Enterprise Session Border Controllers (E-SBC)

AudioCodes Mediant™ Series

Interoperability Lab

Configuration Note

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

Microsoft Partner
Gold Communications

Lync

windstream.

AudioCodes

Version 6.8

August 2014

Document # LTRT-39226

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Notice

This document describes how to connect the Microsoft Lync Server 2013 and Windstream SIP Trunk using AudioCodes Mediant E-SBC product series, which includes the Mediant 500 E-SBC, Mediant 800 Gateway & E-SBC, Mediant 1000B Gateway & E-SBC, Mediant 3000 Gateway & E-SBC, Mediant 2600 E-SBC, and Mediant 4000 E-SBC.

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Date Published: August-26-2014

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1 Introduction

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

1.1 Intended Audience

The document is intended for engineers, or AudioCodes and Windstream Partners who are responsible for installing and configuring Windstream'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
- 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.

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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 3000 Gateway & E-SBC ▪ Mediant 2600 E-SBC ▪ Mediant 4000 E-SBC
Software Version	SIP_6.80A.234.004
Protocol	▪ SIP/UDP (to the Windstream SIP Trunk) ▪ SIP/TCP or TLS (to the Lync FE Server)
Additional Notes	None

2.2 Windstream SIP Trunking Version

Table 2-2: Windstream Version

Vendor/Service Provider	Windstream
SSW Model/Service	BroadSoft
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

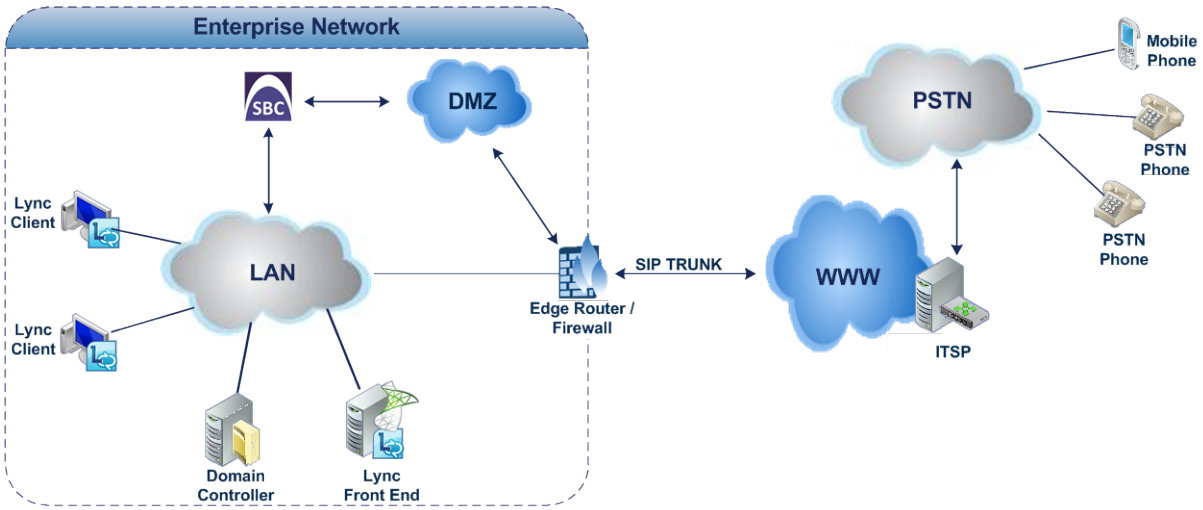
2.4 Interoperability Test Topology

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

- Enterprise deployed with Microsoft Lync Server 2013 in its private network for enhanced communication within the Enterprise.
- Enterprise wishes to offer its employees enterprise-voice capabilities and to connect the Enterprise to the PSTN network using Windstream'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 the IP-based Session Initiation Protocol (SIP).
  - **Border:** IP-to-IP network border between Lync Server 2013 network in the Enterprise LAN and Windstream's SIP Trunk located in the public network.

The figure below illustrates this interoperability test topology:

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



2.4.1 Environment Setup

The interoperability test topology includes the following environment setup:

Table 2-4: Environment Setup

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

2.4.2 Known Limitations

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

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3 Configuring Lync Server 2013

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

Note: Dial plans, voice policies, and PSTN usages are also necessary for Enterprise voice deployment; however, they are beyond the scope of this document.

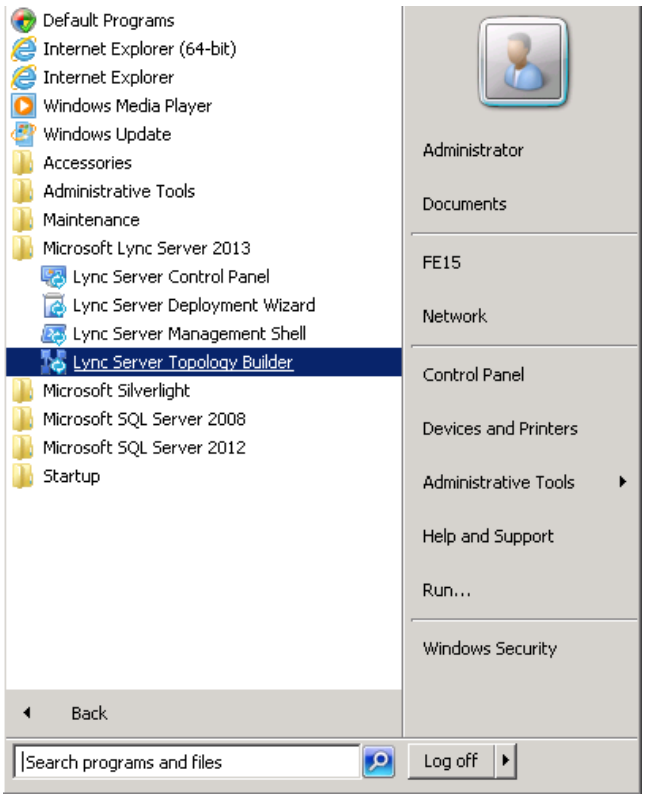
3.1 Configuring the E-SBC as an IP / PSTN Gateway

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:**

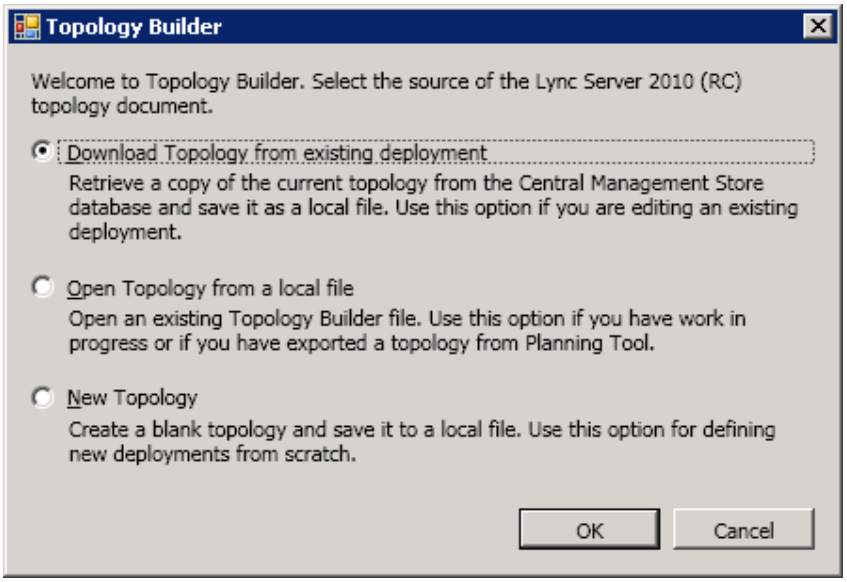
1. On the server where the Topology Builder is installed, start the Lync Server 2013 Topology Builder (Windows **Start** menu > **All Programs** > **Lync Server Topology Builder**), as shown below:

Figure 3-1: Starting the Lync Server Topology Builder



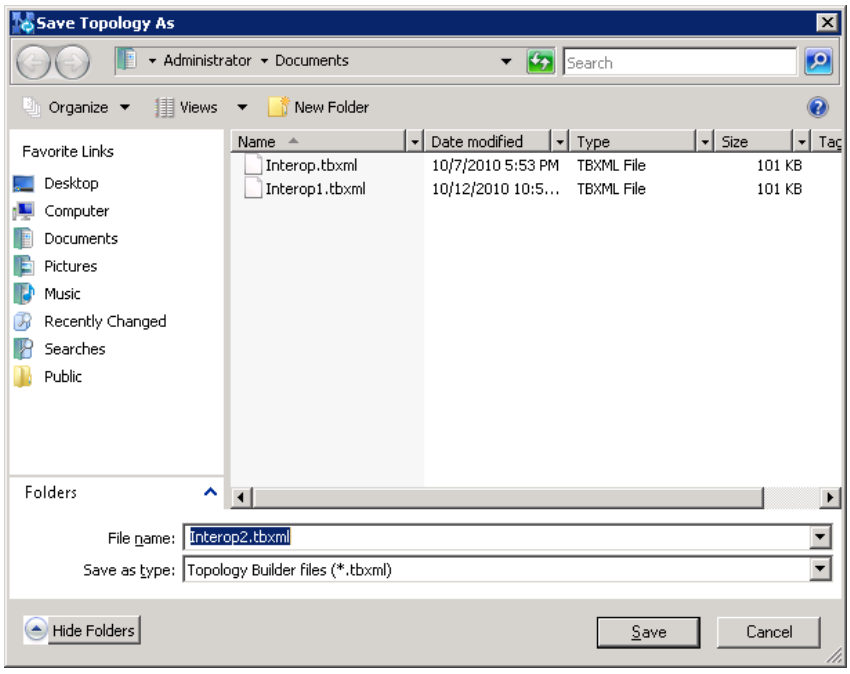
The following is displayed:

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:

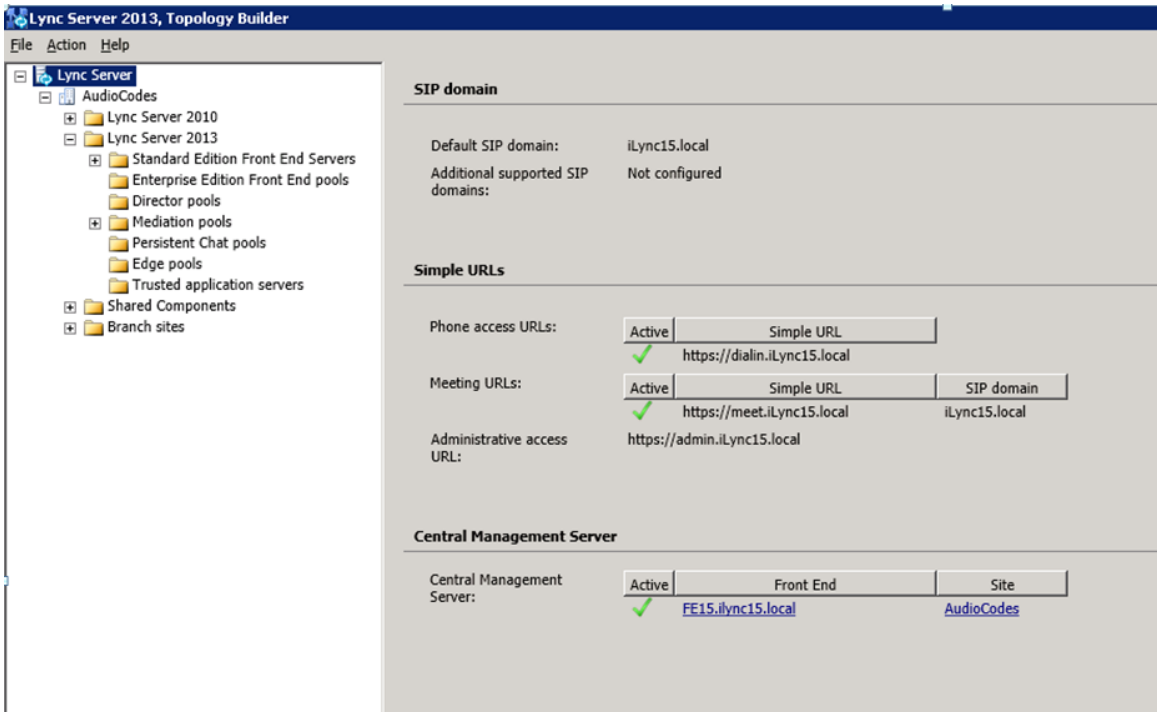
Figure 3-3: Save Topology Dialog Box



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.

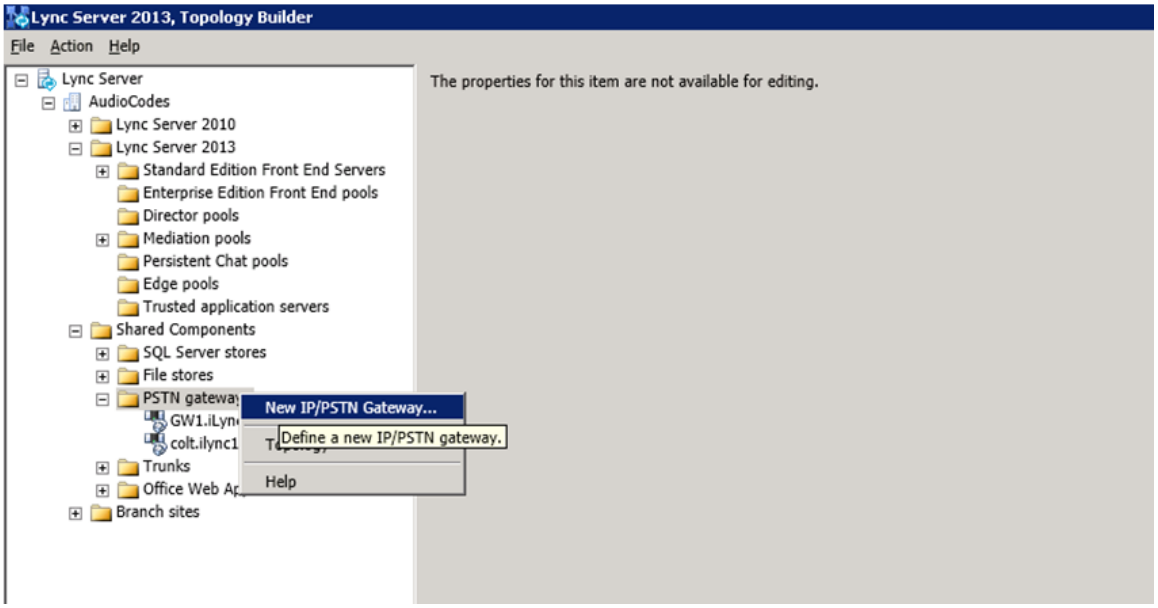
   The Topology Builder screen with the downloaded Topology is displayed:

Figure 3-4: Downloaded Topology



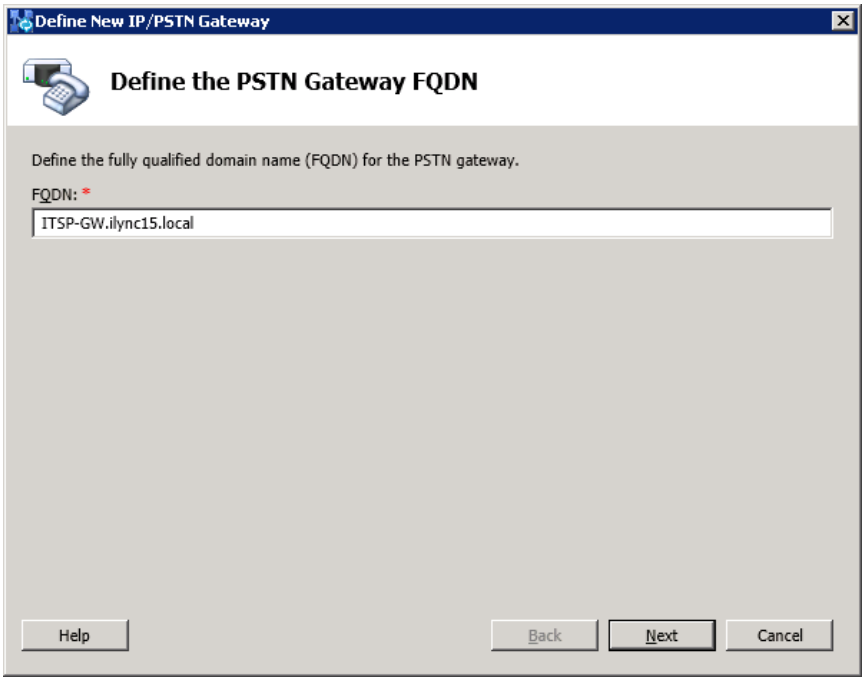
4. Under the **Shared Components** node, right-click the **PSTN gateways** node, and then from the shortcut menu, choose **New IP/PSTN Gateway**, as shown below:

Figure 3-5: Choosing New IP/PSTN Gateway



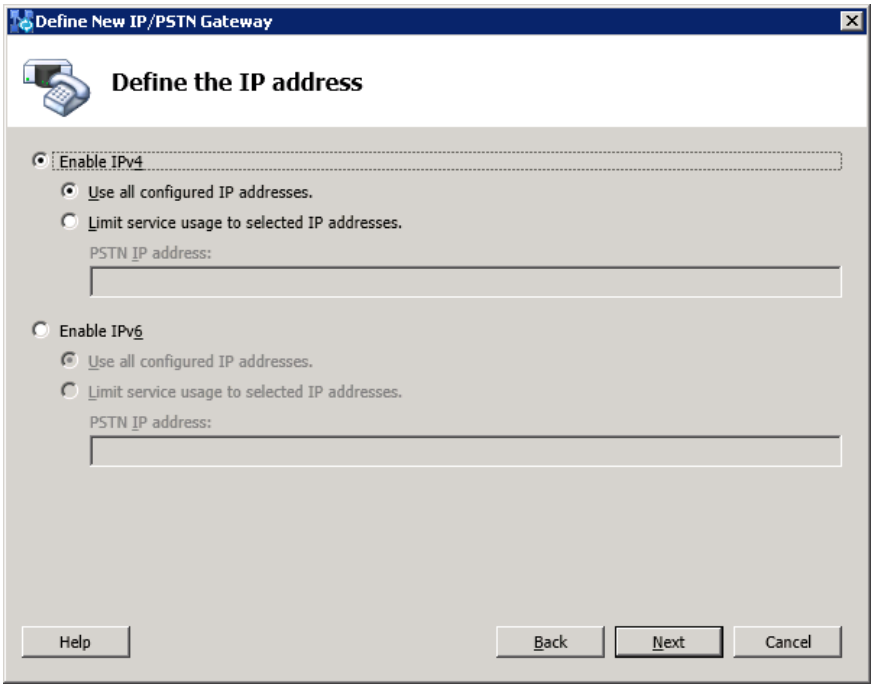
The following is displayed:

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



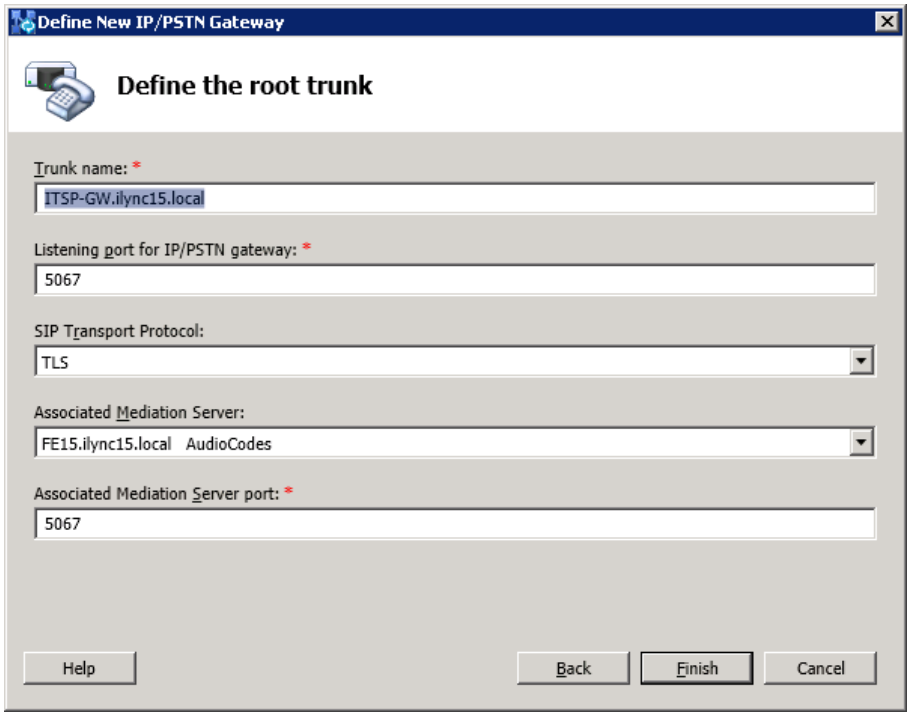
6. Define the listening mode (IPv4 or IPv6) of the IP address of your new PSTN gateway, and then click **Next**.

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.

Notes:

- 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.

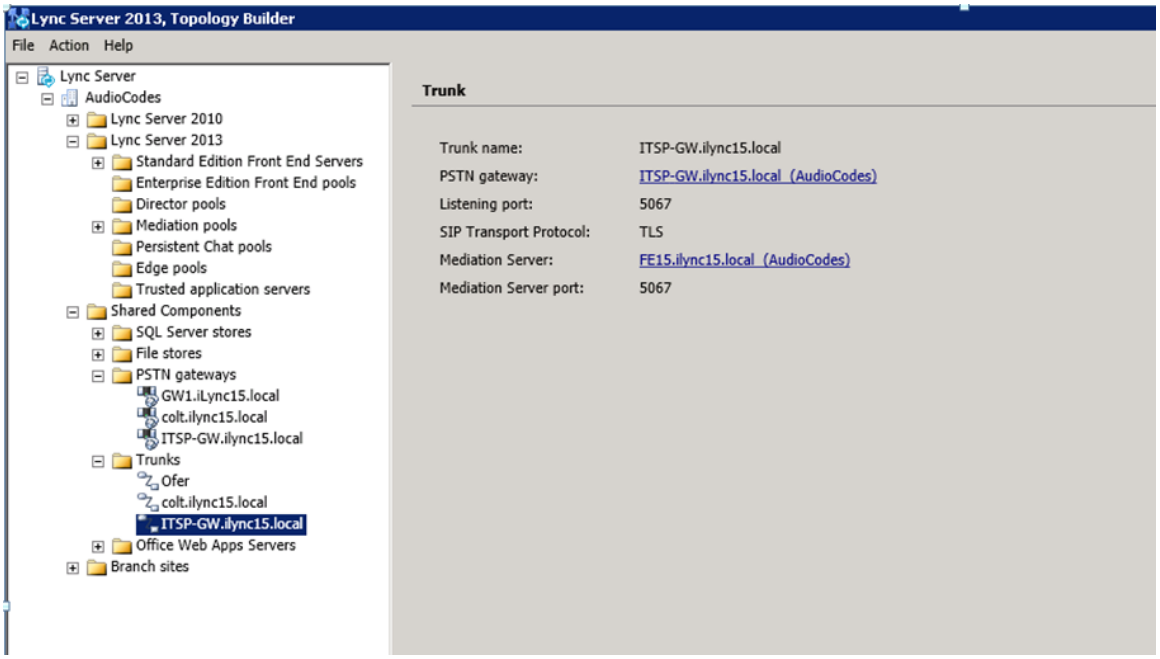
Figure 3-8: Define the Root Trunk



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**.

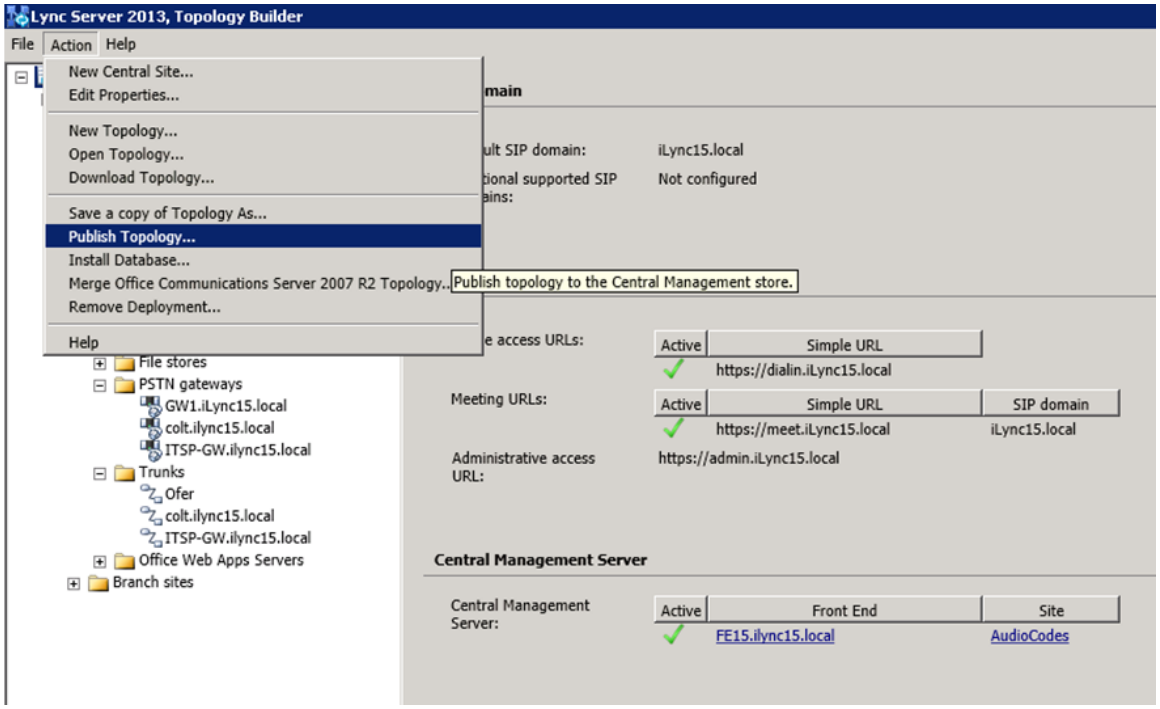
The E-SBC is added as a PSTN gateway, and a trunk is created as shown below:

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



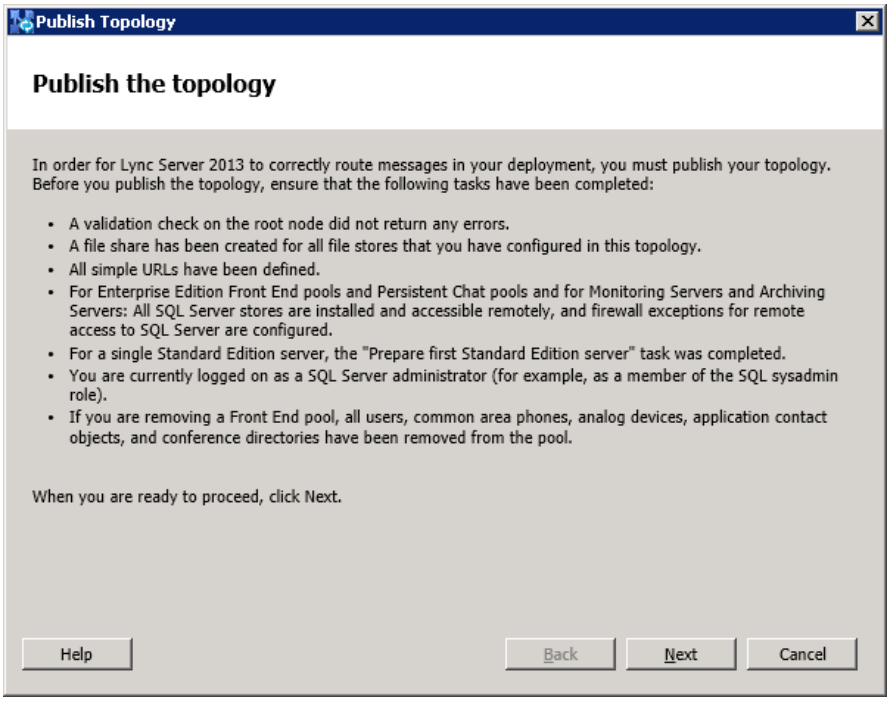
8. Publish the Topology: In the main tree, select the root node **Lync Server**, and then from the **Action** menu, choose **Publish Topology**, as shown below:

