Enterprise Session Border Controllers (E-SBC)

AudioCodes Mediant™ Series

Interoperability Lab

Configuration Note

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

Version 6.8

June 2014

Document # LTRT-12320

Table of Contents

Reader's Notes

Notice

This document describes how to connect the Microsoft Lync Server 2013 and Netia 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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Reader's Notes

1 Introduction

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

1.1 Intended Audience

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

1.2 About AudioCodes E-SBC Product Series

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

The E-SBC provides perimeter defense as a way of protecting Enterprises from malicious VoIP attacks; mediation for allowing the connection of any PBX and/or IP-PBX to any service provider; and Service Assurance for service quality and manageability.

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

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

Reader's Notes

2 Component Information

2.1 AudioCodes E-SBC Version

Table 2-1: AudioCodes E-SBC Version

SBC Vendor	AudioCodes
Models	▪ Mediant 500 E-SBC ▪ Mediant 800 Gateway & E-SBC ▪ Mediant 1000B Gateway & E-SBC ▪ Mediant 3000 Gateway & E-SBC ▪ Mediant 2600 E-SBC ▪ Mediant 4000 E-SBC
Software Version	SIP_6.80A.231.003
Protocol	▪ SIP/UDP (to the Netia SIP Trunk) ▪ SIP/TCP or TLS (to the Lync FE Server)
Additional Notes	None

2.2 Netia SIP Trunking Version

Table 2-2: Netia Version

Vendor/Service Provider	Netia
SSW Model/Service	BroadSoft
Software Version	BroadWorks R19 SP1
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

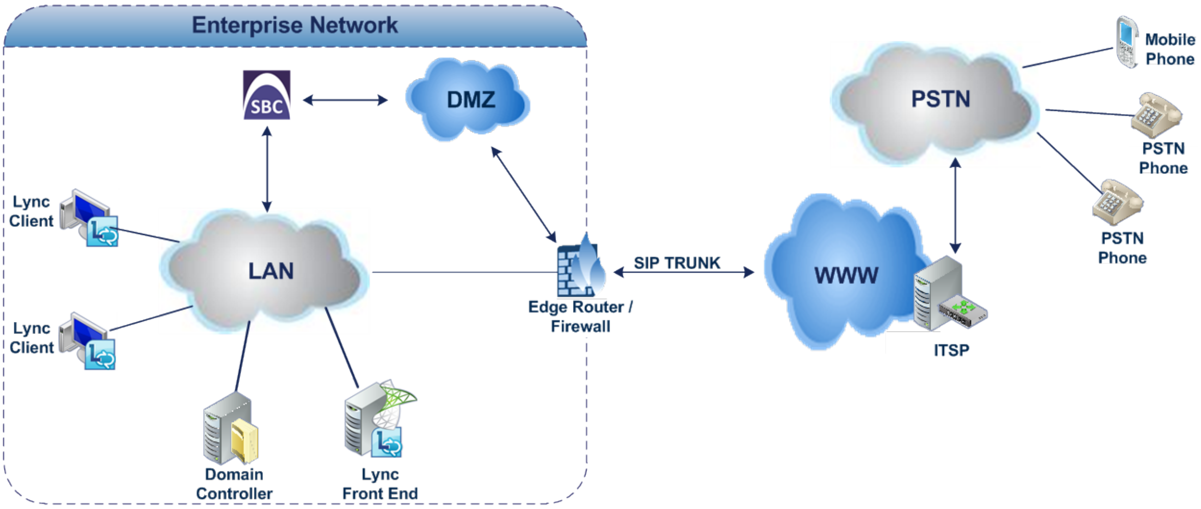
2.4 Interoperability Test Topology

The interoperability testing between AudioCodes E-SBC and Netia 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 Netia'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 Netia's SIP Trunk located in the public network.

The figure below illustrates this interoperability test topology:

Figure 2-1: Interoperability Test Topology - E-SBC and Microsoft Lync with Netia 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 ▪ Netia SIP Trunk is located on the WAN
Signaling Transcoding	▪ Microsoft Lync Server 2013 operates with SIP-over-TLS transport type ▪ Netia SIP Trunk operates with SIP-over-UDP transport type
Codecs Transcoding	▪ Microsoft Lync Server 2013 supports G.711A-law and G.711U-law coders ▪ Netia SIP Trunk supports G.711A-law, G.711U-law, and G.729 coder
Media Transcoding	▪ Microsoft Lync Server 2013 operates with SRTP media type ▪ Netia SIP Trunk operates with RTP media type

2.4.2 Known Limitations

The following limitations were observed in the Interoperability tests done for the AudioCodes E-SBC interworking between Microsoft Lync Server 2013 and Netia's SIP Trunk:

1. If any of following Error Responses are sent from the Lync server:
   - Lync Client set as DnD and send “480 Temporarily Unavailable”
   - Lync Client reject call with “488 Not Acceptable Here”
   - Lync Client response with “503 Service Unavailable”
   - Lync Client reject call with “603 Decline”

   Netia disconnects the call only after a number of additional re-INVITES are sent.
2. 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 Netia 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. Message Manipulation rules are applied to work around this limitation.

Reader's Notes

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.

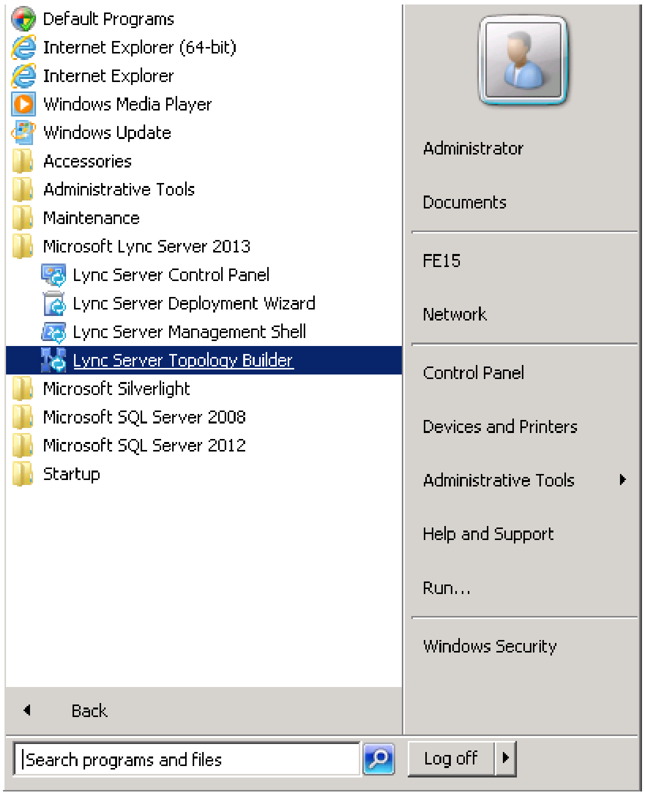
3.1 Configuring the E-SBC as an IP / PSTN Gateway

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:**

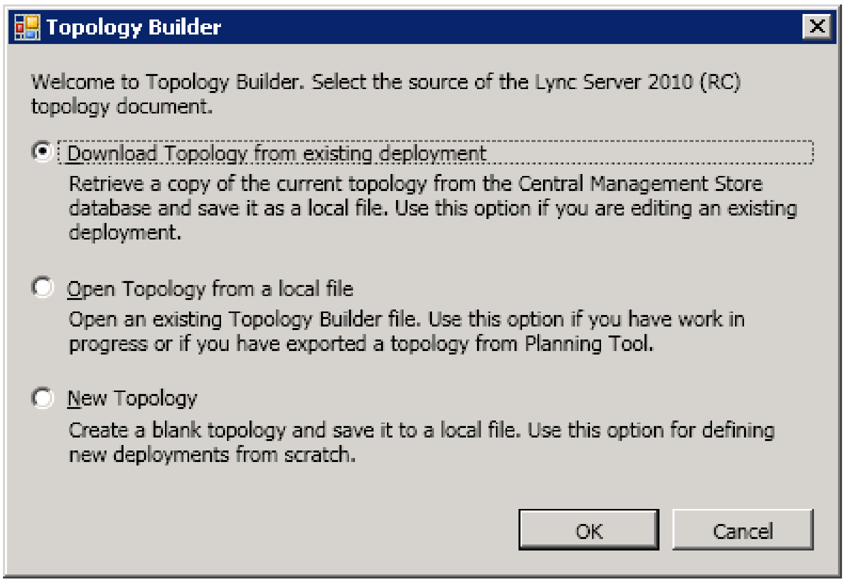
1. On the server where the Topology Builder is installed, start the Lync Server 2013 Topology Builder (Windows **Start** menu > **All Programs** > **Lync Server Topology Builder**), as shown below:

Figure 3-1: Starting the Lync Server Topology Builder



The following is displayed:

Figure 3-2: Topology Builder Dialog Box



Topology Builder

Welcome to Topology Builder. Select the source of the Lync Server 2010 (RC) topology document.

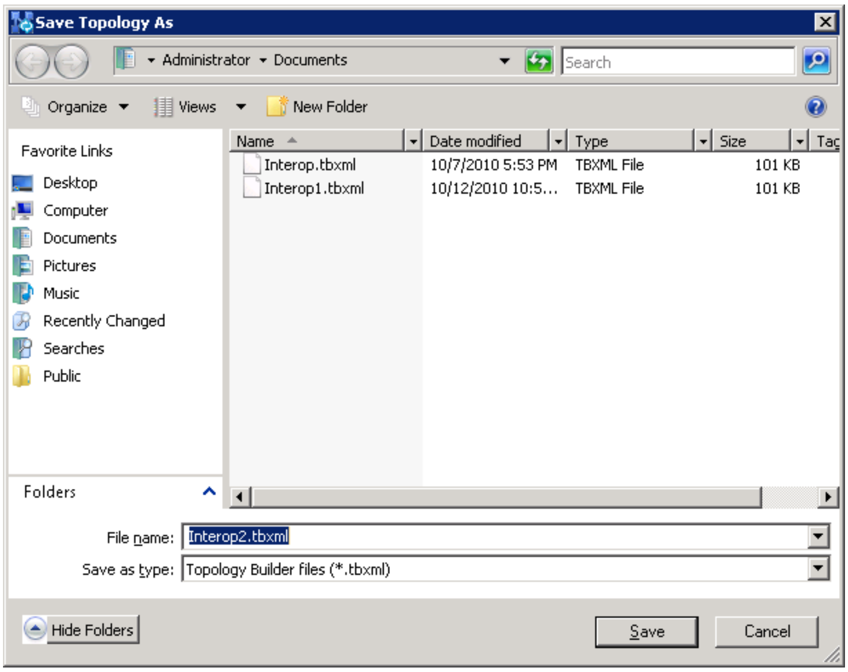
● Download Topology from existing deployment
Retrieve a copy of the current topology from the Central Management Store database and save it as a local file. Use this option if you are editing an existing deployment.

○ Open Topology from a local file
Open an existing Topology Builder file. Use this option if you have work in progress or if you have exported a topology from Planning Tool.

○ New Topology
Create a blank topology and save it to a local file. Use this option for defining new deployments from scratch.

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



Save Topology As

Name	Date modified	Type	Size	Tag
Interop.tbxml	10/7/2010 5:53 PM	TBXML File	101 KB	
Interop1.tbxml	10/12/2010 10:5...	TBXML File	101 KB	

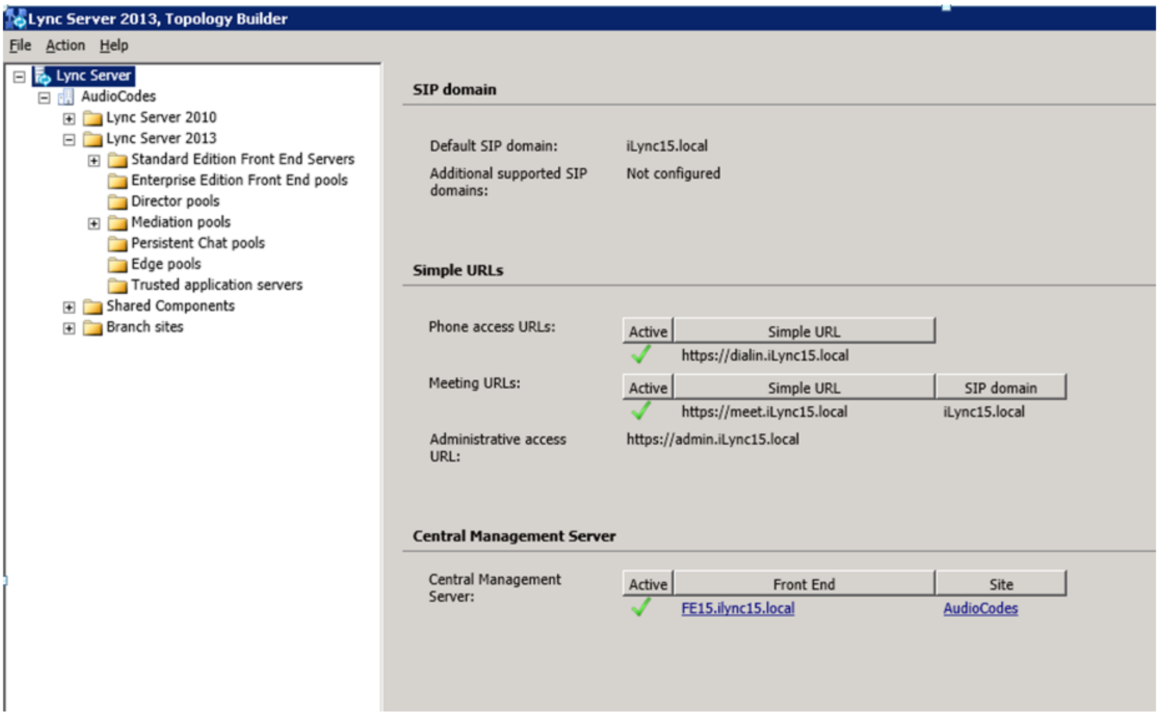
File name: Interop2.tbxml

Save as type: Topology Builder files (*.tbxml)

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



Lync Server 2013, Topology Builder

- Lync Server
  - AudioCodes
    - Lync Server 2010
    - Lync Server 2013
      - Standard Edition Front End Servers
      - Enterprise Edition Front End pools
      - Director pools
      - Mediation pools
      - Persistent Chat pools
      - Edge pools
      - Trusted application servers
    - Shared Components
    - Branch sites

SIP domain

Default SIP domain: iLync15.local

Additional supported SIP domains: Not configured

Simple URLs

Phone access URLs:

Active	Simple URL
✓	https://dialin.iLync15.local

Meeting URLs:

Active	Simple URL	SIP domain
✓	https://meet.iLync15.local	iLync15.local

Administrative access URL: https://admin.iLync15.local

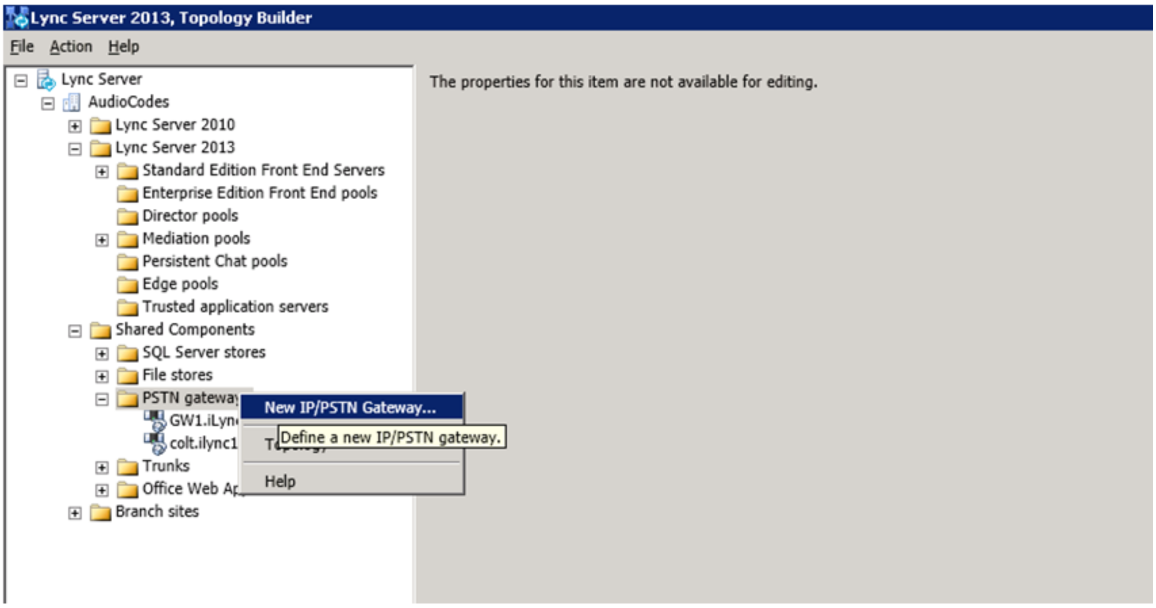
Central Management Server

Central Management Server:

Active	Front End	Site
✓	FE15.ilync15.local	AudioCodes

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



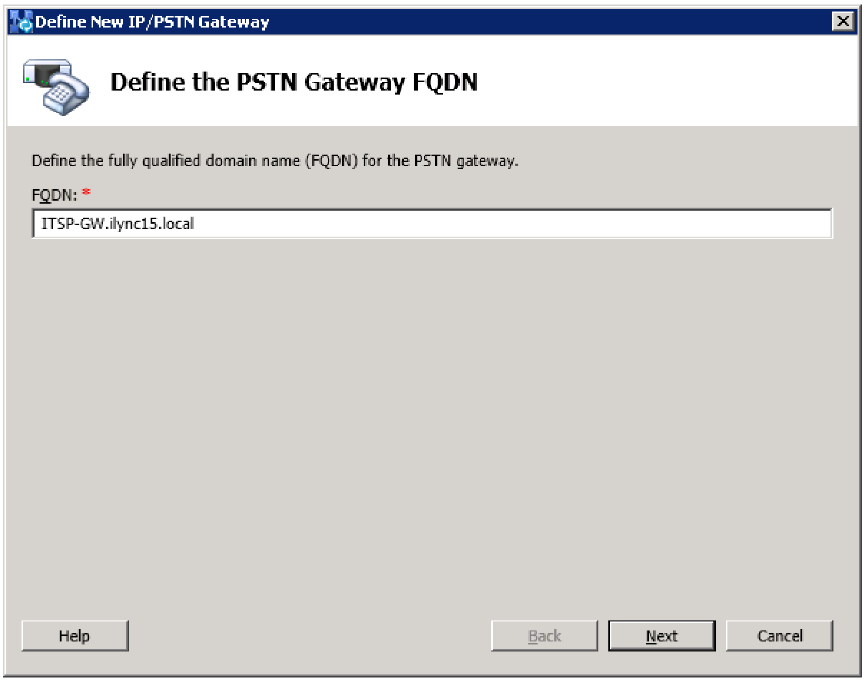
Lync Server 2013, Topology Builder

- Lync Server
  - AudioCodes
    - Lync Server 2010
    - Lync Server 2013
      - Standard Edition Front End Servers
      - Enterprise Edition Front End pools
      - Director pools
      - Mediation pools
      - Persistent Chat pools
      - Edge pools
      - Trusted application servers
    - Shared Components
      - SQL Server stores
      - File stores
      - PSTN gateway
        - GW1.iLyn
        - colt.ilync1
      - Trunks
      - Office Web Ap
    - Branch sites

The properties for this item are not available for editing.

The following is displayed:

Figure 3-6: Define the PSTN Gateway FQDN



Define New IP/PSTN Gateway

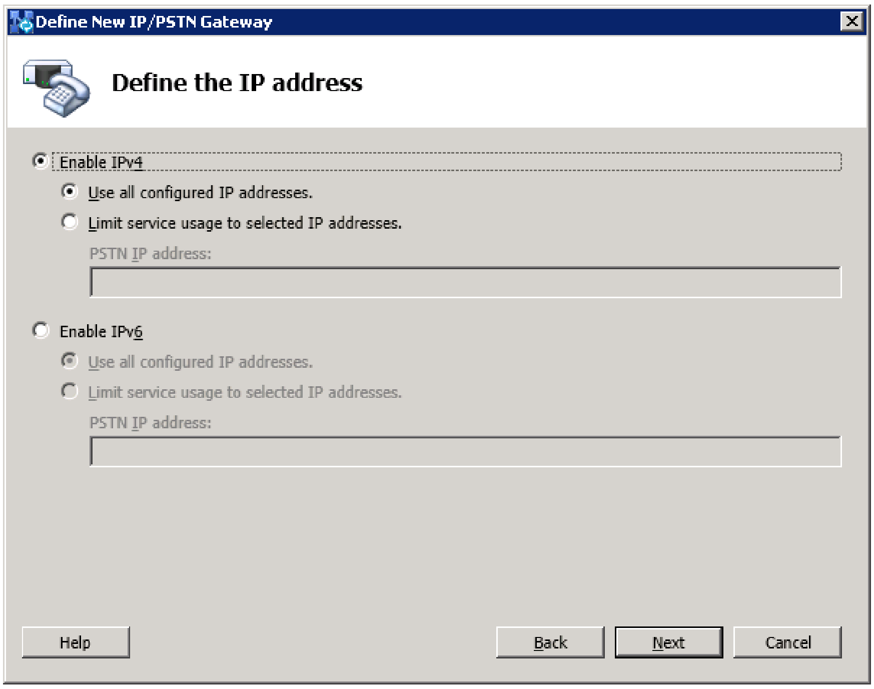
Define the PSTN Gateway FQDN

Define the fully qualified domain name (FQDN) for the PSTN gateway.

FQDN: * ITSP-GW.ilync15.local

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



Define New IP/PSTN Gateway

Define the IP address

● Enable IPv4

● Use all configured IP addresses.

○ Limit service usage to selected IP addresses.

PSTN IP address:

○ Enable IPv6

● Use all configured IP addresses.

○ Limit service usage to selected IP addresses.

PSTN IP address:

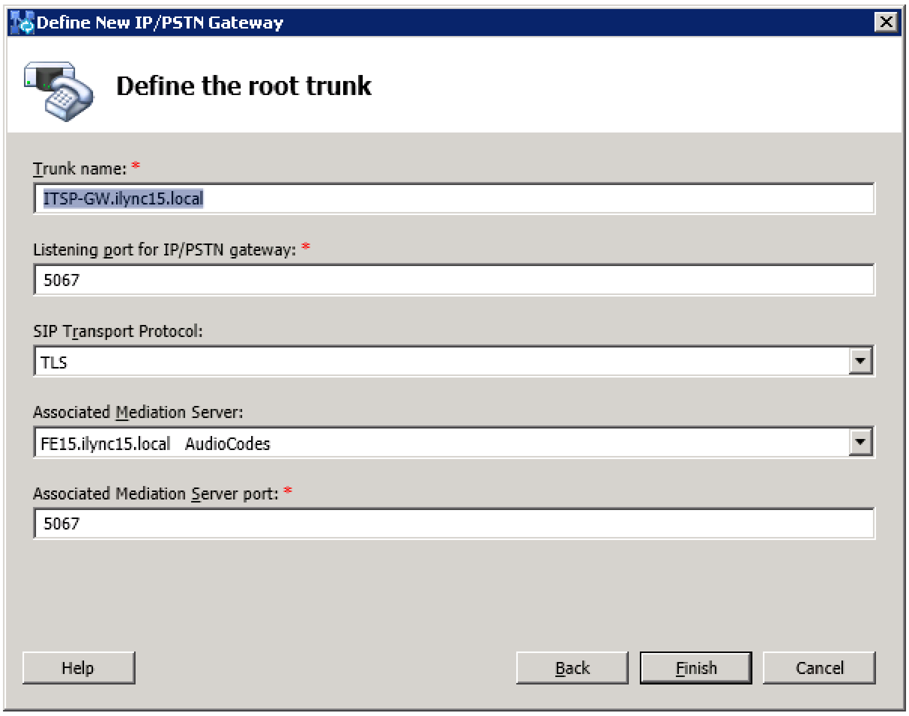
6. Define the listening mode (IPv4 or IPv6) of the IP address of your new PSTN gateway, and then click **Next**.

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



Define New IP/PSTN Gateway

Define the root trunk

Trunk name: * ITSP-GW.ilync15.local

Listening port for IP/PSTN gateway: * 5067

SIP Transport Protocol: TLS

Associated Mediation Server: FE15.ilync15.local AudioCodes

Associated Mediation Server port: * 5067

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.

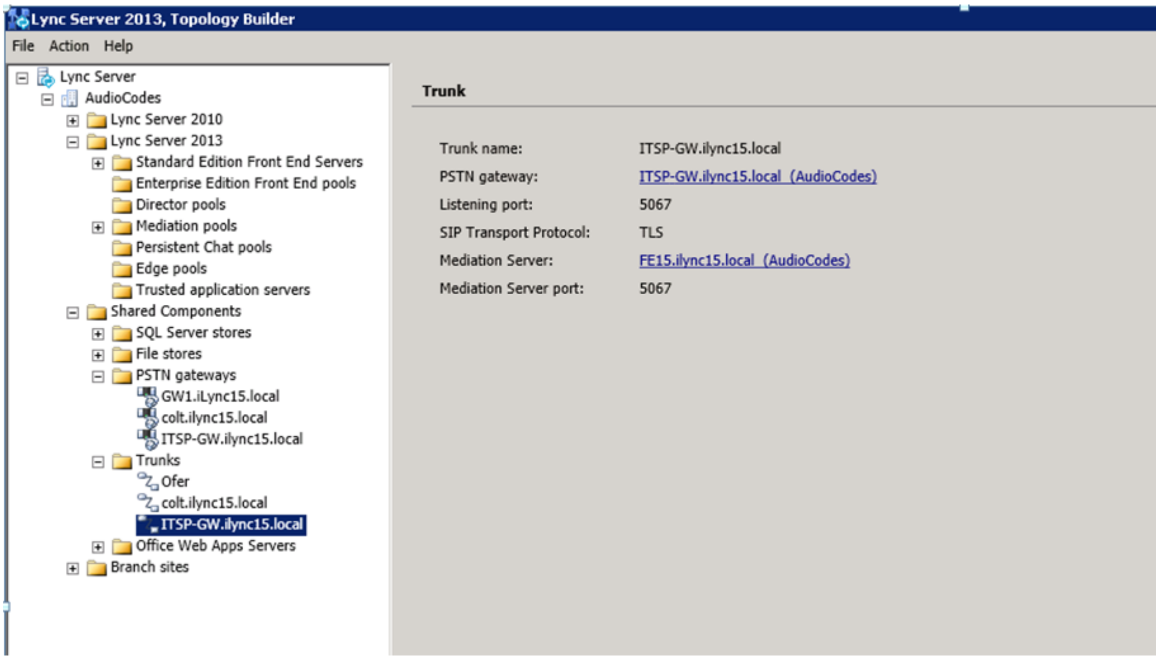
c. In the 'Associated Mediation Server' field, select the Mediation Server pool to associate with the root trunk of this PSTN gateway.

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

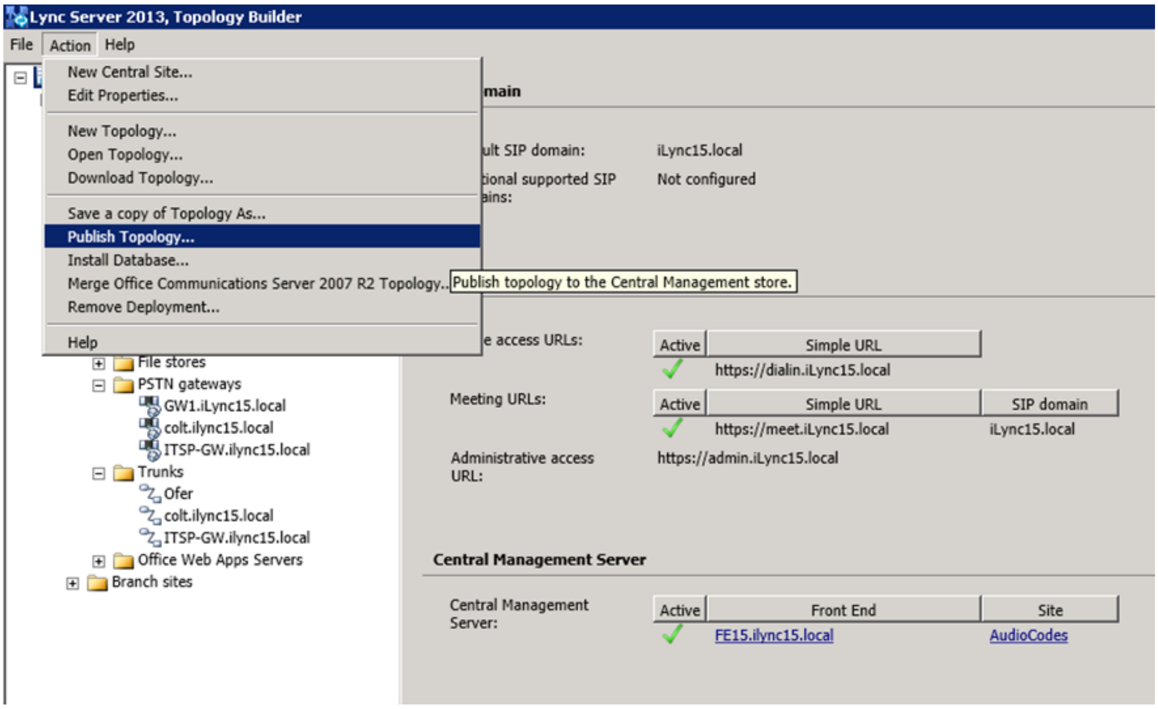


Trunk

Trunk name:	ITSP-GW.ilync15.local
PSTN gateway:	ITSP-GW.ilync15.local (AudioCodes)
Listening port:	5067
SIP Transport Protocol:	TLS
Mediation Server:	FE15.ilync15.local (AudioCodes)
Mediation Server port:	5067

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



main

ult SIP domain:	iLync15.local
tional supported SIP ains:	Not configured

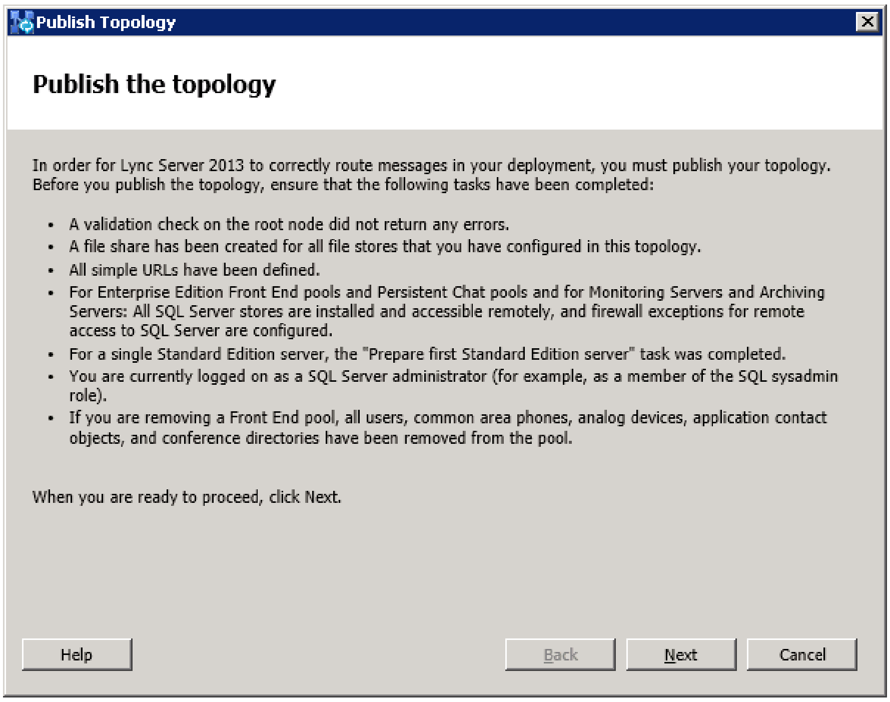
	Active	Simple URL	SIP domain
e access URLs:	✓	https://dialin.iLync15.local	
Meeting URLs:	✓	https://meet.iLync15.local	iLync15.local
Administrative access URL:		https://admin.iLync15.local	

Central Management Server

	Active	Front End	Site
Central Management Server:	✓	FE15.ilync15.local	AudioCodes

The following is displayed:

Figure 3-11: Publish the Topology



Publish Topology

Publish the topology

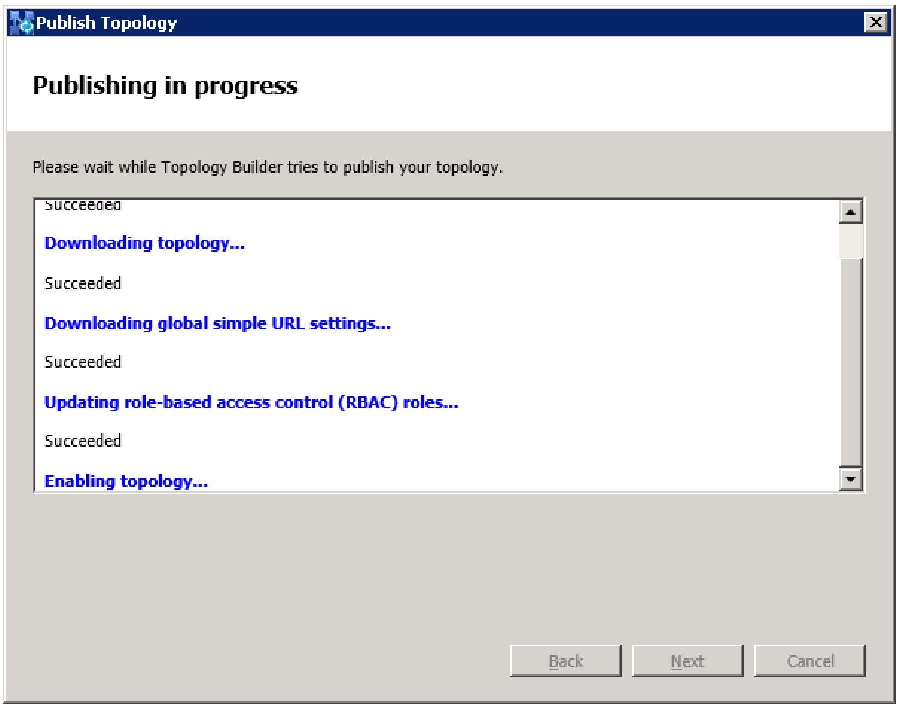
In order for Lync Server 2013 to correctly route messages in your deployment, you must publish your topology. Before you publish the topology, ensure that the following tasks have been completed:

- A validation check on the root node did not return any errors.
- A file share has been created for all file stores that you have configured in this topology.
- All simple URLs have been defined.
- For Enterprise Edition Front End pools and Persistent Chat pools and for Monitoring Servers and Archiving Servers: All SQL Server stores are installed and accessible remotely, and firewall exceptions for remote access to SQL Server are configured.
- For a single Standard Edition server, the "Prepare first Standard Edition server" task was completed.
- You are currently logged on as a SQL Server administrator (for example, as a member of the SQL sysadmin role).
- If you are removing a Front End pool, all users, common area phones, analog devices, application contact objects, and conference directories have been removed from the pool.

