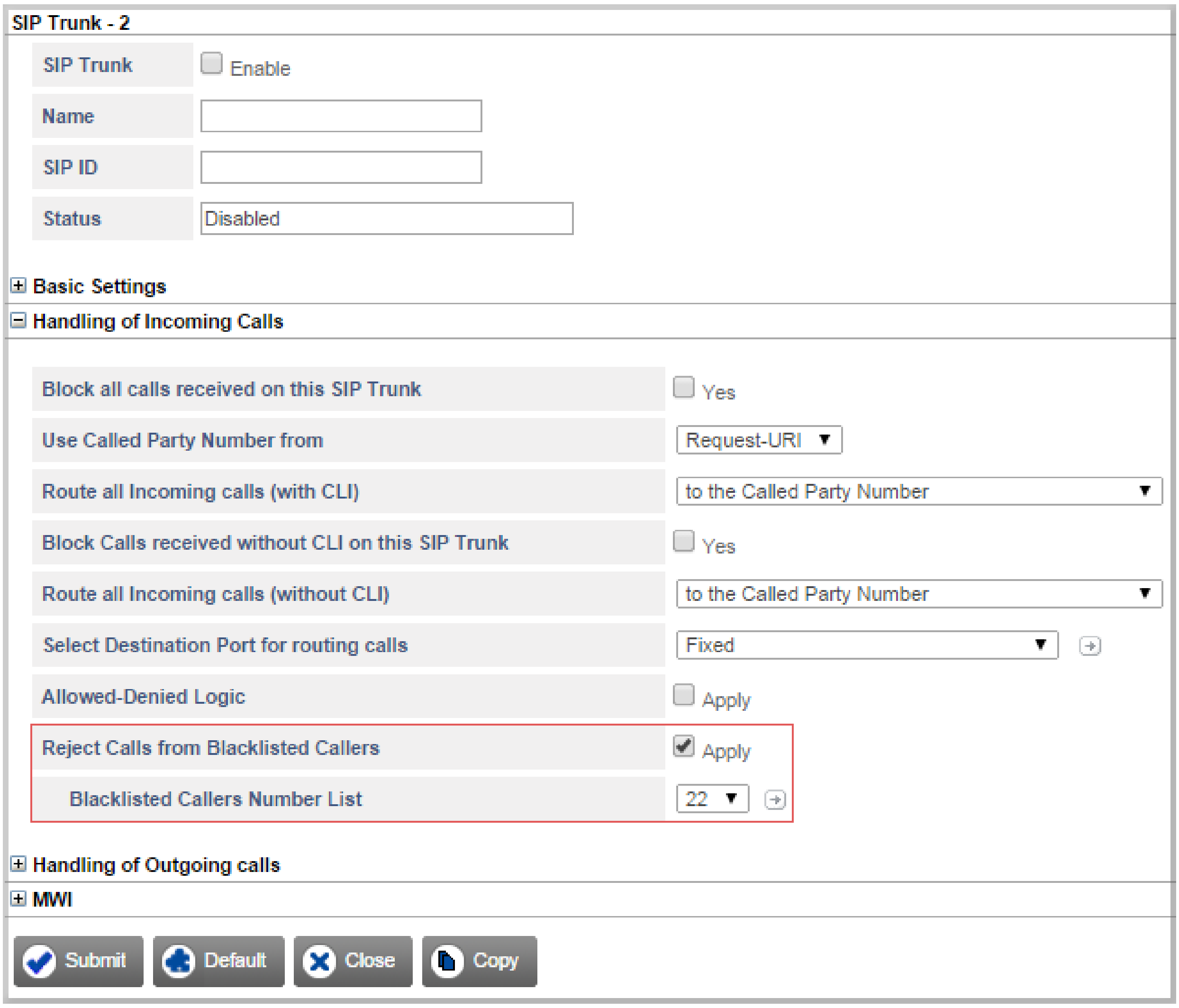
- Click the **Advanced Settings** link to expand.
- Click the **Number List** link.



- List 1 to 4 appears on the page. To select another List number, click the tab on the top of the table.
- Select the list number you want to configure.
- Enter the numbers strings in each list.
- Click **Submit** to save entries.
- Assign the list to the respective features for which you configured them on the various port types.

  For example, if you configured Number List 22 with black listed numbers for the Black Listed Callers feature on SIP Trunk 2,

- Click the **Basic Settings** link to expand.
- Click the **SIP Trunk** link.
- Click **SIP-2**.
- Under Handling of Incoming Calls, select the **Reject Calls from Blacklisted Callers** check box.



- In **Blacklisted Callers Number List**, select **22**.
- Click **Submit**.

You can also configure the Number lists on the respective FXS Port, FXO Port, BRI Port, Mobile Port, SIP Trunk and T1E1 Port pages under the "Basic Settings" link of Jeeves.

Automatic Number Translation (ANT)

Automatic Number Translation (ANT) is used to modify the number string—entire number or part thereof—into the desired number string as per your requirement. ANT is useful when you need to modify the Called/Calling number, before the system routes the call further.

For example, in India the PSTN requires you to dial the prefix 00 for calling international numbers, whereas the ITSP you have subscribed the SIP Trunk with, restricts the dialing of the prefix 00. If you dial this prefix, your call will be rejected by the ITSP. The ANT Table will enable you to modify the Number string as per your requirement so that the calls routed through the SIP Trunk are not rejected.

The Automatic Number Translation feature can be applied on all the FXS Ports, FXO Ports, BRI Ports, SIP Trunks, Mobile Ports and T1E1 Ports.

Automatic Number Translation makes use of Automatic Number Translation Table. The ANT Table consists of three columns:

- **Number:** In this column, enter the numbers that you want the system to modify.
- **Strip Digit:** In this column, enter the number of digit(s) to be stripped off by the system from the Called/ Calling number string. If you do not want any digits to be stripped, enter '0'.
- **Add Prefix:** In this column, enter the digit(s) which are to be added as prefix to the Called/ Calling number string by the system before routing it further.

To apply this feature on the desired port,

- on a piece of paper make a table, in the first column note down the numbers that needs to be modified. In the second column enter the number of digits you want the system to strip off (if required), and in the third column, enter the number you want the system to add as prefix (if required).
- configure the **Automatic Number Translation Table**. You can configure upto 8 different ANT Tables.
- enable **Automatic Number Translation (ANT) for Called Number** and/or **Automatic Number Translation (ANT) for Calling Number** on the respective ports/trunks, on which you want to apply this feature.
- assign the **Automatic Number Translation Table** you configured.
- configure the **Pause Timer**, if applicable.

For instructions, see:

- "General" under "FXS Port"
- "Advanced" under "BRI Port - Network"
- "Advanced" under "BRI Port - Terminal"
- "Advanced" under "T1 Port"
- "Advanced" under "E1 Port"

- “Handling of Outgoing Calls” under “SIP Trunk”
- “Handling of Outgoing Calls” under “FXO Port”
- “Handling of Outgoing Calls” under “Mobile Port”

Now, whenever there is a call on/ from the Port for which you have applied this feature, SARVAM UMG will match the Called/ Calling number with the Number configured in the Automatic Number Translation Table using the best match found logic.

- If a match is found, the system will check whether and how many digits to strip off. It will strip off digits according to the number you have entered in the Strip Digit column. If ‘0’ is configured in the Strip Digit column, it will check the Add Prefix column. If configured, the system will add that prefix. If no prefix is configured, the system will route the same number string further.

  If ~ (Wait for Answer) is configured in the Add Prefix column, the system will wait for the call to mature. Similarly, if ^ (Pause) is configured in the Add Prefix column, the system will wait for the Pause timer and then route the call further.

- If no match is found for the Called/ Calling number in the ANT Table, the system will route the number string, without modifying it.

Automatic Number Translation feature will not be applied when Emergency Numbers are dialed.

Automatic Number Translation also forms the basis of Multi-Stage Dialing. Using of Calling Card for making international calls is the most common example of Multi-Stage Dialing.

While using a Calling Card, you have to dial the digits in the following sequence:

1. Dial the number for using the Calling Card, for example, 160223.
2. After the call is matured, dial the PIN number printed on the Calling Card, for example, 113212.
3. At last, dial the international number you want to call. For example, 0014162357896.

Thus, you will have to dial the Calling Card number and the PIN number every time before dialing the international number. To avoid repetitive dialing of these fixed digits for making a call, you can configure the ANT table as under.

- In **Number**, configure ‘00’, the prefix for international numbers.
- In **Add Prefix**, configure the Calling Card server number and the PIN Number.

  As the system must wait for the Calling Card server to answer before dialing the PIN, you must configure Wait for Answer (**~**) between the Calling Card server number and the PIN number.

  You must also insert a delay by configuring the Pause Timer (**^**) after the PIN number.

- Keep Strip Digit as 00.
- The Automatic Number Translation table would look like this:

Index	Number	Strip Digit	Add Prefix
1	00	00	160223~113212^
2			

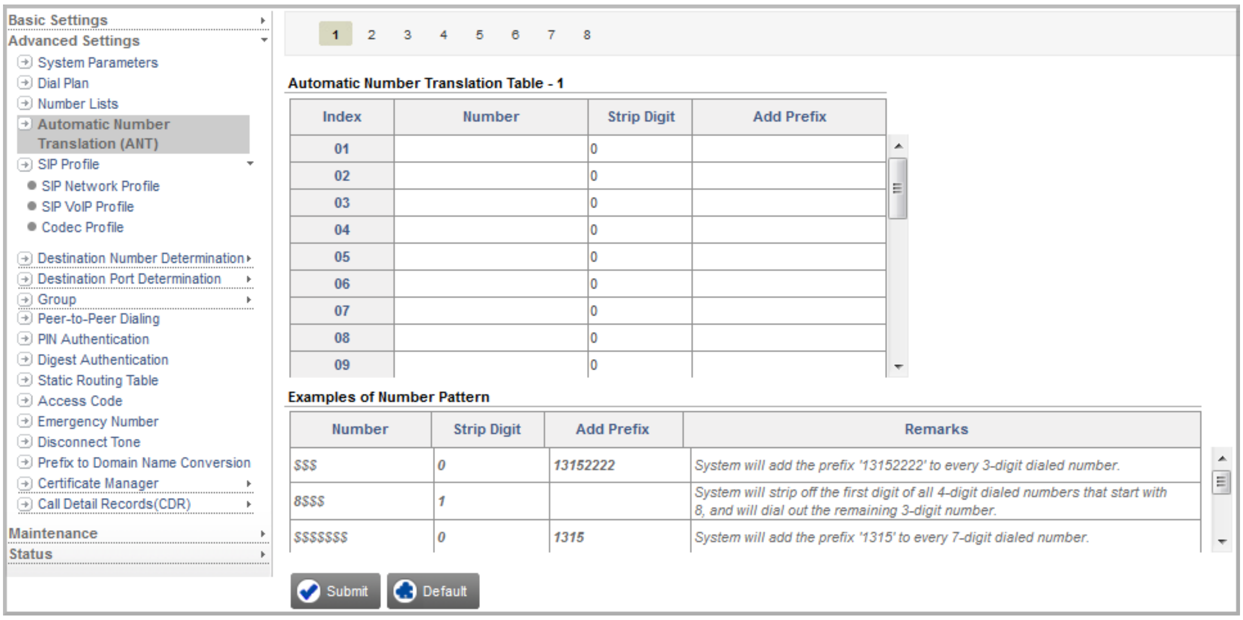
Index	Number	Strip Digit	Add Prefix
3			
4			
5			
6			
:			
24			

- When the Automatic Number Translation table is configured, the user must simply dial the destination number, say, 0014125126508.
- The system matches the Called number with the Number configured in the ANT table. The number matches with the entry ‘00’ stored in the table.
- The system dials the Add Prefix number string 160223 (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 113212 and waits for the Pause Timer before dialing the destination number.

Thus, the user can directly dial the desired destination number and the system dials the rest using the ANT table.

Configuring Automatic Number Translation Table

- Click the **Advanced Settings** link to expand.
- Click the **Automatic Number Translation (ANT)** link.



The Automatic Number Translation Table window opens. In this table, you can store as many as 24 Numbers at Index Numbers 01 to 24.

- In **Number**, enter the Called/ Calling numbers that need to be modified. You can enter maximum 24 digits. Digits 0-9, #, *, + and $ are allowed. Default: Blank.

  To configure a range of numbers you can use the character **$**. Here, **$** is any number from 0 to 9.

  For example, if you want SARVAM UMG to add prefix ‘1’ to all 10 digit numbers dialed by the user, configure Number as $$$$$$$$$$, Strip Digit as 0 and Add Prefix as 1. Now, when the user dials any number between the range of 0000000000 to 9999999999, say 4161231234, the system will add prefix 1 to it and dials out the number as 14161231234.

- In **Strip Digit**, enter the number of digits you want the system to strip off from the Called/Calling Number. You can configure from 00-24. Default: 0.

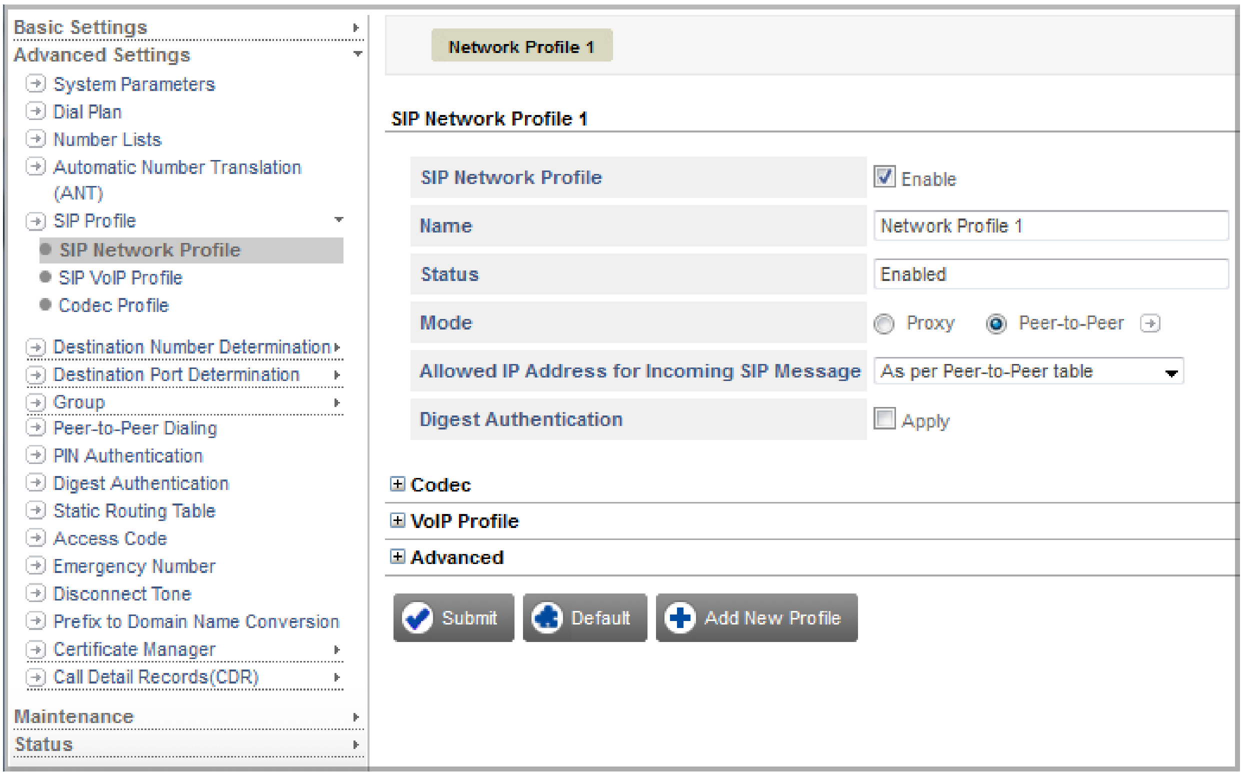
- In **Add Prefix**, enter the number string(s) that you want the system to add as prefix to the Called/ Calling Number. You can enter maximum 24 characters. Characters 0-9, *, #, +, ~ (Wait for Answer), ^ (Pause) are allowed. Default: Blank.

- Click **Submit** to save your entries.

SIP Network Profile

You can either edit the settings of the default **Network Profile 1** or you can add a new profile. To do so, click **Add New Profile**.

- Click **Advanced Settings** link.
- Click **SIP Network Profile** under **SIP Profile**.
- The **SIP Network Profile** page opens.



- Configure the following parameters:
  - Keep the **SIP Network Profile** check box enabled, to use this profile.

    Clear this check box only if you do not want to use this profile.
  - Assign a **Name** to the Network Profile for identification. The Name can be a maximum of 24 characters.
  - **Status** displays the status of this Network Profile.
  - You can select **Proxy** or **Peer-to-Peer** as the **Mode**. Default: Peer-to-Peer.
    - If you select **Proxy,** you must configure the following parameters:
      - Registrar Settings
      - Redundancy Settings
      - Codec
      - VoIP Profile

- Advanced
- Timer

- If you select **Peer-to-Peer,** you must configure the following parameters:
  - Enable Peer-to Peer option and configure the Peer-to-Peer Dialing table
  - Allowed IP Address for Incoming SIP Message
  - Digest Authentication
  - Codec
  - VoIP Profile
  - Advanced

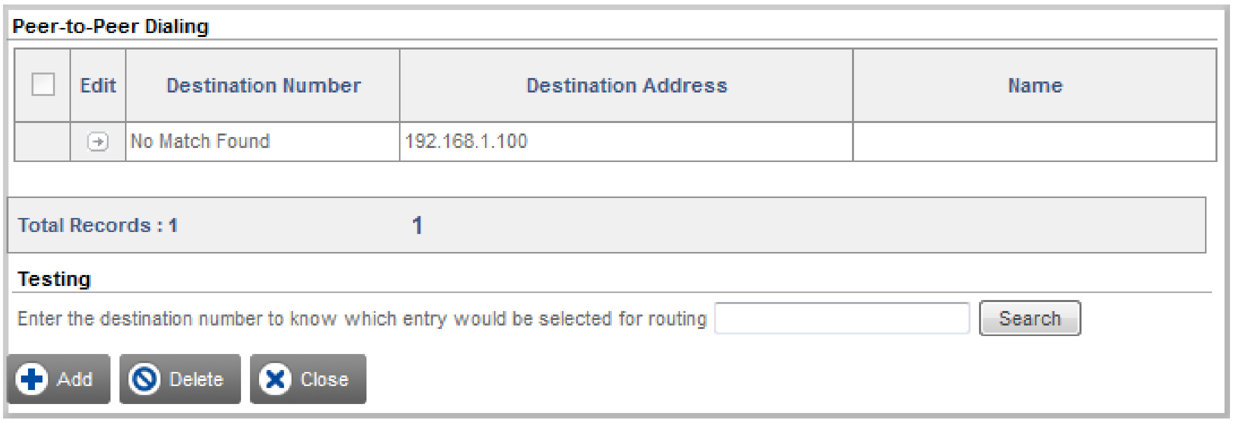
Peer-to-Peer Table

By default **Peer-to-Peer** is selected as the SIP Trunk **Mode**. You must configure the Peer-to-Peer Table.

To do so,

- Click **Settings** .

- The **Peer-to-Peer Dialing** table opens.



- You can add maximum 500 entries. Each entry in the table consists of the Destination Number, Destination Address and Name. For detailed instructions, see “Peer-to-Peer Dialing”.

Allowed IP Address for Incoming SIP Message

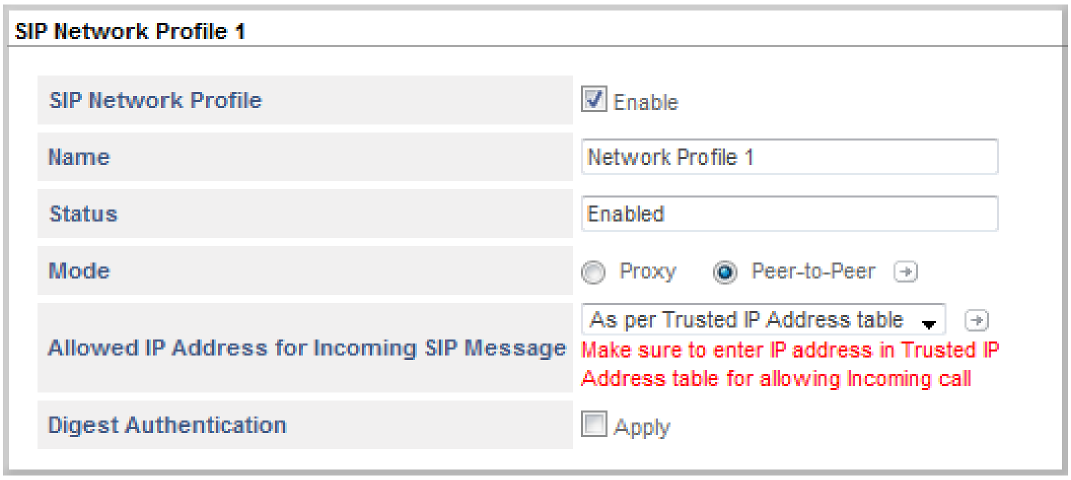
If you have selected the SIP Trunk **Mode** as **Peer-to-Peer**, you must select the desired option in **Allowed IP Address for Incoming SIP Message — As per Trusted IP Address table**, **As per Peer-to-Peer table** or **Any**.

- If you select **As per Trusted IP Address table** option, the system matches the **IP Address: Port** received in the INVITE message (Source IP address from the Network layer and Source Port from the Transport layer) with the entries configured in the Trusted IP Address table. If a match is found, the call will be routed to the desired destination. Else the call will be rejected.

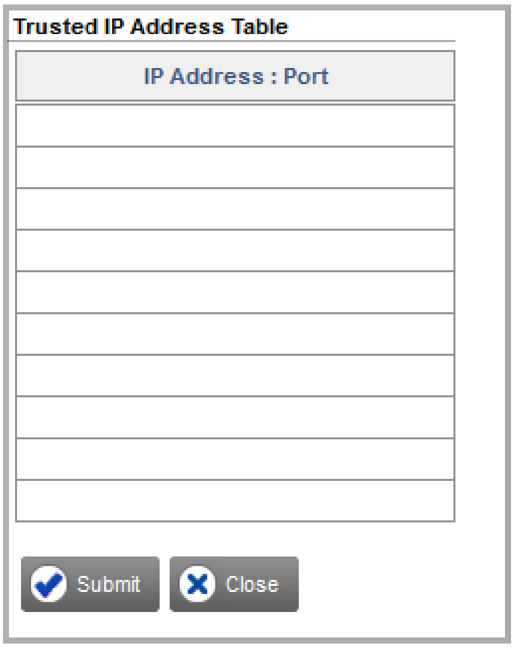
  You must configure the **Trusted IP Address Table** to receive incoming calls on this SIP Trunk. If you do not configure this table, incoming calls on this SIP Trunk will be rejected.

  You can configure maximum 10 entries in the Trusted IP Address table.

To do so, click **Settings** .



The **Trusted IP Address Table** opens in a new window.



- Enter the **IP Address** and the corresponding **Port** from which you want to allow incoming calls on this SIP Trunk. You can configure maximum 21 characters. Allowed characters are **0-9**, **dot** (**.**), **colon** (**:**).

  Do not configure the port, if you want to allow incoming calls from all the ports for a particular IP Address.

- Click **Submit** and close the window.

- If you select **As per Peer-to-Peer table** option, the system matches the **IP Address: Port** received in the INVITE message (Source IP address from the Network layer and Source Port from the Transport layer) with the Destination Address configured in the Peer to Peer table. If a match is found, then the call will be routed to the desired destination. Else the call will be rejected.

- If you select **Any** as the option, **Digest Authentication** will be enabled automatically. The system will allow incoming calls only after the callers authenticates themselves with the correct credentials—User ID and Password. The system matches the User ID and Password entered by the callers with the entries stored in the Digest Authentication table. If a match is found, the call will be routed to the desired destination. Else the call will be rejected.

  Default: **As per Peer-to-Peer table**.

Digest Authentication

If you have selected the SIP Trunk **Mode** as **Peer-to-Peer**,

- you may enable the **Digest Authentication** if you have set *Allowed IP Address for Incoming SIP Message* to *As per Trusted IP Address table* or *As per Peer to Peer table*. Incoming calls on this SIP Trunk will be allowed only after the callers authenticate themselves with their User ID and Password. Default: Disabled.

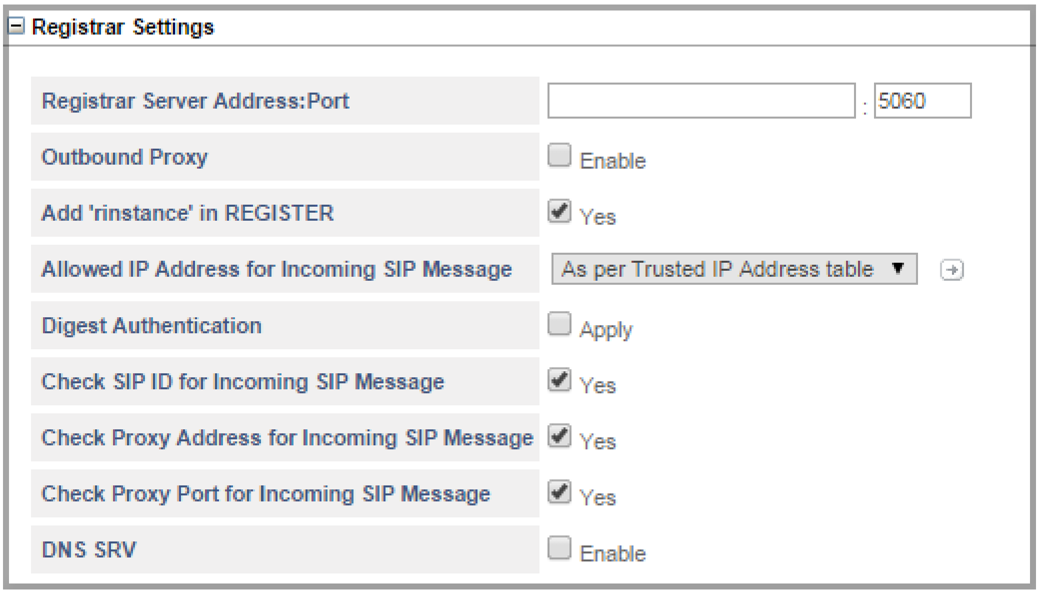
- Digest Authentication is enabled and you must configure the **Digest Authentication** table, if you have set *Allowed IP Address for Incoming SIP Message* to ***Any***.

For detailed instructions, see "Digest Authentication". Default: Disabled.

Registrar Settings

If you have selected the SIP Trunk **Mode** as **Proxy**, configure the Registrar Settings.

- Click **Registrar Settings**.



- In **Registrar Server Address: Port**, enter the Registrar Server Address and the Registrar Server's listening port for SIP messages. The registrar server address may be an IP address or a domain. The Registrar Server Address can be of maximum 64 characters. Valid range is 1025 to 65534. Default: 5060.

- If your Service Provider uses outbound proxy for handling voice calls, select the **Outbound Proxy** check box. Default: Disabled.

- In **Outbound Proxy Server Address: Port**, enter the Outbound Proxy Server's IP Address and the Outbound Proxy Server's Listening Port for SIP. The Outbound Proxy Server Address may be of maximum 64 characters. Valid range is 1025 to 65534. Default: 5060.

- To add 'rinstance' in REGISTER message, keep the **Add 'rinstance' in REGISTER** check box enabled.

  'rinstance' is any random value which can be used by SARVAM UMG to fetch its own contact binding, that is, to know the Registration Expiry Timer assigned by the server.

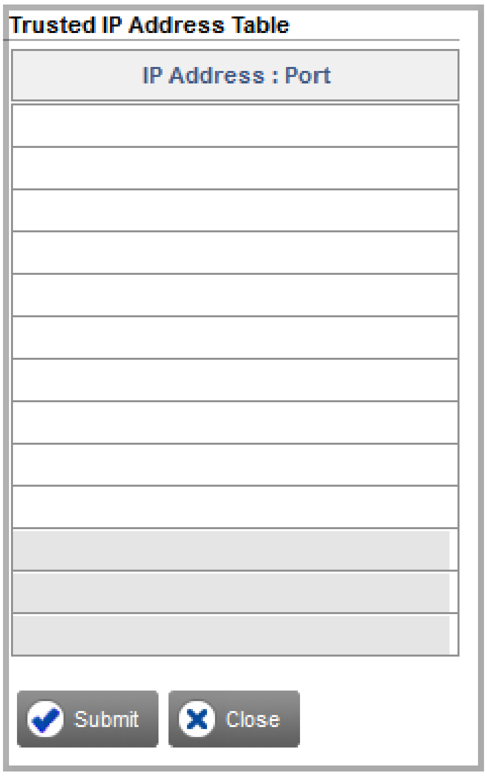
  When you enable 'rinstance' in Register, SARVAM UMG will generate any random value of 'rinstance' and include it in the REGISTER message. The system will use the registration expiry timer of that contact binding.

- By default, the **Allowed IP Address for Incoming SIP Message** is set to **As per Trusted IP Address table** for Proxy SIP Trunk and is non-programmable. You must configure the **Trusted IP Address** table to allow incoming calls from specific IP addresses on this SIP Trunk.

  Trusted IP Address table stores upto 13 entries, from which last three entries are uneditable. The last three entries in the table will display the *Registrar Server Address:Port* or *Outbound Proxy Address: Port* and *Fallback Registrar Server Address:Port1* and *2* or *Fallback Outbound Proxy Server Address:Port1* and *2*, if configured for this SIP Trunk. If you do not configure the Trusted IP Address table, incoming calls will be allowed from the last three IP Addresses only.

  To configure Trusted IP Address table, click **Settings** .

  The **Trusted IP Address Table** opens in a new window.



- Enter the **IP Address** and the corresponding **Port** from which you want to allow incoming calls on this SIP Trunk. You can configure maximum 21 characters. Allowed characters are **0-9**, **dot** (**.**), **colon** (**:**).

  Do not configure the port, if you want to allow incoming calls from all the ports for a particular IP Address.

- Click **Submit** and close the window.

- If you want to allow incoming calls on this SIP Trunk only after the callers authenticate themselves with their User ID and Password, enable **Digest Authentication**. Default: Disabled.

  If you enable Digest Authentication feature on the SIP Trunk, you must configure the Digest Authentication Table. See "Digest Authentication" for more details.

- Keep the **Check SIP ID for Incoming SIP Message** check box enabled, if you want SARVAM UMG to validate the SIP ID during an incoming call. Default: Enabled.

- Keep the **Check Proxy Address for Incoming SIP Message** check box enabled, if you want SARVAM UMG to validate the Proxy Address during an incoming call. Default: Enabled.

- Keep the **Check Proxy Port for Incoming SIP Message** check box enabled, if you want SARVAM UMG to validate the Proxy Port during an incoming call. Default: Enabled.

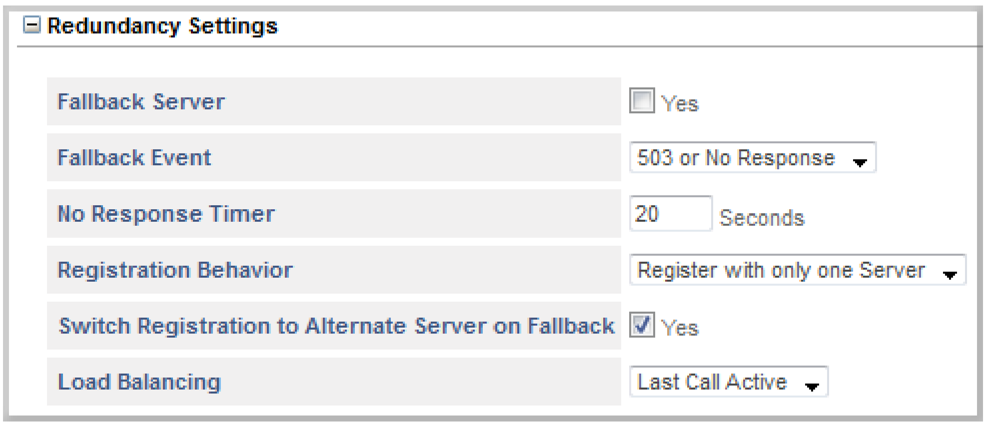
- Enable **DNS SRV**, if you want the system to send DNS SRV query to the configured domain server. When disabled, the system will send DNS A query to the configured domain server. Default: Disabled.

If you enable DNS SRV, Fallback Server logic will not be applicable.

Redundancy Settings

If you have selected the SIP Trunk **Mode** as **Proxy**, configure the Redundancy Settings.

- Click **Redundancy Settings**.



- Select the **Fallback Server** check box, if your Service Provider supports multiple servers in its network. Default: Disabled.

  If you have enabled Fallback Server and Outbound Proxy is disabled,

- In the **Fallback Registrar Server Address 1: Port** and **Fallback Registrar Server Address 2: Port** field, enter addresses of the alternate Registrar Servers and their respective listening ports. The Fallback Registrar Server Address can be of maximum 64 characters. Valid range is 1025 to 65534. Default: 5060.

If you have enabled Fallback Server and Outbound Proxy is enabled,

- In **Fallback Outbound Proxy Server Address 1: Port** and **Fallback Outbound Proxy Server Address 2: Port** field, enter addresses of the alternate Outbound Proxy Servers and their respective listening ports. The Fallback Outbound Proxy Server Address can be of maximum 64 characters. Valid range is 1025 to 65534. Default: 5060.

- In the **Fallback Event** list, select the event on occurrence of which SARVAM UMG should fallback to an alternate Registrar/Outbound Proxy Server, if available.
  - No Response
  - 503 or No Response
  - 5xx or No Response

  Default: 503 or No Response

  In case, the Fallback Server does not respond and the call is not routed to the destination port, the call will be routed to another port type as per the Routing/Fallback Routing Group configured for the SIP Trunk.

- Set the duration of the **No Response Timer.** This timer defines the time period for which SARVAM UMG will wait for the response from the server for any request. If no valid response is received before the expiry of this timer, SARVAM UMG will fallback to alternate Registrar/Outbound Proxy Server or Routing Group/Fallback Routing Group for further processing of the call. Valid range is 01 to 99 seconds. Default: 20 seconds.

If the SIP General Request Timer configured in the System Parameters is less than the No Response Timer, then SARVAM UMG will fallback to alternate Registrar/Outbound Proxy Server or Routing Group/ Fallback Routing Group on the expiry of the SIP General Request Timer and the No Response Timer will stop.

- In the **Registration Behavior**, select the desired option:
  - Register with all Servers
  - Register with only one Server

  If you select **Register with only one Server**, SARVAM UMG will get registered with the Registrar/ Outbound Proxy Server. If registration with the Registrar/Outbound Proxy Server fails, it will get registered with Fallback Registrar/Outbound Proxy Server 1 or Fallback Registrar/Outbound Proxy Server 2 respectively for further processing of call.

  If you select **Register with all Servers**, SARVAM UMG will get registered with Registrar/Outbound Proxy Server as well as Fallback Registrar/Outbound Proxy Servers. It will not apply Fallback logic even if *Fallback Server* is enabled.

*The **Registration Behavior** will be applicable only if, **SIP Registration** is enabled.*

- Keep the **Switch Registration to Alternate Server on Fallback** check box enabled. SARVAM UMG will get unregistered with the current server and will register with the alternate server, if fallback occurs while sending the INVITE message.

*The **Switch Registration to Alternate Server on Fallback** will be applicable only if, **SIP Registration** is **enabled** and **Registration Behavior** is set as **Register with only one Server**.*

- Select the desired option for **Load Balancing** from the following:
  - **Last Call Active**: Each new call will be processed through the Registrar/Outbound Proxy Server through which the last active call has been processed.

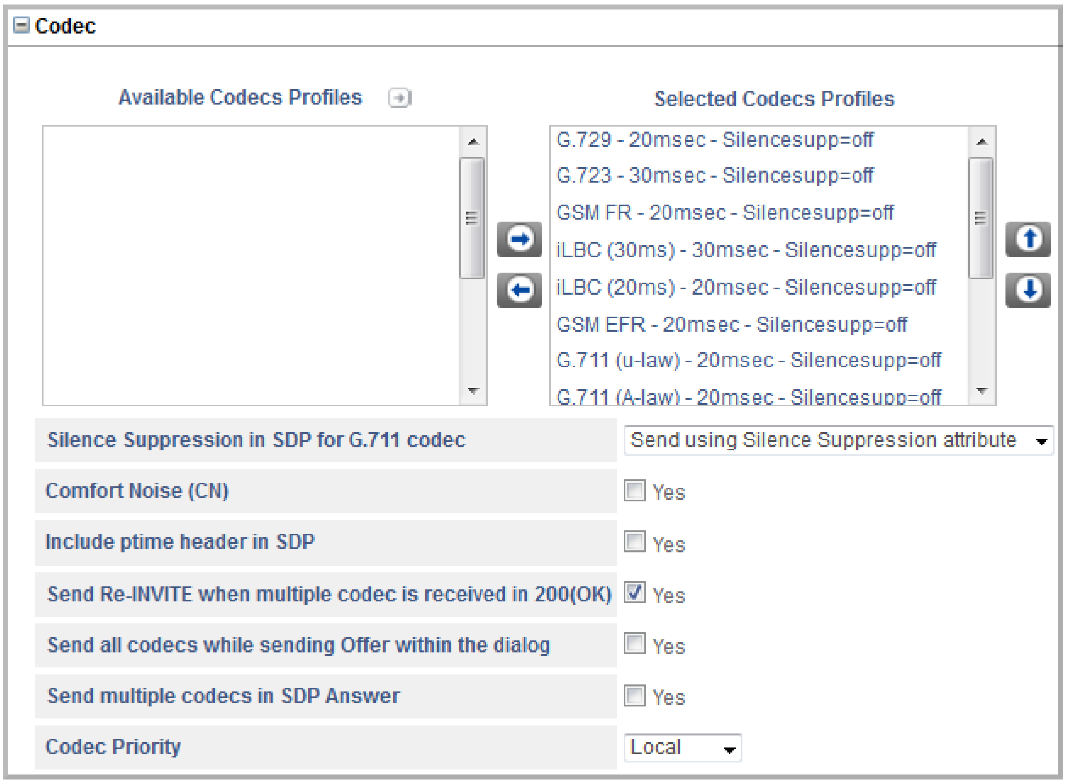
    For example, if the last call has been processed by Fallback Registrar/Outbound Proxy Server 2, the new call will also be processed through Fallback Registrar/Outbound Proxy Server 2 only.
  - **First Active**: Each new call will be processed through the first active Registrar/Outbound Proxy Server only.
  - **Cyclic**: Each new call will be processed through the next active Registrar/Outbound Proxy Server.

    For example, if the last call has been processed by Fallback Registrar/Outbound Proxy Server 1, the new call will be processed through Fallback Registrar/Outbound Proxy Server 2 and the subsequent new call will be processed through the Registrar/Outbound Proxy Server.
    Default: Last Call Active.

Codec Profile

If you have selected the SIP Trunk **Mode** as **Proxy** or **Peer-to-Peer**, configure the Codec Profile.

- Click **Codec**.

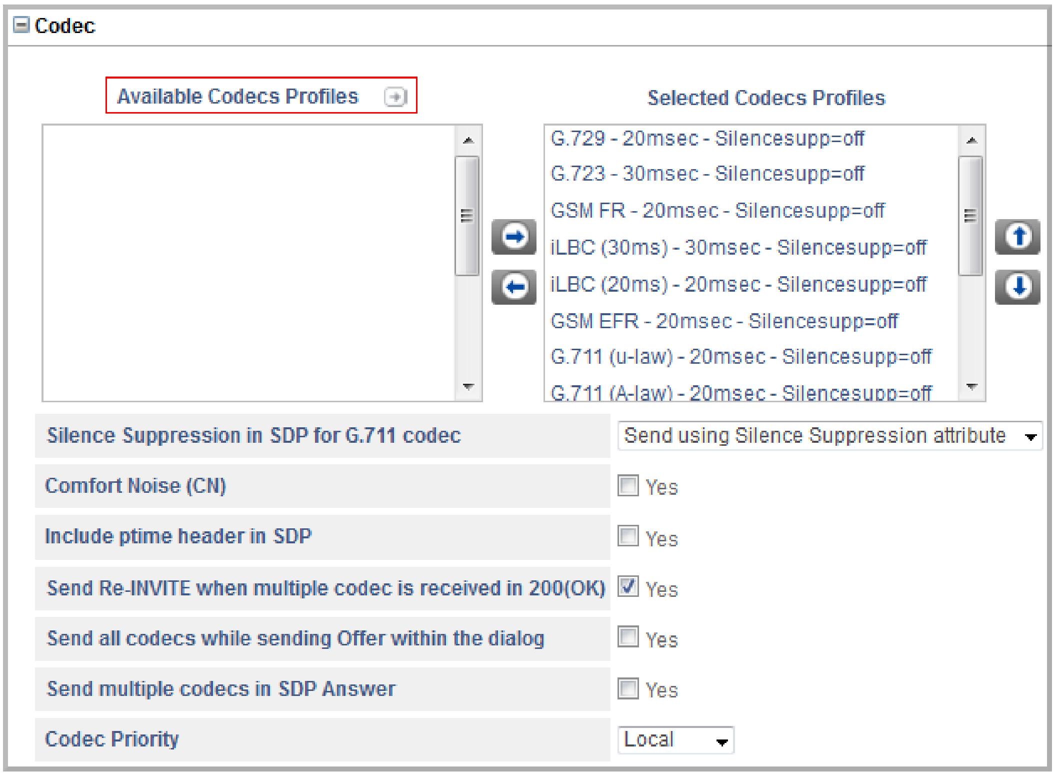


Codecs are used to compress the data in RTP packets to enable quick transmission. It also decompresses the received data.

The codec profiles supported by SARVAM UMG appears in the **Selected Codecs Profiles** list in the following order of preference:

1. G.729 - 20msec - Silencesupp=off
2. G.723 - 30msec - Silencesupp=off
3. GSM FR - 20msec - Silencesupp=off
4. iLBC (30ms) - 30msec - Silencesupp=off
5. iLBC (20ms) - 20msec - Silencesupp=off
6. GSM EFR - 20msec - Silencesupp=off
7. G.711 (u-law) - 20msec - Silencesupp=off
8. G.711 (A-law) - 20msec - Silencesupp=off

- You can change the order of preference by moving the desired Codecs up or down the list. To move a Codec up or down the list, do the following:
  - In the **Selected Codecs Profiles** list, click the Codec you want to move.
  - Click the UP/DOWN ARROW to move the Codec to the desired position in the list.
- To remove a Codec from the **Selected Codecs** list, click the Codec you want to remove, and then click the LEFT ARROW. The Codec is moved to the **Available Codecs Profiles** list.
- To move a Codec from the **Available Codecs Profiles** list to the **Selected Codecs Profiles** list, click the Codec you want to move, and then click the RIGHT ARROW.
- You can edit any existing profile or add a new profile. To do so, click **Settings** .



  - The **Codec Profile** window opens. For detailed instructions, see “Codec Profile”.

- Select the desired **Silence Suppression in SDP for G.711 codec** option. SARVAM UMG suppresses the *Silence* packets and allows only the *Voice* packets to pass through.

  This is used to deactivate certain processes during non-speech section of an audio session to avoid unnecessary coding/ transmission of silence packets in VoIP application. Hence, it results in saving on computation and network bandwidth.

  You can select either *Do Not Send*, *Send using Silence Suppression attribute* or *Send using VAD attribute*.

  If you select *Do Not Send*, SARVAM UMG will not send any “Silence Suppression” media attribute in the SDP offer / answer exchanges.This is not dependant on the Silence Suppression check box.

  If the Silence Suppression check box is disabled. and you select S*end using Silence Suppression attribute,* SARVAM UMG will send *Silence Suppression=OFF* in the SDP offer / answer exchanges.

  If the Silence Suppression check box is enabled. and you select S*end using Silence Suppression attribute,* SARVAM UMG will send *Silence Suppression=ON* in the SDP offer / answer exchanges.

  If the Silence Suppression check box is disabled. and you select *Send using VAD attribute,* SARVAM UMG will send *VAD=NO* in the SDP offer / answer exchanges.

  If the Silence Suppression check box is enabled. and you select *Send using VAD attribute,* SARVAM UMG will send *VAD=YES* in the SDP offer / answer exchanges.

  Default: Send using Silence Suppression attribute

- Select the **Comfort Noise (CN)** check box, if you want SARVAM UMG to negotiate the Comfort Noise received in the SDP body with the remote peer. Default: Disabled.

- Select the **Include ptime header in SDP** check box, if you want SARVAM UMG to add ptime header in the SDP offer / answer exchanges. Default: Disabled.

- Clear the **Send Re-INVITE when multiple codec is received in 200 (OK)** check box, if you do not want SARVAM UMG to send Re-INVITE message and use only the first codec from the multiple codecs received in 200 (OK). Default: Enabled.

- Select the **Send all codecs while sending Offer within the dialog** check box, if you want SARVAM UMG to send all the codec configured on SIP Trunk in same order as configured while sending the RE-INVITE message to hold or unhold the call.

  When RE-INVITE without SDP is received then the system sends SDP offer in 200 OK response with all the codecs configured for that SIP trunk. The codec order is same as configured for the SIP Trunk.

  Default: Disabled.

- Select the **Send multiple codecs in SDP Answer** check box, if you want SARVAM UMG to send all the configured codecs in SDP answer. Default: Disabled.

  The codecs sent will depend on the option you select in **Codec Priority**.

  - When Codec Priority is set as *Local* then system sends all the codecs as per local preference.

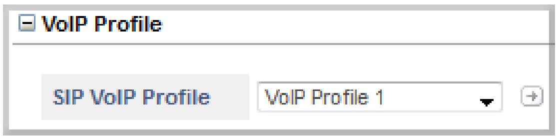
- When Codec Priority is set as *Remote* then system sends all the codecs as per the preference received in SDP offered from remote party. Here, the system sends only configured codecs in remote preference.

- Select the desired **Codec Priority** option using which system can decide which codecs — Local or Remote — should be given preference. Default: Local.

VoIP Profile

If you have selected the SIP Trunk **Mode** as **Proxy** or **Peer-to-Peer**, configure the VoIP Profile.

- Click **VoIP Profile.**



- In **SIP VoIP Profile**, you can either select the default **VoIP Profile 1** or **Add New VoIP Profile** option.

  - Click **Settings** to configure the parameters of the selected VoIP Profile.

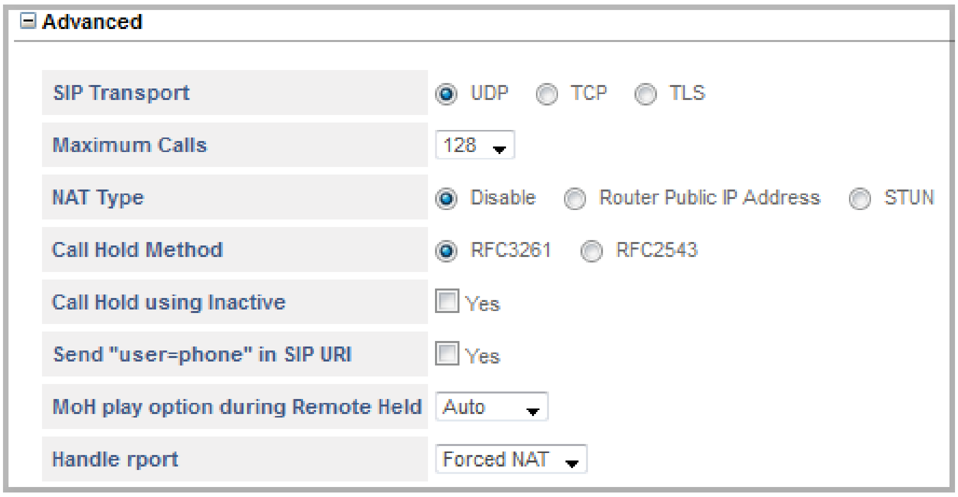
    For detailed instructions, see "SIP VoIP Profile".

    You can also configure the **SIP VoIP Profile** under **SIP Profile** from *Advanced Settings.*

Advanced

If you have selected the SIP Trunk **Mode** as **Proxy** or **Peer-to-Peer**, configure the Advanced Settings.

- Click **Advanced**.



- Select the default **SIP Transport** for outgoing SIP messages from the following options:

  - **UDP:** Outgoing messages are transported using UDP.

  - **TCP:** Outgoing messages are transported using TCP.

- **TCP Fallback to UDP:** TCP is used for outgoing messages. However, if the TCP connection fails, the system will attempt to send the message again over UDP.

- **TLS:** Outgoing messages are transported using TLS.

  Default: UDP

*To use TCP or TCP Fallback to UDP, you must enable **SIP over TCP** in the "System Parameters".*

*To use TLS, you must enable **SIP over TLS** in "System Parameters".*

- For the selected Network Profile in **Maximum Calls**, select the number of simultaneous calls you want to allow on the SIP Trunk.

  The maximum number of simultaneous SIP calls depend upon the number of Vocoder channels[20] supported.

- When the system is installed behind a NAT Router, select specific NAT traversal mechanism to be used as **NAT Type**. Default: Disabled.

  - Select **Router's IP Address,** if your SARVAM UMG is located behind the NAT router (any type).

    Make sure you disable Outbound Proxy on SIP Trunk and have configured the same IP Address under NAT settings in the "System Parameters" page.

  - Select **STUN**, if your system is located behind the NAT router other than Symmetric.

    Make sure you disable Outbound Proxy on SIP Trunk and have configure the STUN Server Address and port under NAT settings in "System Parameters".

- In **Call Hold Method** select the desired option **—** RFC 2543 or RFC 3261 — that is compatible with your ITSP proxy server / remote peer. Default: RFC 3261

- Select the **Call Hold using Inactive** check box, if you want the system to send '*a=inactive'* message instead of *'a=sendonly'* message on the SIP Trunk, when the user puts the call on hold. Default: Disabled.

- Select **Send "user=phone" in SIP URI** check box, if you want SARVAM UMG to add user=phone in the Request URI / From / To header of the INVITE message.

  SARVAM UMG will send user=phone in SIP URI, only if the SIP ID is numeric.

  Default: Disabled.

- In **MoH play option during Remote Held,** select the desired option — Auto, Local or Remote.

  If you select *Local,* SARVAM UMG will play Music-On-Hold to the extension that is put on hold.

20. *During the Demo period, the number of Vocoder channels that will be supported will be equal to the total number of channels available in the Vocoder module/s installed in the System. If the Demo period is paused or gets expired, then the number of supported Vocoder channels will be as per the license you purchase.*

If you select *Remote,* SARVAM UMG will play the Music-On-Hold received from the remote end to the extension that is put on hold.

If you select *Auto*, SARVAM UMG will play Music-On-Hold received from the remote end for 5 seconds and if it receives a silent tone it will play the local Music-On-Hold to the extension that is put on hold.

Default: Auto.

- In **Handle rport** select the desired option — Forced NAT or RFC 3581.

  If you select *Forced NAT,* SARVAM UMG will not check Contact/Via header etc. while sending SIP messages and will follow Symmetric Signaling.

  If you select *RFC 3581,* the system follows Standard RFC while sending SIP messages.

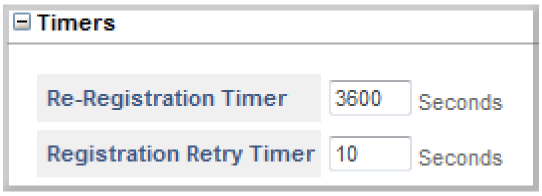
  Default: Forced NAT.

- Click **Submit** to save the changes.

Timers

If you have selected the SIP Trunk **Mode** as **Proxy**, configure the Timers.

- Click **Timers**.



- Set the duration of the **Re-registration Timer**. This is the time period after which the SARVAM UMG will send registration request to maintain registration binding with the Registrar Server. Valid range is 00001 to 65535 seconds. Default: 3600 seconds.

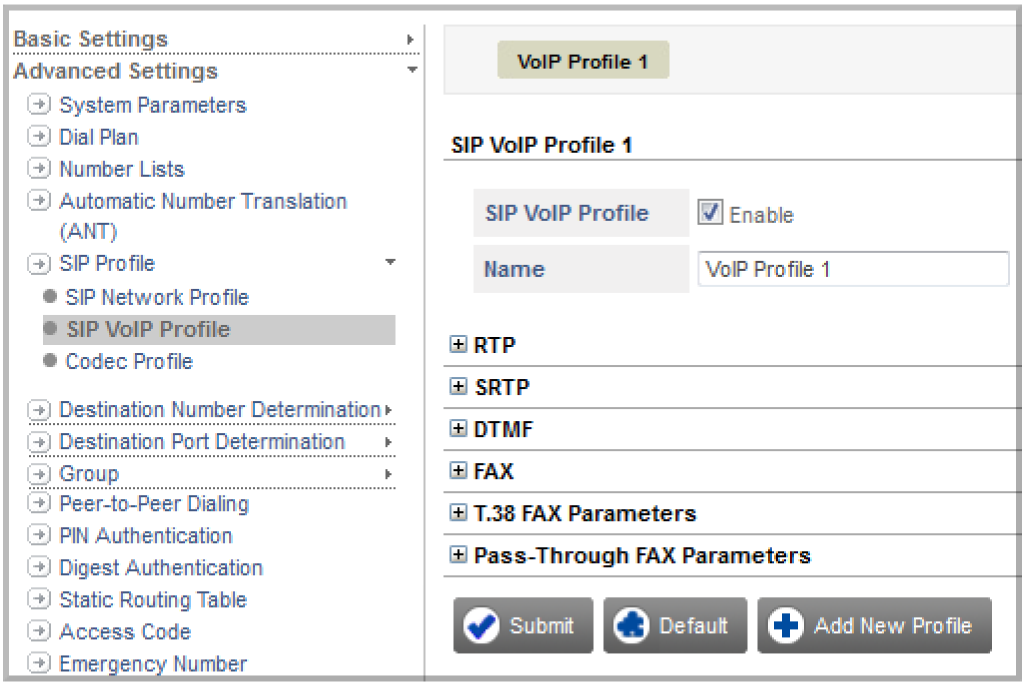
- Set the duration of the **Registration Retry Timer**. When a registration attempt fails, SARVAM UMG will resend registration request to the Registrar Server after the expiry of the Re-registration Timer. Valid range is 00001 to 65535. Default: 10 seconds.

*The above timers will be applicable only if, **SIP Registration** is enabled in **SIP Trunk**.*

SIP VoIP Profile

You can either edit the settings of the default **VoIP Profile 1** or you can add a new profile. To do so, click **Add New Profile**.

- Click **Advanced Settings** to expand.
- Click **SIP VoIP Profile** under **SIP Profile**.
- The **SIP VoIP Profile** page opens.

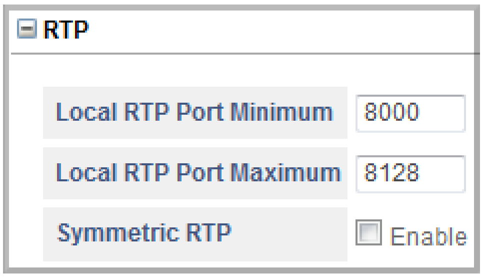


- Configure the following parameters:
  - Keep the **SIP Network Profile** check box enabled to use this profile.

    Clear this check box if you do not want to use this profile.
  - Assign a **Name** to the VoIP Profile for identification. The Name can be a maximum of 24 characters.

RTP

- Click **RTP**.

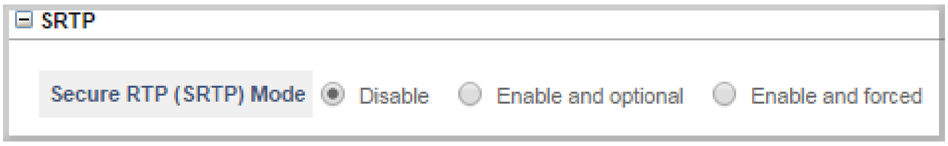


RTP Port is the port on which the SARVAM UMG listens for RTP Packets. This port is also used as the source port for sending RTP packets to the remote peer.

- In **Local RTP Port Minimum**, enter the desired minimum RTP Port Number. Valid range is 1032 to 65535. Default: 8000.
- In **Local RTP Port Maximum**, enter the desired maximum RTP Port Number. Valid range is 1032 to 65535. By default, the **Local RTP Port Maximum** = **Local RTP Port Minimum** + [**Maximum calls**[21] **supported by the product** x 2].
- Select the **Symmetric RTP** check box, if you want the system to send RTP packets to original IP and Port from where RTP packets are received. Default: Disabled.

SRTP

- Click **SRTP**.

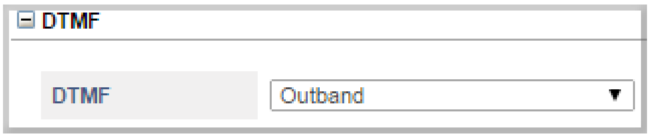


- For secure conversations over SIP, SARVAM UMG supports the **Secure RTP (SRTP)** mode. Default: Disabled. Select the desired option as per your requirement:
  - Select **Disable**, if you want SARVAM UMG to use normal RTP instead of SRTP for transporting the speech packets.
  - Select **Enable and optional**, if you want SARVAM UMG to use SRTP for transporting the speech packets. If the remote user does not support SRTP, normal RTP will be used.
    - If you select this option, you must configure the **SRTP Media Type**. You may select *AVP* or *SAVP*. Default: AVP.
  - Select **Enable and forced**, if you want SARVAM UMG to use only SRTP (SAVP) for transporting the speech packets. If the remote user does not support SRTP, SARVAM UMG will reject incoming calls from and drop outgoing calls made to such users.

21. The maximum number of calls depends upon the number of Vocoder Channels supported.

DTMF

- Click **DTMF**.

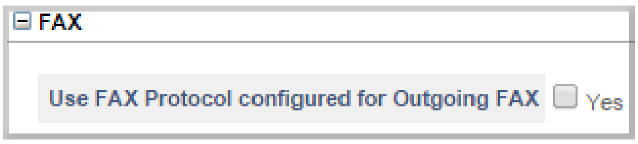


- Select the appropriate **DTMF** sending / receiving mechanism that is compatible with the DTMF sending/ receiving mechanism of your ITSP or remote peer.
- SARVAM UMG supports:
  - **In-band:** System will send and detect digits in In-band only.
  - **Outband:** System will send and detect digits in Outband events only.
  - **SIP INFO:** System will send and detect digits in SIP INFO message only.
  - **Outband-->In-band:** System will send and detect digits in Outband, if negotiated in offer / answer else it will use In-band.
  - **SIP INFO-->In-band:** System will send and detect digits in SIP INFO, if negotiated in offer / answer else it will use In-band.
  - **Outband-->SIP INFO-->In-band:** System will send and detect digits in Outband or SIP INFO, if negotiated in offer / answer else it will use In-band. If both Outband and SIP INFO are negotiated, Outband will have priority over SIP INFO.
  - **SIP INFO-->Outband-->In-band:** System will send and detect digits in SIP INFO or Outband, if negotiated in offer / answer else it will use In-band. If both SIP INFO and Outband are negotiated, SIP INFO will have priority over Outband.

  Default: Outband.

FAX

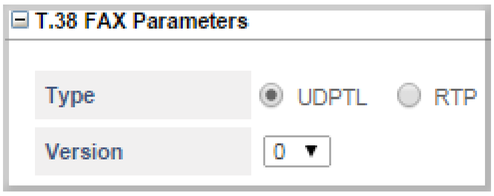
- Click **FAX**.



- Select the **Use FAX Protocol configured for Outgoing FAX** check box, if you want SARVAM UMG to use the Fax Protocol configured for outgoing fax for this SIP Trunk and not the one that is received in RE- INVITE message from the remote end. Default: Disabled.

T.38 FAX Parameters

- Click **T.38 FAX Parameters**.

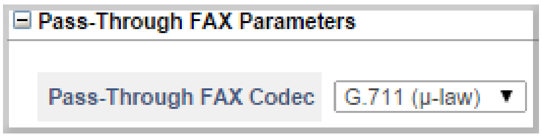


- Select the **Type** of Fax Protocol— UDPTL or RTP — that is compatible with your ITSP proxy server / remote peer. Default: UDPTL.

- For the Type you select, you must select a compatible **Version** — 0, 1, 2 — as supported by your ITSP proxy server / remote peer. Default: 0.

Pass-Through FAX Parameters

- Click **Pass-Through FAX Parameters**.

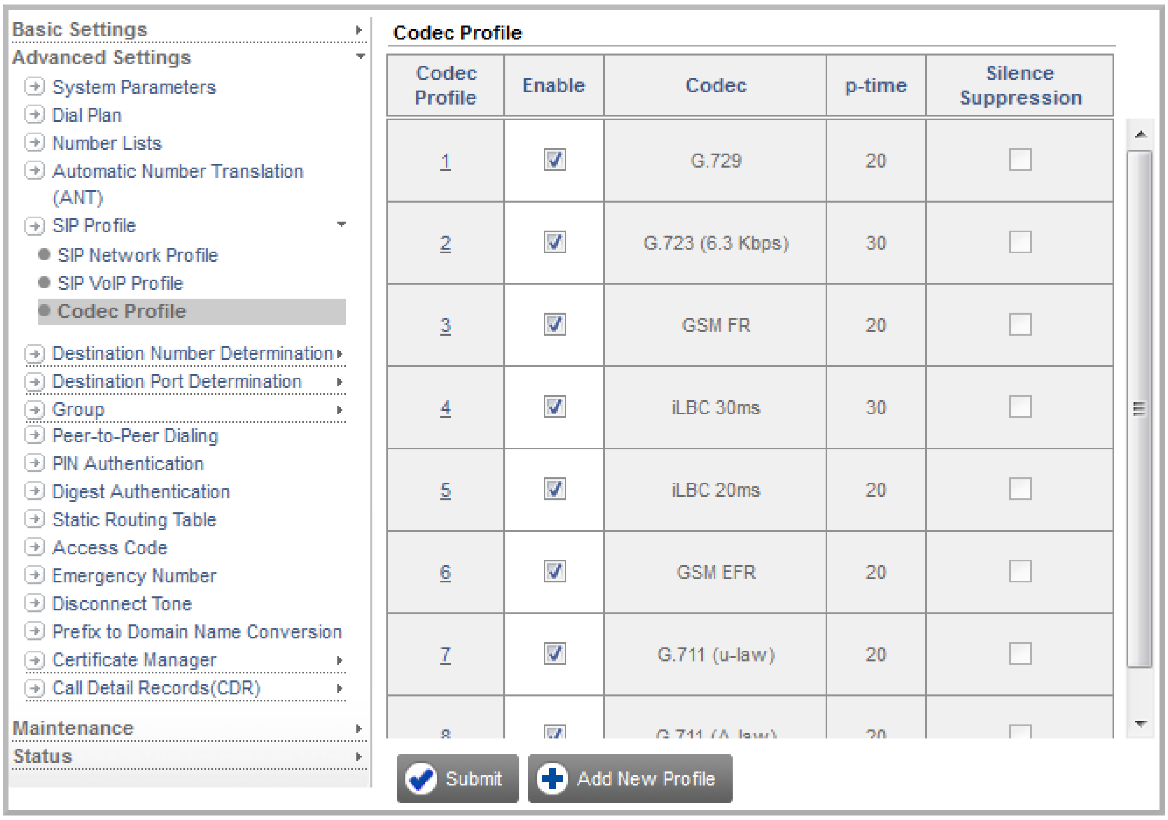


- Select an appropriate **Pass-Through FAX Codec** — G.711 (μ-law) or G.711 (A-law) — that is compatible with your ITSP proxy server/ remote peer. Default: G.711 (μ-law).

Codec Profile

You can either edit the settings of the **Codec Profiles** numbered from 1 to 8 or you can add a new profile.

- Click **Advanced Settings** to expand.
- Click **Codec** under **SIP Profile**.
- The **Codec Profile** page opens.



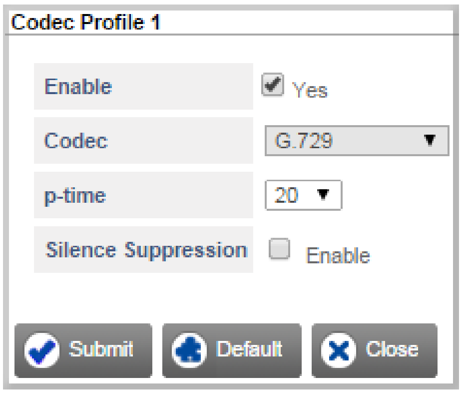
The **Codec Profile** page displays the following parameters:

- **Codec Profile:** It displays the Codec Profile numbers. To configure the Codec Profile Parameters, click on the desired Codec Profile number.
- **Enable:** Click the check box to enable the desired Codec Profile.
- **Codec:** It displays the Codecs assigned to each of the Codec Profile.
- **p-time:** It displays the p-time selected for each of the Codec Profile.
- **Silence Suppression:** It displays whether the Silence Suppression for the respective Codec Profile is enabled or not.

To configure the Codec Profile parameters:

- Click **Add New Profile** to add a new profile.

- To edit the existing profile, click on the **Codec Profile** number you want to edit.
- The respective **Codec Profile** window opens.



- Keep the **Enable** check box enabled to use this profile.

  Clear this check box if you do not want to use this profile.

- The **Codec** is fixed for **Codec Profile 1** to **Codec Profile 8**. You can assign the desired **Codec** — G. 729, G.723, GSM FR, iLBC 30ms, iLBC 20ms, G.711 (u - Law), G.711 (A - Law) — only when you add a new Codec Profile.
- Select the desired **p-time** value, if you have selected codec as — G. 729, G.723, GSM FR, G.711 (u - Law), G.711 (A - Law) or GSM EFR.
- If you have selected codec G.723, select the desired **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. While receiving the RTP packets from the remote end, both the Bit Rates of G.723 will be accepted.

- For the codecs — G.729, G.723 and G.711 (u - Law) — select the **Silence Suppression** check box, if you want SARVAM UMG to suppress the Silence packets and allow only the Voice packets to pass through. Default: Disabled.

Destination Port Determination

The process of routing calls originated on FXS Ports, FXO Ports, BRI Ports, Mobile Ports, SIP Trunks and T1E1 Ports to the destination port in SARVAM UMG takes place in two steps:

- Determination of Destination Number
- Determination of Destination Port

SARVAM UMG supports different methods of determining the destination port for the calls originated on FXS Ports, FXO Ports, BRI Ports, Mobile Ports, SIP Trunks and T1E1 Ports.

Destination Port Determination on FXS Ports

For FXS Port, the system supports the following methods for Destination Port Determination:

- Fixed
- On the basis of Destination Number

To apply Destination Port Determination **on the basis of Destination Number**, you must configure the **FXS Port - Destination Port Determination - Destination Number Based** table.

Destination Port Determination on FXO Ports

For FXO Ports, the system supports the following methods for Destination Port Determination:

- Fixed
- On the basis of Destination Number
- On the basis of Calling Party Number

To apply Destination Port Determination **on the basis of Calling Party Number**, you must configure the **FXO Port - Destination Port Determination - Calling Number Based** table.

To apply Destination Port Determination **on the basis of Destination Number**, you must configure the **FXO Port - Destination Port Determination - Destination Number Based** table.

Destination Port Determination on BRI Ports

For BRI Port with **Orientation Type - Terminal**, you must configure the following destination port determination methods by Port, Channel and MSN Number/DDI Number:

- Fixed
- On the basis of Destination Number
- On the basis of Calling Party Number

For BRI Port with **Orientation Type - Network**, you must configure the following destination port determination methods by Port and Channel:

- Fixed
- On the basis of Destination Number
- On the basis of Calling Party Number

To apply Destination Port Determination **on the basis of Calling Party Number**, you must configure the **BRI Port - Destination Port Determination - Calling Number Based** table.

To apply Destination Port Determination **on the basis of Destination Number**, you must configure the **BRI Port - Destination Port Determination - Destination Number Based** table.

Destination Port Determination on Mobile Ports

For Mobile Port, the system supports the following methods for Destination Port Determination:

- Fixed
- On the basis of Destination Number
- On the basis of Calling Party Number

To apply Destination Port Determination **on the basis of Calling Party Number**, you must configure the **Mobile Port - Destination Port Determination - Calling Number Based** table.

To apply Destination Port Determination **on the basis of Destination Number**, you must configure the **Mobile Port - Destination Port Determination - Destination Number Based** table.

Destination Port Determination on SIP Trunks

For SIP Trunks, the system supports the following methods for Destination Port Determination:

- Fixed
- On the basis of Destination Number
- On the basis of Calling Party Number

To apply Destination Port Determination **on the basis of Calling Party Number**, you must configure the **SIP Trunk - Destination Port Determination - Calling Number Based** table.

To apply Destination Port Determination **on the basis of Destination Number**, you must configure the **SIP Trunk - Destination Port Determination - Destination Number Based** table.

Destination Port Determination on T1/E1 Port

For T1/E1 Port with **Orientation Type - Terminal**, the system allows you to configure the following destination port determination methods by Port, Channel and MSN Number/DDI Number:

- Fixed
- On the basis of Destination Number
- On the basis of Calling Party Number

For T1/E1 Port with **Orientation Type - Network**, the system allows you to configure the following destination port determination methods by Port and Channel:

- Fixed
- On the basis of Destination Number
- On the basis of Calling Party Number

To apply Destination Port Determination **On the basis of Calling Party Number**, you must configure the **T1E1 Port - Destination Port Determination - Calling Number Based** table.

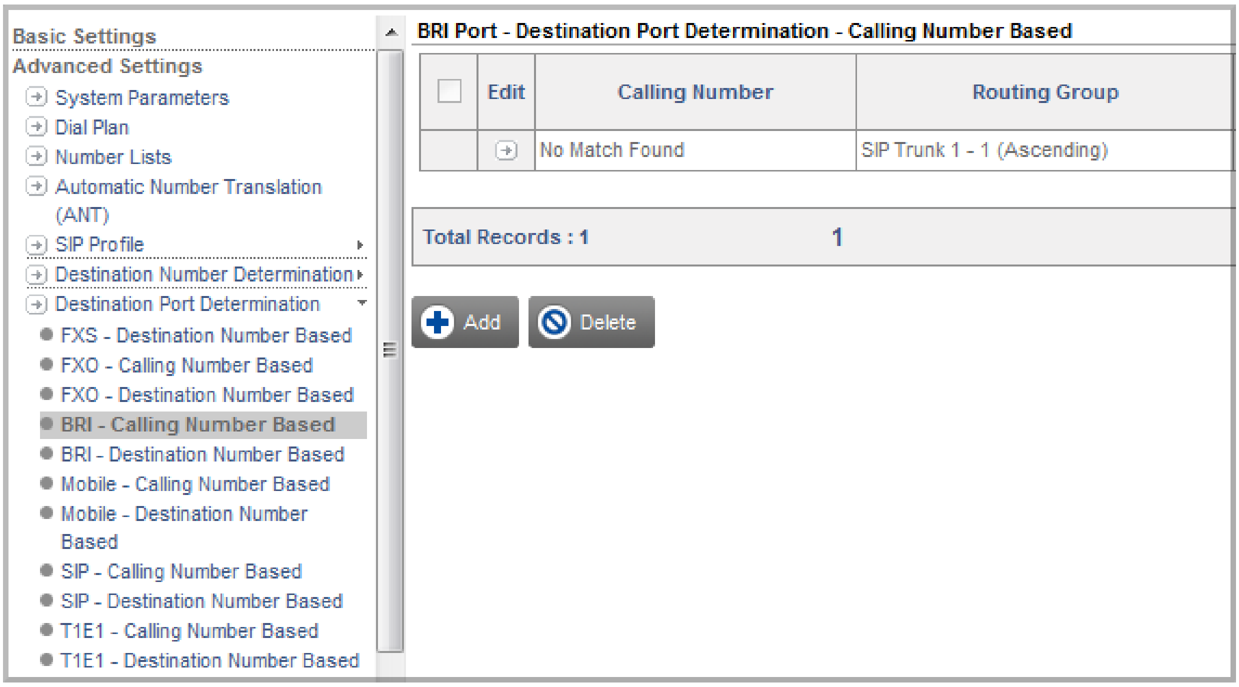
To apply Destination Port Determination **On the basis of Destination Number**, you must configure the **T1E1 Port - Destination Port Determination - Destination Number Based** table.

Configuring BRI Port-Calling Number Based Table

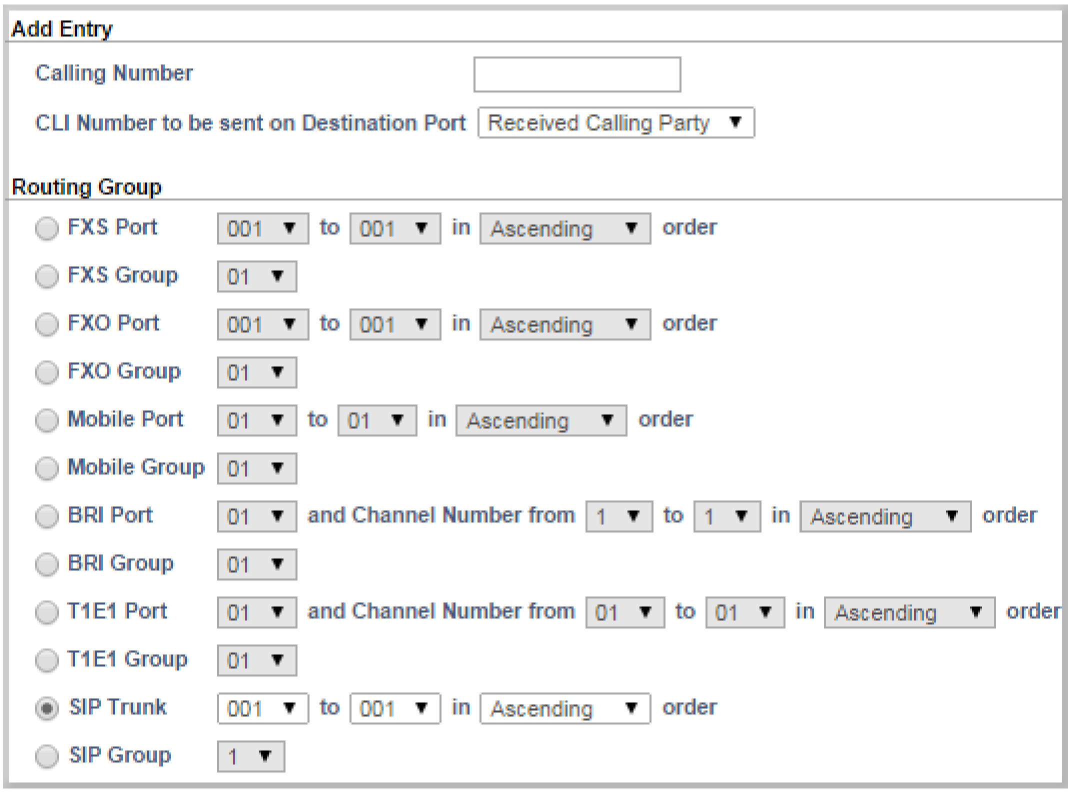
- Click the **Advanced Settings** link to expand.

- Click **BRI-Calling Number Based** under **Destination Port Determination**.

- The **BRI Port - Destination Port Determination - Calling Number Based** page opens.



- To add an entry, click **Add**. A new window opens.



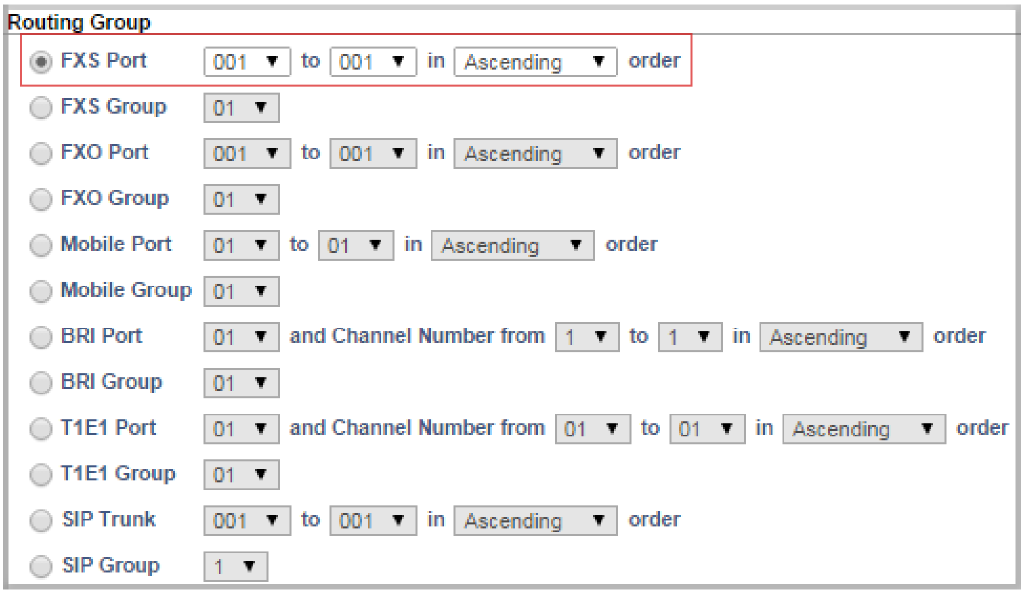
Configure the following parameters:

- In **Calling Number**, enter the number (max. 24 characters) from which you expect calls to be received. Valid digits are 0 to 9, *, #, (dot). Default: Blank.

