Configuration Note

AudioCodes Professional Services - Interoperability Lab

Microsoft® Skype for Business Server 2015 and BroadConnect Telecom's SIP Trunk using AudioCodes Mediant™ E-SBC

Version 7.0

Table of Contents

This page is intentionally left blank.

Notice

This document describes how to connect the Microsoft Skype for Business Server 2015 and BroadConnect SIP Trunk using AudioCodes Mediant E-SBC product series.

Information contained in this document is believed to be accurate and reliable at the time of printing. However, due to ongoing product improvements and revisions, AudioCodes cannot guarantee accuracy of printed material after the Date Published, nor can it accept responsibility for errors or omissions. Updates to this document and other documents as well as software files can be viewed by registered customers at http://www.audiocodes.com/downloads.

This document is subject to change without notice.

Date Published: July-16-2015

Trademarks

AudioCodes, AC, HD VoIP, HD VoIP Sounds Better, IPmedia, Mediant, MediaPack, What's Inside Matters, OSN, SmartTAP, VMAS, VoIPerfect, VoIPerfectHD, Your Gateway To VoIP, 3GX, VocaNOM and One Box 365 are trademarks or registered trademarks of AudioCodes Limited All other products or trademarks are property of their respective owners. Product specifications are subject to change without notice.

WEEE EU Directive

Pursuant to the WEEE EU Directive, electronic and electrical waste must not be disposed of with unsorted waste. Please contact your local recycling authority for disposal of this product.

Customer Support

Customer technical support and services are provided by AudioCodes or by an authorized AudioCodes Service Partner. For more information on how to buy technical support for AudioCodes products and for contact information, please visit our Web site at www.audiocodes.com/support.

Document Revision Record

LTRT	Description
12440	Initial document release for Version 7.0.

Documentation Feedback

AudioCodes continually strives to produce high quality documentation. If you have any comments (suggestions or errors) regarding this document, please fill out the Documentation Feedback form on our Web site at http://www.audiocodes.com/downloads.

1 Introduction

This Configuration Note describes how to set up AudioCodes Enterprise Session Border Controller (hereafter, referred to as *E-SBC*) for interworking between BroadConnect's SIP Trunk and Microsoft's Skype for Business Server 2015 environment.

1.1 Intended Audience

The document is intended for engineers, or AudioCodes and BroadConnect Partners who are responsible for installing and configuring BroadConnect's SIP Trunk and Microsoft's Skype for Business Server 2015 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.

This page is intentionally left blank.

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_7.00A.026.016
Protocol	▪ SIP/UDP (to the BroadConnect SIP Trunk) ▪ SIP/TCP or TLS (to the S4B FE Server)
Additional Notes	None

2.2 BroadConnect SIP Trunking Version

Table 2-2: BroadConnect Version

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

2.3 Microsoft Skype for Business Server 2015 Version

Table 2-3: Microsoft Skype for Business Server 2015 Version

Vendor	Microsoft
Model	Skype for Business
Software Version	Release 2015 6.0.9305.0
Protocol	SIP
Additional Notes	None

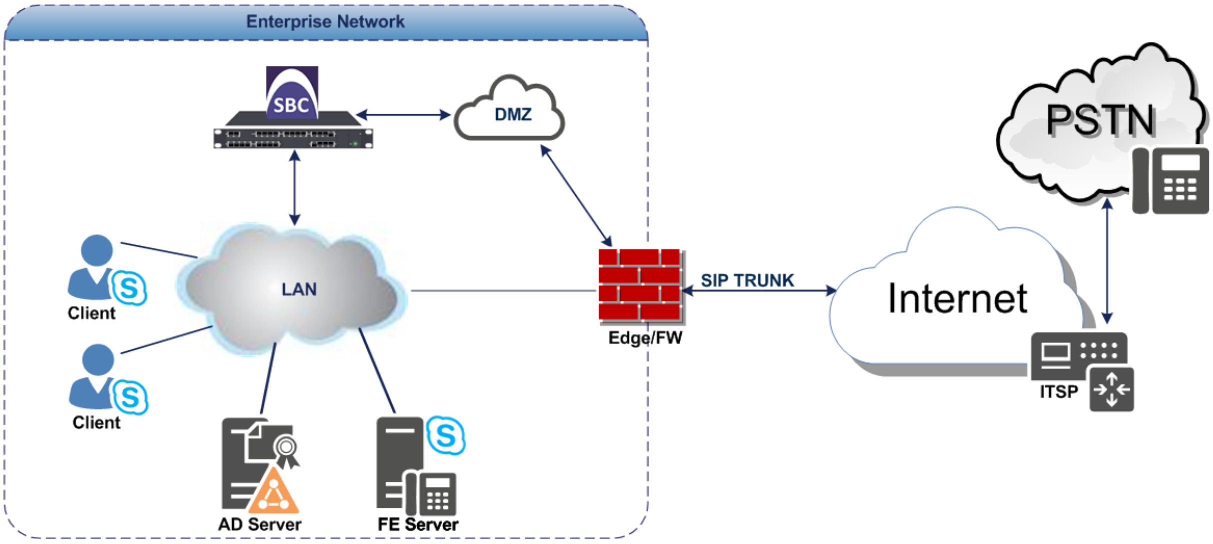
2.4 Interoperability Test Topology

Interoperability testing between AudioCodes' E-SBC and BroadConnect SIP Trunk with Skype for Business 2015 was performed using the following topology setup:

- Enterprise deployed with Microsoft Skype for Business Server 2015 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 BroadConnect's SIP Trunking service.
- AudioCodes E-SBC is implemented to interconnect between the enterprise LAN and the SIP Trunk.
  - **Session:** Real-time voice session using IP-based Session Initiation Protocol (SIP).
  - **Border:** IP-to-IP network border between Skype for Business Server 2015 network in the enterprise LAN and BroadConnect's SIP Trunk located in the public network.

The figure below illustrates this interoperability test topology:

Figure 2-1: Test Topology for Interoperability between E-SBC and Microsoft Skype for Business with BroadConnect 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 Skype for Business Server 2015 environment is located on the enterprise's LAN ▪ BroadConnect SIP Trunk is located on the WAN
Signaling Transcoding	▪ Microsoft Skype for Business Server 2015 operates with SIP-over-TLS transport type ▪ BroadConnect SIP Trunk operates with SIP-over-UDP transport type
Codecs Transcoding	▪ Microsoft Skype for Business Server 2015 supports G.711A-law and G.711U-law coders ▪ BroadConnect SIP Trunk supports G.711U-law and G.729 coder
Media Transcoding	▪ Microsoft Skype for Business Server 2015 operates with SRTP media type ▪ BroadConnect SIP Trunk operates with RTP media type

2.4.2 Known Limitations

The following limitation was observed during interoperability tests performed for AudioCodes' E-SBC interworking between Microsoft Skype for Business Server 2015 and BroadConnect's SIP Trunk:

- If the Microsoft Skype for Business Server 2015 sends one of the following error Responses:
  - 403 Forbidden
  - 406 Not Acceptable
  - 480 Temporarily Unavailable

  BroadConnect SIP Trunk still sends re-INVITEs and does not disconnect the call.

  To disconnect the call, a message manipulation rule is used to replace the above error response with the “486 Busy Here” response (see Section 4.14).

This page is intentionally left blank.

3 Configuring Skype for Business Server 2015

This section shows how to configure Microsoft Skype for Business Server 2015 to operate with AudioCodes' E-SBC.

> **Note:** Dial plans, voice policies, and PSTN usages are also necessary for enterprise voice deployment but they're beyond the scope of this document.

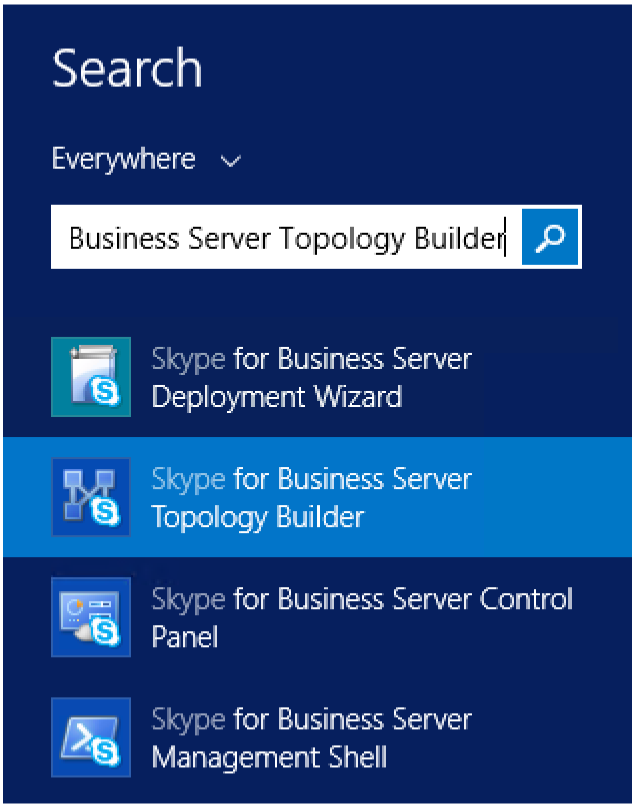
3.1 Configuring the E-SBC as an IP / PSTN Gateway

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 Skype for Business Server 2015 Topology Builder (Windows **Start** menu > search for **Skype for Business Server Topology Builder**), as shown below:

Figure 3-1: Starting the Skype for Business Server Topology Builder



Search

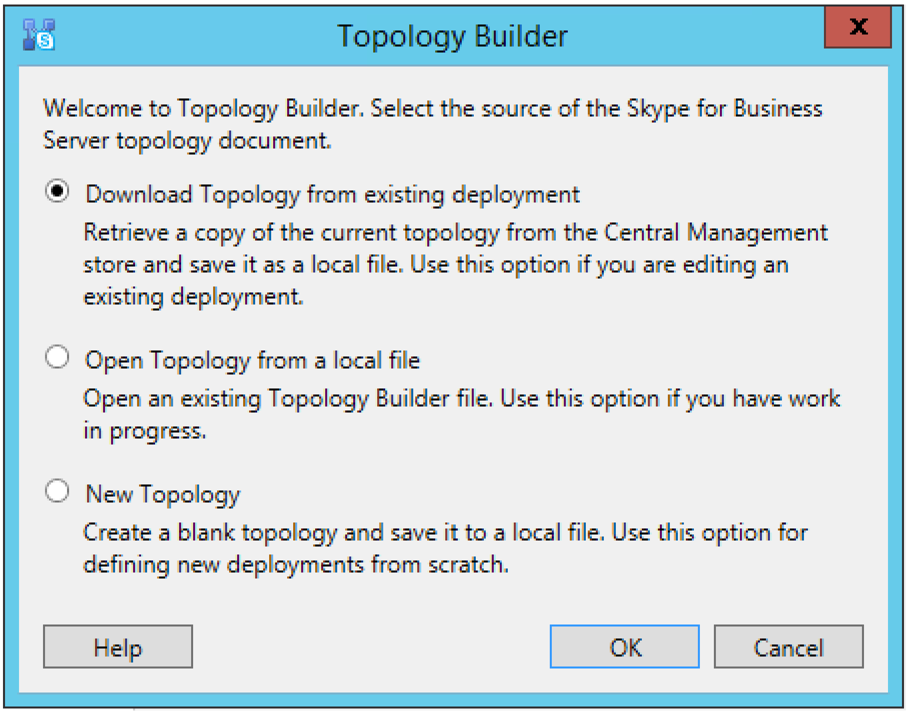
Everywhere

Business Server Topology Builder

Skype for Business Server Deployment Wizard
Skype for Business Server Topology Builder
Skype for Business Server Control Panel
Skype for Business Server Management Shell

The following is displayed:

Figure 3-2: Topology Builder



Topology Builder

Welcome to Topology Builder. Select the source of the Skype for Business Server topology document.

● Download Topology from existing deployment

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

○ Open Topology from a local file

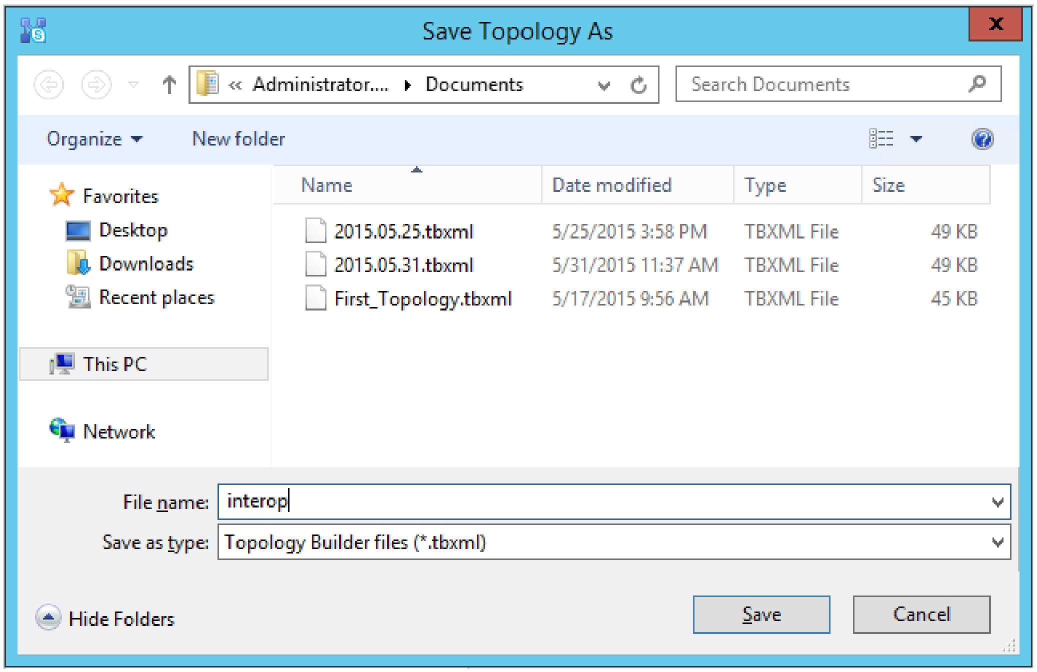
Open an existing Topology Builder file. Use this option if you have work in progress.

