Enterprise Session Border Controllers

Multi-Service Business Gateways

VoIP Media Gateways

Configuration Note

Connecting Microsoft® Lync™ and Verizon SIP-Trunk using AudioCodes® Mediant™ E-SBC Series

Version 6.4

August 2012

Document #: LTRT-31904

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Reader's Notes

Notice

This document describes the procedure for integrating the Verizon Internet Telephony Service Provider (ITSP) SIP Trunk with Microsoft® Lync Server using AudioCodes' Mediant E-SBC series, which includes the Mediant 800 E-SBC, Mediant 1000 E-SBC and Mediant 3000 E-SBC, as was performed during the certification/compliance process.

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Date Published: August-16-2012

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Abbreviations and Terminology

Each abbreviation, unless widely used, is spelled out in full when first used.

Note: Throughout this guide, the term *E-SBC device* refers to the E-SBC family (Mediant 800 Media Gateway and E-SBC, Mediant 800 MSBG, Mediant 1000 MSBG, Mediant 1000B Media Gateway and E-SBC, Mediant 3000 Media Gateway and E-SBC).

1 Introduction

This Configuration Note shows how to set up AudioCodes' networking device to operate with Verizon SIP Trunking Service and the Microsoft Lync Communication platform.

1.1 Intended Audience

The Note is intended as a supporting reference for certification/compliance testing and may be utilized as a configuration example by installation engineers or AudioCodes and Verizon partners installing and configuring Verizon SIP Trunking service and the Microsoft Lync Communication platform for VoIP calls using AudioCodes networking devices for superior voice quality services.

1.2 About AudioCodes Networking Devices

AudioCodes' family of Enterprise Session Border Controllers (E-SBC) enables reliable connectivity and security between enterprises and Service Providers' VoIP networks.

The E-SBC family (Mediant 800 E-SBC, Mediant 1000 E-SBC and Mediant 3000 E-SBC) provides perimeter defense as a way of protecting companies from malicious VoIP attacks; mediation for allowing the connection of any PBX and/or IP-PBX to any Service Provider; and Service Assurance for service quality and manageability.

AudioCodes E-SBC is available as an integrated solution running on top of the field-proven Mediant Media Gateway and Multi-Service Business Gateway platforms or as a software-only solution for deployment on 3rd party hardware.

The native implementation of E-SBC functions on the AudioCodes Mediant Media Gateways and Multi-Service Business Gateways provides a host of additional capabilities that are not possible with standalone SBC appliances, such as VoIP mediation, PSTN Access, data routing, WAN access, data security, survivability, and third party value-added services applications. This enables enterprises to utilize the advantages of converged networks and eliminate the need for standalone appliances.

Mediant 800 E-SBC is based on field-proven VoIP and network services with a native host processor, allowing the creation of purpose-built multiservice appliances, providing smooth connectivity to cloud services, with integrated quality of service, SLA monitoring, security and manageability. The native implementation of SBC provides a host of additional capabilities that are not possible with standalone SBC appliances, such as VoIP mediation, PSTN access survivability, and third party value-added services applications. This enables enterprises to utilize the advantages of converged networks and eliminate the need for standalone appliances.

Note: The scope of this document does not cover security aspects for connecting to the Microsoft Lync environment. Security measures should be implemented in accordance with your organization's security policies. For basic security guidelines, see the document *Recommended Security Guidelines Technical Note*.

Note: The scope of this document does not serve as a replacement or alternative to the AudioCodes *Release Notes*, *User's Manual*, *Installation Guide* or *Product Reference Manual* with regards to specific aspects for connecting to the Microsoft Lync environment. Specific implementation requirements should be implemented in accordance with your organization's topology requirements and utilization requirements. For further capability understanding, see the AudioCodes *Release Notes*, *User's Manual*, *Hardware Installation Guide* and *Product Reference Manual*.

2 Components Information

2.1 AudioCodes E-SBC Version

Table 2-1: AudioCodes E-SBC Version

Gateway Vendor	AudioCodes
Model	The E-SBC family (Mediant 800 Media Gateway and E-SBC, Mediant 800 MSBG, Mediant 1000 MSBG, Mediant 1000B Media Gateway and E-SBC, Mediant 3000 Media Gateway and E-SBC).
Software Version	SIP_6.40A.034.005
Interface Type	SIP/IP
VoIP Protocol	SIP
Additional Notes	With Digital Trunk Module or MPM module for DSP resources **MSFT** – Enable working with Microsoft Lync **IPSEC, MediaEncryption, StrongEncryption and EncryptControlProtocol** – Enable working with TLS transport type **SBC = 120** – Enable the IP-to-IP feature up to 60 sessions

2.2 AudioCodes Gateway Version

Table 2-2: AudioCodes Gateway Version

Gateway Vendor	AudioCodes
Model	AudioCodes MP-11x series.
Software Version	SIP_6.20A.060.001
Interface Type	SIP/IP
VoIP Protocol	SIP
Additional Notes	Analog device FAX gateway

2.3 Verizon SIP Trunking Version

Table 2-3: Verizon Version

Gateway Vendor	Verizon
Models	
Software Version	
VoIP Protocol	SIP
Additional Notes	None

2.4 Microsoft Lync Server 2010 Version

Table 2-4: Microsoft Lync Version

Gateway Vendor	Microsoft
Models	Microsoft Lync Server 2010
Software Version	RTM: Release 2010 4.0.7577.184
VoIP Protocol	SIP
Additional Notes	None

2.5 Mediation Server Version

Table 2-4: Microsoft Lync Version

Gateway Vendor	Microsoft
Models	Mediation Server
Software Version	RTM: Release 2010 4.0.7577.183
VoIP Protocol	SIP
Additional Notes	None

2.6 Microsoft Lync Client Version

Table 2-5: Microsoft Lync Version

Gateway Vendor	Microsoft
Models	Microsoft Lync Client
Software Version	RTM: Release 2010 4.0.7577.4051
VoIP Protocol	SIP
Additional Notes	None

2.7 Lync Server 2010 Specification

For the purposes of this testing, the following server specification was used to host the Microsoft Lync environment in a fully functioning Lync Server 2010 deployment, including Active Directory and DNS.

The Domain Controller (DC) and the Lync standard edition where installed on the OSN3 server hosted on the Mediant 1000B MSBG chassis.

OSN3 Specifications

Parameter	Specification
CPU	Intel® Core™ 2 Duo 1.5 GHz processors L7400 with Intel 3100 Chipset (64-bit)
RAM Memory	2 G DDR2 with ECC
Hard Drives	Up to 2 hard drives (HDMX modules) 160GB
Bus/Chipset	64 Bit
L2 Cache	2 M
Interfaces	▪ Gigabit Ethernet ▪ USB 2.0 via Connection Module ▪ RS-232 COM
Operating System	▪ Windows Server 2008 standard

2.8 Topology

The procedures described in this document describe the following example scenario:

- An enterprise has a deployed Lync server 2010 in its private network for enhanced communication within the company.
- The enterprise decides to offer its employees Enterprise voice and to connect the company to the PSTN network using the Verizon SIP Trunking service.
- AudioCodes Session Border Controller (SBC) is used to manage the connection between the Enterprise LAN and the Verizon SIP trunk.

  The “session" refers to the real-time voice session using IP SIP signaling protocol. The "border" refers to the IP to IP network border between the Microsoft Lync network in the Enterprise LAN and the Verizon SIP trunk in the public network.

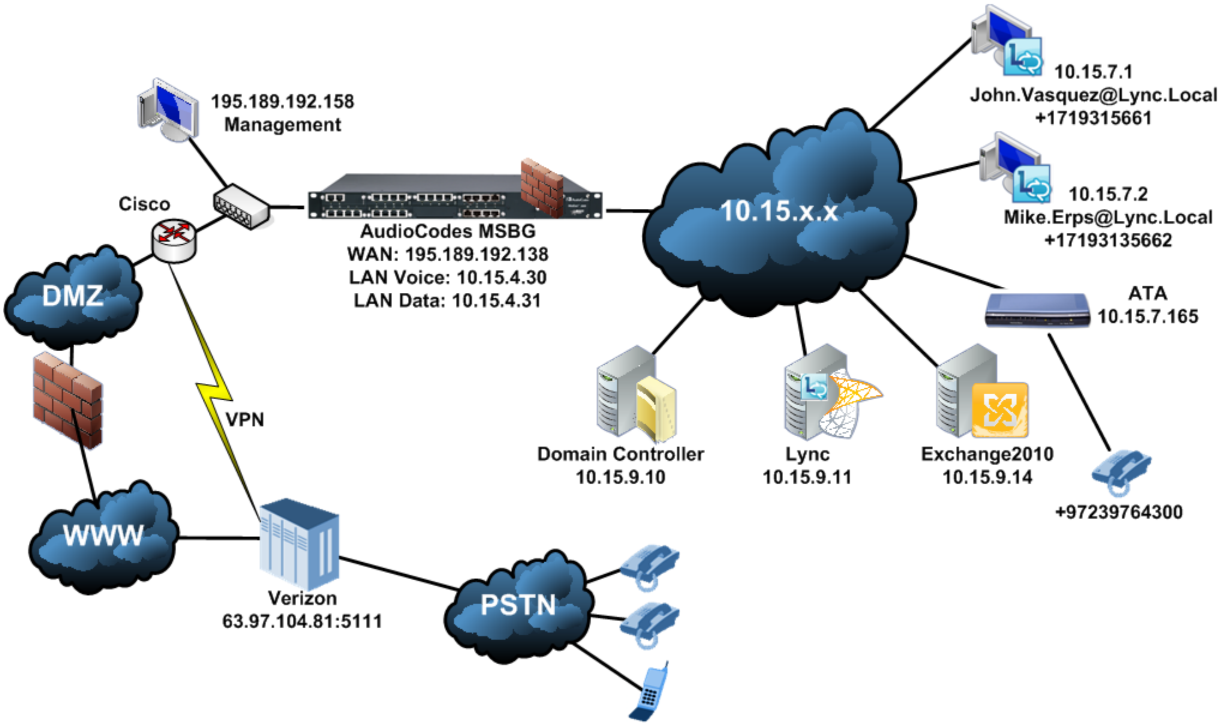
Figure 2-1 below illustrates the interoperability topology between the Lync Server 2010 LAN and the Verizon SIP Trunking site.

The setup requirements are characterized as follows:

- While the Lync Server 2010 environment is located on the Enterprise's Local Area Network (LAN), the Verizon SIP Trunks are located on the WAN.
- The internal data routing capabilities of the Mediant 1000 MSBG device are used. Consequently, a separate WAN interface is configured in the LAN.
- Lync Server 2010 works with the TLS transport type, while the Verizon SIP trunk works on the SIP over UDP transport type.
- Transcoding support: Lync Server 2010 supports G.711A-law and G.711U-law coders, while the Verizon SIP Trunk also supports the G.729 coder type.
- ATA - FAX Support via Analog Media Gateway.
- The AudioCodes E-SBC device can support B2BUA transcoding or non-transcoding interactions. The E-SBC device can also be configured to transparently forward fax calls without intervention (e.g. transcoding).

- Support for early media handling:

Figure 2-1: Topology



Note: Throughout this manual, unless otherwise specified, the term *device* refers to the Mediant 800 E-SBC, Mediant 1000B E-SBC and the Mediant 3000 E-SBC.

2.8.1 Port Topology

This section summarizes the SIP source and destination port settings in this configuration scenario in support of call processing.

Proxy Set	Transport Protocol	Destination (Sending) Port
Microsoft Lync	TCP *or*	5068
	TLS	5067
Verizon	UDP	5111
Fax Gateway	UDP	5060

The Destination Port for Microsoft Lync, Verizon and Fax Gateways are configured in the Proxy Sets table. See Section 4.4 on page 61.

Transport Protocol	Source (Listening) Port
UDP	5060
TCP	5068
TLS	5067

The Source Ports for the Microsoft Lync and Verizon proxy sets are configured in the SIP General Settings screen for the E-SBC device. See Section 4.14 on page 87.

The Source Port for the Fax Gateway is configured in the SIP General Settings for the ATA device. See Section A.4 on page 106.

3 Configuring Lync Server 2010

This section describes how to configure the Lync Server 2010 to operate with the E-SBC device. This section describes the following procedures:

1. Configuring the device as an 'IP/PSTN Gateway'. See Section 3.1 on page 19.
2. Associating the 'IP/PSTN Gateway' with the Mediation Server. See Section 3.2 on page 23.
3. Configuring a 'Route' to utilize the SIP trunk connected to the E-SBC device. See Section 3.3 on page 28.

> **Note:** Dial Plans, Voice Policies, and PSTN usages are also necessary for enterprise voice deployment; however, they are beyond the scope of this document.

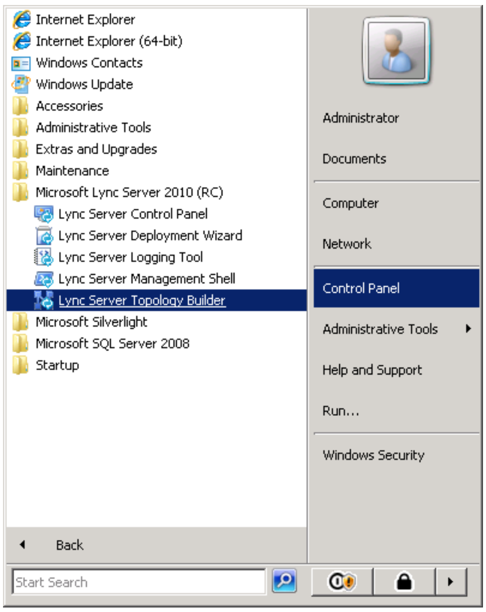
3.1 Configuring the E-SBC device as an 'IP/PSTN Gateway'

This section describes how to configure the E-SBC device as an IP/PSTN Gateway.

➢ To configure the Mediant 1000 MSBG as a IP/PSTN Gateway and associating it with the Mediation Server:

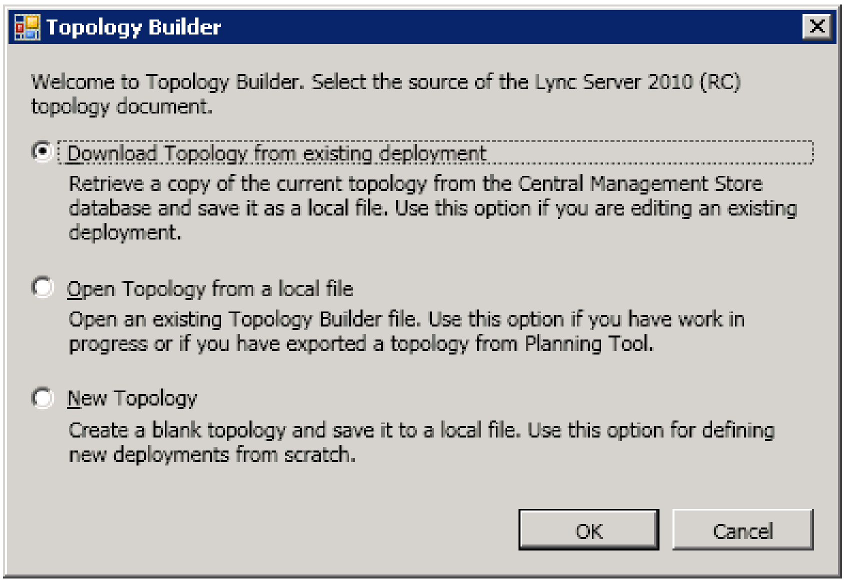
1. On the server where the Topology Builder is located, start the Lync Server 2010 **Topology Builder** (Click **Start**, select **All Programs**, and select **Lync Server Topology Builder**).

Figure 3-1: Starting the Lync Server Topology Builder



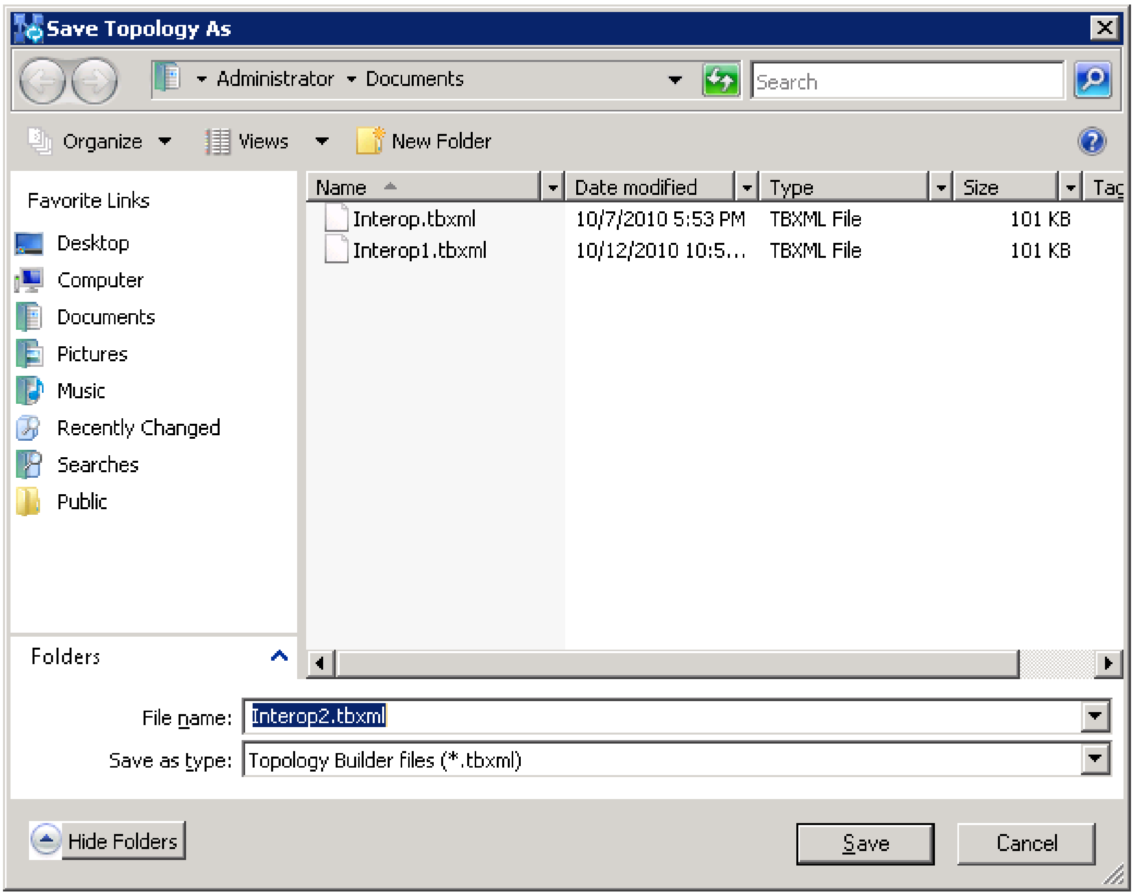
The following screen is displayed:

Figure 3-2: Topology Builder Options



2. Click the **Download Topology from the existing deployment** button and click **OK**; you are prompted to save the Topology which you have downloaded.

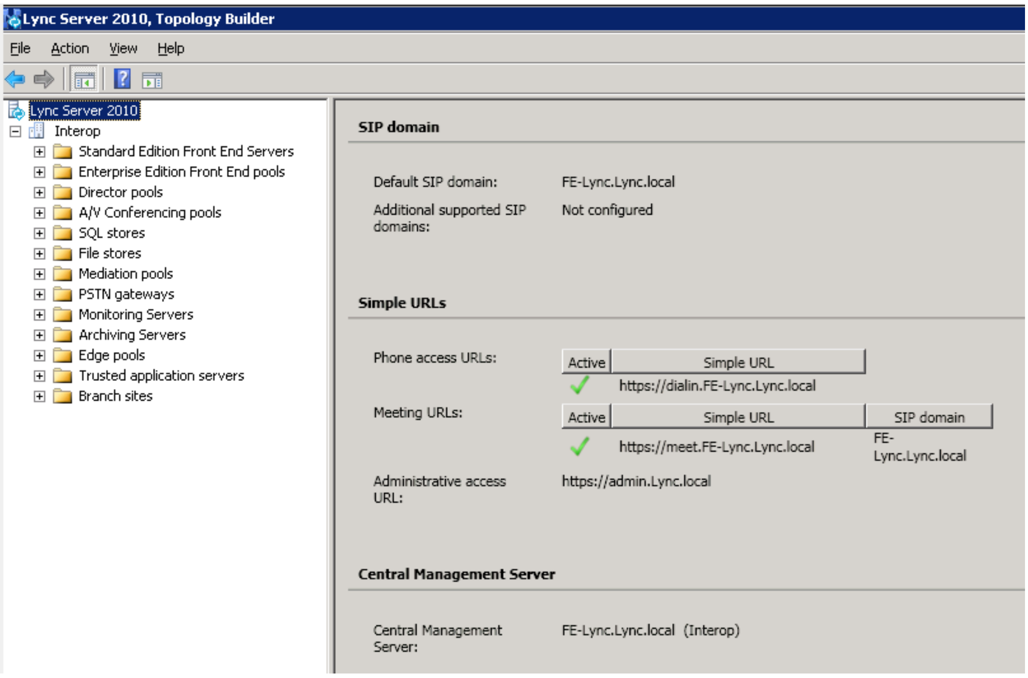
Figure 3-3: Save Topology



3. Enter new File Name and **Save** (this action enables you to rollback from any

changes you make during the installation); the Topology Builder screen with the topology downloaded is displayed.

Figure 3-4: Downloaded Topology



4. Expand the Site, right-click the IP/PSTN Gateway and choose **New IP/PSTN Gateway**.

Figure 3-5: New IP/PSTN Gateway

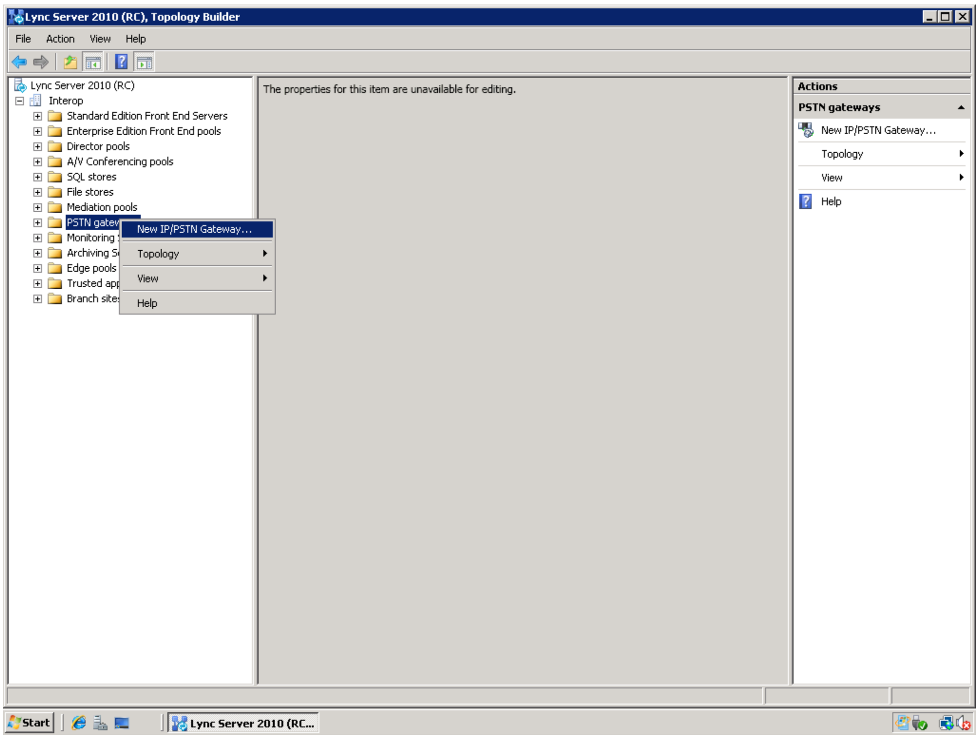
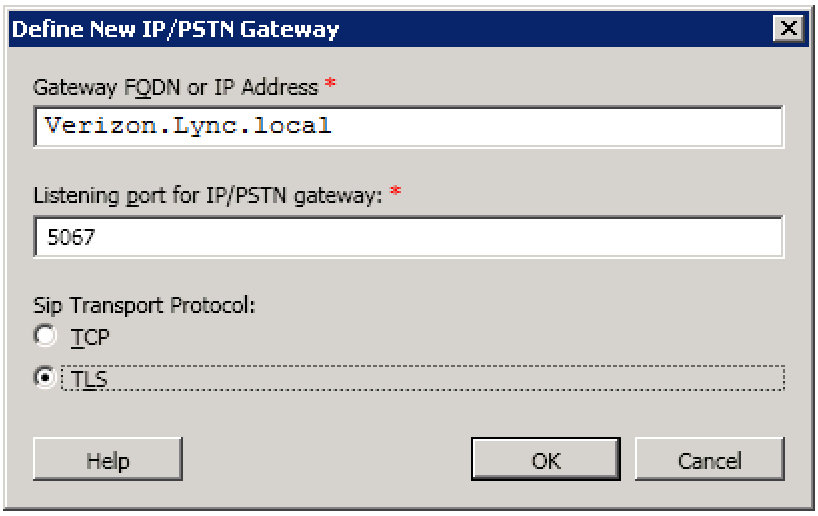


Figure 3-6: Define New IP/PSTN Gateway

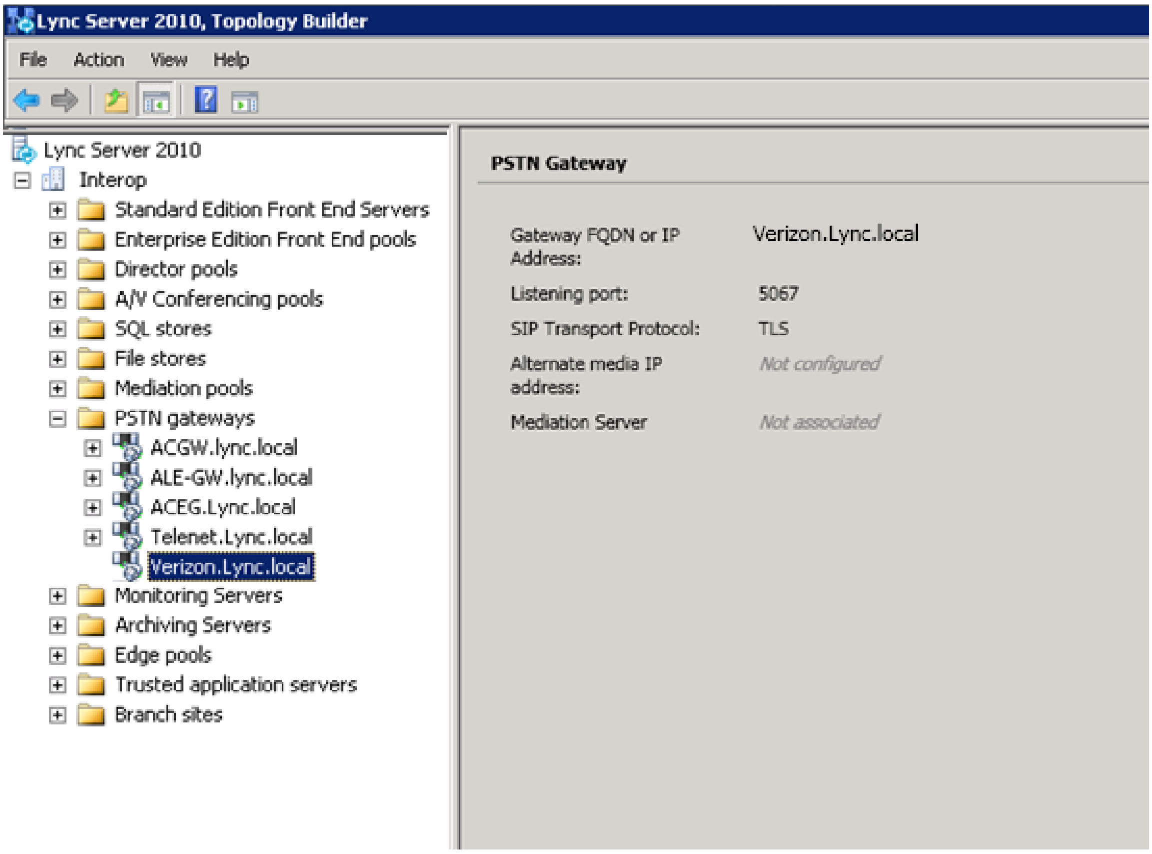


5. Enter the FQDN of the E-SBC device (i.e. ''Verizon.lync.local') and click **OK**.

   Note that the listening port for the Gateway is '5067' and the transport type is 'TLS'.

   The E-SBC device is now added as 'IP/PSTN Gateway'.

Figure 3-7: IP/PSTN Gateway



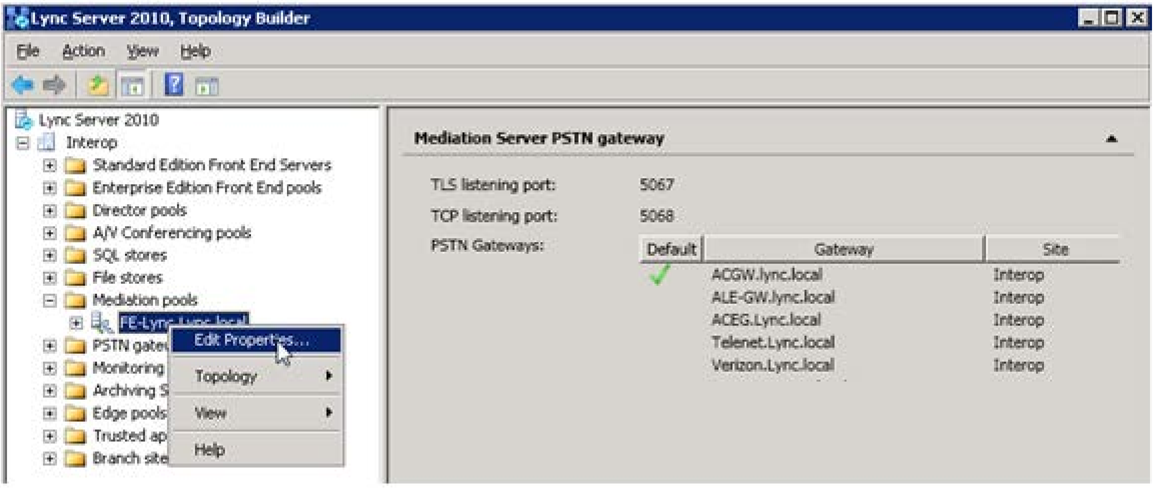
3.2 Associating the 'IP/PSTN Gateway' with the Mediation Server

This section describes how to associate the 'IP/PSTN Gateway' with the Mediation Server.

➢ **To associate the IP/PSTN Gateway with the Mediation Server:**

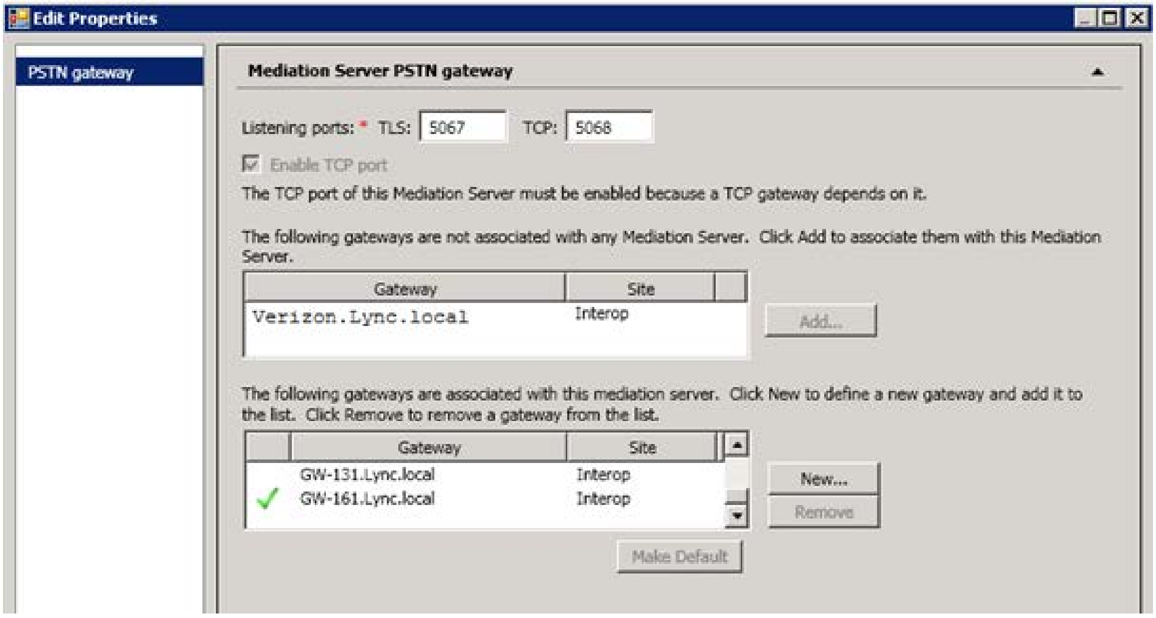
1. Right-click the Mediation server to use with the E-SBC device (i.e. 'FE-Lync.Lync.local') and choose **Edit Properties**.