- Select the **CLI Number to be sent on Destination Port**. You can select Received Calling Party or Received Called Party. Default: Received Calling Party.

CLI to be sent on Destination Port is applicable when FXS Port/ Group is determined as the Destination Port.

- Create the **Routing Group**.
  - To create a group of *sequential* **FXS Ports** as members,

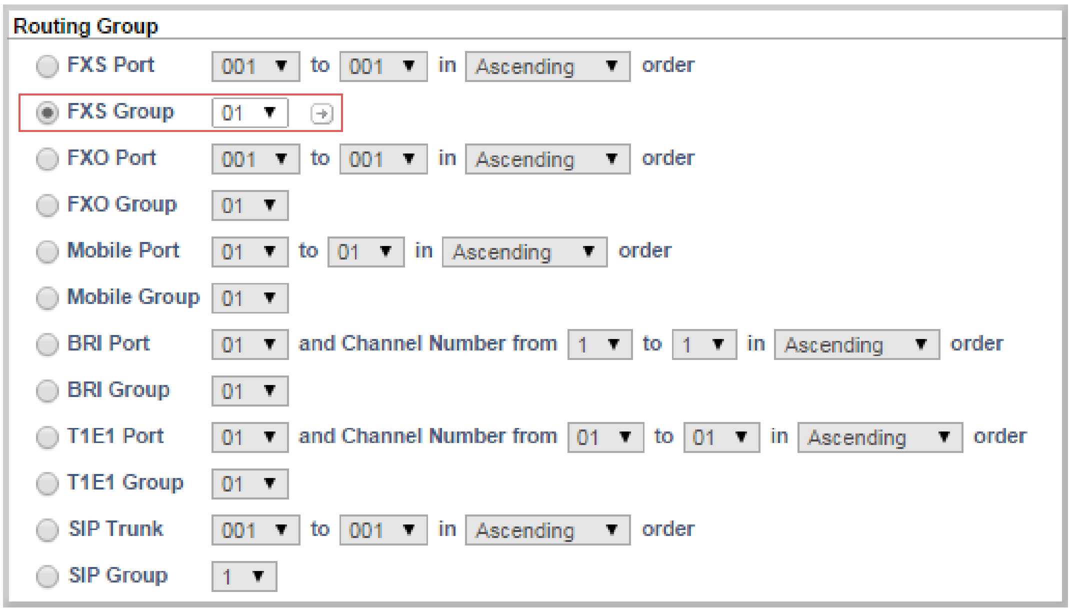


  - Select the desired **FXS Port** numbers as members. Default:1.
  - In **in - order**, select the order in which the system should hunt for a free member FXS Port to route the call.

    Select **Ascending** to start hunting from the first to the last member FXS Port. Select **Descending** to start hunting from the last to the first member FXS Port. Default: Ascending.

  - To create a group of *not-sequential* **FXS Ports** as members,

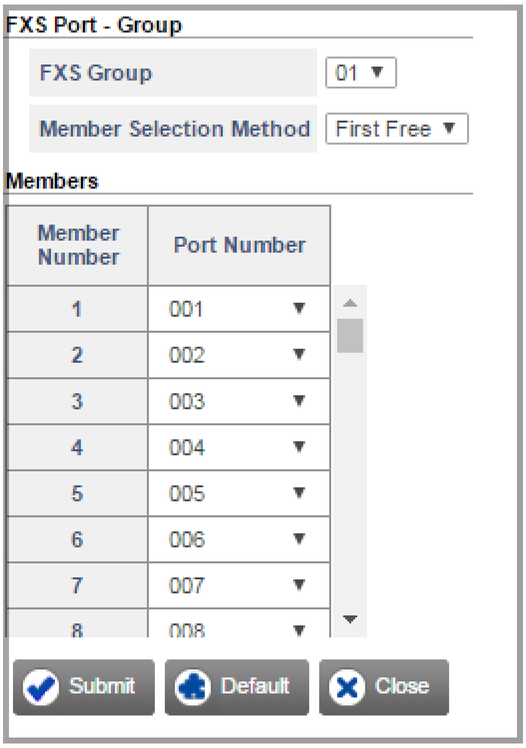
- Select a **FXS Group**.



- Select **FXS Group** number. Default: 1.

- Click **Settings** .

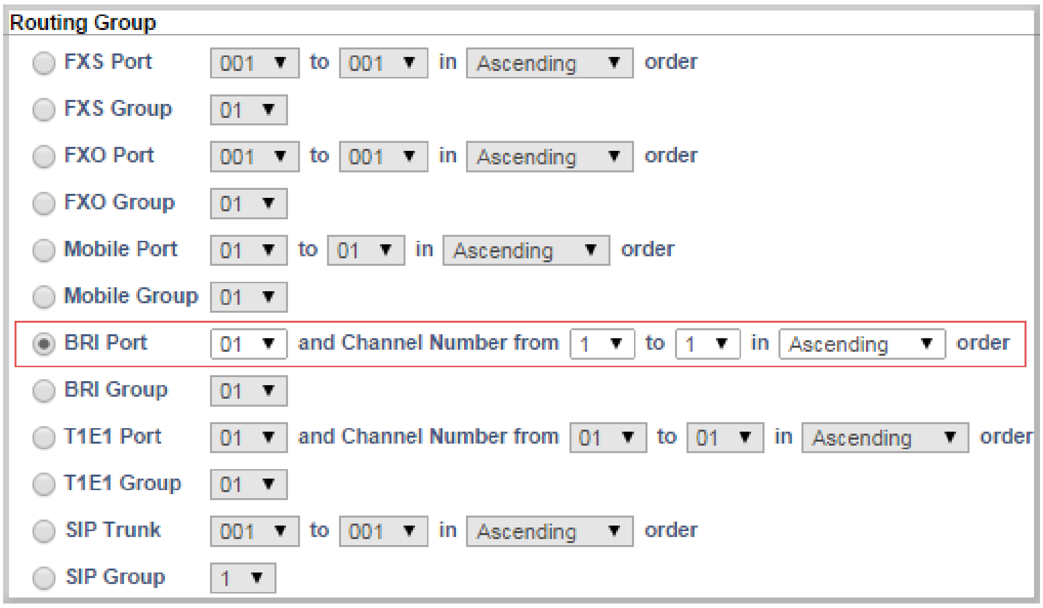
- The **FXS Port - Groups** window opens.



- Create the FXS Group. For detailed instructions on creating groups, see the topic "Group" under *Advanced Settings*.

- Similarly, you can create a group of *sequential* and *not-sequential* FXO Ports, Mobile Ports and SIP Trunks.

- To create a routing group of *sequential* **BRI Channels** as members,



  - Select the **BRI Port** Number. Default: 1.

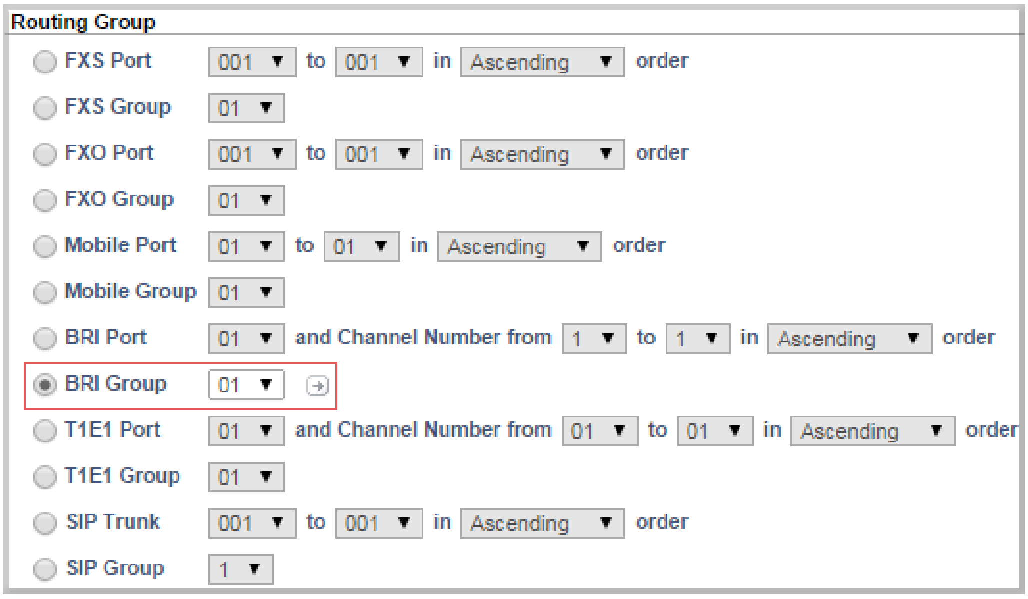
  - In Channel Number **From - to**, select the **Start Channel Number** and the **End Channel Number**, respectively.

  - In **in - order**, select the order in which the system should hunt for a free member Channel to route the call.

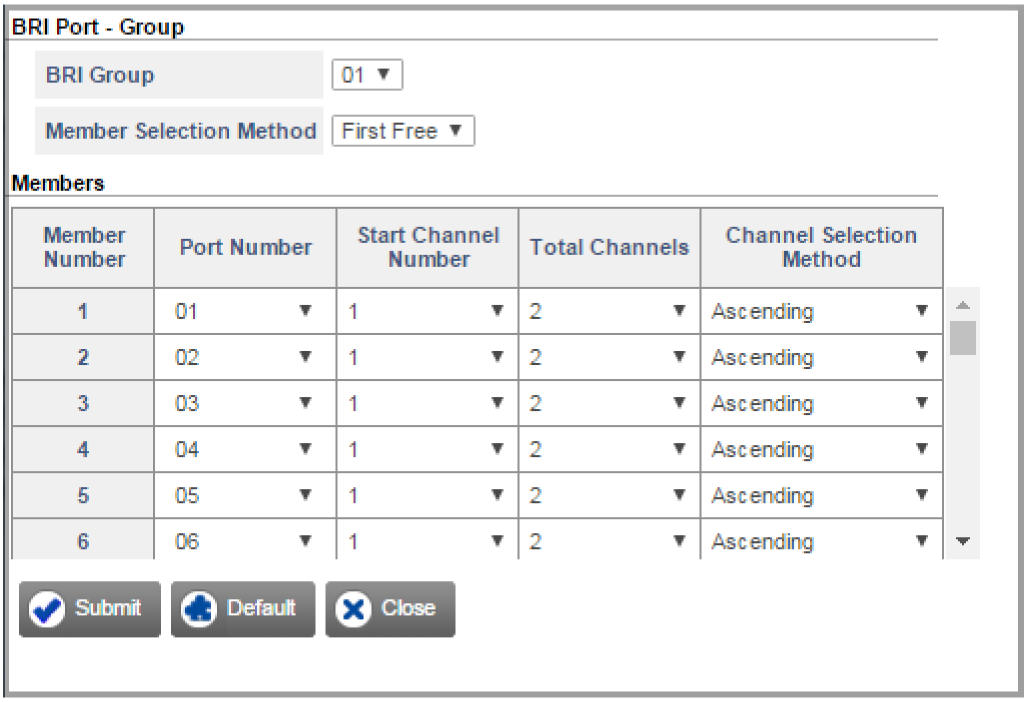
    Select **Ascending** to start hunting from the first to the last member channel. Select **Descending** to start hunting from the last to the first member channel. Default: Ascending.

- To create a group of *not-sequential* **BRI Channels** as members,

- Select **BRI Group**.



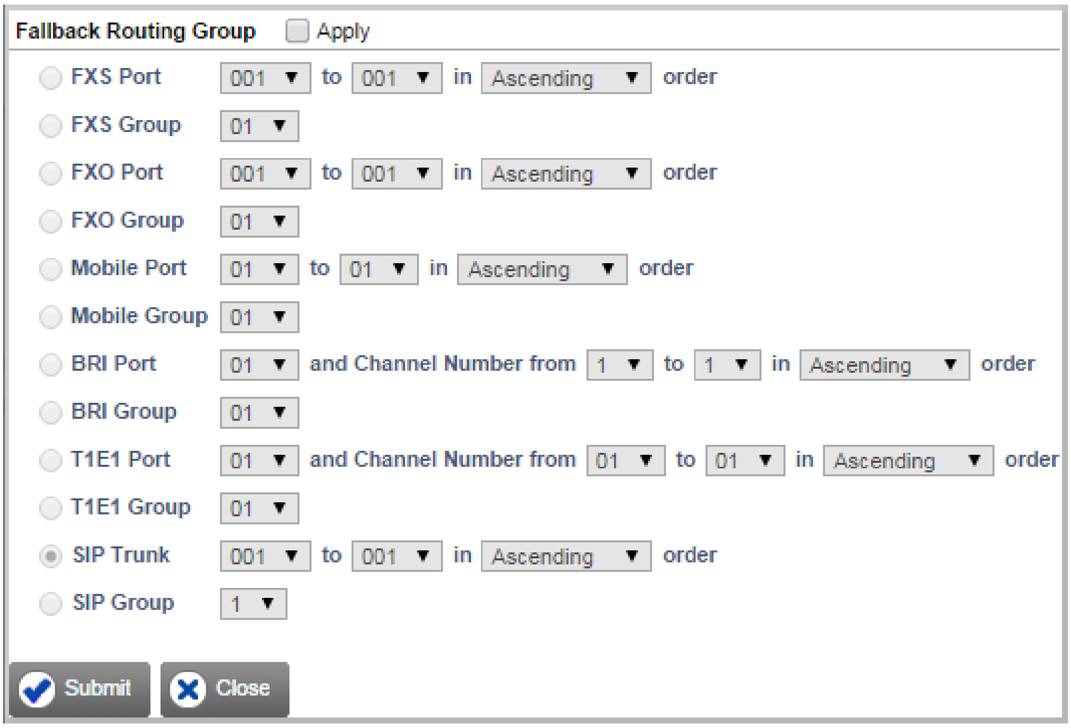
- Select a **BRI Group** number. Default:1.
- Click **Settings** .
- The **BRI Port - Groups** window opens.



- Create the BRI Group. For detailed instructions on creating groups, see the topic "Group" under *Advanced Settings.*

- Similarly, you can create a group of *sequential* and *not-sequential* T1E1 Ports.

- You may create the **Fallback Routing Group**.

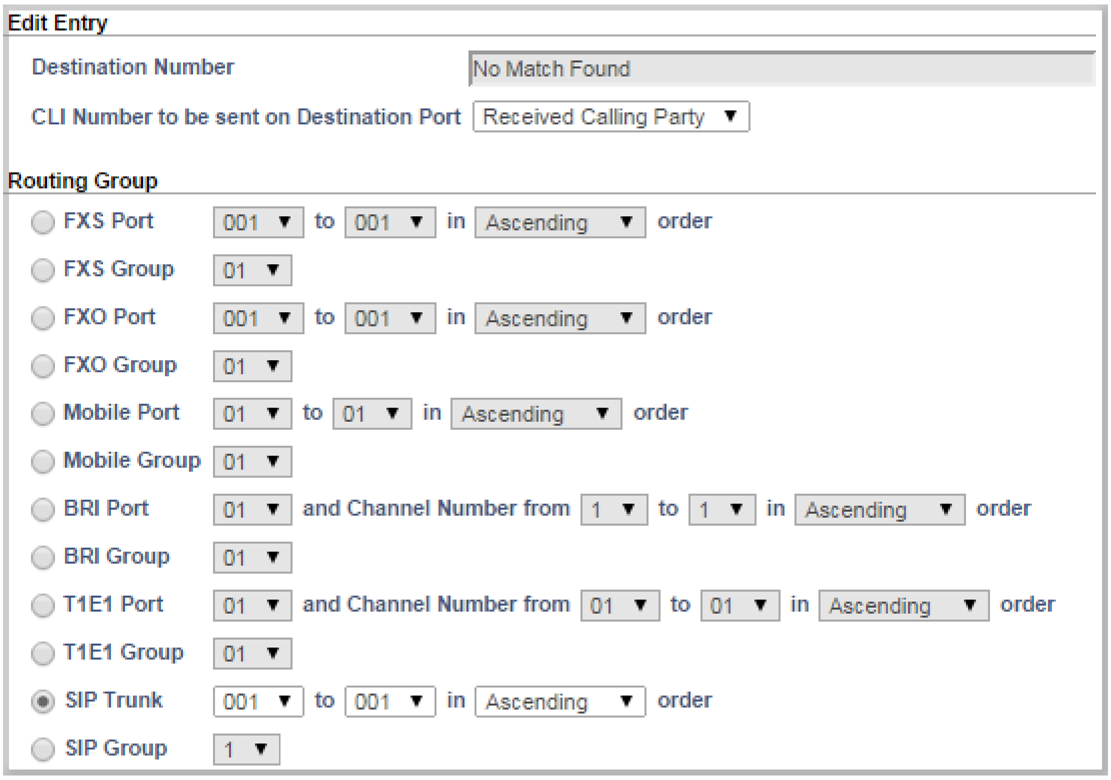


- To do this,
  - Select the **Apply** check box.
  - Follow the same instructions provided earlier for creating *sequential* and *not-sequential* groups of FXS Ports, FXO Ports, Mobile Ports, BRI Ports, T1E1 Ports and SIP Trunks.
- Click **Submit** to save changes. The **Add Entry** window closes.
- The entry you added appears in the **BRI Port - Destination Port Determination - Calling Number Based** table.
- Follow the same steps as above to add another entry to this table.
- To delete an entry, select the check box and click the **Delete** button.
- By default, SIP Trunk 1-1 (Ascending) is assigned as the Routing Group, for routing calls from numbers that do not match with any of the destination numbers you configured (No Match Found).

  To change the default Routing Group and to create the Fallback Routing Group for the No Match Found numbers entry,

  - For the No Match Found entry in the table, click **Edit** .

- The **Edit Entry** window opens.



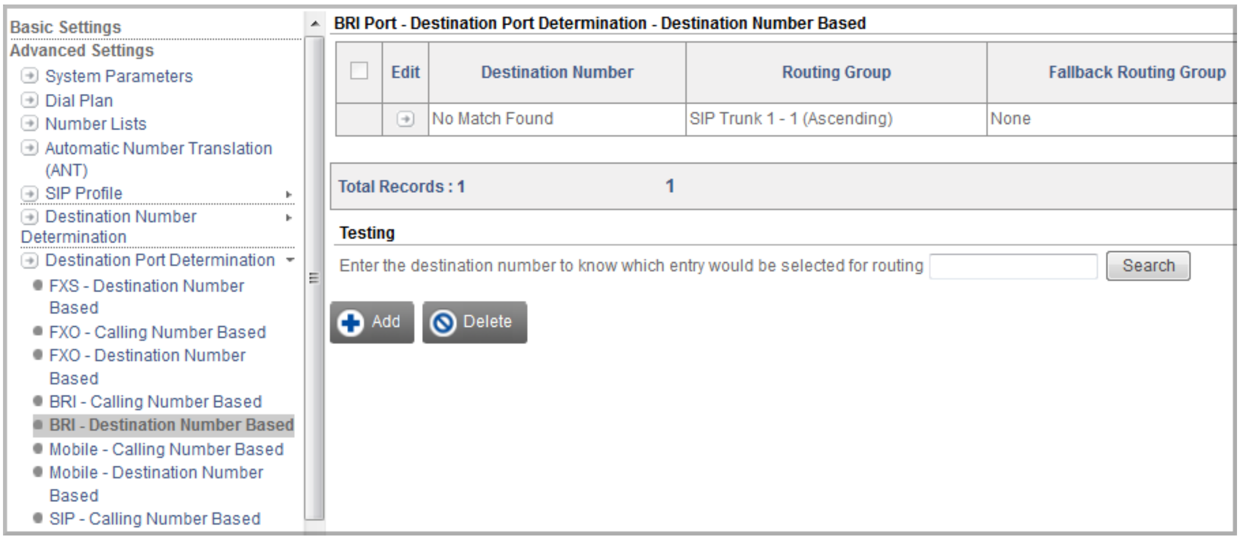
- Create the **Routing Group** and **Fallback Routing Group** as per your requirement.
- Click **Submit** and close the window.

- Close the window if you have finished adding/editing entries.
- To configure the table for the FXO Port, click **FXO-Calling Number Based.**
- To configure the table for the Mobile Port, click **Mobile-Calling Number Based.**
- To configure the table for the SIP Trunk, click **SIP-Calling Number Based.**
- To configure the table for the T1/E1 Port, click **T1E1-Calling Number Based.**

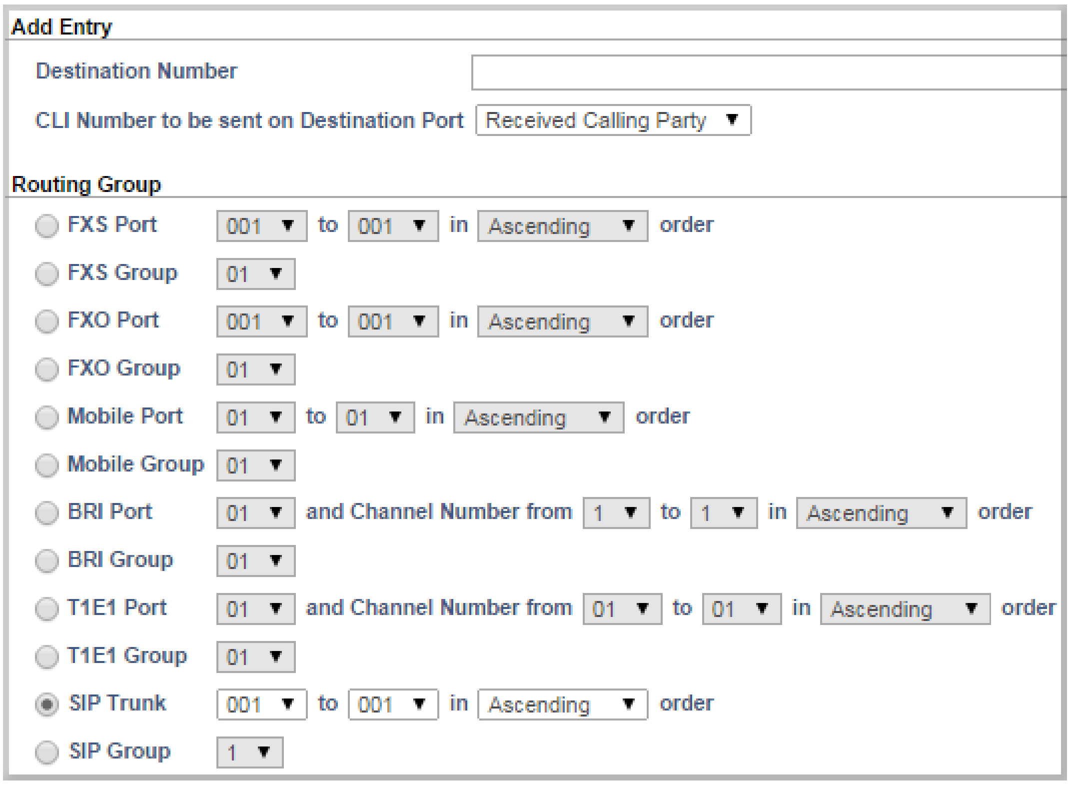
Configuring BRI Port-Destination Number Based Table

- Click the **Advanced Settings** link to expand.
- Click **BRI-Destination Number Based** under **Destination Port Determination.**

The **BRI Port - Destination Port Determination - Destination Number Based** page opens.



- To add an entry, click **Add**. A new window opens.



- In **Destination Number**, enter the number you expect the callers to dial. You may enter upto 64 characters (Digits + "Wildcard Characters") in this field. Valid characters are 0 to 9, *, #, X, T, Comma [,], Hyphen [-], Caret [^]. Default: Blank.

Wildcard Characters

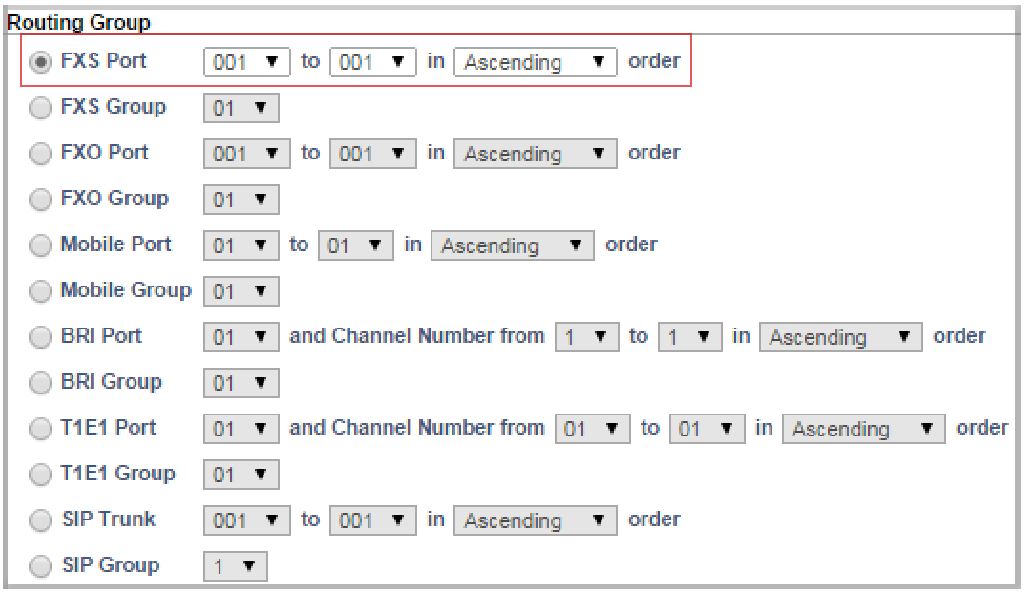
SARVAM UMG supports following characters.

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.

- Select the **CLI Number to be sent on Destination Port**. You can select Received Calling Party or Received Called Party. Default: Received Calling Party.

CLI to be sent on Destination Port is applicable when FXS Port/ Group is determined as the Destination Port.

- Create the **Routing Group**.
  - To create a group of *sequential* **FXS Ports** as members,

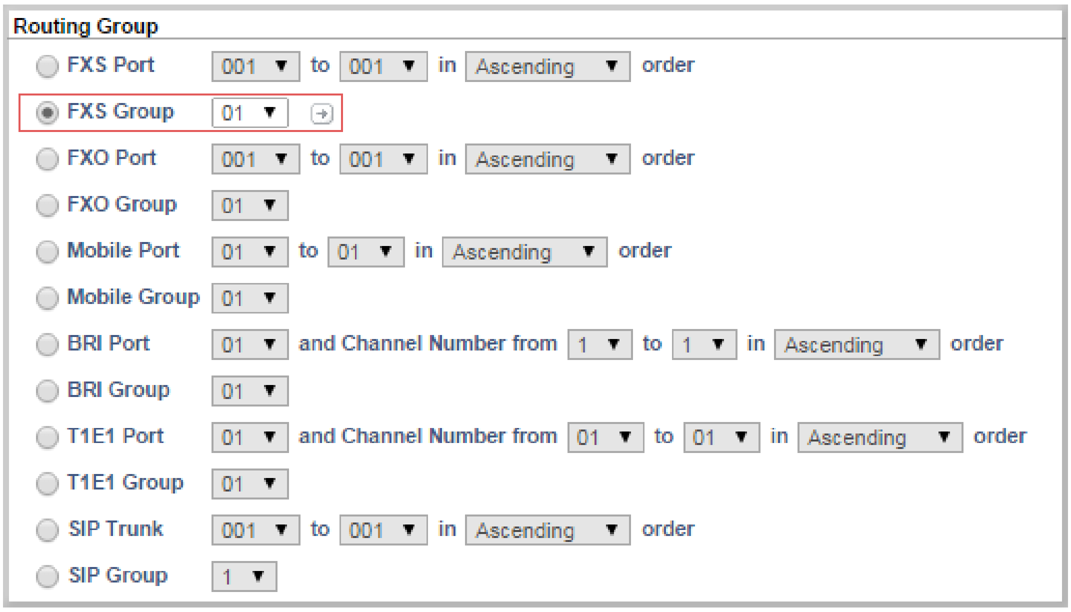


    - Select the desired **FXS Port** numbers as members. Default:1.

- In **in - order**, select the order in which the system should hunt for a free member FXS Port to route the call.

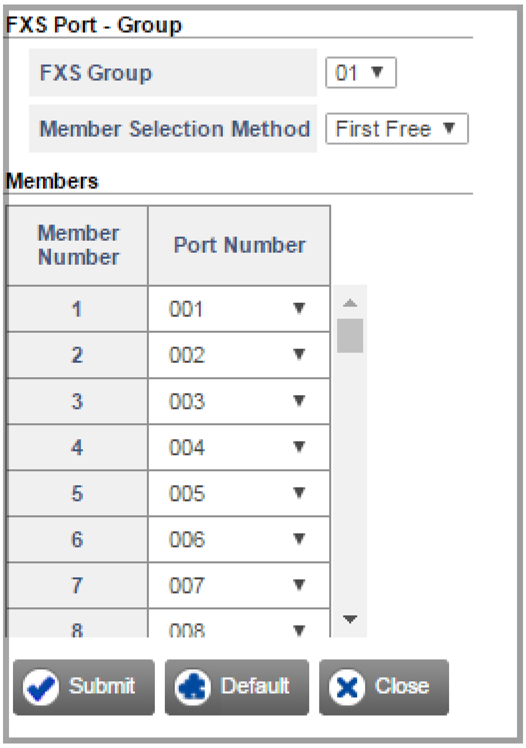
  Select **Ascending** to start hunting from the first to the last member FXS Port. Select **Descending** to start hunting from the last to the first member FXS Port. Default: Ascending.

- To create a group of *not-sequential* **FXS Ports** as members,
  - Select a **FXS Group**.



  - Select **FXS Group** number. Default:1.
  - Click **Settings** .

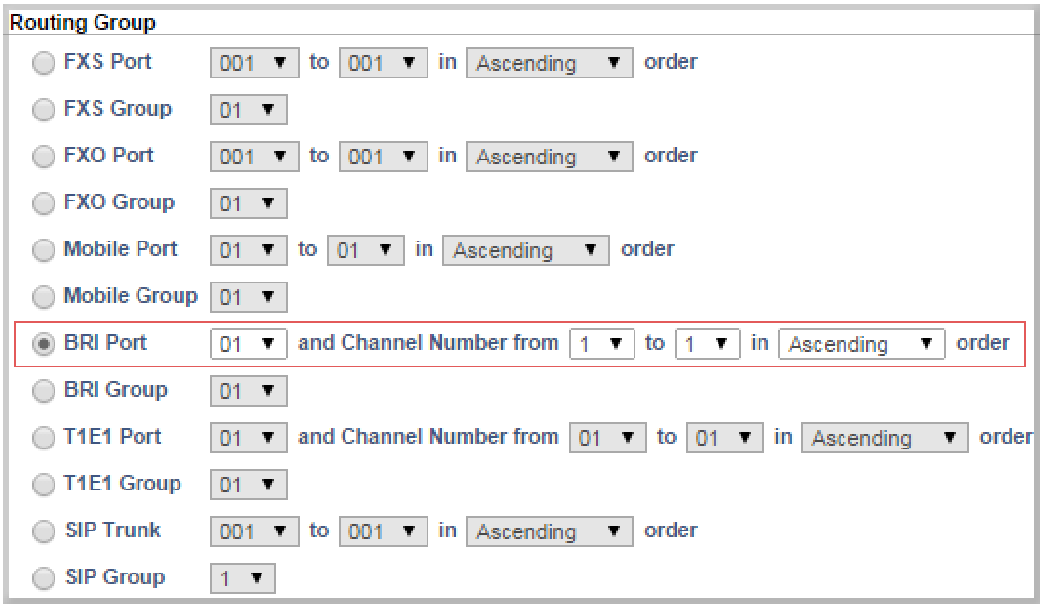
- The **FXS Port - Groups** window opens.



- Create the FXS Group. For detailed instructions on creating groups, see the topic "Group" under *Advanced Settings*.

- Similarly, you can create a group of *sequential* and *not-sequential* FXO Ports, Mobile Ports and SIP Trunks.

- To create a routing group of *sequential* **BRI Channels** as members,

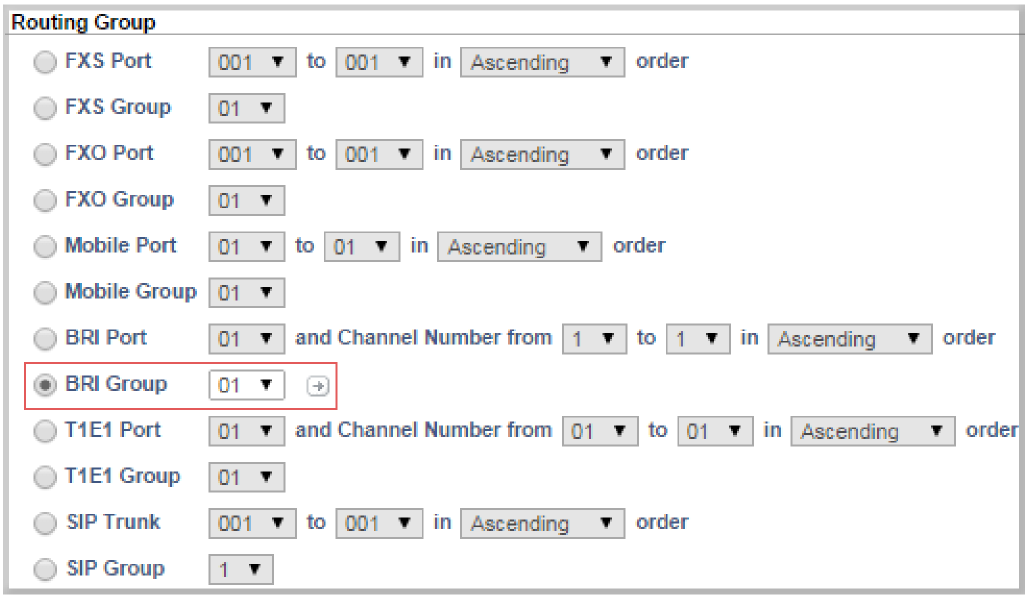


- Select the **BRI Port** Number. Default: 1.

- In Channel Number **From - to**, select the **Start Channel Number** and the **End Channel Number**, respectively.
- In **in - order**, select the order in which the system should hunt for a free member Channel to route the call.

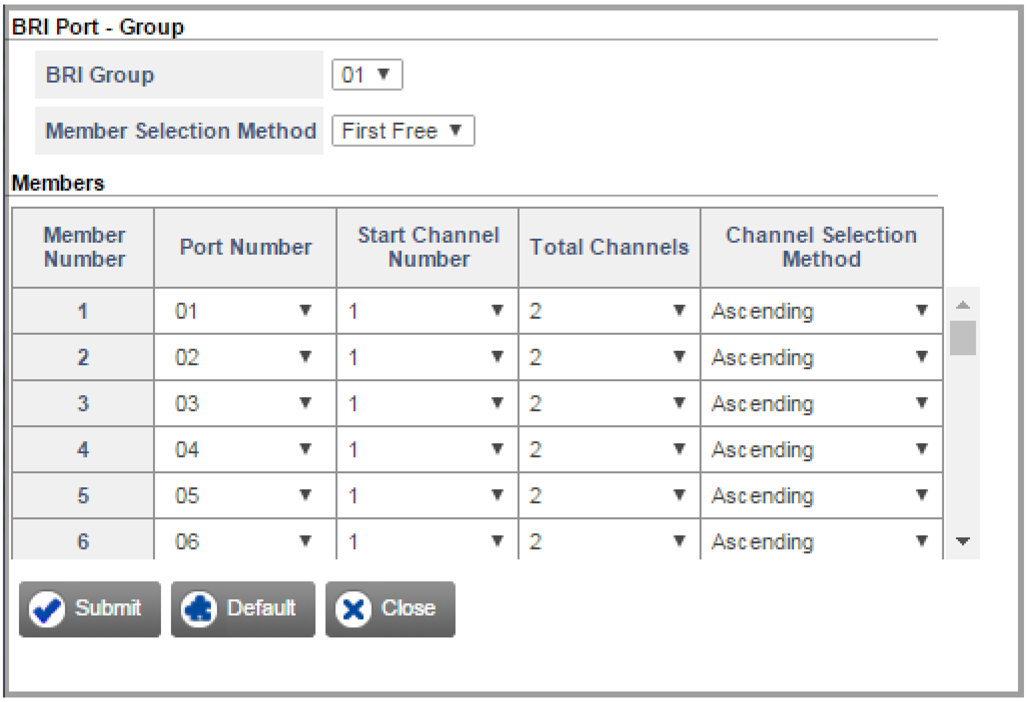
  Select **Ascending** to start hunting from the first to the last member channel. Select **Descending** to start hunting from the last to the first member channel. Default: Ascending.

- To create a group of *not-sequential* **BRI Channels** as members,
  - Select **BRI Group**.



  - Select a **BRI Group** number. Default:1.
  - Click **Settings** .

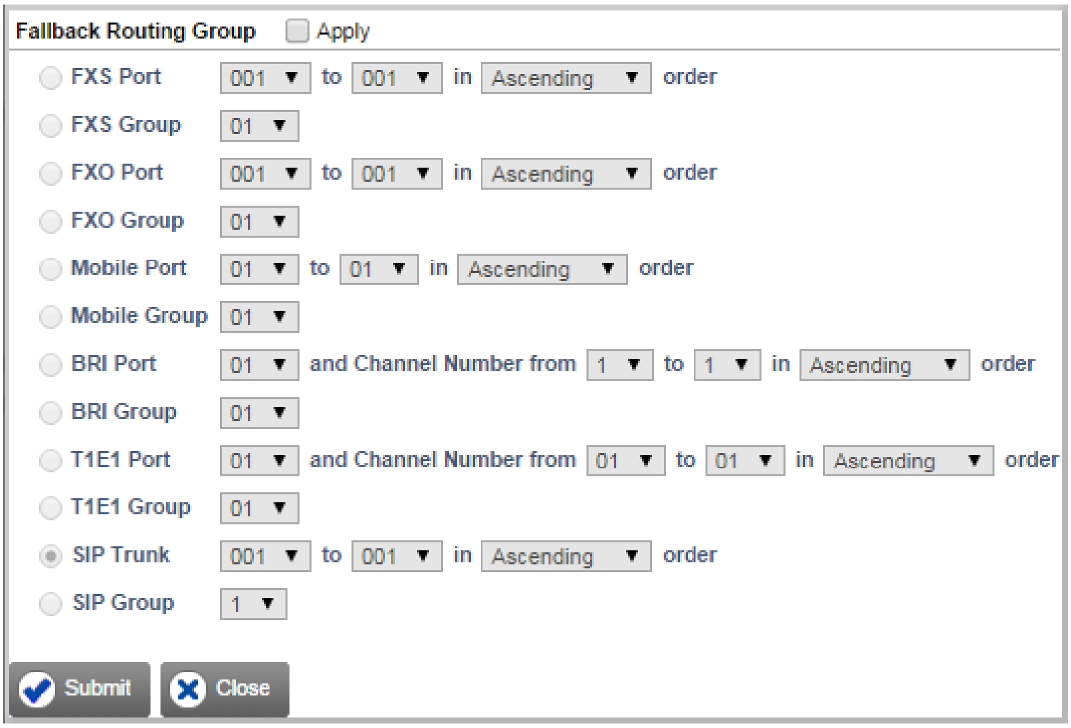
- The **BRI Port - Groups** window opens.



  - Create the BRI Group. For detailed instructions on creating groups, see the topic "Group" under *Advanced Settings.*

- Similarly, you can create a group of *sequential* and *not-sequential* T1E1 Ports.

- You may create the **Fallback Routing Group**.

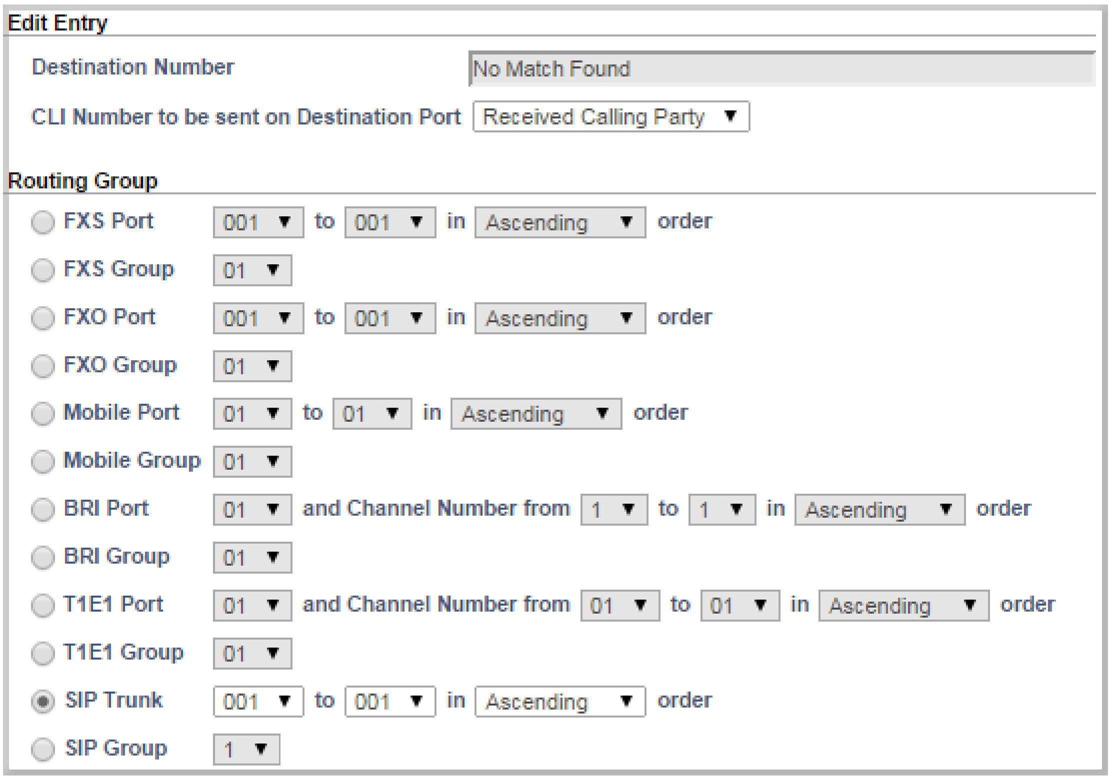


- To do this,
    - Select the **Apply** check box.
    - Follow the same instructions provided earlier for creating *sequential* and *not-sequential* groups of FXS Ports, FXO Ports, Mobile Ports, BRI Ports, T1E1 Ports and SIP Trunks.
- Click **Submit** to save changes. The **Add Entry** window closes.
- The entry you added appears in the **BRI Port - Destination Port Determination - Destination Number Based** table.
- Follow the same steps as above to add another entry to this table.
- To delete an entry, select the check box and click the **Delete** button.

If there are multiple entries in the Destination Number Based table, to search a particular entry in the table, under Testing enter the desired number to know which entry would be selected for routing.

- By default, SIP Trunk 1-1 (Ascending) is assigned as the Routing Group, for routing calls from numbers that do not match with any of the destination numbers you configured (No Match Found).

  To change the default Routing Group and to create the Fallback Routing Group for the No Match Found numbers entry,
    - For the No Match Found entry in the table, click **Edit**.
    - The **Edit Entry** window opens.



- Create the **Routing Group** and **Fallback Routing Group** as per your requirement.
- Click **Submit** and close the window.

- Close the window if you have finished adding/ editing entries.
- To configure the table for the FXS Port, click **FXS-Destination Number Based**.
- To configure the table for the FXO Port, click **FXO-Destination Number Based**.
- To configure the table for the Mobile Port, click **Mobile-Destination Number Based.**
- To configure the table for the SIP Trunk, click **SIP-Destination Number Based**.
- To configure the table for the T1/E1 Port, click **T1E1-Destination Number Based**.

Group

SARVAM UMG supports the following methods for determining the destination port for the calls originated on FXS Ports, FXO Ports, BRI Ports, Mobile Ports, SIP Trunks and T1E1 Ports:

- Fixed
- on the basis of Destination Number
- on the basis of Calling Party Number

The Destination Port Determination Method - on the basis of Calling Party Number is not applicable for the FXS Port.

A Routing Group may have *sequential* or *not-sequential* ports as members.

A Routing Group of *sequential* ports is formed when you select **FXS Port** or **FXO Port** or **BRI port** or **Mobile Port** or **SIP Trunk** or **T1E1 Port** as the destination port.

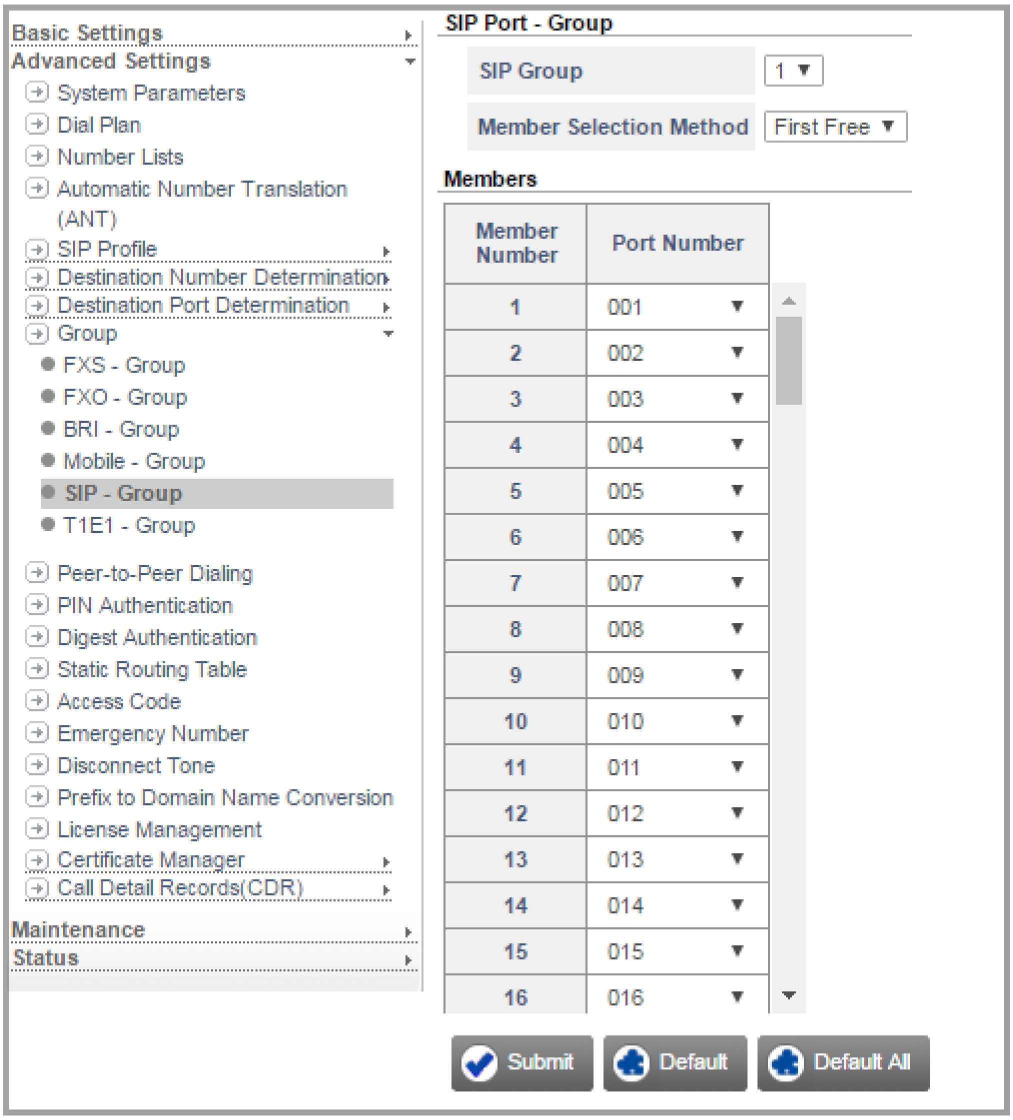
A Routing Group of *not-sequential* ports is formed when you select **FXS - Group** or **FXO - Group** or **BRI- Group** or **Mobile - Group** or **SIP - Group** or **T1E1 - Group** as the destination port. The **FXS/FXO/BRI/SIP/Mobile/T1E1 Group** has members of the same port type, but not in a sequence. For example, a SIP Group can have only SIP Trunks as members, whereas a Mobile Group can have only Mobile Ports as members.

Configuring Groups

To create a Group,

- Click the **Advanced Settings** link to expand.

- Click **SIP - Group** under **Group**.



- You can create 9 SIP Trunk Groups with 120 members in each group.
- Select a SIP Group Number from **1 to 9.**
- To configure **Members** in the Group,
- For each **Member Number**, select a SIP Trunk number as **Port Number**. There can be upto 120 members in a Group.

  If you do not want any more members in a group, select **None** as the **Port Number**. For example, you want two members in a group, select the SIP Trunk numbers for member 1 and 2, and set the remaining members in the group to None.
- Define the **Member Selection Method.** To route a call, the system checks availability of a free port. There are two options for port selection, namely:
  - **First Free:** The first port which is free will be used for routing the call each time. For example, SIP Group Number 1 has four members SIP Trunk 1 (Member 1), 2 (Member 2), 3 (Member 3) and 6

(Member 4). For every incoming call, SARVAM UMG will check the status of Member 1 (SIP Trunk 1) first. If free, the call will be routed using this port else system will check status of Member 2 (SIP Trunk 2) and so on.

- **Rotation:** The first call will be routed through the first member port and the subsequent call through the next member port and so on. For example, SIP Group Number 2 has four members SIP Trunk 6 (Member 1), 7 (Member 2), 8 (Member 3) and 9 (Member 4). For the first incoming call, SARVAM UMG will check the status of Member 1 (SIP Trunk 6). If free, the call will be routed using this port else system will check status of Member 2 (SIP Trunk 7) and so on.

  If the first call has been routed through Member 1 (SIP Trunk 6), then for the next call, system will check the status of Member 2 (SIP Trunk 7) but if the first call has been routed through Member 2 (SIP Trunk 7), then the system will check the status of Member 3 (SIP Trunk 8) to route the call. Similarly, for the subsequent calls the system will check the next member port in the group.

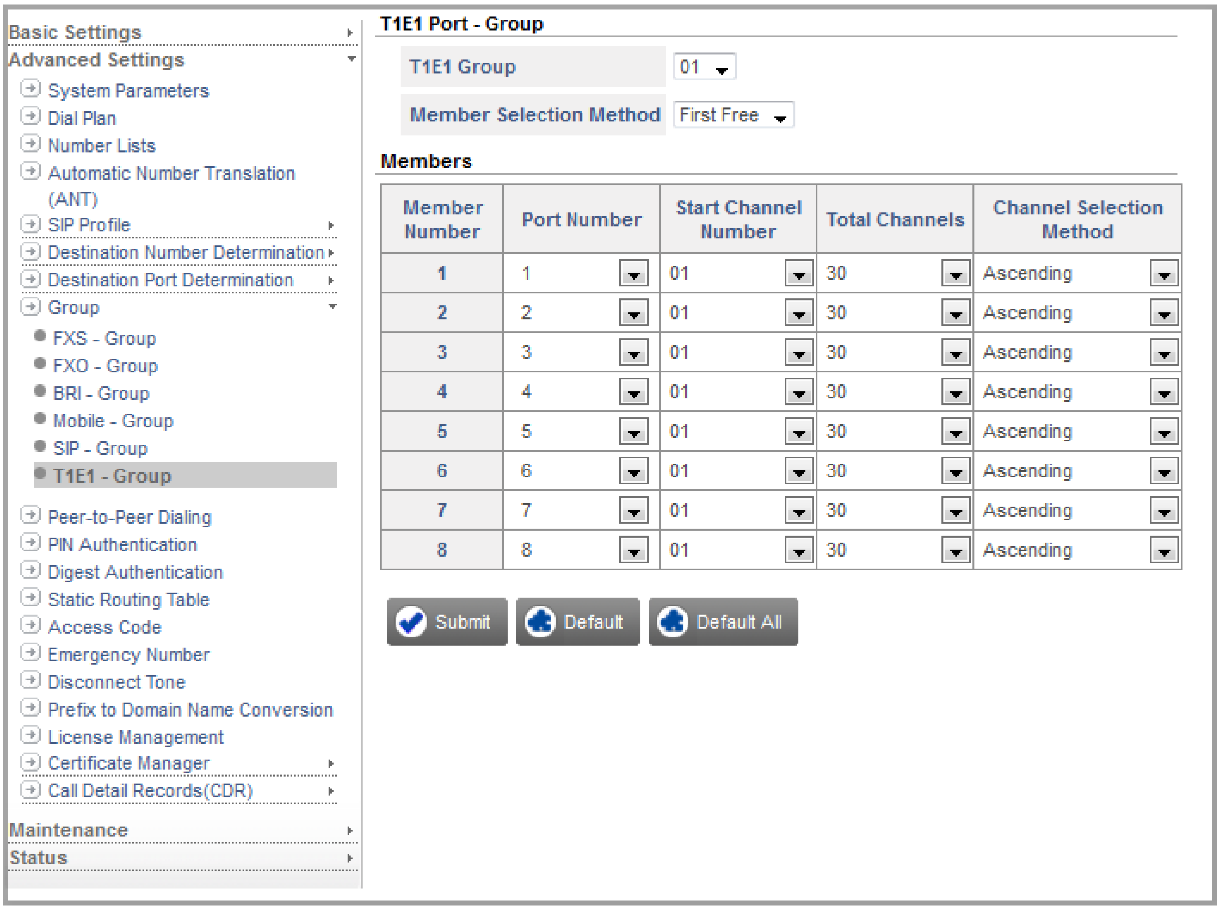
  Default: **First Free**.

- Click **Submit** to save the group.
- Select **Default**, to set the parameters of a particular group to default.
- Select **Default All**, to set the parameters of all the groups to default.
- Similarly, you can create FXS Groups, FXO Groups and Mobile Groups.
  - To create Groups of FXS Ports, click **FXS - Group**.
  - To create Groups of FXO Ports, click **FXO - Group**.
  - To create Groups of Mobile Ports, click **Mobile - Group**.

To create T1E1 Group,

- Click the **Advanced Settings** link to expand.

- Click **T1E1 - Group** under **Group**.



- You can create as many as 10 T1E1 Groups with maximum 8 members in each group.
- Select a T1E1 Group Number from **1 to 10.**
- To configure **Members** in the Group,
- For each **Member Number**, select a T1E1 Port number as **Port Number**. There can be upto 8 members in a Group.
- Define the **Member Selection Method**. To route a call, the system checks availability of a free port. There are two options for port selection, namely:
  - **First Free:** The first port which is free will be used for routing the call each time. For example, T1E1 Group Number 1 has four members T1E1 Port 1 (Member 1), 2 (Member 2), 3 (Member 3) and 6 (Member 4). For every incoming call, SARVAM UMG will check the status of Member 1 first. If free, the call will be routed using this port else system will check status of Member 2 and so on.
  - **Rotation:** The first call will be routed through the first member port and the subsequent call through the next member port and so on. For example, T1E1 Group Number 2 has four members T1E1 Port 3 (Member 1), 4 (Member 2), 5 (Member 3) and 6 (Member 4). For the first incoming call, SARVAM UMG will check the status of Member 1 (T1E1 Port 3). If free, the call will be routed using this port else system will check status of Member 2 (T1E1 Port 4) and so on.

If the first call has been routed through Member 1 (T1E1 Port 3), then for the next call, system will check the status of Member 2 (T1E1 Port 4) but if the first call has been routed through Member 2 (T1E1 Port 4), then the system will check the status of Member 3 (T1E1 Port 5) to route the call. Similarly, for the subsequent calls the system will check the next member port in the group.

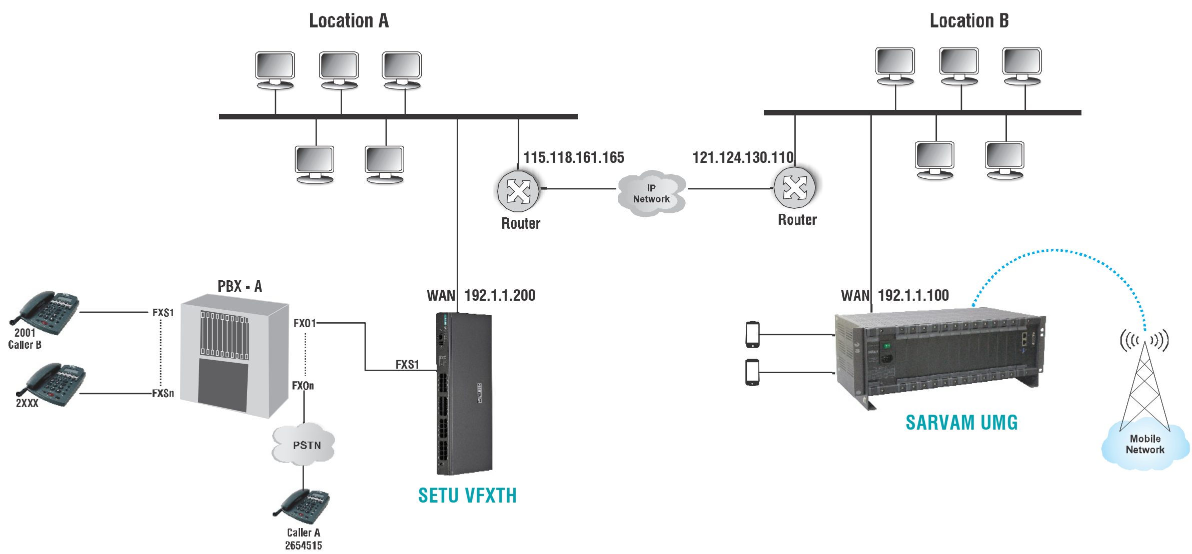
Default: **First Free**.

- Each Member can have multiple channels. For each Member,
  - Select the **Port Number.** If you do not want any more members in a group, select **None** as Port Number. For example, if you want two members, Member 1 and 2 in the Group, set Port Number for Member 3 and 4 to None.
  - Define the **Start Channel Number**. While determining the destination port, SARVAM UMG will select this channel first for routing the calls originated on the T1E1 Port. Valid range is 1 to 30. Default:1.
  - Define the **Total Channels** SARVAM UMG should check while routing the call. Valid range is 1 or 30. Default: 30.
  - Set the sequence in which SARVAM UMG should select the channel for routing the call as **Channel Selection Method.** You may select:
    - **Ascending:** When you select Ascending as Channel Selection Method, SARVAM UMG will route call to first channel. If found busy the call will be routed to the next channel.
    - **Descending:** If you select Descending, SARVAM UMG will route calls in the reverse sequence; starting from last channel to the first.
    - **Cyclic:** If you select Cyclic, SARVAM UMG will route the first call to the first channel; if found busy the call will be routed to the next channel. If the first call has been routed through the first channel then the system will check the second channel for the next call but if the first call has been routed through the second channel, the system will check the third channel to route the call. Similarly, subsequent calls will be routed.
- Click **Submit** to save the group.
- Select **Default**, to set the parameters of a particular group to default.
- Select **Default All**, to set the parameters of all the groups to default.
- Similarly, you can create BRI Groups.
  - To create Groups of BRI Ports, click **BRI- Group**.

Peer-to-Peer Dialing

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.

Let us understand how to use Peer-to-Peer Calling with the following illustration.



- Two offices are connected to the IP network.
- At Location A, a PBX (PBX A) and a Gateway (SETU VFXTH) is installed as shown above.
- SARVAM UMG is installed at Location B.
- Peer-to-Peer calls can be made between the two locations with suitable configuration of SARVAM UMG and the Gateway (SETU VFXTH).
- At **Location A**, you need to do the following configuration in SETU VFXTH:
  - Select a SIP Trunk to be used for this application and enable it. For example, SIP Trunk 1.
  - Set the **SIP Trunk Mode** of this trunk as **Peer-to-Peer**.
  - Keep the **SIP ID** of the SIP Trunk **blank**.

In the Router, you must configure the same SIP and RTP Ports as configured in the SETU VFXTH. In other words, you must configure Port Forwarding for SIP and RTP on the Router.

  - By default, **Allowed IP Address for Incoming SIP Message** is set to **As per Peer to Peer table**. In the Peer to Peer table at Location B, you must configure the IP Address of the Router at Location A.
  - Under *Handling of Incoming Calls* on the SIP Trunk, select the Incoming Call Routing option as **Route all incoming calls (with CLI) - to the Called Party Number**.

- For **SIP-1**, select the **Destination Port for Routing Calls** as **Fixed**, and create **Routing Group** as **FXS Port.**

- For **FXS Port**, select the **Destination Port for Routing Calls** as **Fixed**, and create **Routing Group** as **SIP Trunk 1** only.

For instructions on configuring SIP Trunk parameters, see “SIP Trunk” under *Basic Settings.*

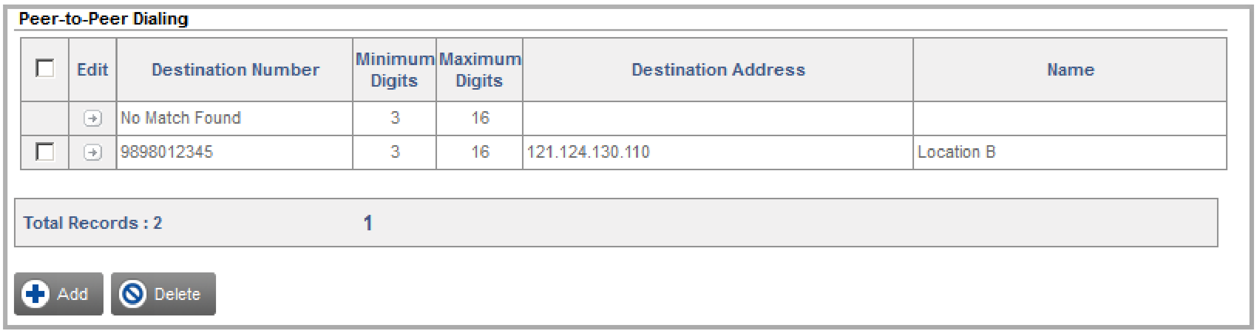
- Now, configure the **Peer-to-Peer Table**.

In this example, you would have to configure the Peer-to-Peer table as follows:

- At Location A, in the Number field of the Peer-to-Peer table, enter the Number you want to dial to call the phone at Location B. In this case, 9898012345.

- For the number you entered, in the Destination Address field in the table, enter the IP Address of the Router connected at Location B. In this case, 121.124.130.110

- The Peer-to-Peer table you configure for SETU VFXTH at Location A would look like this:



Peer-to-Peer Dialing

	Edit	Destination Number	Minimum Digits	Maximum Digits	Destination Address	Name
		No Match Found	3	16		
		9898012345	3	16	121.124.130.110	Location B

Total Records : 2 1

Add Delete

Instead of configuring the complete number string, you may configure only the prefix of the number to be dialed as follows, the system will place all calls that start with '9898' to the IP Address 121.124.130.110.

Destination Number	Destination Address	Name
No Match Found		
9898	121.124.130.110	Location B

- At **Location B**, you need to do the following configuration in SARVAM UMG

  - Select a SIP Trunk to be used for this application and enable it. For example, SIP Trunk 1.

  - Set the **SIP Trunk Mode** of this trunk as **Peer-to-Peer**.

  - Keep the **SIP ID** field of the SIP Trunk **blank**.

In the Router, you must configure the same SIP and RTP Ports as configured in the SARVAM UMG. In other words, you must configure Port Forwarding for SIP and RTP on the Router.

  - By default, **Allowed IP Address for Incoming SIP Message** is set to **As per Peer to Peer table**. In the Peer to Peer table at Location B, you must configure the IP Address of the Router at Location A.

- Under *Handling of Incoming Calls* on the SIP Trunk, select the Incoming Call Routing option as **Route all incoming calls (with CLI) - to the Called Party Number**.

- For **SIP-1**, select the **Destination Port for Routing Calls** as **Fixed**, and create **Routing Group** as **Mobile Port**.

- For **Mobile Port**, select the **Destination Port for Routing Calls** as **Fixed**, and create **Routing Group** as **SIP Trunk 1** only.

For instructions on configuring SIP Trunk parameters, see “SIP Trunk” under *Basic Settings.*

In this example, you would have to configure the **Peer-to-Peer table** as follows:

- At Location B, in the Number field of the Peer-to-Peer table, enter the Number you want to dial to call the phone at Location A. In this case, 2001.

- For the number you entered in the Destination Address field in the table, enter the IP Address of the Router connected at Location A. In this case, 115.118.161.165

- The Peer-to-Peer table you configure for SARVAM UMG at Location B would look like this:

- Configure PBX at location A such that calls received on the FXO Port of the PBX are routed to the FXS Port in sequential order, that is, calls to 2001 are routed to FXS 1 and so on. Similarly, when any FXS Port user dials a number starting with '9898', it should be routed using the FXO Port of the PBX to the FXS Port of the SETU VFXTH.

- When user 2001 of Location-A calls 9898012345, the call is routed using the FXO Port of the PBX to FXS Port of the SETU VFXTH. Further, it will be routed using the SIP Trunk of the SETU VFXTH to the IP address 121.124.130.110, as the system finds a matching entry for the dialed number in the Peer-to-Peer table.

- On receiving a call, the SARVAM UMG at Location B routes this call through the Mobile Port of the SARVAM UMG to the Mobile user 9898012345.

- Similarly, when the Mobile user 9898012345 of Location B calls 2001, the call is received on the SIP Trunk of the SARVAM UMG and is placed to the IP address 115.118.161.165, as the system finds a matching entry for the dialed number in the Peer-to-Peer table.

- On receiving a call, the SETU VFXTH at Location-A routes this call through the FXS Port of the SETU VFXTH to the FXO Port of the PBX, which is further routed to 2001.

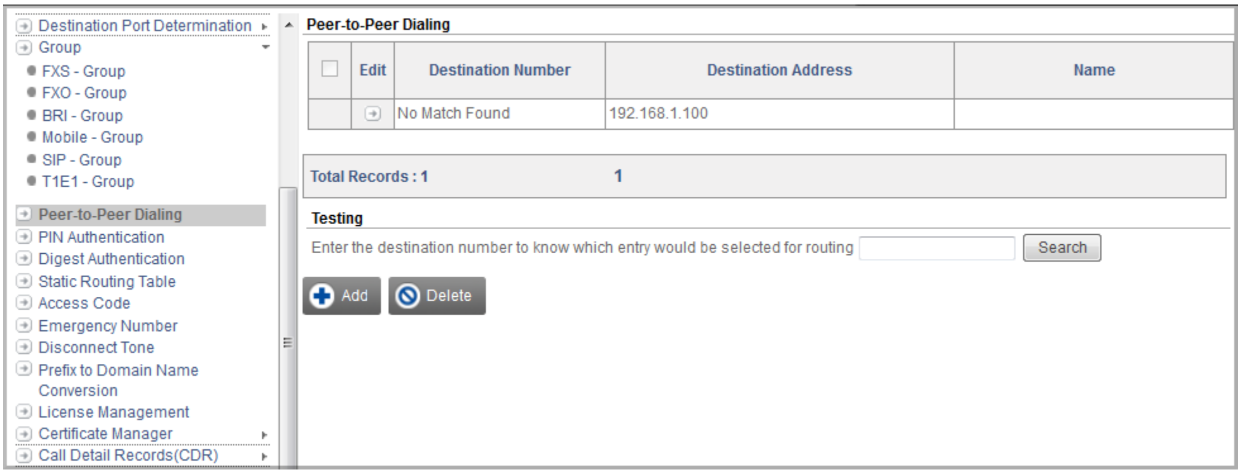
How to Configure

To use Peer-to-Peer calling, you must configure the related SIP Trunk parameters for the Peer-to-Peer application, namely: SIP Trunk Mode, Peer-to-Peer Table, SIP ID and Handling of Incoming Calls. For instructions, see "SIP Trunk" under Basic Settings.

You can also configure the Peer-to-Peer Table from the SIP Trunk page under Basic Settings.

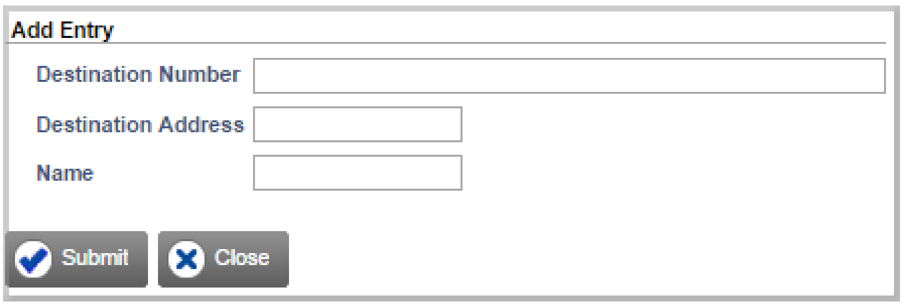
To configure the Peer-to-Peer Table,

- Click the **Advanced Settings** link to expand.
- Click the **Peer-to-Peer Dialing** link.
- The **Peer-to-Peer Dialing** table opens.



In the Peer-to-Peer table, the first entry is reserved for No Match Found.

- Click the **Add** button. The **Add Entry** window opens.



- In **Destination Number**, enter the number you expect the callers to dial. You may enter upto 64 characters (Digits + "Wildcard Characters") in this field. Valid characters: 0 to 9, *, #, X, T, Comma [,], Hyphen [-], Caret [^]. Default: Blank.

If the number to be dialed out is <dialednumber@destination address>, for example, 1234@abc.com, you must enter 1234 in this field.

Wildcard Characters

SARVAM UMG 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.

- In **Destination Address,** enter the domain name or IP Address to where the call is to be placed. The Destination Address may consists of upto 40 characters (maximum). Default: 192.168.1.100.

  For example, if the peer-to-peer number to be dialed out is 1234@abc.com, enter abc.com as Destination Address. If the number is 1234@192.168.1.197, enter 192.168.1.197 as the Destination Address. The Destination Address can also be in the form of Address: Port number.

- In **Name**, enter a name to identify the number string you configured. It may be the name of your contact or any name you wish to assign to the number string. The name may consist of 24 characters (maximum). Default: Blank.

  The name you configure here will not be used in SIP signaling.

- Click **Submit** to save your entries.

If there are multiple entries in the Peer to Peer table, to search a particular entry in the table, under Testing enter the desired number to know which entry would be selected for routing.

PIN Authentication

PIN Authentication is a necessary security feature to restrict access to the system and prevent possible misuse of the resources.

You can use PIN Authentication on the Source Port to establish the identity of callers before their call is processed by SARVAM UMG.

PIN Authentication can be used on the Source Port only if the incoming call routing for the Source Port is set to ***Route calls After Answering the Call and Collecting the Digits***.

To be able to use PIN Authentication, this feature must be enabled on the Source Port and the PIN Authentication table must be configured.

The PIN Authentication table stores upto 500 PIN Numbers and their corresponding authentication Passwords.

When you enable PIN Authentication on the Source Port, SARVAM UMG answers the incoming call on the port and plays the prompt tone. It waits for the caller to dial the PIN Number and the Password. It collects the digits dialed by the caller and matches them with the PIN Authentication table.

When a match is found in the table, SARVAM UMG authenticates the caller and allows the call to be processed.

If the digits dialed by the caller do not match with any entry in this table, SARVAM UMG allows the caller to make two more attempts to dial a valid PIN Number and Password. If the caller fails to dial the correct PIN and Password in all the attempts, the system disconnects the call.

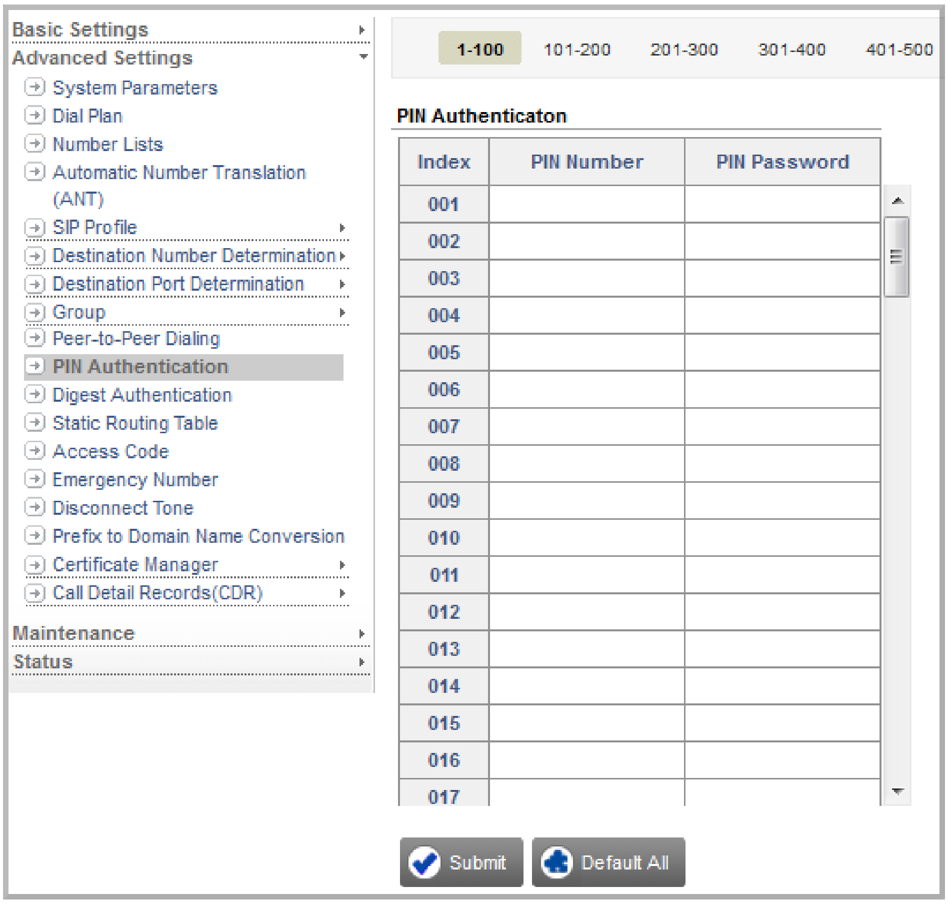
Configuring PIN Authentication

To use this feature, you must enable PIN Authentication on the desired — FXO Ports, SIP Trunks, Mobile Ports, BRI Ports (Terminal) and T1E1 Ports (Terminal) — and configure the PIN Authentication Table.

To configure PIN Authentication table,

- Click the **Advanced Settings** link to expand.

- Click the **PIN Authentication** link.



- Now, configure the **PIN Authentication** table.

  - In **PIN Number**, enter the numbers with which callers will authenticate themselves. Default: Blank. The digits 0 to 9, * and # are allowed in PIN Numbers.

The length of the PIN Number must not exceed four digits. If you enter a PIN Number that is less than 4 digits, the system will add leading zeros. The caller must also dial the PIN Number with the leading zeros to authenticate.

  - For each PIN Number you store, enter an authenticating password in **PIN Password**. The password can be of a maximum of four digits. The digits 0 to 9, * and # allowed. Default: Blank.

- Click **Submit** to save the entries.

- Now enable PIN Authentication on the desired — FXO Ports, SIP Trunks, Mobile Ports, BRI Ports (Terminal) and T1E1 Ports (Terminal) — port.

  To do so, first you need to select **After Answering the Call and Collecting the Digits** as the Incoming Call Routing option under *Handling of Incoming Calls*. Now, enable **Prompt caller to enter PIN** on the respective port.

  Under “Basic Settings”, see “FXO Port”, “SIP Trunk”, “Mobile Port”, “BRI Port - Terminal”, “T1 Port” and “E1 Port” for instructions.

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 authorised to make the request. When the digest credentials of the originator — User Name and Password — in the INVITE message are authenticated and accepted by the recipient, the originator and the recipient are connected.

SARVAM UMG supports Digest Authentication. The Digest Authentication feature works on the basis of the Digest Authentication Table in which the credentials — User Name and Password — of trusted/ authorised calling party SIP devices are stored. You must enable the Digest Authentication on the SIP Trunk and configure the Digest Authentication table.

SARVAM UMG will check the Digest Authentication table,

- when you enable this feature on a SIP Trunk.
- when SIP Trunk mode is Peer to Peer and **Allowed IP Address for Incoming SIP Message** is set to **Any**.