○ 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



Save Topology As

Name	Date modified	Type	Size
2015.05.25.tbxml	5/25/2015 3:58 PM	TBXML File	49 KB
2015.05.31.tbxml	5/31/2015 11:37 AM	TBXML File	49 KB
First_Topology.tbxml	5/17/2015 9:56 AM	TBXML File	45 KB

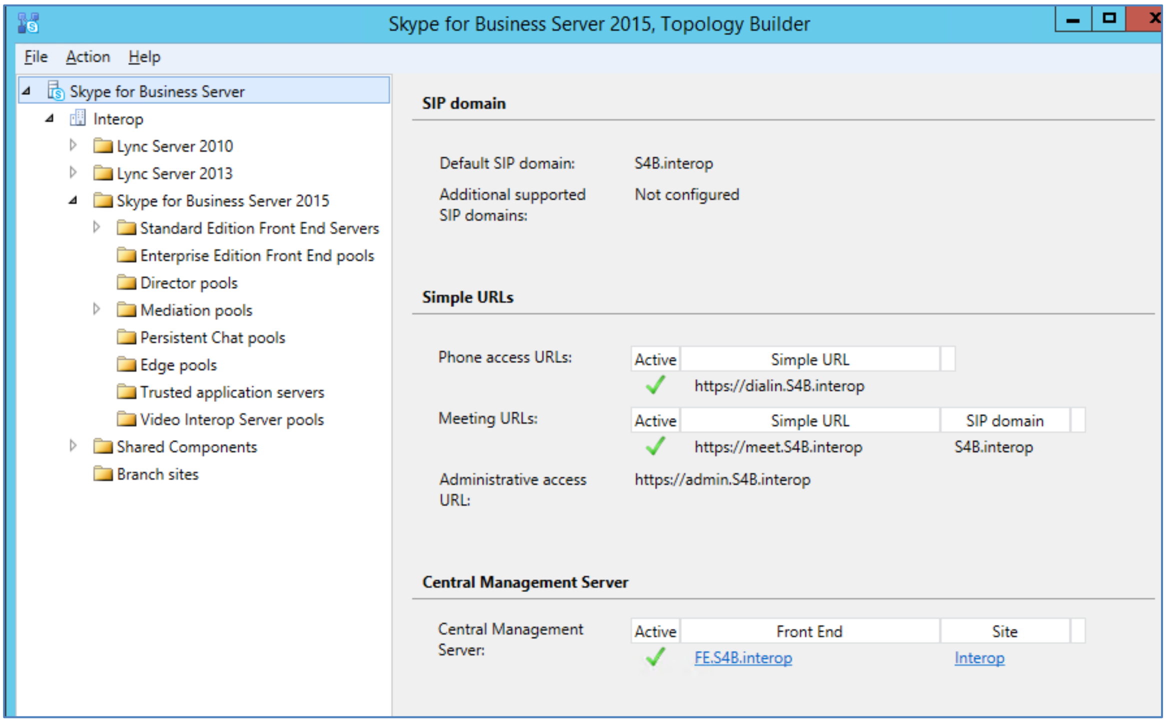
File name: interop

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



- Skype for Business Server
  - Interop
    - Lync Server 2010
    - Lync Server 2013
    - Skype for Business Server 2015
      - Standard Edition Front End Servers
      - Enterprise Edition Front End pools
      - Director pools
      - Mediation pools
      - Persistent Chat pools
      - Edge pools
      - Trusted application servers
      - Video Interop Server pools
    - Shared Components
    - Branch sites

SIP domain

Default SIP domain: S4B.interop

Additional supported SIP domains: Not configured

Simple URLs

Phone access URLs:

Active	Simple URL
✓	https://dialin.S4B.interop

Meeting URLs:

Active	Simple URL	SIP domain
✓	https://meet.S4B.interop	S4B.interop

Administrative access URL: https://admin.S4B.interop

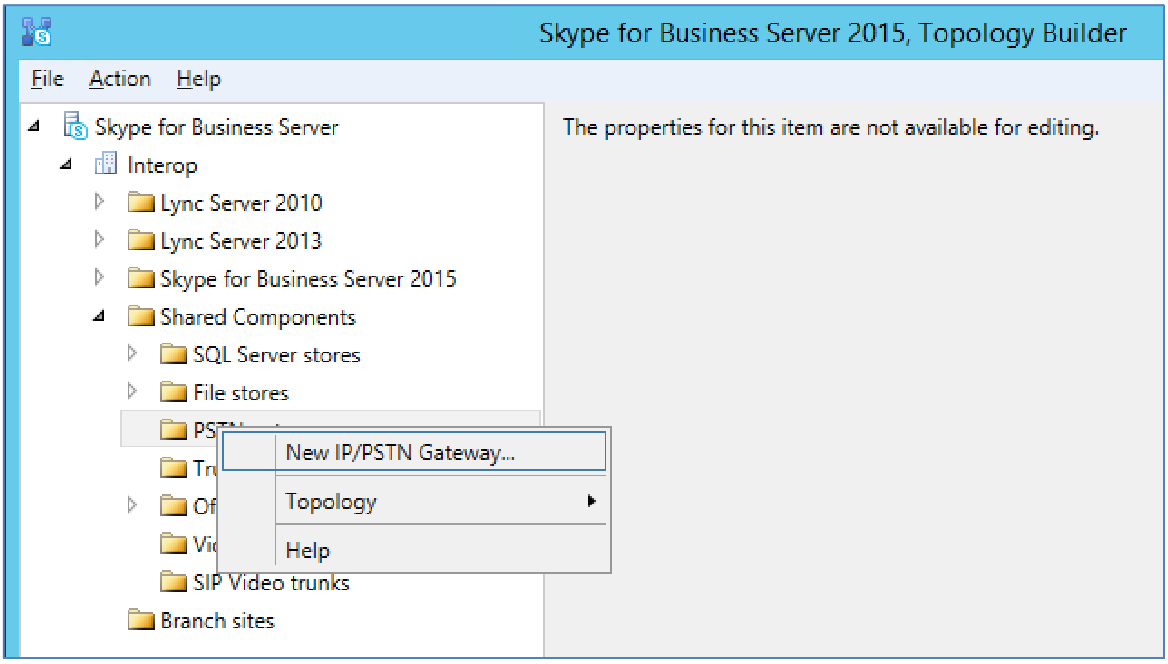
Central Management Server

Central Management Server:

Active	Front End	Site
✓	FE.S4B.interop	Interop

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

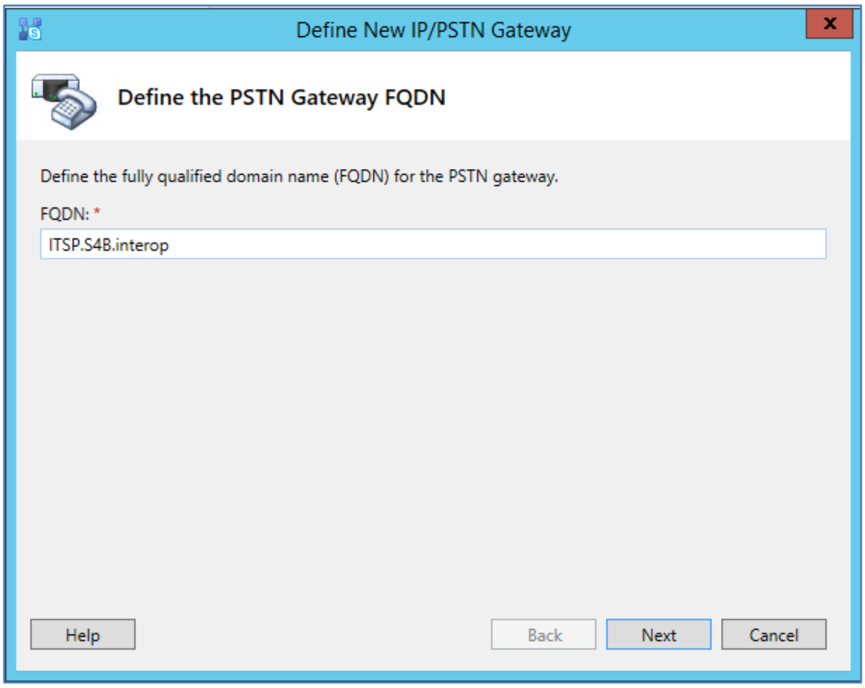


- Skype for Business Server
  - Interop
    - Lync Server 2010
    - Lync Server 2013
    - Skype for Business Server 2015
    - Shared Components
      - SQL Server stores
      - File stores
      - PST
      - Tr
      - Of
      - Vi
      - SIP Video trunks
    - Branch sites

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

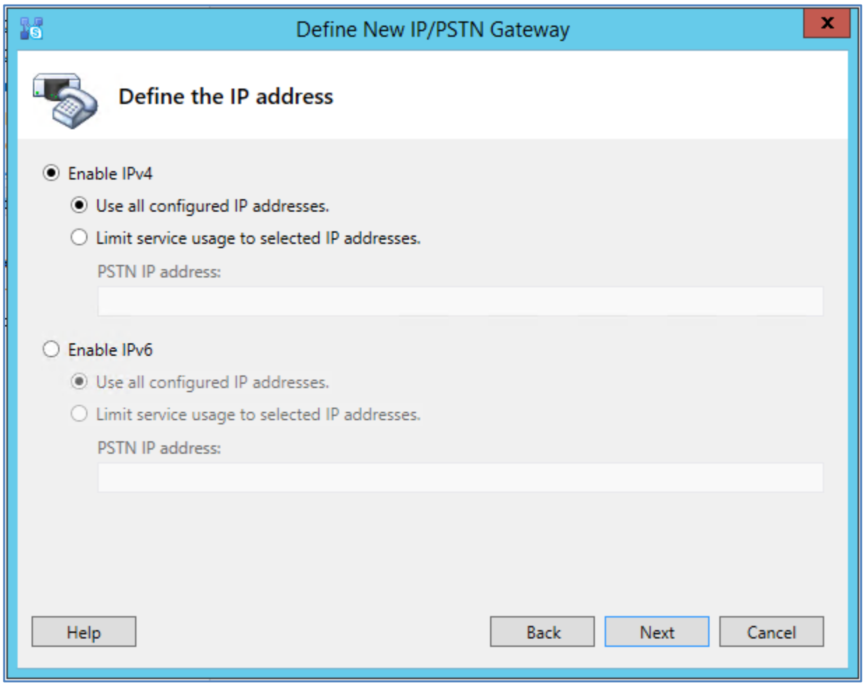
Define the fully qualified domain name (FQDN) for the PSTN gateway.

FQDN: *

ITSP.S4B.interop

5. Enter the Fully Qualified Domain Name (FQDN) of the E-SBC (e.g., **ITSP.S4B.interop**). 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:

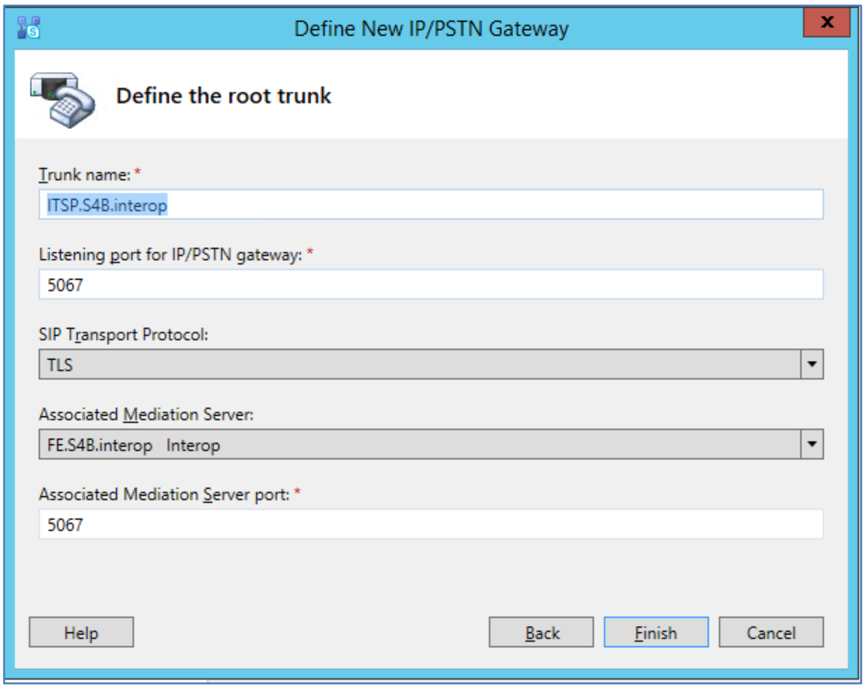
6. Define the listening mode (IPv4 or IPv6) of the IP address of your new PSTN gateway, and then click **Next**.

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.S4B.interop

Listening port for IP/PSTN gateway: * 5067

SIP Transport Protocol: TLS

Associated Mediation Server: FE.S4B.interop Interop

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.

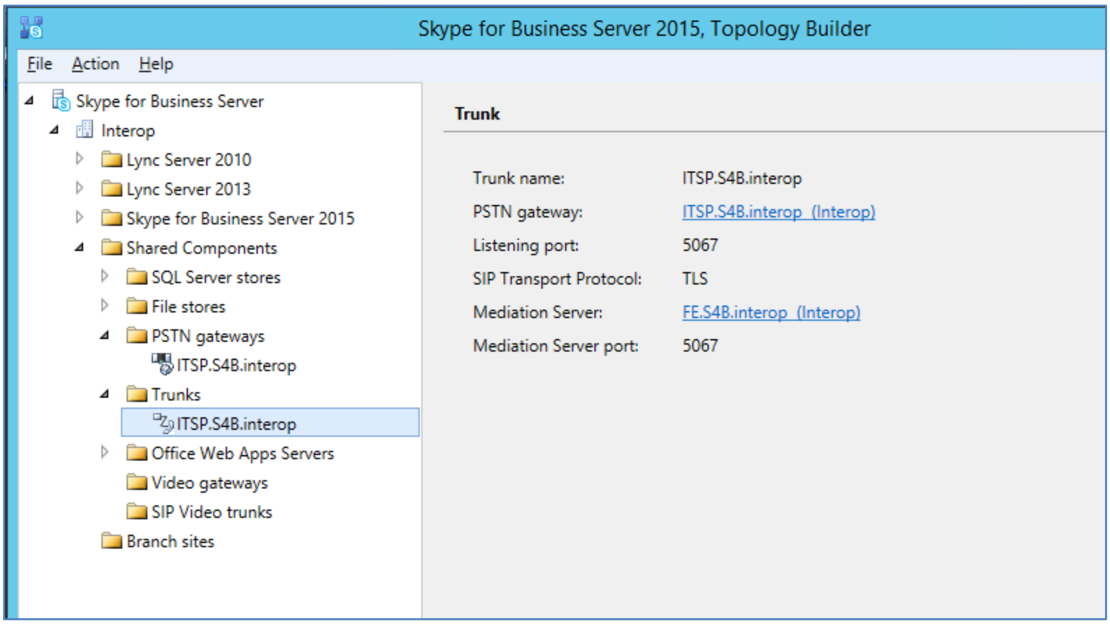
c. In the 'Associated Mediation Server' field, select the Mediation Server pool to associate with the root trunk of this PSTN gateway.