Figure 3-10: Choosing Publish Topology



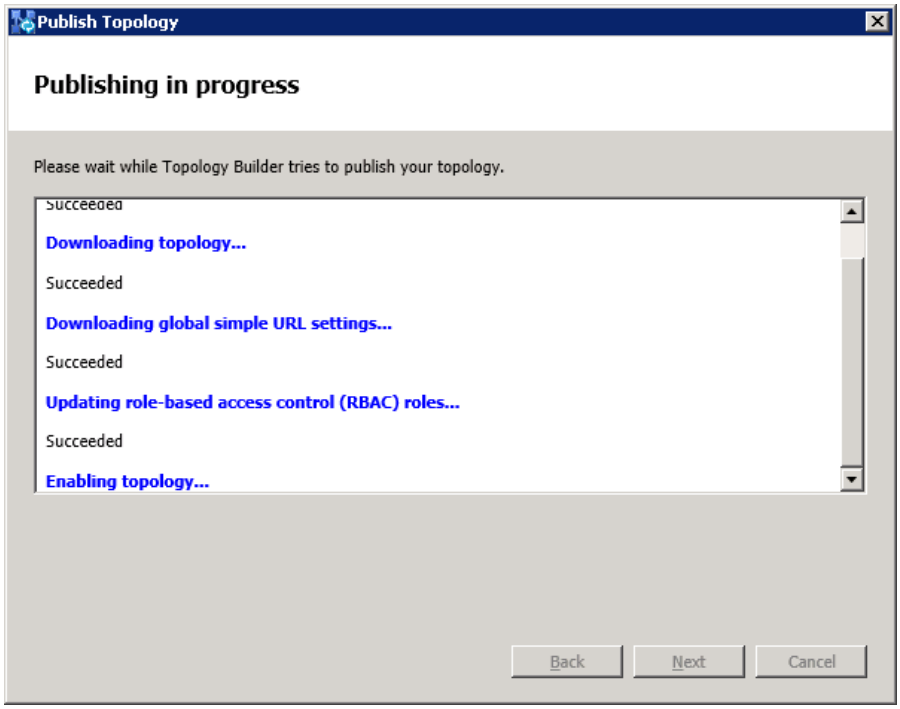
The following is displayed:

Figure 3-11: Publish the Topology



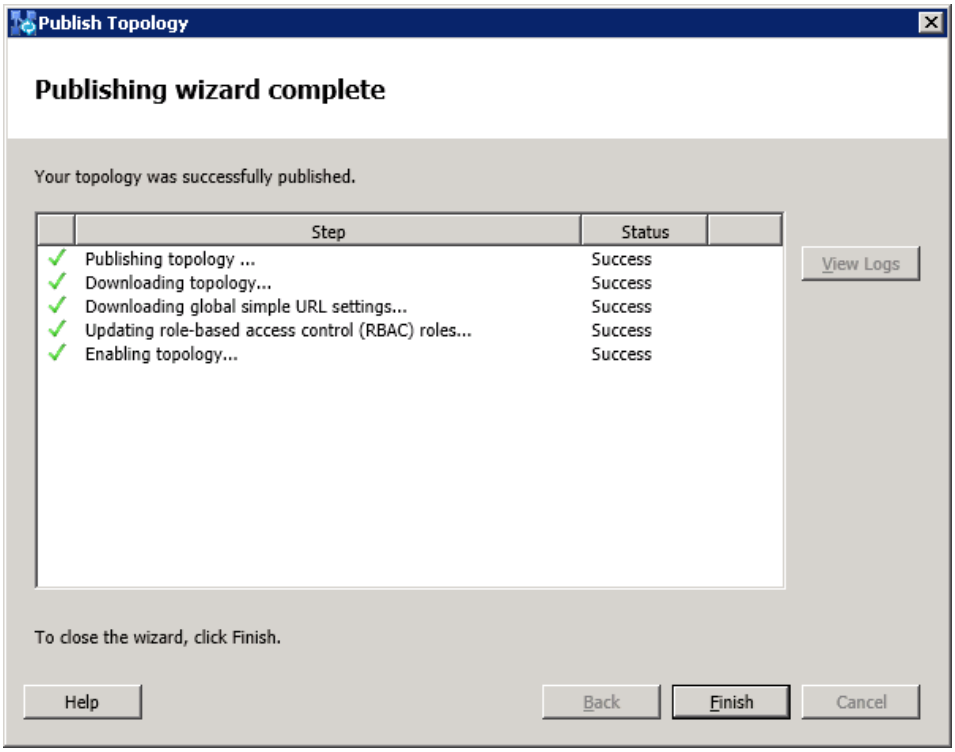
9. Click **Next**; the Topology Builder starts to publish your topology, as shown below:

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

The procedure below describes how to configure a "Route" on the Lync Server 2013 and to associate it with the E-SBC PSTN gateway.

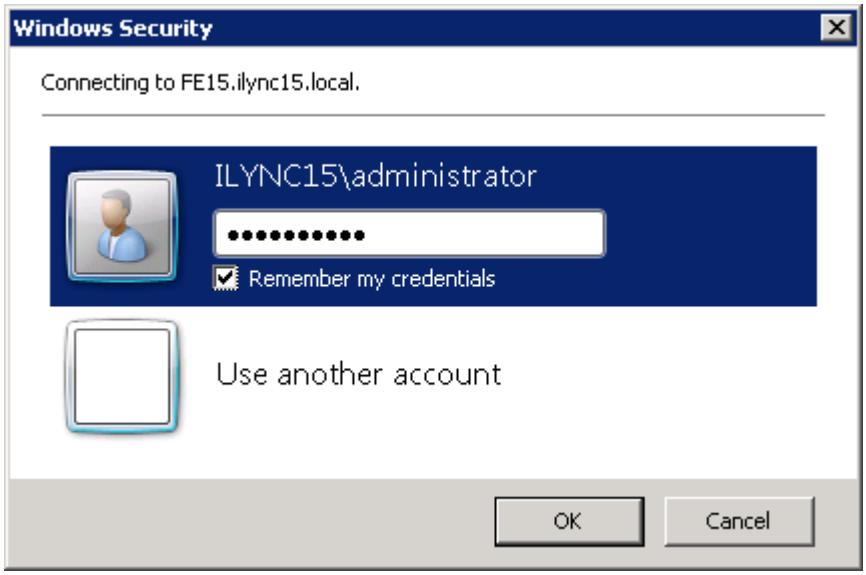
➢ **To configure the "route" on Lync Server 2013:**

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

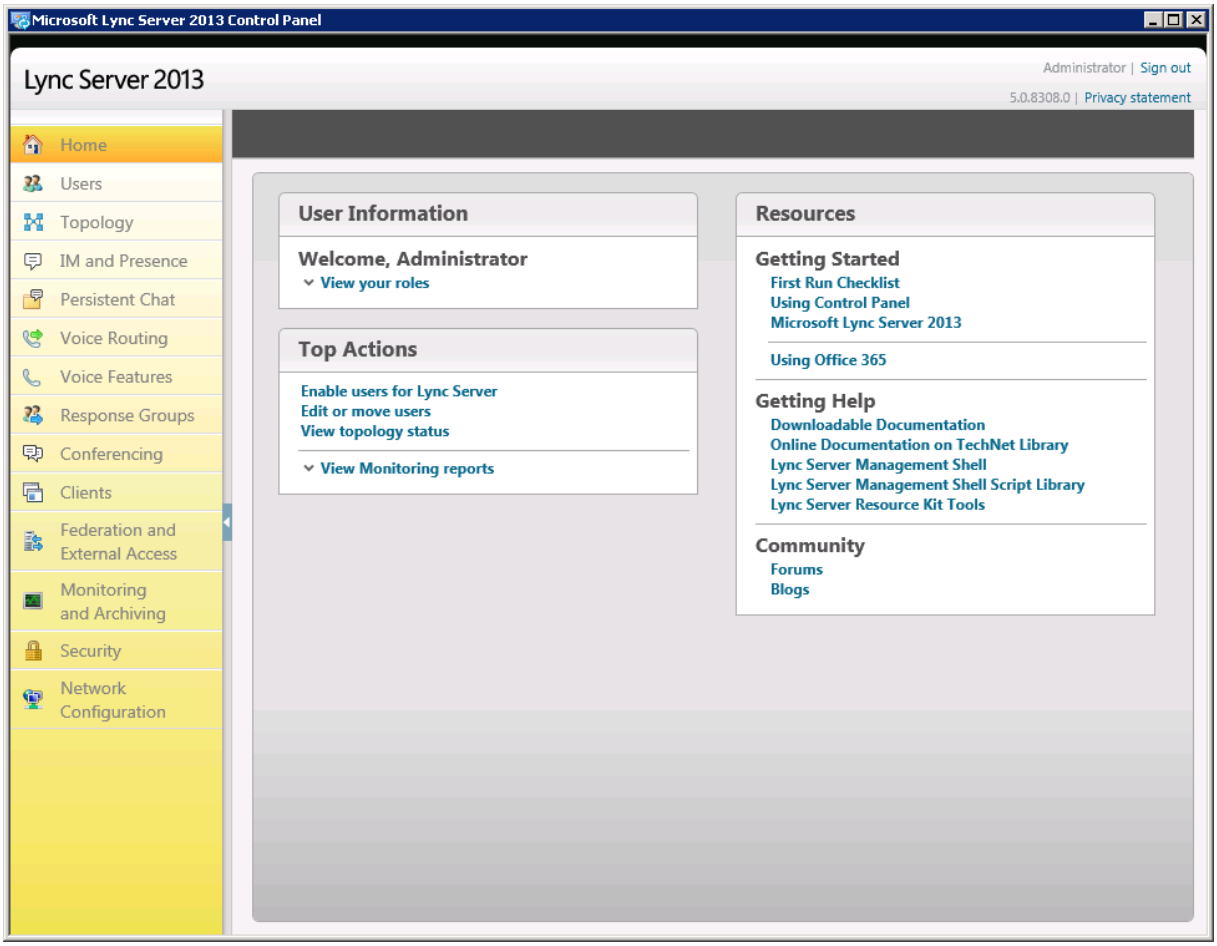
You are prompted to enter your login credentials:

Figure 3-15: Lync Server Credentials



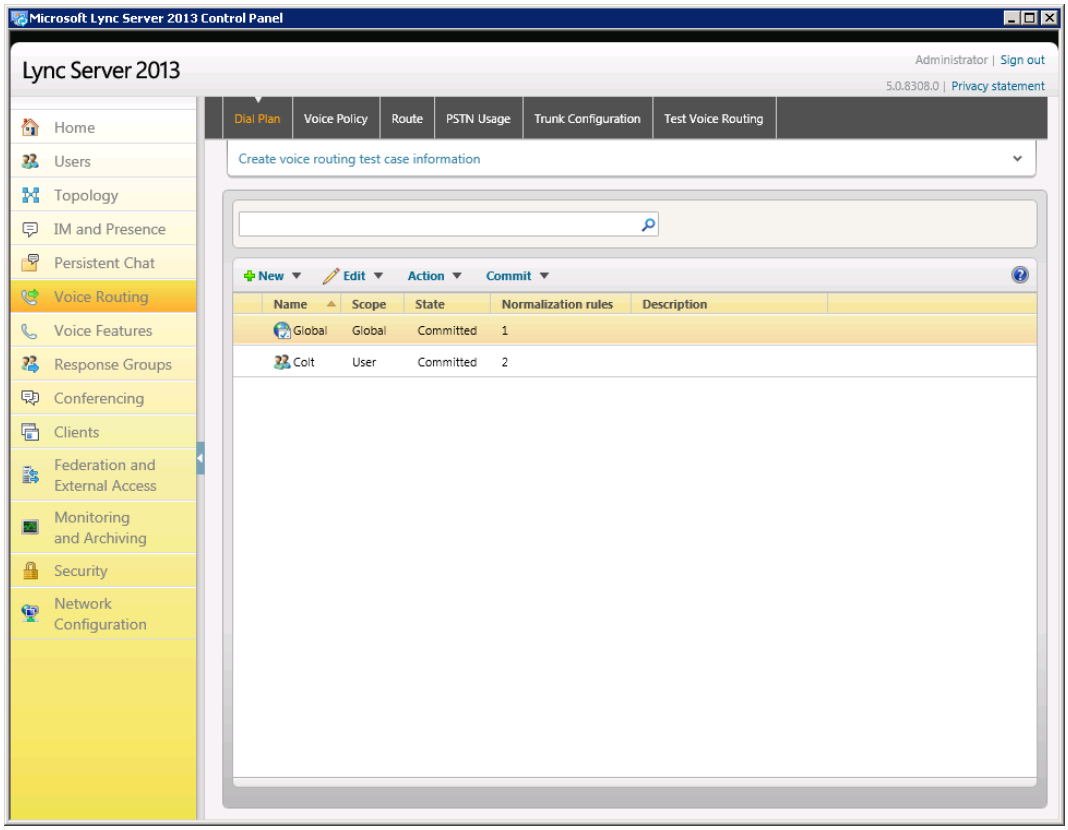
2. Enter your domain username and password, and then click **OK**; the Microsoft Lync Server 2013 Control Panel is displayed:

Figure 3-16: Microsoft Lync Server 2013 Control Panel



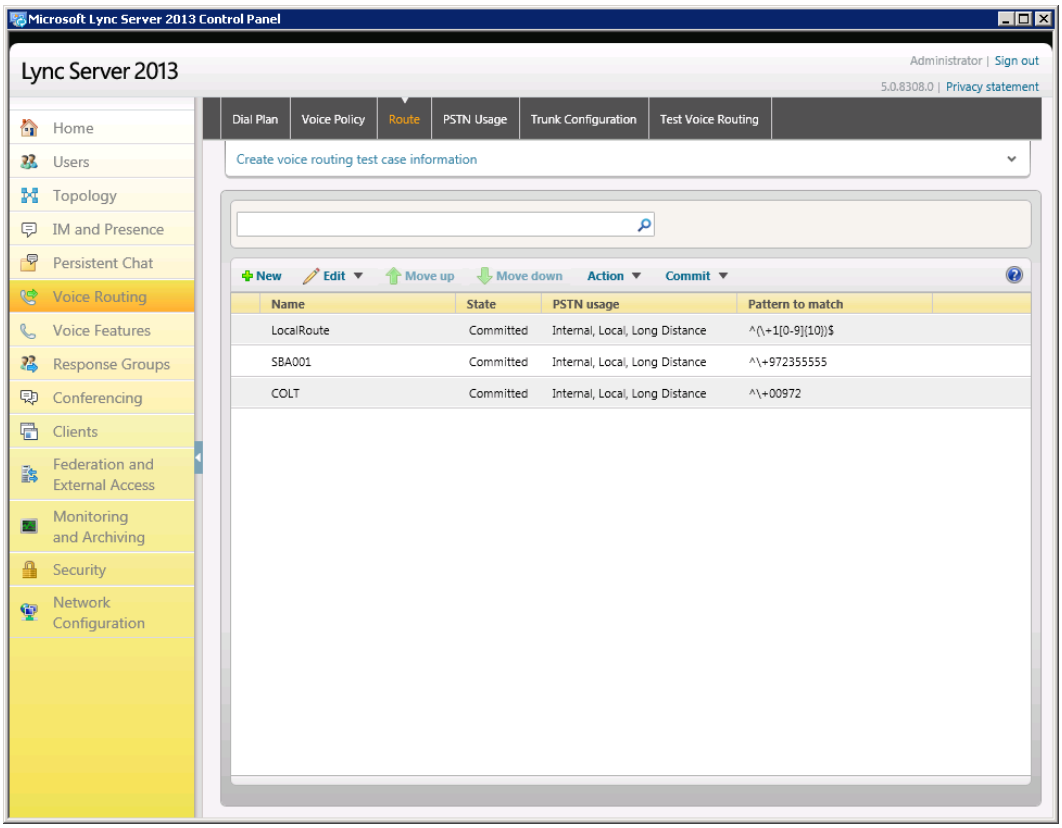
3. In the left navigation pane, select **Voice Routing**.

Figure 3-17: Voice Routing Page



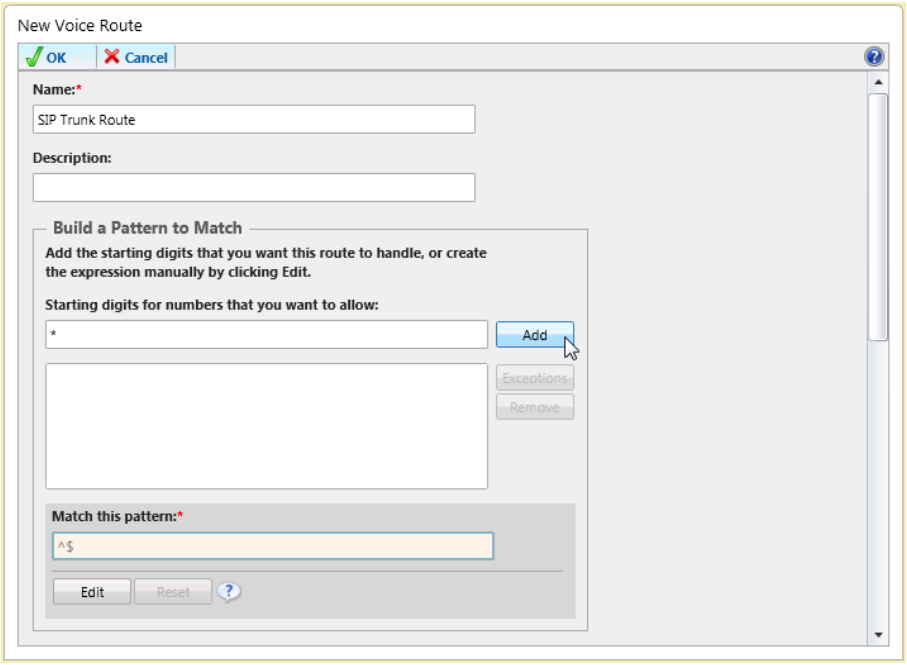
4. In the Voice Routing page, select the **Route** tab.

Figure 3-18: Route Tab



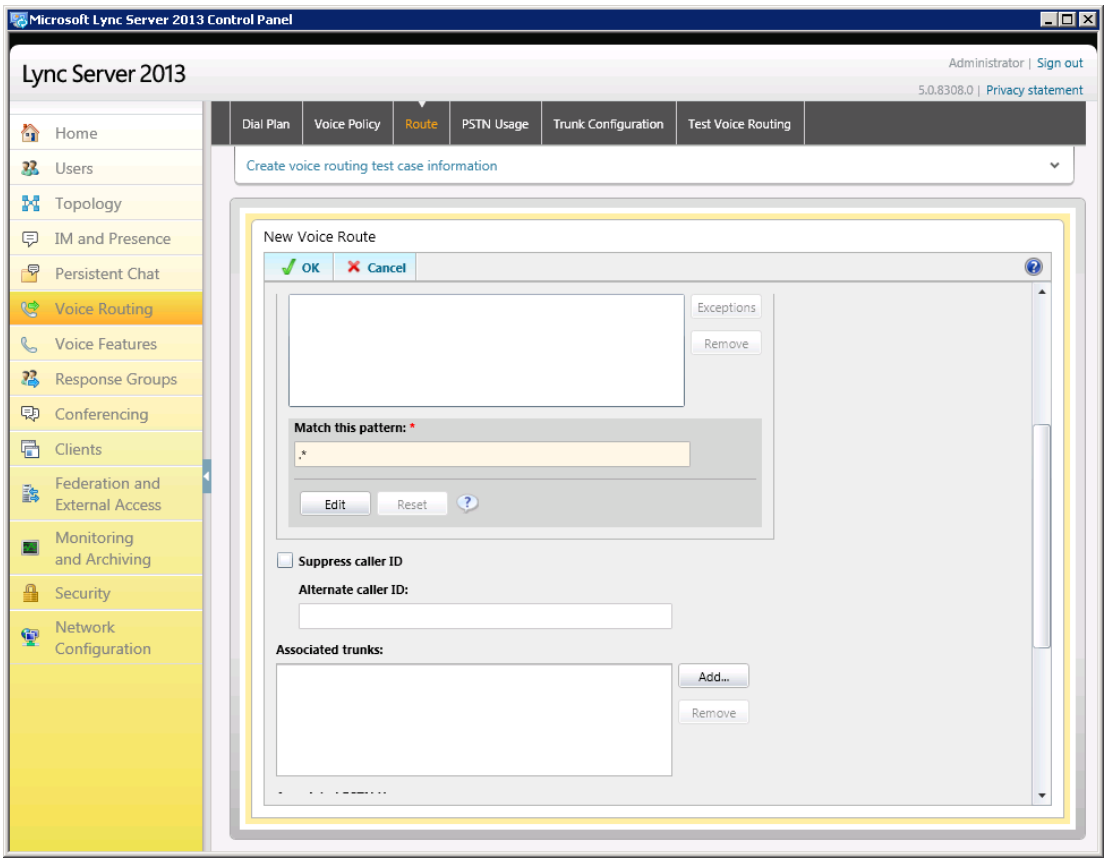
5. Click **New**; the New Voice Route page appears:

Figure 3-19: Adding New Voice Route



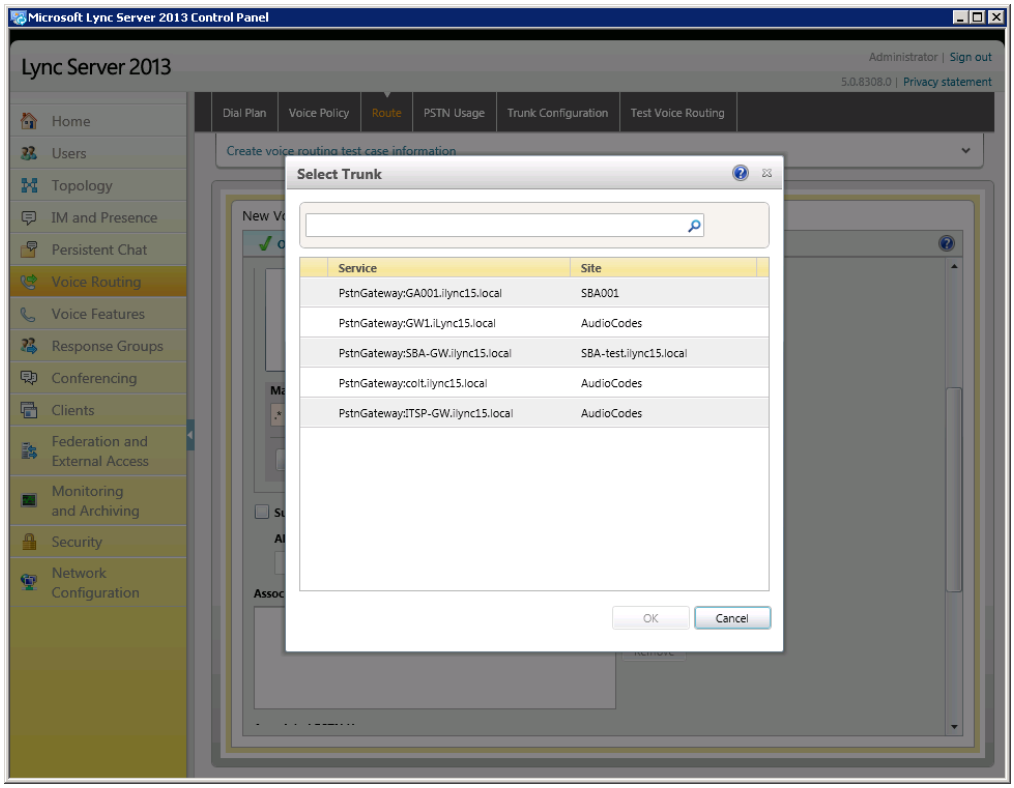
6. In the 'Name' field, enter a name for this route (e.g., **SIP Trunk Route**).
7. In the 'Starting digits for numbers that you want to allow' field, enter the starting digits you want this route to handle (e.g., ***** to match all numbers), and then click **Add**.

Figure 3-20: Adding New Trunk



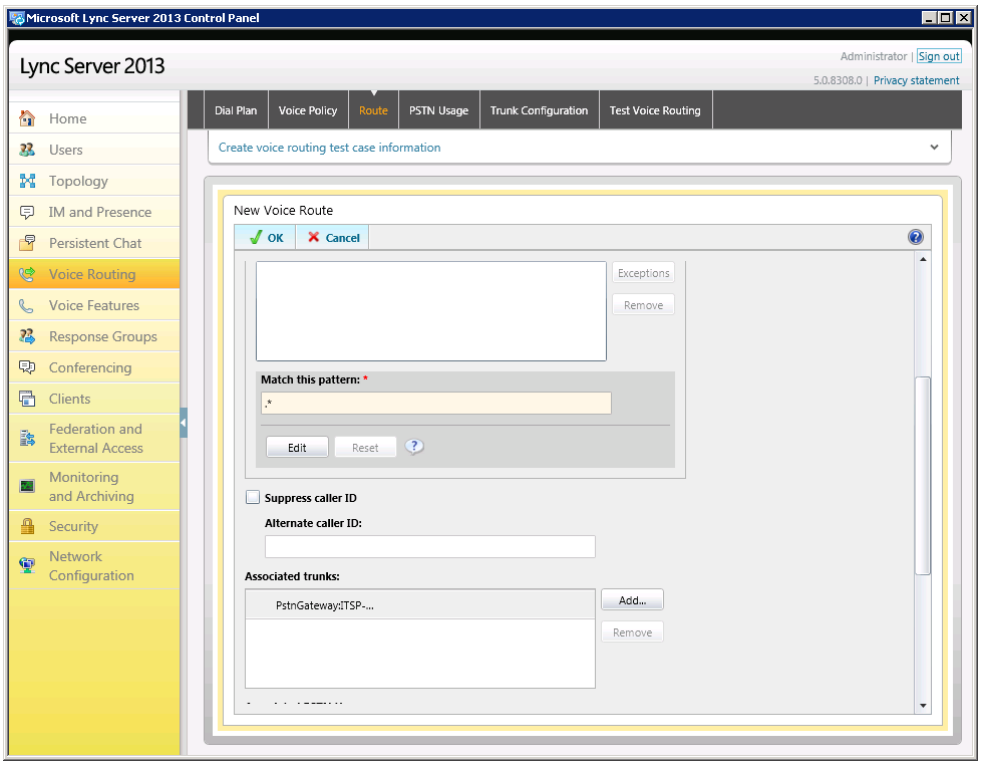
8. Associate the route with the E-SBC Trunk that you created:
   a. Under the 'Associated Trunks' group, click **Add**; a list of all the deployed gateways is displayed:

Figure 3-21: List of Deployed Trunks



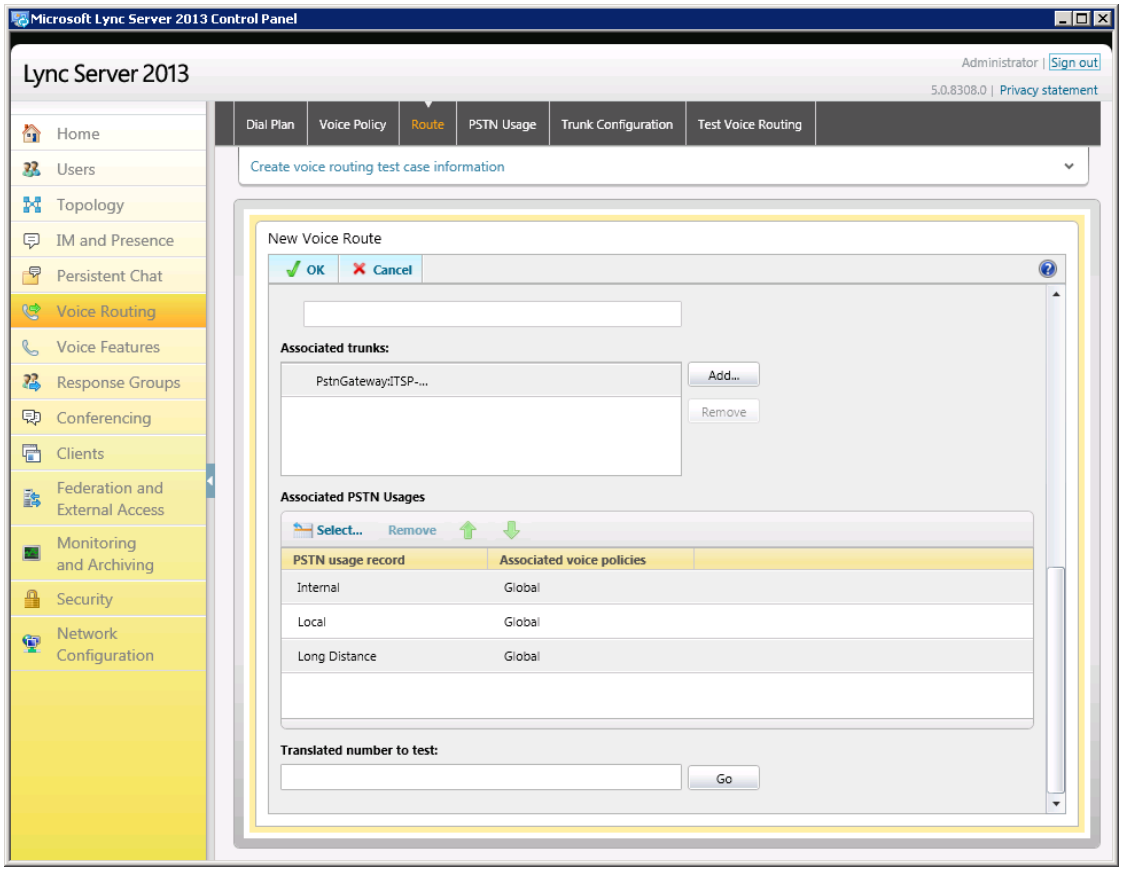
   b. Select the E-SBC Trunk you created, and then click **OK**; the trunk is added to the 'Associated Trunks' group list:

Figure 3-22: Selected E-SBC Trunk



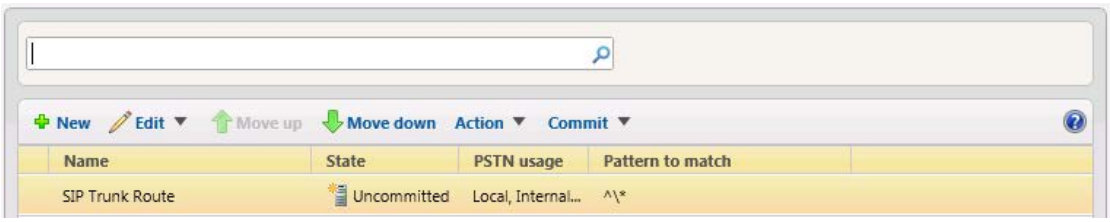
9. Associate a PSTN Usage to this route:
   a. Under the 'Associated PSTN Usages' group, click **Select** and then add the associated PSTN Usage.