When you are ready to proceed, click Next.

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

Figure 3-12: Publishing in Progress



Publish Topology

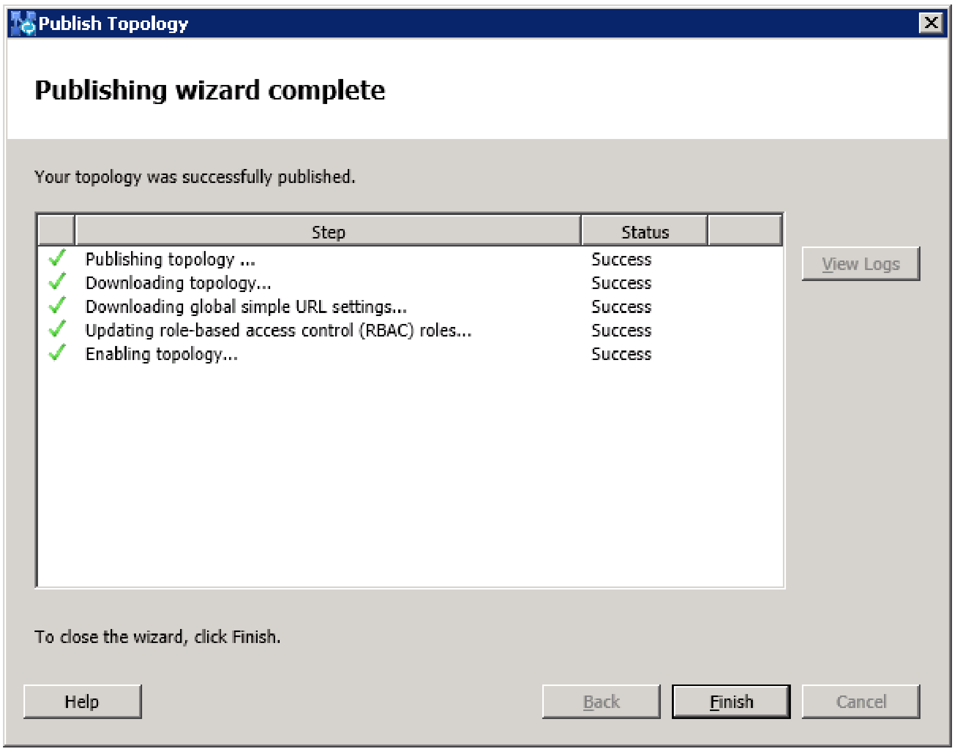
Publishing in progress

Please wait while Topology Builder tries to publish your topology.

```
Succeeded
Downloading topology...
Succeeded
Downloading global simple URL settings...
Succeeded
Updating role-based access control (RBAC) roles...
Succeeded
Enabling topology...
```

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

Figure 3-13: Publishing Wizard Complete



Your topology was successfully published.

	Step	Status
✓	Publishing topology ...	Success
✓	Downloading topology...	Success
✓	Downloading global simple URL settings...	Success
✓	Updating role-based access control (RBAC) roles...	Success
✓	Enabling topology...	Success

To close the wizard, click Finish.

11. Click **Finish**.

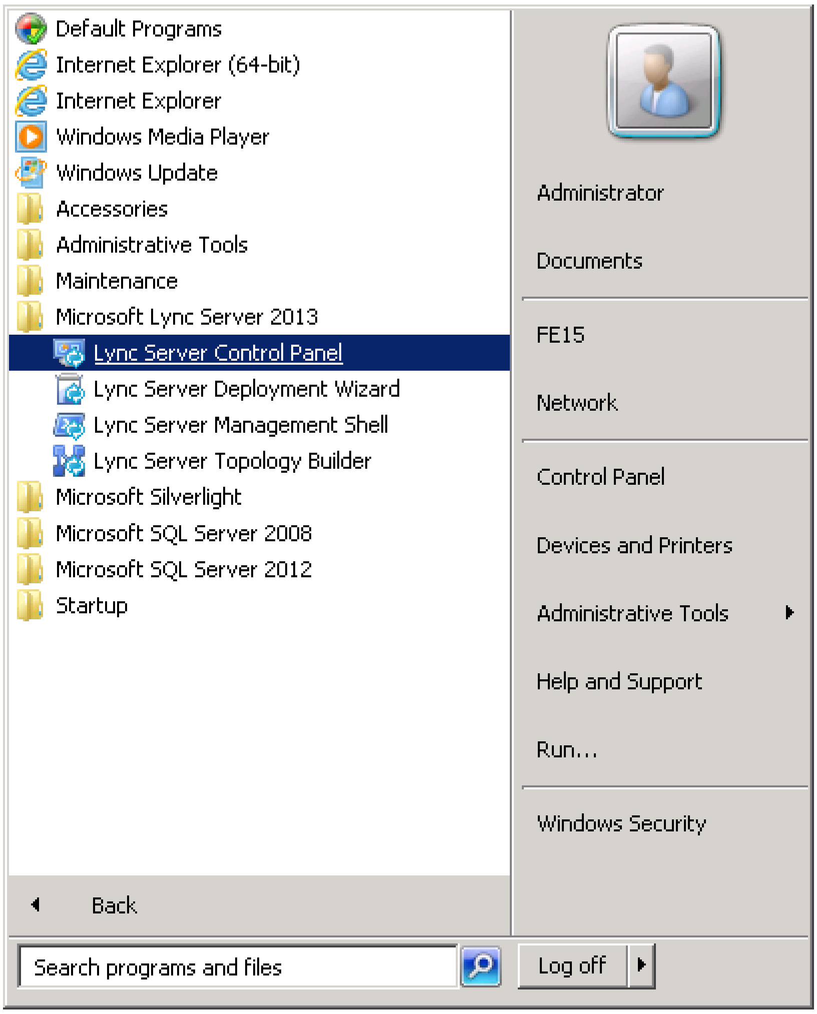
3.2 Configuring the "Route" on Lync Server 2013

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:**

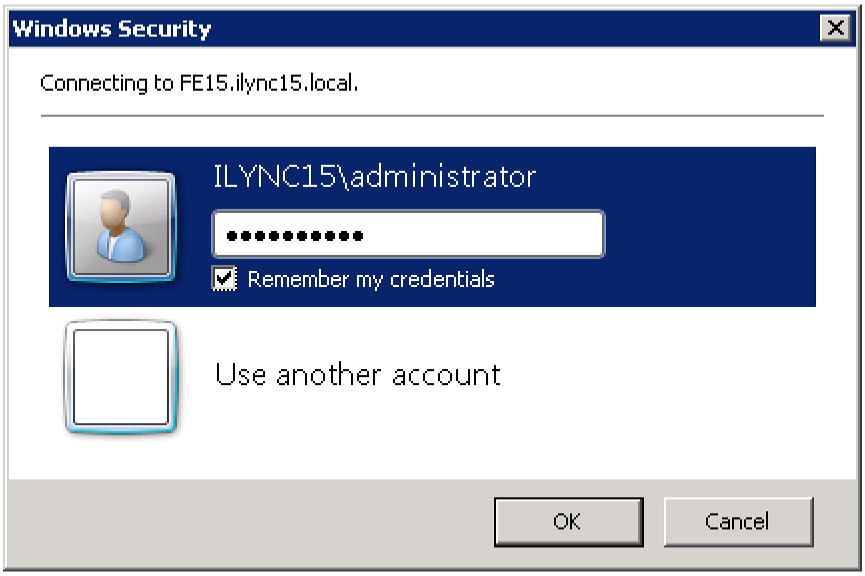
1. Start the Microsoft Lync Server 2013 Control Panel (**Start** > **All Programs** > **Microsoft Lync Server 2013** > **Lync Server Control Panel**), as shown below:

Figure 3-14: Opening the Lync Server Control Panel



You are prompted to enter your login credentials:

Figure 3-15: Lync Server Credentials



Windows Security

Connecting to FE15.ilync15.local.

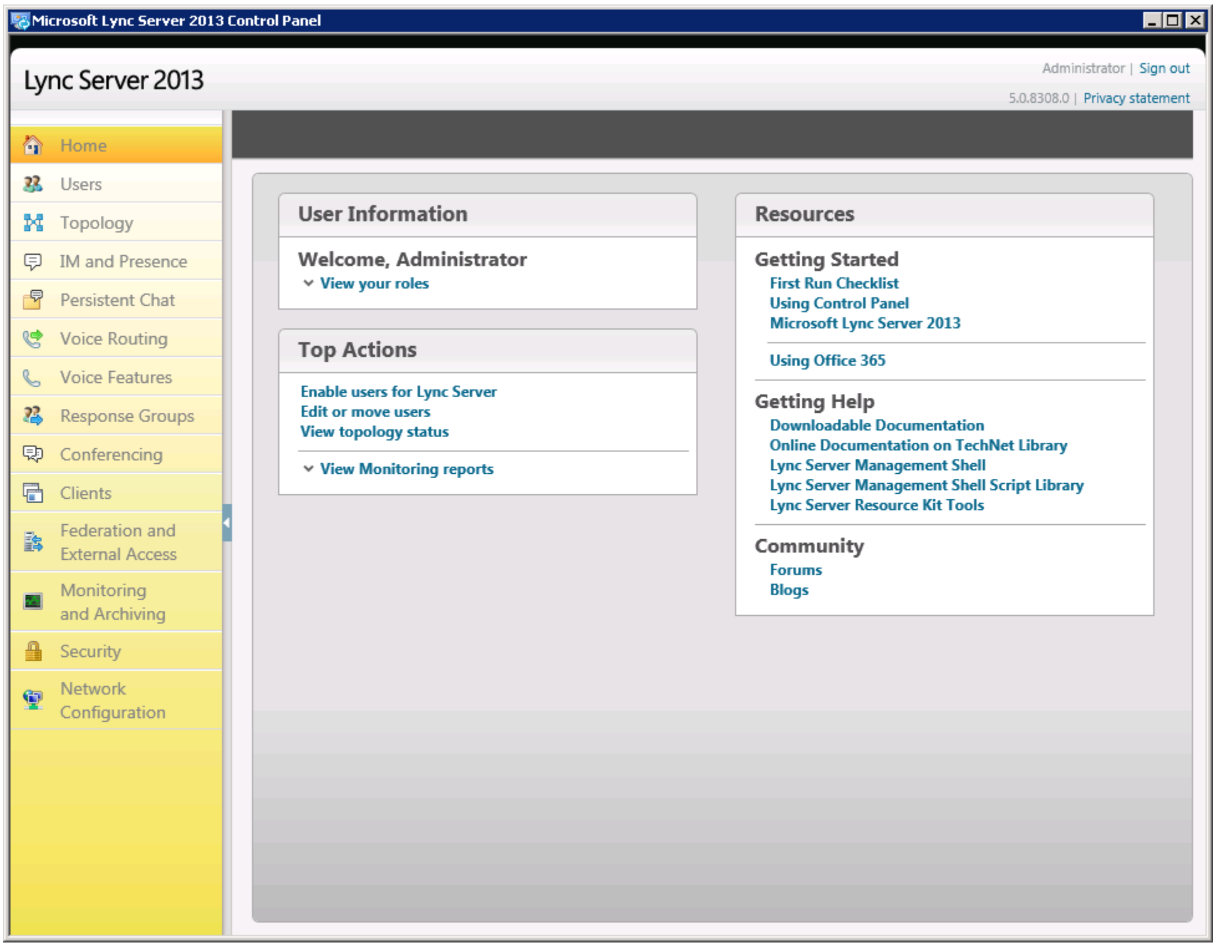
ILYNC15\administrator

●●●●●●●●●●●

☑ Remember my credentials

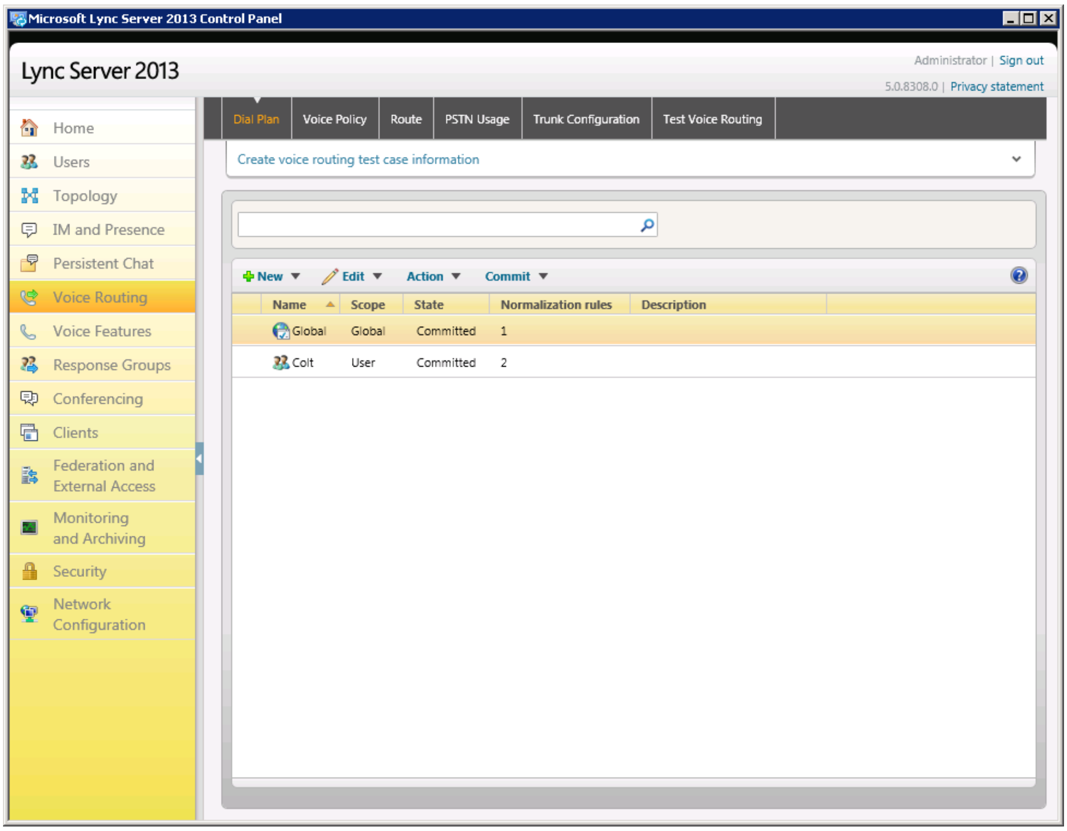
2. Enter your domain username and password, and then click **OK**; the Microsoft Lync Server 2013 Control Panel is displayed:

Figure 3-16: Microsoft Lync Server 2013 Control Panel



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

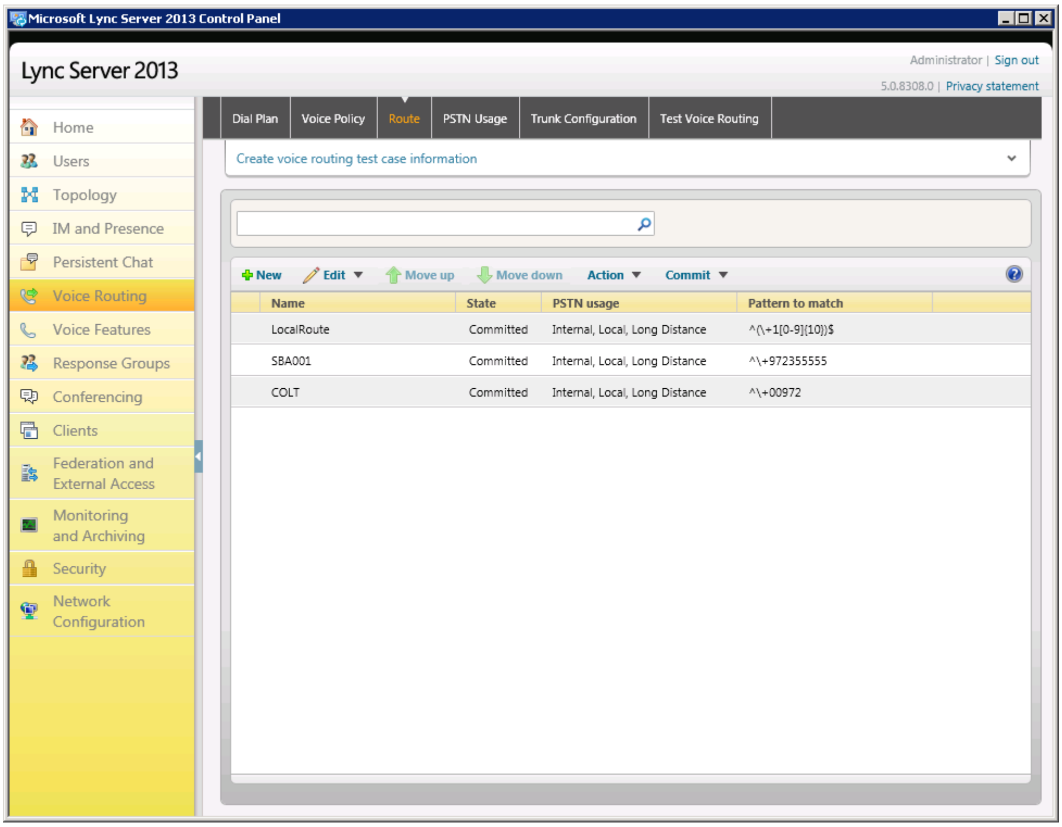
Figure 3-17: Voice Routing Page



Name	Scope	State	Normalization rules	Description
Global	Global	Committed	1	
Colt	User	Committed	2	

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

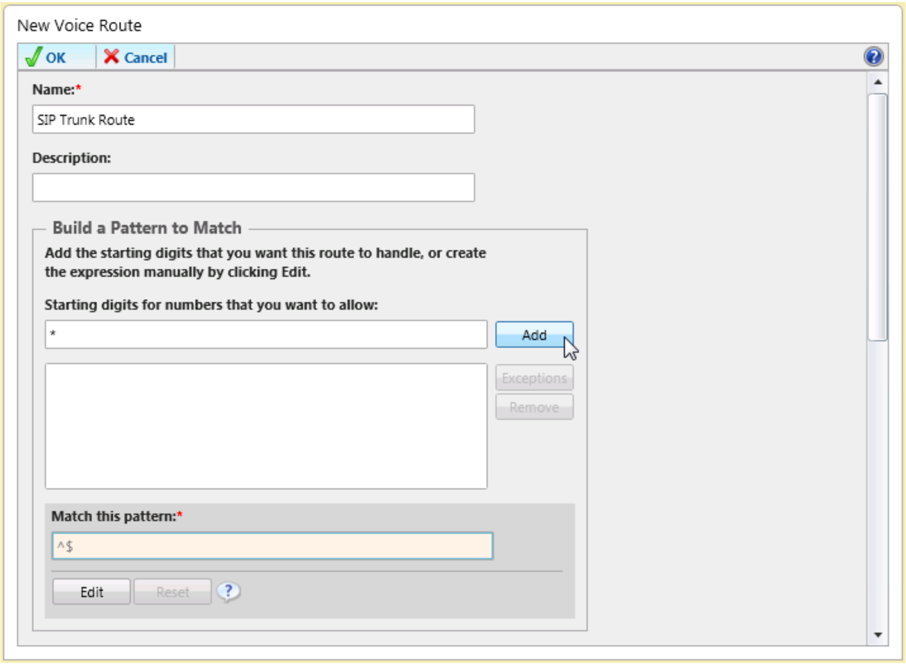
Figure 3-18: Route Tab



Name	State	PSTN usage	Pattern to match
LocalRoute	Committed	Internal, Local, Long Distance	^(\+1[0-9]{10})$
SBA001	Committed	Internal, Local, Long Distance	^\+972355555
COLT	Committed	Internal, Local, Long Distance	^\+00972

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

Figure 3-19: Adding New Voice Route



New Voice Route

Name:* SIP Trunk Route

Description:

Build a Pattern to Match

Add the starting digits that you want this route to handle, or create the expression manually by clicking Edit.

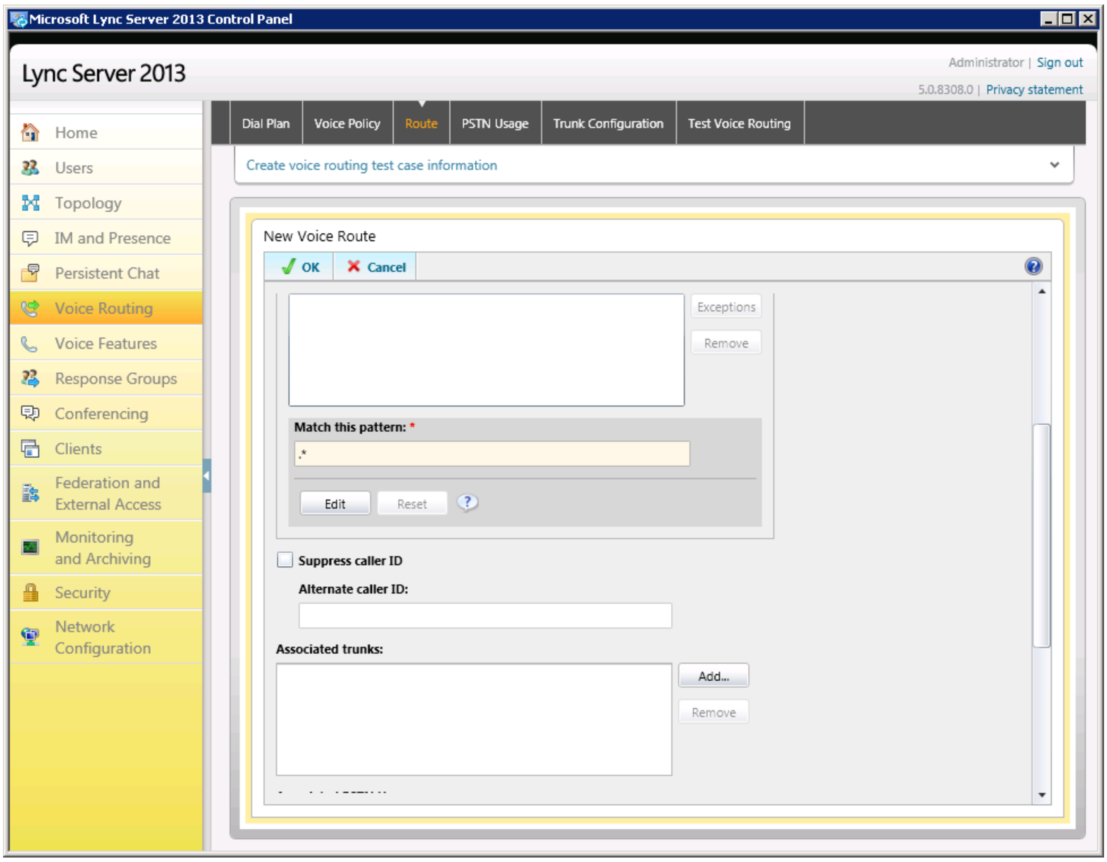
Starting digits for numbers that you want to allow: *

Match this pattern:* ^$

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



New Voice Route

Match this pattern:* .*

□ Suppress caller ID

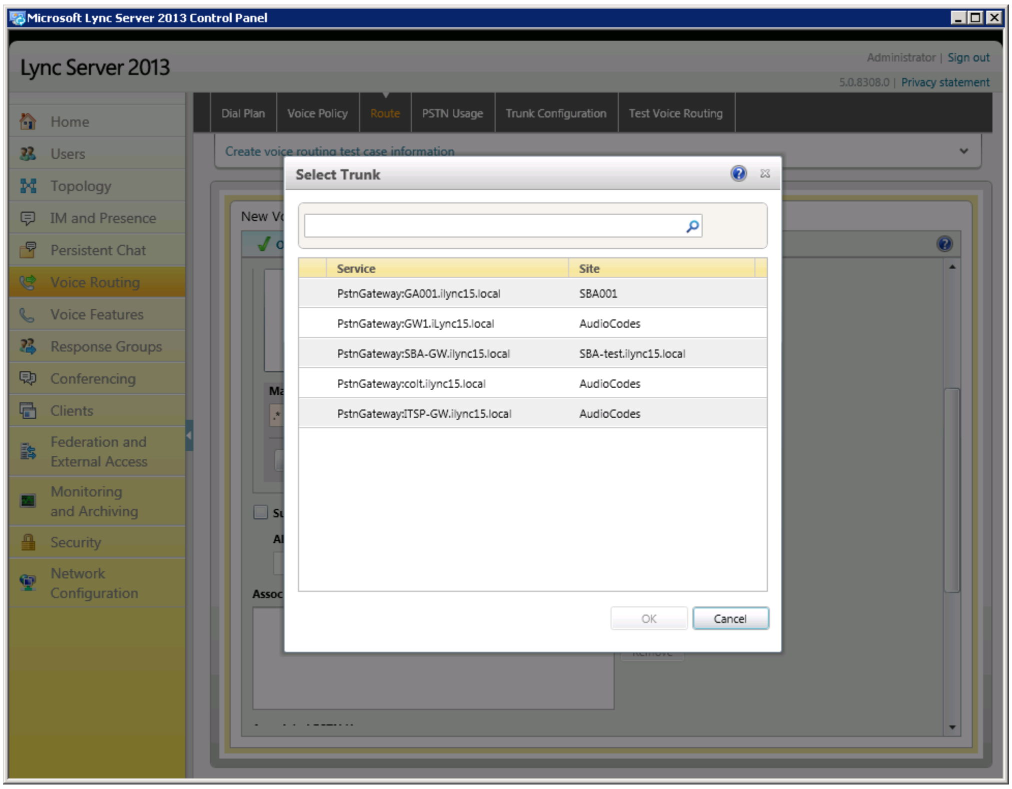
Alternate caller ID:

Associated trunks:

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:

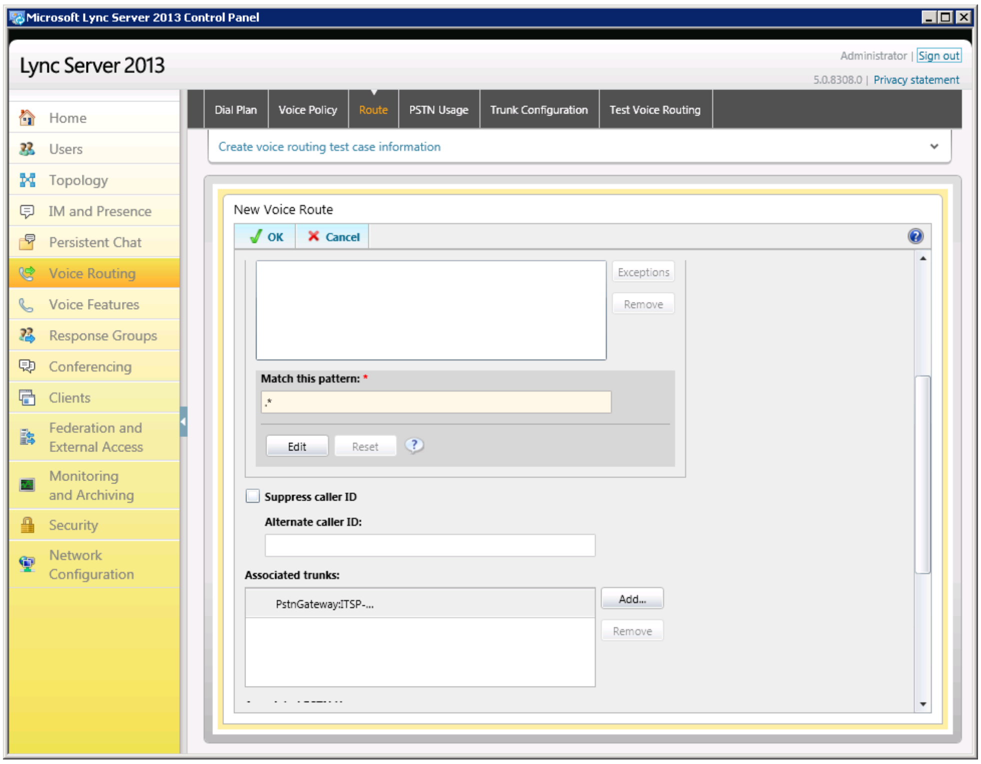
Figure 3-21: List of Deployed Trunks



Service	Site
PstnGateway:GA001.ilync15.local	SBA001
PstnGateway:GW1.iLync15.local	AudioCodes
PstnGateway:SBA-GW.ilync15.local	SBA-test.ilync15.local
PstnGateway:colt.ilync15.local	AudioCodes
PstnGateway:ITSP-GW.ilync15.local	AudioCodes

   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



Match this pattern: * .*

□ Suppress caller ID

Alternate caller ID:

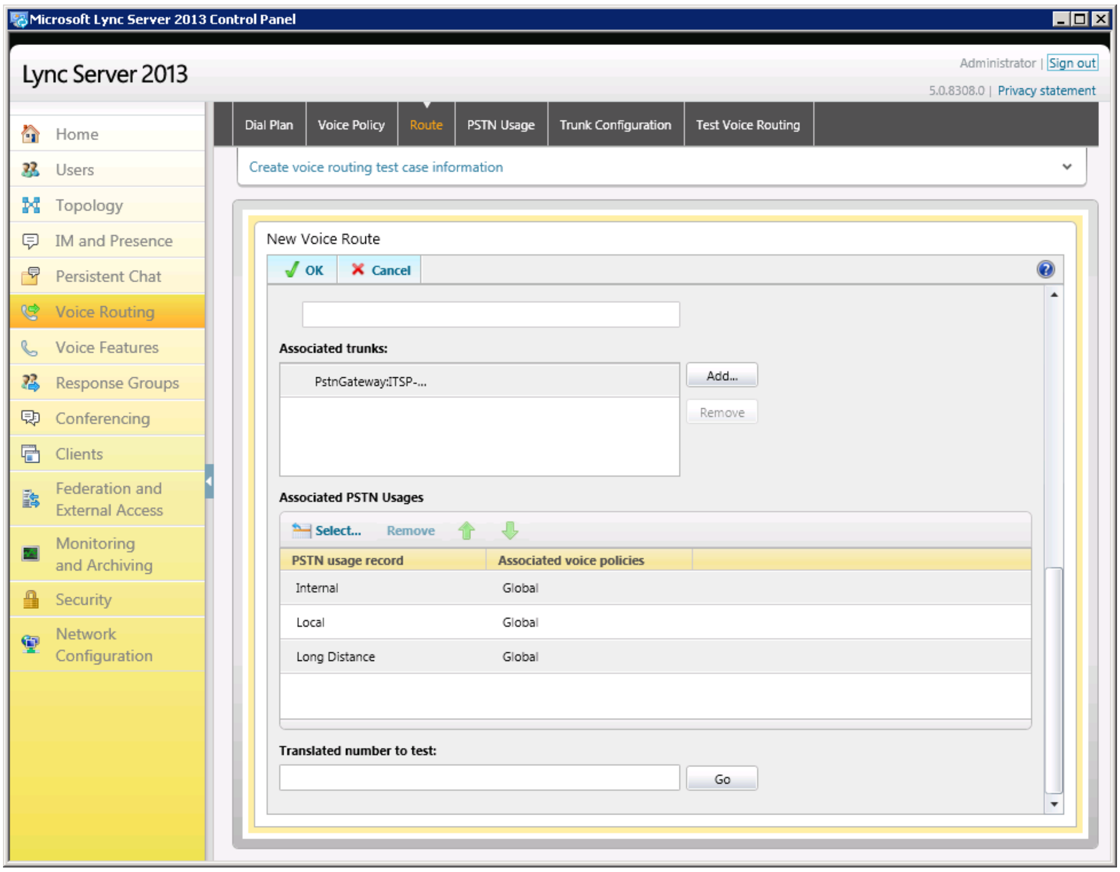
Associated trunks:

PstnGateway:ITSP-...