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



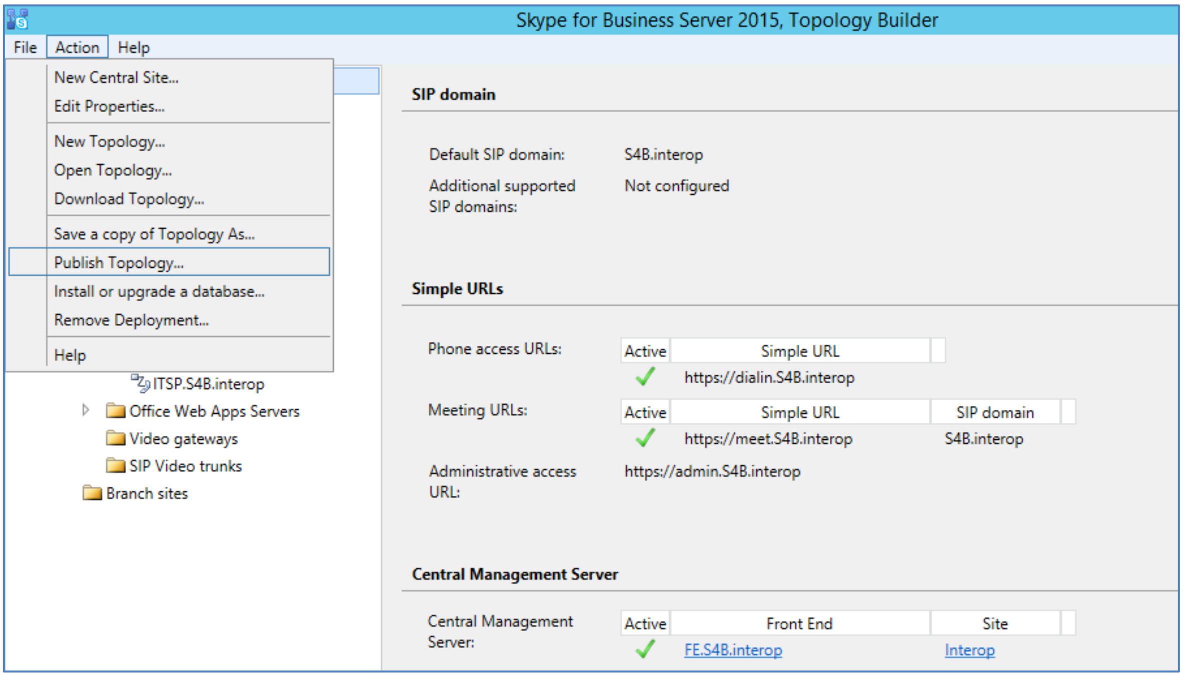
Skype for Business Server 2015, Topology Builder

Trunk

Trunk name:	ITSP.S4B.interop
PSTN gateway:	ITSP.S4B.interop (Interop)
Listening port:	5067
SIP Transport Protocol:	TLS
Mediation Server:	FE.S4B.interop (Interop)
Mediation Server port:	5067

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

Figure 3-10: Choosing Publish Topology



Skype for Business Server 2015, Topology Builder

SIP domain

Default SIP domain:	S4B.interop
Additional supported SIP domains:	Not configured

Simple URLs

Phone access URLs:

Active	Simple URL
✓	https://dialin.S4B.interop

Meeting URLs:

Active	Simple URL	SIP domain
✓	https://meet.S4B.interop	S4B.interop

Administrative access URL: https://admin.S4B.interop

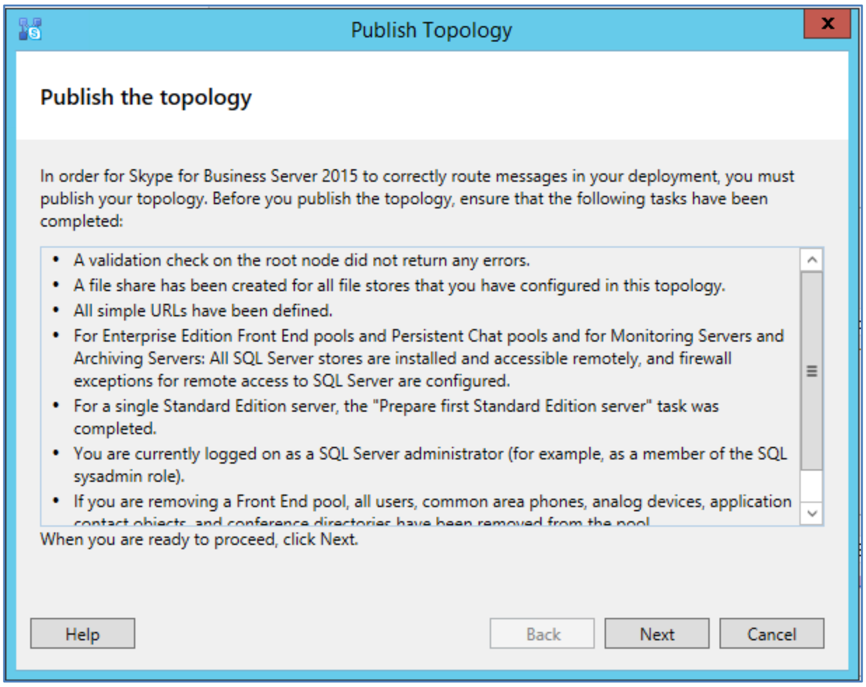
Central Management Server

Central Management Server:

Active	Front End	Site
✓	FE.S4B.interop	Interop

The following is displayed:

Figure 3-11: Publish the Topology



Publish Topology

Publish the topology

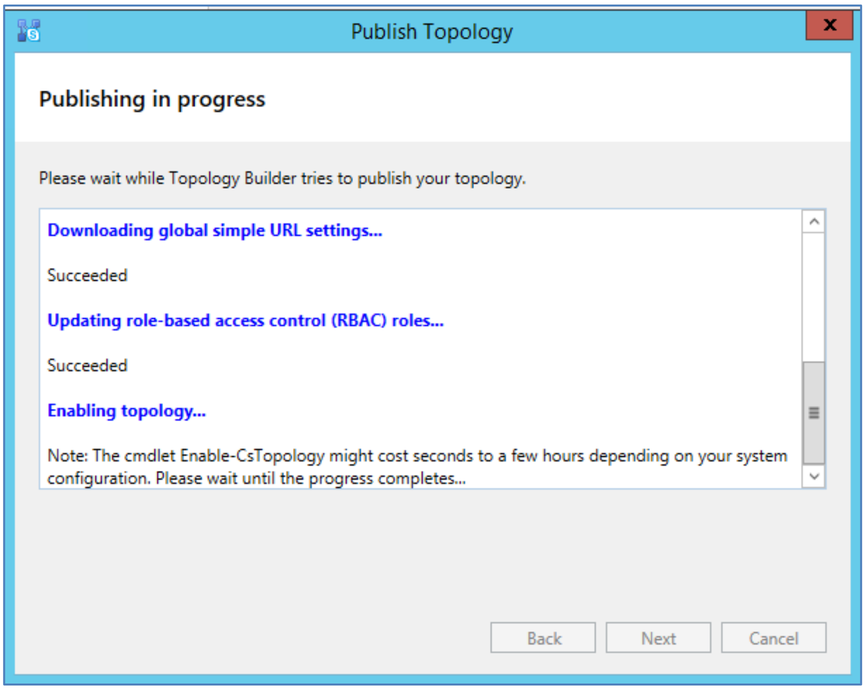
In order for Skype for Business Server 2015 to correctly route messages in your deployment, you must publish your topology. Before you publish the topology, ensure that the following tasks have been completed:

- 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.

Downloading global simple URL settings...

Succeeded

Updating role-based access control (RBAC) roles...

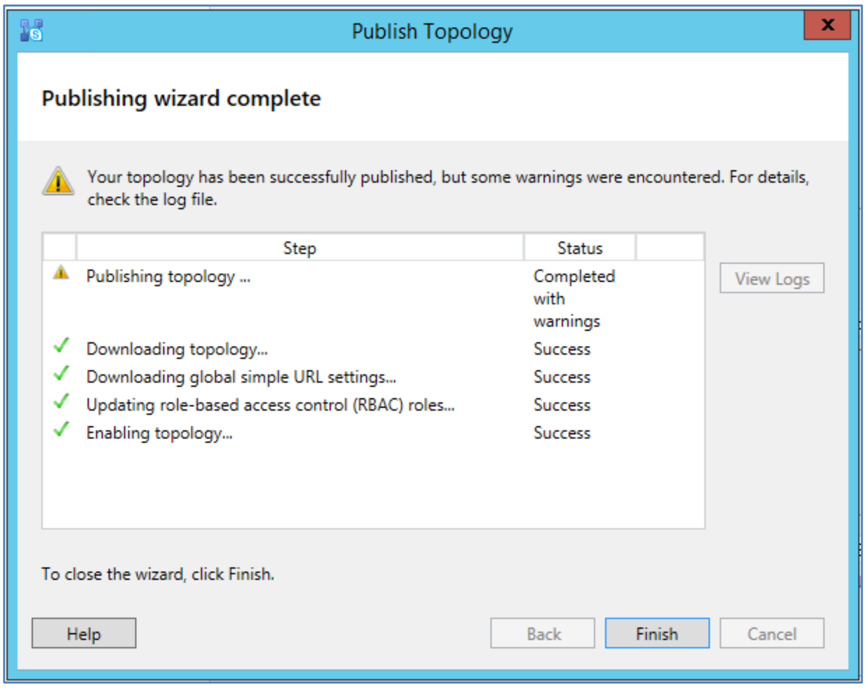
Succeeded

Enabling topology...

Note: The cmdlet Enable-CsTopology might cost seconds to a few hours depending on your system configuration. Please wait until the progress completes...

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

Figure 3-13: Publishing Wizard Complete



Publish Topology

Publishing wizard complete

Your topology has been successfully published, but some warnings were encountered. For details, check the log file.

	Step	Status
⚠	Publishing topology ...	Completed with warnings
✓	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**.

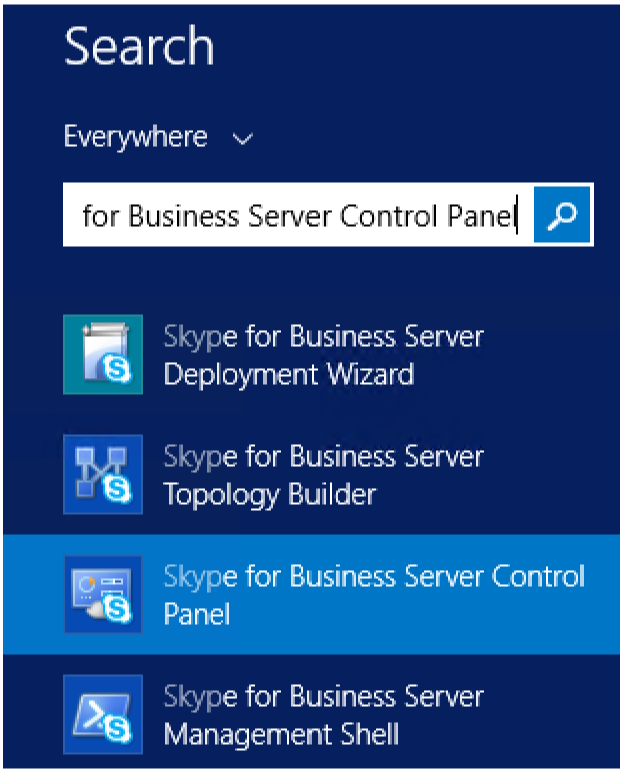
3.2 Configuring a 'Route' on Skype for Business Server 2015

The procedure below describes how to configure a 'Route' on the Skype for Business Server 2015, and to associate it with the E-SBC PSTN gateway.

➢ **To configure a 'route' on Skype for Business Server 2015:**

1. Start the Microsoft Skype for Business Server 2015 Control Panel (**Start** > search for **Microsoft Skype for Business Server Control Panel**), as shown below:

Figure 3-14: Opening the Skype for Business Server Control Panel



Search

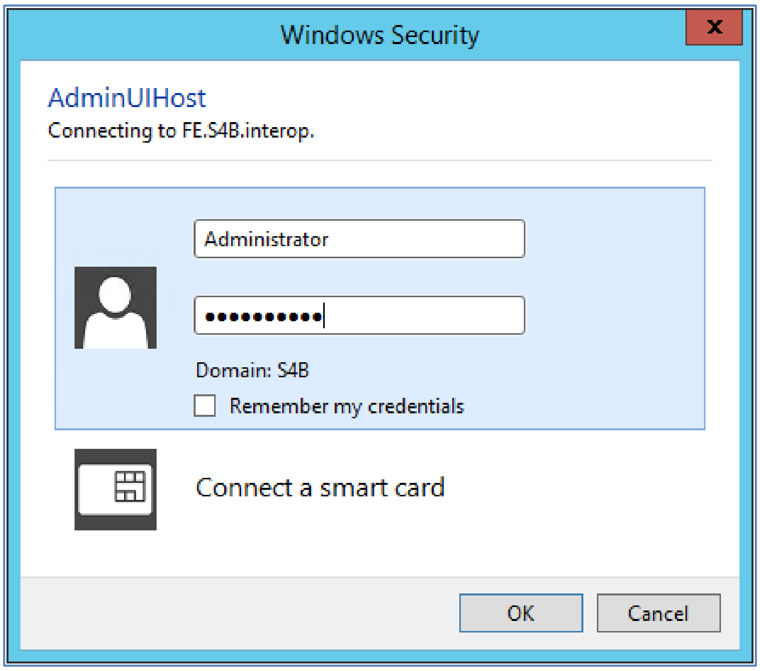
Everywhere

for Business Server Control Panel

Skype for Business Server Deployment Wizard
Skype for Business Server Topology Builder
Skype for Business Server Control Panel
Skype for Business Server Management Shell

2. You're prompted to enter your login credentials:

Figure 3-15: Skype for Business Server Credentials



Windows Security

AdminUIHost
Connecting to FE.S4B.interop.

Administrator

••••••••••

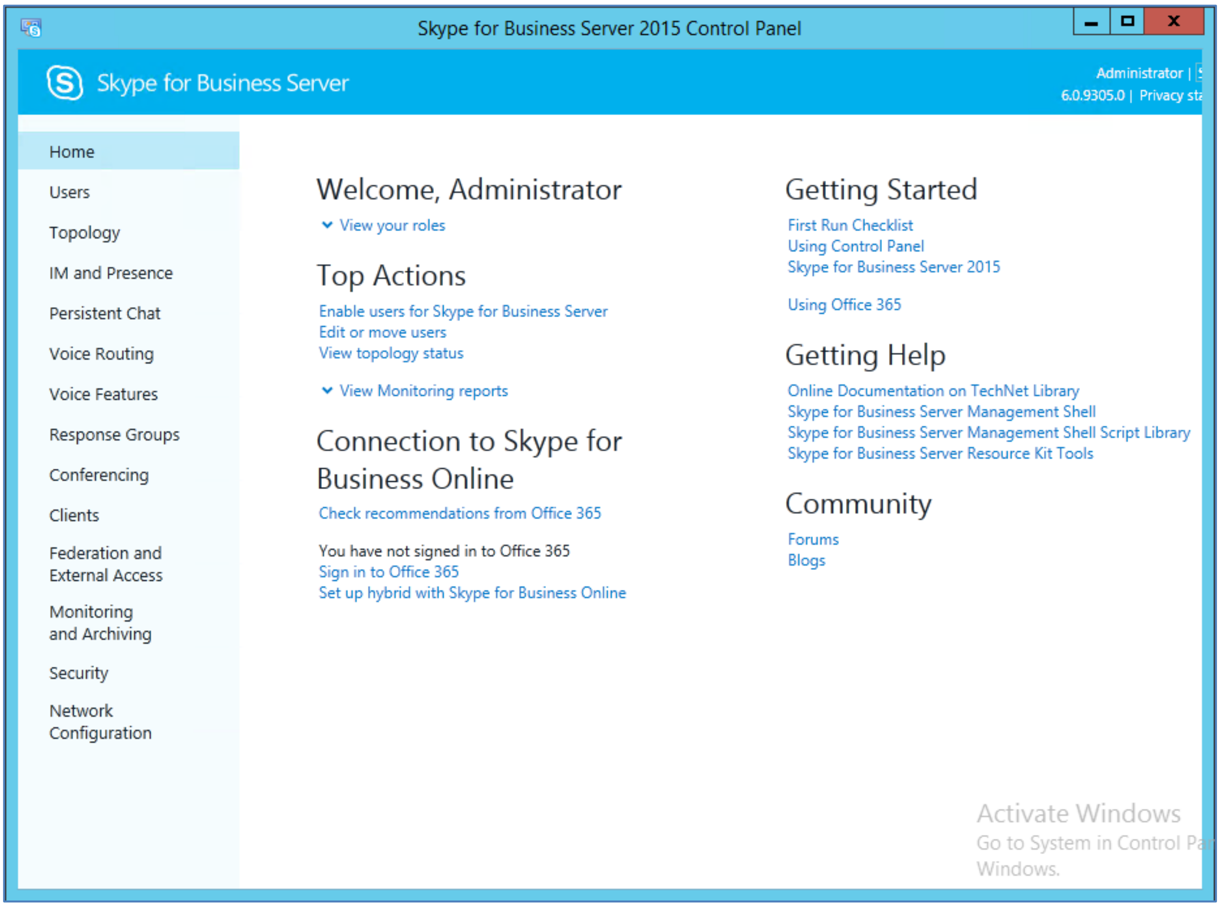
Domain: S4B

□ Remember my credentials

Connect a smart card

3. Enter your domain username and password, and then click **OK**; the Microsoft Skype for Business Server 2015 Control Panel is displayed:

Figure 3-16: Microsoft Skype for Business Server 2015 Control Panel



Skype for Business Server 2015 Control Panel

Skype for Business Server

Administrator | S
6.0.9305.0 | Privacy sta

Welcome, Administrator
View your roles

Top Actions
Enable users for Skype for Business Server
Edit or move users
View topology status