Figure 3-23: Associating PSTN Usage to Route



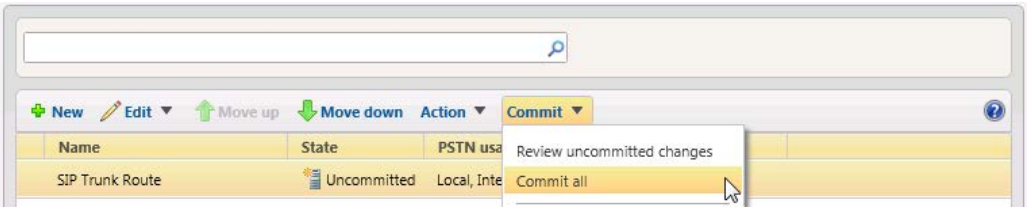
10. Click **OK** (located on the top of the New Voice Route page); the New Voice Route (Uncommitted) is displayed:

Figure 3-24: Confirmation of New Voice Route



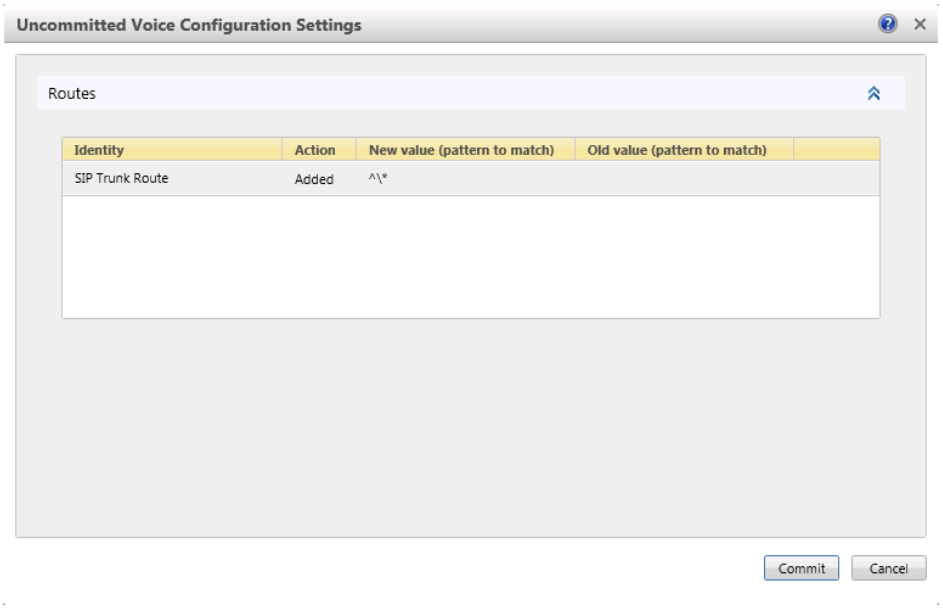
11. From the **Commit** drop-down list, choose **Commit all**, as shown below:

Figure 3-25: Committing Voice Routes



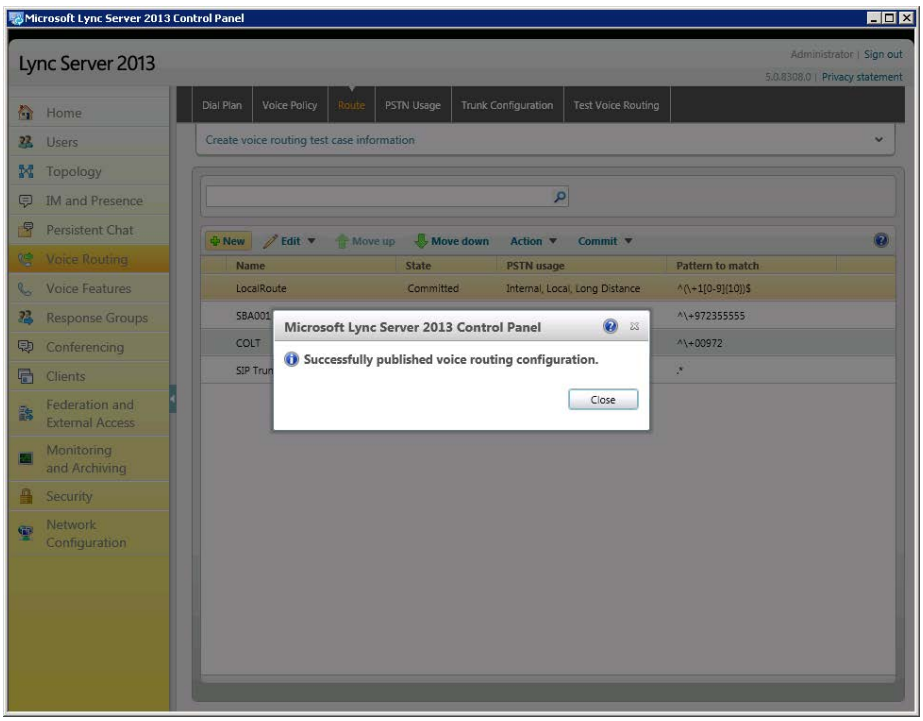
The Uncommitted Voice Configuration Settings page appears:

Figure 3-26: Uncommitted Voice Configuration Settings



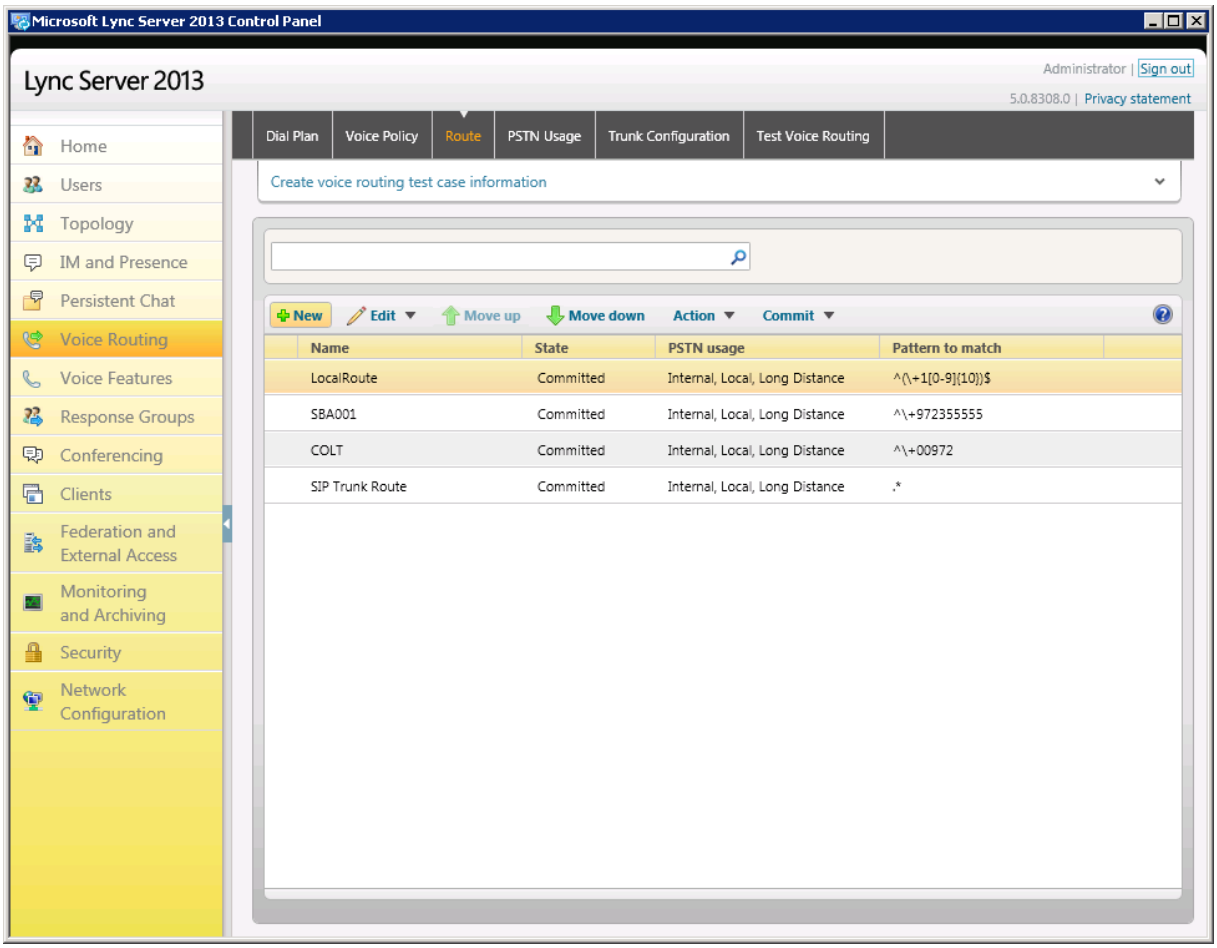
12. Click **Commit**; a message is displayed confirming a successful voice routing configuration, as shown below:

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

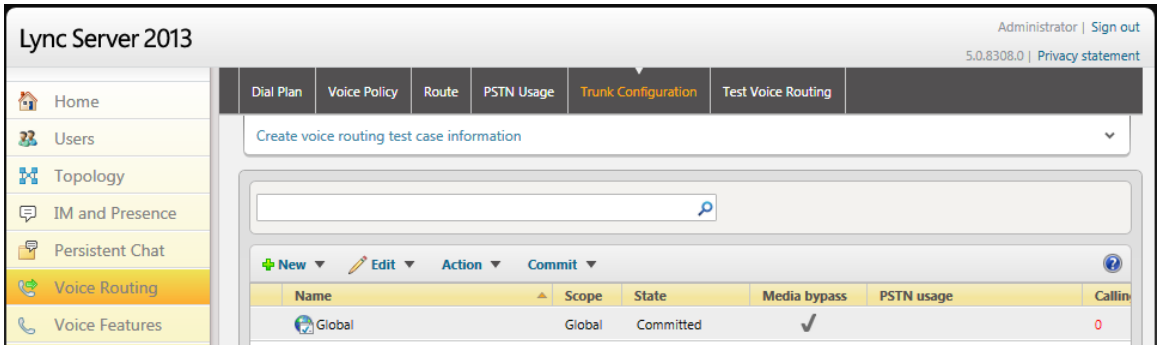


14. For ITSPs that implement a call identifier, continue with the following steps:

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 Windstream SIP Trunk in the P-Asserted-Identity header.

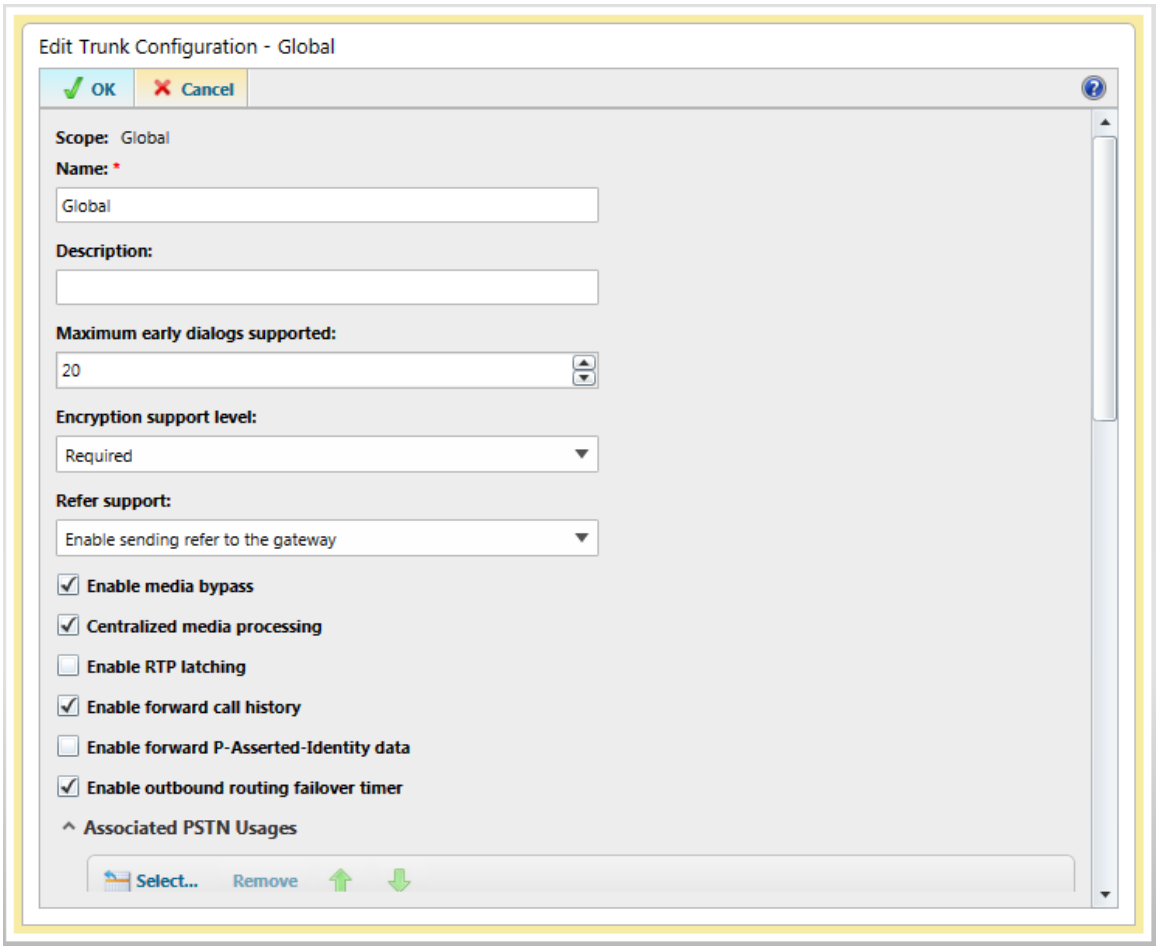
a. In the Voice Routing page, select the **Trunk Configuration** tab. Note that you can add and modify trunk configuration by site or by pool.

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** check box, and then click **OK**.

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

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4 Configuring AudioCodes E-SBC

This chapter provides step-by-step procedures on how to configure AudioCodes E-SBC for interworking between Microsoft Lync Server 2013 and the Windstream SIP Trunk. These configuration procedures are based on the interoperability test topology described in Section 2.4 on page 10, and includes the following main areas:

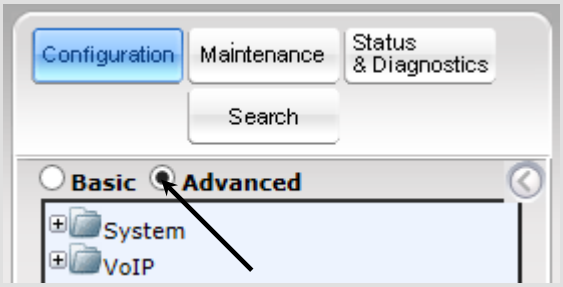
- E-SBC WAN interface - Windstream SIP Trunking environment
- E-SBC LAN interface - Lync Server 2013 environment

This configuration is done using the E-SBC's embedded Web server (hereafter, referred to as *Web interface*).

Notes:

- For implementing Microsoft Lync and Windstream 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 interoperability test and document does **not** cover all security aspects for connecting the SIP Trunk to the Microsoft Lync environment. Comprehensive security measures should be implemented per your organization's security policies. For security recommendations on AudioCodes' products, 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 Full-menu display mode. To do this, select the **Full** option, as shown below:



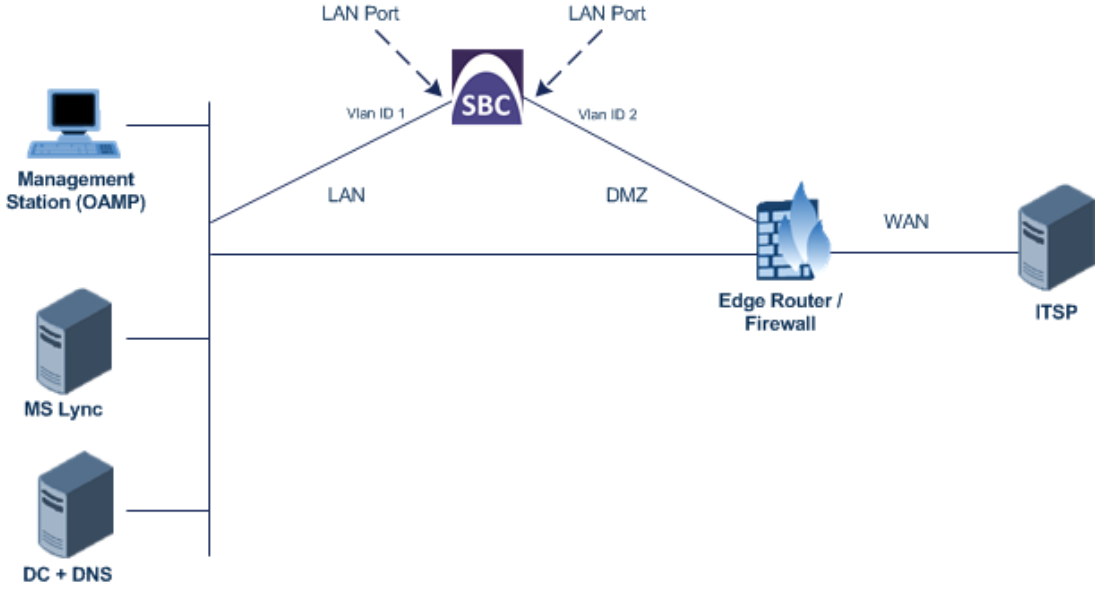
Note that when the E-SBC is reset, the Navigation tree reverts to Basic-menu display.

4.1 Step 1: IP Network Interfaces Configuration

This step describes how to configure the E-SBC's IP network interfaces. There are several ways to deploy the E-SBC; however, this interoperability test topology employs the following deployment method:

- E-SBC interfaces with the following IP entities:
  - Lync servers, located on the LAN
  - Windstream 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 describes how to define VLANs for each of the following interfaces:

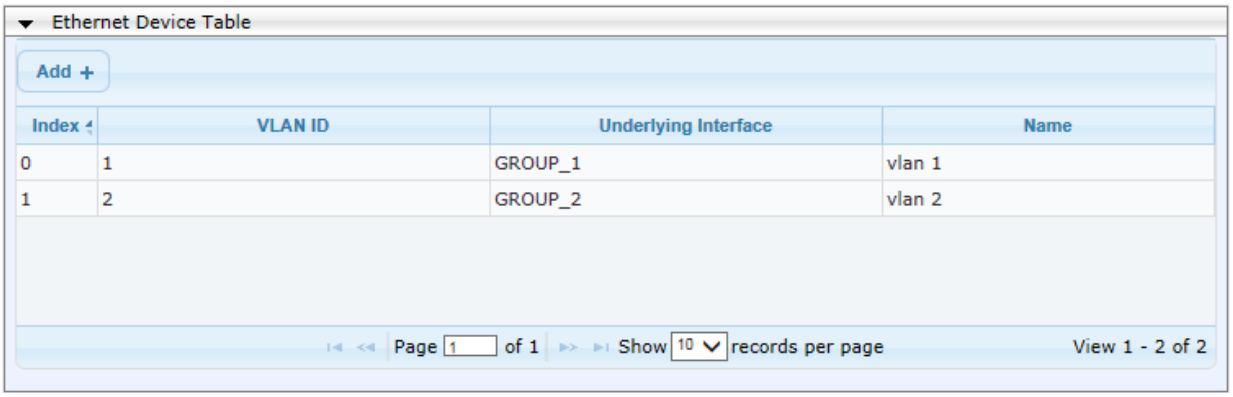
- LAN VoIP (assigned the name "Voice")
- WAN VoIP (assigned the name "WANSP")

➢ **To configure the VLANs:**

1. Open the Ethernet Device Table page (**Configuration** tab > **VoIP** menu > **Network** > **Ethernet Device Table**).
2. There will be one existing row for VLAN ID 1 and underlying interface GROUP_1.
3. 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 describes how to configure the IP network interfaces for each of the following interfaces:

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

➢ **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	**10.15.17.55** (IP address of E-SBC)
Prefix Length	**16** (subnet mask in bits for 255.255.0.0)
Gateway	**10.15.0.1**
VLAN ID	**1**
Interface Name	**Voice** (arbitrary descriptive name)
Primary DNS Server IP Address	**10.15.25.1**
Underlying Device	**vlan 1**

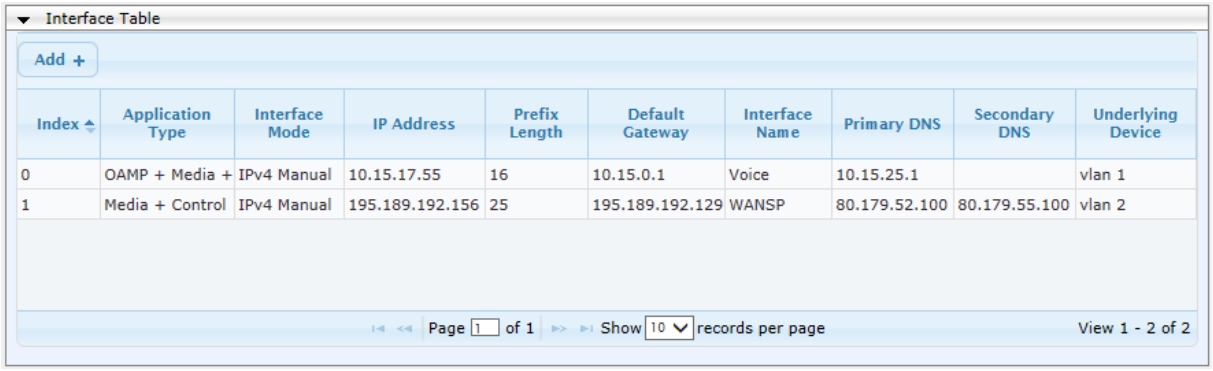
3. Add a network interface for the WAN side:
   a. Enter **1**, and then click **Add Index**.
   b. Configure the interface as follows:

Parameter	Value
Application Type	**Media + Control**
IP Address	**195.189.192.156** (WAN IP address)
Prefix Length	**25** (for 255.255.255.128)
Gateway	**195.189.192.129** (router's IP address)
VLAN ID	**2**
Interface Name	**WANSP**
Primary DNS Server IP Address	**80.179.52.100**
Secondary DNS Server IP Address	**80.179.55.100**
Underlying Device	**vlan 2**

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

The configured IP network interfaces are shown below:

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



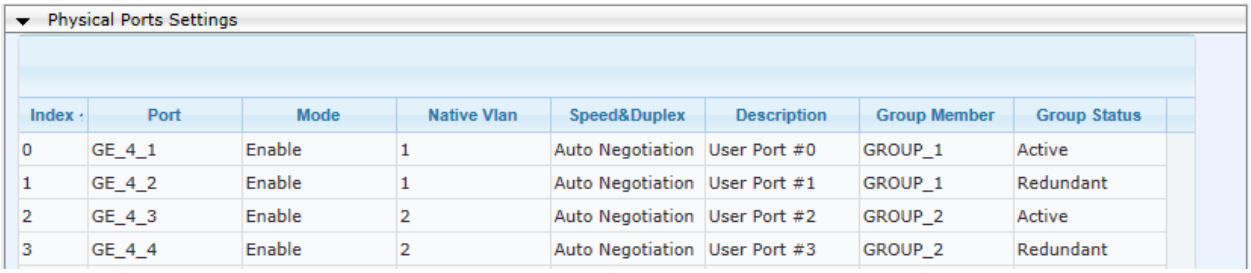
4.1.3 Step 1c: Configure the Native VLAN ID

This step describes 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 "WANSP".

Figure 4-4: Configured Port Native VLAN


Physical Ports Settings

Index	Port	Mode	Native Vlan	Speed&Duplex	Description	Group Member	Group Status
0	GE_4_1	Enable	1	Auto Negotiation	User Port #0	GROUP_1	Active
1	GE_4_2	Enable	1	Auto Negotiation	User Port #1	GROUP_1	Redundant
2	GE_4_3	Enable	2	Auto Negotiation	User Port #2	GROUP_2	Active
3	GE_4_4	Enable	2	Auto Negotiation	User Port #3	GROUP_2	Redundant

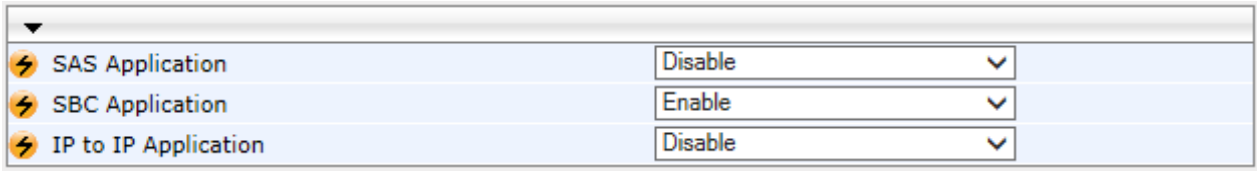
4.2 Step 2: Enable the SBC Application

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 4.16 on page 88).

4.3 Step 3: Signaling Routing Domains Configuration

This step describes how to configure Signaling Routing Domains (SRD). 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 would be required for each one.

The SRD is composed of the following:

- 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

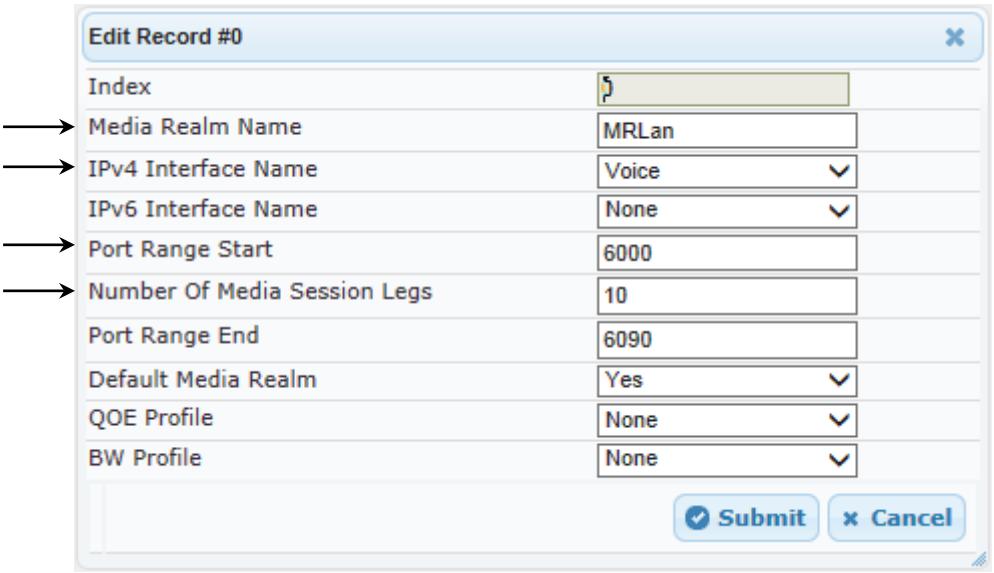
This step describes how to configure Media Realms. The simplest configuration is to create two Media Realms - one for internal (LAN) traffic and one for external (WAN) traffic.