Figure 3-8: Associating Mediation Server with IP/PSTN Gateway



The following screen is displayed:

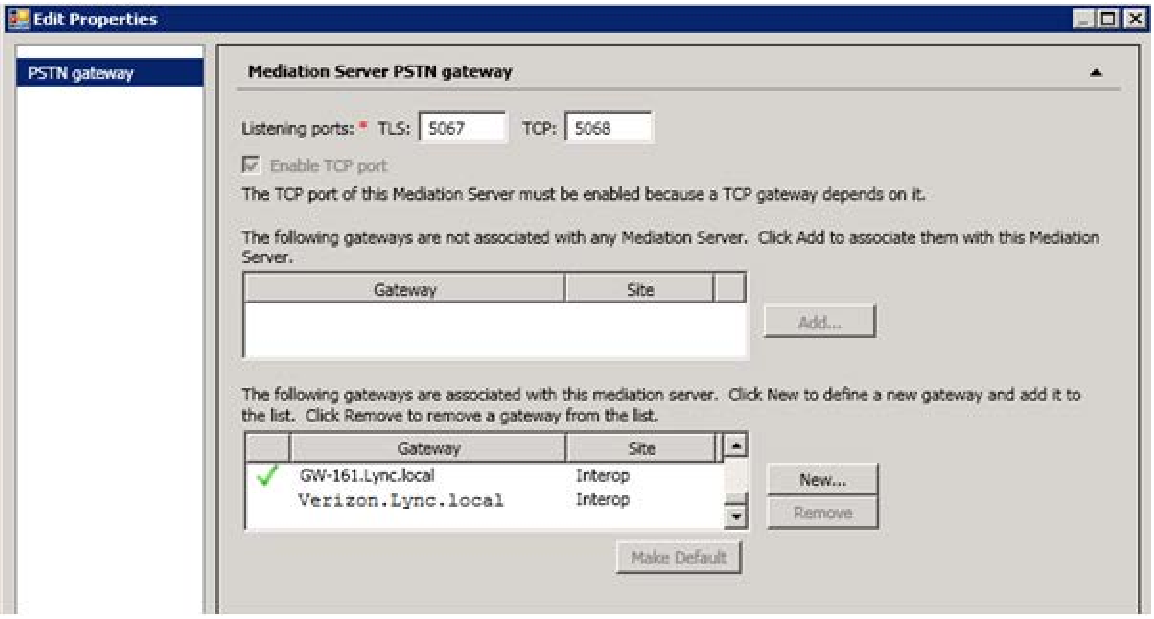
Figure 3-9: Before Associating IP/PSTN Gateway to a Mediation Server Associations



2. In the top-left corner, choose **PSTN gateway** and in the Mediation Server PSTN gateway pane, mark the gateway-SBC device (i.e. 'Verizon.lync.local') and click **Add** to associate it with this Mediation Server.

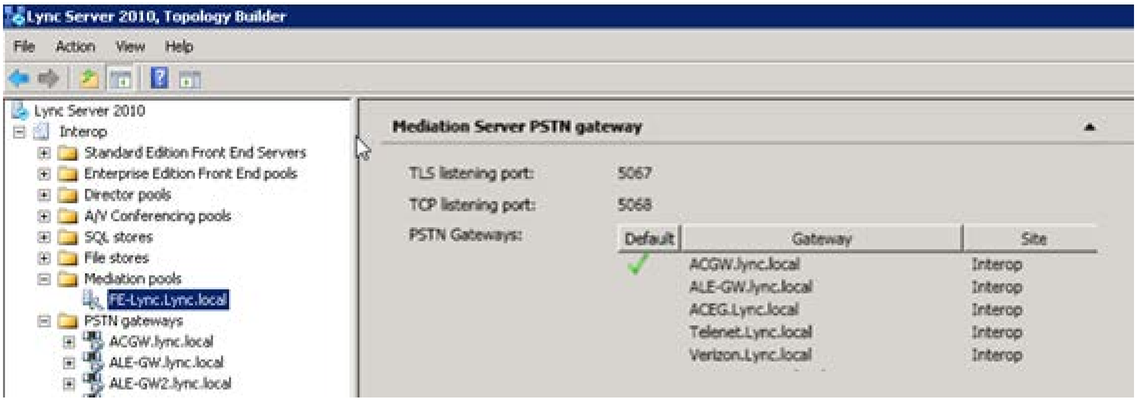
   Note that there are two sub-panes, one including a list of gateways not associated with the Mediation server and one including a list of gateways associated with the Mediation server.

Figure 3-10: After Associating IP/PSTN Gateway to Mediation Server



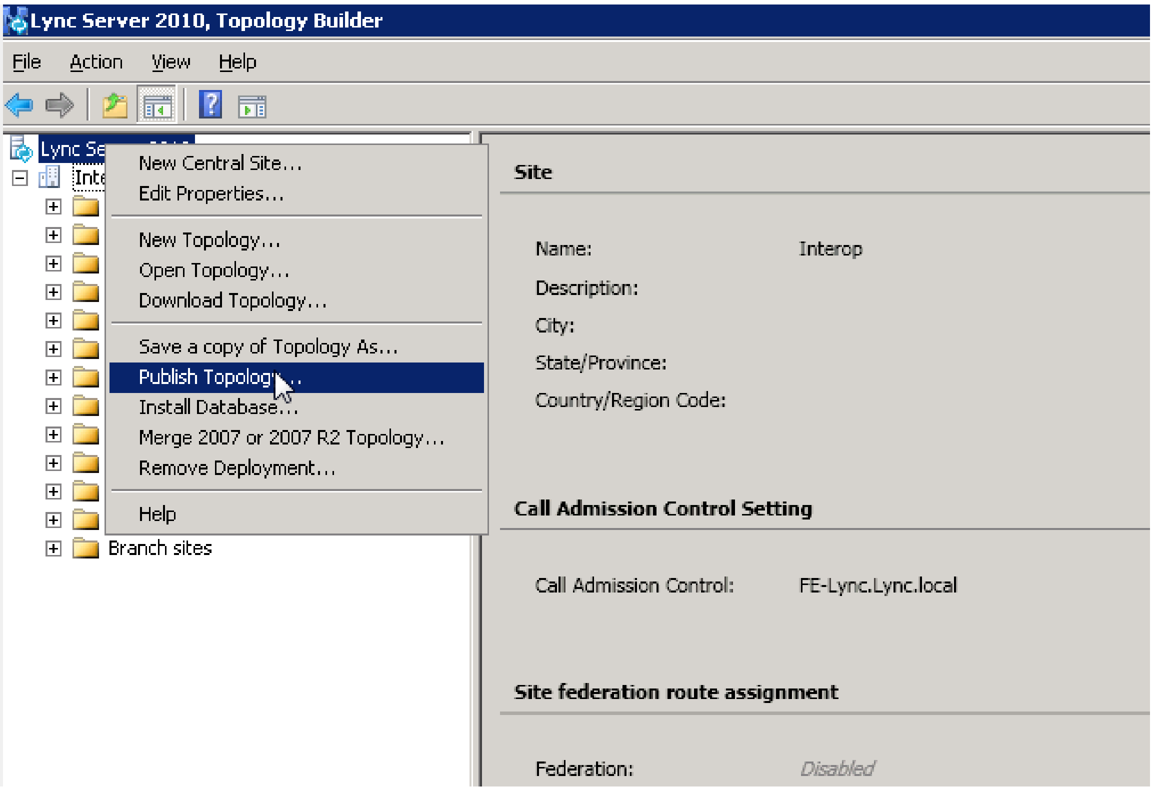
3. Click **OK**.

Figure 3-11: Media Server PSTN Gateway Association Properties



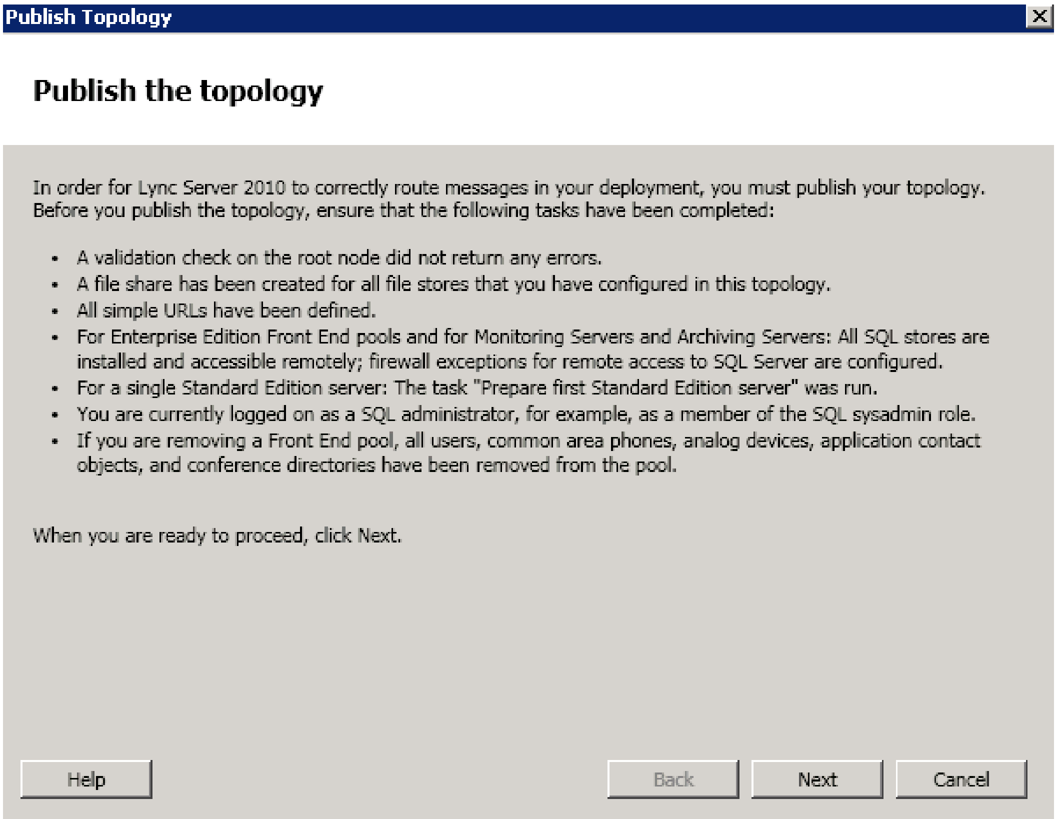
4. In the Lync Server main menu, choose **Action > Publish Topology**.

Figure 3-12: Publishing Topology



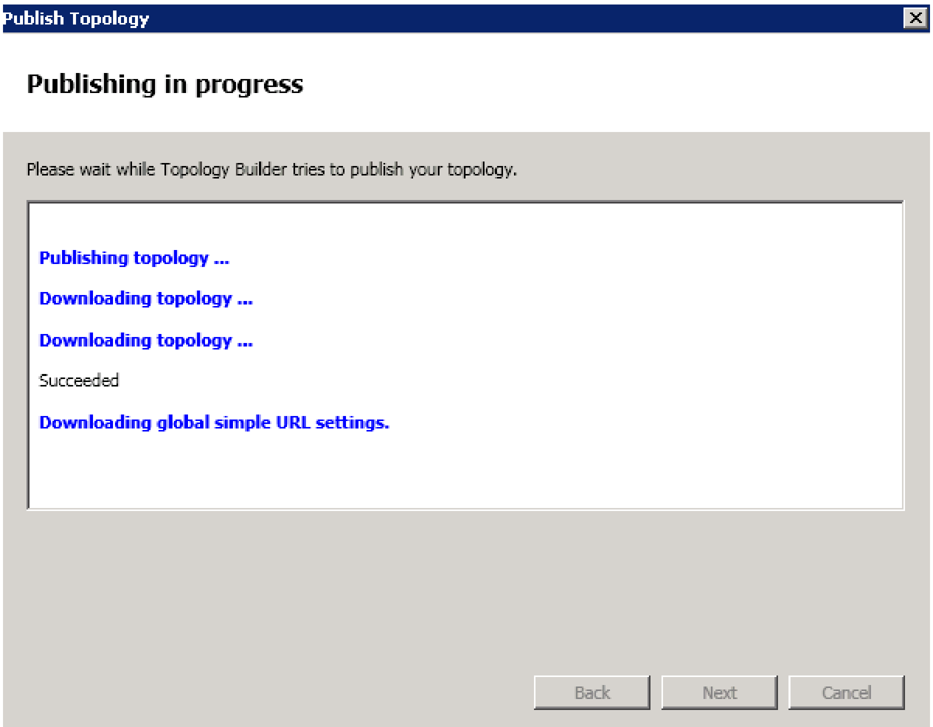
The Publish Topology screen is displayed.

Figure 3-13:Publish Topology Confirmation



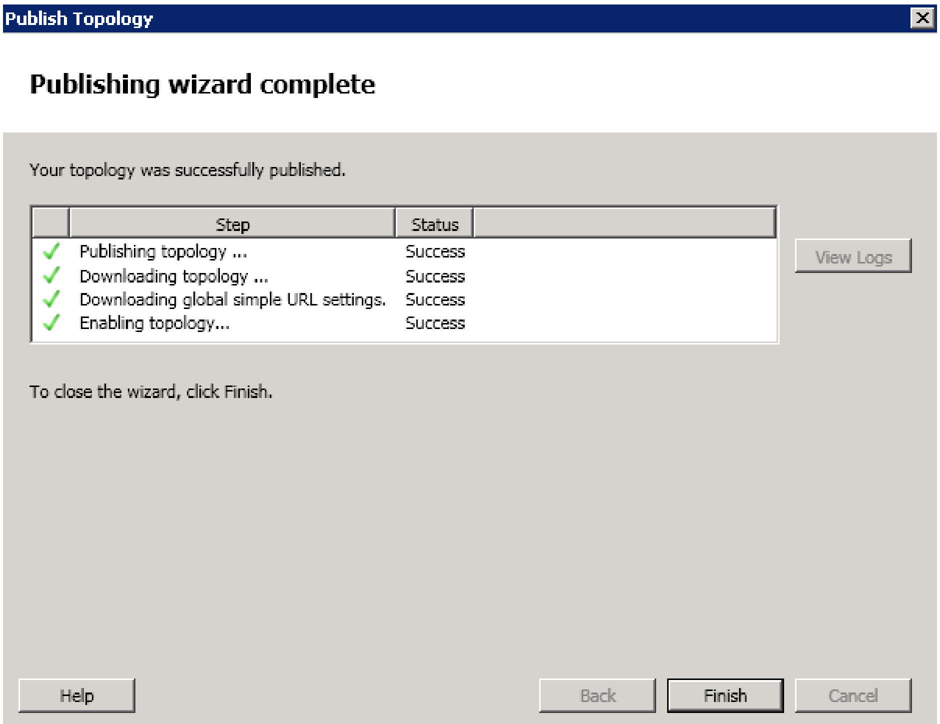
5. Click **Next**; the Topology Builder attempts to publish your topology.

Figure 3-14: Publish Topology Confirmation screen



Wait until the publish topology process has ended successfully.

Figure 3-15: Publish Topology Successfully Completed



6. Click **Finish**.

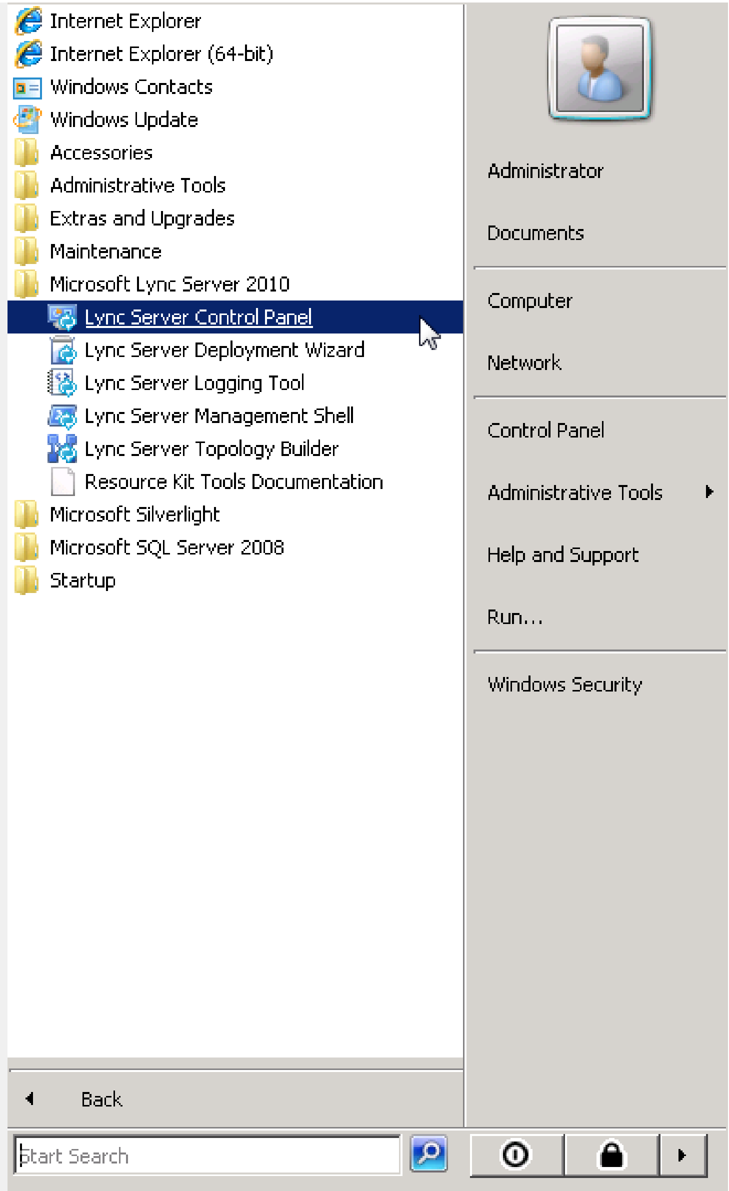
3.3 Configuring the 'Route' on the Lync Server 2010

This section describes how to configure a 'Route' on the Lync server and associate it with the E-SBC device PSTN gateway.

➢ **To configure the 'route' on the Lync server:**

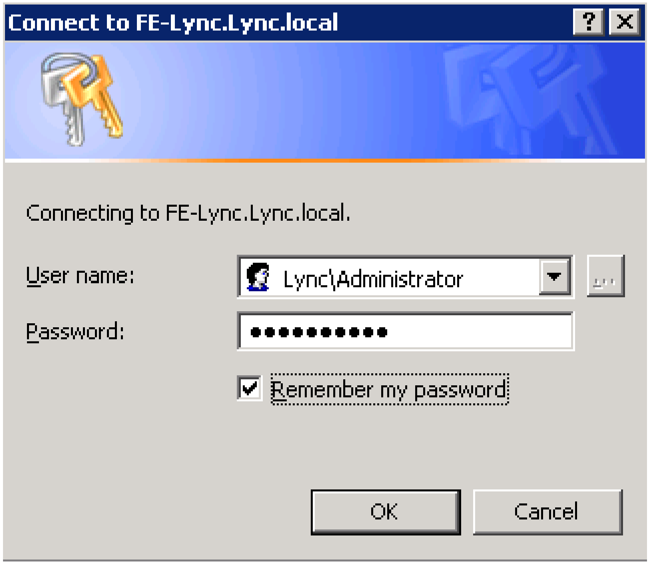
1. Open the Communication Server Control Panel (CSCP), click **Start**, select **All Programs**, and select **Lync Server Control Panel**.

Figure 3-16: Lync Server Control Panel



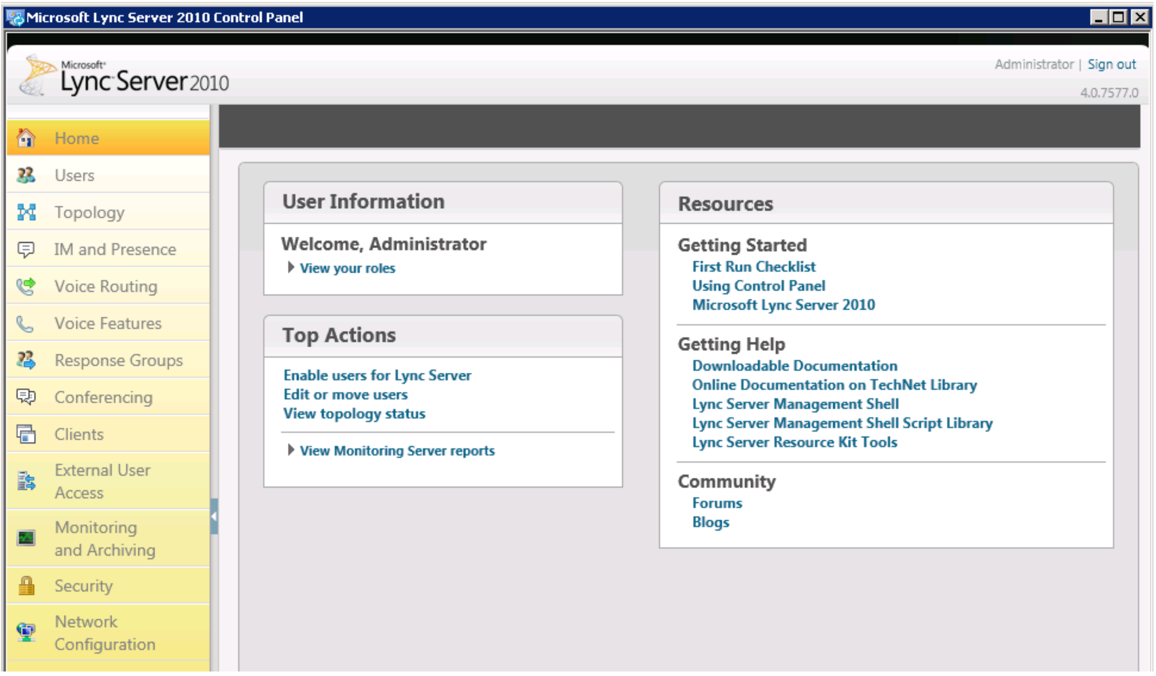
2. You are prompted for credentials; enter your domain username and password.

Figure 3-17: Lync Server Credentials



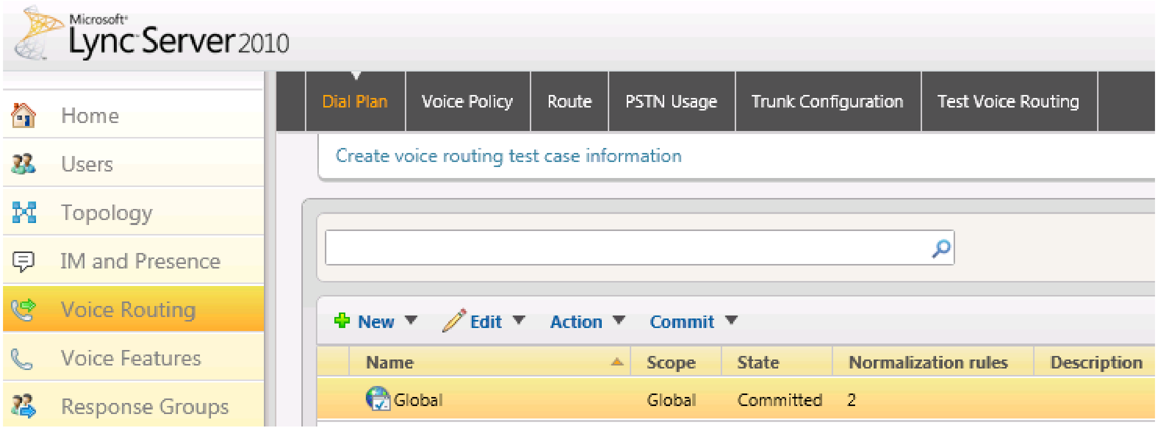
The CSCP Home page is displayed.

Figure 3-18: CSCP Home page



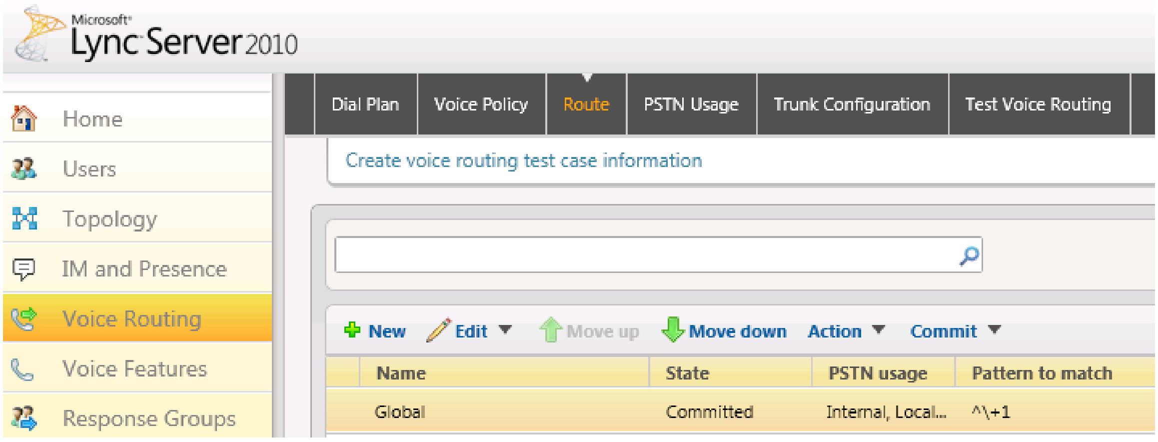
3. In the Navigation pane, select the **Voice Routing** option.

Figure 3-19: Voice Routing Option



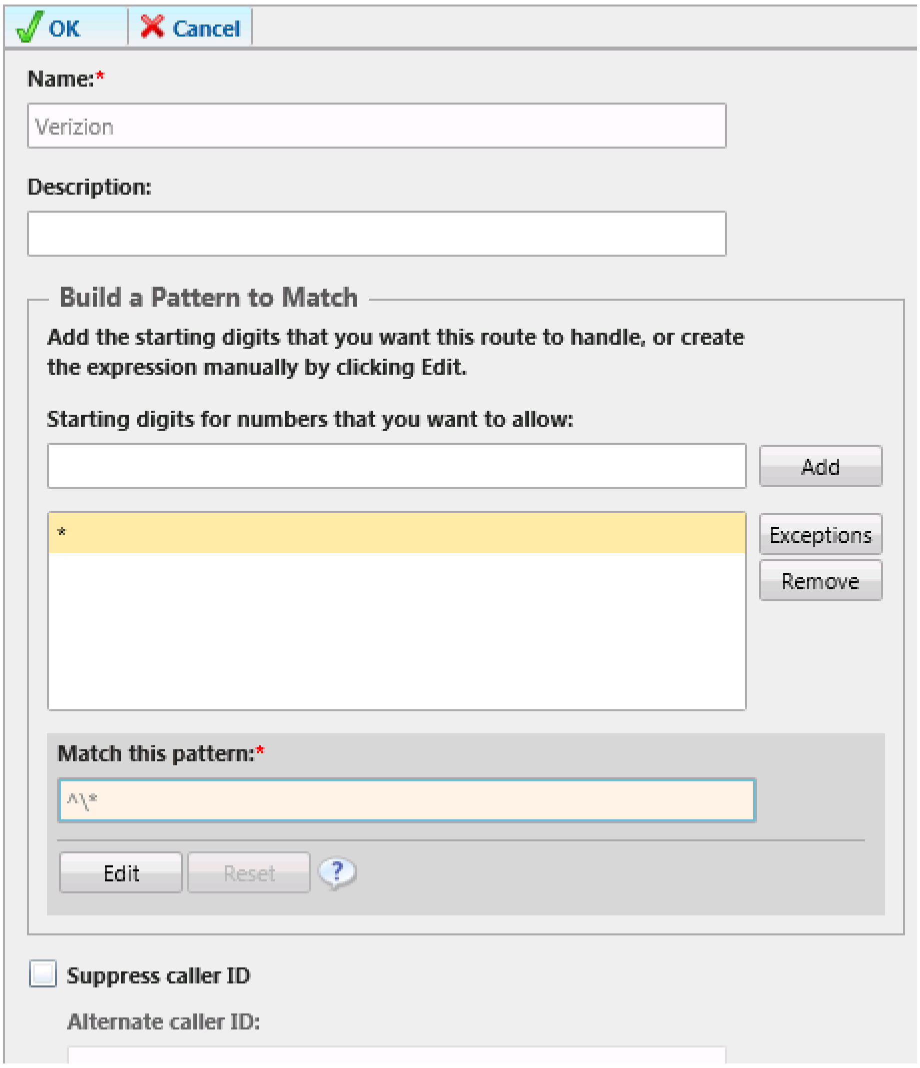
4. In the Voice Routing menu at the top of the page, select the **Route** option.

Figure 3-20: Route Option



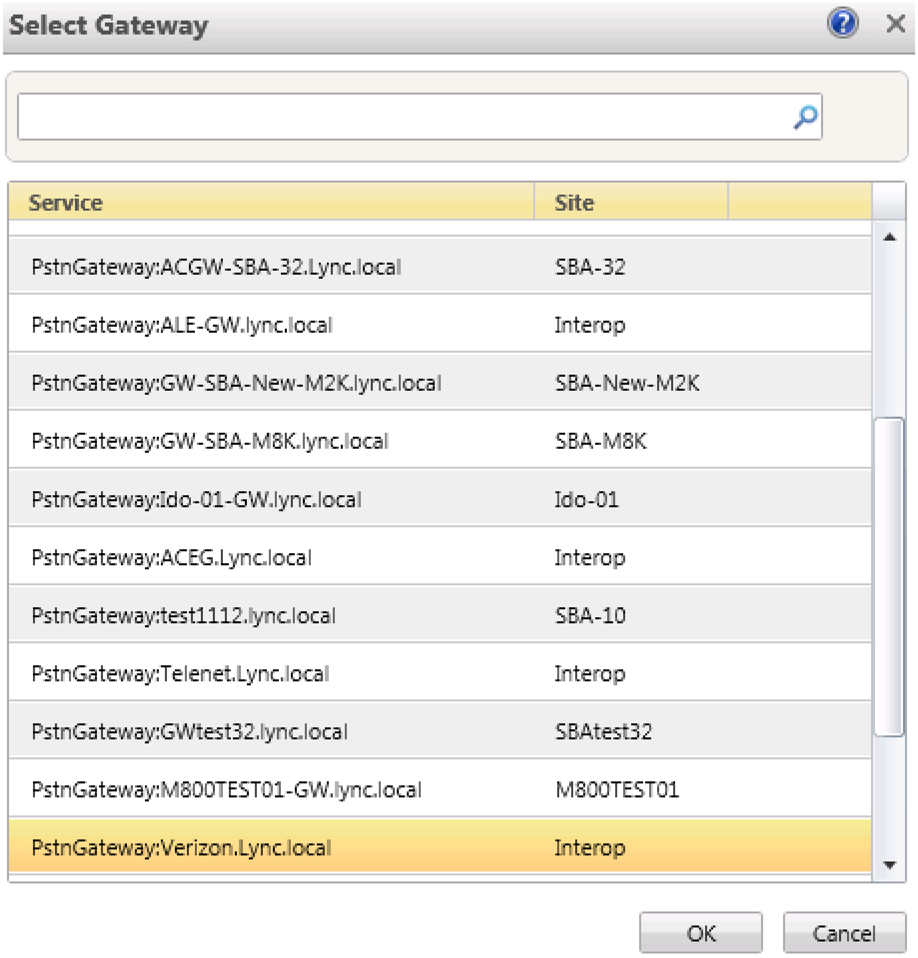
5. In the content area toolbar, click **New**..
6. In the Build a Pattern to Match pane, fill in a Name for this route (i.e. Verizon Route) and a Pattern to Match for the phone numbers you wish this route to handle. In this example, the pattern to match is '*', which means "to match all numbers".
7. Click **Add**.

Figure 3-21: Adding New Voice Route



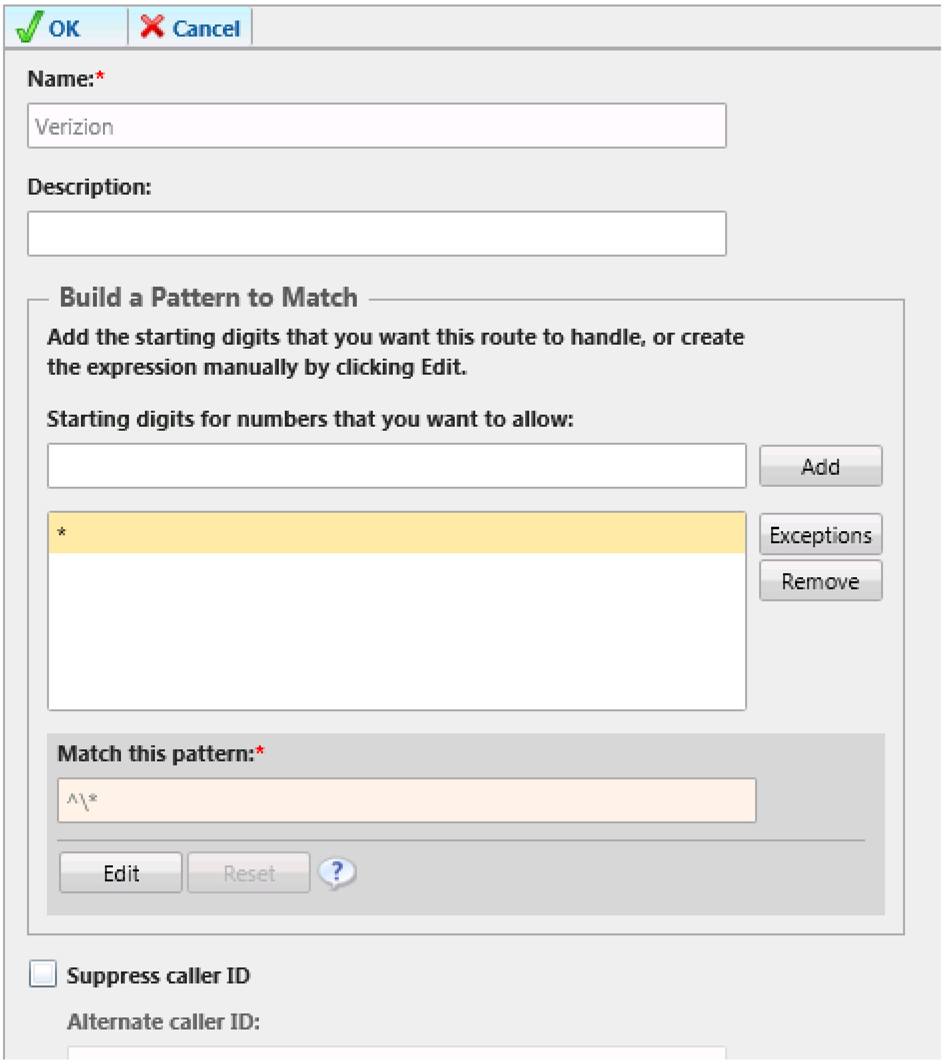
8. Associate the route with the E-SBC device IP/PSTN gateway you created above; scroll down to the Associated Gateways pane and click **Add**; a list of all the deployed gateways is displayed.