View Monitoring reports

Connection to Skype for Business Online
Check recommendations from Office 365

You have not signed in to Office 365
Sign in to Office 365
Set up hybrid with Skype for Business Online

Getting Started
First Run Checklist
Using Control Panel
Skype for Business Server 2015

Using Office 365

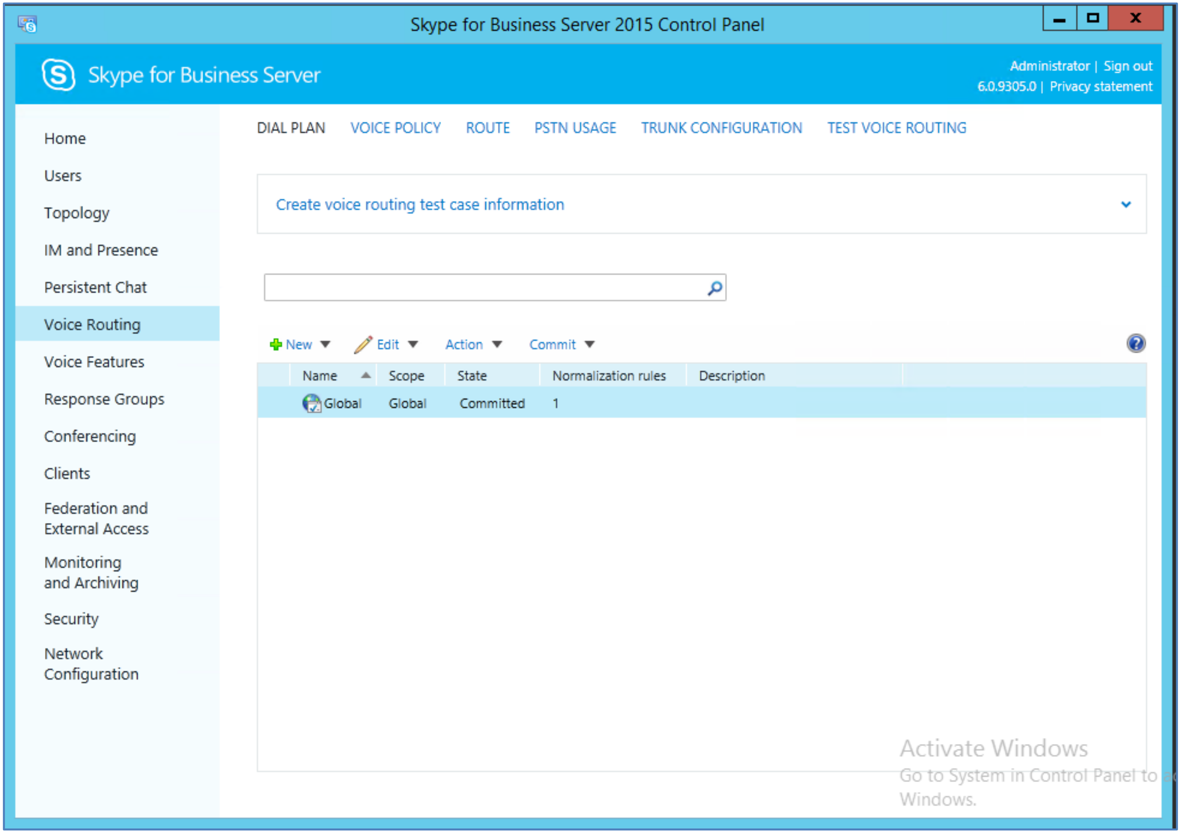
Getting Help
Online Documentation on TechNet Library
Skype for Business Server Management Shell
Skype for Business Server Management Shell Script Library
Skype for Business Server Resource Kit Tools

Community
Forums
Blogs

Activate Windows
Go to System in Control Pa
Windows.

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

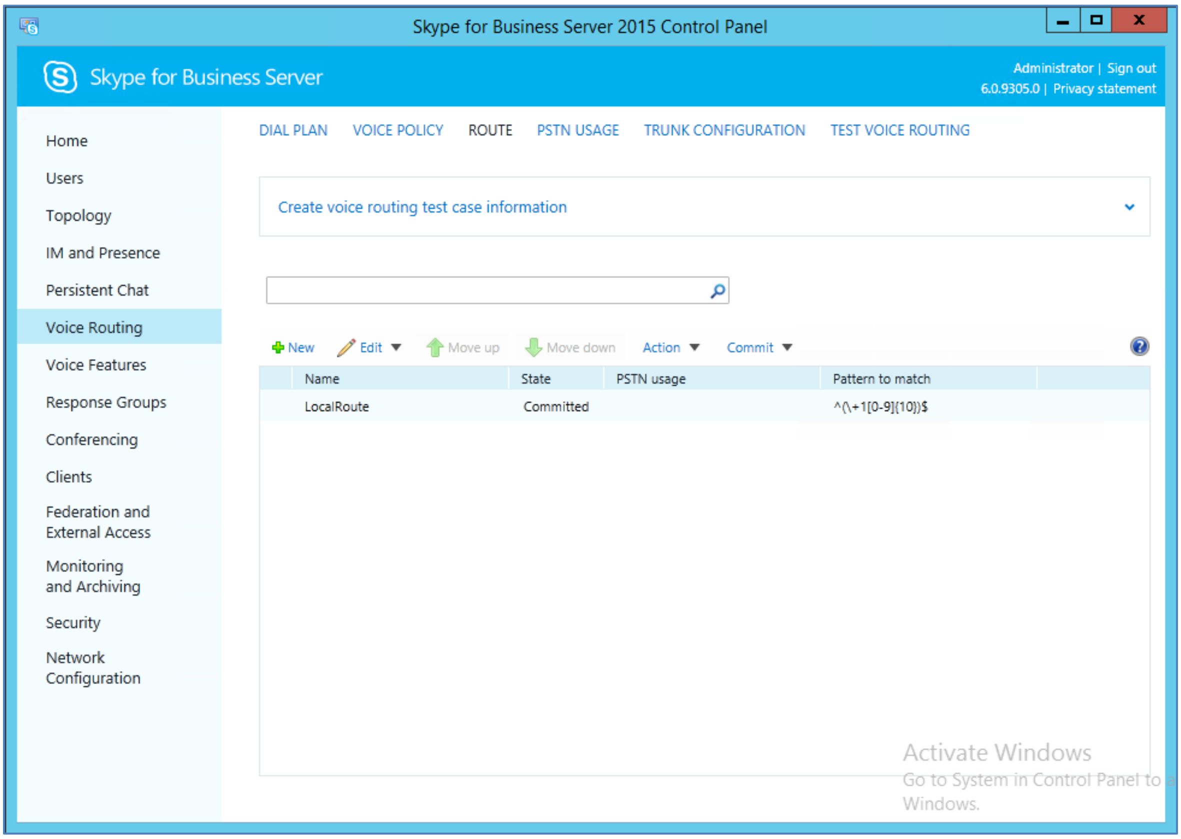
Figure 3-17**: Voice Routing Page**



Name	Scope	State	Normalization rules	Description
Global	Global	Committed	1	

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

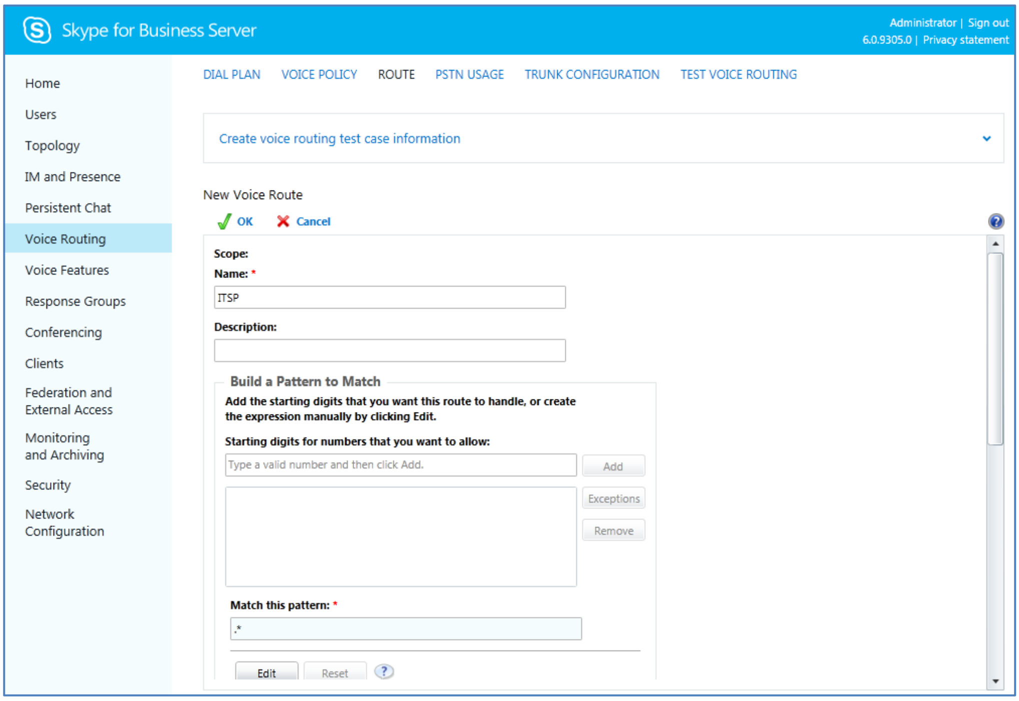
Figure 3-18: Route Tab



Name	State	PSTN usage	Pattern to match
LocalRoute	Committed		^(\+1[0-9]{10})$

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

Figure 3-19: Adding New Voice Route



Name: * ITSP

Match this pattern: * .*

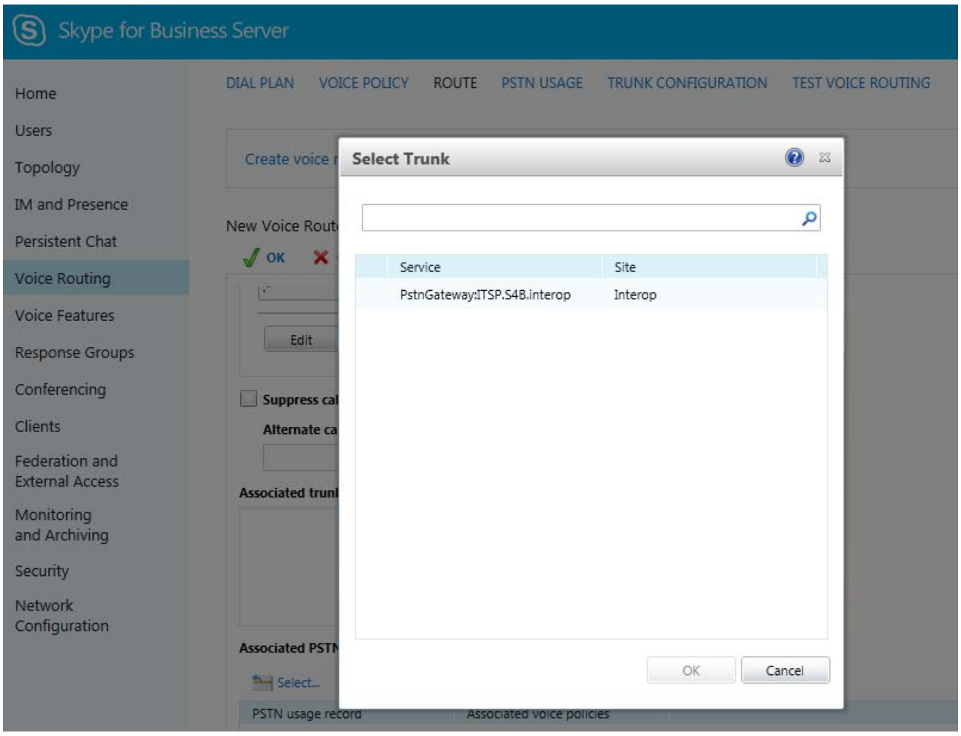
7. In the 'Name' field, enter a name for this route (e.g., **ITSP**).

8. 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**.

9. 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:

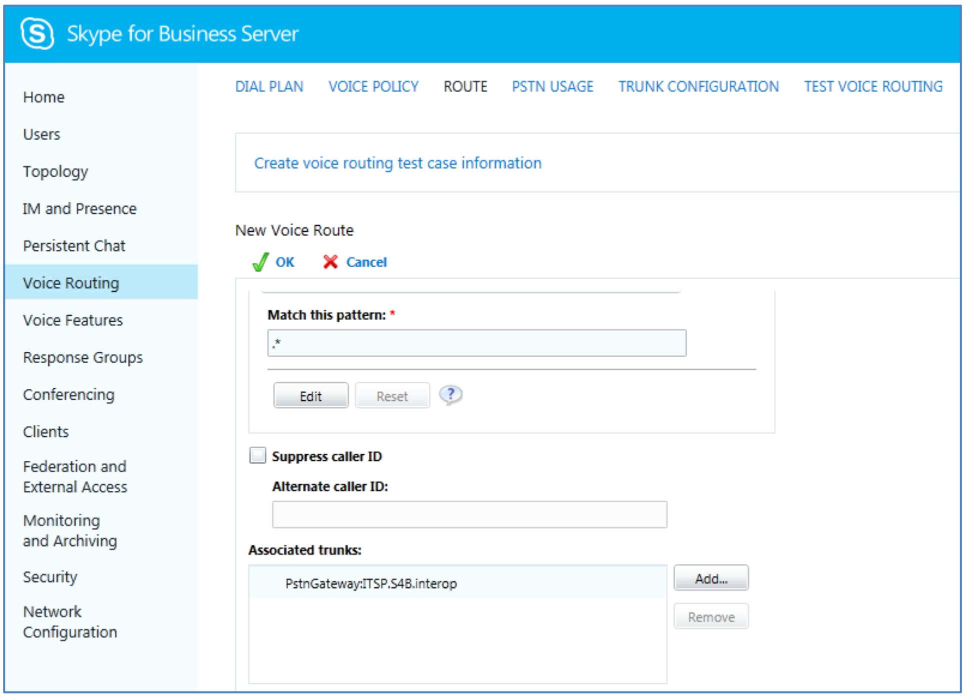
Figure 3-20: List of Deployed Trunks



Service	Site
PstnGateway:ITSP.S4B.interop	Interop

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

Figure 3-21: Selected E-SBC Trunk



Match this pattern: * .*

☐ Suppress caller ID

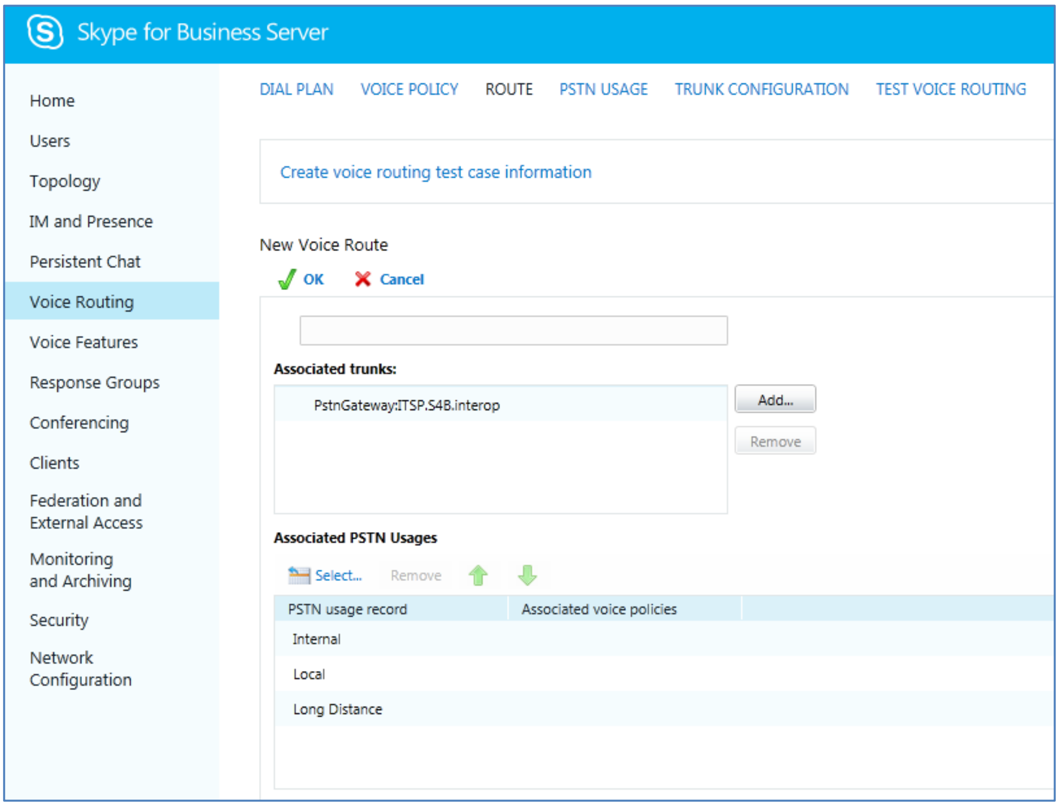
Alternate caller ID:

Associated trunks: PstnGateway:ITSP.S4B.interop

10. Associate a PSTN Usage to this route:

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

Figure 3-22: Associating PSTN Usage to Route



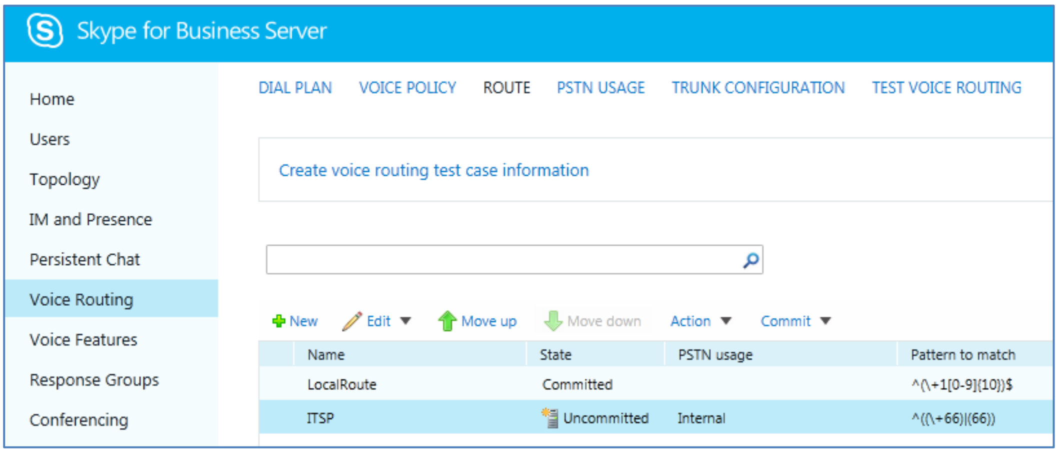
Associated trunks: PstnGateway:ITSP.S4B.interop

Associated PSTN Usages

PSTN usage record	Associated voice policies
Internal	
Local	
Long Distance	

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

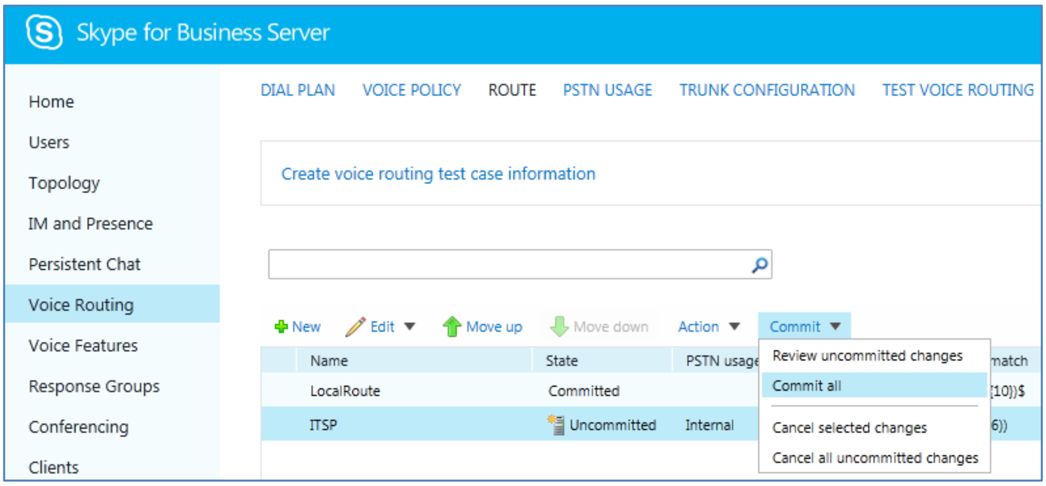
Figure 3-23: Confirmation of New Voice Route



Name	State	PSTN usage	Pattern to match
LocalRoute	Committed		^(\+1[0-9]{10})$
ITSP	Uncommitted	Internal	^((\+66)|(66))

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

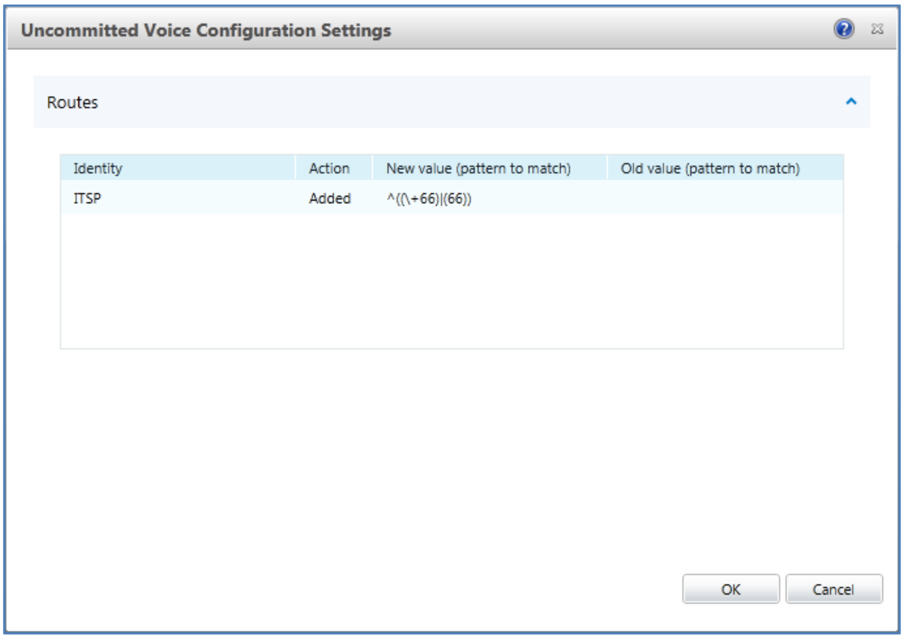
Figure 3-24: Committing Voice Routes



Name	State	PSTN usage	match
LocalRoute	Committed		{10})$
ITSP	Uncommitted	Internal	6))

The Uncommitted Voice Configuration Settings page appears:

Figure 3-25: Uncommitted Voice Configuration Settings



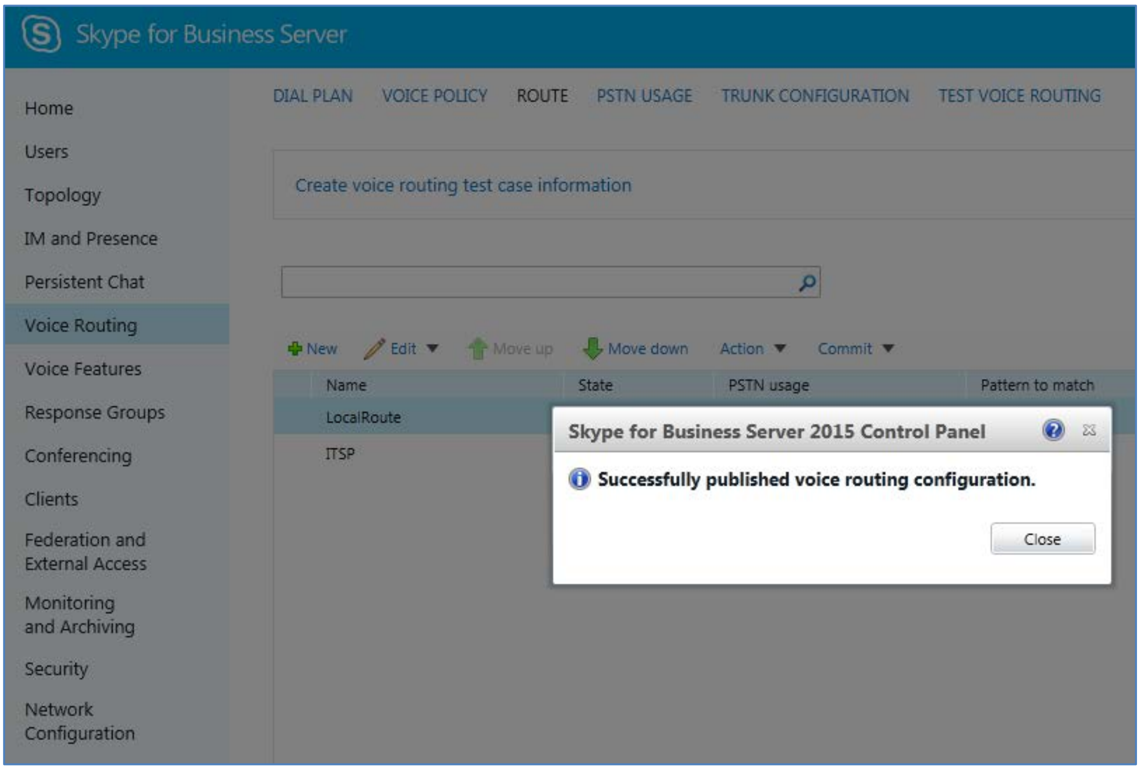
Uncommitted Voice Configuration Settings

Routes

Identity	Action	New value (pattern to match)	Old value (pattern to match)
ITSP	Added	^((\+66)|(66))	

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

Figure 3-26: Confirmation of Successful Voice Routing Configuration

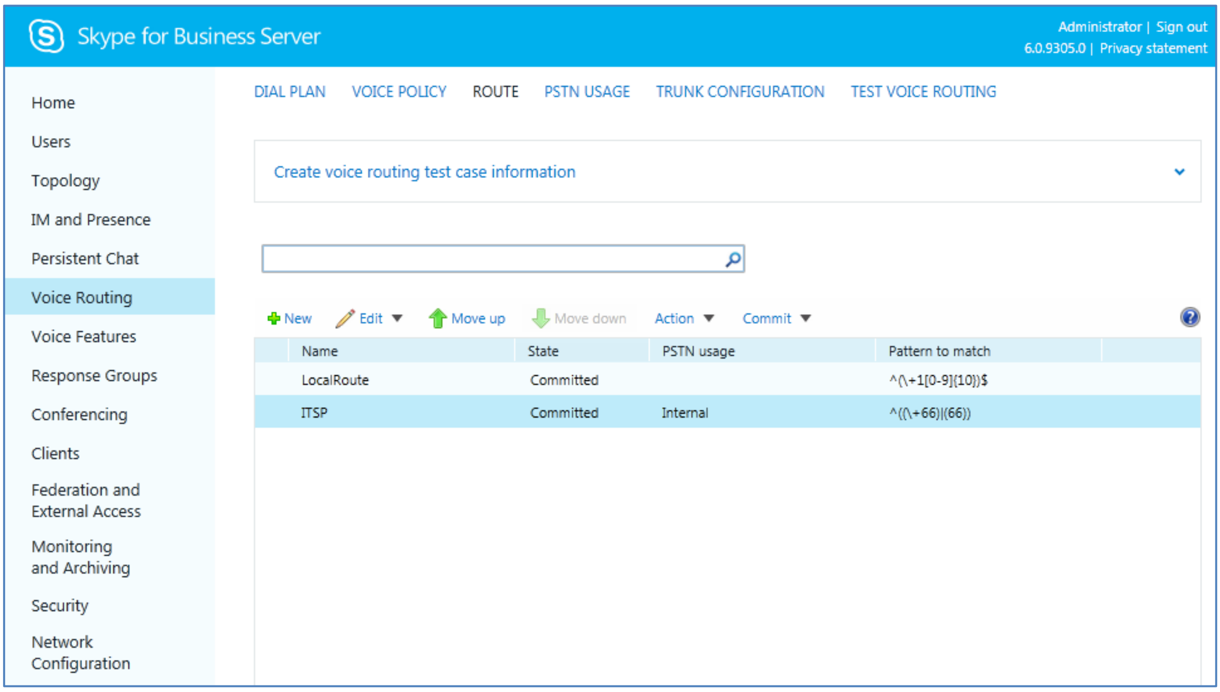


Skype for Business Server 2015 Control Panel

Successfully published voice routing configuration.

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

Figure 3-27: Voice Routing Screen Displaying Committed Routes



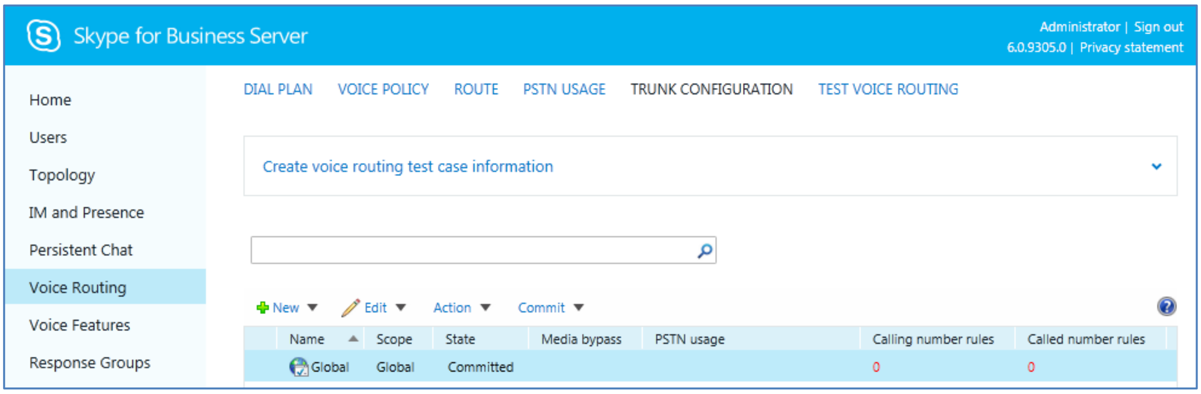
Name	State	PSTN usage	Pattern to match
LocalRoute	Committed		^(\+1[0-9]{10})$
ITSP	Committed	Internal	^((\+66)|(66))

15. 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 S4B user number). This ID is required by BroadConnect SIP Trunk in the P-Asserted-Identity header. The device adds this ID to the P-Asserted-Identity header in the sent INVITE message, using the IP Profile (see Section 4.6).

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.

