Enterprise Session Border Controllers (E-SBC)

AudioCodes Mediant™ Series

Interoperability Lab

Configuration Note

Microsoft® Lync™ Server 2013 & Verizon SIP Trunk using AudioCodes Mediant E-SBC

Microsoft Partner
Gold Communications

Lync

verizon

AudioCodes

July 2014

Document #: LTRT-31906

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Notice

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Reader's Notes

1 Introduction

This Configuration Note describes how to set up AudioCodes Enterprise Session Border Controller (referred to in this document as *E-SBC*) for interworking between Verizon's SIP Trunk and Microsoft's Lync Server 2013 environment.

1.1 Intended Audience

The document is intended for engineers or AudioCodes and Verizon Partners responsible for installing and configuring Verizon's SIP Trunk and Microsoft's Lync Server 2013 for enabling VoIP calls using AudioCodes E-SBC.

1.2 About AudioCodes E-SBC Product Series

AudioCodes' family of E-SBC devices enables reliable connectivity and security between the Enterprise's and the service provider's VoIP networks.

The E-SBC provides perimeter defense as a way of protecting Enterprises 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.

Designed as a cost-effective appliance, the 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.

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

Reader's Notes

2 Component Information

2.1 AudioCodes E-SBC Version

Table 2-1: AudioCodes E-SBC Version

SBC Vendor	AudioCodes
Models	▪ Mediant 500 E-SBC ▪ Mediant 800 Gateway & E-SBC ▪ Mediant 1000B Gateway & E-SBC ▪ Mediant 2600 E-SBC ▪ Mediant 3000 Gateway & E-SBC ▪ Mediant 4000 E-SBC ▪ Mediant 9000 SBC ▪ Mediant Software SBC (Server Edition and Virtual Edition)
Software Version	SIP_6.80A.018.005
Protocol	▪ SIP/UDP (to the Verizon SIP Trunk) ▪ SIP/TCP or TLS (to the Lync FE Server)
Additional Notes	Fax is supported by analog devices (ATAs). FXS ports on devices that support FXS ports can also be used. Mediant 9000 SBC and Mediant Software SBC can be utilized if no transcoding is required for the deployment.

2.2 Verizon SIP Trunking Version

Table 2-2: Verizon Version

Vendor/Service Provider	Verizon
SSW Model/Service	-
Software Version	-
Protocol	SIP
Additional Notes	None

2.3 Microsoft Lync Server 2013 Version

Table 2-3: Microsoft Lync Server 2013 Version

Vendor	Microsoft
Model	Microsoft Lync
Software Version	Release 2013 5.0.8308.0
Protocol	SIP
Additional Notes	None

2.4 AudioCodes Gateway Version

Table 2-4: AudioCodes Gateway Version

Gateway Vendor	AudioCodes
Model	AudioCodes MP-11x series.
Software Version	SIP_6.60A.270.010
Interface Type	SIP/IP
VoIP Protocol	SIP
Additional Notes	Fax is supported by analog devices (ATAs). FXS ports on devices that support FXS ports can also be used. CNG detection enabled to support interwork with Verizon SIP Trunking network.

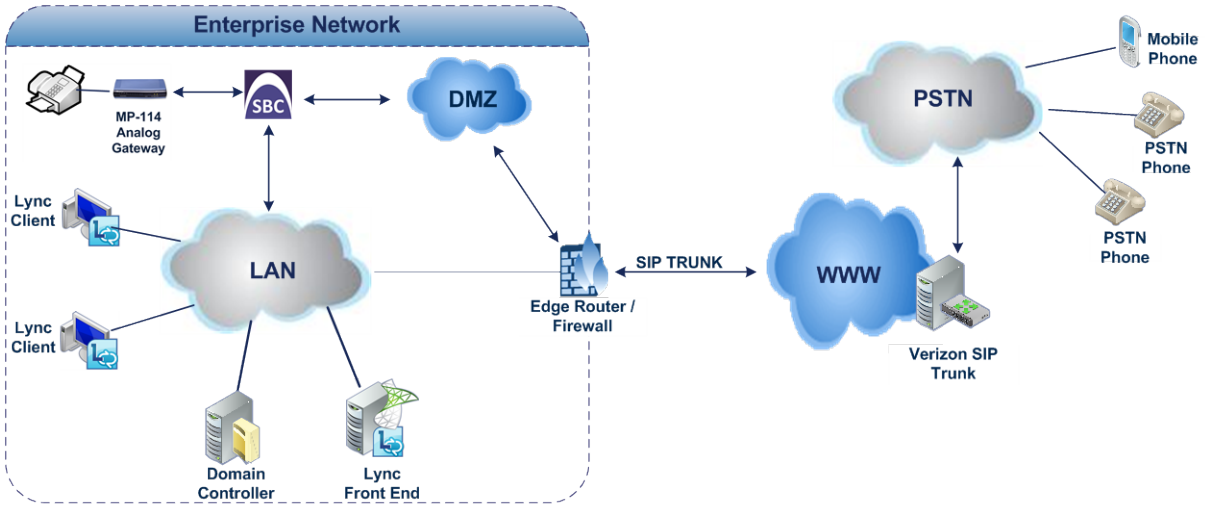
2.5 Interoperability Test Topology

Interoperability testing between AudioCodes E-SBC and Verizon SIP Trunk with Lync 2013 was performed using the following topology setup:

- The enterprise deployed Microsoft Lync Server 2013 in its private network for enhanced communication within the enterprise.
- The enterprise wants to offer its employees enterprise-voice capabilities and to connect the enterprise to the PSTN network using Verizon's SIP Trunking service.
- AudioCodes E-SBC is implemented to interconnect between the enterprise LAN and the SIP Trunk.
  - Session: Real-time voice session using IP-based Session Initiation Protocol (SIP).
  - Border: IP-to-IP network border between Lync Server 2013 network in the enterprise LAN and Verizon's SIP Trunk located in the public network.
  - Microsoft Lync Server 2013 works with TLS transport type while Verizon SIP trunk works on the SIP over UDP transport type.
- Transcoding support: Microsoft Lync Server 2013 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. Alternatively to the ATA, FXS ports on devices that support FXS ports can be used. Verizon Network cannot initiate T.38 FAX relay reliably through all network devices. Because of this, CNG detection and activation of T.38 relay is enabled.

Figure 2-1 illustrates this interoperability test topology.

Figure 2-1: Test Topology: Interoperability between Microsoft Lync and Verizon SIP Trunk using E-SBC



2.5.1 Environment Setup

The interoperability test topology includes the following environment setup:

Table 2-5: Environment Setup

Area	Setup
Network	▪ Microsoft Lync Server 2013 environment is located on the enterprise's LAN ▪ Verizon SIP Trunk is located on the WAN
Signaling Transcoding	▪ Microsoft Lync Server 2013 operates with SIP-over-TLS transport type ▪ Verizon SIP Trunk operates with SIP-over-UDP transport type
Codecs Transcoding	▪ Microsoft Lync Server 2013 supports G.711A-law and G.711U-law coders ▪ Verizon SIP Trunk supports G.711A-law, G.711U-law, and G.729 coder
Media Transcoding	▪ Microsoft Lync Server 2013 operates with SRTP media type ▪ Verizon SIP Trunk operates with RTP media type

2.5.2 Known Limitations

There were no limitations observed in the interoperability tests performed for the AudioCodes E-SBC interworking between Microsoft Lync Server 2013 and Verizon's SIP Trunk.

Reader's Notes

3 Configuring Lync Server 2013

This section describes how to configure Microsoft Lync Server 2013 to operate with AudioCodes E-SBC.

Note: Dial plans, voice policies, and PSTN uses are also necessary for enterprise voice deployment but are beyond the scope of this document.

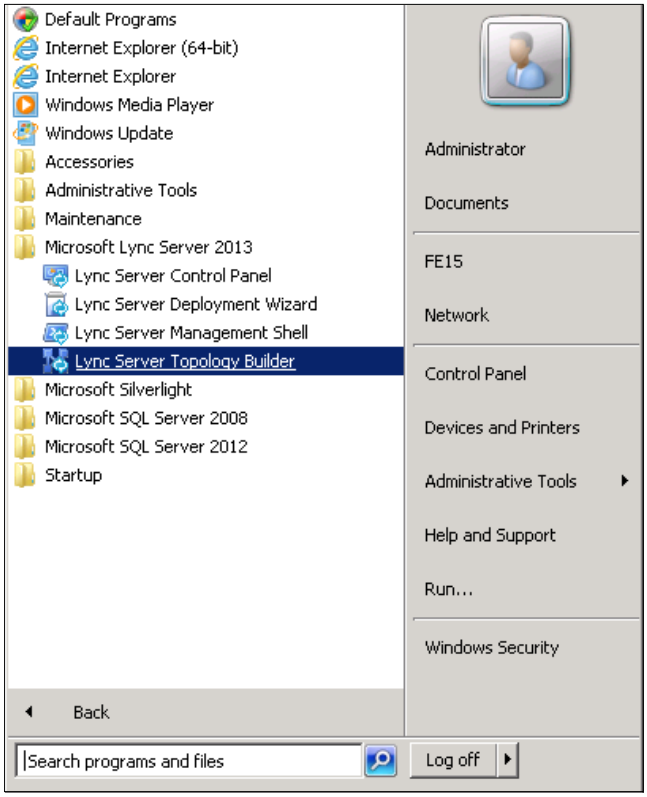
3.1 Configuring the E-SBC as an IP / PSTN Gateway

The procedure below describes how to configure the E-SBC as an IP / PSTN Gateway.

➢ **To configure E-SBC as IP/PSTN Gateway and associate it with Mediation Server:**

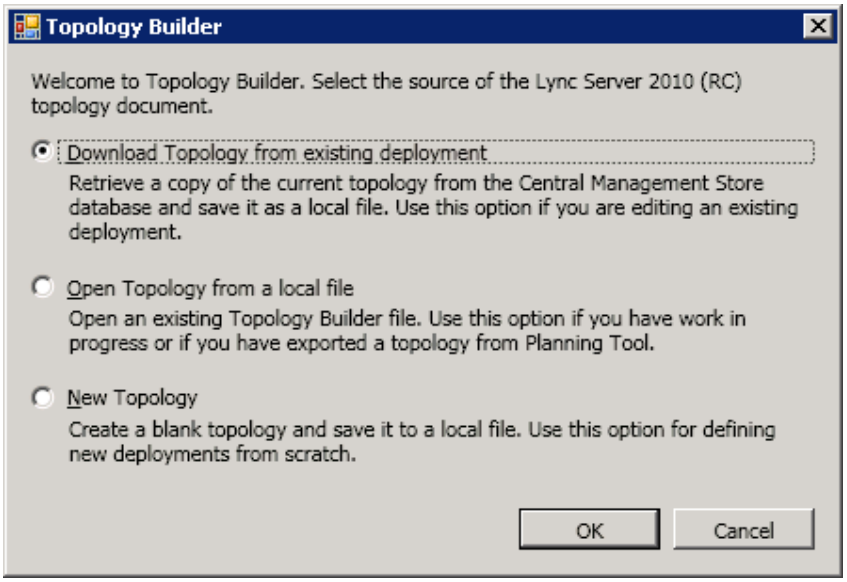
1. On the server where the Topology Builder is installed, start the Lync Server 2013 Topology Builder (Windows **Start** menu > **All Programs** > **Lync Server Topology Builder**), as shown below:

Figure 3-1: Starting the Lync Server Topology Builder



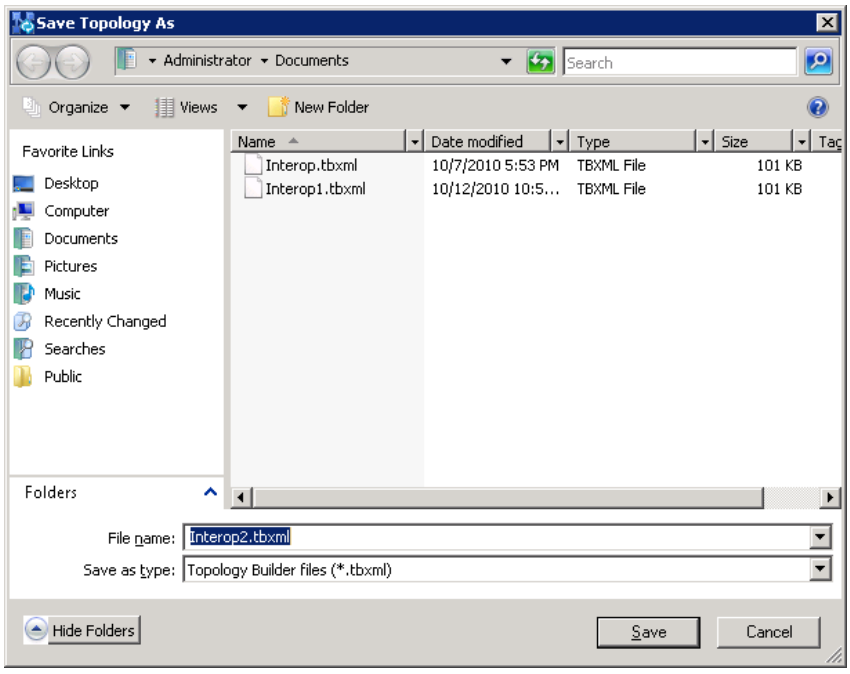
The following is displayed:

Figure 3-2: Topology Builder Dialog Box



2. Select the **Download Topology from existing deployment** option, and then click **OK**; you are prompted to save the downloaded Topology:

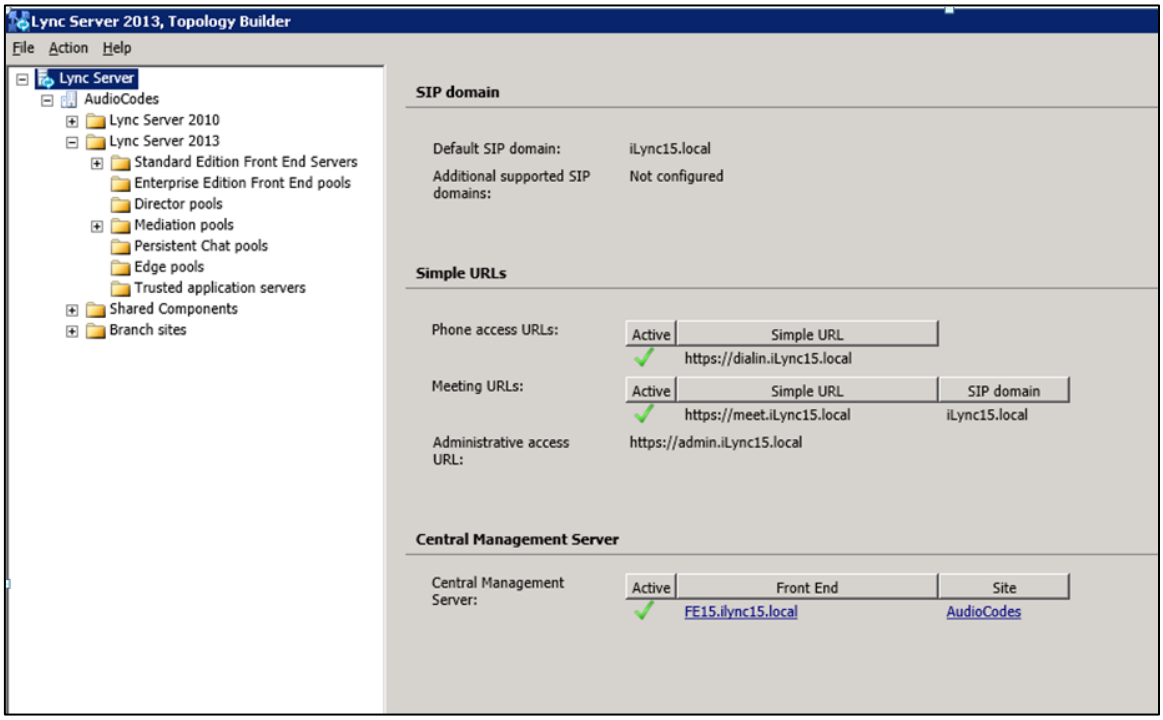
Figure 3-3: Save Topology Dialog Box



3. Enter a name for the Topology file, and then click **Save**. This step enables you to roll back from any changes you make during the installation.

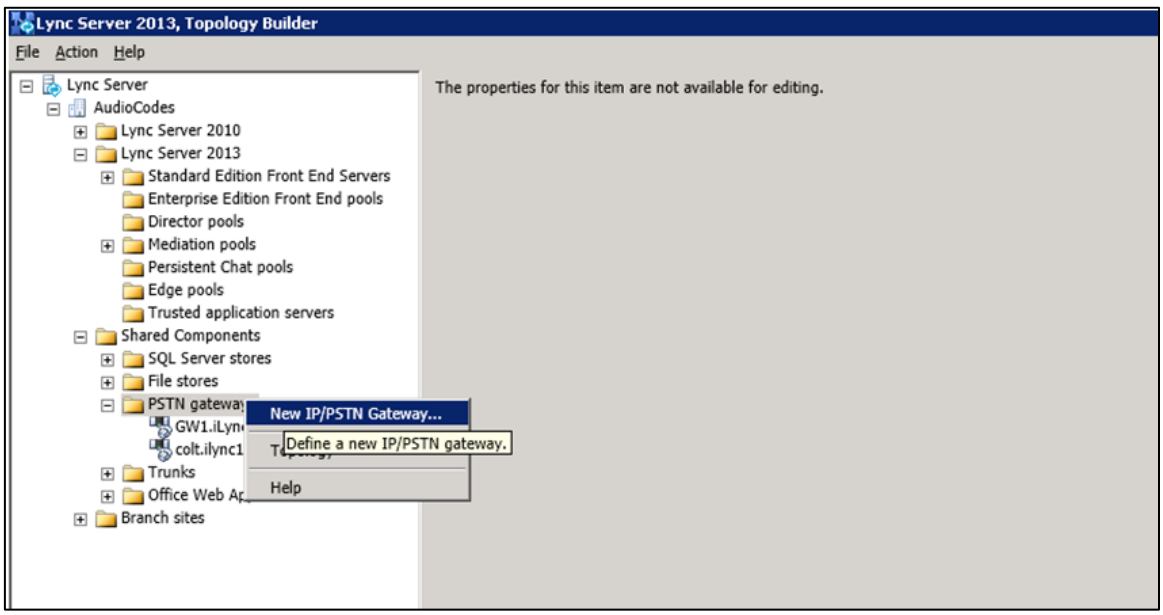
The Topology Builder screen with the downloaded Topology is displayed:

Figure 3-4: Downloaded Topology



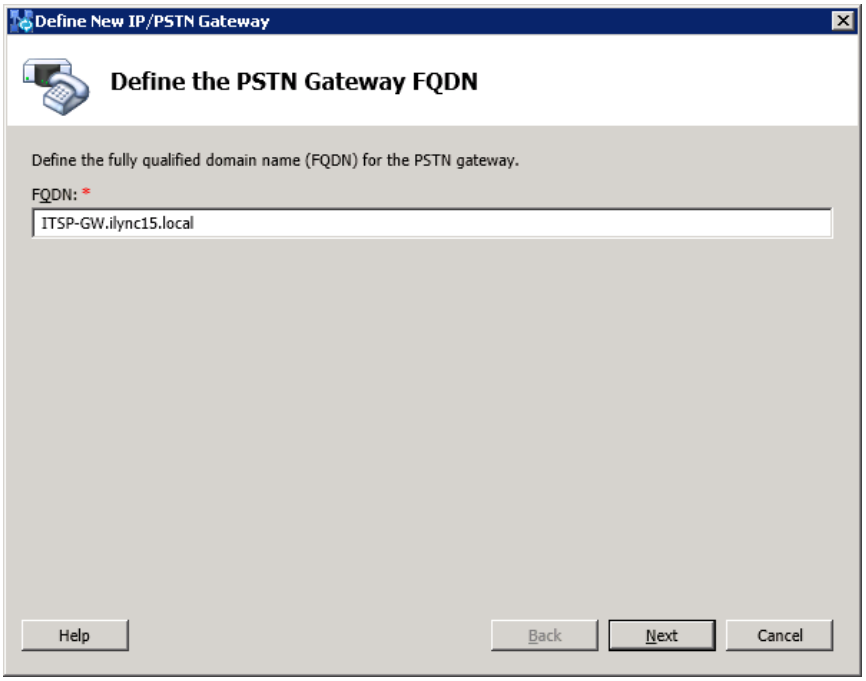
4. Under the **Shared Components** node, right-click the **PSTN gateways** node, and then from the shortcut menu, choose **New IP/PSTN Gateway**, as shown below:

Figure 3-5: Choosing New IP/PSTN Gateway



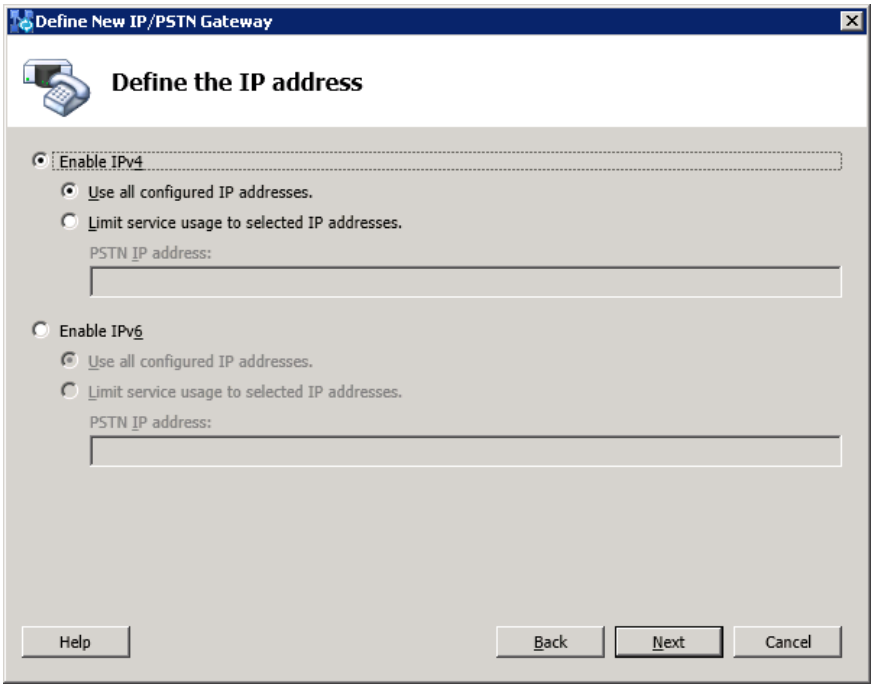
The following is displayed:

Figure 3-6: Define the PSTN Gateway FQDN



5. Enter the Fully Qualified Domain Name (FQDN) of the E-SBC (e.g., **ITSP-GW.ilync15.local**). Update this FQDN in the relevant DNS record, and then click **Next**; the following is displayed:

Figure 3-7: Define the IP Address

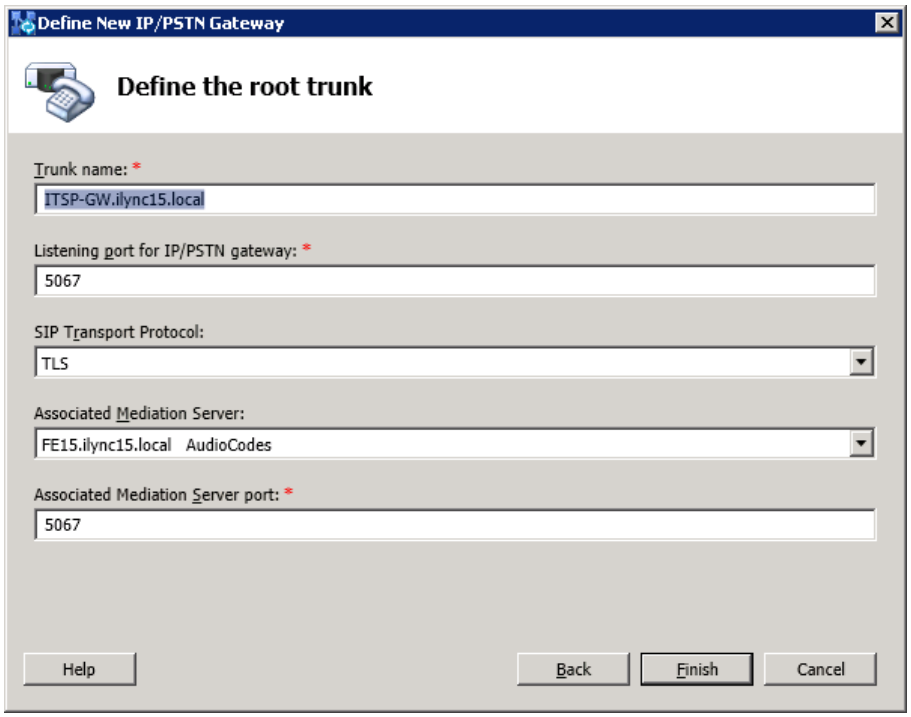


6. Define the listening mode (IPv4 or IPv6) of the IP address of your new PSTN gateway, and then click **Next**.
7. Define a *root trunk* for the PSTN gateway. A trunk is a logical connection between the Mediation Server and a gateway uniquely identified by the following combination: Mediation Server FQDN, Mediation Server listening port (TLS or TCP), gateway IP and FQDN, and gateway listening port.

Note:

- When defining a PSTN gateway in Topology Builder, you must define a root trunk to successfully add the PSTN gateway to your topology.
- The root trunk cannot be removed until the associated PSTN gateway is removed.

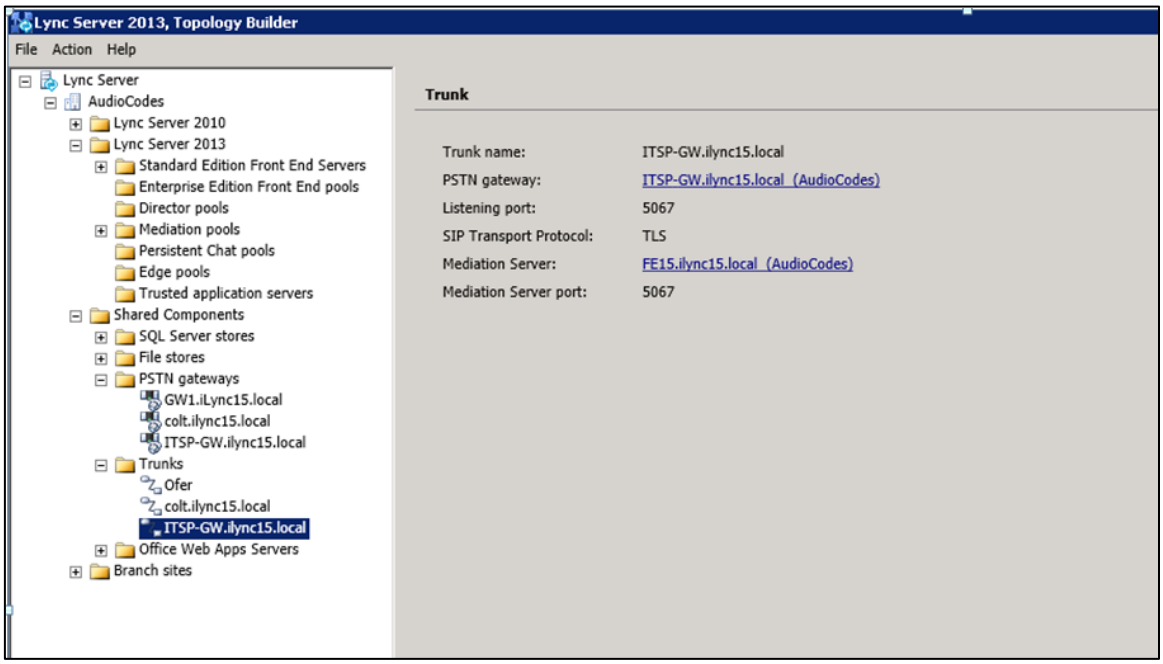
Figure 3-8: Define the Root Trunk



a. In the 'Listening Port for IP/PSTN Gateway' field, enter the listening port that the E-SBC will use for SIP messages from the Mediation Server that will be associated with the root trunk of the PSTN gateway (e.g., **5067**).
b. In the 'SIP Transport Protocol' field, select the transport type (e.g., **TLS**) that the trunk uses.
c. In the 'Associated Mediation Server' field, select the Mediation Server pool to associate with the root trunk of this PSTN gateway.
d. In the 'Associated Mediation Server Port' field, enter the listening port that the Mediation Server will use for SIP messages from the SBC (e.g., **5067**).
e. Click **Finish**.

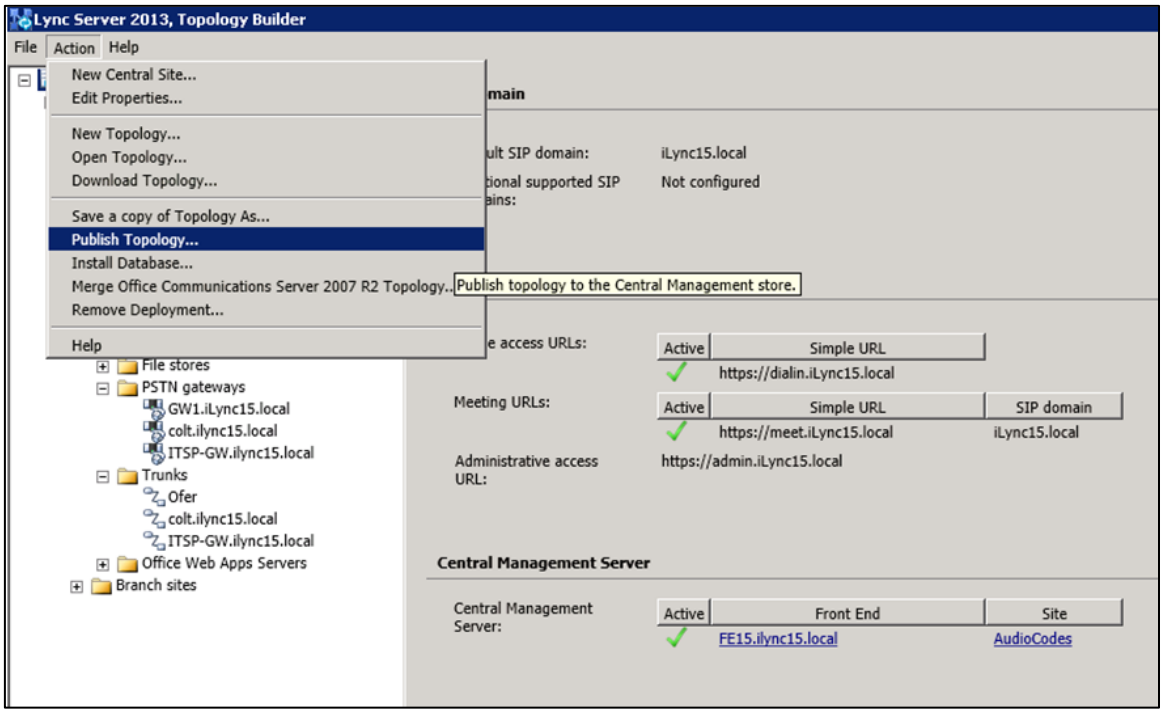
The E-SBC is added as a PSTN gateway, and a trunk is created as shown below:

Figure 3-9: E-SBC added as IP/PSTN Gateway and Trunk Created



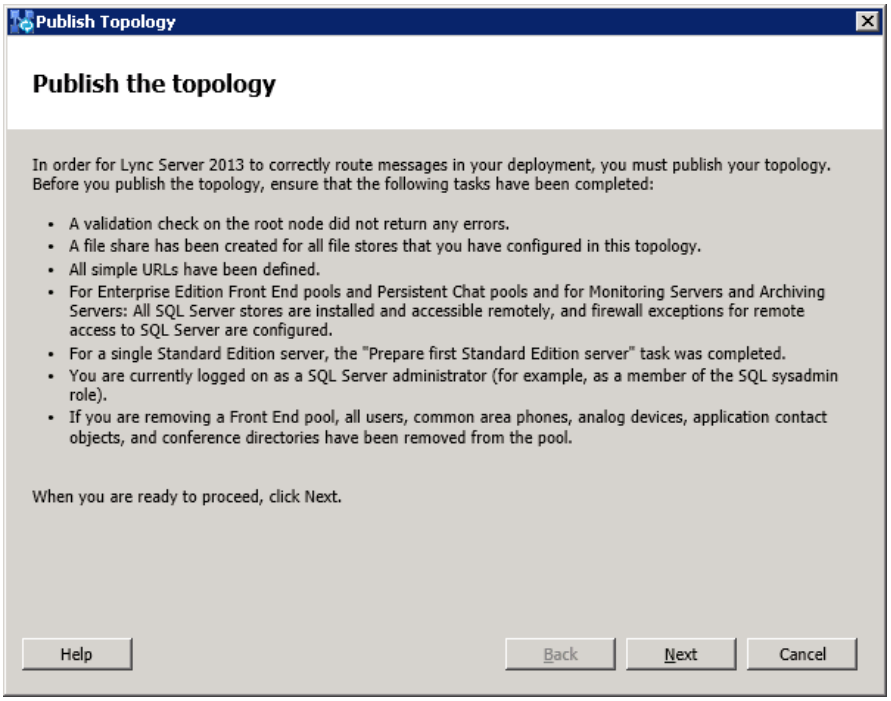
8. Publish the Topology: In the main tree, select the root node **Lync Server**, and then from the **Action** menu, choose **Publish Topology**, as shown below:

Figure 3-10: Choosing Publish Topology



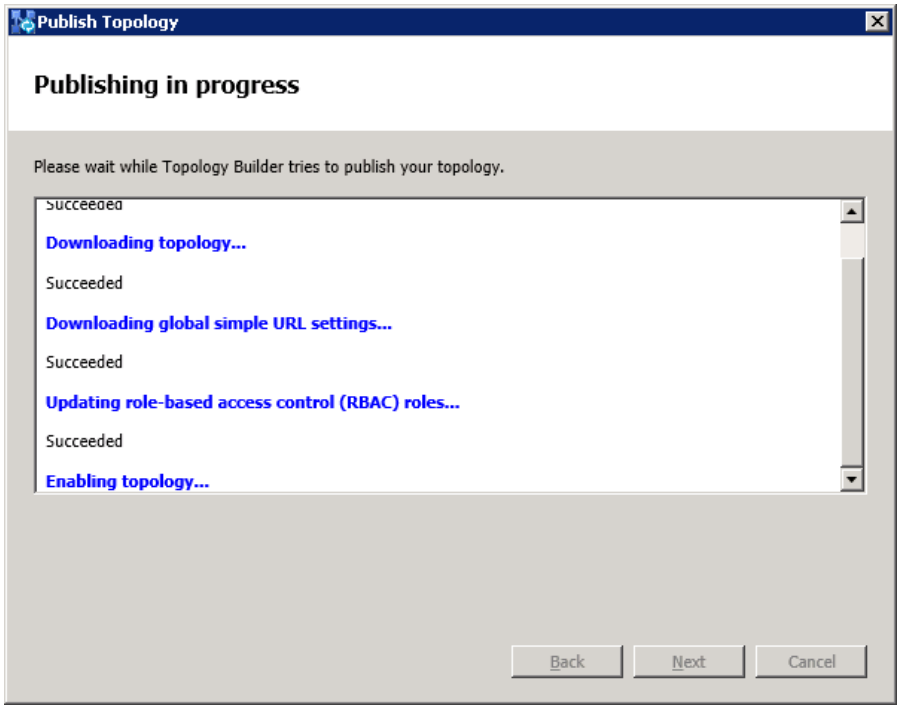
The following is displayed:

Figure 3-11: Publish the Topology



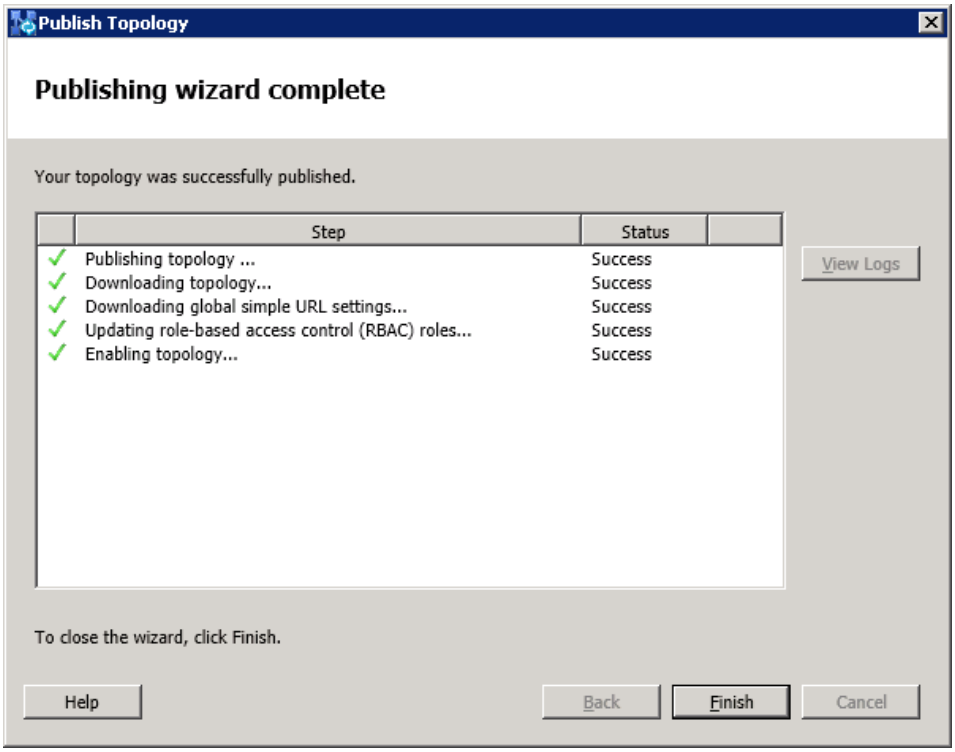
9. Click **Next**; the Topology Builder starts to publish your topology, as shown below:

Figure 3-12: Publishing in Progress



10. Wait until the publishing topology process completes successfully, as shown below:

Figure 3-13: Publishing Wizard Complete



11. Click **Finish**.

3.2 Configuring the "Route" on Lync Server 2013

This section shows how to configure a "Route" on the Lync Server 2013, and to associate it with the E-SBC PSTN gateway.