➢ **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	**0**
Media Realm Name	**MRLan** (descriptive name)
IPv4 Interface Name	**Voice**
Port Range Start	**6000** (represents lowest UDP port number used for media on LAN)
Number of Media Session Legs	**10** (media sessions assigned with port range)

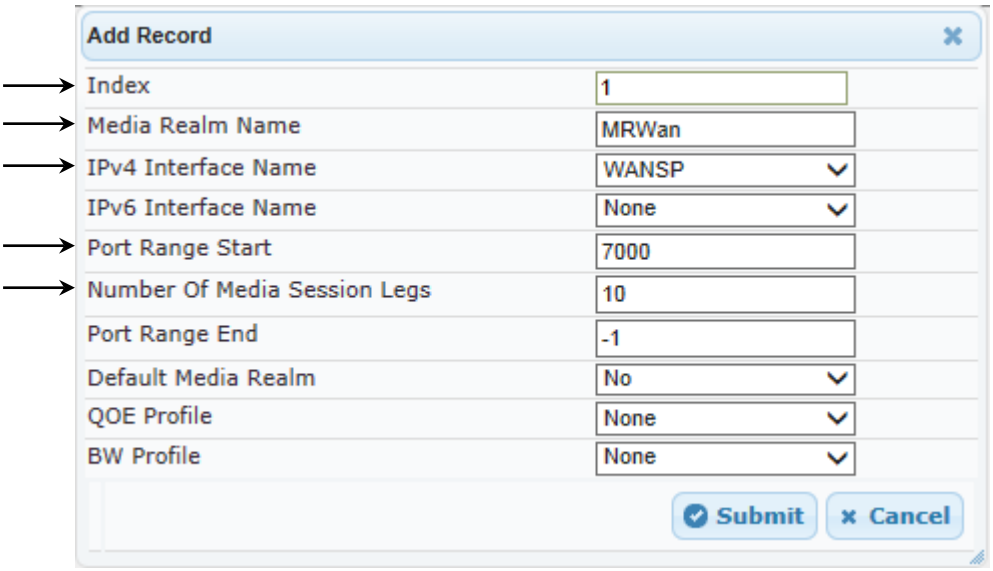
Figure 4-6: Configuring Media Realm for LAN



3. Configure a Media Realm for WAN traffic:

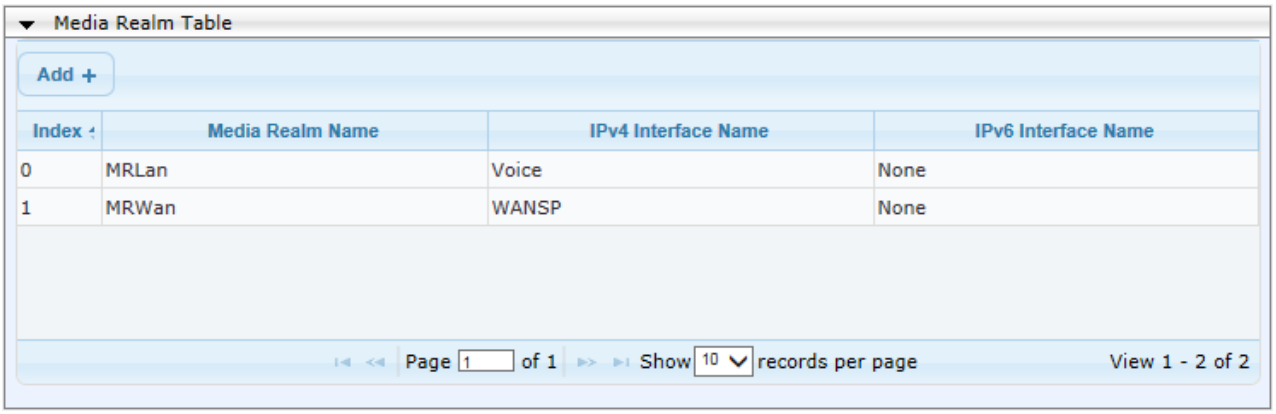
Parameter	Value
Index	**1**
Media Realm Name	**MRWan** (arbitrary name)
IPv4 Interface Name	**WANSP**
Port Range Start	**7000** (represents lowest UDP port number used for media on WAN)
Number of Media Session Legs	**10** (media sessions assigned with port range)

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

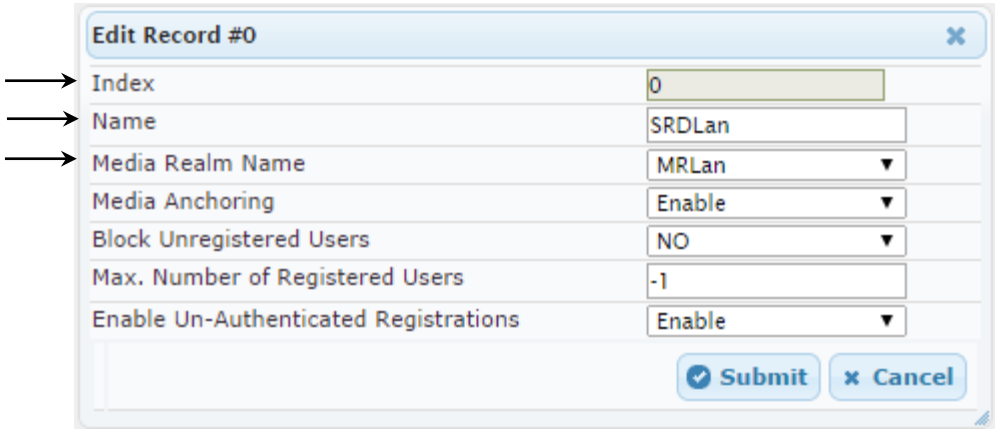
This step describes how to configure the SRDs.

➢ **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
Index	**0**
Name	**SRDLan** (descriptive name for SRD)
Media Realm Name	**MRLan** (associates SRD with Media Realm)

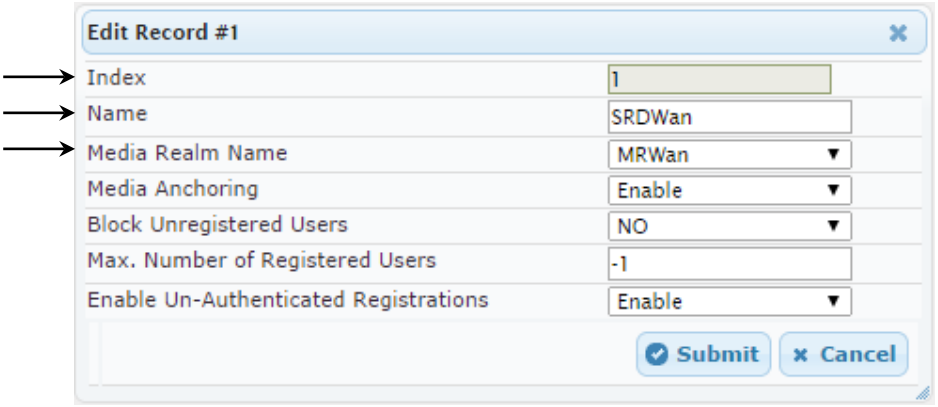
Figure 4-9: Configuring LAN SRD



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

Parameter	Value
Index	**1**
Name	**SRDWan**
Media Realm Name	**MRWan**

Figure 4-10: Configuring WAN SRD



The configured SRDs are shown in the figure below:

Figure 4-11: Configured SRDs in SRD Table

4.3.3 Step 3c: Configure SIP Signaling Interfaces

This step describes 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:

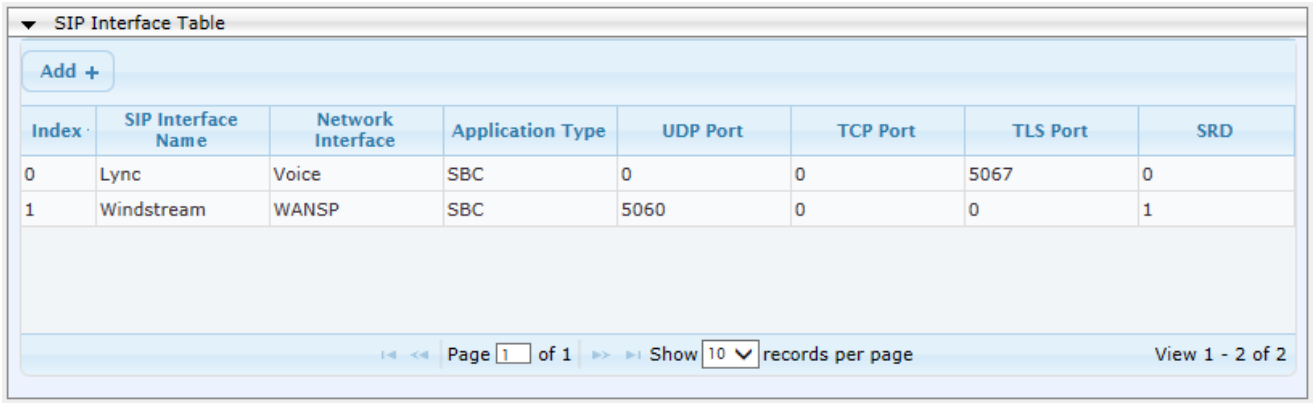
Parameter	Value
Index	**0**
Interface Name	**Lync** (arbitrary descriptive name)
Network Interface	**Voice**
Application Type	**SBC**
TLS Port	**5067**
TCP and UDP	**0**
SRD	**0**

3. Configure a SIP interface for the WAN:

Parameter	Value
Index	**1**
Interface Name	**Windstream** (arbitrary descriptive name)
Network Interface	**WANSP**
Application Type	**SBC**
UDP Port	**5060**
TCP and TLS	**0**
SRD	**1**

The configured SIP Interfaces are shown in the figure below:

Figure 4-12: Configured SIP Interfaces in SIP Interface Table



4.4 Step 4: Configure Proxy Sets

This step describes 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 need to be configured for the following IP entities:

- Microsoft Lync Server 2013
- Windstream SIP Trunk

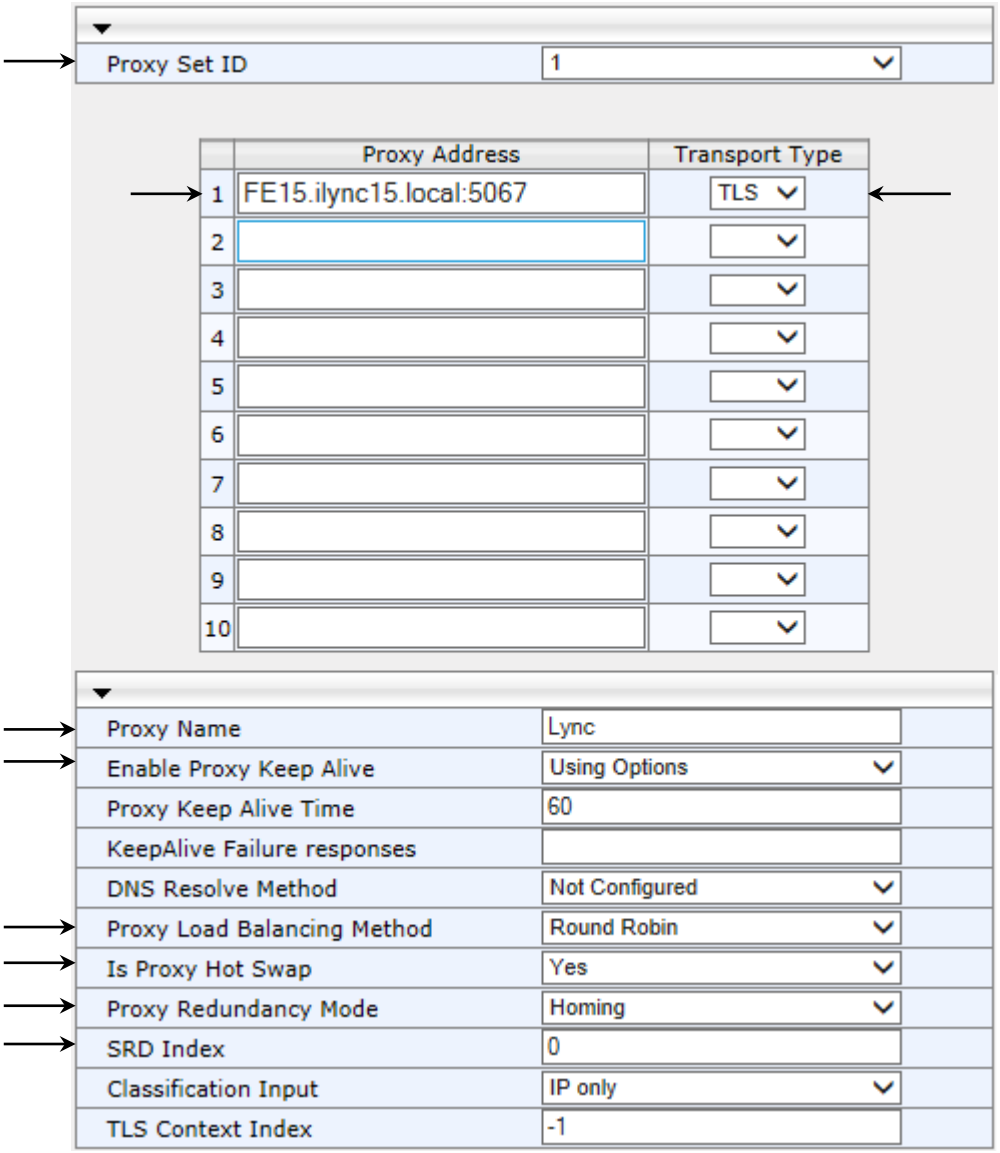
These Proxy Sets will later be associated with IP Groups.

➢ **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	**Lync** (arbitrary descriptive name)
Enable Proxy Keep Alive	**Using Options**
Proxy Load Balancing Method	**Round Robin**
Is Proxy Hot Swap	**Yes**
Proxy Redundancy Mode	**Homing**
SRD Index	**0**

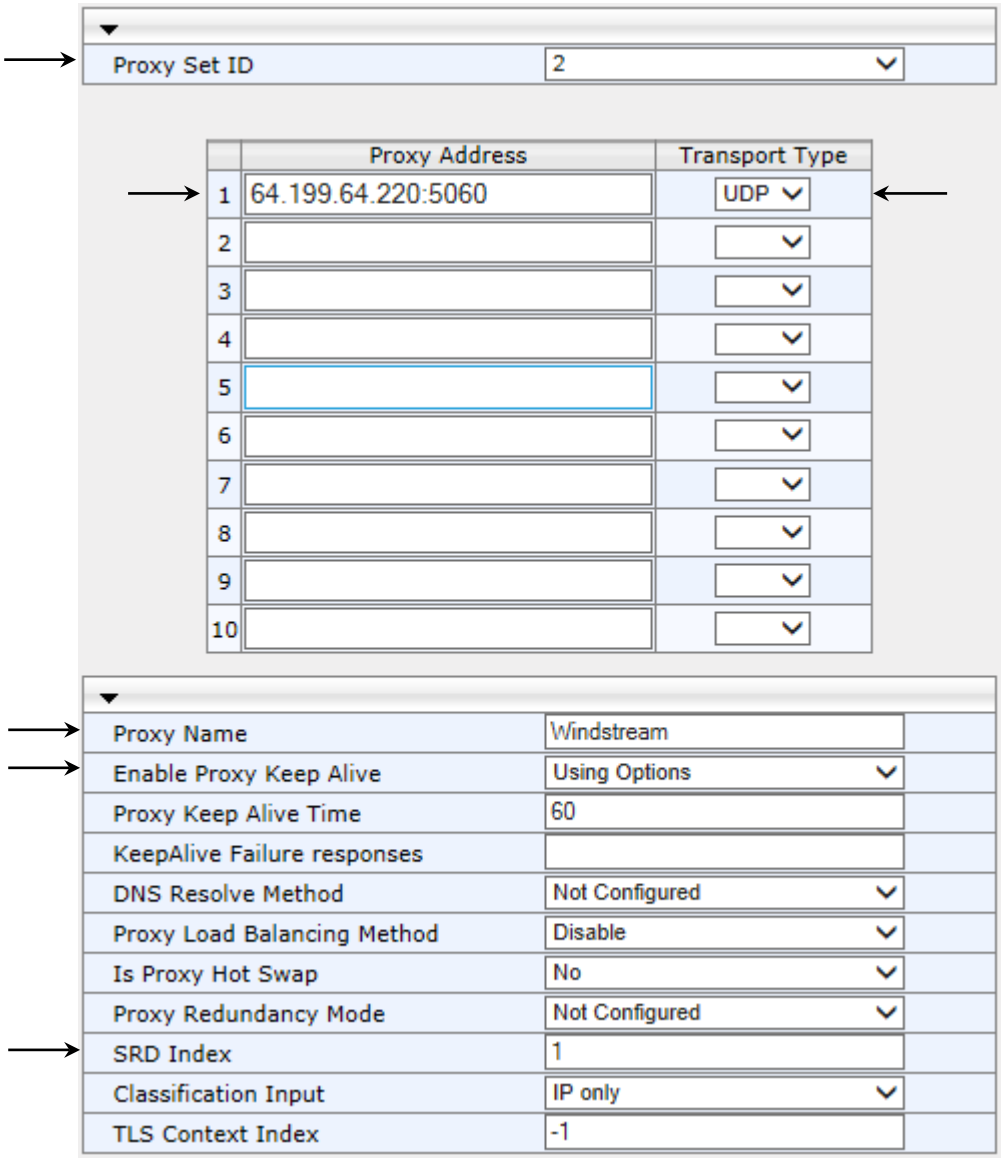
Figure 4-13: Configuring Proxy Set for Microsoft Lync Server 2013



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

Parameter	Value
Proxy Set ID	**2**
Proxy Address	**64.199.64.220:5060** (Windstream IP address / FQDN and destination port)
Transport Type	**UDP**
Proxy Name	**Windstream** (arbitrary descriptive name)
Enable Proxy Keep Alive	**Using Options**
SRD Index	**1** (enables classification by Proxy Set for SRD of IP Group belonging to Windstream SIP Trunk)

Figure 4-14: Configuring Proxy Set for Windstream SIP Trunk



4. Reset the E-SBC with a burn to flash for these settings to take effect (see Section 4.16 on page 88).

4.5 Step 5: Configure IP Groups

This step describes 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. Once IP Groups are configured, they are 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 LAN
- Windstream SIP Trunk located on 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:

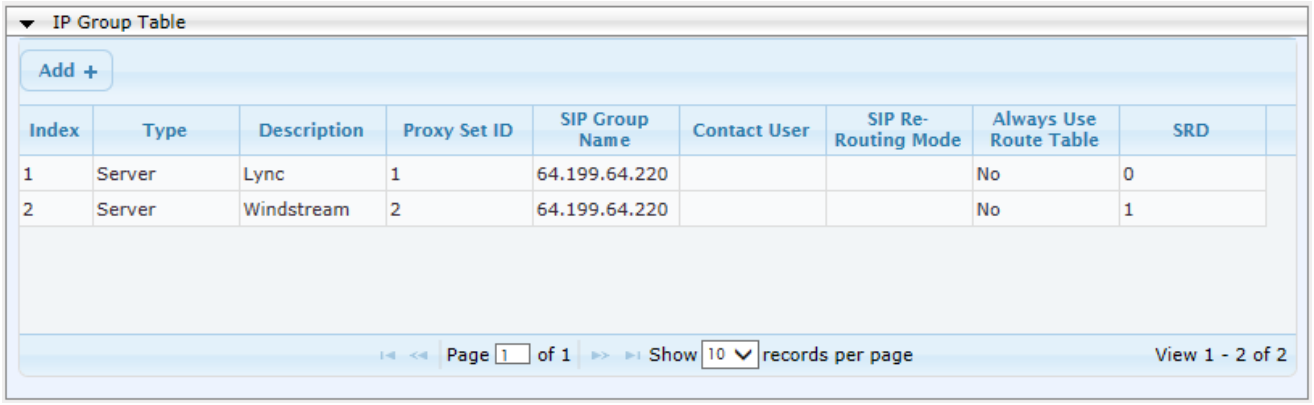
Parameter	Value
Index	**1**
Type	**Server**
Description	**Lync** (arbitrary descriptive name)
Proxy Set ID	**1**
SIP Group Name	**64.199.64.220** (according to ITSP requirement)
SRD	**0**
Media Realm Name	**MRLan**
IP Profile ID	**1**

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

Parameter	Value
Index	**2**
Type	**Server**
Description	**64.199.64.220** (arbitrary descriptive name)
Proxy Set ID	**2**
SIP Group Name	**Windstream** (according to ITSP requirement)
SRD	**1**
Media Realm Name	**MRWan**
IP Profile ID	**2**

The configured IP Groups are shown in the figure below:

Figure 4-15: Configured IP Groups in IP Group Table


IP Group Table

Add +

Index	Type	Description	Proxy Set ID	SIP Group Name	Contact User	SIP Re-Routing Mode	Always Use Route Table	SRD
1	Server	Lync	1	64.199.64.220			No	0
2	Server	Windstream	2	64.199.64.220			No	1

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

4.6 Step 6: Configure IP Profiles

This step describes 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 need to be configured for the following IP entities:

- Microsoft Lync Server 2013 - to operate in secure mode using SRTP and TLS
- Windstream SIP trunk - to operate in non-secure mode using RTP and UDP

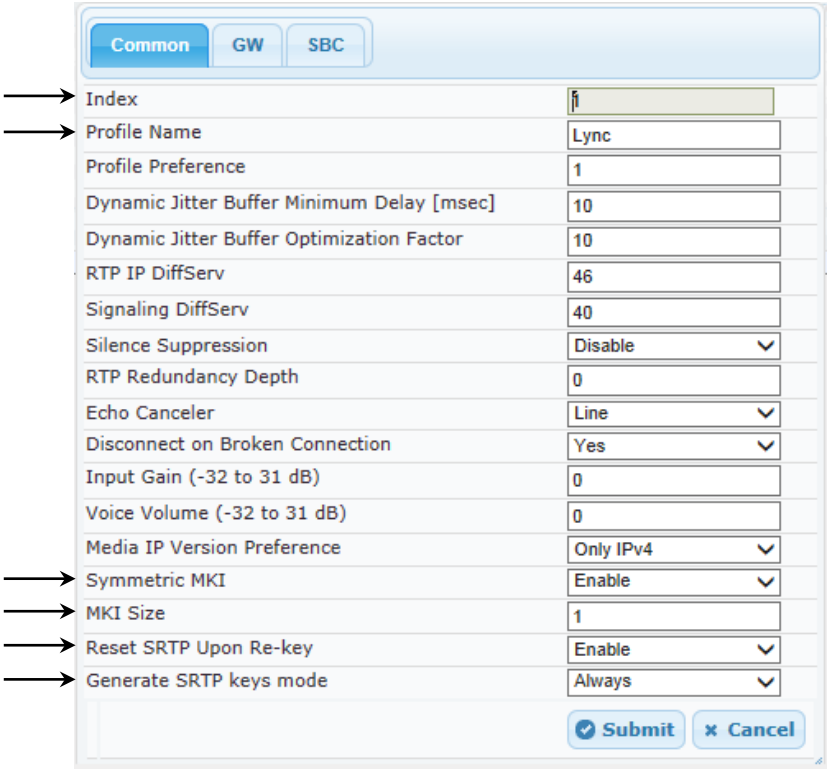
Note that the IP Profiles were assigned to these entities (i.e., IP Groups) in the previous step (see Section 4.5 on page 46).

➢ **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	**Lync** (arbitrary descriptive name)
Symmetric MKI	**Enable**
MKI Size	**1**
Reset SRTP State Upon Re-key	**Enable**
Generate SRTP keys mode:	**Always**

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



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

Parameter	Value
SBC Media Security Behavior	**SRTP**
PRACK Mode	**Optional** (required, as Windstream does not generate PRACK)
Remote Update Support	**Supported Only After Connect**
Remote Re-INVITE	**Supported Only With SDP**
Remote Delayed Offer Support	**Not Supported**
Remote REFER Behavior	**Handle Locally** (required, as Lync Server 2013 does not support receipt of SIP REFER)
Remote 3xx Behavior	**Handle Locally** (required, as Lync Server 2013 does not support receipt of SIP 3xx responses)
Enforce MKI Size	**Enforce**
Remote Early Media RTP Behavior	**Delayed** (required, as Lync Server 2013 does not send RTP immediately to remote side when it sends a SIP 18x response)

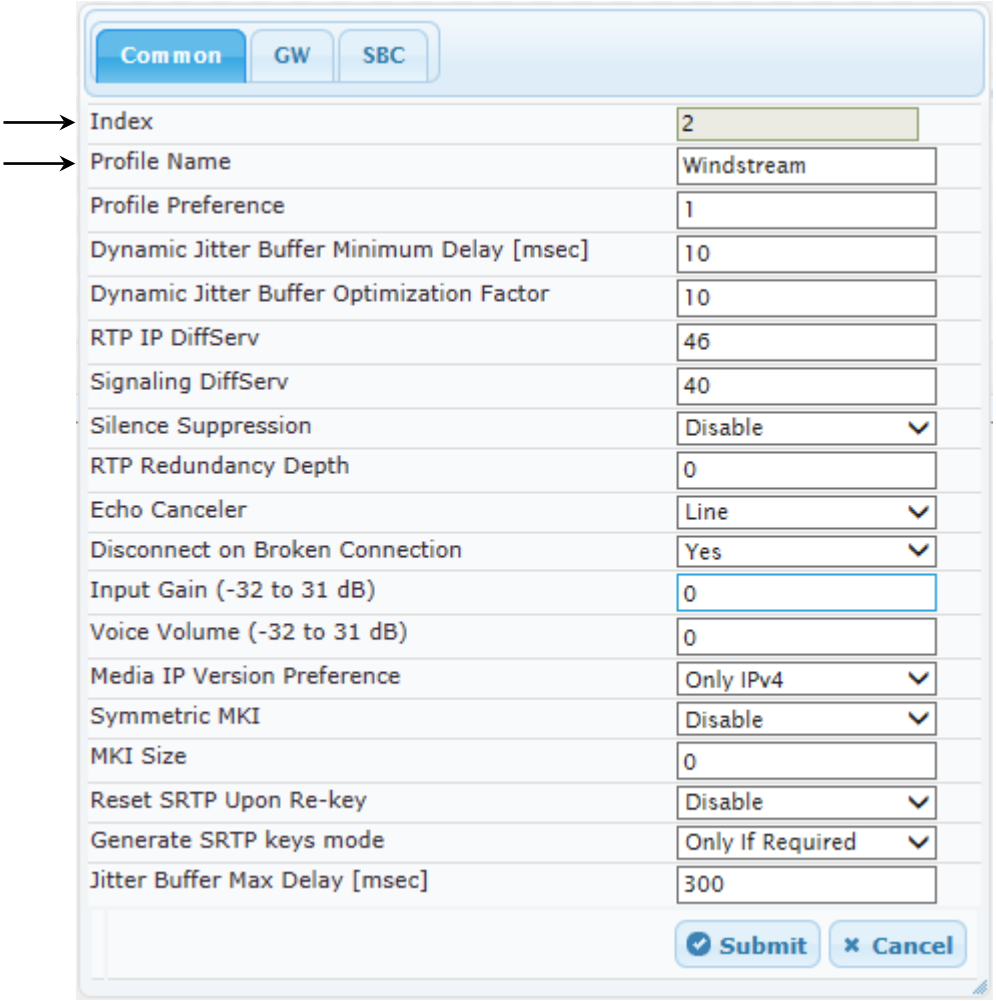
Figure 4-17: Configuring IP Profile for Lync Server 2013 – SBC Tab

➢ **To configure an IP Profile for the Windstream SIP Trunk:**

1. Click **Add**.
2. Click the **Common** tab, and then configure the parameters as follows:

Parameter	Value
Index	**2**
Profile Name	**Windstream** (arbitrary descriptive name)