Figure 3-22: List of Deployed Gateways



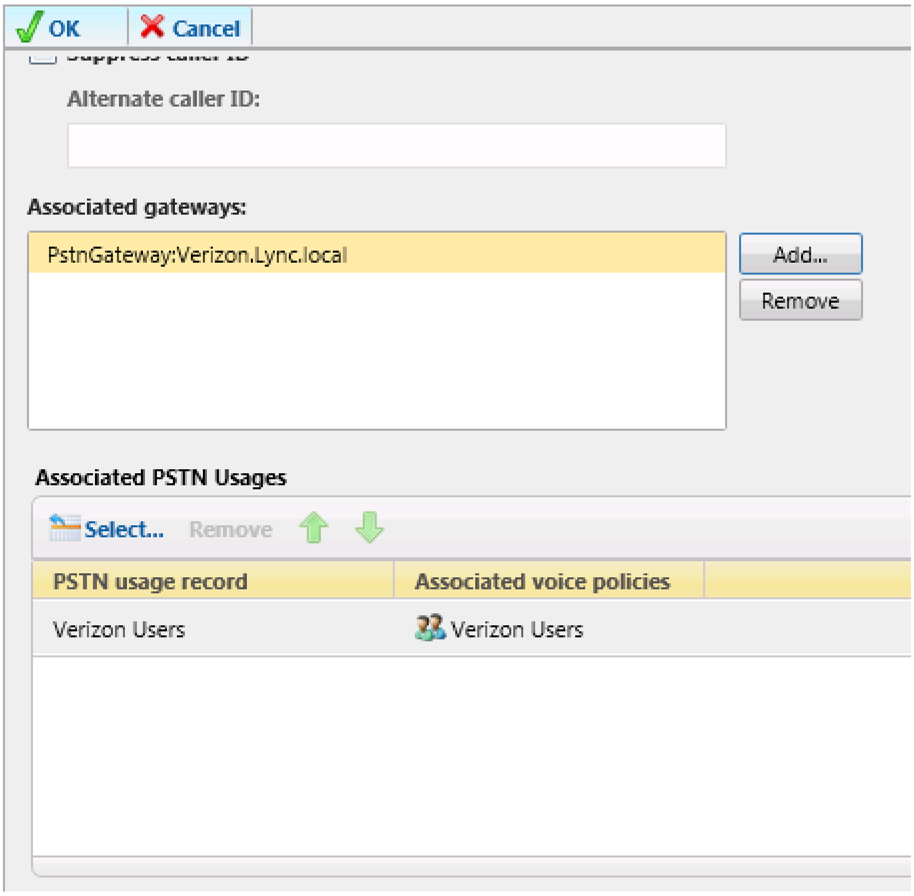
9. Select the E-SBC device PSTN Gateway you created above and click **OK**.

Figure 3-23: Selecting the E-SBC device PSTN Gateway



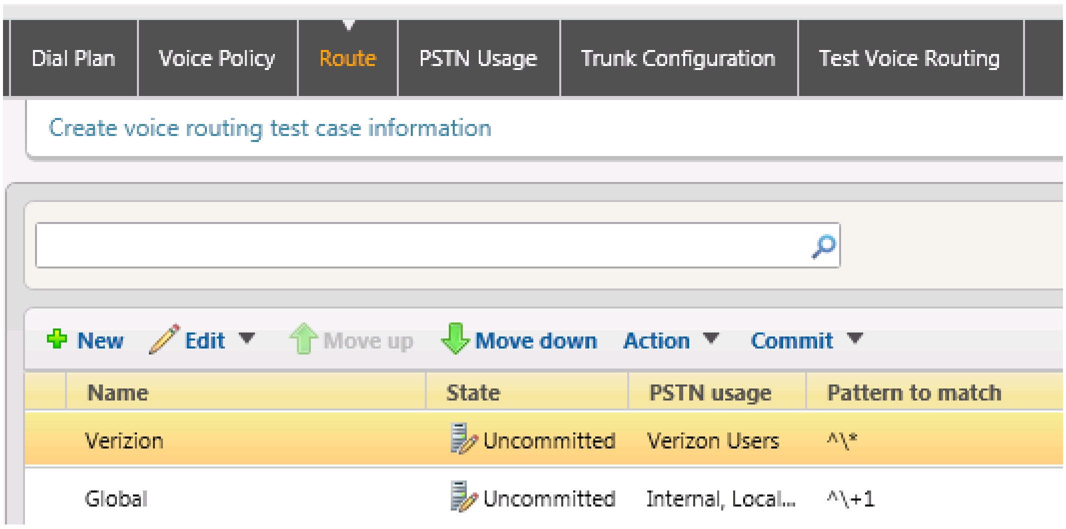
10. Associate a PSTN Usage to this route. In the **Associated PSTN Usages** toolbar, click **Select** and add the associated PSTN Usage.

Figure 3-24: Associating PSTN Usage to E-SBC Device



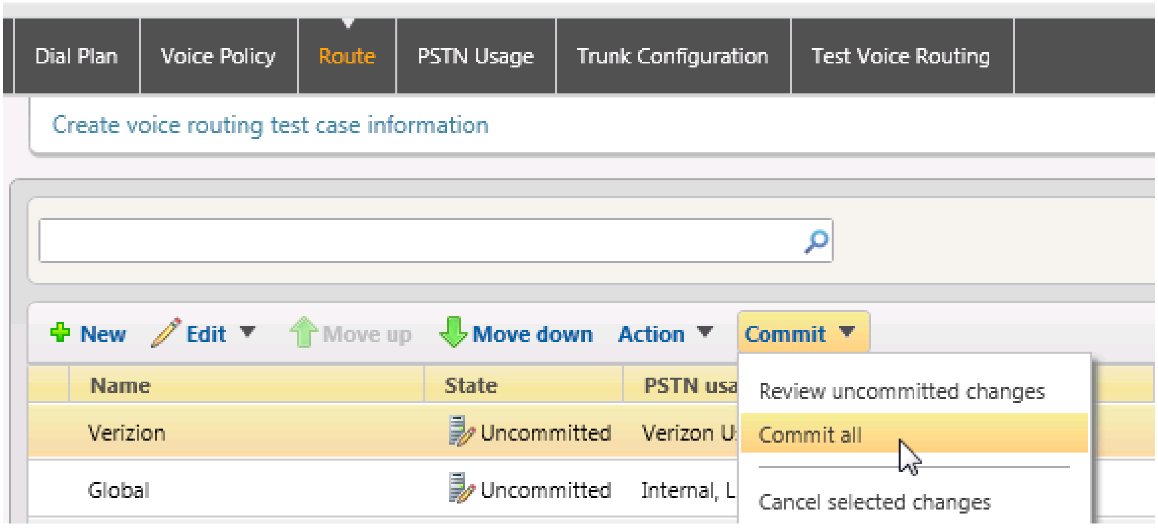
11. Click the **OK** button in the toolbar at the top of the New Voice Route pane.

Figure 3-25: Confirmation of New Voice Route



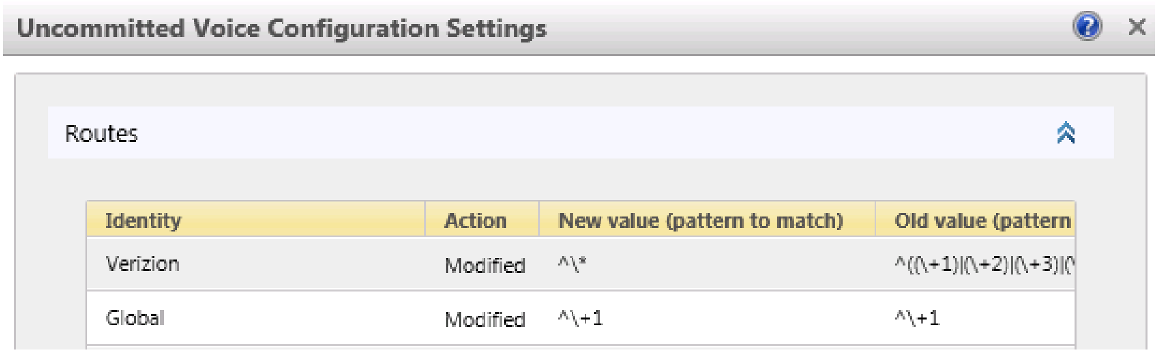
12. In the Content area Toolbar, click on the arrow adjacent to the **Commit** button; a drop-down menu is displayed; select the **Commit All** option.

Figure 3-26: Committing Voice Routes



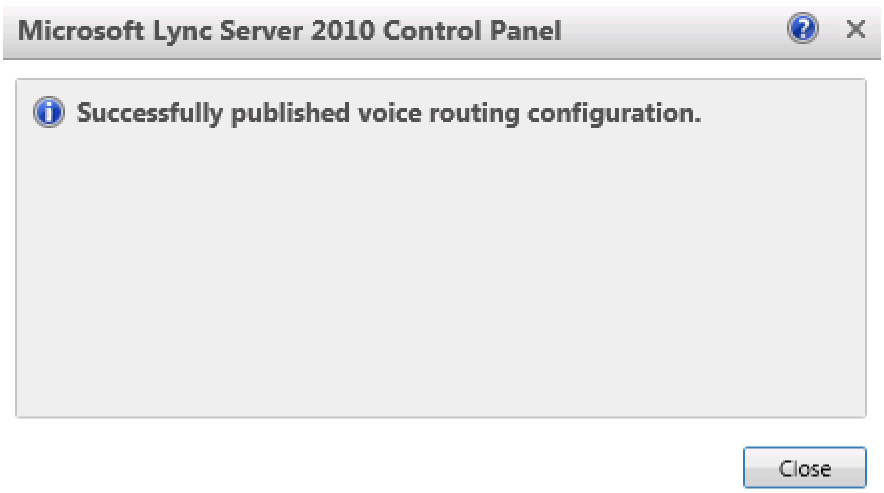
13. In the Uncommitted Voice Configuration Settings window, click **Commit**.

Figure 3-27: Uncommitted Voice Configuration Settings



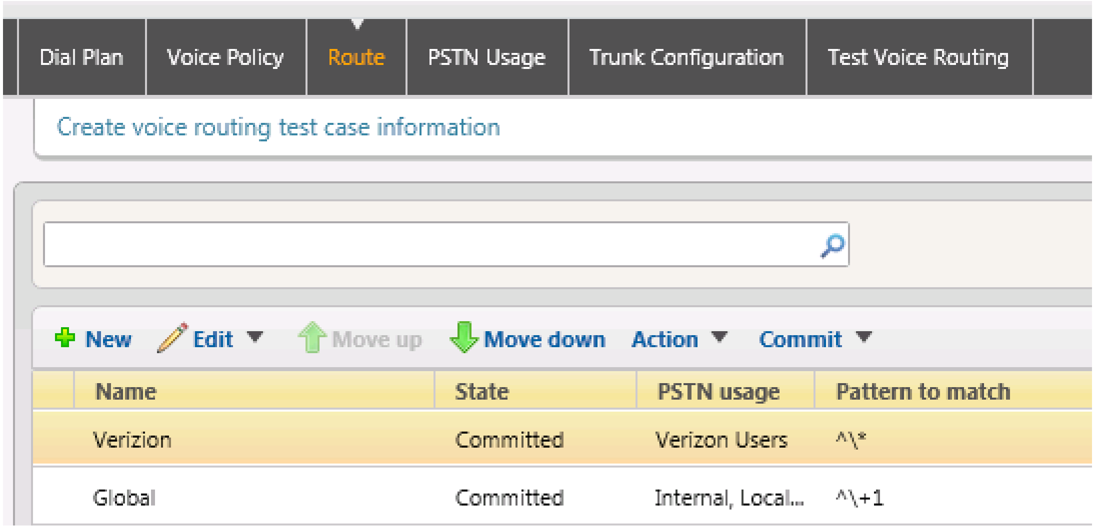
14. A message is displayed, confirming a successful voice routing configuration; in the **Microsoft Lync Server 2010 Control Panel** prompt; click **Close**.

Figure 3-28: Voice Routing Configuration Confirmation



The new committed Route is now displayed in the Voice Routing screen.

Figure 3-29: Voice Routing Screen Displaying Committed Routes



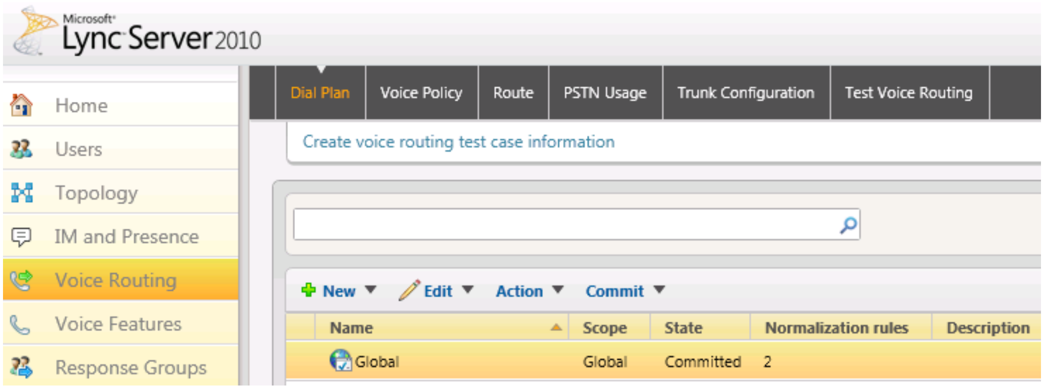
3.4 Configuring the 'Dial Plan' on the Lync Server 2010

This section describes how to configure a 'Dial Plan' on the Lync server and associate it with 'Normalization' Rules.

➢ **To configure the 'Dial Plan on the Lync Server 2010:**

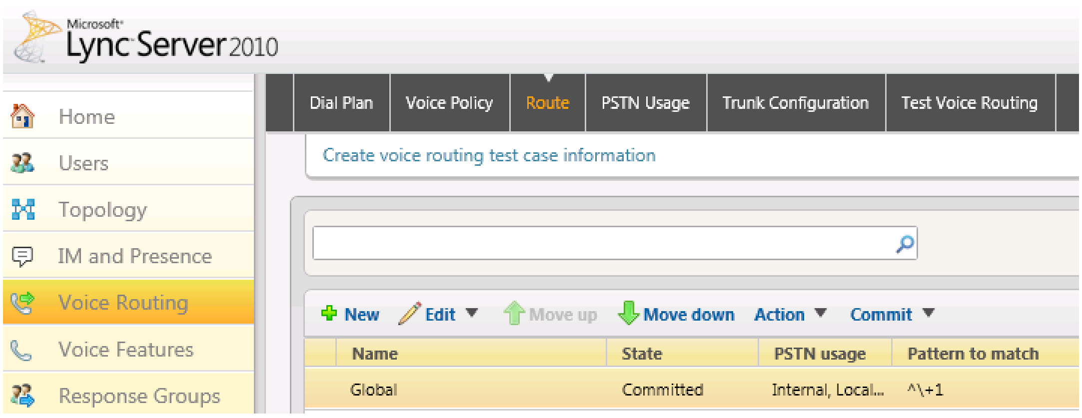
1. In the Navigation pane, select the **Voice Routing** option.

Figure 3-30: Voice Routing Option



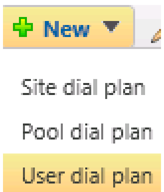
2. In the Voice Routing menu at the top of the page, select the **Dial Plan** option.

Figure 3-31: Dial Plan Option



3. In the content area toolbar, click 

. and select the Dial Plan type.



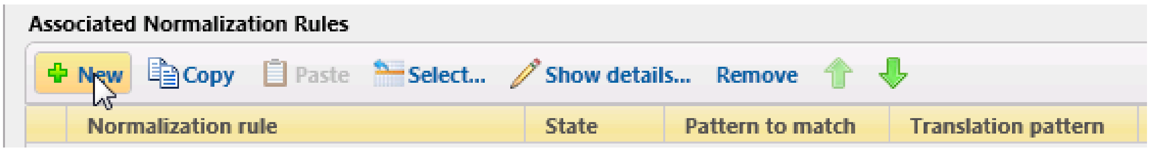
In this example, we create a 'User Dial Plan'.

4. In the New Dial Plan pane, fill in a Name for this Dial Plan (i.e. Verizon Users) and create new 'Associated Normalization Rules' to match for the dialing phone numbers you wish this Dial Plan to handle.

5. Click **Add**.

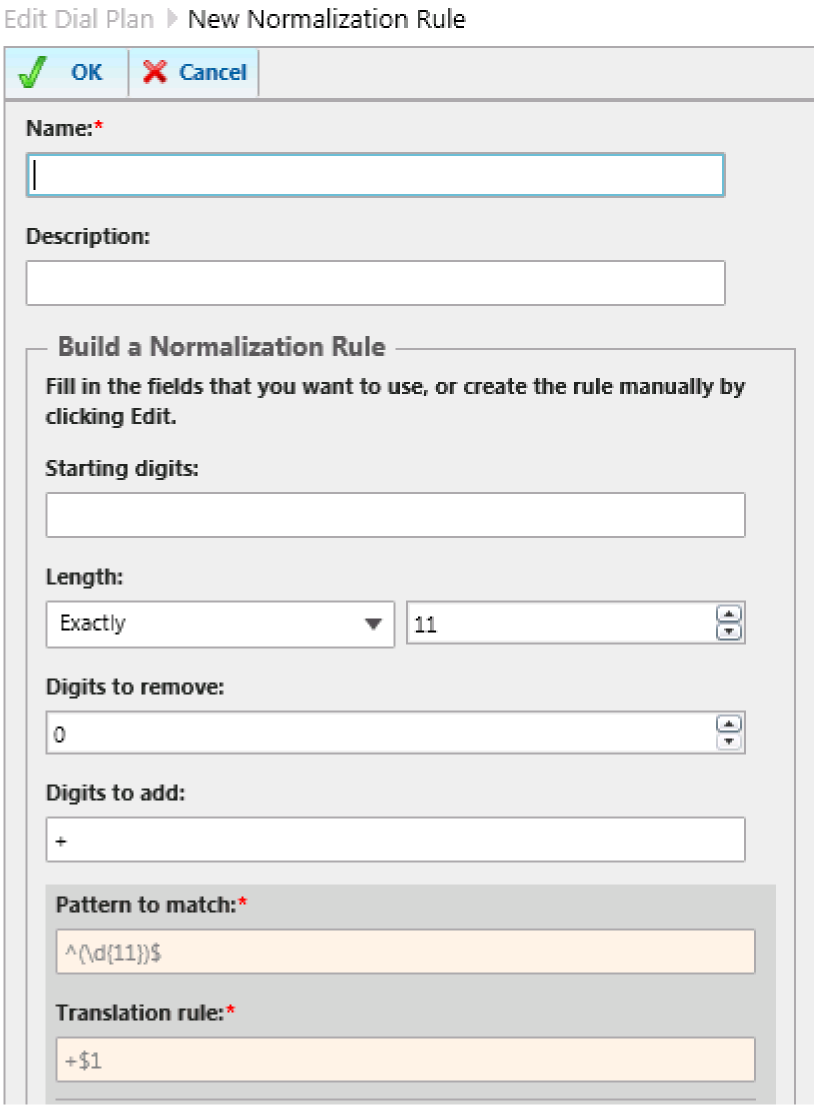
6. For creating 'New Normalization Rules' in the content area toolbar, click the  option.

Figure 3-32: New Normalization Rules Option



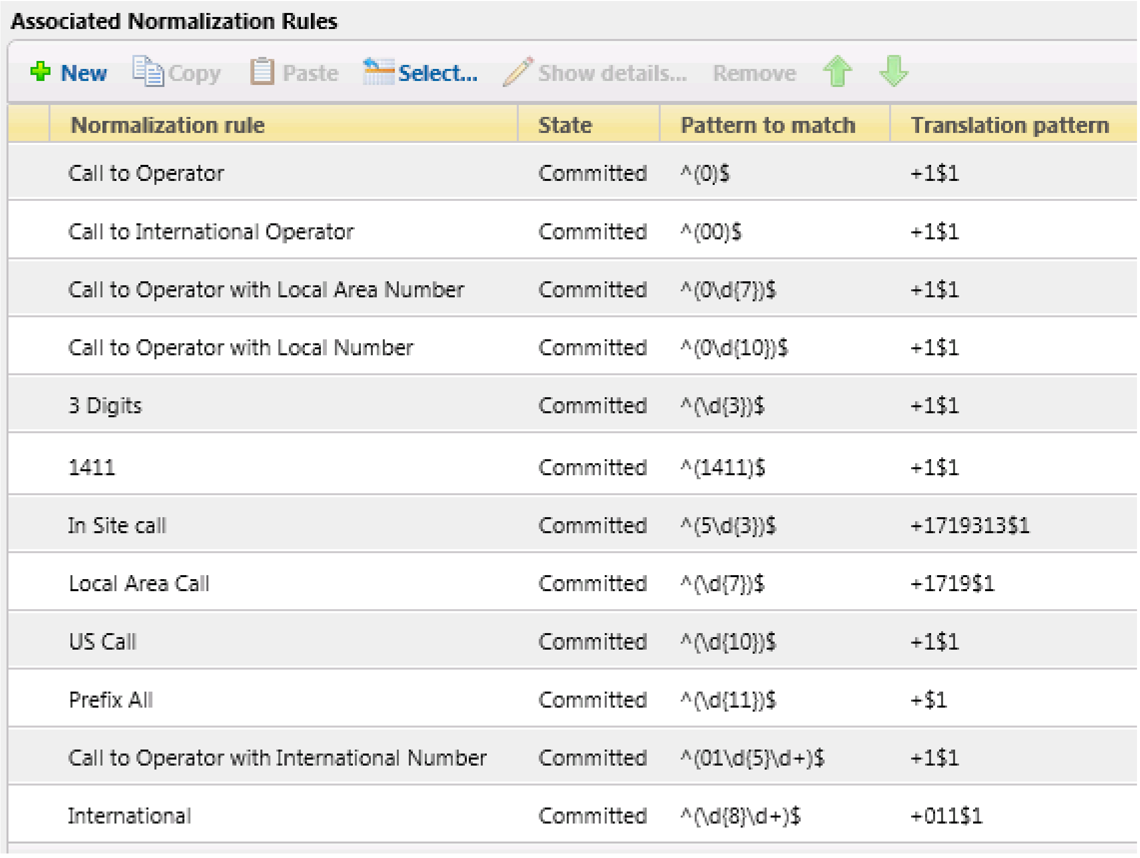
7. In the New Normalization Rules pane, fill in a name for these Normalization rules (i.e. Verizon Users) and the Normalization rules for the phone numbers you wish this device to handle.

Figure 3-33: New Normalization Rules Option



8. In this example, we created the follow **Normalization Rules**:

Figure 3-34: Normalization Rules



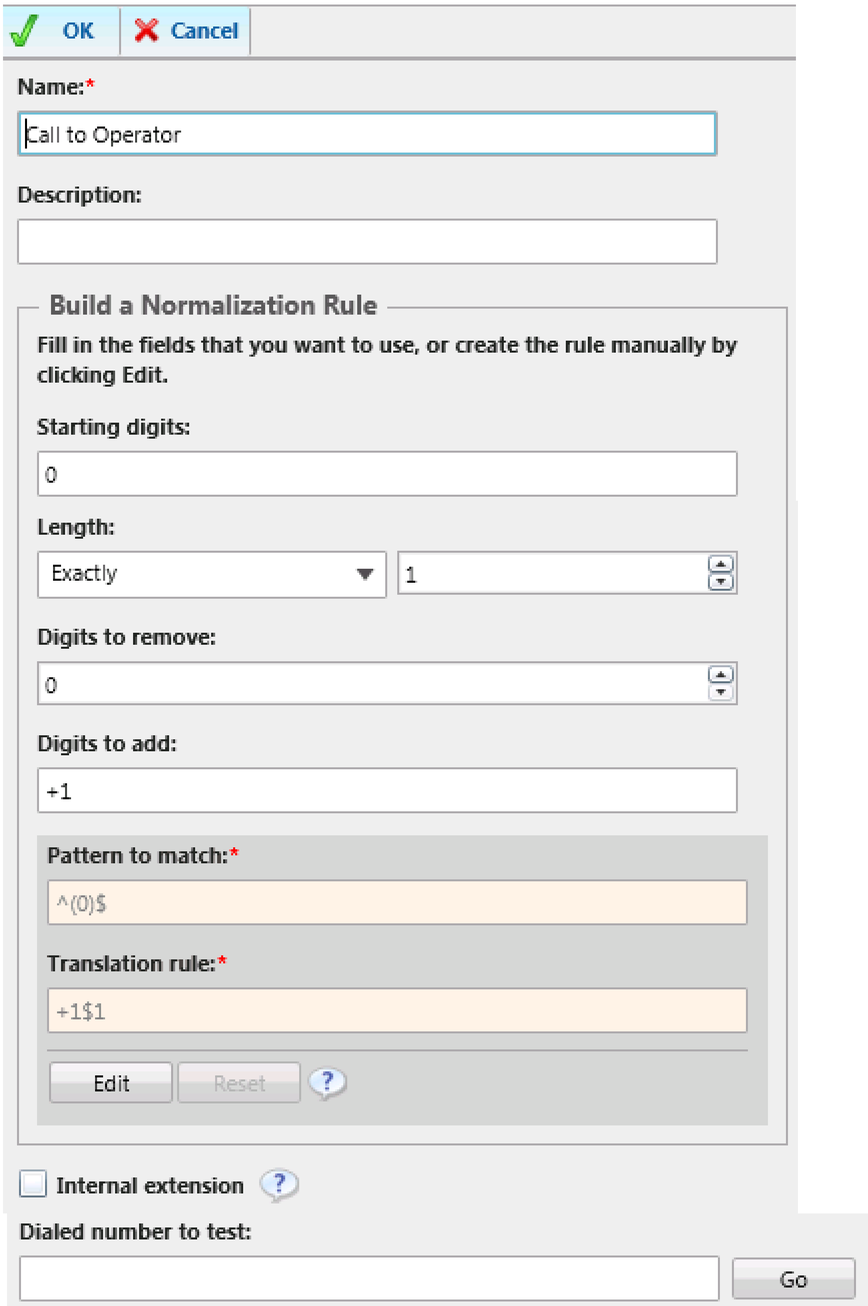
Normalization rule	State	Pattern to match	Translation pattern
Call to Operator	Committed	^(0)$	+1$1
Call to International Operator	Committed	^(00)$	+1$1
Call to Operator with Local Area Number	Committed	^(0\d{7})$	+1$1
Call to Operator with Local Number	Committed	^(0\d{10})$	+1$1
3 Digits	Committed	^(\d{3})$	+1$1
1411	Committed	^(1411)$	+1$1
In Site call	Committed	^(5\d{3})$	+1719313$1
Local Area Call	Committed	^(\d{7})$	+1719$1
US Call	Committed	^(\d{10})$	+1$1
Prefix All	Committed	^(\d{11})$	+$1
Call to Operator with International Number	Committed	^(01\d{5}\d+)$	+1$1
International	Committed	^(\d{8}\d+)$	+011$1

9. Example (and Translation) of **Normalization Rules**:
   - **Call to Operator:** Dial exactly '0' the rule add '+1' → Lync will send to the GW '+10'.
   - **Call to International Operator:** Dial exactly '00' the rule add '+1' → Lync will send to the GW '+100'.
   - **Call to Operator with Local Area Number:** Dial number that start with '0' and more 7 digits ('0xxxxxxx'), the rule add '+1' → Lync will send to the GW '+10xxxxxxx'.
   - **Call to Operator with Local Number:** Dial number that start with '0' and more 11 digits ('0xxxxxxxxxxx'), the rule add '+1' → Lync will send to the GW '+10xxxxxxxxxxx'.
   - **3 Digits:** Dial 3 digits ('xxx') the rule add '+1' → Lync will send to the GW '+1xxx'. This rule using for emergency call (e.g., 911).
   - 1411: Dial exactly '1411' the rule add '+1' → Lync will send to the GW '+11411'.
   - **In Site call:** Dial 4 digits ('xxxx') the rule complete the number to E-164 standard number and add '+1719313' → This rule affect only in site calls and not sending to the GW.
   - **Local Area Call:** Dial 7 digits ('xxxxxxx') the rule add '+1719' → Lync will send to the GW '+1719xxxxxxx'.

- **US Call:** Dial 10 digits ('xxxxxxxxxx') the rule will add '+1' → Lync will send to the GW '+1xxxxxxxxxx'.
- **Prefix All:** Dial 11 digits ('xxxxxxxxxxx') the rule will add '+' → '+1xxxxxxxxxx' will send to the GW.
- **Call to Operator with International Number:** Dial 8 or more digits that starting with '01' ('01xxxxxxxxxxxx') the rule add '+1' → '+101xxxxxxxxxxxx' will send to the GW.
- **International:** Dial 9 or more digits ('xxxxxxxxxxx') the rule add '+011' → '+011xxxxxxxxxxx' will send to the GW.

10. Example of Normalization Rules:

Figure 3-35: Example: Normalization Rules



- **Name:** The Rule name.
- **Description:** Rule description.
- **Starting digits:** The Digit/s Number/s that the Normalization Rule starting.
- **Length:** Three options: 'At Least', 'Exactly', and 'Any'. And then the numbers of dial digits (e.g., exactly 1 digit).
- **Digits to remove:** The number of Digits that need to remove from the dialed number. (e.g., Don't remove digits)
- **Digits to add:** The number of Digits that need to add to the dialed number. (e.g., add '+1' as prefix)
- **Pattern to match:** This field automatic filled, based on the previous fields.
- **Translation rule:** This field automatic filled, based on the previous fields.
- **Dialed number to test:** Use for test the rule before approve it.

11. Gateway number manipulation. The number manipulation configuration of the AudioCodes device is later described in Section 4.13 on page 83.

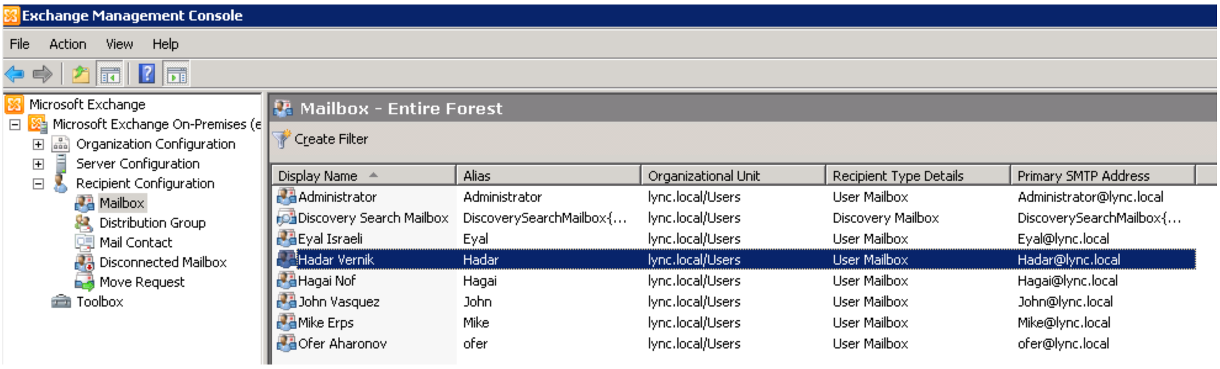
3.5 Enabling Voice Mail in the Exchange Server

This section describes how to enable voice mail for the user on the Exchange Server.

➢ **to enable user for voice mail in the Exchange Server:**

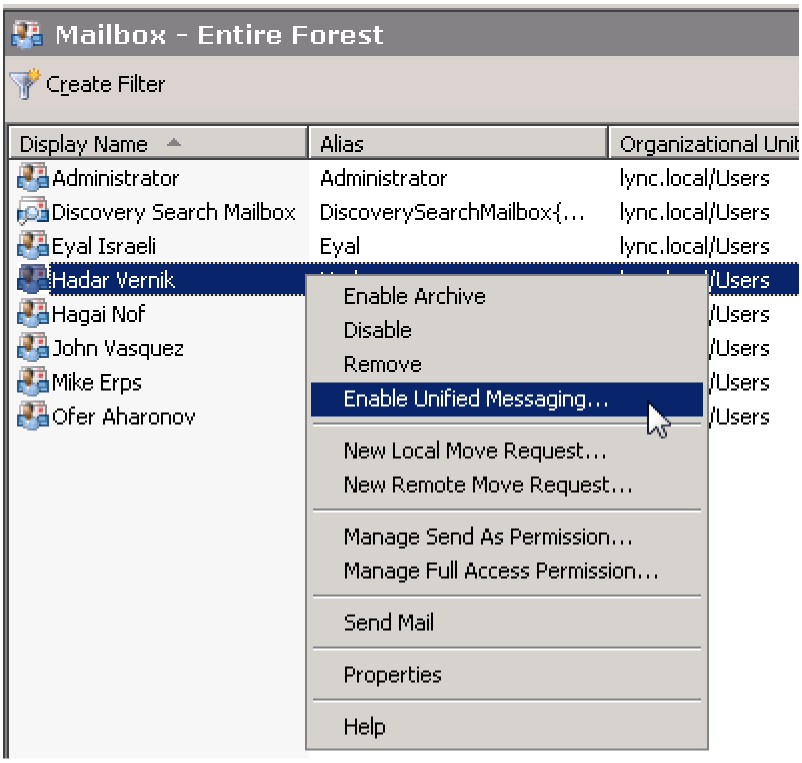
1. Open the Exchange Management Console.