For all incoming calls (SIP requests),

- SARVAM UMG will challenge the identity of the calling party, that is, the SIP device initiating the request to send its digest credentials.
- When the calling party sends its credentials, SARVAM UMG 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 UMG will consider it as invalid authentication information and reject the call.

You may use Digest Authentication to:

- restrict access to SARVAM UMG to specific callers.
- prevent unwanted or malicious calls.

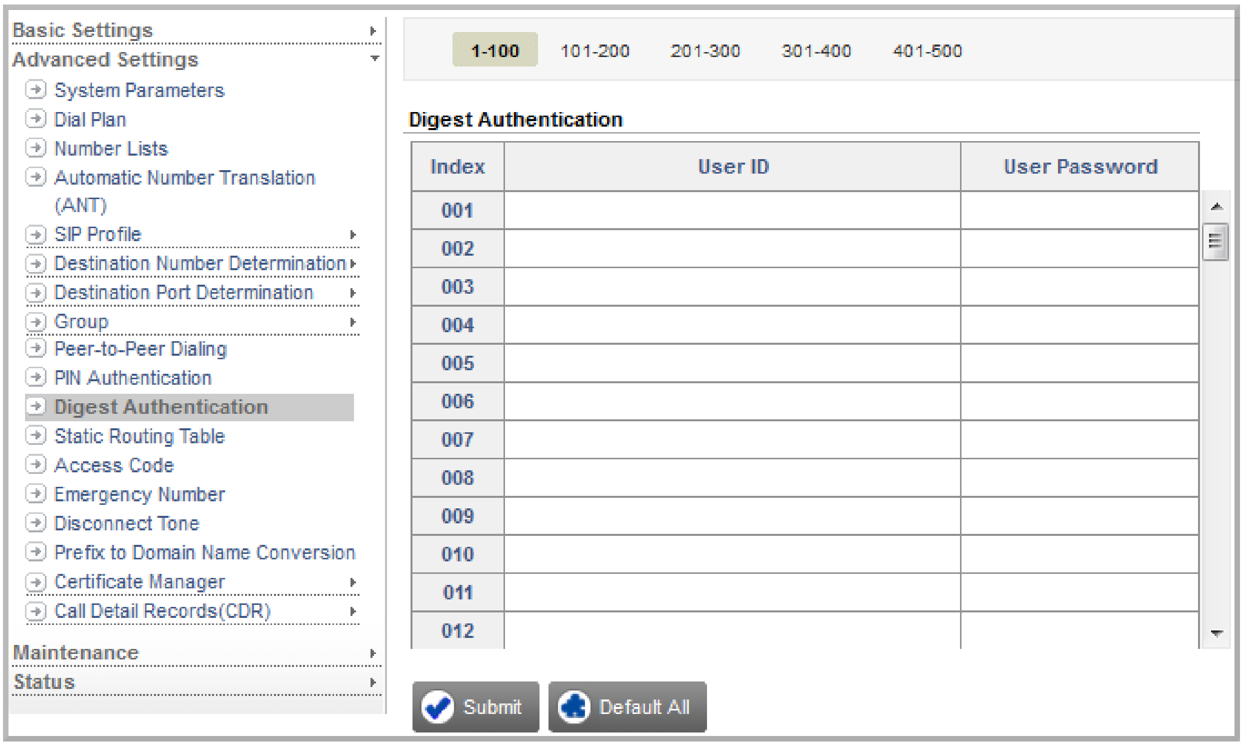
Configuring Digest Authentication

To use this feature, you must enable **Digest Authentication** on the desired SIP Trunk and configure the Digest Authentication Table.

To configure Digest Authentication Table,

- Click the **Advanced Settings** link to expand.
- Click the **Digest Authentication** link.

The **Digest Authentication** Table window opens. You can configure upto 500 entries in this table. This Table is common for all SIP Trunks.



- Enter the user name assigned to the caller/ calling device in the **User ID**. SARVAM UMG will use this User ID to match the digest credentials sent by the caller/ calling devices when challenged.

  Make sure the User ID you enter here and the User ID assigned at the *calling end* are the same. The User ID may consist of a maximum of 40 characters. Default: Blank.

- Enter the password to authenticate the user ID in **User Password**. The password may consist of a maximum of 24 characters. Default: Blank.

  Make sure the User Password you enter here and the User Password assigned at the calling end are the same.

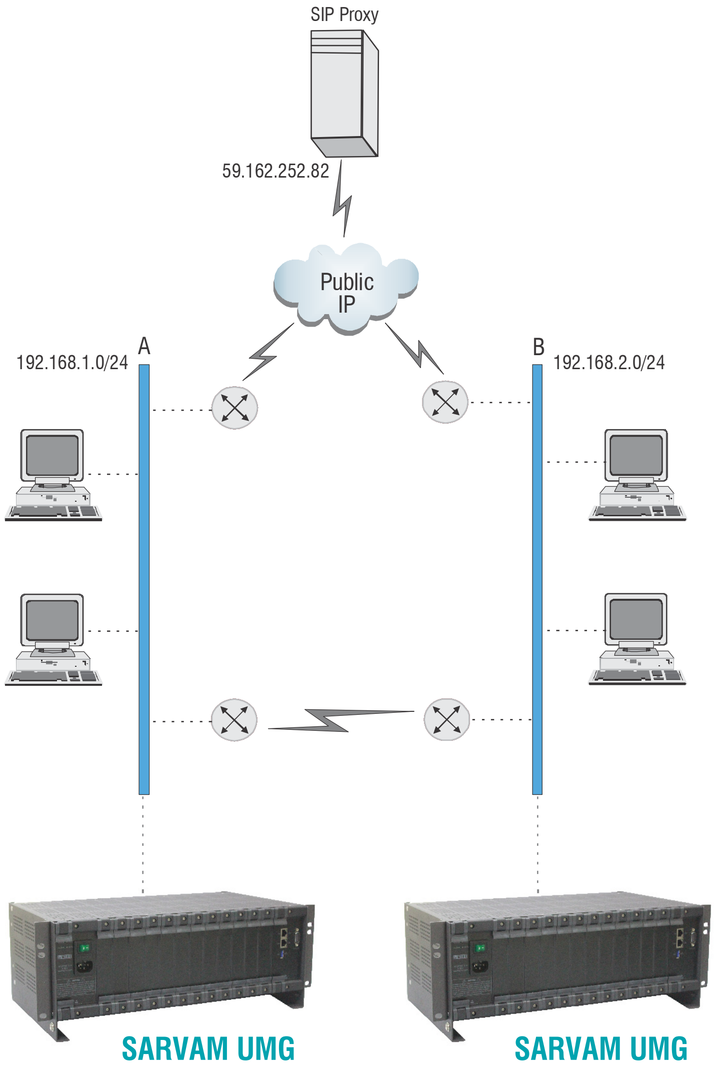
- Click **Submit** to save the entries.
- Make sure you enable Digest Authentication on the desired SIP Trunk. For instructions, see “SIP Trunk” under “Basic Settings”.

Static Routing

Static Routing Table is required when you have more than one router (gateway) in your network and you want SARVAM UMG 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.

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. SARVAM UMG is connected at both sites 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 UMG sends packets, if the final destination IP Address and SARVAM UMG are not in the same Subnet, the system will check the Static Routing Table.

If a perfect match is found, SARVAM UMG will start sending the IP packets to the corresponding Gateway Address configured in the table.

If no match is found, SARVAM UMG will send the IP Packets to the **Default Gateway Address** (Network Connection Type) you configured in the "Network" page.

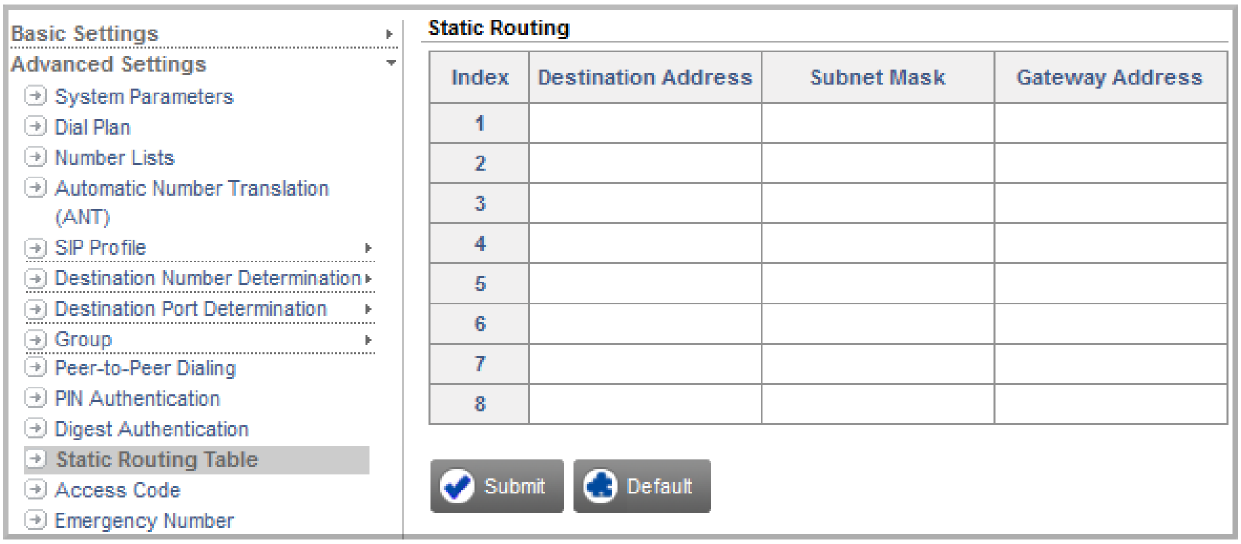
- *The Static Routing Table is common for all SIP Trunks.*
- *The Static Routing Table is applicable only when the Network Connection is established through WAN.*

Configuring Static Routing Table

The Static Routing Table must be configured at each location where SARVAM UMG is installed. To configure the Static Routing Table,

- Click the **Advanced Settings** link to expand.
- Click the **Static Routing Table** link.

- The Static Routing Table page opens.



The Static Routing Table allows you to configure upto 8 entries. Each entry is stored against an Index number.

For each entry, you must configure the following:

- In **Destination Address**, enter the address of the final destination where the call is to be made. This can be a device IP Address or Network Address.
- In **Subnet Mask**, enter the subnet mask to be applied on the destination address.
- In **Gateway Address**, enter 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 UMG.

- Click **Submit** to save your entries.

As per the above example, the Static Routing Table of SARVAM UMG 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			

- 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 UMG at Location A.

With the Static Routing Table configured, all calls made by SARVAM UMG 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 UMG to addresses other than 192.168.2.0/ 24 will be routed through the Default Gateway.

Similarly, configure the Static Routing Table in SARVAM UMG at location B to enable calling from Location B to Location A.

Access Code

Access Code is a string of digits dialed to use a feature. SARVAM UMG users can access the following features and facilities by dialing their Access Codes from a phone.

Feature/Function	Default Access Code
System Engineer (SE) Programming	#19[a]
Call Waiting - Set/Cancel	#16
Do Not Disturb (DND) - Set/Cancel	#18
Hotline - Set/Cancel	#151
Hotline - Number	#152
Hotline - Timer	#153
Call Forward Unconditional - Set/Cancel	#131
Call Forward Unconditional - Number	#135
Call Forward Busy - Set/Cancel	#132
Call Forward Busy - Number	#136
Call Forward No-Reply - Set/Cancel	#133
Call Forward No-Reply - Number	#137
Call Forward No-Reply - Ring Timer	#139
Call Pick-up	#5
Call Hold/Retrieve	Flash[a]
Call Toggle (Call Split)	#2
Reject the Waiting Call and Speech with Current Call	#31
Ignore the Waiting Call and Speech with Current Call	#32
Accept the Waiting Call and Hold Current Call	#33
Accept the Waiting Call and Release Current Call	#34
Blind Transfer	#6
Attended Transfer	^
Conference	#8
Using Supplementary Services of Service Provider	#4
Using Voice Mail of the Service Provider	#7
Making a New Call	#91
Disconnect Call	#92

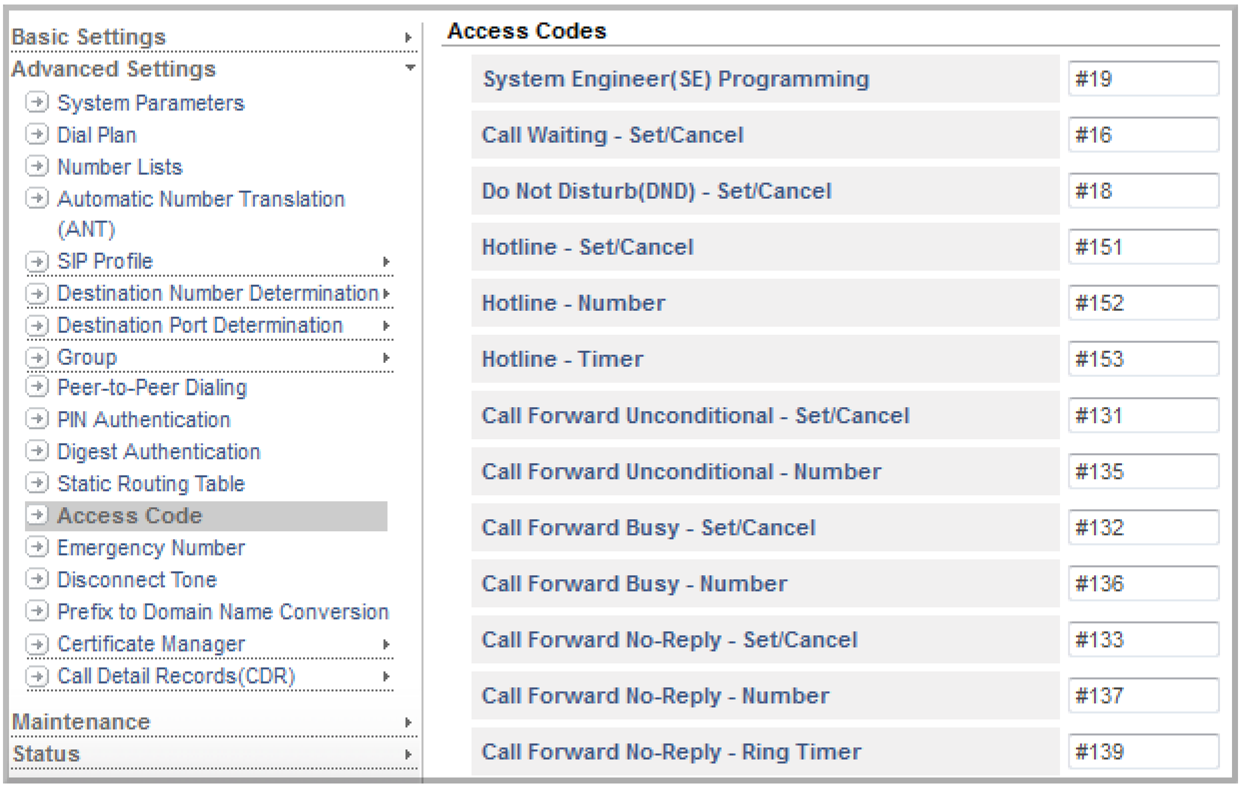
a. Non-programmable.

You can change the default access codes assigned to the respective features and facilities as per your requirement.

Configuring Access Codes

To change the default Access Codes assigned to the features and facilities,

- Click the **Advanced Settings** link to expand.
- Click the **Access Code** link.



- Change the default access code for the feature/ facility as per your requirement. Access Codes can be a maximum of 4 digits and digits 0-9, *, # and ^ are allowed.

Do not configure Access Codes that may conflict with the Emergency Numbers.

- Click **Submit** to save changes.

Virtual Access Code

Virtual Access Code is a string of digits dialed to use the Virtual User features. Virtual users can access the following features and facilities by dialing their Virtual Access Codes from a phone.

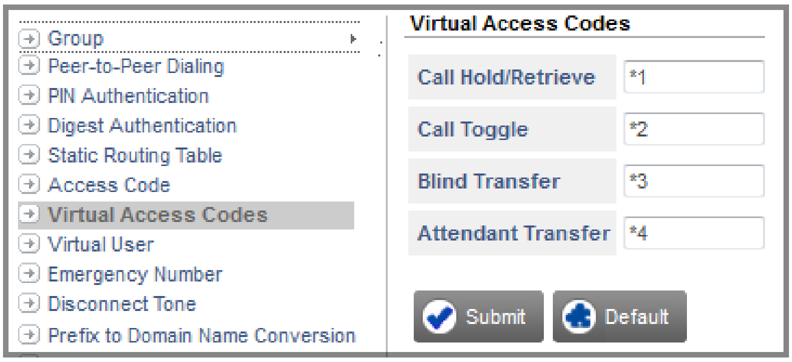
You can change the default access codes assigned to the respective features and facilities as per your requirement.

- *If you wish to use the Virtual features on Mobile Port then make sure that you have activated the “Virtual User” License in the system.*
- *Only one Virtual User can access the Virtual Feature on the Mobile Port at a time.*
- *Making a new Call and Disconnecting a call using Access Code will not be applicable to Virtual Users if **Allow Virtual Feature** option is configured as Allow All or As per Virtual User Table in General settings of the “Mobile Port”.*

Configuring Virtual Access Codes

To change the default Access Codes assigned to the features,

- Click the **Advanced Settings** link to expand.
- Click the **Virtual Access Codes** link.



- Change the default Virtual Access Code for the feature as per your requirement. Virtual Access Codes can be a maximum of 4 digits and digits 0-9, # and * are allowed.

  If Virtual Access Code is configured as Blank then that specific feature will be disabled for the all the Virtual Users.

Do not configure Virtual Access Codes that may conflict with the Emergency Numbers or Other Access Codes.

- Click **Submit** to save changes.

Virtual User

Virtual Users are the external users who use the Mobile phones. The details of the Virtual Users needs to be configured if *As per Virtual User Table* option is selected for **Allow Virtual feature** parameter in "General" settings of the Mobile Port.

The Virtual User table determines:

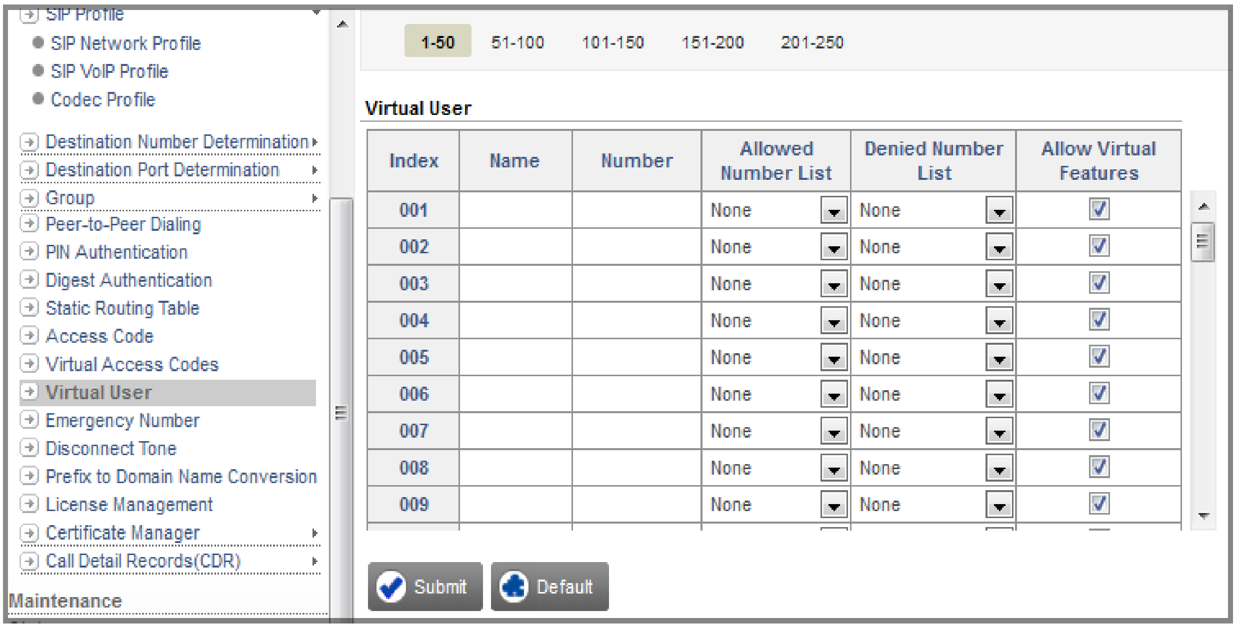
- the routing of incoming calls from the Mobile Port to the destination port.
- the routing of outgoing calls to the Mobile phone users.
- the access of Virtual Features to the Virtual Users.

To use the Virtual User features, make sure the "Pre-requisites" are fulfilled.

Configuring Virtual User Table

- Click the **Advanced Settings** link to expand.

- Click the **Virtual User** link. The Virtual User Table opens.



You can configure up to 250 entries in the table. This table is common for all the Mobile Ports.

- In **Name**, you may assign a Name to the Virtual User. When there is an incoming call on the Mobile port, the name you assign will appear on the phone of the called party, if the phone supports CLI display.

  During outgoing calls the name will not be sent.

  The name you assign may consist of a maximum of 24 characters. Default: Blank.

- In **Number**, you can assign a Number corresponding to the name of the Virtual User. When a call is placed or received through the Mobile port, the number you assign will appear on the phone of the called party, if the phone supports CLI display. Users configured here will be allowed to use the Virtual features only if the **Allow Virtual Feature** check box is enabled.

- This is the number provided by the Service Provider to the Mobile phone users or the numbers you wish to store for outgoing as well as incoming calls.

  The number you assign can have a maximum of 24 digits. Valid characters are 0 to 9, *, # and +. Default: Blank.

- In **Allowed Number List,** select the list number you have configured with numbers which you want Virtual User to dialed out from the Mobile Port. Default: None.

- In **Denied Number List**, select the list number you have configured with numbers you do not want Virtual User to dial out from the Mobile Port. Default: None.

- Select the **Allow Virtual Feature** check box if you want to allow the Virtual User to access the Virtual features during an ongoing call on the Mobile Port. Default: Enabled.

  This parameter is applicable only when *As per Virtual User Table* option is selected for **Allow Virtual feature** parameter in “General” settings of the Mobile Port.

  After verifying the Virtual User Table, the call is routed to the destination port according to the option selected in **Select Destination Port for Routing Calls**.

- Click **Submit** to save changes.

Emergency Number

SARVAM UMG supports the dialing of Emergency Numbers from all ports. Emergency numbers and their respective Routing Groups (through which they are to be routed) must be configured in the Emergency Number Table.

When you select “Region”, the system loads the Emergency Numbers used in the country you select. These numbers are included in the Emergency Number Table.

For each of these numbers, the system assigns a default Routing Group to route the number. You may reassign the Routing Group as per your requirement.

You may also add numbers of emergency services as per your requirement and assign the Routing Group for the same in the Emergency Number Table.

The Emergency Number Table stores upto 10 numbers, including those loaded by default.

- *For a few Regions, the system may not load default Emergency numbers in the Emergency Table. You may add the numbers as per your requirement.*
- *Emergency number Dialing will not work if Mains power to SARVAM UMG fails.*
- *Emergency Numbers have priority over Destination Number Table, PIN Number and Access Codes.*
- *The system does not apply End-of-Dialing when dialing Emergency Numbers.*
- *The system does not check Allowed-Denied Logic and Automatic Number Translation table when dialing an Emergency Number.*

SARVAM UMG will dial out an emergency number only if — FXO Port, SIP Trunk, Mobile Port, BRI Port or T1E1 Port — included as the Routing Group for the respective numbers are enabled.

Emergency Number can be dialed even when the option **Block all calls through this FXS Port** is enabled on the FXS Ports.

SARVAM UMG can dial out the numbers available in the Emergency Number Table even in the following situations:

- When SIM is absent
- When SIM is invalid
- When wrong SIM PIN is entered
- When SIM is blocked
- When GSM module is not registered

Some countries do not allow dialing of Emergency Number without SIM. As per TEC standard, India allows dialing of Emergency Number without SIM.

When 2G GSM engine Sierra Wireless SL6087 is installed, SARVAM UMG will support the following Emergency Numbers with and without SIM in the system.

*Emergency Numbers supported **with SIM** present in the system:*
- 112
- 911
- 000
- 08
- 110
- 999
- 118
- 119

*Emergency Numbers supported **without SIM** in the system:*
- 112
- 911

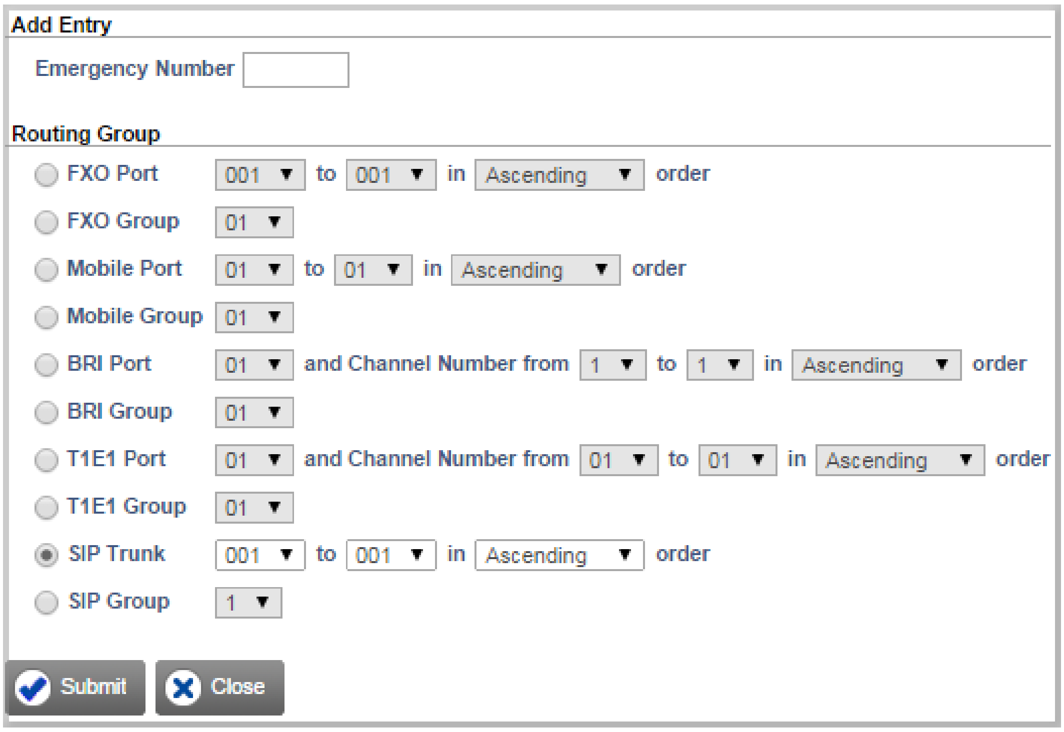
Configuring Emergency Numbers

To configure the Emergency Number Table,

- Click the **Advanced Settings** link to expand.
- Click the **Emergency Number** link.

- To **Add** an Emergency Number in the table, click the **Add** button.
- To **Edit** an Emergency Number and or assign a Routing Group, click **Settings** of that number.

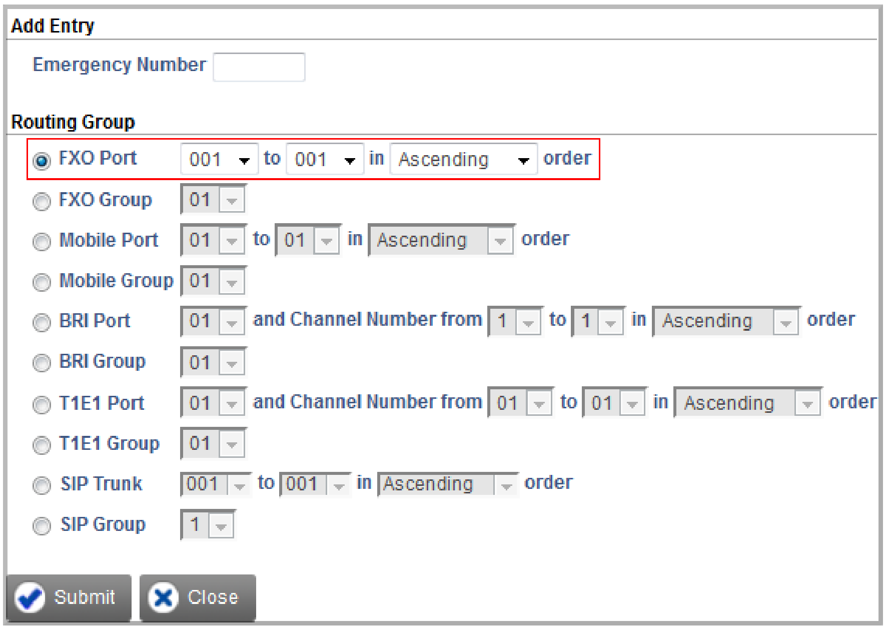
A new window opens, to allow you to add/ edit the entry.



- In **Emergency Number**, enter the emergency number used in your country/region.

Make sure that Access Codes you have configured do not conflict with the Emergency Numbers.

- Create the **Routing Group**.
  - To create a group of *sequential* **FXO Ports** as members,



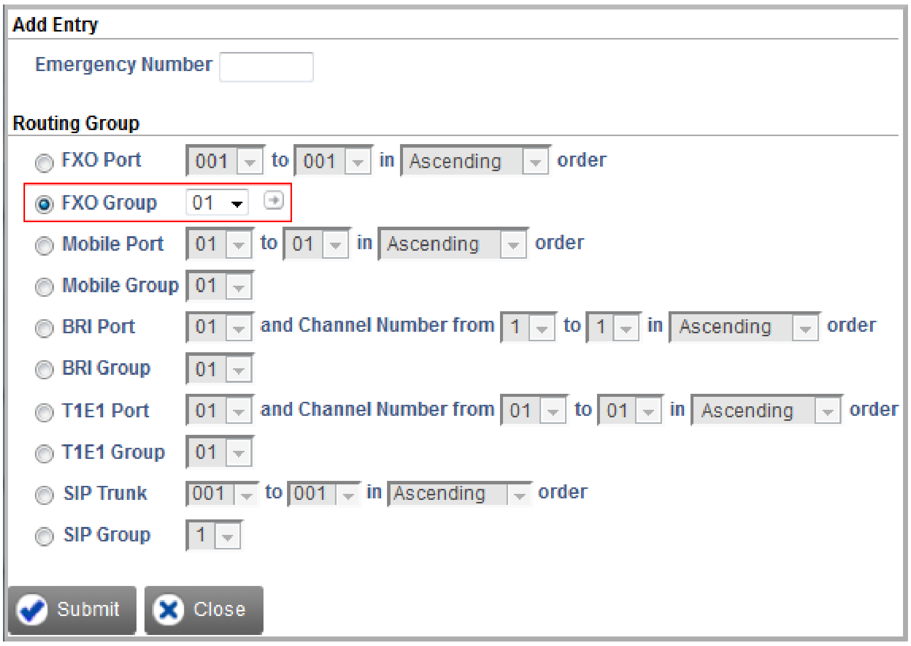
- Select the desired **FXO Port** numbers as members. Default:1.

- In **in - order**, select the order in which the system should hunt for a free member FXO Port to route the call.

  Select **Ascending** to start hunting from the first to the last member FXO Port. Select **Descending** to start hunting from the last to the first member FXO Port. Default: Ascending.

- To create a group of *not-sequential* **FXO Ports** as members,

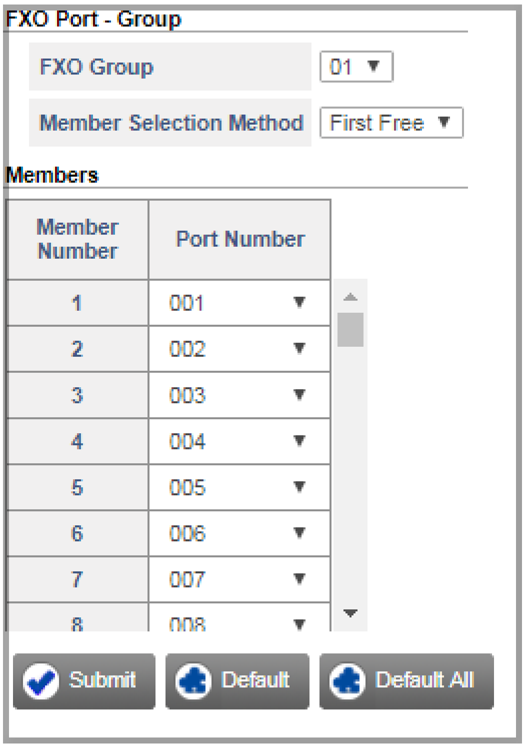
  - Select a **FXO Group**.



  - Select **FXO Group** number. Default:1.

  - Click **Settings** .

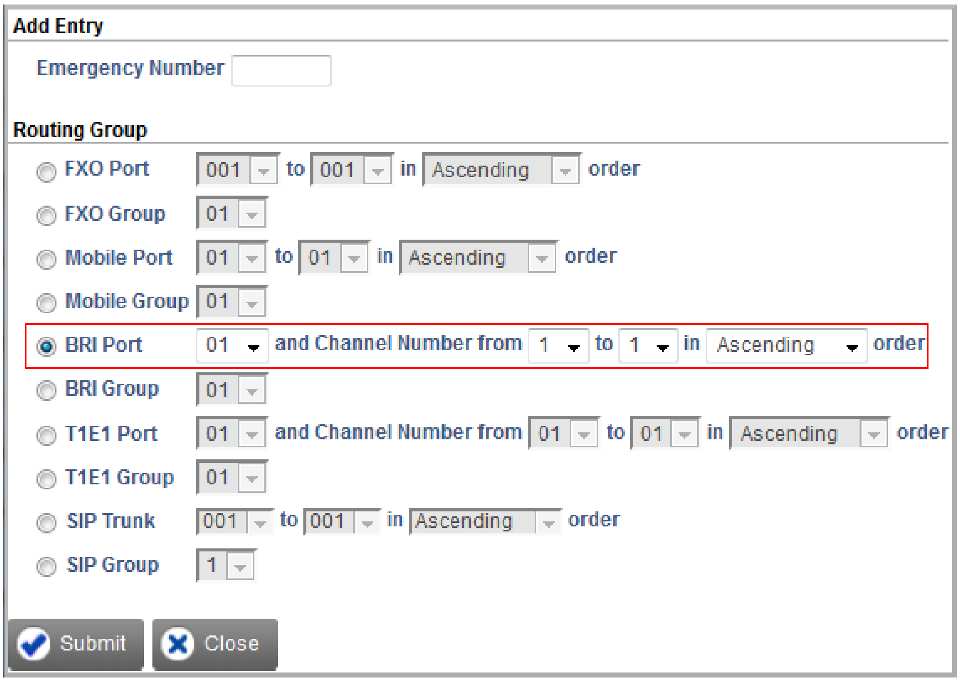
- The **FXO Port - Groups** window opens.



  - Create the FXO Group. For detailed instructions on creating groups, see the topic "Group" under *Advanced Settings*.

- Similarly, you can create a group of *sequential* and *not-sequential* Mobile Ports and SIP Trunks.

- To create a routing group of *sequential* **BRI Channels** as members,



  - Select the **BRI Port** Number. Default: 1.

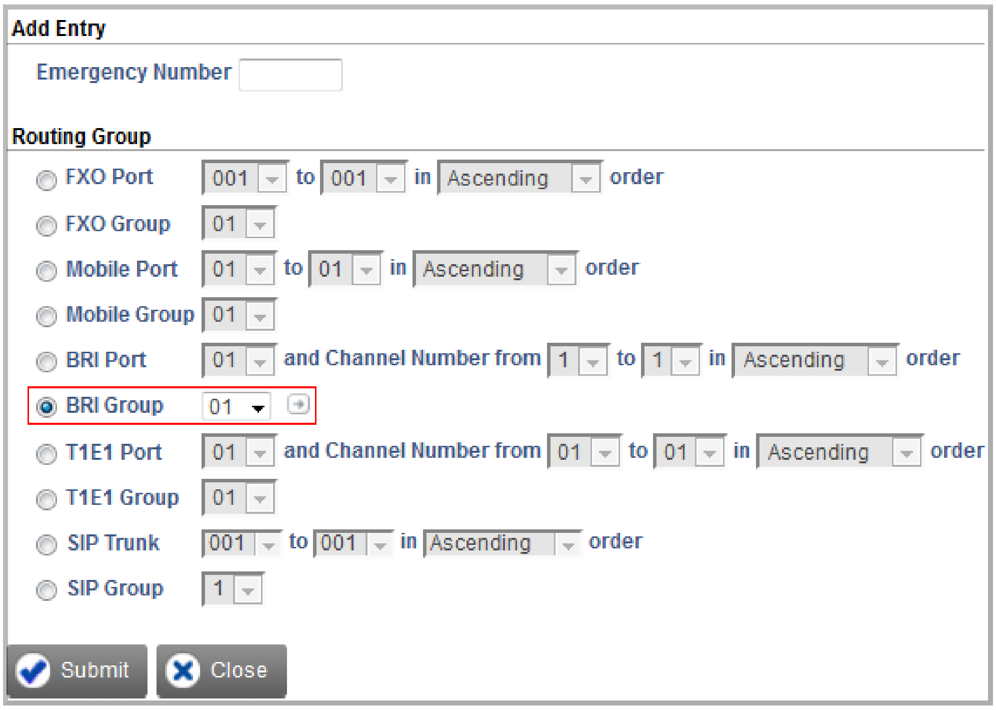
- In Channel Number **From - to**, select the **Start Channel Number** and the **End Channel Number**, respectively.

- In **in - order**, select the order in which the system should hunt for a free member Channel to route the call.

  Select **Ascending** to start hunting from the first to the last member channel. Select **Descending** to start hunting from the last to the first member channel. Default: Ascending.

- To create a group of *not-sequential* **BRI Channels** as members,

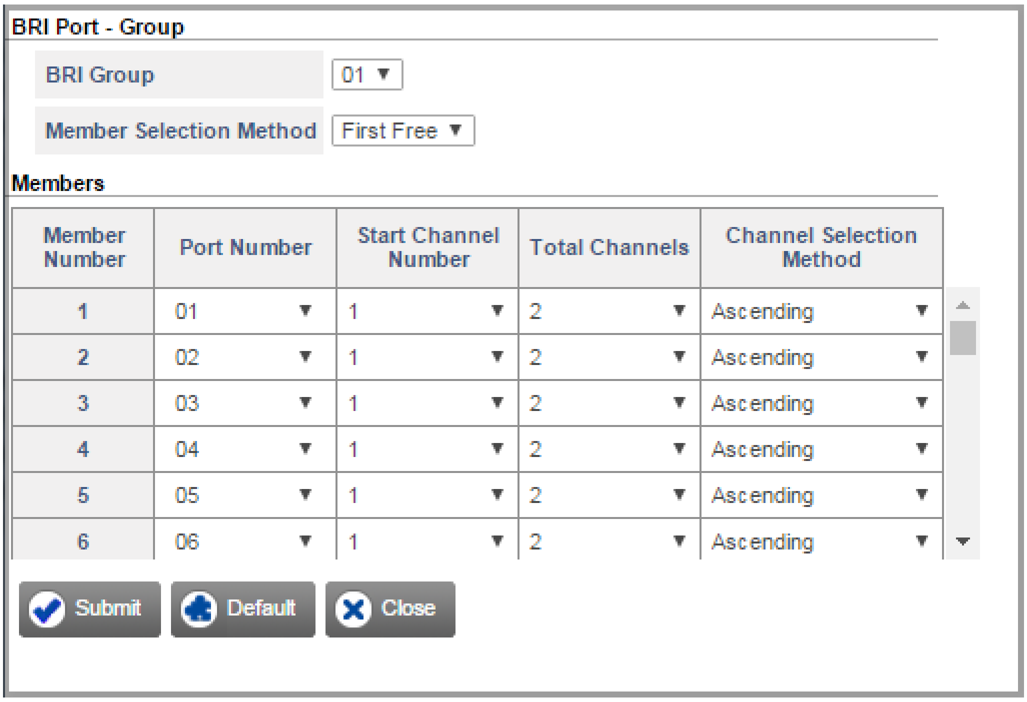
  - Select **BRI Group**.



  - Select a **BRI Group** number. Default:1.

  - Click **Settings**.

- The **BRI Port - Groups** window opens.



- Create the BRI Group. For detailed instructions on creating groups, see the topic “Group” under *Advanced Settings.*

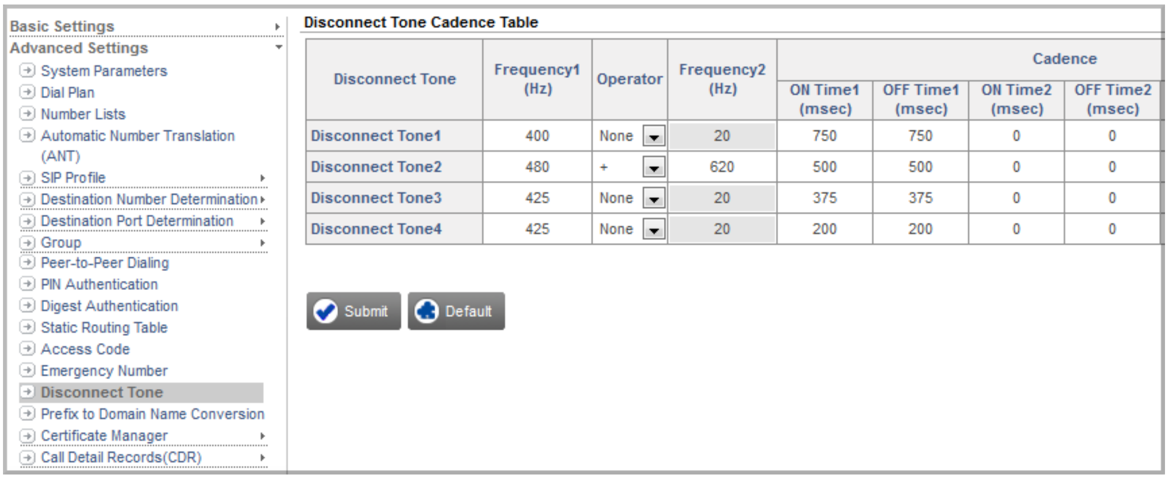
- Similarly, you can create a group of *sequential* and *not-sequential* T1E1 Ports.
- Click **Submit** to save. Close the **Add Entry/Edit Entry** window. The entries you added appear on the Emergency Numbers page.

Disconnect Tone

If Call Disconnection is signalled by your CO Network in the form of Disconnect Tone, configure **Disconnect Tone** parameters on the FXO Ports. You must enable *Disconnect Tone Detection* on the *FXO Port* and select the *Disconnect Tone Type*.

To enable the system to detect the Disconnect Tone accurately, you must configure the Cadence (ON-OFF time) and Frequency of the Disconnect Tone Type you selected, as supported by the CO Network. To do so,

- Click the **Advanced Settings** link to expand.
- Click the **Disconnect Tone** link.
- The **Disconnect Tone Cadence Table** opens.



- For each Disconnect Tone, set the following parameters:
  - Set the **Frequency 1 (Hz)**. Valid range is 300 Hz to 1400 Hz.
  - Select the **Operator**. You may select — Modulation (*), Addition (+) or None.

    If Modulation is selected, frequency 1 and frequency 2 will be used as modulation (F1* F2).

    If Addition is selected, frequency 1 and frequency 2 will be used as addition (F1 + F2).

    If None is selected, frequency 2 will not be applicable.
  - Set the **Frequency 2 (Hz)** if you selected Modulation or Addition as the Operator. Frequency 2 is applicable only when the Disconnect Tone supported by the CO network consists of Dual Frequency. Valid range is 20 to 1400 Hz.
  - In **Cadence**, configure the ON Time1-OFF Time1, ON Time2-OFF Time2 and ON Time3-OFF Time-3 for each Disconnect Tone. Valid range is 0000 to 9999 msec.

    When the system detects disconnect tone on the FXO Port and if it matches with the Frequency and Cadences you have set, the call will be disconnected and the FXO Port will be released.
- Click **Submit** to save.

Prefix to Domain Name Conversion

Prefix to Domain Name Conversion is used when a user sets Call Forward or makes a Blind Transfer on SIP. This feature is applicable only when the destination port is SIP.

SARVAM UMG supports multiple SIP Trunks and FXS Ports. When the FXS Port user dials a SIP number, SARVAM UMG routes the call to the IP network using the SIP Trunk determined by the routing mechanism. The FXS Port user can dial only numbers, not domain names. Therefore, it becomes necessary to assign Prefix codes to domain names, which the FXS user can dial.

Now, it is necessary that the number string dialed by SARVAM UMG is understood by the ITSP through which the call is routed. So, an appropriate Prefix Code is assigned to the Domain of the ITSP through which the calls are to be routed.

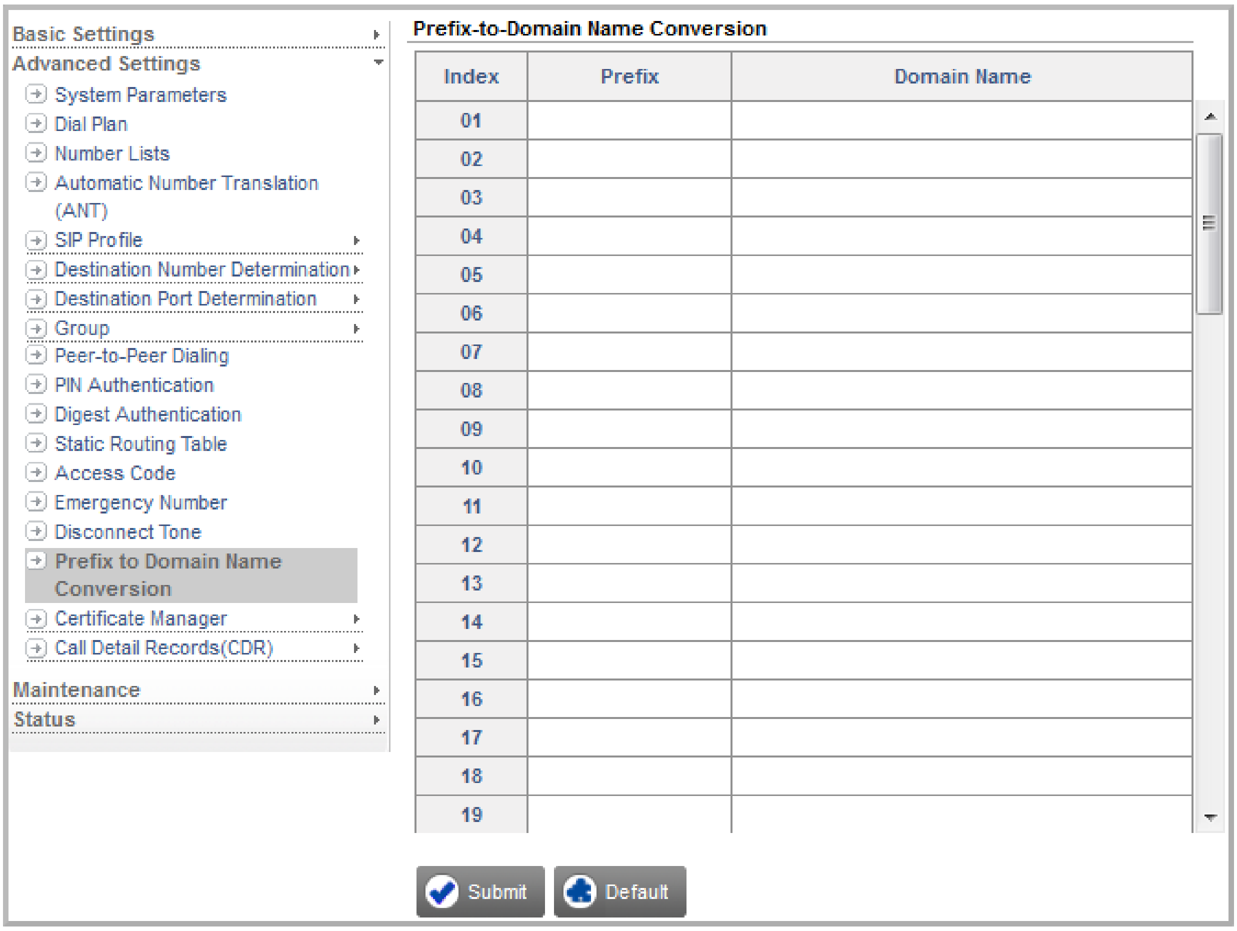
However, when the FXS Port user sets Call Forward or Blind Transfer, the Prefix Code and the number are sent to the calling party in the redirect message, without the domain name. So the calling party will not be able to reach the FXS user at the forwarded/transfer destination. The 'Prefix to Domain Name Conversion' feature resolves this.

Let us understand this feature with the help of an example:

- Assume that SARVAM UMG is configured to route calls made to the domain 'abc.com' from the FXS Port, through the SIP Trunk subscribed with the ITSP 'Pulver.com'.
- Since the FXS Port user cannot dial the domain name, a prefix code must be assigned to the domain name.
- The Prefix code, *234 is assigned to the domain 'abc.com'.
- When the FXS Port user wants to dial the SIP ID 9874@abc.com, the user must dial *234 followed by 9874.
- SARVAM UMG determines that the called party is the subscriber of abc.com and converts *2349874 to 9874@abc.com and routes the call to the desired destination through 'Pulver.com'.
- Now, given the above scenario, assume that the FXS Port user sets Call Forward to *2349874 (that is, 9874@abc.com).
- When an external caller calls the FXS Port user, the caller will receive *2349874 in the Redirect message.
- However, to reach the FXS user at the forwarded destination, the caller must have a domain name in the contact address. Since the caller will not recognize *234 (the prefix code assigned to abc.com), the caller will not be able to reach the FXS user at the forwarded destination address.
- With Prefix to Domain Name Conversion, SARVAM UMG converts this prefix code to the domain name abc.com (9874@abc.com), and sends it in the Redirect message to the external caller informing the caller of the new contact address.
- On receiving this information in the Redirect message, the external caller can call the new contact number, 9874@abc.com, and talk to the FXS user.

To configure Prefix to Domain Name Conversion,

- Click the **Advanced Settings** link to expand.
- Click the **Prefix to Domain Name Conversion** link.
- The Prefix to Domain Name Conversion table opens. You can store upto 64 entries in this table.



- In **Prefix**, enter the Prefix Code you want to assign to the Domain Server Names. The Prefix code must not exceed four digits. Valid digits are 0 to 9, * and #
- For each Prefix Code you assigned, enter the corresponding **Domain Name**. The Domain Name may consist of a maximum of 40 characters.
- Click **Submit** to save the entries.

License Management

What's this?

The application SARVAM UMG and Vocoder Channels that it supports, require the purchase of licenses. When you buy the ETERNITY GENX Platform, you get a default license key for the platform.

SARVAM UMG Application License

To use ETERNITY GENX as the Universal Media Gateway, you need to purchase the **SARVAM UMG SME** Application software license.

If you do not have the license for the SARVAM UMG SME Application and you do not start the Demo Period, the system will allow configuring and making of calls, but the connected calls[22] from any port will be disconnected after 60 seconds. For details, refer *"Demo Provision".*

Described below are the features of SARVAM UMG for which you require licenses. To know the name of the license you need to purchase, see "Supported Licenses".

Expansion Slots

Expansion Slots License is required to expand the number of functional universal slots in the system. This license is applicable only for Firmware version V1R4 and later.

If you have upgraded the system firmware to V1R4 and later in the old ETERNITYGENX system, the Expansion Slots license will be applicable for the universal slots. No universal slots will be functional by default. You must purchase the license to activate the universal slots as required.

After upgrading the system with V1R4, if required you can again downgrade the system to V1R3.In this, case all the universal slots will be functional.

If you have purchased the new ETERNITY GENX system with the firmware V1R4 and later, the Expansion Slots license will be applicable for the universal slots. The first four universal slots after the power supply card will be functional by default.

System purchased with V1R4 firmware must not be downgraded to earlier versions.

If you require more functional universal slots, you must purchase the Expansion Slots License to activate the same.

Matrix provides the expansion slot license — SARVAM EXP4 SME for ETERNITY GENX system.

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

VOCODER Channels

Vocoder Channels license is required for SIP calls. The number of Vocoder channels that will be supported in demo period will be equal to the total number of channels available in the Vocoder module/s installed in the System.

22. *Connected calls means where speech is connected between the calling party port and the called party port even if the called party port is not matured.*

If the Demo period is paused or gets expired, then the maximum number of SIP calls that will be supported would be as per the license you purchase.

The system supports two NX DBM VOCODER64 Modules. You must purchase the module separately. Each NX DBM VOCODER64 module supports a maximum of 64 Vocoder channels.

The system provides 4 pre-activated Vocoder channels by default which can be used after installing NX DBM VOCODER64 module. If you require more channels, you can purchase the licenses accordingly.

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

During a SIP call, VOCODER channel is required and if no free channel is available, the system will reject the call.

Virtual User

This license is required for Virtual Users to access the Virtual Features on the Mobile Port remotely.

It is also required in the countries where the GSM Service Provider supports Called Number with Calling Number using Separator.

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 UMG SME	Universal Media Gateway License for Small-Medium Enterprise.
SARVAM EXP4 SME	License for SARVAM UMG SME to activate four Expansion Slots.
SARVAM VOCODER CHNL4	License for Vocoder Channels to support 4 simultaneous calls with transcoding.
SARVAM VOCODER CHNL16	License for Vocoder Channels to support 16 simultaneous calls with transcoding.
SARVAM Virtual User	License to support 250 Virtual Users, Virtual features and Separator feature.

Demo Provision

Demo provision is useful, when the customer’s system cannot be repaired on-site and a standby system needs to be installed Or when end users demand to use the licensed feature on trial basis before actually purchasing the license.

Demo Provision enables you to use the SARVAM UMG application, free of cost for a period of 60 days. In this case, the application starts even if you have not purchased the SARVAM UMG SME license.

All the Universal Slots will be functional during the Demo period irrespective of the number of activated SARVAM EXP4 SME licenses.

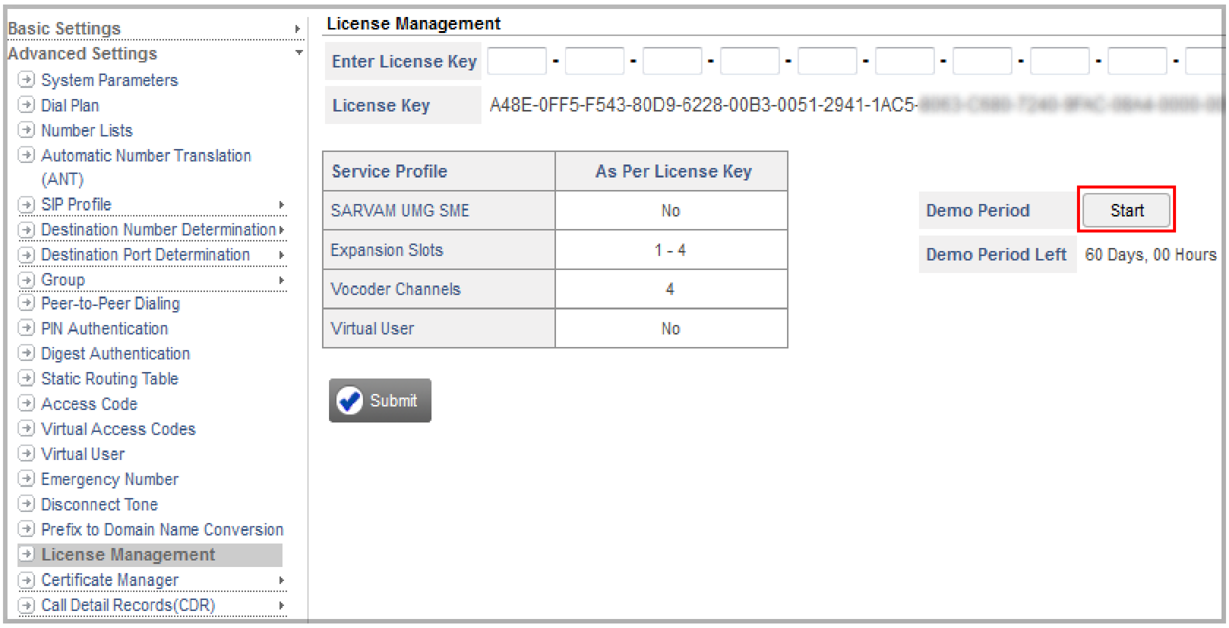
During the Demo Provision you can access and use all the features and functionalities[23] supported by the application.

23. *The number of Vocoder channels that will be supported in demo period will be equal to the total number of channels available in the Vocoder module/s installed in the System.*

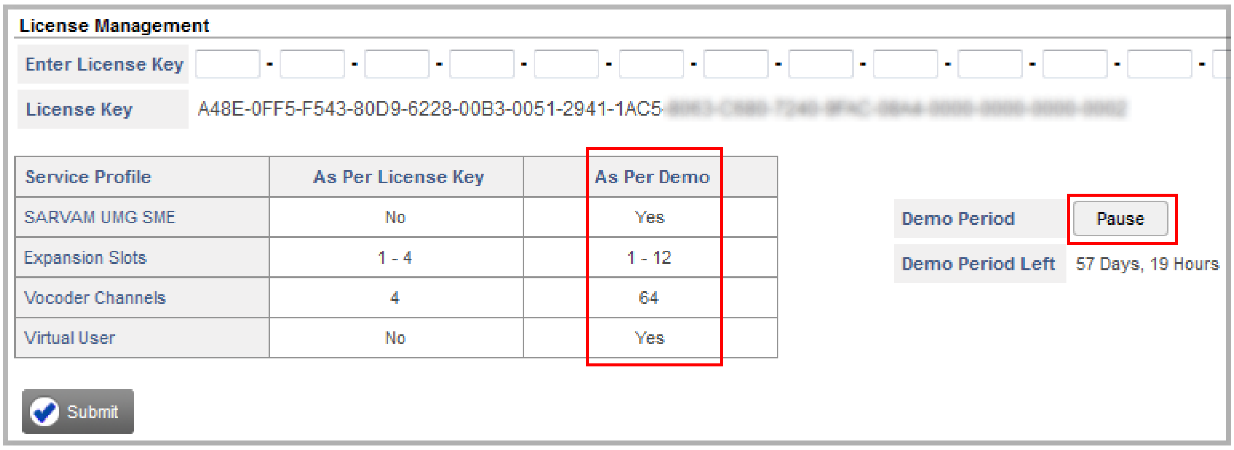
To activate the Demo,

- Open Jeeves.
- Log in as System Engineer.

Under **Advanced Settings**, click **License Management** link. The License Management page opens.



- Click the **Demo Period** Start button.



The demo period for using licensed feature in your system starts and the Start button will change to Pause. The demo period is of 60 days.

You may Pause the Demo Period, if required. When you pause the demo period, all licensed features will work as per the license key installed in the system.

If you want to use licensed feature after the expiry of the demo period, you must purchase the license key and activate it in your system.

When you default the SARVAM UMG, the demo period will not be reset.

How to activate your License

For the functional modules and features described above, you would need to activate a valid License Key.

Instructions for Matrix Channel Partners

Your license voucher may be a paper or a PDF (protected) file.

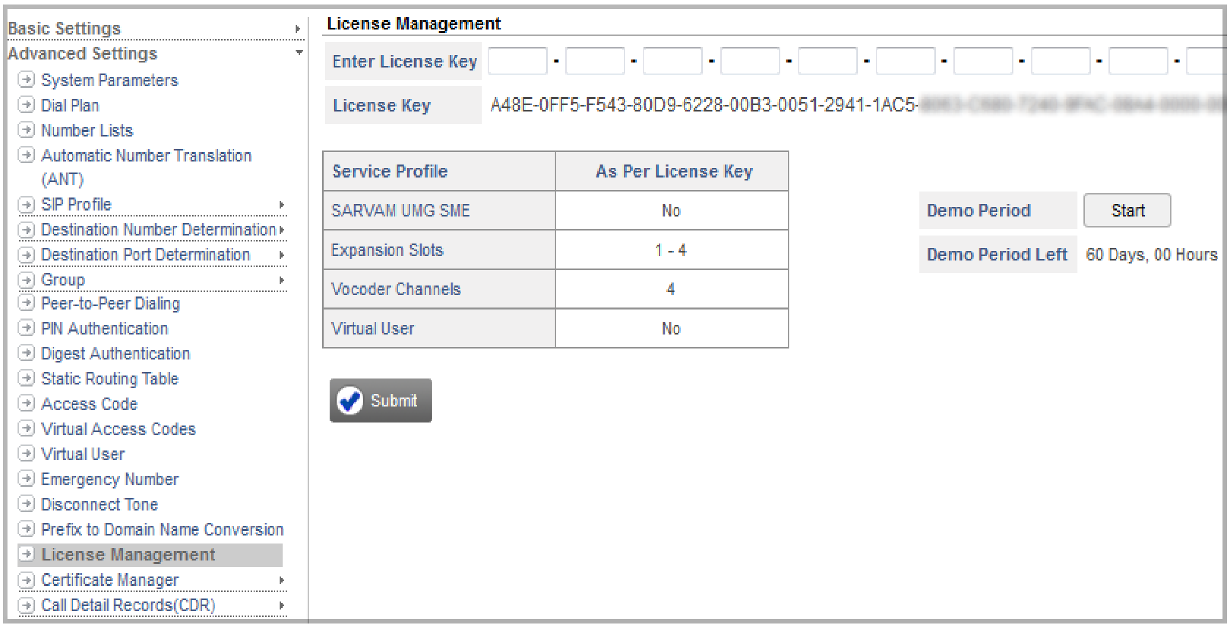
You must activate your License Online. For this, keep the following items ready:

- The License Voucher containing the 16-digit PIN.
- A valid, unique User ID and Password from the Matrix License Support Centre.
- Access to Internet.
- Current License Key of the system.

To activate the License Key online,

- Open Jeeves.
- Log in as System Engineer.
- Under **Advanced Settings**, click **License Management** link.

  The License Management page opens.



- Note down or copy the current **License Key** on this page.

  You can view the features and functions that are currently available to you under **Service Profile**.

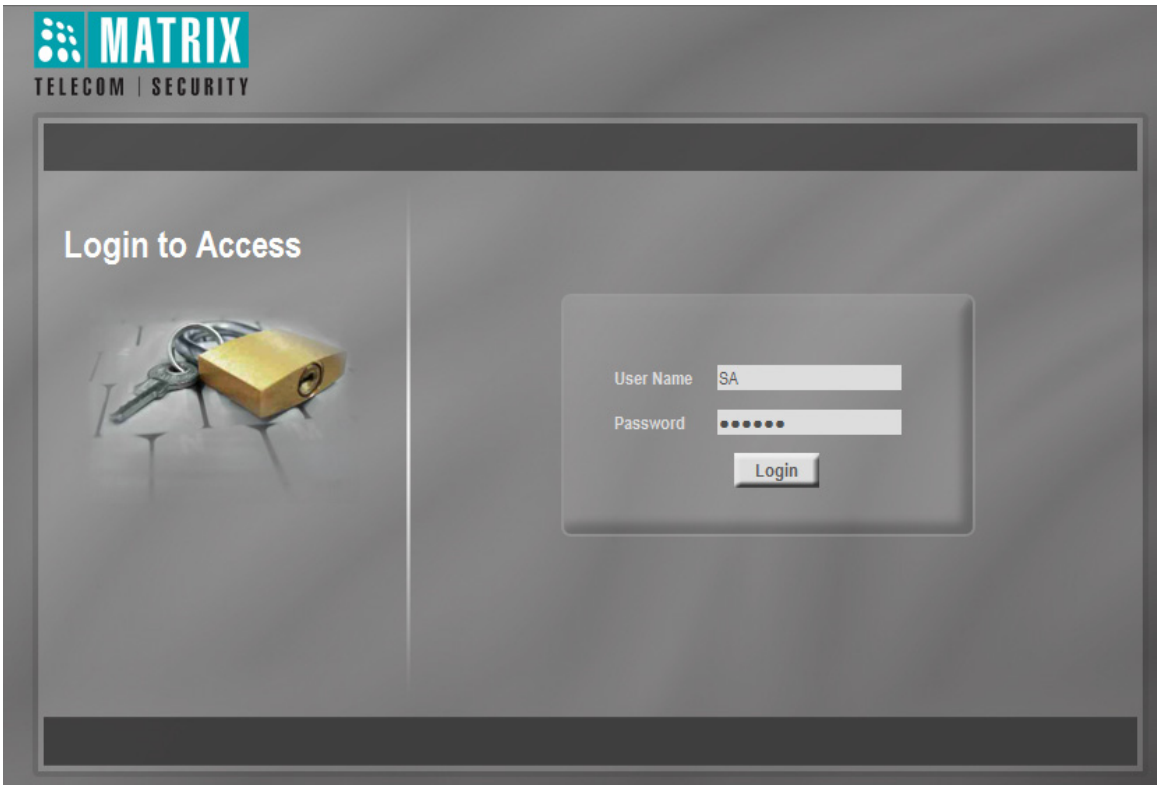
- Keep your Current License Key and the License Voucher (paper or PDF) ready.
- Open a new window on your browser.

- Enter **http://www.matrixcomsec.com/MatrixLicense/** in the address bar.

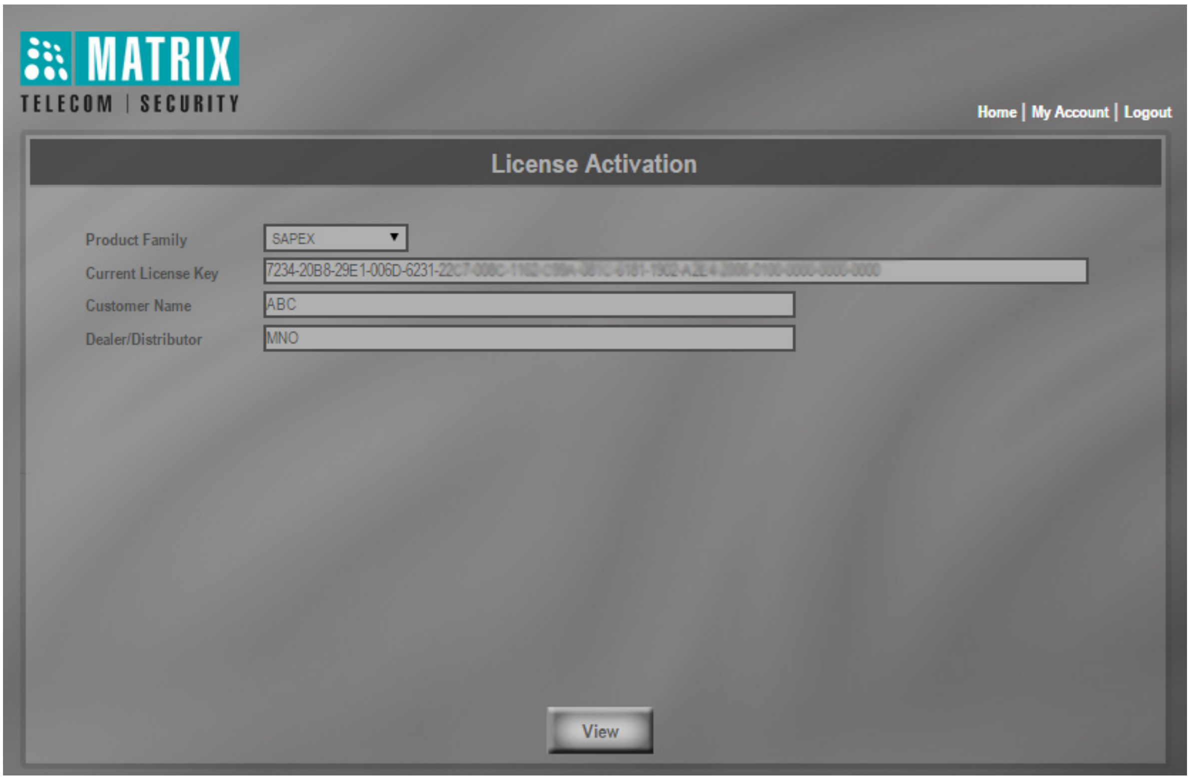
- Click **Login to Access**.

  The Login to Access page will open.

- Enter your **User Name** and **Password** provided by Matrix and click the **Login** button.



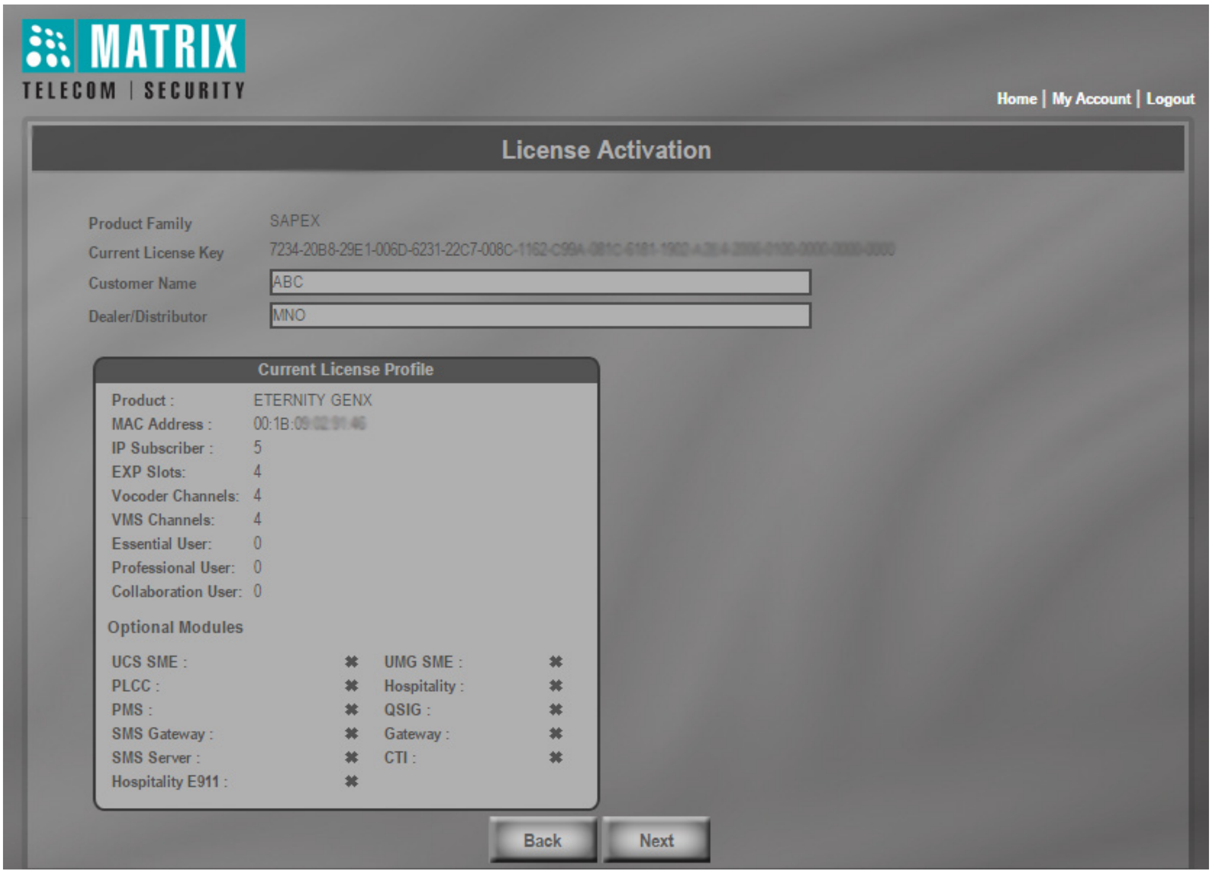
On successful login, the **License Activation** page will open.



- As **Product Family**, select the option **SAPEX**.

- In the **Current License Key** field, type the current product license key you noted or paste the key you copied from the *License Management* page of Jeeves.

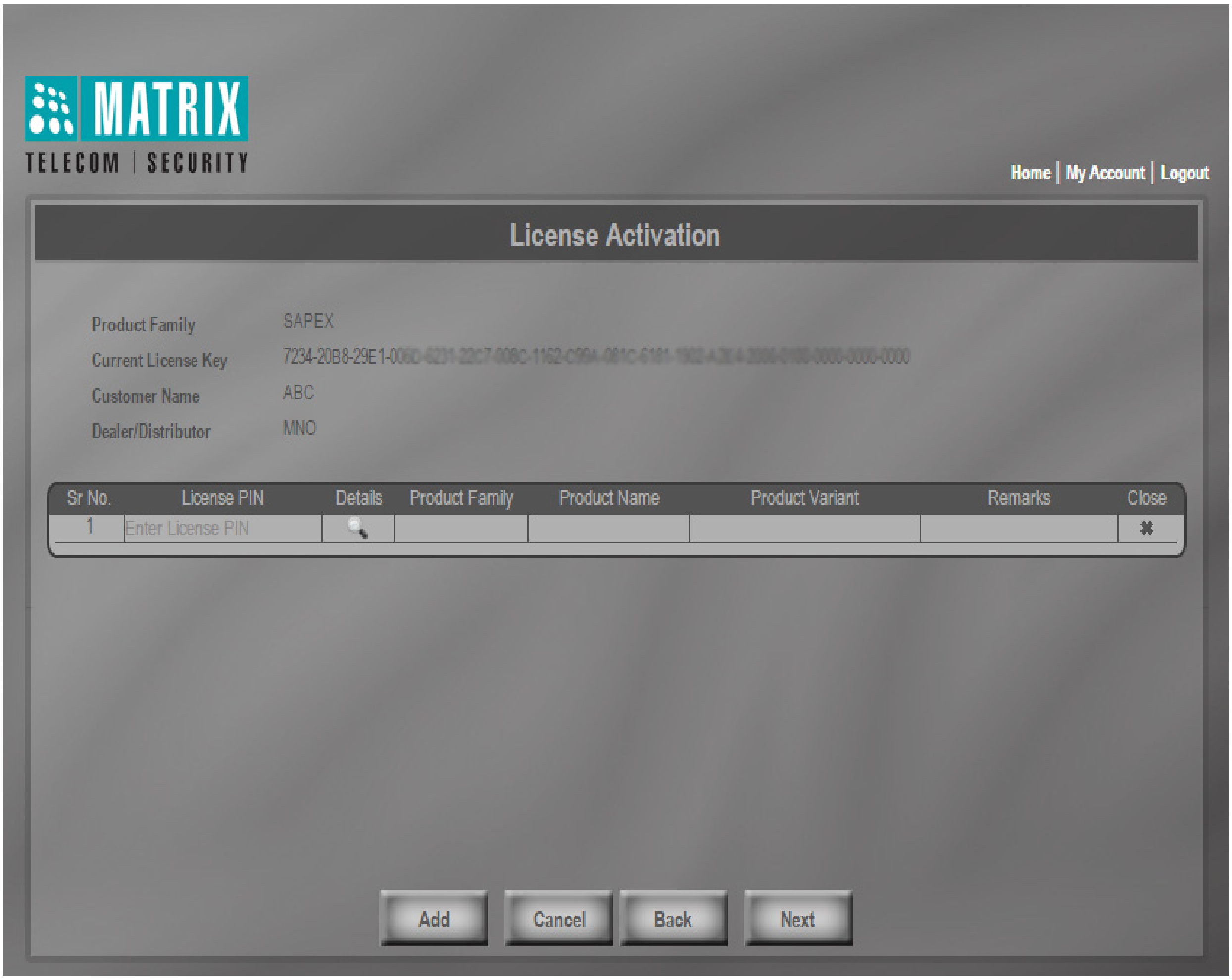
- Click the **View** button.