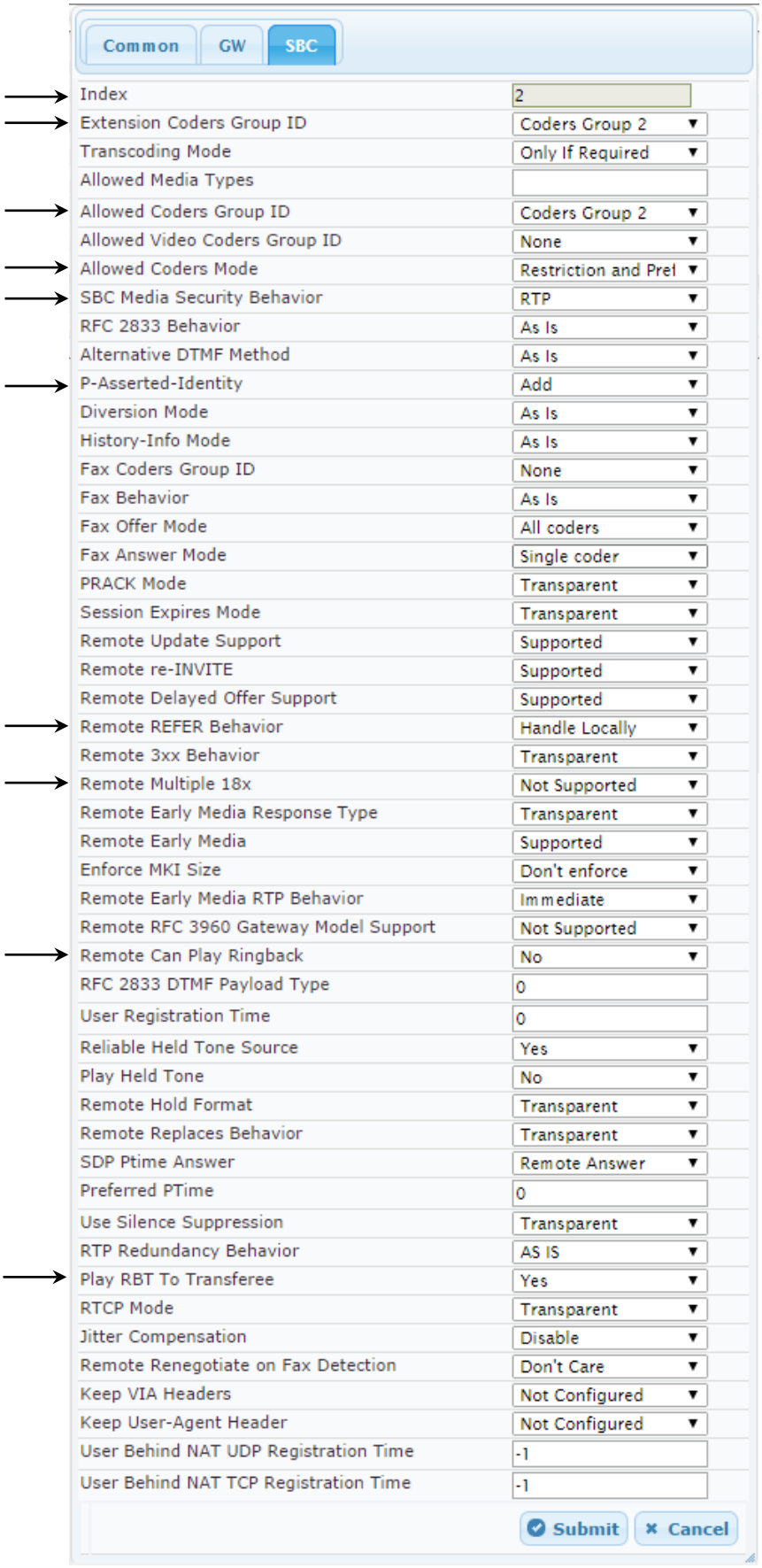
Figure 4-18: Configuring IP Profile for Windstream SIP Trunk – Common Tab



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

Parameter	Value
Index	**2**
Extension Coders Group ID	**Coders Group 2**
Allowed Coders Group ID	**Coders Group 2**
Allowed Coders Mode	**Preference** (lists Allowed Coders first and then original coders in received SDP offer)
SBC Media Security Behavior	**RTP**
P-Asserted-Identity	**Add** (required for anonymous calls)
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, as Lync Server 2013 does not provide a ringback tone for incoming calls)
Play RBT To Transferee	**Yes**

Figure 4-19: Configuring IP Profile for Windstream SIP Trunk – SBC Tab



4.7 Step 7: Configure Coders

This step describes how to configure coders (termed *Coder Group*). As Lync Server 2013 supports the G.711 coder while the network connection to Windstream 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 Windstream 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 48).

➢ **To configure coders:**

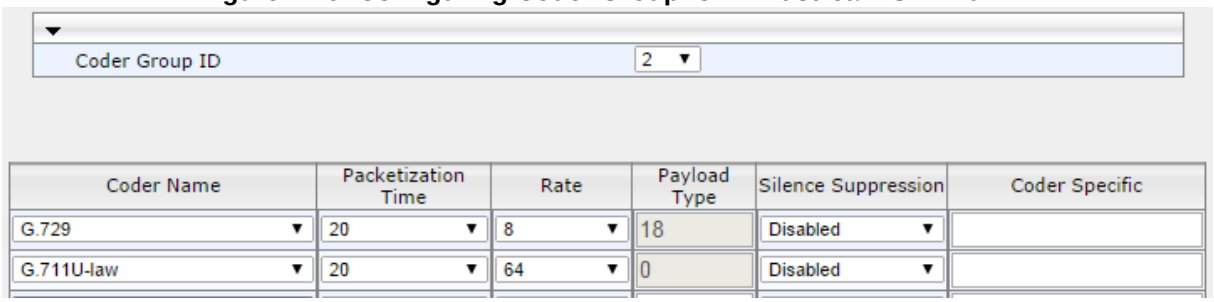
1. Open the Coder Group Settings (**Configuration** tab > **VoIP** menu > **Coders and Profiles** > **Coders Group Settings**).
2. Configure a Coder Group for Windstream SIP Trunk:

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

3. Configure a Coder Group for Windstream SIP Trunk:

Parameter	Value
Coder Group ID	**2**
Coder Name	**G.729**

Figure 4-20: Configuring Coder Group for Windstream SIP Trunk



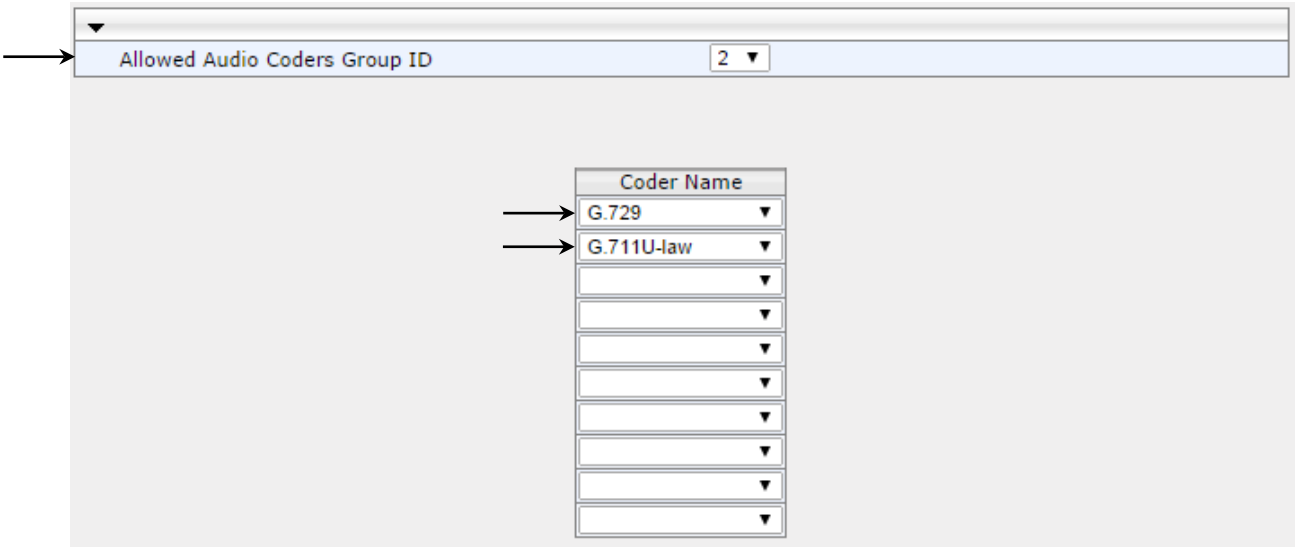
The procedure below describes how to configure an Allowed Coders Group to ensure that voice sent to the Windstream SIP Trunk uses the G.729 coder whenever possible. Note that this Allowed Coders Group ID was assigned to the IP Profile belonging to the Windstream SIP Trunk in the previous step (see Section 4.6 on page 48).

➢ **To set a preferred coder for the Windstream SIP Trunk:**

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

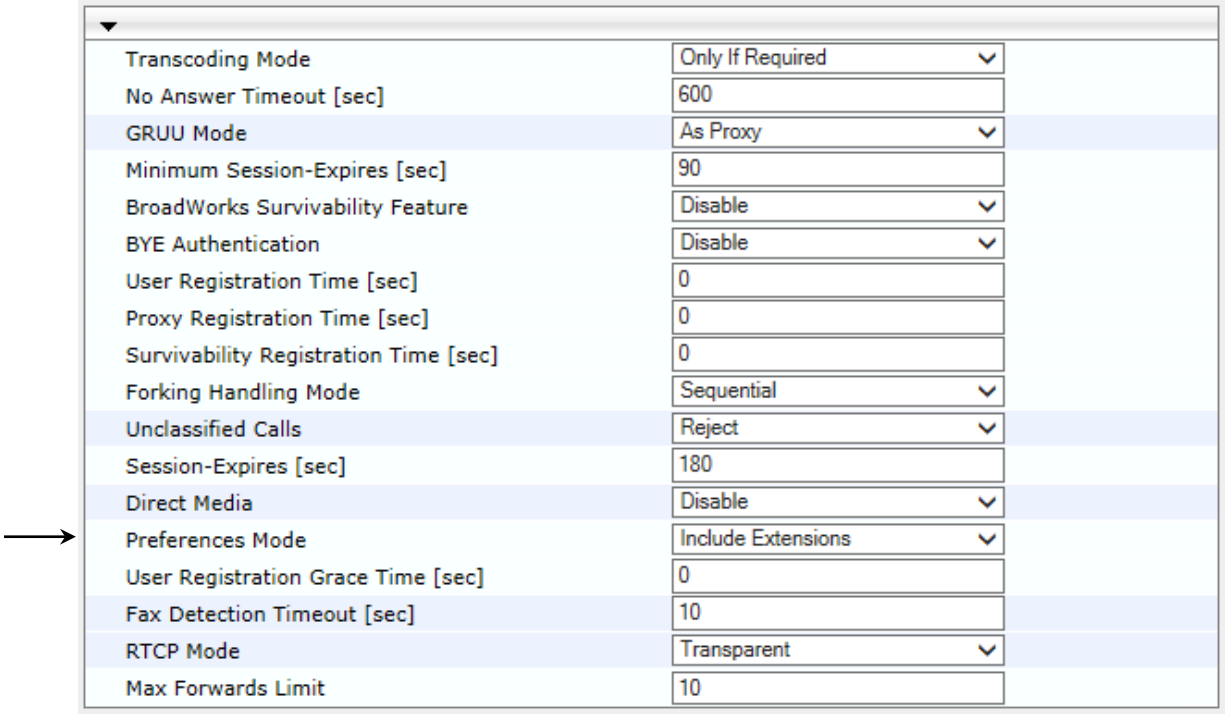
Parameter	Value
Allowed Coders Group ID	**2**
Coder Name	**G.729**
Coder Name	**G.711U-law**

Figure 4-21: Configuring Allowed Coders Group for Windstream SIP Trunk



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

Figure 4-22: SBC Preferences Mode



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

4.8 Step 8: SIP TLS Connection Configuration

This section describes 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.

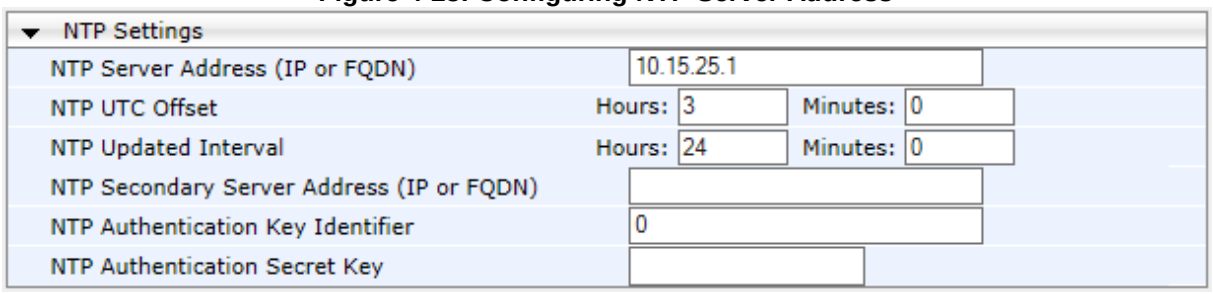
4.8.1 Step 8a: Configure the NTP Server Address

This step describes 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 Address' field, enter the IP address of the NTP server (e.g., **10.15.25.1**).

Figure 4-23: Configuring NTP Server Address



3. Click **Submit**.

4.8.2 Step 8b: Configure a Certificate

This step describes 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.

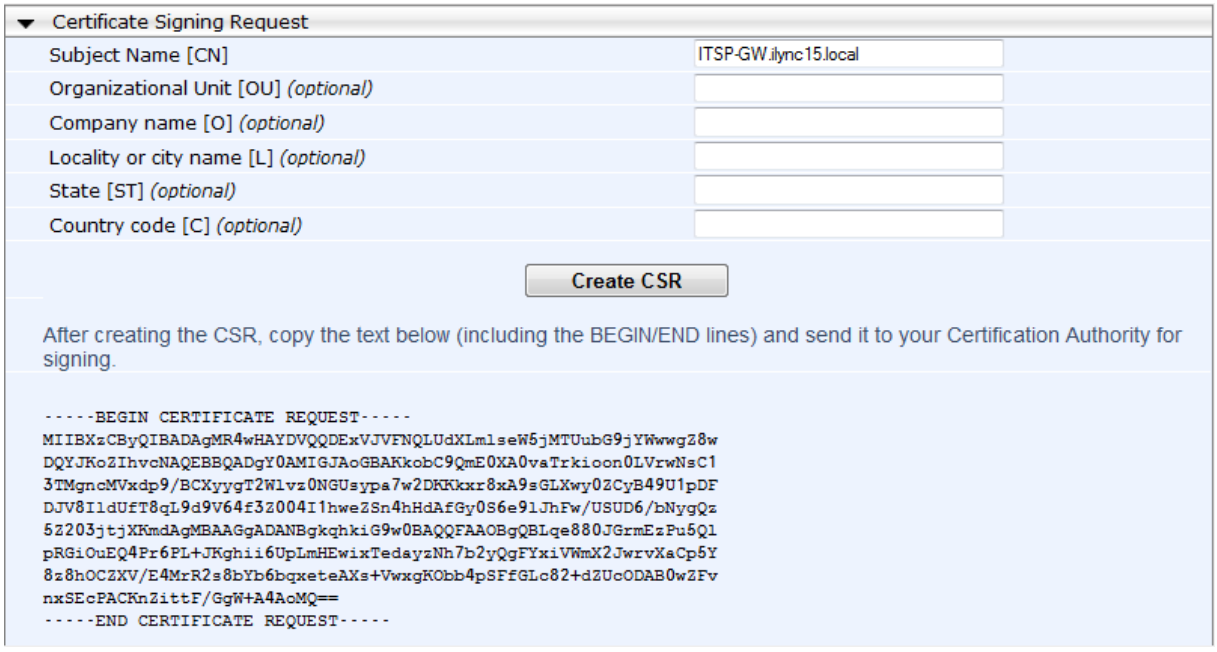
The procedure involves the following main steps:

a. Generating a Certificate Signing Request (CSR).
b. Requesting Device Certificate from CA.
c. Obtaining Trusted Root Certificate from CA.
d. Deploying Device and Trusted Root Certificates on E-SBC.

➢ **To configure a certificate:**

1. Open the TLS Contexts page (**Configuration** tab > **System** menu > **TLS Contexts**).
2. In the TLS Contexts table, select the required TLS Context index row, and then click the **Context Certificates** button, located at the bottom of the TLS Contexts page; the Context Certificates page appears.
3. Under the **Certificate Signing Request** group, do the following:
   a. In the 'Subject Name [CN]' field, enter the E-SBC FQDN name (e.g., **ITSP-GW.ilync15.local**).
   b. Fill in the rest of the request fields according to your security provider's instructions.
4. Click the **Create CSR** button; a textual certificate signing request is displayed in the area below the button:

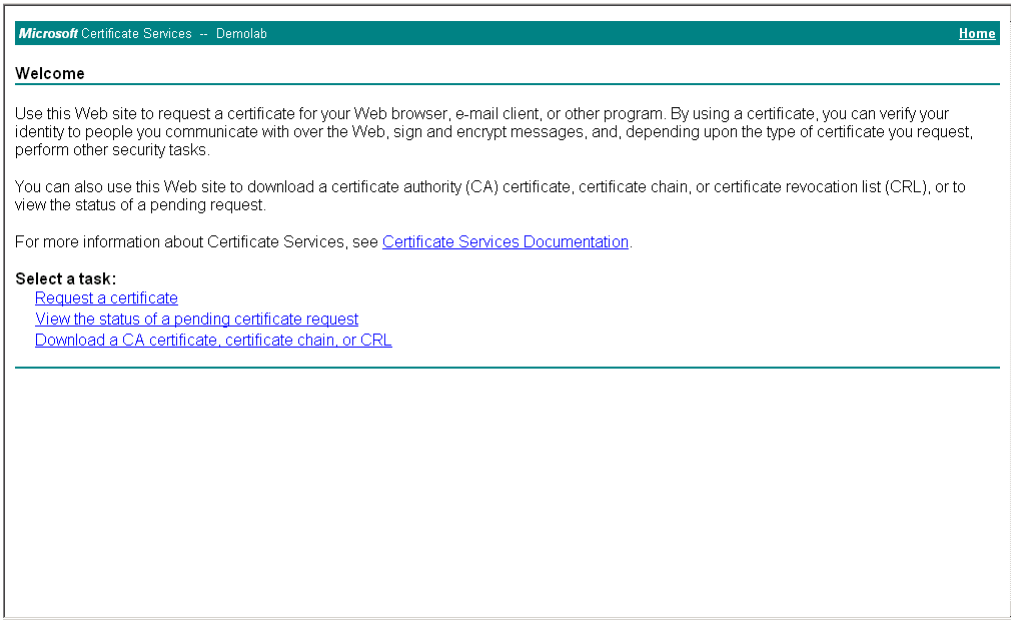
Figure 4-24: Certificate Signing Request – Creating CSR



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.

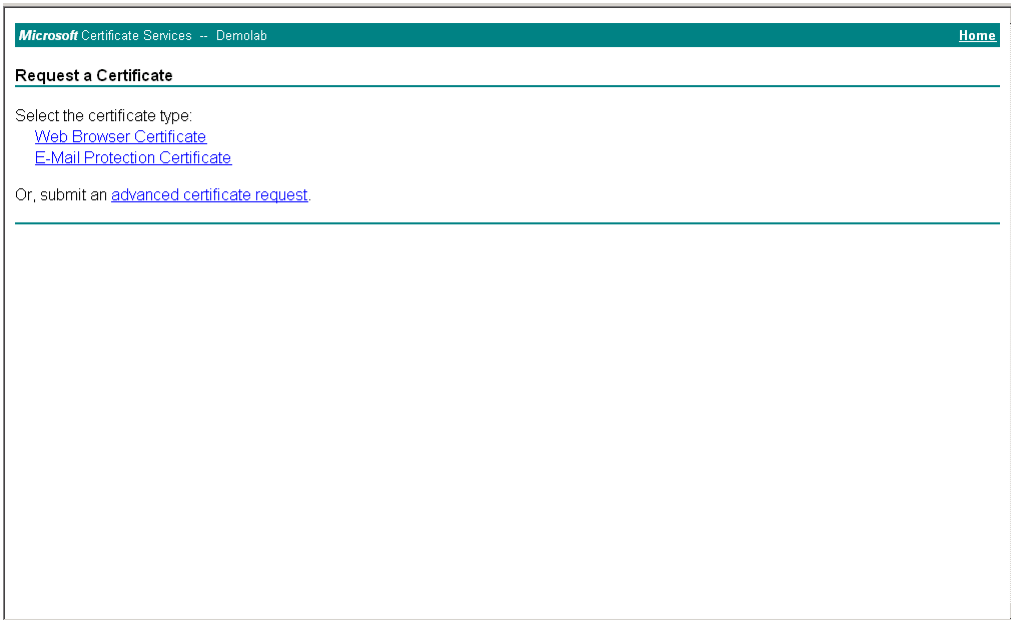
5. Copy the CSR from the line **"----BEGIN CERTIFICATE" to "END CERTIFICATE REQUEST----**" to a text file (such as Notepad), and then save it to a folder on your computer with the file name, *certreq.txt*.
6. Open a Web browser and navigate to the Microsoft Certificates Services Web site at http://<certificate server>/CertSrv.

Figure 4-25: Microsoft Certificate Services Web Page



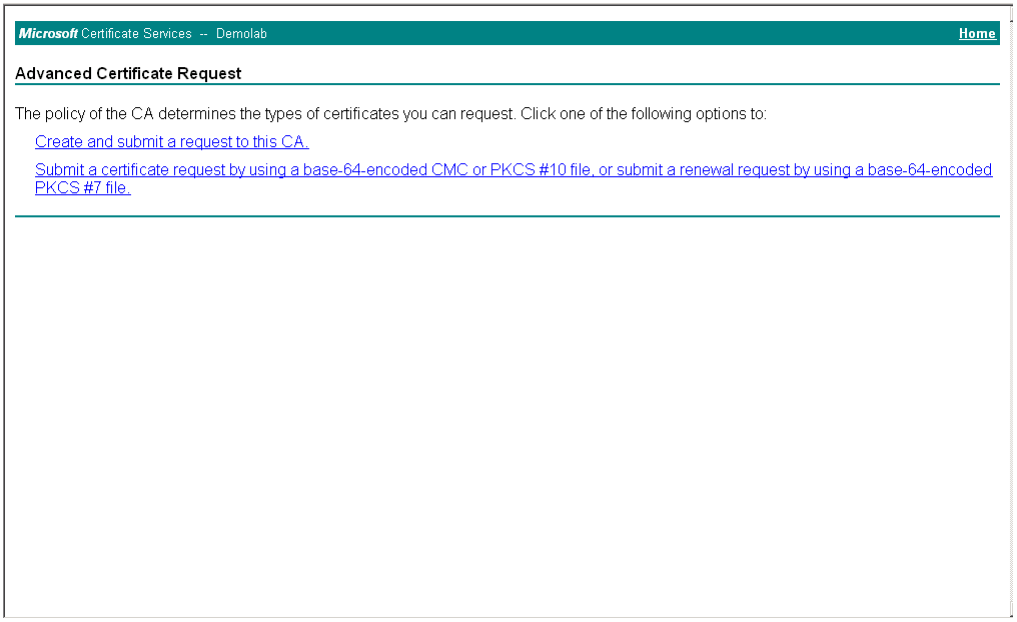
7. Click **Request a certificate**.

Figure 4-26: Request a Certificate Page



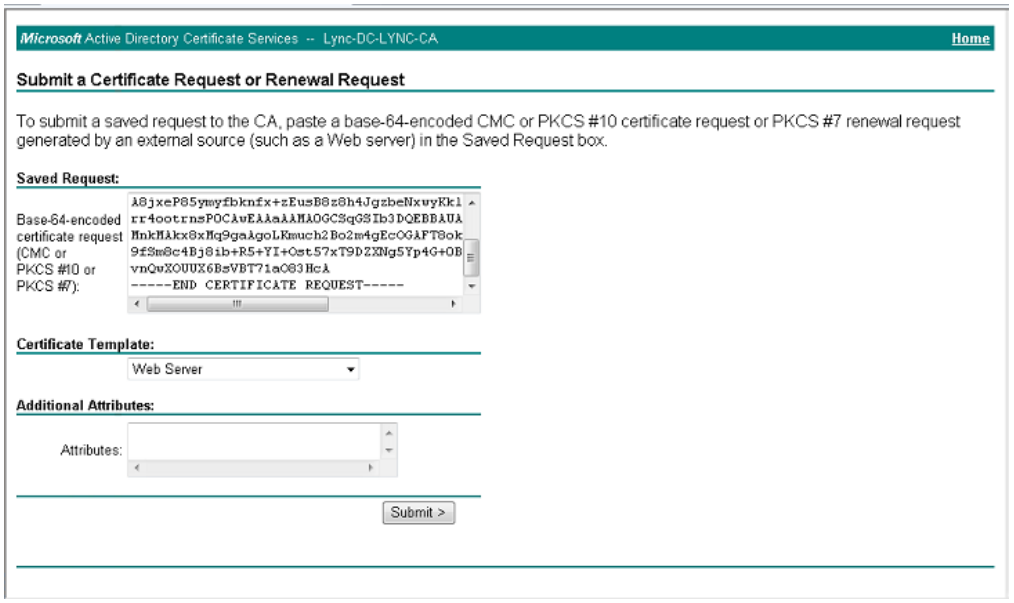
8. Click **advanced certificate request**, and then click **Next**.

Figure 4-27: Advanced Certificate Request Page



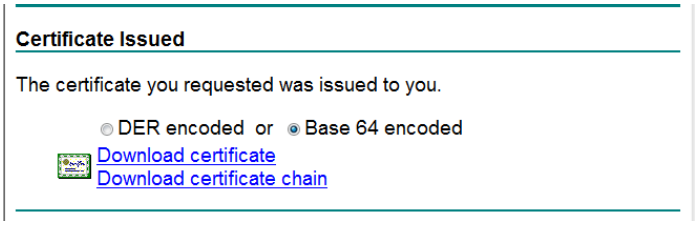
9. Click **Submit a certificate request ...**, and then click **Next**.

Figure 4-28: Submit a Certificate Request or Renewal Request Page



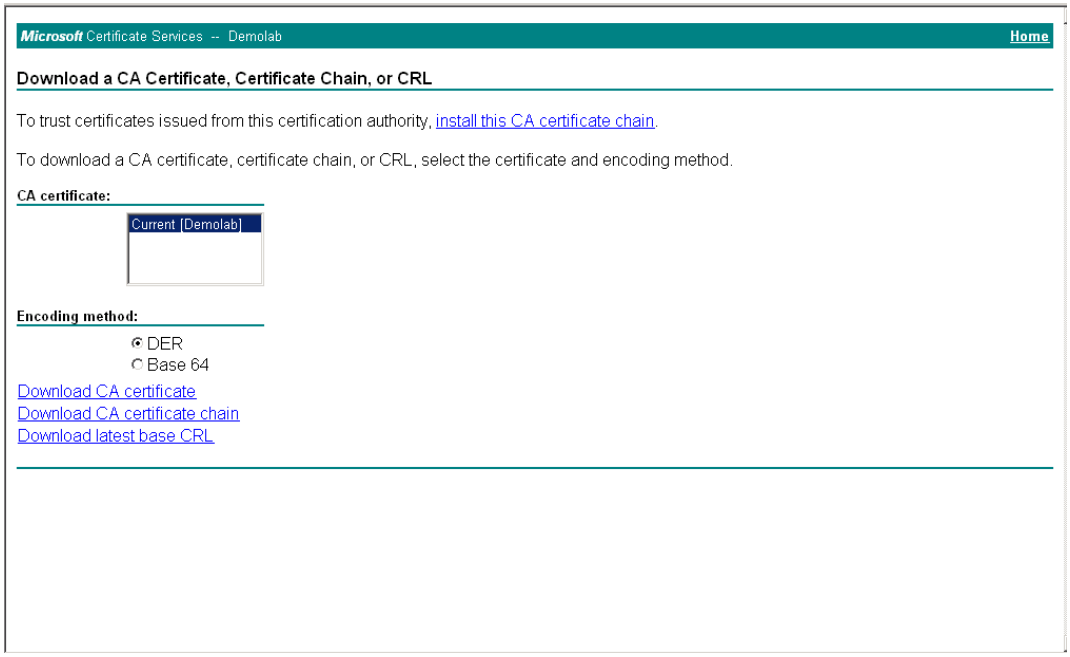
10. Open the *certreq.txt* file that you created and saved in Step 5, and then copy its contents to the 'Saved Request' field.
11. From the 'Certificate Template' drop-down list, select **Web Server**.
12. Click **Submit**.