Figure 3-36: Exchange Management Console



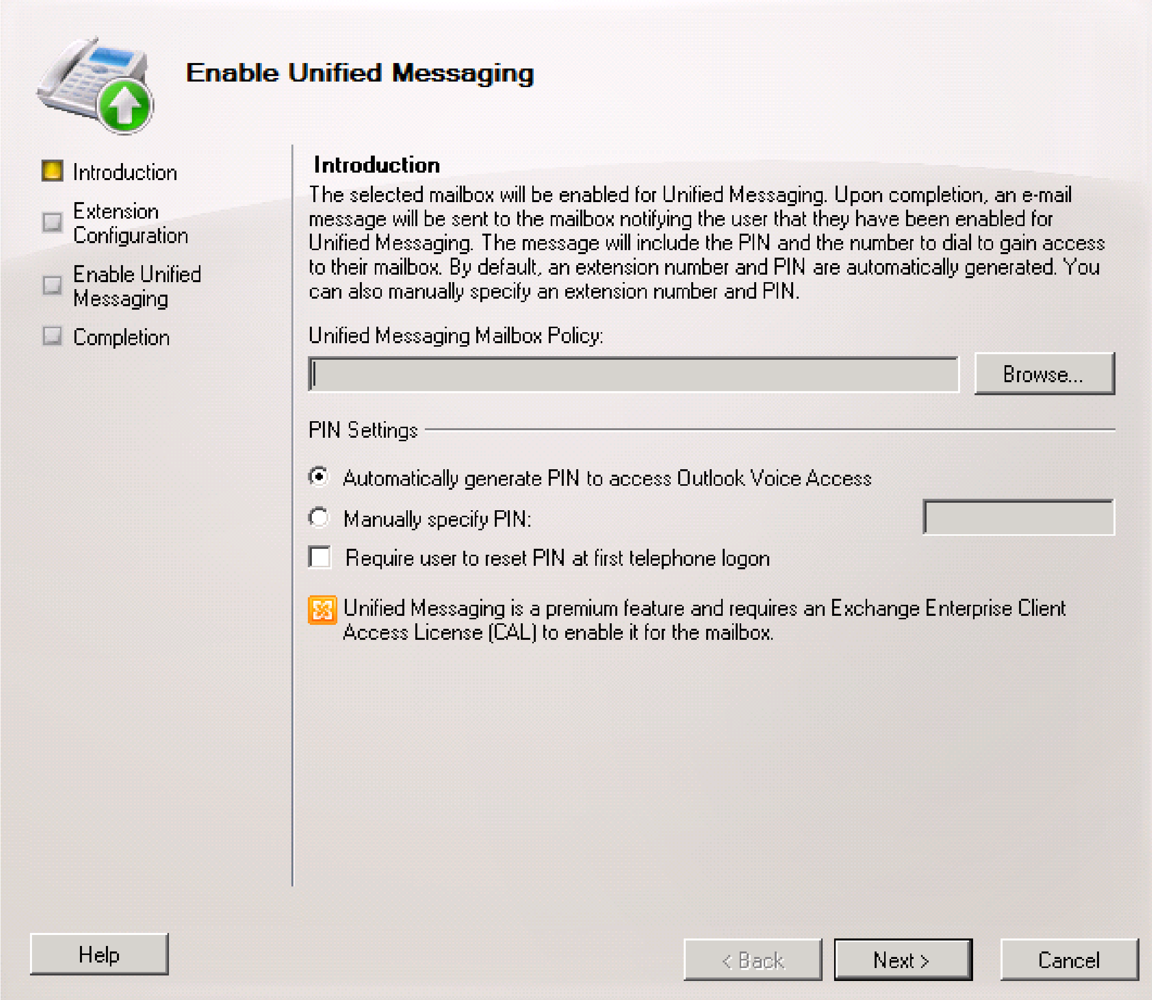
2. In the Navigation pane, select the **'Recipient Configuration >> Mailbox'** Tree folder.
3. Select user for enabling voice mail from the list.
4. From the dropdown list, right-click on the selected user and choose **Enable Unified Messaging**.

Figure 3-37: Mailbox – Entire Forest



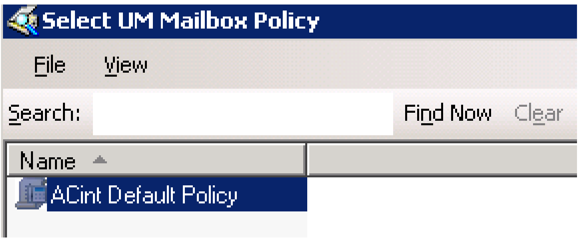
The Enable Unified Messaging Wizard – Introduction screen is displayed.

Figure 3-38: Enable Unified Messaging



5. In the new dialog pane, click the **Browse** button; the Select UM Mailbox Policy dialog is displayed.

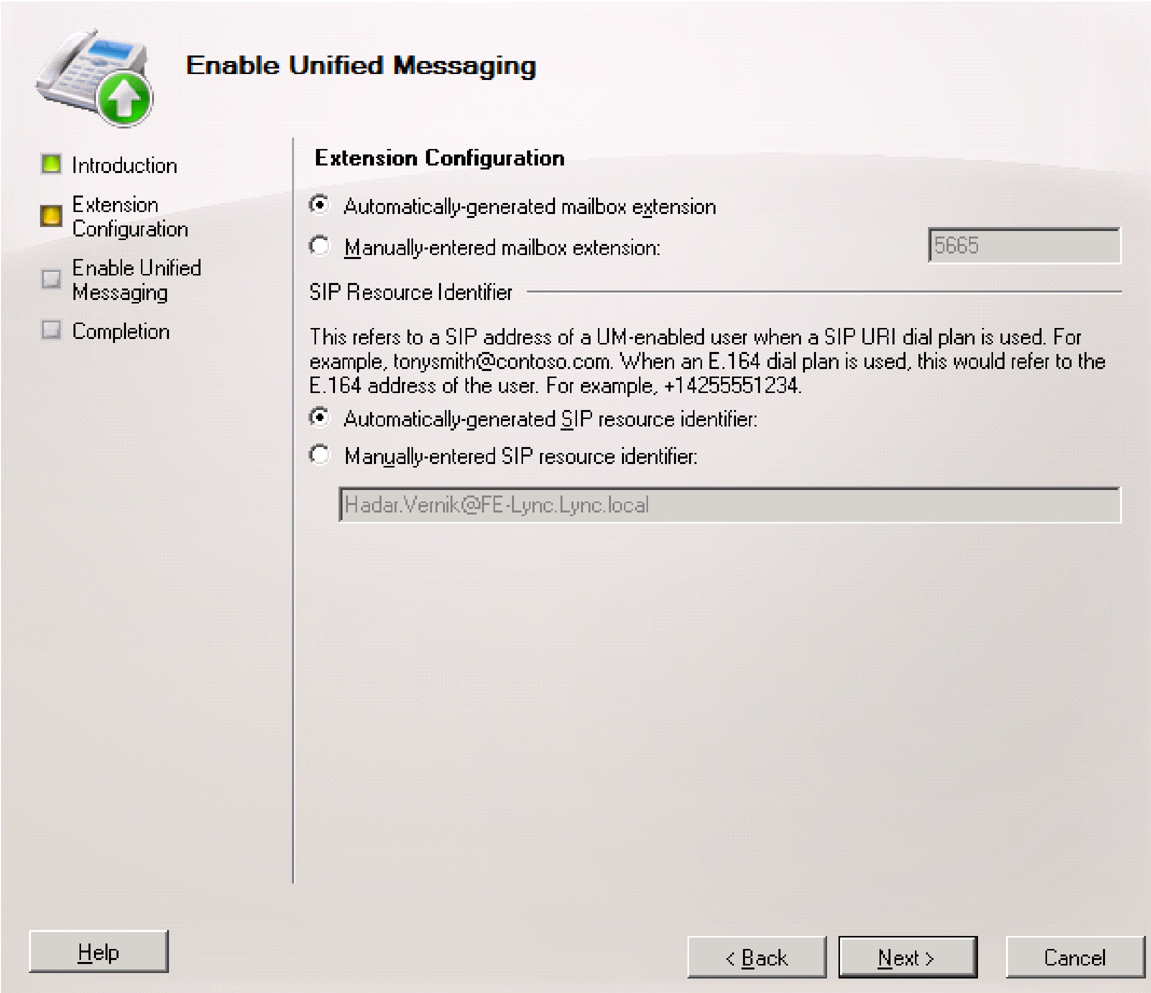
Figure 3-39: Select UM Mailbox Policy



6. Select **ACint Default Policy**.

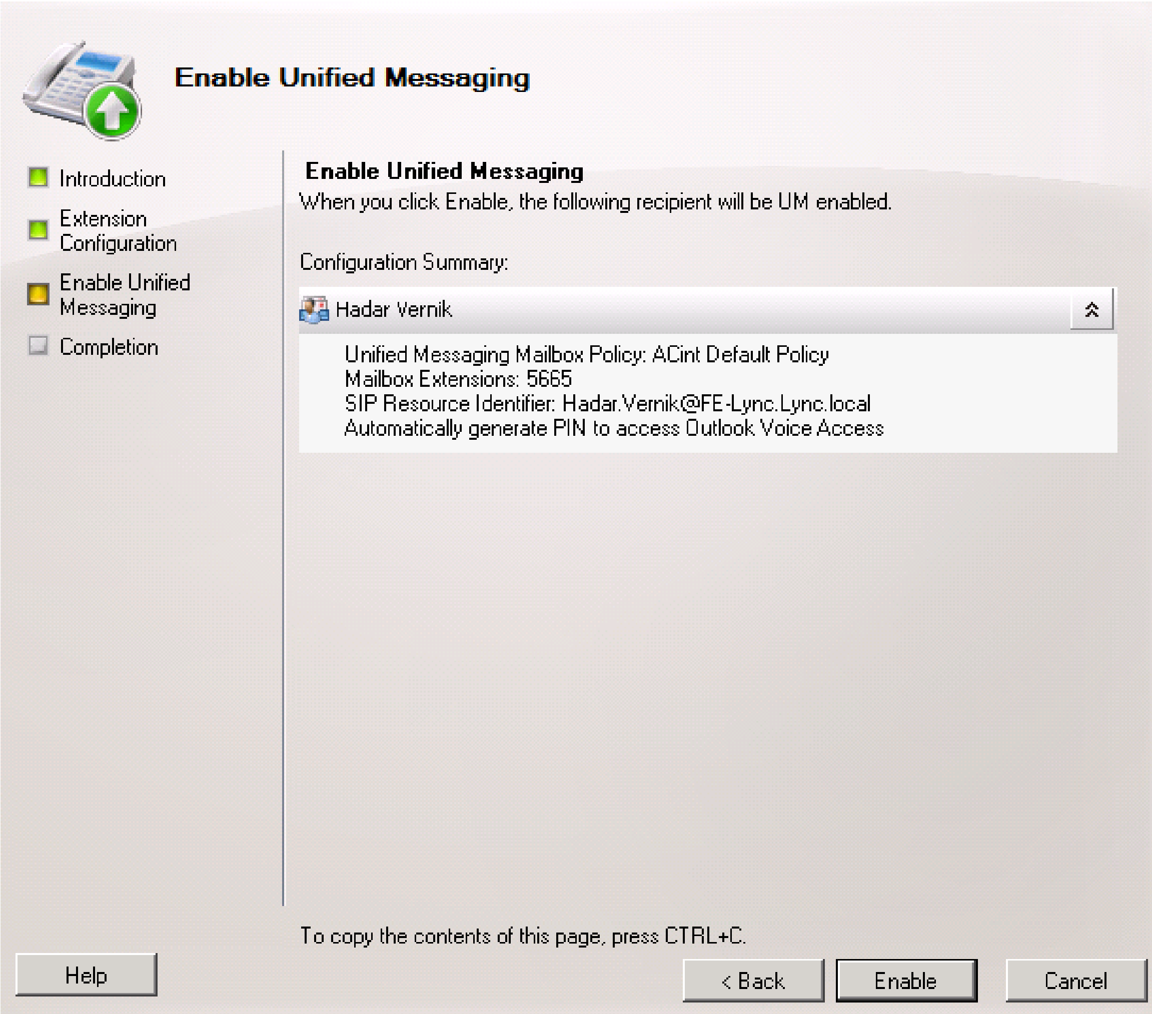
7. Click **Next**; the Enable Unified Messaging Wizard – Extension Configuration screen is displayed.

Figure 3-40: Enable Unified Messaging Wizard – Extension Configuration



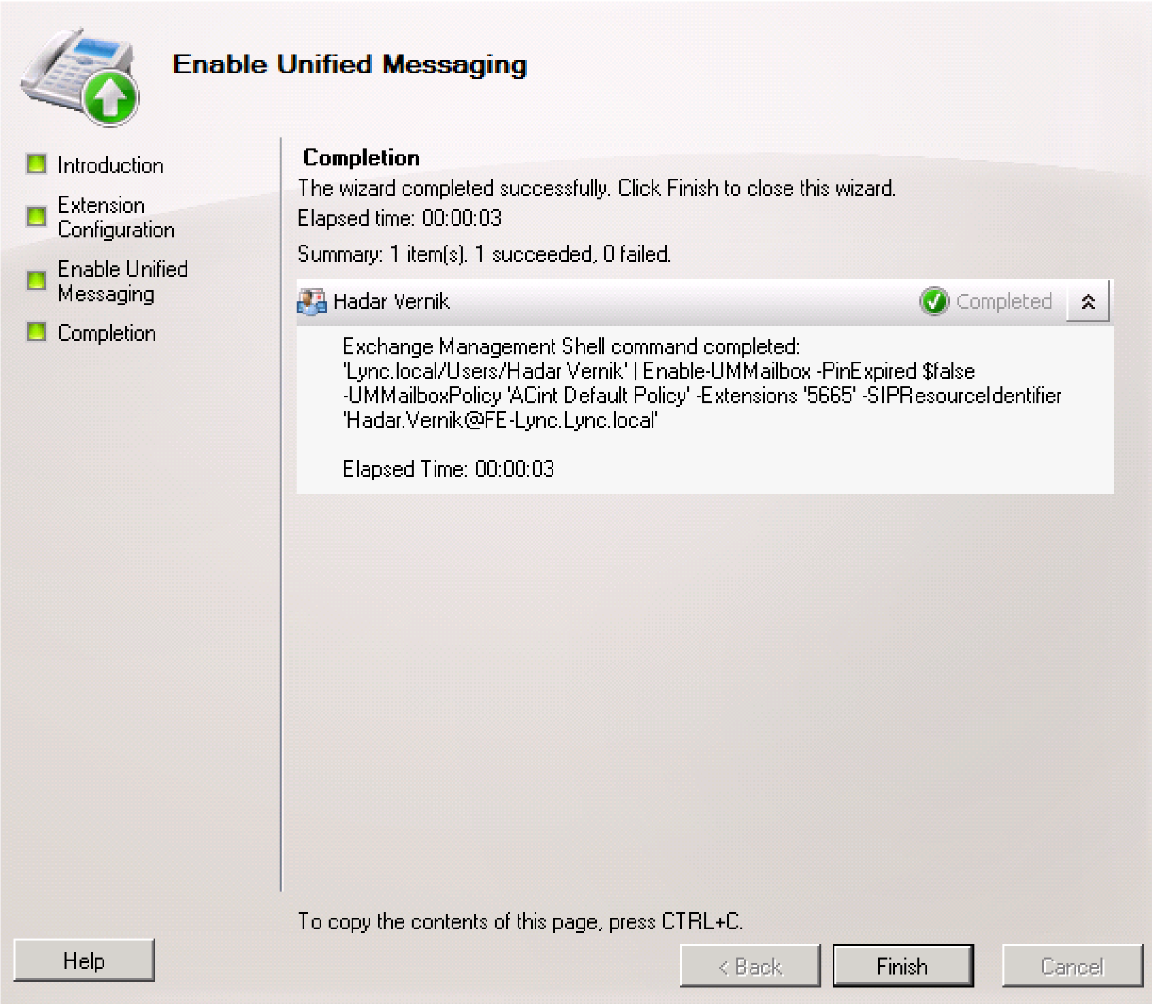
8. Click **Next**; the Enable Unified Messaging Wizard Configuration Summary screen is displayed.

Figure 3-41: Enable Unified Messaging Wizard – Configuration Summary



9. Click **Enable**; the Enable Unified Messaging Wizard – Completion screen is displayed.

Figure 3-42: Enable Unified Messaging Wizard – Completion



10. Click **Finish**.

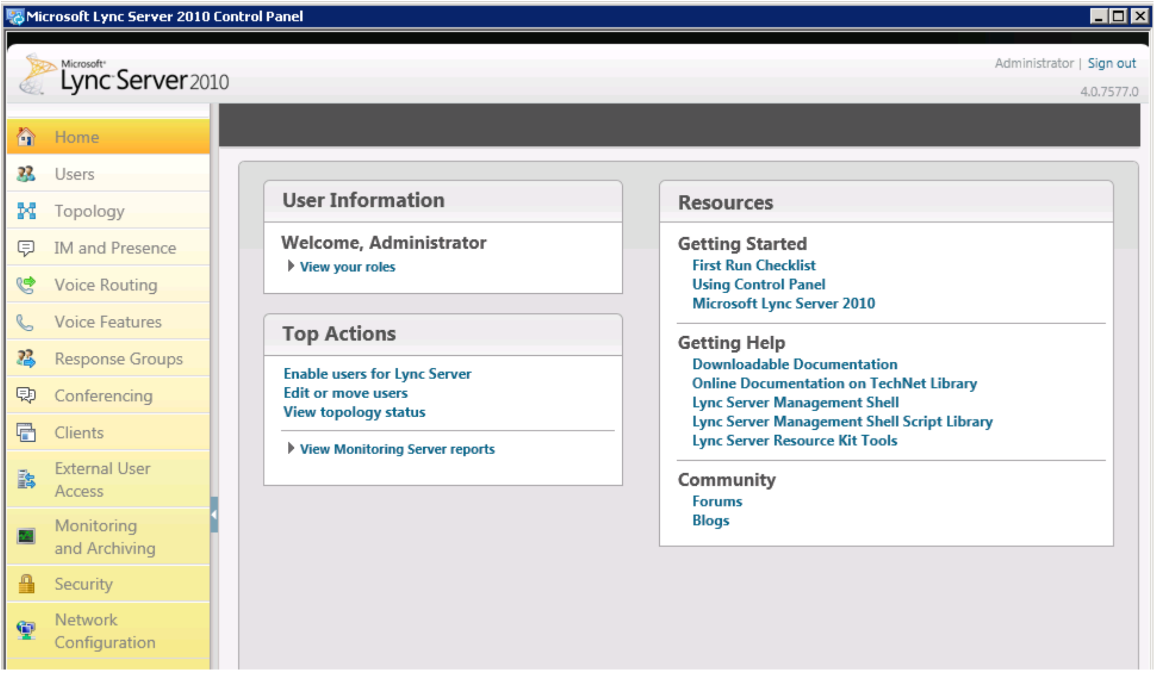
3.6 Enabling User for Lync Voice

This section describes how to enable Lync Voice Mail for a specific Lync user.

➢ To enable Lync Voice Mail:

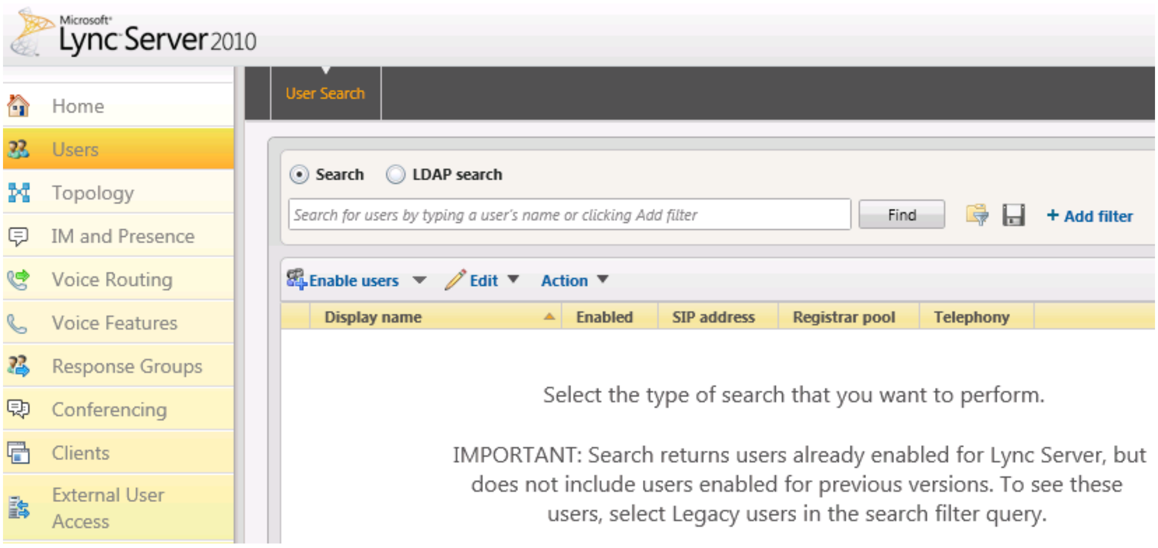
1. Open the Lync Server 2010 (Start menu > Programs > Microsoft Lync Server 2010 > Lync Server Control Panel); the CSCP Home page is displayed.

Figure 3-43: CSCP Home page



2. In the Navigation pane, select the **'Users'** option.

Figure 3-44: Users Option

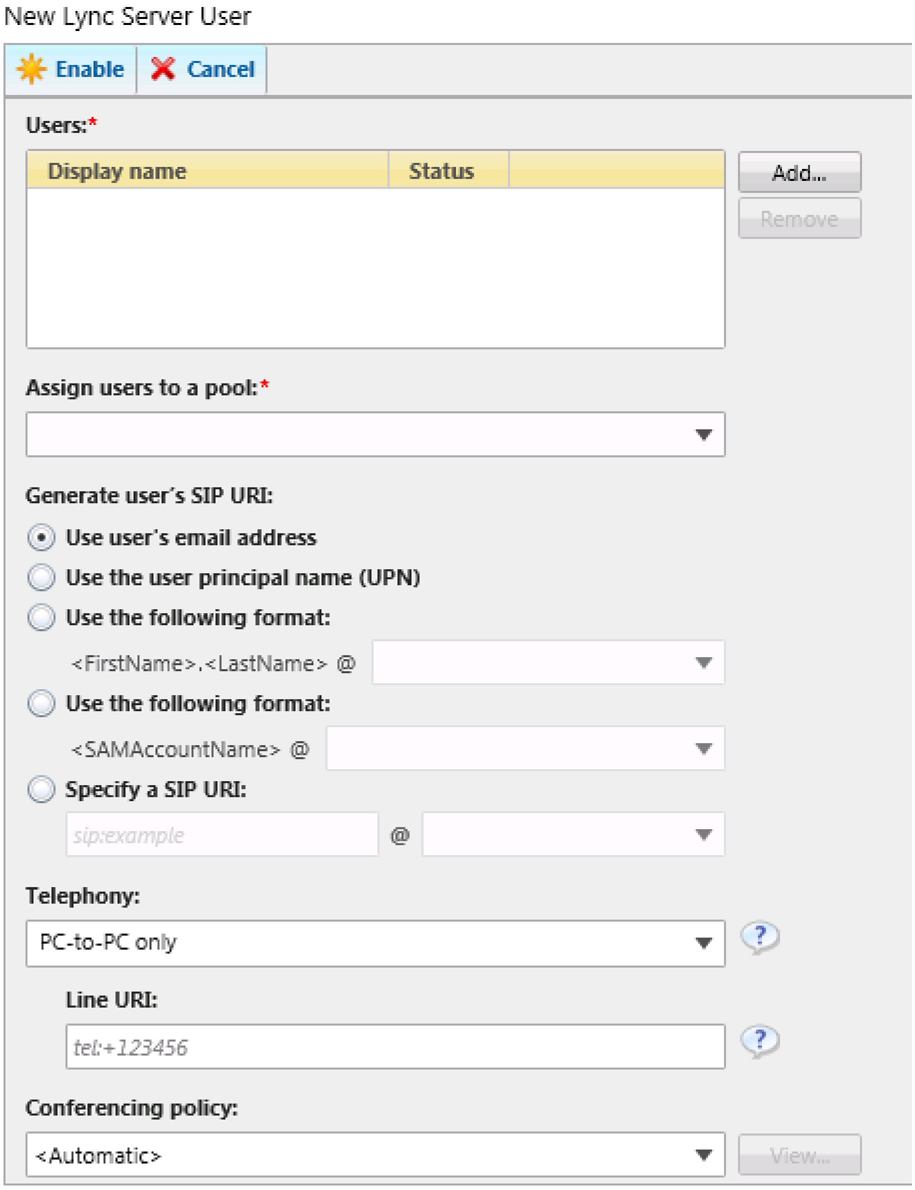


3. In the content area toolbar, click 

.

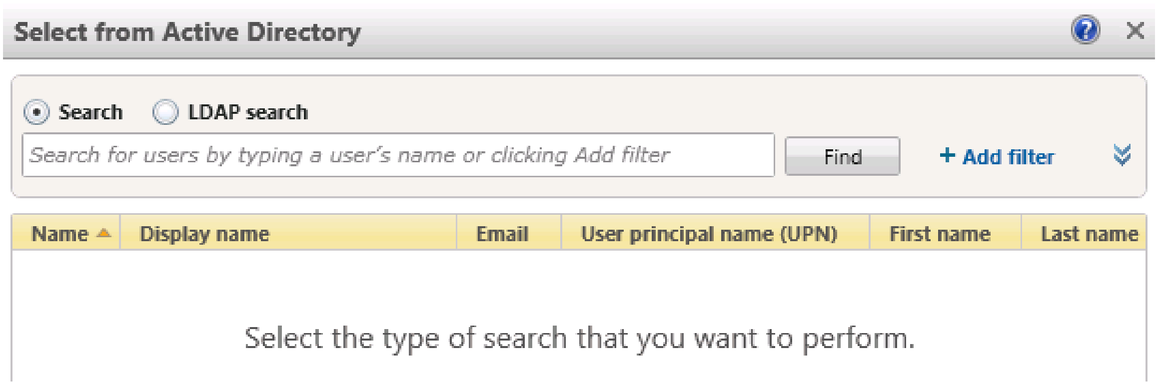
The New Lync Server User screen is displayed.

Figure 3-45: New Lync Server User



4. In the Users: click  ; the Lync search engine opens.

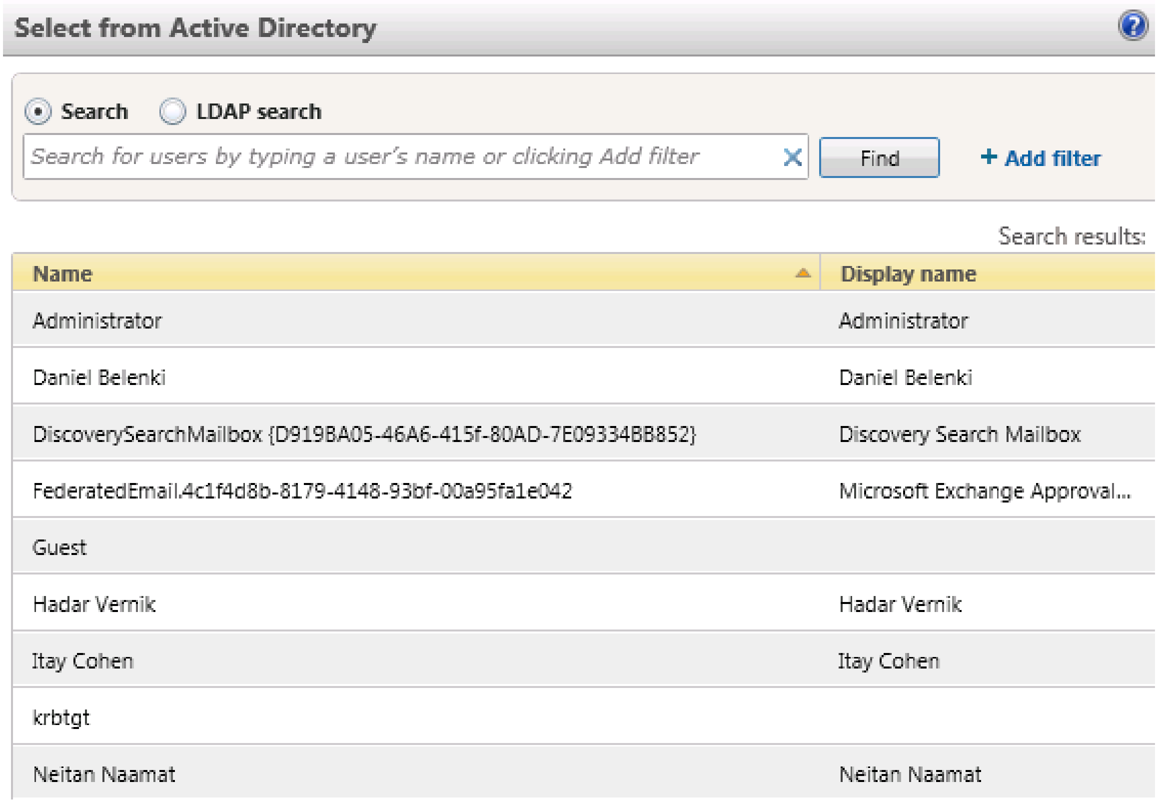
Figure 3-46: Lync Search Engine



5. Click 

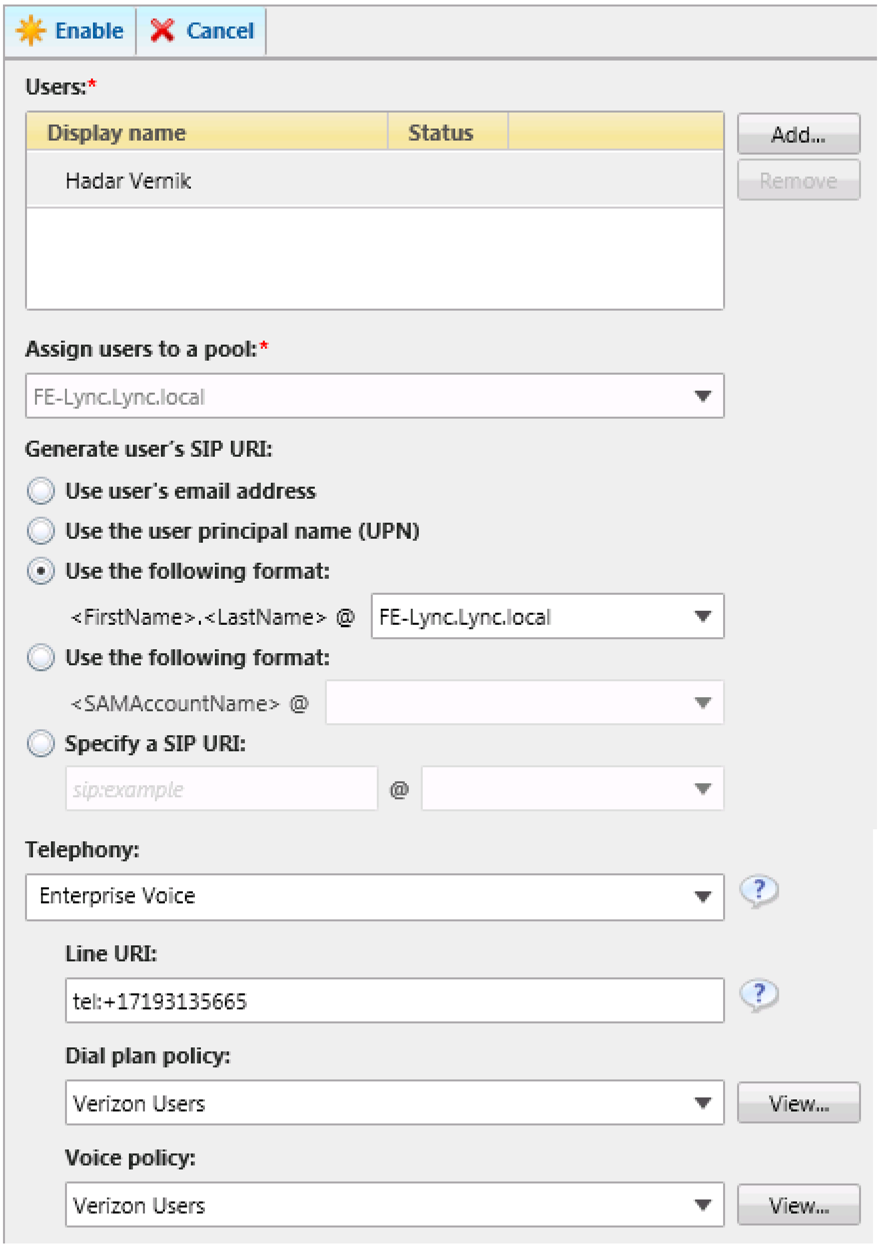
for searching un-enable voice user in the Active Directory.

Figure 3-47: Search Un-enable Voice Users



6. Select one of the users from the list and click **OK**.

Figure 3-48: User Properties



7. In the 'Assign users to a pool' field, select the Pools that you working with it.
8. In the 'Generate user's SIP URI' field, select the format.
9. In the 'Telephony' field, select **Enterprise Voice**.
10. In the 'Line URI' field, add the **tel:<phone number> (e.g., tel:17193135665)**.
11. In the 'Dial plan policy' field, select the dial plan.
12. In the 'Voice policy' select the voice policy.
13. Click **Enable**.