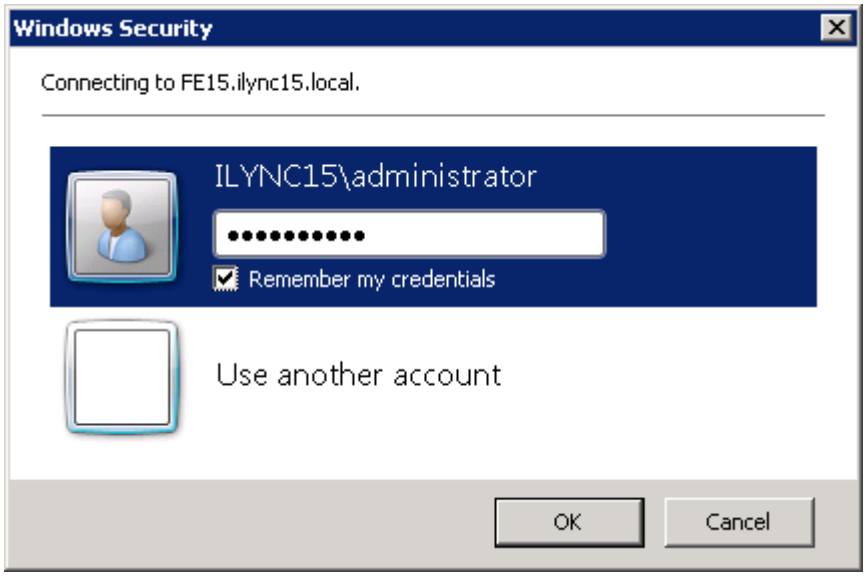
➢ **To configure the "route" on Lync Server 2013:**

1. Start the Microsoft Lync Server 2013 Control Panel (**Start** > **All Programs** > **Microsoft Lync Server 2013** > **Lync Server Control Panel**), as shown below:

Figure 3-14: Opening the Lync Server Control Panel

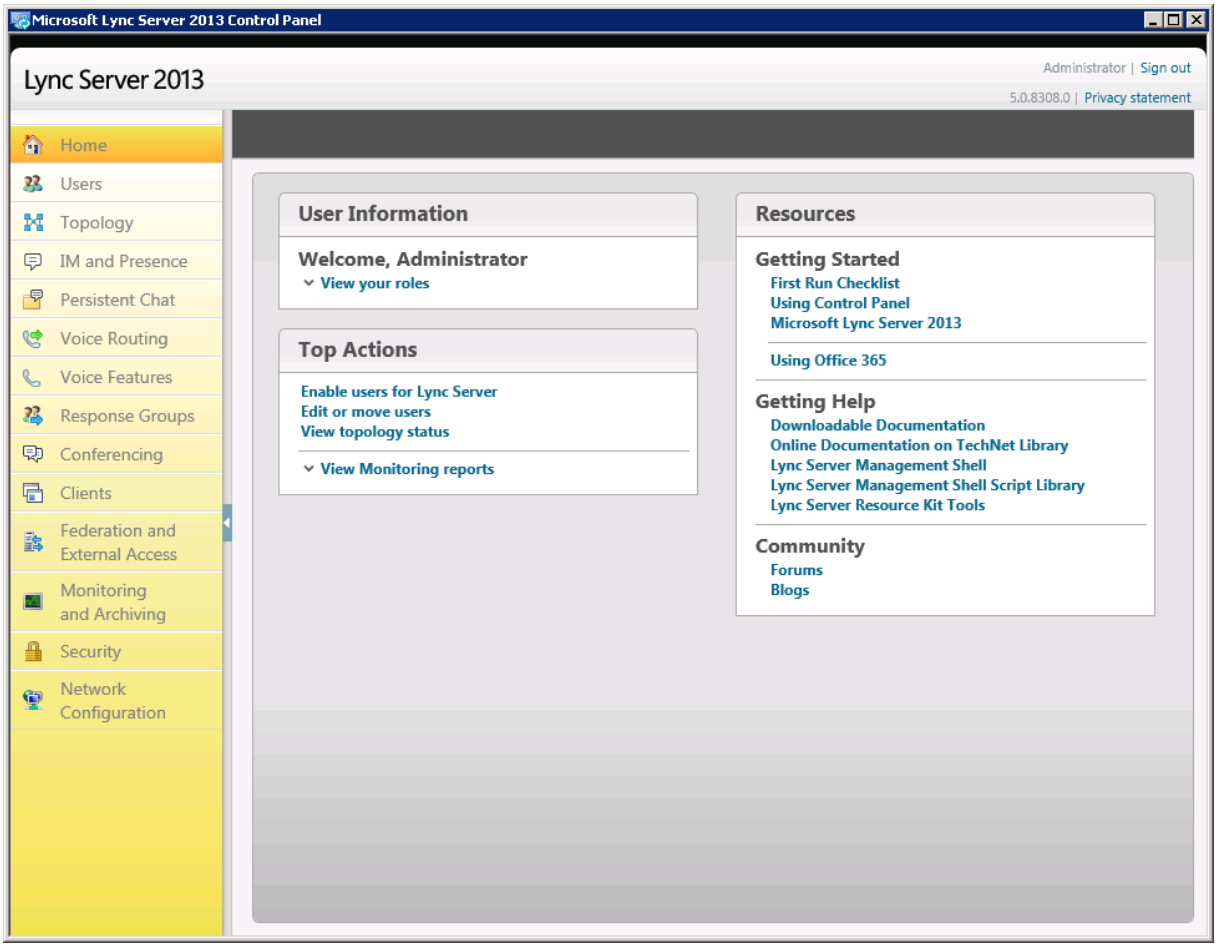
You're prompted to enter your login credentials:

Figure 3-15: Lync Server Credentials



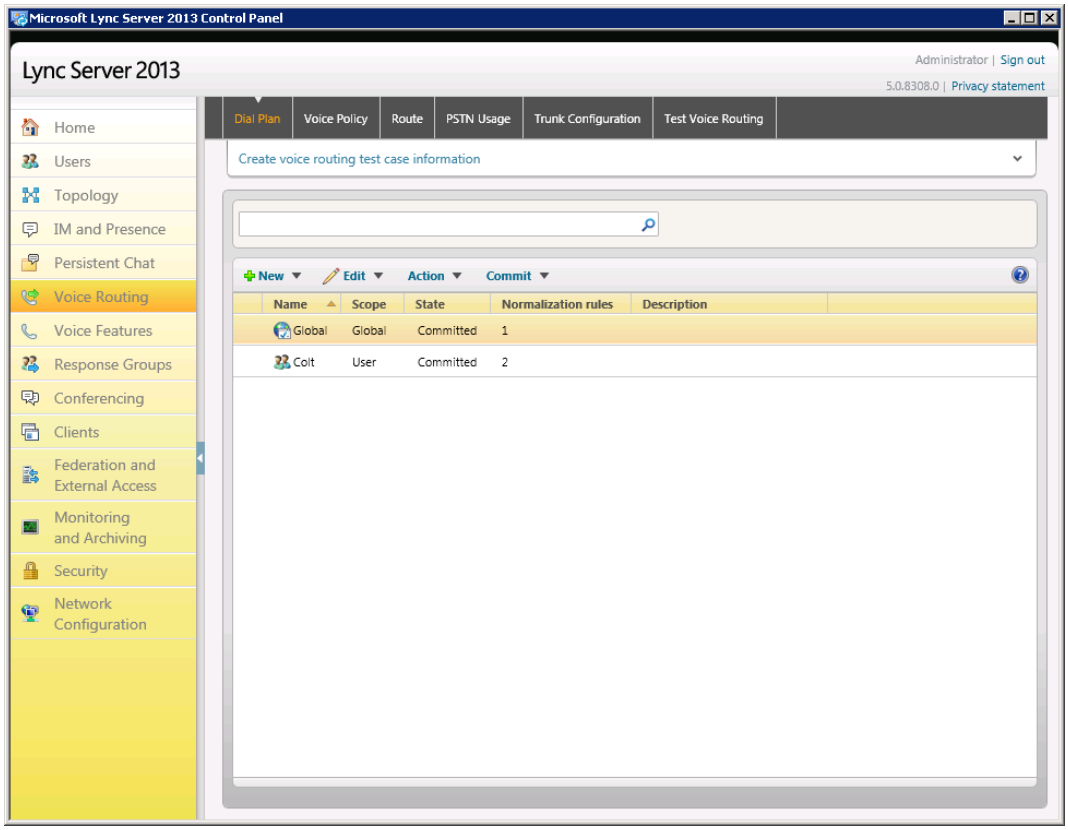
2. Enter your domain username and password, and then click **OK**; the Microsoft Lync Server 2013 Control Panel is displayed:

Figure 3-16: Microsoft Lync Server 2013 Control Panel



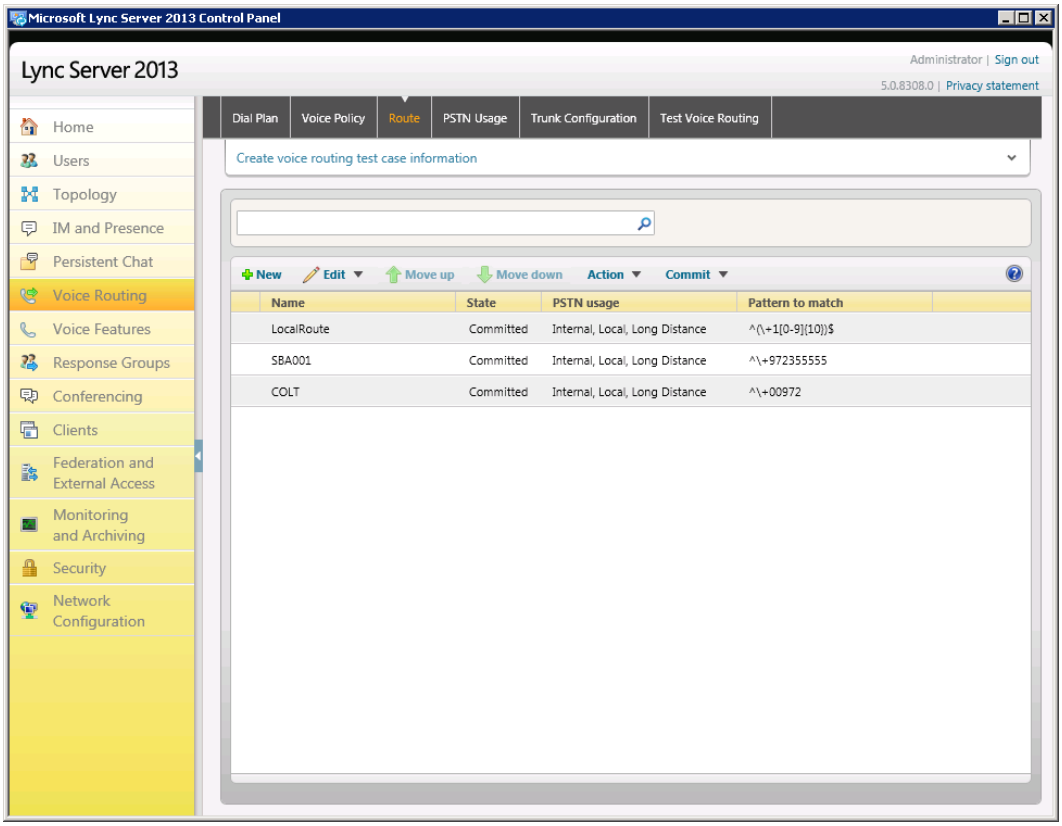
3. In the left navigation pane, select **Voice Routing**.

Figure 3-17: Voice Routing Page



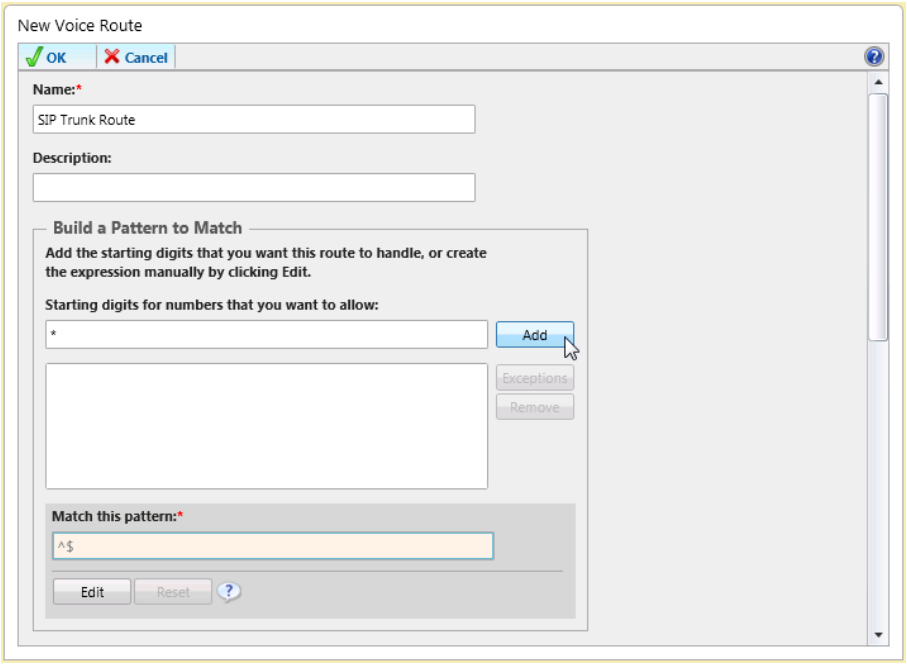
4. In the Voice Routing page, select the **Route** tab.

Figure 3-18: Route Tab



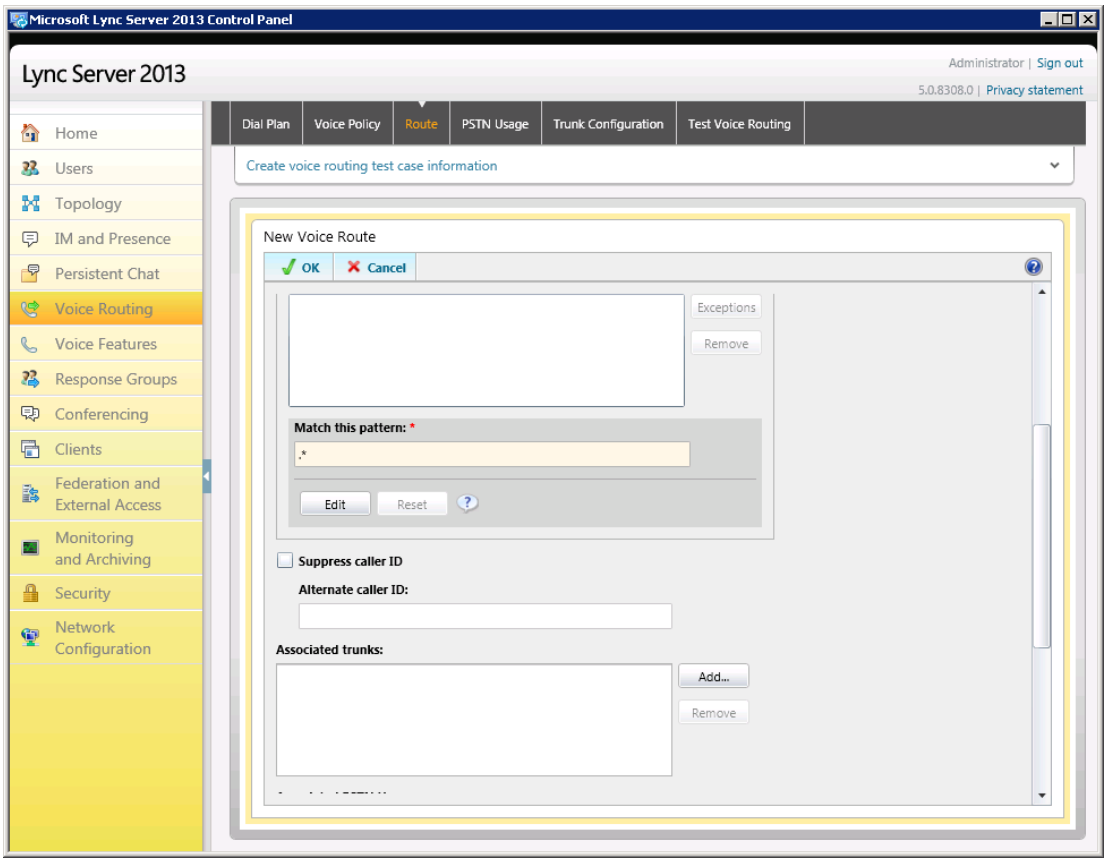
5. Click **New**; the New Voice Route page appears:

Figure 3-19: Adding New Voice Route



6. In the 'Name' field, enter a name for this route (e.g., **SIP Trunk Route**).
7. In the 'Starting digits for numbers that you want to allow' field, enter the starting digits you want this route to handle (e.g., ***** to match all numbers), and then click **Add**.

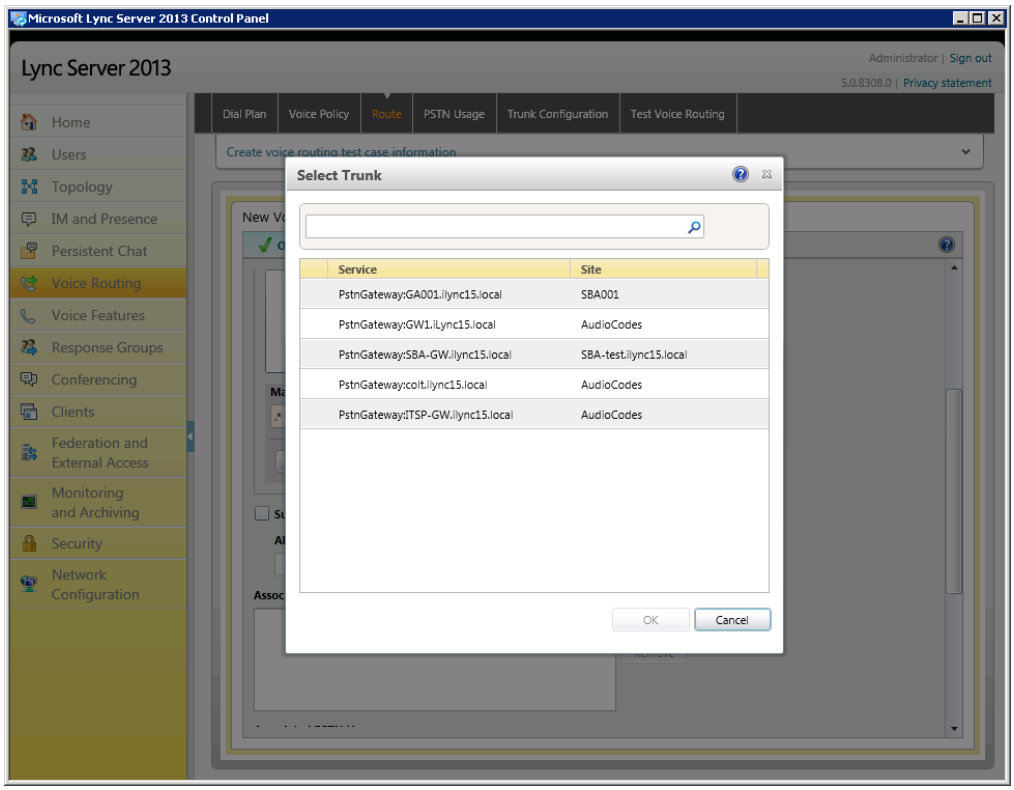
Figure 3-20: Adding New Trunk



8. Associate the route with the E-SBC Trunk that you created:

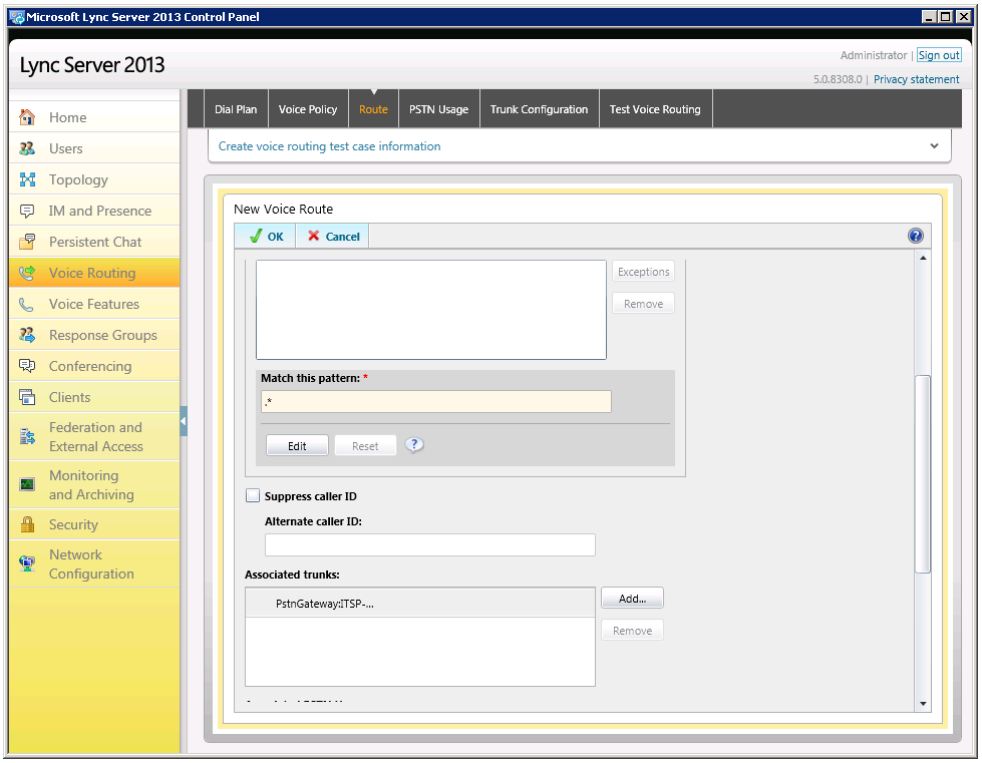
a. Under the 'Associated Trunks' group, click **Add**; a list of all the deployed gateways is displayed:

Figure 3-21: List of Deployed Trunks



b. Select the E-SBC Trunk you created, and then click **OK**; the trunk is added to the 'Associated Trunks' group list:

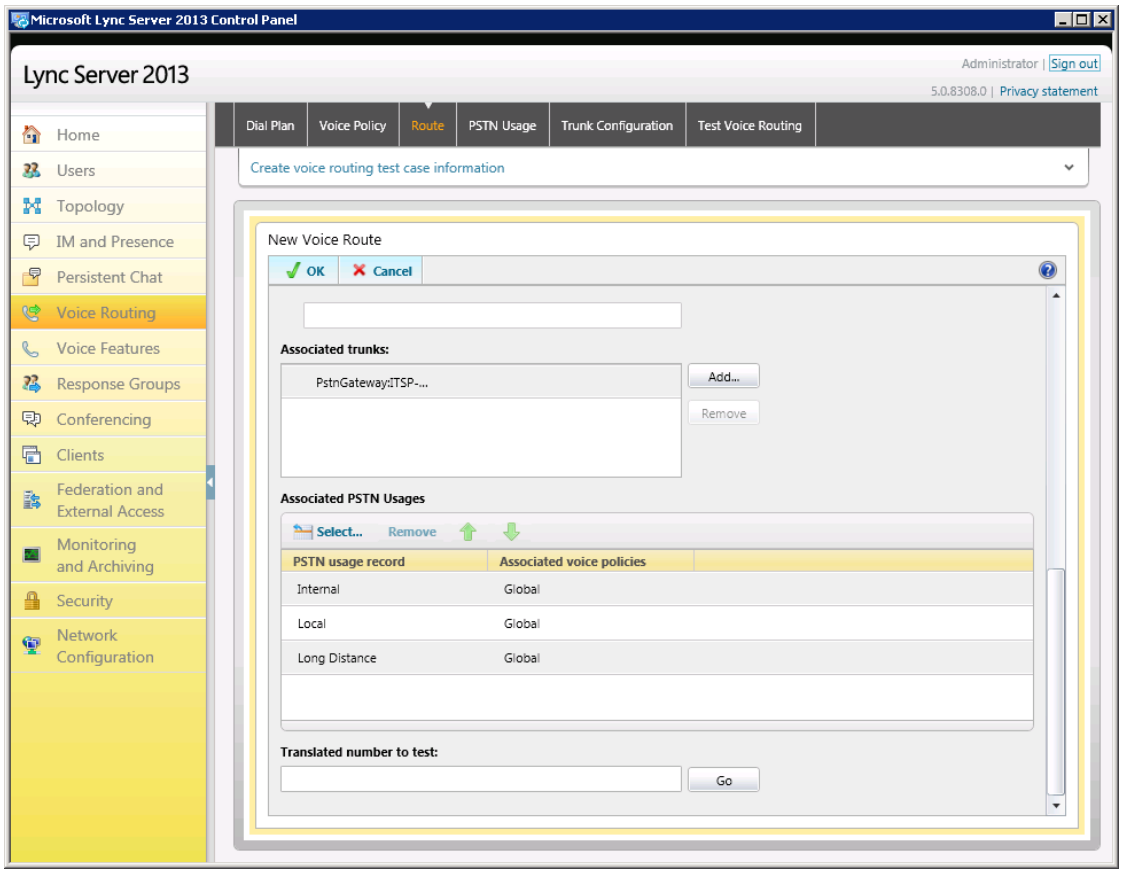
Figure 3-22: Selected E-SBC Trunk



9. Associate a PSTN Usage with this route:

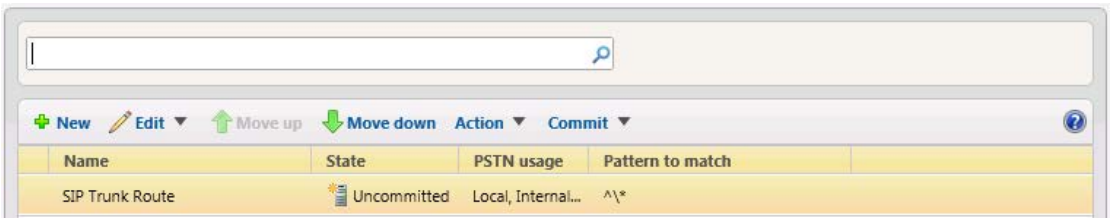
a. Under the 'Associated PSTN Usages' group, click **Select** and then add the associated PSTN Usage.

Figure 3-23: Associating PSTN Usage with the Route



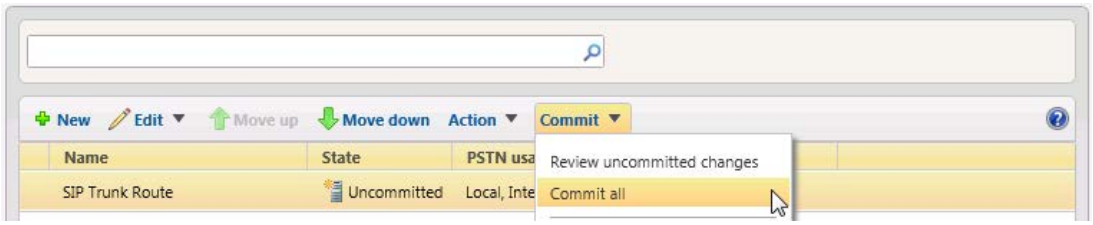
10. Click **OK** (located on the top of the New Voice Route page); the New Voice Route (Uncommitted) is displayed:

Figure 3-24: Confirmation of New Voice Route



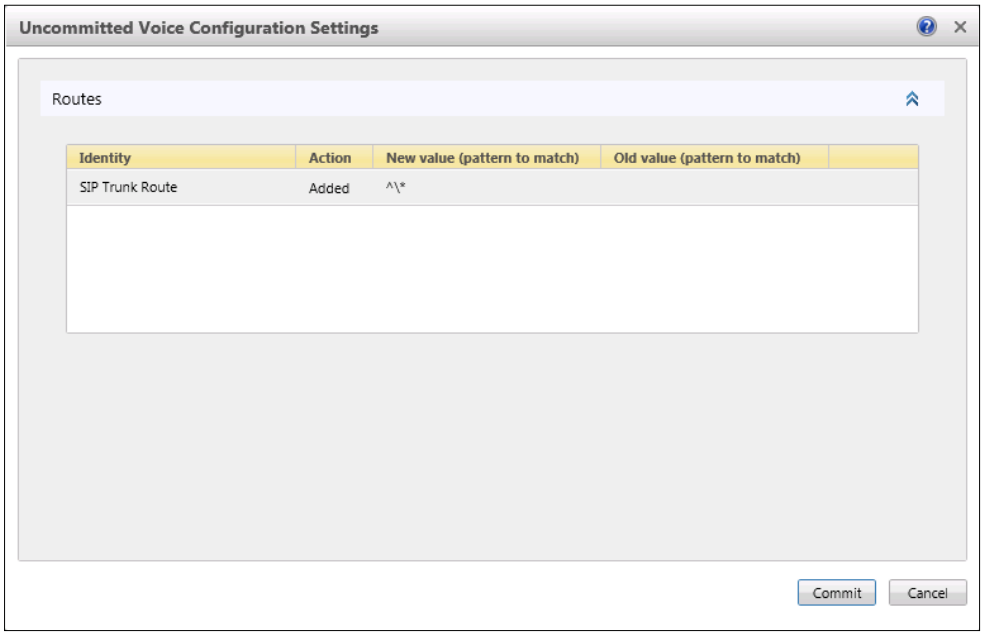
11. From the **Commit** drop-down list, choose **Commit all**, as shown below:

Figure 3-25: Committing Voice Routes



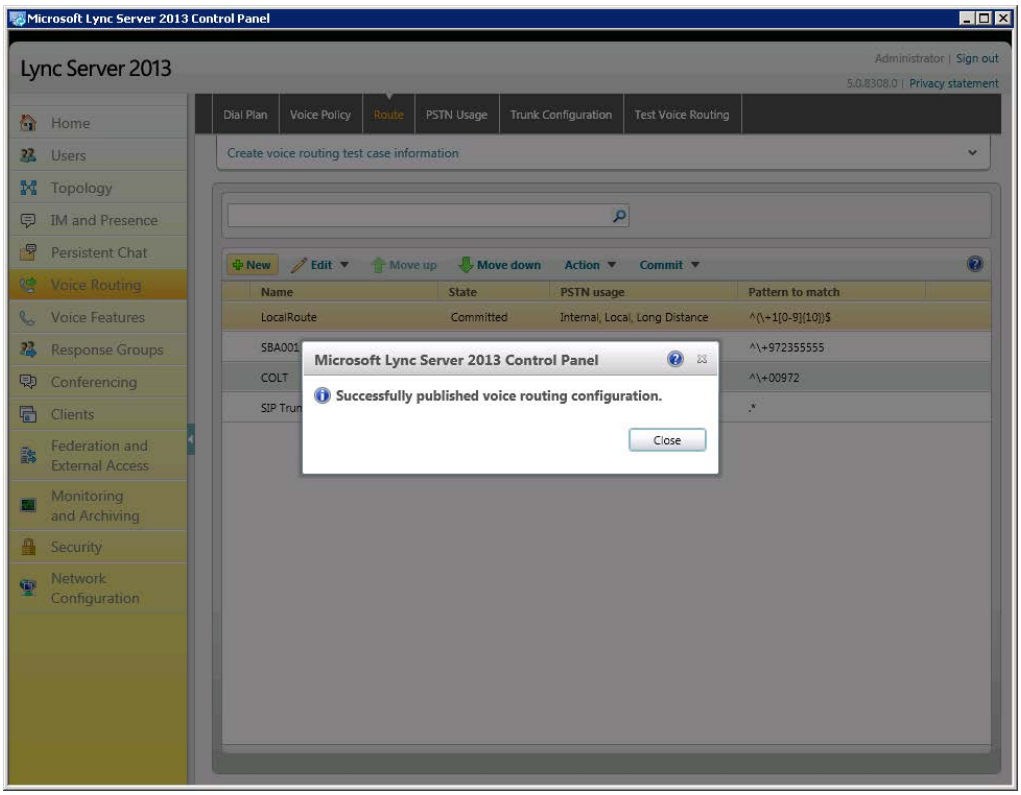
The Uncommitted Voice Configuration Settings page appears:

Figure 3-26: Uncommitted Voice Configuration Settings



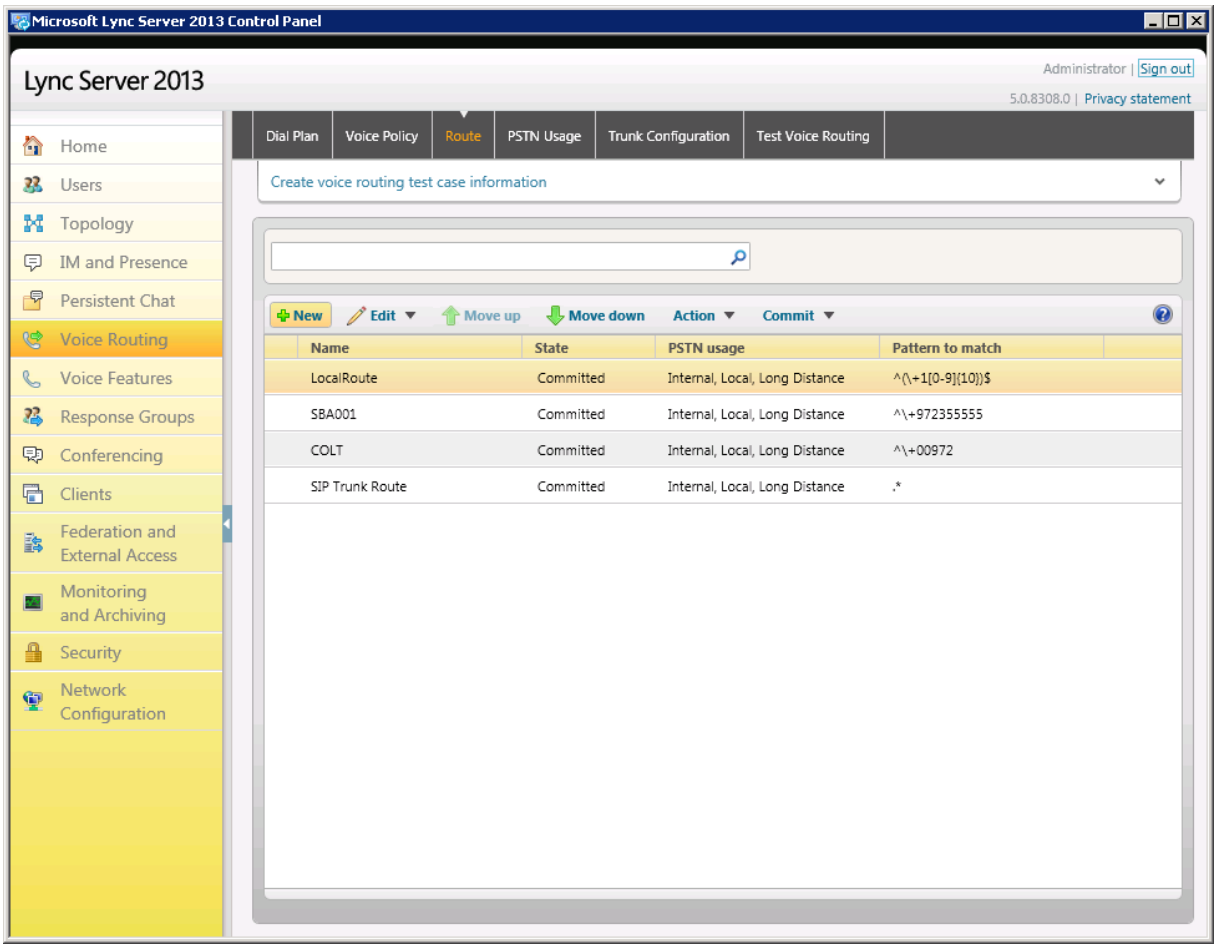
12. Click **Commit**; a message is displayed confirming a successful voice routing configuration, as shown below:

Figure 3-27: Confirmation of Successful Voice Routing Configuration



13. Click **Close**; the new committed Route is displayed in the Voice Routing page, as shown below:

Figure 3-28: Voice Routing Screen Displaying Committed Routes

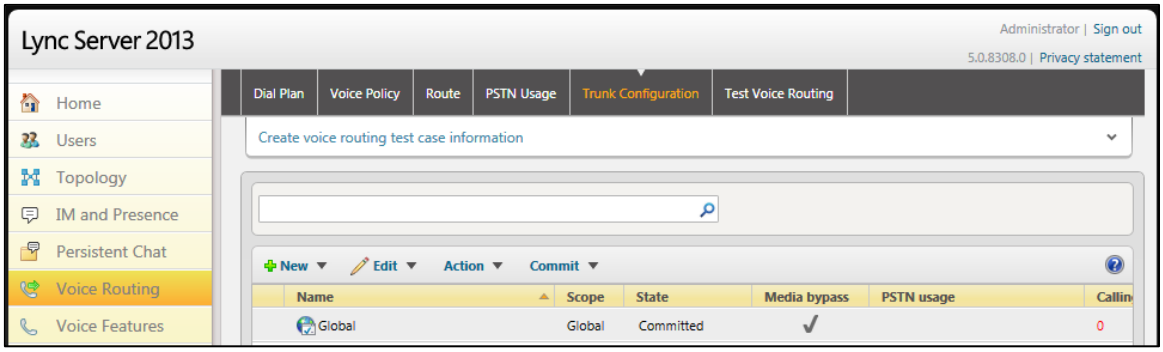


14. For ITSPs that implement a call identifier, continue with the steps below.

Note: The SIP History-Info header provides a method to verify the identity (ID) of the call forwarder (i.e., the Lync user number). This ID is required by Verizon SIP Trunk in the Diversion header. Using the IP Profile settings assigned to the Verizon SIP Trunk (see Section 4.13 on page 82), the device adds this ID to an added Diversion header in the sent INVITE message while removing the History-Info header on the Verizon side of the call session.