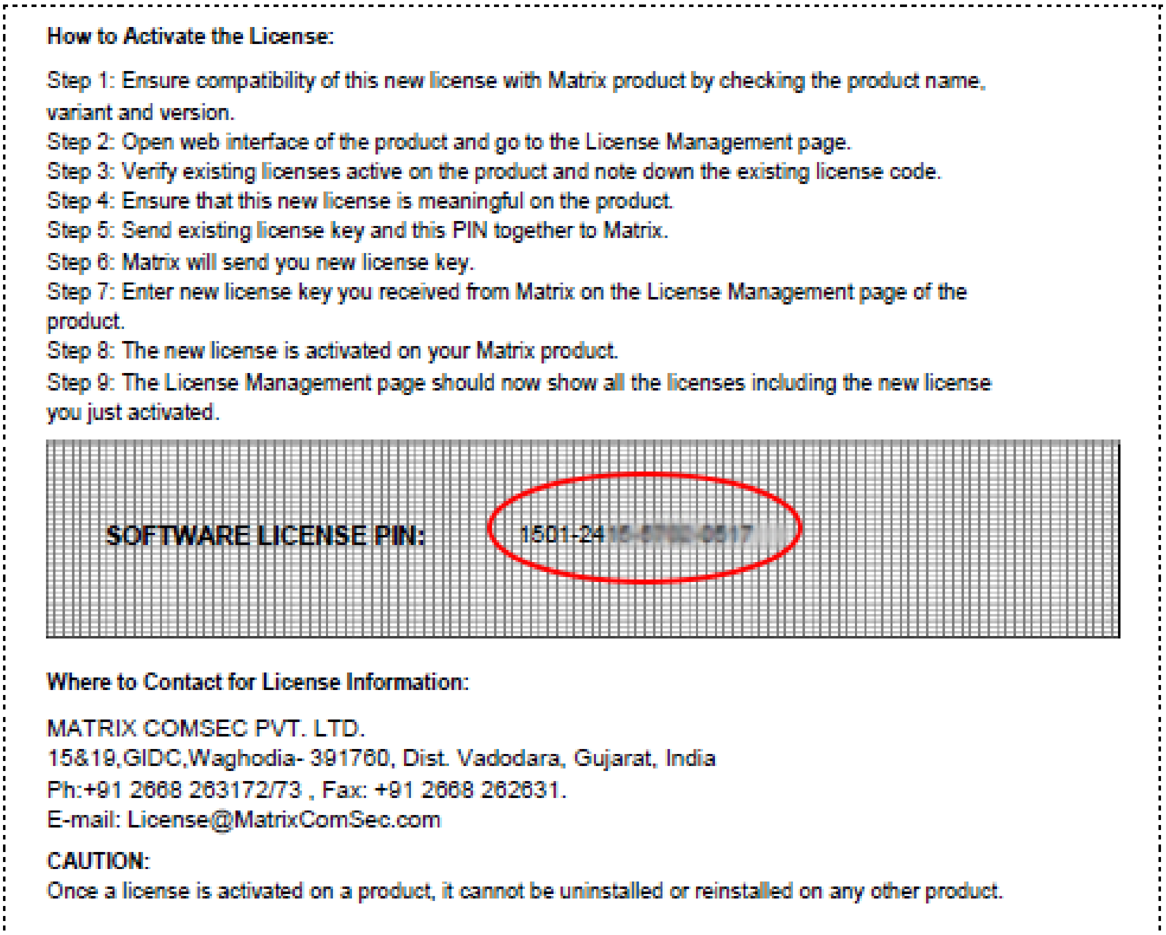
- The page displays the current License Profile[24] on ETERNITY GENX. Click the **Next** button to continue.

  The **License Activation** page opens.

24. *When ETERNITY GENX is used as the Universal Media Gateway, only SARVAM UMG SME and Vocoder Channels Licenses are applicable. Other licenses are applicable when you run the ETERNITY GENX as the Unified Communication Server.*


MATRIX
TELECOM | SECURITY
Home | My Account | Logout
License Activation
Product Family
SAPEX
Current License Key
Customer Name
ABC
Dealer/Distributor
MNO
Sr No.
License PIN
Details
Product Family
Product Name
Product Variant
Remarks
Close
1
Enter License PIN
Add
Cancel
Back
Next

In the **License PIN** field on this page, enter the 16-digit License PIN from the Voucher.



How to Activate the License:

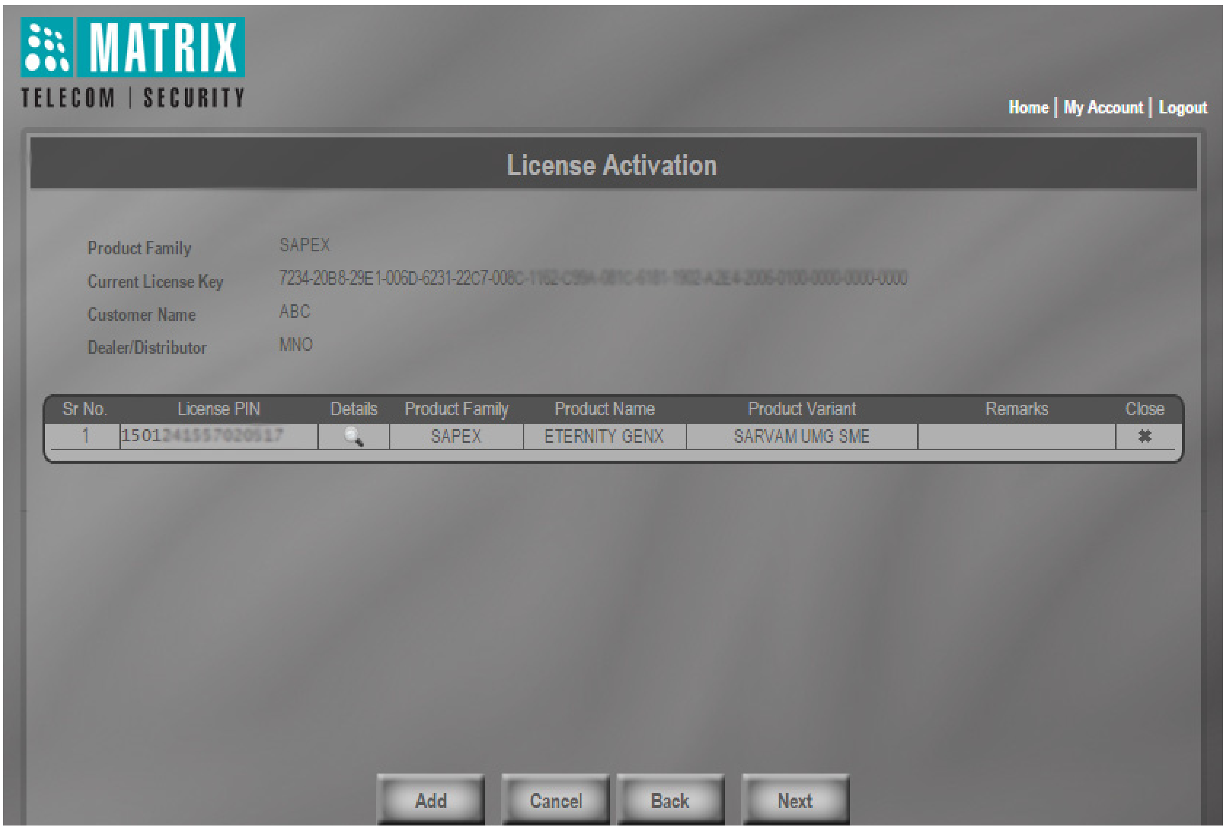
Step 1: Ensure compatibility of this new license with Matrix product by checking the product name, variant and version.
Step 2: Open web interface of the product and go to the License Management page.
Step 3: Verify existing licenses active on the product and note down the existing license code.
Step 4: Ensure that this new license is meaningful on the product.
Step 5: Send existing license key and this PIN together to Matrix.
Step 6: Matrix will send you new license key.
Step 7: Enter new license key you received from Matrix on the License Management page of the product.
Step 8: The new license is activated on your Matrix product.
Step 9: The License Management page should now show all the licenses including the new license you just activated.

SOFTWARE LICENSE PIN: 1501-24

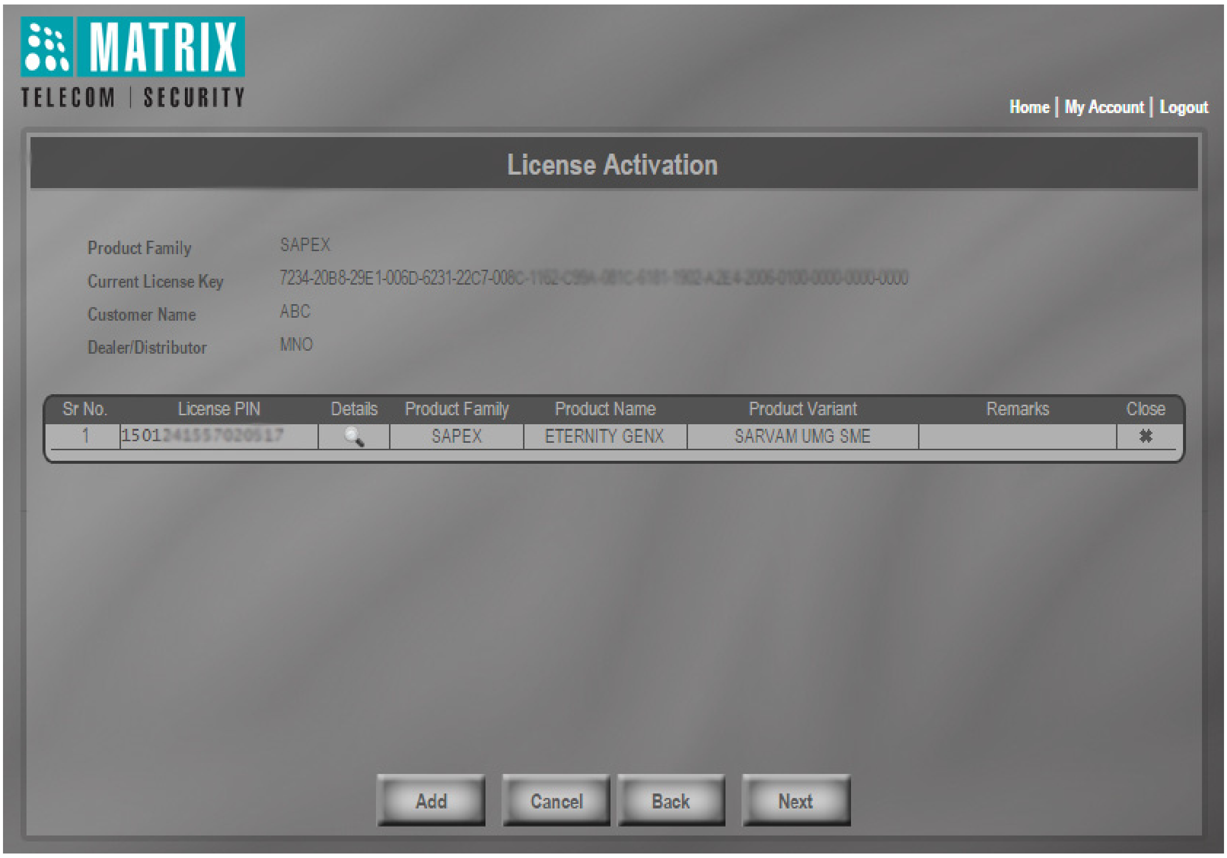
Where to Contact for License Information:

MATRIX COMSEC PVT. LTD.
15&19,GIDC,Waghodia- 391760, Dist. Vadodara, Gujarat, India
Ph:+91 2668 263172/73 , Fax: +91 2668 262631.
E-mail: License@MatrixComSec.com

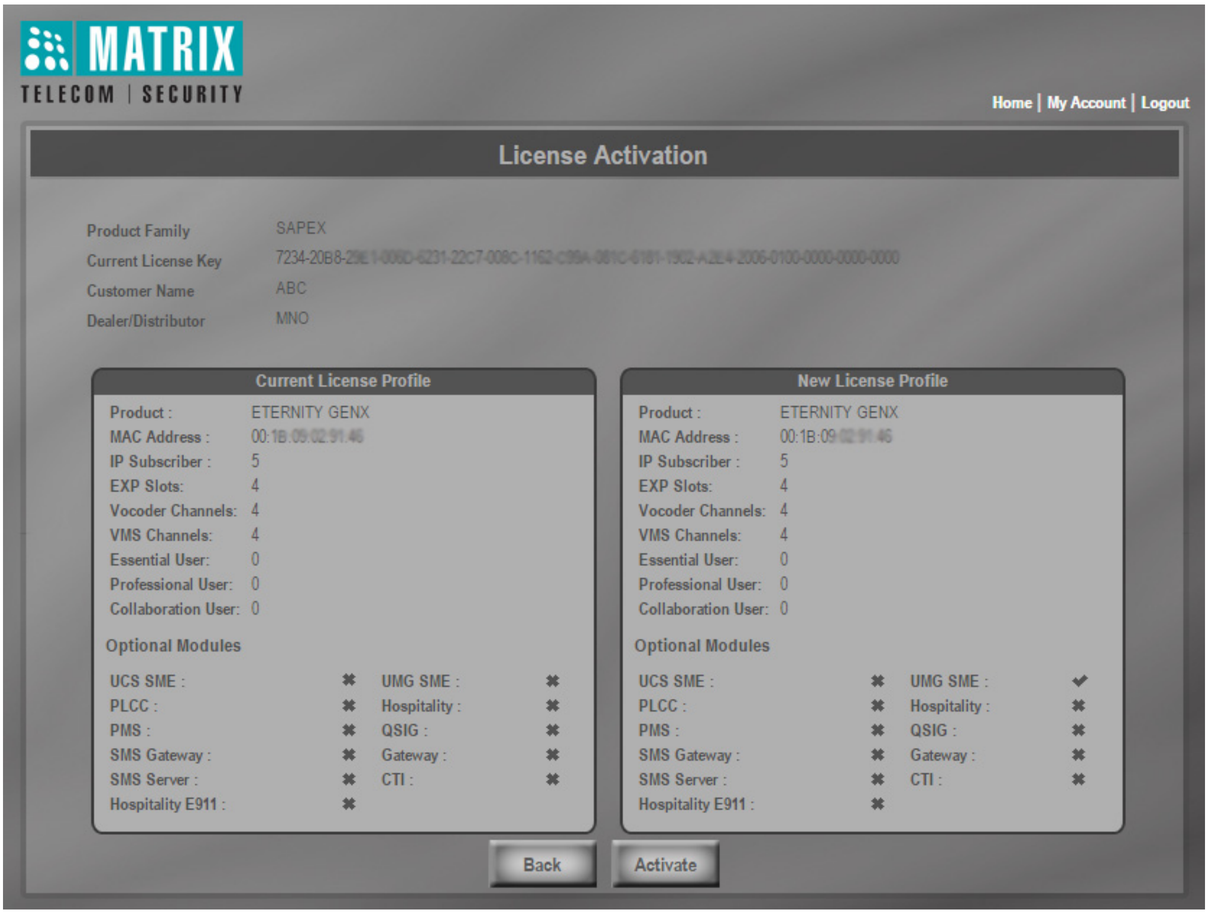
CAUTION:
Once a license is activated on a product, it cannot be uninstalled or reinstalled on any other product.



- Click **Details.** The details appear in the fields **Product Family, Product Name**, **Product Variant.**



- Click the **Next** button. Your **Current License Profile** and your **New License Profile** will appear on this page.



- Click the **Activate** button and wait for a few seconds, as the activation is initiated.

  On successful activation, the confirmation message will appear on your screen along with the activation date and time.

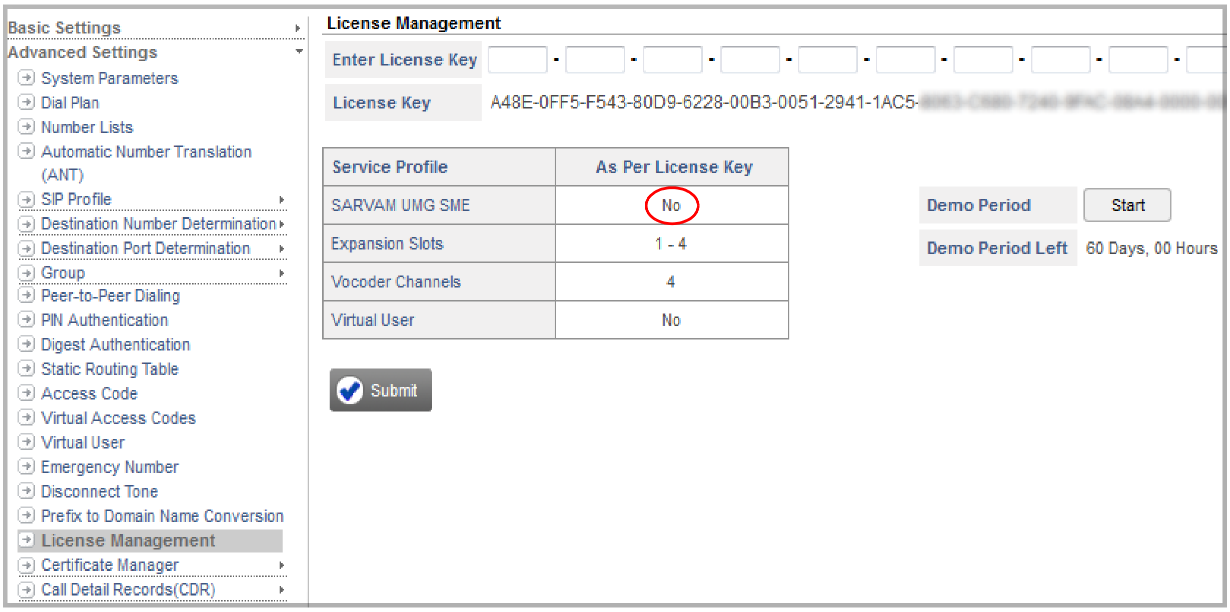
*The screens for **Current License Profile** and the **New License Profile** may differ according to the licenses purchased by you.*

A confirmation mail will also be sent to your e-mail ID (registered with Matrix).

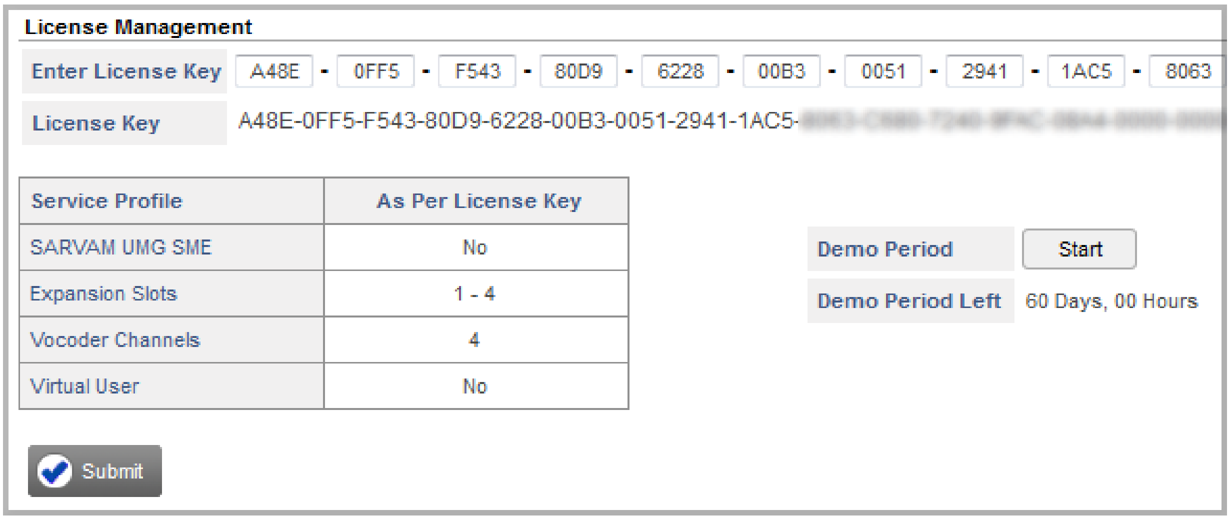
You may **Save**, **Print**, or **Email** this information for your records, by clicking the relevant button.

- Note down or copy the New License Key generated on this page.
- Go back to the Jeeves window (or log in as System Engineer again, if your session has ended).

- Click the **License Management** page again.

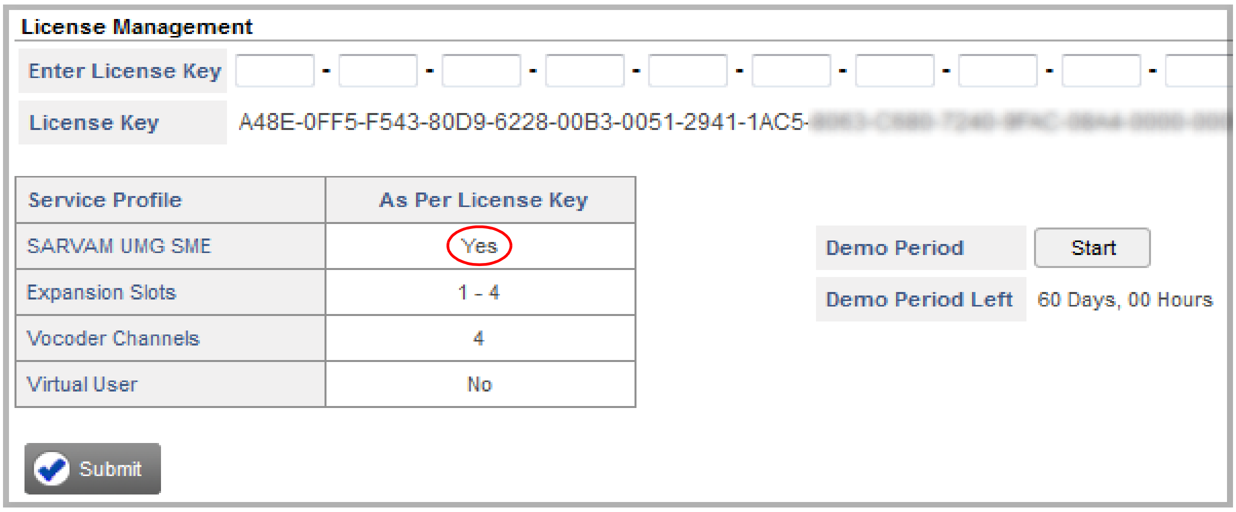


- In **Enter License Key**, paste or enter the new License Key generated.



- Click the **Submit** button.

The **Service Profile** on this page will be updated according to the license.



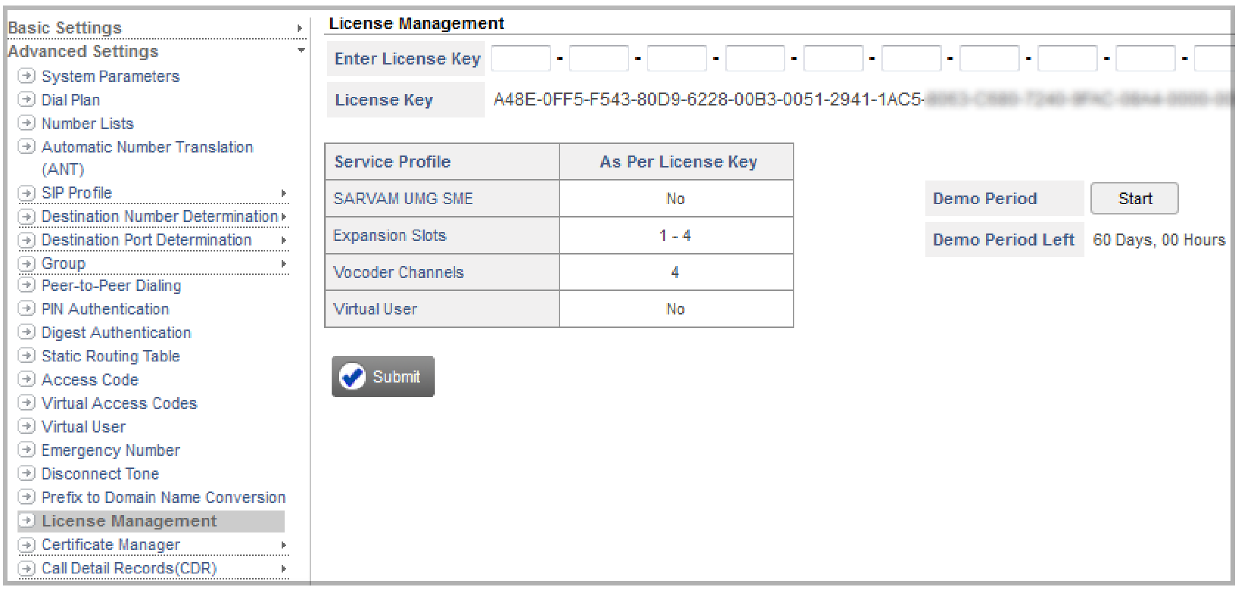
- To log off, click the **Logout** button.

If you are unable to use Online Activation of the License Key or have no internet access, contact the Matrix License Support Centre for assistance in generating the new License key.

Instructions for Customers

To activate your License, you would need the License Voucher containing the 16-digit License PIN. Contact your Dealer/Distributor in this regard. Your License Voucher may be a paper or a protected PDF file.

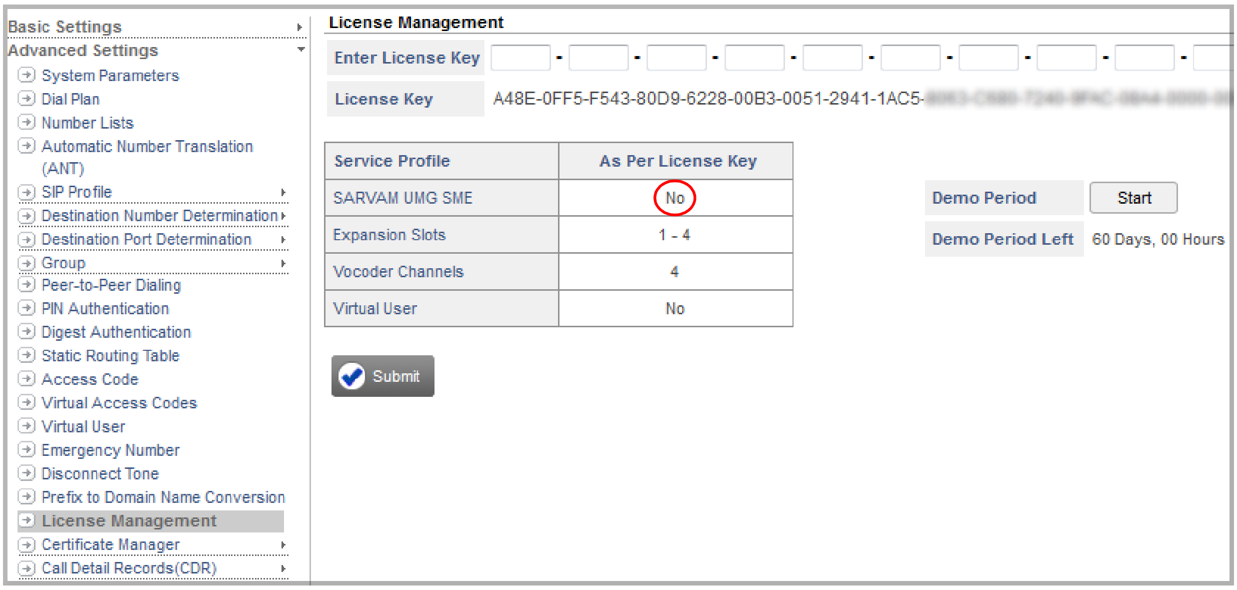
- Open Jeeves.
- Log in as System Engineer.
- Under **Advanced Settings**, click **License Management**. The License Management page opens.



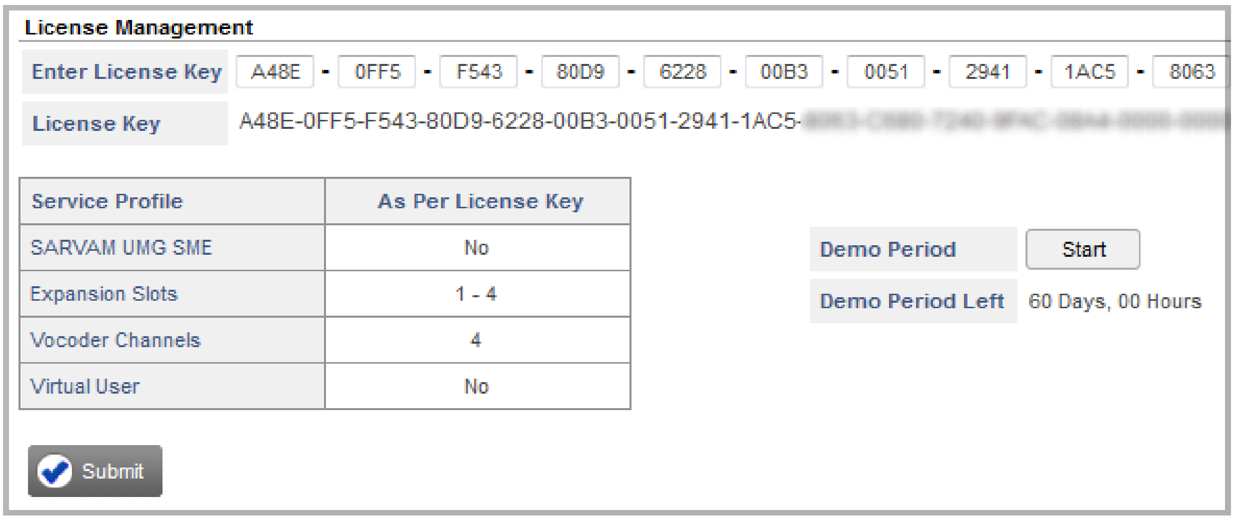
- Note down the current **License Key** on this page.

Under **Service Profile**, you can view the functional modules and features that are currently available to you.

- Send your Current License Key and the License PIN (on the Voucher) to the Matrix License Support Centre.
- You will receive a new License Key.
- Open Jeeves again.
- Log in as System Engineer.
- Click the **License Management** link.

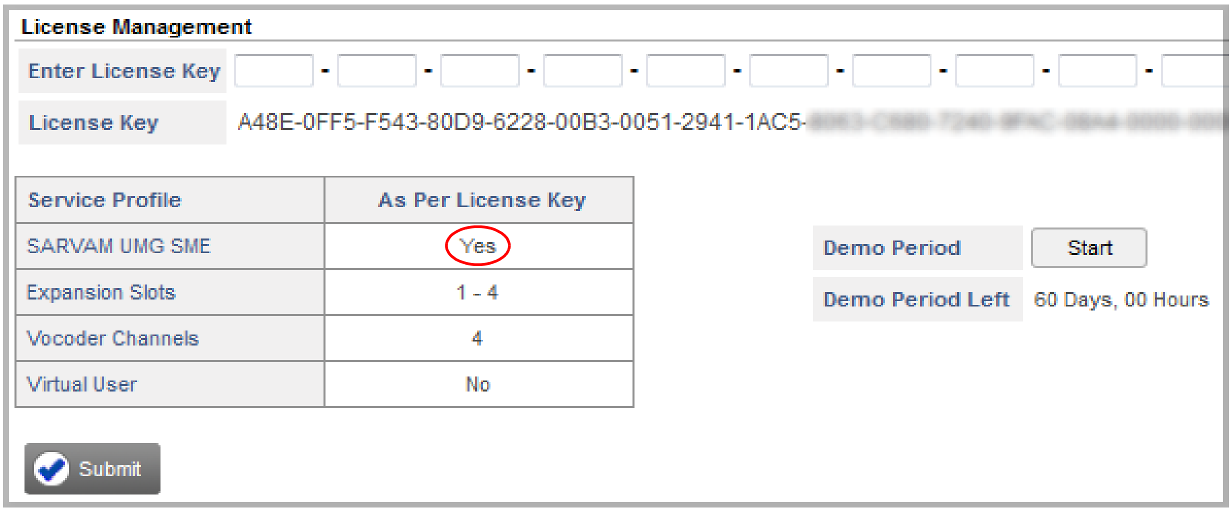


- In **Enter License Key**, enter the New License Key you obtained from Matrix.



- Click the **Submit** button.

The **Service Profile** on this page will be updated accordingly.



- 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.

Call Detail Records (CDR)

SARVAM UMG enables you to generate reports of Call Detail Records of calls using various filters such as:

- The port from which the calls originate (Source Port)
- The port on which the calls terminate (Destination Port)
- Calls made on particular dates
- Calls made at a particular time
- Calls of a certain duration
- Calls of certain Called Party Numbers
- Calls of certain Calling Party Numbers
- Calls made with PIN Authentication
- Calls made without PIN Authentication

You can set the different filters as required and generate Call Detail Record Report. The reports can be used for analyzing the call records for different purposes like cost savings, productivity enhancement, security and privacy.

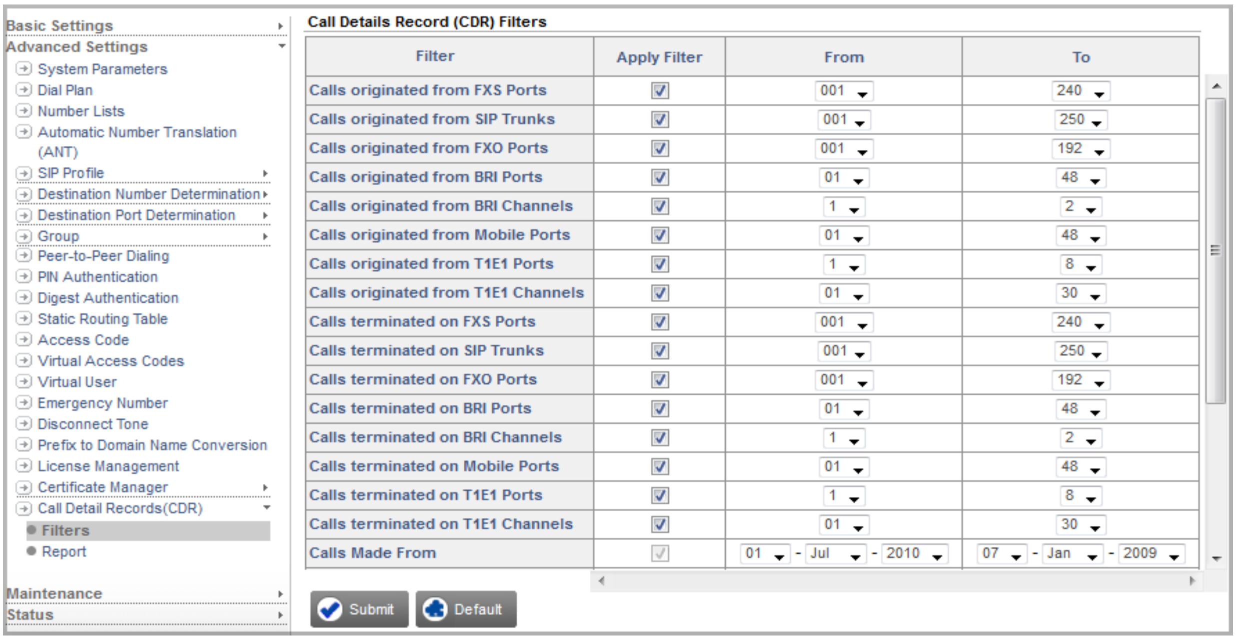
The system stores records of matured calls only and it generates reports only of the filters that are set. For example, if you have not enabled the filter for *Calls Originated from SIP Trunks*, the system will not generate report for calls originated from SIP Trunks.

SARVAM UMG supports upto 2000 call record entries and these entries are stored using the First In First Out (FIFO) method. Call records remain stored even when the system is set to default or the Firmware is upgraded.

Call records can be cleared manually or downloaded at any time.

Configuring Call Detail Record Filters

- Click the **Advanced Settings** link to expand.
- Click **Filters** under **Call Detail Record (CDR)** link.



Setting Filters

- To set the filters, click the **Filters** link under Call Detail Records (CDR).

  By default, all the filters are enabled. You may disable the filter you do not want to use by clearing the related **Apply Filter** check box.

  Some of these filters are enabled by default, you cannot disable them, but you can set them.

- Set the following filters as required:

The filters you set are not applied on the downloaded report. The CSV and TXT files will contain all the records, irrespective of the filters you set.

  - **Calls originated from FXS Ports:** The system will generate report of outgoing calls made from the FXS Ports of SARVAM UMG. To generate report using this filter for both FXS Ports, set the range of the FXS Ports in the **From** and **To** fields.

    You can also generate report for a single FXS Port, by setting the same port number in the **From** and **To** fields.

  - **Calls originated from SIP Trunks:** The system will generate report of calls that were received on the SIP Trunks of SARVAM UMG for further routing. To generate report using this filter for a range of SIP Trunks, select the range of the SIP Trunks in the **From** and **To** fields.

    You can also generate report for a single trunk, by setting the same trunk number in the **From** and **To** fields.

- **Calls originated from FXO Ports:** The system will generate report of calls that originated from the FXO Ports. Set the range of the FXO Ports in the **From** and **To** fields.
  You can generate report for a single FXO Port, by setting the same port number in the **From** and **To** fields.

- **Calls originated from BRI Ports:** The system will generate report of calls that originated from the BRI Ports. To generate report using this filter for a range of ports, set the range of the BRI Ports in the **From** and **To** fields.

  You can also generate report for a single BRI Port, by setting the same port number in the **From** and **To** fields.

- **Calls originated from BRI Channels:** The system will generate report of calls that originated from each BRI Channel. To generate report using this filter for a range of channels, set the range of the BRI Channels in the **From** and **To** fields.

  You can also generate report for a single channel, by setting the same channel number in the **From** and **To** fields.

- **Calls originated from Mobile Ports:** The system will generate report of calls that originated from the Mobile Ports. Set the range of the Mobile Ports in the **From** and **To** fields.

  You can generate report for a single Mobile Port by setting the same port number in the **From** and **To** fields.

- **Calls originated from T1E1 Ports:** The system will generate report of calls that originated from the T1E1 Ports. To generate report using this filter for a range of ports, set the range of the T1E1 Ports in the **From** and **To** fields.

  You can also generate report for a single T1E1 Port, by setting the same port number in the **From** and **To** fields.

- **Calls originated from T1E1 Channels:** The system will generate report of calls that originated from each T1E1 Channel. To generate report using this filter for a range of channels, set the range of the T1E1 Channels in the **From** and **To** fields.

  You can also generate report for a single T1E1 channel, by setting the same channel number in the **From** and **To** fields.

- **Calls terminated on FXS Ports:** The system will generate report of calls that terminated on FXS Ports. To generate report using this filter for both FXS Ports, set the range of the FXS Ports in the **From** and **To** fields.

  You can also generate report for a single FXS Port, by setting the same port number in the **From** and **To** fields.

- **Calls terminated on SIP Trunks:** The system will generate report of calls terminated on the SIP Trunks. To generate report using this filter for a range of SIP Trunks, set the range of the SIP Trunks in the **From** and **To** fields.

  To generate report for calls terminated on a single SIP Trunk, set the same trunk number in both fields.

- **Calls terminated on FXO Ports:** The system will generate report of calls that terminated on the FXO Ports. Set the range of the FXO Ports in the **From** and **To** fields.

  You can generate report for a single FXO Port, by setting the same port number in the **From** and **To** fields.

- **Calls terminated on BRI Ports:** The system will generate report of calls that terminated the BRI Ports. To generate report using this filter for a range of ports, set the range of the BRI Ports in the **From** and **To** fields.

  You can also generate report for a single BRI Port, by setting the same port number in the **From** and **To** fields.

- **Calls terminated on BRI Channels:** The system will generate report of calls that terminated on each BRI Channel. To generate report using this filter for a range of channels, set the range of the BRI Channels in the **From** and **To** fields.

  You can also generate report for a single channel, by setting the same channel number in the **From** and **To** fields.

- **Calls terminated on Mobile Ports:** The system will generate report of calls that terminated on the Mobile Ports. Set the range of ports in the **From** and **To** fields. Set the same port number in both fields, if you want to generate report for calls terminated on a particular Mobile Port.

- **Calls terminated from T1E1 Ports:** The system will generate report of calls that terminated the T1E1 Ports. To generate report using this filter for a range of ports, set the range of the T1E1 Ports in the **From** and **To** fields.

  You can also generate report for a single T1E1 Port, by setting the same port number in the **From** and **To** fields.

- **Calls terminated on T1E1 Channels:** The system will generate report of calls that terminated on each T1E1 Channel. To generate report using this filter for a range of channels, set the range of the T1E1 Channels in the **From** and **To** fields.

  You can also generate report for a single channel, by setting the same channel number in the **From** and **To** fields.

- **Calls made From:** The system will generate report of calls made between particular dates. Enter the start date and end date in the corresponding **From** and **To** fields.

- **Calls made Between:** The system will generate report of calls made between a particular time period. Enter the start time and end time in the corresponding **From** and **To** fields.

- **Called Party Number Matching with Number List:** The system generates report for calls made to specific numbers.

  Select the Number List you want to assign to this filter. Make sure that you also configure this Number List with the Called Party Numbers which you want the system to match. See “Number Lists” for instructions.

- **Calling Party Numbers Matching with Number List:** The system generates report for calls received from specific numbers.

Select a Number List you want to assign to this filter. Make sure that you also configure this Number List with the Calling Party Numbers which you want the system to match. See “Number Lists” for instructions.

- **Call Duration equal to and greater than (HH: MM: SS):** The system generates report for calls of a specific time duration. Select the call duration in HH: MM: SS format.

- **Calls without PIN Number:** The system will generate report for calls without PIN Authentication.

- **Calls with PIN Number:** The system will generate a report for calls that were made using PIN Authentication. You can generate report of calls of specific PIN Numbers.

  Enter the range of PIN Numbers in the **From** and **To** fields. PIN Numbers can be in the range of 0000 to 9999. The system will generate Report of all calls having PIN Numbers within the range you have set and display them under the ‘PIN Numbers’ column of the report.

  If you want to generate report of a particular PIN Number, enter the same PIN Number in the **From** and **To** fields.

- Click **Submit** to save the settings.

Clear Call Records

- You can clear the call detail records any time you want by clicking the **Clear Call Records** button.

Calls Made From		01 - Jul - 2010	30 - Apr - 2015
Calls Made Between		00 : 00	23 : 59
Called Party Numbers Matching with Number List		01	
Calling Party Numbers Matching with Number List		01	
Call Duration equal to and greater than (HH:MM:SS)		00 : 00 : 00	
Calls without PIN Number			
Calls with PIN Number		0001	9999
Clear Call Records Download Call Records			

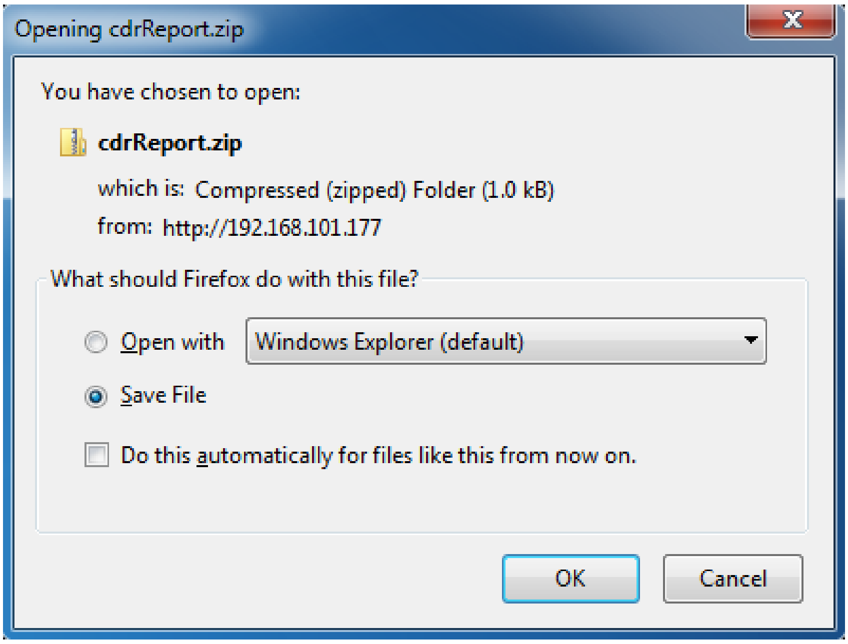
When call records are cleared, the **From** field of the filter **Calls Made Between** will change to the date of clearing of the records.

Download Call Records

- If you want to open/ save Call Detail Record Report on your computer, click the **Download Call Records** button.

*If you are using Mozilla Firefox (version 3.5 recommended), set the Downloads option of your browser as **Always ask me where to save the files**.*

- You will get a prompt with the option to open the **cdrReport.zip** file or save the file to a location. Save the file on the local disk.



- Open the cdrReport.zip file from the location you saved. The zip file contains the CDR report in Excel and Text format.

Printing Call Detail Record Report

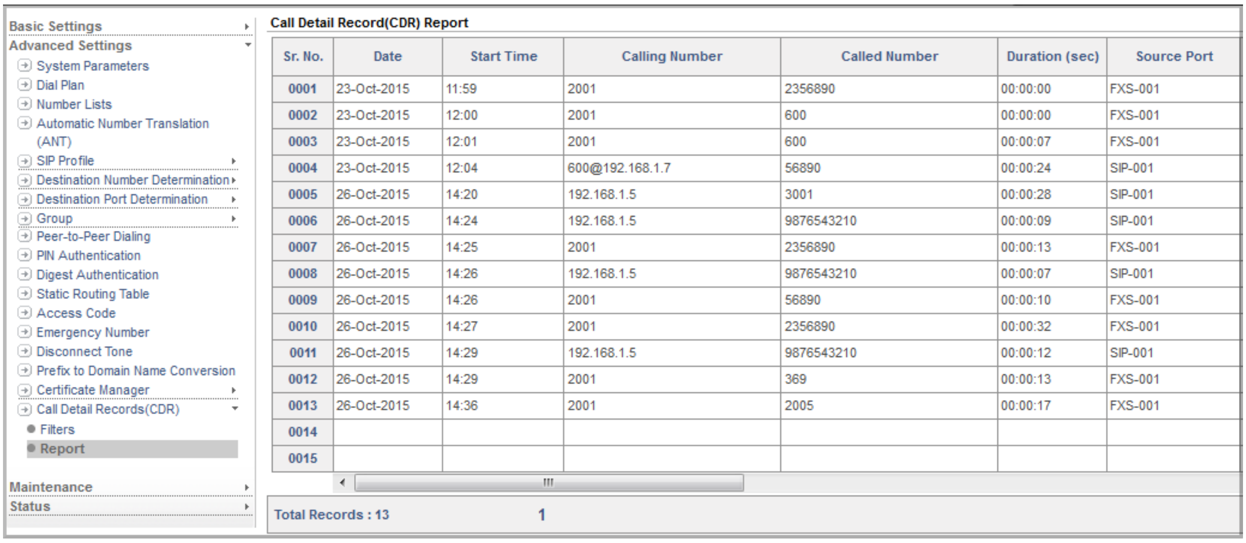
- You can also print the Call Detail Record Report, if required.
  - To print the CDR report in Excel format, open the file **CdrReport.csv**
  - To print the CDR report in text format, open the file **CdrReport.txt**
  - Print the file you opened. You may change the formatting of the text in the files before printing.

The filters you set are not applied on the downloaded report. The CSV and TXT files will contain all the records, without filters.

A sample **Call Detail Record Report** is presented at the end of this topic.

Viewing Call Detail Report

Click the **Report** link under Call Detail Record, to view the report generated by the system for the filters you have set.



Sr. No.	Date	Start Time	Calling Number	Called Number	Duration (sec)	Source Port
0001	23-Oct-2015	11:59	2001	2356890	00:00:00	FXS-001
0002	23-Oct-2015	12:00	2001	600	00:00:00	FXS-001
0003	23-Oct-2015	12:01	2001	600	00:00:07	FXS-001
0004	23-Oct-2015	12:04	600@192.168.1.7	56890	00:00:24	SIP-001
0005	26-Oct-2015	14:20	192.168.1.5	3001	00:00:28	SIP-001
0006	26-Oct-2015	14:24	192.168.1.5	9876543210	00:00:09	SIP-001
0007	26-Oct-2015	14:25	2001	2356890	00:00:13	FXS-001
0008	26-Oct-2015	14:26	192.168.1.5	9876543210	00:00:07	SIP-001
0009	26-Oct-2015	14:26	2001	56890	00:00:10	FXS-001
0010	26-Oct-2015	14:27	2001	2356890	00:00:32	FXS-001
0011	26-Oct-2015	14:29	192.168.1.5	9876543210	00:00:12	SIP-001
0012	26-Oct-2015	14:29	2001	369	00:00:13	FXS-001
0013	26-Oct-2015	14:36	2001	2005	00:00:17	FXS-001
0014						
0015						

Call Detail Record Report generated as per the filters you set will appear in the following columns:

- **Date:** Calls made between particular dates.
- **Start Time**: Calls made during a particular time period.
- **Calling Number:** Calls received from specific numbers.
- **Called Number:** Calls made to specific numbers.
- **Duration**: Calls of a specific time duration.
- **Source Port**: Calls originated from the SIP Trunks/FXO Ports/FXS Ports/Mobile Ports/BRI Ports/T1E1 Ports.
- **Destination Port**: Calls terminated on the SIP Trunks/FXO Ports/FXS Ports/Mobile Ports/BRI Ports/T1E1 Ports.
- **Disconnected By**: The port that disconnected the call.
- **Cause**: The cause for disconnection.
- **PIN Number**: Calls made using PIN Authentication, the PIN Number dialed by the caller.
- **Remarks**: The type of call. A for Anonymous, U for Unanswered and N for Normal.
- **By Port**: Displays the number of the FXS Port using the supplementary services as Call Forward, Blind Transfer, Attended Transfer etc.
- **By Number**: Displays the number assigned to the FXS Port using the supplementary services.

The total number of the records is displayed below the table.

On each page, 15 records are displayed. Click the page number at the bottom of the report to view the next 15 records.

The Alert message **No Calls to Display** will appear, if there are no records to be displayed.

Certificate Manager

SARVAM UMG supports certification for TLS, Web Server, Firmware Upgrade and Configuration Upgrade.

The two types of Certificates supported are: **Self-Signed Certificate** and **CA Signed Certificate**.

Self-Signed Certificate

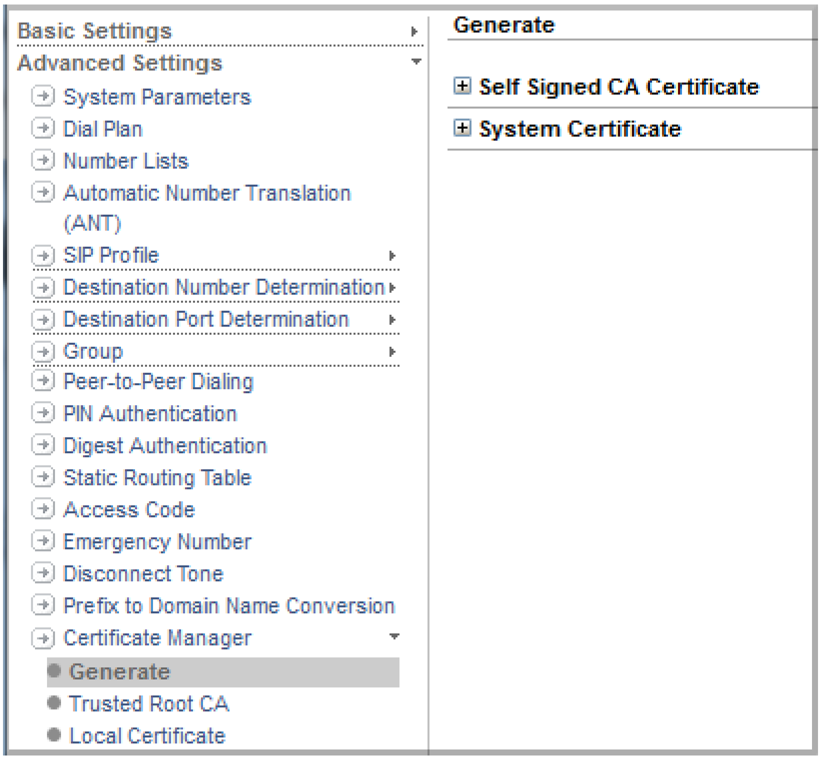
A self-signed certificate is created by the clients themselves or by the Servers and then given to their clients. It means that you yourself become the Certificate Authority (CA), create a CA Certificate and sign it. The self-signed certificate is faster to create but is not signed by a trusted CA Organization. The self-signed certificate must be installed in the trusted list of clients that connects over TLS with the Server. Because the certificate has been self-signed, the signature is not likely to be in the clients' trust file, hence, they need to add it.

If you select **Self-Signed Certificate**, you need to do the following:

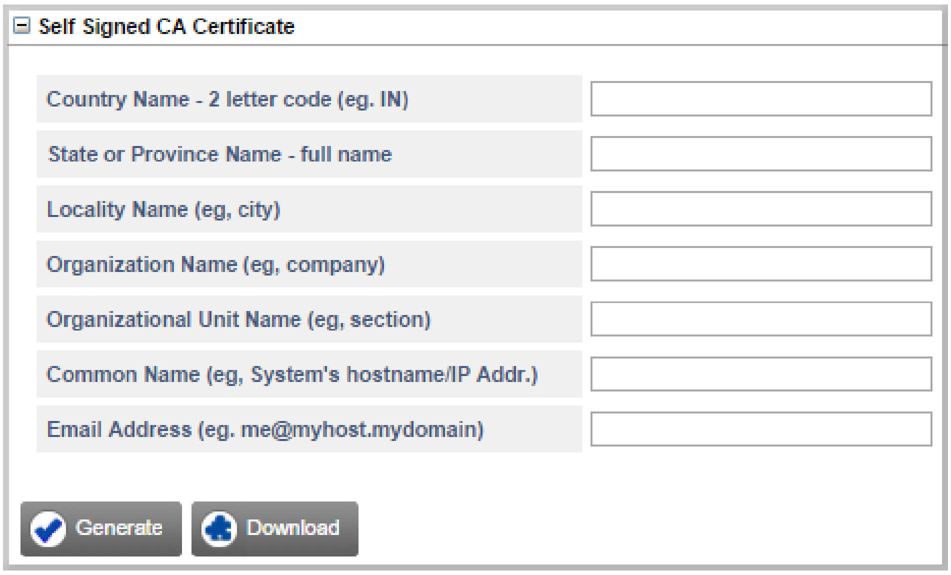
1. Create a Self-Signed CA Certificate.
2. Create a System Certificate (Self-Signed Certificate).

Generating a Self-Signed CA Certificate

- Click **Generate** under the **Certificate Manager** link.



- Click **Self Signed CA Certificate** to expand and configure the following parameters.

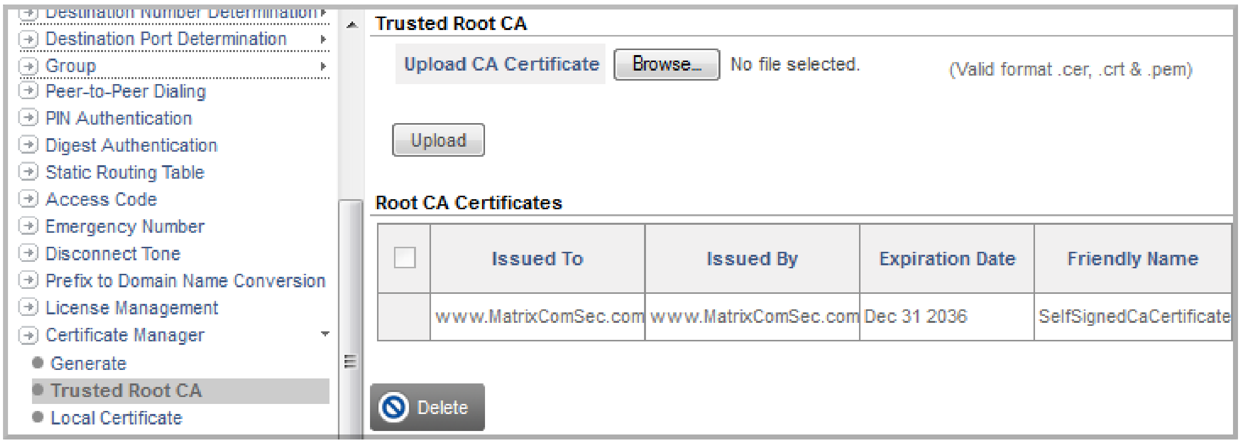


  - In **Country Name - 2 letter code (e.g. IN)**, enter the name of your country.
  - In **State or Province Name - full name**, enter the full name of your state or province.
  - In **Locality Name (e.g. city)**, enter the name of your city.
  - In **Organization Name (e.g. company)**, enter the name of your organization where SARVAM UMG is installed.
  - In **Organizational Unit Name (e.g. section)**, enter the name of the unit or section or domain of your organization, where SARVAM UMG is installed.
  - In **Common Name (e.g. System's hostname/IP Addr.)**, enter your Server's (SARVAM UMG) host name or IP Address. This Common Name serves as the distinguishing factor.
  - In **Email Address (e.g. me@myhost.mydomain)**, enter your host's e-mail address.
- Click **Generate**, to generate this self-signed CA Certificate.

Once you generate self-signed certificate, you must send it to your clients so that they install it in their trusted list.

- To do this, click **Download**. Save the file at the desired location.

- Click the **Trusted Root CA** under **Certificate Manager** link. The CA Certificate you created appears in the **Root CA Certificate** table.



- If you want to upload other CA Certificates, in **Upload CA Certificate** browse the location at which the certificate is saved and click **Upload**. The CA Certificate you uploaded appears in the **Root CA Certificate** table. Valid format are .cer, .crt and .pem.

- To delete a CA Certificate, select the check box of the respective Root CA Certificate and click **Delete.**

A sample Self-Signed CA Certificate is as follows:

```
Certificate:
    Data:
        Version: 3 (0x2)
        Serial Number: 2 (0x2)
    Signature Algorithm: sha1WithRSAEncryption
        Issuer: C=IN, ST=Gujarat, L=Vadodara, O=MATRIX COMSEC PVT. LTD., OU=R&D, CN=www.MatrixComSec.com/emailAddress=Support@MatrixComSec.com
        Validity
            Not Before: Apr 30 07:42:32 2015 GMT
            Not After : Dec 31 07:42:32 2036 GMT
        Subject: C=IN, ST=Gujarat, L=Vadodara, O=MATRIX COMSEC PVT. LTD., OU=R&D, CN=www.MatrixComSec.com/emailAddress=Support@MatrixComSec.com
        Subject Public Key Info:
            Public Key Algorithm: rsaEncryption
                Public-Key: (2048 bit)
                Modulus:
                    00:a2:93:45:0d:40:0b:97:90:a5:c5:83:ae:2a:25:
                    d9:8d:55:05:06:09:d5:03:ed:b1:9f:53:34:b7:8f:
                    2c:43:f6:d0:ff:a1:4e:30:ec:ed:8c:c8:b8:28:53:
                    ad:25:19:19:a6:e4:da:3d:b1:75:79:c4:c7:e1:5f:
                    58:ad:68:55:ae:f2:08:f8:82:f4:cc:be:cd:28:7a:
                    8f:57:99:d7:41:e0:c4:57:a7:02:e9:7a:e1:95:1a:
                    f2:9d:d0:66:49:17:60:a0:c9:51:eb:cd:ff:87:e0:
                    1f:3f:5c:5d:34:46:67:67:22:99:2a:46:c6:16:3d:
                    c1:3e:fc:f6:65:62:43:9e:d8:b1:99:96:c1:a3:47:
                    03:53:78:17:77:22:fd:b5:c2:4a:94:9b:ec:b8:f3:
                    52:c9:c7:cf:95:7c:a9:df:bc:c3:d2:2f:b7:26:1f:
                    c5:0c:06:29:ae:c3:25:69:d2:eb:40:4a:0a:d4:cc:
                    de:60:be:c2:73:33:3c:d1:cd:b7:a6:9f:37:0c:86:
                    c8:28:52:b9:06:8f:3e:58:c1:e7:f4:7c:a0:0d:b9:
                    91:40:53:b8:ac:42:99:2b:7c:12:ae:4c:57:34:c4:
                    77:83:95:2b:be:3b:04:d7:58:3c:87:cf:f4:10:cf:
                    80:15:44:f4:17:5d:c7:eb:73:42:01:88:b2:c8:10:
                    9c:8b
                Exponent: 65537 (0x10001)
```

In the above Self-Signed CA Certificate:

- C = Country
- ST = State
- L = Location
- O = Organization
- OU = Organization Unit
- CN = Common Name

- **Issuer** represents the details of the CA issuing the Certificate. Here, the Organization itself is the CA (issuer), hence, the O, OU and CN of both Issuer and Subject is same.

- **Validity** represents the valid period of this certificate.

- **Subject** represents the credentials of the Server / User requesting for certification.

- **Public Key** represents the public key of the certificate.

Generating a System Certificate (Self-Signed Certificate)

After creating a Self-Signed CA Certificate, you can either,

- generate a System Certificate for your clients. These System Certificates can then be given to the respective clients.
  or
- the Clients can prepare their own System Certificates. For this you need to send them the CA Certificate created by you.
  or
  generate a Certificate Signing Request (CSR), if you want the Certificate to be signed by a third party.

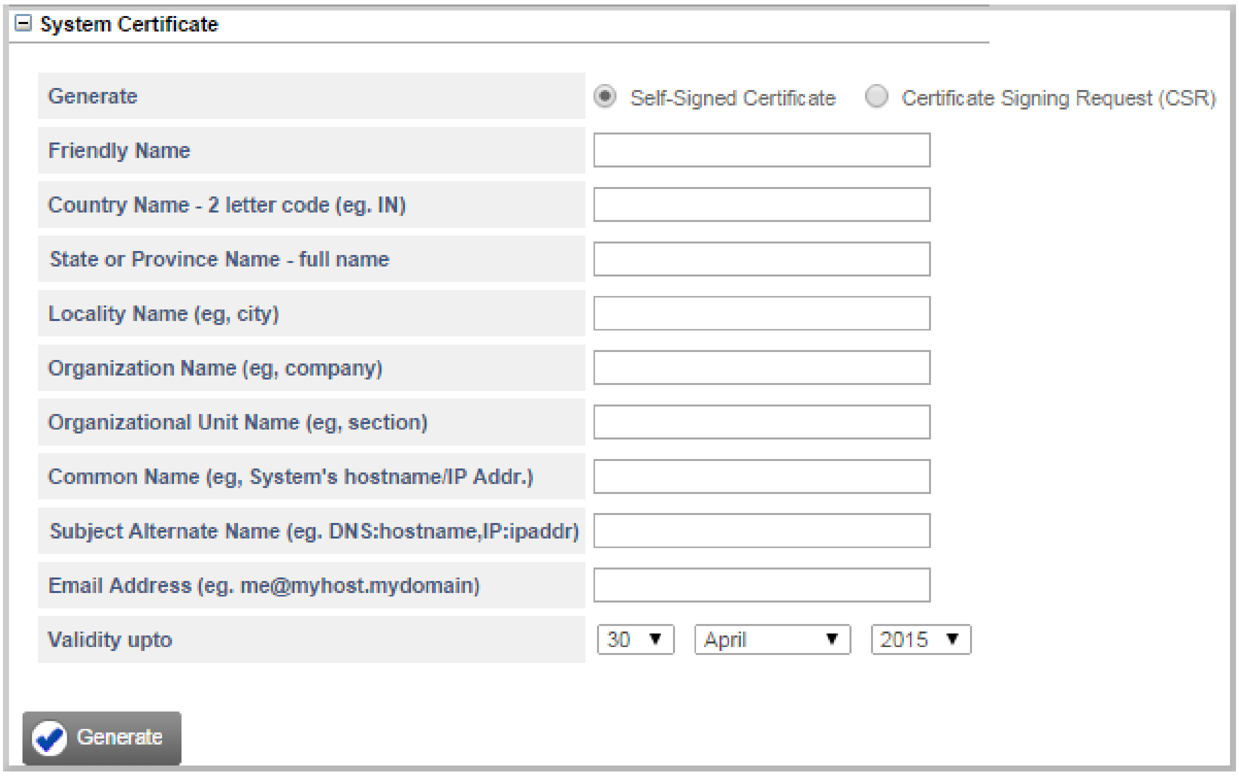
If the clients prepare their own certificates, you need to send your CA Certificate to all the clients. The clients must upload the same in their system. Similarly, all the clients must send their CA Certificates to you and you must upload the same in your system. To avoid this, it is recommended that you create the Certificates and then provide it to your clients.

To create the System Certificate,

- Click the **Advanced Settings** link to expand.

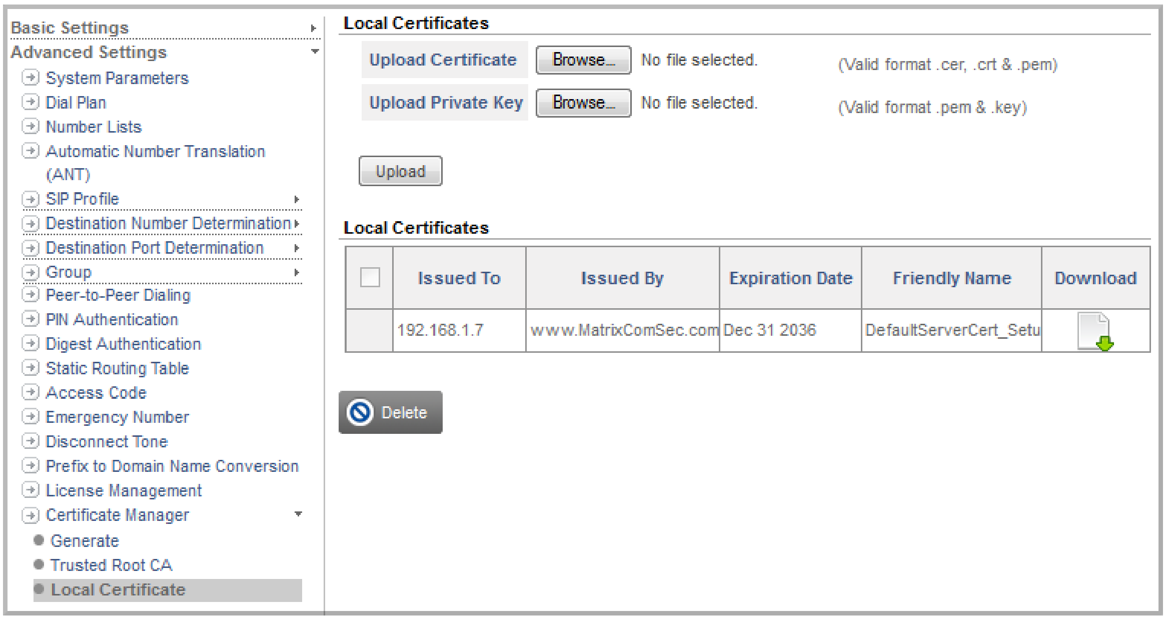
- Click **Generate** under the **Certificate Manager** link to expand.

- Click **System Certificate** to expand and configure the following parameters.



  - In **Generate**, select the type of certificate you want to create. You must select **Self-Signed Certificate**.
  - In **Friendly Name**, enter the name you want to assign to the certificate.
  - In **Country Name - 2 letter code (e.g. IN)**, enter the name (two letter code) of your country.
  - In **State or Province Name - full name**, enter the full name of your state or province.
  - In **Locality Name (e.g. city)**, enter the name of your city.
  - In **Organization Name (e.g. company)**, enter the name of your organization where SARVAM UMG is installed.
  - In **Organizational Unit Name (e.g. section)**, enter the name of the unit or section or domain of your organization, where SARVAM UMG is installed.
  - In **Common Name (e.g. System's hostname/IP Addr.)**, enter your Server's (SARVAM UMG) host name or IP Address. This Common Name serves as the distinguishing factor.
  - In **Subject Alternate Name (e.g. DNS:hostname,IP:ipaddr)**, enter the name of the multiple domain separated by comma (if the same certificate is to be issued for multiple domain of the organization).
  - In **Email Address**, enter the your host's e-mail address.
  - In **Validity Upto**, select the date till which this certificate will be valid.
- Click **Generate**, to generate this System Certificate.

- Under **Certificate Manager**, click the **Local Certificate** link. The generated certificate appears in the **Local Certificates** table.



- If you want to upload other System Certificates, in **Upload Certificate** browse the location at which the certificate is saved. Along with the certificate you also need to upload the Private Key, in **Upload Private Key** browse the location at which the key is saved and click **Upload**.

  The System Certificate you uploaded appears in the **Local Certificates** table.Valid formats for certificate are .cer, .crt and .pem. Valid format for key are .pem and .key (Base64 encoded ASCII file).

- To delete a System Certificate, select the check box of the respective Certificate and click **Delete**.
- To download the System Certificate, click **Download** .

A sample Default Server System Certificate is as follows:

```
Certificate:
    Data:
        Version: 3 (0x2)
        Serial Number: 1 (0x1)
    Signature Algorithm: sha1WithRSAEncryption
        Issuer: C=IN, ST=Gujarat, L=Vadodara, O=MATRIX COMSEC PVT. LTD., OU=R&D, CN=www.MatrixComSec.com/emailAddress=Support@MatrixComSec.com
        Validity
            Not Before: Apr 30 07:59:06 2015 GMT
            Not After : Dec 31 07:59:06 2036 GMT
        Subject: C=IN, ST=Gujarat, L=Vadodara, O=MATRIX COMSEC PVT. LTD., OU=R&D, CN=www.MatrixComSec.com/emailAddress=Support@MatrixComSec.com
        Subject Public Key Info:
            Public Key Algorithm: rsaEncryption
                Public-Key: (2048 bit)
                Modulus:
                    00:ae:e3:26:c7:54:a9:37:f7:11:42:19:0f:21:bd:
                    3b:6e:9d:fe:9c:83:48:da:74:c9:d7:3f:2f:f3:bc:
                    cb:33:e4:29:8c:e7:b5:5d:19:9d:e7:bd:27:e7:6e:
                    16:60:60:35:13:cf:fb:e6:e4:48:e3:3a:a7:27:ae:
                    4e:82:6a:05:9d:ec:e9:b6:bb:a1:80:56:73:39:28:
                    a0:30:e3:23:f0:d9:31:19:57:cc:d7:a2:74:99:4d:
                    cf:86:b6:d5:b8:0e:4e:5d:9c:ad:b4:56:dd:1e:37:
                    1d:d0:49:03:32:a7:e7:02:64:bb:36:09:19:b0:43:
                    34:f3:6b:83:61:9a:64:53:63:19:3d:e4:80:1b:39:
                    f8:93:22:c6:f3:60:7a:1a:e2:de:81:eb:90:fa:f7:
                    3b:88:4a:c6:38:62:32:93:b7:6b:d2:ea:87:6c:9f:
                    7b:86:3f:fd:dd:0e:2b:a6:68:38:09:11:66:5f:e1:
                    35:4c:d5:af:20:1a:b4:66:b9:30:f5:8b:0a:63:cc:
                    30:4d:c6:e6:05:51:0e:62:ff:d7:b5:24:42:57:98:
                    47:85:74:b8:7c:51:66:f7:42:9f:ce:62:f1:fb:07:
                    b6:5b:74:3d:fe:a6:42:d1:80:1c:4a:10:51:9e:ee:
                    e6:f1:7b:1b:31:05:35:fc:45:ed:1d:e1:5a:e2:e6:
                    55:b5
                Exponent: 65537 (0x10001)
```

In the above Server Certificate,

- **Issuer** represents the details of the CA issuing the Certificate. Here, the Organization itself is the CA (issuer), hence, the O and CN of both Issuer and Subject is same.
- **Validity** represents the valid period of this certificate.
- **Subject** represents the credentials of the Server / User requesting for certification. Here, OU=R&D i.e. for whom the certificate is signed.
- **Public Key** represents the public key of the certificate.

CA Signed Certificate

Certificate Authority (CA) is a trusted organization which creates and sells TLS Certificates to websites. *CA Signed Certificates* are the TLS Certificates which are created by such trusted CAs, signed and sold to any applicant. These certificates contain a public key and the identity of the owner; and it is upto the CA to verify the owner's (applicant's) credentials. 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 certain period of time.

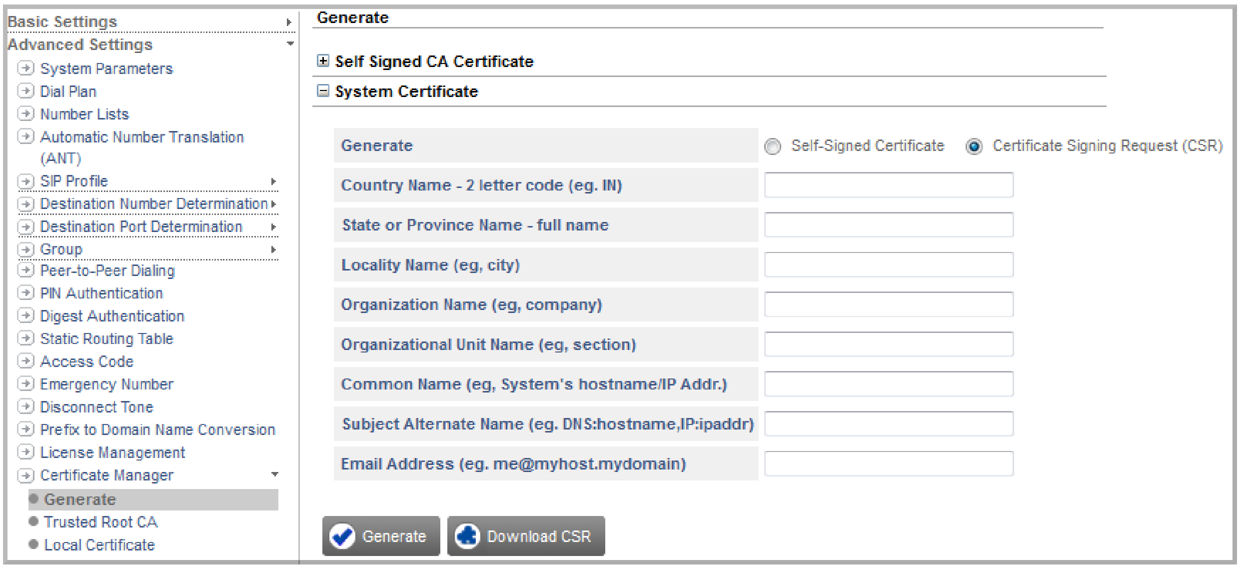
If you want to get a **CA Signed Certificate**, you need to do the following:

1. Generate and enroll the Certificate Signing Request (CSR).
2. Get the Certificate Signing Request (CSR) verified and signed by the Certified Authority (CA).

Generating the Certificate Signing Request

- Click the **Advanced Settings** link to expand.
- Click **Generate** under the **Certificate Manager** link.

- Click **System Certificate** to expand and configure the following parameters.



  - In **Generate**, select the type of certificate you want to create. You must select **Certificate Signing Request (CSR)**.
  - In **Country Name - 2 letter code (e.g. IN)**, enter the name (two letter code) of your country.
  - In **State or Province Name - full name**, enter the full name of your state or province.
  - In **Locality Name (e.g. city)**, enter the name of your city.
  - In **Organization Name (e.g. company)**, enter the name of your organization where SARVAM UMG is installed.
  - In **Organizational Unit Name (e.g. section)**, enter the name of the unit or section or domain of your organization, where your SARVAM UMG is installed.
  - In **Common Name (e.g. System's hostname/IP Addr.)**, enter your Server's (SARVAM UMG) host name or IP Address. This Common Name serves as the distinguishing factor.
  - In **Subject Alternate Name (e.g. DNS:hostname,IP:ipaddr)**, enter the name of the multiple domain separated by comma (if the same certificate is to be issued for multiple domain of the organization).
  - In **Email Address (e.g. me@myhost.mydomain)**, enter your host's e-mail address.
- Click **Generate**, to generate this System Certificate.
- To send the certificate to the signing authority, click **Download CSR**. The Certificate and the Key downloads.