4 Configuring AudioCodes E-SBC device

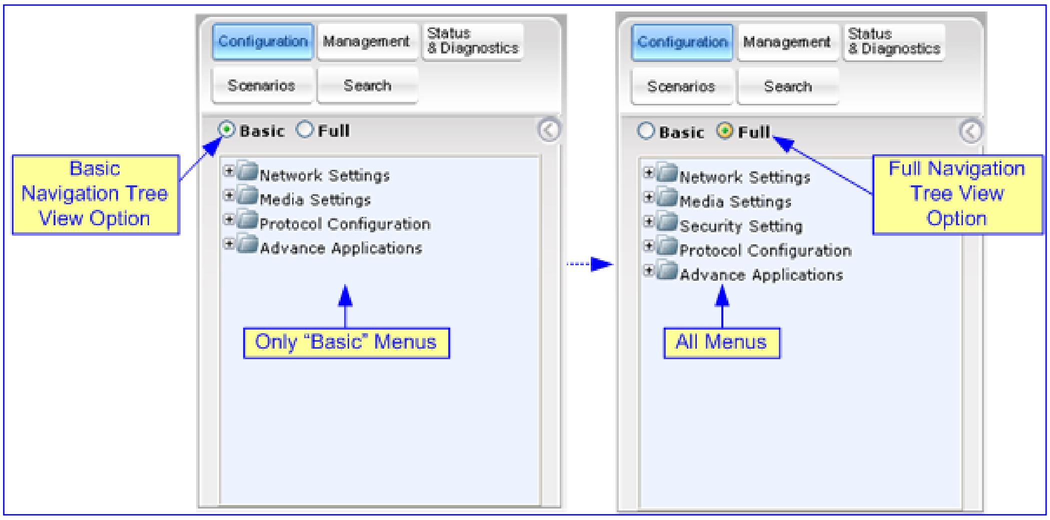
This section provides step-by-step procedures for configuring AudioCodes' E-SBC device. These procedures are based on the setup example described in the following sections.

The steps for configuring the gateway can be summarized as follows:

- **Step 1**: Configure IP Addresses. See Section 4.1 on page 53.
- **Step 2**: Enable the SBC Capabilities. See Section 4.2 on page 59.
- **Step 3**: Configure the Number of Media Channels. See Section 4.3 on page 60.
- **Step 4**: Configure the Proxy Sets. See Section 4.4 on page 61.
- **Step 5**: Configure the IP Groups. See Section 4.5 on page 63.
- **Step 6**: Configure the Voice Coders. See Section 4.6 on page 66.
- **Step 7**: Define Silence Suppression and Comfort Noise. See Section 4.7 on page 68.
- **Step 8**: Configure IP Profile Settings. See Section 4.8 on page 69.
- **Step 9:** Configure Header Manipulation for Call Forwarding. See Section 4.9 on page 73.
- **Step 10:** Configure Unscreened ANI for P-Asserted Identity. See Section 4.10 on page 77.
- **Step 11:** Configuring Unscreened ANI using Diversion Header. See Section 4.11 on page 78.
- **Step 12:** Configure IP-to-IP Routing Setup. See Section 4.12 on page 79.
- **Step 13:** Configure Number Manipulation. See Section 4.13 on page 83.
- **Step 14:** Configuring SIP General Parameters. See Section 4.14 on page 87.
- **Step 15:** Configuring SIP Supplementary Services. See Section 4.15 on page 89.
- **Step 16:** Defining Reasons for Alternative Routing (see Section 4.16 on page 90).
- **Step 17:** Configuring Call Progress Tones file for Regional support (see Section 4.17 on page 92.

The procedures described in this section are performed using the E-SBC device's Web-based management tool (i.e., embedded Web server). Before you begin configuring the device, ensure that the Web interface's Navigation tree is in full menu display mode (i.e., the **Full** option on the Navigation bar is selected), as displayed below:

Figure 4-1: Web Interface Showing Basic/Full Navigation Tree Display



4.1 Step 1: Configuring IP Addresses

This step shows how to configure LAN IP addresses when the internal data-routing capabilities of the E-SBC device are used to connect to the Verizon SIP Trunking service. In this case, configure a separate WAN interface as described below.

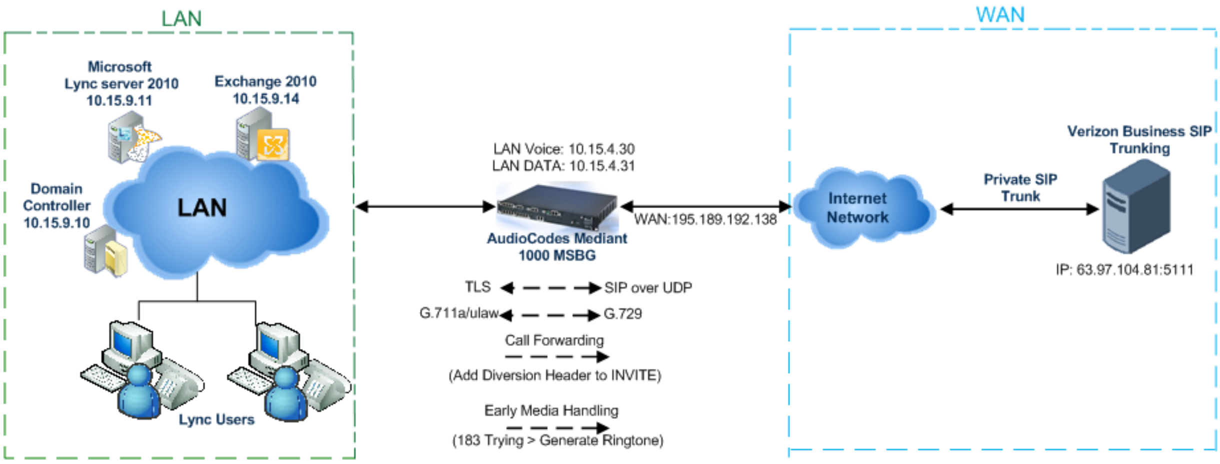
4.1.1 LAN and WAN Interface Separation

This section describes how to configure IP addresses when the internal data-routing capabilities of the E-SBC device are used in order to connect to the Verizon Business SIP Trunk. In this case, you must configure a separate WAN interface as shown in the figure below.

Notes:

- The VoIP and Management interface must be in the same subnet as the data-routing interface as shown in the figure below.
- When operating with both VoIP and data-routing functionalities, it is recommended to define the Default Gateway IP address for the VoIP network interface in the same subnet and with the same VLAN ID as the IP address for the data-routing LAN interface as shown in the figure below.

Figure 4-2: Physical Interface Separation



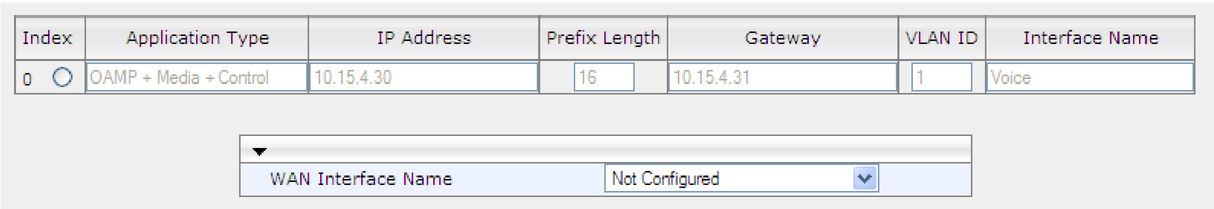
4.1.1.1 Configuring the LAN IP Addresses

This section describes how to assign the LAN IP addresses.

➢ **To assign a LAN VoIP and Management IP address using the Web interface:**

1. Open the ‘IP Settings’ page (**Configuration** tab > **VoIP** menu > **Network** sub-menu > **IP Settings**).
2. Select the 'Index' radio button corresponding to the Application Type "**OAMP + Media + Control** (i.e., VoIP and management interface), and then click **Edit**.
3. Configure the new IP address and prefix length so that it corresponds to your network IP addressing scheme (e.g., 10.15.4.30).
4. Configure additional IP interfaces, if required.

Figure 4-3: Multiple Interface Table Page



5. Click **Apply**, and then **Done** to apply and validate settings. If validation fails, the E-SBC device does not reboot.
6. Save your settings to flash memory and reset the E-SBC device.

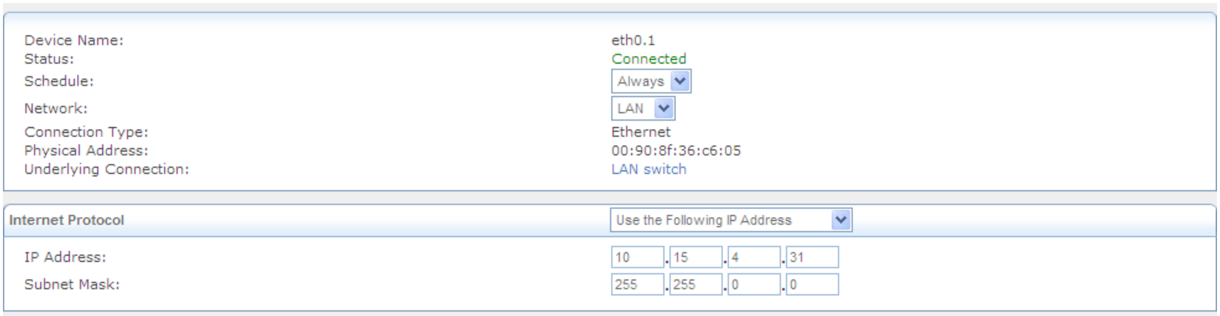
➢ To define the E-SBC device's LAN data-routing IP address:

1. Access the E-SBC device's Web interface with the IP address that you assigned to the VoIP and Management interface.
2. Access the 'Connections' page (**Configuration** tab > **Data** menu > **Data System** > **Connections**).

Figure 4-4: Connections Page

3. Click the **Edit** icon corresponding to the "LAN Switch VLAN 1" connection, and then click the **Settings** tab.
4. In the 'IP Address' and 'Subnet Mask' fields, enter the required IP address (e.g., 10.15.4.31) and subnet respectively, and then click **OK**.

Figure 4-5: Defining LAN Data-Routing IP Address



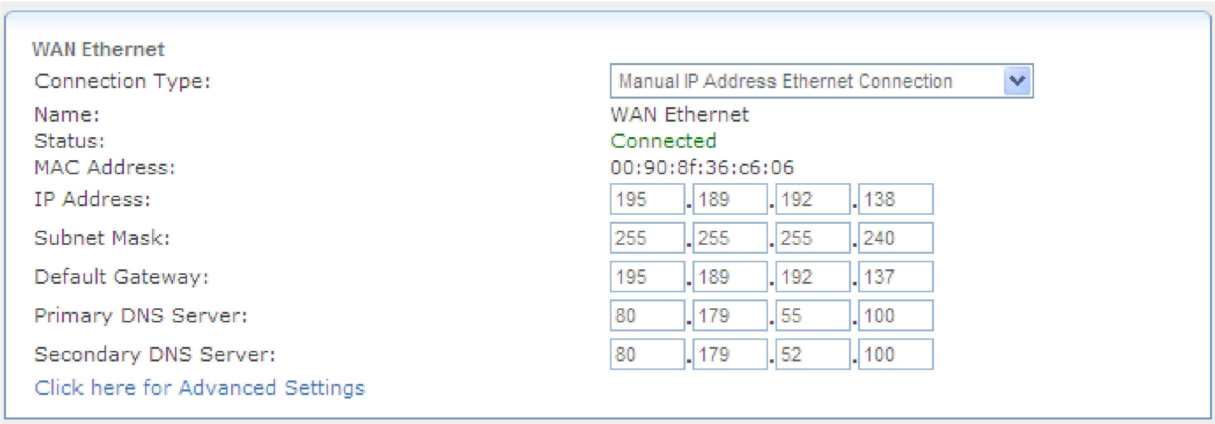
4.1.1.2 Configuring the WAN IP Addresses

This section describes how to assign the WAN IP addresses.

➢ **To assign a WAN IP address:**

1. Cable the E-SBC device to the WAN network (i.e., ADSL or Cable modem), using the WAN port.
2. Access the E-SBC device's Web interface with the Voice and Management IP address.
3. Access the 'Settings' page (**Configuration** tab > **Data** menu > **WAN Access** > **Settings** tab).

Figure 4-6: Configuring the WAN IP Address

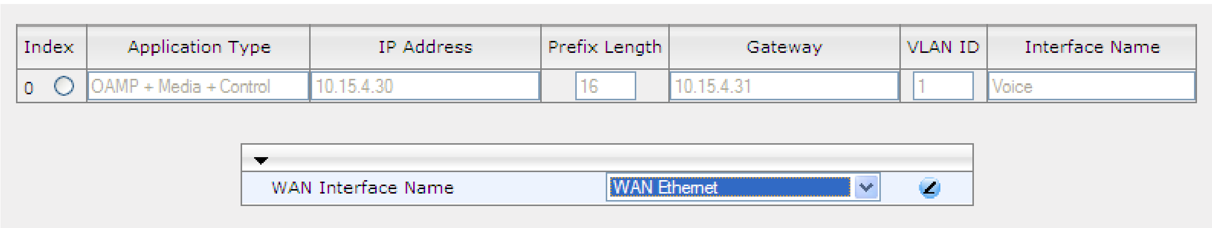


4. From the 'Connection Type' drop-down list, select the required connection type for the WAN, and then configure the IP address (e.g., 195.189.192.138).

➢ **To assign a WAN interface for VoIP traffic:**

1. Select the WAN interface.
2. Open the 'Multiple Interface Table' page (**Configuration** tab > **VoIP** menu > **Network** sub-menu > **IP Settings**).

Figure 4-7: Selecting WAN Interface for VoIP Traffic in Multiple Interface Table Page

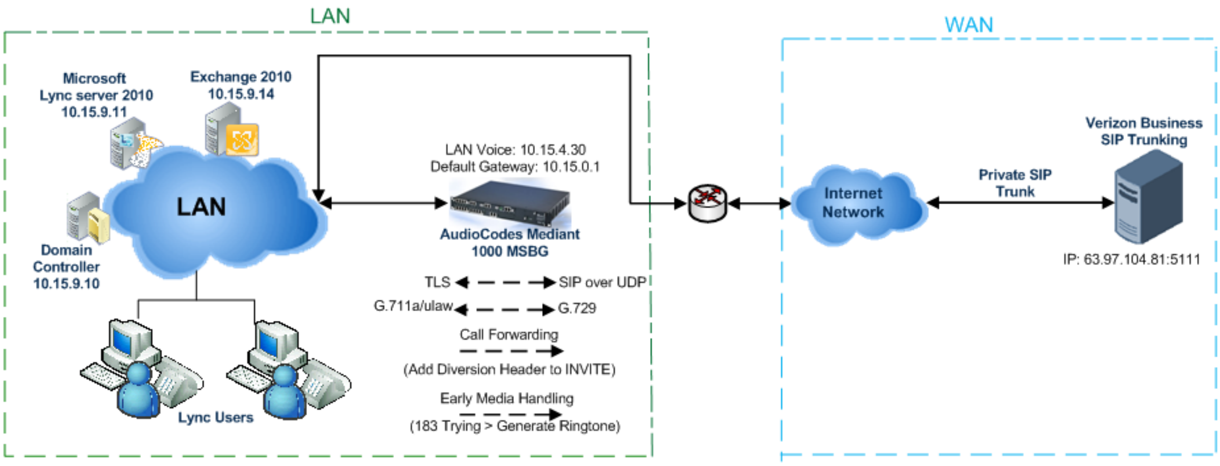


3. From the 'WAN Interface Name' drop-down list, select the WAN interface for VoIP traffic.
4. Click **Done**, and then reset the E-SBC device for your setting to take effect.

4.1.2 Configuring a Single LAN Interface

This section describes how to configure IP addresses when a single LAN interface is used to connect to the Verizon Business SIP Trunk. In this configuration, the internal data-routing capabilities of the E-SBC device are not used as shown in the figure below. As a consequence, you must disable the internal data-routing interface as described in the procedure below.

Figure 4-8: Single LAN Interface

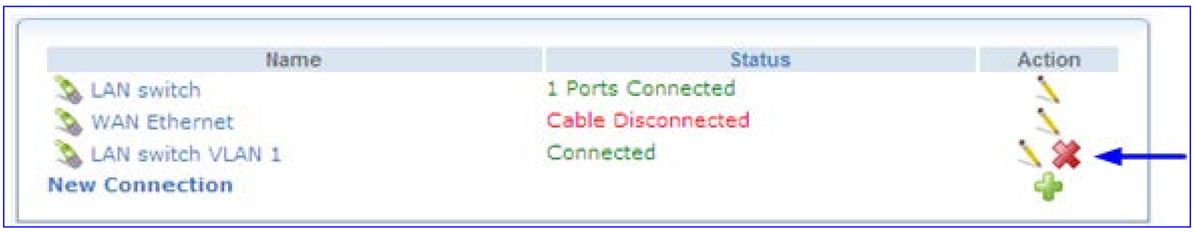


Note: When operating in LAN VoIP-only mode, do not use the E-SBC device's WAN port.

➢ **To operate the E-SBC device as a LAN VoIP gateway only:**

1. Disconnect the network cable from the WAN port and then connect one of the E-SBC device's LAN ports to the network.
2. Disable or remove the data-routing IP network interface:
   - Access the 'Connections' page (**Configuration** tab > **Data** menu > **Data System** > **Connections**).
   - Delete the "LAN Switch VLAN 1" connection by clicking the corresponding **Remove**  button, and then clicking **OK** to confirm deletion.

Figure 4-9: Removing Data-Routing Connection Interface



3. Configure VoIP IP network interfaces in the ‘Multiple Interface’ table (**Configuration** tab > **VoIP** menu > **Network** > **IP Settings**).
4. In the ‘Multiple Interface’ table, define a single IP network interface for application types “OAMP + Media + Control”.

Figure 4-10: Multiple Interface Table

Index	Application Type	IP Address	Prefix Length	Gateway	VLAN ID	Interface Name	Primary DNS Server IP Address	Secondary DNS Server IP Address	Underlying Interface
0	OAMP + Media + Control	10.15.4.30	16	10.15.0.1	0	Mng	0.0.0.0	0.0.0.0	None

5. Click **OK** to save settings.

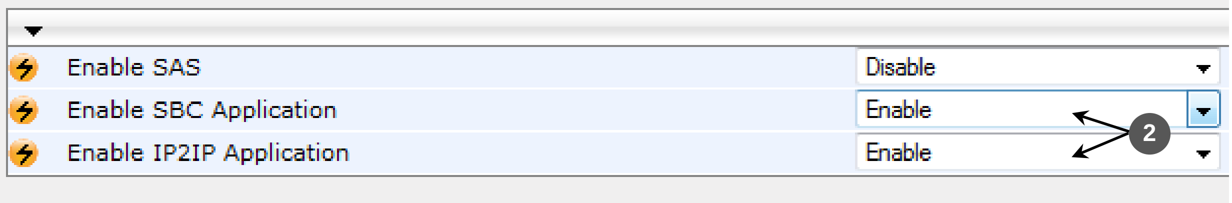
4.2 Step 2: Enabling the SIP SBC and IP2IP Application

This step describes how to enable the device's SIP IP2IP application.

➢ **To enable the SIP SBC and IP2IP application:**

1. Open the 'Application Enabling' page (**Configuration** tab > **VoIP** menu > **Applications Enabling** > **Applications Enabling**).

Figure 4-11: Application Enabling



2. From the 'Enable SBC Application' and the 'Enable IP2IP Application' drop-down lists, select **Enable**.

   Reset with BURN to FLASH is required.

Note: To enable the SBC and IP2IP capabilities on the AudioCodes gateway-SBC device, your device must be loaded with the feature key that includes the SBC feature and also the MSBG device must be running SIP version 6.2 or later.

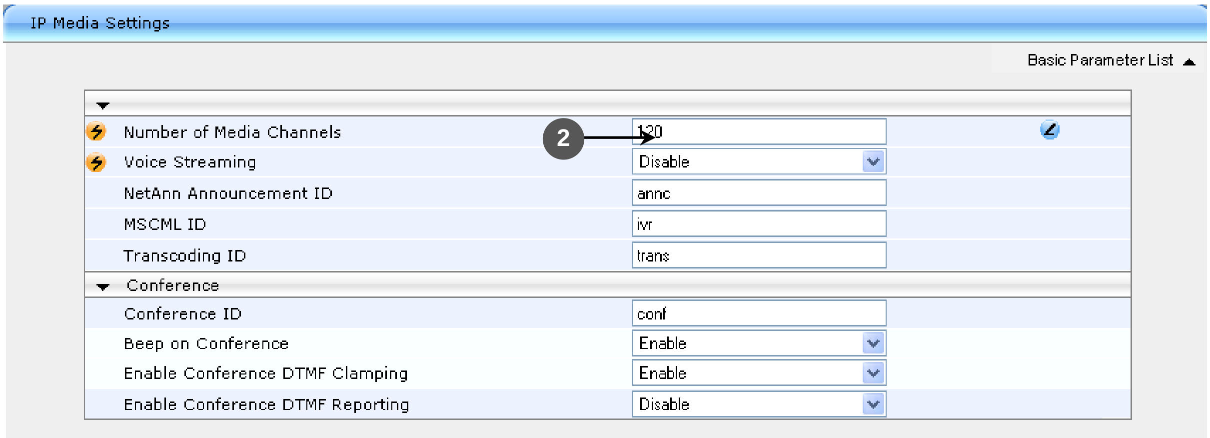
4.3 Step 3: Configuring the Number of Media Channels

In order to perform the coder transcoding, you need to define DSP channels. The number of media channels represents the number of digital signaling processors (DSP) channels that the gateway allocates to IP-to-IP calls (the remaining DSP channels can be used for PSTN calls). Two IP media channels are used per IP-to-IP call. The maximum number of media channels available on the E-SBC device is 120 (i.e., up to 60 IP-to-IP calls for the Mediant 1000B E-SBC, and for the Mediant 3000 E-SBC, 2016 (i.e., up to 1008 IP-to-IP calls).

➢ **To configure the number of media channels:**

1. Open the 'IP Media Settings' page (**Configuration** tab > **VoIP** menu > **IP Media** > **IP Media Settings**).

Figure 4-12: IP Media Channels Settings



2. In the 'Number of Media Channels parameter, do one of the following:
   - For Mediant 1000B E-SBC: Enter **120** to enable up to 60 IP-to-IP calls with transcoding or; click **Apply New Value**.
   - For the Mediant 3000 E-SBC: Enter **2016** to enable up to 1008 IP-to-IP calls with transcoding or; click **Apply New Value**.

4.4 Step 4: Configuring the Proxy Sets

This step describes how to configure the Proxy Sets. The Proxy Sets represent the IP addresses (or FQDN), which are required for communicating with the entities in the network:

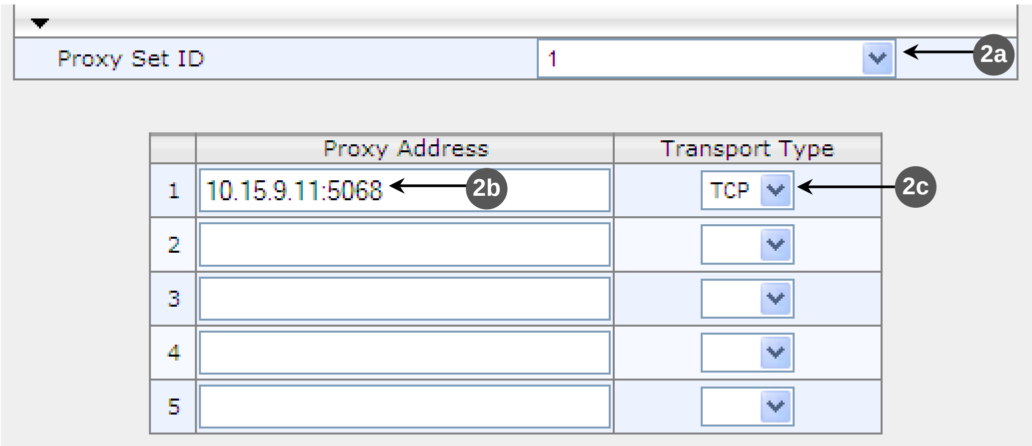
- Proxy Set ID #1 is assigned with the IP address of Lync Mediation server.
- Proxy Set ID #2 is assigned with the IP address of Verizon Business SIP Trunk.
- Proxy Set ID #3 is assigned with the IP address of ATA Media Gateway supporting the Fax machine.

These Proxy Sets are later assigned to IP Groups (see Section 4.5 on page 63).

➢ To configure proxy sets:

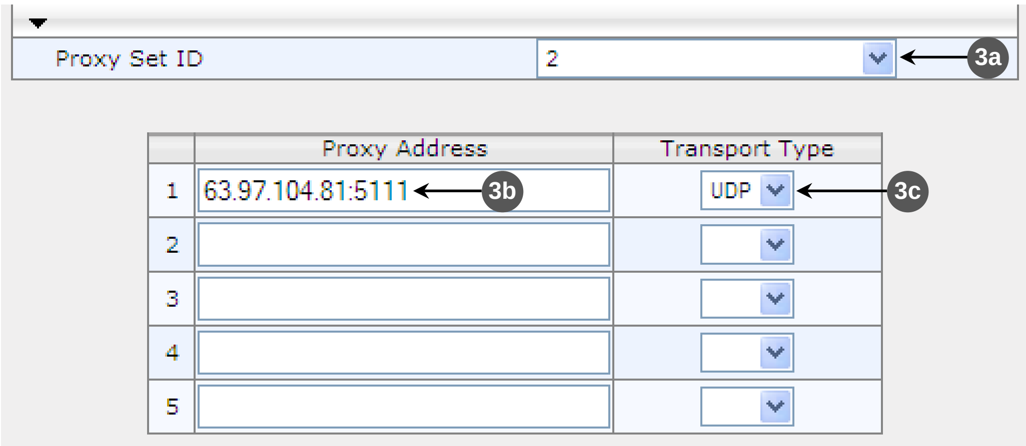
1. Open the 'Proxy Sets Table' page (**Configuration** tab > **VoIP** menu > **Control Network** > **Proxy Sets Table**).
2. Configure the Proxy Set for Lync Mediation Server:
   a. From the 'Proxy Set ID' drop-down list, select **1**.
   b. In the 'Proxy Address' column, enter the IP address or FQDN and the listening port of the Lync Mediation Server.
   c. From the 'Transport Type' drop-down list, corresponding to the IP address entered above, select **TCP** (port 5068) or **TLS** (port 5067) depending on the deployed Mediation Server Transport Type.

Figure 4-13: Proxy Set ID 1 for Lync Mediation server



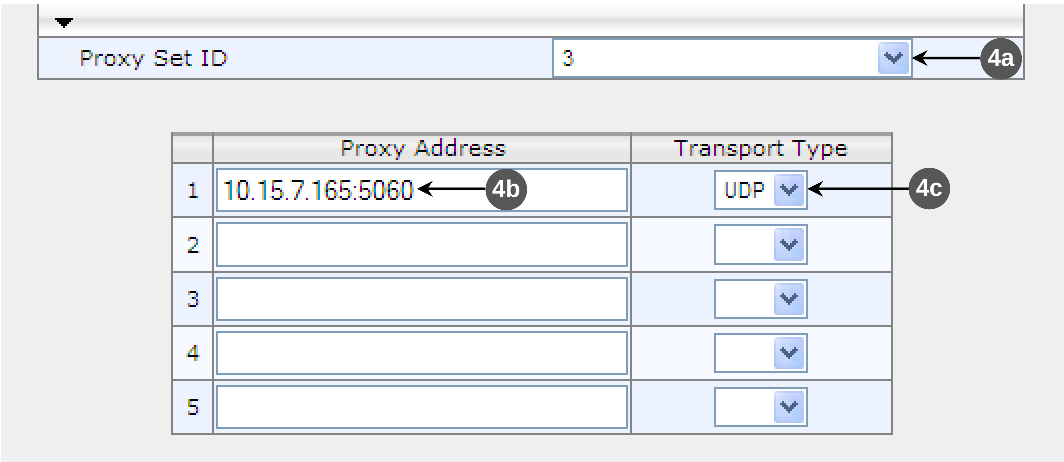
3. Configure the Proxy Set for the Verizon Business SIP Trunk:
   a. From the 'Proxy Set ID' drop-down list, select **2**.
   b. In the 'Proxy Address' column, enter the IP address or the FQDN of the Verizon Business SIP Trunk and the listening port of the Verizon Business SIP Trunk.
   c. From the 'Transport Type' drop-down list corresponding to the IP address entered above, select **UDP**.

Figure 4-14: Proxy Set ID 2 for Verizon Business SIP Trunk



4. Configure the Proxy Set for the Fax supporting Media Gateway:
   a. From the 'Proxy Set ID' drop-down list, select **3**.
   b. In the 'Proxy Address' column, enter the IP address or the FQDN and the listening port of the Fax Supporting Media Gateway.
   c. From the 'Transport Type' drop-down list corresponding to the IP address entered above, select **UDP** Transport Type.

Figure 4-15: Proxy Set ID 3 for Fax Supporting Media Gateway



4.5 Step 5: Configuring the IP Groups

This step describes how to create IP groups. Each IP group represents a SIP entity in the gateway's network. You need to create IP groups for the following entities:

- Verizon Business SIP Trunk
- Lync Server 2010 – Mediation Server
- ATA - FAX Support via Analog Media Gateway.

These IP groups are later used by the IP2IP application for routing calls as well as the SBC application for Header Manipulation.

➢ To configure IP Groups:

1. Open the 'IP Group Table' page (**Configuration** tab > **VoIP** menu > **Control Network**> **IP Group Table**).
2. Define IP Group #**2** for the Verizon ITSP as follows:
   a. IP Group Index **2**
   b. Type: **SERVER**
   c. Description: arbitrary name. (e.g., "Verizon")
   d. Proxy Set ID: **2** (represents the IP address, configured in Section 4.4 on page 61, for communicating with this IP Group).
   e. SIP Group Name: The SIP Request-URI host name used in INVITE messages sent to the IP Group, or the host name in the From header of INVITE messages received from the IP Group. Enter the WAN IP address.
   f. IP Profile ID: **2**: Different IP profile is used for the Verizon Business SIP Trunk and the Mediation Server. See Section 4.8 on page 69.

Figure 4-16: IP Group 2 Table

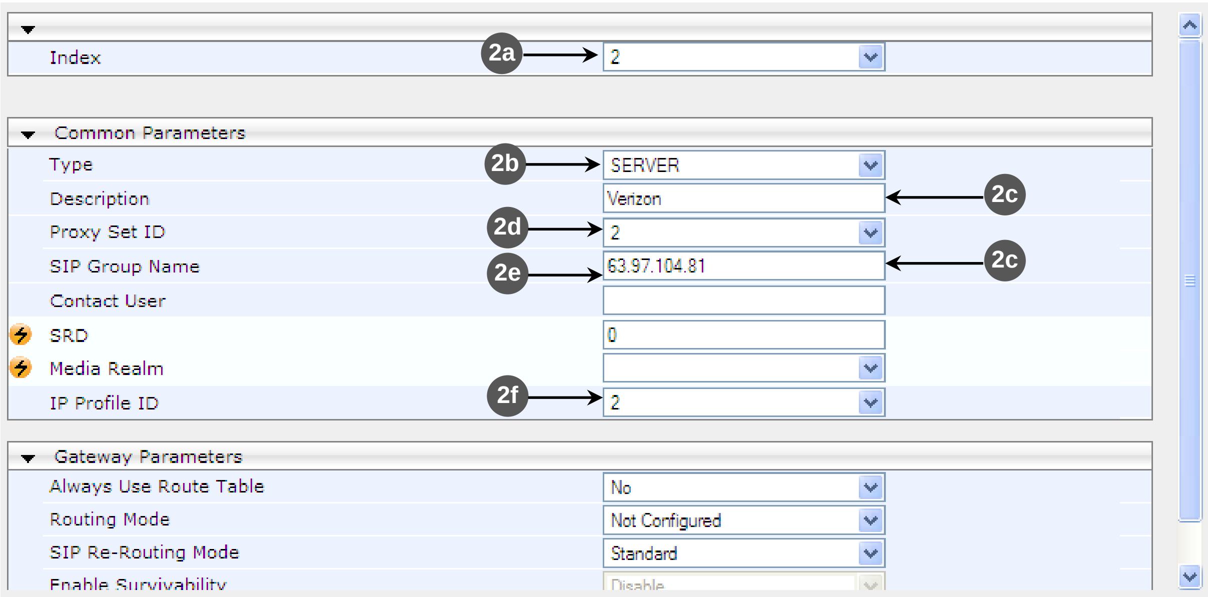
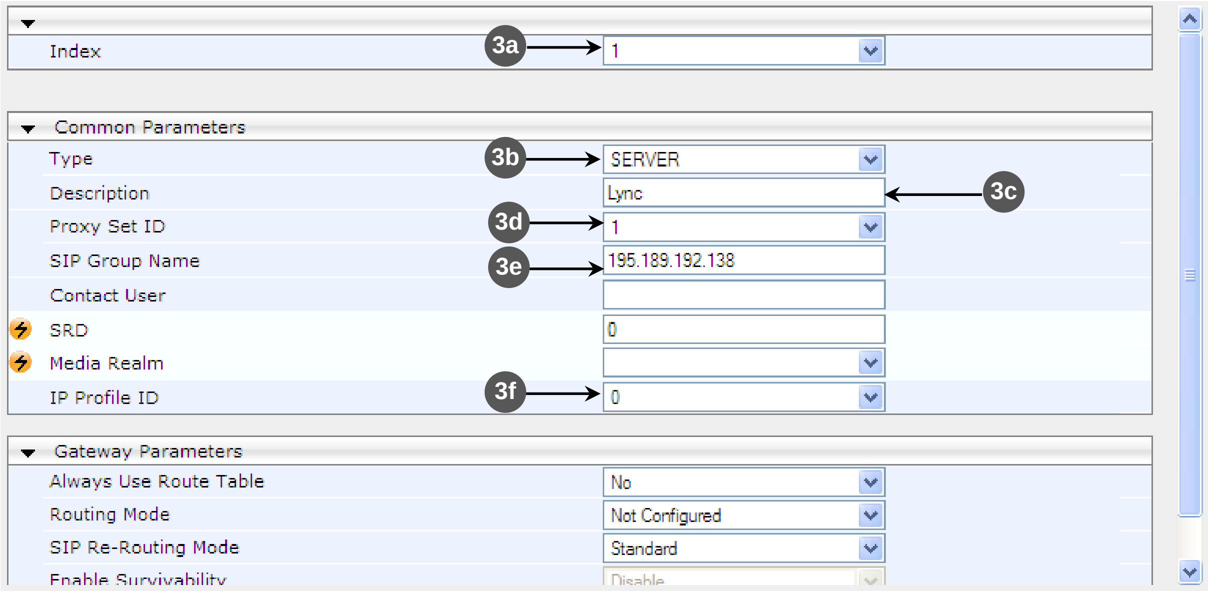
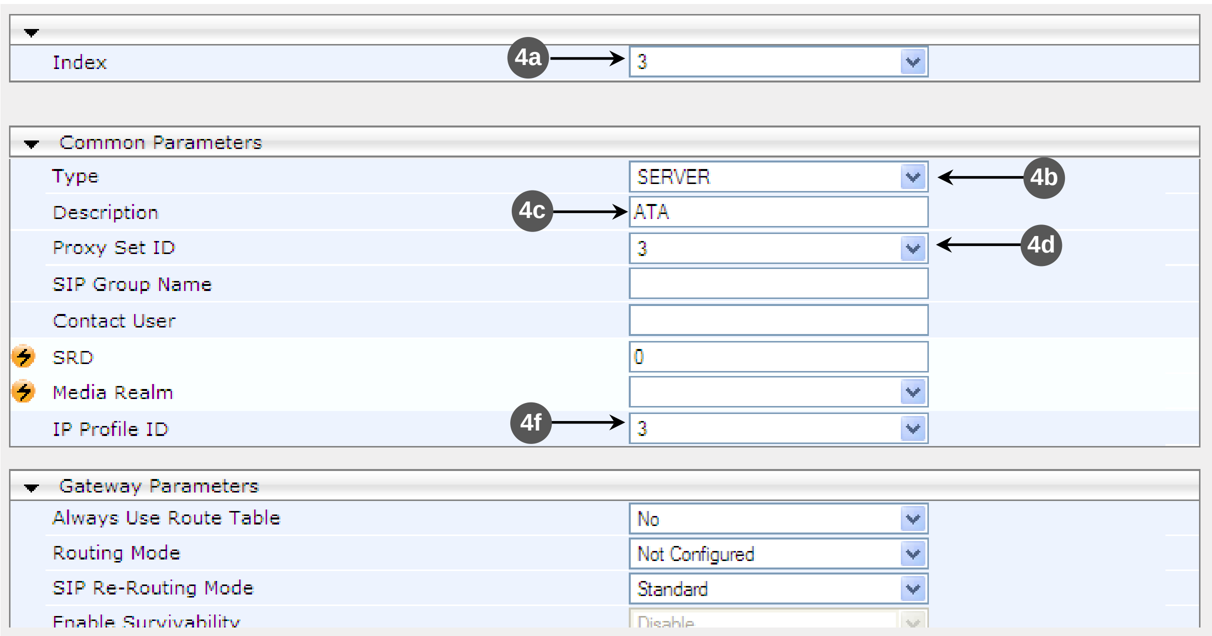


Figure 4-17: IP Group 1 Table Page



3. Define IP Group #**1** for Mediation Server as follows:
   a. Select IP Group Index **1**
   b. Type: **SERVER**
   c. Description: <Free Description> (e.g., "Lync Mediation Server")
   d. Proxy Set ID: **1**
   e. SIP Group Name: The SIP Request-URI host name used in INVITE messages sent to the IP Group, or the host name in the From header of INVITE messages received from the IP Group. Enter the Gateway Name.
   f. IP Profile ID: **0** (see Section 4.8 on page 69).