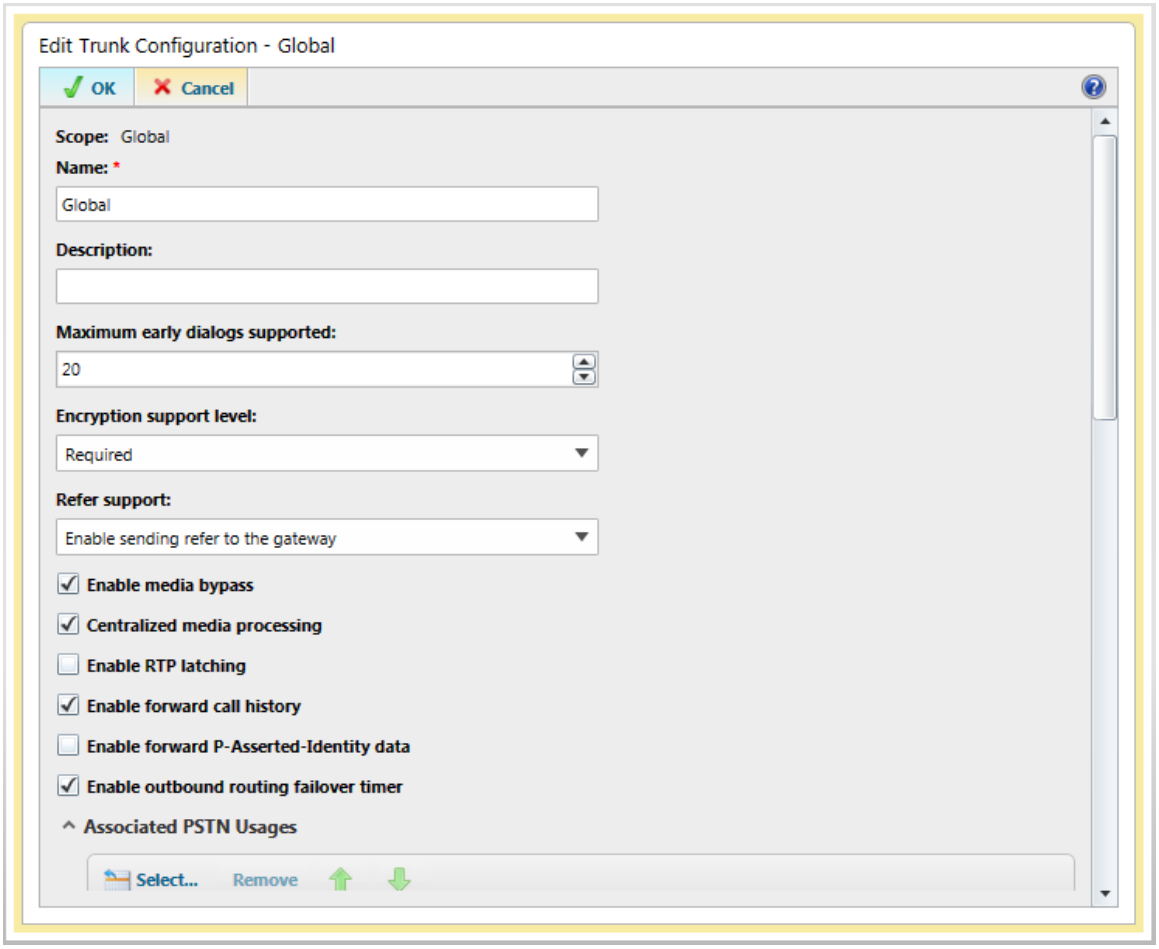
a. In the Voice Routing page, select the **Trunk Configuration** tab. Note that you can add and modify trunk configuration by site or by pool.

Figure 3-29: Voice Routing Screen – Trunk Configuration Tab



b. Click **Edit**; the Edit Trunk Configuration page appears:

Figure 3-30: Edit Trunk Configuration



c. Select the **Enable forward call history** option, and then click **OK**.

d. Repeat Steps 11 through 13 to commit your settings.

Reader's Notes

4 Configuring AudioCodes E-SBC

This section shows how to configure AudioCodes E-SBC for interworking between Microsoft Lync Server 2013 and the Verizon SIP Trunk. These configuration procedures are based on the interoperability test topology described in Section 2.5 on page 10, and includes the following main areas:

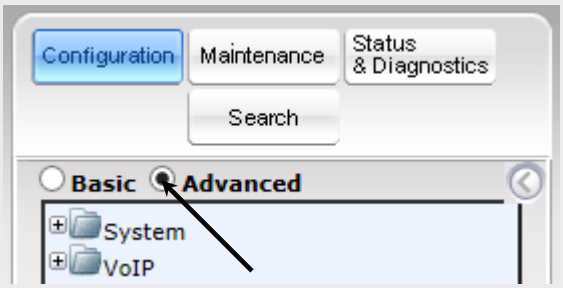
- E-SBC WAN interface - Verizon SIP Trunking environment
- E-SBC LAN interface - Lync Server 2013 environment

Configuration is performed using the E-SBC's embedded Web server (referred to in this document as *Web interface*).

Notes:

- To implement Microsoft Lync and Verizon SIP Trunk based on the configuration described in this section, AudioCodes E-SBC must be installed with a Software License Key that includes the following software features:
  - √ **Microsoft**
  - √ **SBC**
  - √ **Security**
  - √ **DSP**
  - √ **RTP**
  - √ **SIP**

  For more information about the Software License Key, contact your AudioCodes sales representative.
- The scope of this document does *not* cover security aspects for connecting the SIP Trunk to the Microsoft Lync environment. Security measures should be implemented in accordance with your organization's security policies. For basic security guidelines, refer to the *Recommended Security Guidelines* document.
- Before you begin configuring the E-SBC, ensure that the E-SBC's Web interface navigation tree is in **Advanced** display mode. To do this, select the **Advanced** option, as shown below:



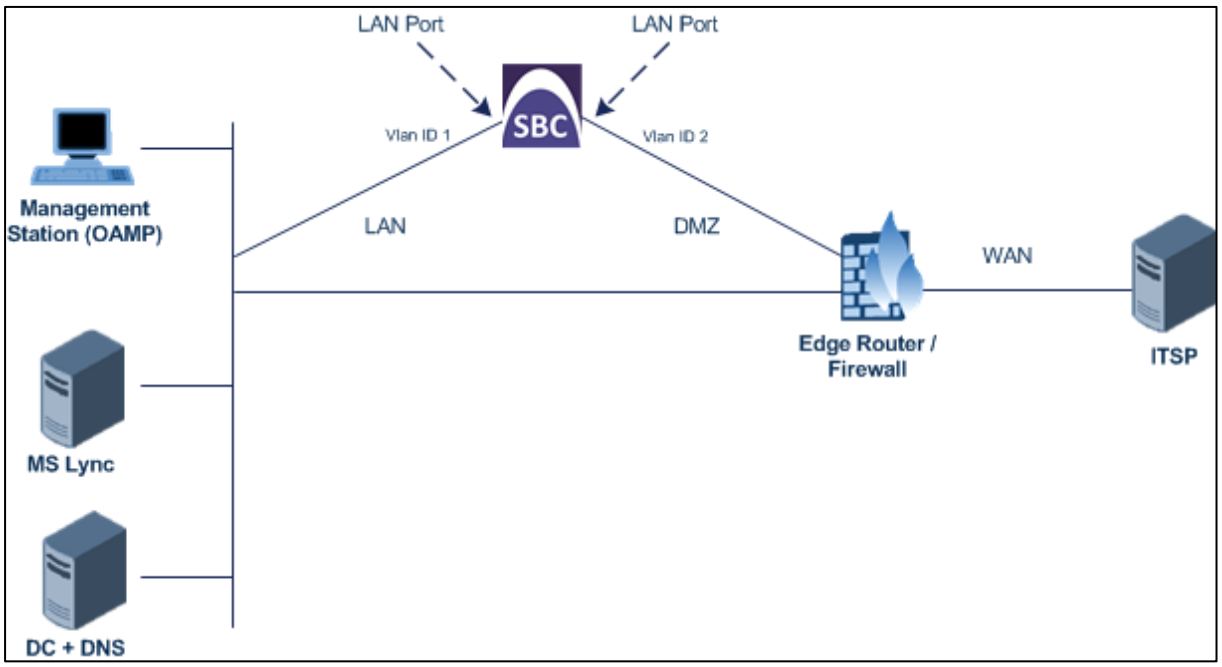
Note that when the E-SBC is reset, the navigation tree reverts to **Basic** menu display.

4.1 Step 1: Configure IP Network Interfaces

This step shows how to configure the E-SBC's IP network interfaces. There are several methods of deploying the E-SBC; this interoperability test topology employs the following deployment method:

- E-SBC interfaces with the following IP entities:
  - Lync servers, located on the LAN
  - Verizon SIP Trunk, located on the WAN
- E-SBC connects to the WAN through a DMZ network
- Physical connection: The type of physical connection to the LAN depends on the method used to connect to the enterprise's network. In the interoperability test topology, E-SBC connects to the LAN and WAN using dedicated LAN ports (i.e., two ports and two network cables are used).
- E-SBC also uses two logical network interfaces:
  - LAN (VLAN ID 1)
  - WAN (VLAN ID 2)

Figure 4-1: Network Interfaces in Interoperability Test Topology



4.1.1 Step 1a: Configure VLANs

This step shows how to define VLANs for each of the following interfaces:

- LAN VoIP (assigned the name "Voice")
- WAN VoIP (assigned the name "Public")

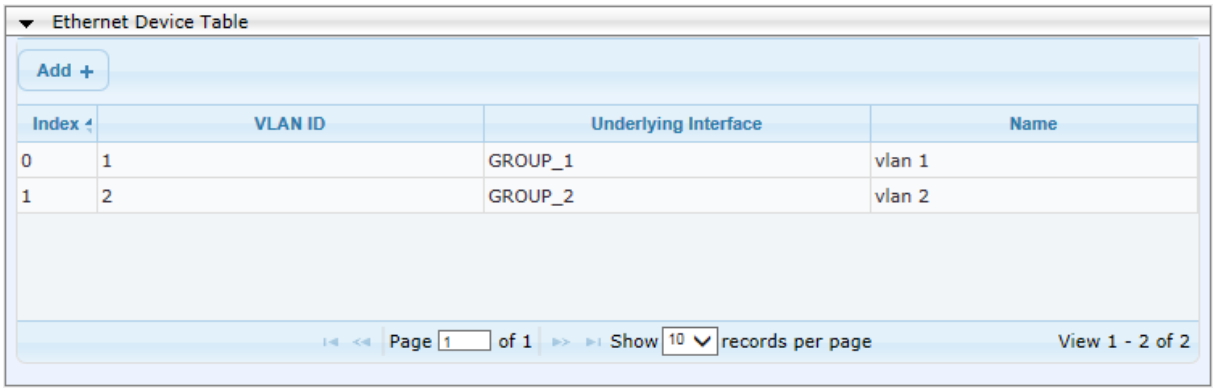
➢ **To configure the VLANs:**

1. Open the Ethernet Device Table page (**Configuration** tab > **VoIP** menu > **Network** > **Ethernet Device Table**).

   There will be one existing row for VLAN ID 1 and underlying interface GROUP_1.
2. Add another VLAN ID 2 for the WAN side as follows:

Parameter	Value
Index	**1**
VLAN ID	**2**
Underlying Interface	**GROUP_2** (Ethernet port group)
Name	**vlan 2**

Figure 4-2: Configured VLAN IDs in Ethernet Device Table



4.1.2 Step 1b: Configure Network Interfaces

This step shows how to configure the IP network interfaces for each of the following interfaces:

- LAN VoIP (assigned the name "Voice")
- WAN VoIP (assigned the name "Public")

➢ **To configure the IP network interfaces:**

1. Open the IP Interfaces Table page (**Configuration** tab > **VoIP** menu > **Network** > **IP Interfaces Table**).
2. Modify the existing LAN network interface:
   a. Select the 'Index' radio button of the **OAMP + Media + Control** table row, and then click **Edit**.
   b. Configure the interface as follows:

Parameter	Value
IP Address	**10.133.4.43** (IP address of E-SBC)
Prefix Length	**16** (subnet mask in bits for 255.255.0.0)
Gateway	**10.133.4.1**
Interface Name	**Voice** (arbitrary descriptive name)
Primary DNS Server IP Address	**10.133.4.40**
Underlying Device	**vlan 1**

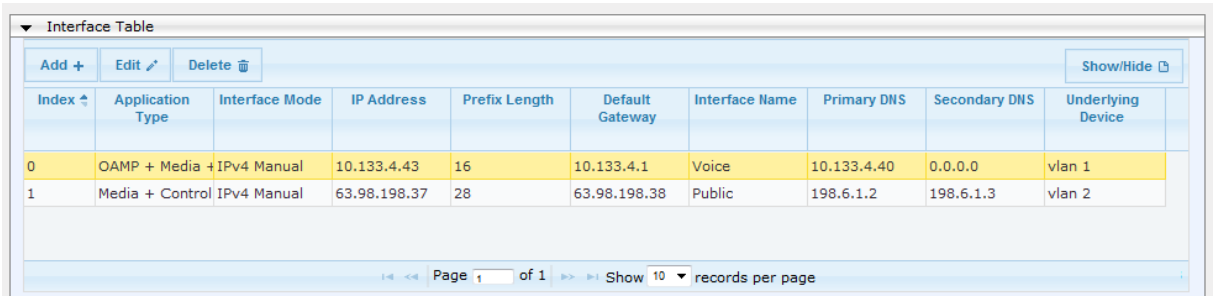
3. Add a network interface for the WAN side:
   a. Enter **1**, and then click **Add Index**.
   b. Configure the interface as follows:

Parameter	Value
Application Type	**Media + Control**
IP Address	**63.98.198.37** (WAN IP address)
Prefix Length	**28** (for 255.255.255.240)
Gateway	**63.98.198.38** (router's IP address)
Interface Name	**Public**
Primary DNS Server IP Address	**192.6.1.2**
Secondary DNS Server IP Address	**192.6.1.3**
Underlying Device	**vlan 2**

4. Click **Apply**, and then **Done**.

The configured IP network interfaces are shown in the figure below:

Figure 4-3: Configured Network Interfaces in IP Interfaces Table



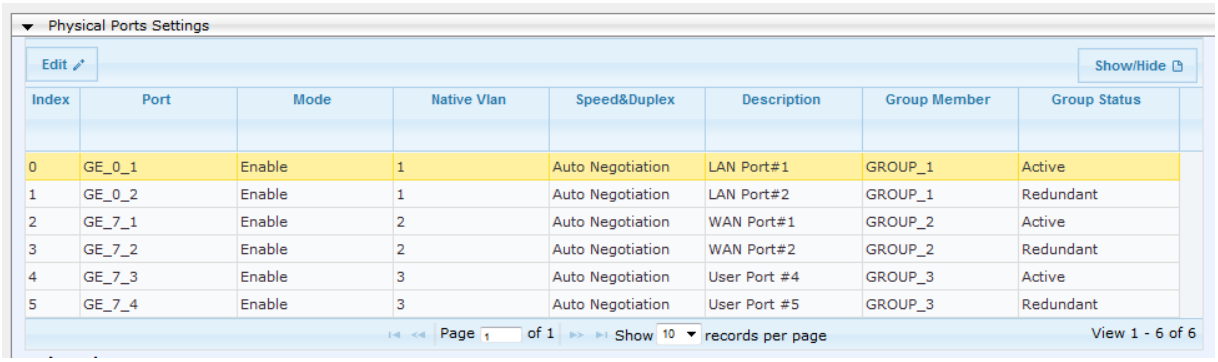
4.1.3 Step 1c: Configure the Native VLAN ID

This step shows how to configure the Native VLAN ID for the LAN and WAN interfaces.

➢ **To configure the Native VLAN ID for the IP network interfaces:**

1. Open the Physical Ports Settings page (**Configuration** tab> **VoIP** menu > **Network** > **Physical Ports Table**).
2. For the **GROUP_1** member ports, set the 'Native Vlan' field to **1**. This VLAN was assigned to network interface "Voice".
3. For the **GROUP_2** member ports, set the 'Native Vlan' field to **2**. This VLAN was assigned to network interface "Public".

Figure 4-4: Configured Port Native VLAN



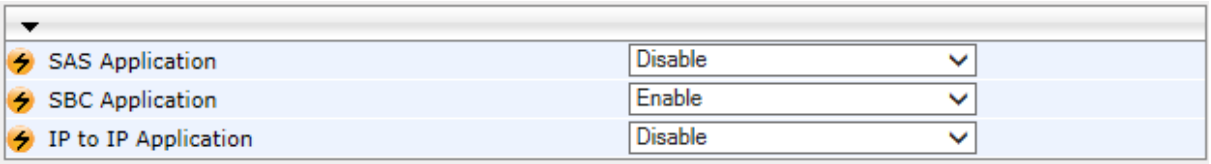
4.2 Step 2: Enable the SBC Application

This step describes how to enable the SBC application.

➢ **To enable the SBC application:**

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

Figure 4-5: Enabling SBC Application



2. From the 'SBC Application' drop-down list, select **Enable**.
3. Click **Submit**.
4. Reset the E-SBC with a burn to flash for this setting to take effect (see Section 4.15 on page 98).

4.3 Step 3: Configure SRDs

This step describes how to configure Signaling Routing Domains (SRDs). The SRD represents a logical VoIP network. Each logical or physical connection requires an SRD, for example, if the E-SBC interfaces with both the LAN and WAN, a different SRD is required for each.

The SRD comprises:

- Media Realm: defines a UDP port range for RTP/SRTP (media) traffic on a specific logical IP network interface of the E-SBC.
- SIP Interface: defines a listening port and type (UDP, TCP, or TLS) for SIP signaling traffic on a specific logical IP network interface of the E-SBC.

4.3.1 Step 3a: Configure Media Realms

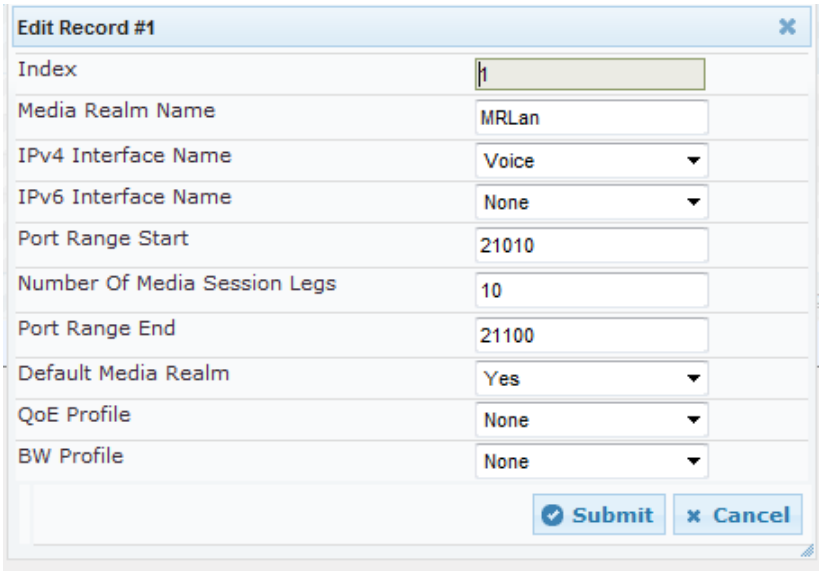
This step shows how to configure Media Realms. The simplest configuration is to create two Media Realms - one for internal (LAN) traffic and one for external (WAN) traffic.

➢ **To configure Media Realms:**

1. Open the Media Realm Table page (**Configuration** tab > **VoIP** menu > **VoIP Network** > **Media Realm Table**).
2. Modify the existing Media Realm for LAN traffic:

Parameter	Value
Index	**1**
Media Realm Name	**MRLan** (descriptive name)
IPv4 Interface Name	**Voice**
Port Range Start	**21010** (represents lowest UDP port number used for media on LAN.)
Number of Media Session Legs	**10** (media sessions assigned with port range)

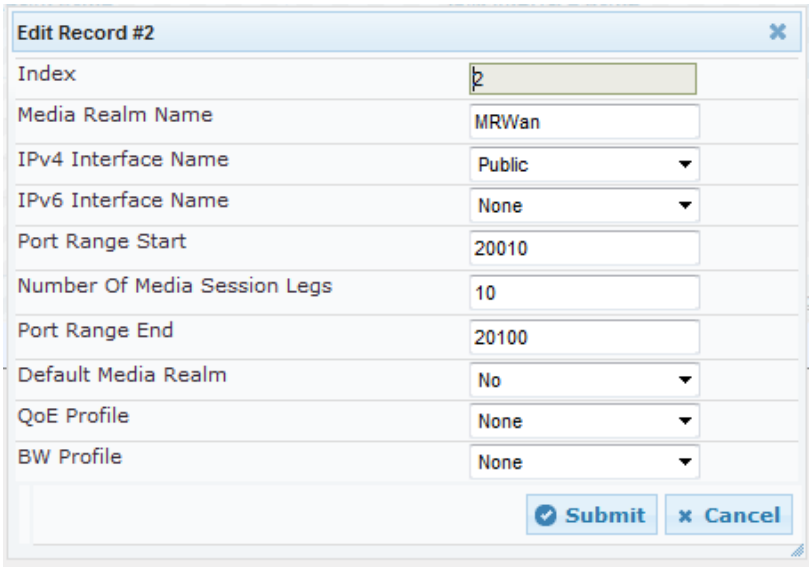
Figure 4-6: Configuring Media Realm for LAN



3. Configure a Media Realm for WAN traffic:

Parameter	Value
Index	**2**
Media Realm Name	**MRWan** (arbitrary name)
IPv4 Interface Name	**Public**
Port Range Start	**20010** (represents the lowest UDP port number used for media on WAN. This was a requirement of certification testing)
Number of Media Session Legs	**10** (media sessions assigned with port range)

Figure 4-7: Configuring Media Realm for WAN



The configured Media Realms are shown in the figure below:

Figure 4-8: Configured Media Realms in Media Realm Table

4.3.2 Step 3b: Configure SRDs

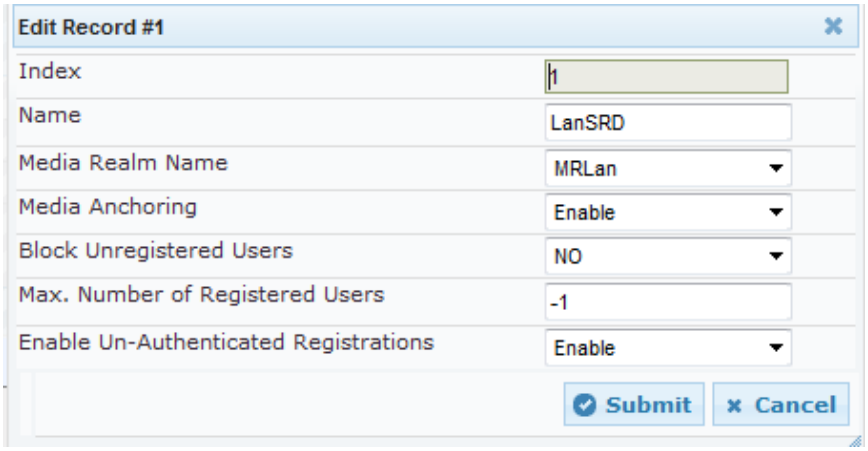
This step shows how to configure the SRDs.

➢ **To configure SRDs:**

1. Open the SRD Settings page (**Configuration** tab > **VoIP** menu > **VoIP Network** > **SRD Table**).
2. Configure an SRD for the E-SBC's internal interface (toward Lync Server 2013):

Parameter	Value
SRD Index	**1**
SRD Name	**LanSRD** (descriptive name for SRD)
Media Realm	**MRLan** (associates SRD with Media Realm)

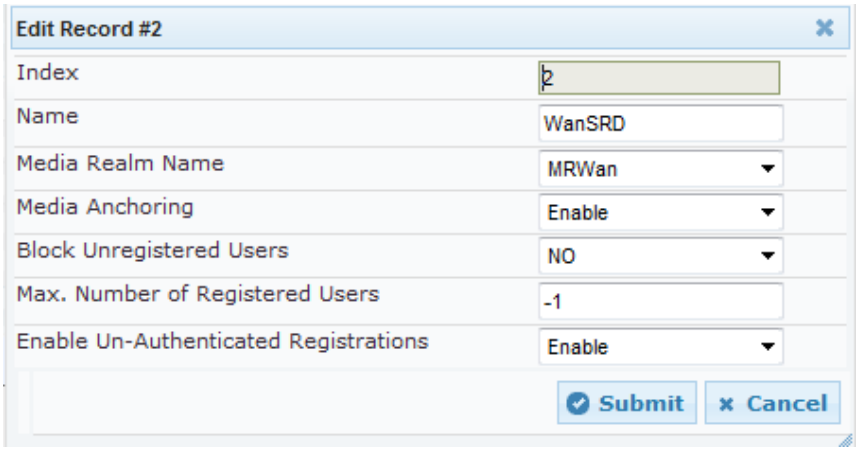
Figure 4-9: Configuring LAN SRD



3. Configure an SRD for the E-SBC's external interface (toward the Verizon SIP Trunk):

Parameter	Value
SRD Index	**2**
SRD Name	**WanSRD**
Media Realm	**MRWan**

Figure 4-10: Configuring WAN SRD



4.3.3 Step 3c: Configure SIP Signaling Interfaces

This step shows how to configure SIP Interfaces. For the interoperability test topology, an internal and external SIP Interface must be configured for the E-SBC.

➢ **To configure SIP Interfaces:**

1. Open the SIP Interface Table page (**Configuration** tab > **VoIP** menu > **VoIP Network** > **SIP Interface Table**).
2. Configure a SIP interface for the LAN:

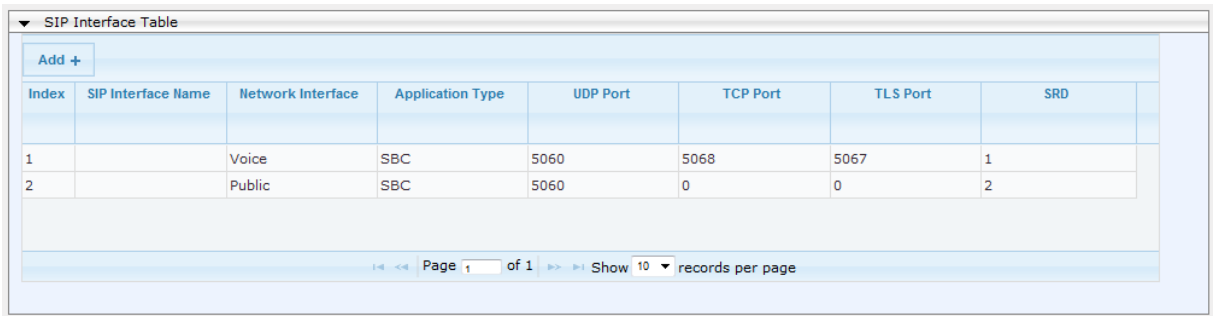
Parameter	Value
Index	**1**
Interface Name	Arbitrary descriptive name
Network Interface	**Voice**
Application Type	**SBC**
TLS Port	**5067 (recommended)**
TCP Port	**5068 (if TCP is used)**
UDP Port	**5060**
SRD	**1**

3. Configure a SIP interface for the WAN:

Parameter	Value
Index	**2**
Interface Name	Arbitrary descriptive name
Network Interface	**Public**
Application Type	**SBC**
UDP Port	**5060**
TCP and TLS	**0**
SRD	**2**

The configured SIP Interfaces are shown in the figure below:

Figure 4-11: Configured SIP Interfaces in SIP Interface Table



4.4 Step 4: Configure Proxy Sets

This step shows how to configure Proxy Sets. The Proxy Set defines the destination address (IP address or FQDN) of the IP entity server. Proxy Sets can also be used to configure load balancing between multiple servers.

For the interoperability test topology, two Proxy Sets must be configured for the following IP entities:

- Microsoft Lync Server 2013
- Verizon SIP Trunk

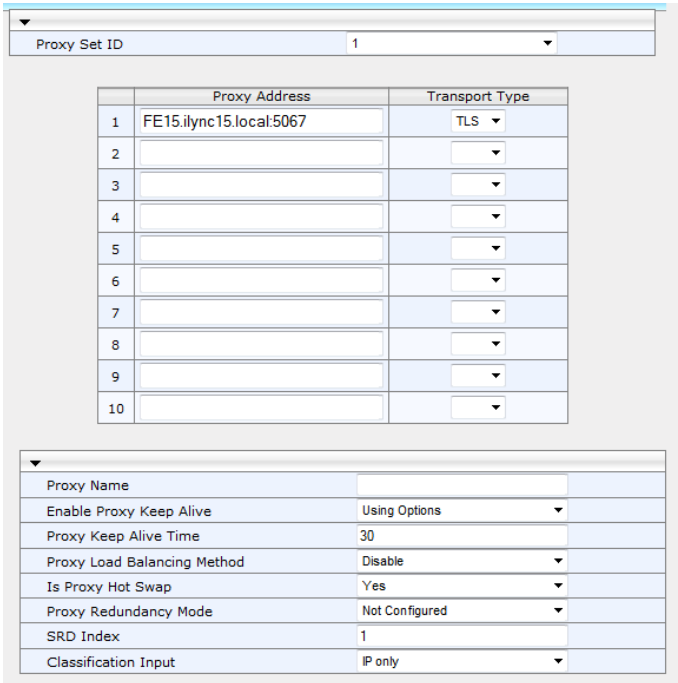
These Proxy Sets will later be associated with IP Groups.

➢ **To configure Proxy Sets:**

1. Open the Proxy Sets Table page (**Configuration** tab > **VoIP** menu > **VoIP Network** > **Proxy Sets Table**).
2. Configure a Proxy Set for Lync Server 2013:

Parameter	Value
Proxy Set ID	**1**
Proxy Address	**FE15.ilync15.local:5067** (Lync Server 2013 IP address / FQDN and destination port)
Transport Type	**TLS**
Proxy Name	Arbitrary descriptive name
Enable Proxy Keep Alive	**Using Options**
Is Proxy Hot Swap	**Yes**
SRD Index	**1**

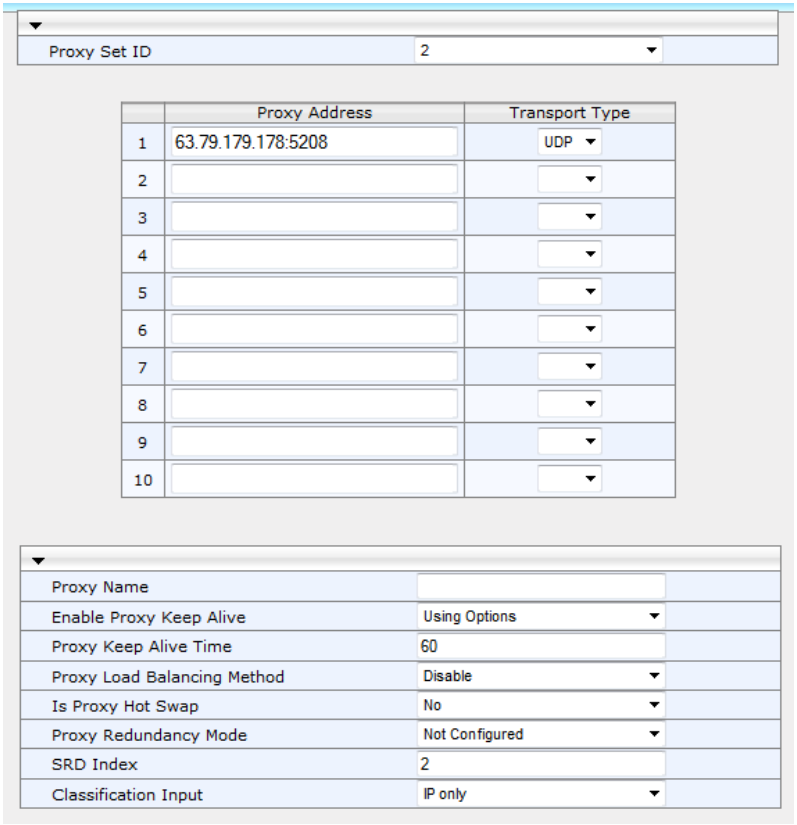
Figure 4-12: Configuring Proxy Set for Microsoft Lync Server 2013



3. Configure a Proxy Set for the Verizon SIP Trunk:

Parameter	Value
Proxy Set ID	**2**
Proxy Address	**63.79.179.178:5208** (Verizon IP address / FQDN and destination port)
Transport Type	**UDP**
Proxy Name	Arbitrary descriptive name
Enable Proxy Keep Alive	**Using Options**
SRD Index	**2** (enables classification by Proxy Set for SRD of IP Group belonging to Verizon SIP Trunk)

Figure 4-13: Configuring Proxy Set for Verizon SIP Trunk



4. Reset the E-SBC with a burn to flash for these settings to take effect (see Section 4.15 on page 98).

4.5 Step 5: Configure IP Groups

This step shows how to configure IP Groups. The IP Group represents an IP entity on the network with which the E-SBC communicates. This can be a server (e.g., IP PBX or ITSP) or it can be a group of users (e.g., LAN IP phones). For servers, the IP Group is typically used to define the server's IP address by associating it with a Proxy Set. A typical deployment consists of multiple IP Groups associated with the same SRD. For example, you can have two LAN IP PBXs sharing the same SRD, and two ITSPs / SIP Trunks sharing the same SRD. After IP Groups are configured, they're used to configure IP-to-IP routing rules for denoting source and destination of the call.