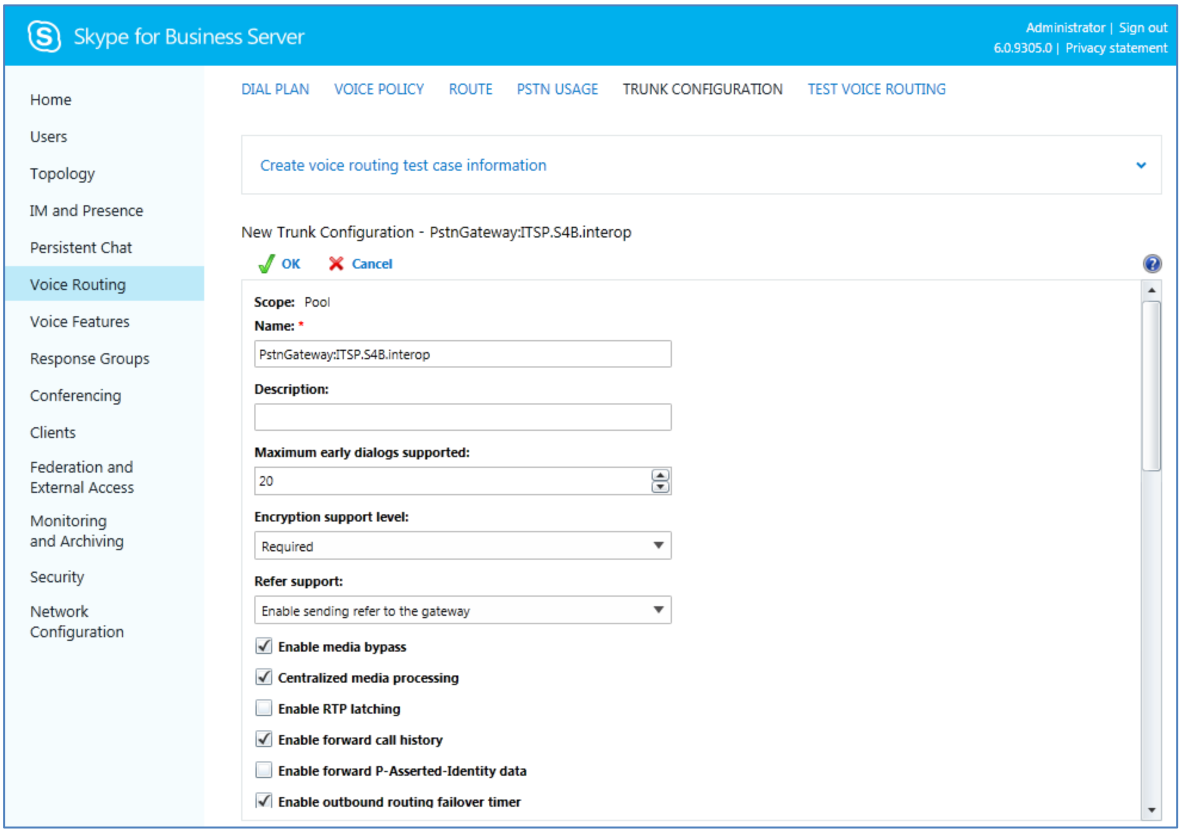
Figure 3-28: Voice Routing Screen – Trunk Configuration Tab



Name	Scope	State	Media bypass	PSTN usage	Calling number rules	Called number rules
Global	Global	Committed			0	0

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

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



New Trunk Configuration - PstnGateway:ITSP.S4B.interop

Scope: Pool

Name: * PstnGateway:ITSP.S4B.interop

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** option, and then click **OK**.

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

This page is intentionally left blank.

4 Configuring AudioCodes' E-SBC

This section shows how to configure AudioCodes' E-SBC for interworking between Microsoft Skype for Business Server 2015 and the BroadConnect SIP Trunk. The configuration procedures are based on the interoperability test topology described in Section 2.4, including the following main areas:

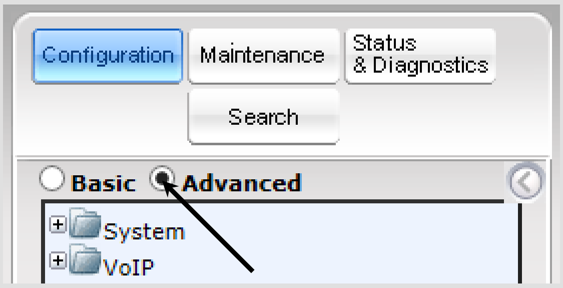
- E-SBC WAN interface - BroadConnect SIP Trunking environment
- E-SBC LAN interface - Skype for Business Server 2015 environment

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

Notes:

- For implementing Microsoft Skype for Business and BroadConnect 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 Skype for Business 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 Advanced-menu display mode. To do this, select the **Advanced** option, as shown below:



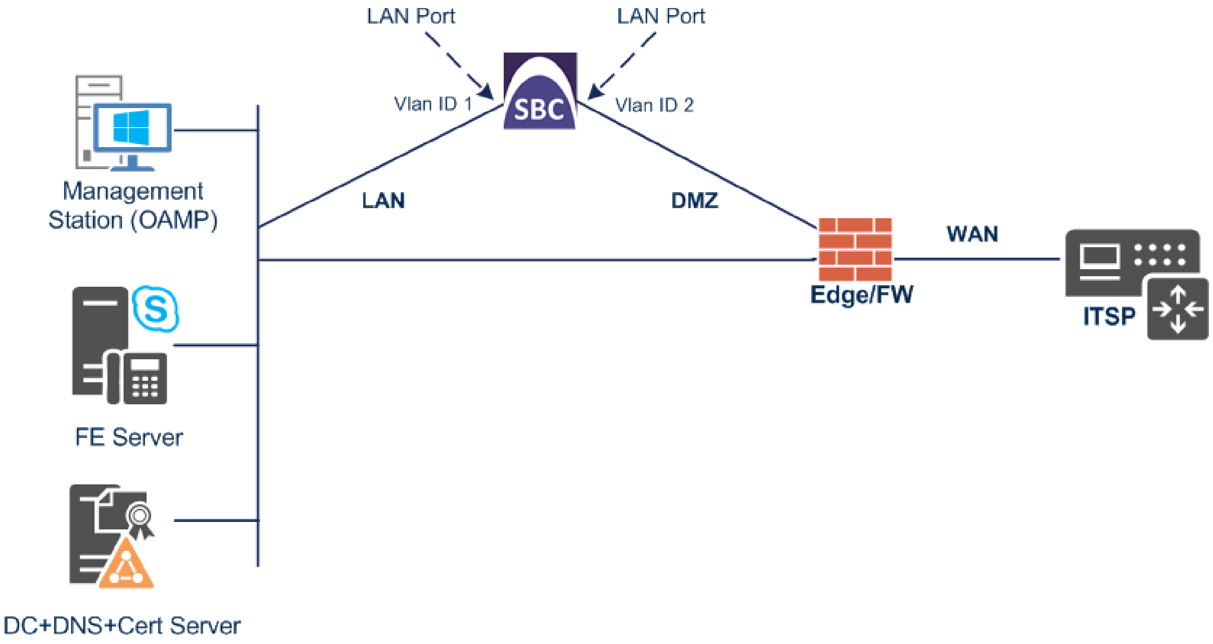
Note that when the E-SBC is reset, the navigation tree reverts to **Basic** menu display.

4.1 Step 1: IP Network Interfaces Configuration

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

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

Figure 4-1: Network Interfaces in Interoperability Test Topology



4.1.1 Step 1a: Configure VLANs

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

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

➢ **To configure the VLANs:**

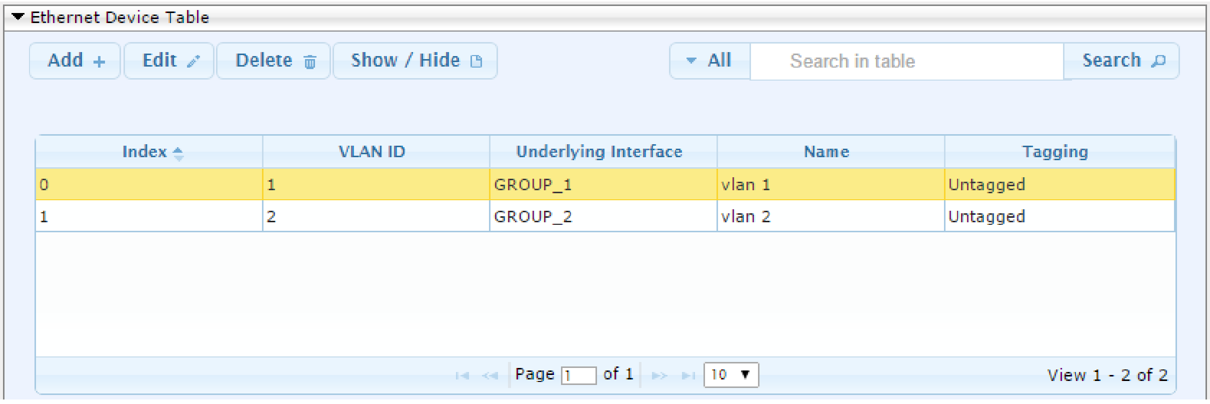
1. Open the Ethernet Device Table page (**Configuration** tab > **VoIP** menu > **Network** > **Ethernet Device Table**).

   There will be one existing row for VLAN ID 1 and underlying interface GROUP_1.

2. Add another VLAN ID 2 for the WAN side as follows:

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

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



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

4.1.2 Step 1b: Configure Network Interfaces

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

- LAN VoIP (assigned the name "Voice")
- WAN VoIP (assigned the name "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' option of the **OAMP + Media + Control** table row, and then click **Edit**.
   b. Configure the interface as follows:

Parameter	Value
IP Address	**10.15.17.55** (IP address of E-SBC)
Prefix Length	**16** (subnet mask in bits for 255.255.0.0)
Default Gateway	**10.15.0.1**
VLAN ID	**1**
Interface Name	**Voice** (arbitrary descriptive name)
Primary DNS Server IP Address	**10.15.27.1**
Underlying Device	**vlan 1**

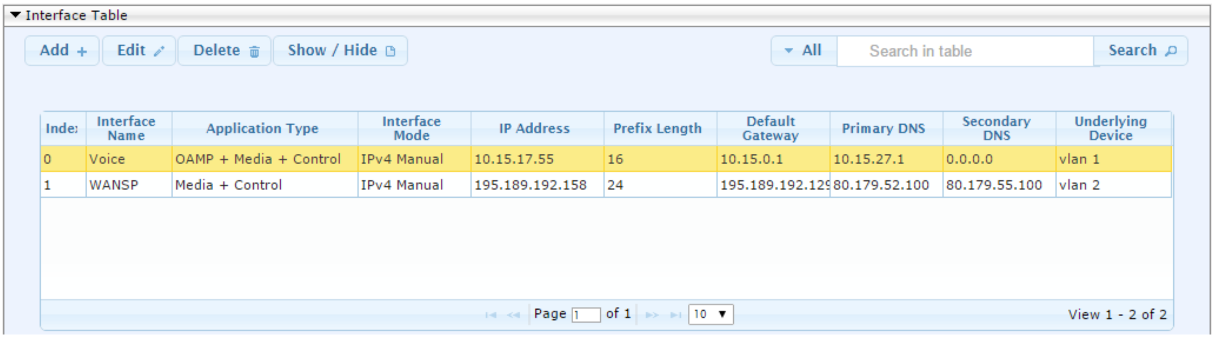
3. Add a network interface for the WAN side:
   a. Enter **1**, and then click **Add Index**.
   b. Configure the interface as follows:

Parameter	Value
Application Type	**Media + Control**
IP Address	**195.189.192.158** (WAN IP address)
Prefix Length	**25** (for 255.255.255.128)
Default 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



Inde:	Interface Name	Application Type	Interface Mode	IP Address	Prefix Length	Default Gateway	Primary DNS	Secondary DNS	Underlying Device
0	Voice	OAMP + Media + Control	IPv4 Manual	10.15.17.55	16	10.15.0.1	10.15.27.1	0.0.0.0	vlan 1
1	WANSP	Media + Control	IPv4 Manual	195.189.192.158	24	195.189.192.12(	80.179.52.100	80.179.55.100	vlan 2

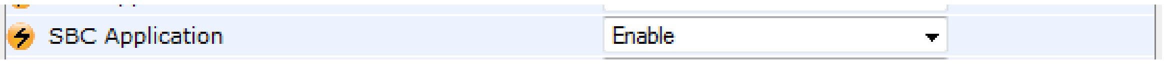
4.2 Step 2: Enable the SBC Application

This step shows 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 4-4: Enabling SBC Application



SBC Application: Enable

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 82).

4.3 Step 3: Configure Media Realms

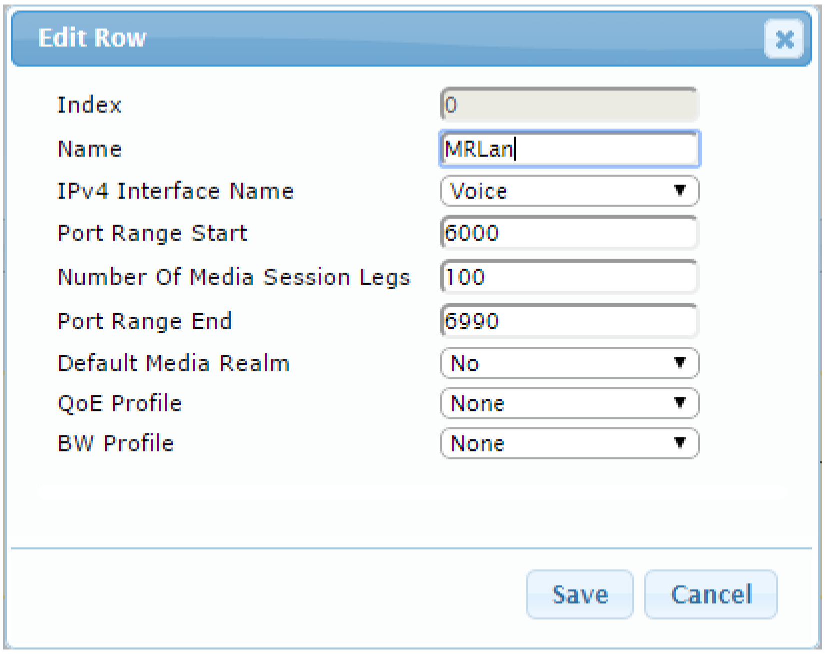
This step shows how to configure Media Realms. The simplest configuration is to create two Media Realms - one for internal (LAN) traffic and one for external (WAN) traffic.