Figure 4-18: IP Group 3 Table Page



4. Define IP Group #**3** for supporting FAX Media Gateway as follows:
   a. Select IP Group Index **3**.
   b. Type: **SERVER**
   c. Description: <Free Description> (e.g., “ATA”)
   d. Proxy Set ID: **3**
   e. SIP Group Name: The SIP Request-URI host name used in INVITE messages sent to the IP Group, or the host name in the From header of INVITE messages received from the IP Group. Enter the Gateway Name or leave blank for default.
   f. IP Profile ID: **3** (see Section 4.8 on page 69).

4.6 Step 6: Configuring the Voice Coders

This step describes how to configure the Voice Coders. Since the Mediation Server supports both the G.711A-law and G.711U-law voice coders, while the Verizon SIP trunk supports the G.729 and G.711U-law coders, you can configure a single coder table reference for both services by utilizing the G711U-law coder as a single voice coder (see Figure 4-20) or you can create a more dynamic servicing interworking based on commonality of supported vocoders via the default Coders and Coders Group tables (see Figure 4-19).

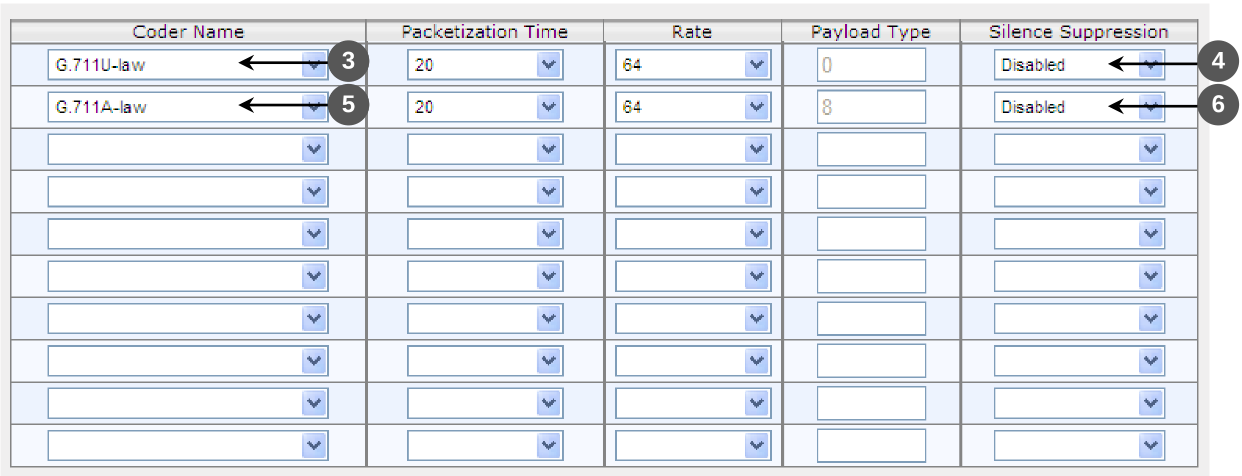
The Coder table is associated with the following IP Profiles:

- IP Profile index 1 and 3 (see Section 4.8 on page 69) references ‘Coders Table’ (see Figure 4-19) which is associated with the IP Groups 1 and 3 (see Section 4.5 on page 63).
- IP Profile index 2 (see Section 4.8 on page 69) references ‘Coder Group 1’ (see Figure 4-20), which is associated with IP Group 2 (see Section 4.5 on page 63).

➢ To configure Coder Table for Mediation Server and in-band FAX usage:

1. Open the 'Coders Table' page (**Configuration** tab > **VoIP** menu > **Coders And Profiles** > **Coders**).

Figure 4-19: Coders Table – Lync Mediation Server



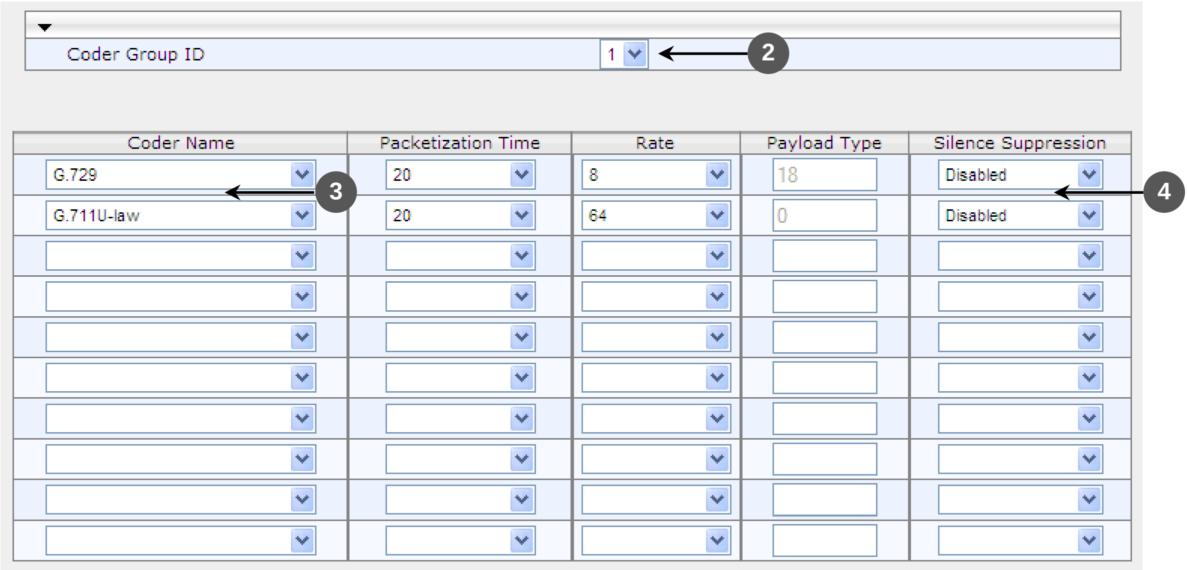
Coder Name	Packetization Time	Rate	Payload Type	Silence Suppression
G.711U-law	20	64	0	Disabled
G.711A-law	20	64	8	Disabled

2. From the 'Coders Table', select via the drop-down list, coder and attributes in the manner of preference with the first entry as being the preferred entry, and all others following in descending order.
3. Select the **G.711U-law** coder, as shown in the figure above.
4. From ‘Silence Suppression’ drop-down list, select **Enable** or **Disabled** as shown in the figure above.
5. Select the **G.711 A-law** coder, as shown in the figure above.
6. From ‘Silence Suppression’ drop-down list, select **Enable** or **Disabled** as shown in the figure above.

➢ To configure Coder Table for Verizon SIP Trunk usage:

1. Open the 'Coders Table' page (**Configuration** tab > **VoIP** menu > **Coders And Profiles** > **Coders Group Settings**).

Figure 4-20: Coder Group Table 1 – Verizon SIP Trunk



2. From the ‘Coder Group Setting’ table, select via drop-down list, the index **1**. This index, **1**, is referenced by the datafill of parameter 'Coders Group index' of IP Profile index **2**. This allows a user to list the allowed vocoders in a supported group to be referenced and utilized. This points to table 'Coder Group' for IP Profile index 2, where Coder Group 1 is explicitly referenced. As shown above, Coder Group 1 is declared to support G.729 and G.711U-law.
3. Select the **G.729** and **G.711U-law** coder, in the specific order as shown in the figure above. This will be the preference as advertised in the SDP for proper ‘Offer/Answer’ interworking.
4. From ‘Silence Suppression’ drop-down list, select **Enabled** or **Disabled** as shown in the figure above.

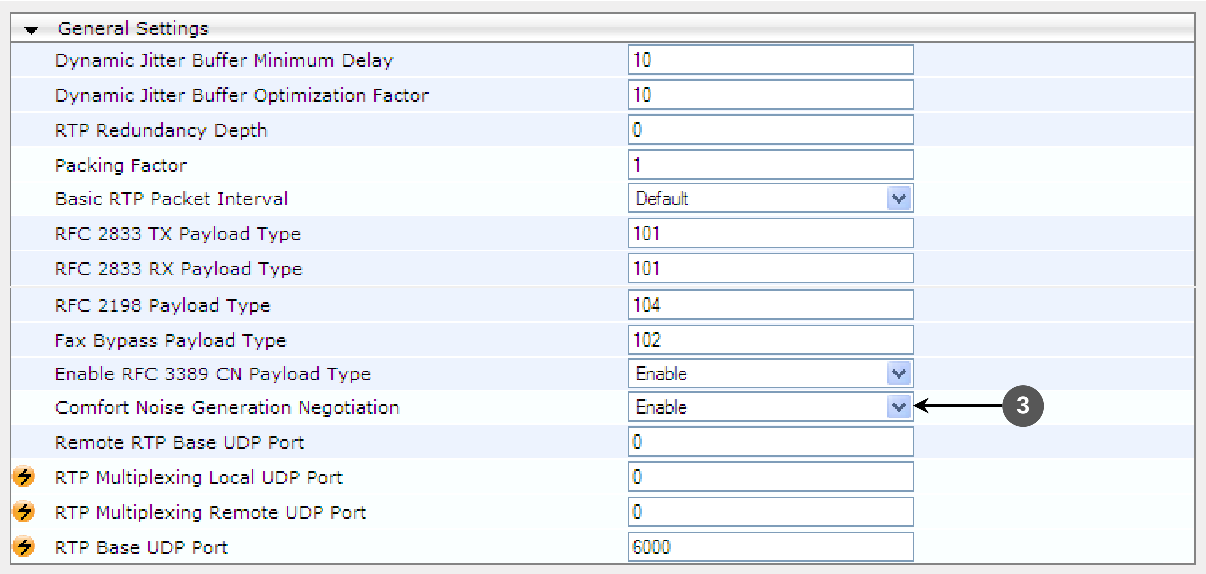
4.7 Step 7: Defining Silence Suppression and Comfort Noise

Overall voice quality has been significantly improved for the Microsoft Lync 2010 environment. These improvements include suppression of typing noise during calls and improved generation of “comfort noise,” which reduces hissing and smoothes over the discontinuous flow of audio packets. You may need to change the MSBG Silence Suppression and Comfort Noise parameters to achieve this goal. Note that the Echo canceller is enabled by default.

➢ To configure silence suppression parameters:

1. Silence Suppression is configured per coder type. (See Section 4.6 on page 66 above to enable Silence Suppression per coder.)
2. Open the 'RTP/RTCP Settings' page (**Configuration** tab > **Media** menu > **RTP / RTCP Settings**).

Figure 4-21: RTP/RTCP Settings Page



3. From the 'Comfort Noise Generation Negotiation' drop-down list, select **Enable**. This enables negotiation and usage of Comfort Noise (CN).
4. Click **Submit**.

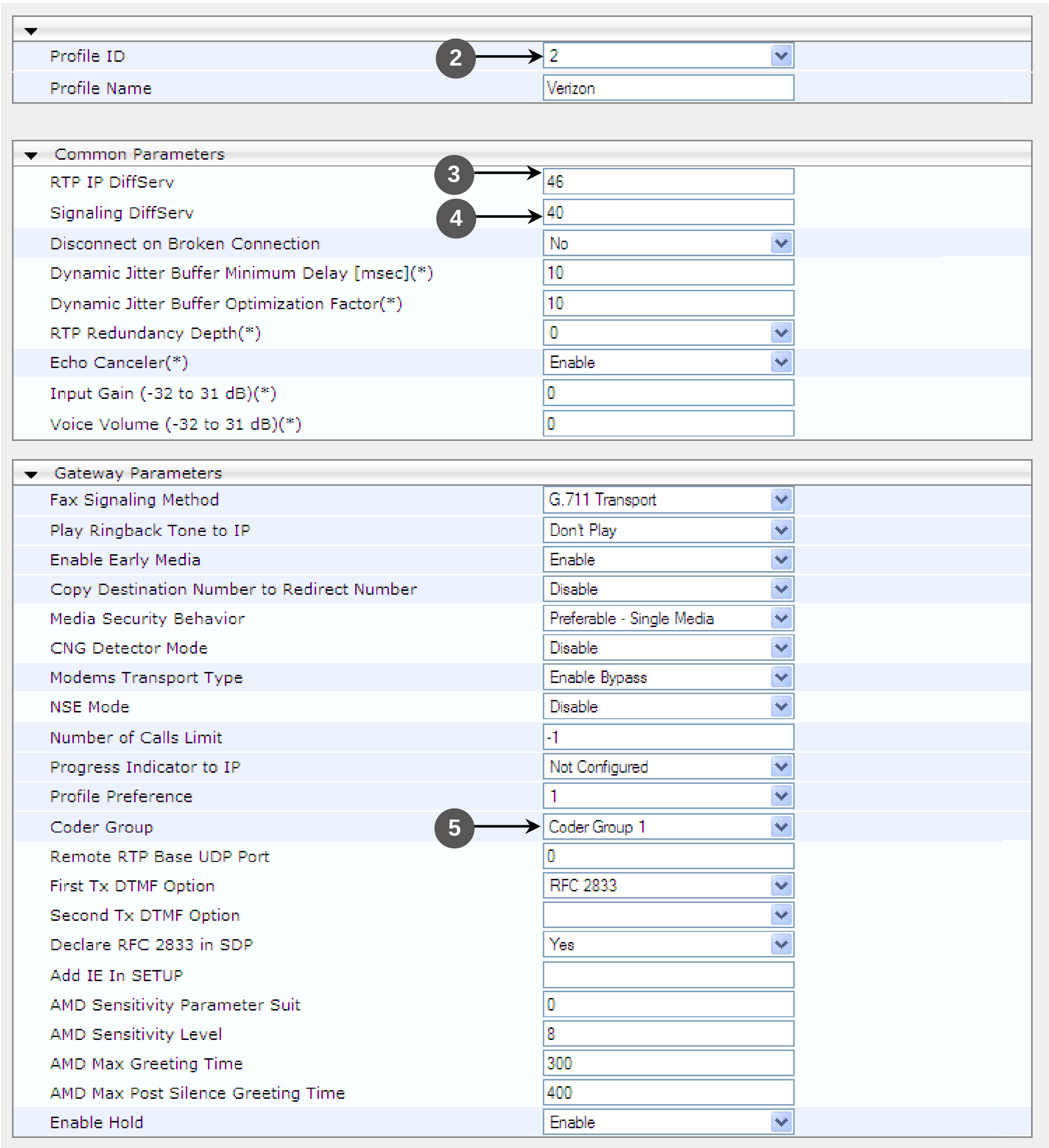
4.8 Step 8: Configuring IP Profile Settings

This section describes how to configure the IP Profile Settings.

➢ **To configure IP Profile for Verizon SIP Trunk Server :**

1. Open the 'IP Profile Settings' page (**Configuration** tab > **VoIP** menu > **Coders and Profiles** > **IP Profile Settings**).

Figure 4-22: IP Profile Page-Verizon SIP Trunk Server



2. From the 'Profile ID' drop-down list, select **2**.
3. In the 'RTP IP DiffServ' field, enter the required value for the DSCP (see example in figure below).
4. In the 'Signaling DiffServ' field, enter the required value for the DSCP (see example in figure below).

Figure 4-23:DiffServ Service Classes

Service Class Name	DSCP Name	DSCP Value	Example Apps	AudioCodes Value
Telephony	EF	101110	RTP / IP Telephony bearer	46
Signaling	CS5	101000	RTP / SIP / IP Telephony signaling	40
Multimedia Streaming	AF31 AF32 AF33	011010 011100 011110	SIP / Streaming video and audio on demand	26 28 30
Broadcast Video	CS3	011000	SIP / Broadcast TV & live events	24
Standard	DF (CS0)	000000	Undifferentiated applications	0
Low-priority data	CS1	001000	Any flow that has no BW assurance	8

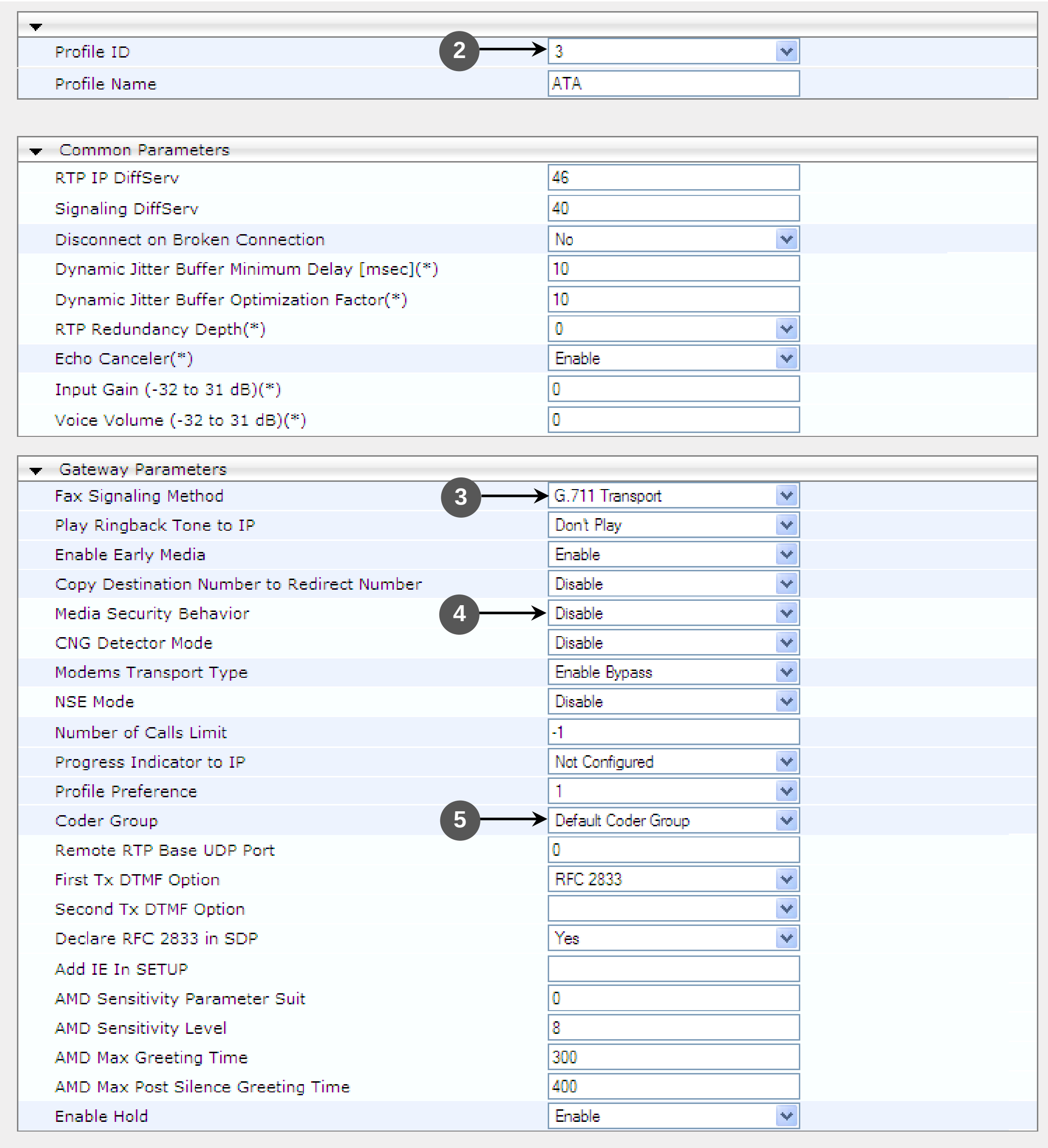
Source: RFC 4594. *Configuration Guidelines for DiffServ Service Classes.*

5. From the 'Coder Group' drop-down list, select **Coder Group 1**.

➢ To configure IP Profile for FAX supporting ATA:

1. Open the 'IP Profile Settings' page (**Configuration** tab > **VoIP** menu > **Coders and Profiles** > **IP Profile Settings**).

Figure 4-24: IP Profile Page-FAX Supporting ATA



2. From the 'Profile ID' drop-down list, select **3**.
3. From the 'Fax Signaling Method' drop-down list, select **G.711 Transport**.

4. From the ‘Media Security Behavior’ drop-down list, select one of the following options:
   - **Mandatory** if Mediation Server is configured to SRTP Required
   - **Preferable-Single media** if Mediation Server is configured to SRTP Optional.
   - **Disable** if the Mediation Server is configured to SRTP disabled.
5. From the ‘Coder Group’ drop-down list, select **Default Coder Group**. The call associated with this profile utilizes the base coders that were assigned in the 'Coders' table as opposed to coders that were assigned via an index into the Coder Group Table. The usage of G.711 U-law coder will be based on the previous datafill.

4.9 Step 9: Configuring Message Manipulation for Call Forwarding

This step describes how to configure Message Manipulation for Call Forwarding.

➢ **To configure SBC Header Manipulation Rules:**

1. Open the Manipulation Table page (**Configuration** tab > **VoIP** menu > **SBC** > **Manipulations SBC** > **Message**).

Figure 4-25: Message Manipulations Table

Index	Manipulation Set ID	Message Type	Condition	Action Subject	Action Type	Action Value	Row Role
0	0	Any.Request		Header.Diversion.URL.host	Modify	'195.189.192.138'	Use Current Condition
1	1	Any.Request	header.referred-by exists	Header.Diversion	Add	'<'+header.referred-by.URL+'>'	Use Current Condition

2. **Index #0** defines manipulation of calls to modify a header. For any request received modify the Header.Diversion.URL.host with ‘195.189.192.138’.
3. **Index #1** defines manipulation of calls to add a header. For any Invite request received and IF a Referred-by header exists, then Add the Diversion Header with what was received within the referred-by header, '<'+header.referred-by.URL+'>'.

Each of these rules bind to a specific Manipulation Set ID as represented within this table. Indexes 0 binds to Manipulation Set ID 0 and Index 1 is bound to Manipulation Set ID 1.

Note the following:

- **Index 0** was created for a specific call scenario where a PSTN user originated a call on the Verizon SIP Trunk to an Lync 2010 DID, which has its DID forwarded to the PSTN as well as have its presentation Restricted in the Source number manipulation tables. This was set to change the Diversion URL host from 'anonymous.invalid' to ‘195.189.192.138’, which is the WAN address of the device.
- **Index 1** is utilized to dynamically capture the original dialed number for Lync DIDs and present them back to Verizon when a call is forwarded back to the PSTN when creating a Diversion Header. This functionality can only be used if Lync Server 2010 is set up to send Referred-by headers. This specific index creates a new Diversion header for the call using what was presented in the Referred-by header, thus allowing the Verizon service to receive the original dialed DID extension that forwarded the call back to the PSTN SIP trunk.

These newly created message manipulation sets can now be assigned for use. There are several methods to perform this function. One method is to assign the functionality to the entire application via the *ini* file or via the AdminPage.

This functionality is configured using the parameters GWINBOUNDMANIPULATIONSET = [1] and GWOUTBOUNDMANIPULATIONSET = [-1].

When using these parameter settings, the respective Manipulation Set ID is assigned for all inbound and outbound manipulations. When using the IP to IP application, such as in this example, the manipulation for inbound messages **must be** performed at the gateway level, so therefore set **GWINBOUNDMANIPULATIONSET = 1**. The outbound message manipulation can be performed either at the device level or more dynamically, at the IP Group level.

This section describes the following procedures:

- Assign inbound manipulation to the entire device via the *ini* file or via the AdminPage. See Section 4.9.1 below.
- Assign outbound manipulation to a specific IP group index. See Section 4.9.2 below.

4.9.1 Assigning Inbound Message Manipulation Set to Entire Gateway

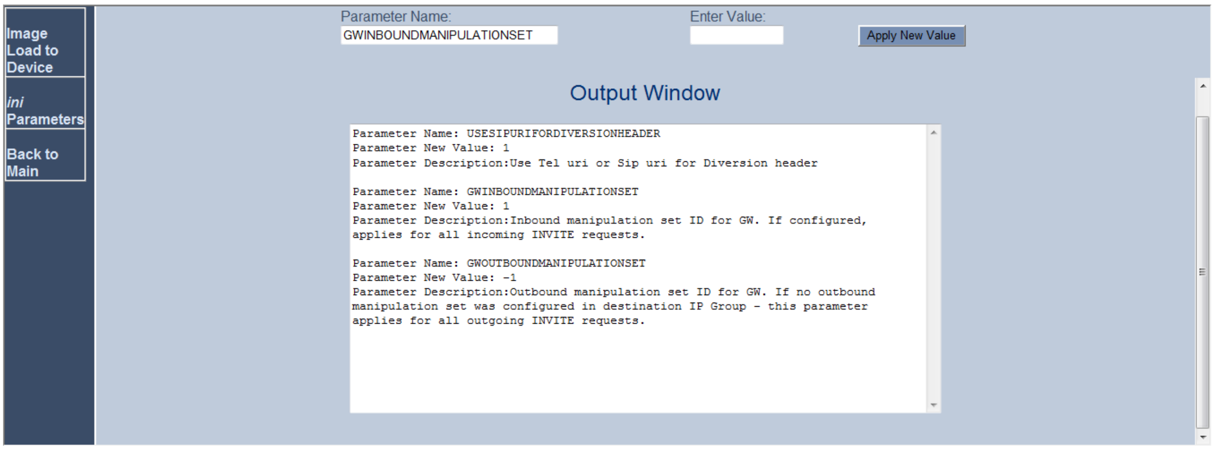
This step describes how to assign a message manipulation set to the entire device via the *INI* file.

This assignment is configured using the parameters GWINBOUNDMANIPULATIONSET = [1] and GWOUTBOUNDMANIPULATIONSET = [-1]. These parameter settings are applied for all inbound and outbound manipulations respectively. For IP2IP applications, GWINBOUNDMANIPULATIONSET must be used for header manipulation requirements on incoming messages.

➢ **To assign inbound message manipulation set to gateway:**

1. Open the Admin page, by appending the case-sensitive suffix 'AdminPage' to the gateway-SBC device's IP address in your Web browser's URL field (e.g., http://10.15.9.118/AdminPage).
2. On the left pane, click ***ini* Parameters**.

Figure 4-30: Output Window



2. In the 'Parameter Name' field, enter **GWINBOUNDMANIPULATIONSET** or **GWOUTBOUNDMANIPULATIONSET**.
3. In the 'Enter Value' field, enter the respective Manipulation Set ID you wish for the specific direction. e.g., **1**.
4. Click **Apply New Value** (GWINBOUNDMANIPULATIONSET should be set to **1** for proper interworking).

4.9.2 Assigning Outbound Message Manipulation Set to an IP Group

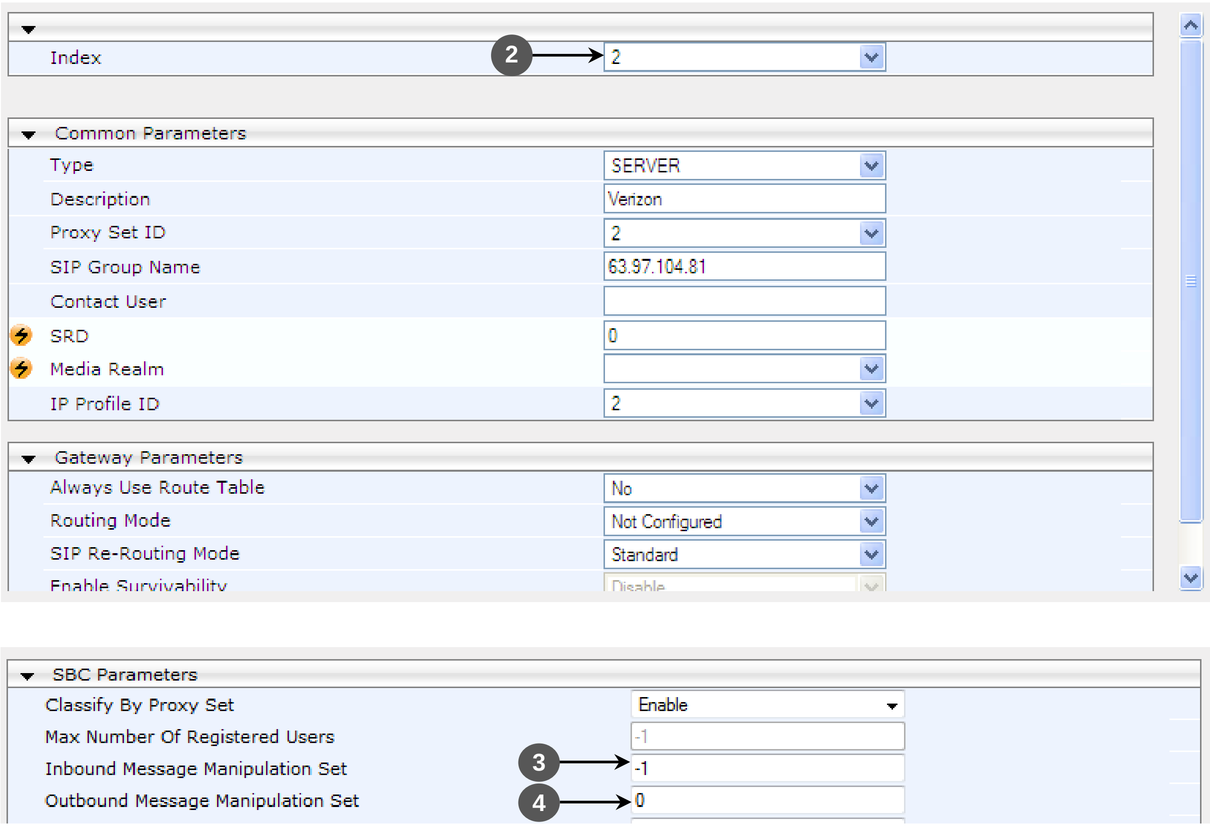
This step describes how to assign outbound message manipulation rules to IP groups to support outbound message manipulation. Each IP group represents a SIP entity in the device's network. You may assign manipulation rules to IP groups for the Verizon SIP Trunk entity.

When completed, the IP group is used by the IP2IP, as well as the SBC application for routing calls with advanced outbound header manipulation support.

➢ **To assign outbound message manipulation set to IP Group Table 2 (Verizon SIP Trunk):**

1. Open the IP Group Table page (**Configuration** tab > **VoIP** menu > **Control Network**> **IP Group Table**).

Figure 4-26: IP Group Table 2 (Verizon SIP Trunk)



2. Set Index to **2**.
3. Set 'Inbound Message Manipulation Set' to **-1**. This means that the device does not perform message manipulation at this stage (for IP2IP applications, inbound message manipulation can only be performed at the gateway level).
4. Set 'Outbound Message Manipulation Set' to **0**.