In this interoperability test topology, IP Groups must be configured for the following IP entities:

- Lync Server 2013 (Mediation Server) located on the LAN
- Verizon SIP Trunk located on the WAN

➢ **To configure IP Groups:**

1. Open the IP Group Table page (**Configuration** tab > **VoIP** menu > **VoIP Network** > **IP Group Table**).
2. Configure an IP Group for the Lync Server 2013 Mediation Server:

Parameter	Value
Index	**1**
Type	**Server**
Description	**Microsoft Lync** (arbitrary descriptive name)
Proxy Set ID	**1**
SIP Group Name	**63.98.198.37** (according to ITSP requirement)
SRD	**1**
Media Realm Name	**MRLan**
IP Profile ID	**1**

3. Configure an IP Group for the Verizon SIP Trunk:

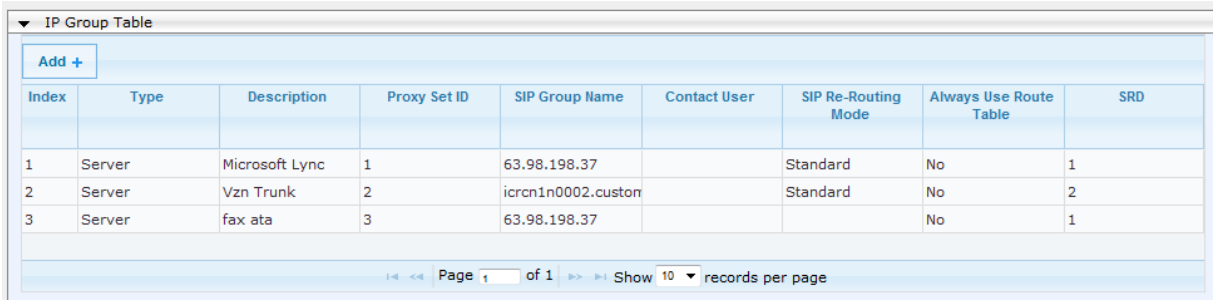
Parameter	Value
Index	**2**
Type	**Server**
Description	**Vzn Trunk** (arbitrary descriptive name)
Proxy Set ID	**2**
SIP Group Name	**icrcn1n0002.customer08.tsengr.com** (according to ITSP requirement)
SRD	**2**
Media Realm Name	**MRWan**
IP Profile ID	**2**

4. Configure an IP Group for the Fax supporting ATA:

Parameter	Value
Index	**3**
Type	**Server**
Description	**fax ata** (arbitrary descriptive name)
Proxy Set ID	**3**
SIP Group Name	**63.97.198.37** (according to ITSP requirement)
SRD	**1**
Media Realm Name	**MRLan**
IP Profile ID	**3**

The configured IP Groups are shown in the figure below:

Figure 4-14: Configured IP Groups in IP Group Table



4.6 Step 6: Configure IP Profiles

This step shows how to configure IP Profiles. The IP Profile defines a set of call capabilities relating to signaling (e.g., SIP message terminations such as REFER) and media (e.g., coder and transcoding method).

In this interoperability test topology, IP Profiles must be configured for the following IP entities:

- Microsoft Lync Server 2013 - to operate in secure mode using SRTP and TLS
- Verizon SIP trunk - to operate in non-secure mode using RTP and UDP
- Fax supporting ATA

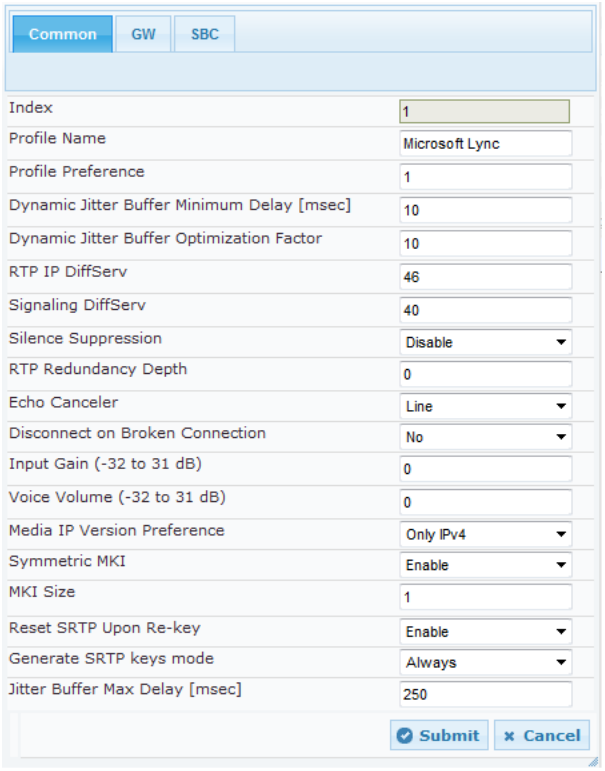
Note that the IP Profiles were assigned to these entities (i.e., IP Groups) in the previous step (see Section 4.5 on page 43).

➢ **To configure IP Profiles:**

1. Open the IP Profile Settings page (**Configuration** tab > **VoIP** > **Coders and Profiles** > **IP Profile Settings**).
2. Click **Add**.
3. Click the **Common** tab, and then configure the parameters as follows:

Parameter	Value
Index	**1**
Profile Name	**Microsoft Lync** (arbitrary descriptive name)
Symmetric MKI	**Enable**
MKI Size	**1**
Reset SRTP State Upon Re-key	**Enable**
Generate SRTP keys mode:	**Always**

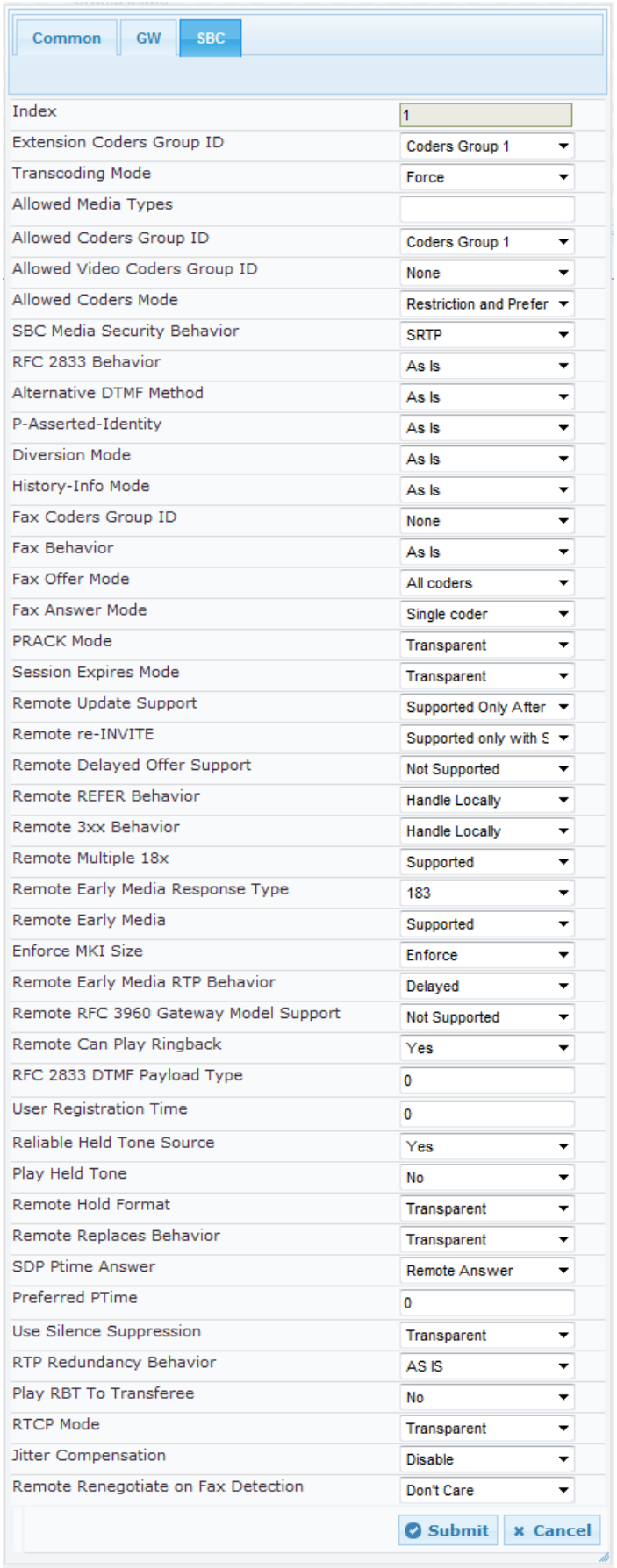
Figure 4-15: Configuring IP Profile for Lync Server 2013 – Common Tab



4. Click the **SBC** tab, and then configure the parameters as follows:

Parameter	Value
Extension Coders Group ID	**Coders Group 1**
Transcoding Mode	**Force**
Allowed Coders Group ID	**Coders Group 1**
Allowed Coders Mode	**Restriction and Preference**
Media Security Behavior	**SRTP**
Remote Update Support	**Supported Only After Connect**
Remote Re-Invite Support	**Supported Only With SDP**
Remote Delayed Offer Support	**Not Supported**
Remote Refer Behavior	**Handle Locally** (required because Lync Server 2013 does not support receipt of SIP REFER)
Remote 3xx Behavior	**Handle Locally** (required because Lync Server 2013 does not support receipt of SIP 3xx responses)
Enforce MKI Size	**Enforce**
Remote Early Media RTP Behavior	**Delayed** (required because Lync Server 2013 does not send RTP immediately to the remote side when it sends a SIP 18x response)

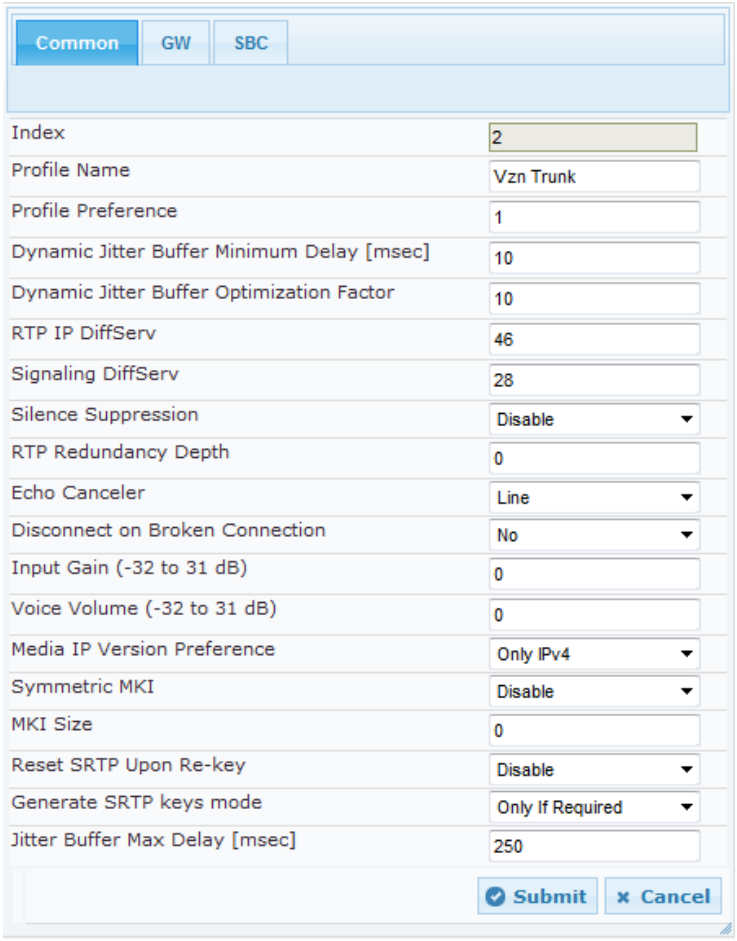
Figure 4-16: Configuring IP Profile for Lync Server 2013 – SBC Tab



5. Configure an IP Profile for the Verizon SIP Trunk:
   a. Click **Add**.
   b. Click the **Common** tab, and then configure the parameters as follows:

Parameter	Value
Index	**2**
Profile Name	**Vzn Trunk** (arbitrary descriptive name)
RTP IP DiffServ	**46**
Signaling Diffserv	**28**

Figure 4-17: Configuring IP Profile for Verizon SIP Trunk – Common Tab



6. Verizon requires the SIP Signaling and RTP packets to be treated independently and marked with the proper settings to be delivered to the Verizon SIP Trunking network.

Table 4-1: DiffServ Service Classes

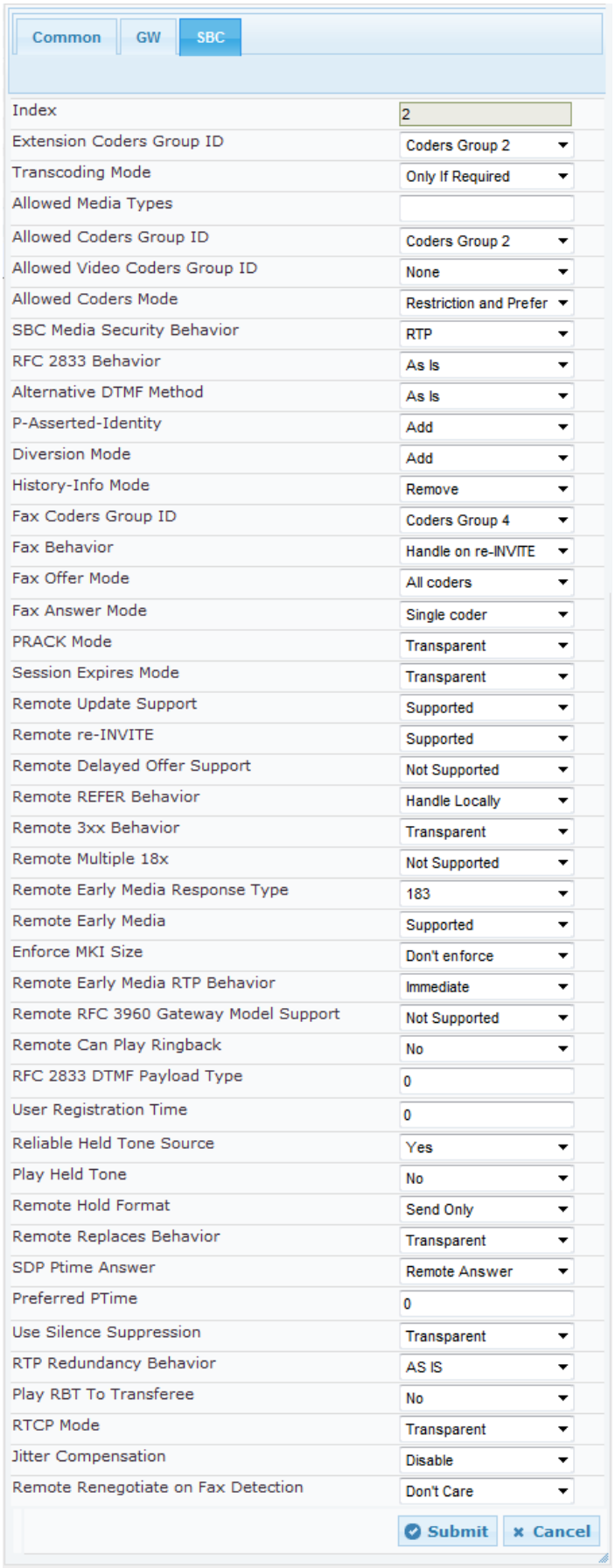
Service Class Name	DSCP Name	DSCP Value	Example Apps	AudioCodes Value
Telephony	EF	101110	RTP/IP Telephony bearer	46
Signaling	CS5	101000	RTP/SIP/IP Telephony signaling	40
Multimedia Streaming	AF31 AF32 AF33	011010 011100 011110	SIP/Streaming video and audio on demand	26 28 30
Broadcast Video	CS3	011000	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.

7. Click the **SBC** tab, and then configure the parameters as follows:

Parameter	Value
Profile ID	**2**
Extension Coders Group ID	**Coders Group 2**
Allowed Coders Group ID	**Coders Group 2**
Allowed Coders Mode	**Restriction and Preference** (lists Allowed Coders first and then original coders in received SDP offer)
Media Security Behavior	**RTP**
P-Asserted-Identity	**Add** (required for anonymous calls)
Diversion Mode	**Add**
History-Info Mode	**Remove**
Fax Coders Group ID	**Coder Group 4**
Fax Behavior	**Handle on re-INVITE**
Remote REFER Behavior	**Handle Locally** (E-SBC handles / terminates incoming REFER requests instead of forwarding them to SIP Trunk)
Remote Multiple 18x	**Not Supported**
Remote Can Play Ringback	**No** (required because Lync Server 2013 does not provide a ringback tone for incoming calls)
Remote Hold Format	**Send Only**

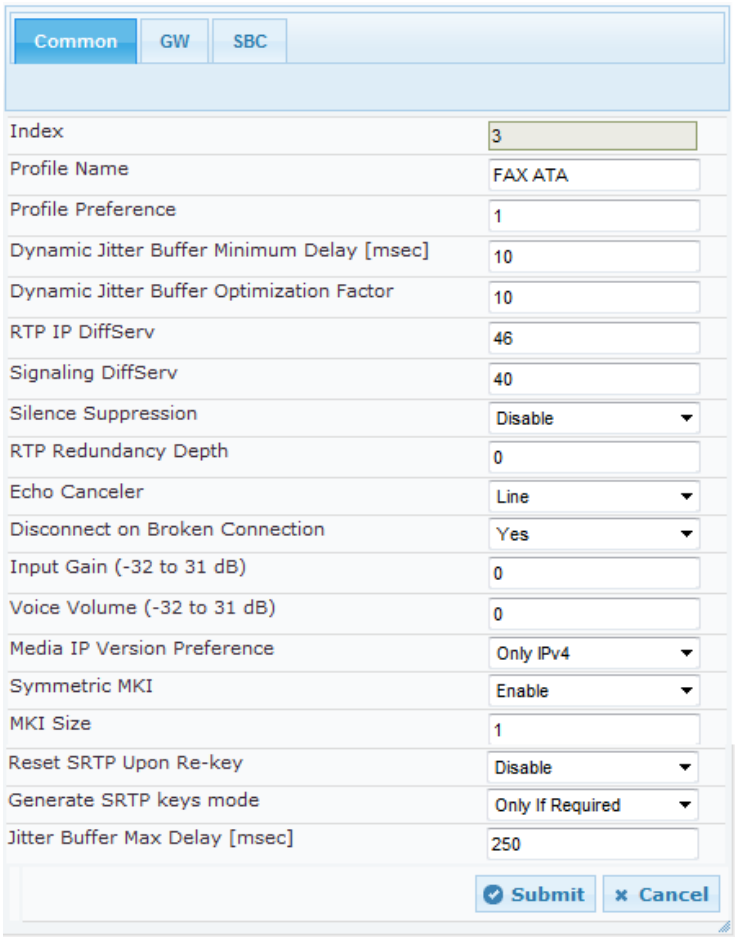
Figure 4-18: Configuring IP Profile for Verizon SIP Trunk – SBC Tab



8. Configure an IP Profile for the Fax supporting ATA:
   a. Click **Add**.
   b. Click the **Common** tab, and then configure the parameters as follows:

Parameter	Value
Index	**3**
Profile Name	**FAX ATA** (arbitrary descriptive name)

Figure 4-19: Configuring IP Profile for FAX supporting ATA – Common Tab



9. Click the **SBC** tab, and then configure the parameters as follows:

Parameter	Value
Profile ID	**3**
Allowed Coders Mode	**Restriction** (lists Allowed Coders first and then original coders in received SDP offer)
Fax Coders Group ID	**Coders Group 4**
FAX Behavior	**Handle on re-INVITE**

Figure 4-20: Configuring IP Profile for Fax supporting ATA – SBC Tab

4.7 Step 7: Configure Coders

This step shows how to configure coders (termed *Coder Group*). As Lync Server 2013 supports the G.711 coder while the network connection to Verizon SIP Trunk may restrict operation with a lower bandwidth coder such as G.729, you need to add a Coder Group with the G.729 coder for the Verizon SIP Trunk.

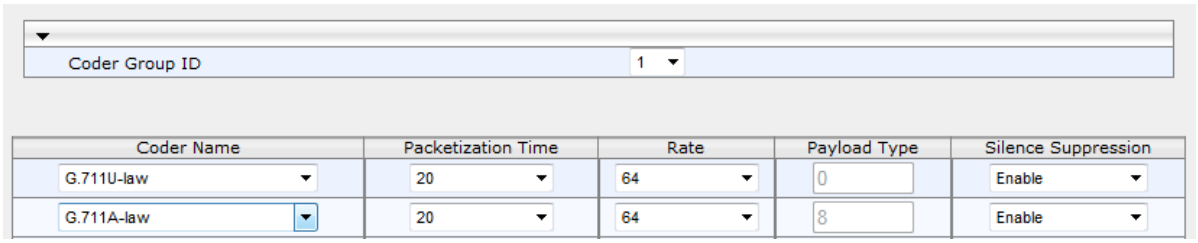
Note that the Coder Group ID for this entity was assigned to its corresponding IP Profile in the previous step (see Section 4.6 on page 45).

➢ **To configure coders:**

1. Open the Coder Group Settings (**Configuration** tab > **VoIP** menu > **Coders and Profiles** > **Coders Group Settings**).
2. Configure a Coder Group for Lync Server 2013:

Parameter	Value
Coder Group ID	**1**
Coder Name	**G.711 U-law** G.711 A-law
Silence Suppression	**Enable**

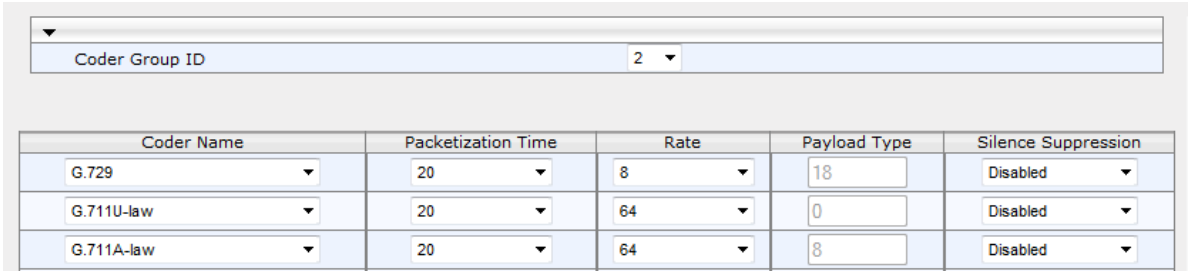
Figure 4-21: Configuring Coder Group for Lync Server 2013



3. Configure a Coder Group for Verizon SIP Trunk:

Parameter	Value
Coder Group ID	**2**
Coder Name	**G.729** G.711 U-law G.711 A-law

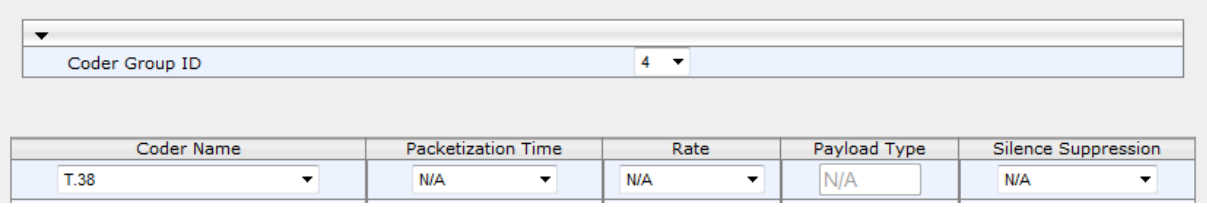
Figure 4-22: Configuring Coder Group for Verizon SIP Trunk



4. Configure a Coder Group for the FAX supporting ATA:

Parameter	Value
Coder Group ID	**4**
Coder Name	**T.38**

Figure 4-23: Configuring Coder Group for the FAX Supporting ATA



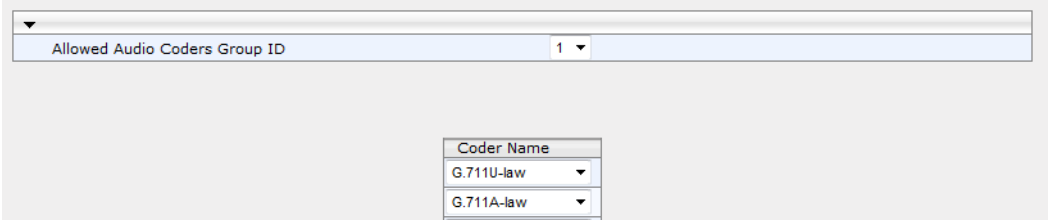
The procedure below describes how to configure an Allowed Coders Group to ensure that voice sent to the Verizon SIP Trunk uses the G.729 coder whenever possible. Note that this Allowed Coders Group ID was assigned to the IP Profile belonging to the Verizon SIP Trunk in the previous step (see Section 4.6 on page 45).

➢ **To set a preferred coder for Microsoft Lync:**

1. Open the Allowed Coders Group page (**Configuration** tab > **VoIP** menu > **SBC** > **Allowed Coders Group**).
2. Configure an Allowed Coder as follows:

Parameter	Value
Allowed Coders Group ID	**1**
Coder Name	**G.711 U-law** **G.711 A-law**

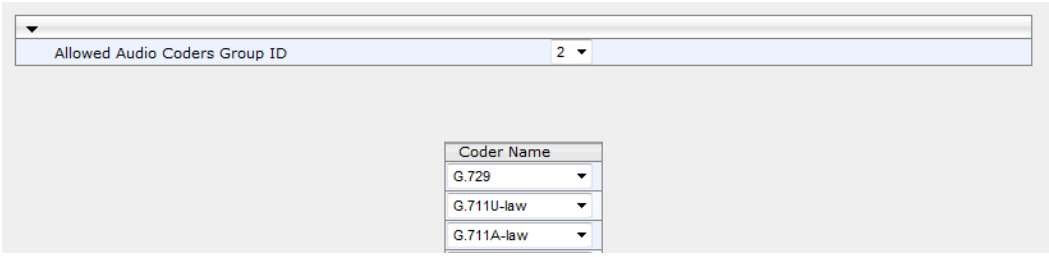
Figure 4-24: Configuring Allowed Coders Group for Microsoft Lync



3. Configure an Allowed Coder as follows:

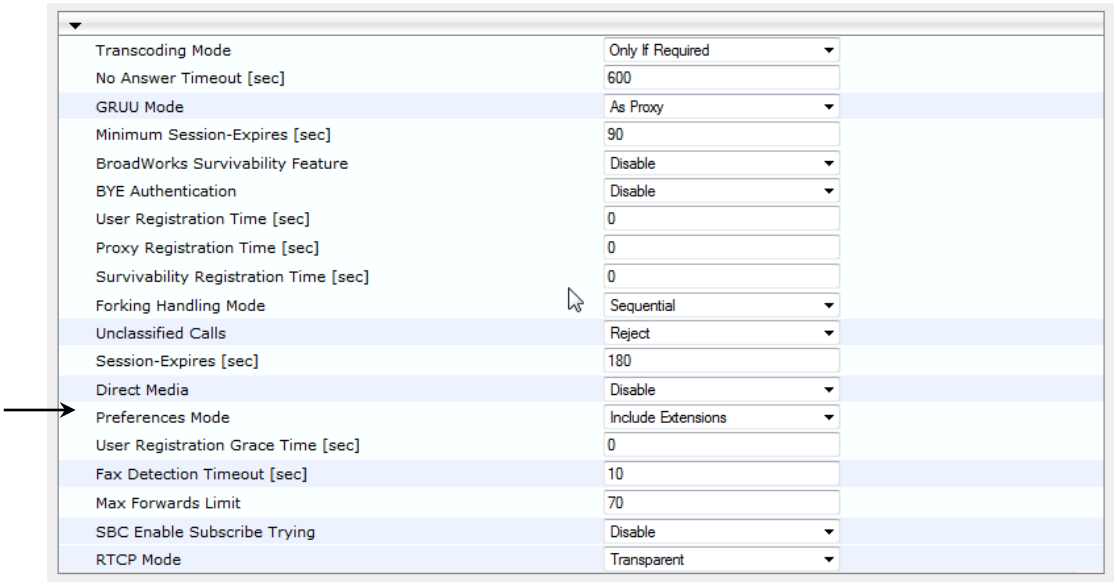
Parameter	Value
Allowed Coders Group ID	**2**
Coder Name	**G.729** **G.711 U-law** **G.711 A-law**

Figure 4-25: Configuring Allowed Coders Group for Verizon SIP Trunk



4. Open the General Settings page (**Configuration** tab > **VoIP** menu > **SBC** > **General Settings**).

Figure 4-26: SBC Preferences Mode



5. From the '**Preferences Mode**' drop-down list, select **Include Extensions**.
6. Click **Submit**.

4.8 Step 8: Configure a SIP TLS Connection

This section shows how to configure the E-SBC for using a TLS connection with the Lync Server 2013 Mediation Server. This is essential for a secure SIP TLS connection.

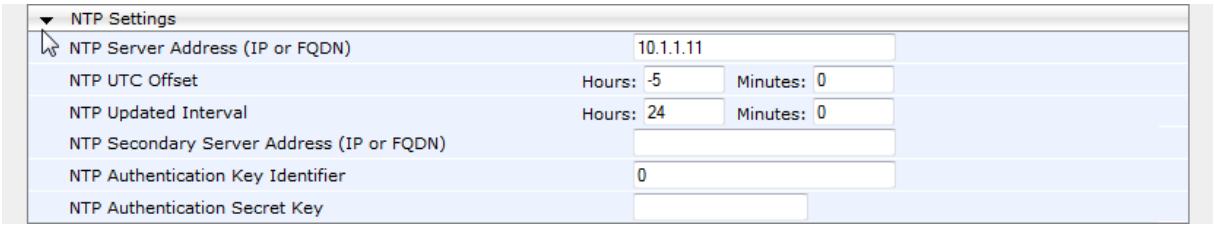
4.8.1 Step 8a: Configure the NTP Server Address

This step shows how to configure the NTP server's IP address. It is recommended to implement an NTP server (Microsoft NTP server or a third-party server) to ensure that the E-SBC receives the accurate and current date and time. This is necessary for validating certificates of remote parties.

➢ **To configure the NTP server address:**

1. Open the Application Settings page (**Configuration** tab > **System** > **Application Settings**).
2. In the 'NTP Server IP Address' field, enter the IP address of the NTP server (e.g., **10.1.1.11**).

Figure 4-27: Configuring NTP Server Address



3. Click **Submit**.

4.8.2 Step 8b: Configure a Certificate

This step shows how to exchange a certificate with Microsoft Certificate Authority (CA). The certificate is used by the E-SBC to authenticate the connection with Lync Server 2013.

Follow these main steps:

a. Generate a Certificate Signing Request (CSR).

b. Request a Device Certificate from CA.

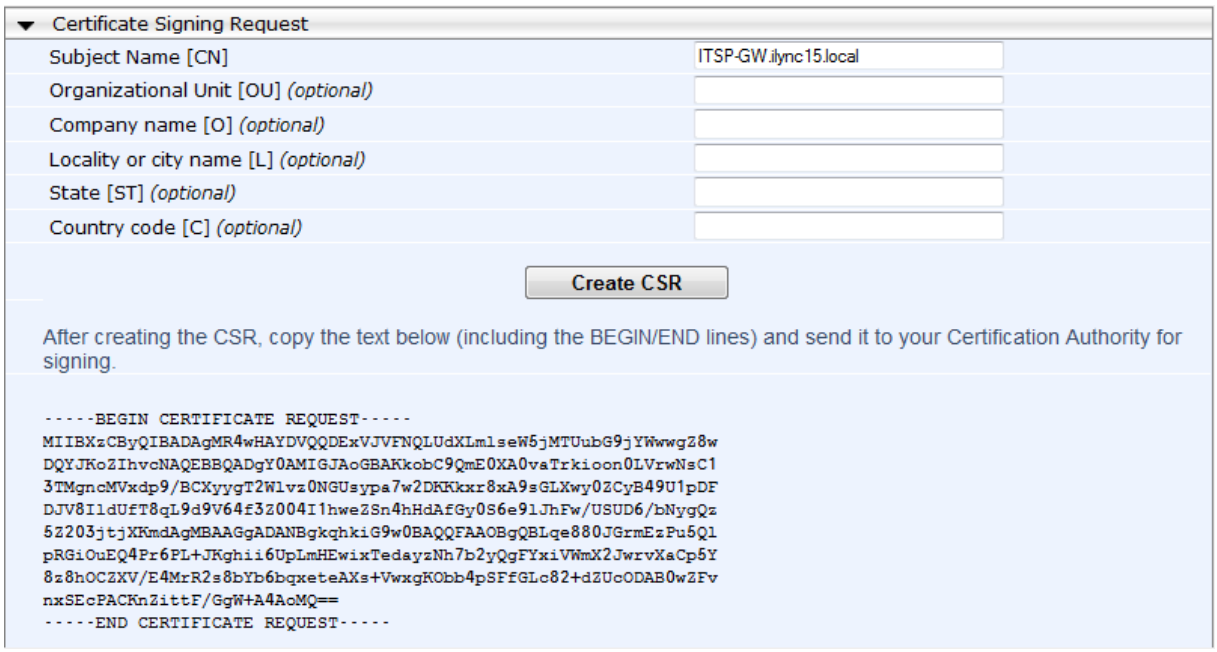
c. Obtain a Trusted Root Certificate from CA.

d. Deploy Device and Trusted Root Certificates on the E-SBC.

➢ **To configure a certificate:**

1. Open the Certificates page (**Configuration** tab > **System** > **Certificates**).

Figure 4-28: Certificates Page - Creating CSR



2. In the 'Subject Name' field, enter the media gateway name (e.g., **ITSP-GW.ilync15.local**).

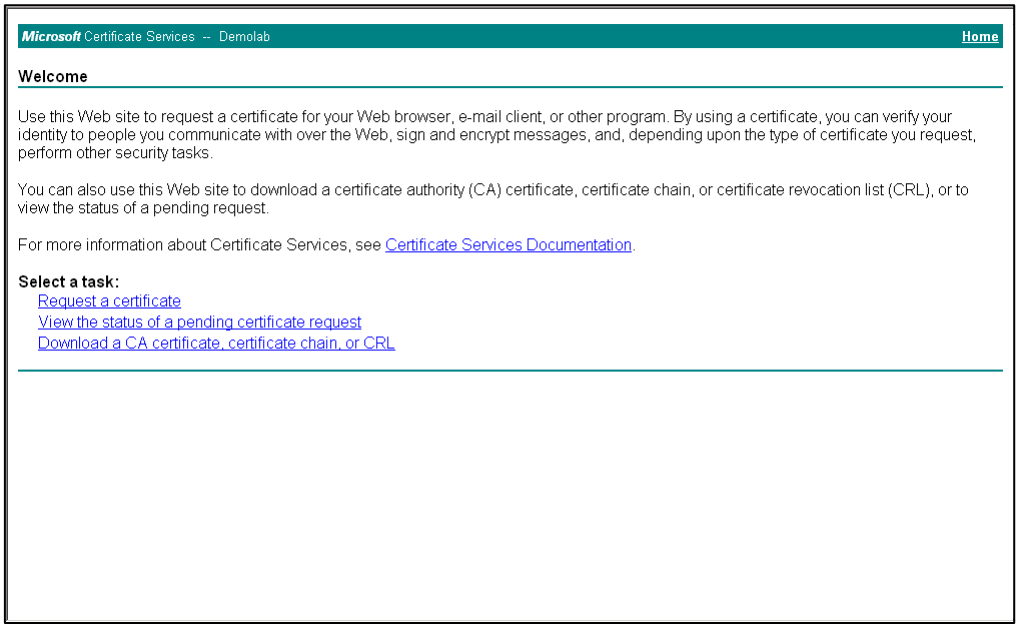
Note: The value entered in this field must be identical to the gateway name configured in the Topology Builder for Lync Server 2013 (see Section 3.1 on page 13.