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



Associated trunks:

PstnGateway:ITSP-...

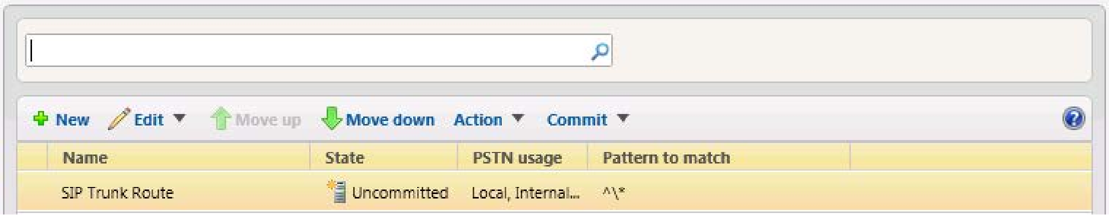
Associated PSTN Usages

PSTN usage record	Associated voice policies
Internal	Global
Local	Global
Long Distance	Global

Translated number to test:

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

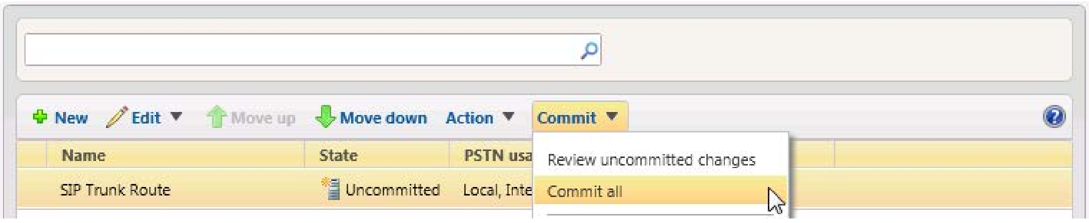
Figure 3-24: Confirmation of New Voice Route



Name	State	PSTN usage	Pattern to match
SIP Trunk Route	Uncommitted	Local, Internal...	^*

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

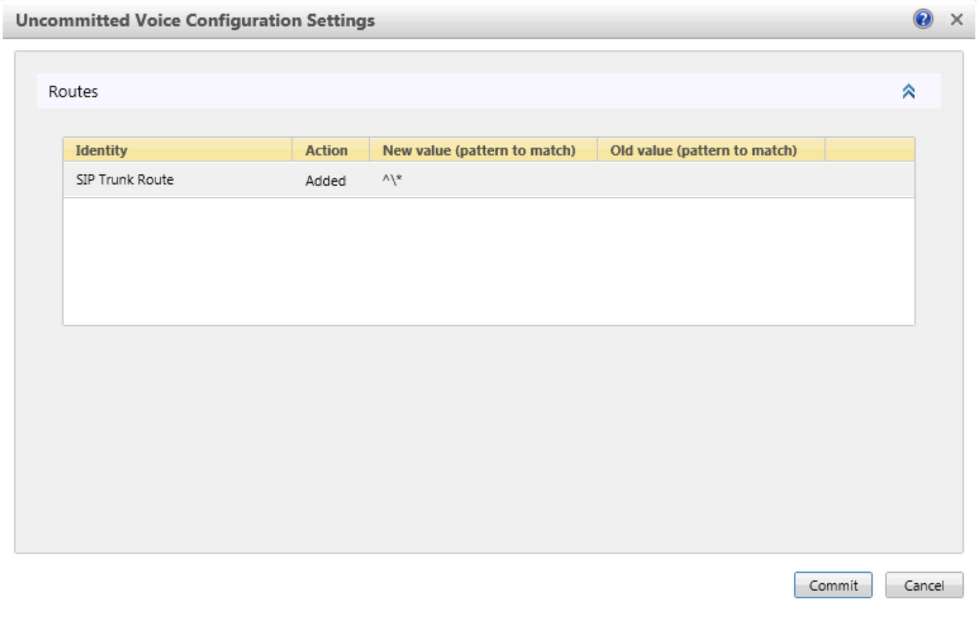
Figure 3-25: Committing Voice Routes



Name	State	PSTN usa
SIP Trunk Route	Uncommitted	Local, Inte

The Uncommitted Voice Configuration Settings page appears:

Figure 3-26: Uncommitted Voice Configuration Settings



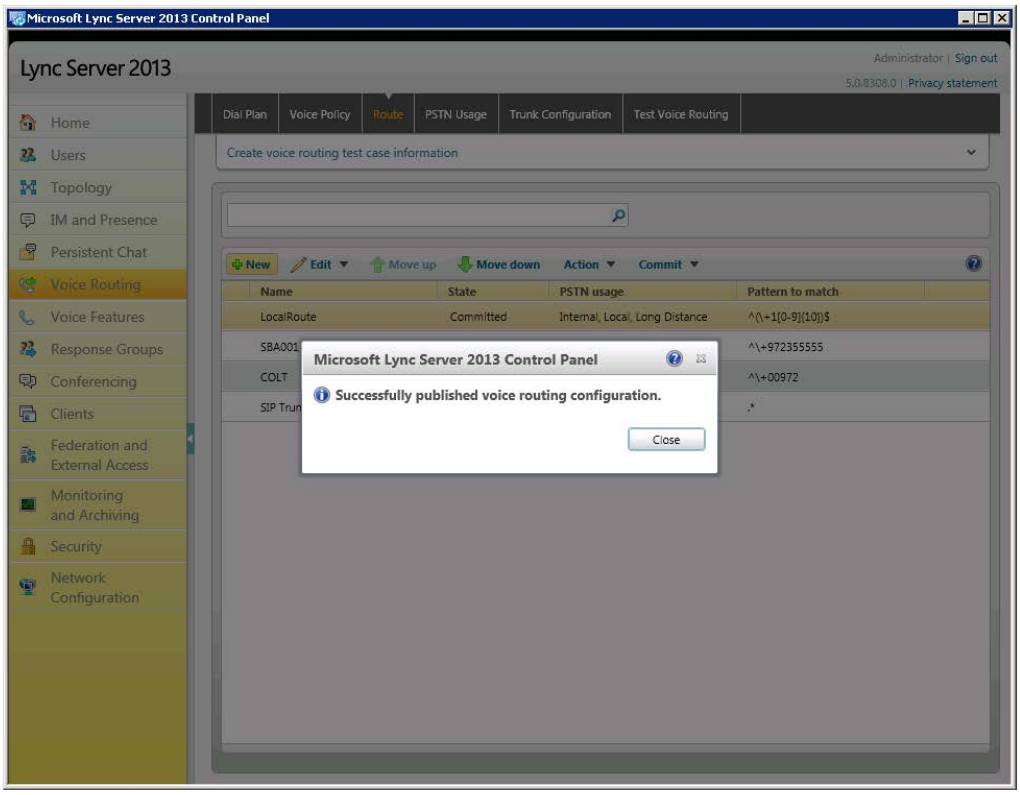
Uncommitted Voice Configuration Settings

Routes

Identity	Action	New value (pattern to match)	Old value (pattern to match)
SIP Trunk Route	Added	^*	

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



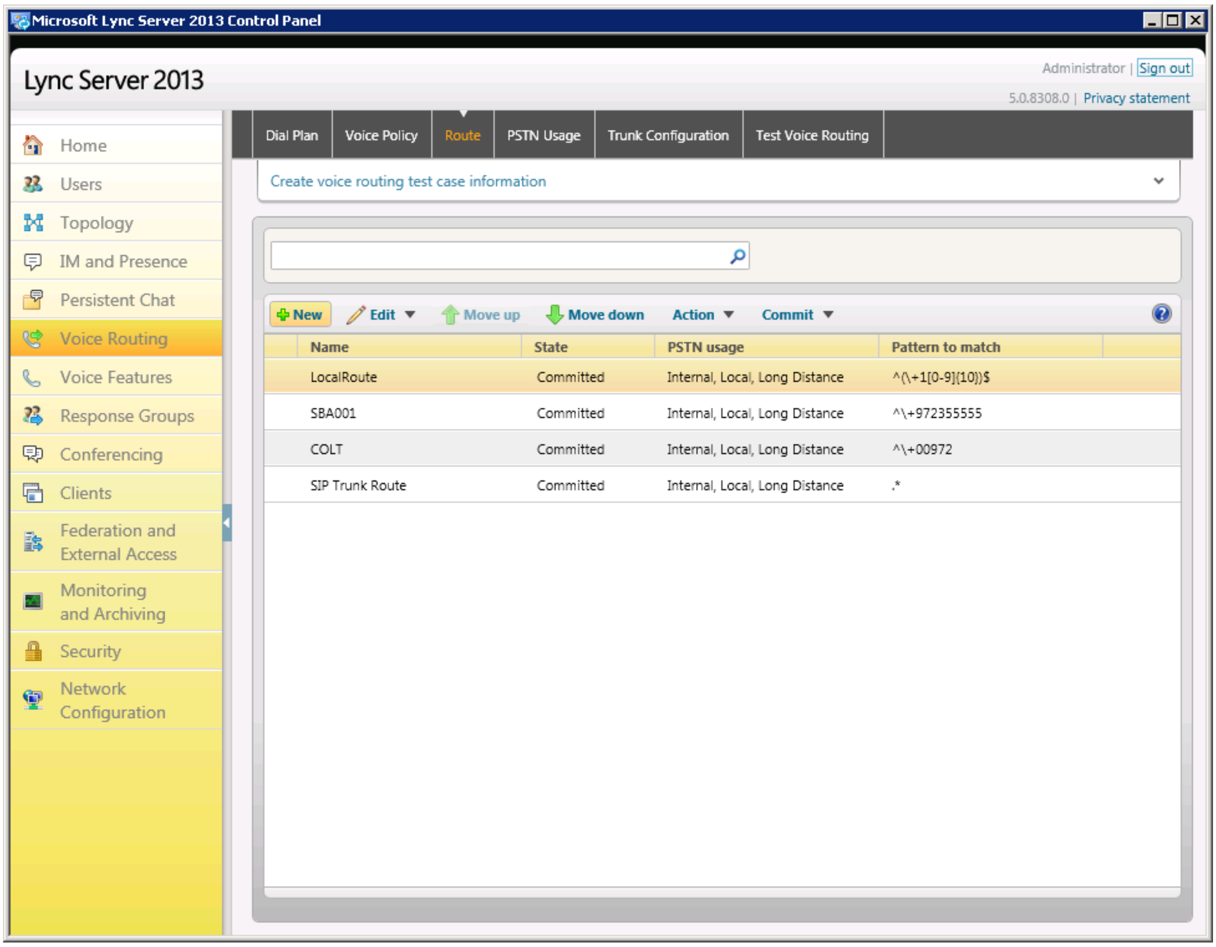
Name	State	PSTN usage	Pattern to match
LocalRoute	Committed	Internal, Local, Long Distance	^(\+1[0-9]{10})$
SBA001			^\+972355555
COLT			^\+00972
SIP Trun			.*

Microsoft Lync Server 2013 Control Panel

Successfully published 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



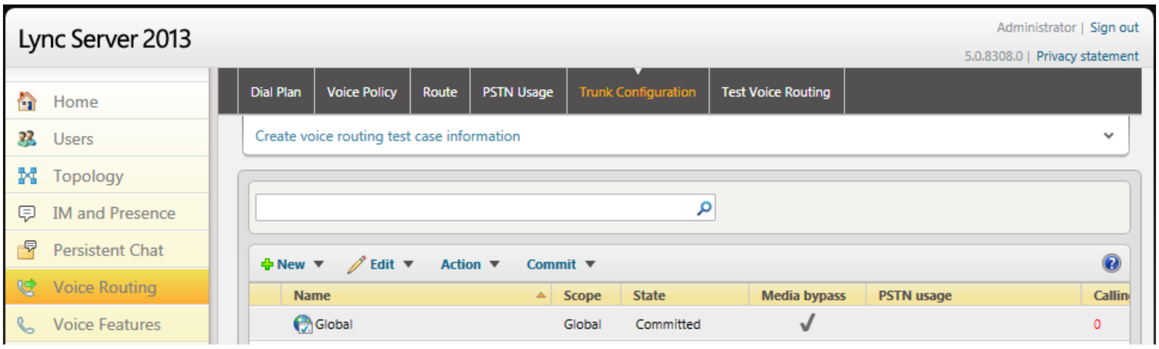
Name	State	PSTN usage	Pattern to match
LocalRoute	Committed	Internal, Local, Long Distance	^(\+1[0-9]{10})$
SBA001	Committed	Internal, Local, Long Distance	^\+972355555
COLT	Committed	Internal, Local, Long Distance	^\+00972
SIP Trunk Route	Committed	Internal, Local, Long Distance	.*

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 Netia SIP Trunk in the P-Asserted-Identity header. Using a Message Manipulation rule (see Section 4.13 on page 72), the device adds this ID to the P-Asserted-Identity header in the sent INVITE message.

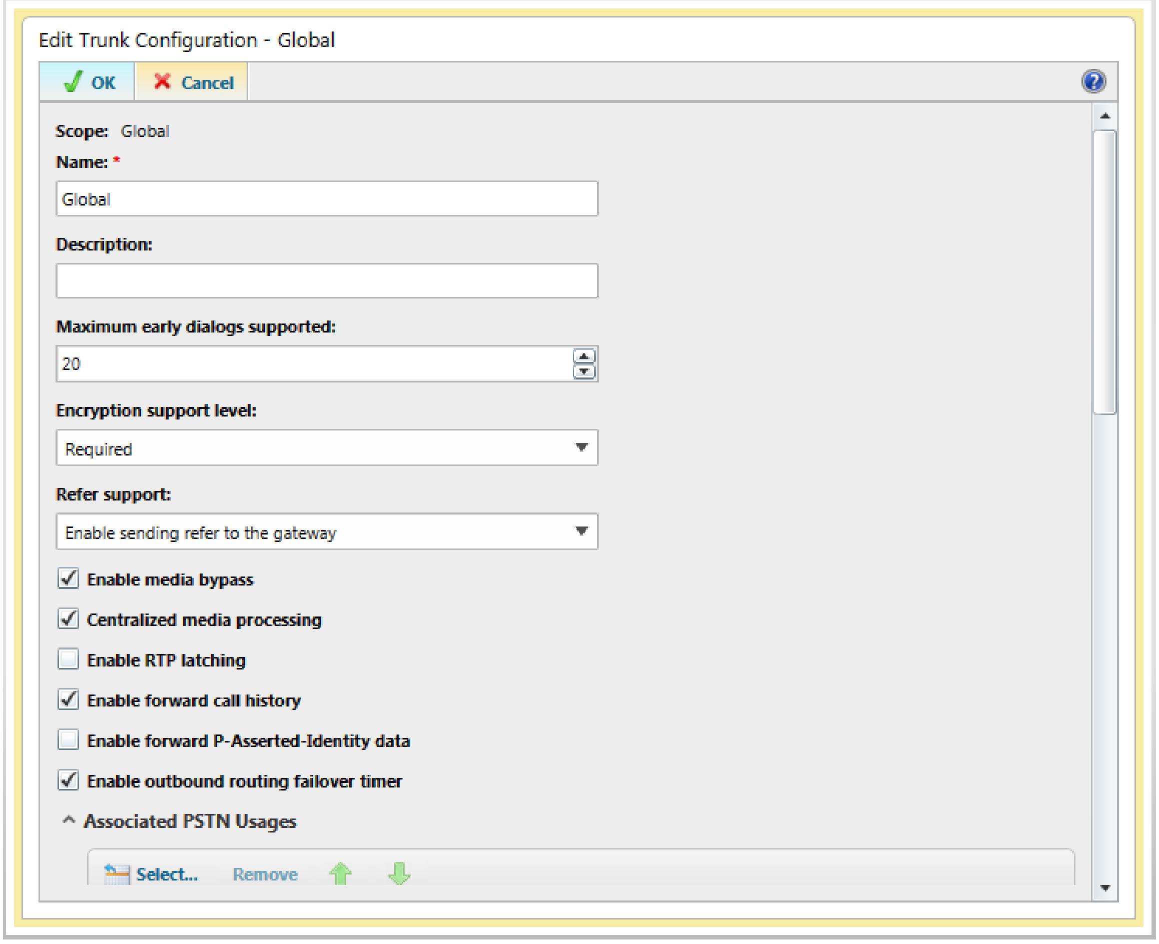
a. In the Voice Routing page, select the **Trunk Configuration** tab. Note that you can add and modify trunk configuration by site or by pool.

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



Name	Scope	State	Media bypass	PSTN usage	Callin
Global	Global	Committed	✓		0

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



Scope: Global

Name: * Global

Description:

Maximum early dialogs supported: 20

Encryption support level: Required

Refer support: Enable sending refer to the gateway

☑ Enable media bypass

☑ Centralized media processing

☐ Enable RTP latching

☑ Enable forward call history

☐ Enable forward P-Asserted-Identity data

☑ Enable outbound routing failover timer

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

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

Reader's Notes

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 Netia 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:

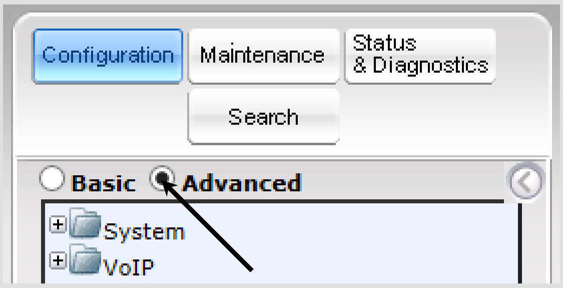
- E-SBC WAN interface - Netia SIP Trunking environment
- E-SBC LAN interface - Lync Server 2013 environment

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

Notes:

- For implementing Microsoft Lync and Netia 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:



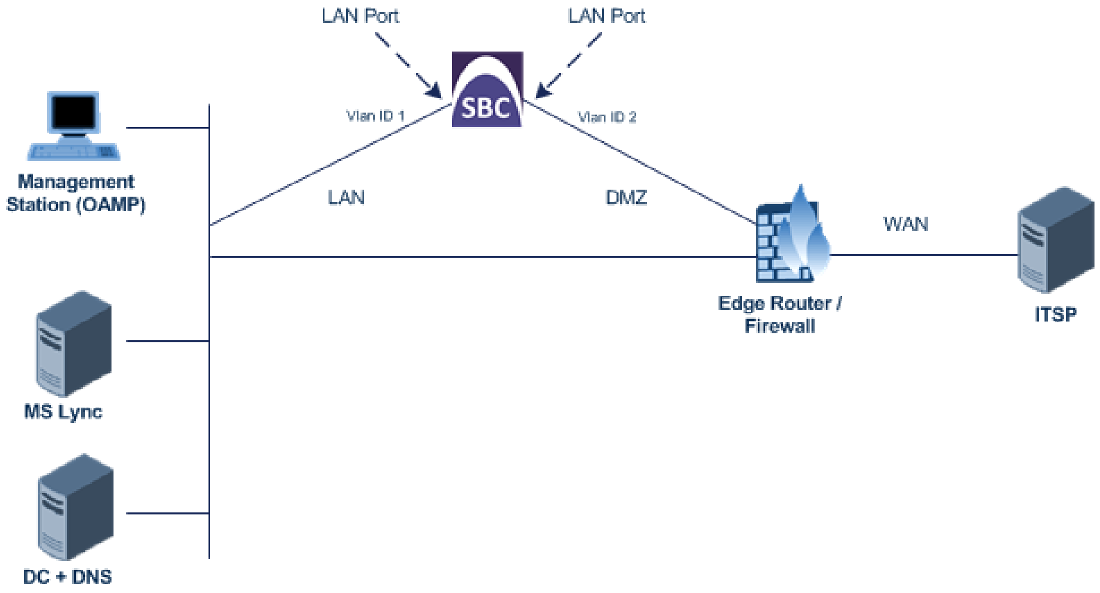
Note that when the E-SBC is reset, the Navigation tree reverts to Basic-menu display.

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
  - Netia 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:

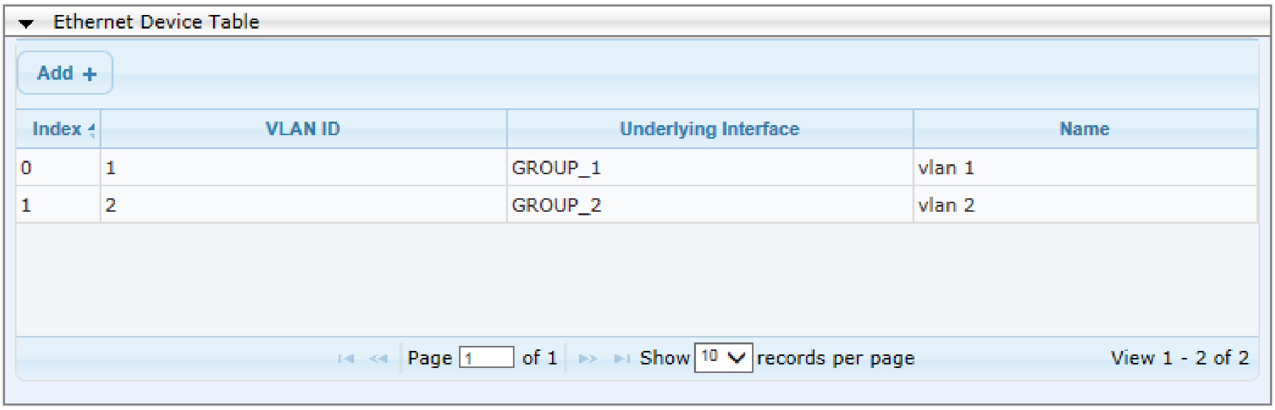
- LAN VoIP (assigned the name "Voice")
- WAN VoIP (assigned the name "WANSP")

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



Index	VLAN ID	Underlying Interface	Name
0	1	GROUP_1	vlan 1
1	2	GROUP_2	vlan 2

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	**10.15.17.10** (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**

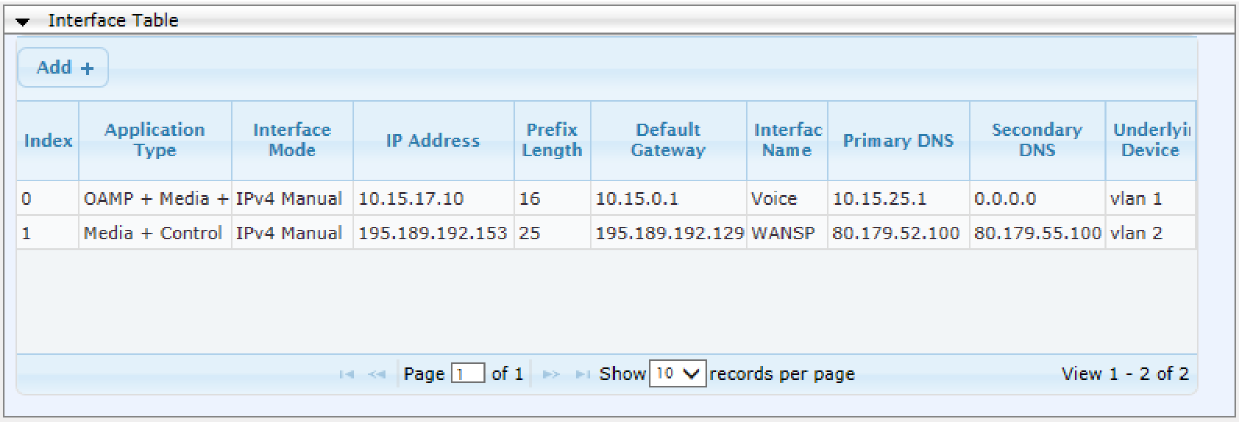
3. Add a network interface for the WAN side:
   a. Enter **1**, and then click **Add Index**.
   b. Configure the interface as follows:

Parameter	Value
Application Type	**Media + Control**
IP Address	**195.189.192.153** (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



Index	Application Type	Interface Mode	IP Address	Prefix Length	Default Gateway	Interfac Name	Primary DNS	Secondary DNS	Underlyi Device
0	OAMP + Media +	IPv4 Manual	10.15.17.10	16	10.15.0.1	Voice	10.15.25.1	0.0.0.0	vlan 1
1	Media + Control	IPv4 Manual	195.189.192.153	25	195.189.192.129	WANSP	80.179.52.100	80.179.55.100	vlan 2

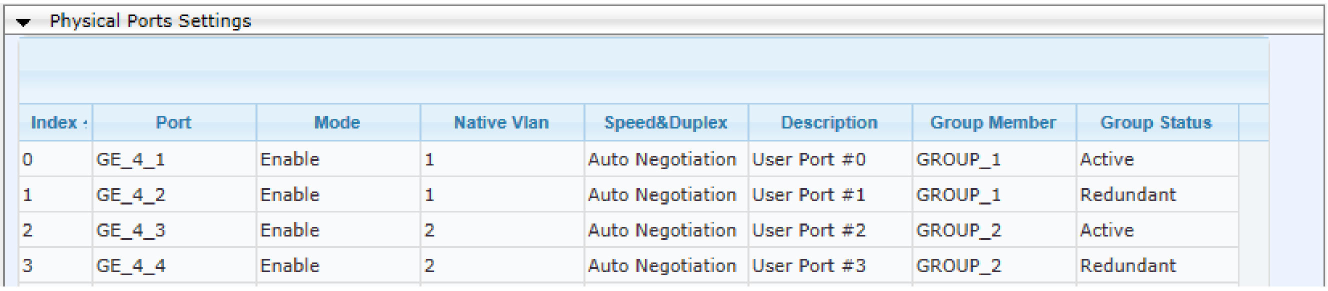
4.1.3 Step 1c: Configure the Native VLAN ID

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



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

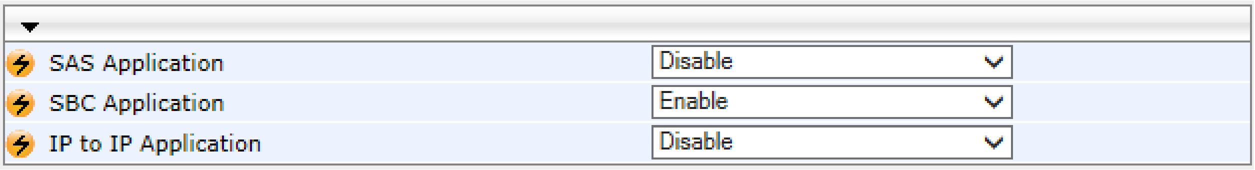
4.2 Step 2: Enable the SBC Application

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



SAS Application: Disable
SBC Application: Enable
IP to IP Application: Disable

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 86).