➢ **To configure Media Realms:**

1. Open the Media Realm Table page (**Configuration** tab > **VoIP** menu > **VoIP Network** > **Media Realm Table**).
2. Add a Media Realm for the LAN interface. You can use the default Media Realm (Index 0), but modify it as shown below:

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	**100** (media sessions assigned with port range)

Figure 4-5: Configuring Media Realm for LAN



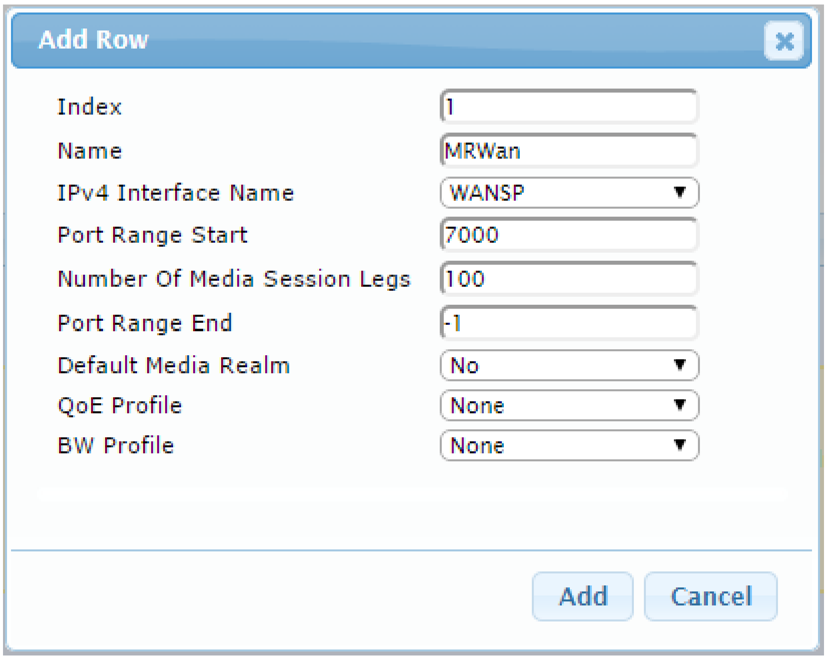
Edit Row

Index	0
Name	MRLan
IPv4 Interface Name	Voice
Port Range Start	6000
Number Of Media Session Legs	100
Port Range End	6990
Default Media Realm	No
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	**100** (media sessions assigned with port range)

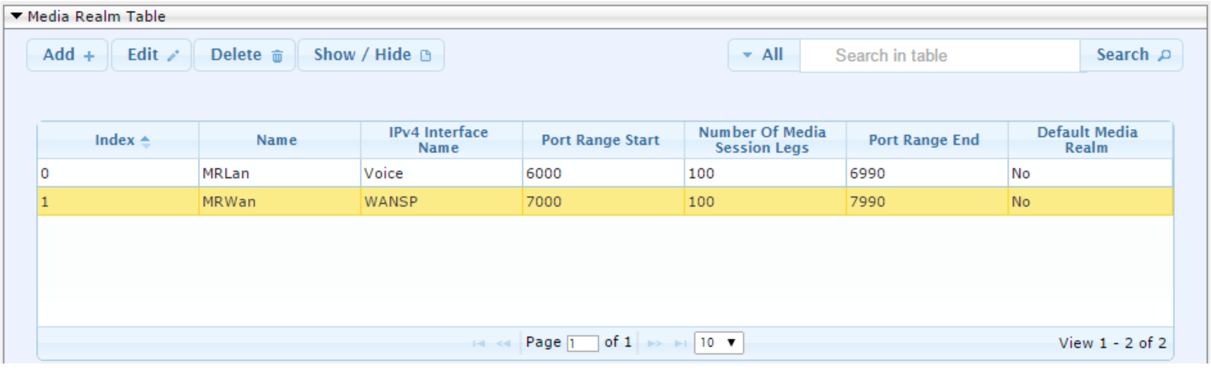
Figure 4-6: Configuring Media Realm for WAN



Index	1
Name	MRWan
IPv4 Interface Name	WANSP
Port Range Start	7000
Number Of Media Session Legs	100
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 4-7: Configured Media Realms in Media Realm Table



Index	Name	IPv4 Interface Name	Port Range Start	Number Of Media Session Legs	Port Range End	Default Media Realm
0	MRLan	Voice	6000	100	6990	No
1	MRWan	WANSP	7000	100	7990	No

4.4 Step 4: Configure SIP Signaling Interfaces

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

➢ **To configure SIP Interfaces:**

1. Open the SIP Interface Table page (**Configuration** tab > **VoIP** menu > **VoIP Network** > **SIP Interface Table**).
2. Add a SIP Interface for the LAN interface. You can use the default SIP Interface (Index 0), but modify it as shown below:

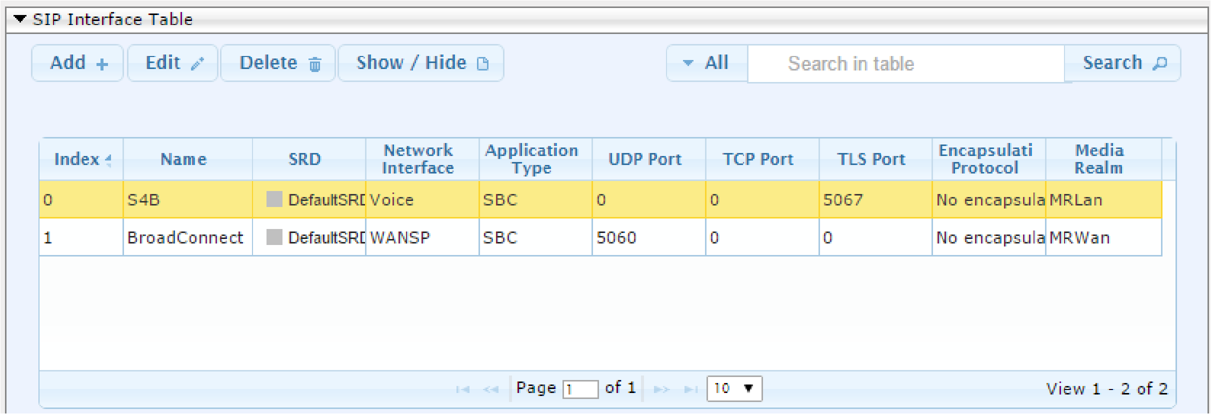
Parameter	Value
Index	**0**
Interface Name	**S4B** (see Note below)
Network Interface	**Voice**
Application Type	**SBC**
TLS Port	**5067**
TCP and UDP	**0**
Media Realm	**MRLan**

3. Configure a SIP Interface for the WAN:

Parameter	Value
Index	**1**
Interface Name	**BroadConnect** (see Note below)
Network Interface	**WANSP**
Application Type	**SBC**
UDP Port	**5060**
TCP and TLS	**0**
Media Realm	**MRWan**

The configured SIP Interfaces are shown in the figure below:

Figure 4-8: Configured SIP Interfaces in SIP Interface Table



Index	Name	SRD	Network Interface	Application Type	UDP Port	TCP Port	TLS Port	Encapsulati Protocol	Media Realm
0	S4B	DefaultSRD	Voice	SBC	0	0	5067	No encapsula	MRLan
1	BroadConnect	DefaultSRD	WANSP	SBC	5060	0	0	No encapsula	MRWan

Note: Unlike in previous software releases where configuration entities (e.g., SIP Interface, Proxy Sets, and IP Groups) were associated with each other using table row indices, Version 7.0 uses the string **names** of the configuration entities. Therefore, it is recommended to configure each configuration entity with meaningful names for easy identification.

4.5 Step 5: Configure Proxy Sets

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

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

- Microsoft Skype for Business Server 2015
- BroadConnect SIP Trunk

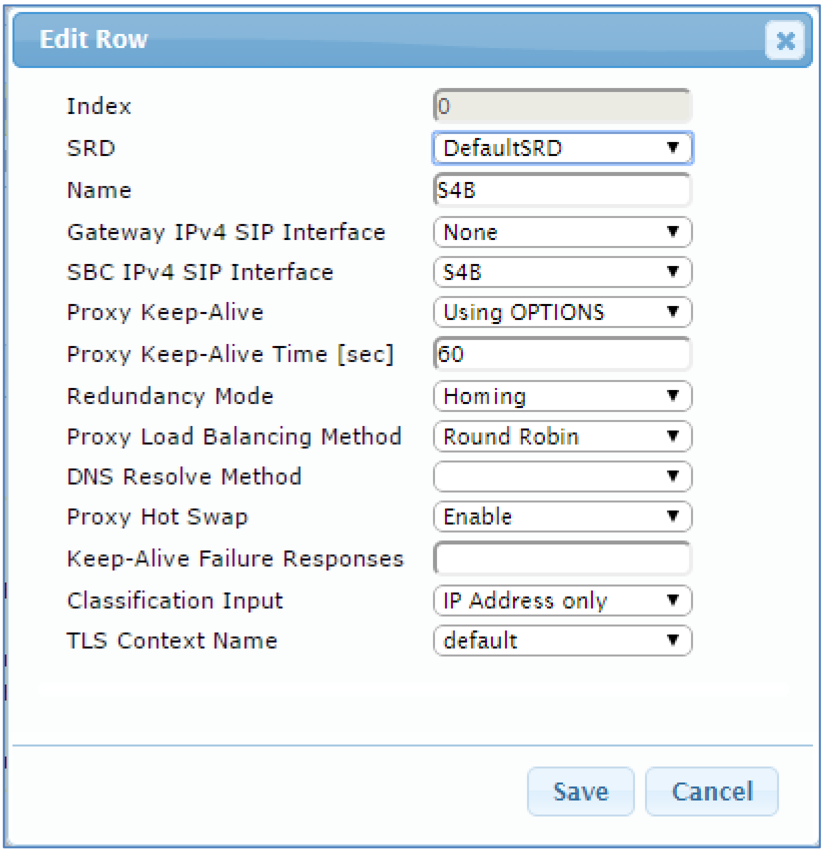
The Proxy Sets will later be applied to the VoIP network by assigning them to IP Groups.

➢ To configure Proxy Sets:

1. Open the Proxy Sets Table page (**Configuration** tab > **VoIP** menu > **VoIP Network** > **Proxy Sets Table**).
2. Add a Proxy Set for the Skype for Business Server 2015. You can use the default Proxy Set (Index 0), but modify it as shown below:

Parameter	Value
Proxy Set ID	**0**
Proxy Name	**S4B** (see the Note on page 39)
SBC IPv4 SIP Interface	**S4B**
Proxy Keep Alive	**Using Options**
Redundancy Mode	**Homing**
Load Balancing Method	**Round Robin**
Proxy Hot Swap	**Enable**
TLS Context Name	**default**

Figure 4-9: Configuring Proxy Set for Microsoft Skype for Business Server 2015



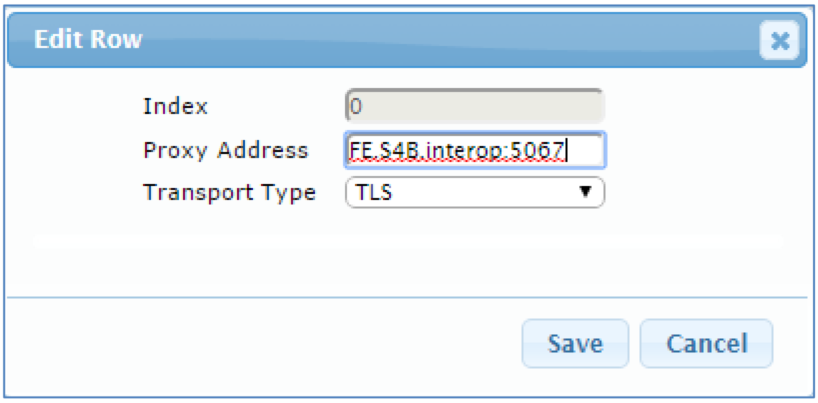
Index	0
SRD	DefaultSRD
Name	S4B
Gateway IPv4 SIP Interface	None
SBC IPv4 SIP Interface	S4B
Proxy Keep-Alive	Using OPTIONS
Proxy Keep-Alive Time [sec]	60
Redundancy Mode	Homing
Proxy Load Balancing Method	Round Robin
DNS Resolve Method	
Proxy Hot Swap	Enable
Keep-Alive Failure Responses	
Classification Input	IP Address only
TLS Context Name	default

3. Configure a Proxy Address Table for Proxy Set for Skype for Business Server 2015:

   a. Go to **Configuration** tab > **VoIP** menu > **VoIP Network** > **Proxy Sets Table** > **Proxy Address Table**.

Parameter	Value
Index	**0**
Proxy Address	**FE.S4B.interop:5067** (Skype for Business Server 2015 IP address / FQDN and destination port)
Transport Type	**TLS**

Figure 4-10: Configuring Proxy Address for Microsoft Skype for Business Server 2015

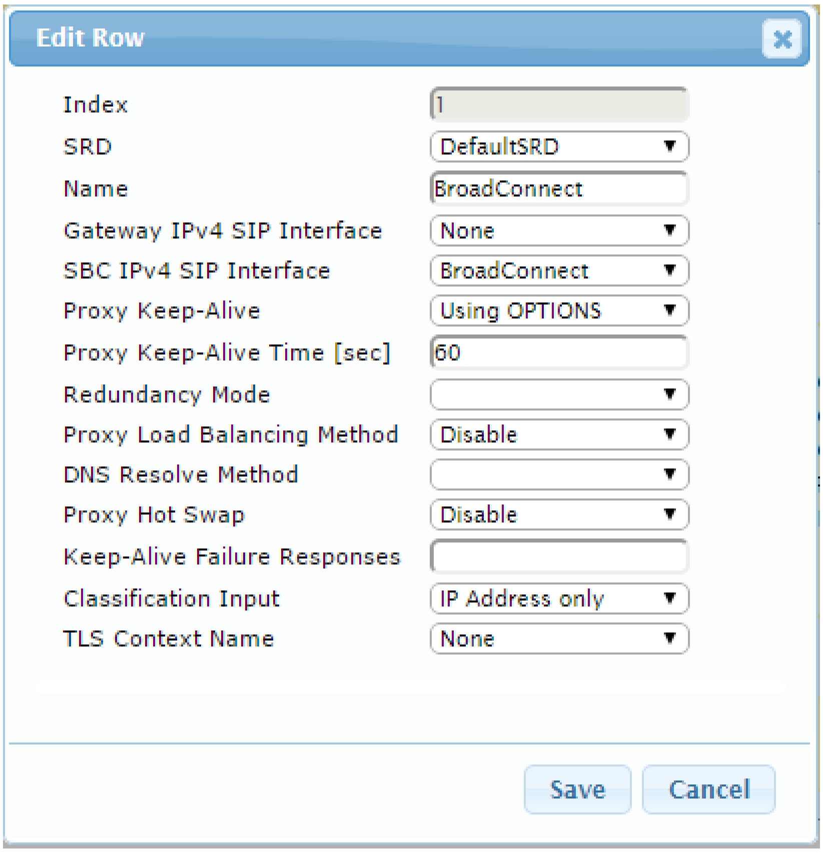


Index	0
Proxy Address	FE.S4B.interop:5067
Transport Type	TLS

4. Configure a Proxy Set for the BroadConnect SIP Trunk:

Parameter	Value
Proxy Set ID	**1**
Proxy Name	**BroadConnect** (see the Note on page 39)
SBC IPv4 SIP Interface	**BroadConnect**
Proxy Keep Alive	**Using Options**

Figure 4-11: Configuring Proxy Set for BroadConnect SIP Trunk



Edit Row

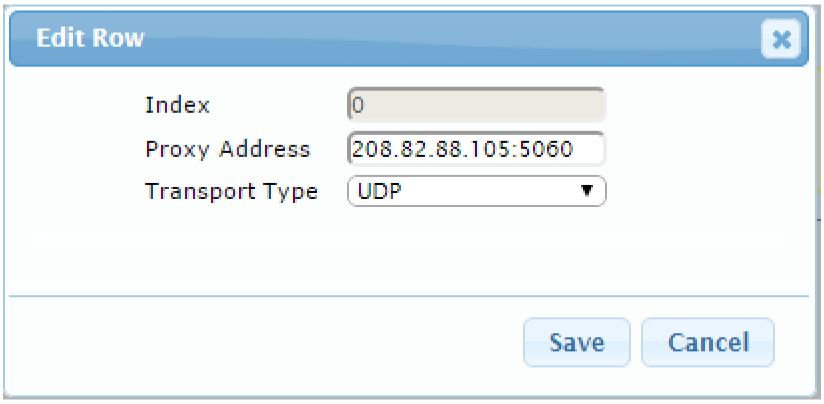
Index	1
SRD	DefaultSRD
Name	BroadConnect
Gateway IPv4 SIP Interface	None
SBC IPv4 SIP Interface	BroadConnect
Proxy Keep-Alive	Using OPTIONS
Proxy Keep-Alive Time [sec]	60
Redundancy Mode	
Proxy Load Balancing Method	Disable
DNS Resolve Method	
Proxy Hot Swap	Disable
Keep-Alive Failure Responses	
Classification Input	IP Address only
TLS Context Name	None

a. Configure a Proxy Address Table for Proxy Set 1:

b. Go to **Configuration** tab > **VoIP** menu > **VoIP Network** > **Proxy Sets Table** > **Proxy Address Table**.