The Certificate Signing Request (CSR) to be sent to any trusted CA, appears as under:


```
-----BEGIN CERTIFICATE REQUEST-----
MIIDLDCCAhQCAQAwgaoxCzAJBgNVBAYTAklOMRAwDgYDVQQIEwdHdWphcmF0MREw
DwYDVQQHEwhWYWRvZGFyYTEgMB4GA1UEChMXTUFUUklYIENPTVNFQyBQV1QuIExU
RC4xDDAKBgNVBAsUA1ImRDEdMBsGA1UEAxMUd3d3Lk1hdHJpeENvbVNlYy5jb20x
JzAlBgkqhkiG9w0BCQEWGFN1cHBvcnRATWF0cml4Q29tU2VjLmNvbTCCASIwDQYJ
KoZIhvcNAQEBBQADggEPADCCAQoCggEBAPutA1/cZcz/qZe3soIITiVpPI8PIZ6d
9RvInx4haqVob7Ml10dYWvN2rLmFod3ZtEu9dX645crC4NXn9pxKXmkp5iNdBVca
rm1qedZ63SlcR3m4YhL2dUc7DQ9T1GNTPbXLrlA4sQk+nVwO+C+XU/jPlpqiR0sn
Idh2/eLWVOauRgY3qdGjPaN8ndq8xVieY+v1/XpLQa4Oyd6aP+xn+z4pSWK4YLeP
36/CRh5q4f3vfMpuQTfegxGA+UB1V3qPMSqI0jBr7r1jptDxlmwzXkwz5w1rovh8
ZNP+1sIYPyZ9zrZm+eyhxpSX8o09jCcEm/R816x6GHEER7UGdZR1HvUCAwEAAaA8
MDoGCSqGSIb3DQEJDjEtMCswCQYDVR0TBAIwADALBgNVHQ8EBAMCBeAwEQYDVR0R
BAowCIIGTWF0cml4MA0GCSqGSIb3DQEBBQUAA4IBAQCQtMJnA13HAWYa9wlJGbKW
YjoC/gbrhSUwgbR4Jh+13guInViTyJ5YDt9pLc8xzJe23MV2XDv4ImSSUSkRojcg
IpVTqNPgf9lk50WmJHTIT0JJGEUXvzKE71V0kuf0XTelW0o81QYpjGn8GaSQqCDV
q746F0i84zwsejY+/jL+pDMpczxvbnotWg+wCkMXwdkAk0InqL+DuSTEnuBEcW82
UFe0rqoMdt90XpS9YZpjIsotRYgTRNIFaBFF4LxQa1bYQl5pZ79MxWJIZQZTnqHf
MbwSosS/QM7ZjE147b13m9Lk69jdzfSAPmCW4Adu1Be7PENGGI+MMzfAVyYSwdkw
-----END CERTIFICATE REQUEST-----
```


Enrolling the Certificate Signing Request with CA

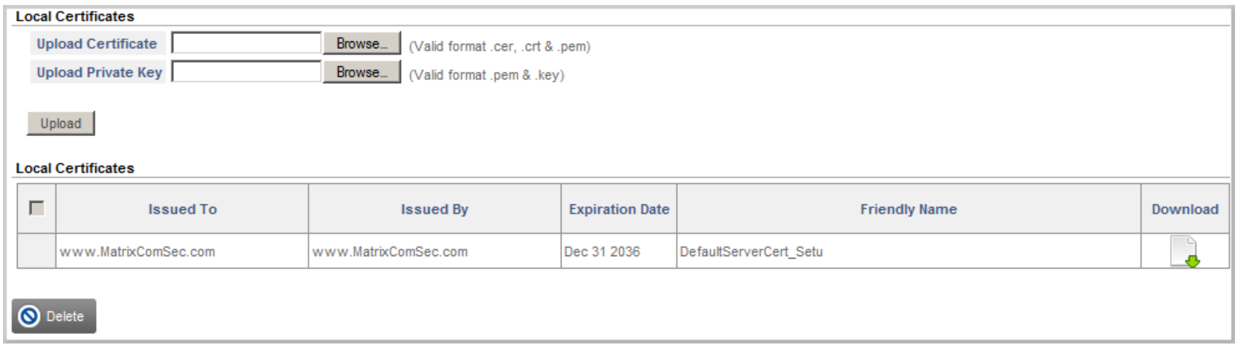
Enrollment is a process of obtaining a certificate from any trusted third party (CA). After you have generated the Certificate Signing Request (CSR), you must contact any authorized third party that issues TLS Certificates to companies or web owners, such as Thawte, VeriSign, etc. and enroll the Certificate 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 year. After the Certificate Signing Request (CSR) has been validated and signed by the CA, it becomes the CA Signed Certificate.

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.

After you receive the signed certificate, you must:

- Click the **Certificate Manager** link to expand.
- Click the **Local Certificate** link.



- In **Upload Certificate** browse the location at which the certificate is saved. Along with the certificate you also need to upload the Private Key, in **Upload Private Key** browse the location at which the key is saved and click **Upload**

  The System Certificate you uploaded appears in the **Local Certificates** table.Valid formats for certificate are .cer, .crt and .pem. Valid format for key are .pem and .key (Base64 encoded ASCII file).

  To delete a System Certificate, select the check box of the respective Certificate and click **Delete**.

  To download the System Certificate, click **Download** .

CHAPTER 10

Features of SARVAM UMG

SARVAM UMG offers users the following telephony features, which they can access by dialing Access Codes.

- Call Hold
- Making a Second Call
- Call Toggle
- Call Transfer - Attended and Blind
- Call Forward
- Conference
- DND
- Call Waiting
- Hotline
- Supplementary Services of Service Providers
- Making a new call using Access Code
- Disconnecting a call using Access Code

You can change the default access codes assigned to these features and facilities as per your requirement. See “Access Code”.

In addition to these, SARVAM UMG offers users IP Dialing and Fax over IP (FoIP). It supports *Remote Call Forward*, *Remote Held*, and *Remote Transfer* using SIP Signaling.

Call Hold

Call Hold enables you to put an on-going conversation on hold and call another person, or receive a call from another person. You can retrieve the call you put on hold after the conversation with the other party has ended. You can also retrieve the call you put on hold in the middle of the conversation with the other party.

Call Hold is also used in the following features:

- Retrieve Held Call
- Make a Second Call
- Call Toggle
- Call Conference
- Call Transfer
- Call Waiting

Configuring Call Hold

To use this feature make sure,

- Call Hold is enabled in the **Class of Service** of the FXS Port.
- The **Subscriber Type** of the FXS Port is configured as **Gateway**.

For detailed instructions, see "FXS Port" under *Basic Settings*.

How to use Call Hold

- You are in speech with party A.
- To put your call with A on hold, dial ***Flash***.
- A is put on hold. You get feature tone for 7 seconds, followed by an error tone.
- To retrieve the call you put on hold, dial ***Flash*** again during feature tone or during error tone.
- You will be in speech with A.

- *If you go On-hook during the feature tone, your call with party A will be disconnected.*
- *If you go On-hook during error tone, you will get ring back.*
- *If you go Off-hook when your phone rings, you will get connected with party A again.*

Making a Second Call

You can make a second call, by putting the current call on hold.

Configuring Making a Second Call

To use this feature, **Call Hold** must be enabled in the **Class of Service** of the FXS Port. For instructions, see "FXS Port" under Basic Settings.

How to make a Second Call

- You are in speech with party A.
- You want to talk to party B.
- Dial ***Flash*** to put party A on Hold.
- You get feature tone.
- Dial the number of the desired party B.
- When party B answers the call, you are in speech with party B.

- *Making a second call feature can also be used with other features such as Call Transfer-Attended, Call Toggle and 3-Party Conference.*
- *For example, after making a second call, you can toggle between the first and the second call using the Call Toggle access code; you can also conduct a Conference with both parties, by dialing the Conference access code.*

Call Toggle

Call Toggle (Call Split) allows you to have two simultaneous telephone conversations, talking to two people alternately.

The parties for Call Toggle can be:

- Two outgoing calls
- Two incoming calls
- One outgoing call and one incoming call.

You must dial the Call Toggle Access Code to switch between the held call and the active call. You can use Call Toggle only when there are two held calls on the FXS Port.

Configuring Call Toggle

To use this feature, **Call Toggle** must be enabled in the **Class of Service** of the FXS Port, see "FXS Port" under Basic Settings.

How to Toggle between calls

- You are in speech with A and you want to talk to B.
- Dial ***Flash,*** A is put on hold. You get feature tone.
- Dial the number of B. When B answers the call, you are in speech with B.
- To talk to A, dial ***Flash-#2****.*
- You are in speech with A and B is put on hold.
- To talk to B, dial ***Flash-#2*** again.
- You are in speech with B. A is put on hold.
- This way, you can talk alternately to A and B, by dialing ***Flash-#2*** again.

- *When toggling between calls, you can disconnect the call you are currently in speech with, by going On-hook.*
- *You can also use Call Toggle (Call Split) during a Conference call.*

Call Transfer-Attended

Attended Call Transfer is when you transfer the call to the desired party after consulting the party and or obtaining their consent for transfer.

In the case of SIP to SIP Attended Transfer SARVAM UMG enables the transferor to know the result of the transfer activity, whether the call has been transferred successfully or not.

As soon as the transferor goes On-hook or dials the Attended Transfer access code (if configured) to transfer the call, SARVAM UMG loads the Transfer Notification Timer.

SARVAM UMG indicates to the transferor of the result of the transfer activity within this Timer. The Transfer Notification Timer is configurable.

This is how Attended Transfer works:

- A (transferor) is in speech with B. A wants to transfer B's (transferee) call to C (transfer target).
- A dials Flash (Call Hold access code) to put B on hold and then dials C's number.
- A goes On-hook (the default Access Code for attended call transfer) while the C's number is ringing or after speech with C.
- B is in speech with C.

- *In Attended Call Transfer, the Transfer Notification Timer does not stopped even if the transferor goes On-hook.*
- *The default Attended Transfer access code* **^** *(On-hook) can be changed as per user requirement. If an access code other than* **^** *(On-hook) is assigned to Attended Transfer, then the transferor (A, in this case) must dial the Attended Transfer access code after dialing the transfer target's number (C's number).*

Configuring Attended Transfer

To use this feature, **Attended Transfer** and **Call Hold** must be enabled in the **Class of Service** of the FXS Port. For instructions, see "FXS Port" under *Basic Settings*.

Set the duration of the **Transfer Notification Timer**. Valid range is 001 to 999 seconds. Default: 60 seconds. To change the duration of this timer, see "System Parameters".

How to use Attended Transfer

- You are in speech with party A.
- Party A wishes to speak to party B.
- Dial ***Flash*** to put party A on Hold.
- Dial Party B's Number.
- When Party B answers the call, go On-hook to transfer.

- You can go On-hook even while Party B’s number is ringing.

Call Transfer - Blind

Blind Transfer is when you transfer the call to the desired party, without informing the party of the transfer.

SARVAM UMG enables the transferor to know the result of the transfer activity; whether the call has been transferred successfully or not. As soon as the transferor dials the Blind Call Transfer access code, SARVAM UMG loads the Transfer Notification Timer. This timer stops if the transferor goes On-hook. SARVAM UMG indicates to the transferor the result of the transfer activity within this Timer. The Transfer Notification Timer is configurable.

This is how Blind Call Transfer works:

- A (transferor) is in speech with B (transferee). A wants to transfer B's (transferee) call to C (transfer target), without informing C.
- A dials Flash (or Call Hold access code) to put B on hold and then dials Blind Call Transfer access code.
- A dials the number of C and B's call is transferred to C.

Now, to know the result of the Blind call transfer, A should remain Off-hook after dialing C's number. One of the following results may occur:

- If transfer is successful, A (transferor) gets confirmation tone for the duration of the Confirmation Tone Timer, followed by error tone for the duration of the Error Tone Timer, followed by system standby.
- If the transfer target C is busy, A (transferor) gets busy tone, and speech is established between A and the transferee B.
- If no message is received during the Transfer Notification Timer, A (transferor) gets error tone for Error Tone Timer, followed by system standby.

Blind Call Transfer is not allowed if you dial the Blind Transfer access code after putting the remote held call on hold.

Configuring Blind Transfer

To use this feature, you must enable **Blind Transfer** and **Call Hold** in the **Class of Service** of the FXS Port. For instructions, see “FXS Port” under *Basic Settings*.

Set the duration of the **Transfer Notification Timer**. Valid range is 001 to 999 seconds. Default: 60 seconds. To change the duration of this timer, see “System Parameters”.

How to use Blind Transfer

- You are in speech with party A.
- Party A wishes to speak to party B.
- Dial ***Flash***
- Dial ***#6*** Blind Transfer Access Code.
- Dial the number of party B.

- On successful transfer, you will get confirmation tone.
- Party A will get connected with party B.
- Replace the handset of your phone.

When you transfer a call, remain Off-hook, i.e. do not replace your handset, after dialing the transfer destination's number, until you know the result of the call transfer. One of the following results may occur:

- *If the transfer is successful, you will get confirmation tone, followed by error tone.*
- *If the transfer target is busy, you will get busy tone, and you will be connected to the transferee (party A, in this case).*
- *If the transfer has timed out, i.e. the Notification Timer has expired and no notification has been received from the transfer target (party B, in this case) you will get error tone, followed by system standby.*

Call Forward

When you are away from your phone, but would like to answer your calls, you can use the Call Forward feature to have SARVAM UMG forward incoming calls on the FXS Ports to another number.

SARVAM UMG supports the following Call Forward options:

- **Call Forward-Unconditional**: All incoming calls received on the FXS Port are forwarded to the desired destination number, automatically without waiting for a response from your phone.
- **Call Forward- Busy**: All incoming calls received on the FXS Port are forwarded to the destination number, when your phone is busy.
- **Call Forward- No Reply:** All incoming calls received on the FXS Port are forwarded to the destination number, when you do not answer your phone.

  When you set Call Forward-No Reply, SARVAM UMG waits for the duration of the **Call Forward-No Reply Timer** for the phone to answer. This timer is configurable, and is set to 45 seconds as default.

  If the phone does not answer before the timer expires, SARVAM UMG considers it as No Reply and forwards it to the desired destination number.

You can set a different destination number for each Call Forward option.

- *Call Forward-Unconditional has priority over Call Forward-Busy and Call Forward-No Reply.*
- *Call Forward-When Busy has priority over Call Waiting feature in all conditions except in call mature state, that is, when the FXS Port is in speech with another port.*

Configuring Call Forward

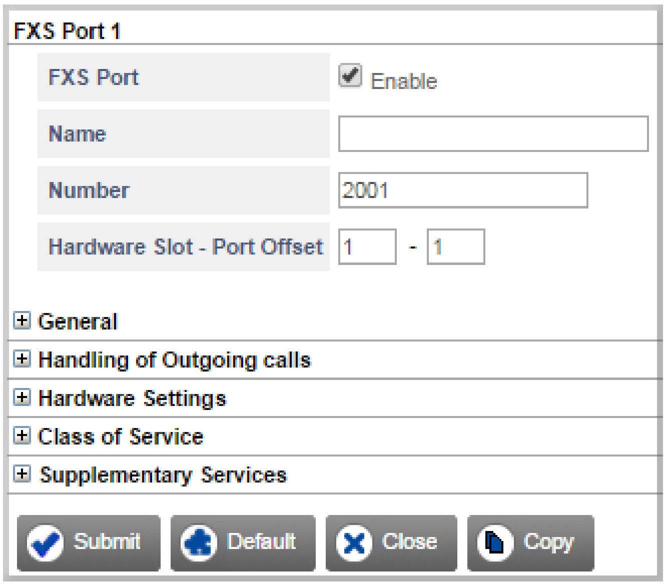
To use this feature, **Call Forward** must be enabled in the **Class of Service** of the FXS Port. For instructions, see "FXS Port" under *Basic Settings*.

How to use Call Forward

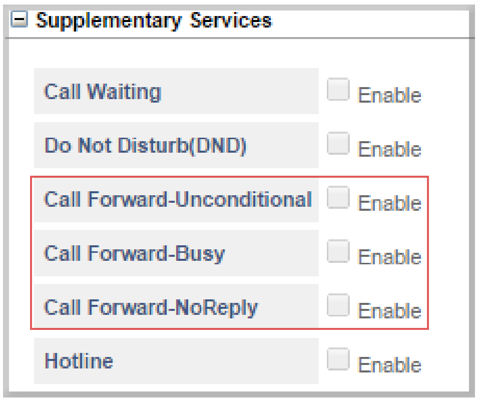
You can set/cancel Call Forward from Jeeves or from the phone connected to the FXS Port.

To set or cancel Call Forward from Jeeves,

- Click the **Basic Settings** link to expand.
- Click the **FXS Port** link.
- Click the FXS Port number tab, on which you want to use this feature. The page of the selected port opens.



- On the FXS Port page, click **Supplementary Services** link to expand.



- You may select — **Call Forward-Unconditional**, **Call Forward-Busy** or **Call Forward-No Reply.** To do so, select the respective check box to enable. Default: Disabled.

- For the Call Forward option you select, in the corresponding **Number** field that appears, enter the desired destination number (upto 40 characters) you want the calls to be forwarded to.

- If you selected **Call Forward- No Reply**, if required, you may also change the duration of the no-reply **Ring Timer**. The range of the Call Forward No-Reply Timer is 01 to 99 seconds. Default: 15 seconds.

- Click **Submit** to save.

To cancel the Call Forward option you selected, clear the respective check box.

To set or cancel Call Forward from the Phone,

- Lift handset of your phone.

Call Forward - Unconditional

- Dial ***#131-1*** to set.
- Dial ***#131-0*** to cancel.
- Dial ***#135-Destination Number-End-of-Dialing*** to configure destination number.

  Where, Destination number can be of maximum 24 digits.

  End of Dialing may be # or * as configured in the system.

Call Forward - Busy

- Dial ***#132-1*** to set.
- Dial ***#132-0*** to cancel.
- Dial ***#136-Destination Number-End-of-Dialing*** to configure destination number.

  Where, Destination number can be of maximum 24 digits.

  End of Dialing may be # or * as configured in the system.

Call Forward - No Reply

- Dial ***#133-1*** to set.
- Dial ***#133-0*** to cancel.
- Dial ***#137-Destination Number-End-of-Dialing*** to configure destination number.

  Where, Destination number can be of maximum 24 digits.

  End of Dialing may be # or * as configured in the system.
- Dial ***#139-Timer*** to set the Call Forward No Reply Ring Timer.

  Where, Timer value is from 1 to 99 seconds.

  Default: 15 seconds.
- Replace the handset.

Conference

Three-party Conference, also referred to as Three-Way Calling, is a telephone call, in which you can have two other persons participate in the call.

The maximum number of three-party conferences supported by SARVAM UMG is 4.

You can initiate a conference by calling the first person, and then put the first person on hold to call the second person. You can also include another person when you in the middle of speech with a person.

The parties to a conference can be:

- Two outgoing calls
- Two incoming calls
- One outgoing call and one incoming call.

- *A 3-Party Conference can be converted to Call Toggle by dialing Call Toggle (Call Split) access code **Flash-#2**.User will be connected to one of the parties and the other party goes on hold.*
- *During conference, if user dials Flash, user will be in speech with last held party and the other party will go on hold.*
- *Conference is not allowed to the FXS Port, which is already in conference.*
- *Conference can be initiated only from the FXS Ports. To add a SIP party to a conference the PCM Companding Type selected on the Region page and the Vocoder selected on SIP should be same.*

Configuring Conference

To use this feature, **Conference** must be enabled in the **Class of Service** of the FXS Port. For instructions, see "FXS Port" under *Basic Settings*.

How to use Conference

- Lift handset.
- Dial the number of party A.
- When A answers the call, dial ***Flash***.
- Party A is put on hold, and you will hear feature tone.
- Now, dial the number of party B.
- When B answers the call and you are in speech with B, dial ***Flash-#8***.
- Three-way speech will be established between you, party A and party B.
- Replace the handset at the end of your conversation.

- The conference will be terminated.

Call Waiting

When your phone is busy, the Call Waiting feature notifies you about another incoming call in the form of beeps.

The Call Waiting feature of SARVAM UMG allows you to:

- reject the waiting call.
- ignore the waiting call.
- hold the current call and answer the waiting call.
- disconnect the current call and answer the waiting call.

To use any of these options, you may dial the respective Access Code.

- *Call Waiting feature has priority over Call Forward-When Busy in call mature state i.e. when the FXS Port is in speech with another port.*
- *If your SARVAM UMG has a 3G module, disable Call Waiting in the SIM Card before inserting it in the Mobile Port to prevent current calls from being disconnected.*
- *Call Waiting feature does not apply:*
  - *If Call Waiting feature is disabled.*
  - *If Waiting Call is ignored.*
  - *If already one Waiting Call is present.*
  - *In the Programming Mode.*
  - *In Conference.*
  - *In Remote Held condition.*
  - *In Dial state, Routing state and in Disconnect state.*

Configuring Call Waiting

To receive Call Waiting Beeps as waiting call indication, **Call Waiting** must be enabled in the **Class of Service** of the FXS Port. For instructions, see “FXS Port” under *Basic Settings*.

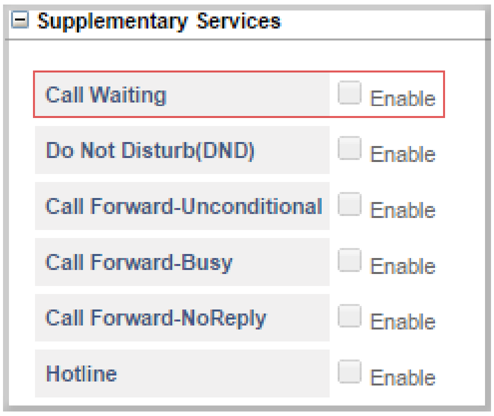
How to use Call Waiting

You can set or cancel Call Waiting from Jeeves or from the phone connected to the FXS Port.

To set or cancel Call Waiting from Jeeves,

- Click the **Basic Settings** link to expand.
- Click the **FXS Port** link.
- Click the FXS Number tab on which you want to use this feature. The page of the selected port opens.

- On the FXS Port page, click **Supplementary Services**.



- To set Call Waiting, select the **Call Waiting Enable** check box. Default: Disabled.

  To cancel, clear the check box.

- Click **Submit** to save.

- Log out of Jeeves.

To set or cancel Call Waiting from the Phone,

- Lift handset of your phone.
- Dial ***#16-1*** to set.
- Dial ***#16-0*** to cancel.
- Replace handset.

When Call Waiting is enabled and you are in speech with Party A, you will hear call waiting beeps indicating another incoming call from Party B. You may Reject, Ignore or Accept the Waiting Call by dialing the relevant Access Code.

- To Reject the Waiting Call and Speech with Current Call, dial **Flash-#31**.

  The beeps will stop, you will remain in speech with Party A.

  OR

- To Ignore the Waiting Call and Speech with Current Call, dial **Flash-#32**.

  The beeps will continue, you will remain in speech with Party A.

  OR

- To Accept the Waiting Call and Hold Current Call, dial **Flash-#33**.

  The beeps will stop, Party A will be put on hold and you will be in speech with Party B.

OR

- To Accept the Waiting Call and Release Current Call, dial **Flash-#34**.

  The beeps will stop, Party A will be disconnected and you will be in speech with Party B.

Message Wait Indication on SIP Trunks

SARVAM UMG supports Message Wait Indication (MWI) on SIP Trunks for Voicemail service subscribed from ITSPs.

If you have subscribed Voicemail service from the ITSP of a SIP Trunk, you can subscribe Message Wait Indication for that SIP Trunk to get notification for new messages in your mailbox on the phone connected to an FXS Port.

You can retrieve messages from the phone connected to the FXS Port by dialing an access code. See “How to Retrieve Messages” at the end of this topic.

You can also view the status of MWI on the SIP Trunks Status page on Jeeves. See “How to view Status of Message Wait Indication” later in this topic.

To be able to use Message Wait Indication (MWI) for the voicemail service of the ITSP, you must configure the following parameters on the SIP Trunk and FXS Port:

On the SIP Trunk,

- Enable **Subscribe for MWI** check box.
- Configure the **Message Retrieval Number** provided to you by your ITSP. This number is used for retrieval of voicemail on the SIP Trunk. The Message Retrieval Number may consist of a maximum of 24 characters. Valid range is 0 to 9, * and # are allowed. Default: Blank.
- Configure the **Authentication ID** and **Authentication Password** provided by your ITSP. Default: Blank.
- In **Send Message Notification on**, select the FXS Port on which Message Wait Indication is to be sent whenever there is a new message on the SIP Trunk. Default: FXS Port 1.
- Assign the FXS Port on which you want to receive MWI notification from the SIP Trunk, as the destination port to **Send MWI Notification on**. Whenever a new message arrives in the mailbox of the SIP Trunk, SARVAM UMG gives notification to the FXS Port you have selected as destination, according to the type of Message Wait Notification you have selected on the FXS Port.
- If you have completed the configuration of SIP Trunk-1, click **Submit** to save settings.

On the FXS Port selected as Destination for MWI,

- If you have subscribed to *Message Wait Indication* for the voicemail service from your ITSP, and have selected this FXS Port as the destination for receiving Message Wait Indication[25], you may select the desired type of **Message Wait Notification** from the following options.
    - Select **Stuttered Dial Tone**, if you want new message indication in the form of a stuttered dial tone, whenever the user picks up the phone connected to the FXS Port.
    - Select **LED Lamp (HV),** if the phone connected to the FXS Port is equipped with a 'Message Wait' lamp and you want new messages to be indicated on this LED lamp using High Voltage.

25. *You have selected the number of this FXS Port for the* ***Send Message Notification on*** *parameter, under* ***MWI Parameters*** *you configured on the SIP Trunk.*

- Select **Ring**, if you want the arrival of a new message to be indicated by the *Message Wait Ring* (a Short, Fast ring).

  You can select a different Ring Type to indicate message wait. For instructions, see "Message Wait Ring Type".

  You can also set the duration for which the ring is to be played (Ring Timer), the number of times the ring is to be played (Ring Count) and the interval between rings (Ring Interval). For instructions, see "Message Wait" in "System Parameters".

- Select **LED Lamp (FSK)**, if the phone connected to the FXS Port is equipped with a 'Message Wait' lamp and you want new messages to be indicated on this LED lamp using FSK CLI.

- Select **Stuttered Dial Tone + LED Lamp (HV),** if you want new message indication on the LED Lamp using High Voltage as well as in the form of a stuttered dial tone, when the user picks up the phone connected to the FXS Port.

- Select **Stuttered Dial Tone + LED Lamp (FSK)**, if you want new message indication on the LED Lamp using FSK CLI as well as in the form of a stuttered dial tone, when the user picks up the phone connected to the FXS Port.

Default: LED Lamp (HV)

Whenever a new message arrives in the Mailbox of the SIP Trunk, SARVAM UMG gives notification to this (destination) FXS Port according to the type of *Message Wait Notification* you select.

How to view Status of Message Wait Indication

You can view the status of Message Wait Indication, that is, new messages, old messages, urgent old and new messages on the SIP Trunk Status page of Jeeves. For more information, in *Status,* under "SIP Trunk", see "MWI Status".

How to Retrieve Messages

You can retrieve messages by dialing the access code for *Using Voice Mail of the Service Provider* **#7** (default) from the phone connected to the FXS port designated as destination for MWI. This access code is configurable, for instructions, see "Access Code".

To retrieve messages:

- Dial the Using Voice Mail of the Service Provider Access Code, **#7** from the phone connected to SARVAM UMG.
- SARVAM UMG checks whether the FXS Port to which the phone is connected is configured as the destination for Message Wait Notification (**Send MWI Notification on**) for any SIP Trunk.
- On finding the FXS Port as the destination for Message Wait Notification for the SIP Trunk, SARVAM UMG dials out the Message Wait Retrieval Number configured for the SIP Trunk.
- The FXS Port gets connected to the Voicemail server of the ITSP.
- You can follow the voice mail prompts to retrieve your messages.

*If Message Wait Notification is enabled on more than one SIP Trunk and you have configured same FXS Port in '**Send MWI Notification on**' field, you are recommended not to use Access Code to retrieve the Voice Mail messages. Instead, do the following:*

- *Configure the Destination Number Based table. In the Destination Number Based table, enter the **Message Retrieval Number** as **Destination Number** and select the respective SIP Trunk port as **Routing Group**.*

- *Retrieve your messages, by dialing the Message Retrieval Number of that SIP Trunk from the FXS Port.*

- *You will be connected to the Voice Mail server of the ITSP of that SIP Trunk.*

Call Pick-up

Call Pick-up is a feature that allows you to answer the call on someone else's phone, provided all the phones are connected to SARVAM UMG. You can answer the other phones without physically going over to the ringing phone.The feature can be accessed by dialing the access code for Call Pick-up (default: **#5)**.

SARVAM UMG stores the FXS Port number from where the call was answered as the *Destination Port* in the Call Detail Record (CDR). To know more, see "Call Detail Records (CDR)".

You cannot pick up RCOC calls ringing on extensions in the same Call Pick-up group.

For this feature to work,

- You must create a Call Pick-up group, consisting of the FXS Ports.

  Calls ringing on a phone connected to an FXS Port within a group can be picked up from another phone in the same group by dialing the Call Pick-up access code (programmable; default: **#5**).

- You must enable **Call Pick-up** in the **Class of Service** of all the FXS Ports included in the Call Pick-up group.

Configuring Call Pick-up

To use this feature,

- assign the FXS Ports to a Call Pick-up Group. For instructions, see "General" under *Basic Settings*.

- enable **Call Pick-up** in the **Class of Service** of the FXS Ports you have assigned to Call Pick-up groups. For instructions, see "FXS Port" under *Basic Settings*.

You can also change the default access code for Call Pick-up. For instructions, see "Access Code" under *Advanced Settings*.

How to use Call Pick-up

To pick up calls ringing on the phones in your Call Pick-up group,

- Lift the handset of your phone.

- Dial **#5**

- You are in speech with the caller.

- Talk.

- Replace the handset to disconnect.

Do Not Disturb (DND)

If you do not want to receive any calls on your phone, you may use Do Not Disturb (DND). When you set DND, all incoming calls on your phone will be rejected, but you can make outgoing calls.

If both, Do Not Disturb (DND) and Call Forward-Unconditional are set, Call Forward-Unconditional will have priority over Do Not Disturb (DND).

Do Not Disturb (DND) has priority over Call Forward-No Reply and Call Forward- Busy.

Configuring DND

To use this feature, **Do Not Disturb** must be enabled in the **Class of Service** of the FXS Port. For instructions, see “FXS Port” under *Basic Settings*.

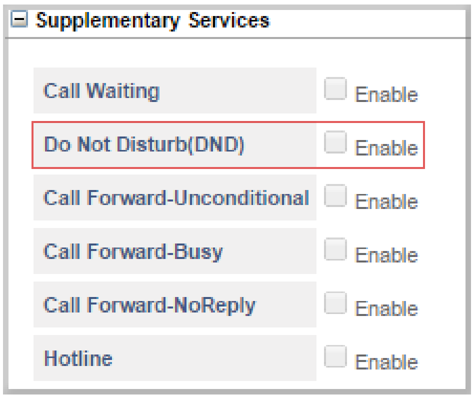
How to use DND

You can set or cancel DND from Jeeves or from the phone connected to the FXS Port.

To set or cancel DND from Jeeves,

- Click the **Basic Settings** link to expand.
- Click the **FXS Port** link.
- Click the FXS Number tab, on which you want to use this feature.

  The page of the selected port opens.
- On the FXS Port page, click **Supplementary Services** link to expand.



- To set DND, select the **Do Not Disturb (DND) Enable** check box. Default: Disabled.
- Click **Submit** to save.

To set or cancel DND from the Phone,

- Lift handset of your phone.

- Dial ***#18-1*** to set.
- Dial ***#18-0*** to cancel.
- Replace handset.

Hotline

The Hotline feature connects the FXS Port user immediately to a particular number, when the user goes OFF-hook. Users can use Hotline to dial the number they call most frequently.

Let us understand how this feature works with an example,

- A is Sales Manager who frequently dials the number of the B, the Sales Coordinator.
- Instead of dialing B's number each time, A sets Hotline.
- A configures B's number as Hotline Number
- A can also set the Hotline Timer, i.e. the time after which B's number should be dialed out. By default, the Hotline Timer is set to 5 seconds.
- Whenever A goes Off-hook, SARVAM UMG plays feature tone and waits for the duration of the Hotline Timer.
- If A does not dial any digit during this Timer, on the expiry of the Timer, SARVAM UMG dials out B's number.

Allowed-denied number logic is applied for the hotline number.

Configuring Hotline

To use this feature, **Hotline** must be enabled in the **Class of Service** of the FXS Port. For instructions, see "FXS Port" under *Basic Settings*.

How to use Hotline

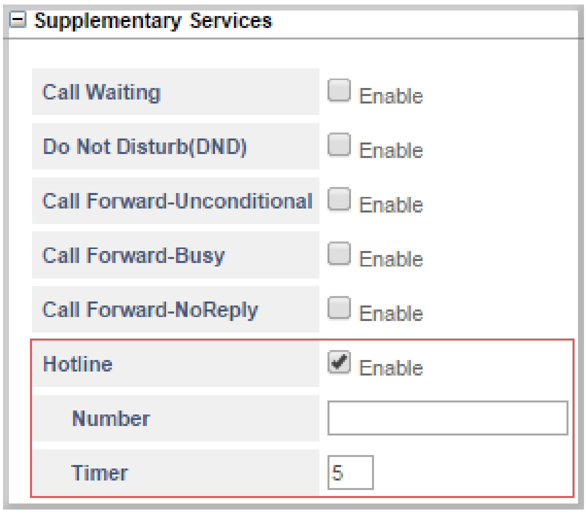
You can set or cancel Hotline from Jeeves or from the phone connected to the FXS Port.

To set or cancel Hotline from Jeeves,

- Click the **Basic Settings** link to expand.
- Click the **FXS Port** link.
- Click the FXS Port Number on which you want to use this feature.

  The selected port number window opens.
- On the FXS Port window, click **Supplementary Services** link to expand.

- Select the **Hotline Enable** check box. Default: Disabled.



- In **Number**, enter the Number to be dialed out from the port when you go Off-hook. Default: Blank.

- Set the duration of the **Timer** after which the Hotline Number is to be dialed out after you go Off-hook. The range of this timer is 1 to 9 seconds. Default: 5 seconds.

To set or cancel Hotline from the Phone,

- Lift handset of your phone.

- Dial ***#151-1*** to set.

- Dial ***#151-0*** to cancel.

- Dial ***#152-Destination Number-End-of-Dialing*** to configure Hotline Number.

  Where, Destination Number can be of maximum 24 digits. Digits 0 to 9, *, #, dot (.) are allowed.
  End of Dialing may be # or * as configured in the system.

- Dial ***#153-Timer*** to set the Hotline Timer.

  Where, Timer value is from 1 to 9 seconds. Default: 5 seconds.

- Replace handset.

Supplementary Services of Service Provider

When SARVAM UMG is interfaced with a service provider (ITSP, VoIP-PSTN adapter, the Matrix Eternity IP-PBX, or any other PBX) that supports supplementary services like Call Hold, Call Transfer, Call Waiting that require dialing of Flash[26], you may choose to access the features of the service provider, or to access primarily the features of SARVAM UMG.

If you want to use the features of the service provider, you must select ***Network*** as the *Subscriber Type* for the FXS Port of SARVAM UMG.

If you want to use the features of SARVAM UMG, you must select ***Gateway*** as the Subscriber Type for the FXS Port of SARVAM UMG. However, you will be able to access the features of the service provider also in the Gateway mode, by dialing Flash, followed by the Access Code **#4**, for using Supplementary Services of Service Provider.

When the SARVAM UMG is set in the Gateway mode, when you dial flash and the access code #4 to access the supplementary services of the service provider, the active call will be put on hold and you will get a feature tone. SARVAM UMG will start the an internal timer, called the **Service Provider Access Code Wait Timer**. This timer has a duration of 10 seconds. Any activity performed on the FXS Port within this timer will be sent to the service provider server. Any activity performed on the FXS Port after the expiry of this timer, will be processed by SARVAM UMG.

Configuring Supplementary Services of Service Provider

To use Supplementary Services of Service Provider, configure the parameter **Subscriber Type** on the FXS Port as **Network** or as **Gateway**. For instructions on configuring Subscriber Type, see "FXS Port" under *Basic Settings*.

How to use Supplementary Services of Service Provider

If SARVAM UMG is set in the **Network Mode**,

- Dial **Flash** during speech.

  You will get feature tone played by your service provider.

- Dial feature access code provided by the service providers server.

If SARVAM UMG is set in the **Gateway Mode**,

- Dial **Flash** during speech. This will place the active call on hold and you will get a feature tone.
- Dial ***#4***.

  You will be in Network mode for 10 seconds.

- Dial the feature access code provided by the service providers server within 10 seconds.

26. To be able to use features of the service provider SARVAM UMG supports dialing of Flash during speech on SIP.

Making a New Call using Access Code

This feature enables callers to disconnect the current call and make a new call using SARVAM UMG without getting disconnected from the system.

This feature is useful when you want to allow users to make multiple calls without getting disconnected each time their call ends.

Let us understand this feature with an example:

- A Cyber-Cafe has installed SARVAM UMG for providing international calling service to the home users.
- The Cyber-Cafe provides home users who have subscribed for this service, a number to call the SARVAM UMG, a PIN Number and a Password.
- To make international calls, home users must call SARVAM UMG, dial the PIN Number, Password and the international number. Thus, each time they want to make a call, they must repeat this process of dialing over and over again.
- Making a New Call feature eliminates repeated dialing of these numbers.
- After calling SARVAM UMG, home users can dial their PIN number and Password once and call the international number. At the end of the call, they can dial the Access Code for Making a New Call. They will remain connected to the system and can make another call.
- If the remote end disconnects the call during speech, SARVAM UMG will play error tone for 4 seconds followed by dial tone. Users can make a new call without dialing the feature Access Code.

*This feature is applicable only on the Source Port and only when **After Answering the Call and Collecting the Digits** is selected as the option to Route all Incoming Calls (with CLI).*

Making New Call Access Code dialed by users will be ignored if any other option is selected to Route all Incoming Calls (with CLI).

*However, if you have enabled **Connect Source Port when number is outdialed** on the FXO Port or Mobile Port or have enabled **Connect Source Port when 183 (Session Progress) is received on SIP** on the SIP Trunk, you will be able to provide this feature to users.*

*Making a new Call using Access Code will not be allowed if **Allow Virtual Feature** option is selected as Allow All or As per Virtual User Table in the General settings of the Mobile Port.*

Configuring Making a New Call

To provide this feature to users, you must enable **Allow making New Call using Access code** on the FXO Ports, SIP Trunks, BRI (Terminal) Ports, Mobile Ports and T1E1 ports. For instructions, see “FXO Port”, “SIP Trunk”, “BRI Port - Terminal”, “Mobile Port”, “T1 Port” and “E1 Port” under *Basic Settings*.

How to make a New Call using Access Code

- When you are in speech during the current call.
- Dial **#91.** Current call will disconnect.

- Dial the new number you want to call.
- While in speech, dial **#91** again to make another call.

Disconnecting a Call using Access Code

SARVAM UMG enables users to disconnect a call using an access code. When the Disconnect Call Access Code is dialed, SARVAM UMG releases the port engaged in the call.

*Disconnecting a Call using Access Code will not be allowed if **Allow Virtual Feature** option is selected as Allow All or As per Virtual User Table in the General settings of the Mobile Port.*

Configuring Call Disconnection using Access Code

To provide this feature to users, you must enable **Allow Call Disconnection using Access code** on the FXO Ports, BRI Ports, SIP Trunks, Mobile Ports and T1E1 Ports. For instructions, see “FXO Port”, “BRI Port - Terminal”, “BRI Port - Network”, “SIP Trunk”, “Mobile Port”, “T1 Port” and “E1 Port” under *Basic Settings.*

How to Disconnect a Call using Access Code

- You are in speech or are at the end of the current call.
- Dial **#92**

IP Dialing

SARVAM UMG supports direct dialing of IP Addresses from the source port. To provide IP Dialing facility to the users, you must configure a SIP Trunk or a SIP Group for IP Dialing.

When a number is dialed out from the source port, SARVAM UMG routes the call to the desired destination as per the routing mechanism configured for that port. However, when an IP Address is dialed from the source port of SARVAM UMG, the system does not check the Destination Port Determination method you have configured for that port, instead it routes the dialed IP Address through the SIP Trunk or the SIP Group you configured for IP Dialing.

When dialing an IP Address, users must press * key (star/asterisk) in place of. (dot/period) in the IP Address.

For example, to call the IP Address **192.167.100.1**, users must dial **192*167*100*1** or **192*167*100*001**

SARVAM UMG interprets the * dialed as a '.' (dot / period).

Configuring IP Dialing

To provide this feature to users,

- you need to select a SIP Trunk or a SIP Group through which the dialed IP Addresses are to be routed.

  If you want to use a SIP Trunk group for IP Dialing, you must configure a SIP Group first. This Group is common for all port types. For instructions, see “Group” under *Advanced Settings*.

  When you assign a SIP Trunk, make sure it is enabled and has the necessary configuration done. For instructions, see “SIP Trunk” under *Basic Settings*.

- assign the SIP Trunk you want or the SIP Group you configured to **SIP Trunk for IP Dialing** in the System Parameters. For instructions, see “System Parameters” under *Advanced Settings*. By default, SIP Group 1 is selected for IP Dialing in the System Parameters.

If the SIP Trunk Group for IP Dialing is programmed as 'None', SARVAM UMG will give error tone to the caller; the call will be rejected.

CHAPTER 11

Virtual User Features

SARVAM UMG offers Virtual Users the following telephony features on the Mobile Port, which they can access by dialing the Virtual Access Codes.

- Call Hold
- Making a Second Call
- Call Toggle
- Call Transfer - Attended and Blind

You can change the default Virtual Access Codes assigned to the respective features as per your requirement. To know more, refer to "Virtual Access Code".

Virtual User Features is not supported on FXS Port.

Pre-requisites

To use the Virtual User features, make sure

- Virtual User License is activated. Refer to "Virtual User" in License Management.
- **Allow Virtual Feature** option is configured as Allow All or As per Virtual User Table in *General* settings of the "Mobile Port".
- **Virtual User Table** is configured if you have selected *As per Virtual User Table* as the **Allow Virtual Feature** option. For more details, refer to "Configuring Virtual User Table".

Call Hold

Call Hold enables you to put an on-going conversation on hold and call another person, or receive a call from another person. You can retrieve the call you put on hold after the conversation with the other party has ended. You can also retrieve the call you put on hold in the middle of the conversation with the other party.

Call Hold is also used in the following features:

- Retrieve Held Call
- Make a Second Call
- Call Toggle
- Call Transfer - Attended and Blind

Configuring Call Hold

To use this feature, make sure the "Pre-requisites" are fulfilled.

How to use Call Hold

- You are in speech with party A.
- To put your call with A on hold, dial ***1**
- A is put on hold. You get dial tone for 7 seconds, followed by error tone.
- To retrieve the call you put on hold, dial ***1** again during dial tone or during error tone.
- You will be in speech with A.

- *If you disconnect the call during dial tone, your call with party A will be also disconnected.*
- *If you disconnect during error tone, you will get ring back. When answered, call will be disconnected.*

Making a Second Call

You can make a second call, by putting the current call on hold.

Configuring Making a Second Call

To use this feature, make sure the “Pre-requisites” are fulfilled.

How to make a Second Call

- You are in speech with party A.
- You want to talk to party B.
- Dial ***1** to put party A on Hold.
- You get dial tone.
- Dial the number of the desired party B.
- When party B answers the call, you are in speech with party B.

- *Making a second call feature can also be used with other features such as Call Transfer-Attended and Call Toggle.*
- *For example, after making a second call, you can toggle between the first and the second call using the Call Toggle Virtual access code.*

Call Toggle

Call Toggle (Call Split) allows you to have two simultaneous telephone conversations, talking to two people alternately.

The parties for Call Toggle can be:

- Two outgoing calls
- Two incoming calls
- One outgoing call and one incoming call.

You must dial the Call Toggle Virtual Access Code to switch between the held call and the active call.

Configuring Call Toggle

To use this feature, make sure the "Pre-requisites" are fulfilled.

How to Toggle between calls

- You are in speech with A and you want to talk to B.
- Dial ***1,** A is put on hold. You get dial tone.
- Dial the number of B. When B answers the call, you are in speech with B.
- To talk to A, dial ***2***.
- You are in speech with A and B is put on hold.
- To talk to B, dial ***2** again.
- You are in speech with B. A is put on hold.
- This way, you can talk alternately to A and B, by dialing ***2** again.

Call Transfer-Attendant

Attendant Call Transfer is when you transfer the call to the desired party after consulting the party and or obtaining their consent for transfer.

This is how Attendant Transfer works:

- A (transferor) is in speech with B. A wants to transfer B's (transferee) call to C (transfer target).
- A dials *1 (Call Hold Virtual access code) to put B on hold and then dials C's number.
- A disconnects while the C's number is ringing or after speech with C.
- B is in speech with C.

Configuring Attendant Transfer

To use this feature, make sure the “Pre-requisites” are fulfilled.

How to use Attendant Transfer

- You are in speech with party A.
- Party A wishes to speak to party B.
- Dial ***1*** to put party A on Hold.
- Dial Party B’s Number.
- Party B answers the call. You are in speech with party B.
- Dial ***4*** or disconnect to transfer the call.
- You are disconnected. Speech is established between party A and party B.

Call Transfer - Blind

Blind Transfer is when you transfer the call to the desired party, without informing the party of the transfer.

This is how Blind Call Transfer works:

- A (transferor) is in speech with B (transferee). A wants to transfer B's (transferee) call to C (transfer target), without informing C.
- A dials Blind Transfer Virtual Access Code. The system puts B on hold and plays dial tone to A.
- A dials the number of C. System disconnects A and B's call is transferred to C.

Configuring Blind Transfer

To use this feature, make sure the “Pre-requisites” are fulfilled.

How to use Blind Transfer

- You are in speech with party A.
- Party A wishes to speak to party B.
- Dial ***3.**
- System puts party A on hold.
- Dial the number of party B.
- On successful transfer, you will get confirmation tone.
- Party A will get connected with party B.

CHAPTER 12 **Maintenance**

Firmware Upgrade

You can upgrade Firmware of SARVAM UMG:

1. From a Provisioning Server
2. From a Personal Computer

Old systems having V1R3 firmware can be upgraded to V1R4. in this case the Expansion Slot License will be applicable. For details refer, *"Expansion Slots" under "License Management".*

System purchased with V1R5 firmware cannot be downgraded to earlier versions.

If a system purchased with firmware V1R5.5 or earlier is upgraded to V1R5.6 or later, then default the system to configure according to the resources available in the latest firmware.

Firmware Upgrade from Provisioning Server

Auto Firmware Upgrade

Using Auto Firmware Upgrade, SARVAM UMG can automatically upgrade its firmware by downloading the firmware files stored at a central location: HTTP Server or HTTPS Server or Provisioning Server.

This feature is useful for ITSPs that have Provisioning Servers to store the firmware files. ITSPs can update the firmware of SARVAM UMG provided to their customers from a centralized location without physically visiting the customer premises.

*For the **Auto Firmware Upgrade File** contact Matrix Support Team.*

To perform Auto Firmware Upgrade,

1. ITSPs must store the following Auto Firmware Upgrade files of SARVAM UMG on the Provisioning Server.
   - matrix_firmware.html file
   - SARVAM UMG_VwRx.y.z.Zip file

2. The following parameters must be configured in the SARVAM UMG.
   - IP Address of the Provisioning Server.
   - Path of the Folder (containing the firmware files) on the Provisioning Server.

- The protocol to be used: HTTP, HTTPS.

3. When SARVAM UMG installed at a customer site gets connected to the ITSP network, it will automatically compare its current firmware with the firmware files stored on the Provisioning Server.

   The matrix_firmware.html file helps SARVAM UMG decide which firmware it should upgrade to.

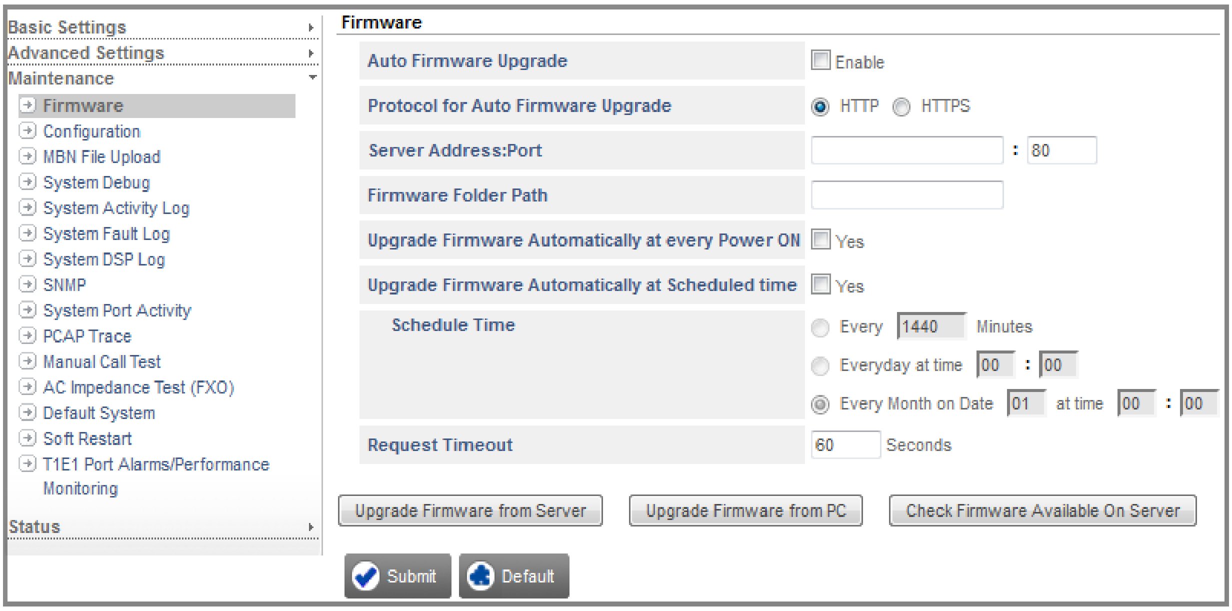
4. After SARVAM UMG decides the Firmware Version/Revision to upgrade to, it will send the request for the firmware files to the Provisioning Server. Once the respective firmware files are received, SARVAM UMG will upgrade its current firmware with the new firmware without the intervention or assistance of a technician.

The table below describes a few possible cases and the corresponding action taken by SARVAM UMG.

Version-Revision of your SARVAM UMG	Version- Revision in the matrix_firmware.html file received from the Provisioning Server	Action Taken by SARVAM UMG
V1R5.3.0	V1R4.1.0	SARVAM UMG will not downgrade its current firmware with V1R4.1.0
	V1R5.3.0	SARVAM UMG will discard the upgrade process as same Version/Revision is found.
	V1R4.1.0 and V1R5.3.1	SARVAM UMG will upgrade its current firmware with V1R5.3.1

To configure Auto Firmware Upgrade parameters,

- Click the **Maintenance** link to expand.
- Click the **Firmware** link.



- Select the **Auto Firmware Upgrade** check box. Default: Disabled.

- Select the **Protocol for Auto Firmware Upgrade** to be used by the Provisioning Server to upgrade the firmware of SARVAM UMG. SARVAM UMG generates file transfer request to the server according to the protocol you select. You may select **HTTP** or **HTTPS**. Default: HTTP.

- In **Server Address: Port,** enter the IP Address/Domain and Port of the Provisioning Server on which the firmware files of SARVAM UMG are stored.

  The Provisioning Server Address can also be obtained by SARVAM UMG using DHCP (using Option 224). To fetch Provisioning Server Address using DHCP, keep the Server Address: Port field blank.

  Make sure that you also set the *Connection Type* on the “Network” page as *DHCP*.

  The default Port differs as per the protocol you select. For HTTP, the Default Port is 80 and for HTTPS, the Default Port is 443. You can also change the port as per your requirement. Valid Port Range: 80/ 443/ 1031 to 65534.

- In **Firmware Folder Path,** specify the path of the folder on the Provisioning Server where the firmware files are stored. Default: Blank.

- Enable the **Upgrade Firmware Automatically at Every Power ON** check box, if you want SARVAM UMG to check for updates in the firmware at each power ON.

At Power ON, if both Auto-Firmware upgrade and Auto-Configuration upgrade are enabled, Auto-Firmware upgrade has priority over Auto-Configuration upgrade.

*While upgrading, if SARVAM UMG has to upgrade itself with the benchmark firmware first then it is recommended that you select **Upgrade Firmware Automatically at Every Power ON.***

- Enable the **Upgrade Firmware Automatically at Scheduled Time** check box, if you want SARVAM UMG to check for updates in the firmware at a scheduled time. You may select any one of the following schedule options:

  - In **Every XX minutes**, enter the minutes after which you want SARVAM UMG to check for firmware updates.

  - In **Everyday at HH:MM**, enter the time in **Hours(00-23)** and **Minutes(00-59)** after which you want SARVAM UMG to check for firmware updates everyday.

  - In **Every Month on DD at HH:MM**, enter the **Date** (01-31) and **Time** in Hours (00-23) and Minutes(00-59) after which you want SARVAM UMG to check for firmware updates every month.

*If SARVAM UMG has to upgrade itself with the benchmark firmware and you have selected **Upgrade Firmware Automatically at Scheduled Time**, SARVAM UMG will first upgrade itself with the benchmark firmware. At the subsequent scheduled time, it will upgrade itself with the final firmware.*

- **Request Timeout** is used when SARVAM UMG tries to connect to the Provisioning Server for TCP/ TLS binding. This timer specifies for how long SARVAM UMG should wait for successful TCP/TLS binding.

  Enter the required time in seconds. The range of Request Timeout is 01-99 seconds. Default: 60 seconds.

If SARVAM UMG fails to connect to the Provisioning Server, it will make 10 attempts at a regular interval of 10 seconds each to establish the binding. Even then, if it is unable to establish the binding, it will abort the Auto upgrade process.

- Click **Submit** to save.
- To view the status of Auto Firmware Upgrade from Jeeves, see “Firmware” under “Status”.

Manual Firmware Upgrade

You can manually upgrade Firmware of SARVAM UMG, whenever you want.

To manually upgrade firmware of SARVAM UMG from server,

- Click the **Upgrade Firmware from Server** button on the Firmware page. SARVAM UMG will automatically upgrade its firmware with the latest firmware available on the server.

Checking Firmware Availability

You can check the firmware files available on the server and then decide whether you want to upgrade SARVAM UMG.

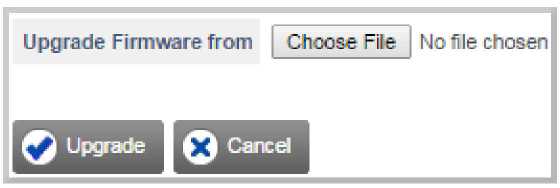
Before upgrading Firmware from server, you can also choose the firmware with which you want to upgrade your SARVAM UMG.

- To view the firmware files available on the Server, click the **Check Firmware Available On Server** button.
- A list of Firmware files available on the server appears in a new window.
- If you want to upgrade SARVAM UMG with the desired Firmware, select the Firmware and click the **Submit** button.
- SARVAM UMG will upgrade itself with the firmware you select.

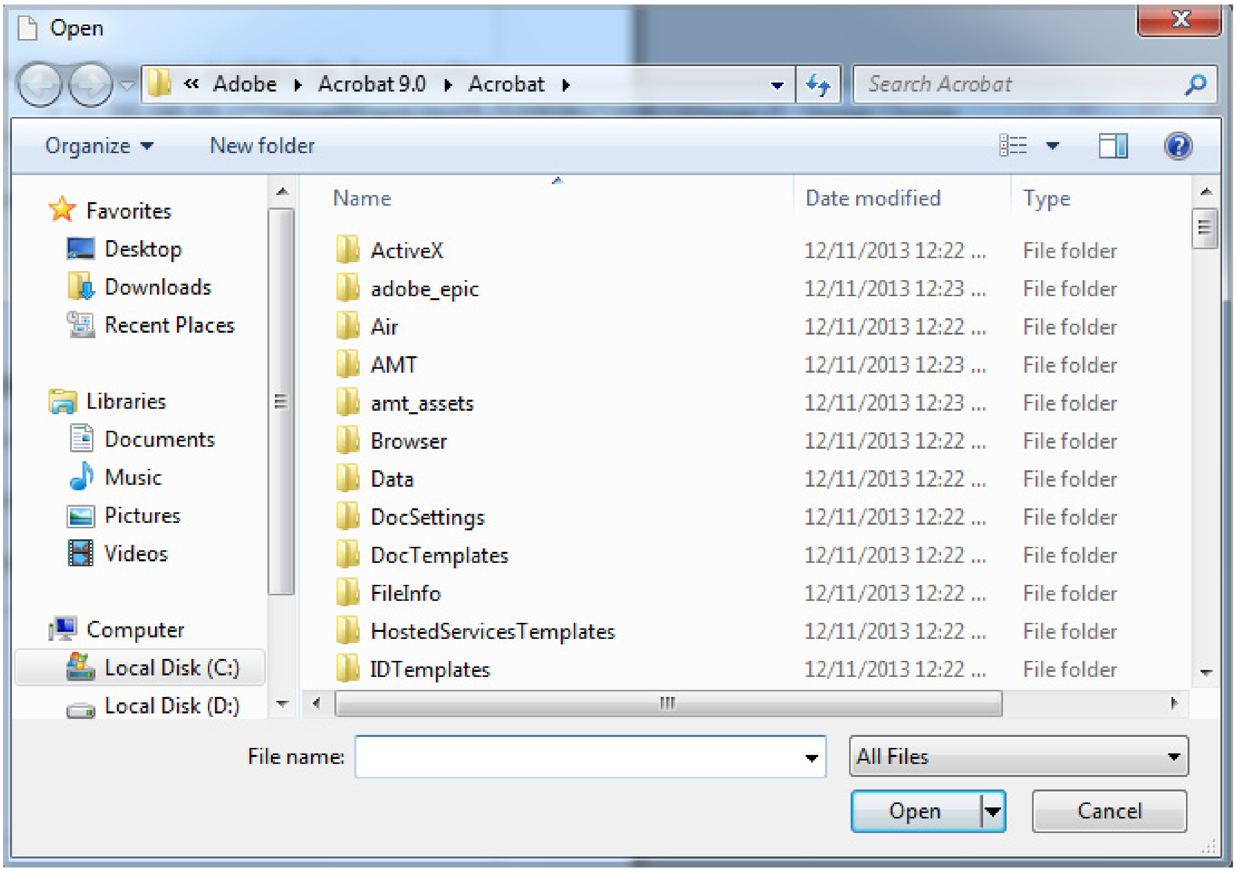
Firmware Upgrade from Personal Computer

You can also upgrade firmware of SARVAM UMG with the firmware files stored on your computer. To do so,

- Click the **Upgrade Firmware from PC** button. A new window - **Firmware Upgrade From** opens.
    - Click the **Choose File** button to reach the location on the local disk on which the firmware files are stored.



- Select the required firmware files from the location on the local disk.



- The path to the file will appear in the **Firmware Upgrade From** box. Click the **Upgrade** button.

Configuration Upgrade

You can upgrade Configuration of SARVAM UMG:

1. From the Auto Configuration Server
2. From a Personal Computer

Upgrading Configuration from the Auto Configuration Server

Auto Configuration Upgrade

Using Auto-Configuration, SARVAM UMG can automatically download the configuration files stored at a central location: Auto Configuration Server (ACS).

This feature is useful for ITSPs that have deployed a large number of SARVAM UMG. ITSPs can store the configuration files of each SARVAM UMG that they have provided to their customers on the Auto Configuration Server (ACS).

*For the **Auto Configuration File** contact Matrix Support Team.*

To perform Auto Configuration,

1. Make sure that the configuration file of SARVAM UMG is stored on the Auto-Configuration Server (ACS).

2. To ensure security, ITSP can encrypt the configuration file stored on the ACS. If the ITSP has encrypted the configuration file, the password to decrypt the file must be provided to you.

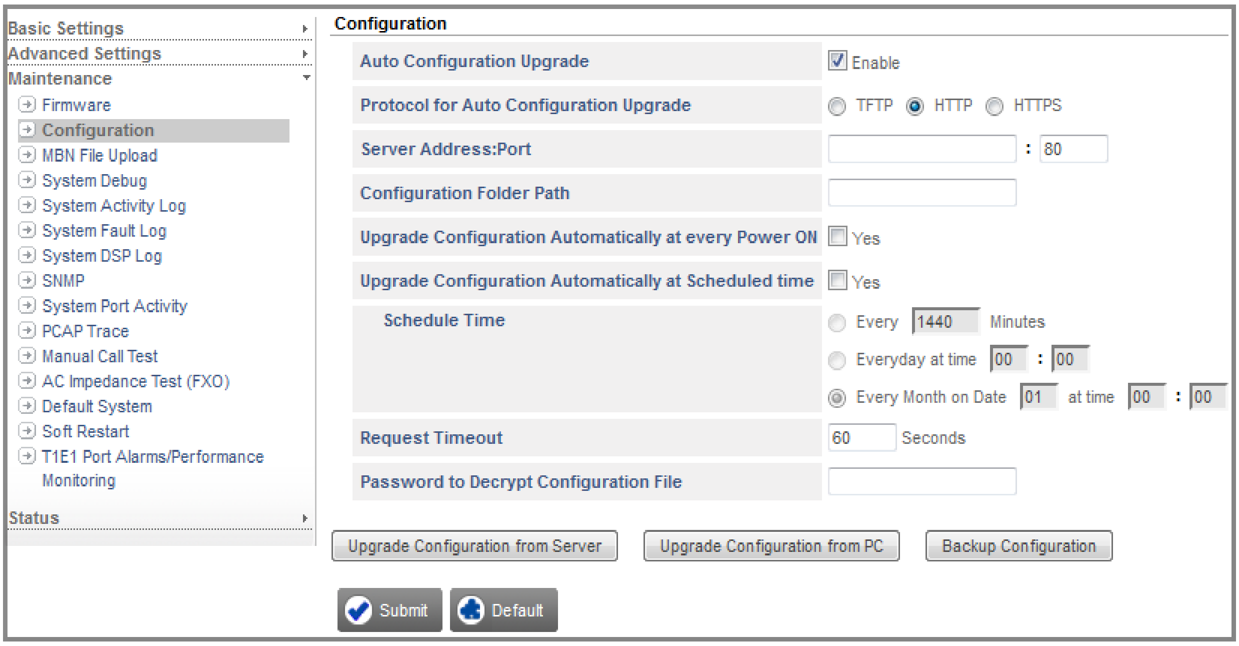
3. The following parameters must be configured in the SARVAM UMG.
   - IP Address of the Auto Configuration Server (ACS).
   - Path of the Folder (containing the configuration file) on the Auto Configuration Server.
   - Password to decrypt the configuration file (if encryption is used).
   - The protocol to be used: TFTP, HTTP, HTTPS.

4. When SARVAM UMG installed at a customer site connects to the ITSP network, it will automatically download its configuration file stored on the Auto-Configuration Server (ACS), without the intervention or assistance of a technician.

To configure Auto Configuration parameters,

- Click the **Maintenance** link to expand.

- Click the **Configuration** link.



- By default, **Auto Configuration Upgrade** check box is enabled. You may clear this check box, if required.

- In **Protocol for Auto Configuration Upgrade**, select the protocol used by the Auto Configuration Server to upgrade the configuration. SARVAM UMG generates file transfer request to the Auto-Configuration Server according to the protocol you select. You may select **TFTP**, **HTTP** or **HTTPS**. Default: HTTP.

- In **Server Address: Port**, enter the IP Address/Domain and the Port of the Auto Configuration Server on which the configuration files of SARVAM UMG are stored.

  The Auto Configuration Server Address can also be obtained by SARVAM UMG using DHCP (using Option 224). To fetch Auto Configuration Server Address using DHCP, keep the Server Address: Port field blank.

  Make sure that you also set the *Connection Type* on the “Network” page as *DHCP*.

  The default Port differs as per the protocol you select. For TFTP, the Default Port is 69. For HTTP, the Default Port is 80. For HTTPS, the Default Port is 443. You can change the port as per your requirement. Valid range is 69/ 80/ 443/ 1031 to 65534.

- In **Configuration Folder Path,** specify the path of the folder on the Auto Configuration Server where the configuration files are stored. Default: Blank.

- Enable **Upgrade Configuration Automatically at Every Power ON** check box, if you want SARVAM UMG to check for updates in the configuration file at each Power ON.

At Power ON, if both Auto Firmware upgrade and Auto Configuration upgrade are enabled, Auto Firmware upgrade has priority over Auto Configuration upgrade.

- Enable **Upgrade Configuration Automatically at Scheduled Time** check box, if you want SARVAM UMG to check for updates in the configuration at a scheduled time. You may select any one of the following schedule options:

  - In **Every XX minutes**, enter the minutes after which you want SARVAM UMG to check for configuration updates.

  - In **Everyday at HH:MM**, enter the time in **Hours(00-23)** and **Minutes(00-59)** after which you want SARVAM UMG to check for configuration updates everyday.

  - In **Every Month on DD at HH:MM**, enter the **Date** (01-31) and **Time** in Hours (00-23) and Minutes(00-59) after which you want SARVAM UMG to check for configuration updates every month.

- **Request Timeout** is the time for which SARVAM UMG will try to connect to the Auto Configuration Server for TCP/TLS binding using HTTP or HTTPS. This timer specifies for how long SARVAM UMG should wait for successful TCP/TLS binding.

  Enter the required time in seconds. The range of Request Timeout is 01-99 seconds. Default: 60 seconds.

  If SARVAM UMG fails to connect to the Auto Configuration Server, it will make 10 attempts at a regular interval of 10 seconds to establish the binding. Even then, if it is unable to establish the binding, it will stop retrying and wait for next event of Auto Configuration upgrade.

- In **Password to Decrypt Configuration File**, enter the password as provided by your ITSP to decrypt the configuration file. During Auto Configuration, if SARVAM UMG receives an encrypted configuration file, it will decrypt the file using this password.

  The password may consist of 40 characters (maximum). Default: Blank.

The password is case-sensitive, make sure you enter the password in the same format as given to you by your ITSP.

- Click **Submit** to save.

- To view the status of Auto Configuration upgrade from Jeeves, see “Configuration” under “Status”.

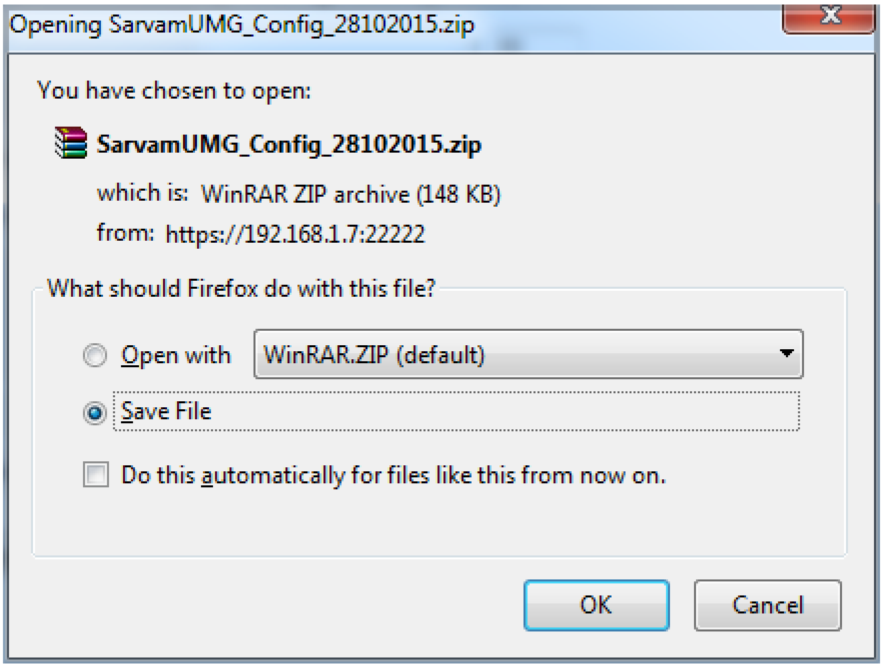
Manual Configuration Upgrade

To manually upgrade configuration of SARVAM UMG, click the **Upgrade Configuration from Server** button.

Backup Configuration

- To save the existing configuration files as backup, click the **Backup Configuration** button.

The **SarvamUMG_Config_ddmmyyyy.zip** window will open; where ddmmyyyy signifies the current date.



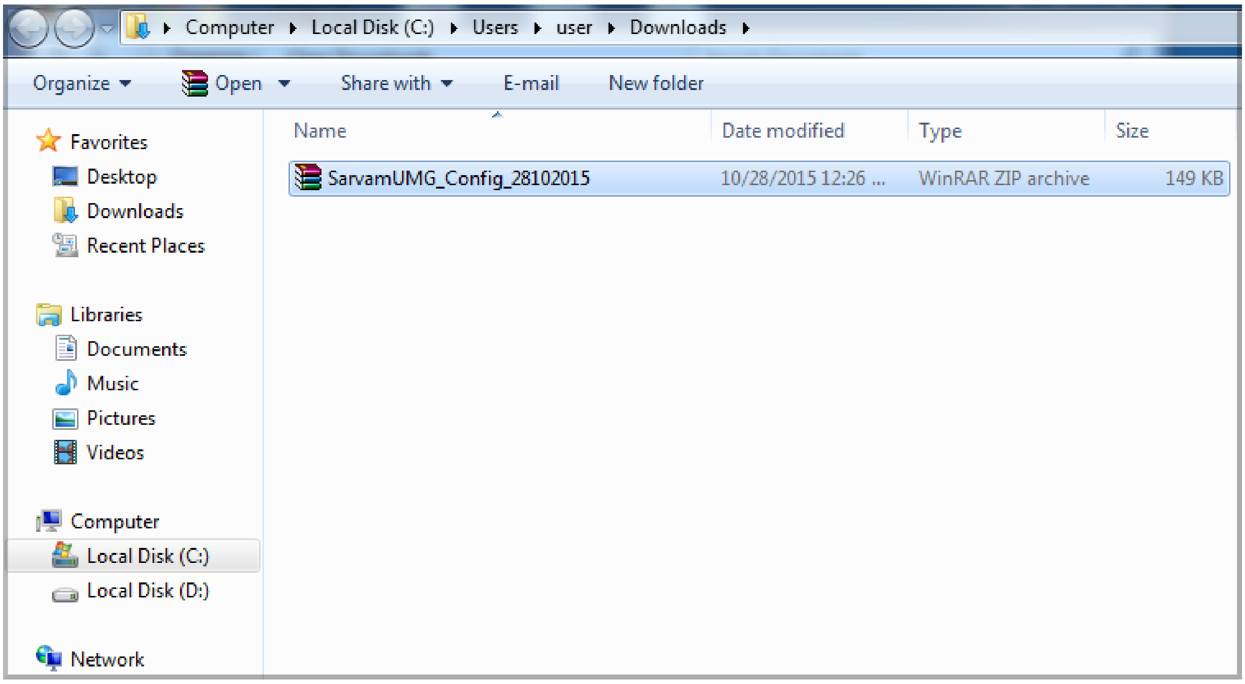
- You can either open the zip file or save the file to a location.

*The above window display depends upon the browser you are using. Check the **Download Settings** of your browser and set the Download path accordingly.*
OR
If your browser does not ask you for the location you want to save your file, it saves it in the default location according to the download path specified for that browser.

*If you are using Mozilla Firefox (version 3.5 recommended), before you save the configuration files, set the **Downloads** option of your browser as **Always ask me where to save files**.*

If you wish to upload the Backup Configuration of firmware V1R5.6 or later in system with firmware V1R5.5 or earlier, then make sure you upgrade the firmware to avoid system malfunction.

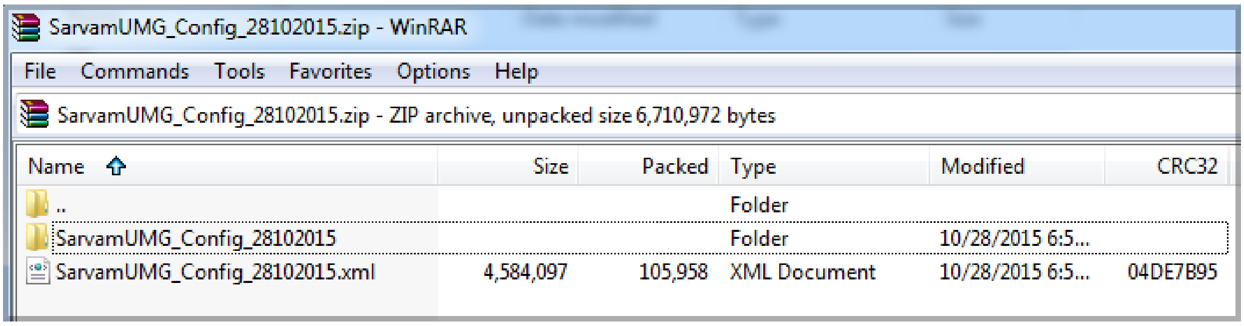
- Save the file on the local disk.



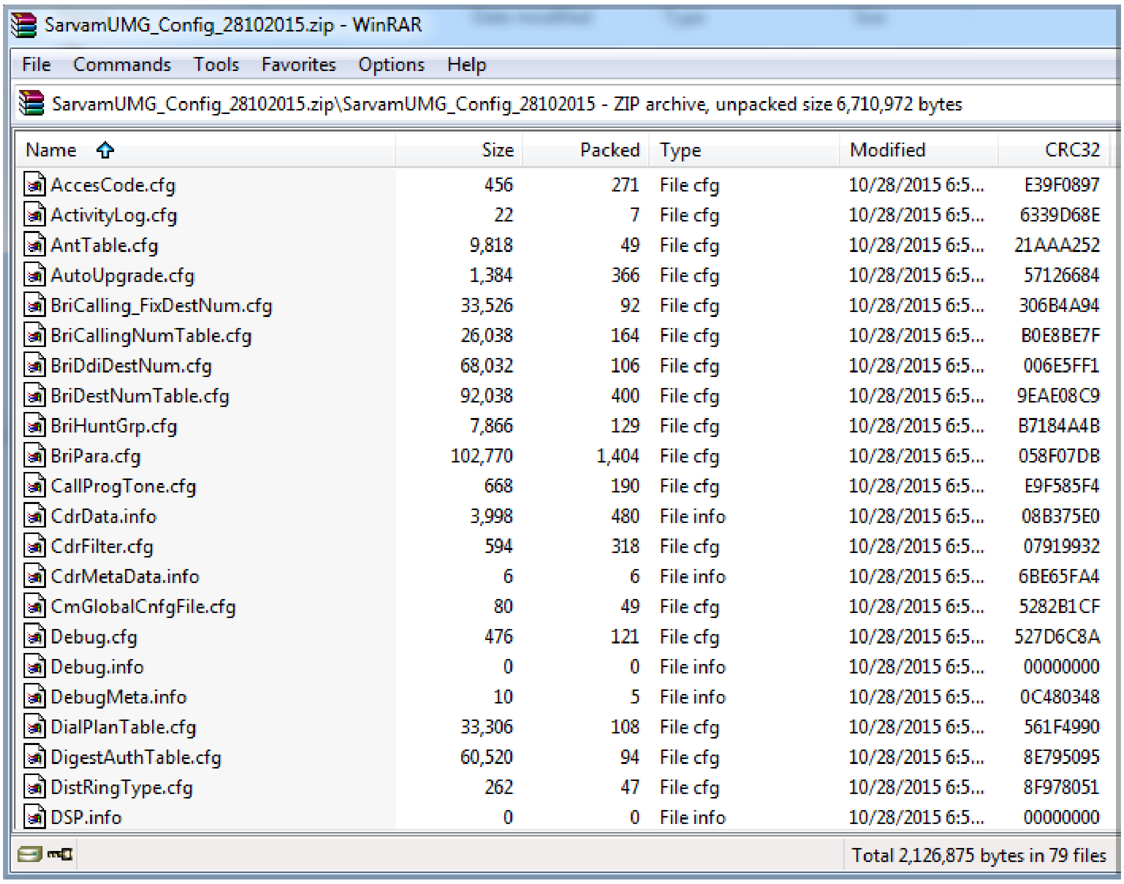
Save the back up configuration files by tagging the file name with the Version-Revision of the Firmware and tag the name of the backup folder on your computer with the date. This will help you at the time of restoring the back up configuration files.

- Open the configuration file (.zip) from the location you saved.

  The zip file contains all the system configuration files in .cfg format and xml format. You cannot edit the configuration files in .cfg format, however you can edit the configuration files in xml format and then upgrade the system with it.



- Open the **SarvamUMG_Config_28102015** 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.

Upgrading Configuration from a Personal Computer

You can upgrade configuration of SARVAM UMG with the configuration files stored on your computer. To do so,

- Click the **Upgrade Configuration from PC** button. A new window - **Upgrade Configuration From** opens.

- Click the **Choose File** button to reach the location on the local disk on which the configuration file is stored.

- Select the required configuration files from the location on the local disk.

- The path to the file will appear in the **Configuration Upgrade From** box.

- Click the **Upgrade** button.

At a time, you can upgrade configuration either:

- *manually or automatically from Auto Configuration Server*
- *manually from a Personal Computer.*

MBN File Upload

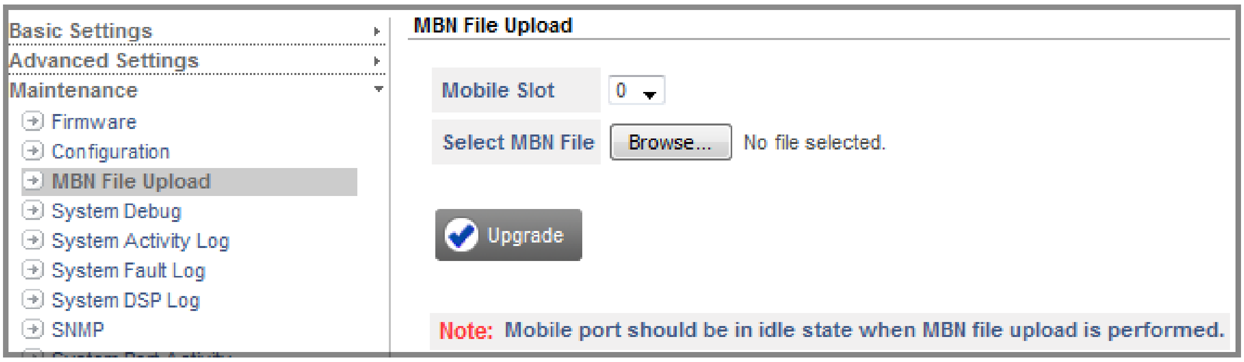
SARVAM UMG gives you the facility to upload MBN file of a newly emerged VoLTE supporting service provider in the market.