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

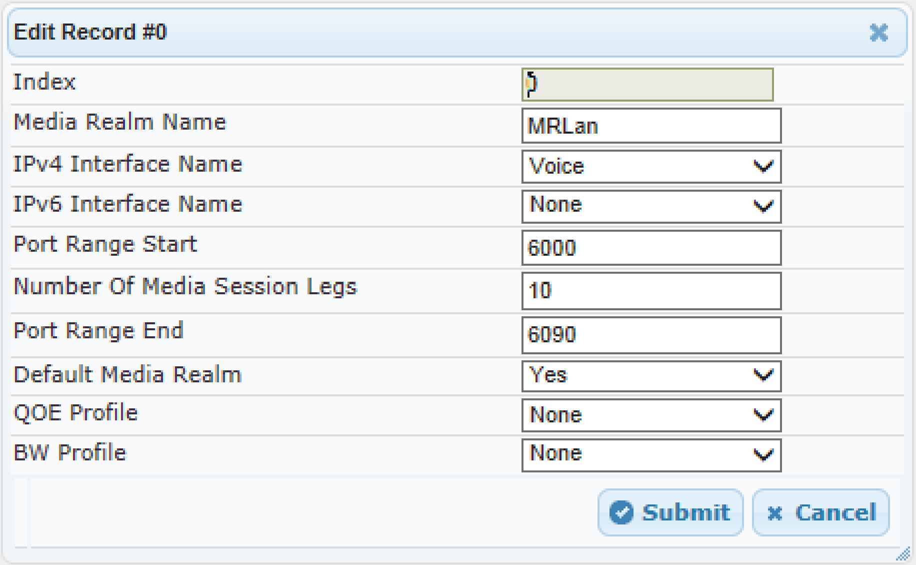
This step describes how to configure Media Realms. The simplest configuration is to create two Media Realms - one for internal (LAN) traffic and one for external (WAN) traffic.

➢ **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



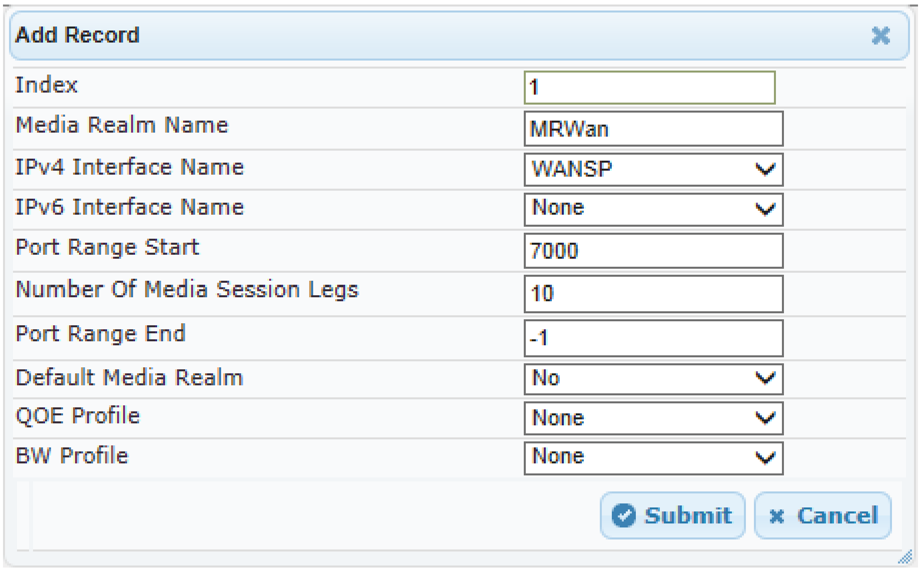
Edit Record #0

Index	0
Media Realm Name	MRLan
IPv4 Interface Name	Voice
IPv6 Interface Name	None
Port Range Start	6000
Number Of Media Session Legs	10
Port Range End	6090
Default Media Realm	Yes
QOE Profile	None
BW Profile	None

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

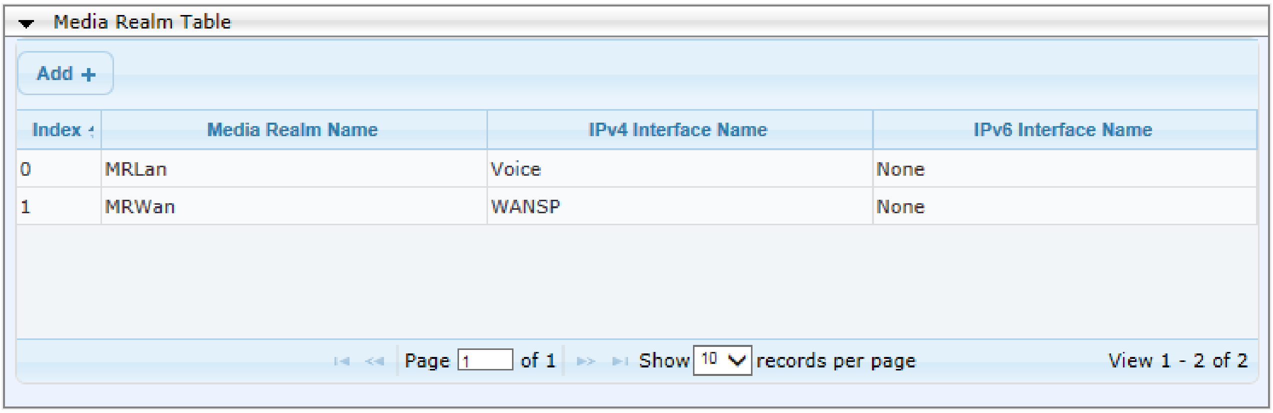


Add Record

Index	1
Media Realm Name	MRWan
IPv4 Interface Name	WANSP
IPv6 Interface Name	None
Port Range Start	7000
Number Of Media Session Legs	10
Port Range End	-1
Default Media Realm	No
QOE Profile	None
BW Profile	None

The configured Media Realms are shown in the figure below:

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



Media Realm Table

Index	Media Realm Name	IPv4 Interface Name	IPv6 Interface Name
0	MRLan	Voice	None
1	MRWan	WANSP	None

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

4.3.2 Step 3b: Configure SRDs

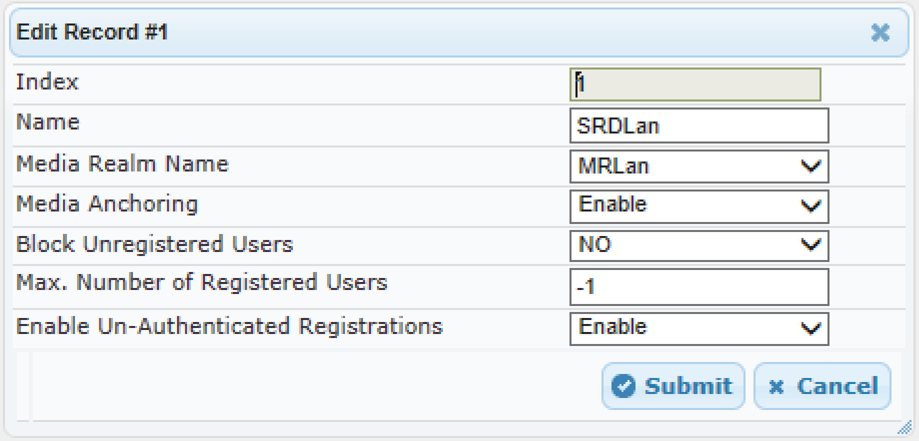
This step describes how to configure the SRDs.

➢ **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
SRD Index	**1**
SRD Name	**SRDLan** (descriptive name for SRD)
Media Realm	**MRLan** (associates SRD with Media Realm)

Figure 4-9: Configuring LAN SRD



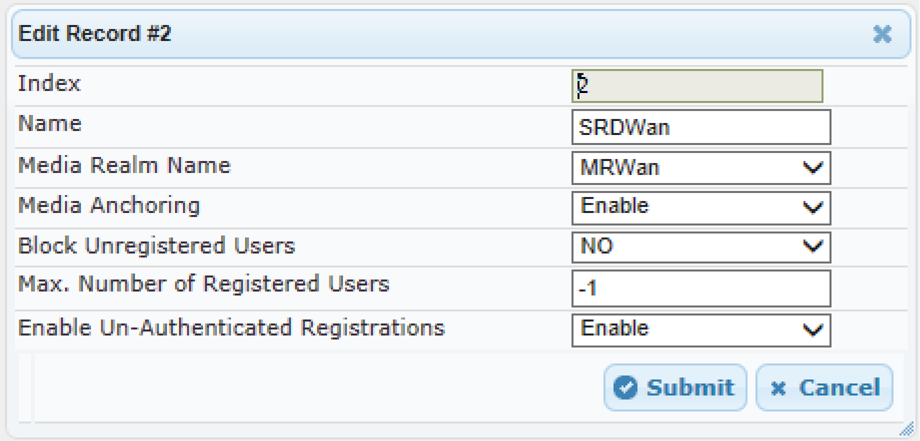
Edit Record #1

Index	1
Name	SRDLan
Media Realm Name	MRLan
Media Anchoring	Enable
Block Unregistered Users	NO
Max. Number of Registered Users	-1
Enable Un-Authenticated Registrations	Enable

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

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

Figure 4-10: Configuring WAN SRD

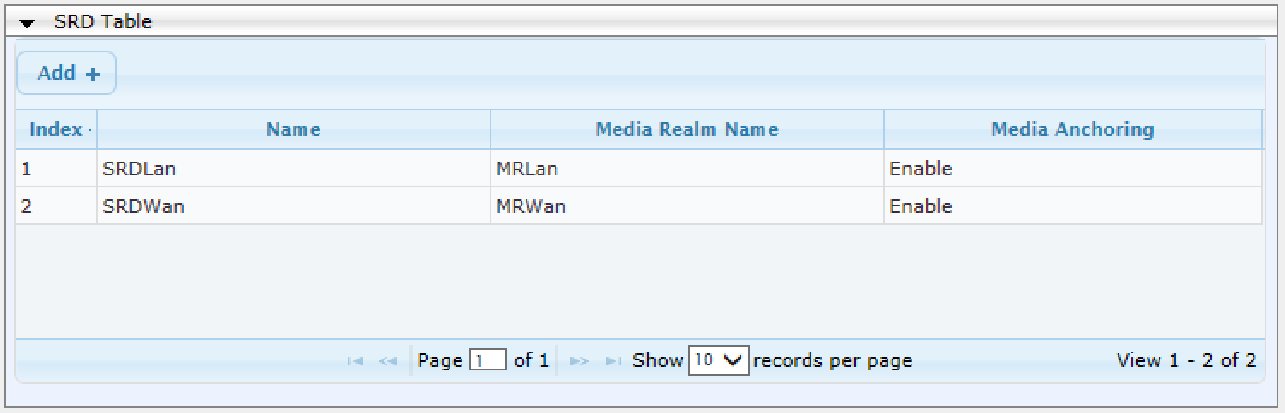


Edit Record #2

Index	2
Name	SRDWan
Media Realm Name	MRWan
Media Anchoring	Enable
Block Unregistered Users	NO
Max. Number of Registered Users	-1
Enable Un-Authenticated Registrations	Enable

The configured SRDs are shown in the figure below:

Figure 4-11: Configured SRDs in SRD Table



SRD Table

Index	Name	Media Realm Name	Media Anchoring
1	SRDLan	MRLan	Enable
2	SRDWan	MRWan	Enable

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

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:

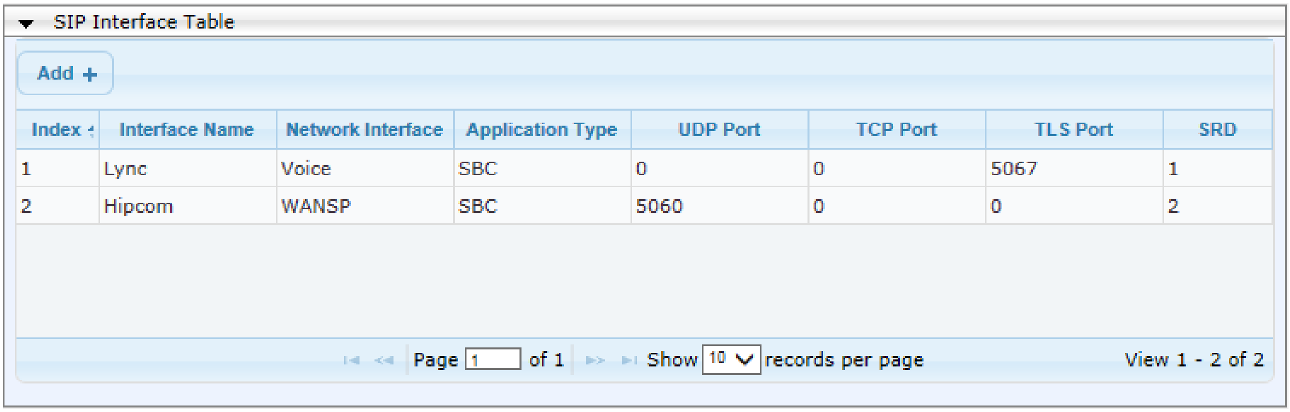
Parameter	Value
Index	**1**
Interface Name	**Lync** (arbitrary descriptive name)
Network Interface	**Voice**
Application Type	**SBC**
TLS Port	**5067**
TCP and UDP	**0**
SRD	**1**

3. Configure a SIP interface for the WAN:

Parameter	Value
Index	**2**
Interface Name	**Netia** (arbitrary descriptive name)
Network Interface	**WANSP**
Application Type	**SBC**
UDP Port	**5060**
TCP and TLS	**0**
SRD	**2**

The configured SIP Interfaces are shown in the figure below:

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



SIP Interface Table

Add +

Index	Interface Name	Network Interface	Application Type	UDP Port	TCP Port	TLS Port	SRD
1	Lync	Voice	SBC	0	0	5067	1
2	Hipcom	WANSP	SBC	5060	0	0	2

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

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
- Netia SIP Trunk

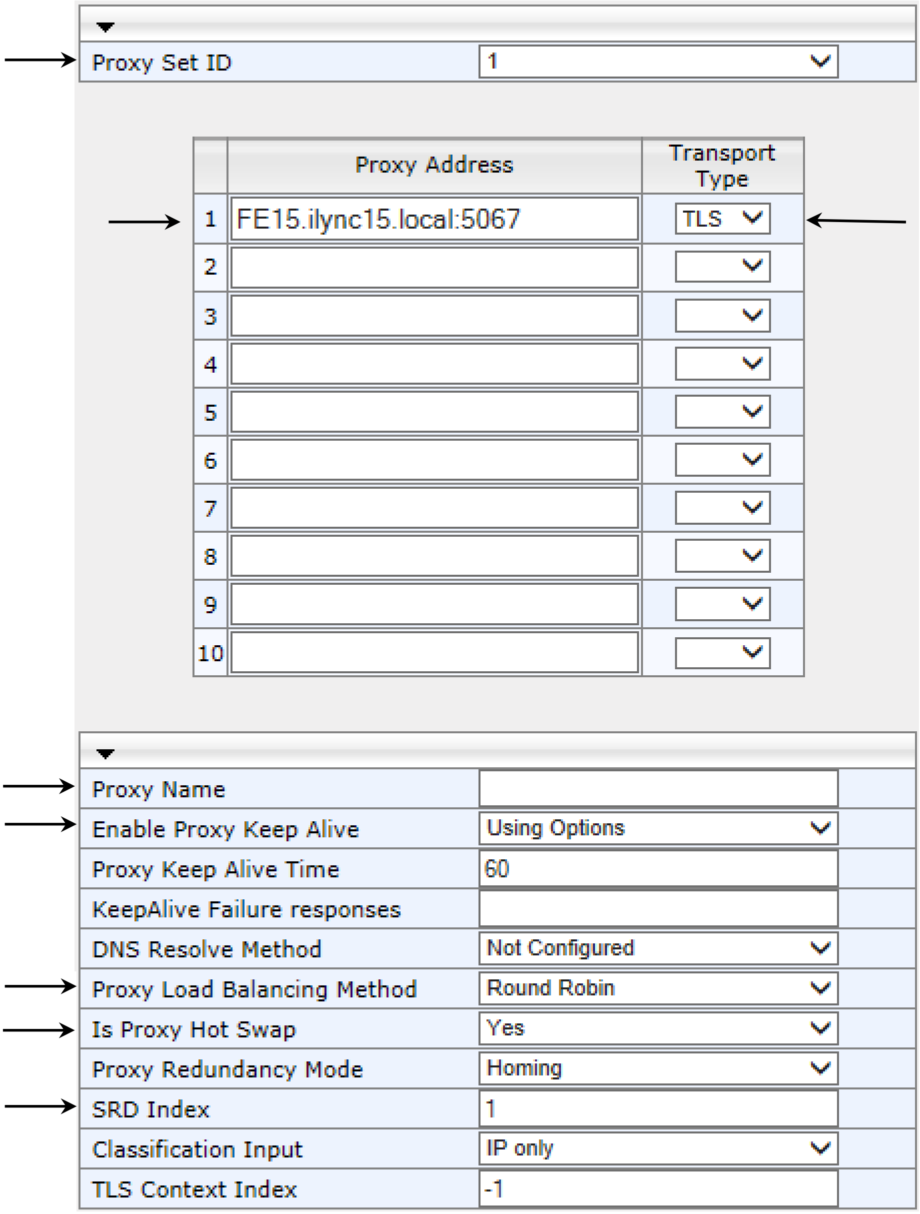
These Proxy Sets will later be associated with IP Groups.

➢ 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**
SRD Index	**1**

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



Proxy Set ID	1

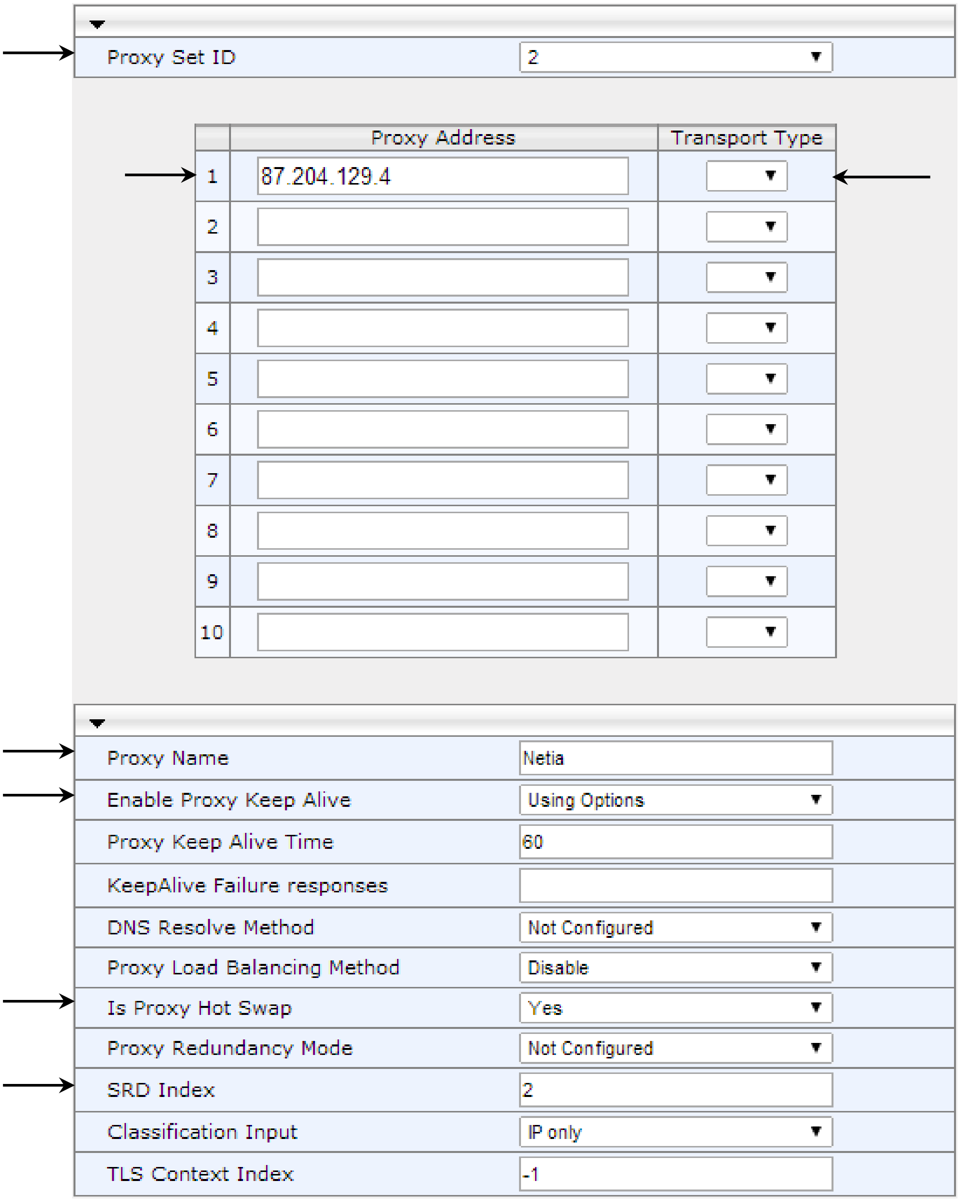
	Proxy Address	Transport Type
1	FE15.ilync15.local:5067	TLS
2		
3		
4		
5		
6		
7		
8		
9		
10		

Proxy Name	
Enable Proxy Keep Alive	Using Options
Proxy Keep Alive Time	60
KeepAlive Failure responses	
DNS Resolve Method	Not Configured
Proxy Load Balancing Method	Round Robin
Is Proxy Hot Swap	Yes
Proxy Redundancy Mode	Homing
SRD Index	1
Classification Input	IP only
TLS Context Index	-1

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

Parameter	Value
Proxy Set ID	**2**
Proxy Address	**87.204.129.4** (Netia IP address / FQDN)
Transport Type	**UDP**
Proxy Name	**Netia** (arbitrary descriptive name)
Enable Proxy Keep Alive	**Using Options**
Is Proxy Hot Swap	**Yes**
SRD Index	**2** (enables classification by Proxy Set for SRD of IP Group belonging to Netia SIP Trunk)

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



Proxy Set ID	2

	Proxy Address	Transport Type
1	87.204.129.4	
2		
3		
4		
5		
6		
7		
8		
9		
10		

Proxy Name	Netia
Enable Proxy Keep Alive	Using Options
Proxy Keep Alive Time	60
KeepAlive Failure responses	
DNS Resolve Method	Not Configured
Proxy Load Balancing Method	Disable
Is Proxy Hot Swap	Yes
Proxy Redundancy Mode	Not Configured
SRD Index	2
Classification Input	IP only
TLS Context Index	-1

4. Reset the E-SBC with a burn to flash for these settings to take effect (see Section 4.16 on page 86).

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
- Netia 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:

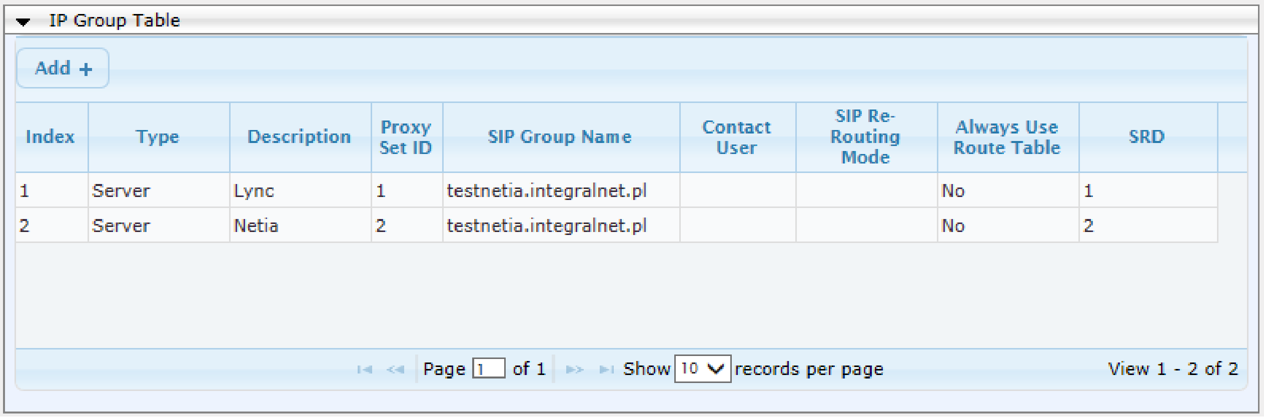
Parameter	Value
Index	**1**
Type	**Server**
Description	**Lync** (arbitrary descriptive name)
Proxy Set ID	**1**
SIP Group Name	**testnetia.integralnet.pl** (according to ITSP requirement)
SRD	**1**
Media Realm Name	**MRLan**
IP Profile ID	**1**

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

Parameter	Value
Index	**2**
Type	**Server**
Description	**Netia** (arbitrary descriptive name)
Proxy Set ID	**2**
SIP Group Name	**testnetia.integralnet.pl** (according to ITSP requirement)
SRD	**2**
Media Realm Name	**MRWan**
IP Profile ID	**2**
Classify By Proxy Set	**Disable** (classification will be done according to the Classification Table)
Destination URI Input	**TO** (routing for this IP Group will be done according to the **TO** SIP Header)

The configured IP Groups are shown in the figure below:

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



Index	Type	Description	Proxy Set ID	SIP Group Name	Contact User	SIP Re-Routing Mode	Always Use Route Table	SRD
1	Server	Lync	1	testnetia.integralnet.pl			No	1
2	Server	Netia	2	testnetia.integralnet.pl			No	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
- Netia SIP trunk - to operate in non-secure mode using RTP and UDP

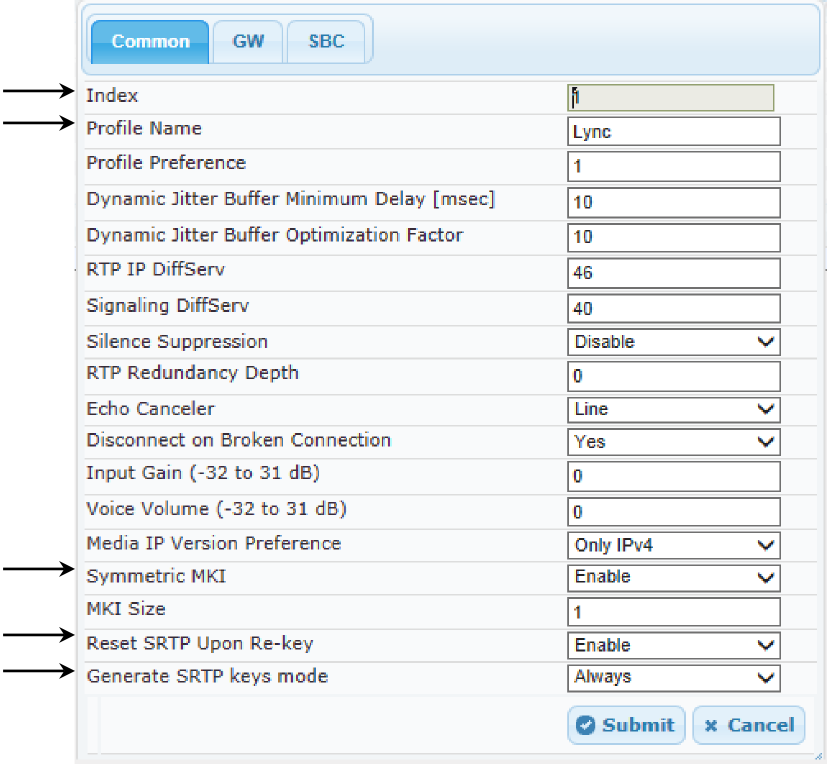
Note that the IP Profiles were assigned to these entities (i.e., IP Groups) in the previous step (see Section 4.5 4.5on page 46).