Figure 4-29: Certificate Issued Page



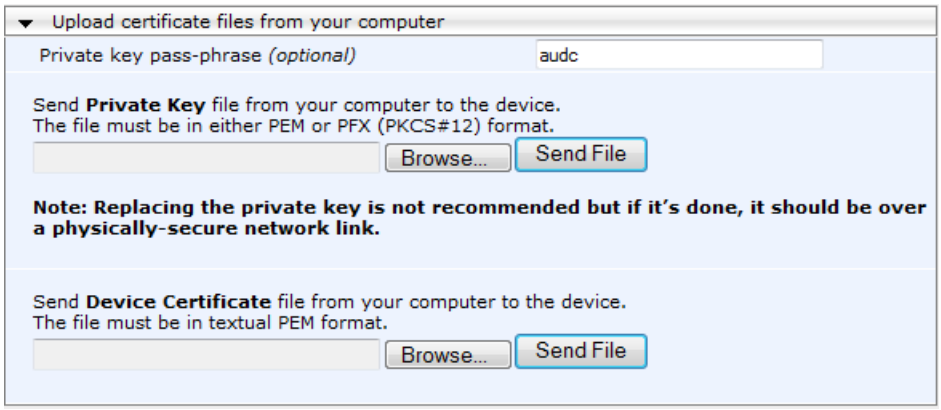
13. Select the **Base 64 encoded** option for encoding, and then click **Download certificate**.
14. Save the file as *gateway.cer* to a folder on your computer.
15. Click the **Home** button or navigate to the certificate server at http://<Certificate Server>/CertSrv.
16. Click **Download a CA certificate**,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

Figure 4-30: Download a CA Certificate, Certificate Chain, or CRL Page



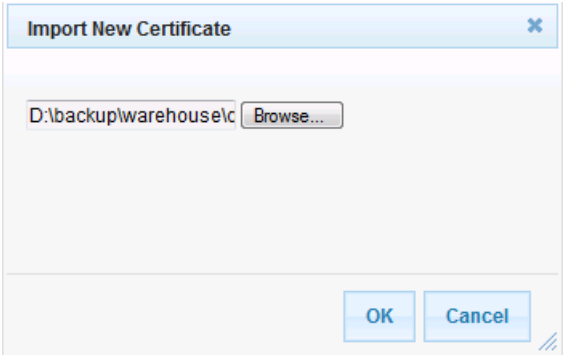
17. Under the 'Encoding method' group, select the **Base 64** option for encoding.
18. Click **Download CA certificate**.
19. Save the file as *certroot.cer* to a folder on your computer.
20. In the E-SBC's Web interface, return to the **TLS Contexts** page and do the following:
    a. Scroll down to the **Upload certificates files from your computer** group, click the **Browse** button corresponding to the 'Send Device Certificate...' field, navigate to the *gateway.cer* certificate file that you saved on your computer in Step 14, and then click **Send File** to upload the certificate to the E-SBC.

Figure 4-31: Upload Device Certificate Files from your Computer Group



b. In the E-SBC's Web interface, return to the **TLS Contexts** page.
c. In the TLS Contexts table, select the required TLS Context index row, and then click the **TLS Context Trusted-Roots Certificates** button, located at the bottom of the TLS Contexts page; the Trusted Certificates page appears.
d. Click the **Import** button, and then select the certificate file to load.

Figure 4-32: Importing Root Certificate into Trusted Certificates Store



21. Click **OK**; the certificate is loaded to the device and listed in the Trusted Certificates store.
22. Reset the E-SBC with a burn to flash for your settings to take effect (see Section 4.16 on page 88).

4.9 Step 9: Configure SRTP

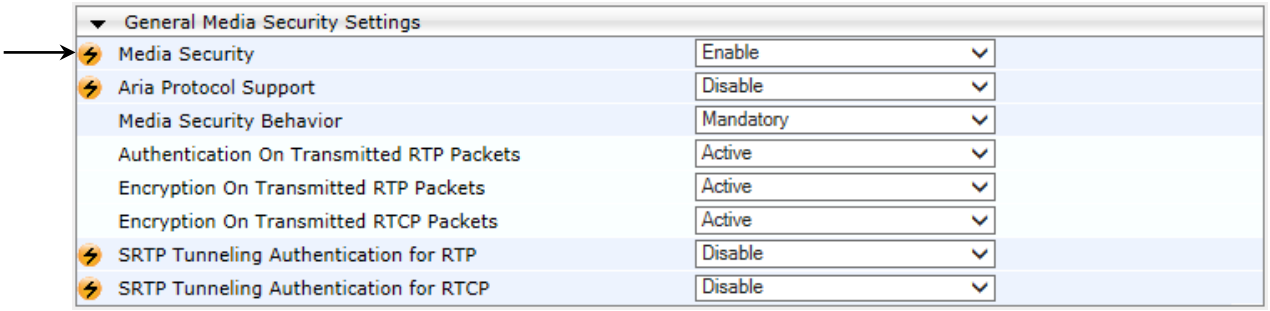
This step describes how to configure media security. If you configure the Microsoft Mediation Server to use SRTP, you need to configure the E-SBC to operate in the same manner. Note that SRTP was enabled for Lync Server 2013 when you configured an IP Profile for Lync Server 2013 (see Section 4.6 on page 48).

➢ **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 4-33: Configuring SRTP



3. Click **Submit**.
4. Reset the E-SBC with a burn to flash for your settings to take effect (see Section 4.16 on page 88).

4.10 Step 10: Configure Maximum IP Media Channels

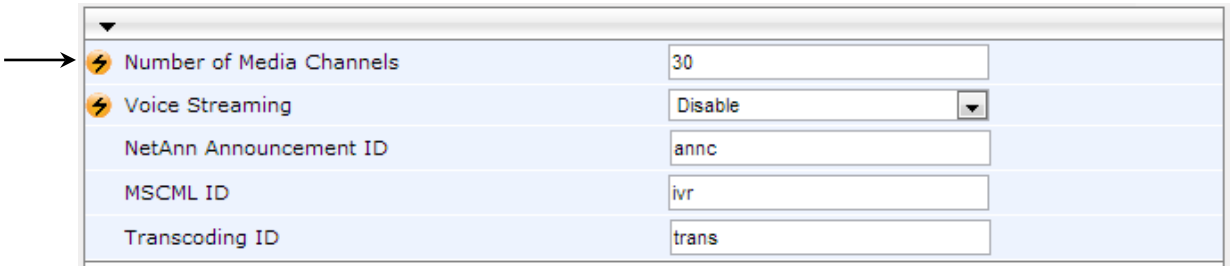
This step describes how to configure the maximum number of required IP media channels. The number of media channels represents the number of DSP channels that the E-SBC allocates to call sessions.

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 4-34: 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 4.16 on page 88).

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

This step describes 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 46, IP Group 1 represents Lync Server 2013, and IP Group 2 represents Windstream SIP Trunk.

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

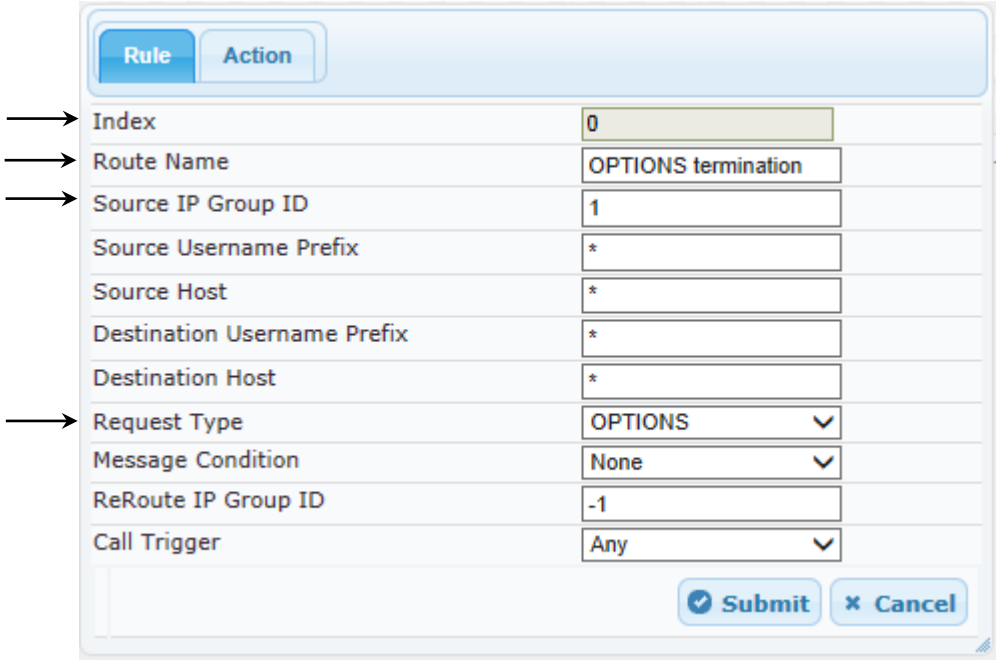
- Terminate SIP OPTIONS messages on the E-SBC that are received from the LAN
- Calls from Lync Server 2013 to Windstream SIP Trunk
- Calls from Windstream SIP Trunk to Lync Server 2013

➢ **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:
3. Click **Add**.
4. Click the **Rule** tab, and then configure the parameters as follows:

Parameter	Value
Index	**0**
Route Name	**OPTIONS termination** (arbitrary descriptive name)
Source IP Group ID	**1**
Request Type	**OPTIONS**

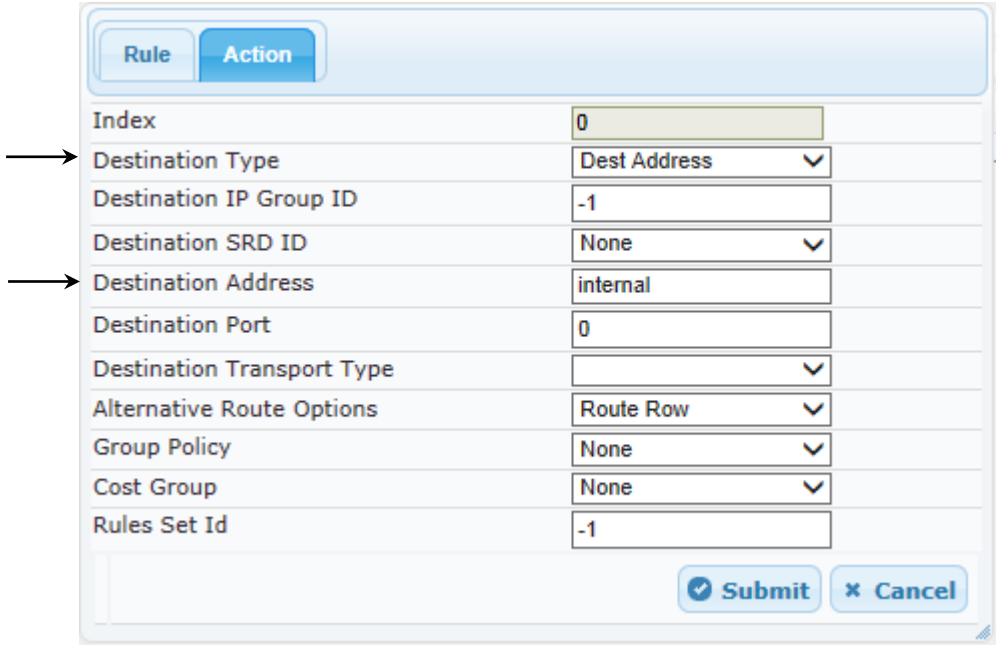
Figure 4-35: Configuring IP-to-IP Routing Rule for Terminating SIP OPTIONS from LAN – Rule Tab



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

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

Figure 4-36: Configuring IP-to-IP Routing Rule for Terminating SIP OPTIONS from LAN – Action Tab

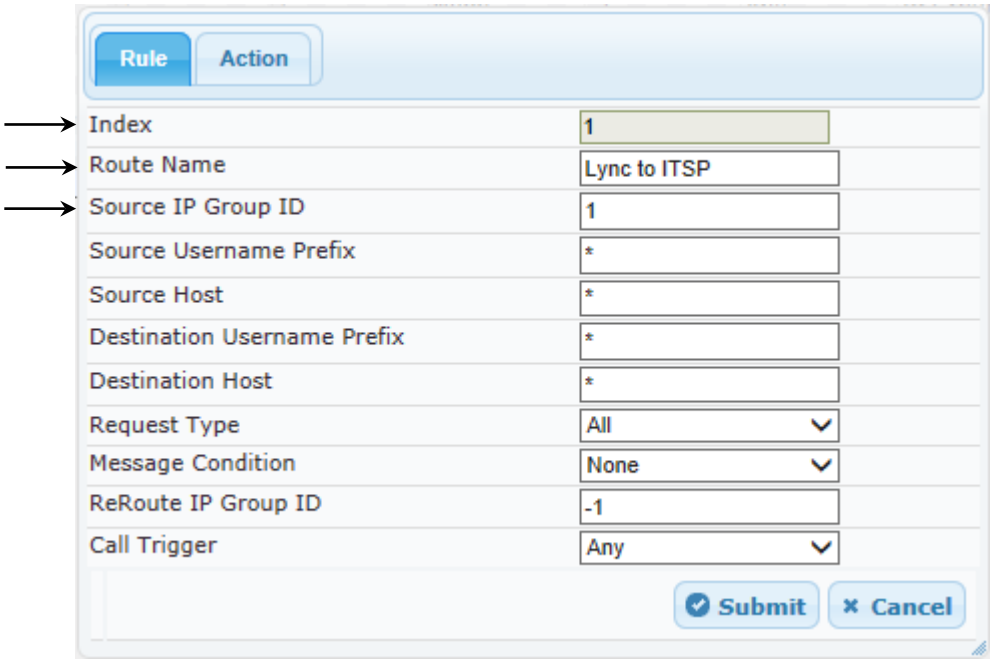


➢ **To configure rule to route calls from Lync Server 2013 to Windstream SIP Trunk:**

1. Click **Add**.
2. Click the **Rule** tab, and then configure the parameters as follows:

Parameter	Value
Index	**1**
Route Name	**Lync to ITSP** (arbitrary descriptive name)
Source IP Group ID	**1**

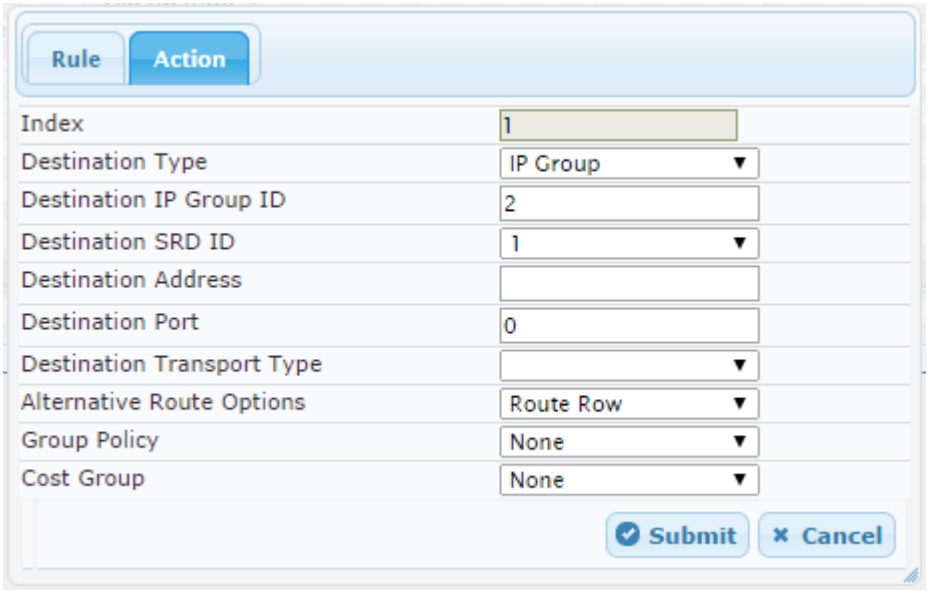
Figure 4-37: Configuring IP-to-IP Routing Rule for Lync to ITSP – Rule tab



3. 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	**1**

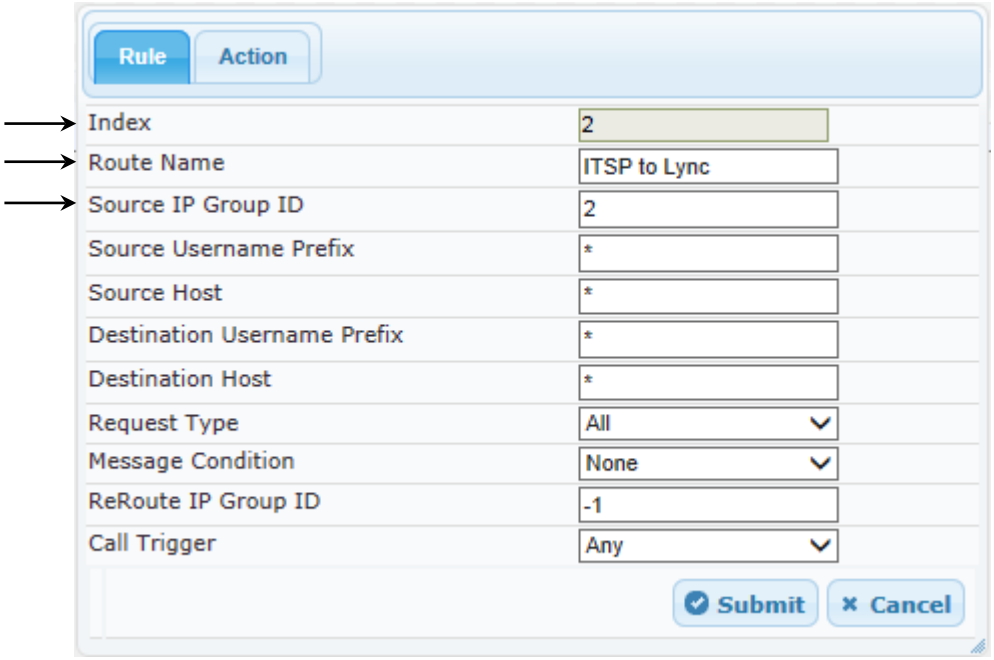
Figure 4-38: Configuring IP-to-IP Routing Rule for Lync to ITSP – Action tab



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

Parameter	Value
Index	**2**
Route Name	**ITSP to Lync** (arbitrary descriptive name)
Source IP Group ID	**2**

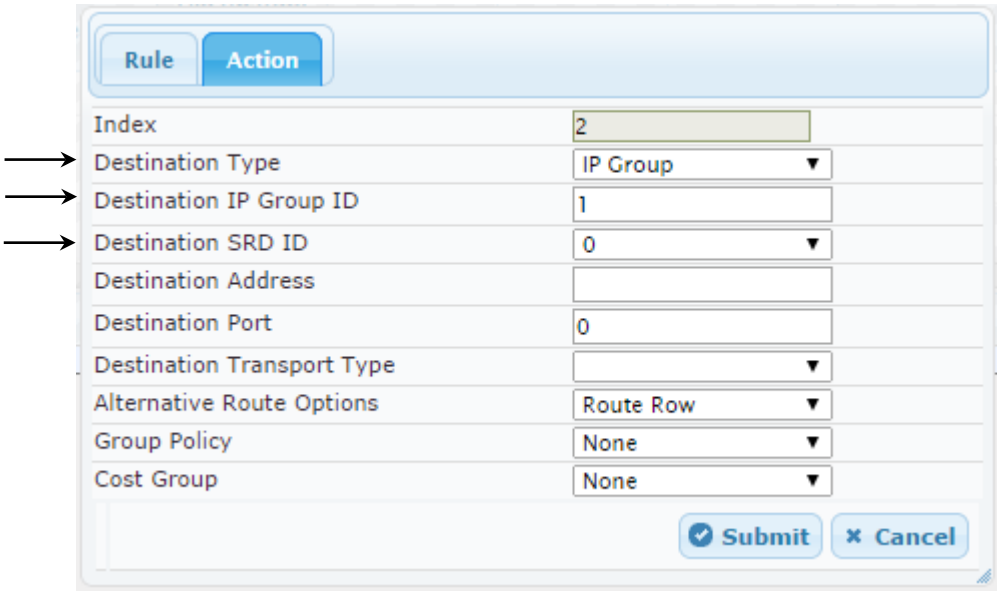
Figure 4-39: Configuring IP-to-IP Routing Rule for ITSP to Lync – Rule tab



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

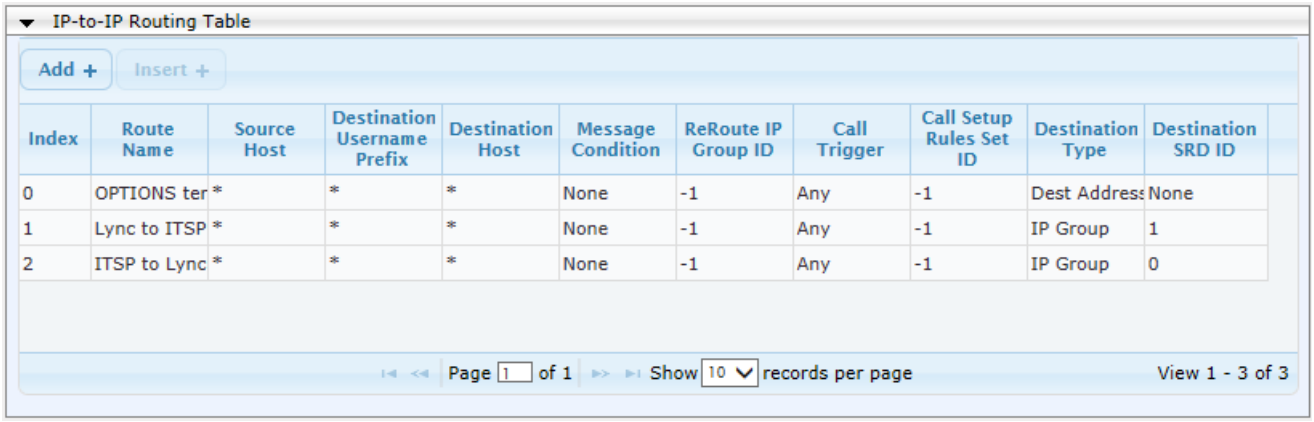
Parameter	Value
Destination Type	**IP Group**
Destination IP Group ID	**1**
Destination SRD ID	**0**

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



The configured routing rules are shown in the figure below:

Figure 4-41: 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 describes 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 46, IP Group 1 represents Lync Server 2013, and IP Group 2 represents Windstream SIP Trunk.

Note: Adapt the manipulation table according to you environment dial plan.

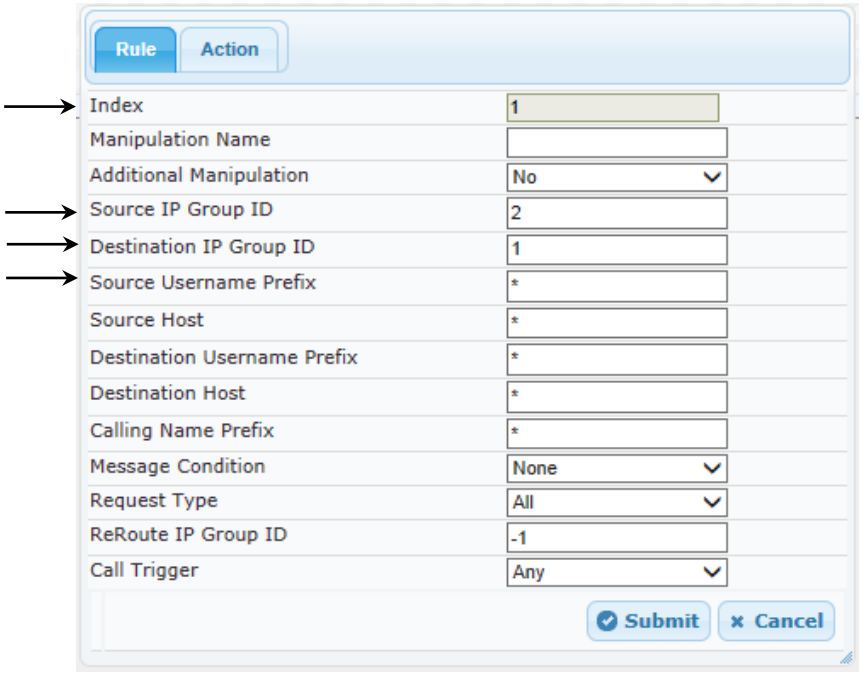
For this interoperability test topology, a manipulation is configured to add the "+" (plus sign) to the destination number for calls from IP Group 2 (Windstream SIP Trunk) to IP Group 1 (i.e., Lync Server 2013) for any destination username prefix.

➢ **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 Outbound**).
2. Click **Add**.
3. Click the **Rule** tab, and then configure the parameters as follows:

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

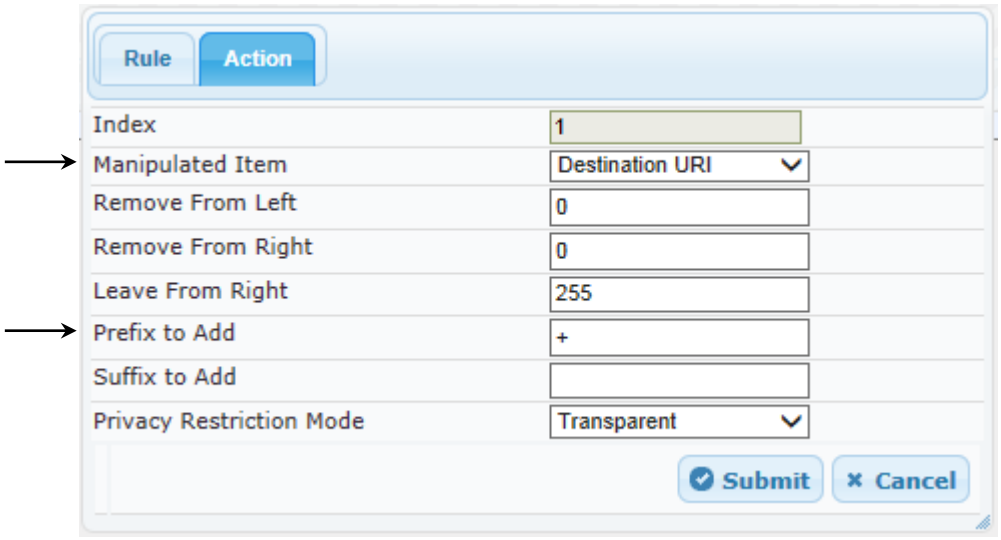
Figure 4-42: Configuring IP-to-IP Outbound Manipulation Rule – Rule Tab



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

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

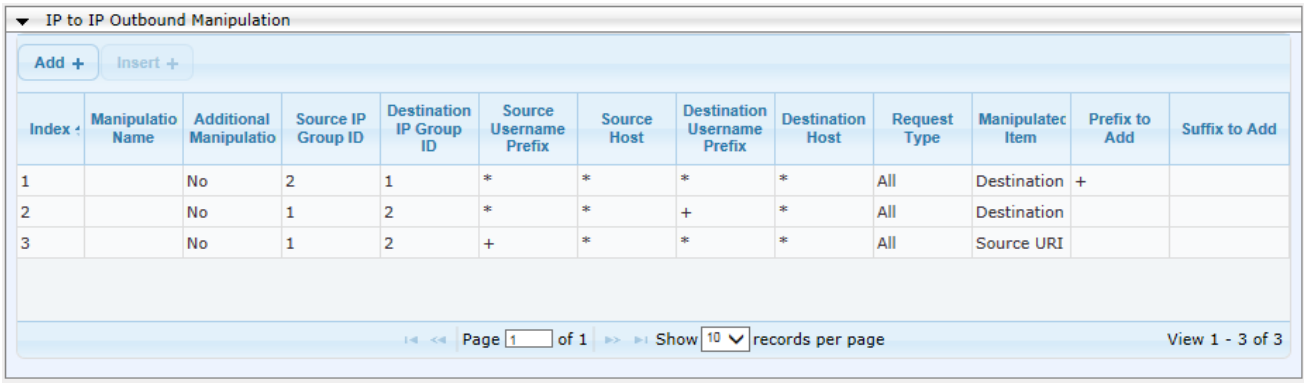
Figure 4-43: Configuring IP-to-IP Outbound Manipulation Rule - Action Tab



5. 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., Windstream SIP Trunk):

Figure 4-44: Example of Configured IP-to-IP Outbound Manipulation Rules



Rule Index	Description
1	Calls from IP Group 2 to IP Group 1 with any destination number (*), add "+" to the prefix of the destination number.
2	Calls from IP Group 1 to IP Group 2 with the prefix destination number "+", remove "+" from this prefix.
3	Calls from IP Group 1 to IP Group 2 with source number prefix "+", remove the "+" from this prefix.