4.10 Step 10: Configuring Unscreened ANI using P-Asserted Identity

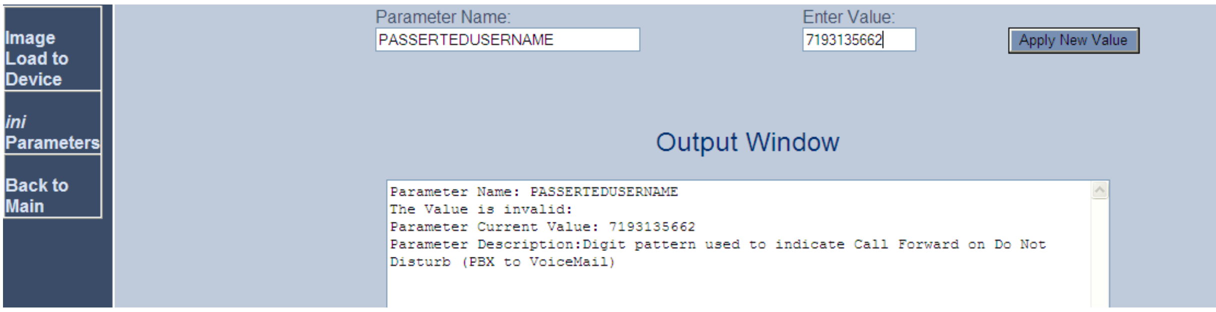
This step describes how to support proper termination of an outbound call with the correct Caller ID when the Screened Telephone Number contained in the P-Asserted Identity Header is different from the CLI contained in the From Header.

This functionality is required to support TC16 of VIT.2010.03154.TPL.001 'Test Suite for Retail VoIP Interoperability IP Trunking' for the Verizon North American retail market.

➢ To configure P-Asserted Identity for Unscreened ANI:

1. Open the 'Admin' page by appending the case-sensitive suffix 'AdminPage' to the E-SBC device's IP address in your Web browser's URL field (e.g., http://10.15.4.15/AdminPage).
2. On the left pane, click ***ini*** Parameters.

Figure 4-27: Output Window



3. In the 'Parameter Name' field, enter the parameter **PASSERTEDUSERNAME**. In the 'Enter Value' field, enter "7193135662" or any valid number which will be utilized in the P-Asserted Identity header.
4. Click **Apply New Value**.

4.11 Step 11: Configuring Unscreened ANI using Diversion Header

This step describes how to support proper termination of an outbound call with the correct Caller ID when the Screened Telephone Number contained in the Diversion Header is different from the Calling Line Identification (CLI) contained in the From Header.

This functionality is required to support TC15 of VIT.2010.03154.TPL.001 'Test Suite for Retail VoIP Interoperability IP Trunking' for the Verizon North American retail market.

➢ To configure diversion header for unscreened ANI:

This step describes how to configure SBC Message Manipulation for creating a Diversion Header to support Unscreened ANI.

➢ To configure SBC header manipulation rules:

1. Open the Manipulation Table page (**Configuration** tab > **VoIP** menu > **SBC** > **Manipulations SBC** > **Message**).

Figure 4-27: Message Manipulations Table

Index	Manipulation Set ID	Message Type	Condition	Action Subject	Action Type	Action Value	Row Role
0	0	Invite.Request		Header.Diversion	Add	'<sip:9727289417@10.15.9.11	Use Current Condition

2. **Index #0** defines manipulation of calls to add a header. For any Invite request received, add a Header.Diversion with the following: <sip:9727289417@10.15.9.118;user=phone>'
3. Click **Apply New Value**.
4. Attaching or assigning to bind the interworking for message manipulation was discussed previously in sections 4.9, 4.9.1 and 4.9.2. Do this either at the gateway level for incoming messages or at the respective Outgoing IP Group based on routing and selection of the terminating route.

4.12 Step 12: Configuring IP-to-IP Routing Setup

The MSBG's IP-to-IP call routing capabilities is performed in two stages:

- **Inbound IP Routing:** Recognizes the received call as an IP-to-IP call, based on the call's source IP address. This stage is configured in the 'Inbound IP Routing Table'
- **Outbound IP Routing:** Once recognized as an IP-to-IP call in the first stage (see above), the call is routed to the appropriate destination (i.e., IP address). This stage is configured in the 'Outbound IP Routing Table'.

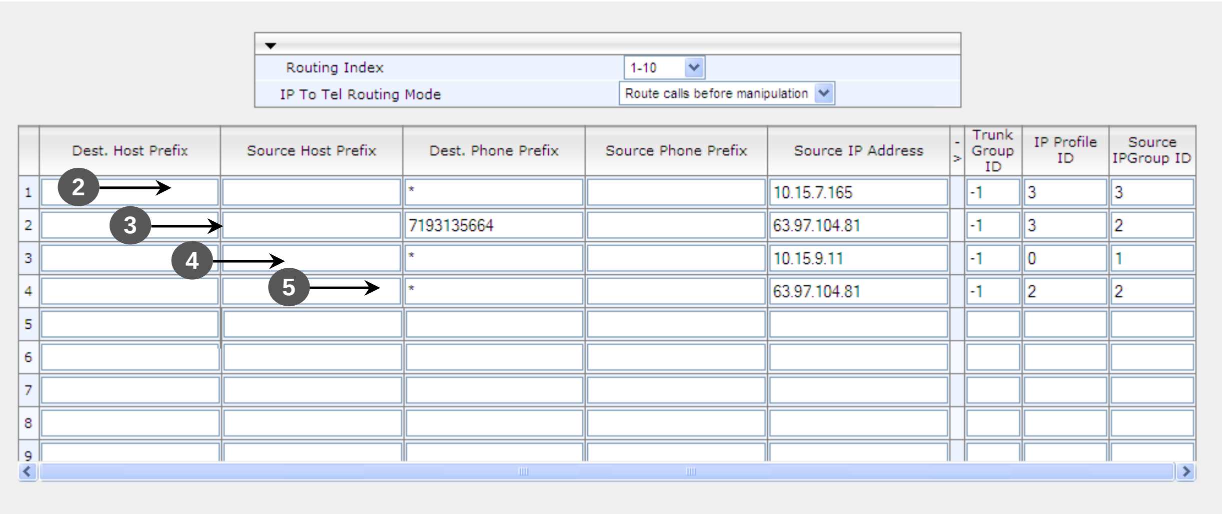
4.12.1 Configure Inbound IP Routing

This step defines how to configure the E-SBC device for routing inbound (i.e., received) IP-to-IP calls to the Microsoft Lync network.

➢ To configure in bound IP routing:

1. Open the 'Inbound IP Routing Table' page (**Configuration** tab > **VoIP** menu > **GW and IP to IP** > **Routing** sub-menu > **IP to Trunk Group Routing**).

Figure 4-28: Inbound IP Routing Table Page



Routing Index: 1-10
IP To Tel Routing Mode: Route calls before manipulation

	Dest. Host Prefix	Source Host Prefix	Dest. Phone Prefix	Source Phone Prefix	Source IP Address	->	Trunk Group ID	IP Profile ID	Source IPGroup ID
1			*		10.15.7.165		-1	3	3
2			7193135664		63.97.104.81		-1	3	2
3			*		10.15.9.11		-1	0	1
4			*		63.97.104.81		-1	2	2
5									
6									
7									
8									
9									

2. **Index #1** configuration identifies all IP calls received from FAX supporting ATA as IP-to-IP calls and assigns them to the IP Group ID configured for the Verizon Business SIP Trunk:
   - *Dest Phone Prefix*: Enter the asterisk (*) symbol to indicate all destinations.
   - *Source IP Address*: Enter the IP address of ATA device.
   - *Trunk Group ID*: Enter **-1** to indicate that these calls are IP-to-IP calls.
   - *IP Profile ID*: Enter **3** to indicate the IP Profile for interworking G.711 in-band fax support to Verizon Business SIP Trunk.
   - *Source IP Group ID*: Enter **3** to assign these calls to the IP Group pertaining to the ATA supporting fax services.

3. **Index #2** configuration identifies all IP calls received from the Verizon Business SIP Trunk in the event of determining a route for a dedicated FAX line call Scenario as IP-to-IP calls. It then assigns these IP calls to the IP Group ID configured for the FAX supporting ATA and also assigns a specific IP Profile to support the FAX call.:
   - *Source Host Prefix*: Enter the asterisk (*) symbol to indicate all destinations.
   - *Dest Phone Prefix*: Enter the asterisk (*) symbol to indicate all destinations.
   - *Source Phone Prefix*: Enter the specific DID assigned to support FAX services.
   - *Source IP Address*: Enter the IP address of Verizon SIP Trunk Server.
   - *Trunk Group ID*: Enter **-1** to indicate that these calls are IP-to-IP calls.
   - *IP Profile ID*: Enter **3** to indicate that the IP Profile supports the fax call for G.711 in-band service.
   - *Source IP Group ID*: Enter **2** to assign these originated calls to the IP Group pertaining to the Verizon SIP Trunk server.
4. **Index #3** configuration identifies all IP calls received from the Mediation Server as IP-to-IP calls and assigns them to the IP Group ID configured for the Lync Mediation Server:
   - *Dest Phone Prefix*: Enter the asterisk (*) symbol to indicate all destinations.
   - *Source Phone Prefix*: Enter asterisk (*) symbol to indicate all sources.
   - *Source IP Address*: Enter the IP address of the Mediation server.
   - *Trunk Group ID*: Enter **-1** to indicate that these calls are IP-to-IP calls.
   - *IP Profile ID*: Enter **0** indicate the IP Profile for Mediation server.
   - *Source IP Group ID*: Enter **1** to assign these calls to the IP Group pertaining to the Lync Mediation server.
5. **Index #4** configuration identifies all IP calls received from Verizon Business SIP Trunk as IP-to-IP calls and assigns them to the IP Group ID configured for the Verizon Business SIP Trunk:
   - *Dest Phone Prefix*: Enter the asterisk (*) symbol to indicate all destinations.
   - *Source IP Address*: Enter the IP address of Verizon Business SIP Trunk.
   - *Trunk Group ID*: Enter "-1" to indicate that these calls are IP-to-IP calls.
   - *IP Profile ID*: Enter ‘2’ indicate the IP Profile for Verizon Business SIP Trunk.
   - *Source IP Group ID*: Enter "2" to assign these calls to the IP Group pertaining to Verizon Business SIP Trunk.

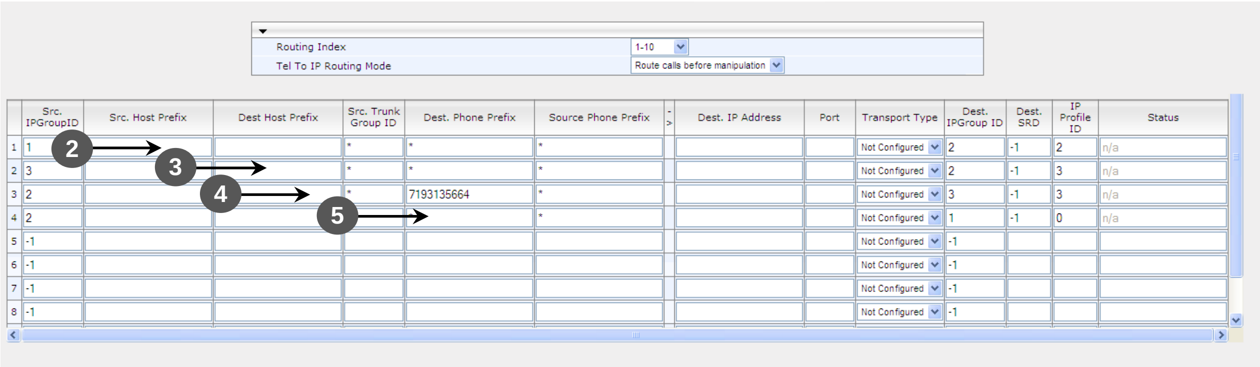
4.12.2 Configuring Outbound IP Routing

This step defines how to configure the gateway for outbound routing (i.e., sent) IP-to-IP calls to the Verizon Business SIP Trunk.

➢ To configure outbound IP routing:

1. Open the 'Outbound IP Routing Table' page (**Configuration** tab > **VoIP** menu > **GW and IP to IP** > **Routing** sub-menu > **Tel to IP Routing**).

Figure 4-29: Outbound IP Routing Table Page



Routing Index	1-10
Tel To IP Routing Mode	Route calls before manipulation

	Src. IPGroupID	Src. Host Prefix	Dest Host Prefix	Src. Trunk Group ID	Dest. Phone Prefix	Source Phone Prefix	->	Dest. IP Address	Port	Transport Type	Dest. IPGroup ID	Dest. SRD	IP Profile ID	Status
1	1			*	*	*				Not Configured	2	-1	2	n/a
2	3			*	*	*				Not Configured	2	-1	3	n/a
3	2			*	7193135664	*				Not Configured	3	-1	3	n/a
4	2				*	*				Not Configured	1	-1	0	n/a
5	-1									Not Configured	-1			
6	-1									Not Configured	-1			
7	-1									Not Configured	-1			
8	-1									Not Configured	-1			

2. **Index #1** defines the routing of IP calls to the Verizon Business SIP Trunk. All calls received from IP Group ID **1** (i.e., Lync 2010 Mediation server) are routed to Destination IP Group ID **2** (i.e., Verizon Business SIP Trunk):
   - *Source IP Group ID*: Select **1** to indicate received (inbound) calls identified as belonging to the IP Group configured for the Lync 2010 Mediation Server.
   - *Dest Phone Prefix*: Enter the asterisk (*) symbol to indicate all destinations.
   - *Source Phone Prefix*: Enter the asterisk (*) symbol to indicate all callers.
   - *Dest IP Group ID*: Select **2** to indicate the destination IP Group to where the calls must be sent, i.e., to the Verizon Business SIP Trunk.
   - *IP Profile*: *Select* **2** to indicate usage of IP Profile supporting Verizon coder preference.
3. **Index #2** defines the routing of FAX IP calls to the Verizon Business SIP Trunk. All calls received from IP Group ID 3 (i.e., ATA support FAX) are routed to Destination IP Group ID 2 (i.e., Verizon Business SIP Trunk):
   - *Source IP Group ID*: Select **3** to indicate received (inbound) calls identified as belonging to the IP Group configured for the ATA supporting FAX service.
   - *Dest Phone Prefix*: Enter the asterisk (*) symbol to indicate all destinations.
   - *Source Phone Prefix*: Enter the asterisk (*) symbol to indicate all callers.
   - *Dest IP* Group ID: Select **2** to indicate the destination IP Group to where the calls must be sent, i.e., to the Verizon Business SIP Trunk
   - *IP Profile*: Select **3** to indicate usage of IP Profile supporting Fax.

4. **Index #3** defines routing of FAX IP calls to the ATA supporting FAX. All calls received from Source IP Group ID 2 (i.e., from the Verizon Business SIP Trunk) are matched against assigned FAX DIDs and then routed to Destination IP Group ID 3 (i.e., to ATA supporting FAX services):
   - *Source IP Group ID*: Select **2** to indicate received (inbound) calls identified as belonging to the IP Group configured for the Verizon Business SIP Trunk.
   - *Dest Phone Prefix*: Enter the assigned Verizon DID associated with the customers fax services.
   - *Source Phone Prefix*: Enter the asterisk (*) symbol to indicate all callers.
   - *Dest IP Group ID*: Select **3** to indicate the destination IP Group to where the calls must be sent, i.e., to ATA supporting FAX services.
   - *IP Profile*: Select **3** to indicate usage of IP Profile supporting Fax.

5. **Index #4** defines routing of IP calls to the Lync 2010 Mediation server. All calls received from Source IP Group ID 1 (i.e., from the Verizon Business SIP Trunk) are routed to Destination IP Group ID 2 (i.e., to Lync 2010 Mediation server):
   - *Source IP Group ID*: Select **2** to indicate received (inbound) calls identified as belonging to the IP Group configured for the Verizon Business SIP Trunk.
   - *Dest Phone Prefix*: Enter the asterisk (*) symbol to indicate all destinations.
   - *Source Phone Prefix*: Enter the asterisk (*) symbol to indicate all callers.
   - *Dest IP Group ID*: Select **1** to indicate the destination IP Group to where the calls must be sent, i.e., to Lync 2010 Mediation server.
   - *IP Profile*: Select **0** to indicate no special usage of IP Profile. Instead, service will be selected from the base provisioning such as ‘Coders’.

4.13 Step 13: Configuring Number Manipulation

The Manipulation Tables sub-menu allows you to configure number manipulation and mapping of NPI/TON to SIP messages. This sub-menu includes the following options:

- Dest Number Tel->IP. See Section 4.13.1 on page 84.
- Source Number Tel->IP. See Section 4.13.2 on page 85.
- Redirect Number Tel->IP. See Section 4.13.3 on page 86

4.13.1 Configuring Destination Phone Number Manipulation

This section describes how to configure the destination phone number manipulation.

➢ **To configure Destination Phone Number Manipulation Table for Tel -> IP Calls Table:**

1. Open the 'Destination Phone Number Manipulation Table for Tel > IP calls' page (**Configuration** tab > **VoIP** menu > **GW and IP to IP** > **Manipulations** sub-menu > **Dest Number Tel** > **IP**).

Figure 4-30: Destination Phone Number Manipulation Table for Tel -> IP Calls Page

Note: Select row index to modify the relevant row.

Add

Index	Source Trunk Group	Source IP Group	Destination Prefix	Source Prefix	Stripped Digits From Left	Stripped Digits From Right	Prefix to Add	Suffix to Add	Number of Digits to Leave
1	-1	1	+011	*	1	0			255
2	-1	1	+1	*	2	0			255
3	-1	1	+	*	1	0			255
4	-1	2	719	*	0	0	+1		255

- **Index #1** defines destination number manipulation of IP calls from Lync Server 2010. All calls received from Source IP Group 1 (i.e., from Lync Server 2010), where the destination number prefix begins with '+011', remove the '+'.
- **Index #2** defines destination number manipulation of IP calls from Lync Server. All calls received from Source IP Group 1 (i.e., from Lync Server 2010), where the destination number prefix begins with '+1', remove the '+1'.
- **Index #3** defines destination number manipulation of IP calls from Lync Server. All calls received from Source IP Group 1 (i.e., from Lync Server 2010), where the destination number prefix begins with '+', remove the '+'.
- **Index #4** defines destination number manipulation of IP calls from Verizon Business SIP Trunk. All calls received from Source IP Group 2 (i.e., from Verizon Business SIP Trunk), where the destination number prefix is '719', add the '+1' prefix to the number.

4.13.2 Configuring Source Phone Number Manipulation

This section describes how to configure the source phone number manipulation.

➢ **To configure Source Phone Number Manipulation Table for Tel -> IP Calls Table:**

1. Open the 'Source Phone Number Manipulation Table for Tel -> IP calls' page (**Configuration** tab > **VoIP** menu > **GW and IP to IP** > **Manipulations** sub-menu > **Source Number Tel** > **IP**).

Figure 4-31: Source Phone Number Manipulation Table for Tel -> IP Calls Page

Index	Source Trunk Group	Source IP Group	Destination Prefix	Source Prefix	Stripped Digits From Left	Stripped Digits From Right	Prefix to Add	Suffix to Add	Number of Digits to Leave	Presentation
1	-1	-1	*	+17193135662	2	0			255	Restricted
2	-1	-1	*	+1	2	0			255	Not Configured
4	-1	-1	*	anonymous	20	0	7193135662		255	Restricted
5	-1	-1	*	+	1	0			255	Not Configured

- **Index #1** defines Source number manipulation of IP calls from any Source IP Group, where the Source number or prefix begins with '+17193135662'. In this case, remove the '+1' prefix from the number and set the Presentation to Restricted. This enables the customer to set restricted presentation for a specific DID or range of DIDs.
- **Index #2** defines Source number manipulation of IP calls from any Source IP, where the Source number prefix begins with '+1'. In this case, remove the '+1' prefix from the number.
- **Index #3** defines Source number manipulation of anonymous calls from any Source IP group. All calls received a Source number 'anonymous' are replaced with a known valid Verizon DID, such as '7193135662' and the presentation should be set to 'restricted'. This is a unique interworking to support the forwarding of calls originating from the PSTN and forwarded back to the PSTN when the original calling party was restricted. In this instance, the Lync Server 2010 sends the source as 'anonymous'.
- **Index #4** defines Source number manipulation of IP calls from any Source IP Group where the Source number prefix is a '+'. In this case, remove the '+' prefix from the number.

4.13.3 Configuring Redirect Number Manipulation

In the event of a call forwarding scenario, a Diversion header needs to be added to the INVITE towards the Verizon Business SIP Trunk (as configured in Section 4.9 on page 73). In this case, the E-SBC device copies and adds the referred-by number to a newly created Diversion header. For a well known number in the Diversion header (for Verizon Business SIP Trunk), a manipulation rule should be defined to remove the (+1) from the redirect number to be presented as a well known number derived from the Referred-by header received from the Lync Server 2010 environment during the forwarding scenario.

➢ **To configure redirect number Tel -> IP Table:**

1. Open the 'Redirect Number Tel -> IP' page (**Configuration** tab > **VoIP** menu > **GW and IP to IP** > **Manipulations** sub-menu> **Redirect Number Tel** > **IP**).

Figure 4-32: Redirect Number Tel -> IP Page

Index	Source Trunk Group	Source IP Group	Destination Prefix	Redirect Prefix	Stripped Digits From Left	Stripped Digits From Right	Prefix to Add	Suffix to Add	Number of Digit Leave
1	-1	-1	*	+1	2	0			255

- Index **#1** defines redirect number manipulation for the call forwarding scenario.

The redirect number is changed to remove the "+1" prefix from the number i.e. "+17193133661" to "7193133661".

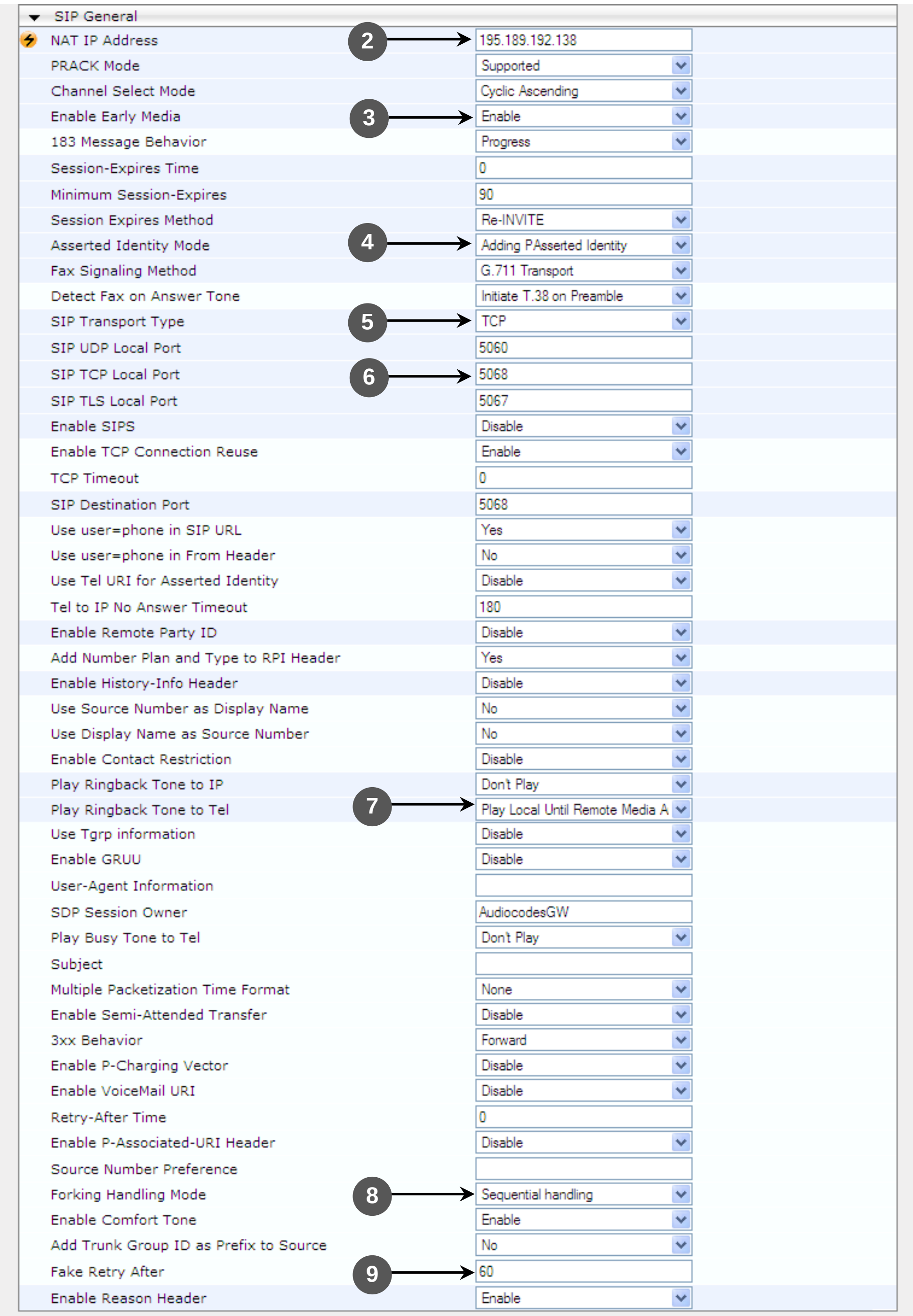
4.14 Step 14: Configuring SIP General Parameters

This section describes how to configure the SIP general parameters.

➢ To configure general SIP parameters:

1. Open the 'SIP General Parameters' page (**Configuration** tab > **VoIP** menu > **SIP Definitions** sub-menu > **General Parameters**).

Figure 4-33: SIP General Parameters Page



2. In the 'NAT IP Address' field, enter the Global (public) IP address of the E-SBC device to enable the static NAT between the E-SBC device and the Internet.

3. From the ‘Enable Early Media’ drop-down list, select **Enable** to enable early media.
4. From the ‘Asserted Identity Mode’ drop-down list, select **Adding PAsserted Identity**.
5. From the ‘SIP Transport Type’ drop-down list, select **TCP** in case the Mediation Server is configured to use TCP transport Type.
6. In the ‘SIP TCP Local Port’ field, enter **5068**; this port is the listening E-SBC port for TCP transport type. This port must match the transmitting port of the Mediation Server.
7. From 'Play Ringback Tone to Tel' drop-down list, select **Play Local Until Remote Media Arrive**. Plays the RBT according to the received media. If a SIP 180 response is received and the voice channel is already open (due to a previous 183 early media response or due to an SDP in the current 180 response), the E-SBC device plays a local RBT if there are no prior received RTP packets. The E-SBC device stops playing the local RBT as soon as it starts receiving RTP packets. At this stage, if the E-SBC device receives additional 18x responses, it does not resume playing the local RBT.
8. From the 'Forking Handling Mode' drop-down list, select **Sequential handling**; this parameter determines whether18x with SDP is received. In this case, the E-SBC device opens a voice stream according to the received SDP. The E-SBC device re-opens the stream according to subsequently received 18x responses with SDP.
9. In the 'Fake Retry After' field, enter **60** sec. This parameter determines whether the E-SBC device, upon receipt of a SIP 503 response without a Retry-After header, behaves as if the 503 response included a Retry-After header and with the period (in seconds) specified by this parameter.
10. Open the 'Admin" page, by appending the case-sensitive suffix ‘AdminPage’ to the E-SBC device's IP address in your Web browser's URL field (e.g., http://10.15.4.15/AdminPage).
11. On the left pane, click ***ini*** Parameters.
12. In the 'Parameter Name' field, enter the parameter **IGNOREALERTAFTEREARLYMEDIA**. In the 'Enter Value' field, enter “1”.
13. Click **Apply New Value**.

Figure 4-34: INI file Output Window

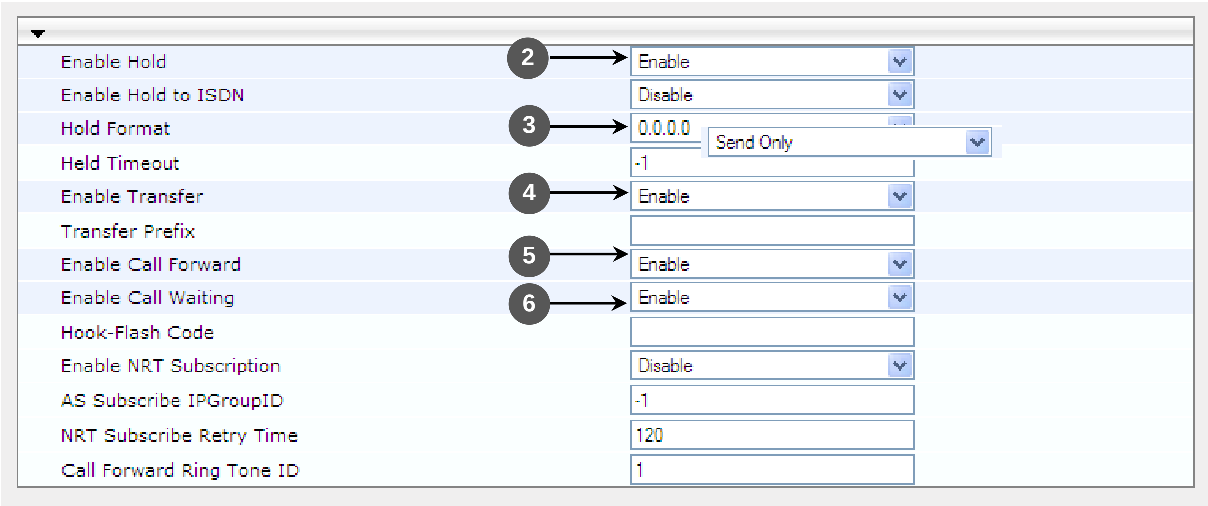
4.15 Step 15: Configuring SIP Supplementary Services

This section describes how to configure the SIP general parameters.

➢ To configure general SIP Supplementary Services parameters:

1. Open the 'Supplementary Services' page (**Configuration** tab > **VoIP** menu > **GW and IP to IP** sub-menu > **DTMF and Supplementary** sub-menu > **Supplementary Services**).

Figure 4-35: SIP Supplementary Services Page



2. In the 'Enable Hold' drop-down list, select **Enable**.
3. From the 'Hold Format' drop-down list, select **0.0.0.0** to enable Hold in the no media in either direction method or select **Send Only** to enable Hold to support one way audio for Music on Hold type service support.
4. From the 'Enable Transfer' drop-down list, select **Enable**.
5. From the 'Enable Call Forward' drop-down list, select **Enable**.
6. In the 'Enable Call Waiting' drop-down list, select **Enable**.

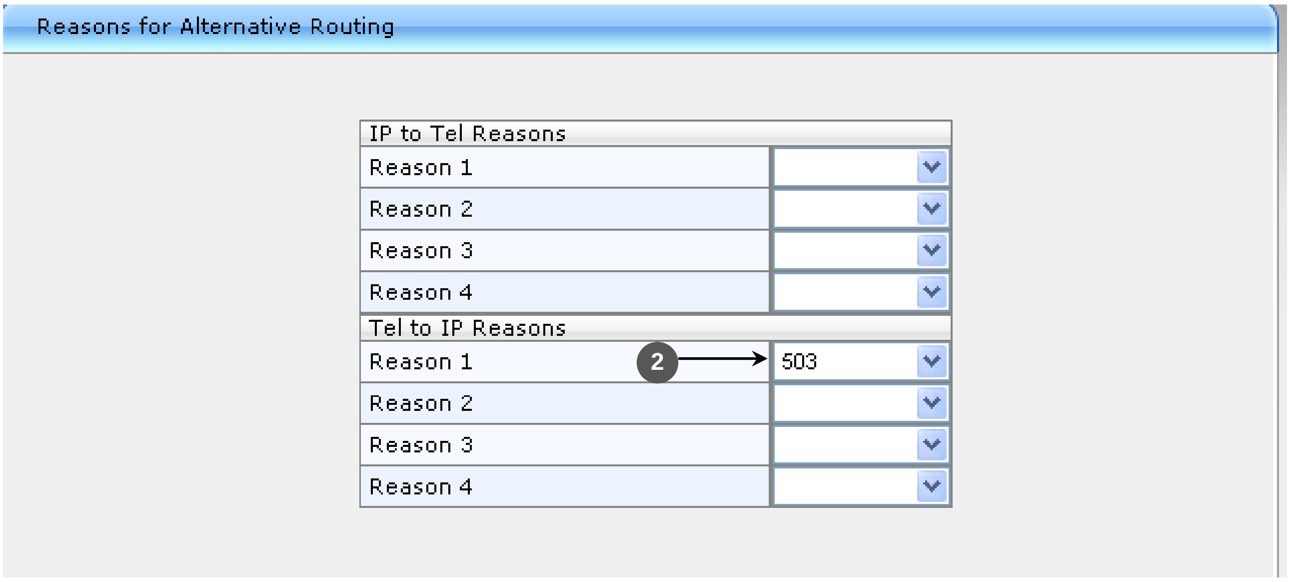
4.16 Step 16: Defining Reasons for Alternative Routing

A 503 SIP response from the Mediation Server to an INVITE must cause the E-SBC device to perform a failover. For this event to occur, you need to configure the Reasons for Alternative Routing for Tel-to-IP calls to be a 503 SIP response.