➢ **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

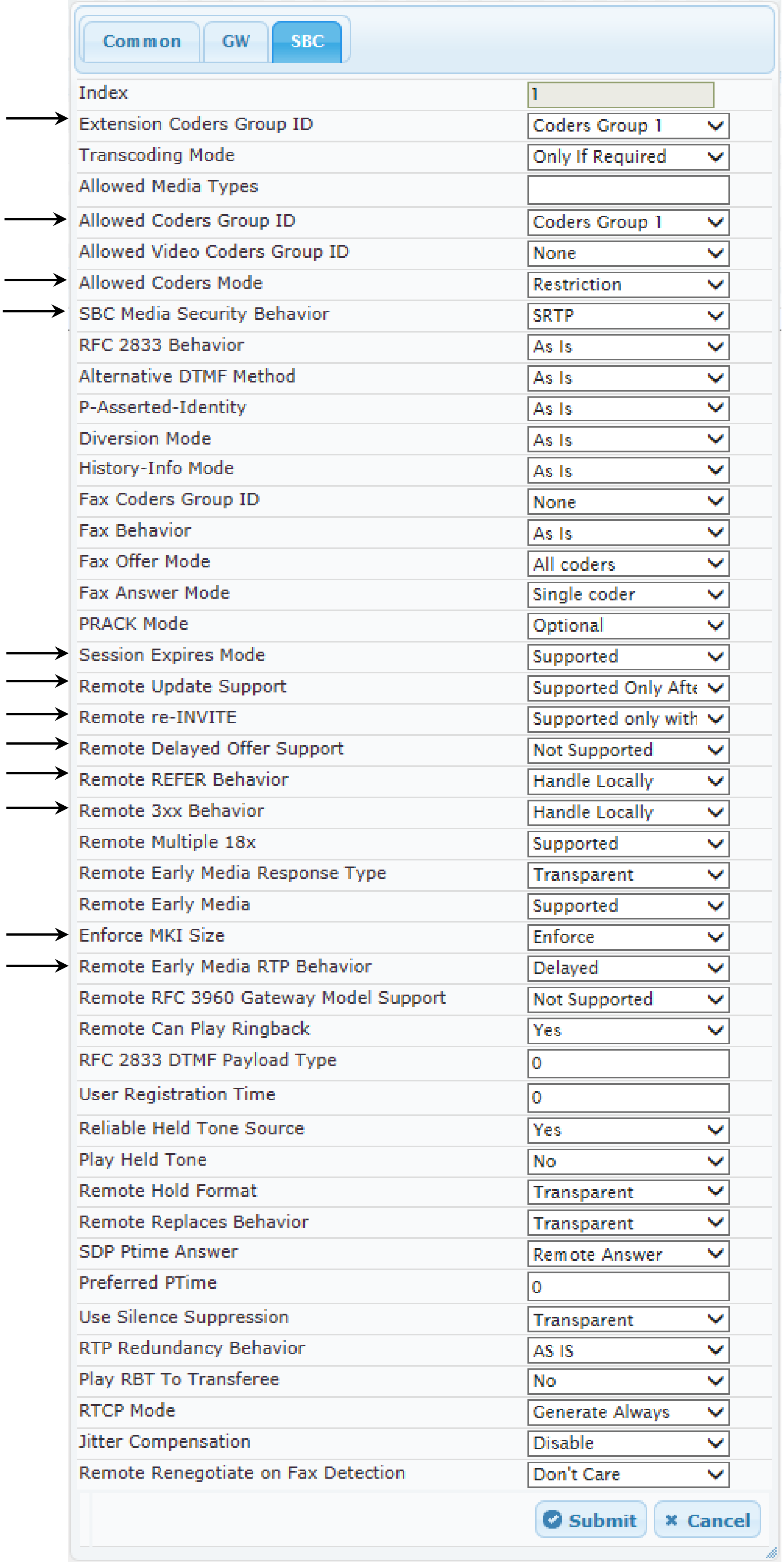


Index	1
Profile Name	Lync
Profile Preference	1
Dynamic Jitter Buffer Minimum Delay [msec]	10
Dynamic Jitter Buffer Optimization Factor	10
RTP IP DiffServ	46
Signaling DiffServ	40
Silence Suppression	Disable
RTP Redundancy Depth	0
Echo Canceler	Line
Disconnect on Broken Connection	Yes
Input Gain (-32 to 31 dB)	0
Voice Volume (-32 to 31 dB)	0
Media IP Version Preference	Only IPv4
Symmetric MKI	Enable
MKI Size	1
Reset SRTP Upon Re-key	Enable
Generate SRTP keys mode	Always

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

Parameter	Value
Extension Coders Group ID	**Coders Group 1**
Allowed Coders Group ID	**Coders Group 1**
Allowed Coders Mode	**Restriction** (only Allowed Coders will be introduced in SDP offer)
SBC Media Security Behavior	**SRTP**
PRACK Mode	**Optional** (required, as Netia SIP Trunk does not support PRACK)
Session Expires Mode	**Supported** (required because Netia's SIP Trunk does not support Session Timer, so SBC should negotiate it with Lync)
Remote Update Support	**Supported Only After Connect**
Remote Re-INVITE Support	**Supported Only With SDP**
Remote Delayed Offer Support	**Not Supported**
Remote REFER Behavior	**Handle Locally** (required, 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

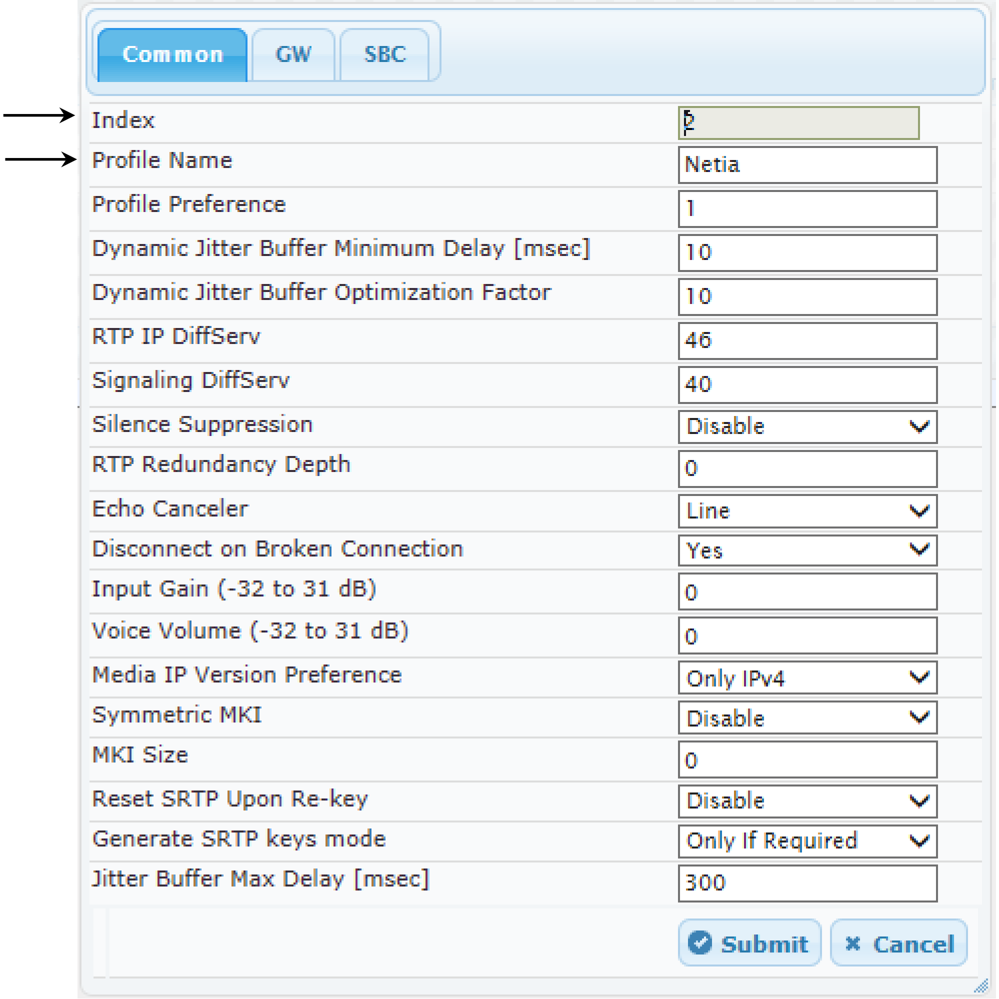


	Index	1
→	Extension Coders Group ID	Coders Group 1
	Transcoding Mode	Only If Required
	Allowed Media Types	
→	Allowed Coders Group ID	Coders Group 1
	Allowed Video Coders Group ID	None
→	Allowed Coders Mode	Restriction
→	SBC Media Security Behavior	SRTP
	RFC 2833 Behavior	As Is
	Alternative DTMF Method	As Is
	P-Asserted-Identity	As Is
	Diversion Mode	As Is
	History-Info Mode	As Is
	Fax Coders Group ID	None
	Fax Behavior	As Is
	Fax Offer Mode	All coders
	Fax Answer Mode	Single coder
	PRACK Mode	Optional
→	Session Expires Mode	Supported
→	Remote Update Support	Supported Only Afte
→	Remote re-INVITE	Supported only with
→	Remote Delayed Offer Support	Not Supported
→	Remote REFER Behavior	Handle Locally
→	Remote 3xx Behavior	Handle Locally
	Remote Multiple 18x	Supported
	Remote Early Media Response Type	Transparent
	Remote Early Media	Supported
→	Enforce MKI Size	Enforce
→	Remote Early Media RTP Behavior	Delayed
	Remote RFC 3960 Gateway Model Support	Not Supported
	Remote Can Play Ringback	Yes
	RFC 2833 DTMF Payload Type	0
	User Registration Time	0
	Reliable Held Tone Source	Yes
	Play Held Tone	No
	Remote Hold Format	Transparent
	Remote Replaces Behavior	Transparent
	SDP Ptime Answer	Remote Answer
	Preferred PTime	0
	Use Silence Suppression	Transparent
	RTP Redundancy Behavior	AS IS
	Play RBT To Transferee	No
	RTCP Mode	Generate Always
	Jitter Compensation	Disable
	Remote Renegotiate on Fax Detection	Don't Care

5. Configure an IP Profile for the Netia SIP Trunk:
6. Click **Add**.
7. Click the **Common** tab, and then configure the parameters as follows:

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

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

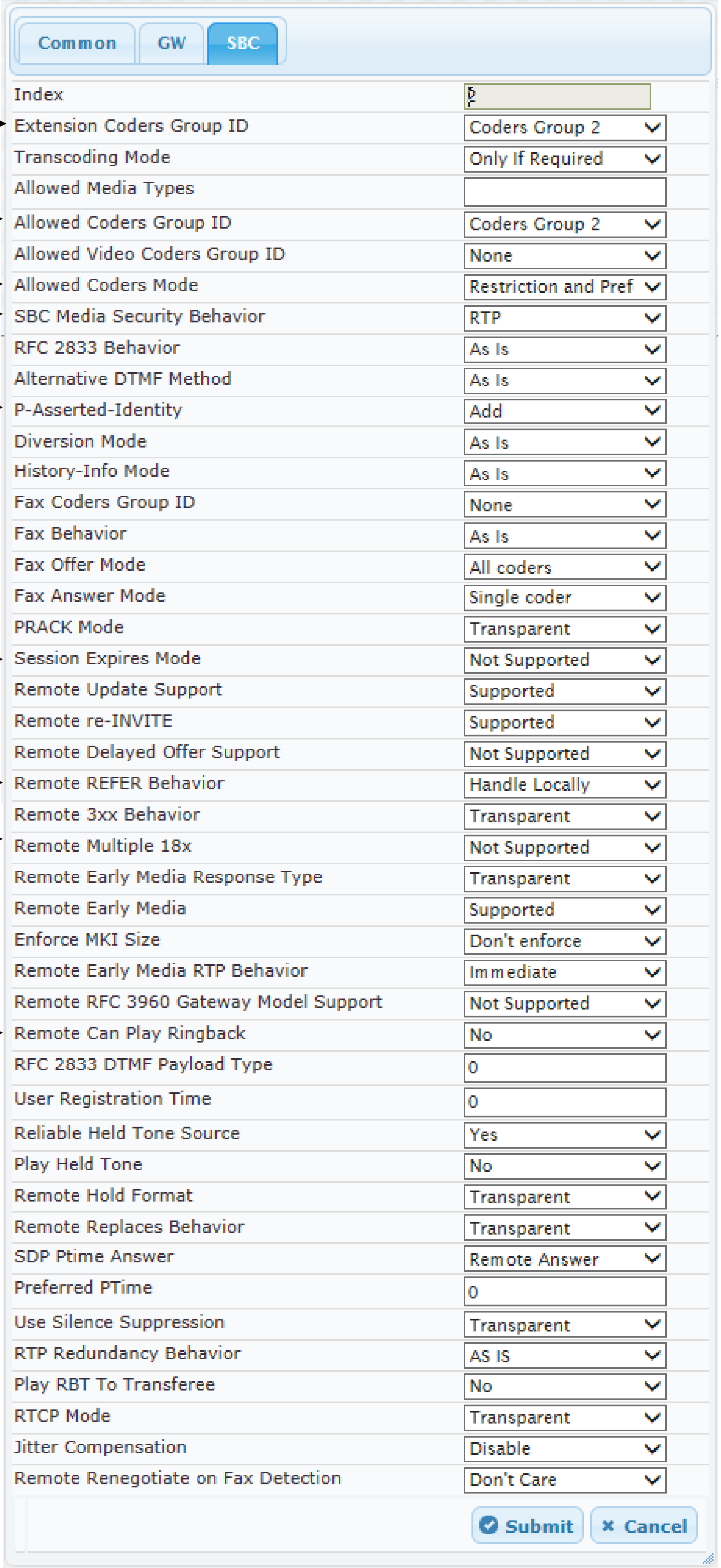


Index	2
Profile Name	Netia
Profile Preference	1
Dynamic Jitter Buffer Minimum Delay [msec]	10
Dynamic Jitter Buffer Optimization Factor	10
RTP IP DiffServ	46
Signaling DiffServ	40
Silence Suppression	Disable
RTP Redundancy Depth	0
Echo Canceler	Line
Disconnect on Broken Connection	Yes
Input Gain (-32 to 31 dB)	0
Voice Volume (-32 to 31 dB)	0
Media IP Version Preference	Only IPv4
Symmetric MKI	Disable
MKI Size	0
Reset SRTP Upon Re-key	Disable
Generate SRTP keys mode	Only If Required
Jitter Buffer Max Delay [msec]	300

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

Parameter	Value
Profile ID	**2**
Extension Coders Group ID	**Coders Group 2**
Allowed Coders Group ID	**Coders Group 2**
Allowed Coders Mode	**Restriction and Preference** (lists Allowed Coders first and then original coders in received SDP offer)
SBC Media Security Behavior	**RTP**
P-Asserted-Identity	**Add** (required for anonymous calls)
Session Expires Mode	**Not Supported**
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)

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



Index	2
Extension Coders Group ID	Coders Group 2
Transcoding Mode	Only If Required
Allowed Media Types	
Allowed Coders Group ID	Coders Group 2
Allowed Video Coders Group ID	None
Allowed Coders Mode	Restriction and Pref
SBC Media Security Behavior	RTP
RFC 2833 Behavior	As Is
Alternative DTMF Method	As Is
P-Asserted-Identity	Add
Diversion Mode	As Is
History-Info Mode	As Is
Fax Coders Group ID	None
Fax Behavior	As Is
Fax Offer Mode	All coders
Fax Answer Mode	Single coder
PRACK Mode	Transparent
Session Expires Mode	Not Supported
Remote Update Support	Supported
Remote re-INVITE	Supported
Remote Delayed Offer Support	Not Supported
Remote REFER Behavior	Handle Locally
Remote 3xx Behavior	Transparent
Remote Multiple 18x	Not Supported
Remote Early Media Response Type	Transparent
Remote Early Media	Supported
Enforce MKI Size	Don't enforce
Remote Early Media RTP Behavior	Immediate
Remote RFC 3960 Gateway Model Support	Not Supported
Remote Can Play Ringback	No
RFC 2833 DTMF Payload Type	0
User Registration Time	0
Reliable Held Tone Source	Yes
Play Held Tone	No
Remote Hold Format	Transparent
Remote Replaces Behavior	Transparent
SDP Ptime Answer	Remote Answer
Preferred PTime	0
Use Silence Suppression	Transparent
RTP Redundancy Behavior	AS IS
Play RBT To Transferee	No
RTCP Mode	Transparent
Jitter Compensation	Disable
Remote Renegotiate on Fax Detection	Don't Care

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 Netia 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 Netia SIP Trunk.

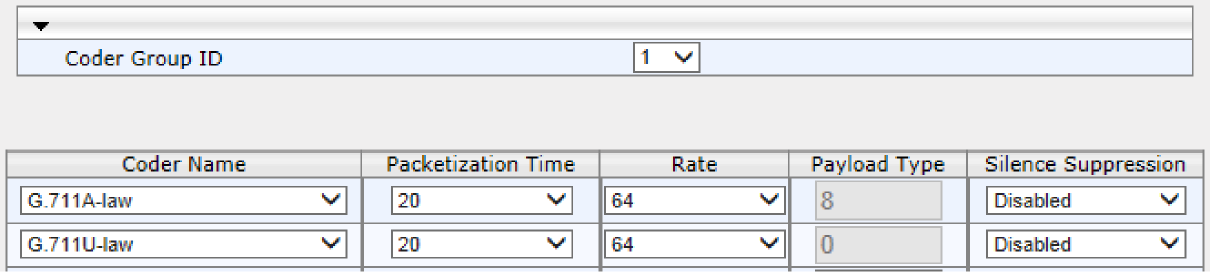
Note that the Coder Group ID for this entity was assigned to its corresponding IP Profile in the previous step (see Section 4.6 on page 48).

➢ **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 Lync Server 2013:

Parameter	Value
Coder Group ID	**1**
Coder Name	▪ **G.711 U-law** ▪ **G.711 A-law**
Silence Suppression	**Disable** (for both coders)

Figure 4-20: Configuring Coder Group for Lync Server 2013



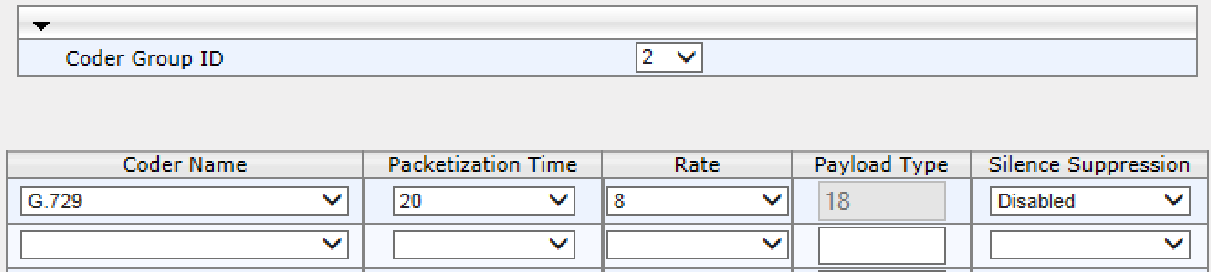
Coder Group ID: 1

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

3. Configure a Coder Group for Netia SIP Trunk:

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

Figure 4-21: Configuring Coder Group for Netia SIP Trunk



Coder Group ID: 2

Coder Name	Packetization Time	Rate	Payload Type	Silence Suppression
G.729	20	8	18	Disabled

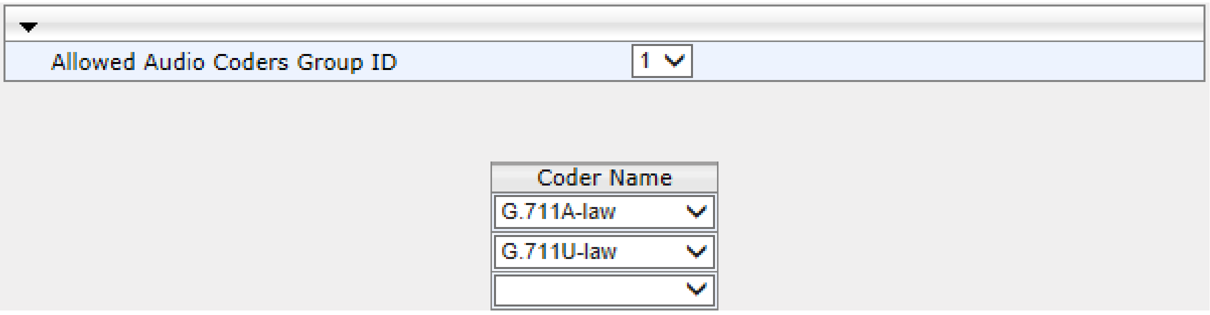
The procedure below describes how to configure an Allowed Coders Group to ensure that voice sent to the Netia SIP Trunk uses the G.729 coder whenever possible. Note that this Allowed Coders Group ID was assigned to the IP Profile belonging to the Netia SIP Trunk in the previous step (see Section 4.6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

➢ **To set a preferred coders:**

1. Open the Allowed Coders Group page (**Configuration** tab > **VoIP** menu > **SBC** > **Allowed Coders Group**).
2. Configure an Allowed Coders for Lync Server 2013:

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

Figure 4-22: Configuring Allowed Coders Group for Lync Server 2013



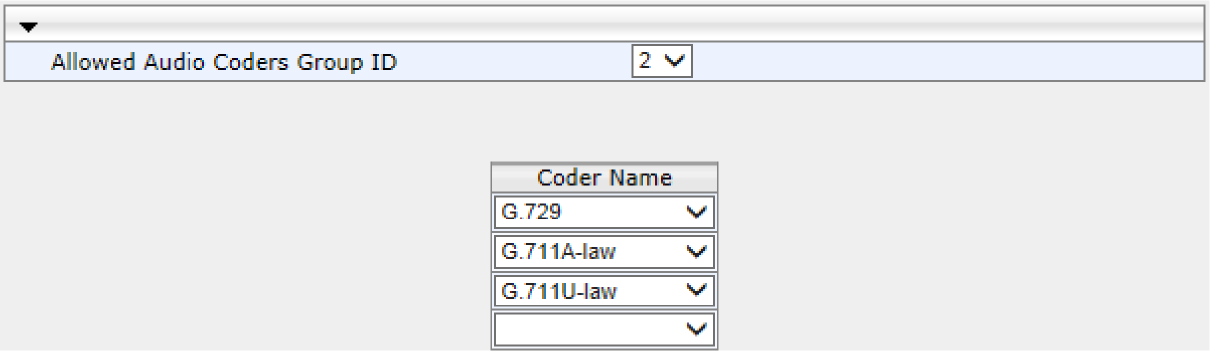
Allowed Audio Coders Group ID: 1

Coder Name
G.711A-law
G.711U-law

3. Configure an Allowed Coder for Netia SIP Trunk:

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

Figure 4-23: Configuring Allowed Coders Group for Netia SIP Trunk



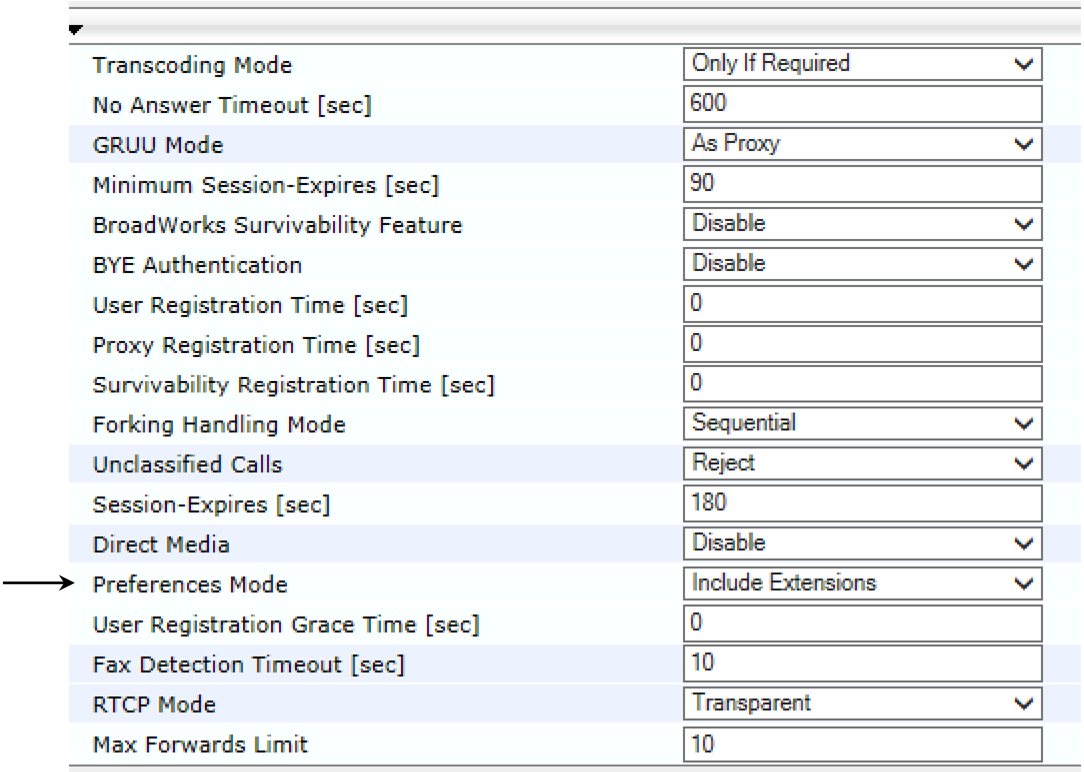
Allowed Audio Coders Group ID: 2

Coder Name
G.729
G.711A-law
G.711U-law

Note: Please consider, that in the current configuration example, when E-SBC performs transcoding on each call, more resources are used – i.e., 2 DSPs per each call.

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

Figure 4-24: SBC Preferences Mode



Transcoding Mode	Only If Required
No Answer Timeout [sec]	600
GRUU Mode	As Proxy
Minimum Session-Expires [sec]	90
BroadWorks Survivability Feature	Disable
BYE Authentication	Disable
User Registration Time [sec]	0
Proxy Registration Time [sec]	0
Survivability Registration Time [sec]	0
Forking Handling Mode	Sequential
Unclassified Calls	Reject
Session-Expires [sec]	180
Direct Media	Disable
Preferences Mode	Include Extensions
User Registration Grace Time [sec]	0
Fax Detection Timeout [sec]	10
RTCP Mode	Transparent
Max Forwards Limit	10

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

6. 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.

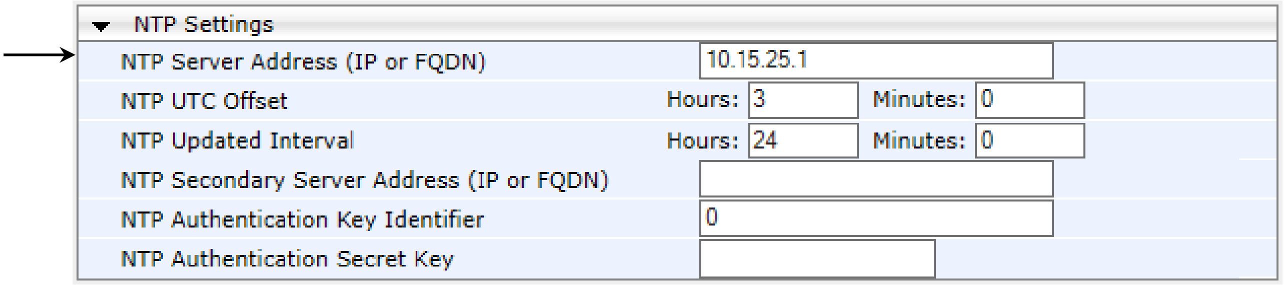
4.8.1 Step 8a: Configure the NTP Server Address

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

Figure 4-25: Configuring NTP Server Address



NTP Settings

NTP Server Address (IP or FQDN)	10.15.25.1			
NTP UTC Offset	Hours:	3	Minutes:	0
NTP Updated Interval	Hours:	24	Minutes:	0
NTP Secondary Server Address (IP or FQDN)				
NTP Authentication Key Identifier	0			
NTP Authentication Secret Key				

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.

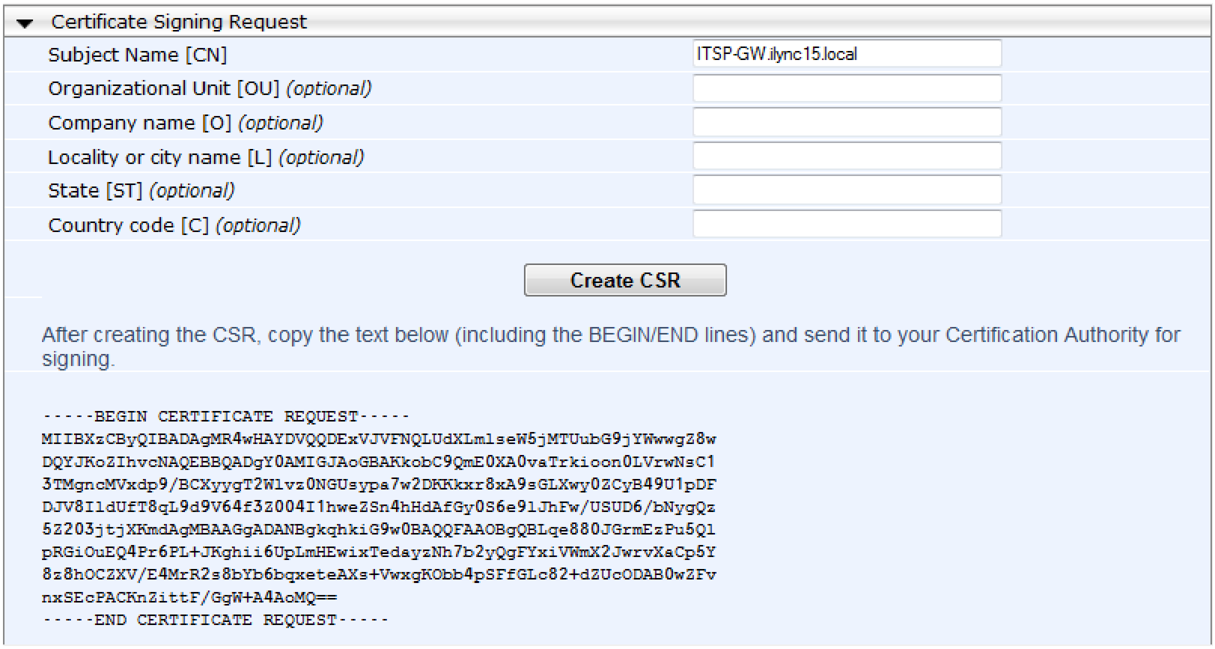
The procedure involves the following main steps:

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 4-26: Certificate Signing Request – Creating CSR



▼ Certificate Signing Request

Subject Name [CN]	ITSP-GW.ilync15.local
Organizational Unit [OU] *(optional)*	
Company name [O] *(optional)*	
Locality or city name [L] *(optional)*	
State [ST] *(optional)*	
Country code [C] *(optional)*	

After creating the CSR, copy the text below (including the BEGIN/END lines) and send it to your Certification Authority for signing.

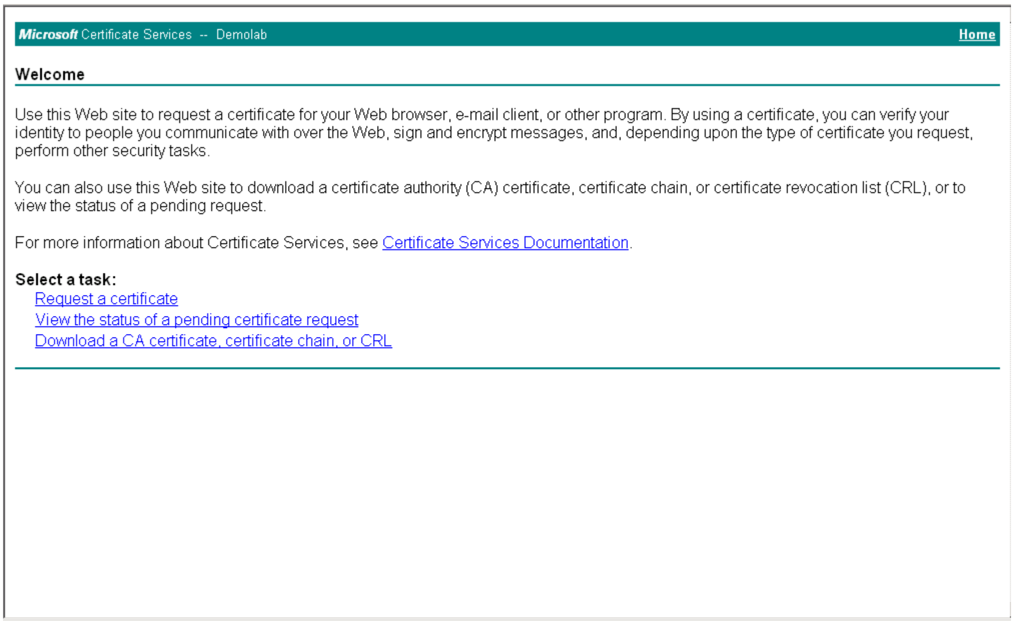
```
-----BEGIN CERTIFICATE REQUEST-----
MIIBXzCByQIBADAgMR4wHAYDVQQDExVJVFNQLUdXLmlseW5jMTUubG9jYWwwgZ8w
DQYJKoZIhvcNAQEBBQADgY0AMIGJAoGBAKkobC9QmE0XA0vaTrkioon0LVrwNsC1
3TMgncMVxdp9/BCXyygT2Wlvz0NGUsypa7w2DKKkxr8xA9sGLXwy0ZCyB49U1pDF
DJV8IldUfT8qL9d9V64f3Z004I1hweZSn4hHdAfGy0S6e9lJhFw/USUD6/bNygQz
5Z203jtjXKmdAgMBAAGgADANBgkqhkiG9w0BAQQFAAOBgQBLqe880JGrmEzPu5Ql
pRGiOuEQ4Pr6PL+JKghii6UpLmHEwixTedayzNh7b2yQgFYxiVWmX2JwrvXaCp5Y
8z8hOCZXV/E4MrR2s8bYb6bqxeteAXs+VwxgKObb4pSFfGLc82+dZUcODAB0wZFv
nxSEcPACKnZittF/GgW+A4AoMQ==
-----END CERTIFICATE REQUEST-----
```

Note: The value entered in the 'Subject Name [CN]' field must be identical to the gateway name configured in the Topology Builder for Lync Server 2013 (see Paragraph 5 in 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 4-27: Microsoft Certificate Services Web Page



Microsoft Certificate Services -- Demolab **Home**

Welcome

Use this Web site to request a certificate for your Web browser, e-mail client, or other program. By using a certificate, you can verify your identity to people you communicate with over the Web, sign and encrypt messages, and, depending upon the type of certificate you request, perform other security tasks.