MBN File Upload can be configured only when 4G Mobile card is inserted.

Uploading MBN File

To upload the new MBN file,

- Click the **Maintenance** link to expand.
- Click **MBN File Upload** link.



- In **Mobile Slot**, select the desired slot number in which you have inserted the 4G card. The new MBN file will be uploaded in this card.
- In **Select MBN File,** click **Browse** to reach the location where MBN file is stored.

Make sure the file to be uploaded is in .mbn format.

- Click **Upload**.

The maximum number of MBN files that can be manually uploaded is 20.

While uploading the MBN, make sure:

- *you do not remove or insert any SIM.*
- *the Mobile Port is in idle state in "System Port Activity" or no SIM is inserted.*

After you have uploaded the MBN files make sure you check all the ports for the updated MBN files.

On successful upload a message is displayed in Jeeves.

The uploaded file will appear in the Manual Selection Table. For details, refer to "VoLTE Configuration".

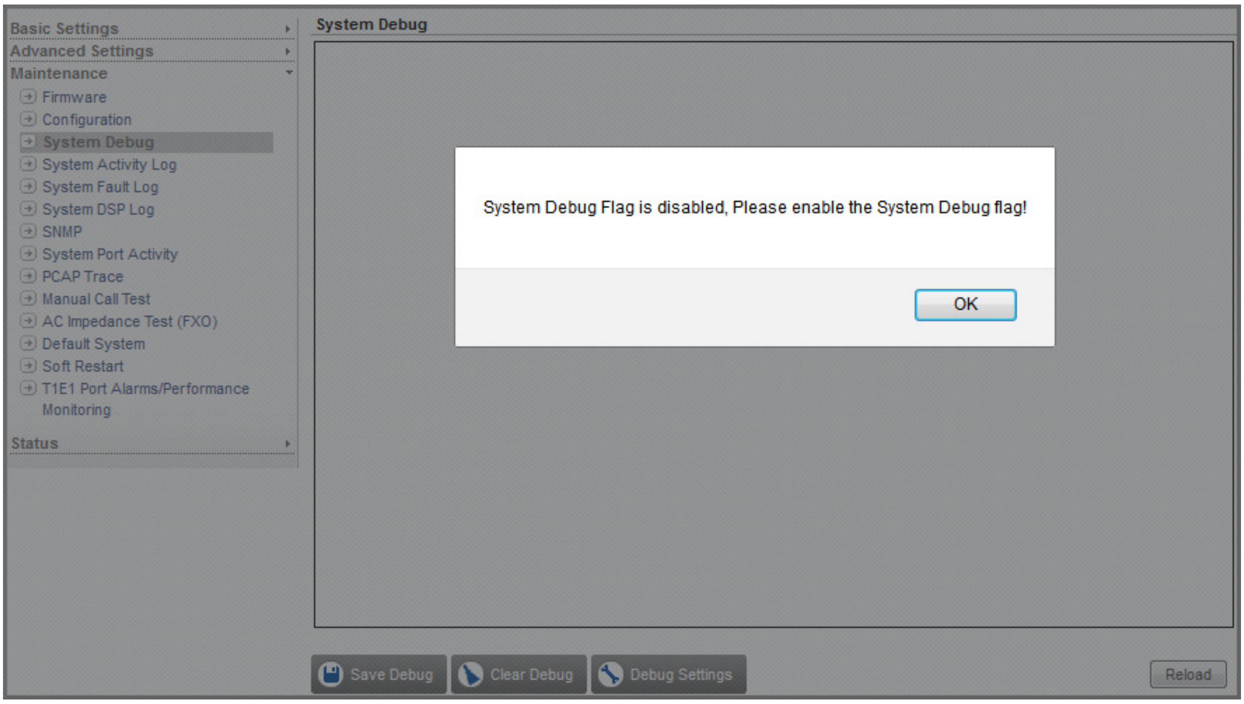
System Debug

Debugging is a method used for recording actions and events of the system. Debugs are the primary record keepers of the system and network activity. Debugging has several benefits which include troubleshooting, security and system administration.

SARVAM UMG supports Syslog Client for sending debug messages to the remote syslog server on the IP network.

Configuring System Debug

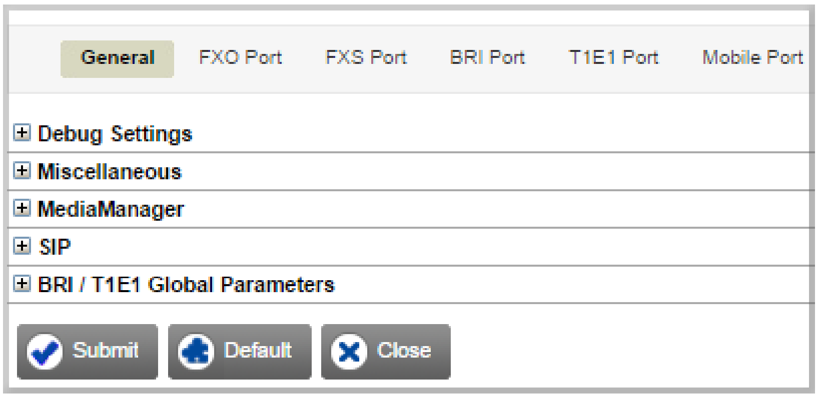
- Click the **Maintenance** link to expand.
- Click the **System Debug** link.



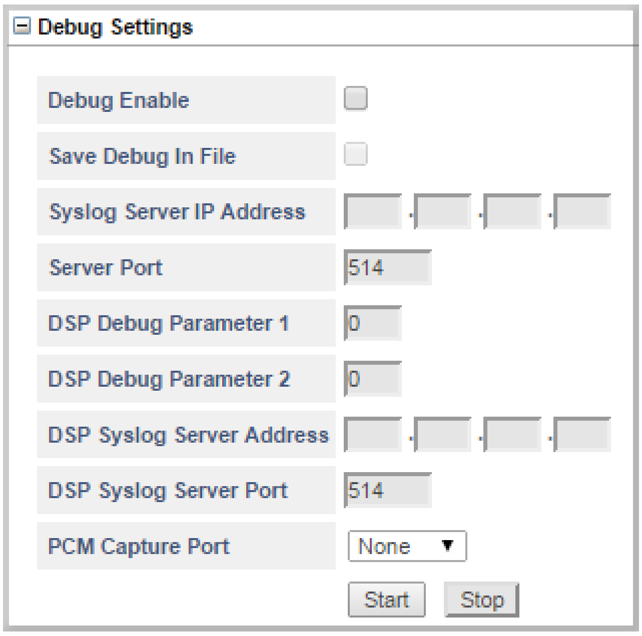
Debug details will be displayed only if you enable System Debug.

- The message appears to enable the Debug check box. Click **OK**.
- Click the **Debug Settings** button.

- The **Debug Settings** window opens.



- Under the **General** tab, click **Debug Settings** to expand.



- Select the **Debug Enable** check box to enable system debug. Default: Disabled.

  You will be able to configure the Debug Settings only after you enable this check box.

- Select the **Save Debug In File** check box, if you want to save the debug file in the system. Default: Disabled.

- In **Syslog Server IP Address**, enter the remote Syslog Server IP Address. Default: Blank.

- In Syslog **Server Port**, enter the port number. The range of the server port is 514, 1024 to 65535. Default: 514.

- Enter the **DSP Debug Parameter 1** value for which you require the debug. You can enter the values from 000 to 255.

Value	Meaning
000	Enables DSP Serial Real Time Debug
001	Enables DSP Serial Slow Debug
002	Enables DSP Ethernet Debug
003 to 255	Enables DSP Serial Slow Debug

Default is 000.

It is not possible to enable all these debugs at a time, only one of these can be enabled at a time.

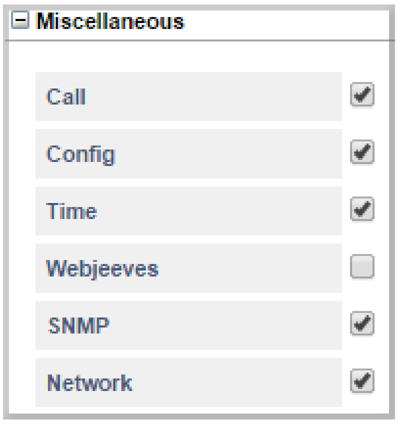
- Enter the **DSP Debug Parameter 2** value for which you require the debug. You can enter the values from 000 to 255.

Value	Meaning
000	Debug Off
001	Enable FXS Debug
002	Enable FXO Debug
004	Enable ISDN Debug
008	Enable GSM Debug
016	Enable Switch Debug
032	Enable Other Debugs (e.g. Slave HPI)

Default is 000.

- In **DSP Syslog Server Address**, enter the remote DSP Syslog Server Address. Default: Blank.
- In **DSP Syslog Server Port**, enter the port number. The range of the server port is 514, 1024 to 65535. Default: 514.
- In **PCM Capture Port**, select the desired port — FXO, Mobile, BRI, T1E1 and the respective port number. Default: None.
  - To start PCM Capturing for the desired port, click **Start**.
  - To stop PCM Capturing for the desired port, click **Stop.**

- Click **Miscellaneous** to expand.



- If you want the debug of these parameters, keep their — Call, Config, Time, SNMP and Network check boxes enabled.

  Select the Webjeeves check box to enable, if required.

  To disable a parameter, clear the respective check box.

- Click **MediaManager** to expand.

- Keep the **DebugLevel** check box enabled. All the debug levels — Error, Warning, Info, Debug — are enabled. To disable a debug level, clear the respective check box.

- Keep the **DebugModule** check box enabled. All debug modules — Media Layer, DSP Layer, RTP Handling — are enabled. To disable a debug module, clear the respective check box.

- In **DSP PCAP**, click the **Start** button to begin the capturing of the DSP Trace.

  Click the **Stop** button to stop DSP Trace capture.

  OR

Wait for the system to stop capturing. The system stops capturing once the maximum allotted memory is utilized.

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 DSP Trace is being used.

  - When the DSP capturing is stopped (by you or the system), click the **Save** button to save the files on your computer or on another computer.

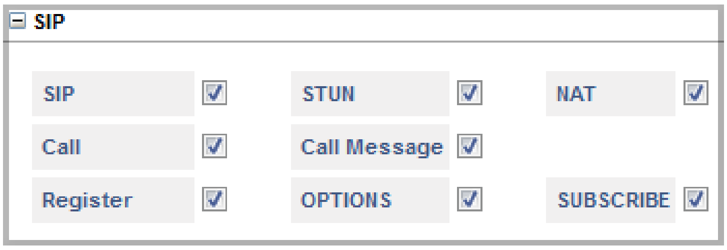
    A dialog box opens. You can select the path for saving the trace file.

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 DSP capture again.

  - After logging out of Jeeves, you can open the trace files using Wireshark/ Ethereal or any other software which supports opening of trace files.

- In **DSP Debug**, click the **Save** button to save the DSP Debug files on your computer or on another computer.

- Click **SIP** to expand.



  - If you want the debug of these parameters, keep the parameters — SIP, STUN, NAT, Call, Call Message, Register, OPTIONS and SUBSCRIBE check boxes enabled.

  - To disable a parameter, clear the respective check box.

- Click **BRI / T1E1 Global Parameters** to expand.

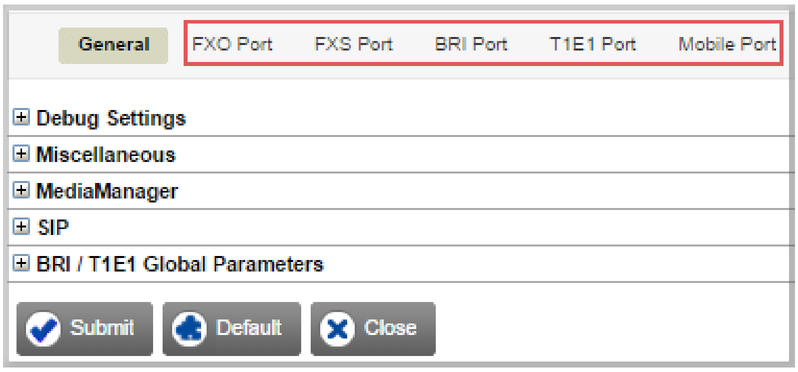
  - Select the check box of the desired level — Level 1 or Level 2 — to enable.

  - Click **Submit**.

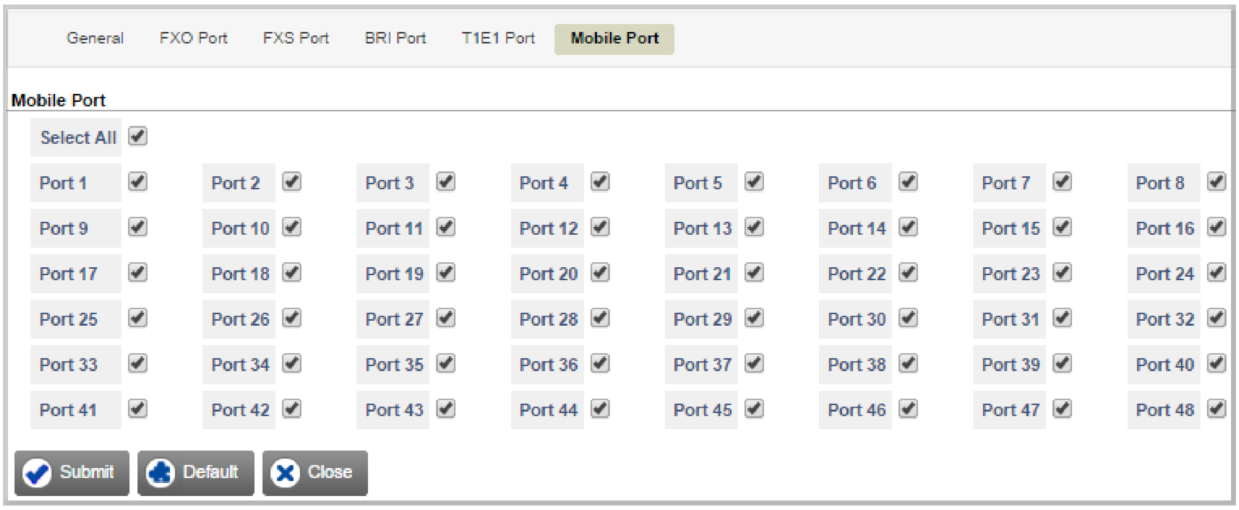
- You can also enable debug for the desired Ports — **FXO**, **FXS**, **Mobile**, **BRI**, **T1E1**.

- To do so,

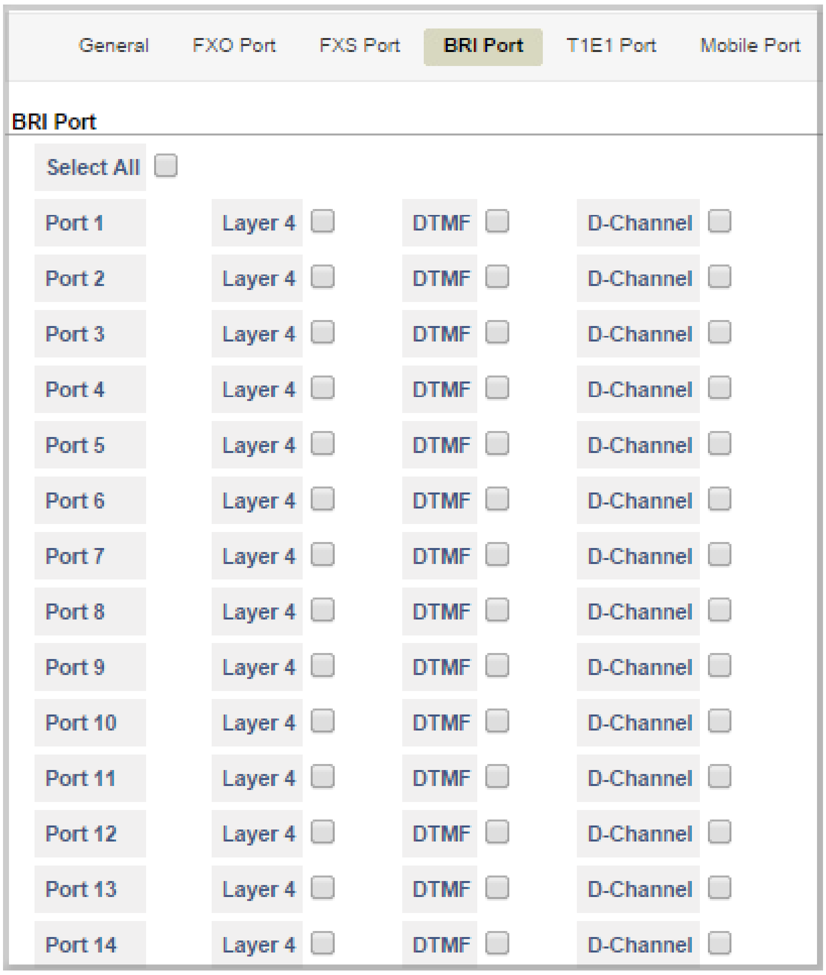
- Click the respective port tab.



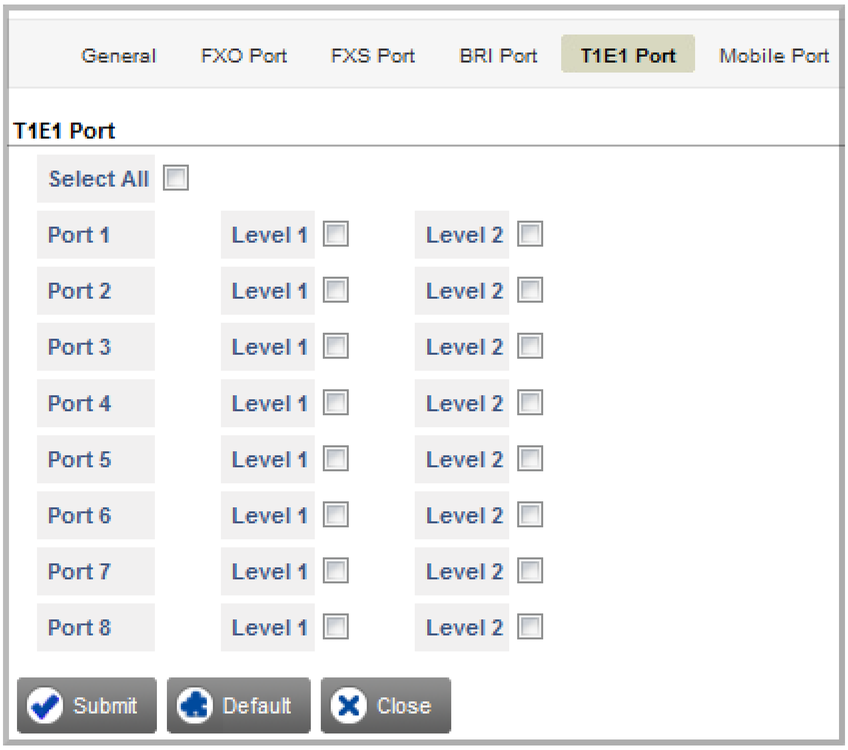
- If you have selected **FXO**, **FXS** or **Mobile**, select the check boxes of the respective port numbers you wish to enable.



- If you have selected **BRI**, select the check boxes of the desired options — Layer 4, DTMF, D-Channel of the respective port numbers you wish to enable.



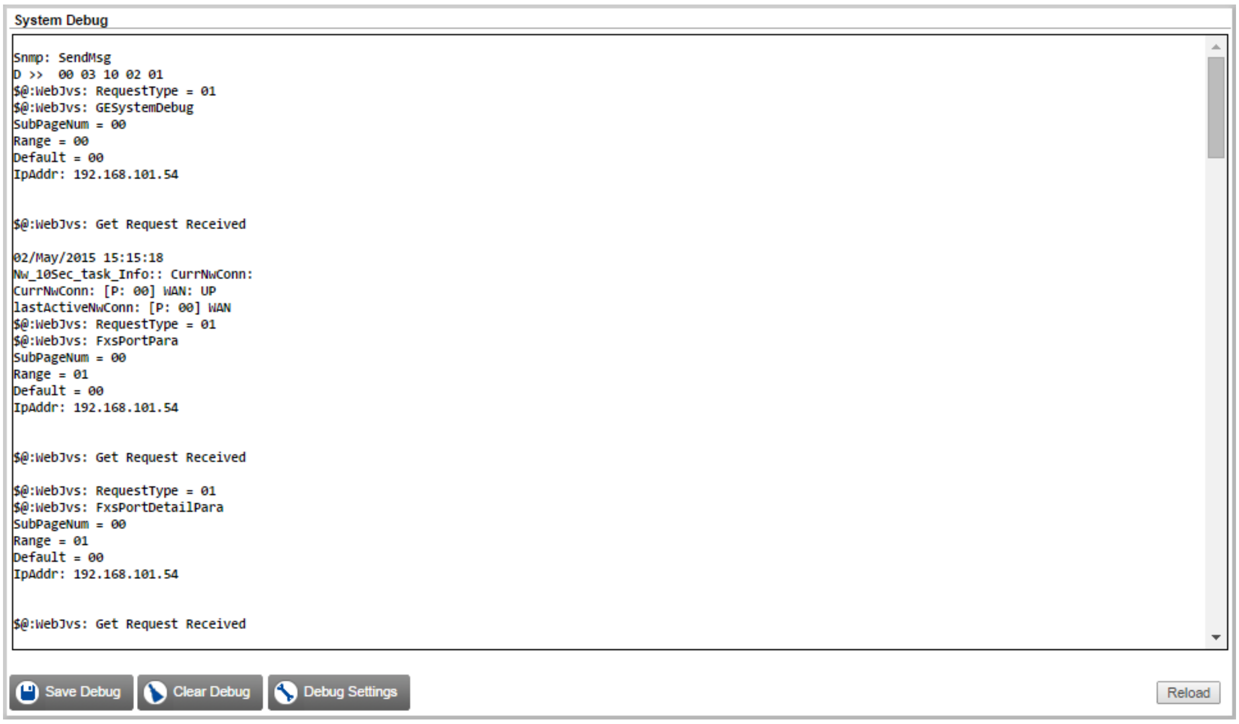
- If you have selected **T1E1**, select the check boxes of the desired options — Level 1, Level 2 of the respective port numbers you wish to enable.



- Click **Submit**.
- Click **Close**, you return to the System Debug page.

If debug is enabled, atleast one debug level should be selected. If no debug level is selected, SARVAM UMG will prompt you to select a debug level.

- All the Debug events appear on the screen.



- Whenever you want the system to fetch an updated debug report, click the **Reload** button.
- If you want to delete all the events, click the **Clear Debug** button.
- If you want to save the Debug events, click the **Save Debug** button.
    - Save the file at the desired location on the local disk.
    - Open the **SaveDebug.zip** file. The zip file contains the system debug file **SaveDebug.txt**.

System Activity Log

The SARVAM UMG 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.

The System Activity Log can be downloaded on a computer in form of a report. SARVAM UMG also supports Syslog Client for System Activity Logs. The Syslog Client enables the system to send activity logs in syslog format to the remote ‘Syslog Server’. You can view the logs on the remote server.

Index of the Type of Activities recorded in the System Activity Log:

Event Index	Activity	Description
1	Matrix SARVAM UMG started:	When the application starts.
2	Default Jumper set in Master Card.	Jumper on the CPU Card is set in default position.
3	Display Configuration Type:	Displays Configuration type used by the system.
4	Card Present: Slot Num:Card Type:	The Slot Number and Type of cards present in the system.
5	Card Status: Slot Num:Card Type:	Card Status during Power ON.
6	System Network functionality restarted	When network module restarts.
7	Network Connection Type change:	When the Network Connection Type is changed.
8	Network IP address change:	When the Network IP Address is changed.
9	Network Subnet Mask change:	When the Network Subnet is changed.
10	Network Gateway address change:	When the Network Gateway Address is changed.
11	DNS Server address change:	When the DND Server Address is changed.
12	Network VLAN/CoS Flag Enabled:	VLAN/CoS Flag status.
13	Network VLAN/CoS ID Change:	VLAN ID configured.
14	SIP Stack Destructed.	When the SIP Stack is Destructed.
15	SIP Stack Constructed.	When the SIP Stack is Constructed.
16	Stun Status:	Displays the STUN status.
17	Sync Date-Time with SNTP Server	SNTP Server sync with configured Server Address.
18	RTC Change	When Date-time is changed in the system.
19	System Default	When the system is set to default.
20	System Restart using:	Displays the reason for the system restart.
21	System Config file change:	Displays config file name when parameter is changed.
22	Network WAN Link UP	When WAN port of CPU card is connected and network link is working.

Event Index	Activity	Description
23	Network WAN Link DOWN	When WAN port of CPU card is connected and network link is not working.
24	DynDNS status:	Displays the DynDNS Status.
25	WAN MAC address change:	When the system inits or MAC address of the system is changed (System to Clone or vice-versa).
26	FXO Line Connected, Slot Num: Port Offset:	When Line is connected to FXO Port and status is found to be connected.
27	FXO Line Not Connected, Slot Num: Port Offset:	When FXO Line status is found to be not connected.
28	BRI Layer 1 UP: Slot Num: Port Offset:	BRI Port Layer 1 link is up.
29	BRI Layer 2 UP: Slot Num: Port Offset:	BRI Port Layer 2 link is up.
30	BRI Layer 1 DOWN: Slot Num: Port Offset:	BRI Port Layer 1 link is down.
31	BRI Layer 2 DOWN: Slot Num: Port Offset:	BRI Port Layer 2 link is down.
32	T1E1 Layer 1 UP: Slot Num: Port Offset:	T1E1 Port Layer 1 link is up.
33	T1E1 Layer 2 UP: Slot Num: Port Offset:	T1E1 Port Layer 2 link is up.
34	T1E1 Layer 1 DOWN: Slot Num: Port Offset:	T1E1 Port Layer 1 link is down.
35	T1E1 Layer 2 DOWN: Slot Num: Port Offset:	T1E1 Port Layer 2 link is down.
36	Call Minutes Expire, Slot Num: Port Offset:	When Mobile Port Call Minutes assigned is used by the port.
37	Web Jeeves, FTP and Telnet Password change.	When SE password is changed.
38	Command Password change	When command password is changed.
39	Server Port Change: HTTP: HTTPS: FTP: Telnet:	When Server port is changed.
40	Web Server Access from WAN:	Displays the status whether enable or disable.
41	FTP Server Access from WAN:	Displays the status whether enable or disable.
42	Telnet Server Access from WAN:	Displays the status whether enable or disable.
43	Allow Server Access from specific IP Address:	Displays the status whether enable or disable.
44	Block ICMP on WAN:	Displays the status whether enable or disable.
45	Block PING on WAN:	Displays the status whether enable or disable.
46	Local Certificate for TLS:	Displays Local Certificate name used for TLS

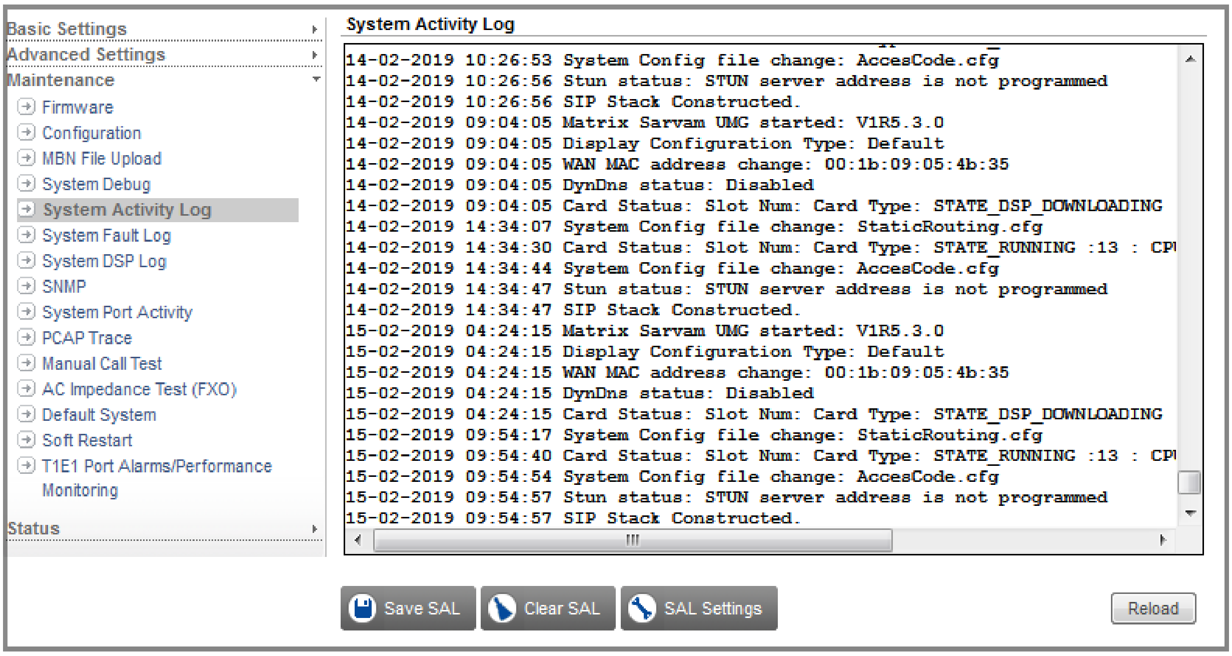
Event Index	Activity	Description
47	Local Certificate for Firmware Upgrade:	Displays Local Certificate name used for Firmware Upgrade.
48	Local Certificate for Configuration Upgrade:	Displays Local Certificate name used for Configuration Upgrade.
49	Reserved	
50	Local Certificate for WebServer:	Displays Local Certificate name used for WebServer.
51	Trusted Root CA Certificate Uploaded:	Displays Trusted Root CA Certificate uploaded.
52	Trusted Root CA Certificate Upload Status:	Displays Trusted Root CA Certificate upload status.
53	Local Certificate Uploaded:	Displays Local Certificate uploaded.
54	Local Certificate Upload Status:	Displays Local Certificate upload status.
55	Trusted Root CA Certificate Deleted:	Displays Trusted Root CA Certificate deleted.
56	Local Certificate Deleted:	Displays Local Certificate deleted.
57	Auto Firmware Upgrade process is started:	Display the reason for which Auto Firmware Upgrade started.
58	Auto Firmware is uploading file name:	Display firmware filename to be uploaded.
59	Auto Firmware Upgrade is stopped due to	Display the reason for upgrade stop.
60	Auto Firmware file is successfully uploaded:	Display filename uploaded successfully.
61	Auto Firmware file upload failed due to	Displays the reason for upload failure.
62	Auto Config process is started:	Displays the reason for which Auto Configuration Upgrade started.
63	Auto Config process is stopped due to	Displays the reason for upgrade being stopped.
64	Auto Config file is successfully uploaded	Displays filename of the config file that has been uploaded successfully.
65	Auto Config process failed due to	Displays the reason for upload failure.
66	Auto Config file parsing failed	Displays the filename of the config file for which parsing got failed.
67	Auto Config file is parsing Done	Displays when the parsing gets completed.
68	MM::Maximum DSP CPU Usage Reached	When the maximum usage limit of DSP CPU is reached.
69	SE Login blocked for IP:	When an IP Address is blocked for SE Login due to continual invalid password entry.
70	MM::	Displays CMM Module status (when PCAP is start/stop).
71	Mobile SIM:	Mobile SIM Activity-Absent/Present

How to configure

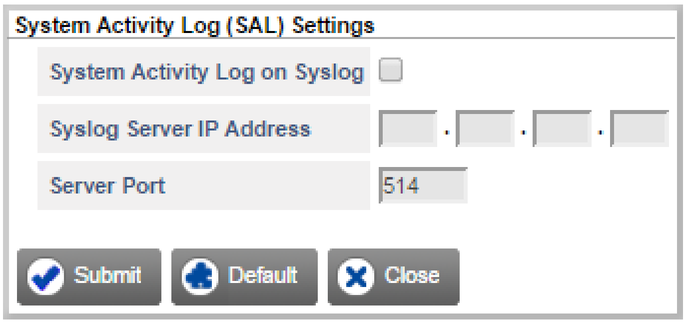
Activity Logs are stored in the system by default. For sending System Activity Log on remote server, Syslog Server setting needs to be done.

Configuring System Activity Log

- Click the **Maintenance** link to expand.
- Click the **System Activity Log** link.



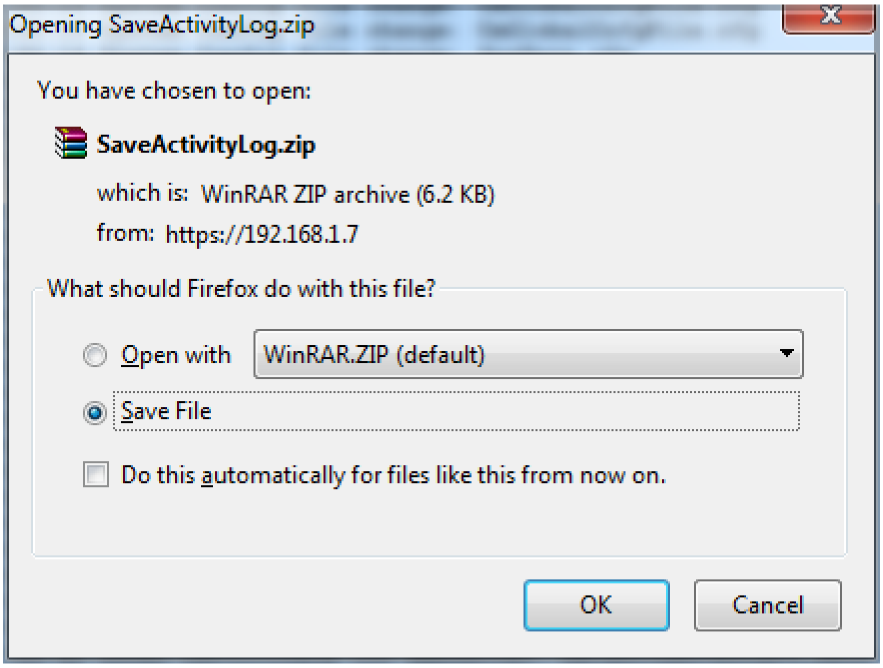
- To configure the System Activity Log settings, click the **SAL Settings** button.
  - The System Activity Log (SAL) Settings page opens.



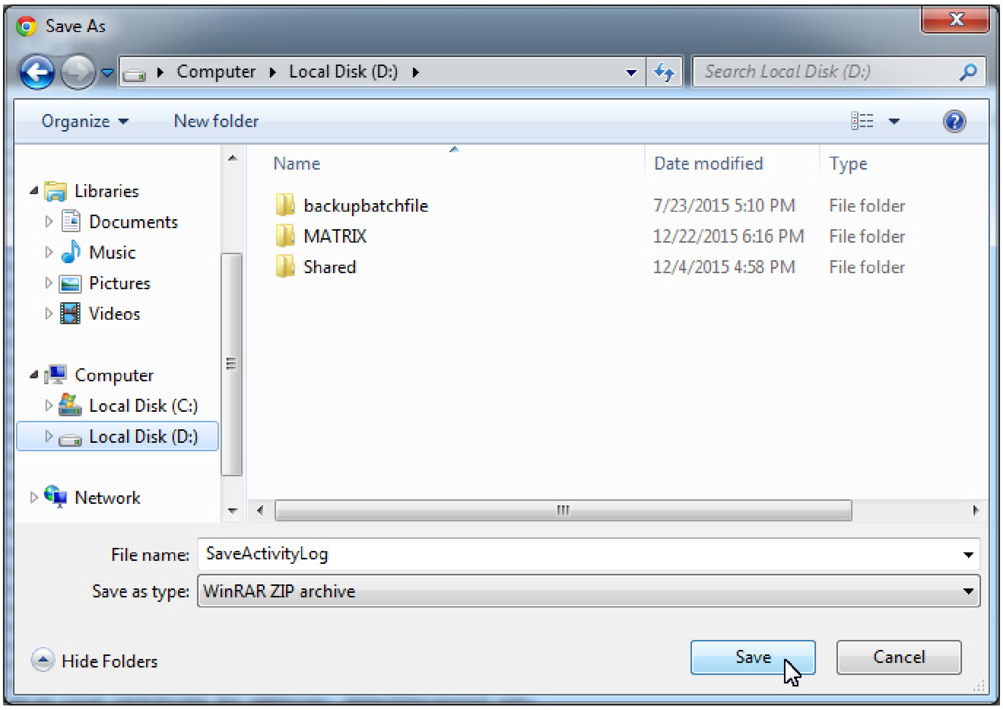
  - Select the **System Activity Log on Syslog** check box to enable the System Activity Log. Default: Disabled. You will be able to configure the SAL Settings only after you enable this check box.
  - In **Syslog Server IP Address**, enter the remote Syslog Server IP Address. Default: Blank.

- In Syslog **Server Port**, enter the port number. The range of the server port is 514, 1024 to 65535. Default: 514.
- Click **Submit**.
- Click **Close**, you return to the System Activity Log page.

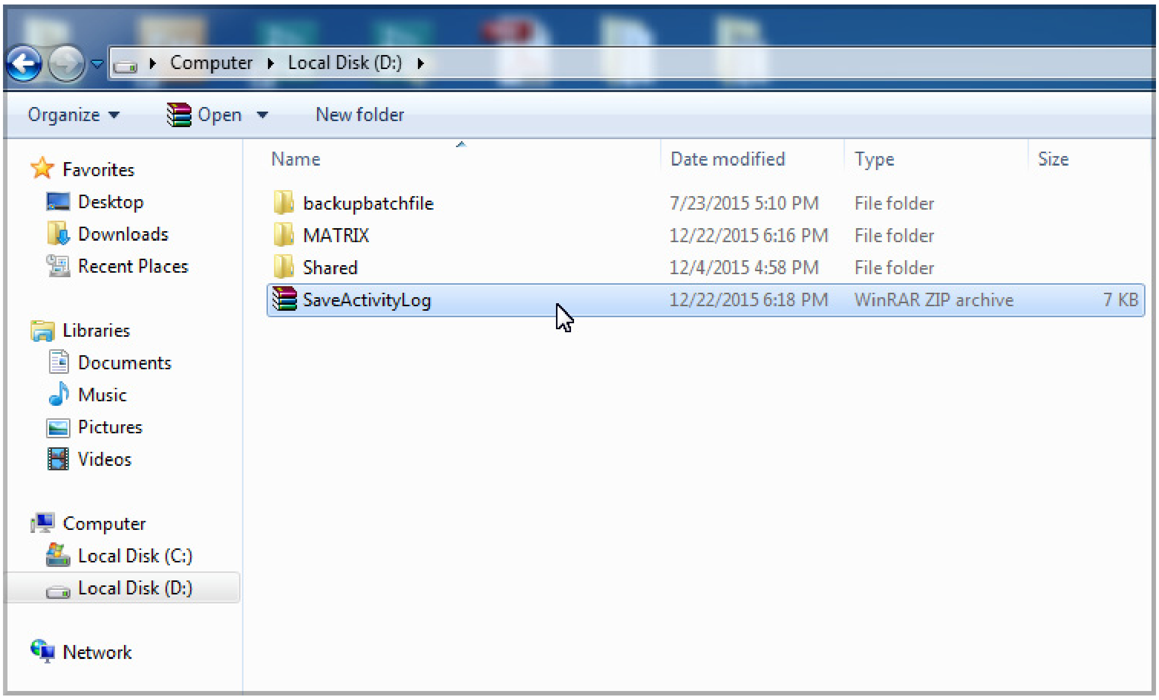
- The list of activities appear on the screen.
- Whenever you want the system to fetch an updated activity report, click the **Reload** button.
- If you want to delete all the activities, click the **Clear SAL** button.
- If you want to Save the activities, click the **Save SAL** button.
  - You will get a prompt with the option to open the **SaveActivityLog.zip** file or save the file to a location.



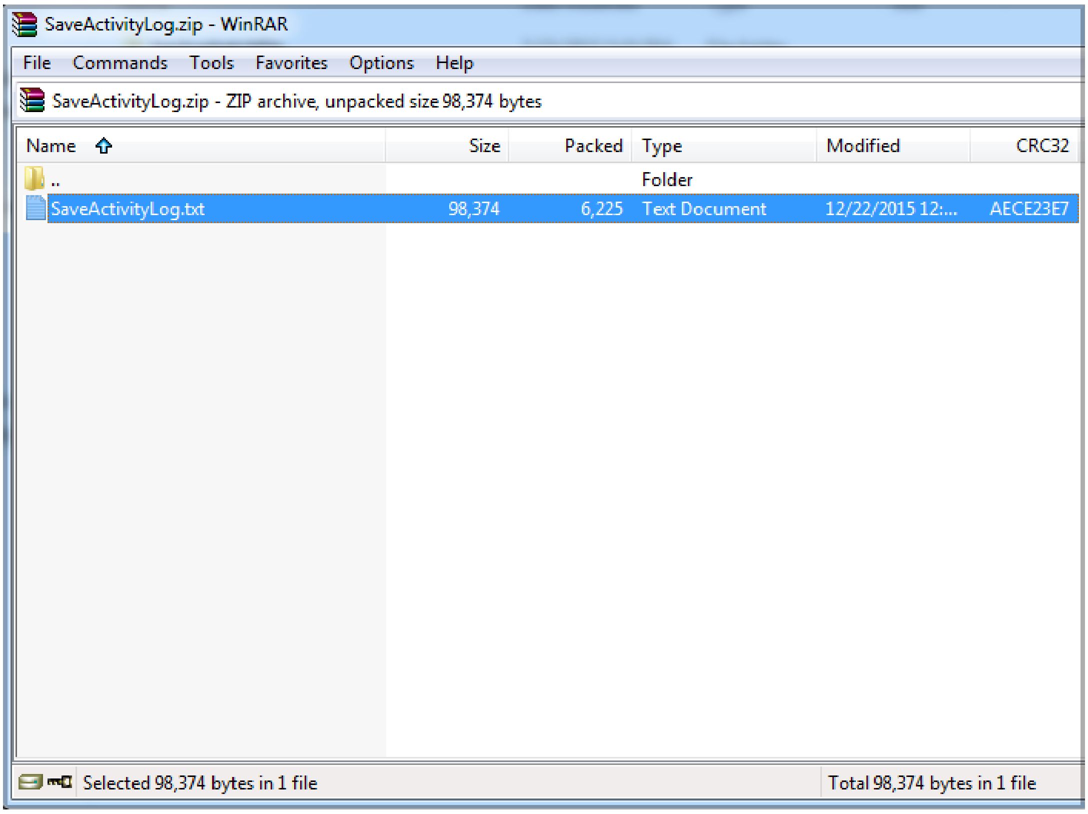
- Save the file at the desired location on the local disk.



- Open the **SaveActivityLog.zip** file.



- The zip file contains the **SaveActivityLog.txt** file.



System Fault Log

The SARVAM UMG maintains a log of all system faults. The system Fault Log has a buffer capacity of 500 records. The Fault Log stores records using the FIFO method.

The System Fault log can be downloaded on a computer in form of a report. SARVAM UMG also supports Syslog Client for System Fault Logs. The Syslog Client enables the system to send fault logs in syslog format to the remote 'Syslog Server'. You can view the logs on the remote server.

The different fault events that are logged are summarized in this table:

Event ID	Event	Description
1	Failed to initialize Syslog Server. Syslog Type:	Failure in Syslog Server initialization.
2	Failed to initialize SNMP Server.	Failure in SNMP Server.
3	Failed to initialize IPC Message Queue	Internal system queue error.
4	Failed to initialize random number generator.	Random number generation failure.
5	Failed to download DSP, Slot Num: Card Type:	When the system fails to download DSP.
6	Keep alive not received Slot Num: Card Type:	Keep Alive failed for CPU DSP or Slave Card.
7	Failed to initialize Call Manager.	Call manager initialization failure.
8	Failed to initialize Port Config.	Config file initialization failure.
9	Failed to initialize Auto Upgrade.	Auto Upgrade failure.
10	SIP stack construct failed, error code:	Error response when SIP Stack construct fails.
11	SIP TLS initialize failed, error code:	Error response when SIP TLS initialize fails.
12	Outgoing Invite send failed, Trunk Num: reason:	When any INVITE message sending fails.
13	Invalid Slave Layer msg recv:	When Invalid message is received from Slave Layer.
14	Invalid Slave Card msg recv, Slot Num: Port Num:	When Invalid message is received from Slave Card.
15	DSP Frame sync error, Slot Num:	Internal system DSP error.
16	DSP DMA Drop error, Slot Num:	Internal system DSP error.
17	DSP DMA Time out error, Slot Num:	Internal system DSP error.
18	DSP DMA address error, Slot Num:	Internal system DSP error.
19	DSP HPI Queue Full error, Slot Num:	Internal system DSP error.
20	MM::Channel Allocation Failed	VoIP DSP channel allocation failure.
21	MM::DSP Command Failed	VoIP DSP command failure.
22	MM::DSP Firmware Download Failed	VoIP DSP Firmware download failure.
23	MM::Invalid Command from Master	VoIP DSP command error.
24	MM::Invalid Event from DSP Layer	VoIP DSP error.

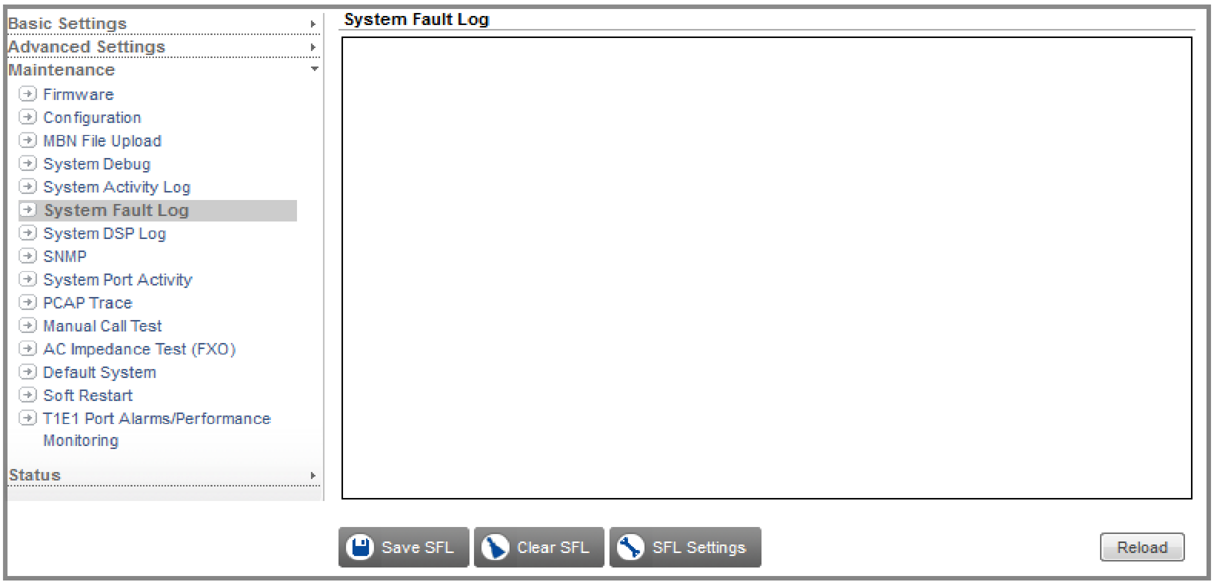
Event ID	Event	Description
25	DspLoader::Init Failed	When DSP loader failed to init.
26	DspLoader::CoffLoader Init Failed, Image: AppId:	Dsp Coff loader init failure.
27	DspLoader::Timer expired, State: AppId:	Dsp Loader timer expired.
28	DspLoader::Incorrect Read Result, State: AppId:	Dsp loader read status failure.
29	CardManager::Saved and Received Card Info not matched	Card status mismatched.
30	MMAccLayer::In Running state, received Invalid State:	When Media manager receives an invalid state.
31	Failed to Open RTC Driver	RTC failure.
32	Invalid Card Type Received, Slot Num: Card Type:	When a card not supported by the system is inserted in any slot.
33	Failed to Init Native Configuration	Failure in initialisation of the Native Configuration.
34	SLT Thermal Shutdown, Slot Num : Port Offset : Port Num	FXS Port Thermal Shutdown
35	Unlicensed slot, Card Inactive, Slot Num	Unlicensed slot inactive card

How to configure

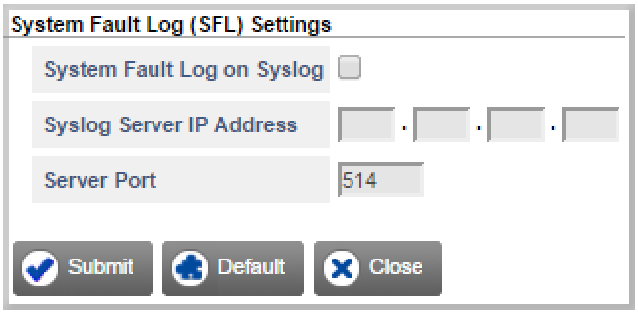
Fault Logs are stored in the system by default. For sending System Fault Log on remote server, Syslog Server setting needs to be done.

Configuring System Fault Log

- Click the **Maintenance** link to expand.
- Click the **System Fault Log** link.

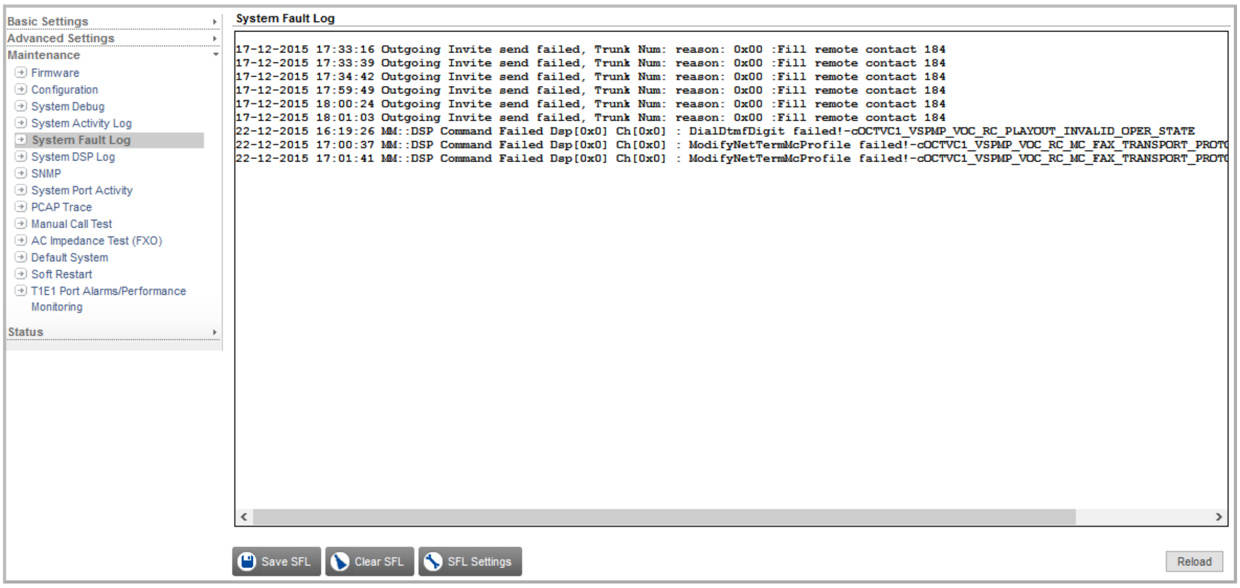


- To configure the System Fault Log settings, click the **SFL Settings** button.

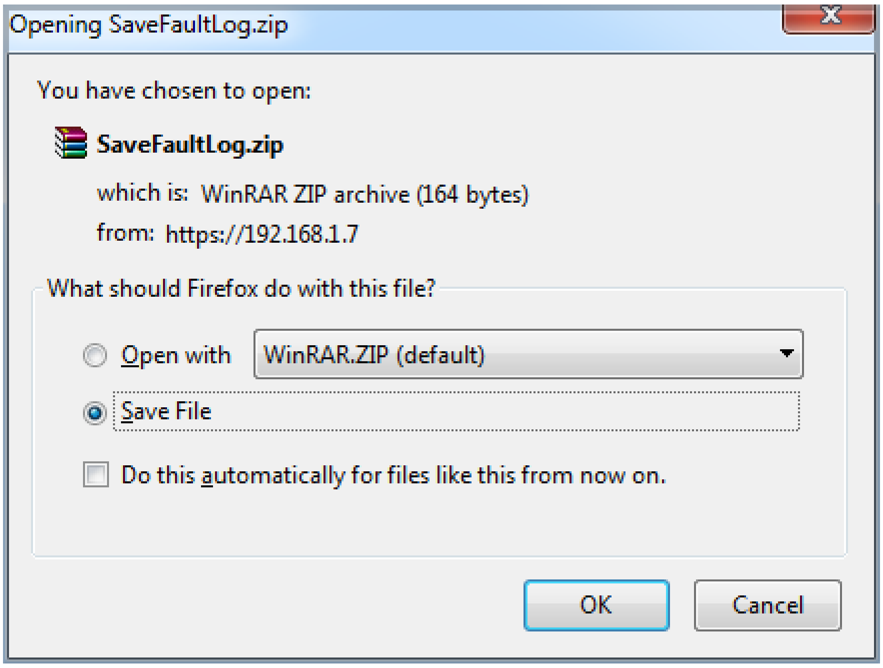


- The System Fault Log (SFL) Settings page opens.
    - Select the **System Activity Log on Syslog** check box to enable the System Fault Log. Default: Disabled. You will be able to configure the SFL Settings only after you enable this check box.
    - In **Syslog Server IP Address**, enter the remote Syslog Server IP Address. Default: Blank.
    - In Syslog **Server Port**, enter the port number. The range of the server port is 514, 1024 to 65535. Default: 514.
    - Click **Submit**.
    - Click **Close**, you return to the System Fault Log page.

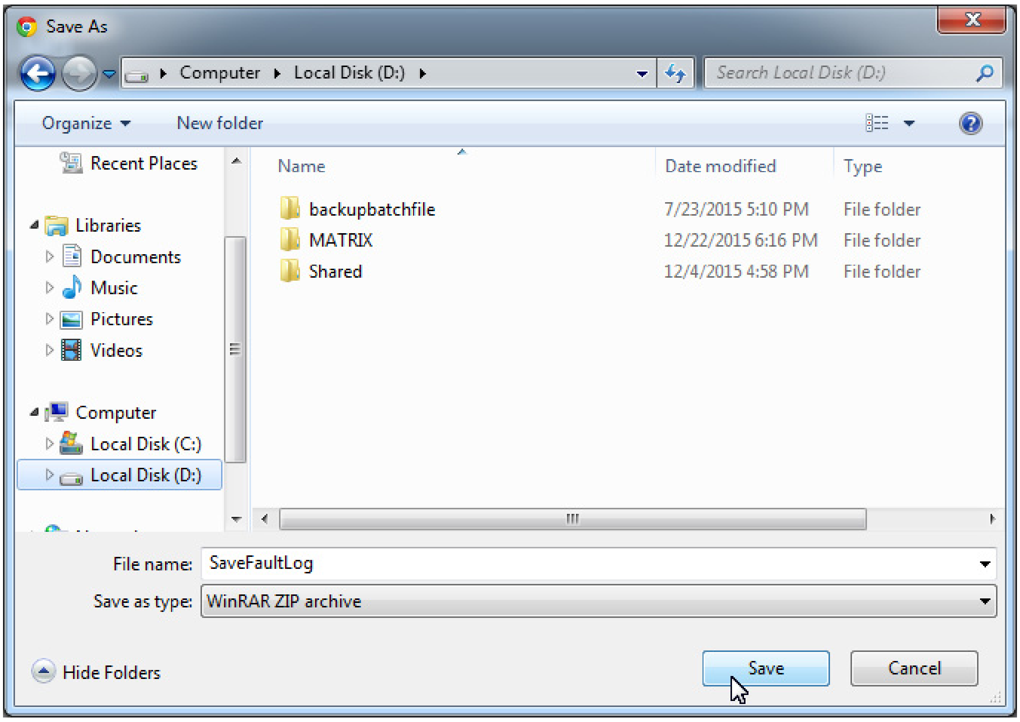
- The list of faults appear on the screen.



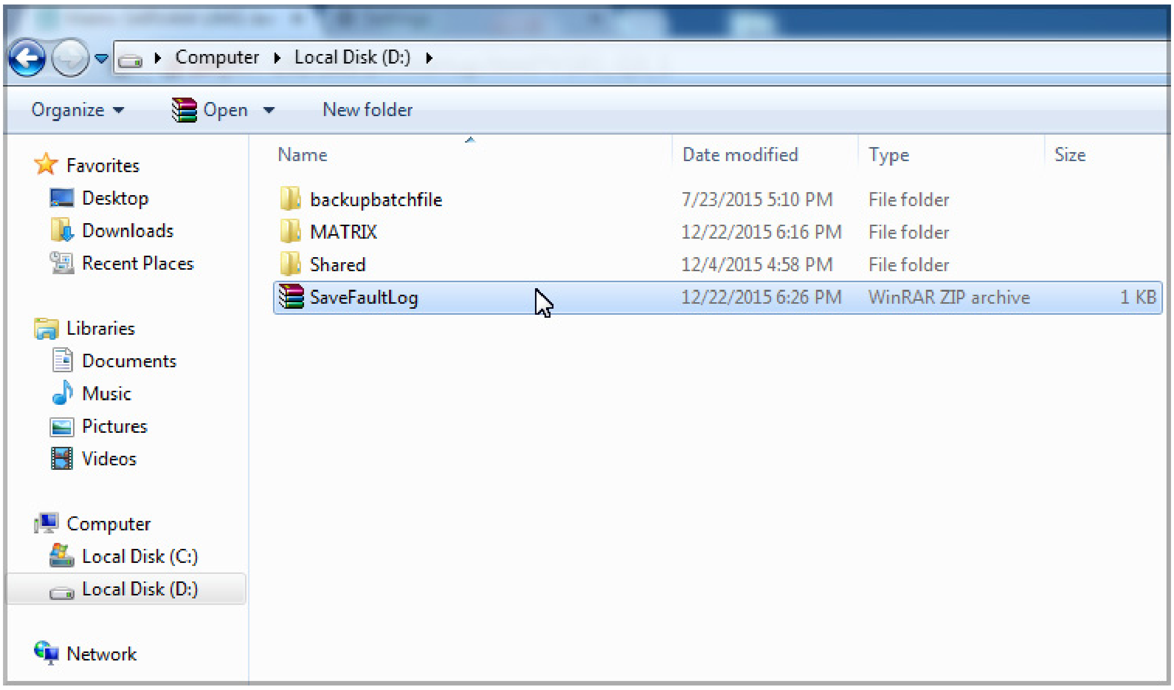
- Whenever you want the system to fetch an updated fault report, click the **Reload** button.
- If you want to delete the entire list of faults, click the **Clear SFL** button.
- If you want to Save the list of faults, click the **Save SFL** button.
    - You will get a prompt with the option to open the **SaveFaultLog.zip** file or save the file to a location.



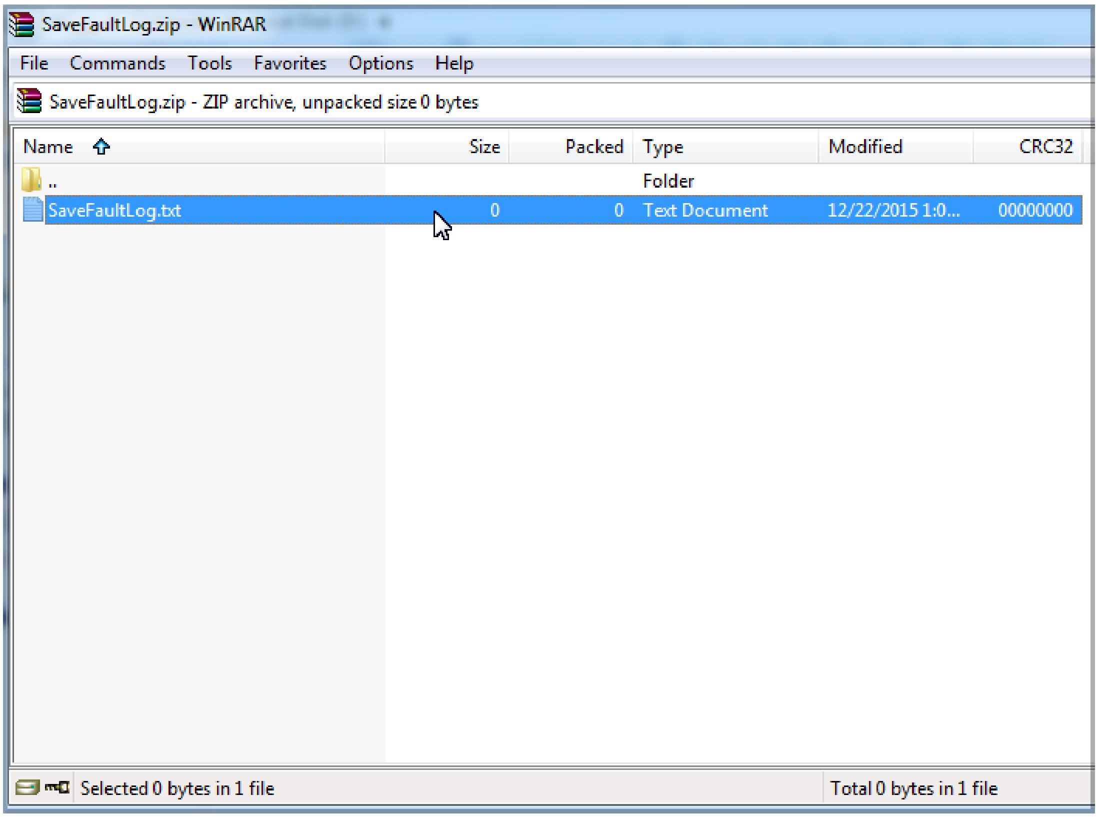
- Save the file at the desired location on the local disk.



- Open the **SaveFaultLog.zip** file.



- The zip file contains the **SaveFaultLog.txt** file.



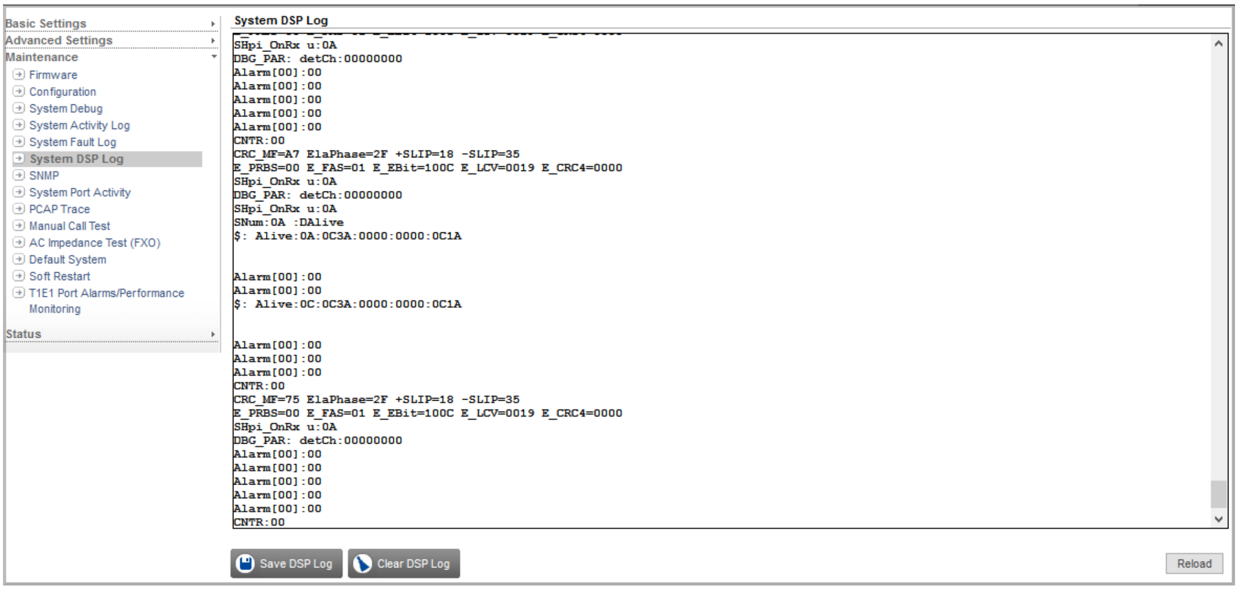
System DSP Log

The SARVAM UMG monitors all the activities of the slave cards inserted in the system and maintains records of these activities in the System DSP Log.

You can download and save the System DSP log on a computer, if required.

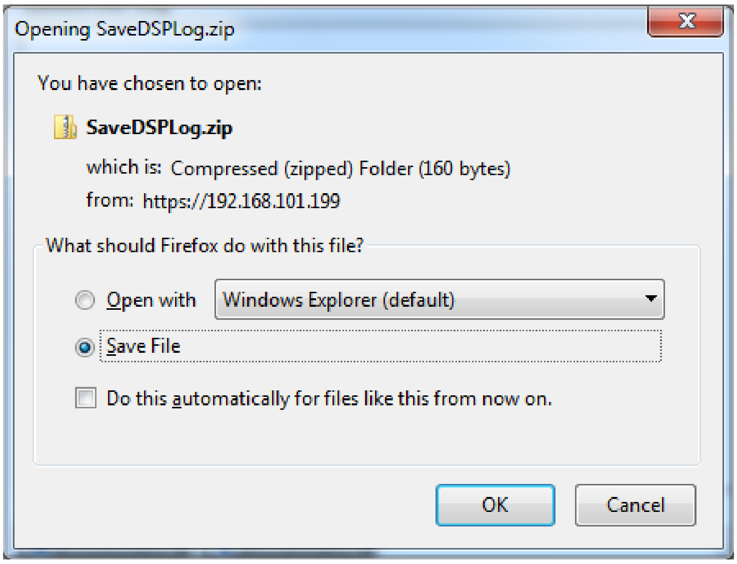
Configuring System DSP Log

- Click the **Maintenance** link to expand.
- Click the **System DSP Log** link.
- The System DSP Log page opens. The list of DSP activities appear on the screen.

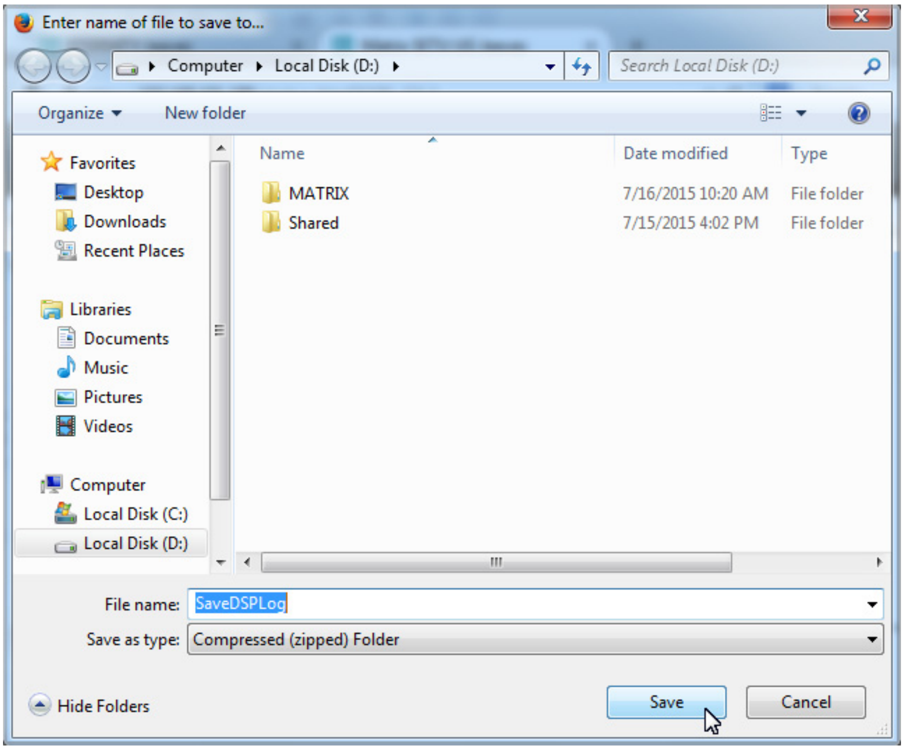


- Whenever you want the system to fetch an updated DSP activity report, click the **Reload** button.
- If you want to delete all the DSP activities, click the **Clear DSP Log** button.
- If you want to Save the DSP activities, click the **Save DSP Log** button.

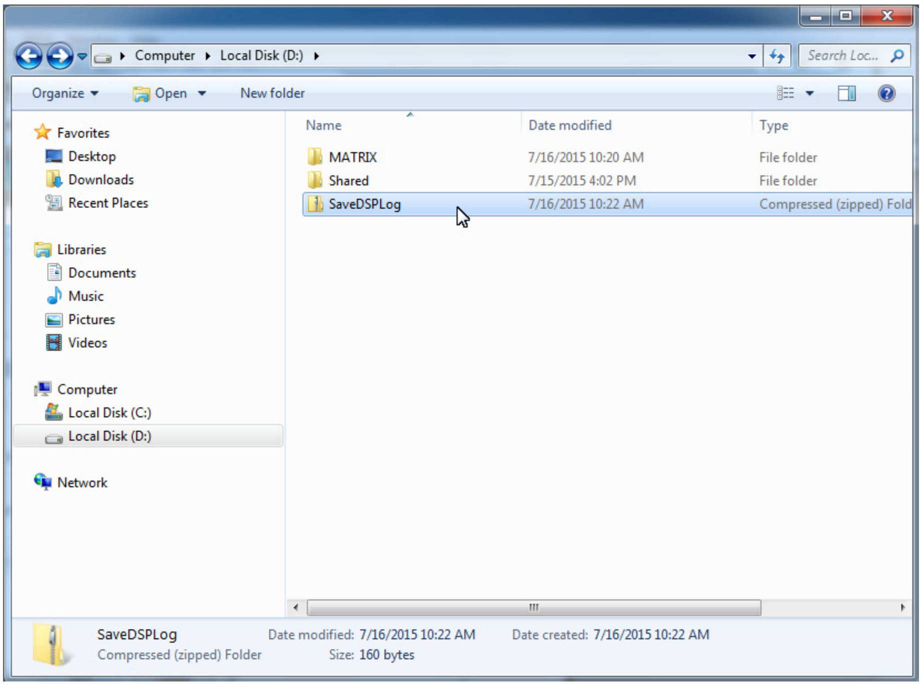
- You will get a prompt with the option to open the **SaveDSPLog.zip** file or save the file to a location.



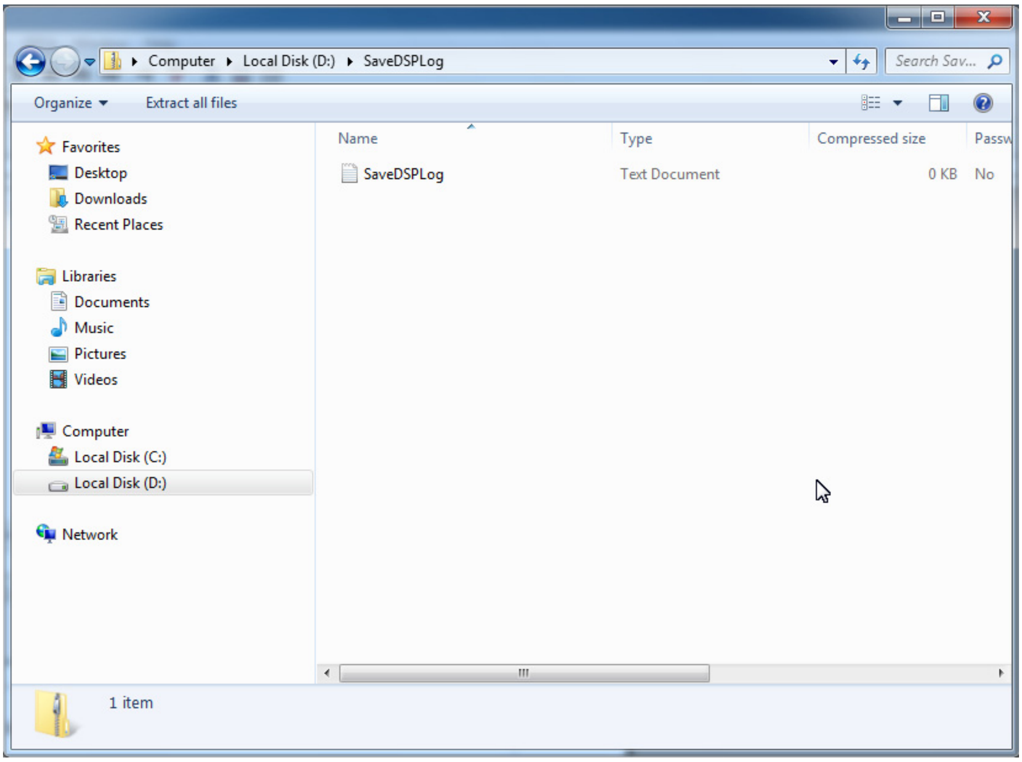
- Save the file at the desired location on the local disk.



- Open the **SaveDSPLog.zip** file.



- The zip file contains the **SaveDSPLog.txt** file.



Simple Network Management Protocol (SNMP)

Simple Network Management Protocol (SNMP) is an application-layer protocol used for exchanging management information between network devices. Using SNMP, you can manage and monitor network elements, audit network usage, detect network faults or inappropriate network access.

The SNMP architecture consists of:

- An **SNMP Agent** is a program that is bundled within the managed device. SNMP agent allows a managed device to collect the Management Information Base from the device and make it available to the SNMP Manager on request. It receives SNMP requests and generates SNMP responses or notifications (traps/informs). The SNMP Agents are SNMP Servers.

- **SNMP Manager**, usually the Network Management Station. The manager communicates with multiple SNMP Agents implemented in the network. It generates SNMP requests and receives SNMP responses and notifications (traps/informs). The SNMP Manager is an SNMP Client.

- **Managed device** or the network element is a part of the network that requires some form of monitoring and management. For example, switch, routers, servers.

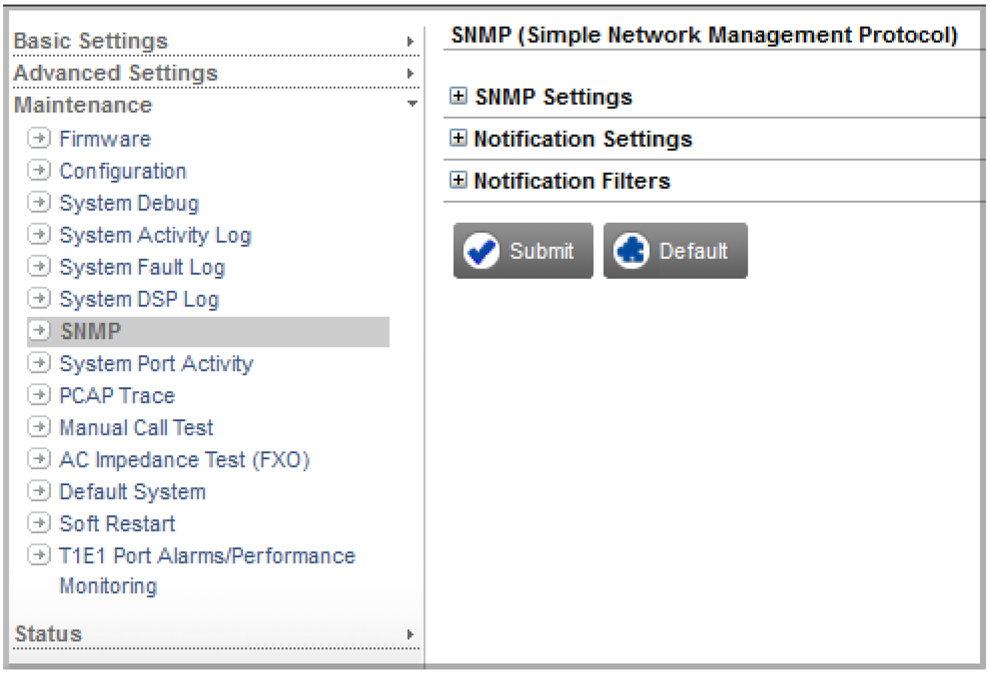
- **Management Information Base** is the commonly shared database between the Agent and the Manager.