➢ To define SIP Reason for Alternative Routing:

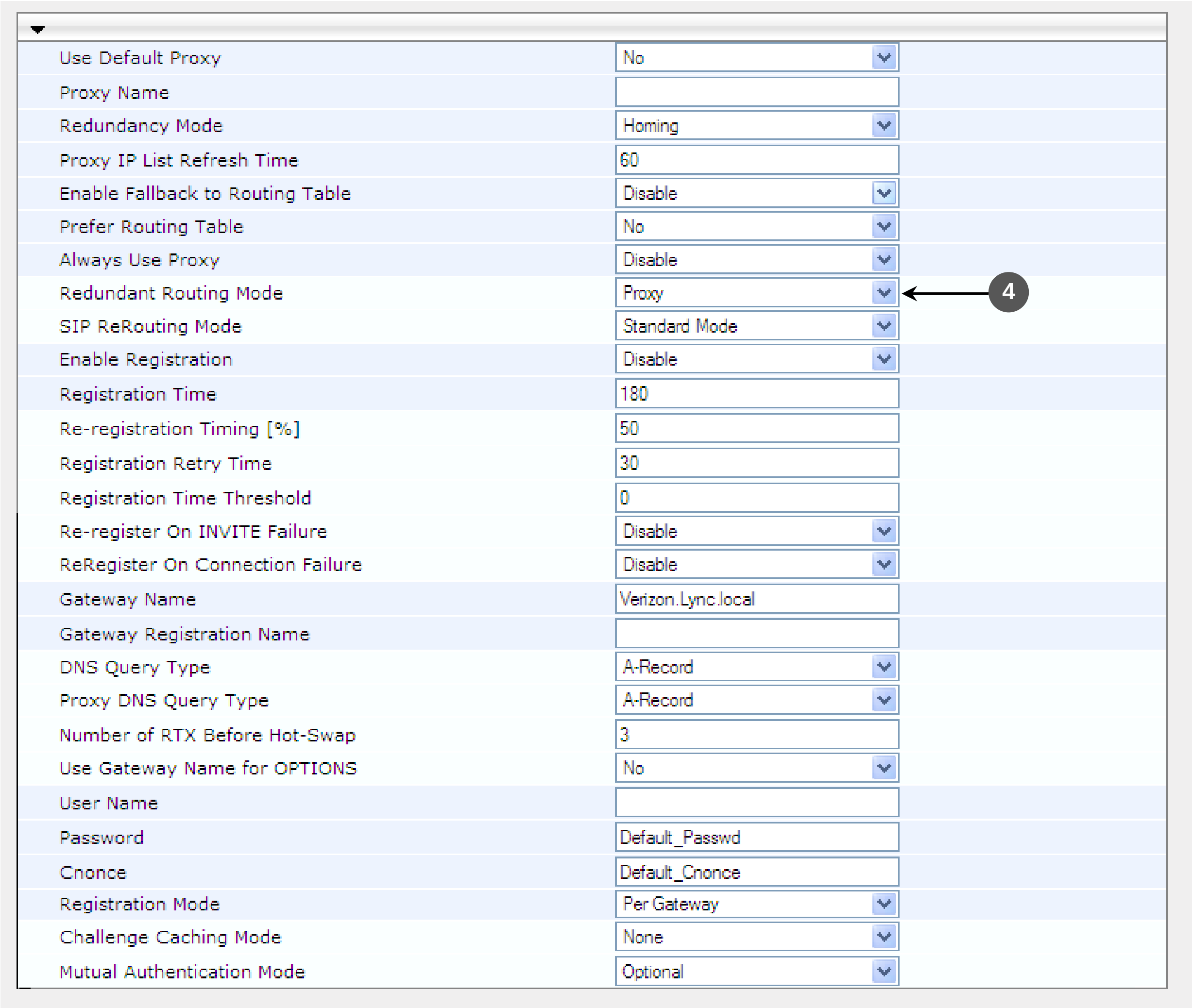
1. Open the 'Reasons for Alternative Routing' page (**Configuration** tab > **VoIP** menu > **GW and IP to IP** > **Routing** sub-menu > **Alternative Routing Reasons**).

Figure 4-36: Reasons for Alternative Routing Page



2. Under the Tel to IP Reasons group, for Reason 1, select **503**.
3. Click **Submit**.
4. Open the 'Proxy & Registration' page (**Configuration** > **VoIP** > **SIP Definitions** > **Proxy & Registration**) and configure the 'Redundant Routing Mode' parameter to **Proxy** as shown below.

Figure 4-37: Proxy & Registration Page



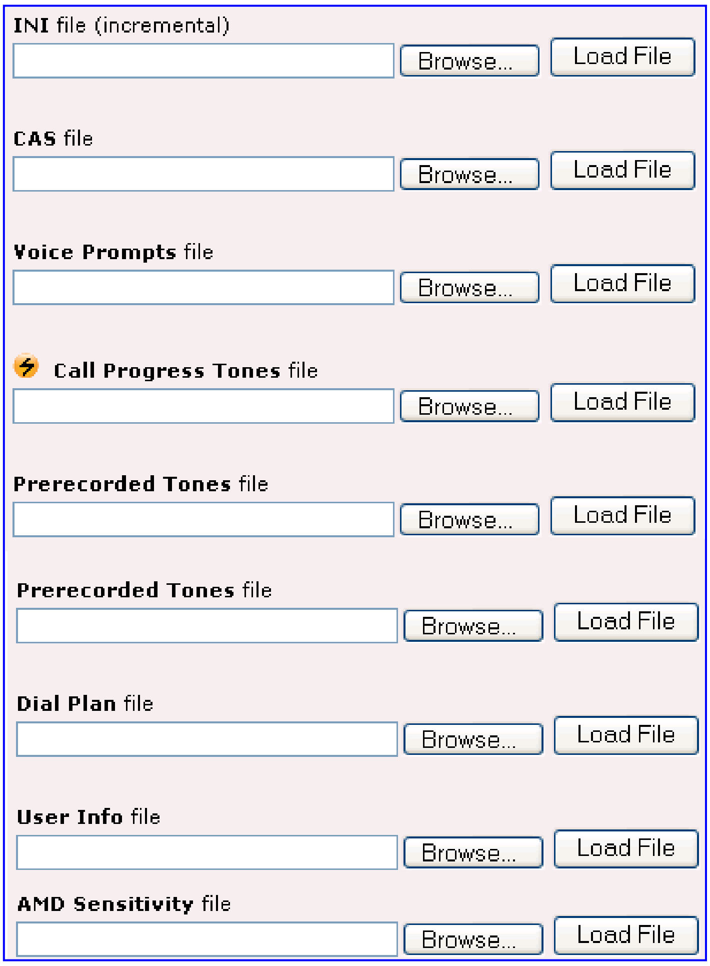
4.17 Step 17: Supporting Regional Call Progress Tone

The 'Load Auxiliary Files' page allows you to load various auxiliary files to the device. For Regional Call Progress Tones support, use the “Call Progress Tones file” area to select the appropriate regional file stored on your local computer to send to the device for proper support and interworking.

➢ **To load an auxiliary file to the device using the Web interface:**

1. Open the 'Load Auxiliary Files' page (**Maintenance** tab > **Software Update** menu > **Load Auxiliary Files**).

Figure 4-38:'Load Auxiliary Files Page



Note: The appearance of certain file load fields depends on the installed Software Upgrade Key.

2. Click the **Browse** button corresponding to the file type that you want to load, navigate to the folder in which the file is located, and then click **Open**; the name and path of the file appear in the field next to the **Browse** button.
3. Click the **Load File** button corresponding to the file you want to load.
4. Repeat steps 2 through 3 for each file you wish to load.
5. Save the loaded auxiliary files to flash memory, and reset the device (if you have loaded a Call Progress Tones file).

Reader's Notes

5 Troubleshooting

This section should provide some tips for troubleshooting problems, including troubleshooting commands and case reporting procedures for trouble escalation.

5.1 Debugging Procedures

This section discusses the following debugging procedures:

- Case Reporting Procedures. See Section 5.1.1 below.
- Syslog. See Section 5.1.2 on page 96.
- Wireshark Network Sniffer. See Section 5.1.3 on page 98.

5.1.1 Case Reporting Procedures

When reporting a problem to AudioCodes' Technical Support department, the following information should be provided:

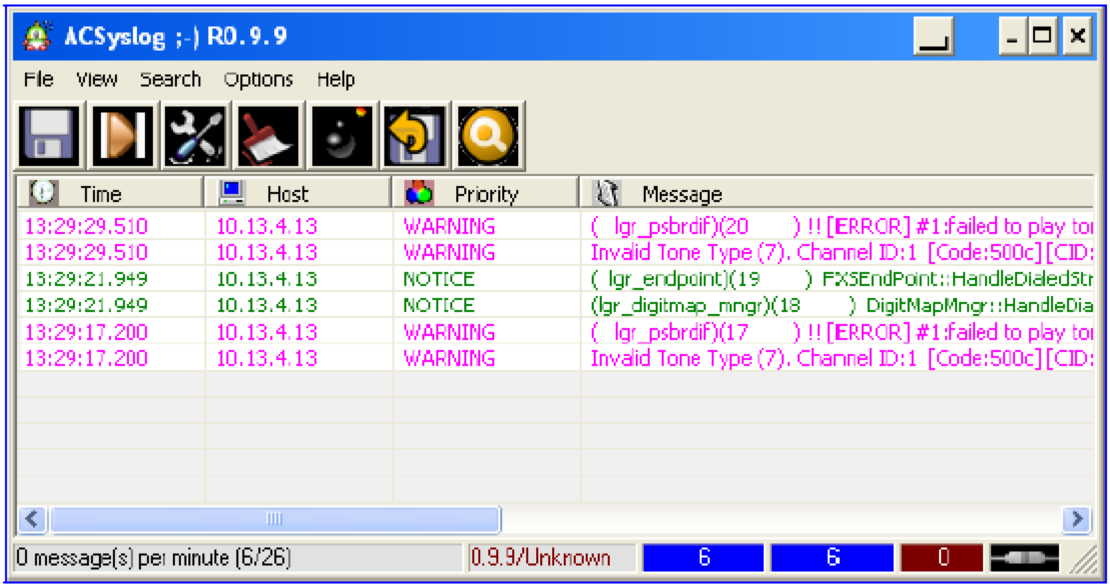
- Basic information (required for all types of problems):
  - Problem description (nature of failure, symptoms, call direction, etc.)
  - Network diagram
  - *ini* configuration file (downloaded to your PC from the device using the Web interface)
  - Syslog trace (without missing messages)
  - Unfiltered IP network trace using the Wireshark application

  (Note: If you are unable to collect all the network traffic, then at least collect the mandatory protocols SIP, RTP, and T38.)
- Advanced information (if required upon request):
  - PSTN message traces - for PSTN problems
  - Media stream traces - for problems related to voice quality, modem\fax, DTMF detection, etc.

5.1.2 Syslog

Syslog is a standard for forwarding log messages in an IP network. A syslog client, embedded in the device sends error reports/events generated by the device to a remote Syslog server using IP/UDP protocol. This information is a collection of error, warning and system messages that record every internal operation of the device. You can use the supplied AudioCodes proprietary Syslog server "ACSyslog" (shown in the figure below) or any other third-party Syslog server for receiving Syslog messages.

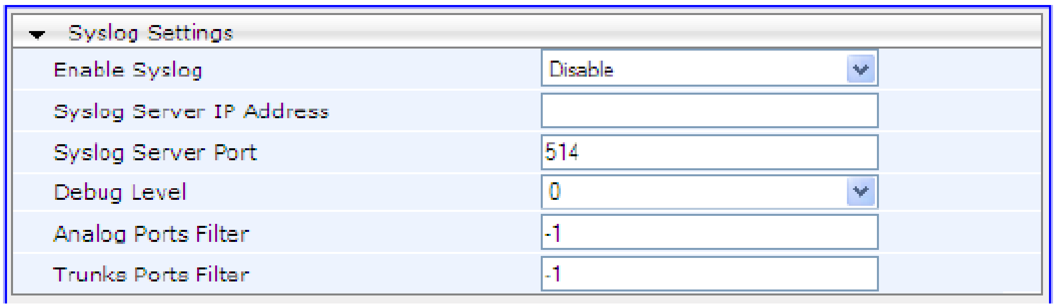
Figure 5-1: AudioCodes' Proprietary Syslog Server



➢ **To activate the Syslog client on the device using the Web interface:**

1. Open the 'Syslog Settings' page (**Configuration** tab > **System** menu > **Syslog Settings**).
2. In the 'Syslog Server IP Address' field, enter the IP address of the Syslog server (*ini* file parameter SyslogServerIP).
3. From the 'Enable Syslog' drop-down list, select 'Enable' to enable the device to send syslog messages to a Syslog server (defined in Step 2).

Figure 5-2: Enabling Syslog



4. From the 'Debug Level' drop-down list, select '5' if debug traces are required.

   To enable syslog reporting, using the *ini* file, load an *ini* file to the device with the following settings:

```
[Syslog]
SyslogServerIP = 192.168.2.35
EnableSyslog = 1
SyslogServerPort = 514
GWDebugLevel = 5
```

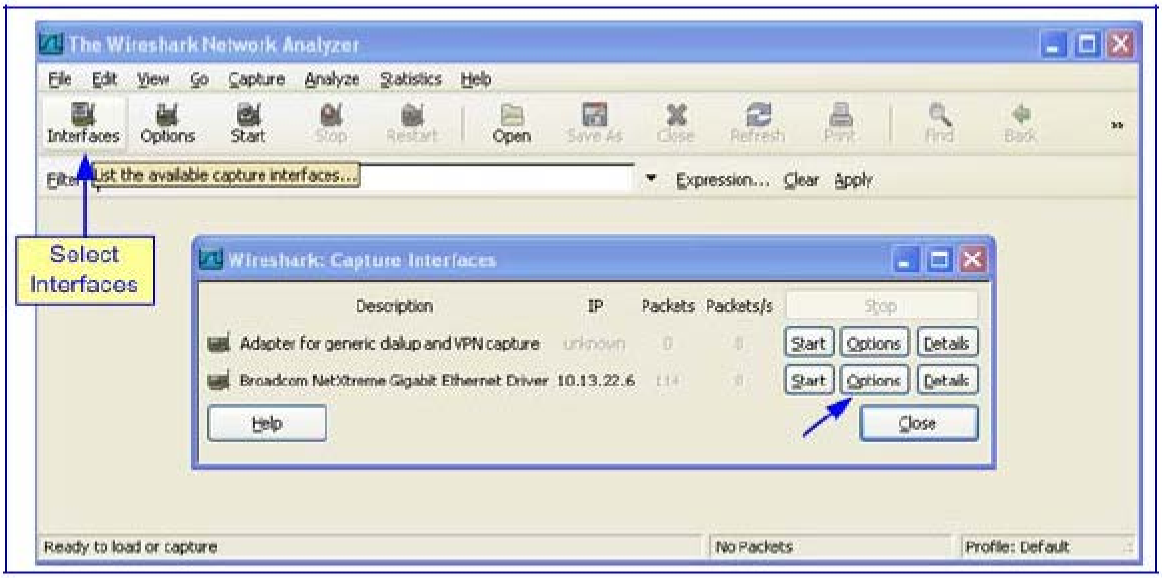
5.1.3 Wireshark Network Sniffer

Wireshark is a freeware packet sniffer application that allows you to view the traffic that is being passed over the network. Wireshark can be used to analyze any network packets. Wireshark can also be used to analyze RTP data streams and extract the audio from the data packets (only for G.711). The audio can be saved as a *.pcm file.

➢ **To record traffic sent to / from the device:**

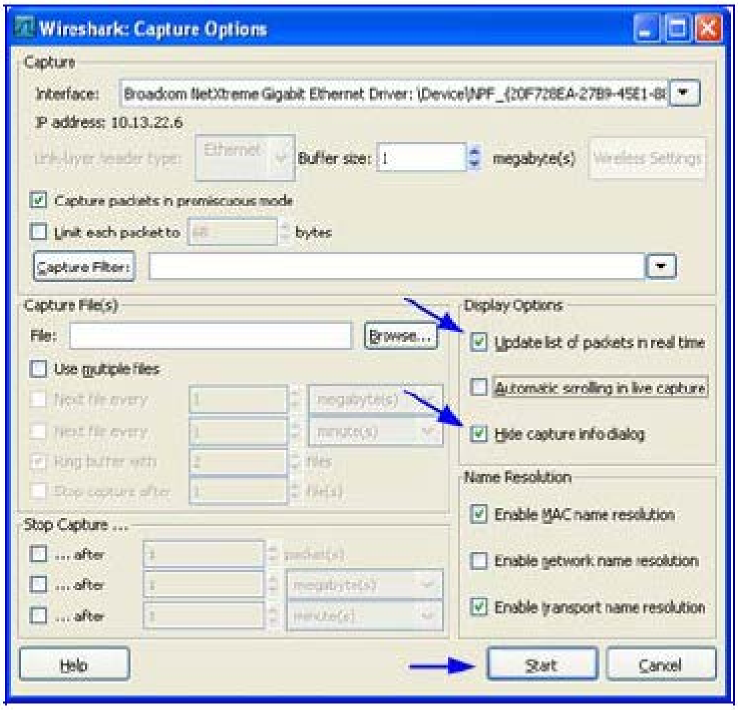
1. Install Wireshark on your PC. (You can download it from the following Web site: http://www.wireshark.org/
2. Connect the PC and the device to the same hub.
3. If you are using a switch, use a switch with port mirroring for the port to which the Wireshark is connected.
4. Start Wireshark.
5. Select the network interface that is currently being used by the PC - on the toolbar,click **Interfaces**, and then in the 'Capture Interfaces' dialog box, click the **Options** button corresponding to the network interface:

Figure 5-3: Selecting Interface Currently used by the PC



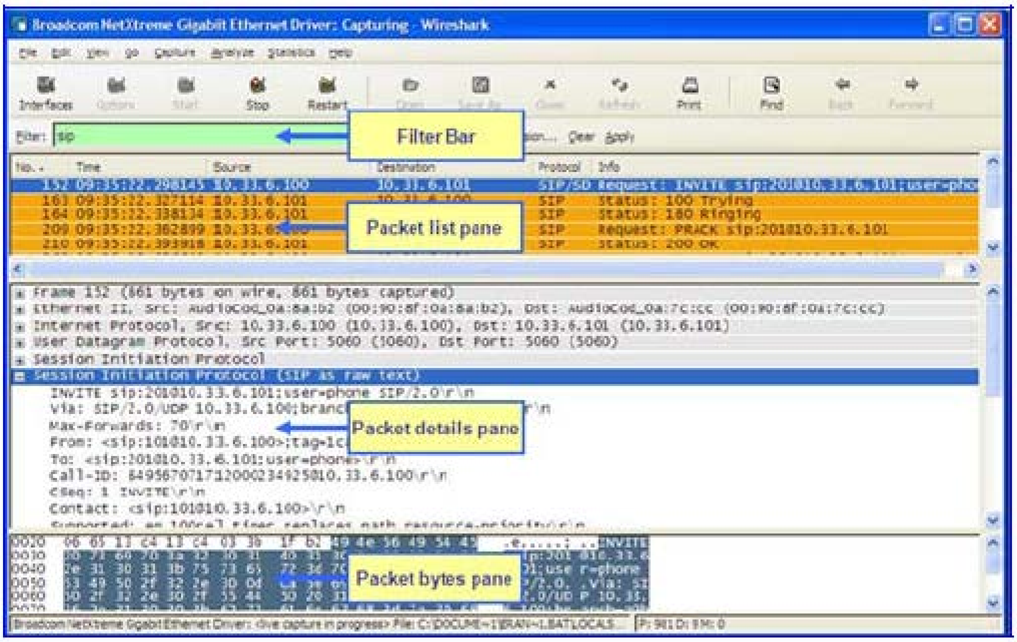
6. In the 'Capture Options' dialog box, select the desired display options:

Figure 5-4: Configuring Wireshark Display Options



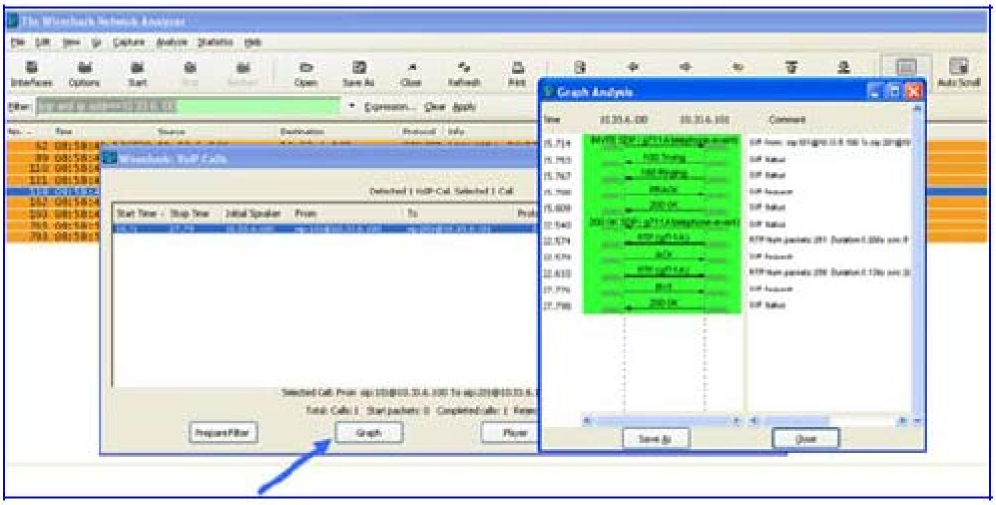
7. Click **Start**.

Figure 5-5: Captures Packets



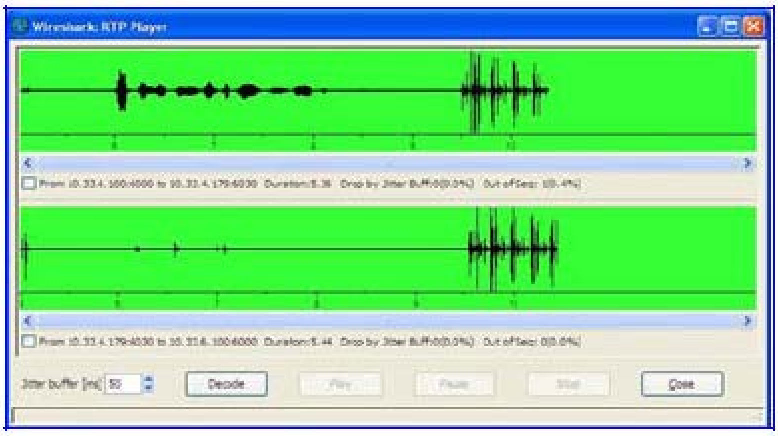
8. To view VoIP call flows, from the **Statistics** menu, choose **VoIP Calls**. You can view the statistics in graph format by clicking **Graph**.

Figure 5-6: Viewing VoIP Call Flows



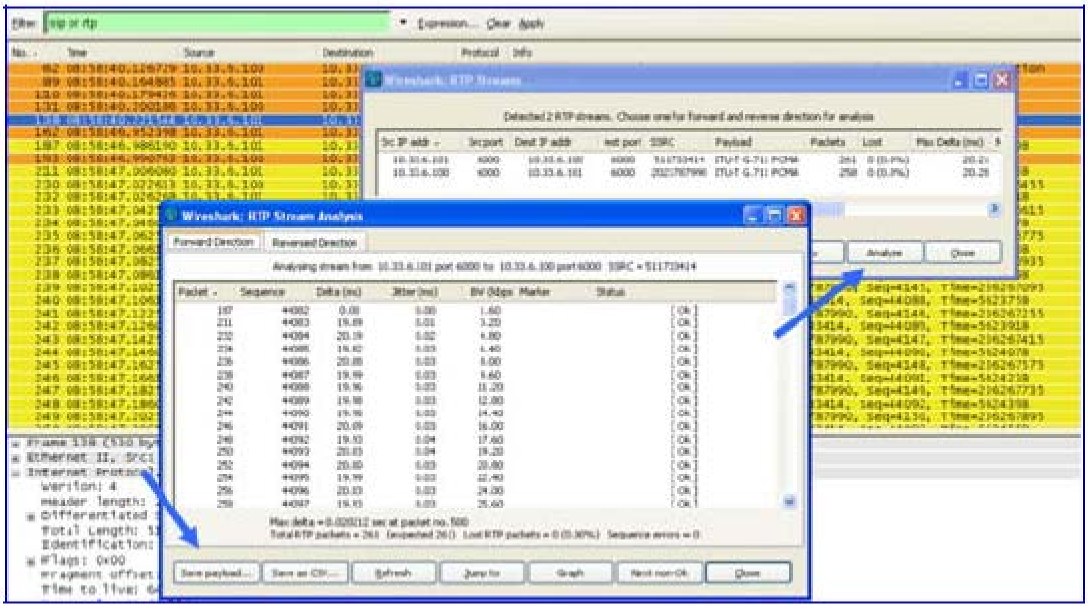
9. To play G.711 RTP streams, click the **Player** button.

Figure 5-7: Playing G.711 RTP Streams



10. To analyze the RTP data stream and extract the audio (which can be played using programs such as CoolEdit) from the data packets (only for G.711), from the **Statistics** menu, point to **RTP**, and then choose **Stream Analysis**.

Figure 5-8: Analyzing the RTP Data



a. Save the audio payload of the RTP stream to a file.

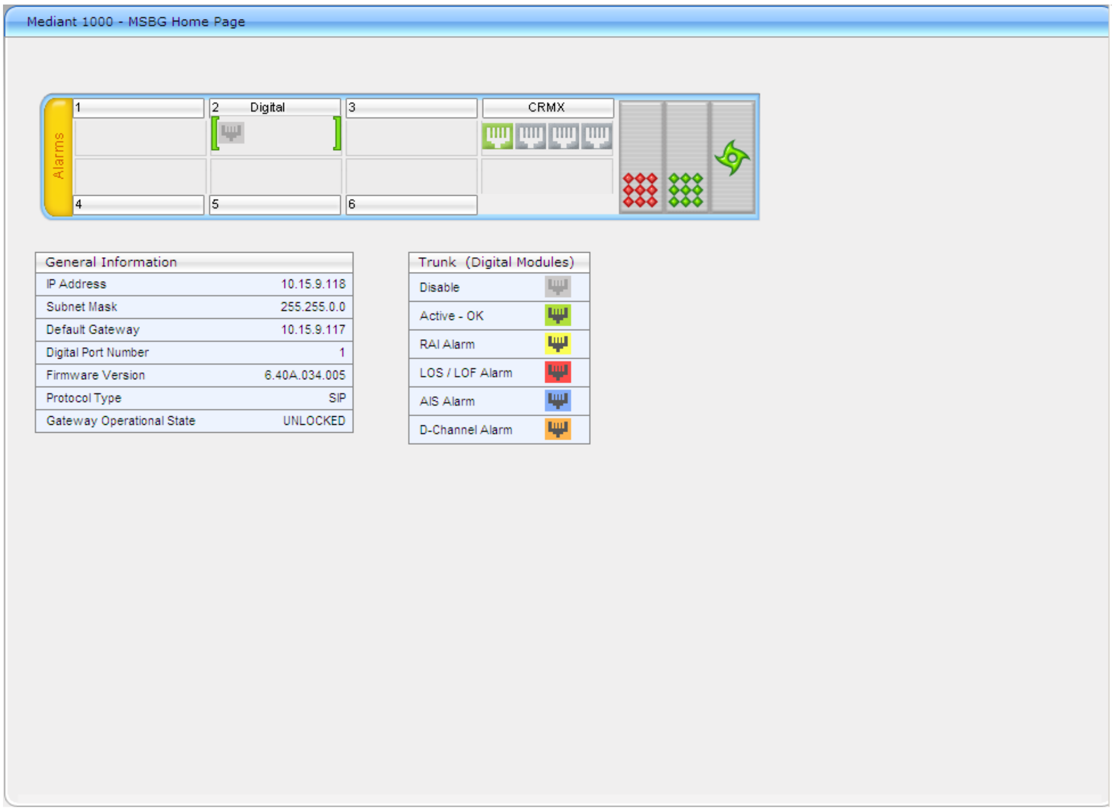
b. Save the Payload as a *.pcm file.

c. Select the 'forward' option.

5.2 Verifying Firmware

To verify the firmware load actively running on the device, log into the device and view the firmware version on the product homepage as shown in the figure below.

Figure 5-9: Viewing active firmware version



A Appendix: Configuring Analog Devices (ATA's) for FAX Support

This section describes how to configure the analog device entity to route its calls to the AudioCodes Media Gateway for supporting faxes. The analog device entity must be configured to send all calls to the Media Gateway without any registration process.

Note: The configuration described in this section is for ATA devices configured for AudioCodes MP-11x series.

A.1 Step 1: Configure the Endpoint Phone Number Table

The 'Endpoint Phone Number Table' page allows you to activate the MP-11x ports (endpoints) by defining telephone numbers. The configuration below uses the example of ATA1 destination phone number ''7193135664'' (IP address 10.15.7.165) with all routing directed to the AudioCodes E-SBC device (10.15.4.30).

➢ **To configure the Endpoint Phone Number table:**

- Open the 'Endpoint Phone Number Table' page (**Configuration** tab > **VoIP** menu > **GW and IP to IP submenu** > **Hunt Group** sub-menu > **Endpoint Phone Number**).

Figure 5-10: Endpoint Phone Number Table Page

Endpoint Phone Number Table

	Channel(s)	Phone Number	Hunt Group ID	Tel Profile ID
1	1	7193135664		0
2				
3				
4				

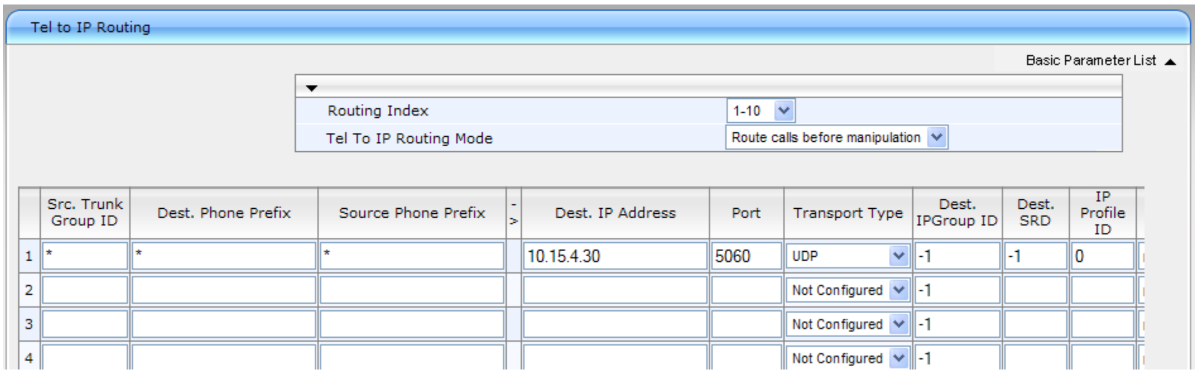
A.2 Step 2: Configure Tel to IP Routing Table

This step describes how to configure the Tel-to-IP routing rules to ensure that the MP-11x device sends all calls to the AudioCodes central E-SBC device.

➢ To configure the Tel to IP Routing table:

- Open the 'Tel to IP Routing' page (**Configuration** tab > **VoIP** menu > **GW and IP to IP** sub-menu > **Routing** sub-menu > **Tel to IP Routing**).

Figure 5-11: Tel to IP Routing Page



A.3 Step 3: Configure Coders Table

This step describes how to configure the coders for the MP-11x device.

➢ To configure MP-11x coders:

- Open the 'Coders' page (**Configuration** tab > **VoIP** menu > **Coders And Profiles** sub-menu > **Coders**).

Figure 5-12: Coders Table Page

Coder Name	Packetization Time	Rate	Payload Type	Silence Suppression
G.729	20	8	18	Disabled
G.711U-law	20	64	0	Disabled

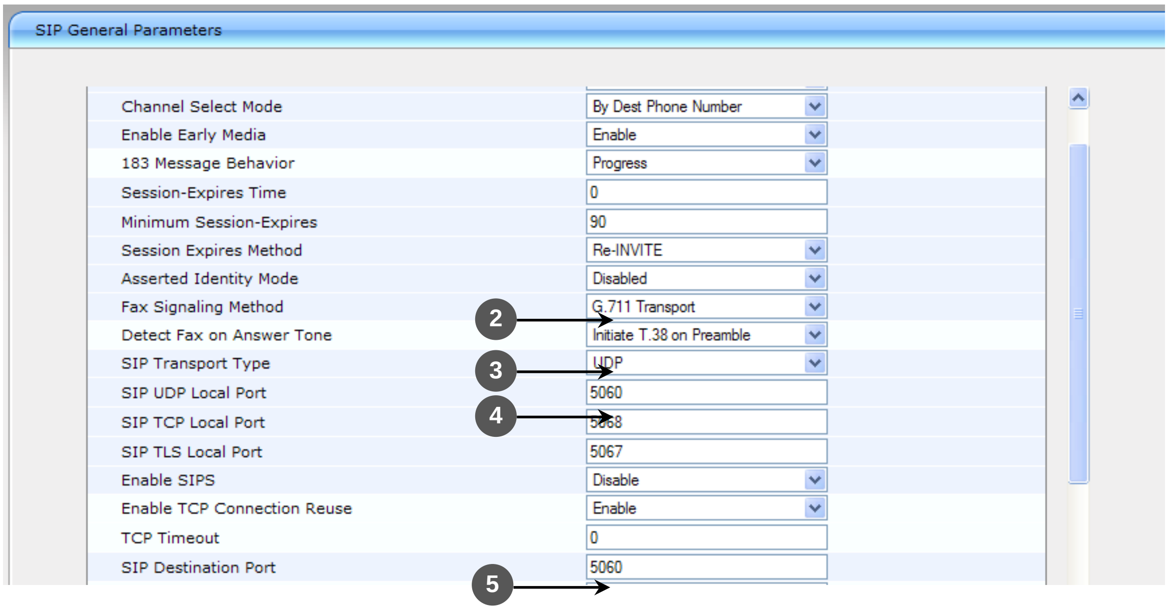
A.4 Step 4: Configure SIP UDP Transport Type and Fax Signaling Method

This step describes how to configure the fax signaling method for the MP-11x device.

➢ **To configure the fax signaling method:**

1. Open the 'SIP General Parameters' page (**Configuration** tab > **VoIP** menu > **SIP Definitions** submenu > **General Parameters**).

Figure 5-13: SIP General Parameters Page



2. From the 'FAX Signaling Method' drop-down list, select **G.711 Transport.**
3. From the 'SIP Transport Type' drop-down list, select **UDP**.
4. In the 'SIP UDP Local Port**'** field, enter **5060** (corresponding to the Central Gateway UDP transmitting port configuration).
5. In the 'SIP Destination Port', enter **5060** (corresponding to the Central Gateway UDP listening port configuration).

Reader's Notes

Configuration Note

www.audiocodes.com