4.13 Step 13: Configure Message Manipulation Rules

This step describes 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).

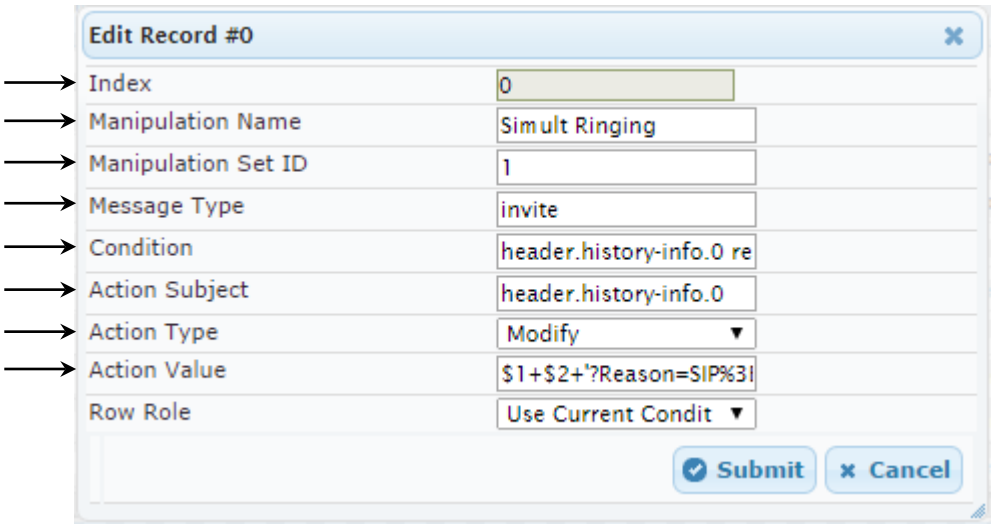
Once you have configured the SIP message manipulation rules, you need to assign them to the relevant IP Group (in the IP Group table) and determine whether they must be applied to inbound or outbound messages.

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

1. Open the Message Manipulations page (**Configuration** tab > **VoIP** menu > **SIP Definitions** > **Msg Policy & Manipulation** > **Message Manipulations**).
2. Configure a new manipulation rule (Manipulation Set 1) for Lync Server 2013. This rule applies to messages received from the Lync Server 2013 (IP Group 1), for simultaneous ringing initiated by the Lync Server 2013 (IP Group 1). This adds an Action Value containing the Reason for the History-Info header, causing the E-SBC to add a Diversion Header towards the SIP Trunk.

Parameter	Value
Index	**0**
Manipulation Name	**Simult Ringing**
Manipulation Set ID	**1**
Message Type	**invite**
Condition	**header.history-info.0 regex (<.*)(user=phone)(>)(.*)**
Action Subject	**header.history-info.0**
Action Type	**Modify**
Action Value	**$1+$2+'?Reason=SIP%3Bcause%3D404'+$3+$4**

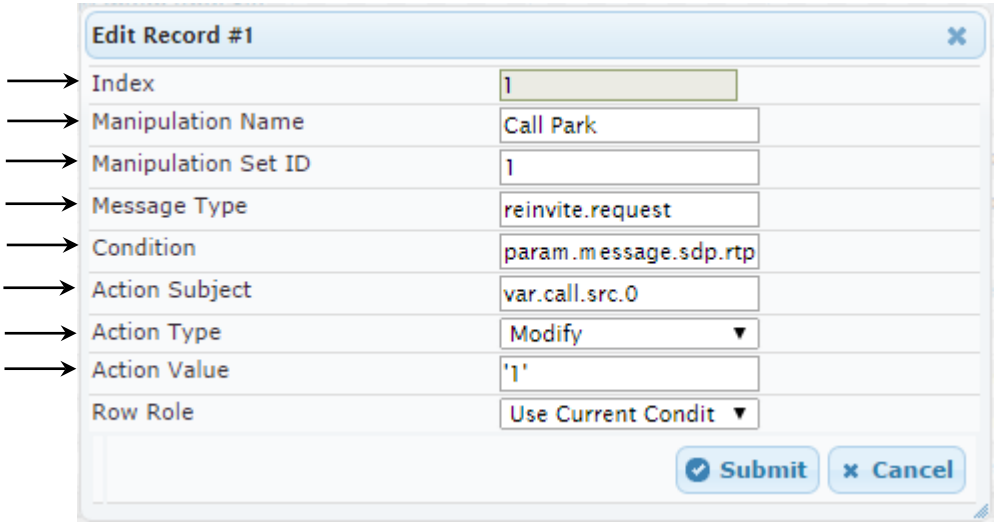
Figure 4-45: Configuring SIP Message Manipulation Rule 0 (for Lync Server 2013)



3. For every SIP Re-INVITE request with SDP, where RTP mode = "sendonly" (occurs in a Lync 2013-initiated Hold), create a variable and set it to **'1'**. This variable manages how the call will be handled in each state (answer, request, etc.).

Parameter	Value
Index	**1**
Manipulation Name	**Call Park**
Manipulation Set ID	**1**
Message Type	**reinvite.request**
Condition	**param.message.sdp.rtpmode=='sendonly'**
Action Subject	**var.call.src.0**
Action Type	**Modify**
Action Value	**'1'**
Row Role	**Use Current Condition**

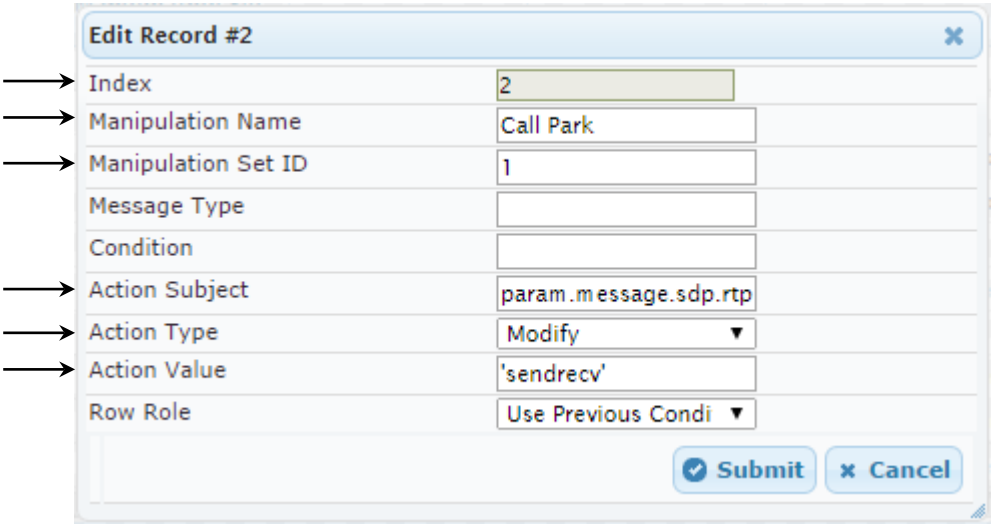
Figure 4-46: Configuring SIP Message Manipulation Rule 1 (for Microsoft Lync)



4. If the manipulation rule Index 1 (above) is executed, then the following rule is also executed on the same SIP message: if RTP mode within the SDP is set to “sendonly” change it to “sendrecv”.

Parameter	Value
Index	**2**
Manipulation Name	**Call Park**
Manipulation Set ID	**1**
Message Type	
Condition	
Action Subject	**param.message.sdp.rtpmode**
Action Type	**Modify**
Action Value	**‘sendrecv’**
Row Role	**Use Previous Condition**

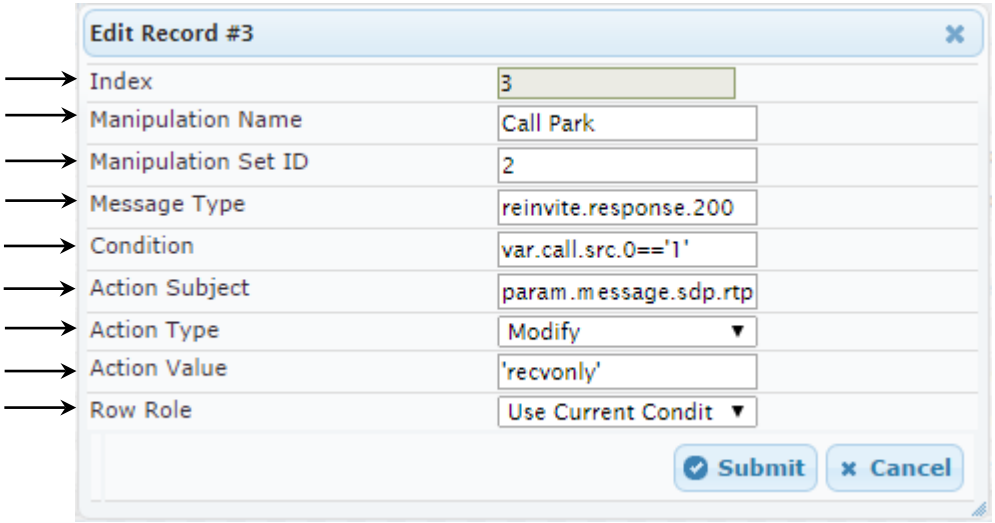
Figure 4-47: Configuring SIP Message Manipulation Rule 2 (for Microsoft Lync)



5. The following rule attempts to normalize the call processing state back to Lync 2013 for the correct reply to the initially received “sendonly”. For every SIP Re-INVITE message with the variable set to **'1'**, change RTP mode to "recvonly". This SIP Re-INVITE message is the response sent from the Windstream SIP Trunk to the Lync initiated Hold.

Parameter	Value
Index	**3**
Manipulation Name	**Call Park**
Manipulation Set ID	**2**
Message Type	**reinvite.response.200**
Condition	**var.call.src.0==”1”**
Action Subject	**param.message.sdp.rtpmode**
Action Type	**Modify**
Action Value	**‘recvonly’**
Row Role	**Use Current Condition**

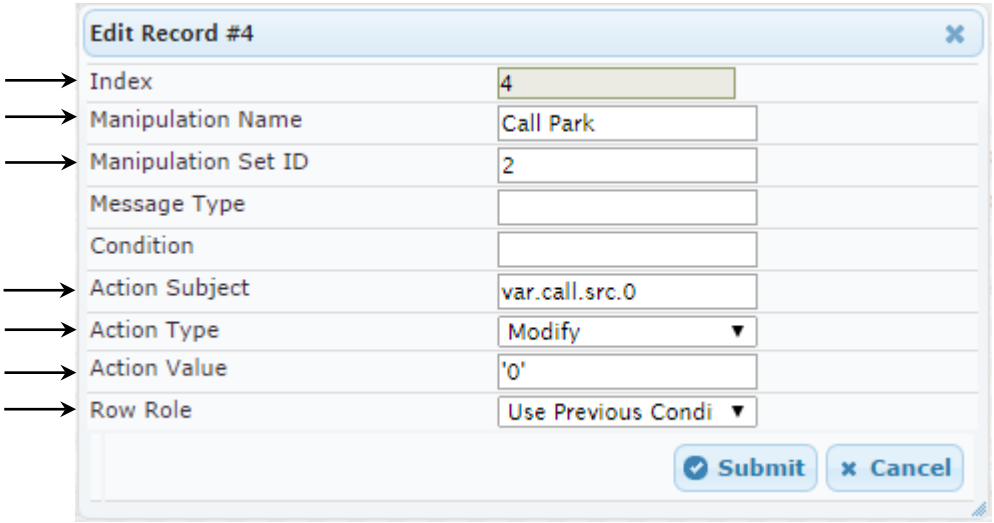
Figure 4-48: Configuring SIP Message Manipulation Rule 3 (for Microsoft Lync)



6. If the manipulation rule Index 3 (above) is executed, then the following rule is also executed. If the variable is determined to be set to **"1"** (in the previous manipulation rule), then set it to **"0"** to normalize the call processing state. Lync now sends Music on Hold to the Windstream SIP Trunk. The call is now truly on hold with Music on Hold.

Parameter	Value
Index	**4**
Manipulation Name	**Call Park**
Manipulation Set ID	**2**
Message Type	
Condition	
Action Subject	**var.call.src.0**
Action Type	**Modify**
Action Value	**'0'**
Row Role	**Use Previous Condition**

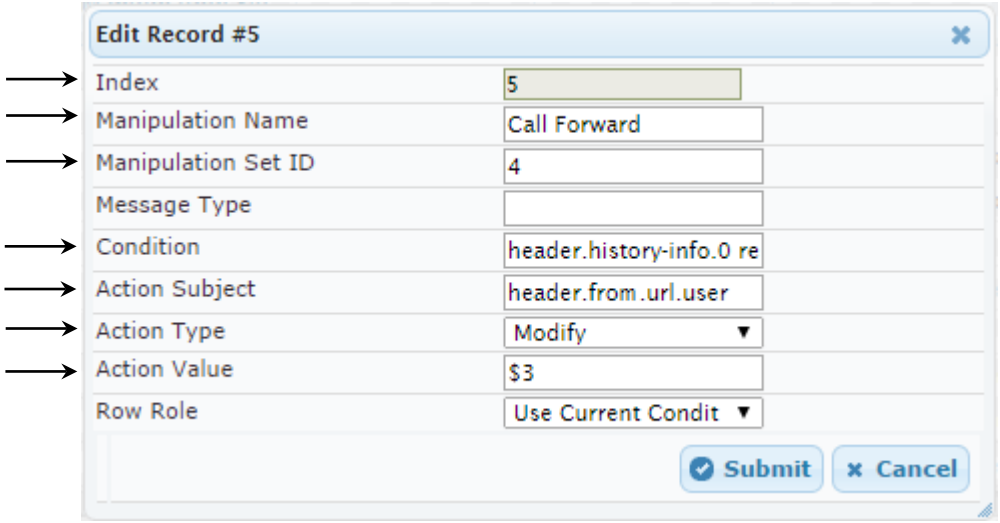
Figure 4-49: Configuring SIP Message Manipulation Rule 4 (for Microsoft Lync)



7. Configure another manipulation rule (Manipulation Set 4) for Windstream SIP Trunk. This rule applies to messages sent to the Windstream SIP Trunk (IP Group 2) in call forward scenario. This replaces the user part of the From Header with the value from History Info Header.

Parameter	Value
Index	**5**
Manipulation Name	**Call Forward**
Manipulation Set ID	**4**
Message Type	
Condition	**header.history-info.0 regex (<sip:)(.)(.*)(@)(.*)**
Action Subject	**header.from.url.user**
Action Type	**Modify**
Action Value	**$3**

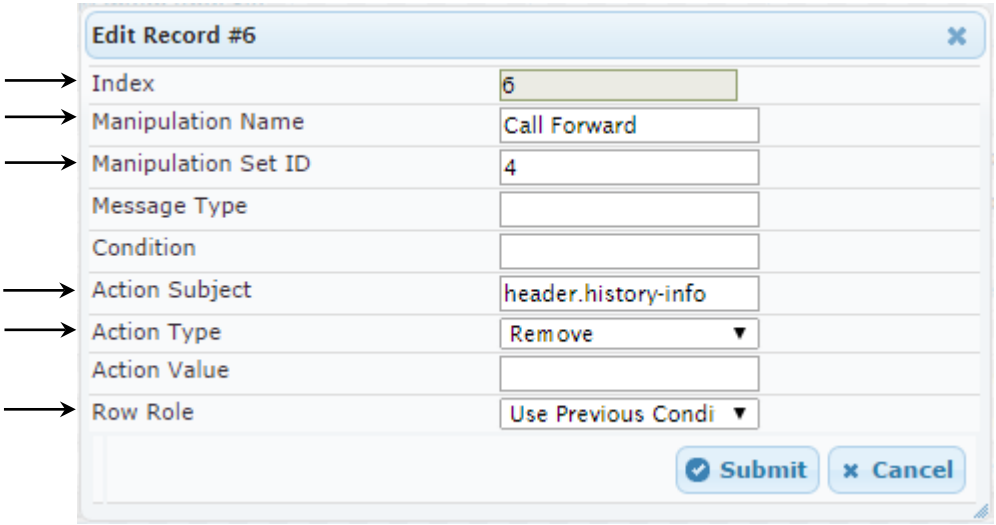
Figure 4-50: Configuring SIP Message Manipulation Rule 5 (for Windstream SIP Trunk)



8. If the manipulation rule Index 5 (above) is executed, then the following rule is also executed. It removes the History Info Header.

Parameter	Value
Index	**6**
Manipulation Name	**Call Forward**
Manipulation Set ID	**4**
Message Type	
Condition	
Action Subject	**header.history-info**
Action Type	**Remove**
Action Value	
Row Role	**Use Previous Condition**

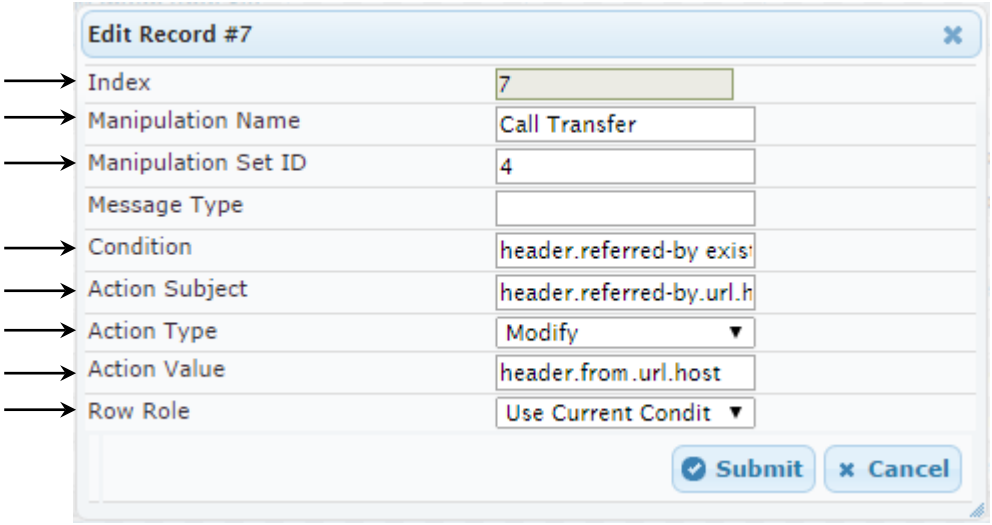
Figure 4-51: Configuring SIP Message Manipulation Rule 6 (for Windstream SIP Trunk)



9. Configure another manipulation rule (Manipulation Set 4) for Windstream SIP Trunk. This rule applies to messages sent to the Windstream SIP Trunk (IP Group 2). This replaces the host part of the Referred-By Header with the value from the SIP From Header.

Parameter	Value
Index	**7**
Manipulation Name	**Call Transfer**
Manipulation Set ID	**4**
Message Type	
Condition	**header.referred-by exists**
Action Subject	**header.referred-by.url.host**
Action Type	**Modify**
Action Value	**header.from.url.host**

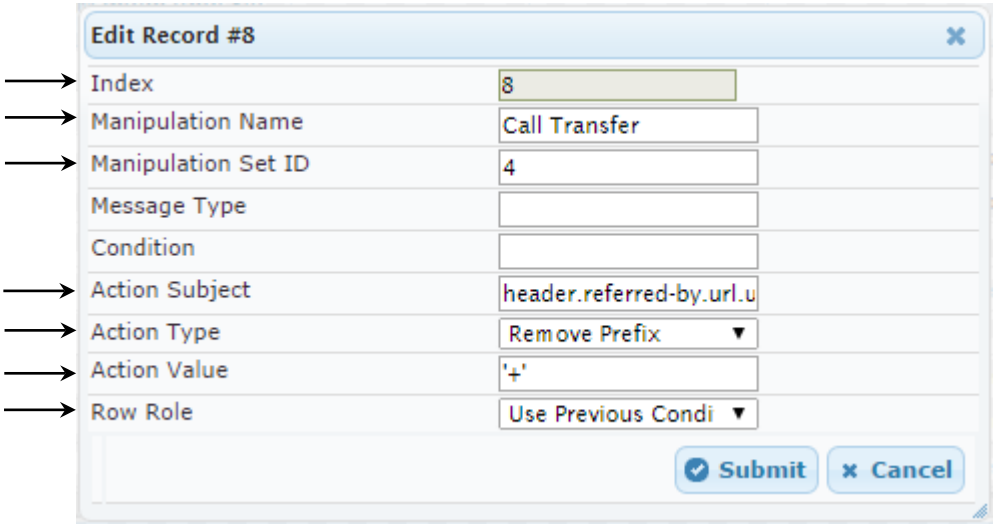
Figure 4-52: Configuring SIP Message Manipulation Rule 7 (for Windstream SIP Trunk)



10. If the manipulation rule Index 7 (above) is executed, then the following rule is also executed. It remove prefix '+' from the Referred-By Header.

Parameter	Value
Index	**8**
Manipulation Name	**Call Transfer**
Manipulation Set ID	**4**
Message Type	
Condition	
Action Subject	**header.referred-by.url.user**
Action Type	**Remove Prefix**
Action Value	**'+'**
Row Role	**Use Previous Condition**

Figure 4-53: Configuring SIP Message Manipulation Rule 8 (for Windstream SIP Trunk)



11. Configure another manipulation rule (Manipulation Set 4) for Windstream SIP Trunk. This rule applies to messages sent to the Windstream SIP Trunk (IP Group 2). This rule replaces the **user** part of the **From** Header with the value from Referred-By Header.

Parameter	Value
Index	**9**
Manipulation Name	**Call Transfer**
Manipulation Set ID	**4**
Message Type	
Condition	**header.referred-by exists**
Action Subject	**header.from.url.user**
Action Type	**Modify**
Action Value	**header.referred-by.url.user**

Figure 4-54: Configuring SIP Message Manipulation Rule 9 (for Windstream SIP Trunk)

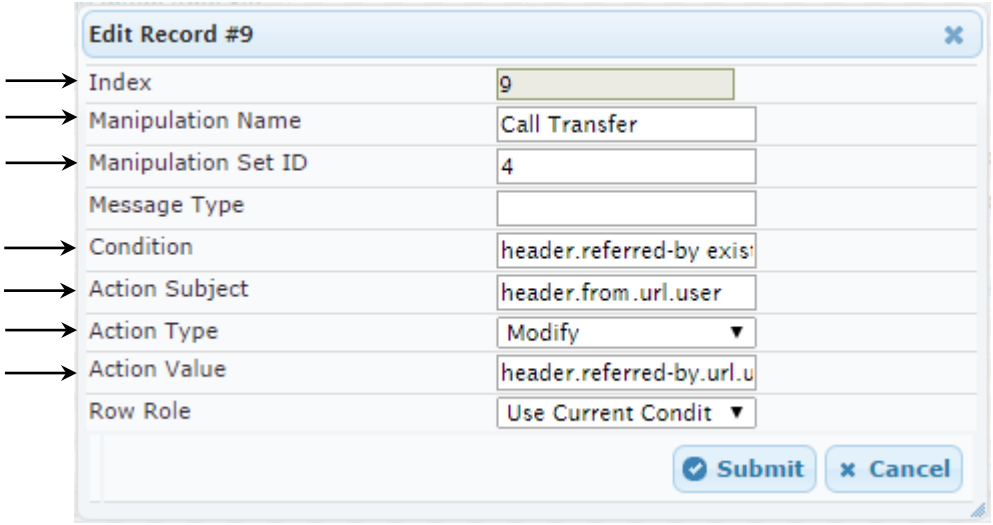
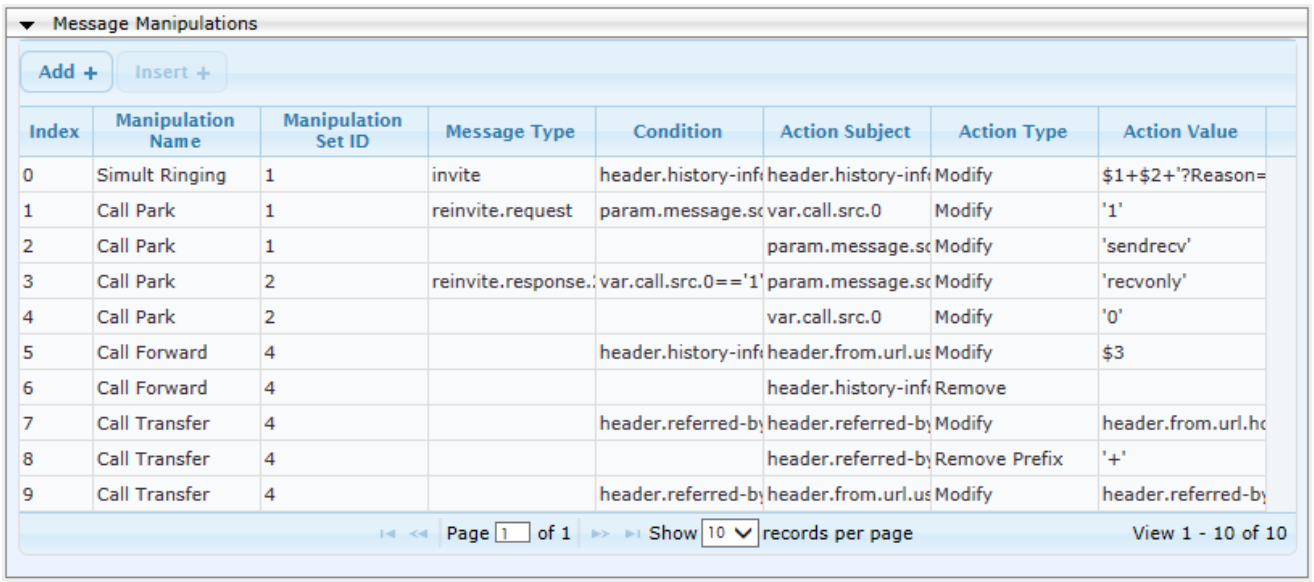


Figure 4-55: Configured SIP Message Manipulation Rules



The table below includes SIP message manipulation rules which are bound together by commonality via the Manipulation Set IDs (Manipulation Set IDs 1, 2, and 4) which are executed for messages sent to and from the Windstream SIP Trunk (IP Group 2) as well as the Lync Server 2013 (IP Group 1). These rules are specifically required to enable proper interworking between Windstream SIP Trunk and Lync Server 2013. The specific items are needed to support Music on Hold (rules 1-4). Refer to the *User's Manual* for further details concerning the full capabilities of header manipulation.