SNMP uses UDP (User Datagram Protocol) as the transport protocol for passing information between Managers and Agents. The Agent listens on UDP port 161 for requests from Manager and the Manager listens on UDP port 162 for notification from Agent.

To configure SNMP parameters,

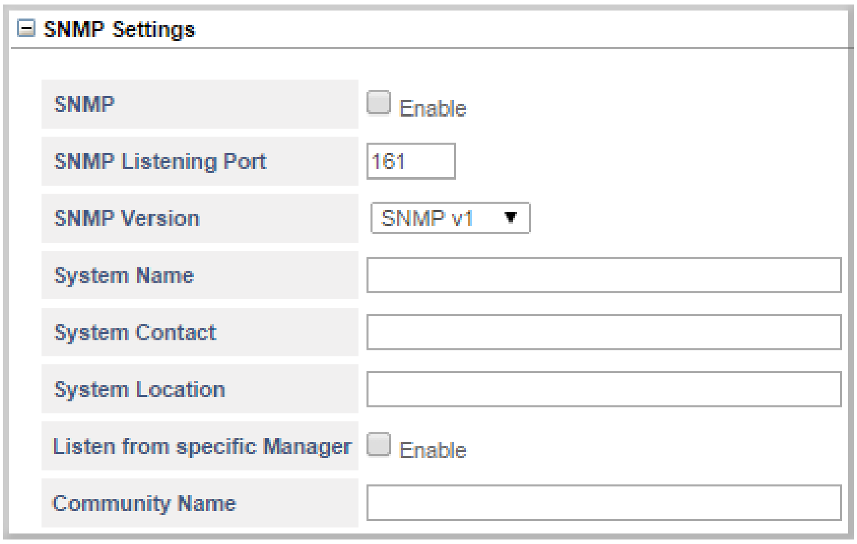
- Click the **Maintenance** link to expand.

- Click the **SNMP** link.



SNMP Settings

- Click **SNMP Settings** to expand.



- Select the **SNMP** check box to enable. Default: Disabled.
- Configure the **SNMP Listening Port**. Valid range is 161/ 1031 to 65535. Default: 161.
- Select the **SNMP Version** as supported by your SNMP Manager. You can select— SNMPv1, SNMPv2c, SNMPv3.

  For enhanced security, you must select SNMPv3.

- Configure the **System Name**. When there are multiple devices connected in the same network, the name configured helps to identify the SNMP Agent within the network. The System Name can be a maximum of 40 characters. Default: Blank.

- Configure the **System Contact**. It is the name and number of the person to be contacted, in case of notification. The System Contact can be of a maximum of 40 characters. Default: Blank.

- Configure the **System Location**. This is the physical location of SARVAM UMG. This information is helpful to the administrator. The System Location may consist of a maximum of 40 characters. Default: Blank.

- Select the **Listen from Specific Manager** check box, if you want the system to listen to the incoming SNMP messages from a specific manager. Default: Disabled.

  - If you have enabled **Listen from Specific Manager** check box, you must configure the specific **Manager's Address.**

    The Manager's Address can be a Domain Name or an IP Address. It can be a maximum of 64 characters. Default: Blank.

- If SNMP version is set as **SNMPv1** or **SNMPv2c,** configure **Community Name**.

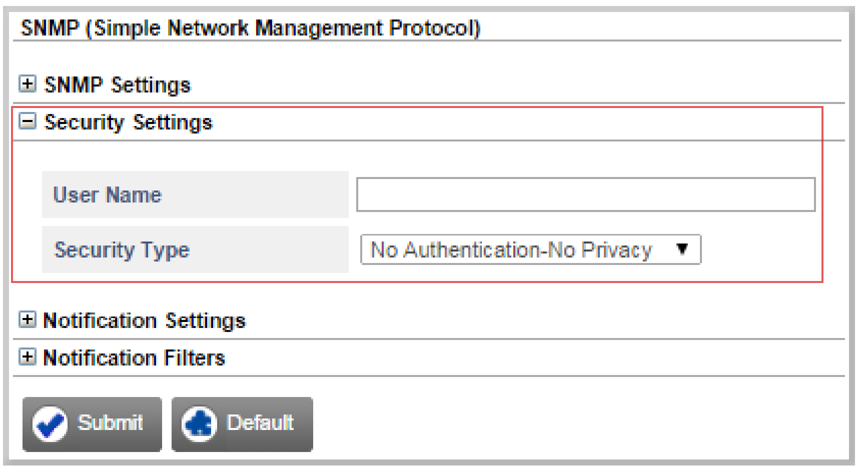
  Community Name identifies the SNMP community in which the sender and recipient of the message are located. It enables communication between SARVAM UMG and the Manager. The Community Name can be a maximum of 40 characters. Default: Blank.

- If SNMP version is set as **SNMPv3,** the **System's Engine ID** is displayed in this field. This is a unique identification of the system. It is a hexadecimal field with length of 22 characters.

  The ID consists of:
  - Enterprise Number (800086df03 which is fixed)
  - MAC Address of the system (MAC address of Network port)

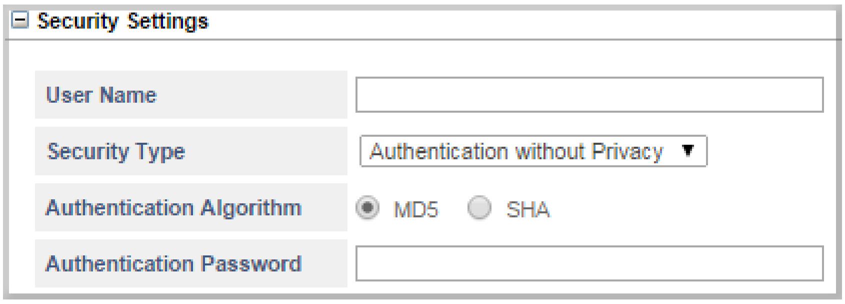
Security Settings

- If SNMP version is set as **SNMPv3**, click **Security Settings** to expand and configure the following.



  - Enter the **User Name**. The User Name can be a maximum of 40 characters. User Name will be used for authentication and privacy in SNMPV3.

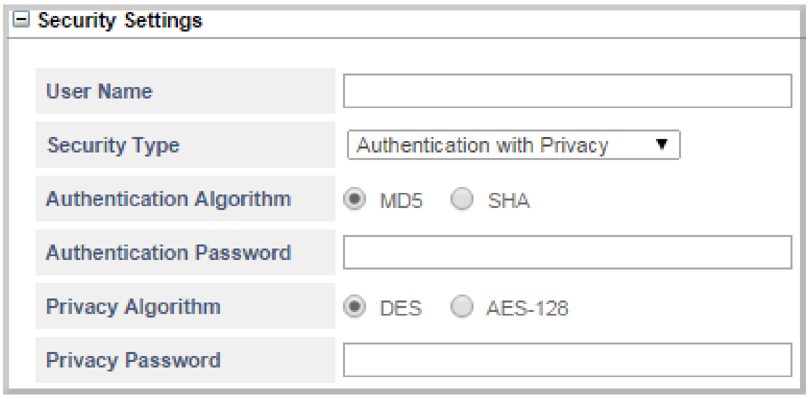
- Select the appropriate **Security Type** as per your requirement. Security Type defines the level of security.
  - When Authentication and Privacy are not required, select **No Authentication-No Privacy**
  - When only Authentication is required, select **Authentication without Privacy**. Incoming SNMP Messages will require authentication.



If you select this method, select the **Authentication Algorithm** as **MD5** or **SHA**. Default: MD5.

In the **Authentication Password**, enter a password of your choice as Authentication Password for the User Name you have assigned.The Authentication Password must be a minimum of 8 characters and may have upto 24 characters. Default: Blank.

  - When both Authentication and Privacy are required, select **Authentication with Privacy**. Incoming SNMP Message will require authentication and these messages will be encrypted, which will be decrypted at the receivers end only.

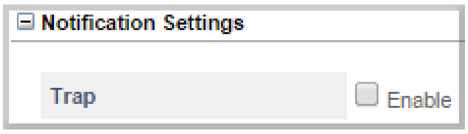


If you select this method,

- Select the **Authentication Algorithm** as **MD5** or **SHA**. Default: MD5.
- Enter **Authentication Password** for the User Name you have assigned. The Authentication Password must be a minimum of 8 characters and may have upto 24 characters. Default: Blank.
- Select the **Privacy Algorithm** as **DES** or **AES-128**. Default: DES.
- Enter the **Privacy Password** of your choice.The Privacy Password must be a minimum of 8 characters and may have upto 24 characters. Default: Blank.

Notification Settings

- Click **Notification Settings** to expand.



If SNMP version is set as **SNMPv1**, configure the following parameters.

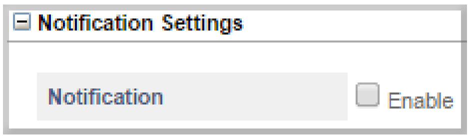
- If you want SARVAM UMG to generate Trap message for an error, select the **Trap** check box to enable. Default: Disabled.

    - You must configure the **Notification Destination**. SARVAM UMG will send the notification (error message) to the destination configured.

      The Notification Destination can be an IP Address or a Domain Name and the Port of the Manager or of any other device where you want to receive the trap messages. IP Address/Domain Name can be a maximum of 64 characters. Valid range is 0 to 65535. Default: 162.

- Click **Submit** button to save the settings.

If SNMP version is set as **SNMPv2c or SNMPv3**, configure the following parameters.



- Select the **Notification** check box to enable, if you want SARVAM UMG to generate Trap or Inform message for an error.

- Select the **Notification Type**. You may select **Trap** or **Inform**.

  If you want the system to send notification message without acknowledgement, select **Trap**.

  If you want the system to send notification message with acknowledgement, select **Inform**.

    - If you select **Inform** as the *Notification Type*, you must configure Retry Attempts and Retry Interval.

      If acknowledgement is not received from the Manager for the notification sent, the system will keep retransmitting the message for the number of attempts you have configured as the **Retry Attempts**. Default: 3.

The system will retransmit the messages at regular time intervals you have configured as **Retry Interval**. Default: 10 seconds.

- Configure the **Notification Destination**. SARVAM UMG will send the notification (error message) to the destination configured.

  The Notification Destination can be an IP Address or a Domain Name and the Port of the Manager or of any other device where you want to receive the trap messages. IP Address/Domain Name can be a maximum of 64 characters. Valid range is 0 to 65535. Default: 162.

- Click **Submit** button to save the settings.

Notification Filters

By default, you get error notifications, information and warnings for events related to the Application, Network and all Port Types. See table at the end of this topic for the event list. You can choose the type of notification you want by setting the notification filters.

To set filters,

Click **Notification Filters** to expand.

To disable any filter, clear the respective check box.

Make sure that you have uploaded the MIB in your SNMP Manager to get the status and notifications for SNMP. Contact Matrix Support Team for the MIB files.

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, analyze network problems, debug client/ server communications, debug network protocol implementations.

SARVAM UMG supports PCAP Trace, which you can use to detect and diagnose network related problems; for example, when the SIP account is not getting registered, or a SIP related feature is not functioning.

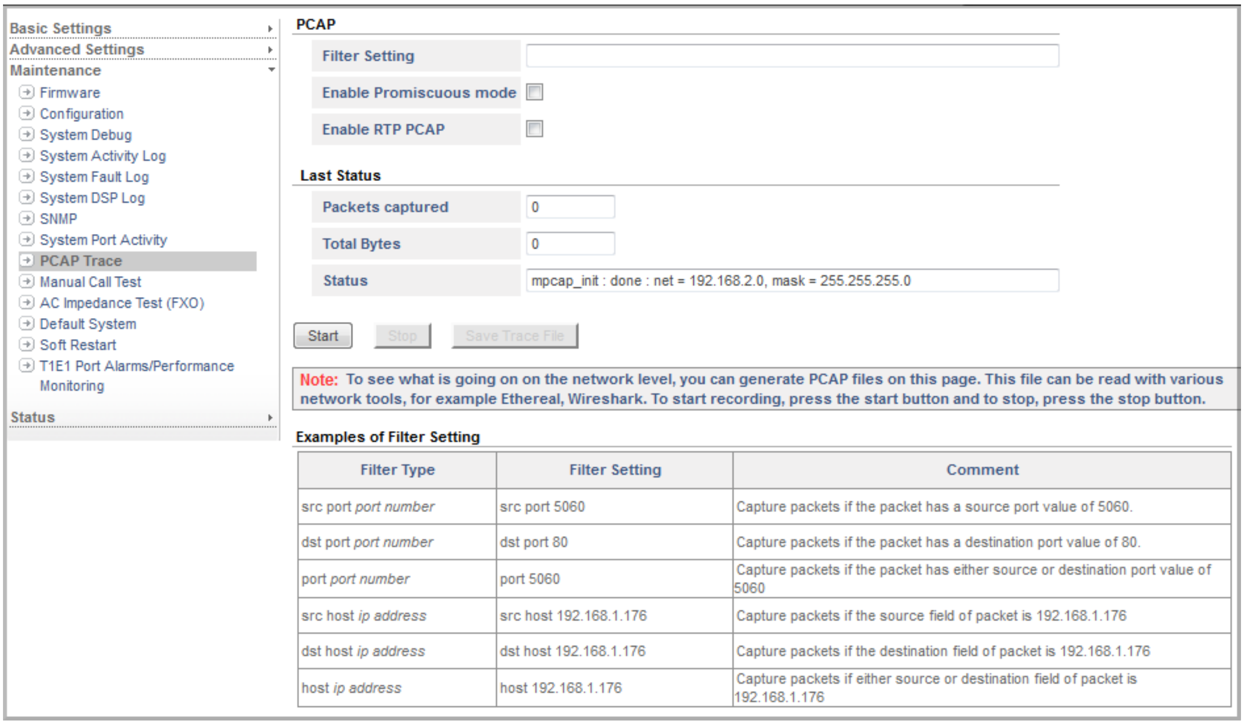
Packets traveling over a network are captured and saved in the system. You can save these trace files (packets captured by the system) on a computer and open these trace files using a graphical packet capture and protocol analysis tool such as Wireshark or Ethereal.

A maximum of 10 MB of packets can be captured and stored in the system.

SARVAM UMG also supports Filters and Promiscuous mode for capturing packets, which you can use to specify the types of data packets to be captured.

To use PCAP Trace,

- Click the **Maintenance** link to expand.
- Click the **PCAP** link.



- Decide the type of packets to be captured and set the Filters accordingly. The **Filter Settings** must be within 60 characters. By default, this field is blank. So, all packets will be captured.

Few examples of the Filter Settings are provided to you on this page to help you set the Filters as per your requirement.

It is not mandatory to set Filters. When the Filter Settings left blank, the system will capture all packets.

- You may enable **Promiscuous Mode** by selecting the check box. Default: Disabled.

  When you enable Promiscuous Mode, the SARVAM UMG will capture all network traffic. However, this will work only in a non-switched environment.

  When Promiscuous Mode is disabled, the system will capture only traffic that is directly related to it. Only traffic to, from or routed through the SARVAM UMG will be picked up by the PCAP Trace.

'Filter Settings' and 'Promiscuous Mode' (enabled) will not be cleared during power down.

- You may enable RTP PCAP by selecting the **Enable RTP PCAP** check box. Default: Disabled.

  If you want SARVAM UMG to capture RTP packets during PCAP for SIP calls, make sure you enable RTP PCAP check box.

- 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.

  The Number of Packets and bytes captured as per the filter setting will be displayed in the fields **Packets Captured** and **Total Bytes** respectively.

  The **Status** field displays the current activity of packet capturing.

Capturing of packets will not stop if you open any other page of Jeeves. So, you may continue using Jeeves for any other purpose while PCAP Trace is being used.

- When the packet capturing is stopped (by you or the system), click the **Save Trace File** button to save the files on your computer or on another computer.

  A dialog box opens. You can select the path for saving the trace file.

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.

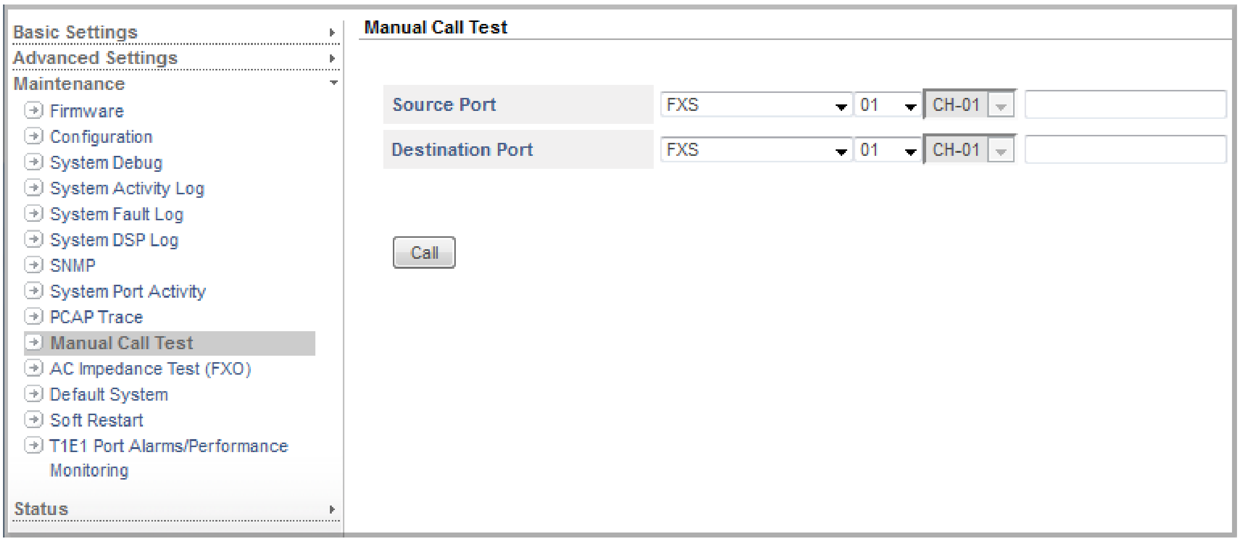
- After logging out of Jeeves, you can open the trace files using Wireshark/ Ethereal or any other software which supports opening of trace files.

Manual Call Test

Manual Call Test enables you to check the quality of Speech between two ports — Source Port and Destination Port — of SARVAM UMG without altering the existing call routing configuration.

To conduct Manual Call Test,

- Click the **Maintenance** link to expand.
- Click the **Manual Call Test** link.



In **Source Port**,
- Select the **Port Type** you want to test from the list.
- Select the **Port Number** you want to test from the list.
- Select the **Channel Number** for T1E1 and BRI Ports.
- Enter the **Phone Number** in the corresponding field. The phone number can be of maximum 16 characters. Valid characters are 0-9, *, #, + and dot (.).

In **Destination Port**,
- Select the **Port Type** you want to test from the list.
- Select the **Port Number** you want to test from the list.
- Select the **Channel Number** for T1E1 and BRI Ports.
- Enter the **Phone Number** in the corresponding field. The phone number must be a valid number that the system can outdial. It can be of maximum 16 characters. Valid characters are 0-9, *, #, + and dot (.).

- Click the **Call** button. SARVAM UMG will out dial the phone number you entered to make a test call between the Source Port and the Destination Port.
- As soon as the test call is made, the **System Port Activity** page will open. You can view the call states and status of the ports you are testing on this page.

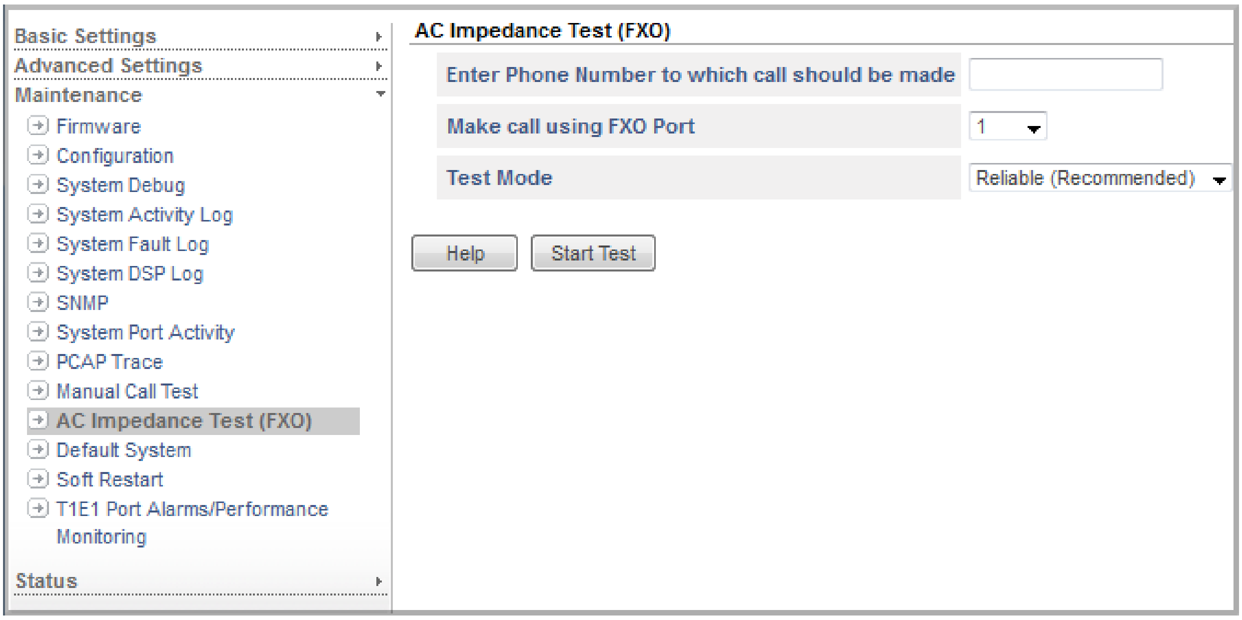
For more information on Call States and Port Status, see “System Port Activity”.

AC Impedance Test (FXO)

SARVAM UMG supports the AC Impedance Test for clear, audible and echo-free speech over FXO Ports. This test helps you to set the most appropriate values for the FXO Port Parameters — AC Termination Impedance, CO Termination and CO Line Type — to correct the line impedance mismatch between the AC Termination Impedance presented by the FXO Port of SARVAM UMG to the line and the CO Termination Impedance presented by the Central Office to the line.

To conduct the AC Impedance Test,

- Click the **Maintenance** link to expand.
- Click the **AC Impedance Test (FXO)** link.



- In **Enter Phone Number to which call should be made**, enter the phone number on which you want to make a test call. The number can be a landline or a mobile number. We recommend you to use a mobile number for the test call.

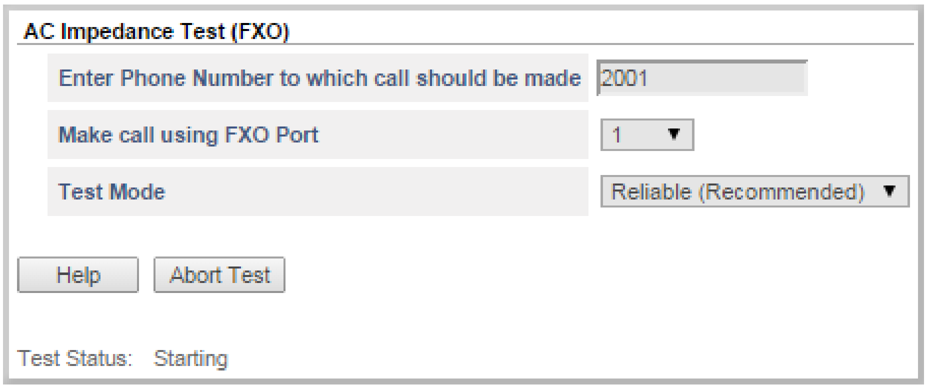
If you are using a mobile phone number, make sure the handset of the configured mobile number supports the Mute function.

- In **Make call using FXO Port**, select the FXO Port using which you want to make the test call. This must be the same FXO Port for which AC Impedance is to be set.
- Select the **Test Mode**. You may select **Reliable (Recommended)** or **Accurate**.

  The **Reliable Test** mode suggests the AC Impedance settings on the basis of most commonly used AC Impedances, CO Terminations and CO Line Types across the globe. The test using Reliable Test mode takes approximately 5 minutes to complete.

  The **Accurate Test** mode suggests the AC Impedance settings on the basis of all the possible AC Impedances, CO Terminations and CO Line Types across the globe. The test using the Accurate Test mode takes approximately 1 hour and 20 minutes to complete.

- Click the **Start Test** button. The system will call the phone number, you have configured. The message '*Starting* ' appears on your screen.



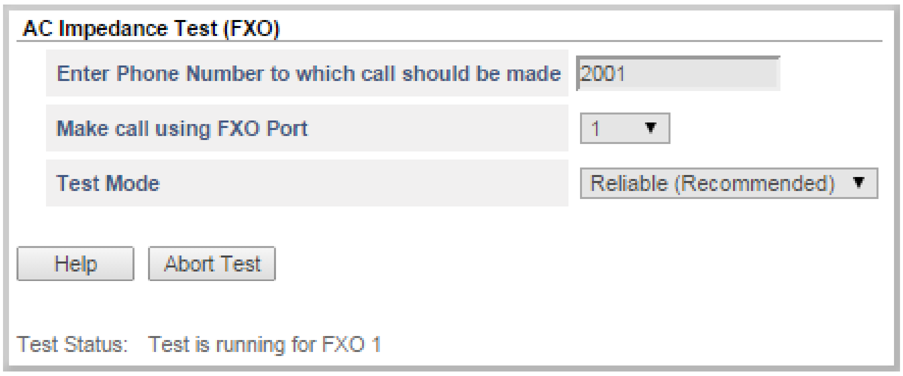
While the test is being conducted, you will hear pulsating tone on all the ports of the same card.

- Answer the test call from the telephone, you have configured.

  If you are using a Mobile phone, 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.

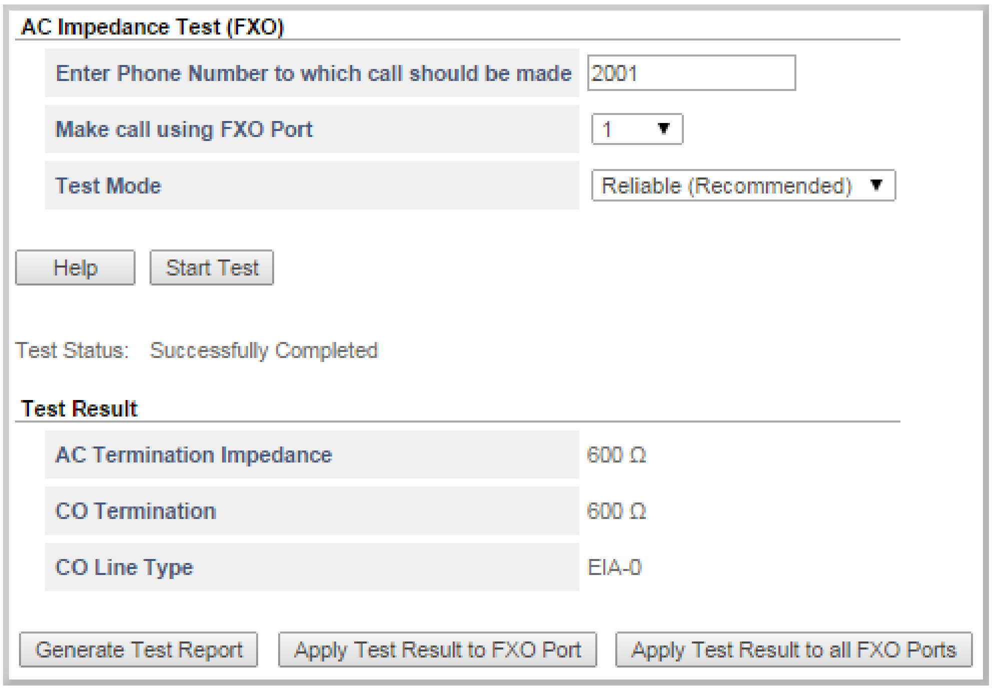
- After approximately 5 seconds, you will hear the test signals being transmitted by the system for the duration of the test. The message '*Test is running for FXO 1...*' appears on your screen.



  If you wish to abort the test midway, you may click the **Abort Test** button.

- On completion of the test, the system will automatically disconnect the call. The message '*Test Status: Successfully completed*' appears on the screen.

- At the end of the test, the page displays the **Test Result**. Suggested Impedance Settings for the AC Termination Impedance, CO Termination and CO Line Type to be configured for the FXO Port you have tested, appears on the screen as shown below.



- Click the **Apply Test Result to FXO Port** button, to apply the test result to the FXO Port you have tested.

- Click the **Apply Test Result to all FXO Ports** button, to apply the test result to all the FXO Ports of the system.

- Verify the settings by making a trial call. There should be no echo and speech should be audible and clear.

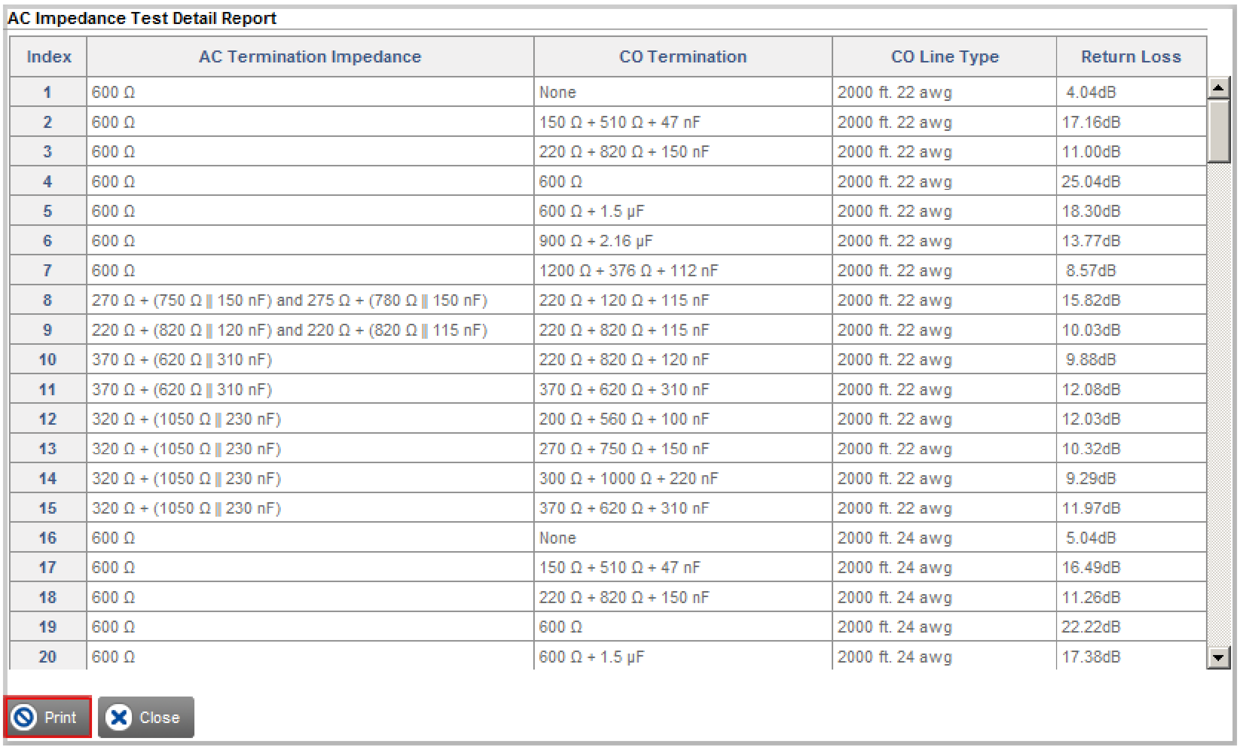
  If there is no echo/ mild echo, and the volume level is low/high, you may adjust the **Rx Gain** and **Tx Gain** of the FXO Port manually. See “Hardware Settings” under the “FXO Port” for details.

  If you still hear echo during the trial call, you may re-run the test using the **Accurate Test** mode.

It is possible that the AC Impedance Settings may differ for different CO Trunks subscribed from the same exchange. In such a case, you must run the test for each CO Trunk connected to the FXO Port separately and configure the settings accordingly.

- To generate the detailed test report, click the **Generate Test Report** button.

The detailed test report appears in a new window.

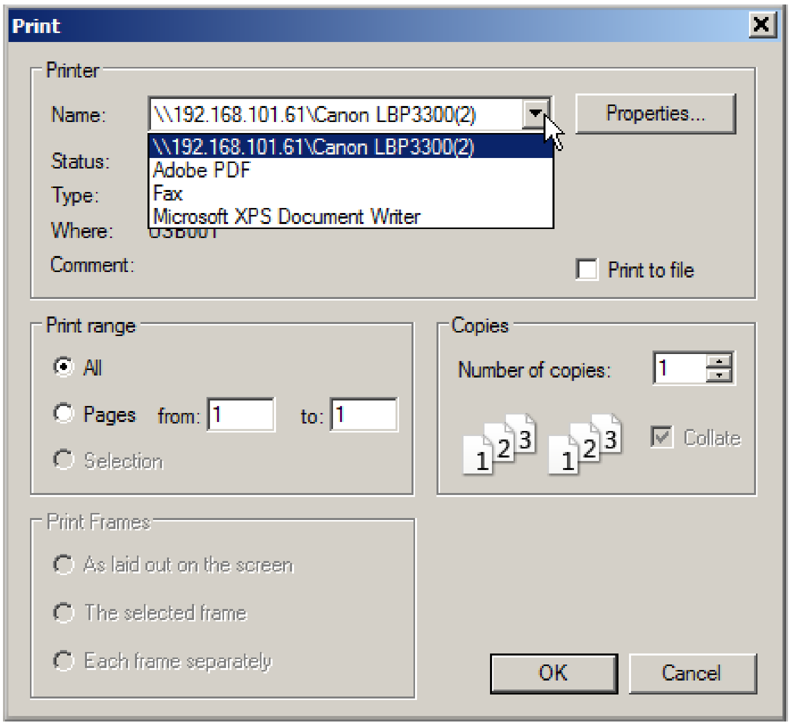


AC Impedance Test Detail Report

Index	AC Termination Impedance	CO Termination	CO Line Type	Return Loss
1	600 Ω	None	2000 ft. 22 awg	4.04dB
2	600 Ω	150 Ω + 510 Ω + 47 nF	2000 ft. 22 awg	17.16dB
3	600 Ω	220 Ω + 820 Ω + 150 nF	2000 ft. 22 awg	11.00dB
4	600 Ω	600 Ω	2000 ft. 22 awg	25.04dB
5	600 Ω	600 Ω + 1.5 µF	2000 ft. 22 awg	18.30dB
6	600 Ω	900 Ω + 2.16 µF	2000 ft. 22 awg	13.77dB
7	600 Ω	1200 Ω + 376 Ω + 112 nF	2000 ft. 22 awg	8.57dB
8	270 Ω + (750 Ω ‖ 150 nF) and 275 Ω + (780 Ω ‖ 150 nF)	220 Ω + 120 Ω + 115 nF	2000 ft. 22 awg	15.82dB
9	220 Ω + (820 Ω ‖ 120 nF) and 220 Ω + (820 Ω ‖ 115 nF)	220 Ω + 820 Ω + 115 nF	2000 ft. 22 awg	10.03dB
10	370 Ω + (620 Ω ‖ 310 nF)	220 Ω + 820 Ω + 120 nF	2000 ft. 22 awg	9.88dB
11	370 Ω + (620 Ω ‖ 310 nF)	370 Ω + 620 Ω + 310 nF	2000 ft. 22 awg	12.08dB
12	320 Ω + (1050 Ω ‖ 230 nF)	200 Ω + 560 Ω + 100 nF	2000 ft. 22 awg	12.03dB
13	320 Ω + (1050 Ω ‖ 230 nF)	270 Ω + 750 Ω + 150 nF	2000 ft. 22 awg	10.32dB
14	320 Ω + (1050 Ω ‖ 230 nF)	300 Ω + 1000 Ω + 220 nF	2000 ft. 22 awg	9.29dB
15	320 Ω + (1050 Ω ‖ 230 nF)	370 Ω + 620 Ω + 310 nF	2000 ft. 22 awg	11.97dB
16	600 Ω	None	2000 ft. 24 awg	5.04dB
17	600 Ω	150 Ω + 510 Ω + 47 nF	2000 ft. 24 awg	16.49dB
18	600 Ω	220 Ω + 820 Ω + 150 nF	2000 ft. 24 awg	11.26dB
19	600 Ω	600 Ω	2000 ft. 24 awg	22.22dB
20	600 Ω	600 Ω + 1.5 µF	2000 ft. 24 awg	17.38dB

Print Close

- You may print the report by clicking the **Print** button in the test report window.
- Select your Printer in the Printer options.



- You can also save the report in PDF format by selecting the **Adobe PDF** in the Printer options.

Default System

You can restore the system configuration to default values:

- using the Web Jeeves.
- using the System Command.
- by changing the Jumper position.

Restoring Default Settings using Web Jeeves

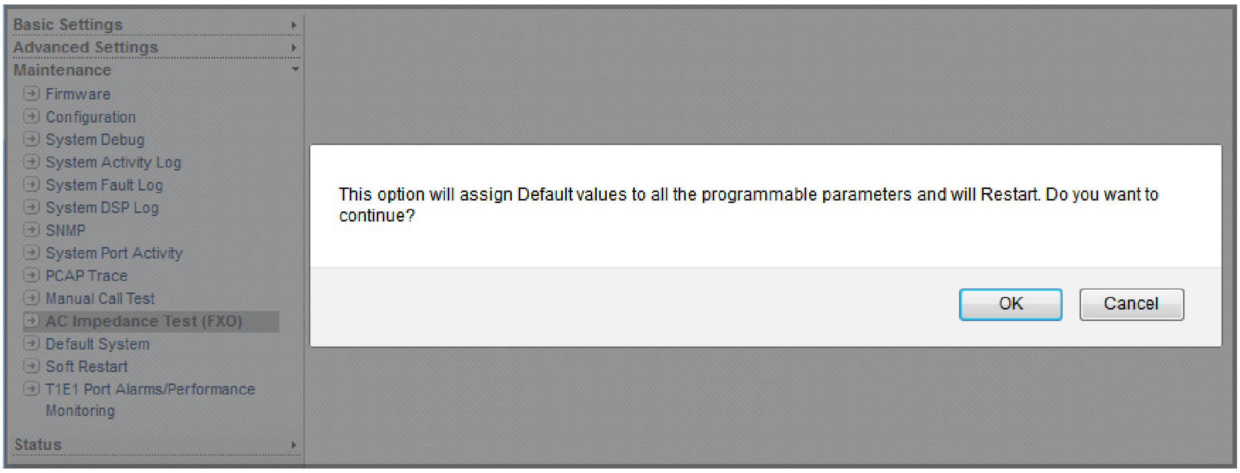
When you restore default settings using the Web Jeeves, all the parameters will be assigned default values **except** the following:

- Real Time Clock
- Call Detail Records
- Region
- Language
- Network
  - Connection Type
  - DNS Settings
  - DYN DNS
- System Parameters - NAT
  - Route Public IP Address
  - STUN Server Address
  - STUN Server Port
- System Parameters - Server Ports
  - HTTP Web Server Port
  - HTTPS Web Server Port
  - FTP Server Port
  - Telnet Server Port
- SIP Trunk
  - Allowed IP Address for Incoming SIP Message.
  - NAT Type
- Mobile Port
  - SIM PIN
- Firmware Parameters
- Configuration Parameters
- Login Password (Jeeves and Command)

To restore the default settings using the Web Jeeves,

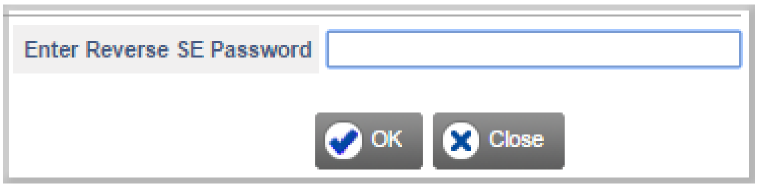
- Click the **Maintenance** link to expand.

- Click the **Default System** link.



- An alert message will appear, "**This option will assign default values to all the programmable parameters and will Restart. Do you want to continue?**".

- Click **OK**.



- You will be prompted to enter the reverse SE password. Enter reverse SE password. For example, if your password is Matrix@1234, enter 4321@xirtam. Click **OK**. The system will restart.

Restoring Default Settings using System Command

When you restore default settings using the System Command, all the parameters will be assigned default values **except** the following:

- Real Time Clock
- Call Detail Records
- Region
- Language
- Network
  - Connection Type
  - DNS Settings
  - DYN DNS
- System Parameters - NAT
  - Route Public IP Address
  - STUN Server Address
  - STUN Server Port
- System Parameters - Server Ports
  - HTTP Web Server Port
  - HTTPS Web Server Port
  - FTP Server Port
  - Telnet Server Port

- SIP Trunk
  - Allowed IP Address for Incoming SIP Message.
  - NAT Type
- Mobile Port
  - SIM PIN
- Firmware Parameters
- Configuration Parameters
- Login Password (Jeeves and Command)

To restore the default settings by dialing the system command,

- Lift the handset of phone connected to the FXS Port.
- Dial ***#19-Command Password*** to enter the Programming Mode.
- You will get programming tone.
- Dial ***51-Reverse Command Password-#****

  For example, if your password is 5699, enter 9965.
- Replace handset of the phone.
- The system will restart.

Restoring Default Settings by changing the Jumper Position

By changing the position of **Jumper J1** on the PCB on the CPU Card, you can restore the following parameters to default values:

- SE Password
- LAN Port Parameters
  - IP Address
  - Subnet Mask
- System Parameters - Server Ports
  - HTTP Web Server Port
  - HTTPS Web Server Port
  - FTP Server Port
  - Telnet Server Port

To restore the Default Jeeves/Command Password by changing the Jumper **(J1)** Settings on the CPU Card,

- Switch off the power supply.
- Remove the CPU Card from the slot.
- Locate Jumper **(J1)** on the PCB of the CPU Card.
- Change the position of the Jumper from ***BC*** to ***AB*** (that is, from 'Normal' to 'Reset Login Password')
- Reinsert the CPU Card into the slot.
- Switch ON the system and wait for the system to initialize.
- Switch OFF the system and remove the CPU Card from the slot.
- Change the Jumper position from ***AB*** to the original position ***BC.***
- Insert the CPU Card back into the slot.
- Switch ON the system.

Soft Restart

If you need to restart SARVAM UMG, you may do it

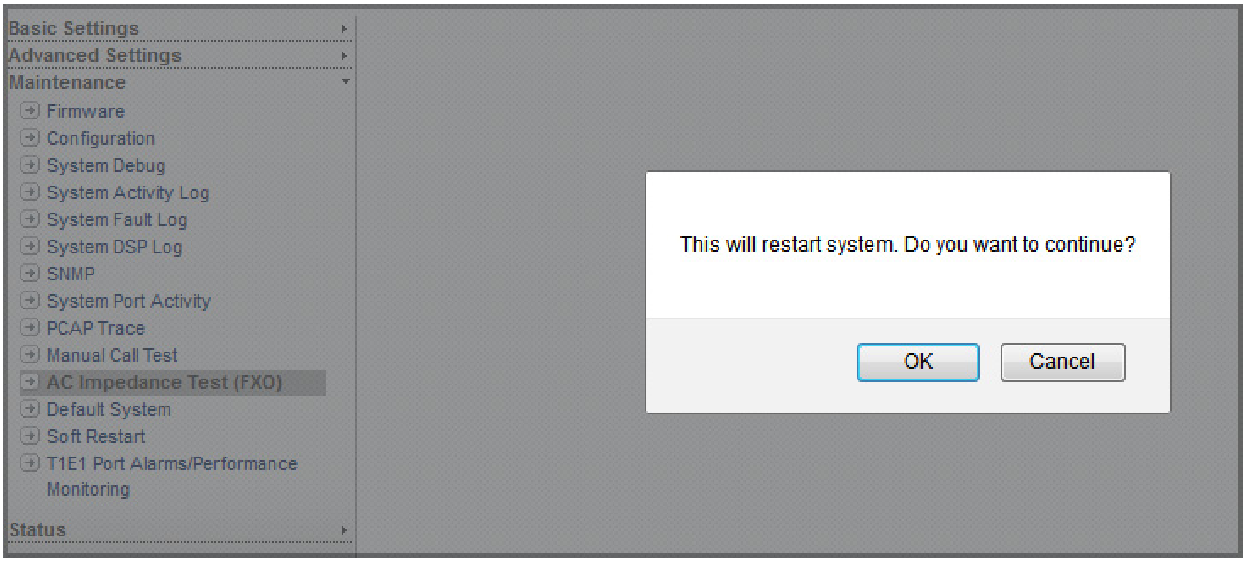
- using the Jumper
- using *Soft Restart* from Jeeves

When you restart the system, all active calls will be disconnected and the ports in use will be released. The system configuration however, will remain unaffected.

To use Soft Restart,

- Click the **Maintenance** link to expand.

- Click the **Soft Restart** link.



- An alert message will appear, “**This will Restart System. Do you want to continue?”**

- Click **OK** to restart the system.

T1E1 Port Alarm/ Performance Monitoring

T1E1 Port Alarms

SARVAM UMG supports the following alarms to detect the errors occurring on the T1E1 Port.

- RED Alarm (Loss of Signal)
- YELLOW Alarm (Remote Alarm Indication)
- BLUE Alarm (Alarm Indication Signal)

RED Alarm

- This alarm is generated if Loss of Signal persists for 2.5 seconds.
- When RED Alarm is declared, Yellow Alarm is sent to the far end within 12ms of detection of Loss of Signal.
- RED Alarm is declared if,
  - received signal is more than 20 dB or 40 dB below nominal for at least 1ms.
  - 10 consecutive zeroes are received.
  - Loss of frame alignment occurs.
- This alarm is cleared when the signal is acquired back and persists for 10 seconds.

YELLOW Alarm

- This Alarm is also known as Remote Alarm Indication.
- This Alarm is generated when Yellow Alarm is sent by the far end (Yellow Alarm is sent by the far end to indicate that it has lost the incoming signal).
- Yellow Alarm is declared when the signal corresponding to Yellow Alarm persists for 0.5 seconds.
- This alarm is cleared when No Yellow Alarm signal persists for 0.5 seconds.

BLUE Alarm

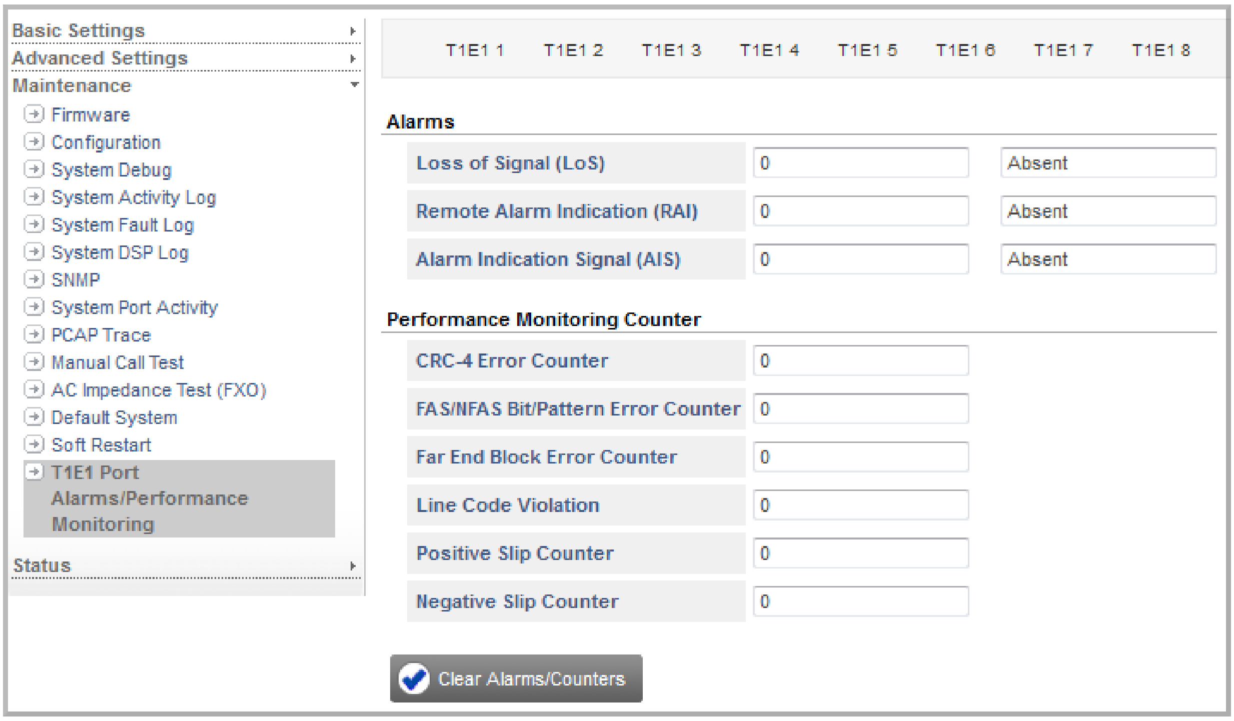
- It is also known as Alarm Indication Signal (AIS).
- This alarm is generated when AIS persists for 2.5 seconds.
- Blue Alarm (AIS) is declared if less than six zeroes are received on the incoming line data during a 3 msec interval. AIS is cleared if the above condition does not exist for 3 msec. This interval of 3 msec can extend upto a maximum of 75 msec.
- When BLUE Alarm is declared, Yellow Alarm is sent to the far end.
- This alarm is cleared when clearance of AIS is detected for continuous 10 seconds.

You can view the status of these alarms in the Jeeves. To view the status,

- Click the **Maintenance** link to expand.

- Click the **T1E1 Port Alarms/Performance Monitoring** link.

- Click the tab of the desired T1E1 port.



- Whenever, any alarm is detected by the system, it will increment the counter for that alarm and display its status as **Present**. When that alarm is cleared, the system will change the status of that alarm to **Absent**.

  By default, the value of each counter is 0. The maximum value for each counter is 255. Once the counter value reaches 255, the system will continue to display this value until you clear it.

- Click the **Clear Alarms/Counters** button to clear the counter value. You must clear the counter value, if you change any settings of the system or the line connection.

The page refreshes automatically after every 10 seconds to display the new status.

Performance Monitoring Counter

All errors do not generate an alarm. A few severe errors generate alarms while for others, error counters are supported by the ISDN chip in the system hardware. This error counter is used for Performance Monitoring.

Various Error Counters supported by the ISDN chip in E1 Mode are given in the table below.

Counter	Meaning
Frame Alignment Signal Error Counter	This counter is incremented on receipt of each errored FAS.
Far End Block Error Counter	This counter is incremented when either E1 or E2 bit is set in the transmit frame.
CRC-4 Error Counter	This counter is incremented when the received frame has CRC-4 errors.
Line Code Violation	This counter is incremented when a line code violation error occurs.
Positive Slip Counter	This counter is incremented every time a positive slip occurs.

Counter	Meaning
Negative Slip Counter	This counter is incremented every time a negative slip occurs.

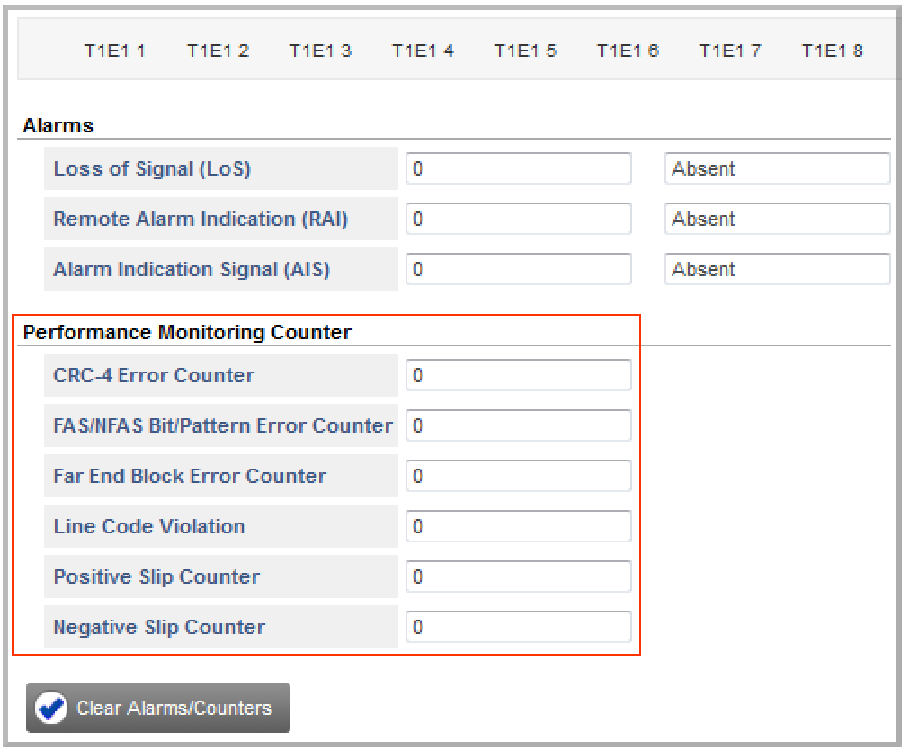
Various Error Counters supported by the ISDN chip in T1 Mode are given in the table below.

Counter	Meaning
Framing Alignment Bit Error Counter	This counter is incremented on receipt of any error in the framing pattern. In D4, Ft errors are counted. (Fs errors are counted if enabled) In ESF, any error in the 001011 framing pattern increments this counter.
Out of Frame Synchronization Error Counter	Out Of Frame (OOF) - Out of Frame is the occurrence of a particular density of framing error events. For D4 framing, OOF is declared when the receiver detects two or more framing errors within 0.75 msec or two or more errors out of five or fewer consecutive framing bits. It ends when there are fewer than two frame bit errors within a 0.75 msec period. For ESF framing, OOF is declared when the receiver detects two or more framing errors within 3ms or two or more errors out of five or fewer consecutive framing bits. It ends when there are fewer than two frame bit errors within a 3 msec period.
CRC-6 Error Counter	This counter is incremented when the received frame has CRC-6 errors. This is applicable for ESF framing only.
Line Code Violation	This counter is incremented when a bipolar violation error occurs or when excessive zeroes event occurs.
Positive Slip Counter	This counter is incremented every time a positive slip occurs.
Negative Slip Counter	This counter is incremented every time a negative slip occurs.

You can view these Error Counters in the Jeeves for monitoring the performance of T1E1 Port.

- Click the **Maintenance** link to expand.

- Click the **T1E1 Port Alarms/Performance Monitoring** link.



- Whenever an error is detected by the system, it will increment the counter for that error.

  By default, the value of each counter is 0. The maximum value for each counter is 255. Once the counter value reaches 255, the system will continue to display this value until you clear it.

- Click the **Clear Alarms/Counters** button to clear the counter value. You must clear the counter value, if you change any settings of the system or the line connection.

The page refreshes automatically after every 10 seconds to display the new status.

CHAPTER 13 Status

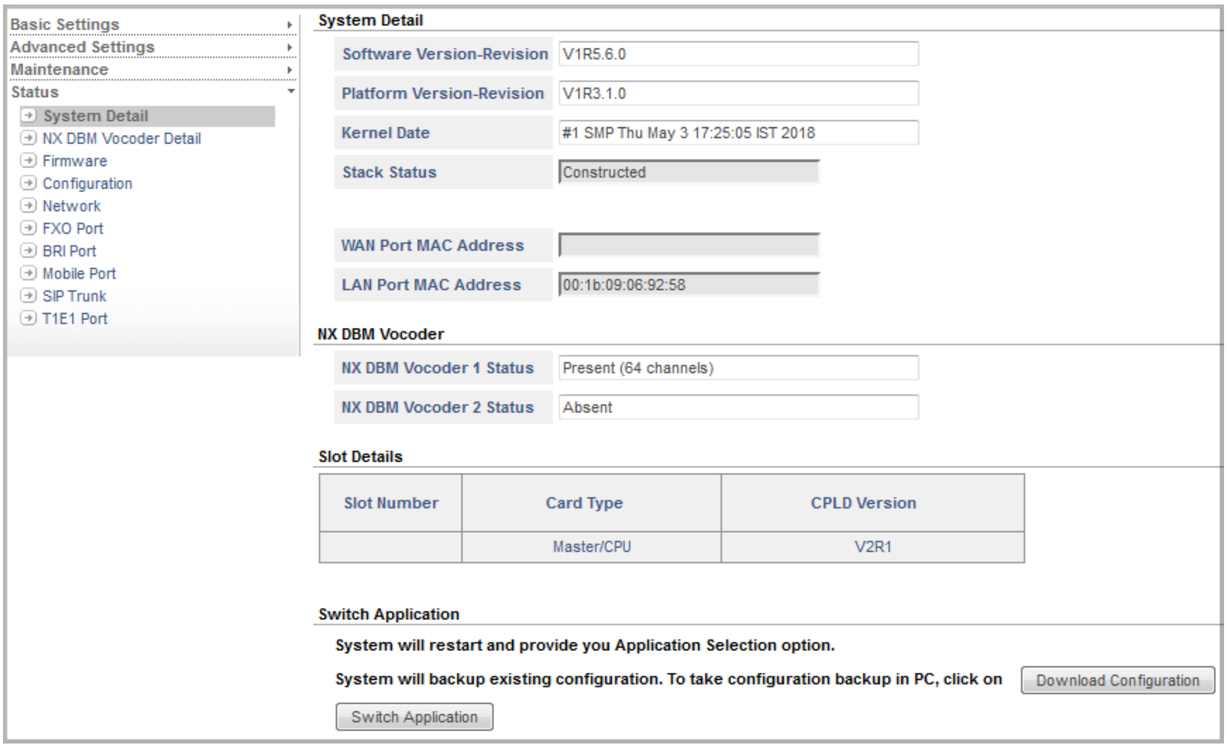
You can view the System Details, NX DBM Vocoder Details and the status of Auto Firmware upgrade, Auto Configuration upgrade, LAN Port, WAN Port, SIP Trunks, Mobile Ports, FXO Ports, BRI Ports and T1E1 Ports from Jeeves.

To view status,

- Click the **Status** link to expand.

System Detail

- Click the **System Detail** link.



The following **System Details** will be displayed on this page.

- **Software Version-Revision:** This displays the current version and revision of the firmware of SARVAM UMG.

- **Kernel Date:** This displays the Kernel compilation date.

- **Stack Status:** This displays the SIP Stack Status.

- **WAN Port MAC Address:** This displays the factory set MAC Address of the WAN Port.

If you have cloned the MAC Address of the WAN Port, you can view it in Network Status.

- **LAN Port MAC Address:** This displays the factory set MAC Address of the LAN Port.

If you have cloned the MAC Address of the LAN Port, you can view it in Network Status.

NX DBM Vocoder

The NX DBM Vocoder displays the status of the Vocoder modules present in the system.

NX DBM Vocoder 1 Status: This displays the status of NX DBM Vocoder 1.

NX DBM Vocoder 2 Status: This displays the status of NX DBM Vocoder 2.

Slot Details

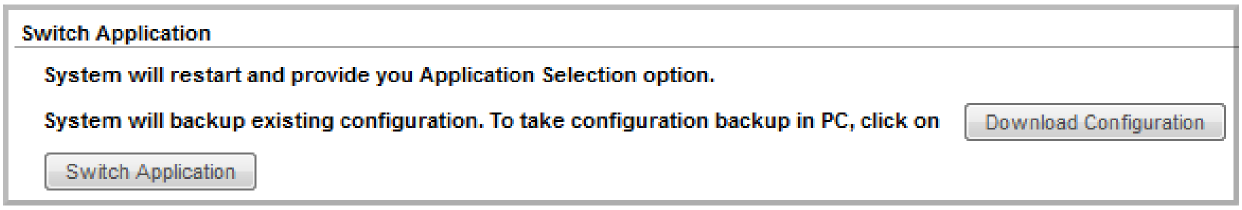
The Slot Details displays the cards which are installed in the system along with the Slot Number they are installed in. It also displays their respective CPLD Version.

Slot Details

Slot Number	Card Type	CPLD Version
	Master/CPU	V1R1
1	FXS 20	
4	FXO 4, FXS 16	
6	BRI 4	
7	GSM 4	
9	FXS 16	

Switch Application

If you want to change the application running on the ETERNITY GENX platform, click on the **Switch Application** button.



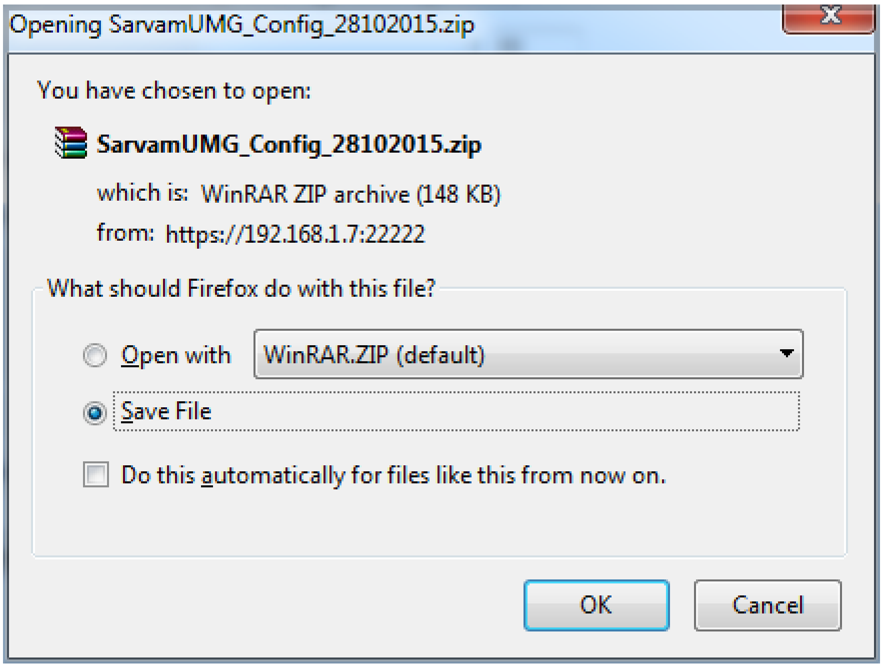
By clicking on the **Switch Application**, you will be diverted to the ETERNITY GENX Login page.

Select the Application Selection link and then the desired application you wish to run on the ETERNITY GENX Platform. Click on Next button. See "Application Selection".

You must login with the same password you kept for the Application.

The system will backup the existing configuration you did on this application. If you wish to take the backup of configuration in your PC, click on the **Download Configuration** button.

The **SarvamUMG_Config_ddmmyyyy.zip** window will open; where ddmmyyyy signifies the current date.

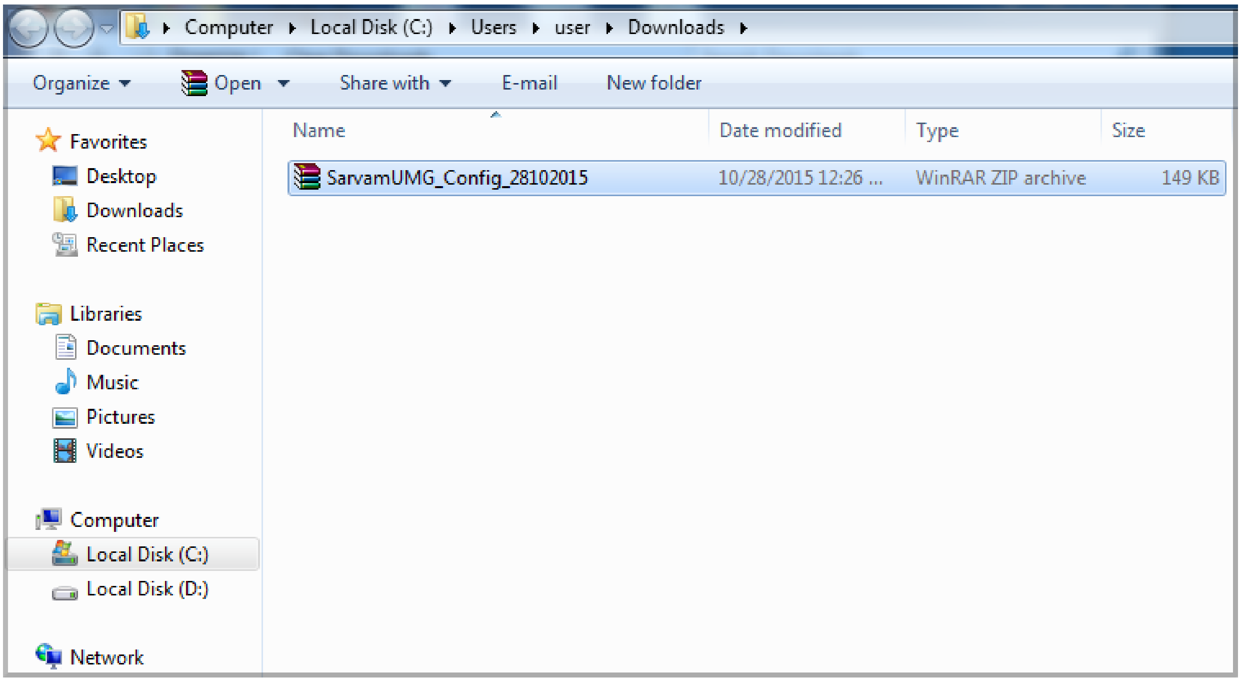


- You can either open the zip file or save the file to a location.

*The above window display depends upon the browser you are using. Check the **Download Settings** of your browser and set the Download path accordingly.*
OR
If your browser does not ask you for the location you want to save your file, it saves it in the default location according to the download path specified for that browser.

*If you are using Mozilla Firefox (version 3.5 recommended), before you save the configuration files, set the **Downloads** option of your browser as **Always ask me where to save files**.*

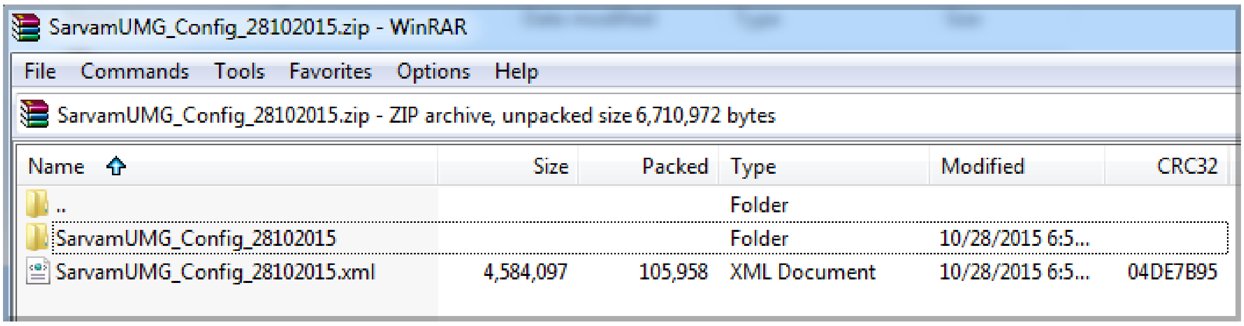
- Save the file on the local disk.



Save the back up configuration files by tagging the file name with the Version-Revision of the Firmware and tag the name of the backup folder on your computer with the date. This will help you at the time of restoring the back up configuration files.

- Open the configuration file (.zip) from the location you saved.

  The zip file contains all the system configuration files in .cfg format and xml format. You cannot edit the configuration files in .cfg format, however you can edit the configuration files in xml format and then upgrade the system with it.



- Open the SarvamUMG_Config_28102015 folder to view the configuration files.

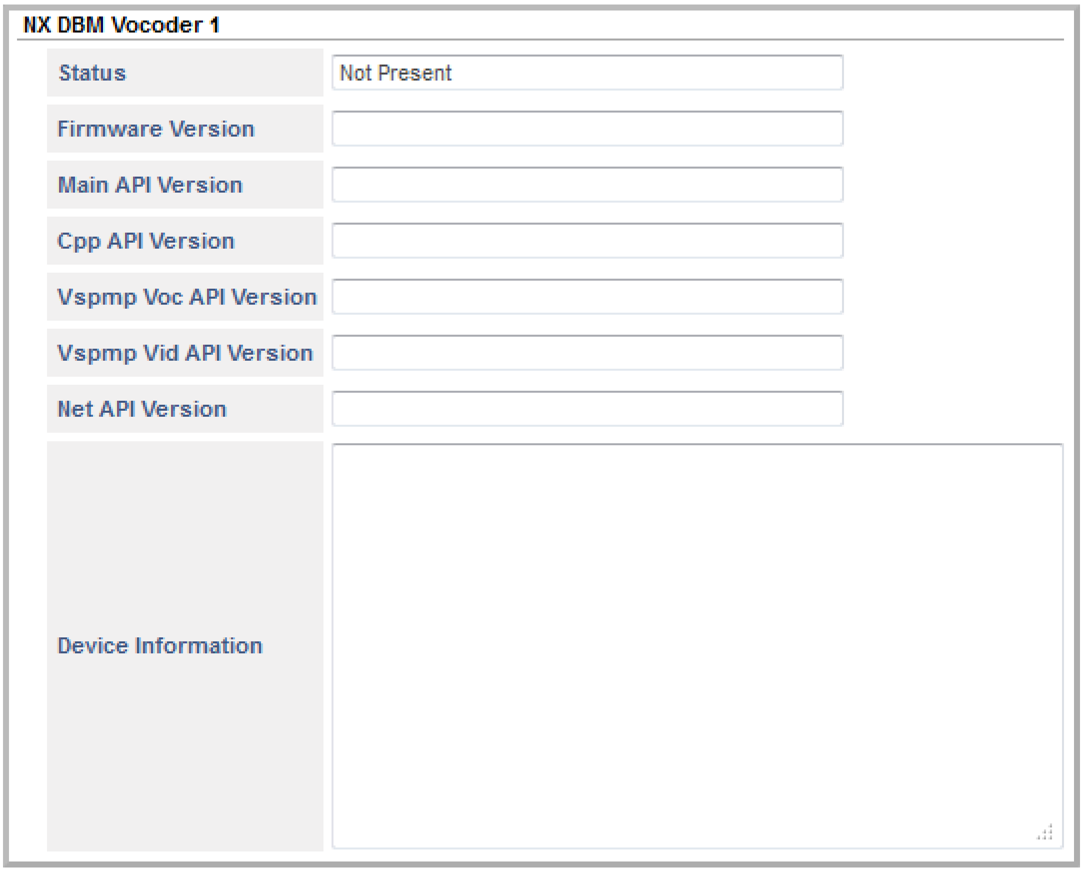
SarvamUMG_Config_27102015.zip\SarvamUMG_Config_27102015 - ZIP archive, unpacked size 6,618,867 bytes

Name	Size	Packed	Type	Modified	CRC32
AccesCode.cfg	456	271	File cfg	10/27/2015 7:0...	E39F0897
ActivityLog.cfg	22	7	File cfg	10/27/2015 7:0...	6339D68E
AntTable.cfg	9,818	49	File cfg	10/27/2015 7:0...	21AAA252
AutoUpgrade.cfg	1,384	366	File cfg	10/27/2015 7:0...	57126684
BriCalling_FixDestNum.cfg	33,526	92	File cfg	10/27/2015 7:0...	306B4A94
BriCallingNumTable.cfg	26,038	164	File cfg	10/27/2015 7:0...	B0E8BE7F
BriDdiDestNum.cfg	52,032	92	File cfg	10/27/2015 7:0...	B8073AA3
BriDestNumTable.cfg	92,038	400	File cfg	10/27/2015 7:0...	9EAE08C9
BriHuntGrp.cfg	7,866	129	File cfg	10/27/2015 7:0...	B7184A4B
BriPara.cfg	102,770	1,399	File cfg	10/27/2015 7:0...	C1EF043A
CallProgTone.cfg	668	190	File cfg	10/27/2015 7:0...	E9F585F4
CdrData.info	1,406	231	File info	10/27/2015 7:0...	6C1D4C45
CdrFilter.cfg	594	318	File cfg	10/27/2015 7:0...	07919932
CdrMetaData.info	6	6	File info	10/27/2015 7:0...	0E60D875
CmGlobalCnfgFile.cfg	80	49	File cfg	10/27/2015 7:0...	5282B1CF
Debug.cfg	476	122	File cfg	10/27/2015 7:0...	C7B5595C
Debug.info	0	0	File info	10/27/2015 7:0...	00000000
DebugMeta.info	10	5	File info	10/27/2015 7:0...	0C480348
DialPlanTable.cfg	33,306	108	File cfg	10/27/2015 7:0...	561F4990
DigestAuthTable.cfg	60,520	94	File cfg	10/27/2015 7:0...	8E795095
DistRingType.cfg	262	47	File cfg	10/27/2015 7:0...	8F978051
DSP.info	0	0	File info	10/27/2015 7:0...	00000000
DSPMeta.info	10	5	File info	10/27/2015 7:0...	0C480348
Dst.cfg	126	50	File cfg	10/27/2015 7:0...	E415CF4C
EmrgncyNum.cfg	140	29	File cfg	10/27/2015 7:0...	A6C51367
FaultLog.cfg	22	7	File cfg	10/27/2015 7:0...	6339D68E
FxoCalling_FixDestNum.cfg	33,526	92	File cfg	10/27/2015 7:0...	306B4A94
FxoCallingNumTable.cfg	26,038	164	File cfg	10/27/2015 7:0...	B0E8BE7F

- 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.

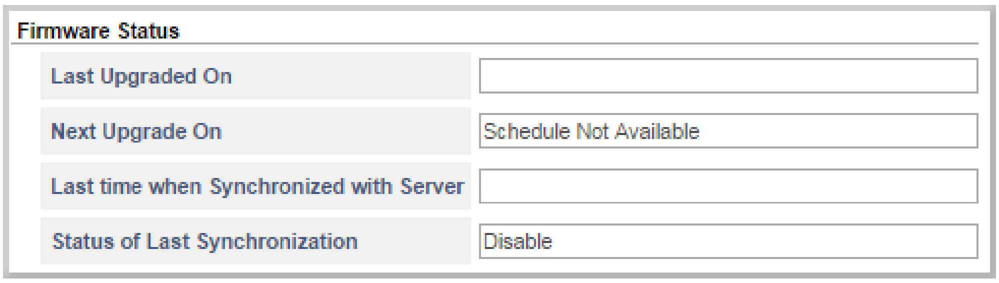
NX DBM Vocoder Detail

NX DBM Vocoder Module status is displayed on this page.



Firmware

- Click the **Firmware** link.



The following information related to Auto-Firmware upgrade will appear on your screen.

- **Last Upgraded On:** This displays the firmware with which SARVAM UMG last upgraded itself through the provisioning server, along with the date (DD:MM:YYYY) and time (HH:MM) of the upgradation.

- **Next Upgrade On:** This displays the date (DD:MM:YYYY) and time (HH:MM), when SARVAM UMG will again check for new firmware updates on the server.

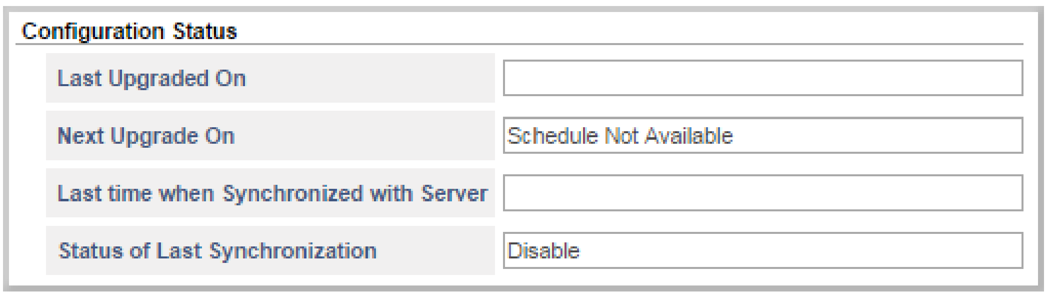
- **Last time when Synchronized with Server:** This displays the date (DD:MM:YYYY) and time (HH:MM), when SARVAM UMG last synchronized with the server for new firmware updates.
- **Status of Last Synchronization:** This displays the status of last synchronization. The possible status messages that will be displayed are listed in the table below.

Possible Responses	Event
Invalid Parameters	When parameters are not valid.
Local Failure	When internal error occurs, like Thread Creation failed.
Resolving Server Address	When IP Address is not found using DNS query.
Server Not Found	When server is not connected after the expiry of Retry Timer and Retry Counter.
Send Request Failed	When there is Curl Internal Error
Connecting to Server	When system is establishing TCP connection with server until the expiry of Retry Timer and Retry Counter.
TCP Connection Failed	When no response is received for TCP connection until expiry of Retry Timer and Retry Counter.
Connection Failed	When no response is received for TCP connection after expiry of Retry Timer and Retry Counter.
	When there is an open SSL error. When the maximum file size is exceeded. When there are too many Redirect or illegal operation from curl response.
Permission Denied	When access is denied. When there is permission problem on the server. When login fails.
Downloading Firmware Index File	When the system is retrieving Firmware Index file.
Downloading Firmware	When the system is retrieving Firmware zip file.
File Not Found	When the remote file is not found.
Waiting for Firmware File Name	When *Check Firmware Available on Server* button is clicked manually and the list of available firmware is presented.