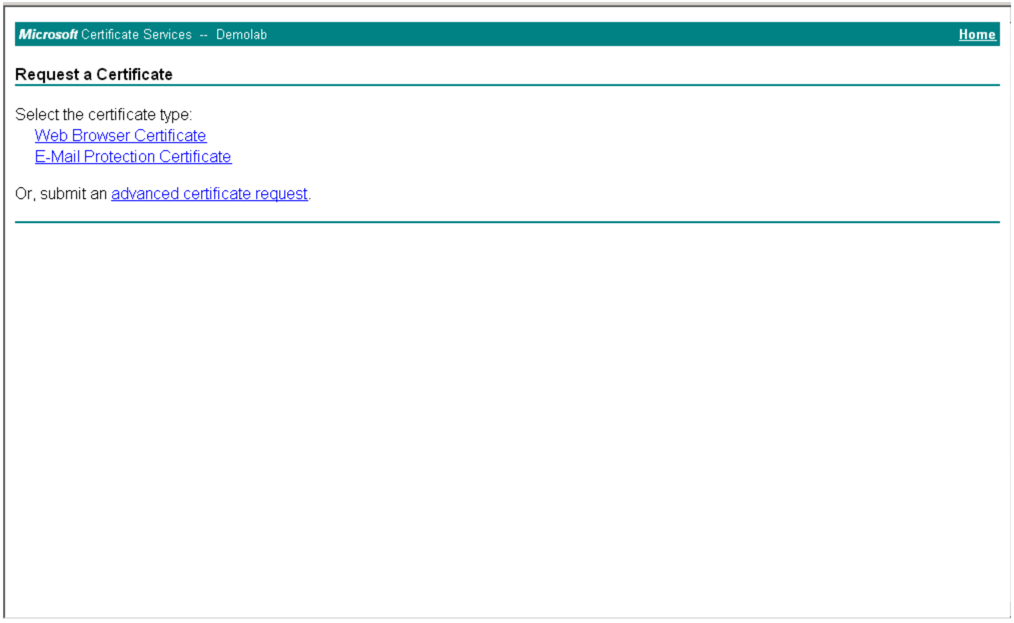
You can also use this Web site to download a certificate authority (CA) certificate, certificate chain, or certificate revocation list (CRL), or to view the status of a pending request.

For more information about Certificate Services, see Certificate Services Documentation.

Select a task:
- Request a certificate
- View the status of a pending certificate request
- Download a CA certificate, certificate chain, or CRL

7. Click **Request a certificate**.

Figure 4-28: Request a Certificate Page



Microsoft Certificate Services -- Demolab **Home**

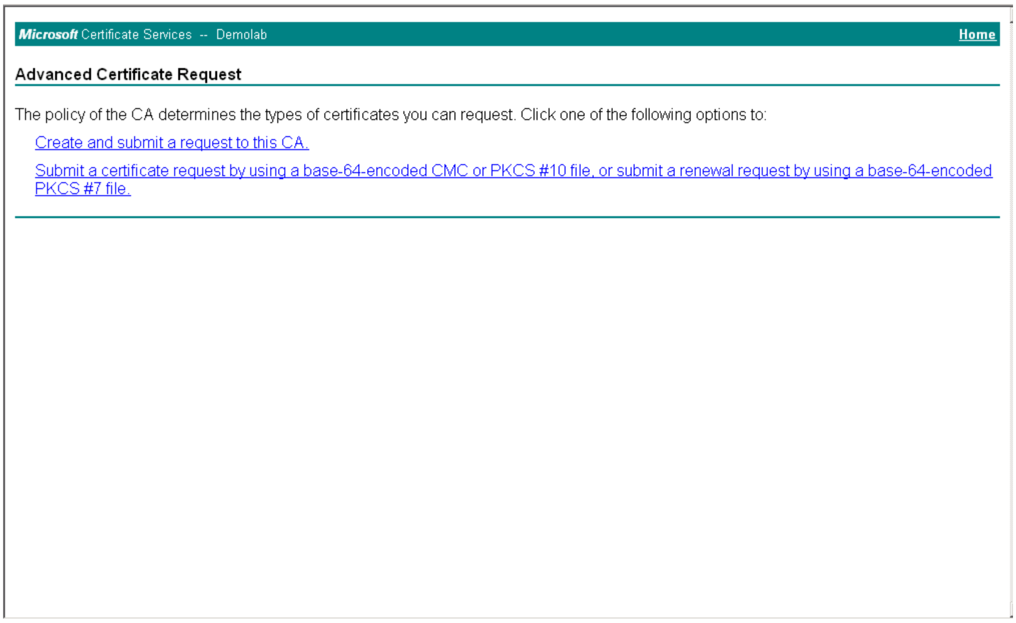
Request a Certificate

Select the certificate type:
- Web Browser Certificate
- E-Mail Protection Certificate

Or, submit an advanced certificate request.

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

Figure 4-29: Advanced Certificate Request Page



Microsoft Certificate Services -- Demolab

Advanced Certificate Request

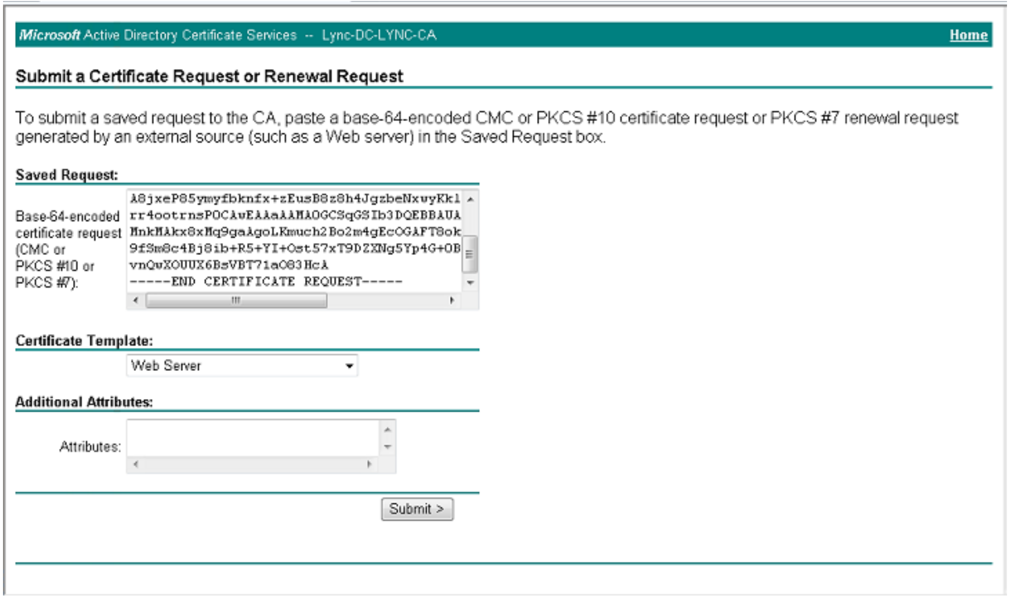
The policy of the CA determines the types of certificates you can request. Click one of the following options to:

Create and submit a request to this CA.

Submit a certificate request by using a base-64-encoded CMC or PKCS #10 file, or submit a renewal request by using a base-64-encoded PKCS #7 file.

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

Figure 4-30: Submit a Certificate Request or Renewal Request Page



Microsoft Active Directory Certificate Services -- Lync-DC-LYNC-CA

Submit a Certificate Request or Renewal Request

To submit a saved request to the CA, paste a base-64-encoded CMC or PKCS #10 certificate request or PKCS #7 renewal request generated by an external source (such as a Web server) in the Saved Request box.

Saved Request:

Base-64-encoded certificate request (CMC or PKCS #10 or PKCS #7):

```
A8jxeP85ymyfbknfx+zEusB8z8h4JgzbeNxuyKkl
rr4ootrnsPOCAuEAAaAAMAOGCSqGSIb3DQEBBAUA
MnkMAkx8xMq9gaAgoLKmuch2Bo2m4gEcOGAFT8ok
9fSm8c4Bj8ib+R5+YI+Ost57xT9DZXNg5Yp4G+OB
vnQuXOUUX6BsVBT71aO83HcA
-----END CERTIFICATE REQUEST-----
```

Certificate Template:

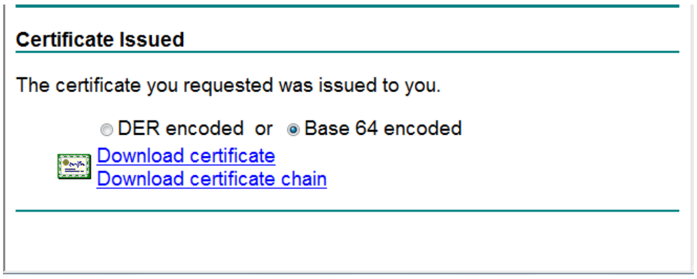
Web Server

Additional Attributes:

Attributes:

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 4-31: Certificate Issued Page



Certificate Issued

The certificate you requested was issued to you.

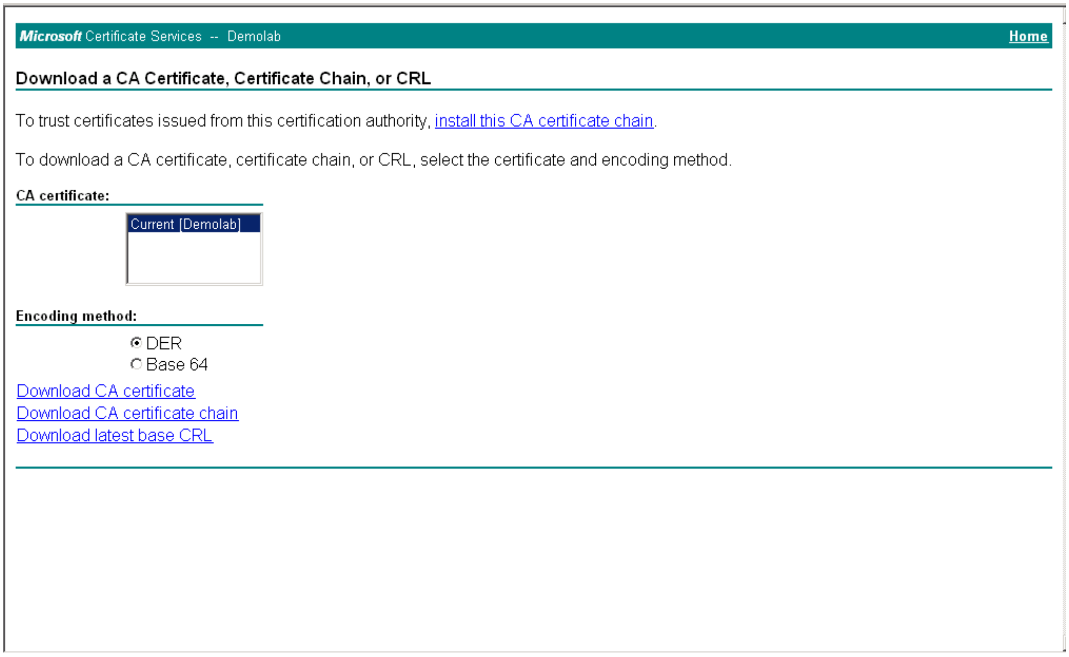
○ DER encoded or ● Base 64 encoded

Download certificate

Download certificate chain

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 4-32: Download a CA Certificate, Certificate Chain, or CRL Page



Microsoft Certificate Services -- Demolab Home

Download a CA Certificate, Certificate Chain, or CRL

To trust certificates issued from this certification authority, install this CA certificate chain.

To download a CA certificate, certificate chain, or CRL, select the certificate and encoding method.

CA certificate:

Current [Demolab]

Encoding method:

● DER

○ Base 64

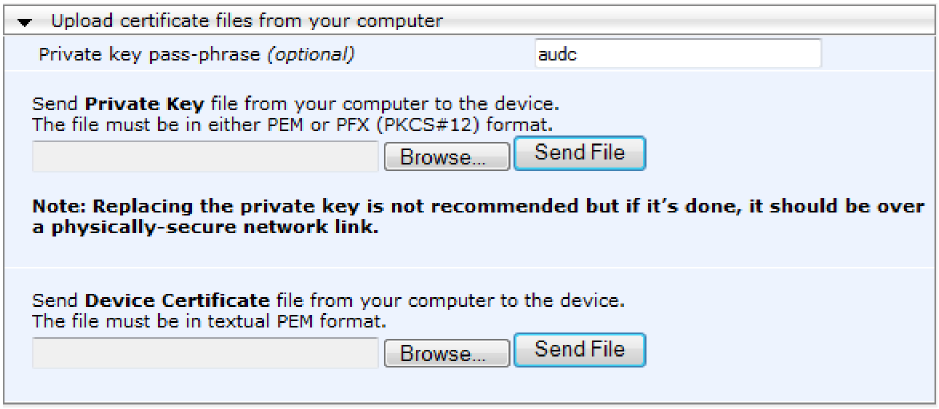
Download CA certificate

Download CA certificate chain

Download latest base CRL

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 4-33: Upload Device Certificate Files from your Computer Group



Upload certificate files from your computer

Private key pass-phrase *(optional)* audc

Send **Private Key** file from your computer to the device.
The file must be in either PEM or PFX (PKCS#12) format.

Note: Replacing the private key is not recommended but if it's done, it should be over a physically-secure network link.

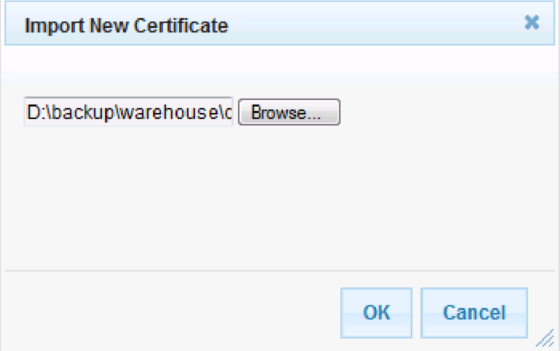
Send **Device Certificate** file from your computer to the device.
The file must be in textual PEM format.

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 4-34: Importing Root Certificate into Trusted Certificates Store



Import New Certificate

D:\backup\warehouse\c

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 86).

4.9 Step 9: Configure SRTP

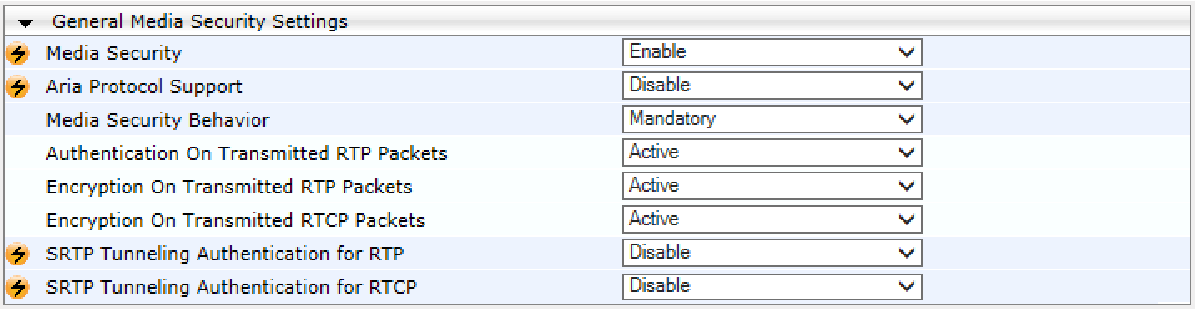
This step describes how to configure media security. If you configure the Microsoft Mediation Server to use SRTP, you need to configure the E-SBC to operate in the same manner. Note that SRTP was enabled for Lync Server 2013 when you configured an IP Profile for Lync Server 2013 (see Section 4.6 on page 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 4-35: Configuring SRTP



General Media Security Settings

Media Security	Enable
Aria Protocol Support	Disable
Media Security Behavior	Mandatory
Authentication On Transmitted RTP Packets	Active
Encryption On Transmitted RTP Packets	Active
Encryption On Transmitted RTCP Packets	Active
SRTP Tunneling Authentication for RTP	Disable
SRTP Tunneling Authentication for RTCP	Disable

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 86).

4.10 Step 10: Configure Maximum IP Media Channels

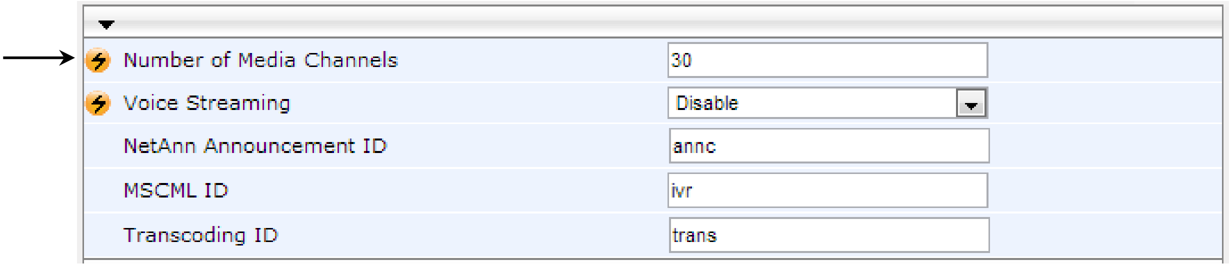
This step describes how to configure the maximum number of required IP media channels. The number of media channels represents the number of DSP channels that the E-SBC allocates to call sessions.

Note: This step is necessary **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 4-36: Configuring Number of IP Media Channels



Number of Media Channels	30
Voice Streaming	Disable
NetAnn Announcement ID	annc
MSCML ID	ivr
Transcoding ID	trans

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 86).

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 Netia 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 Netia SIP Trunk (WAN):

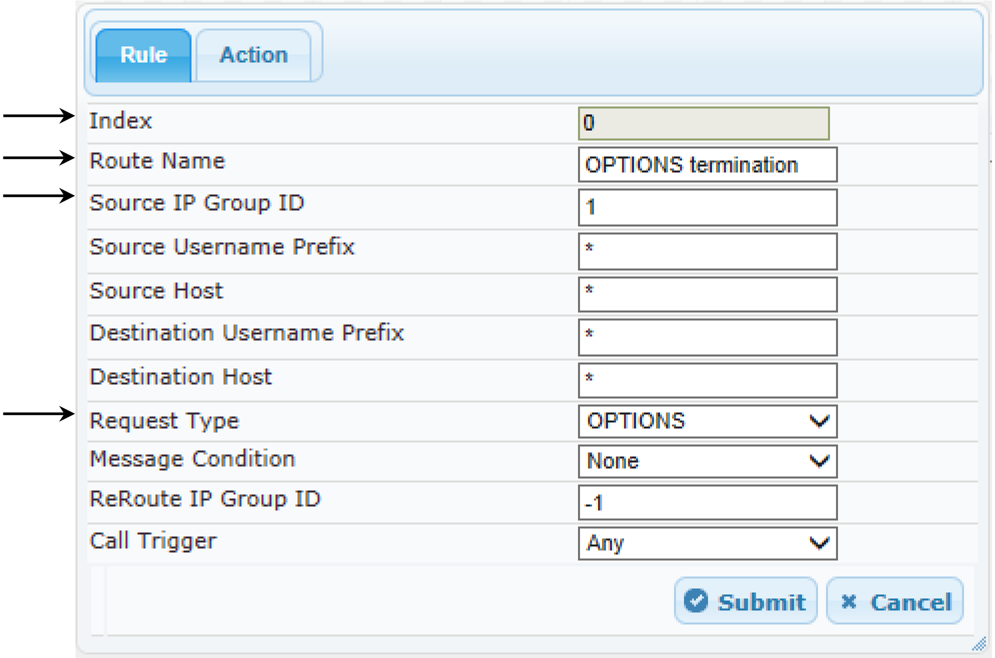
- Terminate SIP OPTIONS messages on the E-SBC that are received from the LAN
- Calls from Lync Server 2013 to Netia SIP Trunk
- Calls from Netia SIP Trunk to Lync Server 2013

➢ **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**
Destination Type	**Dest Address**
Destination Address	**internal**

Figure 4-37: Configuring IP-to-IP Routing Rule for Terminating SIP OPTIONS from LAN – Rule Tab

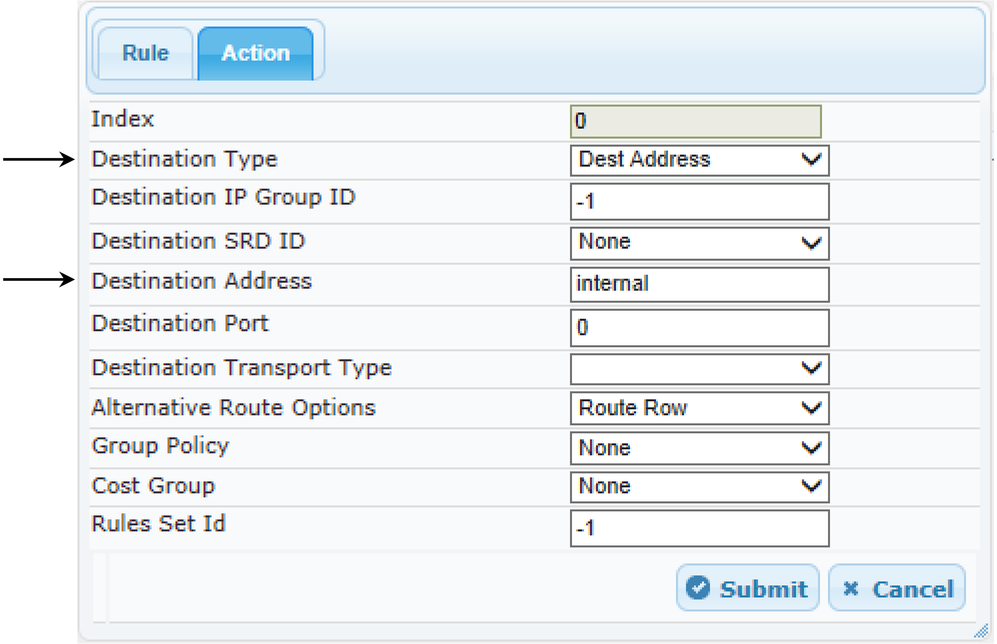


Index	0
Route Name	OPTIONS termination
Source IP Group ID	1
Source Username Prefix	*
Source Host	*
Destination Username Prefix	*
Destination Host	*
Request Type	OPTIONS
Message Condition	None
ReRoute IP Group ID	-1
Call Trigger	Any

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

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

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



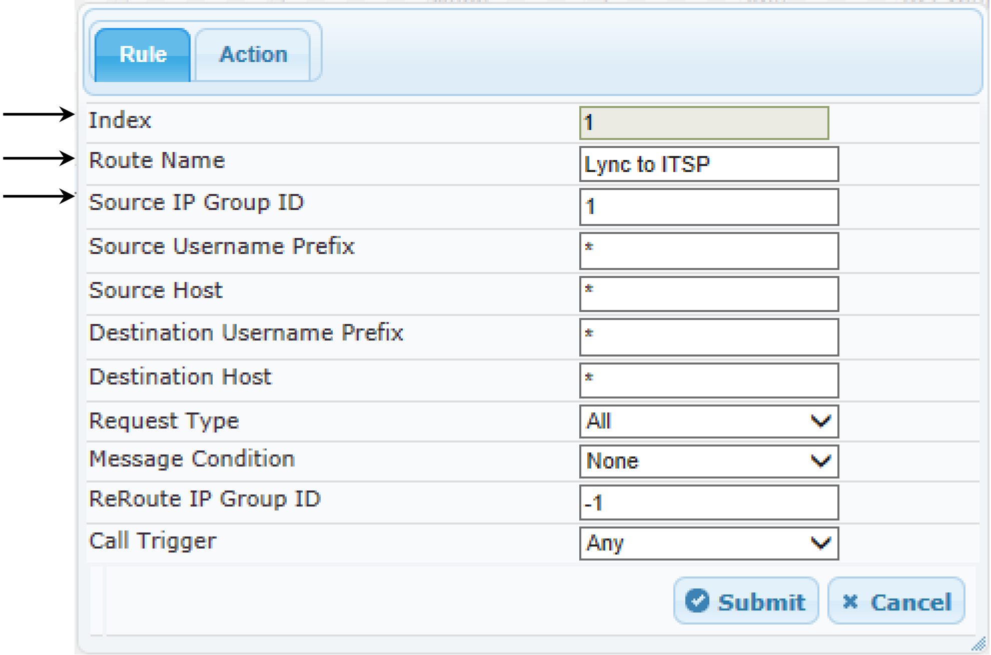
Index	0
Destination Type	Dest Address
Destination IP Group ID	-1
Destination SRD ID	None
Destination Address	internal
Destination Port	0
Destination Transport Type	
Alternative Route Options	Route Row
Group Policy	None
Cost Group	None
Rules Set Id	-1

6. Configure a rule to route calls from Lync Server 2013 to Netia SIP Trunk:
7. Click **Add**.

8. 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 4-39: Configuring IP-to-IP Routing Rule for Lync to ITSP – Rule tab

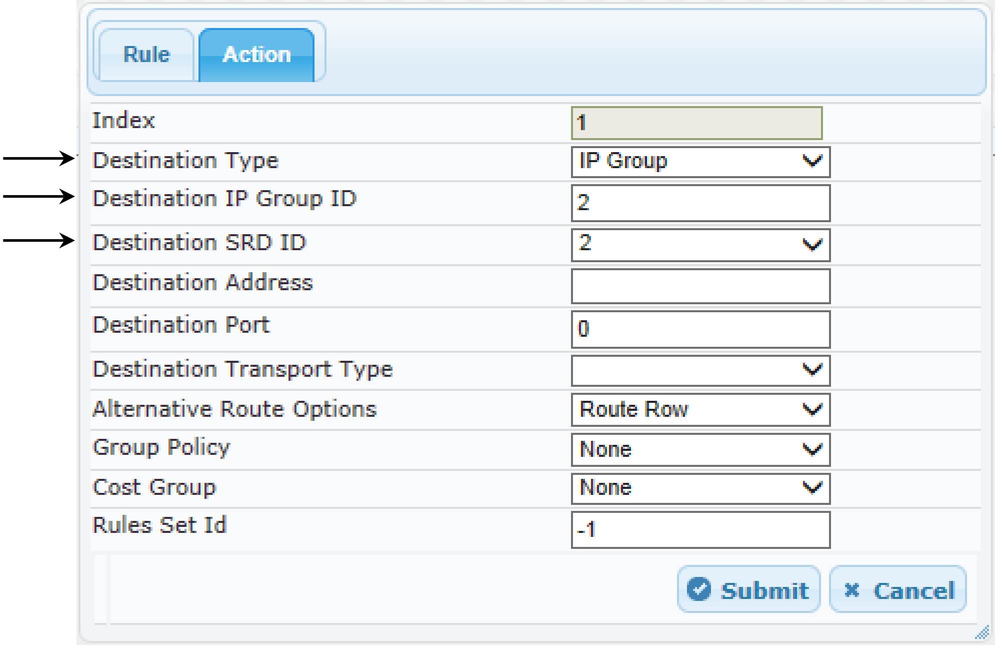


Index	1
Route Name	Lync to ITSP
Source IP Group ID	1
Source Username Prefix	*
Source Host	*
Destination Username Prefix	*
Destination Host	*
Request Type	All
Message Condition	None
ReRoute IP Group ID	-1
Call Trigger	Any

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

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

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

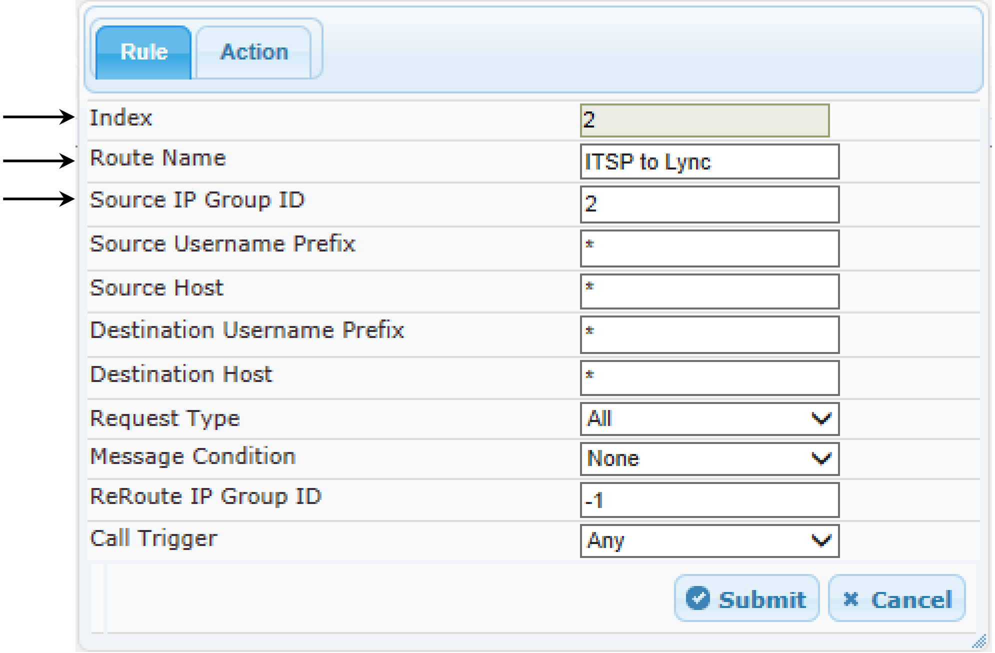


Index	1
Destination Type	IP Group
Destination IP Group ID	2
Destination SRD ID	2
Destination Address	
Destination Port	0
Destination Transport Type	
Alternative Route Options	Route Row
Group Policy	None
Cost Group	None
Rules Set Id	-1

10. Configure a rule to route calls from Netia SIP Trunk to Lync Server 2013:
11. Click **Add**.
12. 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 4-41: Configuring IP-to-IP Routing Rule for ITSP to Lync – Rule tab

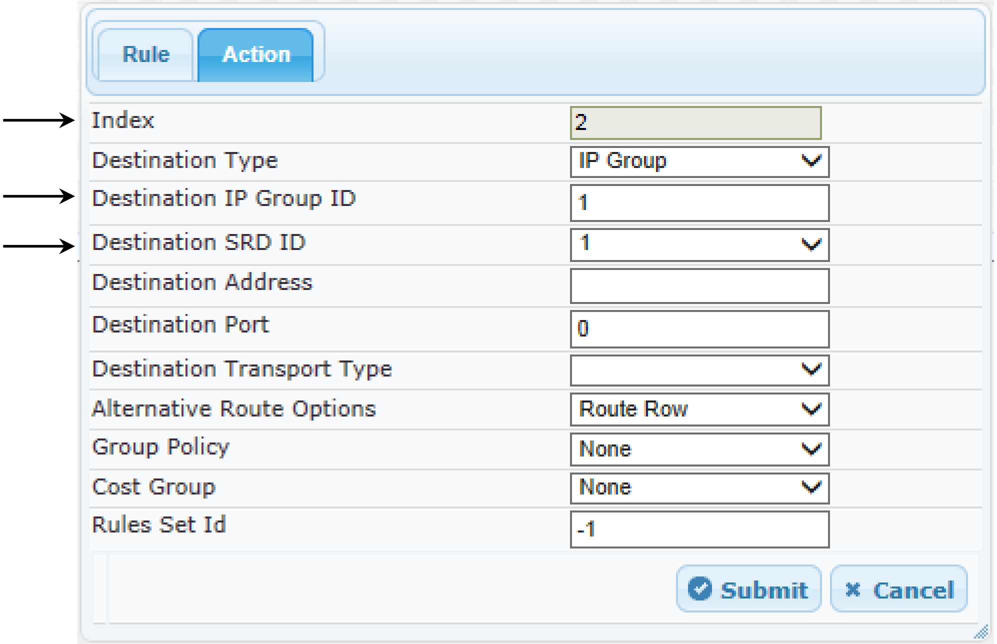


Index	2
Route Name	ITSP to Lync
Source IP Group ID	2
Source Username Prefix	*
Source Host	*
Destination Username Prefix	*
Destination Host	*
Request Type	All
Message Condition	None
ReRoute IP Group ID	-1
Call Trigger	Any

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

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

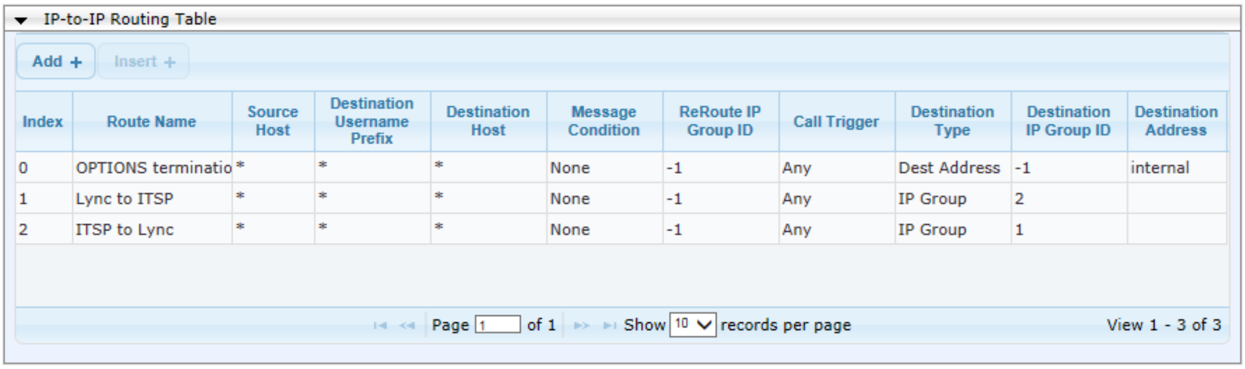
Figure 4-42: Configuring IP-to-IP Routing Rule for ITSP to Lync – Action tab



Index	2
Destination Type	IP Group
Destination IP Group ID	1
Destination SRD ID	1
Destination Address	
Destination Port	0
Destination Transport Type	
Alternative Route Options	Route Row
Group Policy	None
Cost Group	None
Rules Set Id	-1

The configured routing rules are shown in the figure below:

Figure 4-43: Configured IP-to-IP Routing Rules in IP-to-IP Routing Table



Index	Route Name	Source Host	Destination Username Prefix	Destination Host	Message Condition	ReRoute IP Group ID	Call Trigger	Destination Type	Destination IP Group ID	Destination Address
0	OPTIONS terminatio	*	*	*	None	-1	Any	Dest Address	-1	internal
1	Lync to ITSP	*	*	*	None	-1	Any	IP Group	2	
2	ITSP to Lync	*	*	*	None	-1	Any	IP Group	1	

Page 1 of 1 Show 10 records per page View 1 - 3 of 3

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 Netia SIP Trunk.