3. Click **Create CSR**; a certificate request is generated.

4. Copy the CSR from the line **"----BEGIN CERTIFICATE" to "END CERTIFICATE REQUEST----**" to a text file (such as Notepad), and then save it to a folder on your computer with the file name *certreq.txt*.

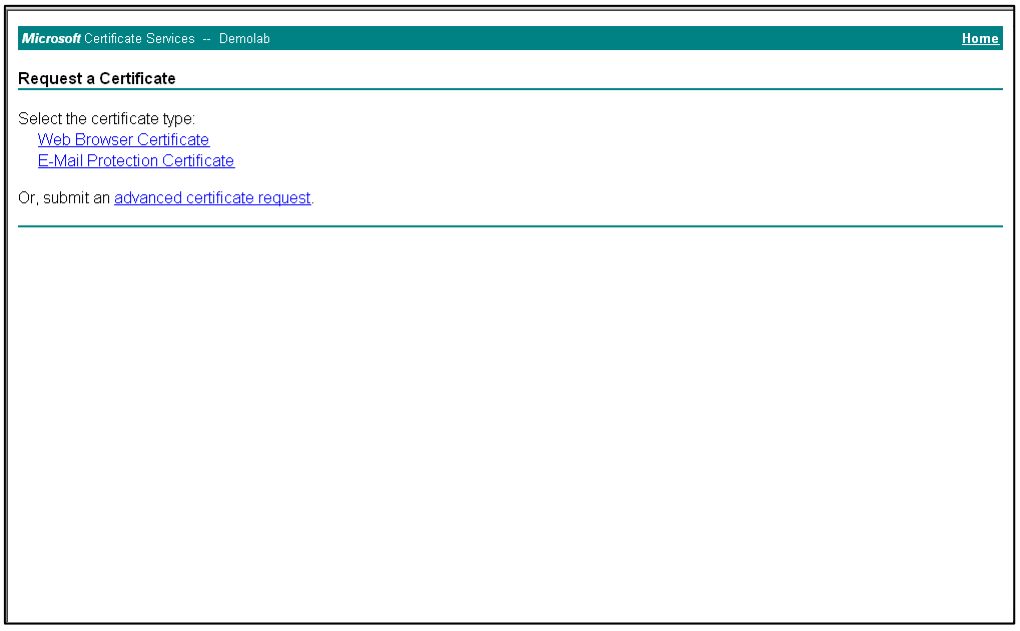
5. Open a Web browser and navigate to the Microsoft Certificates Services Web site at http://<certificate server>/CertSrv.

Figure 4-29: Microsoft Certificate Services Web Page



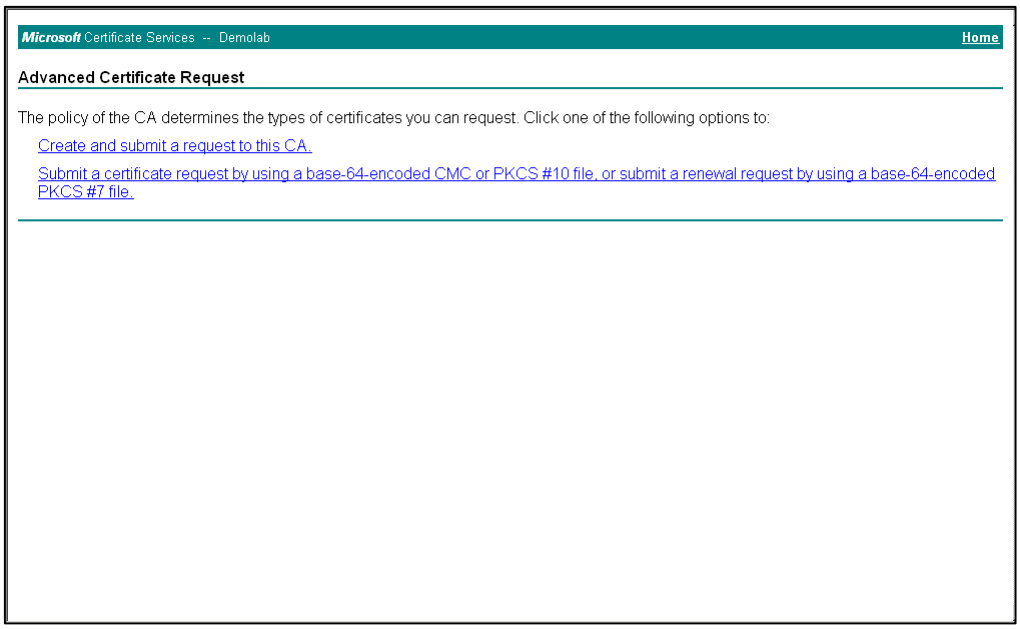
6. Click **Request a certificate**.

Figure 4-30: Request a Certificate Page



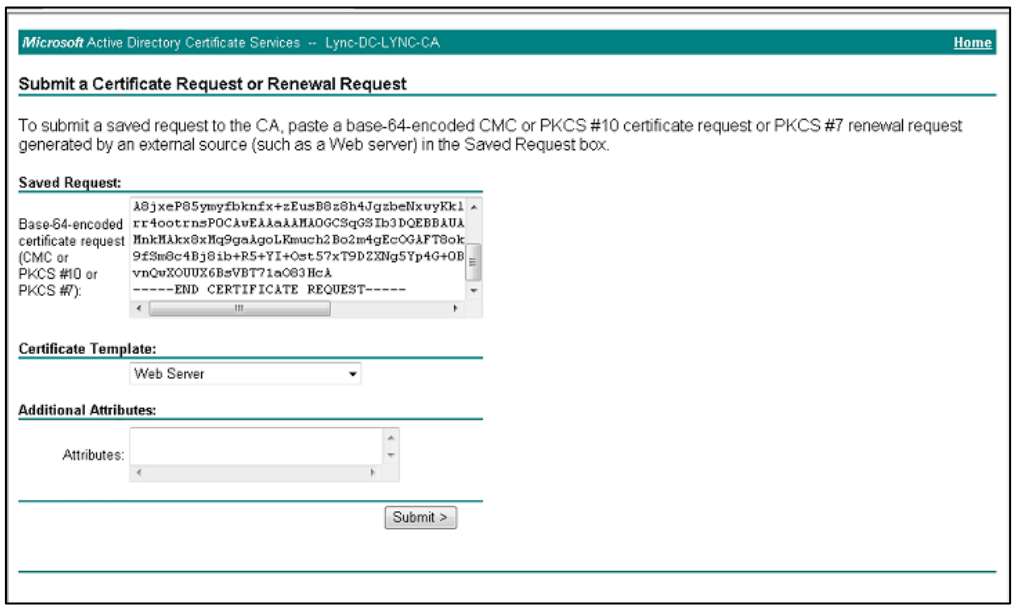
7. Click **advanced certificate request**, and then click **Next**.

Figure 4-31: Advanced Certificate Request Page



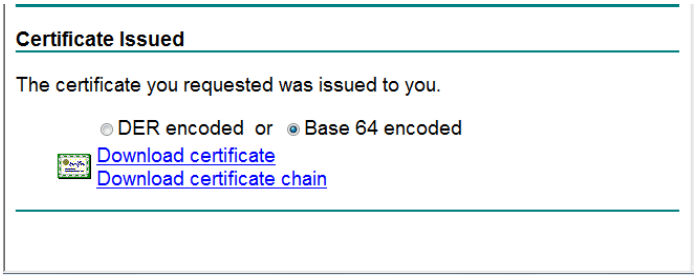
8. Click **Submit a certificate request ...**, and then click **Next**.

Figure 4-32: Submit a Certificate Request or Renewal Request Page



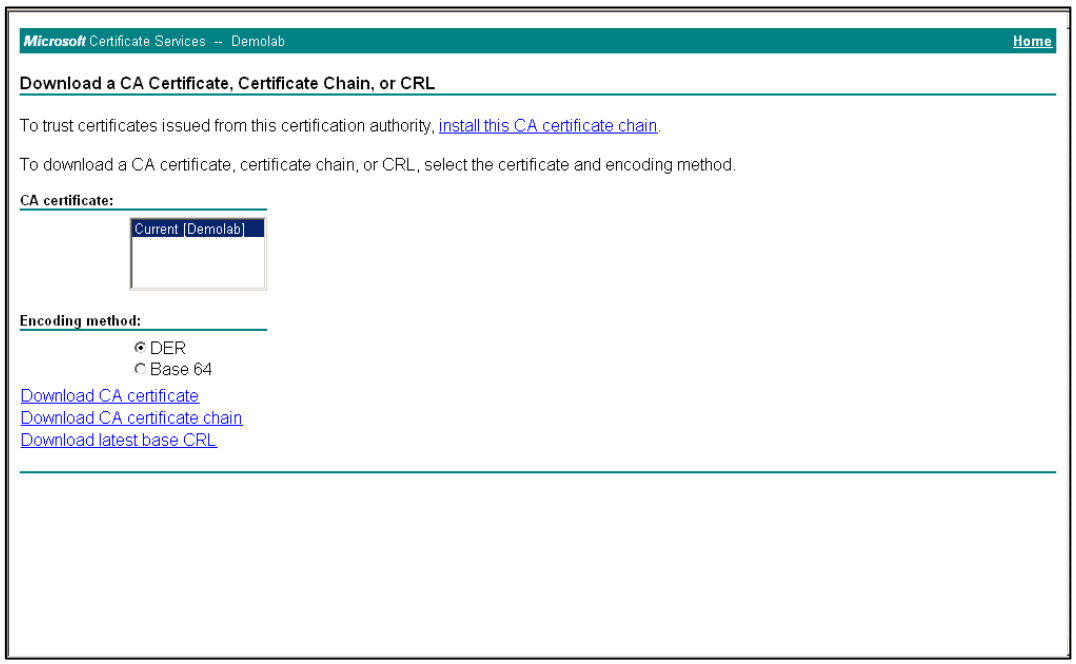
9. Open the *certreq.txt* file that you created and saved in Step 4, and then copy its contents to the 'Saved Request' field.
10. From the 'Certificate Template' drop-down list, select **Web Server**.
11. Click **Submit**.

Figure 4-33: Certificate Issued Page



12. Select the **Base 64 encoded** option for encoding, and then click **Download certificate**.
13. Save the file as *gateway.cer* to a folder on your computer.
14. Click the **Home** button or navigate to the certificate server at http://<Certificate Server>/CertSrv.
15. Click **Download a CA certificate**, **certificate chain, or CRL**.

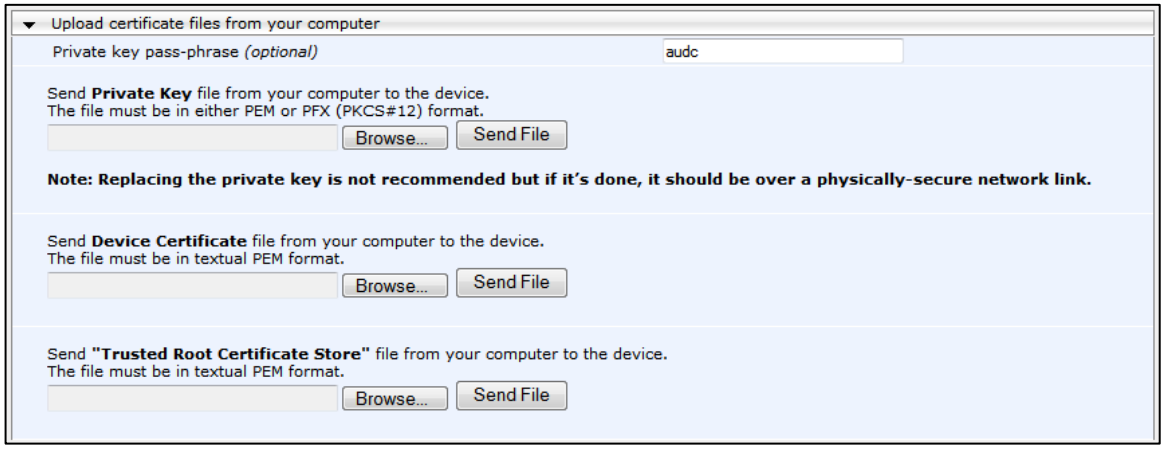
Figure 4-34: Download a CA Certificate, Certificate Chain, or CRL Page



16. Under the 'Encoding method' group, select the **Base 64** option for encoding.
17. Click **Download CA certificate**.
18. Save the file as *certroot.cer* to a folder on your computer.

19. In the E-SBC's Web interface, return to the Certificates page and do the following:
    a. In the 'Device Certificate' field, click **Browse** and select the *gateway.cer* certificate file that you saved on your computer in Step 13, and then click **Send File** to upload the certificate to the E-SBC.
    b. In the 'Trusted Root Certificate Store' field, click **Browse** and select the *certroot.cer* certificate file that you saved on your computer in Step 18, and then click **Send File** to upload the certificate to the E-SBC.

Figure 4-35: Certificates Page (Uploading Certificate)



20. Reset the E-SBC with a burn to flash for your settings to take effect (see Section 4.15 on page 98).

4.9 Step 9: Configure SRTP

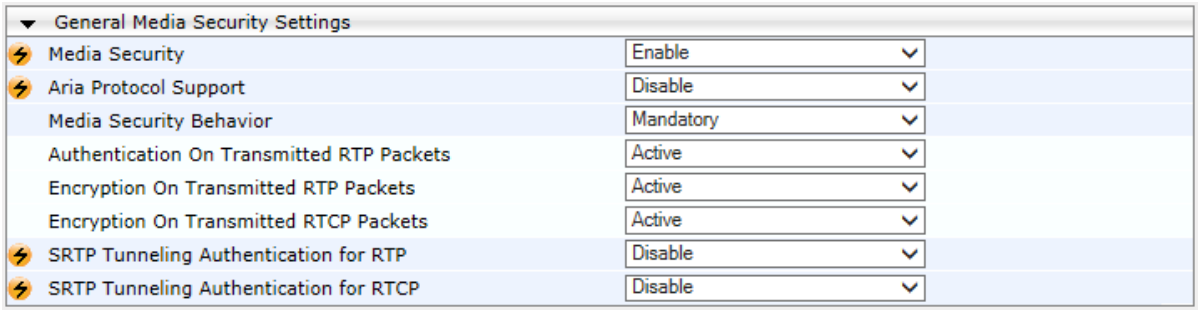
This step shows how to configure media security. If you configure the Microsoft Mediation Server to use SRTP, you must configure the E-SBC to operate in the same manner. Note that SRTP was enabled for Lync Server 2013 when you configured an IP Profile for Lync Server 2013 (see Section 4.6 on page 45).

➢ **To configure media security:**

1. Open the Media Security page (**Configuration** tab > **Media** menu > **Media Security**).
2. Configure the parameters as follows:

Parameter	Value
Media Security	**Enable**

Figure 4-36: Configuring SRTP



3. Click **Submit**.
4. Reset the E-SBC with a burn to flash for your settings to take effect (see Section 4.15 on page 98).

4.10 Step 10: Configure Maximum IP Media Channels

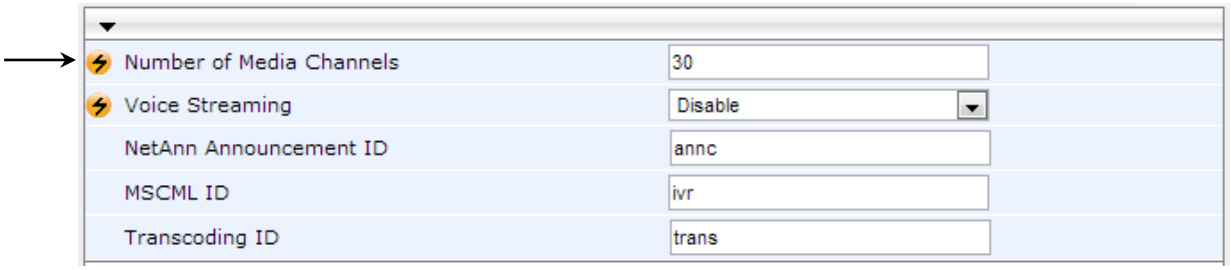
This step shows how to configure the maximum number of required IP media channels. The number of media channels represents the number of DSP channels that the E-SBC allocates to call sessions.

Note: This step is required *only* if transcoding is required.

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

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

Figure 4-37: Configuring Number of IP Media Channels



2. In the 'Number of Media Channels' field, enter the number of media channels according to your environments transcoding calls (e.g., **30**).
3. Click **Submit**.
4. Reset the E-SBC with a burn to flash for your settings to take effect (see Section 4.15 on page 98).

4.11 Step 11: Configure IP-to-IP Call Routing Rules

This step shows how to configure IP-to-IP call routing rules. These rules define the routes for forwarding SIP messages (e.g., INVITE) received from one IP entity to another. The E-SBC selects the rule whose configured input characteristics (e.g., IP Group) match those of the incoming SIP message. If the input characteristics do not match the first rule in the table, they are compared to the second rule, and so on, until a matching rule is located. If no rule is matched, the message is rejected. The routing rules use the configured IP Groups to denote the source and destination of the call. As configured in Section 4.5 on page 43, IP Group 1 represents Lync Server 2013, and IP Group 2 represents Verizon SIP Trunk.

For the interoperability test topology, the following IP-to-IP routing rules must be configured to route calls between Lync Server 2013 (LAN) and Verizon SIP Trunk (WAN):

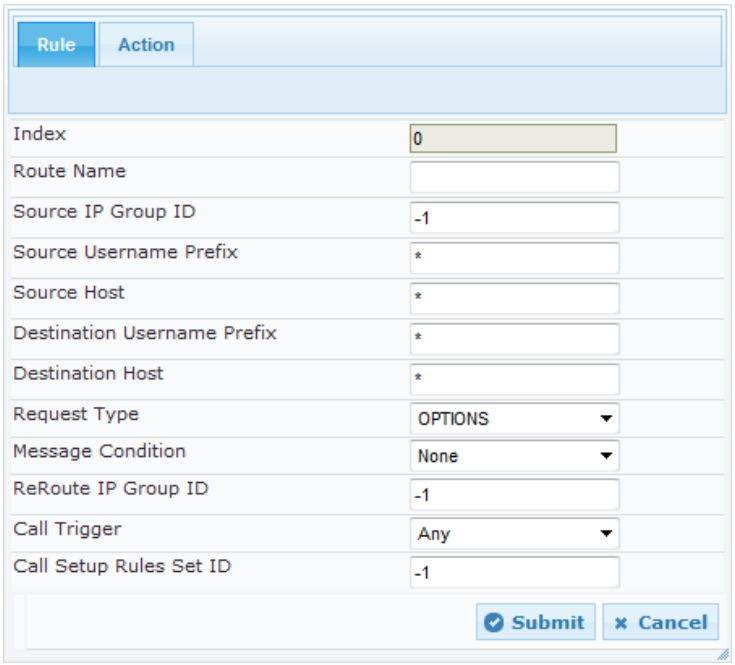
- Terminate SIP OPTIONS messages on the E-SBC that are received from the LAN
- Calls from Lync Server 2013 to Verizon SIP Trunk
- Calls from Verizon SIP Trunk to Lync Server 2013
- Calls from Verizon SIP Trunk to Fax supporting ATA
- Calls from Fax supporting ATA to Verizon SIP trunk

➢ **To configure IP-to-IP routing rules:**

1. Open the IP-to-IP Routing Table page (**Configuration** tab > **VoIP** menu > **SBC** > **Routing SBC** > **IP-to-IP Routing Table**).
2. Configure a rule to terminate SIP OPTIONS messages received from the LAN:
   a. Click **Add**.
   b. Click the **Rule** tab, and then configure the parameters as follows:

Parameter	Value
Index	**0**
Request Type	**OPTIONS**
Destination Type	**Dest Address**
Destination Address	**internal**

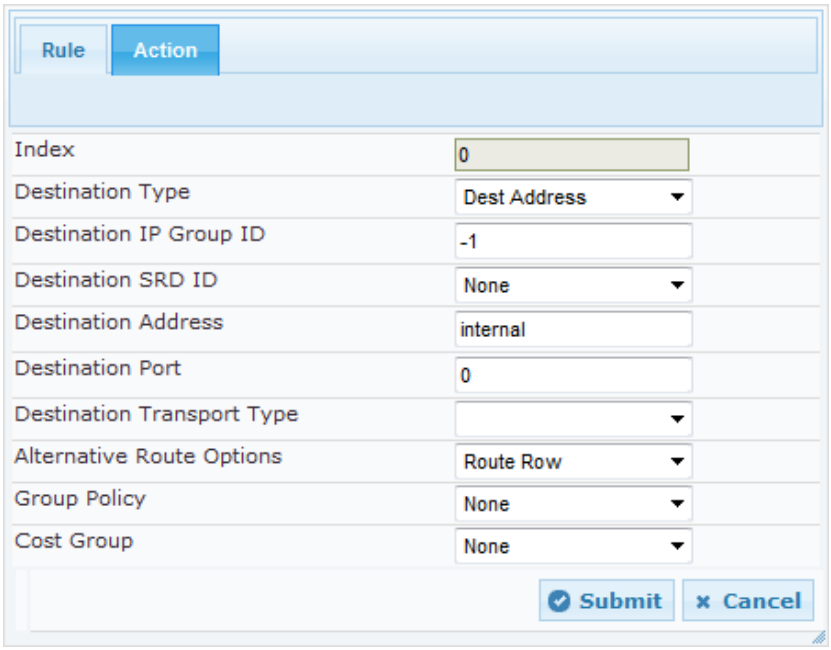
Figure 4-38: Configuring IP-to-IP Routing Rule for Terminating SIP OPTIONS from LAN – Rule Tab



3. Click the **Action** tab, and then configure the parameters as follows:

Parameter	Value
Destination Type	**Dest Address**
Destination Address	**internal**

Figure 4-39: Configuring IP-to-IP Routing Rule for Terminating SIP OPTIONS – Action Tab

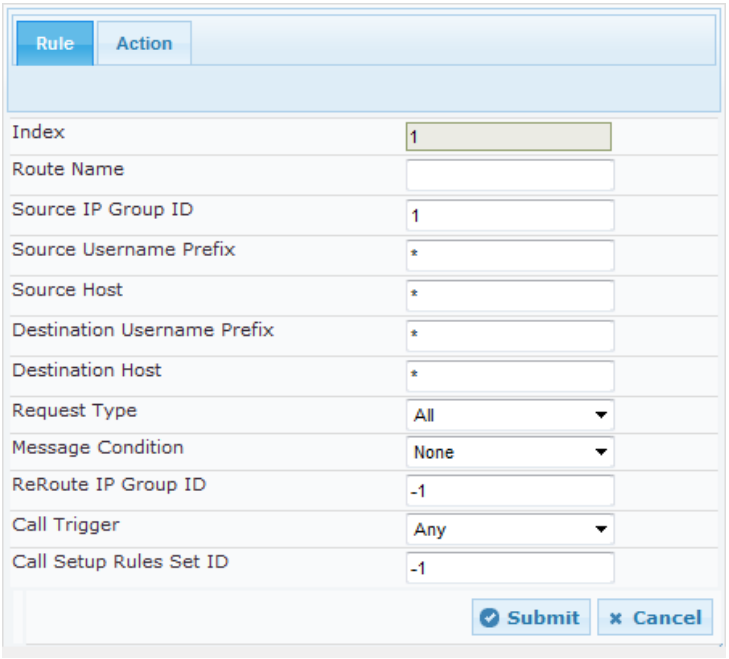


4. Click the **Submit** button.

5. Configure a rule to route calls from Lync Server 2013 to Verizon SIP Trunk:
   a. Click **Add**.
   b. Click the **Rule** tab, and then configure the parameters as follows:

Parameter	Value
Index	**1**
Route Name	Arbitrary descriptive name
Source IP Group ID	**1**

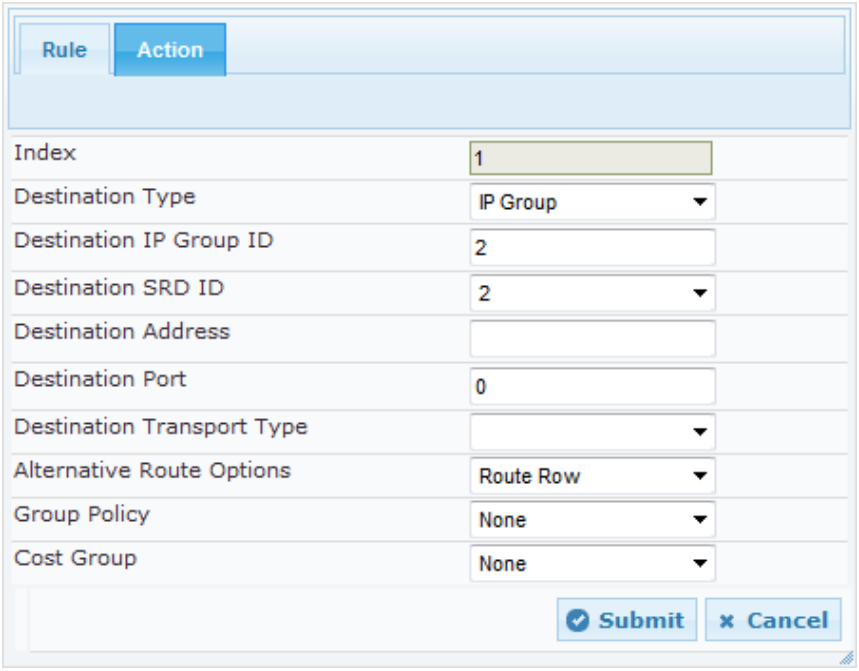
Figure 4-40: Configuring IP-to-IP Routing Rule for Lync to ITSP – Rule tab



6. Click the **Action** tab, and then configure the parameters as follows:

Parameter	Value
Destination Type	**IP Group**
Destination IP Group ID	**2**
Destination SRD ID	**2**

Figure 4-41: Configuring IP-to-IP Routing Rule for Lync to ITSP – Action tab

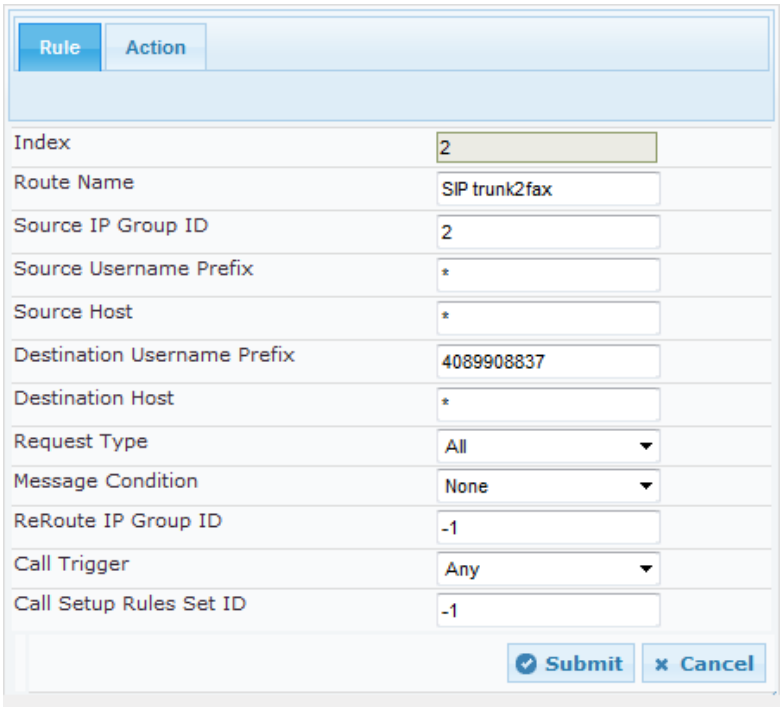


7. Click the **Submit** button.

8. Configure a rule to route calls from Verizon SIP Trunk to Fax supporting ATA:
   a. Click **Add**.
   b. Click the **Rule** tab, and then configure the parameters as follows:

Parameter	Value
Index	**2**
Route Name	**SIPtrunk2fax** (arbitrary descriptive name)
Source IP Group ID	**2**
Destination Username Prefix	**4089908837**

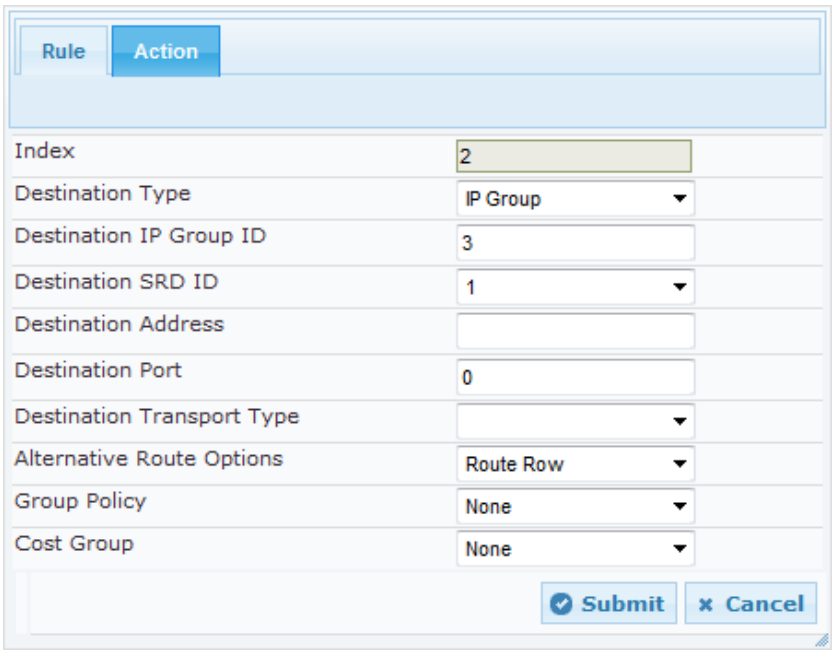
Figure 4-42: Configuring IP-to-IP Routing Rule for ITSP to Fax ATA – Rule tab



9. Click the **Action** tab, and then configure the parameters as follows:

Parameter	Value
Destination Type	**IP Group**
Destination IP Group ID	**3**
Destination SRD ID	**1**

Figure 4-43: Configuring IP-to-IP Routing Rule for ITSP to FAX ATA – Action tab

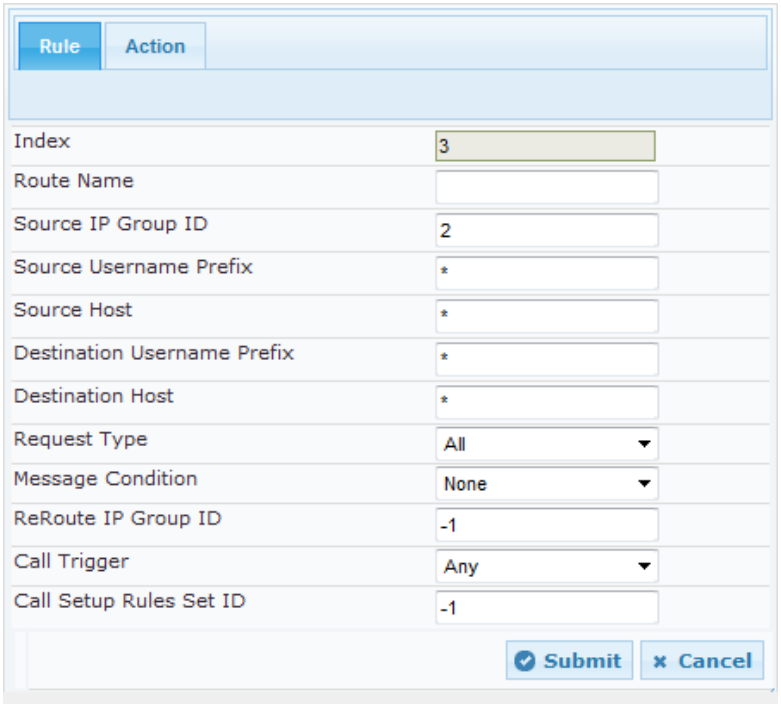


10. Click the **Submit** button.

11. Configure a rule to route calls from Verizon SIP Trunk to Lync Server 2013:
    a. Click **Add**.
    b. Click the **Rule** tab, and then configure the parameters as follows:

Parameter	Value
Index	**3**
Route Name	Arbitrary descriptive name
Source IP Group ID	**2**

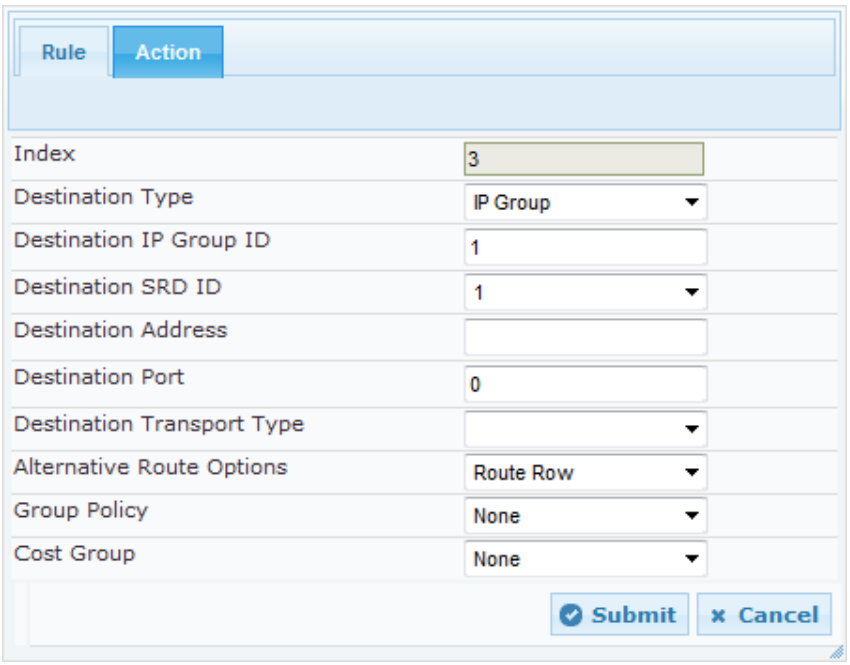
Figure 4-44: Configuring IP-to-IP Routing Rule for ITSP to Lync – Rule tab



12. Click the **Action** tab, and then configure the parameters as follows:

Parameter	Value
Destination Type	**IP Group**
Destination IP Group ID	**1**
Destination SRD ID	**1**

Figure 4-45: Configuring IP-to-IP Routing Rule for ITSP to Lync – Action tab

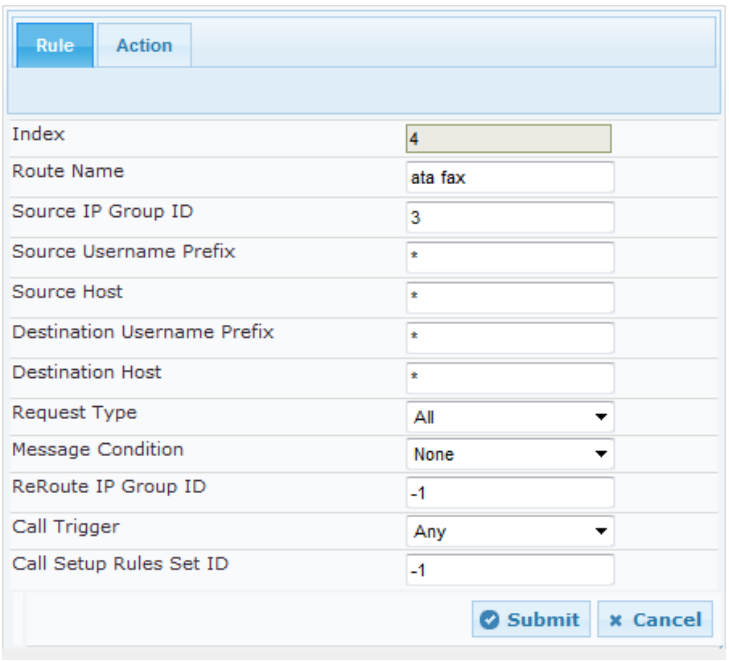


13. Click the **Submit** button.

14. Configure a rule to route calls from Fax ATA to Verizon SIP Trunk:
    a. Click **Add**.
    b. Click the **Rule** tab, and then configure the parameters as follows:

Parameter	Value
Index	**4**
Route Name	**ata fax** (arbitrary descriptive name)
Source IP Group ID	**3**

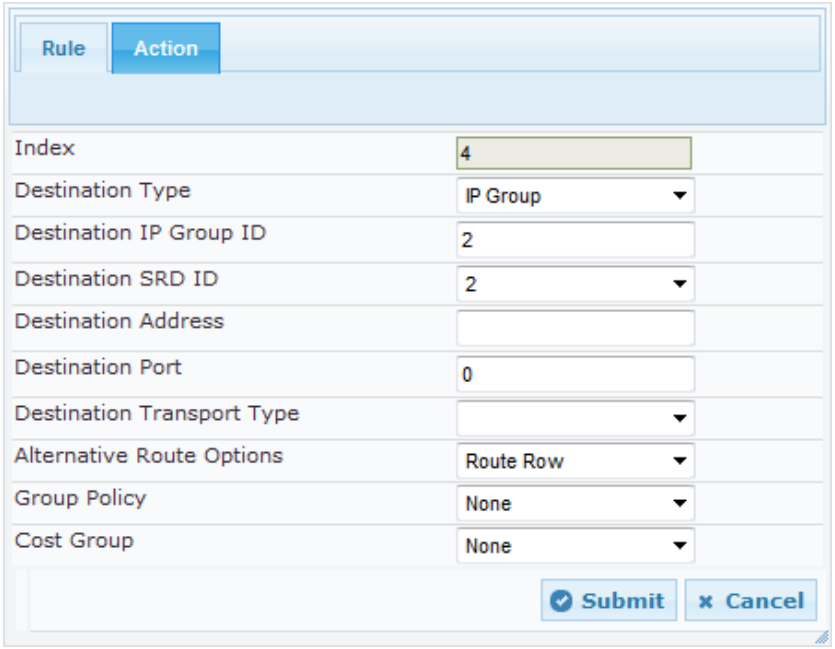
Figure 4-46: Configuring IP-to-IP Routing Rule for FAX ATA to ITSP Trunk – Rule tab



15. Click the **Action** tab, and then configure the parameters as follows:

Parameter	Value
Destination Type	**IP Group**
Destination IP Group ID	**2**
Destination SRD ID	**2**

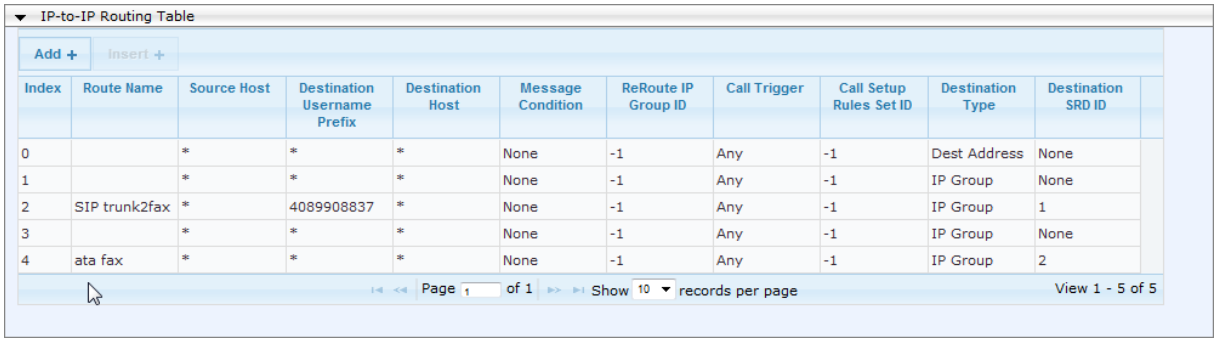
Figure 4-47: Configuring IP-to-IP Routing Rule for FAX ATA to ITSP Trunk – Action tab



16. Click the **Submit** button.

The configured routing rules are shown in the figure below:

Figure 4-48: Configured IP-to-IP Routing Rules in IP-to-IP Routing Table



Note: The routing configuration may change according to your specific deployment topology.