Possible Responses	Event
No File Found for Up-gradation	When selected firmware benchmark is not found. When user does not select the firmware name manually. When matrix_firmware.html file is received but current product name is not found from this file. Single firmware name is received in matrix_firmware.html but this benchmark file does not match with current firmware benchmark. Multiple firmware names are received but all files are below the current firmware.
Firmware Version Below	When the received firmware version is below the current firmware version.
Firmware Version Same	When the received firmware version is same as the current firmware version.
Firmware Decryption Failed	When the firmware zip file decryption has failed. When the firmware file name does not match or benchmark is less than the current firmware version-revision in the text file.
Auto Upgrade stop due to parameter change	When Auto Upgrade is in process and the Firmware parameters are changed.
Auto Upgrade stop by system	When Auto Upgrade process is stopped due to network restart.
Auto Upgrade Stop on User request	When Firmware upgrade process is started manually but user clicks Cancel button after display of list of firmware files.
Successfully Updated	When firmware is updated successfully.

Configuration

- Click the **Configuration** link.



The following information related to Auto-Configuration upgrade will appear on your screen.

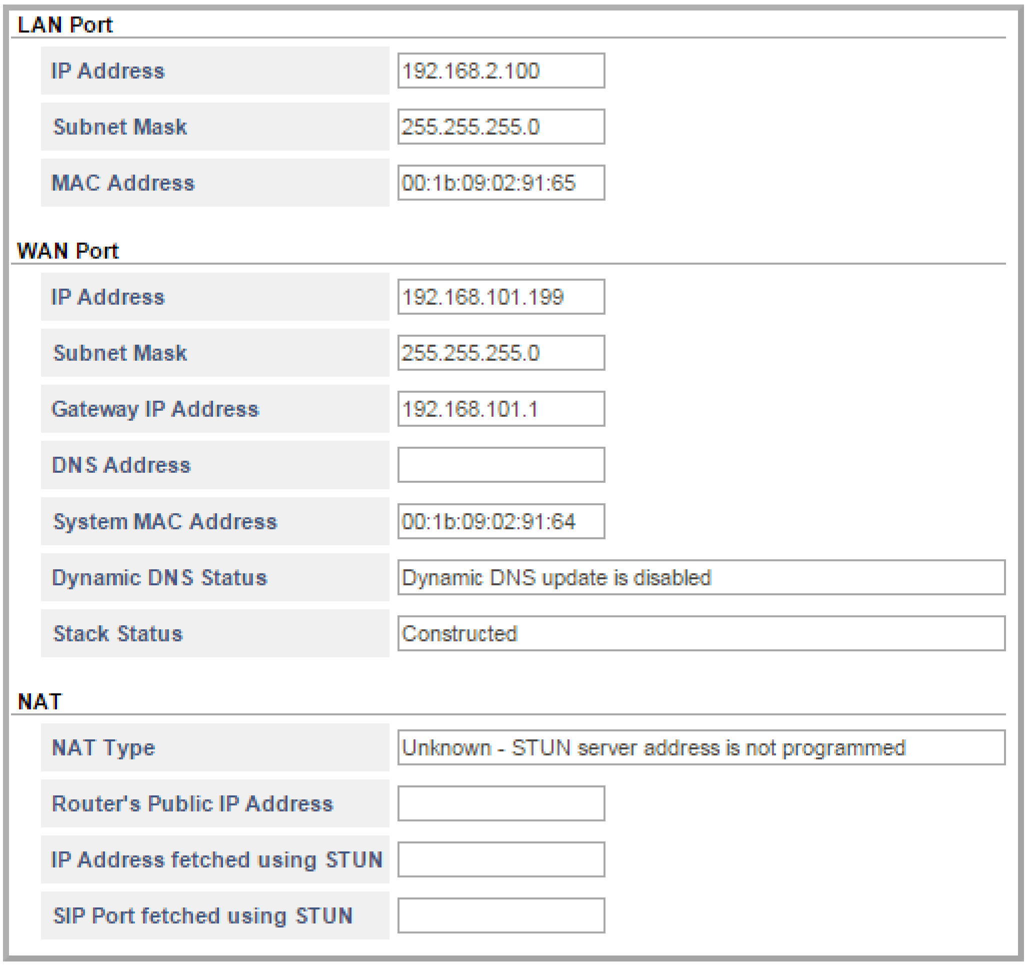
- **Last Upgraded On:** This displays the date (DD:MM:YYYY) and time (HH:MM), when SARVAM UMG last upgraded its configuration through the server.

- **Next Upgrade On:** This displays the date (DD:MM:YYYY) and time (HH:MM), when SARVAM UMG will again check for new configuration on the server.
- **Last time when Synchronized with Server:** This displays the date (DD:MM:YYYY) and time (HH:MM), when SARVAM UMG last resynchronized with the server for new configuration.
- **Status of Last Synchronization:** This displays the status of last synchronization. The possible status messages that may appear are listed in the table below.

Possible Responses	Event
Invalid Parameters	When parameters are not valid.
Local Failure	When internal error occurs, like Thread Creation failed.
Resolving Server Address	When IP Address is not found using DNS query.
Server Not Found	When server is not connected after the expiry of Retry Timer and Retry Counter.
Send Request Failed	When there is Curl Internal Error
Connecting to Server	When system is establishing TCP connection with server until the expiry of Retry Timer and Retry Counter.
TCP/TFTP Connection Failed	When no response is received for TCP/TFTP connection until expiry of Retry Timer and Retry Counter.
Connection Failed	When no response is received for TCP connection after expiry of Retry Timer and Retry Counter. When there is an open SSL error. When the maximum file size is exceeded. When there are too many Redirect or illegal operation from curl response.
Permission Denied	When access is denied. When there is permission problem on the server. When login fails.
Downloading Config File	When the system is retrieving config file.
File Not Found	When the remote file is not found.
Config Decryption Failed	When the config decryption has failed.
Config Parsing Failed	When the file parsing has failed. When the root tag is not found.
Successfully Updated	When configuration is updated successfully.

Network

- Click the **Network** link.



The current values of the following parameters will appear on your screen:

LAN Port

- **IP Address:** This displays the current IP address assigned to the LAN Port of SARVAM UMG.
- **Subnet Mask:** This displays current Subnet Mask assigned to the LAN Port of SARVAM UMG.
- **MAC Address:** This displays the MAC Address assigned to the LAN Port of SARVAM UMG.

WAN Port

- **Status:** This displays the status of the Ethernet Port of SARVAM UMG.
- **IP Address:** This displays the IP address assigned to the Ethernet Port of SARVAM UMG.
- **Subnet Mask:** This displays the Subnet Mask assigned to the Ethernet Port of SARVAM UMG.

- **Gateway IP Address:** This displays the Gateway Address assigned to the Ethernet Port of SARVAM UMG.

- **DNS Address:** This displays the DNS address.

- **System MAC Address:** This displays the MAC Address assigned to the Ethernet Port of SARVAM UMG.

*If you have cloned the MAC Address, **System MAC Address** will display the cloned MAC Address. You can view the factory set MAC Address in System Detail.*

- **Dynamic DNS Status:** This displays the response received from DDNS server while sending the IP Address update request to the server. The following are the responses which you may receive:

Possible Responses	Event
Please Wait….!!	When system is waiting for error/ successful response from DDNS server
Updated Successfully - IP Address	IP Address updated successfully in DDNS server
Host has been blocked	When 'abuse' is received
Authentication Fail	When authentication check is failed either problem in user id or password
No such host in the system	When 'no host' is received
Invalid hostname format	When 'notfqdn' is received
Host not in this account	When '!Yours' is received
DNS error encountered	When 'dnserr' is received
Server goes under schedule maintenance	When '911' is received
No Response	No response is received from DDNS server due to any reason
DDNS Failed	For all remaining cases
In all remaining cases, the default messages supported by DDNS client will appear in this.	

- **Stack Status:** This displays the SIP Stack Status.

NAT

- **NAT Type:** This displays the NAT Type, if STUN is enabled in SARVAM UMG. The commonly used NAT types are:
  - Unknown
  - Open
  - Conenat
  - Restrictednat
  - Portrestrictednat
  - Symmetricnat
  - Symmetricfirewall
  - Blocked

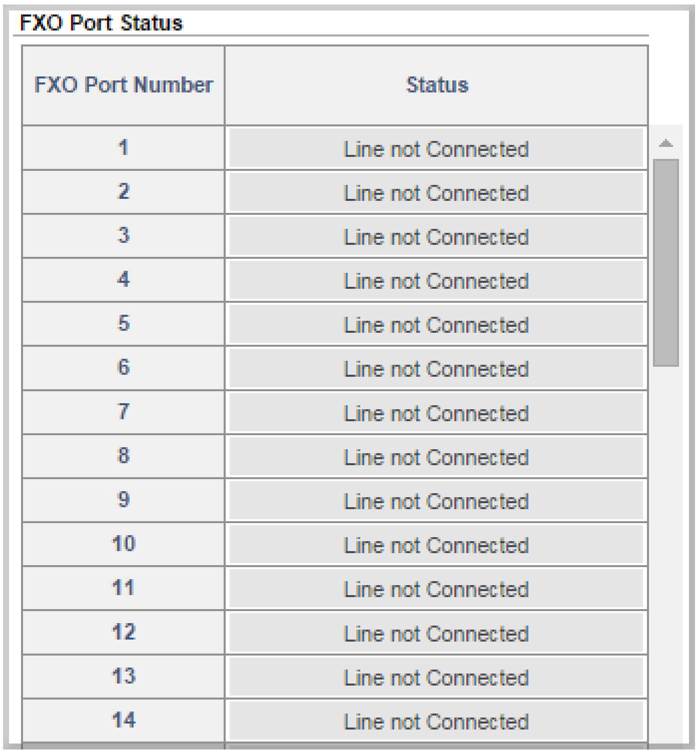
- **Router's Public IP Address:** This displays the Router's Public IP address programmed in the System Parameters. See “NAT” under *System Parameters.*

- **IP Address fetched using STUN:** This displays the IP address fetched using STUN, if STUN server address is programmed in the system.

- **SIP Port fetched using STUN:** This displays the SIP Port fetched using STUN, if STUN server address is programmed in the system.

FXO Port

- Click the **FXO Port** link.



FXO Port Status

FXO Port Number	Status
1	Line not Connected
2	Line not Connected
3	Line not Connected
4	Line not Connected
5	Line not Connected
6	Line not Connected
7	Line not Connected
8	Line not Connected
9	Line not Connected
10	Line not Connected
11	Line not Connected
12	Line not Connected
13	Line not Connected
14	Line not Connected

- The status of the FXO Ports appear on this page, as FXO Port ‘Line not Connected’ or ‘Line Connected’.

BRI Port

- Click the **BRI Port** link.

BRI Port Status

BRI Port Number	Layer 1	Layer 2
1	DOWN	DOWN
2	DOWN	DOWN
3	DOWN	DOWN
4	DOWN	DOWN
5	DOWN	DOWN
6	DOWN	DOWN
7	DOWN	DOWN
8	DOWN	DOWN
9	DOWN	DOWN
10	DOWN	DOWN
11	DOWN	DOWN
12	DOWN	DOWN
13	DOWN	DOWN
14	DOWN	DOWN
15	DOWN	DOWN
16	DOWN	DOWN

The following parameters will be displayed for the BRI Port.

- **Layer 1:** Displays if the link is up or down.

- **Layer 2:** Displays if the link is up or down.

Mobile Port

- Click the **Mobile Port** link.

Mobile Port

1-32 33-48

Port	Hardware Slot-Port	Name	Activity Status	Signal Strength	Registered with Network	Network Operator Code	IMEI
Mobile 1	0 - 0		Module Initialization	-0	Not Registered		
Mobile 2	0 - 0		Module Initialization	-0	Not Registered		
Mobile 3	0 - 0		Module Initialization	-0	Not Registered		
Mobile 4	0 - 0		Module Initialization	-0	Not Registered		
Mobile 5	0 - 0		Module Initialization	-0	Not Registered		
Mobile 6	0 - 0		Module Initialization	-0	Not Registered		
Mobile 7	0 - 0		Module Initialization	-0	Not Registered		
Mobile 8	0 - 0		Module Initialization	-0	Not Registered		
Mobile 9	0 - 0		Module Initialization	-0	Not Registered		
Mobile 10	0 - 0		Module Initialization	-0	Not Registered		

The following parameters will be displayed for the Mobile Ports.

- **Port:** This displays the Mobile Port numbers.
- **Hardware Slot-Port**: This displays the Hardware Slot and Port numbers assigned to the Mobile software ports.
- **Name**: This displays the name assigned to the Mobile Port for identification.
- **Activity Status:** This displays port activity status listed below:
  - Module Initialization
  - SIM PUK Required
  - SIM PIN Wrong
  - SIM Absent
  - SIM Present

- Network Absent
- Network Present

- **Signal Strength:** This displays the current signal strength.
- **Registered with Network:** This displays the type of network with which the SIM is registered, that is 4G, 3G, GSM. If the SIM is not registered this displays the status as Not Registered.
- **Network Operator Code:** This displays the code of the network with which the Mobile Port is registered.

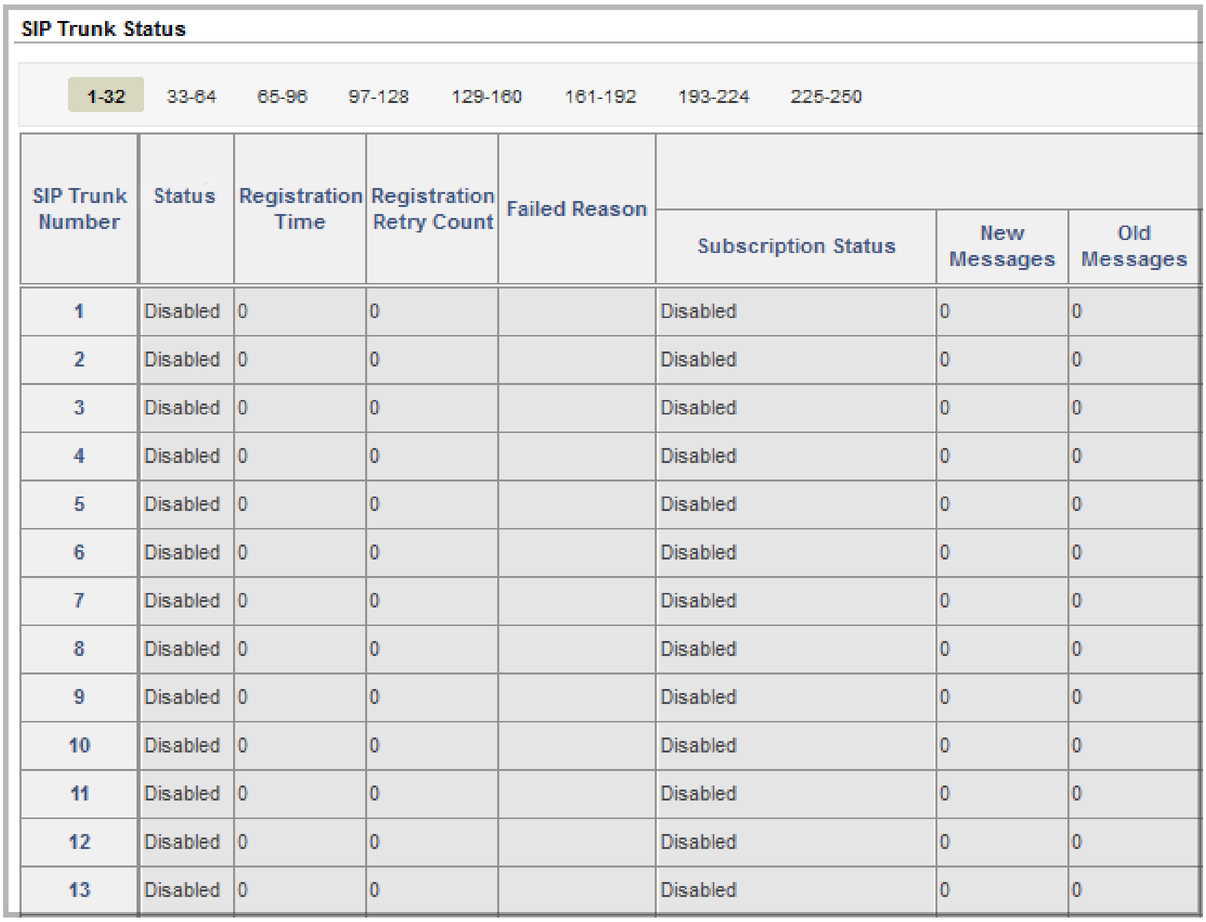
IMEI	IMSI	Module Firmware	SMS Service Center Number	Allowed Call Minutes	Consumed Minutes	Reset Consumed Minutes
						Reset
						Reset
						Reset
						Reset
						Reset
						Reset
						Reset
						Reset
						Reset
						Reset

- **IMEI:** This displays the International Mobile Equipment Identity (IMEI) Number, the unique identity number of the GSM engine.
- **IMSI**: This displays the International Mobile Subscriber Identity (IMSI), the unique identity number of the SIM Card present in the Mobile Port.
- **Module Firmware:** This displays the current version-revision of the engine's firmware.
- **SMS Service Center Number:** This displays the Number of the SMS Service Center of the Service Provider.
- **Allowed Call Minutes:** This displays the Call Minutes allowed to the Mobile Port. To know more about this feature, see "Call Minutes".

- **Consumed Minutes:** This displays the Call Minutes used up by the Mobile Port.
- **Reset Consumed Minutes:** You can reset the consumed minutes by clicking the **Reset** link. To know more, see "Call Minutes".

SIP Trunk

- Click the **SIP Trunk** link.



SIP Trunk Status

1-32 33-64 65-96 97-128 129-160 161-192 193-224 225-250

SIP Trunk Number	Status	Registration Time	Registration Retry Count	Failed Reason	Subscription Status	New Messages	Old Messages
1	Disabled	0	0		Disabled	0	0
2	Disabled	0	0		Disabled	0	0
3	Disabled	0	0		Disabled	0	0
4	Disabled	0	0		Disabled	0	0
5	Disabled	0	0		Disabled	0	0
6	Disabled	0	0		Disabled	0	0
7	Disabled	0	0		Disabled	0	0
8	Disabled	0	0		Disabled	0	0
9	Disabled	0	0		Disabled	0	0
10	Disabled	0	0		Disabled	0	0
11	Disabled	0	0		Disabled	0	0
12	Disabled	0	0		Disabled	0	0
13	Disabled	0	0		Disabled	0	0

The following status indications will appear for the SIP Trunks.

- **SIP Trunk Number:** This displays the SIP Trunk number.
- **Status:** The possible status indications that will be displayed in this column for the respective SIP Trunk numbers are described in the table below.

Status Message	Meaning
Disable	The SIP Trunk is disabled.
Registering	The SIP Trunk is enabled and is waiting for response from the SIP server.
Active	The SIP Trunk is registered with the SIP server.
Failed	Some error has occurred in the SIP Trunk and no calls can be made using the SIP Trunk (applicable only if the SIP Trunk mode is configured as 'Proxy').

Status Message	Meaning
Network Connection Disable	The SIP Trunk is enabled but the active *Network Connection* does not match the option selected for *Use SIP Trunk for Network Connection* parameter.
Inactive	The Proxy Server is unavailable (no response is received from the server).

- **Registration Time:** The SIP Trunk is registered with the Registrar Server for a particular time period, after which it has to be re-registered. The registrar server indicates the time remaining for re-registration of the SIP Trunk. The same is displayed in this field as Registration Time.

- **Registration Retry Count:** This displays the total number of register messages which are sent to the registrar server for registering the SIP Trunk.

- **Failed Reason:** This displays the reason for failure of SIP Trunk registration with the registrar server. The different reasons for registration failure that may appear are:

Failure Message	Description
Message send fail	This reason is displayed when registration request sent to registrar server fails.
Failed to create Register client	This reason is displayed when SIP stack has memory constraints, or resource limitation or the number of SIP clients to register is greater than the number programmed in the stack.
Failed to detach register client	This reason is displayed when SIP stack has memory constraint/ resource limitation/ the number of SIP clients to register is greater than the number programmed in the stack.
Failed to send request	This reason is displayed when DNS server is not programmed.
Local Failure	This reason is displayed when DNS query fails.
Response timeout	This reason is displayed on the expiry of the General Request Timer.
Error Response- 4xx to 6xx	This is the error response code.
No contact header in 2xx	This reason is displayed when no contact address is received in the 2xx response from the SIP server.
Authentication Failed	This reason is displayed when the SIP server does not authenticate the client.
STUN address is not programmed	This reason is displayed when STUN is enabled but address is not configured.
STUN query fail	This reason is displayed when a query to the STUN server fails.
Outbound address is not programmed	This reason is displayed when Outbound is enabled but Outbound address is not configured.
Router's IP address is not programmed	This reason is displayed when Router's IP Address is to be used in signaling but the address is not programmed.
Remote Peer Not Alive	This reason is displayed when no response is received from the remote peer for the OPTIONS message.

*If for a SIP Trunk, you have enabled **Fallback Server** and **Registration Behavior** is set to **Register with all Servers**, the SIP Trunk Status page will display status of all the servers for that SIP Trunk as shown below.*

MWI Status

If you have subscribed for Message Wait Indication service from your ITSP, you can view the status of the messages received on the SIP Trunk.

SIP Trunk Number	MWI						
	Subscription Status	New Messages	Old Messages	Urgent New Messages	Urgent Old Messages	Message Notification-ON	Failed Reason
1	Disabled	0	0	0	0	No	
2	Disabled	0	0	0	0	No	
3	Disabled	0	0	0	0	No	
4	Disabled	0	0	0	0	No	
5	Disabled	0	0	0	0	No	
6	Disabled	0	0	0	0	No	
7	Disabled	0	0	0	0	No	
8	Disabled	0	0	0	0	No	
9	Disabled	0	0	0	0	No	
10	Disabled	0	0	0	0	No	
11	Disabled	0	0	0	0	No	
12	Disabled	0	0	0	0	No	
13	Disabled	0	0	0	0	No	
14	Disabled	0	0	0	0	No	

The following status indications will appear for the MWI subscription on SIP Trunks.

- **Subscription Status:** This displays the MWI Subscription Status. The possible status indications that will be displayed in this column for the respective SIP Trunk numbers are described in the table below.

Status Message	Description
Active	When 200 OK with Event as message-summary and Subscription-State as active is received against SUBSCRIBE for MWI sent from SIP Trunk.
Active	When the NOTIFY with the Event header field as message-summary, Subscription-State as active and the Message Body containing current status of the pending messages, is received on SIP Trunk.
Corresponding 4xx/5xx/6xx response along with the text of the error message as it is received with 4xx/5xx/6xx response.	When any 4xx/5xx/6xx response is received against the SUBSCRIBE for MWI sent from SIP Trunk.

Status Message	Description
Corresponding internal error message (same error messages for relative condition, displayed in case of REGISTER failure)	When any internal error occurs.
Disable	When “Subscribe for MWI” Flag is disabled for SIP Trunk.

- **New Messages:** This displays the number of new messages waiting for the SIP Trunk.
- **Old Messages:** This displays the number of old messages for the SIP Trunk.
- **Urgent New Messages:** This displays the urgent new messages waiting on the SIP Trunk.
- **Urgent Old Messages:** This displays the urgent old messages for the SIP Trunk.
- **Message Notification-ON:** This displays the value of the Message-Waiting field — *Yes* or *No* — depending on which it notifies the FXS Port as per the *Message Wait Notification Type* programmed for that FXS port.
- **Failed Reason:** This displays the reason for failure of MWI registration with the registrar server. The different reasons for registration failure that may appear are:

Failure Message	Description
Error Response- 4xx to 6xx	This is the error response code.

T1E1 Port

- Click the **T1E1 Port** link.

T1E1 Port Status		
T1E1 Port Number	Layer 1	Layer 2
1	DOWN	DOWN
2	DOWN	DOWN
3	DOWN	DOWN
4	DOWN	DOWN
5	DOWN	DOWN
6	DOWN	DOWN
7	DOWN	DOWN
8	DOWN	DOWN

The following parameters will be displayed for the T1E1 Port.

- **Layer 1:** Displays if the link is up or down.
- **Layer 2:** Displays if the link is up or down.

Appendix

Acronyms

AIS	Alarm Indication Signal
ANT	Automatic Number Translation
ANSI	American National Standard Institute
BRI	Basic Rate Interface (2 B-Channels@64Kbps + D-Channel@64Kbps)
CAS	Channel Associate Signaling
CDR	Call Detail Record
CLI	Caller Line Identification
CLIP	Caller Line Identification and Presentation
CLIR	Calling Line Identification Restriction
CO	Call Outgoing
COS	Class of Service
CPT	Call Progress Tone
CPTG	Call Progress Tone (Generation)
CRC	Cyclic Redundancy Check
CUG	Closed User Group
DDI	Direct Dialing In
DHCP	Dynamic Host Control Protocol
DNS	Domain Name Service
DTMF	Dual Tone Multi-Frequency
E1	E-Carrier1 (30B+D)
FCBC	Float cum Boost Charger
FIFO	First In First Out
FoIP	Fax over IP
FSK	Frequency Shift Keying
FTP	File Transfer Protocol

GMT	Greenwich Mean Time
GSM	Global System for Mobile
IC	Incoming call
IP	Internet Protocol
ISDN	Integrated Service Digital Network
ITSP	Internet Telephony Service Provider
ITU	International Telecommunication Union
LAN	Local Area Network
LCD	Liquid Crystal Display
LCR	Least Cost Routing
LED	Light Emitting Diodes
LOS	Loss of signal
MAC	Media Access Control
MDF	Main Distribution Frame
MFA	Multi-Frame Alignment
MOH	Music on Hold
MSN	Multiple Subscribers Numbers
NAT	Network Address Translation
NPI	Numbering Plan Identification
NT	Network Terminal
PBX	Private Branch Exchange
PC	Personal Computer
PCM	Primary Interface Compounding
PIN	Personal Identification Number
PMS	Property Management Software
POTS	Plain Old Telephone Systems
PPM	Primary Protection Module
PPPoE	Point-to-Point Protocol over Ethernet
PRI	Primary Rate Interface
PS	Power Supply
PSTN	Public Switched Telephone Network
PUK	Personal Unlock Key
RBS	Robbed Bit Signaling
RCOC	Returned Call to Original Caller
RF	Radio Frequency
RTC	Real Time Clock

RTP	Real Time Protocol
SAL	System Activity Log
SE	System Engineer
SFL	System Fault Log
SIP	Session Initiation Protocol
SIM	Subscriber Identity Module
SLT	2 wire Analog Station, Single Line Telephone
SMDR	Station Message Detail Recording
SMPS	Switch Mode Power Supply
SNMP	Simple Network Management Protocol
SNTP	Simple Network Time Protocol
T1	T-Carrier (23B+D)
TE	Terminal Equipment / Device
TCP/IP	Transmission Control Protocol/Internet Protocol
TON	Type of Numbering Plan
UDP	User Datagram Protocol
UPS	Un-interrupted Power Supply
URI	Uniform Resource Identifier
VoIP	Voice over IP
WAN	Wide Area Network

Default Region Table

The country-specific default settings of various parameters that will be loaded on changing the **Region** are presented in the table below.

Region Code	Country/ Region	Default Language	Default Time Zone	Default DST Type	Default CPTG	Default Ring Type	Country Code	Companding Type	FXS Port - CLI Type	FXO Port - CLI Type	T1E1 Carrier Type	System Clock Synchro-nization
1	Afghanistan	English	GMT+04:30				93					
2	Algeria	English	GMT+01:00				213	A-law				
3	Antigua and Barbuda	English	GMT-04:00				1 268					
4	Argentina	Spanish	GMT-03:00		4		54	A-law				
5	Australia (Perth)	English	GMT+08:00	2	5	8	61					
6	Australia (Adelaide)	English	GMT+09:30	2	5	8	61					
7	Australia (Brisbane, Canberra, Melbourne, Sydney)	English	GMT+10:00		5	8	61					
8	Austria	German	GMT+01:00	1			43					
9	Bahamas	English	GMT-05:00				1 242					
10	Bahrain	English	GMT+03:00	3			973					
11	Bangladesh	English	GMT+06:00				880					
12	Belarus	English	GMT+02:00				375					
13	Belgium	French	GMT+01:00	2	39	11	32	A-law				
14	Bhutan	English	GMT+06:00				975					
15	Bolivia	Spanish	GMT-04:00				591					
16	Bosnia and Herzegovina	English	GMT+01:00				387					
17	Botswana	English	GMT+02:00				267					
18	Brunei	English	GMT+08:00				673					
19	Brazil (Fernando De Noronha)	Portuguese	GMT-02:00		6	6	55	A-law				
20	Brazil (Brasilia, Rio de Janeiro, Sao Paulo)	Portuguese	GMT-03:00	4	6	6	55	A-law				
21	Brazil (Manaus)	Portuguese	GMT-04:00		6	6	55	A-law				
22	Brazil (Acre)	Portuguese	GMT-05:00		6	6	55	A-law				
23	Bulgaria	English	GMT+02:00				359					
24	Cambodia	English	GMT+07:00				855					
25	Cameroon	English	GMT+01:00				237					
26	Canada (St. John's)	English	GMT-03:30	5	7	7	1	U-law	FSK Bellcore	FSK Bellcore	T1	1.54MHz
27	Canada (Halifax)	English	GMT-04:00	5	7	7	1	U-law	FSK Bellcore	FSK Bellcore	T1	1.54MHz
28	Canada (Montreal, Ottawa, Toronto)	English	GMT-05:00	5	7	7	1	U-law	FSK Bellcore	FSK Bellcore	T1	1.54MHz
29	Canada (Winnipeg)	English	GMT-06:00	5	7	7	1	U-law	FSK Bellcore	FSK Bellcore	T1	1.54MHz
30	Canada (Calgary)	English	GMT-07:00	5	7	7	1	U-law	FSK Bellcore	FSK Bellcore	T1	1.54MHz
31	Canada (Vancouver)	English	GMT-08:00	5	7	7	1	U-law	FSK Bellcore	FSK Bellcore	T1	1.54MHz
32	Chile	Spanish	GMT-04:00	6			56					
33	China	English	GMT+08:00		8	11	86	A-law				
34	Colombia	Spanish	GMT-05:00				57					
35	Costa Rica	Spanish	GMT-06:00				506					
36	Croatia	English	GMT+01:00				385					
37	Cuba	Spanish	GMT-05:00	18			53	A-law				
38	Cyprus	English	GMT+02:00				357					
39	Czech Republic	English	GMT+01:00				420					
40	Denmark	English	GMT+01:00	7			45	A-law				
41	Egypt	English	GMT+02:00	11	9	7	20	A-law				
42	Fiji	English	GMT+12:00				679					
43	Finland	English	GMT+02:00	8			358	A-law				

Region Code	Country/ Region	Default Language	Default Time Zone	Default DST Type	Default CPTG	Default Ring Type	Country Code	Companding Type	FXS Port - CLI Type	FXO Port - CLI Type	T1E1 Carrier Type	System Clock Synchro-nization
44	France	French	GMT+01:00	2	10	14	33	A-law				
45	Germany	German	GMT+01:00	2	11	6	49	A-law				
46	Greece	English	GMT+02:00	2	12	6	30					
47	Guyana	English	GMT-04:00				592					
48	Hong Kong	English	GMT+08:00				852					
49	Hungary	English	GMT+02:00	2			36					
50	India	English	GMT+05:30		13	8	91	A-law				
51	Indonesia	English	GMT+07:00		14		62					
52	Iran	English	GMT+03:30		15		98					
53	Iraq	English	GMT+03:00	9	16		964					
54	Ireland	English	GMT	7			353					
55	Israel	English	GMT+02:00		17	15	972					
56	Italy	Italian	GMT+01:00	2	18	6	39					
57	Japan	English	GMT+09:00		19	10	81	U-law				
58	Jordan	English	GMT+02:00				962	A-law				
59	Kazakhstan	English	GMT+06:00				7					
60	Kenya	English	GMT+03:00		20		254					
61	Korea – North	English	GMT+09:00		21	11	850					
62	Korea – South	English	GMT+09:00		21	11						
63	Kuwait	English	GMT+03:00				965					
64	Kyrgyzstan	English	GMT+06:00	10			996					
65	Lebanon	English	GMT+02:00	12			961					
66	Libya	English	GMT+02:00				218					
67	Malaysia	English	GMT+08:00		22	15	60					
68	Maldives	English	GMT+05:00				960					
69	Mauritius	English	GMT+04:00				230					
70	Mexico (Mexico City)	Spanish	GMT-06:00	3	23		52	A-law				
71	Mexico (Chihuahua)	Spanish	GMT-07:00	3	23		52	A-law				
72	Mexico (Tijuana)	Spanish	GMT-08:00	3	23		52	A-law				
73	Mongolia	English	GMT+08:00				976					
74	Mozambique	Portuguese	GMT+02:00				258					
75	Myanmar	English	GMT+06:30				95					
76	Namibia	English	GMT+01:00	13			264					
77	Nepal	English	GMT+05:45				977					
78	Netherlands	English	GMT+01:00				31	A-law				
79	New Zealand	English	GMT+12:00	14	24	15	64					
80	Nigeria	English	GMT+01:00				234					
81	Norway	English	GMT+01:00	15			47	A-law				
82	Oman	English	GMT+04:00				968					
83	Pakistan	English	GMT+05:00				92					
84	Paraguay	Spanish	GMT-04:00	16			595					
85	Peru	Spanish	GMT-05:00				51					
86	Philippines	English	GMT+08:00		25		63	A-law				
87	Poland	English	GMT+01:00	1	26	15	48					
88	Portugal	Portuguese	GMT	7	27	12	351					
89	Qatar	English	GMT+03:00				974					
90	Romania	English	GMT+02:00				40					
91	Russia (Moscow, St. Petersburg)	English	GMT+03:00	1	28	11	7					
92	Russia (Novosibirsk)	English	GMT+06:00	1	28	11	7					
93	Russia (Vladivostok)	English	GMT+10:00	1	28	11	7					

Region Code	Country/ Region	Default Language	Default Time Zone	Default DST Type	Default CPTG	Default Ring Type	Country Code	Companding Type	FXS Port - CLI Type	FXO Port - CLI Type	T1E1 Carrier Type	System Clock Synchro-nization
94	Singapore	English	GMT+08:00		30	8	65	A-law				
95	Slovakia	English	GMT+01:00				421					
96	South Africa	English	GMT+02:00		31	8	27					
97	Spain	Spanish	GMT+01:00	1	32	13	34	A-law				
98	Sri Lanka	English	GMT+05:30				94					
99	Sudan	English	GMT+03:00				249					
100	Sweden	English	GMT+01:00	2			46	A-law				
101	Switzerland	German	GMT+01:00	2			41					
102	Syria	English	GMT+02:00	17			963					
103	Taiwan	English	GMT+08:00				886					
104	Tajikistan	English	GMT+05:00				992					
105	Thailand	English	GMT+07:00		33	15	66	A-law				
106	Turkey	English	GMT+02:00		34		90					
107	Uganda	English	GMT+03:00				256					
108	Ukraine	English	GMT+02:00				380					
109	United Arab Emirates	English	GMT+04:00		35	15	971	A-law				1.54MHz
110	United Kingdom	English	GMT	7	36	8	44	A-law				1.54MHz
111	United States (Atlanta, Augusta, Boston, Charlotte, Columbus, Detroit, Indiapolis, Miami, NY, Philadelphia, Washington)	English	GMT-05:00	3	37	7	1	U-law	FSK Bellcore	FSK Bellcore	T1	1.54MHz
112	United States (Chicago, Dallas, Des Moines, Memphis, Minneapolis, New Orleans, Oklahoma, Omaha, St. Louis)	English	GMT-06:00	3	37	7	1	U-law	FSK Bellcore	FSK Bellcore	T1	1.54MHz
113	United States (Albuquerque, Boise, Cheyenne, Denver, Salt Lake City)	English	GMT-07:00	3	37	7	1	U-law	FSK Bellcore	FSK Bellcore	T1	1.54MHz
114	United States (Las Vegas, Los Angeles, Phoenix, San Francisco, Seattle)	English	GMT-08:00	3	37	7	1	U-law	FSK Bellcore	FSK Bellcore	T1	1.54MHz
115	United States (Juneau)	English	GMT-09:00	3	37	7	1	U-law	FSK Bellcore	FSK Bellcore	T1	
116	United States (Hawaii)	English	GMT-10:00		37	7	1	U-law	FSK Bellcore	FSK Bellcore	T1	
117	Uzbekistan	English	GMT+05:00				998					
118	Venezuela	Spanish	GMT-04:30				58					
119	Vietnam	English	GMT+07:00				84					
120	Yemen	English	GMT+03:00				967					
121	Yugoslavia	English	GMT+02:00				381					
122	Zambia	English	GMT+02:00				260				**T1E1 Carrier Type**	
123	Zimbabwe	English	GMT+02:00				263					

Call Progress Tones

Call Progress Tones (CPT) are audible tones sent by switching systems such as PSTN or PBX, to calling parties to show the status of the phone call.

Each CPT has a distinctive tone frequency and cadence assigned to it, for which some standards have been established by the ETSI.

On the basis of specific frequency, modulating frequency and cadence, the CPTs generated by SARVAM UMG are categorized as:

- Dial Tone
- Ring Back Tone
- Busy Tone
- Error Tone 1
- Confirmation Tone
- Feature Tone/ Programming Tone
- Intrusion Tone
- Error Tone 2
- Routing Tone
- Stuttered Dial Tone

CPT standards are applied differently in different situations and in different countries.You can match call progress tones of SARVAM UMG to that of the country standard where it is installed.

See the table for the **CPTG Type** (frequency and cadence of the different tones) supported by SARVAM UMG. The table shows the CPTG Types supported for different countries.

When you select 'Region', the Call Progress Tones matching the country standards of the selected Region/Country will be automatically loaded. However, you may select a different CPTG Type, if required. You can also customize the frequency and cadence. For instructions, see "Region" under *Basic Settings*.

Remote Hold Tone is fixed for all the countries; it is non-programmable.

CPTG Types (as per ETSI standard) supported by SARVAM UMG

CPTG Type	Country	Feature / Programming / Prompt Tone		Routing Tone		Intrusion Tone	
		Freq.(Hz)	Cadence (Seconds)	Freq.(Hz)	Cadence (Seconds)	Freq.(Hz)	Cadence (Seconds)
1	Type1	350+440	0.1on 0.9off	350+440	0.1on 1.5off	440	0.1on 2.9off
2	Type2	400	1.5on 0.1off	400	0.1on 1.5off	400	0.2on 4.8off
3	Type3	350+440	0.1on 0.9off	350+440	0.1on 1.5off	440	0.1on 2.9off
4	Argentina	425	0.1on 0.9off	425	0.1on 1.5off	425	0.1on 2.9off
5	Australia	425*25	0.1on 0.9off	425*25	0.1on 1.5off	425	Continuous
6	Brazil	425	0.1on 0.9off	425	0.1on 1.5off	425	0.1on 2.9off
7	Canada	350+440	0.1on 0.9off	350+440	0.1on 1.5off	480+620	0.5on 0.5off

CPTG Type	Country	Feature / Programming / Prompt Tone		Routing Tone		Intrusion Tone	
		Freq.(Hz)	Cadence (Seconds)	Freq.(Hz)	Cadence (Seconds)	Freq.(Hz)	Cadence (Seconds)
8	China	450	0.1on 0.9off	450	0.1on 1.5off	450	0.2on 0.2off 0.2on 0.6off
9	Egypt	425*50	0.1on 0.9off	425*50	0.1on 1.5off	450	0.5on 0.5off
10	France	440	0.1on 0.9off	440	0.1on 1.5off	440	0.1on 2.9off
11	Germany	425	0.1on 0.9off	425	0.1on 1.5off	425	0.1on 2.9off
12	Greece	425	0.1on 0.9off	425	0.1on 1.5off	425	0.15on 0.25off 0.15on 1.45off
13	India	400*25	0.1on 0.9off	400*25	0.1on 1.5off	400	0.15on 4.85off
14	Indonesia	425	0.1on 0.9off	425	0.1on 1.5off	425	0.1on 2.9off
15	Iran	425	0.1on 0.9off	425	0.1on 1.5off	425	0.1on 2.9off
16	Iraq	400	0.1on 0.9off	400	0.1on 1.5off	400	0.1on 2.9off
17	Israel	400	0.1on 0.9off	400	0.1on 1.5off	400	0.1on 2.9off
18	Italy	425	0.1on 0.9off	425	0.1on 1.5off	425	0.1on 2.9off
19	Japan	400	0.1on 0.9off	400	0.1on 1.5off	400*25	0.1on 2.9off
20	Kenya	425	0.1on 0.9off	425	0.1on 1.5off	425	0.1on 2.9off
21	Korea	350+440	0.1on 0.9off	350+440	0.1on 1.5off	350+440	0.1on 2.9off
22	Malaysia	425	0.1on 0.9off	425	0.1on 1.5off	425	0.1on 2.9off
23	Mexico	425	0.1on 0.9off	425	0.1on 1.5off	425	0.1on 2.9off
24	New Zealand	400	0.1on 0.9off	400	0.1on 1.5off	425	0.1on 2.9off
25	Phillippines	425	0.1on 0.9off	425	0.1on 1.5off	440	0.1on 2.9off
26	Poland	425	0.1on 0.9off	425	0.1on 1.5off	425	0.1on 2.9off
27	Portugal	425	0.1on 0.9off	425	0.1on 1.5off	425	0.2on 1.4off
28	Russia	425	0.1on 0.9off	425	0.1on 1.5off	425	0.1on 2.9off
29	Saudi Arabia	425	0.1on 0.9off	425	0.1on 1.5off	425	0.1on 2.9off
30	Singapore	425	0.1on 0.9off	425	0.1on 1.5off	425	0.25on 2.0off
31	South Africa	400*33	0.1on 0.9off	400*33	0.1on 1.5off	400	0.15on 0.25off 0.15on 1.45off
32	Spain	425	0.1on 0.9off	425	0.1on 1.5off	425	0.1on 2.9off
33	Thailand	400*50	0.1on 0.9off	400*50	0.1on 1.5off	400	0.1on 2.9off

CPTG Type	Country	Feature / Programming / Prompt Tone		Routing Tone		Intrusion Tone	
		Freq.(Hz)	Cadence (Seconds)	Freq.(Hz)	Cadence (Seconds)	Freq.(Hz)	Cadence (Seconds)
34	Turkey	450	0.1on 0.9off	450	0.1on 1.5off	450	0.1on 2.9off
35	UAE	350+440	0.1on 0.9off	350+440	0.1on 1.5off	350+440	0.1on 2.9off
36	UK	350+440	0.1on 0.9off	350+440	0.1on 1.5off	400	0.2on 4.8off
37	USA	350+440	0.1on 0.9off	350+440	0.1on 1.5off	480+620	0.5on 0.5off
38	Type4	400	1.75on 0.1off	400	0.1on 1.5off	400	0.2on 0.2off 0.2on 2.5off
39	Belgium	425	0.1on 0.9off	425	0.1on 1.5off	440	0.1on 2.9off
40	Type5	350+440	0.1on 0.9off	350+440	0.1on 1.5off	350+440	0.5on 0.5off 1.0on 5.0off

Stuttered Dial Tone

- **Frequency:** 425 Hz (applicable for all Regions)
- **Cadence (msec):** 100-100-1000-1000 (applicable for all Regions)

Remote Hold

- **Frequency:** 400 Hz (applicable for all Regions)
- **Cadence (msec):** 500-1500 (applicable for all Regions)

Features at Glance

Feature Description	Feature Code
To Enter Programming Mode	***#19-Command Password (Default=1234)***
To Exit Programming Mode	***00#****
To Set Hotline	***#151-1***
To Cancel Hotline	***#151-0***
To Enable Call Waiting	***#16-1***
To Disable Call Waiting	***#16-0***
To Set DND	***#18-1***
To Cancel DND	***#18-0***
To Set Call Forward Unconditional	***#131-1***
To Cancel Call Forward Unconditional	***#131-0***
To Set Call Forward Busy	***#132-1***
To Cancel Call Forward Busy	***#132-0***
To Set Call Forward No Reply	***#133-1***
To Cancel Call Forward No Reply	***#133-0***
To Program Hotline Number	***#152-Destination Number-End-of-Dialing***[a]
To Program Hotline Timer	***#153-X (X is the timer value)***
To Program Call Forward Unconditional Number	***#135-Destination number-End-of-Dialing***[a]
To Program Call Forward Busy Number	***#136-Destination Number-End-of-Dialing***[a]
To Program Call Forward No Reply Number	***#137-Destination Number-End-of-Dialing***[a]
To Program No-Reply Timer	***#139-XX (XX is time in seconds)***
For Call Pick-up	***#5***
For Call Hold	***Flash***
To Retrieve Held Call	***Flash***
For Call Toggle (Call Split)	***#2***
To Reject the Waiting Call and Speech with Current Call	***#31***
To Ignore the Waiting Call and Speech with Current Call	***#32***
To Accept the Waiting Call and Hold Current Call	***#33***
To Accept the Waiting Call and Release Current Call	***#34***
For Blind Transfer	***#6***
For Conference	***#8***
For Using Supplementary Services of Service Provider	***#4***

Feature Description	Feature Code
For Using Voice Mail of the Service Provider	***#7***
For Attended Transfer	**^**
For Making a New Call	***#91***
To Disconnect Call	***#92***

a. Dial # as end of dialing, if it has been configured by you or the system will wait till the expiry of the inter digit wait timer.

System Commands

Certain parameters of SARVAM UMG can be configured by dialing System Commands from a telephone connected to the FXS Port. You can configure certain Network Parameters, like IP Address, Subnet Mask, Connection Type, set the system to default and also view current IP Address, Subnet Mask, Connection Type, DNS and Gateway Address by dialing System Commands.

To be able to view these details, the telephone connected to the FXS Port must have an LCD display.

To dial System Commands,

- Pick up the handset of the telephone connected to the FXS Port of SARVAM UMG.
- Dial ***#19-Command Password*** to enter programming mode.
  You will get programming tone.
- To change WAN Port IP Address, dial ***11-IP Address-#****
  For example, to change IP address to 192.168.1.120, dial ***11-192168001120-#****
  Default: 192.168.1.100
- To change WAN Port Subnet Mask, dial ***12-Subnet Mask-#****
  Where,
  Valid options are 0, 128, 192, 224, 240, 248, 252, 254 and 255.
  For example, to change Subnet Mask to 255.255.254.0, dial ***12-255255254000-#****
  Default: 255.255.255.000
- To configure WAN Port Gateway IP Address, dial ***13-Gateway Address-#****
  Where,
  Gateway IP Address is of 12 digits.
  For example, to change Gateway Address to 192.168.9.10, dial ***13-192168009010-#****
  Default: Blank

  To change Gateway IP Address to Blank, dial **13-#***
- To change LAN Port IP Address, dial ***41-IP Address-#****
  For example, to change IP address to 192.168.1.140, dial ***41-192168001140-#****
  Default: 192.168.2.100
- To change LAN Port Subnet Mask, dial ***42-Subnet Mask-#****
  Where,
  Valid options are 0, 128, 192, 224, 240, 248, 252, 254 and 255.
  For example, to change Subnet Mask to 255.255.253.0, dial ***42-255255253000-#****
  Default: 255.255.255.000
- To select the Connection type, dial ***10-Code-#****
  Where,
  Code is 1 for Static, 2 for DHCP, 3 for PPPoE
  Default: Static
- To enable/disable VLAN Tag, dial ***31-Code-#****
  Where,
  Code is 1 for Enable, 2 for Disable

Default: Disable

- To restore factory defaults, dial ***51-Reverse Command Password-#****
  Replace the handset. The system will restart.

- To view Connection Type, dial ***20-#**** and go On-hook

- To view the Network IP Address, dial ***21-#****and go On-hook

- To view the Subnet Mask, dial ***22-#****and go On-hook

- To view the Gateway Address, dial ***23-#****and go On-hook

- To view the DNS Address, dial ***24-#****and go On-hook

- To display the LAN Port IP address on Phone, dial **46-#*** and go On-hook.

- To display the LAN Port Subnet Mask on Phone, dial **47-#*** and go On-hook.

- To view the Status of SIP Trunks, dial ***27-SIP Trunk number-#**** and go On-hook. SIP Trunk Number is from 001 to 250.

  The value of the parameter will be displayed on the LCD of the telephone instrument.

- To exit programming mode, dial ***00-#****

Technical Specifications - ETERNITY GENX

System Resource	Specifications
Processor Type	32 bit RISC processor
Processor speed	900 MHz dual core
Universal Slots	12
Number of USB Ports	1 Internal USB 2.0 1 External USB 3.0
USB Storage Capacity	Internal USB = 32 GB External USB = 1 TB
Ethernet Ports	1 Gbps for WAN 1 Gbps for LAN
Communication Ports	1
Vocoder Module for VoIP[a]	64 Channels
Mounting Options	1. 19” Rack Mountable 2. Table Top 3. Wall Mountable
Dimensions (WxHxD)	436.5 x 226 x 173.5 mm
Unit Weight (Kg)	5.00 Kg
Shipping Weight (Kg)	8.20 Kg
Shipping Material Type	Corrugated Box
Power Supply Options	AC PS DC PS
Power Supply Input Range	AC PS : 100 - 240 VAC DC PS : 48 VDC (+20% & -15%)
Power Consumption[b]	For AC PS: 140 W maximum For DC PS: 115 W maximum For PSBB PS: 140 W maximum
Operating Temperature	0^{o}C to 45^{o}C
Operating Humidity	5 - 95% RH, Non-Condensing
Storage Temperature	- 20^{o}C to +70^{o}C
Storage Humidity	0 - 95% RH, Non-Condensing

a. Max number of NX DBM VOCODER64 modules that can be placed on the board are 2.
b. Considering 30% SLT off-hook when 240 SLTs are connected. The power consumption may vary depending upon the system configuration and number of SLTs in off-hook state.

Technical Specifications - SARVAM UMG

System Resource	Specifications
FXS Ports[a]	240 Max
FXO Ports	192 Max
T1E1 Ports	8 Max
BRI Ports	48 Max
Mobile Ports	48 Max
SIP Trunks	250 Max
VoIP Channels[b]	128 Max

a. *The maximum number of simultaneous off-hook FXS ports supported are 120.*
b. *Two NX DBM VOCODER64 Modules can be installed.*

Supported Cards

- ETERNITY GE CARD SLT8
- ETERNITY GE CARD SLT16
- ETERNITY GE CARD SLT20
- ETERNITY GE CARD CO8
- ETERNITY GE CARD CO16
- ETERNITY GE CARD CO4+SLT16
- ETERNITY GE CARD GSM4 (2G/3G/4G)
- ETERNITY GE CARD T1E1PRI SINGLE
- ETERNITY GE CARD BRI4

FXS (Analog Station)

Signaling	Loop Start
Dialing	DTMF and Pulse (10/20PPS)
Off Hook AC Impedance	600/900/Complex
Off Hook Current	39mA max
Loop Limit	1800Ω max (excluding Telephone)
On-Hook Voltage (Tip/Ring)	-48V nominal
DTMF Detection	ITU-T Q.24
Return Loss	>18dB
Longitudinal Balance	>50dB
Transmission Level Adjust	Tx Gain: -3dB to +6dB, Rx Gain: -3dB to 6dB
Ringing	Trapezoidal 60VRMS/25Hz and Sinusoidal 52VRMS/25Hz
REN	3

CLI Reception	DTMF, FSK ITU-T V.23 and FSK Bellcore 202
Protection	Over Voltage Secondary Protection
Physical Connector	RJ45

FXO (Central Office) / Two-Wire Trunk (TWT)

Signaling	Loop Start
Loop Limit	1200Ω
Off Hook AC Impedance	600/900/Complex
Pulse Dialing	10/20 PPS
DTMF Dialing and Reception	ITU-T Q.23 and Q.24
Return Loss	>18dB
Longitudinal Balance	>50dB
Transmission Level Adjust	Tx Gain: -15dB to +10dB, Rx Gain: -15dB to 10dB
CLI Reception	DTMF, FSK ITU-T V.23 and FSK Bellcore 202
Call Maturity	Delay and Polarity Reversal
Protection	Over Voltage and Over Current Secondary Protection
Physical Connector	RJ45

ISDN BRI

Channels	2B+D
Personality	Network (NT) and Terminal (TE)
Switch Variant	ETSI - EURO ISDN NET3 BRI (BRI NET3)
Protection	Solid state (Over Voltage and Over Current) Built-in Secondary Protection
Physical Connector	RJ45 (120Ω)

ISDN PRI

Channels	23B+D and 30B+D
Personality	Network (NT) and Terminal (TE)
Line Coding	AMI/B8ZS for T1 and HDB3 for E1
Framing	ESF for T1 and CEPT1 (with/without CRC) for E1
Switch Variant	AT&T 5ESS, DMS, US NI2 (National ISDN 2), ETSI NET5
Protection	Solid State (Over Voltage and Over Current) Built-in Secondary Protection

Physical Connector	RJ45 (Impedance Selectable)

E1 CAS

Bit Rate	2048 kbps +/-50 ppm
Line Coding	HDB3
Framing	CEPT1 (with/without CRC) with CAS MF
Line Signaling	ITU-T Q.400 - Q. 490
Register Signaling	MFC-R2
Alarms	I.431. G.732, ETSI 300-233
Protection	Solid State (Over Voltage and Over Current) Built-in Secondary Protection
Physical Connector	RJ45 (Impedance Selectable), Fiber Optic

T1 RBS

Bit Rate	1544 kbps +/- 50 ppm
Line Coding	AMI and B8ZS
Line Signaling	FXS Loop Start, FXO Loop Start, FXS Ground Start, FXO Ground Start, E&M (Immediate, Wink Start, Wink Start FGD)
Framing	D4, ESF
Digit Dialing	DTMF
Alarms	ANSI T1.231
Performance	ANSI T1.403, ANSI T1.231, AT&T TR54016
Protection	Solid state (Over Voltage and Over Current) Built-in Secondary Protection
Physical Connector	RJ45 (Impedance Selectable)

VoIP

Connector	RJ45
VoIP Protocols	SIP v2, SDP, RTP (RFC 2833), SRTP
Network Protocols	IPv4, TCP, UDP, DHCP, PPPoE, SNTP, NAT, STUN, HTTP, TLS, DynDNS
SIP	Maximum 250 SIP Accounts per system, Outbound Proxy Support, Display Name, User Name, Password, URL, Proxy URL, Register URL, Register Interval
NAT	STUN and NAT Keep Alive
Voice Codecs	G.729, G.723, GSM FR, iLBC (30ms), iLBC (20ms), GSM EFR, G.711 (u-Law), G.711 (A-Law)
Line Echo Cancellation	G.168 with 128ms Tail Length

Call Progress Tones	Dial Tone, Ring Back Tone, Busy Tone, Error Tone
Voice	Dynamic Jitter Buffer (Adaptive), Comfort Noise Generation and Voice Activity Detection
Fax	T.38(UDPTL), T.38(RTP) and Pass Through
Quality of Service	Layer 3 Diffserv and TOS
Security	Password Protected Administration

GSM(2G)

Frequency Band

Module	GSM 850	EGSM 900	DCS 1800	PCS 1900	UMTS 850	UMTS 900	UMTS 1900	UMTS 2100	Rx-diversity	GNSS
Quectel UC20-G	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓
Quectel M95	✓	✓	✓	✓	×	×	×	×	×	×

Compliant	ETSI GSM Phase2/2+
SIM Card	One SIM per GSM Port
SIM Interface	1.8V, 3V
RF Transmission Power	Class 4 (2W) at GSM850MHz and EGSM900MHz band Class 1 (1W) at DCS1800MHz and PCS1900MHz band
RF Sensitivity	Better than -102dBm
Protocol	AT Command Interface
External Antenna	One Antenna per 4 GSM Ports,1.8/3.0*dBi, 50Ω, SMA (Male) Connector, Omni-Directional with cable of 3 meters length

** Depends on GSM Frequency Band.*

UMTS(3G)

Frequency Band

Module	GSM 850	EGSM 900	DCS 1800	PCS 1900	UMTS 850	UMTS 900	UMTS 1900	UMTS 2100	Rx-diversity	GNSS
Quectel UC20-G	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓
Quectel M95	✓	✓	✓	✓	×	×	×	×	×	×

Compliant	ETSI GSM Phase2/2+
SIM Card	One SIM per GSM Port
SIM Interface	1.8V, 3V

Transmission Power	Output Power Class 4 (2W) at GSM850/GSM900 Class 1 (1W) at DCS1800/PCS1900 Class 3 (0.25W) at UMTS 850/900/1900/2100
RF Sensitivity	< -106dBm at GSM850, EGSM900, DCS1800, PCS1900 < -108dBm at WCDMA2100, WCDMA1900 < -106dBm at WCDMA850
Protocol	AT Command Interface
External Antenna	One Antenna per 4 GSM Ports,1.8/3.0*dBi, 50Ω, SMA (Male) Connector, Omni-Directional with cable of 3 meters length

** Depends on GSM Frequency Band.*

Antenna for 2G and 3G

Antenna Type	:	Fixed Omni Directional Swivel Antenna
Antenna Gain	:	Dipole = 1.8/2.5 dBi
Antenna Connector	:	SMA (Male), 50Ω

Antenna for 4G LTE

Antenna Type	:	Monopole Omni Directional Swivel Antenna
Antenna Gain	:	1/3 dBi
Antenna Connector	:	SMA (Male), 50Ω

Technical Specifications of GSM (2G-M95, 3G-UC20G, 4G-EC25) Modules

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)

<table>
<tr><th colspan="3">Module Supported : Quectel UC20-G</th></tr>
<tr><td rowspan="6">Standards and directive</td><td colspan="2">Applied / Complied Harmonized Standards</td></tr>
<tr><td>RE Directive 2014/53 EU,
Article 3(1)(a)
■ Safety</td><td>EN 60950-1:2006+A11:2009+A1:2010+A12:2011+A2:2013</td></tr>
<tr><td>RE Directive 2014/53 EU,
Article 3(1)(a)
■ Health</td><td>EN 62311:2008</td></tr>
<tr><td>RE Directive 2014/35 EU,
Article 3(1)(b)
■ EMC</td><td>ETSI EN 301 489-1 V2.1.1, ETSI EN 301- 408-52 V1.1.0
ETSI EN 301 489- 19 V2.1.0</td></tr>
<tr><td>RE Directive 2014/53 EU,
Article 3(2)
■ Radio</td><td>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*</td></tr>
<tr><td colspan="2">* Note: This is non-harmonized radio standard accepted by the RED (Radio Equipment Directive)</td></tr>
<tr><td>FCC Identifier</td><td colspan="2">XMR201510UC20</td></tr>
<tr><td>Moducations Supported</td><td colspan="2">GSM: GMSK, 8PSK, WCDMA: BPSK, QPSK, 16QAM
GPS: BPSK
GLONASS: OFDM</td></tr>
</table>

<table>
<tr><th colspan="2">Module Supported: Quectel EC25-A</th></tr>
<tr><td>Technology</td><td>LTE / VoLTE
LTE Version : 3GPP E-UTRA Release 10</td></tr>
<tr><td>Frequency Bands</td><td>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.)</td></tr>
<tr><td>FCC Identifier</td><td>XMR201605EC25A</td></tr>
<tr><td>Modulation Supported</td><td>QPSK, 16QAM and 64QAM</td></tr>
</table>

<table>
<tr><th colspan="3">Module Supported: Quectel EC25-AUT</th></tr>
<tr><td rowspan="8">Standards and Directives</td><td rowspan="2">SAFETY (RCM)</td><td>EN60950-1:2006/A11:2009/A1:2010/A12:2011/A2:2013</td></tr>
<tr><td>AS/NZS 60950.1 2011</td></tr>
<tr><td rowspan="3">EMC</td><td>AS/NZS CISPR 32-2015</td></tr>
<tr><td>ETSI EN 301908-13 (ETSI TS 136 521-1 V13.4.0)</td></tr>
<tr><td>ETSI EN 301 908-1</td></tr>
<tr><td rowspan="2">FCC</td><td>FCC CFR47 Part 2(2017)/FCC CFR 47 Part 22H (2017)</td></tr>
<tr><td>AS/CA S042.1:2015, AS/CA S042.4:2015</td></tr>
<tr><td>HEALTH</td><td>AS&NZS 2772.2-2016/ARPANSA standard RPS3-2002</td></tr>
<tr><td>Technology</td><td colspan="2">LTE / VoLTE
LTE Version : 3GPP E-UTRA Release 10</td></tr>
<tr><td>Frequency Bands</td><td colspan="2">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.)</td></tr>
<tr><td>FCC Identifier</td><td colspan="2">NA</td></tr>
<tr><td>Modulation Supported</td><td colspan="2">(WCDMA)QPSK , (LTE)QPSK 16QAM</td></tr>
</table>

<table>
<tr><th colspan="2">Module Supported: Quectel EC25-V</th></tr>
<tr><td>Technology</td><td>LTE / VoLTE
LTE Version : 3GPP E-UTRA Release 10</td></tr>
<tr><td>Frequency Bands</td><td>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</td></tr>
<tr><td>FCC Identifier</td><td>XMR201607EC25V</td></tr>
<tr><td>Modulation Supported</td><td>QPSK, 16QAM and 64QAM</td></tr>
</table>

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)	

EG25-G

Module Supported: EG25-G

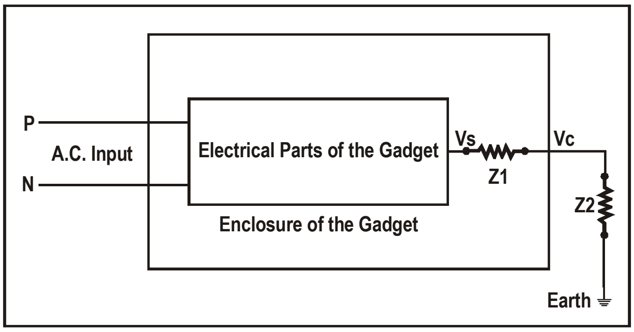
Technology LTE/VoLTE

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

How to Make the Telecom Earth

The Earth (Ground) is the most important safety procedure to prevent electrical shocks and fires. It protects from lightning strikes, electrical transients, static discharges, electromagnetic interference and electrical hazards.

A proper earth must be in place to protect people and the system. The following explanation shows how a perfect electrical earth can save lives.



In the above diagram, Vc = VS *Z2 / Z1 + Z2
Where,

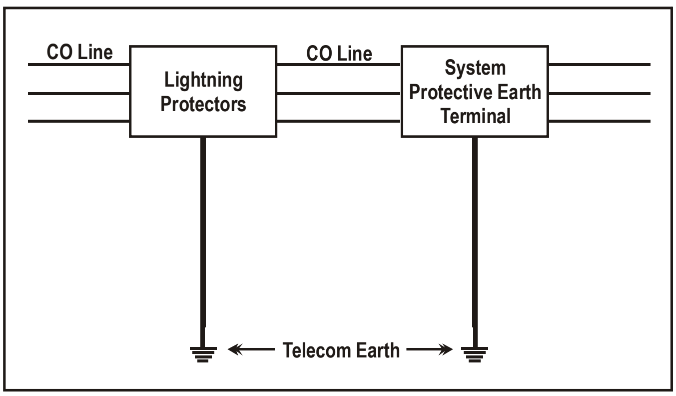
- Z1 is the stray impedance between the electrical parts of the Gadget and the Chassis.
- Z2 is the stray impedance between the Chassis and the Earth.
- If Z2 = 0 then VC = 0

This formula implies that if the impedance between the Chassis and the Earth is reduced to 0 then the Voltage on the Chassis, that is, VC, would be Zero and hence any person touching the enclosure will not get an electric shock. Hence Z2 should be made Zero.

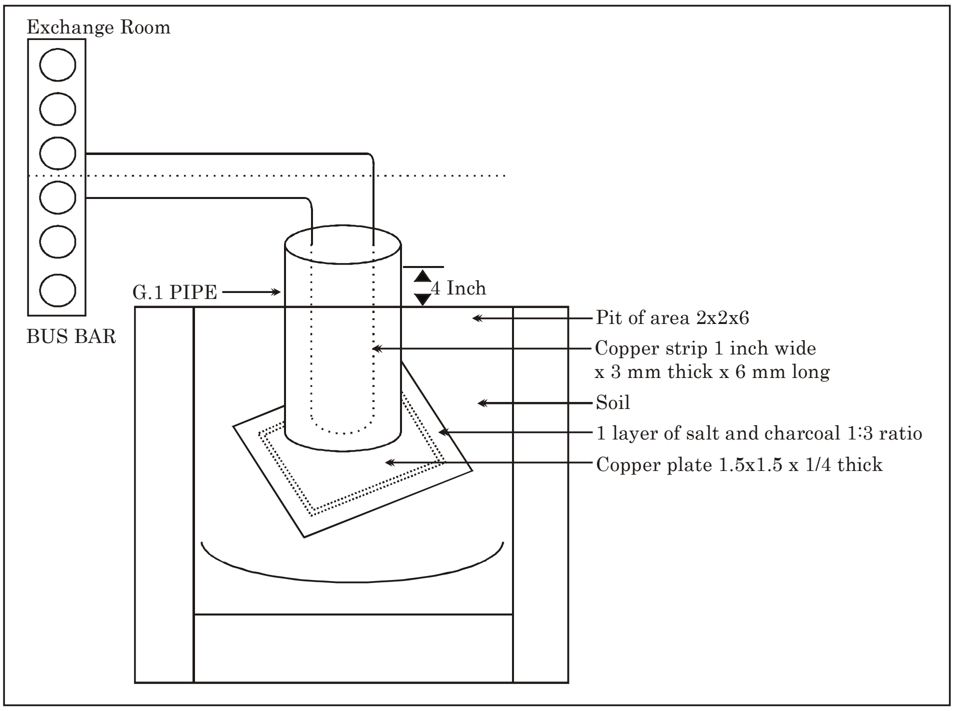
This can be done by providing a perfect earth to the electrical equipment.

It is recommended that you provide a dedicated earth for the *ETERNITY GENX*/any other telecom equipment. This dedicated earth is called the Telecom Earth (Ground).

Providing a separate Telecom Earth to the telecom equipment eliminates the possibility of any back-voltage on the earth.



How to make a perfect Earth



- Dig a pit of area 2 feet x 2 feet x 6 feet (L x B x D).
- Get a copper plate of size 1.5 feet x 1.5 feet x 0.25 feet.
- Connect a copper strip of size 1-inch wide, 3 mm thick and 6 feet length at the center of the copper plate by welding or nuts and bolts.
- Insert a G.I. pipe onto the copper strip till it reaches the copper plate.
- Place this set up into the pit. Make sure that at least 4 inches of the G.I. pipe is above the ground level.
- Fill the bottom of the pit with a 1-inch layer of charcoal and salt in the proportion of 3:1 (3 parts charcoal, 1 part salt) and then cover with the soil.
- Connect a bare 14 SWG copper wire (double) on the top of the copper strip and run it to the exchange room and connect it on the bus bar.
- The Bus bar is a copper strip, 4 inches long with 6 screws and nuts mounted on it. It has to be fixed on the wall in the exchange room.
- The earth wire of the Primary Protection Modules (PPM) should be connected to this Bus bar.
- Water the earth at regular intervals.

Connecting ETERNITY GENX to the Earth

Follow the instructions given below to connect the ETERNITY GENX to the Earth. The instructions are given with reference to the above diagram:

1. Loosen the screw.
2. Insert an earthing wire into the lug.
3. Crimp the earth wire with an appropriate tool.
4. Tighten the screw.
5. Connect the earthing wire to the earth.

*Make sure you use an earthing wire that has a conductor with a **cross-sectional area greater than 1.0 mm2 or AWG less than or equal to 16 AWG** with Green and Yellow insulation.*

- *Make sure you comply with all applicable laws, regulations and guidelines.*
- *Proper earthing is very important to protect the ETERNITY GENX from external noise and to reduce the risk of electrocution in the event of a lightning strike.*
- *The AC cable's earthing pin may not be enough to protect the ETERNITY GENX from external noise and lightning strikes. A permanent connection must be made between earth and the earth terminal of the main unit.*

Packing List

Verify contents of the package shipped to you with the contents listed below. If any of the items is missing or damaged, contact your Dealer/Reseller

You can view the documentation of the system by scanning the QR code printed on the Product Label/Packaging Label of the product.

Sr.No.	Item Name	Qty
1	ETERNITY GENX12S[a]	1
2	3-pin Power Cord, MC-4 Black[b]	1
3	3-pin DC Input Cable[c]	1
4	Screws M 7/30 for Wall Mounting	2
5	Screw Grips for Wall Mounting	2
6	Side Clamp	2
7	Screw M4X12 CSK for the Side Clamp	4
8	Mounting Templates	1

a. ETERNITY GENX12S AC with factory fitted AC Power Supply Card and CPU card.
ETERNITY GENX12S DC with factory fitted DC Power Supply Card and CPU card.
b. Supplied with ETERNITY GENX12S AC.
c. Supplied with ETERNITY GENX12S DC.

ETERNITY GENX Cards

ETERNITY GE Card PS48VDC 250W

Sr. No.	Item	Quantity
1.	ETERNITY GE Card PS48VDC	1
2.	3-pin DC Input Cable	1

ETERNITY GE Card PSUNI 250W

Sr. No.	Item	Quantity
1.	ETERNITY GE Card PSUNI	1
2.	3-pin Power Cord, MC-4 Black	1

ETERNITY GENX Card CPU

Sr. No.	Item	Quantity
1.	ETERNITY GENX Card CPU	1

ETERNITY GE Card SLT20

Sr. No.	Item	Quantity
1.	ETERNITY GE Card SLT20	1
2.	Cable with RJ45 connector on one end	6

ETERNITY GE Card SLT16

Sr. No.	Item	Quantity
1.	ETERNITY GE Card SLT16	1
2.	Cable with RJ45 connector on one end	4

ETERNITY GE Card SLT8

Sr. No.	Item	Quantity
1.	ETERNITY GE Card SLT8	1
2.	Cable with RJ45 connector on one end	2

ETERNITY GE Card CO4+SLT16

Sr. No.	Item	Quantity
1.	ETERNITY GE Card CO4+SLT16	1
2.	Cable with RJ45 connector on one end	6

ETERNITY GE Card CO16

Sr. No.	Item	Quantity
1.	ETERNITY GE Card CO16	1
2.	Cable with RJ45 connector on one end	4

ETERNITY GE Card CO8

Sr. No.	Item	Quantity
1.	ETERNITY GE Card CO8	1
2.	Cable with RJ45 connector on one end	2

ETERNITY GE Card BRI4

Sr. No.	Item	Quantity
1.	ETERNITY GE Card BRI4	1
2.	Cable with RJ45 connectors on both ends	4

ETERNITY GE Card T1E1PRI Single

Sr. No.	Item	Quantity
1.	ETERNITY GE Card T1E1PRI Single	1
2.	Cable with RJ45 connectors on both ends	1

ETERNITY GE Card GSM4

Sr. No.	Item	Quantity
1.	ETERNITY GE Card GSM4	1
2.	GSM Antenna External SMA	1

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-ACTA

Federal Communications Commission Statement

Part 15:

Note: This equipment has been tested and found to comply with the limits for a Class A digital device, pursuant to Part15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference when the equipment is operated in a commercial environment. This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with the instruction manual, may cause harmful interference to radio communications. Operation of this equipment in a residential area is likely to cause harmful interference in this case the user will be required to correct the interference at his own expense.

Using Automatic Dialers:

This equipment supports automatic dialing of Emergency number for using the feature 'Security Dialing and Reporting'. Hence the user should take care about following point:
WHEN PROGRAMMING EMERGENCY NUMBERS AND (OR) MAKING TEST CALLS TO EMERGENCY NUMBERS:

- Remain on the line and briefly explain to the dispatcher the reason for the call.
- Perform such activities in the off-peak hours, such as early morning or late evenings.

Using Direct Inward Dialing ("DID"):

The equipment supports a feature 'direct inward dialing' (DID). Hence while operating it, following points must be considered:
ALLOWING THIS EQUIPMENT TO BE OPERATED IN SUCH A MANNER AS TO NOT PROVIDE FOR PROPER ANSWER SUPERVISION IS A VIOLATION OF PART 68 OF THE FCC'S RULES. PROPER ANSWER SUPERVISION IS WHEN:

a. This equipment returns answer supervision to the public switched telephone network (PSTN) when DID calls are:

   - Answered by the called station
   - Answered by the attendant
   - Routed to a recorded announcement that can be administered by the customer premises equipment (CPE) user.
   - Routed to a dial prompt

b. This equipment returns answer supervision on all DID calls forwarded to the PSTN. Permissible exceptions are:

   - A call is unanswered
   - A busy tone is received
   - A reorder tone is received

Equal Access:

This equipment is capable of providing the end user equal access to the carrier of the user's choice. This equipment is capable of providing users access to interstate providers of operator services through the use of access codes. Modification of this equipment by call aggregators to block access dialing codes is a violation of the Telephone Operator Consumers Act of 1990.

Electrical Safety:

Telephone companies report that electrical surges, typically lightning transients, are very destructive to customer terminal equipment connected to AC power sources. This has been identified as a major nationwide problem.

However Matrix provides all protection against lightning transients in the equipment; the user must provide a suitable surge arrestor while integrating the equipment with other networking equipments.

Using FAX Capability:

The Telephone Consumer Protection Act of 1991 makes it unlawful for any person to use a computer or other electronic device, including FAX machines, to send any message unless such message clearly contains in a margin at the top or bottom of each transmitted page or on the first page of the transmission, the date and time it is sent and an identification of the business or other entity, or other individual sending the message and the telephone number of the sending machine or such business, other entity, or individual. (The telephone number provided may not be a 900 number or any other number for which charges exceed local or long-distance transmission charges.) In order to program this information into your FAX machine, Refer user's Guide for the software of the Fax operation, such as 'Win Fax'.

Installation and Repairs

Before installing this equipment, users should ensure that it is permissible to be connected to the facilities of the local telecommunications company. The equipment must also be installed using an acceptable method of connection. Repairs to certified equipment should be coordinated by a representative designated by the dealer/ supplier. Contact the support at Support@MatrixTeleSol.com

Any repairs or alterations made by the user to this equipment, or equipment malfunctions, may give the telecommunications company cause to request the user to disconnect the equipment.

CE Certificate:

MATRIX
TELECOM | SECURITY

Declaration of Conformity

Manufacturer's Name	: MATRIX COMSEC PVT LTD
Manufacturer's Address	: 15 & 19 - GIDC, WAGHODIA, VADODARA - 391760 (GUJARAT, INDIA)

Declares that the product/s

Product	: IP-PABX
Model/Type	: ETERNITY GENX
Trade Name	: MATRIX

Confirms to the following product specification:

EMI/EMC Standard(s):

EN 55032 : 2015
EN 61000-3-2 : 2014
EN 61000-3-3 : 2013
EN 55024 : 2010 + A1: 2015
IEC 61000-4-2 : 2008
IEC 61000-4-3 : 2006 + A1: 2007 + A2: 2010
IEC 61000-4-4 : 2012
IEC 61000-4-5 : 2014 + A1: 2017
IEC 61000-4-6 : 2013
IEC 61000-4-8 : 2009
IEC 61000-4-11 : 2004 + A1: 2017

SAFETY Standard:

EN 60950-1: 2006 + Am11:2009 + Am 1: 2010 + Am12: 2011 + Am 2: 2013

Supplementary Information:

The product herewith complies with the following directives:

EMC Directive	2014/30/EU
Low Voltage Directive	2014/35/EU
RoHS Directive (RoHS2)	2011/65/EU (as per EN 50581:2012)

Mr. Ganesh Jivani
Director
Date: 13/06/2018

Matrix Comsec Pvt. Ltd. Vadodara

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

Open Source Licensing Terms and Conditions

- 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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    Preamble

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rights.
```

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Also, for each author's protection and ours, we want to make certain that everyone understands that there is no warranty for this free software. If the software is modified by someone else and passed on, we want its recipients to know that what they have is not the original, so that any problems introduced by others will not reflect on the original authors' reputations.

Finally, any free program is threatened constantly by software patents. We wish to avoid the danger that redistributors of a free program will individually obtain patent licenses, in effect making the program proprietary. To prevent this, we have made it clear that any patent must be licensed for everyone's free use or not licensed at all.

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MATRIX COMSEC

Head Office
394-GIDC, Makarpura, Vadodara - 390010, India.
Ph: (+91)1800-258-7747
E-mail: Customer.Care@MatrixComSec.com

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