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

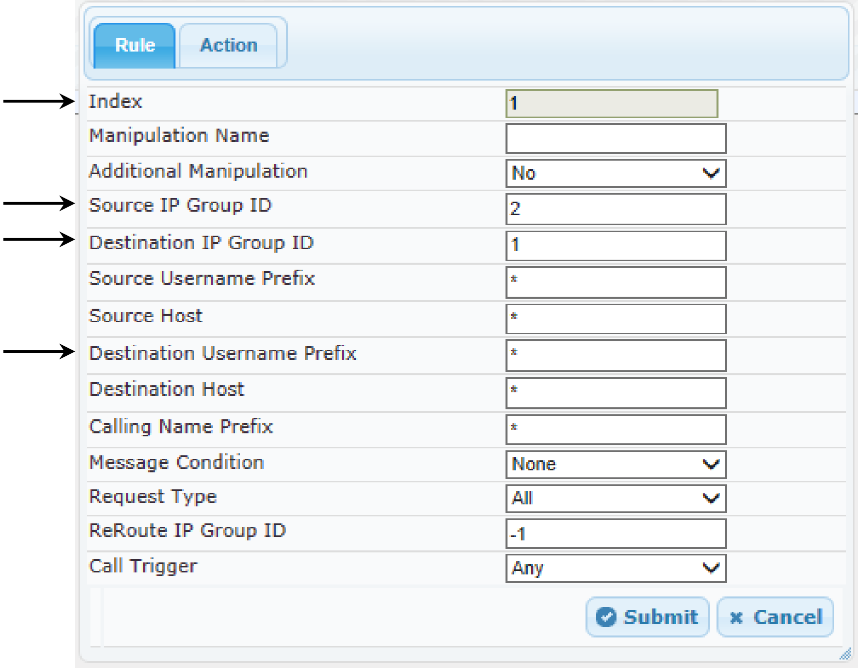
For this interoperability test topology, a manipulation is configured to add the "+" (plus sign) to the destination number for calls from IP Group 2 (Netia SIP Trunk) to IP Group 1 (i.e., Lync Server 2013) for any destination username prefix.

➢ **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	**2**
Destination IP Group ID	**1**
Destination Username Prefix	***** (asterisk sign)

Figure 4-44: Configuring IP-to-IP Outbound Manipulation Rule – Rule Tab

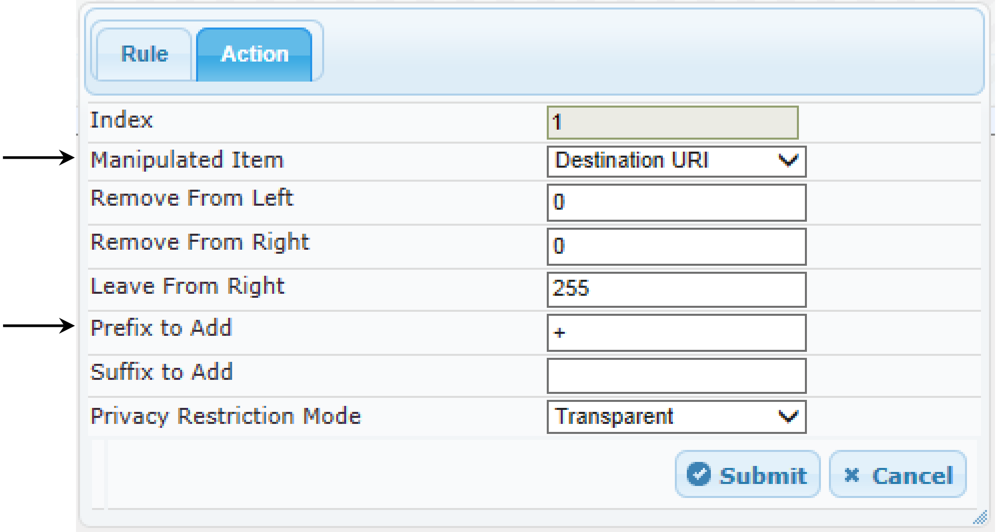


Index	1
Manipulation Name	
Additional Manipulation	No
Source IP Group ID	2
Destination IP Group ID	1
Source Username Prefix	*
Source Host	*
Destination Username Prefix	*
Destination Host	*
Calling Name Prefix	*
Message Condition	None
Request Type	All
ReRoute IP Group ID	-1
Call Trigger	Any

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

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

Figure 4-45: Configuring IP-to-IP Outbound Manipulation Rule - Action Tab

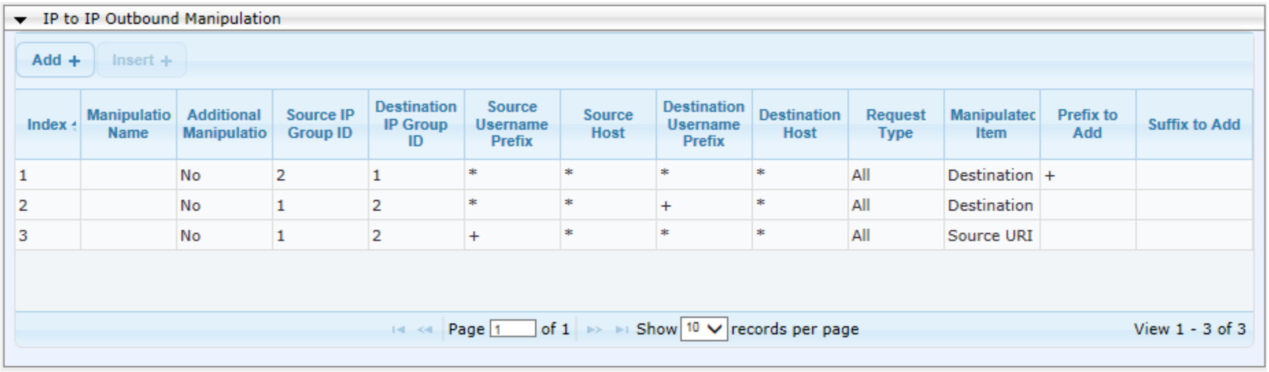


Index	1
Manipulated Item	Destination URI
Remove From Left	0
Remove From Right	0
Leave From Right	255
Prefix to Add	+
Suffix to Add	
Privacy Restriction Mode	Transparent

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., Netia SIP Trunk):

Figure 4-46: Example of Configured IP-to-IP Outbound Manipulation Rules



Index	Manipulatio Name	Additional Manipulatio	Source IP Group ID	Destination IP Group ID	Source Username Prefix	Source Host	Destination Username Prefix	Destination Host	Request Type	Manipulatec Item	Prefix to Add	Suffix to Add
1		No	2	1	*	*	*	*	All	Destination	+	
2		No	1	2	*	*	+	*	All	Destination		
3		No	1	2	+	*	*	*	All	Source URI		

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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).

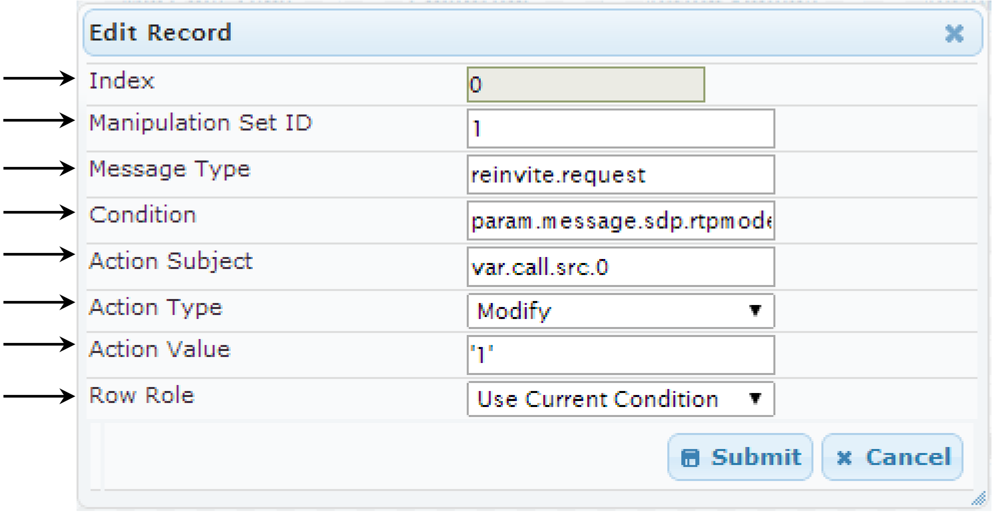
Once you have configured the SIP message manipulation rules, you need to assign them to the relevant IP Group (in the IP Group table) and determine whether they must be applied to inbound or outbound messages.

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

1. Open the Message Manipulations page (**Configuration** tab > **VoIP** menu > **SIP Definitions** > **Msg Policy & Manipulation** > **Message Manipulations**).
2. 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	**0**
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 4-47: Configuring SIP Message Manipulation Rule 0 (for Microsoft Lync)



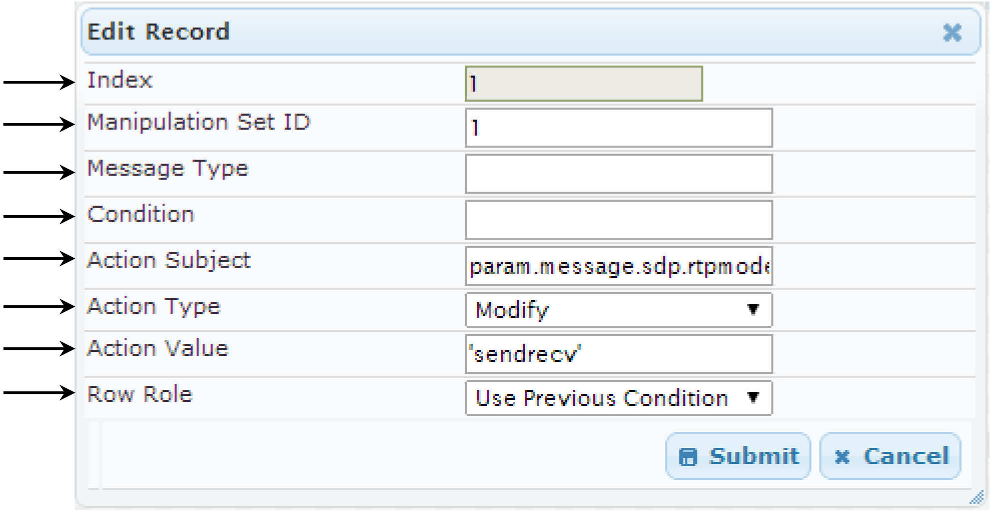
Edit Record

Index	0
Manipulation Set ID	1
Message Type	reinvite.request
Condition	param.message.sdp.rtpmode
Action Subject	var.call.src.0
Action Type	Modify ▼
Action Value	'1'
Row Role	Use Current Condition ▼

3. If the manipulation rule Index 0 (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	**1**
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 4-48: Configuring SIP Message Manipulation Rule 1 (for Microsoft Lync)



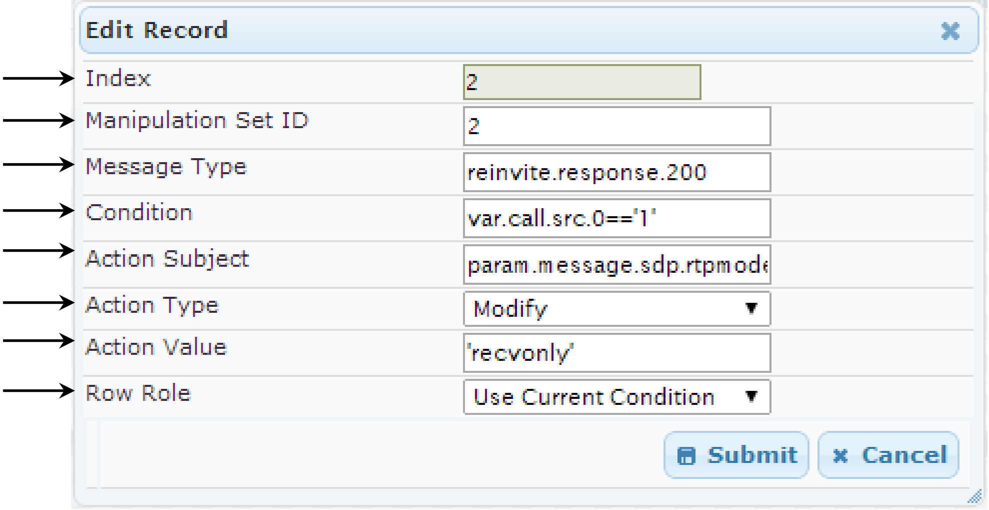
Edit Record

Index	1
Manipulation Set ID	1
Message Type	
Condition	
Action Subject	param.message.sdp.rtpmode
Action Type	Modify ▼
Action Value	'sendrecv'
Row Role	Use Previous Condition ▼

4. 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 Netia SIP Trunk to the Lync initiated Hold.

Parameter	Value
Index	**2**
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 4-49: Configuring SIP Message Manipulation Rule 2 (for Microsoft Lync)



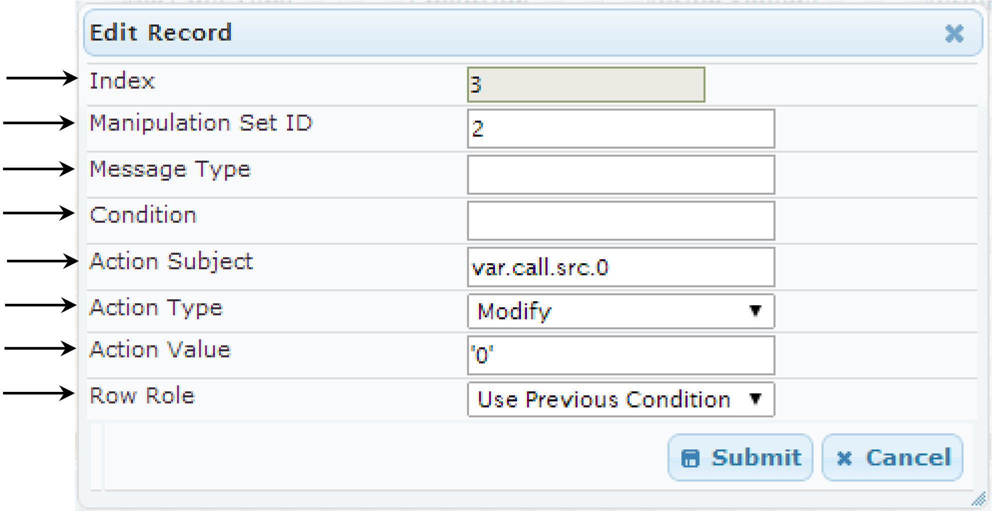
Edit Record

Index	2
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

5. 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"** in order to normalize the call processing state back. Lync now sends Music on Hold to the Netia SIP Trunk even without the Netia SIP Trunk knowing how to receive Music on Hold. The call is now truly on hold with Music on Hold.

Parameter	Value
Index	**3**
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 4-50: Configuring SIP Message Manipulation Rule 3 (for Microsoft Lync)

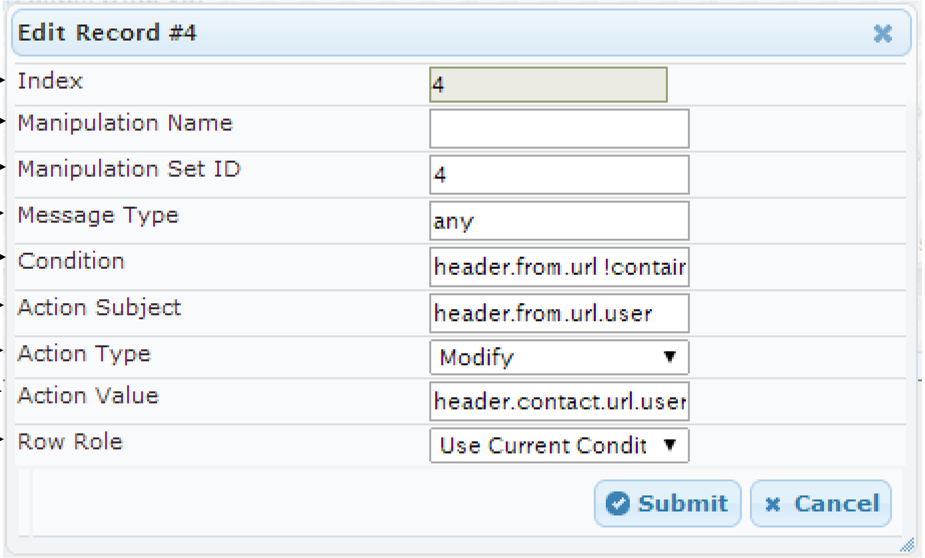


Index	3
Manipulation Set ID	2
Message Type	
Condition	
Action Subject	var.call.src.0
Action Type	Modify
Action Value	'0'
Row Role	Use Previous Condition

6. According to Netia's request, the pilot number should always be used in "From" header for all messages. Following rule replace user part of “From” header by user part from “Contact” header.

Parameter	Value
Index	**4**
Manipulation Set ID	**2**
Message Type	**any**
Condition	**header.from.url !contains 'anonymous'**
Action Subject	**header.from.url.user**
Action Type	**Modify**
Action Value	**header.contact.url.user**
Row Role	**Use Current Condition**

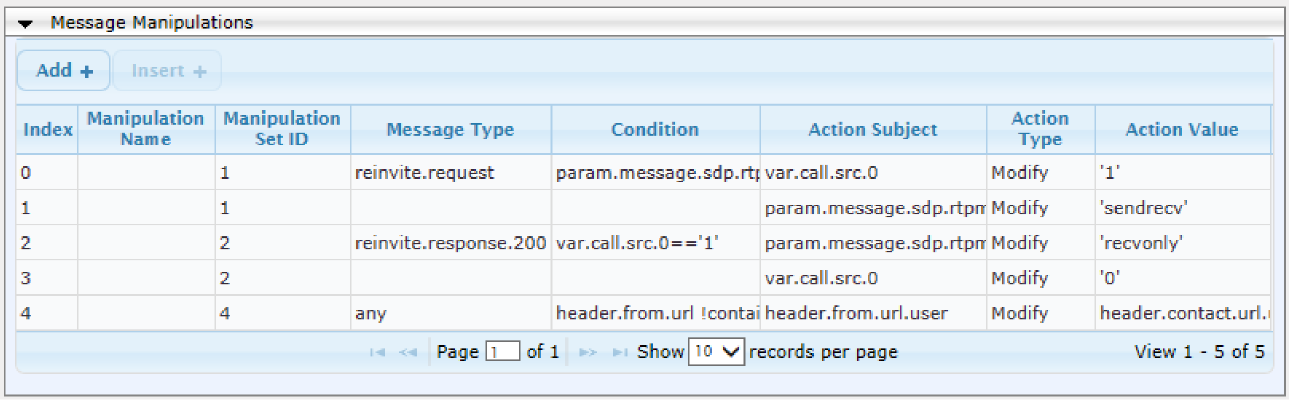
Figure 4-51: Configuring SIP Message Manipulation Rule 4 (for Netia SIP Trunk)



Edit Record #4

Index	4
Manipulation Name	
Manipulation Set ID	4
Message Type	any
Condition	header.from.url !contair
Action Subject	header.from.url.user
Action Type	Modify
Action Value	header.contact.url.user
Row Role	Use Current Condit

Figure 4-52: Configured SIP Message Manipulation Rules



Message Manipulations

Index	Manipulation Name	Manipulation Set ID	Message Type	Condition	Action Subject	Action Type	Action Value
0		1	reinvite.request	param.message.sdp.rtţ	var.call.src.0	Modify	'1'
1		1			param.message.sdp.rtpm	Modify	'sendrecv'
2		2	reinvite.response.200	var.call.src.0=='1'	param.message.sdp.rtpm	Modify	'recvonly'
3		2			var.call.src.0	Modify	'0'
4		4	any	header.from.url !contai	header.from.url.user	Modify	header.contact.url.(

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The table displayed 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 Netia 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 Netia SIP Trunk and Lync Server 2013. The specific items are needed to support Music On Hold (Rules 0-3). 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	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 Netia 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.
1	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”.	
2	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 Netia SIP Trunk to the Lync initiated Hold.	
3	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”** in order to normalize the call processing state back. Lync now sends Music on Hold to the Netia SIP Trunk even without the Netia SIP Trunk knowing how to receive MoH. The call is now truly on hold with MoH.	
4	This rule replace user part of “From” header by user part from “Contact” header.	According to Netia's request, the pilot number should always be used in "From" header for all messages.

7. 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**).

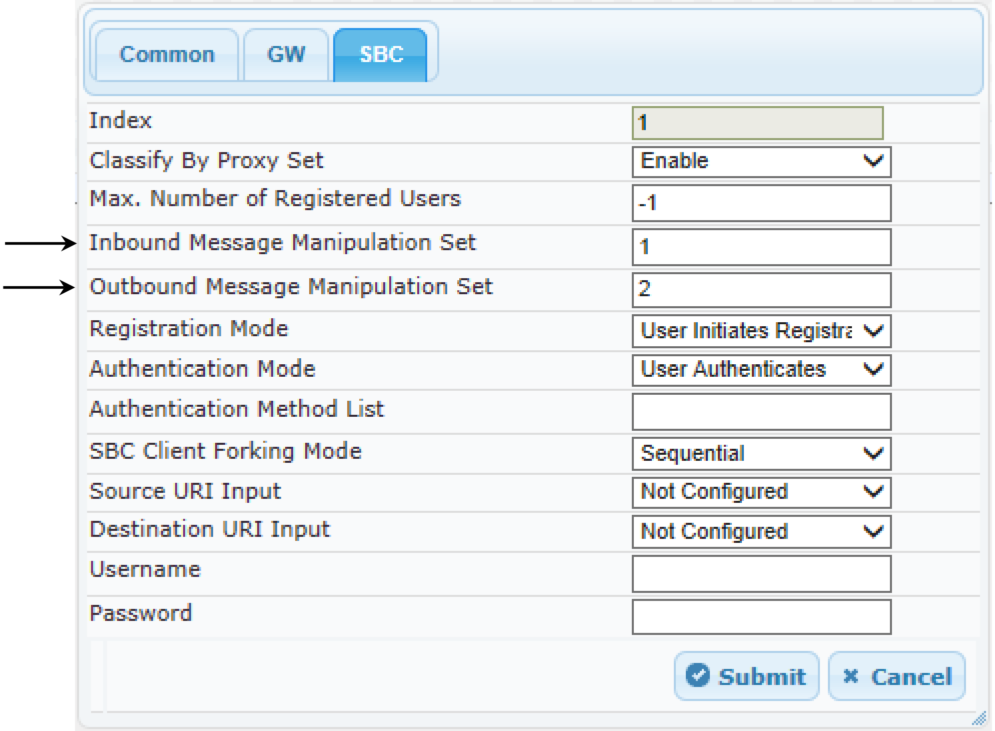
   b. Select the row of IP Group 1, and then click **Edit**.

   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 4-53: Assigning Manipulation Set to IP Group 1



Index	1
Classify By Proxy Set	Enable
Max. Number of Registered Users	-1
Inbound Message Manipulation Set	1
Outbound Message Manipulation Set	2
Registration Mode	User Initiates Registra
Authentication Mode	User Authenticates
Authentication Method List	
SBC Client Forking Mode	Sequential
Source URI Input	Not Configured
Destination URI Input	Not Configured
Username	
Password	

   f. Click **Submit**.

8. Assign Manipulation Set ID 4 to IP Group 2:

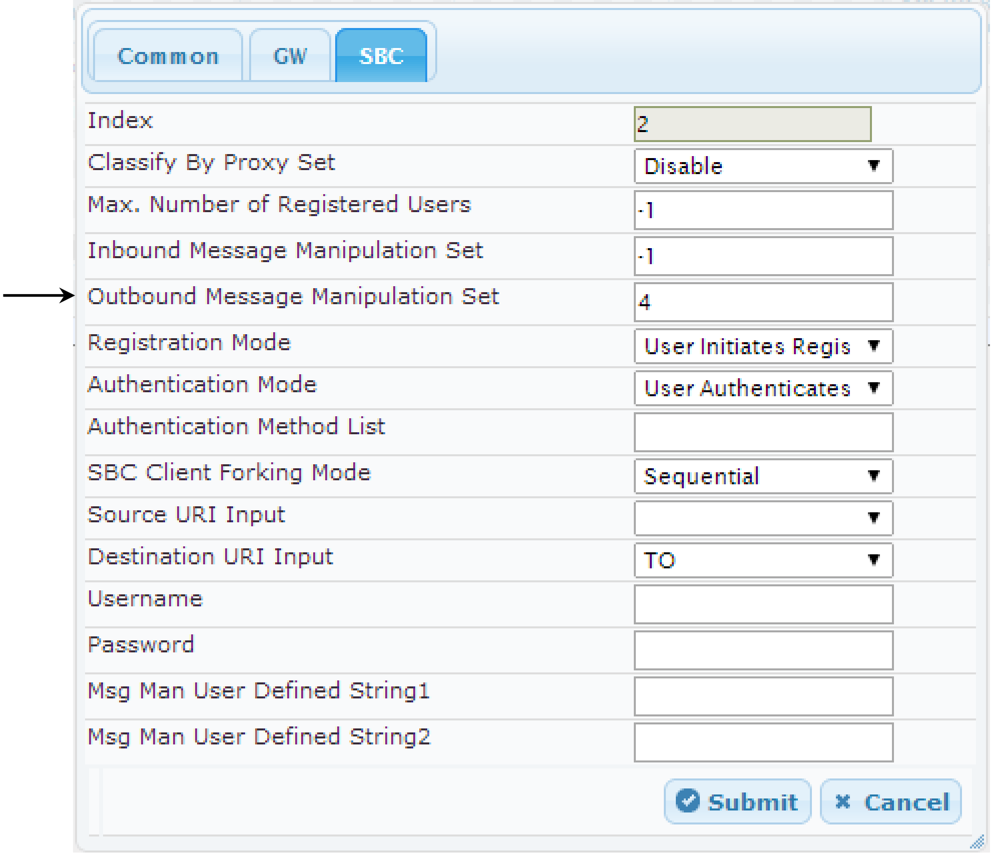
   a. Open the IP Group Table page (**Configuration** tab > **VoIP** menu > **VoIP Network** > **IP Group Table**).

   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 4-54: Assigning Manipulation Set 4 to IP Group 2



Index	2
Classify By Proxy Set	Disable
Max. Number of Registered Users	-1
Inbound Message Manipulation Set	-1
Outbound Message Manipulation Set	4
Registration Mode	User Initiates Regis
Authentication Mode	User Authenticates
Authentication Method List	
SBC Client Forking Mode	Sequential
Source URI Input	
Destination URI Input	TO
Username	
Password	
Msg Man User Defined String1	
Msg Man User Defined String2	

   e. Click **Submit**.

4.14 Step 14: Configure Registration Accounts

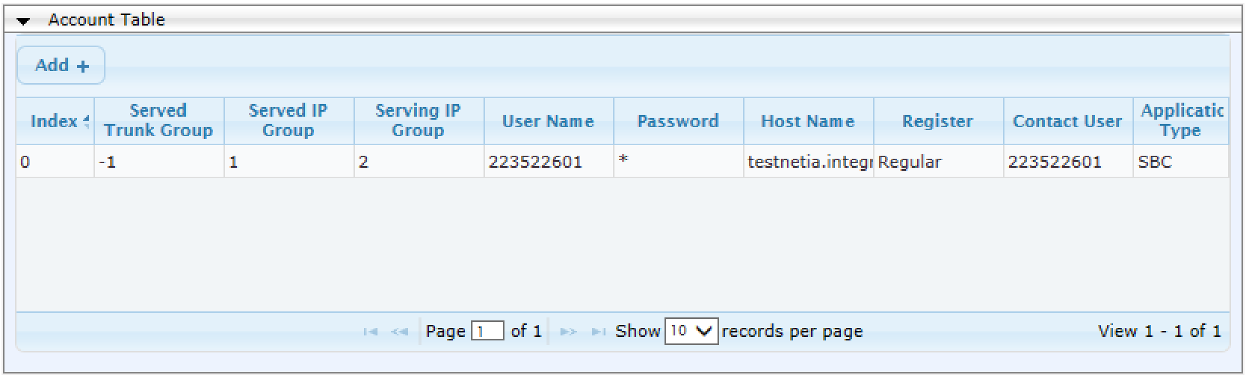
This step describes how to configure SIP registration accounts. This is required so that the E-SBC can register with the Netia SIP Trunk on behalf of Lync Server 2013. The Netia SIP Trunk requires registration and authentication to provide service.