4.12 Step 12: Configure IP-to-IP Manipulation Rules

This step shows how to configure IP-to-IP manipulation rules. These rules manipulate the source and / or destination number. The manipulation rules use the configured IP Groups to denote the source and destination of the call. As configured in Section 4.5 on page 43, IP Group 1 represents Lync Server 2013, and IP Group 2 represents Verizon SIP Trunk. These manipulations can be utilized on the incoming message and/or the outgoing message.

Note: Adapt the manipulation table according to your environment's dial plan.

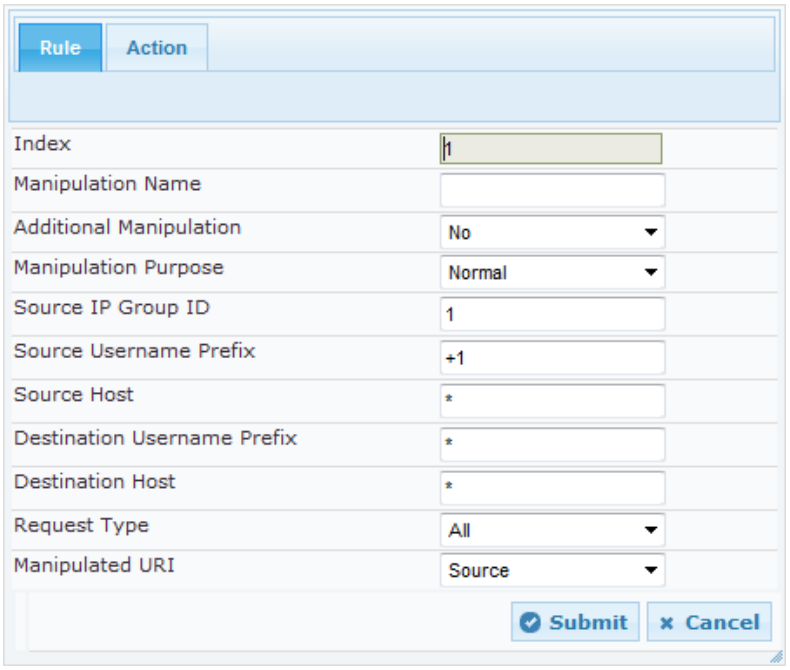
For this interoperability test topology, a set of normalization manipulation rules are configured to remove the '+1' from the Source and Destination numbers presented by Microsoft Lync. These were performed in the IP-to-IP Inbound manipulation rules. While using a set of IP-to-IP Outbound manipulation rules, add the "+" (plus sign) to the destination number for calls from IP Group 2 (Verizon SIP Trunk) to IP Group 1 (i.e., Lync Server 2013) for any destination username prefix.

➢ **To configure a number manipulation rule:**

1. Open the IP-to-IP Outbound Manipulation page (**Configuration** tab > **VoIP** menu > **SBC** > **Manipulations SBC** > **IP-to-IP Inbound**).
2. Click **Add**.
3. Click the **Rule** tab, and then configure the parameters as follows:

Parameter	Value
Index	**1**
Source IP Group	**1**
Source Username Prefix	**+1**
Destination Username Prefix	***** (asterisk sign)
Manipulated URI	**Source**

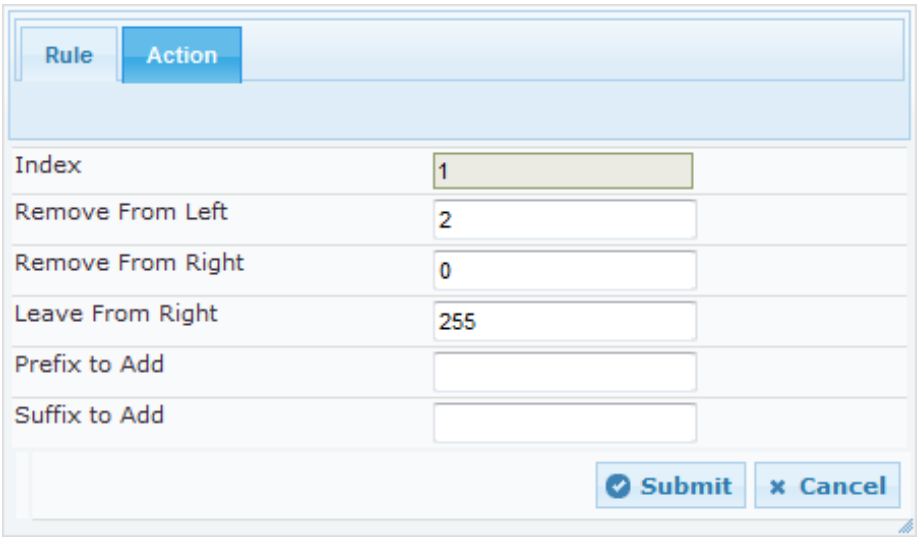
Figure 4-49: Configuring IP-to-IP Outbound Manipulation Rule – Rule Tab



4. Click the **Action** tab, and then configure the parameters as follows:

Parameter	Value
Remove From Left	**2** (this removes the '+1')

Figure 4-50: Configuring IP-to-IP Outbound Manipulation Rule - Action Tab

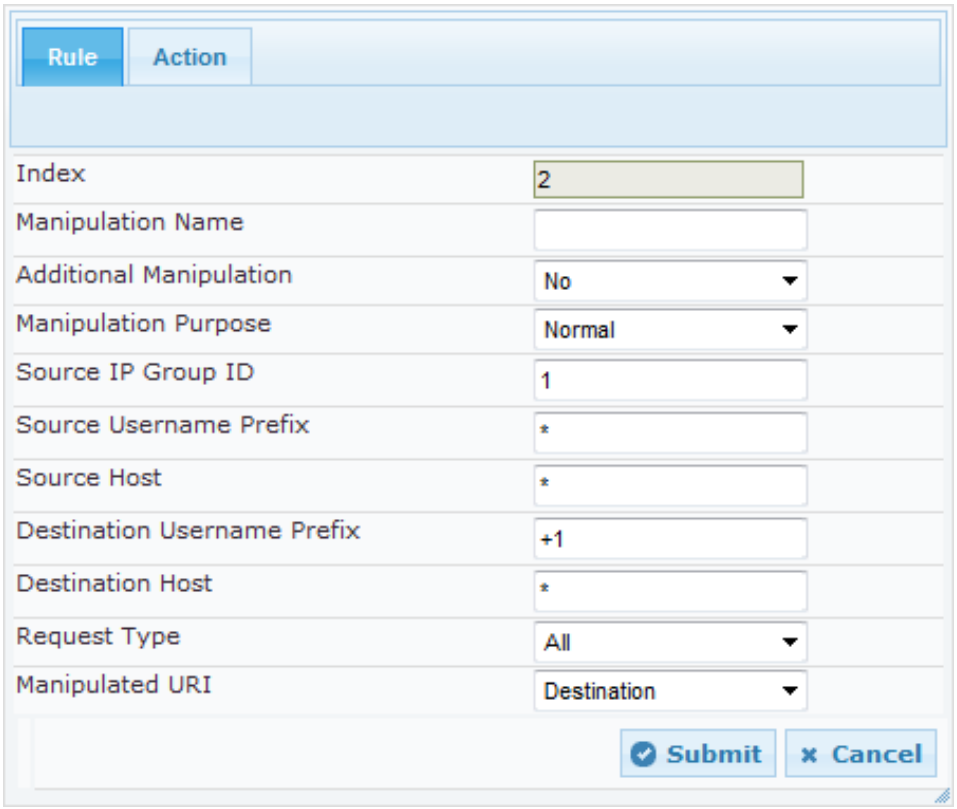


5. Click **Submit**.
6. Click **Add**.

7. Click the **Rule** tab, and then configure the parameters as follows:

Parameter	Value
Index	**2**
Source IP Group	**1**
Source Username Prefix	***** (asterisk sign)
Destination Username Prefix	**+1**
Manipulation URI	**Destination**

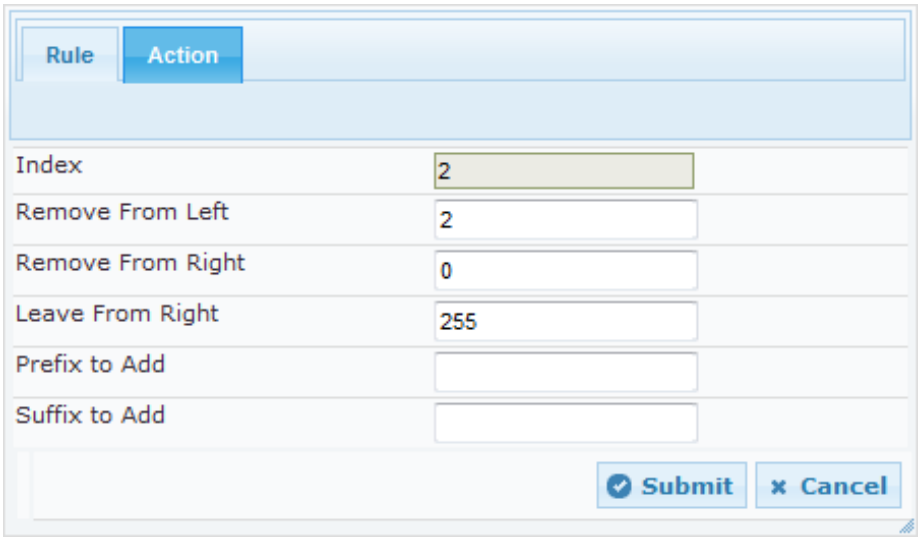
Figure 4-51: Configuring IP-to-IP Outbound Manipulation Rule – Rule Tab



8. Click the **Action** tab, and then configure the parameters as follows:

Parameter	Value
Remove From Left	**2** (this removes the '+1')

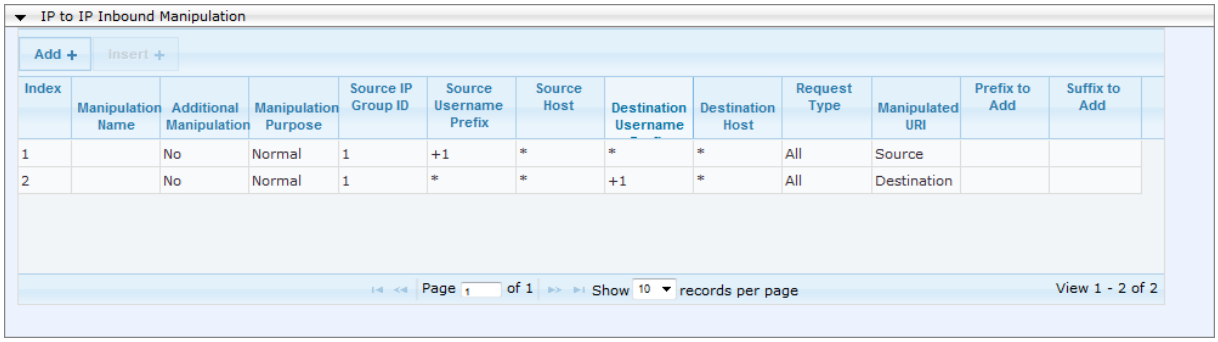
Figure 4-52: Configuring IP-to-IP Outbound Manipulation Rule - Action Tab



9. Click **Submit**.

The figure below shows an example of configured IP-to-IP inbound manipulation rules for calls from IP Group 1 (i.e., Lync Server 2013):

Figure 4-53: Example of Configured IP-to-IP Inbound Manipulation Rules



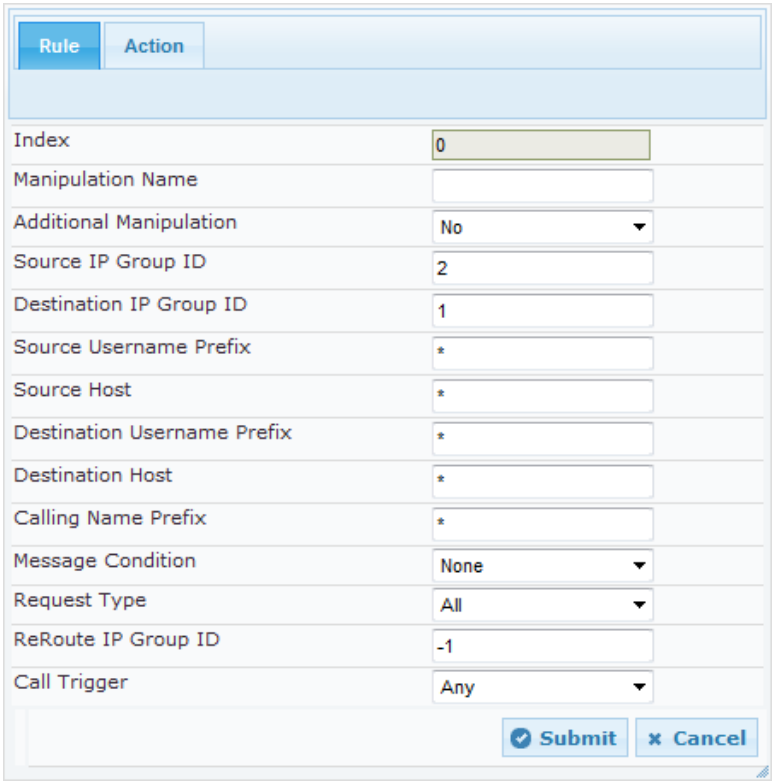
Rule Index	Description
1	Calls from IP Group 1 with a source number that contains a prefix of '+1' and with any destination number (*), remove the "+1" from the prefix of the source number.
2	Calls from IP Group 1 with a destination number that contains a prefix of '+1', remove "+1" from the destination number.

➢ **To configure an IP-to-IP Outbound number manipulation rule:**

1. Open the IP-to-IP Outbound Manipulation page (**Configuration** tab > **VoIP** menu > **SBC** > **Manipulations SBC** > **IP-to-IP Outbound**).
2. Click **Add**.
3. Click the **Rule** tab, and then configure the parameters as follows:

Parameter	Value
Index	**0**
Source IP Group	**2**
Destination IP Group	**1**
Destination Username Prefix	***** (asterisk sign)

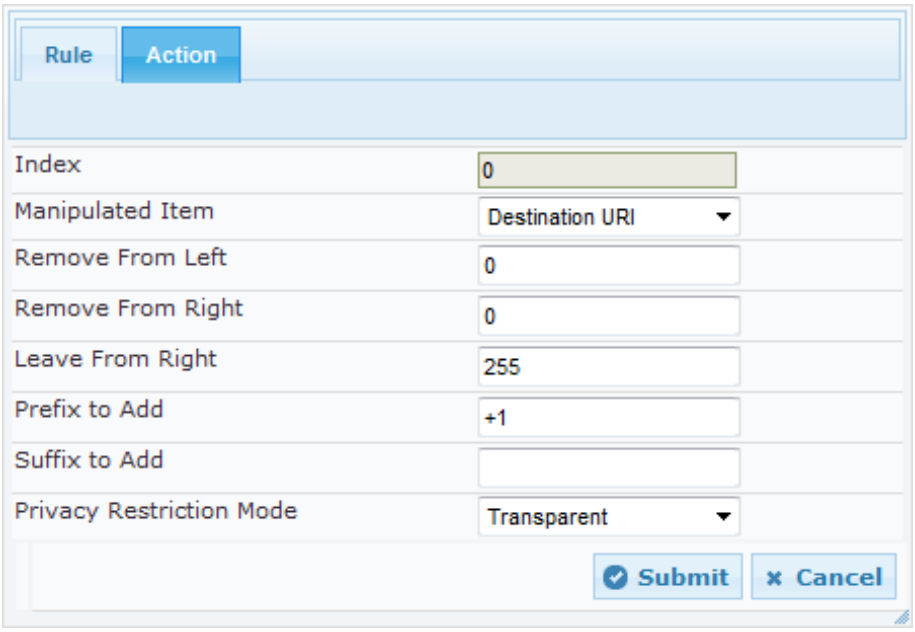
Figure 4-54: Configuring IP-to-IP Outbound Manipulation Rule – Rule Tab



4. Click the **Action** tab, and then configure the parameters as follows:

Parameter	Value
Manipulated Item	**Destination URI**
Prefix to Add	**+1** (plus sign)

Figure 4-55: Configuring IP-to-IP Outbound Manipulation Rule - Action Tab

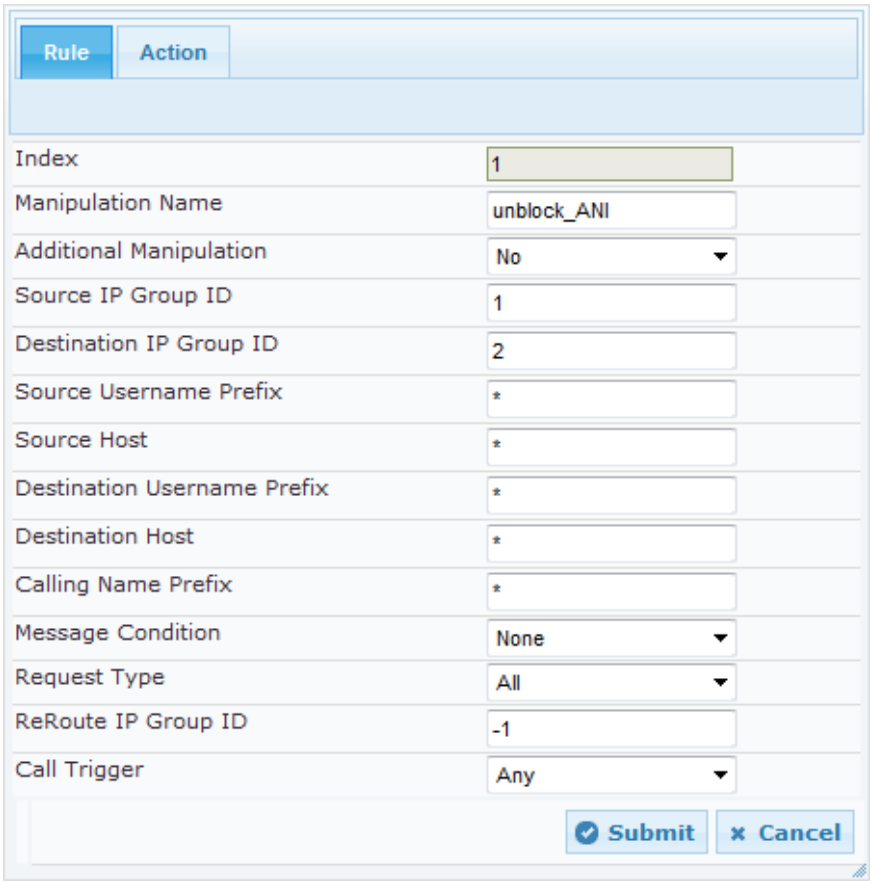


5. Click **Submit**.
6. Click **Add**.

7. Click the **Rule** tab, and then configure the parameters as follows:

Parameter	Value
Index	**1**
Source IP Group	**1**
Destination IP Group	**2**
Destination Username Prefix	***** (asterisk sign)

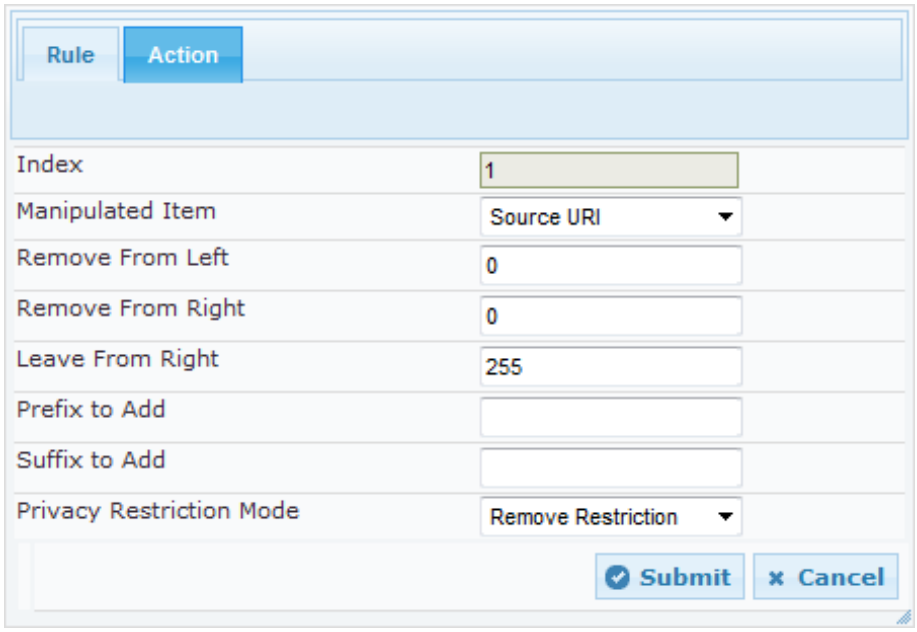
Figure 4-56: Configuring IP-to-IP Outbound Manipulation Rule – Rule Tab



8. Click the **Action** tab, and then configure the parameters as follows:

Parameter	Value
Manipulated Item	**Source URI**
Privacy Restriction Mode	**Remove Restriction**

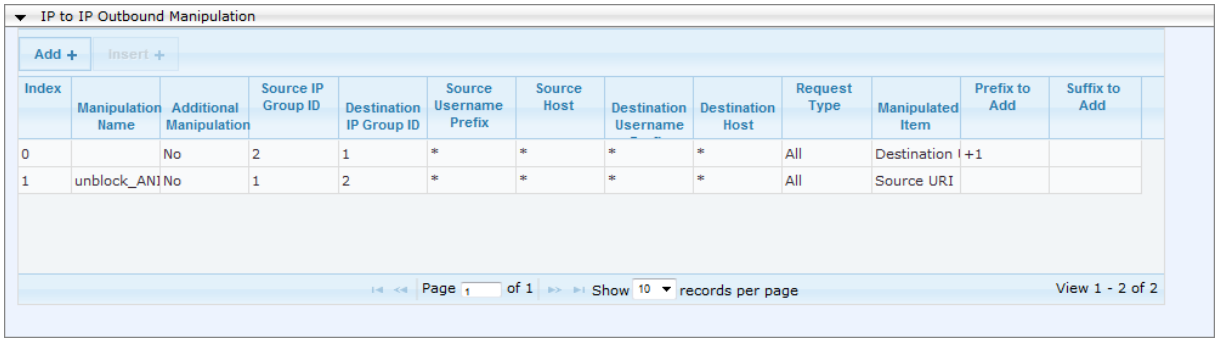
Figure 4-57: Configuring IP-to-IP Outbound Manipulation Rule - Action Tab



9. Click **Submit**.

The figure below shows an example of configured IP-to-IP outbound manipulation rules for calls between IP Group 1 (i.e., Lync Server 2013) and IP Group 2 (i.e., Verizon SIP Trunk):

Figure 4-58: Example of Configured IP-to-IP Outbound Manipulation Rules



Rule Index	Description
0	Calls from IP Group 2 to IP Group 1 with any destination number (*), add "+1" to the prefix of the destination number.
1	Calls from IP Group 1 to IP Group 2 change the Privacy Restriction to allow presentation.

4.13 Step 13: Configure Message Manipulation Rules

This step shows how to configure SIP message manipulation rules. SIP message manipulation rules can include insertion, removal, and/or modification of SIP headers. Manipulation rules are grouped into Manipulation Sets, enabling you to apply multiple rules to the same SIP message (IP entity).

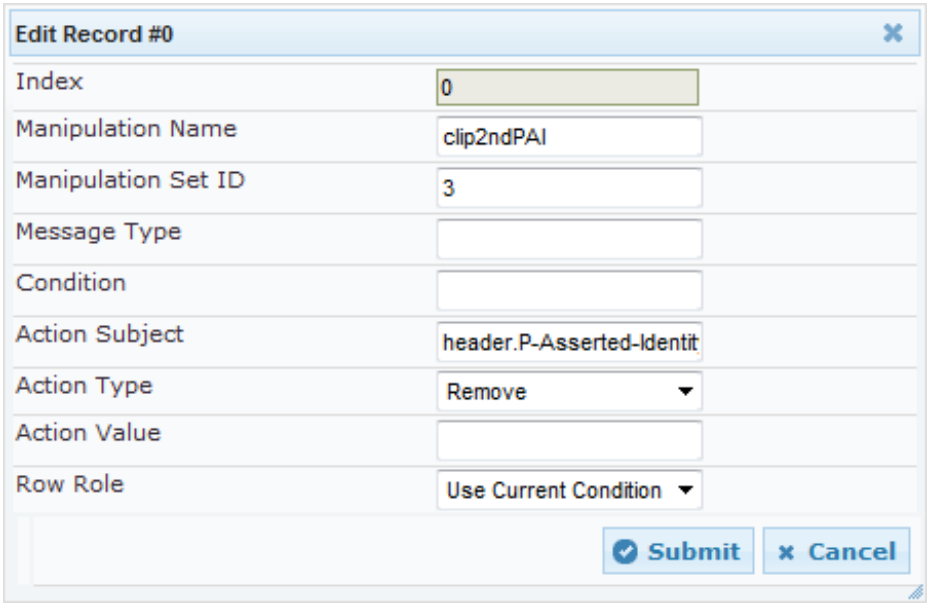
After configuring SIP message manipulation rules, you must assign them to the relevant IP Group (in the IP Group table) and determine whether they must be applied to inbound or outbound messages.

➢ **To configure a SIP message manipulation rule:**

1. Open the Message Manipulations page (**Configuration** tab > **VoIP** menu > **SIP Definitions** > **Msg Policy & Manipulation** > **Message Manipulations**).
2. Select **Add +**
3. Configure a new manipulation rule (Manipulation Set 3) for Verizon SIP Trunk. This rule applies to messages being sent to the Verizon SIP trunk (IP Group 2), calls initiated by the Lync Server 2013 (IP Group 1) which contain a long PAI. The SBC separates the P-Asserted Identity header into 2 separate PAI headers. This removes the second P-Asserted Identity header on the outgoing message towards the Verizon SIP Trunk.

Parameter	Value
Index	**0**
Manipulation Name	Arbitrary descriptive name
Manipulation Set ID	**3**
Action Subject	**header.P-Asserted-Identity.1**
Action Type	**Remove**

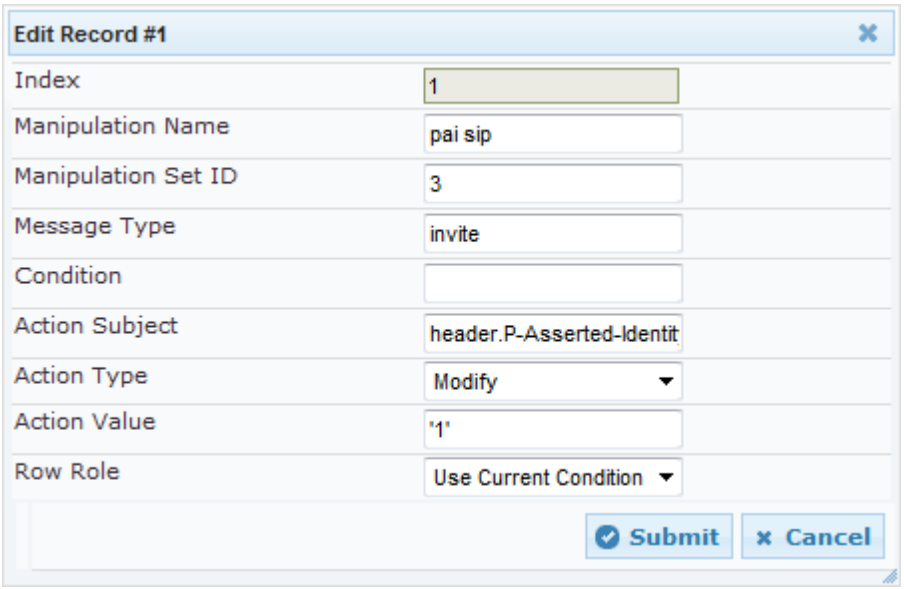
Figure 4-59: Configuring SIP Message Manipulation Rule 0 (for Verizon Sip Trunk)



4. Select **Add +**
5. Configure a new manipulation rule (Manipulation Set 3) for Verizon SIP Trunk. This rule applies to messages being sent to the Verizon SIP trunk (IP Group 2), for calls initiated by the Lync Server 2013 (IP Group 1) with the url type of 'Tel" rather than 'Sip'. This converts the header type for proper interworking towards the Verizon SIP Trunk.

Parameter	Value
Index	**1**
Manipulation Name	Arbitrary descriptive name
Manipulation Set ID	**3**
Message Type	**invite**
Action Subject	**header.P-Asserted-Identity.url.type**
Action Type	**Modify**
Action Value	**'1'**

Figure 4-60: Configuring SIP Message Manipulation Rule 1 (for Verizon SIP Trunk)

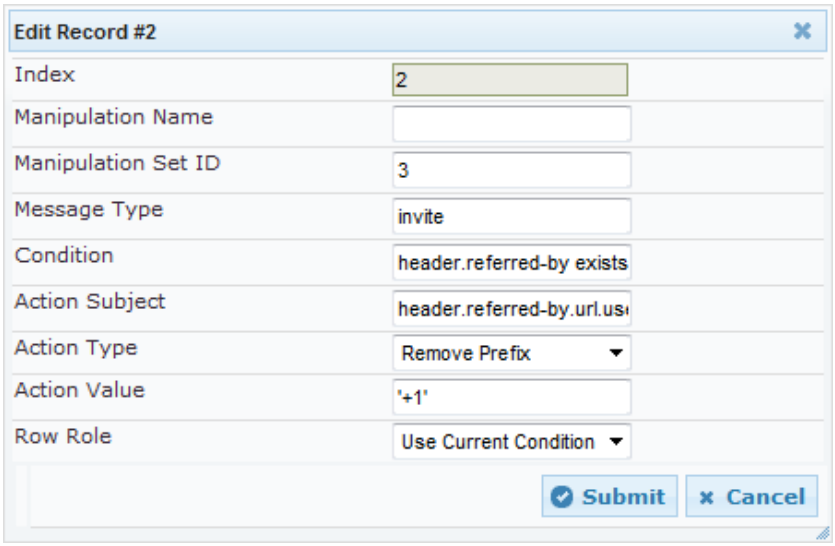


6. Click the **Submit** button.

7. Select **Add +**
8. Configure a new manipulation rule (Manipulation Set 3) for Verizon SIP Trunk. This rule applies to messages being sent to the Verizon SIP trunk (IP Group 2), for calls initiated by the Lync Server 2013 (IP Group 1) that include a referred-by header. This modifying Action Value modifies the Referred-By header, causing the E-SBC to remove the '+1' prefix from the Referred-By Header, if the Referred-By header exists.

Parameter	Value
Index	**2**
Manipulation Name	Arbitrary descriptive name
Manipulation Set ID	**3**
Message Type	**Invite**
Condition	**header.referred-by exists**
Action Subject	**header.referred-by.url.user**
Action Type	**Remove Prefix**
Action Value	**'+1'**

Figure 4-61: Configuring SIP Message Manipulation Rule 2 (for Verizon SIP Trunk)

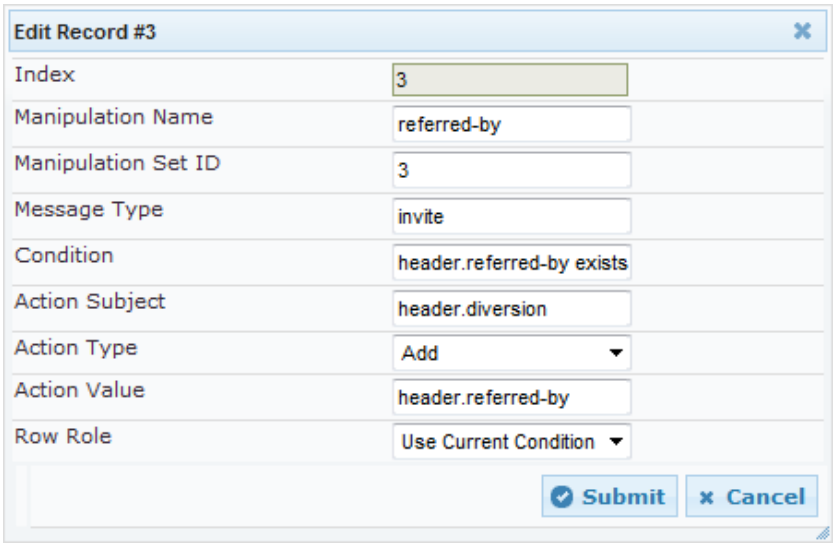


9. Click the **Submit** button.

10. Select **Add +**
11. Configure a new manipulation rule (Manipulation Set 3) for Verizon SIP Trunk. This rule applies to messages being sent to the Verizon SIP trunk (IP Group 2), when an invite containing a referred-by header is initiated by the Lync Server 2013 (IP Group 1). This adds an Action Value to add a Diversion header containing that which was received within the Referred-By header, that causing the E-SBC to add a Diversion Header towards the SIP Trunk.