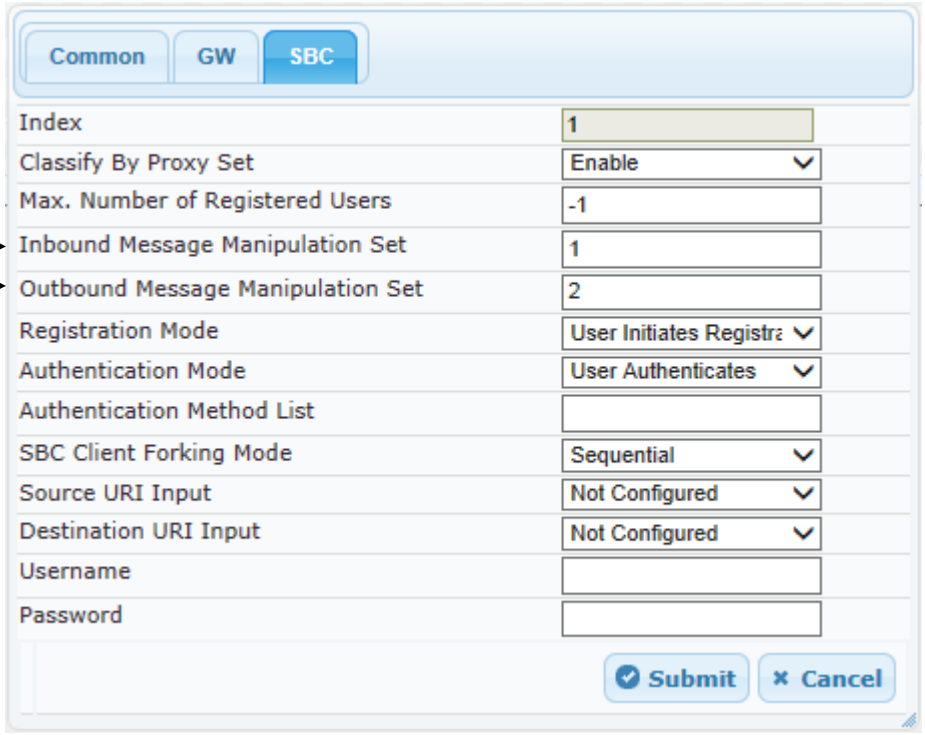
Table 4-1: SIP Message Manipulation Rules

Rule Index	Rule Description	Reason for Introducing Rule
0	This rule applies to messages received from the Lync Server 2013 (IP Group 1), for simultaneous ringing initiated by the Lync Server 2013 (IP Group 1). This rule adds an Action Value containing the Reason for the History-Info header, causing the E-SBC to add a Diversion Header towards the SIP Trunk.	
1	For every SIP Re-INVITE request with SDP, where RTP mode = "sendonly" (occurs in a Lync 2013-initiated Hold), create a variable and set it to **'1'**. This variable manages how the call will be handled in each state (answer, request, etc.).	In the Call Park scenario, Microsoft Lync sends Re-INVITE messages twice. The first message is sent with the SDP, where the RTP mode is set to "a=inactive". The second message is sent with "a=sendonly". The Windstream SIP Trunk has a problem recognizing two sequential Re-INVITE messages with different RTP modes. This causes the loss of the Music On Hold functionality in the Call Park scenario. These four rules are applied to work around this limitation.
2	If the previous manipulation rule (Index 0) is executed, then the following rule is also executed on the same SIP message: if RTP mode within the SDP is set to "sendonly", change it to "sendrecv".	
3	This rule attempts to normalize the call processing state back to Lync 2013 for the correct reply to the initially received "sendonly". For every SIP Re-INVITE message with the variable set to **'1'**, change RTP mode to "recvonly". This SIP Re-INVITE message is the response sent from the Windstream SIP Trunk to the Lync-initiated Hold.	
4	If the manipulation rule Index 2 (above) is executed, then the following rule is also executed. If the variable is determined to be set to **"1"** (in the previous manipulation rule), then set it to **"0"** to normalize the call processing state. Lync now sends Music on Hold to the Windstream SIP Trunk even without the Windstream SIP Trunk knowing how to receive MoH. The call is now truly on hold with MoH.	
5	This rule applies to messages sent to the Windstream SIP Trunk (IP Group 2) in call forward scenario. This replaces the user part of the From Header with the value from History Info Header.	For Call Forward scenario, Windstream SIP Trunk needs to replace the User part of the SIP From Header with the value from History Info Header.
6	This rule removes the History Info Header.	
7	This rule applies to messages sent to the Windstream SIP Trunk (IP Group 2). This rule replaces the host part of the Referred-By Header with the value from the SIP From Header.	For Call Transfer initiated by Lync Server 2013, Windstream SIP Trunk needs to replace the Host part of the SIP Referred-By Header with the value from

8	This rule removes prefix '+' from the Referred-By Header.	the SIP From Header and user part of the From Header with the value from Referred-By Header.
9	This rule applies to messages sent to the Windstream SIP Trunk (IP Group 2). This rule replaces the user part of the From Header with the value from Referred-By Header.	

12. Assign Manipulation Set IDs 1 and 2 to IP Group 1:
    a. Open the IP Group Table page (**Configuration** tab > **VoIP** menu > **VoIP Network** > **IP Group Table**).
    b. Select the row of IP Group 1, and then click **Edit**.
    c. Click the **SBC** tab.
    d. Set the 'Inbound Message Manipulation Set' field to **1**.
    e. Set the 'Outbound Message Manipulation Set' field to **2**.

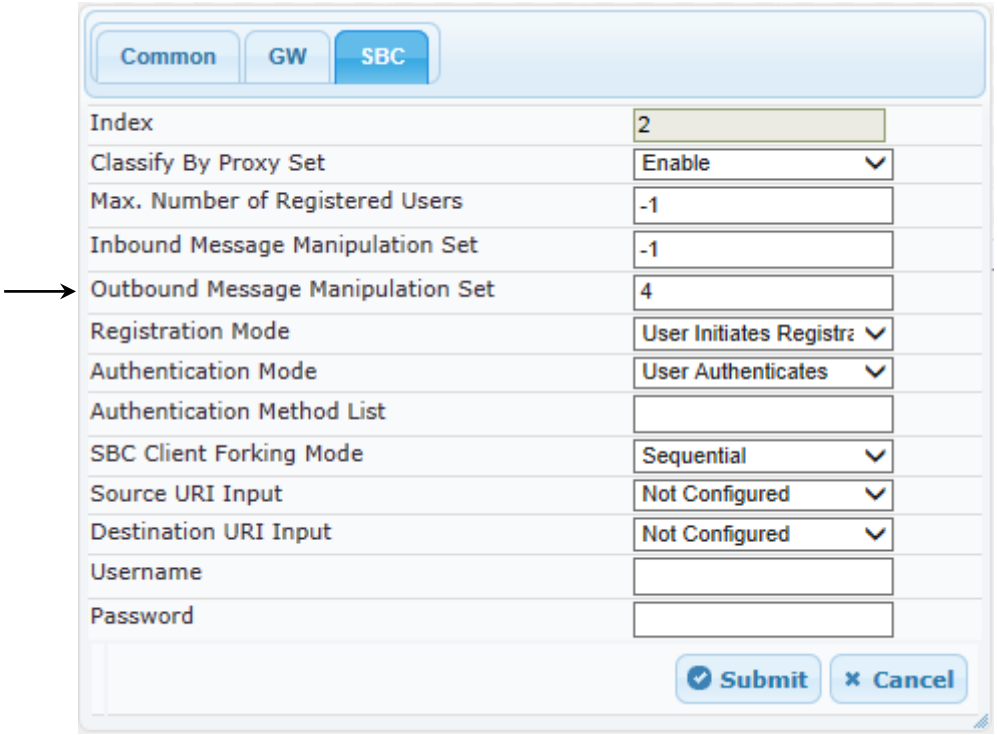
Figure 4-56: Assigning Manipulation Set to IP Group 1



    f. Click **Submit**.

13. Assign Manipulation Set ID 4 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 **4**.

Figure 4-57: Assigning Manipulation Set 4 to IP Group 2



    e. Click **Submit**.

4.14 Step 14: Configure Registration Accounts

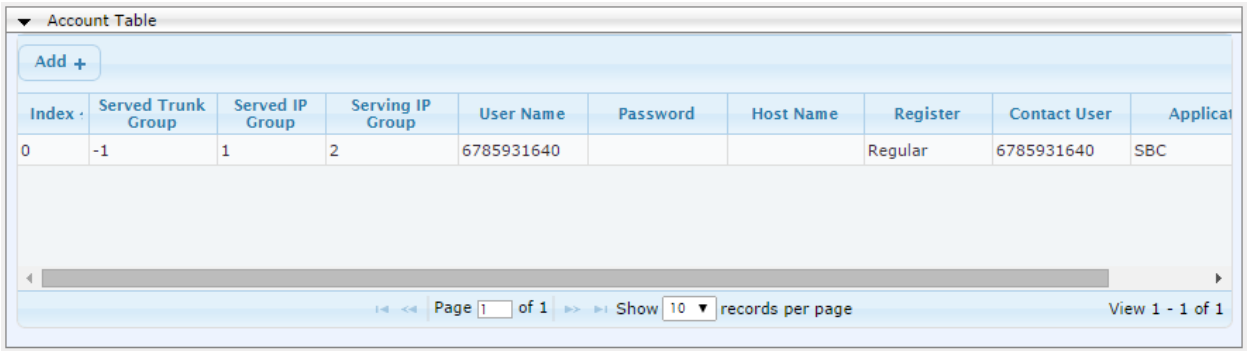
This step describes how to configure SIP registration accounts. This is required so that the E-SBC can register with the Windstream SIP Trunk on behalf of Lync Server 2013. The Windstream SIP Trunk requires registration and authentication to provide service.

In the interoperability test topology, the Served IP Group is Lync Server 2013 (IP Group 1) and the Serving IP Group is Windstream SIP Trunk (IP Group 2).

➢ **To configure a registration account:**

1. Open the Account Table page (**Configuration** tab > **VoIP** menu > **SIP Definitions** > **Account Table**).

Figure 4-58: Configuring SIP Registration Account



2. Enter an index number (e.g., "0"), and then click **Add**.
3. Configure the account according to the provided information from Windstream, for example:

Parameter	Value
Served IP Group	**1** (Lync Server 2013)
Serving IP Group	**2** (Windstream SIP Trunk)
Username	As provided by Windstream
Password	As provided by Windstream
Host Name	As provided by Windstream
Register	**Regular**
Contact User	As provided by Windstream
Application Type	**SBC**

4. Click **Apply**.

4.15 Step 15: Miscellaneous Configuration

This section describes miscellaneous E-SBC configuration.

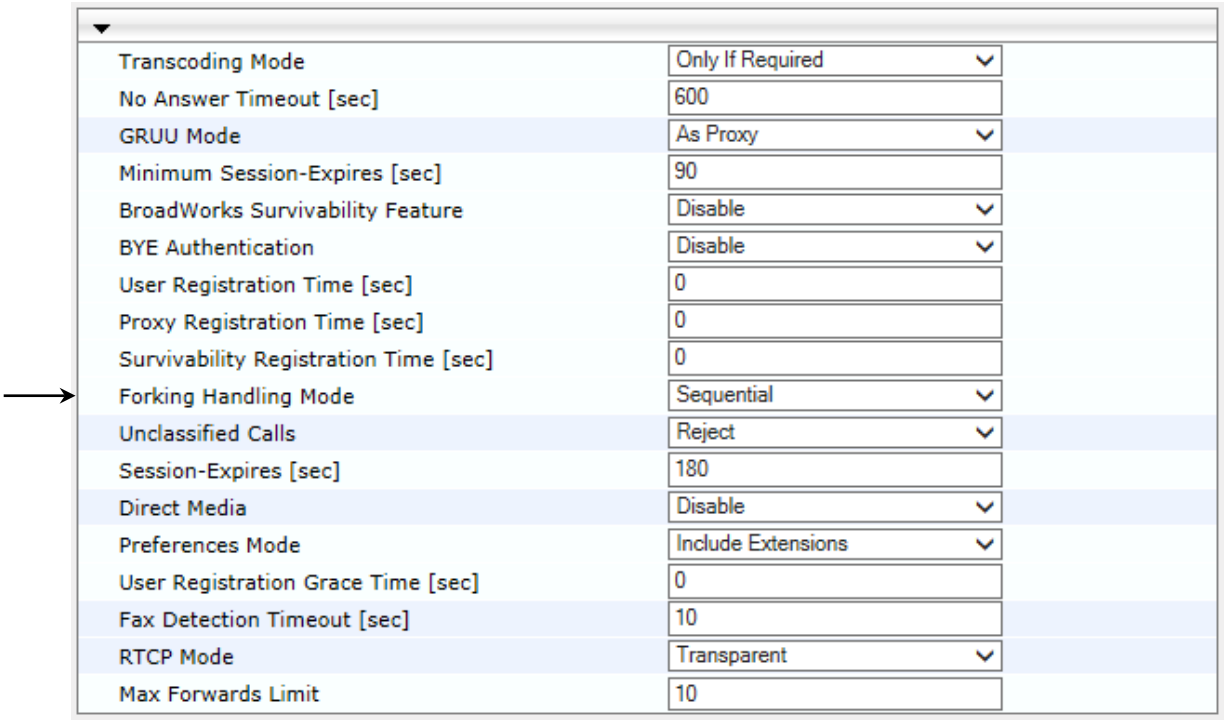
4.15.1 Step 15a: Configure Call Forking Mode

This step describes 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 a SIP 18x message with SDP is received, the E-SBC opens a voice stream according to the received SDP. The E-SBC re-opens the stream according to subsequently received 18x responses with SDP or plays a ringback tone if 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 4-59: Configuring Forking Mode



3. Click **Submit**.

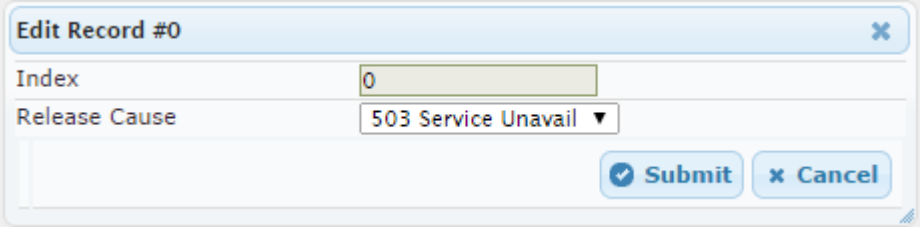
4.15.2 Step 15b: Configure SBC Alternative Routing Reasons

This step describes how to configure the E-SBC's handling of SIP 503 responses received for outgoing SIP dialog-initiating methods, e.g., INVITE, OPTIONS, and SUBSCRIBE messages. In this case E-SBC attempts to locate an alternative route for the call.

➢ **To configure SIP reason codes for alternative IP routing:**

1. Open the SBC Alternative Routing Reasons page (**Configuration** tab > **VoIP** menu > **SBC** > **Routing SBC** > **Alternative Routing Reasons**).
2. Click **Add**; the following dialog box appears:

Figure 4-60: Alternative Routing Reasons Table - Add Record



3. Click **Submit**.

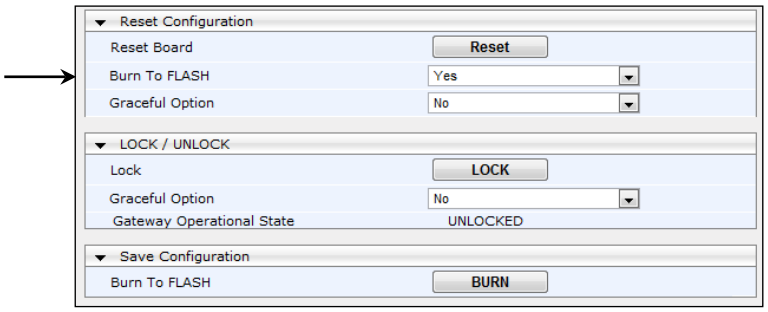
4.16 Step 16: Reset the E-SBC

After you have completed the configuration of the E-SBC described in this chapter, 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 4-61: 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 800 E-SBC
;HW Board Type: 69  FK Board Type: 74
;Serial Number: 5299378
;Slot Number: 1
;Software Version: 6.80A.234.004
;DSP Software Version: 5014AE3_R => 680.23
;Board IP Address: 10.15.17.55
;Board Subnet Mask: 255.255.0.0
;Board Default Gateway: 10.15.0.1
;Ram size: 369M   Flash size: 64M   Core speed: 500Mhz
;Num of DSP Cores: 3  Num DSP Channels: 30
;Num of physical LAN ports: 4
;Profile: NONE
;;Key features:;Board Type: 74 ;IP Media: Conf VXML
VoicePromptAnnounc(H248.9) CALEA TrunkTesting POC ;Channel Type: DspCh=30
IPMediaDspCh=30 ;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 ;DSP Voice features: IpmDetector RTCP-XR
AMRPolicyManagement ;E1Trunks=1 ;FXSPorts=8 ;FXOPorts=0 ;Security: IPSEC
MediaEncryption StrongEncryption EncryptControlProtocol ;DATA features:
;QOE features: VoiceQualityMonitoring MediaEnhancement ;Control
Protocols: MSFT CLI TRANSCODING=30 FEU=100 TestCall=100 MGCP MEGACO H323
SIP TPNCP SASurvivability SBC=50 ;Default features:;Coders: G711 G726;

;------  HW components------
;
; Slot # : Module type : # of ports
;---------------------------------------------
;      1 : FALC56      : 1
;      2 : FXS         : 4
;      3 : FXS         : 4
;---------------------------------------------


[SYSTEM Params]

SyslogServerIP = 10.15.17.100
EnableSyslog = 1
;NTPServerIP_abs is hidden but has non-default value
NTPServerUTCOffset = 7200
;VpFileLastUpdateTime is hidden but has non-default value
NTPServerIP = '10.15.25.1'
```

```
[BSP Params]

PCMLawSelect = 3
UdpPortSpacing = 10
EnterCpuOverloadPercent = 99
ExitCpuOverloadPercent = 95

[Analog Params]

FarEndDisconnectType = 7

[ControlProtocols Params]

AdminStateLockControl = 0

[MGCP Params]


[MEGACO Params]


[PSTN Params]


[SS7 Params]


[Voice Engine Params]

ENABLEMEDIASECURITY = 1
CallProgressTonesFilename = 'usa_tones_13.dat'

[WEB Params]

UseRProductName = 'Mediant 800 E-SBC'
WebLogoText = 'Windstream'
UseWeblogo = 1
;UseLogoInWeb is hidden but has non-default value
UseProductName = 1
HTTPSCipherString = 'RC4:EXP'
;HTTPSCertFileName is hidden but has non-default value
;HTTPSRootFileName is hidden but has non-default value

[SIP Params]

MEDIACHANNELS = 30
REGISTRATIONTIME = 1800
GWDEBUGLEVEL = 5
;ISPRACKREQUIRED is hidden but has non-default value
REGISTRATIONRETRYTIME = 600
ENABLESBCAPPLICATION = 1
MSLDAPPRIMARYKEY = 'telephoneNumber'
SBCPREFERENCESMODE = 1
SBCFORKINGHANDLINGMODE = 1
ENERGYDETECTORCMD = 587202560
```

```
ANSWERDETECTORCMD = 10486144

[SCTP 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_4_1", 1, 1, 4, "User Port #0", "GROUP_1",
"Active";
PhysicalPortsTable 1 = "GE_4_2", 1, 1, 4, "User Port #1", "GROUP_1",
"Redundant";
PhysicalPortsTable 2 = "GE_4_3", 1, 2, 4, "User Port #2", "GROUP_2",
"Active";
PhysicalPortsTable 3 = "GE_4_4", 1, 2, 4, "User Port #3", "GROUP_2",
"Redundant";

[ \PhysicalPortsTable ]


[ EtherGroupTable ]

FORMAT EtherGroupTable_Index = EtherGroupTable_Group,
EtherGroupTable_Mode, EtherGroupTable_Member1, EtherGroupTable_Member2;
EtherGroupTable 0 = "GROUP_1", 2, "GE_4_1", "GE_4_2";
EtherGroupTable 1 = "GROUP_2", 2, "GE_4_3", "GE_4_4";
EtherGroupTable 2 = "GROUP_3", 0, "", "";
EtherGroupTable 3 = "GROUP_4", 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 ]


[ 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, 10.15.17.55, 16, 10.15.0.1, 1, "Voice",
10.15.25.1, , "vlan 1";
InterfaceTable 1 = 5, 10, 195.189.192.156, 25, 195.189.192.129, 2,
"WANSP", 80.179.52.100, 80.179.55.100, "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 0 = "MRLan", "Voice", "", 6000, 10, 6090, 1, "", "";
CpMediaRealm 1 = "MRWan", "WANSP", "", 7000, 10, 7090, 0, "", "";

[ \CpMediaRealm ]


[ SRD ]

FORMAT SRD_Index = SRD_Name, SRD_MediaRealm, SRD_IntraSRDMediaAnchoring,
SRD_BlockUnRegUsers, SRD_MaxNumOfRegUsers,
SRD_EnableUnAuthenticatedRegistrations;
SRD 0 = "SRDLan", "MRLan", 0, 0, -1, 1;
SRD 1 = "SRDWan", "MRWan", 0, 0, -1, 1;

[ \SRD ]


[ SBCAlternativeRoutingReasons ]

FORMAT SBCAlternativeRoutingReasons_Index =
SBCAlternativeRoutingReasons_ReleaseCause;
SBCAlternativeRoutingReasons 0 = 503;

[ \SBCAlternativeRoutingReasons ]


[ ProxyIp ]
```

```
FORMAT ProxyIp_Index = ProxyIp_IpAddress, ProxyIp_TransportType,
ProxyIp_ProxySetId;
ProxyIp 0 = "64.199.64.220:5060", 0, 2;
ProxyIp 1 = "FE15.ilync15.local:5067", 2, 1;

[ \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_SBCKeepVIAHeaders,
IpProfile_SBCKeepUserAgentHeader,
IpProfile_SBCUserBehindUdpNATRegistrationTime,
IpProfile_SBCUserBehindTcpNATRegistrationTime;
IpProfile 1 = "Lync", 1, 0, 0, 10, 10, 46, 40, 0, 0, 0, 0, 2, 0, 0, 0, 0,
-1, 1, 0, 0, -1, 1, 4, -1, 1, 1, 0, 0, "", -1, 0, 0, "", -1, -1, 0, 1, 0,
0, 0, 0, 8, 300, 400, 0, 0, 0, -1, 0, 0, 1, 1, 0, 1, 1, 0, 3, 2, 1, 0, 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, 300, -1, -1, -1, -1;
IpProfile 2 = "Windstream", 1, 0, 0, 10, 10, 46, 40, 0, 0, 0, 0, 2, 0, 0,
0, 0, -1, 1, 0, 0, -1, 1, 4, -1, 1, 1, 0, 0, "", 2, 0, 0, "", 2, -1, 2,
```

```
2, 0, 0, 1, 0, 8, 300, 400, 0, 0, 0, -1, 0, 0, 1, 3, 0, 2, 2, 1, 3, 0, 0,
0, 1, 0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 1, 0, 0, 0, 0, 0, 0, 0, 0, 1,
0, 0, 0, 300, -1, -1, -1, -1;

[ \IpProfile ]


[ ProxySet ]

FORMAT ProxySet_Index = ProxySet_ProxyName,
ProxySet_EnableProxyKeepAlive, ProxySet_ProxyKeepAliveTime,
ProxySet_ProxyLoadBalancingMethod, ProxySet_IsProxyHotSwap, ProxySet_SRD,
ProxySet_ClassificationInput, ProxySet_TLSContext,
ProxySet_ProxyRedundancyMode, ProxySet_DNSResolveMethod,
ProxySet_KeepAliveFailureResp;
ProxySet 0 = "", 0, 60, 0, 0, 0, 0, "-1", -1, -1, "";
ProxySet 1 = "Lync", 1, 60, 1, 1, 0, 0, "-1", 1, -1, "";
ProxySet 2 = "Windstream", 1, 60, 0, 0, 1, 0, "-1", -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_SIPConnect;
IPGroup 1 = 0, "Lync", 1, "64.199.64.220", "", 0, -1, -1, 0, -1, 0,
"MRLan", 1, 1, -1, 1, 2, 0, 0, "", 0, -1, -1, "", "", "$1$gQ==", 0, "",
"", "", 0, "", "", 0;
IPGroup 2 = 0, "Windstream", 2, "64.199.64.220", "", 0, -1, -1, 0, -1, 1,
"MRWan", 1, 2, -1, -1, 4, 0, 0, "", 0, -1, -1, "", "", "$1$gQ==", 0, "",
"", "", 0, "", "", 0;

[ \IPGroup ]


[ Account ]

FORMAT Account_Index = Account_ServedTrunkGroup, Account_ServedIPGroup,
Account_ServingIPGroup, Account_Username, Account_Password,
Account_HostName, Account_Register, Account_ContactUser,
Account_ApplicationType;
Account 0 = -1, 1, 2, "6785931640", "$1$gQ==", "", 1, "6785931640", 2;

[ \Account ]


[ 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 = "OPTIONS termination", 1, "*", "*", "*", "*", 6, "", -1,
0, -1, 1, -1, "", "internal", 0, -1, 0, 0, "";
IP2IPRouting 1 = "Lync to ITSP", 1, "*", "*", "*", "*", 0, "", -1, 0, -1,
0, 2, "1", "", 0, -1, 0, 0, "";
IP2IPRouting 2 = "ITSP to Lync", 2, "*", "*", "*", "*", 0, "", -1, 0, -1,
0, 1, "0", "", 0, -1, 0, 0, "";

[ \IP2IPRouting ]


[ TLSContexts ]

FORMAT TLSContexts_Index = TLSContexts_Name, TLSContexts_TLSVersion,
TLSContexts_ServerCipherString, TLSContexts_ClientCipherString,
TLSContexts_OcspEnable, TLSContexts_OcspServerPrimary,
TLSContexts_OcspServerSecondary, TLSContexts_OcspServerPort,
TLSContexts_OcspDefaultResponse;
TLSContexts 0 = "default", 0, "RC4:EXP", "ALL:!ADH", 0, 0.0.0.0, 0.0.0.0,
2560, 0;

[ \TLSContexts ]


[ SIPInterface ]

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

[ \SIPInterface ]


[ 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, "+", "", 0;
IPOutboundManipulation 1 = "", 0, 1, 2, "*", "*", "+", "*", "*", "", 0, -
1, 0, 1, 1, 0, 255, "", "", 0;
IPOutboundManipulation 2 = "", 0, 1, 2, "+", "*", "*", "*", "*", "", 0, -
1, 0, 0, 1, 0, 255, "", "", 0;

[ \IPOutboundManipulation ]


[ CodersGroup0 ]

FORMAT CodersGroup0_Index = CodersGroup0_Name, CodersGroup0_pTime,
CodersGroup0_rate, CodersGroup0_PayloadType, CodersGroup0_Sce,
CodersGroup0_CoderSpecific;
CodersGroup0 0 = "g711Alaw64k", 20, 255, -1, 0, "";

[ \CodersGroup0 ]


[ CodersGroup2 ]

FORMAT CodersGroup2_Index = CodersGroup2_Name, CodersGroup2_pTime,
CodersGroup2_rate, CodersGroup2_PayloadType, CodersGroup2_Sce,
CodersGroup2_CoderSpecific;
CodersGroup2 0 = "g729", 20, 0, -1, 0, "";
CodersGroup2 1 = "g711Ulaw64k", 20, 0, -1, 0, "";

[ \CodersGroup2 ]


[ AllowedCodersGroup2 ]

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

[ \AllowedCodersGroup2 ]


[ MessageManipulations ]

FORMAT MessageManipulations_Index =
MessageManipulations_ManipulationName, MessageManipulations_ManSetID,
MessageManipulations_MessageType, MessageManipulations_Condition,
MessageManipulations_ActionSubject, MessageManipulations_ActionType,
MessageManipulations_ActionValue, MessageManipulations_RowRole;
MessageManipulations 0 = "Simult Ringing", 1, "invite", "header.history-
info.0 regex (<.*)(user=phone)(>)(.*)", "header.history-info.0", 2,
"$1+$2+'?Reason=SIP%3Bcause%3D404'+$3+$4", 0;
MessageManipulations 1 = "Call Park", 1, "reinvite.request",
"param.message.sdp.rtpmode=='sendonly'", "var.call.src.0", 2, "'1'", 0;
```

```
MessageManipulations 2 = "Call Park", 1, "", "",
"param.message.sdp.rtpmode", 2, "'sendrecv'", 1;
MessageManipulations 3 = "Call Park", 2, "reinvite.response.200",
"var.call.src.0=='1'", "param.message.sdp.rtpmode", 2, "'recvonly'", 0;
MessageManipulations 4 = "Call Park", 2, "", "", "var.call.src.0", 2,
"'0'", 1;
MessageManipulations 5 = "Call Forward", 4, "", "header.history-info.0
regex (<sip:)(.)(.*)(@)(.*)", "header.from.url.user", 2, "$3", 0;
MessageManipulations 6 = "Call Forward", 4, "", "", "header.history-
info", 1, "", 1;
MessageManipulations 7 = "Call Transfer", 4, "", "header.referred-by
exists", "header.referred-by.url.host", 2, "header.from.url.host", 0;
MessageManipulations 8 = "Call Transfer", 4, "", "", "header.referred-
by.url.user", 6, "'+'", 1;
MessageManipulations 9 = "Call Transfer", 4, "", "header.referred-by
exists", "header.from.url.user", 2, "header.referred-by.url.user", 0;

[ \MessageManipulations ]


[ RoutingRuleGroups ]

FORMAT RoutingRuleGroups_Index = RoutingRuleGroups_LCREnable,
RoutingRuleGroups_LCRAverageCallLength, RoutingRuleGroups_LCRDefaultCost;
RoutingRuleGroups 0 = 0, 0, 1;

[ \RoutingRuleGroups ]


[ ResourcePriorityNetworkDomains ]

FORMAT ResourcePriorityNetworkDomains_Index =
ResourcePriorityNetworkDomains_Name,
ResourcePriorityNetworkDomains_Ip2TelInterworking;
ResourcePriorityNetworkDomains 1 = "dsn", 0;
ResourcePriorityNetworkDomains 2 = "dod", 0;
ResourcePriorityNetworkDomains 3 = "drsn", 0;
ResourcePriorityNetworkDomains 5 = "uc", 1;
ResourcePriorityNetworkDomains 7 = "cuc", 0;

[ \ResourcePriorityNetworkDomains ]
```

AudioCodes

Configuration Note

www.audiocodes.com