In the interoperability test topology, the Served IP Group is Lync Server 2013 (IP Group 1) and the Serving IP Group is Netia 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 4-55: Configuring SIP Registration Account



Index	Served Trunk Group	Served IP Group	Serving IP Group	User Name	Password	Host Name	Register	Contact User	Applicatic Type
0	-1	1	2	223522601	*	testnetia.integr	Regular	223522601	SBC

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

Parameter	Value
Served IP Group	**1** (Lync Server 2013)
Serving IP Group	**2** (Netia SIP Trunk)
Username	As provided by Netia
Password	As provided by Netia
Host Name	**testnetia.integralnet.pl**
Register	**Regular**
Contact User	**223522601** (trunk main line)
Application Type	**SBC**

4. Click **Apply**.

4.15 Step 15: Miscellaneous Configuration

This section describes miscellaneous E-SBC configuration.

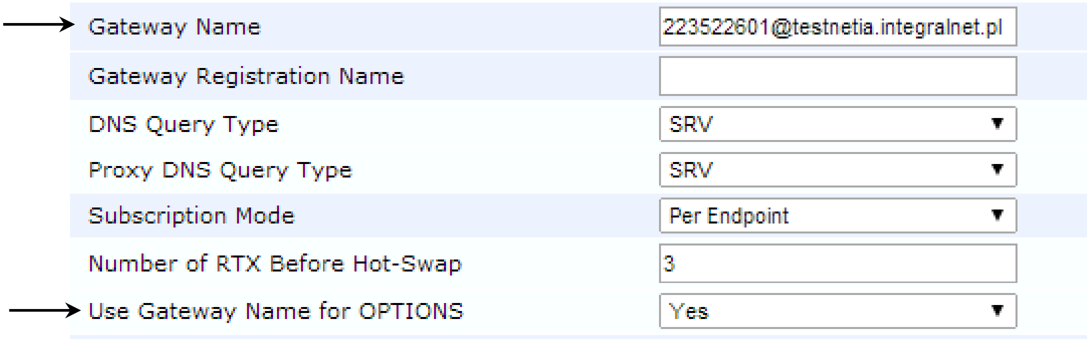
4.15.1 Step 15a: Configure E-SBC to send Domain Name in SIP OPTIONS Request

For the interoperability test topology with Netia SIP Trunk, it's necessary to send Domain Name in Request URI of SIP OPTIONS keep-alive messages. This step shows how to configure the E-SBC to do this.

➢ **To configure:**

1. Open the Proxy & Registration page (**Configuration** tab > **VoIP** menu > **SIP Definitions** > **Proxy & Registration**).
2. Configure the ‘Gateway Name’ parameter with appropriated information (For example, **223522601@testnetia.integralnet.pl**)
3. From the 'Use Gateway Name for OPTIONS' drop-down list, select **Yes**.

Figure 4-56: Configuring Forking Mode



Gateway Name	223522601@testnetia.integralnet.pl
Gateway Registration Name	
DNS Query Type	SRV ▼
Proxy DNS Query Type	SRV ▼
Subscription Mode	Per Endpoint ▼
Number of RTX Before Hot-Swap	3
Use Gateway Name for OPTIONS	Yes ▼

4. Click **Submit**.

4.15.2 Step 15b: Configure Classification Table

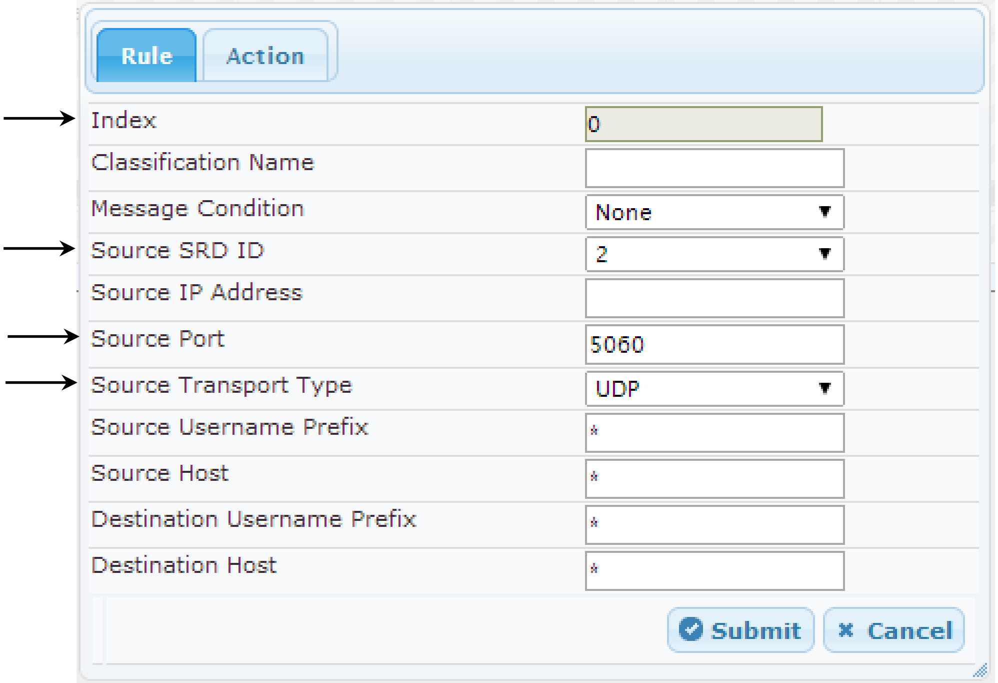
This step shows how to configure the E-SBC Classification Table. For security, it is highly recommended to disable the Classify by Proxy Set feature so that the device can use the Classification Table instead. This enables "strict" classification of incoming calls to IP Groups. For the interoperability test topology with Netia SIP Trunk, it's necessary to allow messages to be received from Netia. The Classification Table does this.

➢ **To configure Classification Table:**

1. Open the Classification Table page (**Configuration** tab > **VoIP** menu > **SBC** > **Routing SBC** > **Classification Table**).
2. Click **Add**.
3. Click the **Rule** tab, and then configure the parameters as follows:

Parameter	Value
Index	**0**
Source SRD ID	**2**
Source Port	**5060**
Source Transport Type	**UDP**

Figure 4-57: Classification Table Page – Rule Tab

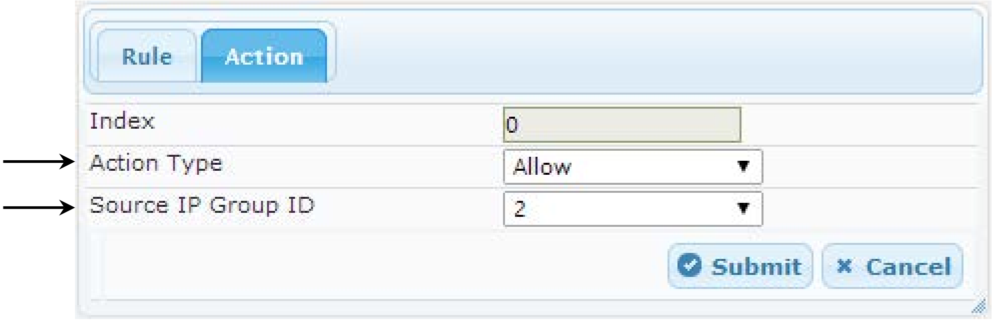


Index	0
Classification Name	
Message Condition	None
Source SRD ID	2
Source IP Address	
Source Port	5060
Source Transport Type	UDP
Source Username Prefix	*
Source Host	*
Destination Username Prefix	*
Destination Host	*

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

Parameter	Value
Action Type	**Allow**
Source IP Group ID	**2**

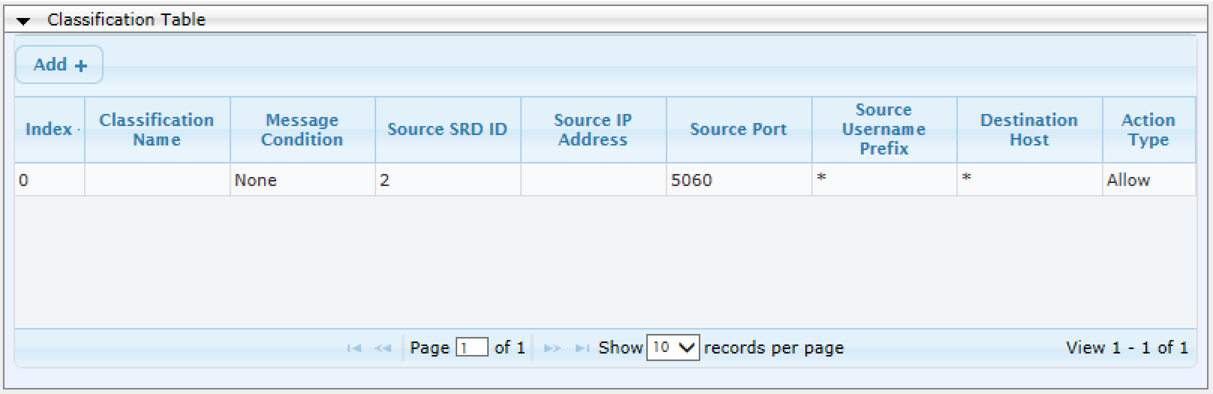
Figure 4-58: Classification Table Page – Action Tab



Index	0
Action Type	Allow
Source IP Group ID	2

5. Click **Submit**.

Figure 4-59: Example of Classification Table



Index	Classification Name	Message Condition	Source SRD ID	Source IP Address	Source Port	Source Username Prefix	Destination Host	Action Type
0		None	2		5060	*	*	Allow

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

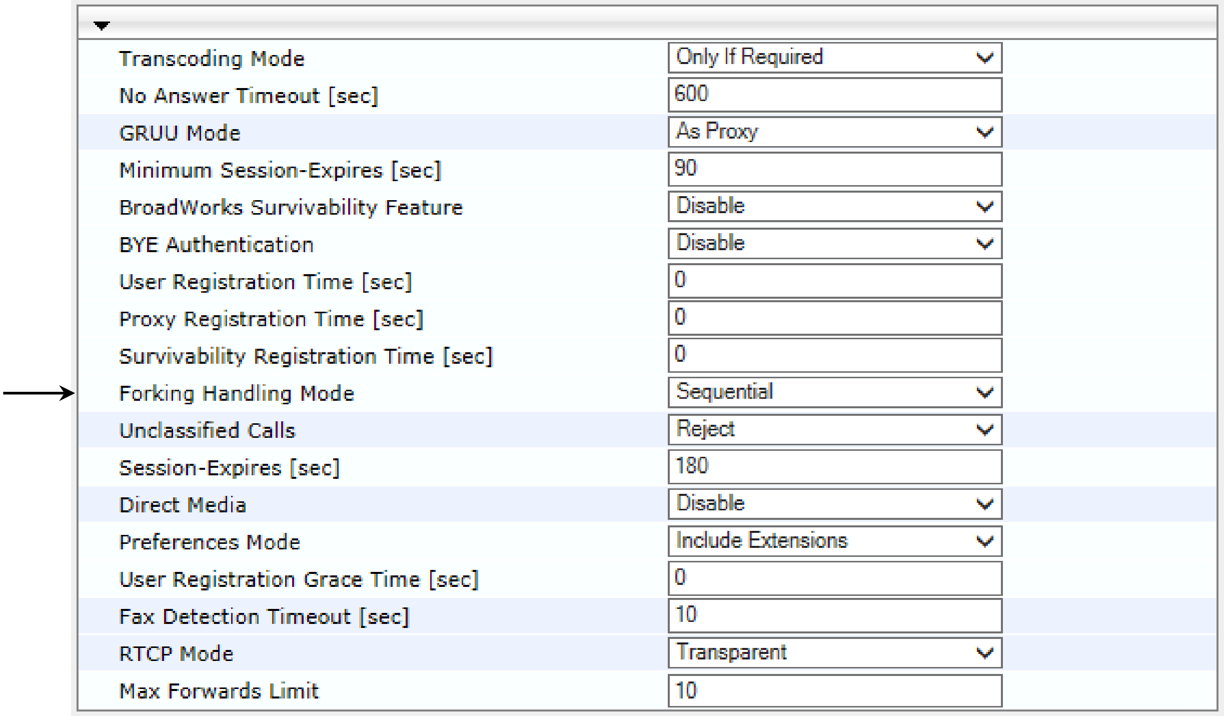
4.15.3 Step 15c: Configure Call Forking Mode

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 response 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 4-60: Configuring Forking Mode



Transcoding Mode	Only If Required
No Answer Timeout [sec]	600
GRUU Mode	As Proxy
Minimum Session-Expires [sec]	90
BroadWorks Survivability Feature	Disable
BYE Authentication	Disable
User Registration Time [sec]	0
Proxy Registration Time [sec]	0
Survivability Registration Time [sec]	0
Forking Handling Mode	Sequential
Unclassified Calls	Reject
Session-Expires [sec]	180
Direct Media	Disable
Preferences Mode	Include Extensions
User Registration Grace Time [sec]	0
Fax Detection Timeout [sec]	10
RTCP Mode	Transparent
Max Forwards Limit	10

3. Click **Submit**.

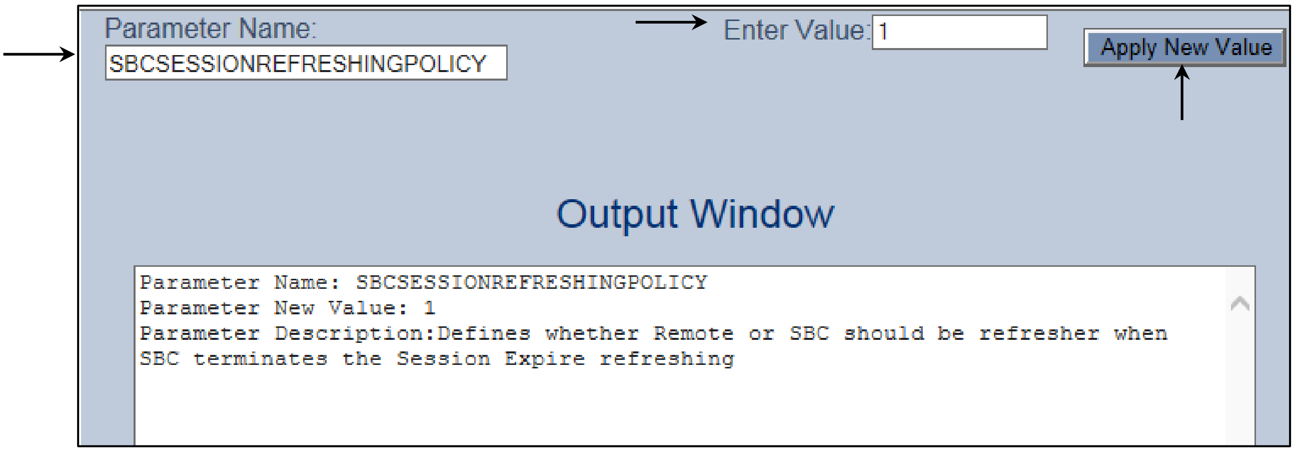
4.15.4 Step 15d: Configure SBC Session Refreshing Policy

This step shows how to configure the 'SBC Session Refreshing Policy' parameter. In some cases, Microsoft Lync does not perform a refresh of Session Timer even when it confirms that it will be refresher. To resolve this issue, the SBC is configured as Session Expire refresher.

➢ **To configure SBC Session Refreshing Policy:**

1. Open the Admin page: Append the case-sensitive suffix ‘AdminPage’ to the device's IP address in your Web browser's URL field (e.g., http://10.15.17.10/AdminPage).
2. In the left pane of the page that opens, click ***ini* Parameters**.

Figure 4-61: Configuring SBC Session Refreshing Policy in AdminPage



Parameter Name: SBCSESSIONREFRESHINGPOLICY

Enter Value: 1

```
Parameter Name: SBCSESSIONREFRESHINGPOLICY
Parameter New Value: 1
Parameter Description:Defines whether Remote or SBC should be refresher when
SBC terminates the Session Expire refreshing
```

3. Enter these values in the 'Parameter Name' and 'Enter Value' fields:

Parameter	Value
SBCSESSIONREFRESHINGPOLICY	**1** (enables SBC as refresher of Session Timer)

4. Click the **Apply New Value** button.

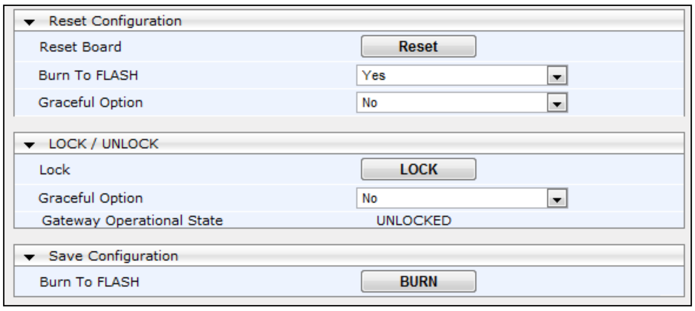
4.16 Step 16: Reset the E-SBC

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 4-62: Resetting the E-SBC



Reset Configuration

Reset Board: Reset

Burn To FLASH: Yes

Graceful Option: No

LOCK / UNLOCK

Lock: LOCK

Graceful Option: No

Gateway Operational State: UNLOCKED

Save Configuration

Burn To FLASH: BURN

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: 72
;Serial Number: 2265355
;Slot Number: 1
;Software Version: 6.80A.231
;DSP Software Version: 5014AE3_R => 680.22
;Board IP Address: 10.15.17.10
;Board Subnet Mask: 255.255.0.0
;Board Default Gateway: 10.15.0.1
;Ram size: 369M   Flash size: 64M   Core speed: 300Mhz
;Num of DSP Cores: 3  Num DSP Channels: 62
;Num of physical LAN ports: 12
;Profile: NONE
;;Key features:;Board Type: 72 ;IP Media: Conf VXML
VoicePromptAnnounc(H248.9) CALEA TrunkTesting POC ;System features: POE-
AF ;DSP Voice features: IpmDetector RTCP-XR AMRPolicyManagement ;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
;QOE features: VoiceQualityMonitoring MediaEnhancement ;Channel Type: RTP
DspCh=62 IPMediaDspCh=62 ;PSTN FALLBACK Supported ;E1Trunks=2 ;T1Trunks=2
;FXSPorts=4 ;FXOPorts=4 ;Security: IPSEC MediaEncryption StrongEncryption
EncryptControlProtocol ;Control Protocols: MGCP MEGACO H323 SIP TPNCP
SASurvivability SBC=60 MSFT CLI TRANSCODING=60 FEU=60 TestCall=60
SIPRec=60 CODER-TRANSCODING=60 EMS SBC-SIGNALING=60 SBC-MEDIA=60 ;Default
features:;Coders: G711 G726;

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


[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'
;PM_gwINVITEDialogs is hidden but has non-default value
;PM_gwSUBSCRIBEDialogs is hidden but has non-default value
;PM_gwSBCRegisteredUsers is hidden but has non-default value
;PM_gwSBCMediaLegs is hidden but has non-default value

[BSP Params]

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

[Analog Params]

FarEndDisconnectType = 7

[ControlProtocols Params]

AdminStateLockControl = 0

[MGCP Params]


[MEGACO Params]

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

[PSTN Params]


[SS7 Params]


[Voice Engine Params]

ENABLEMEDIASECURITY = 1
CallProgressTonesFilename = 'usa_tones_13.dat'

[WEB Params]

UseRProductName = 'Mediant 800 E-SBC'
WebLogoText = 'Netia'
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
GWDEBUGLEVEL = 5
;ISPRACKREQUIRED is hidden but has non-default value
SIPGATEWAYNAME = '223522601@testnetia.integralnet.pl'
USEGATEWAYNAMEFOROPTIONS = 1
;ENABLEPROXYSRVQUERY is hidden but has non-default value
;ENABLESRVQUERY is hidden but has non-default value
DNSQUERYTYPE = 1
PROXYDNSQUERYTYPE = 1
ENABLESBCAPPLICATION = 1
MSLDAPPRIMARYKEY = 'telephoneNumber'
SBCPREFERENCESMODE = 1
SBCFORKINGHANDLINGMODE = 1
ENERGYDETECTORCMD = 587202560
ANSWERDETECTORCMD = 10486144
SBCSESSIONREFRESHINGPOLICY = 1

[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 4 = "FE_5_1", 0, 1, 4, "User Port #4", "None", "  ";
PhysicalPortsTable 5 = "FE_5_2", 0, 1, 4, "User Port #5", "None", "  ";
PhysicalPortsTable 6 = "FE_5_3", 0, 1, 4, "User Port #6", "None", "  ";
PhysicalPortsTable 7 = "FE_5_4", 0, 1, 4, "User Port #7", "None", "  ";
PhysicalPortsTable 8 = "FE_5_5", 1, 1, 4, "User Port #8", "GROUP_5",
"Active";
PhysicalPortsTable 9 = "FE_5_6", 1, 1, 4, "User Port #9", "GROUP_5",
"Redundant";
PhysicalPortsTable 10 = "FE_5_7", 1, 1, 4, "User Port #10", "GROUP_6",
"Active";
PhysicalPortsTable 11 = "FE_5_8", 1, 1, 4, "User Port #11", "GROUP_6",
"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 4 = "GROUP_5", 2, "FE_5_5", "FE_5_6";
EtherGroupTable 5 = "GROUP_6", 2, "FE_5_7", "FE_5_8";
EtherGroupTable 6 = "GROUP_7", 0, "", "";
EtherGroupTable 7 = "GROUP_8", 0, "", "";
EtherGroupTable 8 = "GROUP_9", 0, "", "";
EtherGroupTable 9 = "GROUP_10", 0, "", "";
EtherGroupTable 10 = "GROUP_11", 0, "", "";
EtherGroupTable 11 = "GROUP_12", 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, 10.15.17.10, 16, 10.15.0.1, 1, "Voice",
10.15.25.1, 0.0.0.0, "vlan 1";
InterfaceTable 1 = 5, 10, 195.189.192.153, 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 1 = "SRDLan", "MRLan", 0, 0, -1, 1;
SRD 2 = "SRDWan", "MRWan", 0, 0, -1, 1;

[ \SRD ]


[ ProxyIp ]

FORMAT ProxyIp_Index = ProxyIp_IpAddress, ProxyIp_TransportType,
ProxyIp_ProxySetId;
ProxyIp 0 = "FE15.ilync15.local:5067", 2, 1;
ProxyIp 1 = "87.204.129.4", -1, 2;

[ \ProxyIp ]


[ IpProfile ]

FORMAT IpProfile_Index = IpProfile_ProfileName, IpProfile_IpPreference,
IpProfile_CodersGroupID, IpProfile_IsFaxUsed,
IpProfile_JitterBufMinDelay, IpProfile_JitterBufOptFactor,
IpProfile_IPDiffServ, IpProfile_SigIPDiffServ, IpProfile_SCE,
IpProfile_RTPRedundancyDepth, IpProfile_RemoteBaseUDPPort,
IpProfile_CNGmode, IpProfile_VxxTransportType, IpProfile_NSEMode,
IpProfile_IsDTMFUsed, IpProfile_PlayRBTone2IP,
IpProfile_EnableEarlyMedia, IpProfile_ProgressIndicator2IP,
IpProfile_EnableEchoCanceller, IpProfile_CopyDest2RedirectNumber,
IpProfile_MediaSecurityBehaviour, IpProfile_CallLimit,
IpProfile_DisconnectOnBrokenConnection, IpProfile_FirstTxDtmfOption,
IpProfile_SecondTxDtmfOption, IpProfile_RxDTMFOption,
IpProfile_EnableHold, IpProfile_InputGain, IpProfile_VoiceVolume,
IpProfile_AddIEInSetup, IpProfile_SBCExtensionCodersGroupID,
IpProfile_MediaIPVersionPreference, IpProfile_TranscodingMode,
IpProfile_SBCAllowedMediaTypes, IpProfile_SBCAllowedCodersGroupID,
IpProfile_SBCAllowedVideoCodersGroupID, IpProfile_SBCAllowedCodersMode,
IpProfile_SBCMediaSecurityBehaviour, IpProfile_SBCRFC2833Behavior,
IpProfile_SBCAlternativeDTMFMethod, IpProfile_SBCAssertIdentity,
IpProfile_AMDSensitivityParameterSuit, IpProfile_AMDSensitivityLevel,
IpProfile_AMDMaxGreetingTime, IpProfile_AMDMaxPostSilenceGreetingTime,
IpProfile_SBCDiversionMode, IpProfile_SBCHistoryInfoMode,
IpProfile_EnableQSIGTunneling, IpProfile_SBCFaxCodersGroupID,
IpProfile_SBCFaxBehavior, IpProfile_SBCFaxOfferMode,
IpProfile_SBCFaxAnswerMode, IpProfile_SbcPrackMode,
IpProfile_SBCSessionExpiresMode, IpProfile_SBCRemoteUpdateSupport,
IpProfile_SBCRemoteReinviteSupport,
IpProfile_SBCRemoteDelayedOfferSupport, IpProfile_SBCRemoteReferBehavior,
IpProfile_SBCRemote3xxBehavior, IpProfile_SBCRemoteMultiple18xSupport,
IpProfile_SBCRemoteEarlyMediaResponseType,
IpProfile_SBCRemoteEarlyMediaSupport, IpProfile_EnableSymmetricMKI,
IpProfile_MKISize, IpProfile_SBCEnforceMKISize,
IpProfile_SBCRemoteEarlyMediaRTP, IpProfile_SBCRemoteSupportsRFC3960,
IpProfile_SBCRemoteCanPlayRingback, IpProfile_EnableEarly183,
IpProfile_EarlyAnswerTimeout, IpProfile_SBC2833DTMFPayloadType,
IpProfile_SBCUserRegistrationTime, IpProfile_ResetSRTPStateUponRekey,
IpProfile_AmdMode, IpProfile_SBCReliableHeldToneSource,
IpProfile_GenerateSRTPKeys, IpProfile_SBCPlayHeldTone,
IpProfile_SBCRemoteHoldFormat, IpProfile_SBCRemoteReplacesBehavior,
IpProfile_SBCSDPPtimeAnswer, IpProfile_SBCPreferredPTime,
IpProfile_SBCUseSilenceSupp, IpProfile_SBCRTPRedundancyBehavior,
IpProfile_SBCPlayRBTToTransferee, IpProfile_SBCRTCPMode,
IpProfile_SBCJitterCompensation,
IpProfile_SBCRemoteRenegotiateOnFaxDetection,
IpProfile_JitterBufMaxDelay;
IpProfile 1 = "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, 3, 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, 1, 0,
0, 300;
IpProfile 2 = "Netia", 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, 2, 2, 2, 0, 3, 0, 0, 0,
1, 0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 1, 0, 0, 0, 0, 0, 0, 0, 0, 0, 0,
0, 0, 300;

[ \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, 1, 0, "-1", -1, -1, "";
ProxySet 2 = "Netia", 1, 60, 0, 1, 2, 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 1 = 0, "Lync", 1, "testnetia.integralnet.pl", "", 0, -1, -1, 0, -
1, 1, "MRLan", 1, 1, -1, 1, 2, 0, 0, "", 0, -1, -1, "", "", "$1$gQ==", 0,
"", "", "", 0, "", "";
IPGroup 2 = 0, "Netia", 2, "testnetia.integralnet.pl", "", 0, -1, -1, 0,
-1, 2, "MRWan", 0, 2, -1, -1, 4, 0, 0, "", 0, -1, 1, "", "", "$1$gQ==",
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, "223522601", "$1$dR0DHglBQxMY",
"testnetia.integralnet.pl", 1, "223522601", 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, "", "", 0, -1, 0, 0, "";
IP2IPRouting 2 = "ITSP to Lync", 2, "*", "*", "*", "*", 0, "", -1, 0, -1,
0, 1, "", "", 0, -1, 0, 0, "";

[ \IP2IPRouting ]


[ Classification ]

FORMAT Classification_Index = Classification_ClassificationName,
Classification_MessageCondition, Classification_SrcSRDID,
Classification_SrcAddress, Classification_SrcPort,
Classification_SrcTransportType, Classification_SrcUsernamePrefix,
Classification_SrcHost, Classification_DestUsernamePrefix,
Classification_DestHost, Classification_ActionType,
Classification_SrcIPGroupID;
Classification 0 = "", "", "2", "", 5060, 0, "*", "*", "*", "*", 1, "2";

[ \Classification ]


[ 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 1 = "Lync", "Voice", 2, 0, 0, 5067, 1, "", "", -1, 0, 500, -
1;
SIPInterface 2 = "Netia", "WANSP", 2, 5060, 0, 0, 2, "", "", -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 1 = "", 0, 2, 1, "*", "*", "*", "*", "*", "", 0, -
1, 0, 1, 0, 0, 255, "+", "", 0;
IPOutboundManipulation 2 = "", 0, 1, 2, "*", "*", "+", "*", "*", "", 0, -
1, 0, 1, 1, 0, 255, "00", "", 0;
IPOutboundManipulation 3 = "", 0, 1, 2, "+", "*", "*", "*", "*", "", 0, -
1, 0, 0, 1, 0, 255, "", "", 0;

[ \IPOutboundManipulation ]


[ CodersGroup1 ]
```

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

[ \CodersGroup1 ]


[ CodersGroup2 ]

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

[ \CodersGroup2 ]


[ AllowedCodersGroup1 ]

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

[ \AllowedCodersGroup1 ]


[ AllowedCodersGroup2 ]

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

[ \AllowedCodersGroup2 ]


[ MessageManipulations ]

FORMAT MessageManipulations_Index =
MessageManipulations_ManipulationName, MessageManipulations_ManSetID,
MessageManipulations_MessageType, MessageManipulations_Condition,
MessageManipulations_ActionSubject, MessageManipulations_ActionType,
MessageManipulations_ActionValue, MessageManipulations_RowRole;
MessageManipulations 0 = "", 1, "reinvite.request",
"param.message.sdp.rtpmode=='sendonly'", "var.call.src.0", 2, "'1'", 0;
MessageManipulations 1 = "", 1, "", "", "param.message.sdp.rtpmode", 2,
"'sendrecv'", 1;
MessageManipulations 2 = "", 2, "reinvite.response.200",
"var.call.src.0=='1'", "param.message.sdp.rtpmode", 2, "'recvonly'", 0;
MessageManipulations 3 = "", 2, "", "", "var.call.src.0", 2, "'0'", 1;
MessageManipulations 4 = "", 4, "any", "header.from.url !contains
'anonymous'", "header.from.url.user", 2, "header.contact.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 ]
```

Reader's Notes

Configuration Note

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