Parameter	Value
Index	**3**
Manipulation Name	Arbitrary descriptive name
Manipulation Set ID	**3**
Message Type	**Invite**
Condition	**header.referred-by exists**
Action Subject	**header.diversion**
Action Type	**Add**
Action Value	**header.referred-by**

Figure 4-62: Configuring SIP Message Manipulation Rule 3 (for Verizon SIP Trunk)

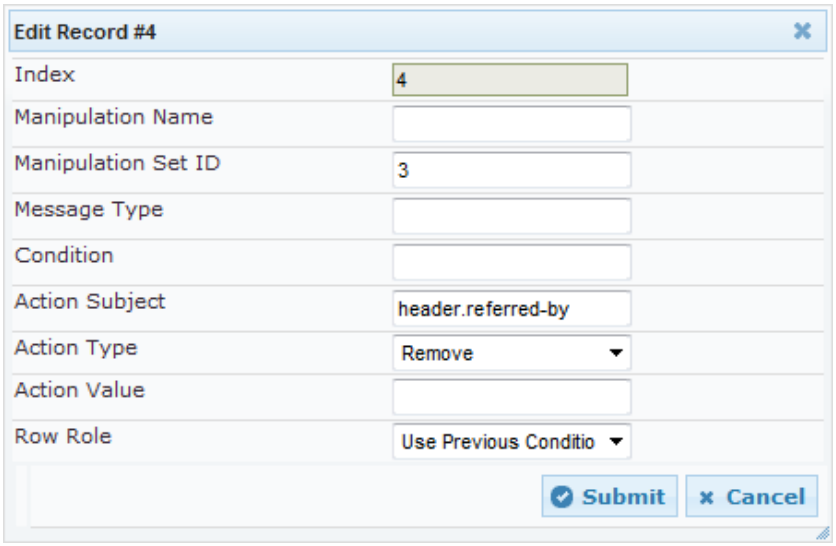


12. Click the **Submit** button.

13. Select **Add +**
14. Configure a new rule as an extension manipulation rule to the previous rule (Manipulation Set 3) for Verizon SIP Trunk. This rule applies to messages being sent to the Verizon SIP trunk (IP Group 2), only if the previous rule was also applied. This removes the referred_by header from the outgoing invite towards the SIP Trunk.

Parameter	Value
Index	**4**
Manipulation Name	Arbitrary descriptive name
Manipulation Set ID	**3**
Action Subject	**header.referred-by**
Action Type	**Remove**
Row Rule	**Use Previous Rule**

Figure 4-63: Configuring SIP Message Manipulation Rule 4 (for Verizon SIP Trunk)

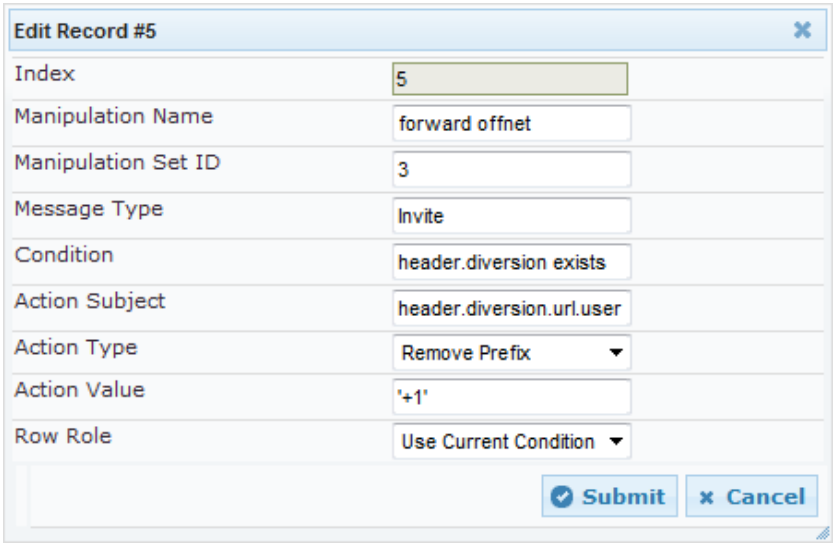


15. Click the **Submit** button.

16. Select **Add +**
17. Configure a new manipulation rule (Manipulation Set 3) for Verizon SIP Trunk. This rule applies to messages being sent to the Verizon SIP trunk (IP Group 2), for PSTN originated call being forwarded back to the PSTN initiated by the Lync Server 2013 (IP Group 1). This removes a prefix Action Value containing '+1' from the Diversion header, causing the E-SBC to manipulate the Diversion Header towards the SIP Trunk.

Parameter	Value
Index	**5**
Manipulation Name	Arbitrary descriptive name
Manipulation Set ID	**3**
Message Type	**Invite**
Condition	**header.diversion exists**
Action Subject	**header.diversion.url.user**
Action Type	**Remove Prefix**
Action Value	**'+1'**

Figure 4-64: Configuring SIP Message Manipulation Rule 5 (for Verizon SIP Trunk)

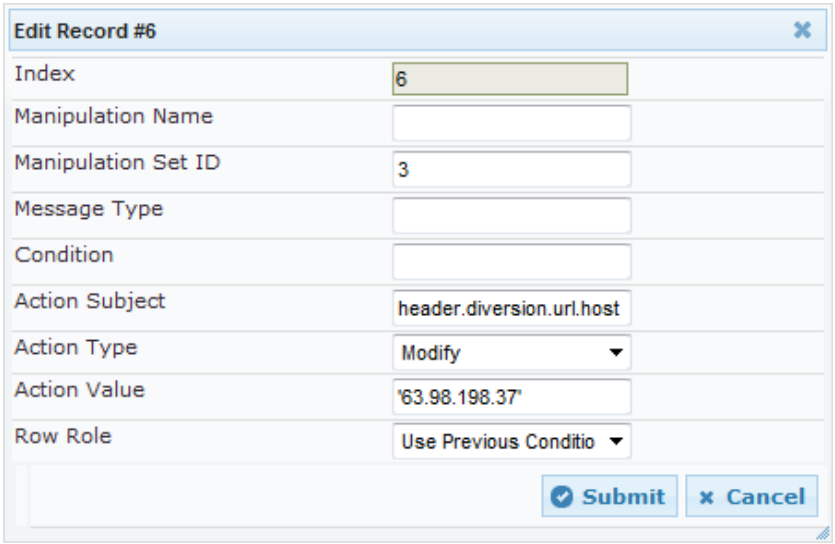


18. Click the **Submit** button.

19. Select **Add +**
20. Configure a new manipulation rule as a continuation rule to the previous rule (Manipulation Set 3) for Verizon SIP Trunk. This rule applies only if the previous rule was applied. This modifies an Action Value to change the URL host of the outgoing Diversion header to reflect that of the E-SBC host towards the SIP Trunk.

Parameter	Value
Index	**6**
Manipulation Name	Arbitrary descriptive name
Manipulation Set ID	**3**
Action Subject	**header.diversion.url.host**
Action Type	**Modify**
Action Value	**'63.98.198.37'**
Row Rule	**User Previous Condition**

Figure 4-65: Configuring SIP Message Manipulation Rule 6 (for Verizon SIP Trunk)

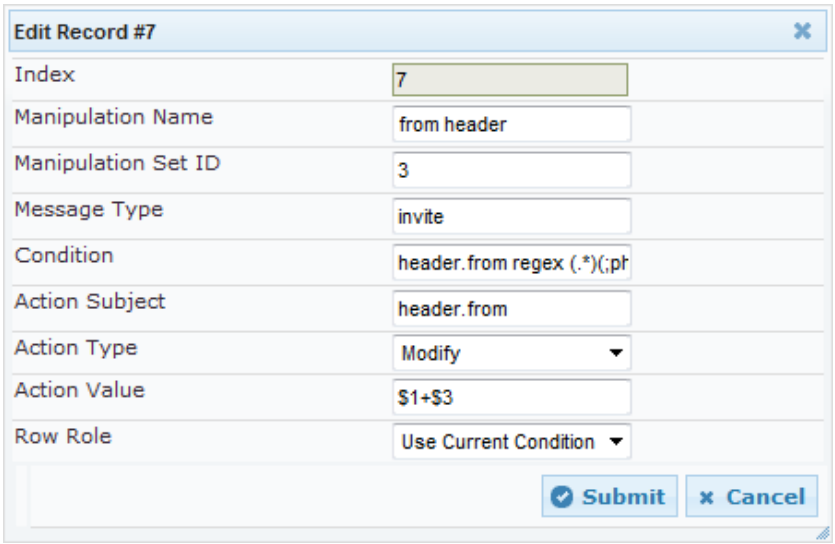


21. Click the **Submit** button.

22. Select **Add +**
23. Configure a new manipulation rule (Manipulation Set 3) for Verizon SIP Trunk. This rule applies to messages being sent to the Verizon SIP trunk (IP Group 2), for calls initiated by the Lync Server 2013 (IP Group 1). This modifies an Action Value of the From header, causing the E-SBC to remove the 'phone-context=enterprise' from the From Header towards the SIP Trunk.

Parameter	Value
Index	**7**
Manipulation Name	Arbitrary descriptive name
Manipulation Set ID	**3**
Message Type	**invite**
Condition	**header.from regex (.*)(;phone-context=enterprise)(.*)**
Action Subject	**header.from**
Action Type	**Modify**
Action Value	**$1+$3**

Figure 4-66: Configuring SIP Message Manipulation Rule 7 (for Verizon SIP Trunk)

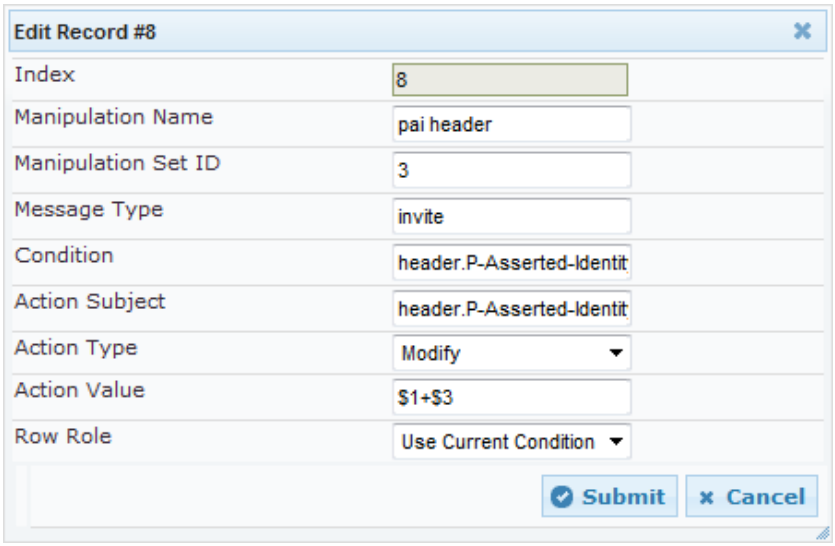


24. Click the **Submit** button.

25. Select **Add +**
26. Configure a new manipulation rule (Manipulation Set 3) for Verizon SIP Trunk. This rule applies to messages being sent to the Verizon SIP trunk (IP Group 2), for calls initiated by the Lync Server 2013 (IP Group 1). This modifies an Action Value of the P-Asserted-Identity header, causing the E-SBC to remove the 'phone-context=enterprise' from the P-Asserted-Identity Header towards the SIP Trunk.

Parameter	Value
Index	**8**
Manipulation Name	Arbitrary descriptive name
Manipulation Set ID	**3**
Message Type	**invite**
Condition	**header.P-Asserted-Identity regex (.*)(;phone-context=enterprise)(.*)**
Action Subject	**header.P-Asserted-Identity**
Action Type	**Modify**
Action Value	**$1+$3**

Figure 4-67: Configuring SIP Message Manipulation Rule 8 (for Verizon SIP Trunk)

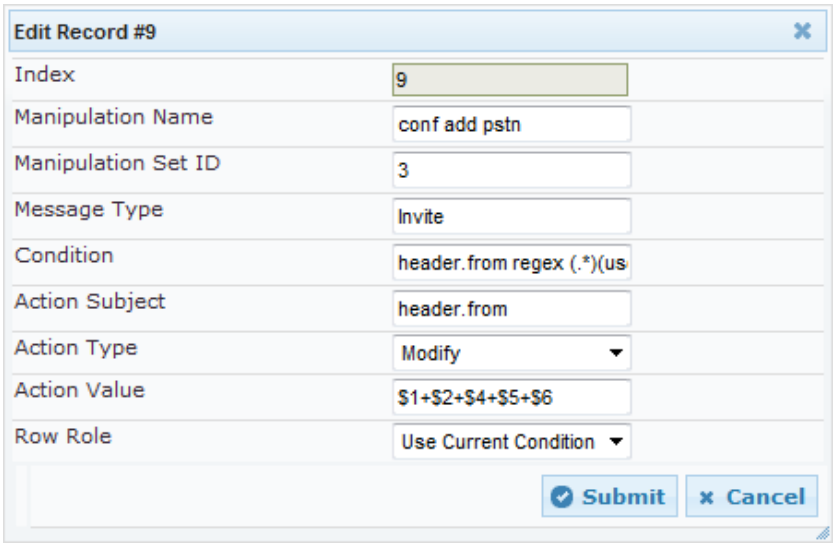


27. Click the **Submit** button.

28. Select **Add +**
29. Configure a new manipulation rule (Manipulation Set 3) for Verizon SIP Trunk. This rule applies to messages being sent to the Verizon SIP trunk (IP Group 2), for conference bridge initiated calls to add a PSTN user by the Lync Server 2013 (IP Group 1). This modifying Action Value modifies the From header into a format which the Verizon SIP Trunk accepts as the invite is sent towards the SIP Trunk.

Parameter	Value
Index	**9**
Manipulation Name	Arbitrary descriptive name
Manipulation Set ID	**3**
Message Type	**Invite**
Condition	**header.from regex (.*)(user=phone)(.*)(>)(;tag=)(.*)**
Action Subject	**header.from**
Action Type	**Modify**
Action Value	**$1+$2+$4+$5+$6**

Figure 4-68: Configuring SIP Message Manipulation Rule 9 (for Verizon SIP Trunk)

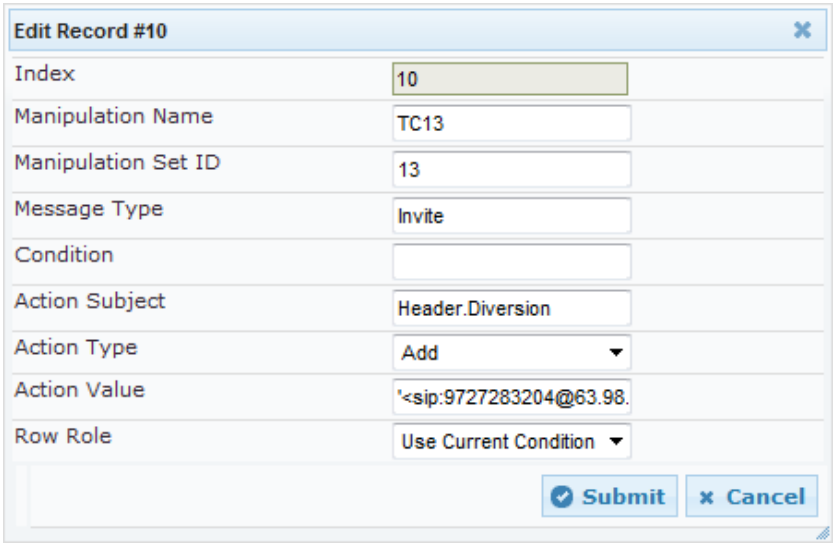


30. Click the **Submit** button.

31. Select **Add +**
32. Configure a new manipulation rule (Manipulation Set 3) for Verizon SIP Trunk. This rule applies to Invite messages being sent to the Verizon SIP trunk (IP Group 2), when Unscreened ANI Delivery is required for calls initiated by the Lync Server 2013 (IP Group 1). This adds an Action Value to include an Unscreened ANI to be delivered within a Diversion header towards the SIP Trunk.

Parameter	Value
Index	**10**
Manipulation Name	Arbitrary descriptive name
Manipulation Set ID	**13** (set to **3** if required, otherwise it will not be used if not associated)
Message Type	**invite**
Condition	**header.history-info.0==regex.(<.*)(user=phone)(>)(.*)**
Action Subject	**Header.Diversion**
Action Type	**Add**
Action Value	**'<sip:9727283204@63.98.198.37>;reason=unknown;privacy="off"'**

Figure 4-69: Configuring SIP Message Manipulation Rule 10 (for Verizon SIP Trunk)

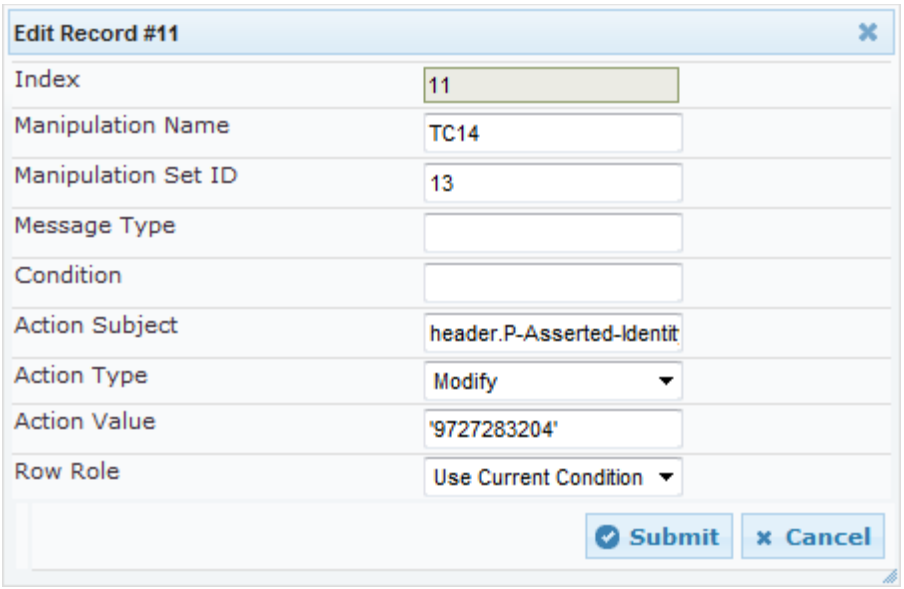


33. Click the **Submit** button.

34. Configure another manipulation rule (Manipulation Set 3) for Verizon SIP Trunk. This rule applies to messages being sent to the Verizon SIP trunk (IP Group 2), when Unscreened ANI Delivery is required for calls initiated by the Lync Server 2013 (IP Group 1). This adds an Action Value to include an Unscreened ANI to be delivered within a P-Asserted-Identity header towards the SIP Trunk.

Parameter	Value
Index	**11**
Manipulation Name	Arbitrary descriptive name
Manipulation Set ID	**13** (set to **3** if required, otherwise it will not be used if not associated)
Action Subject	**header.P-Asserted-Identity.url.user**
Action Type	**Modify**
Action Value	**'9727283204'**

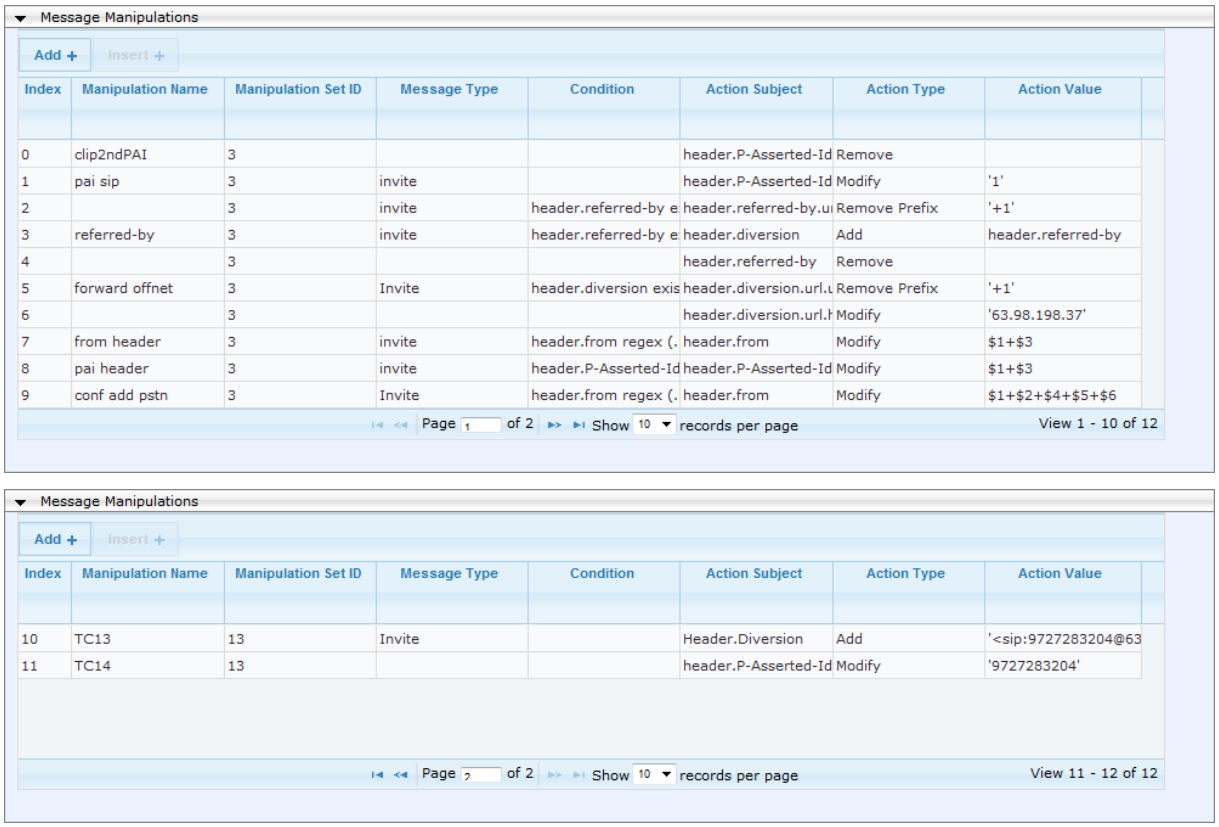
Figure 4-70: Configuring SIP Message Manipulation Rule 11 (for Verizon SIP Trunk)



35. Click the **Submit** button.

Figure 4-71 shows an example of configured SIP Message Manipulation rules.

Figure 4-71: Example of Configured SIP Message Manipulation Rules



Message Manipulations

Add + | Insert +

Index	Manipulation Name	Manipulation Set ID	Message Type	Condition	Action Subject	Action Type	Action Value
0	clip2ndPAI	3			header.P-Asserted-Id	Remove	
1	pai sip	3	invite		header.P-Asserted-Id	Modify	'1'
2		3	invite	header.referred-by e	header.referred-by.u	Remove Prefix	'+1'
3	referred-by	3	invite	header.referred-by e	header.diversion	Add	header.referred-by
4		3			header.referred-by	Remove	
5	forward offnet	3	Invite	header.diversion exis	header.diversion.url.u	Remove Prefix	'+1'
6		3			header.diversion.url.h	Modify	'63.98.198.37'
7	from header	3	invite	header.from regex (.	header.from	Modify	$1+$3
8	pai header	3	invite	header.P-Asserted-Id	header.P-Asserted-Id	Modify	$1+$3
9	conf add pstn	3	Invite	header.from regex (.	header.from	Modify	$1+$2+$4+$5+$6

Page 1 of 2 Show 10 records per page View 1 - 10 of 12

Message Manipulations

Add + | Insert +

Index	Manipulation Name	Manipulation Set ID	Message Type	Condition	Action Subject	Action Type	Action Value
10	TC13	13	Invite		Header.Diversion	Add	'<sip:9727283204@63
11	TC14	13			header.P-Asserted-Id	Modify	'9727283204'

Page 2 of 2 Show 10 records per page View 11 - 12 of 12

The table below includes SIP message manipulation rules which are bound together by commonality via the Manipulation Set IDs which are executed for messages sent to the Verizon SIP Trunk (IP Group 2). These rules are specifically required to enable correct interworking between Verizon SIP Trunk and Lync Server 2013. The specific items are needed to support the interoperability. Refer to the *User's Manual* for further details concerning the full capabilities of header manipulation.

Rule Index	Reason	Description
0	Removal of second PAI	Microsoft Lync 2013 sends a long P-Asserted-Identity which the E-SBC changes into two separate PAI headers. This rule is utilized to remove the second PAI.
1	PAI header URL type to SIP	This rule is utilized to change the URL type from 'TEL' to 'SIP' for the P-Asserted-Identity Header.
2	Remove '+1' prefix from referred-by header	Since Verizon SIP Trunk does not support "Referred-By" Header in SIP INVITE messages, the rule is needed to first modify it. To perform this, the prefix of '+1' is removed.
3	Add diversion header for transferred calls	Since Verizon SIP Trunk does not support "Referred-By" Header in SIP INVITE messages, the rule is needed to copy the contents of it to a new Diversion header.
4	Verizon SIP Trunk not supports "Referred-By" Header in SIP INVITE messages, so it is needed to remove it.	Since Verizon SIP Trunk does not support "Referred-By" Header in SIP INVITE messages, the rule is needed to remove it. To perform this rule; manipulation rule 3 must first be utilized for the Verizon SIP Trunk.

Rule Index	Reason	Description
5	Remove '+1' prefix from Diversion header	If a Diversion Header exists within an Invite message, the following rule removes prefix '+1' from the User part in Diversion Header if it is found to have this.
6	Set the URL host of the Diversion header to that of the E-SBC WAN	Rule replaces Host part in Diversion Header with value of the WAN address of the E-SBC that is recognized by the Verizon SIP Trunk. To perform this rule; manipulation rule 5 must first be utilized for the Verizon SIP Trunk.
7	From header normalization	This rule normalizes the From header by removing the '**phone-context=enterprise'** from the header.
8	P-Asserted-Identity normalization from header normalization	This rule normalizes the P-Asserted-Identity header by removing the '**phone-context=enterprise'** from the header.
9	From header normalization	This rule normalizes the From header when adding a PSTN user from a conference bridge.
10	Verizon Unscreened ANI delivery Option 1	Optional usage to insert an Unscreened ANI within a Diversion Header. To utilize this method, set the Message Manipulation ID to **3** for this rule.
11	Verizon Unscreened ANI delivery Option 2	Optional usage to insert an Unscreened ANI within a P-Asserted-Identity Header. To utilize this method, set the Message Manipulation ID to **3** for this rule.

36. Assign Manipulation Set IDs 3 to IP Group 2:

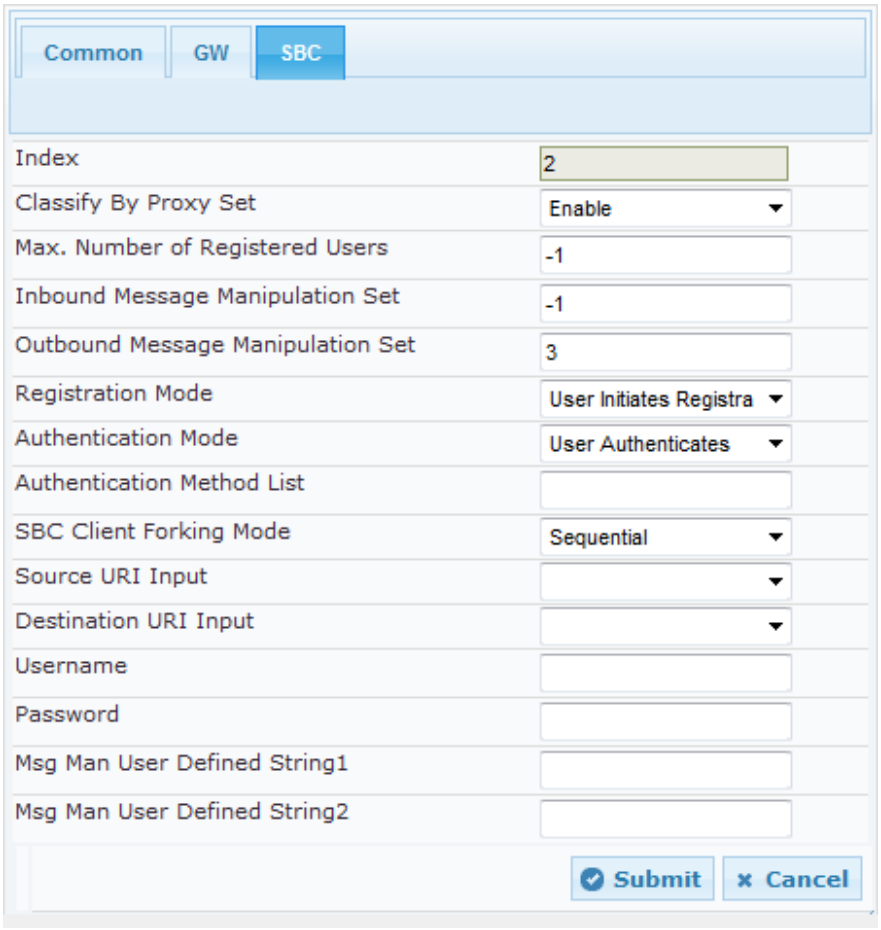
a. Open the IP Group Table page (**Configuration** tab > **VoIP** menu > **VoIP Network** > **IP Group Table**).

b. Select the row of IP Group 2, and then click **Edit**.

c. Click the **SBC** tab.

d. Set the 'Outbound Message Manipulation Set' field to **3**.

Figure 4-72: Assigning Manipulation Set to IP Group 2



e. Click the **Submit** button.

4.14 Step 14: Configure Miscellaneous E-SBC Settings

This section shows how to configure miscellaneous E-SBC settings.

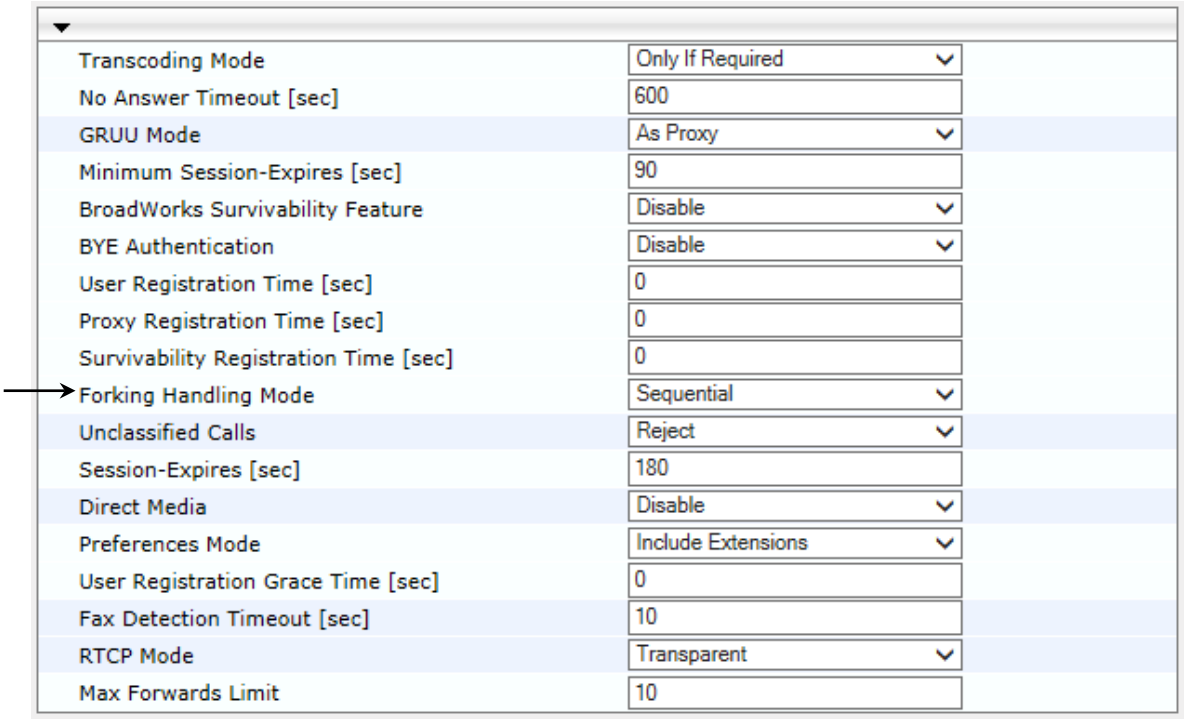
4.14.1 Step 14a: Configure Call Forking Mode

This step shows how to configure the E-SBC's handling of SIP 18x responses received for call forking of INVITE messages. For the interoperability test topology, if 18x with SDP is received, the E-SBC opens a voice stream according to the received SDP. The E-SBC reopens the stream according to subsequently received 18x responses with SDP, or plays a ringback tone if a 180 response without SDP is received. It's mandatory to set this field for the Lync Server 2013 environment.

➢ **To configure call forking:**

1. Open the General Settings page (**Configuration** tab > **VoIP** menu > **SBC** > **General Settings**).
2. From the 'SBC Forking Handling Mode' drop-down list, select **Sequential**.

Figure 4-73: Configuring Forking Mode



3. Click **Submit**.

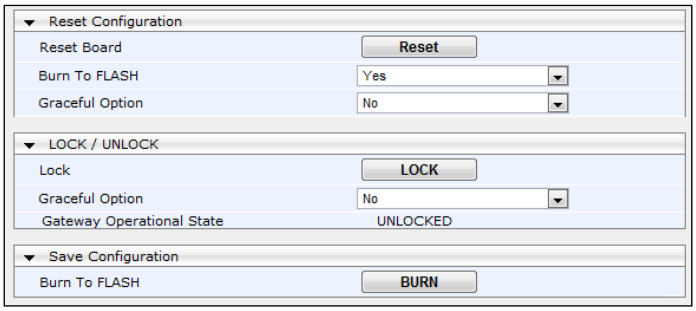
4.15 Step 15: Reset the E-SBC

After completing configuration of the E-SBC, save ("burn") the configuration to the E-SBC's flash memory with a reset for the settings to take effect.

➢ **To save the configuration to flash memory:**

1. Open the Maintenance Actions page (**Maintenance** tab > **Maintenance** menu > **Maintenance Actions**).

Figure 4-74: Resetting the E-SBC



2. Ensure that the 'Burn to FLASH' field is set to **Yes** (default).
3. Click the **Reset** button.

A AudioCodes *ini* File

The *ini* configuration file of the E-SBC, corresponding to the Web-based configuration as described in Section 4 on page 31, is shown below:

Note: To load and save an *ini* file, use the Configuration File page (**Maintenance** tab > **Software Update** menu > **Configuration File**).

```
;**************
;** Ini File **
;**************

;Board: Mediant 1000
;HW Board Type: 47  FK Board Type: 71
;Serial Number: 2967088
;Slot Number: 1
;Software Version: 6.80A.018.005
;DSP Software Version: 624AE3=> 660.10
;Board IP Address: 10.133.4.43
;Board Subnet Mask: 255.255.0.0
;Board Default Gateway: 10.133.4.1
;Ram size: 497M   Flash size: 64M
;Num of DSP Cores: 8  Num DSP Channels: 30
;Num of physical LAN ports: 7
;Profile: NONE
;Key features:;Board Type: Mediant 1000 ;Coders: G723 G729 G728 NETCODER GSM-FR
GSM-EFR AMR EVRC-QCELP G727 ILBC EVRC-B AMR-WB G722 EG711 MS_RTA_NB MS_RTA_WB
SILK_NB SILK_WB SPEEX_NB SPEEX_WB ;PSTN FALLBACK Supported ;E1Trunks=4 ;T1Trunks=4
;DSP Voice features: IpmDetector RTCP-XR ;Eth-Port=6 ;DATA features: ;IP Media:
Conf VoicePromptAnnounc(H248.9) TrunkTesting ;Channel Type: RTP DspCh=30
IPMediaDspCh=30 ;PSTN Protocols: IUA=4 ;Security: IPSEC MediaEncryption
StrongEncryption EncryptControlProtocol ;Control Protocols: MSFT CLI TRANSCODING=20
TestCall=10 CODER-TRANSCODING=531 MGCP MEGACO H323 SIP TPNCP SASurvivability
SBC=120 ;Default features:;Coders: G711 G726;

;--------- Mediant 1000 HW components ---------
;
; Slot # : Module type : # of ports : # of DSPs
;----------------------------------------------
;      1 : Empty
;      2 : FALC56      :          1 :         2
;      3 : Empty
;      4 : Empty
;      5 : Empty
;      6 : Empty
;----------------------------------------------



[BSP Params]

PCMLawSelect = 3
BaseUDPPort = 20010
PREMIUMSERVICECLASSCONTROLDIFFSERV = 28
UdpPortSpacing = 10


[MGCP Params]


[MEGACO Params]
```

```
EP_Num_0 = 0
EP_Num_1 = 1
EP_Num_2 = 1
EP_Num_3 = 0
EP_Num_4 = 0

[PSTN Params]


[SS7 Params]


[Voice Engine Params]

CNGDetectorMode = 1
DisableRTCPRandomize = 1
ENABLEMEDIASECURITY = 1
SRTPTxPacketMKISize = 1
CallProgressTonesFilename = 'usa_tones_13.dat'

[WEB Params]

LogoWidth = '145'
HTTPSCipherString = 'RC4:EXP'
;HTTPSCertFileName is hidden but has non-default value
;HTTPSRootFileName is hidden but has non-default value

[SIP Params]

MEDIACHANNELS = 30
SIPDESTINATIONPORT = 5067
GWDEBUGLEVEL = 5
PROXYREDUNDANCYMODE = 1
TCPLOCALSIPPORT = 5068
TLSLOCALSIPPORT = 5067
MEDIASECURITYBEHAVIOUR = 3
FAXCNGMODE = 1
ENABLESBCAPPLICATION = 1
MSLDAPPRIMARYKEY = 'telephoneNumber'
SBCMAXFORWARDSLIMIT = 70
T38FAXSESSIONIMMEDIATESTART = 1
ENABLESYMMETRICMKI = 1
SBCPREFERENCESMODE = 1
SBCFORKINGHANDLINGMODE = 1
ENERGYDETECTORCMD = 587202560
ANSWERDETECTORCMD = 10485760

[SCTP Params]


[VXML Params]


[IPsec Params]


[Audio Staging Params]


[SNMP Params]


[ PhysicalPortsTable ]

FORMAT PhysicalPortsTable_Index = PhysicalPortsTable_Port, PhysicalPortsTable_Mode,
PhysicalPortsTable_NativeVlan, PhysicalPortsTable_SpeedDuplex,
```

```
PhysicalPortsTable_PortDescription, PhysicalPortsTable_GroupMember,
PhysicalPortsTable_GroupStatus;
PhysicalPortsTable 0 = "GE_0_1", 1, 1, 4, "LAN Port#1", "GROUP_1", "Active";
PhysicalPortsTable 1 = "GE_0_2", 1, 1, 4, "LAN Port#2", "GROUP_1", "Redundant";
PhysicalPortsTable 2 = "GE_7_1", 1, 2, 4, "WAN Port#1", "GROUP_2", "Active";
PhysicalPortsTable 3 = "GE_7_2", 1, 2, 4, "WAN Port#2", "GROUP_2", "Redundant";
PhysicalPortsTable 4 = "GE_7_3", 1, 3, 4, "User Port #4", "GROUP_3", "Active";
PhysicalPortsTable 5 = "GE_7_4", 1, 3, 4, "User Port #5", "GROUP_3", "Redundant";

[ \PhysicalPortsTable ]


[ EtherGroupTable ]

FORMAT EtherGroupTable_Index = EtherGroupTable_Group, EtherGroupTable_Mode,
EtherGroupTable_Member1, EtherGroupTable_Member2;
EtherGroupTable 0 = "GROUP_1", 2, "GE_0_1", "GE_0_2";
EtherGroupTable 1 = "GROUP_2", 2, "GE_7_1", "GE_7_2";
EtherGroupTable 2 = "GROUP_3", 2, "GE_7_3", "GE_7_4";
EtherGroupTable 3 = "GROUP_4", 0, "", "";
EtherGroupTable 4 = "GROUP_5", 0, "", "";
EtherGroupTable 5 = "GROUP_6", 0, "", "";

[ \EtherGroupTable ]


[ DeviceTable ]

FORMAT DeviceTable_Index = DeviceTable_VlanID, DeviceTable_UnderlyingInterface,
DeviceTable_DeviceName;
DeviceTable 0 = 1, "GROUP_1", "vlan 1";
DeviceTable 1 = 2, "GROUP_2", "vlan 2";
DeviceTable 2 = 3, "GROUP_3", "vlan 3";

[ \DeviceTable ]


[ InterfaceTable ]

FORMAT InterfaceTable_Index = InterfaceTable_ApplicationTypes,
InterfaceTable_InterfaceMode, InterfaceTable_IPAddress,
InterfaceTable_PrefixLength, InterfaceTable_Gateway, InterfaceTable_VlanID,
InterfaceTable_InterfaceName, InterfaceTable_PrimaryDNSServerIPAddress,
InterfaceTable_SecondaryDNSServerIPAddress, InterfaceTable_UnderlyingDevice;
InterfaceTable 0 = 6, 10, 10.133.4.43, 16, 10.133.4.1, 1, "Voice", 10.133.4.40,
0.0.0.0, "vlan 1";
InterfaceTable 1 = 5, 10, 63.98.198.37, 28, 63.98.198.38, 2, "Public", 198.6.1.2,
198.6.1.3, "vlan 2";

[ \InterfaceTable ]


[ DspTemplates ]

;
;  *** TABLE DspTemplates ***
; This table contains hidden elements and will not be exposed.
; This table exists on board and will be saved during restarts.
;

[ \DspTemplates ]


[ CpMediaRealm ]

FORMAT CpMediaRealm_Index = CpMediaRealm_MediaRealmName, CpMediaRealm_IPv4IF,
CpMediaRealm_IPv6IF, CpMediaRealm_PortRangeStart, CpMediaRealm_MediaSessionLeg,
```

```
CpMediaRealm_PortRangeEnd, CpMediaRealm_IsDefault, CpMediaRealm_QoeProfile,
CpMediaRealm_BWProfile;
CpMediaRealm 1 = "MRLan", "Voice", "", 21010, 10, 21100, 1, "", "";
CpMediaRealm 2 = "MRWan", "Public", "", 20010, 10, 20100, 0, "", "";

[ \CpMediaRealm ]


[ WebUsers ]

FORMAT WebUsers_Index = WebUsers_Username, WebUsers_Password, WebUsers_Status,
WebUsers_PwAgeInterval, WebUsers_SessionLimit, WebUsers_SessionTimeout,
WebUsers_BlockTime, WebUsers_UserLevel, WebUsers_PwNonce;
WebUsers 0 = "Admin",
"$1$na/8r5WWlJKSnZTCnpGcz8nMnZyIhIrShtaFgYuKjt2Ej9+OpPTy9Kb896Opr/L4+qms/rHgt+u2t+H
v6rvrvu0=", 1, 0, 2, 15, 60, 200, "491d805ea8ee4c45b389866b777e9061";
WebUsers 1 = "User",
"$1$vIyE3N3zpKD2/f3z966t+K/6qff65bOx6rHl5rTvvOjturvq79fW1dWB04fTid7d2NvZj9eVw8rBxMz
Bx5rPzZg=", 1, 0, 2, 15, 60, 50, "88406e8d54c36142d8dee7d55dbc8fac";

[ \WebUsers ]


[ SRD ]

FORMAT SRD_Index = SRD_Name, SRD_MediaRealm, SRD_IntraSRDMediaAnchoring,
SRD_BlockUnRegUsers, SRD_MaxNumOfRegUsers, SRD_EnableUnAuthenticatedRegistrations;
SRD 1 = "LanSRD", "MRLan", 0, 0, -1, 1;
SRD 2 = "WanSRD", "MRWan", 0, 0, -1, 1;

[ \SRD ]


[ ProxyIp ]

FORMAT ProxyIp_Index = ProxyIp_IpAddress, ProxyIp_TransportType,
ProxyIp_ProxySetId;
ProxyIp 0 = "63.79.179.178:5208", 0, 2;
ProxyIp 1 = "FE15.ilync15.local:5067", 2, 1;
ProxyIp 2 = "10.133.4.101:5060", 0, 3;

[ \ProxyIp ]


[ IpProfile ]

FORMAT IpProfile_Index = IpProfile_ProfileName, IpProfile_IpPreference,
IpProfile_CodersGroupID, IpProfile_IsFaxUsed, IpProfile_JitterBufMinDelay,
IpProfile_JitterBufOptFactor, IpProfile_IPDiffServ, IpProfile_SigIPDiffServ,
IpProfile_SCE, IpProfile_RTPRedundancyDepth, IpProfile_RemoteBaseUDPPort,
IpProfile_CNGmode, IpProfile_VxxTransportType, IpProfile_NSEMode,
IpProfile_IsDTMFUsed, IpProfile_PlayRBTone2IP, IpProfile_EnableEarlyMedia,
IpProfile_ProgressIndicator2IP, IpProfile_EnableEchoCanceller,
IpProfile_CopyDest2RedirectNumber, IpProfile_MediaSecurityBehaviour,
IpProfile_CallLimit, IpProfile_DisconnectOnBrokenConnection,
IpProfile_FirstTxDtmfOption, IpProfile_SecondTxDtmfOption, IpProfile_RxDTMFOption,
IpProfile_EnableHold, IpProfile_InputGain, IpProfile_VoiceVolume,
IpProfile_AddIEInSetup, IpProfile_SBCExtensionCodersGroupID,
IpProfile_MediaIPVersionPreference, IpProfile_TranscodingMode,
IpProfile_SBCAllowedMediaTypes, IpProfile_SBCAllowedCodersGroupID,
IpProfile_SBCAllowedVideoCodersGroupID, IpProfile_SBCAllowedCodersMode,
IpProfile_SBCMediaSecurityBehaviour, IpProfile_SBCRFC2833Behavior,
IpProfile_SBCAlternativeDTMFMethod, IpProfile_SBCAssertIdentity,
IpProfile_AMDSensitivityParameterSuit, IpProfile_AMDSensitivityLevel,
IpProfile_AMDMaxGreetingTime, IpProfile_AMDMaxPostSilenceGreetingTime,
IpProfile_SBCDiversionMode, IpProfile_SBCHistoryInfoMode,
IpProfile_EnableQSIGTunneling, IpProfile_SBCFaxCodersGroupID,
```

```
IpProfile_SBCFaxBehavior, IpProfile_SBCFaxOfferMode, IpProfile_SBCFaxAnswerMode,
IpProfile_SbcPrackMode, IpProfile_SBCSessionExpiresMode,
IpProfile_SBCRemoteUpdateSupport, IpProfile_SBCRemoteReinviteSupport,
IpProfile_SBCRemoteDelayedOfferSupport, IpProfile_SBCRemoteReferBehavior,
IpProfile_SBCRemote3xxBehavior, IpProfile_SBCRemoteMultiple18xSupport,
IpProfile_SBCRemoteEarlyMediaResponseType, IpProfile_SBCRemoteEarlyMediaSupport,
IpProfile_EnableSymmetricMKI, IpProfile_MKISize, IpProfile_SBCEnforceMKISize,
IpProfile_SBCRemoteEarlyMediaRTP, IpProfile_SBCRemoteSupportsRFC3960,
IpProfile_SBCRemoteCanPlayRingback, IpProfile_EnableEarly183,
IpProfile_EarlyAnswerTimeout, IpProfile_SBC2833DTMFPayloadType,
IpProfile_SBCUserRegistrationTime, IpProfile_ResetSRTPStateUponRekey,
IpProfile_AmdMode, IpProfile_SBCReliableHeldToneSource, IpProfile_GenerateSRTPKeys,
IpProfile_SBCPlayHeldTone, IpProfile_SBCRemoteHoldFormat,
IpProfile_SBCRemoteReplacesBehavior, IpProfile_SBCSDPPtimeAnswer,
IpProfile_SBCPreferredPTime, IpProfile_SBCUseSilenceSupp,
IpProfile_SBCRTPRedundancyBehavior, IpProfile_SBCPlayRBTToTransferee,
IpProfile_SBCRTCPMode, IpProfile_SBCJitterCompensation,
IpProfile_SBCRemoteRenegotiateOnFaxDetection, IpProfile_JitterBufMaxDelay;
IpProfile 1 = "Microsoft Lync", 1, 0, 0, 10, 10, 46, 40, 0, 0, 0, 0, 2, 0, 0, 0, 1,
-1, 1, 0, 1, -1, 0, 4, -1, 1, 1, 0, 0, "", 1, 0, 1, "", 1, -1, 2, 1, 0, 0, 0, 0, 8,
300, 400, 0, 0, 0, -1, 0, 0, 1, 3, 0, 1, 1, 0, 3, 2, 1, 2, 1, 1, 1, 1, 1, 0, 1, 0,
0, 0, 0, 1, 0, 1, 1, 0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 250;
IpProfile 2 = "Vzn Trunk", 1, 0, 1, 10, 10, 46, 28, 0, 0, 0, 0, 2, 0, 0, 0, 1, -1,
1, 0, 2, -1, 0, 4, -1, 1, 1, 0, 0, "", 2, 0, 0, "", 2, -1, 2, 2, 0, 0, 1, 0, 8,
300, 400, 1, 2, 0, 4, 2, 0, 1, 3, 0, 2, 2, 0, 3, 0, 0, 2, 1, 0, 0, 0, 0, 0, 0, 1,
0, 0, 0, 0, 0, 1, 0, 0, 1, 0, 0, 0, 0, 0, 0, 0, 0, 0, 250;
IpProfile 3 = "FAX ATA", 1, 0, 1, 10, 10, 46, 40, 0, 0, 0, 0, 2, 0, 0, 0, 0, -1, 1,
0, 3, -1, 1, 4, -1, 1, 1, 0, 0, "", -1, 0, 0, "", -1, -1, 0, 0, 0, 0, 0, 0, 8, 300,
400, 0, 0, 0, 4, 2, 0, 1, 3, 0, 2, 2, 1, 0, 0, 1, 0, 1, 1, 1, 0, 0, 0, 1, 0, 0, 0,
0, 0, 0, 1, 0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 250;

[ \IpProfile ]


[ ProxySet ]

FORMAT ProxySet_Index = ProxySet_ProxyName, ProxySet_EnableProxyKeepAlive,
ProxySet_ProxyKeepAliveTime, ProxySet_ProxyLoadBalancingMethod,
ProxySet_IsProxyHotSwap, ProxySet_SRD, ProxySet_ClassificationInput,
ProxySet_ProxyRedundancyMode, ProxySet_DNSResolveMethod,
ProxySet_KeepAliveFailureResp;
ProxySet 0 = "", 0, 60, 0, 0, 0, 0, -1, -1, "";
ProxySet 1 = "", 1, 30, 0, 1, 1, 0, -1, -1, "";
ProxySet 2 = "", 1, 60, 0, 0, 2, 0, -1, -1, "";
ProxySet 3 = "fax ata", 0, 60, 0, 0, 1, 0, -1, -1, "";

[ \ProxySet ]


[ IPGroup ]

FORMAT IPGroup_Index = IPGroup_Type, IPGroup_Description, IPGroup_ProxySetId,
IPGroup_SIPGroupName, IPGroup_ContactUser, IPGroup_EnableSurvivability,
IPGroup_ServingIPGroup, IPGroup_SipReRoutingMode, IPGroup_AlwaysUseRouteTable,
IPGroup_RoutingMode, IPGroup_SRD, IPGroup_MediaRealm, IPGroup_ClassifyByProxySet,
IPGroup_ProfileId, IPGroup_MaxNumOfRegUsers, IPGroup_InboundManSet,
IPGroup_OutboundManSet, IPGroup_RegistrationMode, IPGroup_AuthenticationMode,
IPGroup_MethodList, IPGroup_EnableSBCClientForking, IPGroup_SourceUriInput,
IPGroup_DestUriInput, IPGroup_ContactName, IPGroup_Username, IPGroup_Password,
IPGroup_UUIFormat, IPGroup_QOEProfile, IPGroup_BWProfile,
IPGroup_MediaEnhancementProfile, IPGroup_AlwaysUseSourceAddr,
IPGroup_MsgManUserDef1, IPGroup_MsgManUserDef2;
IPGroup 1 = 0, "Microsoft Lync", 1, "63.98.198.37", "", 0, -1, 0, 0, -1, 1,
"MRLan", 1, 1, -1, -1, -1, 0, 0, "", 0, -1, -1, "", "", "$1$gQ==", 0, "", "", "",
0, "", "";
```

```
IPGroup 2 = 0, "Vzn Trunk", 2, "icrcn1n0002.customer08.tsengr.com", "", 0, -1, 0,
0, -1, 2, "MRWan", 1, 2, -1, -1, 3, 0, 0, "", 0, -1, -1, "", "", "$1$gQ==", 0, "",
"", "", 0, "", "";
IPGroup 3 = 0, "fax ata", 3, "63.98.198.37", "", 0, -1, -1, 0, -1, 1, "MRLan", 1,
3, -1, -1, -1, 0, 0, "", 0, -1, -1, "", "", "$1$gQ==", 0, "", "", "", 0, "", "";

[ \IPGroup ]


[ IP2IPRouting ]

FORMAT IP2IPRouting_Index = IP2IPRouting_RouteName, IP2IPRouting_SrcIPGroupID,
IP2IPRouting_SrcUsernamePrefix, IP2IPRouting_SrcHost,
IP2IPRouting_DestUsernamePrefix, IP2IPRouting_DestHost, IP2IPRouting_RequestType,
IP2IPRouting_MessageCondition, IP2IPRouting_ReRouteIPGroupID, IP2IPRouting_Trigger,
IP2IPRouting_CallSetupRulesSetId, IP2IPRouting_DestType,
IP2IPRouting_DestIPGroupID, IP2IPRouting_DestSRDID, IP2IPRouting_DestAddress,
IP2IPRouting_DestPort, IP2IPRouting_DestTransportType,
IP2IPRouting_AltRouteOptions, IP2IPRouting_GroupPolicy, IP2IPRouting_CostGroup;
IP2IPRouting 0 = "", -1, "*", "*", "*", "*", 6, "", -1, 0, -1, 1, -1, "",
"internal", 0, -1, 0, 0, "";
IP2IPRouting 1 = "", 1, "*", "*", "*", "*", 0, "", -1, 0, -1, 0, 2, "2", "", 0, -1,
0, 0, "";
IP2IPRouting 2 = "SIP trunk2fax", 2, "*", "*", "4089908837", "*", 0, "", -1, 0, -1,
0, 3, "1", "", 0, -1, 0, 0, "";
IP2IPRouting 3 = "", 2, "*", "*", "*", "*", 0, "", -1, 0, -1, 0, 1, "1", "", 0, -1,
0, 0, "";
IP2IPRouting 4 = "ata fax", 3, "*", "*", "*", "*", 0, "", -1, 0, -1, 0, 2, "2", "",
0, -1, 0, 0, "";

[ \IP2IPRouting ]


[ SIPInterface ]

FORMAT SIPInterface_Index = SIPInterface_InterfaceName,
SIPInterface_NetworkInterface, SIPInterface_ApplicationType, SIPInterface_UDPPort,
SIPInterface_TCPPort, SIPInterface_TLSPort, SIPInterface_SRD,
SIPInterface_MessagePolicy, SIPInterface_TLSMutualAuthentication,
SIPInterface_TCPKeepaliveEnable, SIPInterface_ClassificationFailureResponseType,
SIPInterface_PreClassificationManSet;
SIPInterface 1 = "", "Voice", 2, 5060, 5068, 5067, 1, "", -1, 0, 500, -1;
SIPInterface 2 = "", "Public", 2, 5060, 0, 0, 2, "", -1, 0, 500, -1;

[ \SIPInterface ]


[ IPInboundManipulation ]

FORMAT IPInboundManipulation_Index = IPInboundManipulation_ManipulationName,
IPInboundManipulation_IsAdditionalManipulation,
IPInboundManipulation_ManipulationPurpose, IPInboundManipulation_SrcIPGroupID,
IPInboundManipulation_SrcUsernamePrefix, IPInboundManipulation_SrcHost,
IPInboundManipulation_DestUsernamePrefix, IPInboundManipulation_DestHost,
IPInboundManipulation_RequestType, IPInboundManipulation_ManipulatedURI,
IPInboundManipulation_RemoveFromLeft, IPInboundManipulation_RemoveFromRight,
IPInboundManipulation_LeaveFromRight, IPInboundManipulation_Prefix2Add,
IPInboundManipulation_Suffix2Add;
IPInboundManipulation 1 = "", 0, 0, 1, "+1", "*", "*", "*", 0, 0, 2, 0, 255, "",
"";
IPInboundManipulation 2 = "", 0, 0, 1, "*", "*", "+1", "*", 0, 1, 2, 0, 255, "",
"";

[ \IPInboundManipulation ]


[ IPOutboundManipulation ]
```

```
FORMAT IPOutboundManipulation_Index = IPOutboundManipulation_ManipulationName,
IPOutboundManipulation_IsAdditionalManipulation,
IPOutboundManipulation_SrcIPGroupID, IPOutboundManipulation_DestIPGroupID,
IPOutboundManipulation_SrcUsernamePrefix, IPOutboundManipulation_SrcHost,
IPOutboundManipulation_DestUsernamePrefix, IPOutboundManipulation_DestHost,
IPOutboundManipulation_CallingNamePrefix, IPOutboundManipulation_MessageCondition,
IPOutboundManipulation_RequestType, IPOutboundManipulation_ReRouteIPGroupID,
IPOutboundManipulation_Trigger, IPOutboundManipulation_ManipulatedURI,
IPOutboundManipulation_RemoveFromLeft, IPOutboundManipulation_RemoveFromRight,
IPOutboundManipulation_LeaveFromRight, IPOutboundManipulation_Prefix2Add,
IPOutboundManipulation_Suffix2Add, IPOutboundManipulation_PrivacyRestrictionMode;
IPOutboundManipulation 0 = "", 0, 2, 1, "*", "*", "*", "*", "*", "", 0, -1, 0, 1,
0, 0, 255, "+1", "", 0;
IPOutboundManipulation 1 = "unblock_ANI", 0, 1, 2, "*", "*", "*", "*", "*", "", 0,
-1, 0, 0, 0, 0, 255, "", "", 3;

[ \IPOutboundManipulation ]


[ CodersGroup0 ]

FORMAT CodersGroup0_Index = CodersGroup0_Name, CodersGroup0_pTime,
CodersGroup0_rate, CodersGroup0_PayloadType, CodersGroup0_Sce;
CodersGroup0 0 = "g711Alaw64k", 20, 255, -1, 0;

[ \CodersGroup0 ]


[ CodersGroup1 ]

FORMAT CodersGroup1_Index = CodersGroup1_Name, CodersGroup1_pTime,
CodersGroup1_rate, CodersGroup1_PayloadType, CodersGroup1_Sce;
CodersGroup1 0 = "g711Ulaw64k", 20, 0, -1, 1;
CodersGroup1 1 = "g711Alaw64k", 20, 0, -1, 1;

[ \CodersGroup1 ]


[ CodersGroup2 ]

FORMAT CodersGroup2_Index = CodersGroup2_Name, CodersGroup2_pTime,
CodersGroup2_rate, CodersGroup2_PayloadType, CodersGroup2_Sce;
CodersGroup2 0 = "g729", 20, 0, -1, 0;
CodersGroup2 1 = "g711Ulaw64k", 20, 0, -1, 0;
CodersGroup2 2 = "g711Alaw64k", 20, 0, -1, 0;

[ \CodersGroup2 ]


[ CodersGroup4 ]

FORMAT CodersGroup4_Index = CodersGroup4_Name, CodersGroup4_pTime,
CodersGroup4_rate, CodersGroup4_PayloadType, CodersGroup4_Sce;
CodersGroup4 0 = "t38fax", 255, 255, -1, 255;

[ \CodersGroup4 ]


[ AllowedCodersGroup1 ]

FORMAT AllowedCodersGroup1_Index = AllowedCodersGroup1_Name;
AllowedCodersGroup1 0 = "g711Ulaw64k";
AllowedCodersGroup1 1 = "g711Alaw64k";

[ \AllowedCodersGroup1 ]
```

```
[ AllowedCodersGroup2 ]

FORMAT AllowedCodersGroup2_Index = AllowedCodersGroup2_Name;
AllowedCodersGroup2 0 = "g729";
AllowedCodersGroup2 1 = "g711Ulaw64k";
AllowedCodersGroup2 2 = "g711Alaw64k";

[ \AllowedCodersGroup2 ]


[ MessageManipulations ]

FORMAT MessageManipulations_Index = MessageManipulations_ManipulationName,
MessageManipulations_ManSetID, MessageManipulations_MessageType,
MessageManipulations_Condition, MessageManipulations_ActionSubject,
MessageManipulations_ActionType, MessageManipulations_ActionValue,
MessageManipulations_RowRole;
MessageManipulations 0 = "clip2ndPAI", 3, "", "", "header.P-Asserted-Identity.1",
1, "", 0;
MessageManipulations 1 = "pai sip", 3, "invite", "", "header.P-Asserted-
Identity.url.type", 2, "'1'", 0;
MessageManipulations 2 = "", 3, "invite", "header.referred-by exists",
"header.referred-by.url.user", 6, "'+1'", 0;
MessageManipulations 3 = "referred-by", 3, "invite", "header.referred-by exists",
"header.diversion", 0, "header.referred-by", 0;
MessageManipulations 4 = "", 3, "", "", "header.referred-by", 1, "", 1;
MessageManipulations 5 = "forward offnet", 3, "Invite", "header.diversion exists",
"header.diversion.url.user", 6, "'+1'", 0;
MessageManipulations 6 = "", 3, "", "", "header.diversion.url.host", 2,
"'63.98.198.37'", 1;
MessageManipulations 7 = "from header", 3, "invite", "header.from regex
(.*)(;phone-context=enterprise)(.*)", "header.from", 2, "$1+$3", 0;
MessageManipulations 8 = "pai header", 3, "invite", "header.P-Asserted-Identity
regex (.*)(;phone-context=enterprise)(.*)", "header.P-Asserted-Identity", 2,
"$1+$3", 0;
MessageManipulations 9 = "conf add pstn", 3, "Invite", "header.from regex
(.*)(user=phone)(.*)(>)(;tag=)(.*)", "header.from", 2, "$1+$2+$4+$5+$6", 0;
MessageManipulations 10 = "TC13", 13, "Invite", "", "Header.Diversion", 0,
''<sip:9727283204@63.98.198.37>;reason=unknown;privacy="off"'', 0;
MessageManipulations 11 = "TC14", 13, "", "", "header.P-Asserted-
Identity.url.user", 2, "'9727283204'", 0;

[ \MessageManipulations ]
```

B Configuring Analog Devices (ATAs) for Fax Support

This section shows 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.
- Instead of the ATA, FXS ports on devices that support FXS ports can be used.

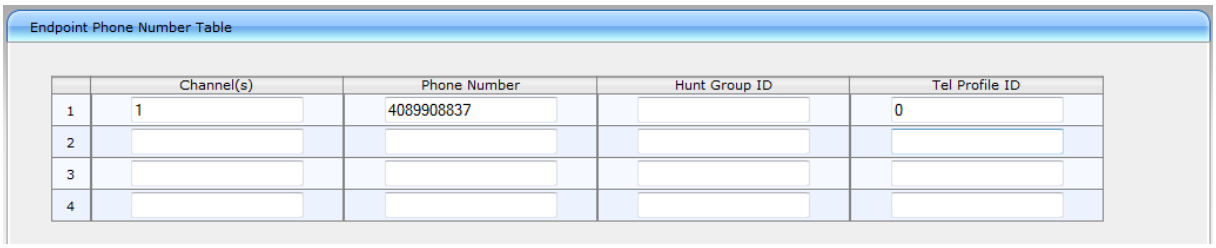
B.1 Step 1: Configure the Endpoint Phone Number Table

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 ATA destination phone number ''4089908877'' (IP address 10.133.4.101) with all routing directed to the AudioCodes E-SBC device (10.133.4.43).

➢ **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 B-1: Endpoint Phone Number Table Page



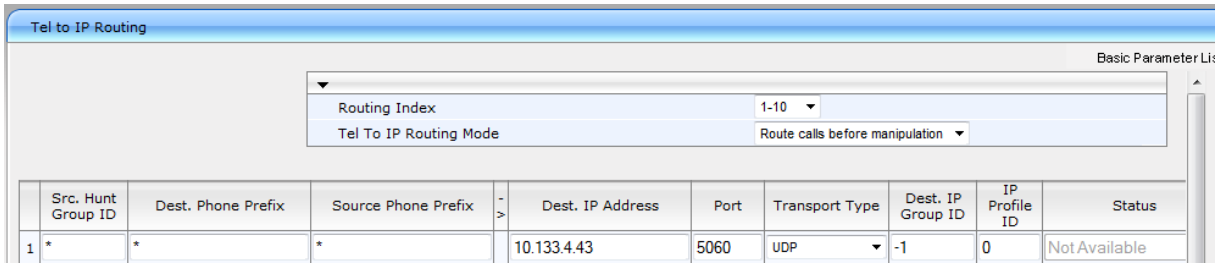
B.2 Step 2: Configure the Tel to IP Routing Table

This step shows 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 B-2: Tel to IP Routing Page



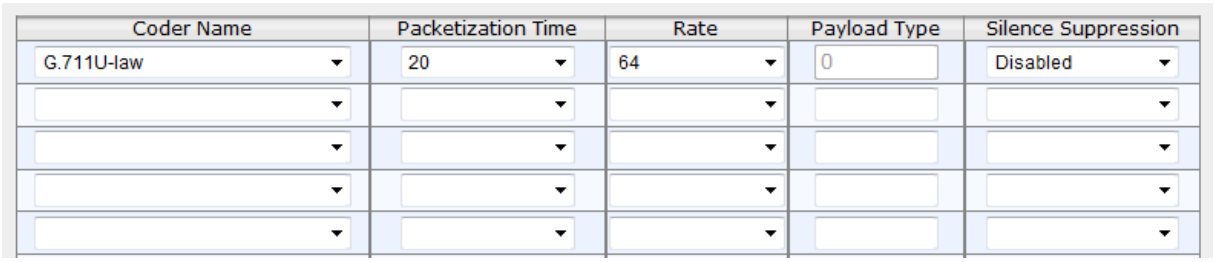
B.3 Step 3: Configure the Coders Table

This step shows 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 B-3: Coders Table Page



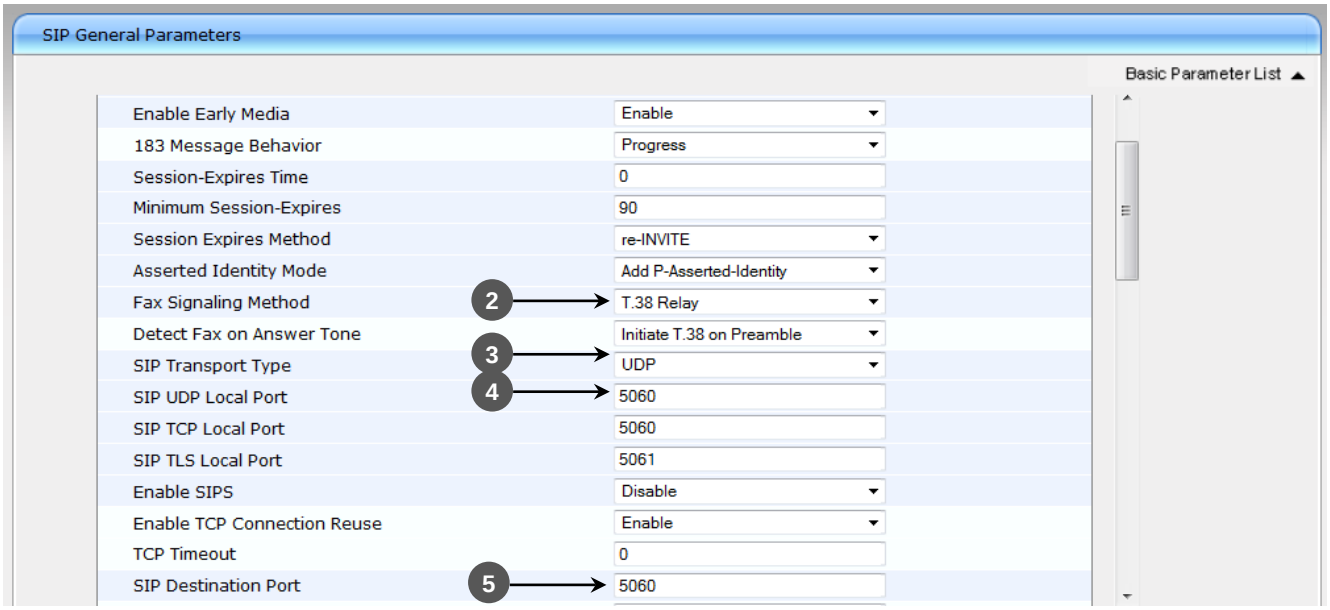
B.4 Step 4: Configure SIP UDP Transport Type and Fax Signaling Method

This step shows 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 B-4: SIP General Parameters Page



2. From the 'FAX Signaling Method' drop-down list, select **T.38 Relay** (or **G.711 Transport** – if G.711 is preferred**).**
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).

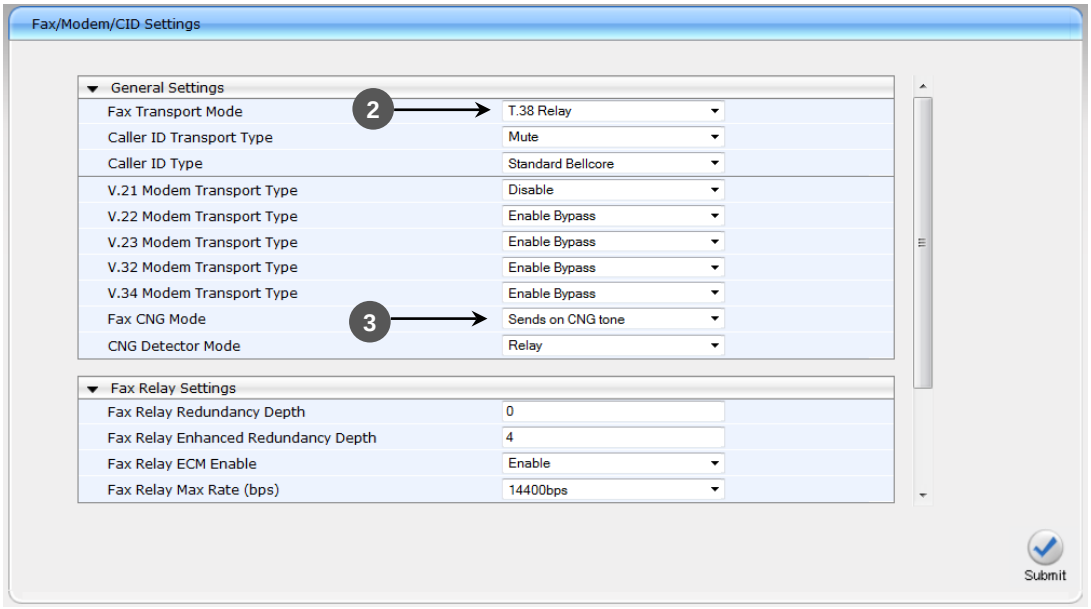
B.5 Step 5: Configure FAX Settings

This step shows how to configure the fax settings for the MP-11x device.

➢ **To configure the fax settings:**

1. Open the 'FAX/Modem/CID Settings' page (**Configuration** tab > **VoIP** menu > **Media** submenu > **FAX/Modem/CID Settings**).

Figure B-5: Fax/Modem/CID Settings Page



2. From the 'FAX Transport Mode' drop-down list, select **T.38 Relay.**
3. From the 'Fax CNG Mode' drop-down list, select **Sends on CNG tone**.

   Because of interworking issues with the Verizon network, the ATA must initiate T.38 fax on both the CED as well as the CNG tone. Verizon does not initiate T.38 FAX relay within the Verizon SIP Trunking Network.

AudioCodes

Configuration Note

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