Parameter	Value
Index	**0**
Proxy Address	**208.82.88.105:5060** (IP address / FQDN and destination port)
Transport Type	**UDP**

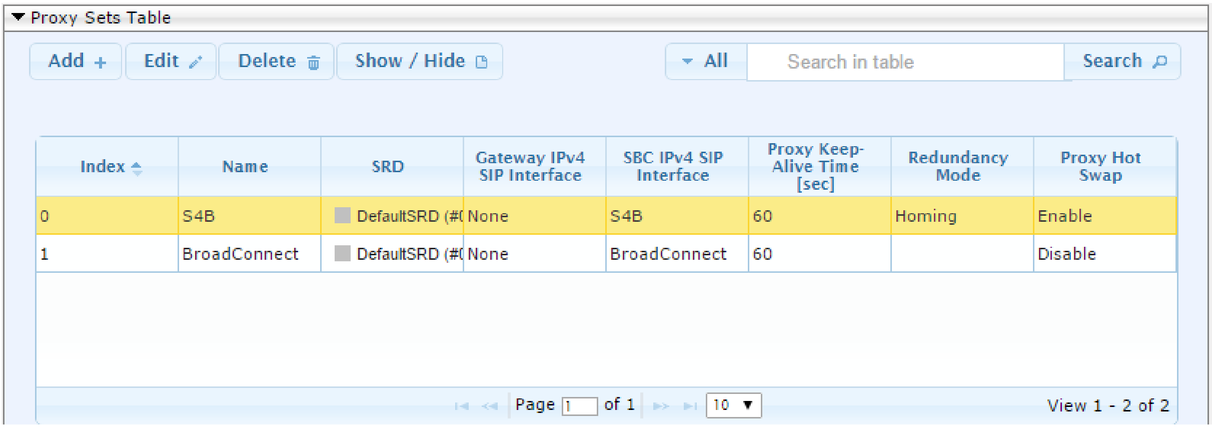
Figure 4-12: Configuring Proxy Address for



Index	0
Proxy Address	208.82.88.105:5060
Transport Type	UDP

The configured Proxy Sets are shown in the figure below:

Figure 4-13**: Configured Proxy Sets in Proxy Sets Table**



Index	Name	SRD	Gateway IPv4 SIP Interface	SBC IPv4 SIP Interface	Proxy Keep-Alive Time [sec]	Redundancy Mode	Proxy Hot Swap
0	S4B	DefaultSRD (#(	None	S4B	60	Homing	Enable
1	BroadConnect	DefaultSRD (#(	None	BroadConnect	60		Disable

4.6 Step 6: Configure IP Profiles

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

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

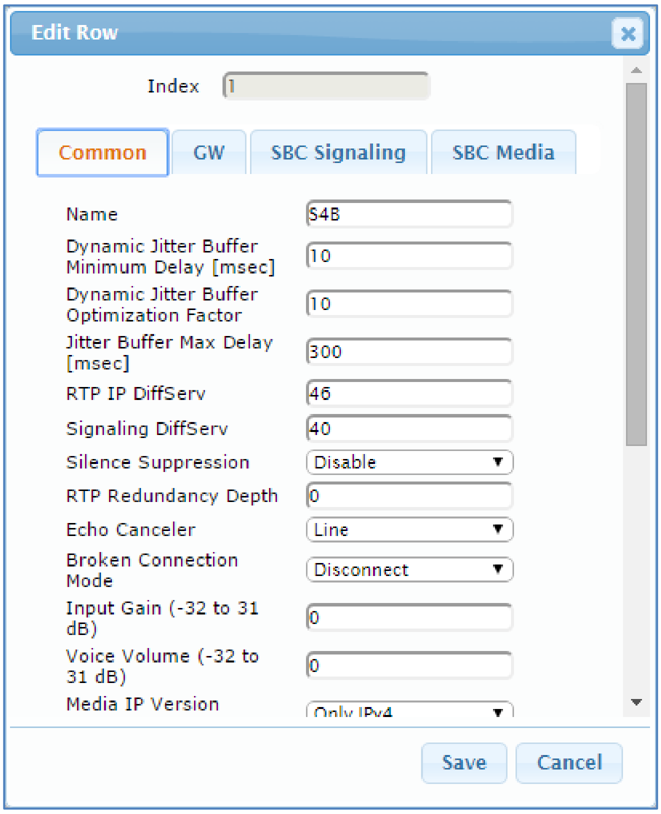
- Microsoft Skype for Business Server 2015 - to operate in secure mode using SRTP and TLS
- BroadConnect SIP trunk - to operate in non-secure mode using RTP and UDP

➢ **To configure IP Profile for the Skype for Business Server 2015:**

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**
Name	**S4B** (see the Note on page 39)
Symmetric MKI	**Enable**
MKI Size	**1**
Reset SRTP State Upon Re-key	**Enable**
Generate SRTP keys mode:	**Always**

Figure 4-14: Configuring IP Profile for Skype for Business Server 2015 – Common Tab

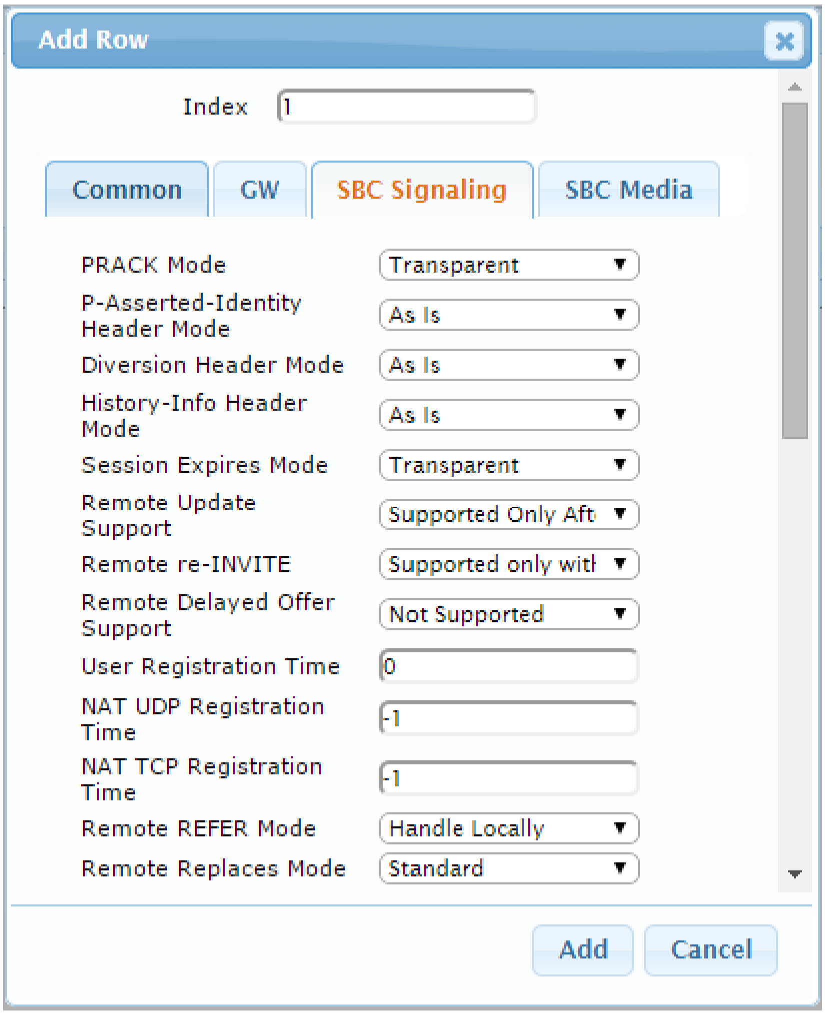


Index: 1
Name: S4B
Dynamic Jitter Buffer Minimum Delay [msec]: 10
Dynamic Jitter Buffer Optimization Factor: 10
Jitter Buffer Max Delay [msec]: 300
RTP IP DiffServ: 46
Signaling DiffServ: 40
Silence Suppression: Disable
RTP Redundancy Depth: 0
Echo Canceler: Line
Broken Connection Mode: Disconnect
Input Gain (-32 to 31 dB): 0
Voice Volume (-32 to 31 dB): 0
Media IP Version: Only IPv4

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

Parameter	Value
Remote Update Support	**Supported Only After Connect**
Remote re-INVITE Support	**Supported Only With SDP**
Remote Delayed Offer Support	**Not Supported**
Remote REFER Mode	**Handle Locally** (required, as Skype for Business Server 2015 does not support receipt of SIP REFER)
Remote 3xx Mode	**Handle Locally** (required, as Skype for Business Server 2015 does not support receipt of SIP 3xx responses)
Remote Early Media RTP Detection Behavior	**By Media** (required, as Skype for Business Server 2015 does not send RTP immediately to the remote side when it sends a SIP 18x response)

Figure 4-15: Configuring IP Profile for Skype for Business Server 2015 – SBC Signaling Tab

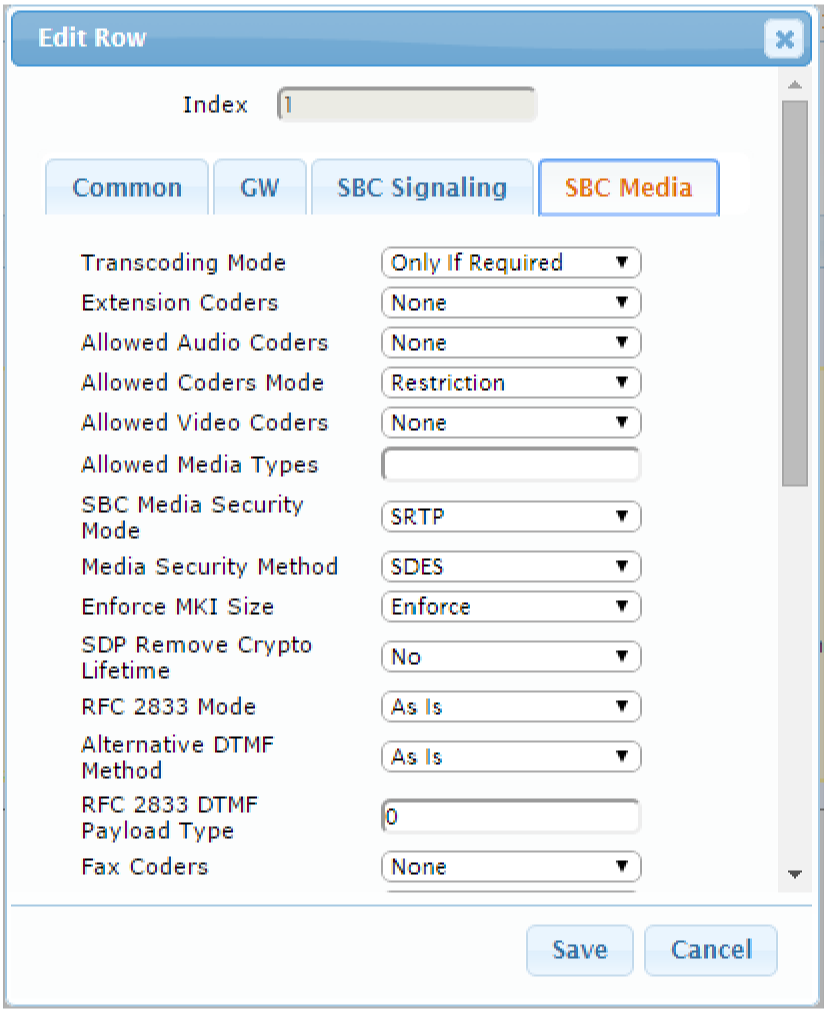


Index	1
PRACK Mode	Transparent
P-Asserted-Identity Header Mode	As Is
Diversion Header Mode	As Is
History-Info Header Mode	As Is
Session Expires Mode	Transparent
Remote Update Support	Supported Only Aft
Remote re-INVITE	Supported only witl
Remote Delayed Offer Support	Not Supported
User Registration Time	0
NAT UDP Registration Time	-1
NAT TCP Registration Time	-1
Remote REFER Mode	Handle Locally
Remote Replaces Mode	Standard

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

Parameter	Value
SBC Media Security Mode	**SRTP**
Enforce MKI Size	**Enforce**
RTCP Mode	**Generate Always** (required, as BroadConnect SIP Trunk does not send RTCP packets in an active call and in a hold call, and in these cases, Microsoft Skype for Business will terminate the call with network problems as the cause)

Figure 4-16: Configuring IP Profile for Skype for Business Server 2015 – SBC Media Tab



Edit Row

Index: 1

Tabs: Common | GW | SBC Signaling | SBC Media

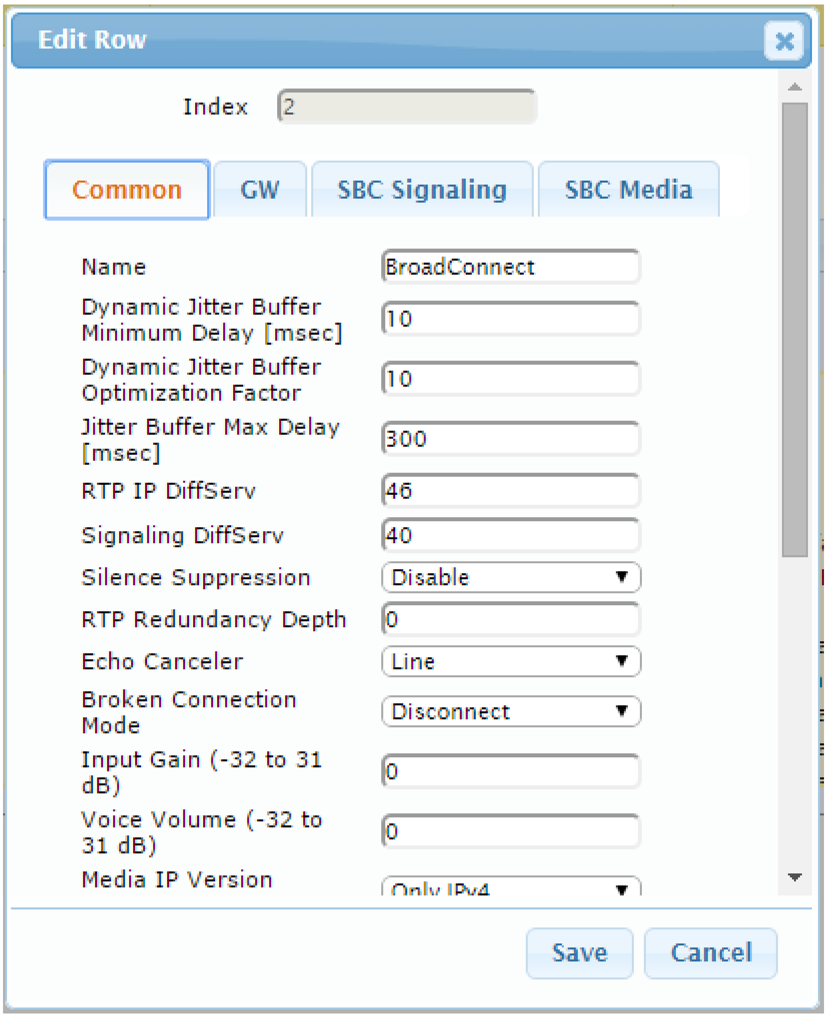
Transcoding Mode: Only If Required
Extension Coders: None
Allowed Audio Coders: None
Allowed Coders Mode: Restriction
Allowed Video Coders: None
Allowed Media Types:
SBC Media Security Mode: SRTP
Media Security Method: SDES
Enforce MKI Size: Enforce
SDP Remove Crypto Lifetime: No
RFC 2833 Mode: As Is
Alternative DTMF Method: As Is
RFC 2833 DTMF Payload Type: 0
Fax Coders: None

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

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

Parameter	Value
Index	**2**
Profile Name	**BroadConnect** (see the Note on page 39)

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



Index 2

Name BroadConnect

Dynamic Jitter Buffer Minimum Delay [msec] 10

Dynamic Jitter Buffer Optimization Factor 10

Jitter Buffer Max Delay [msec] 300

RTP IP DiffServ 46

Signaling DiffServ 40

Silence Suppression Disable

RTP Redundancy Depth 0

Echo Canceler Line

Broken Connection Mode Disconnect

Input Gain (-32 to 31 dB) 0

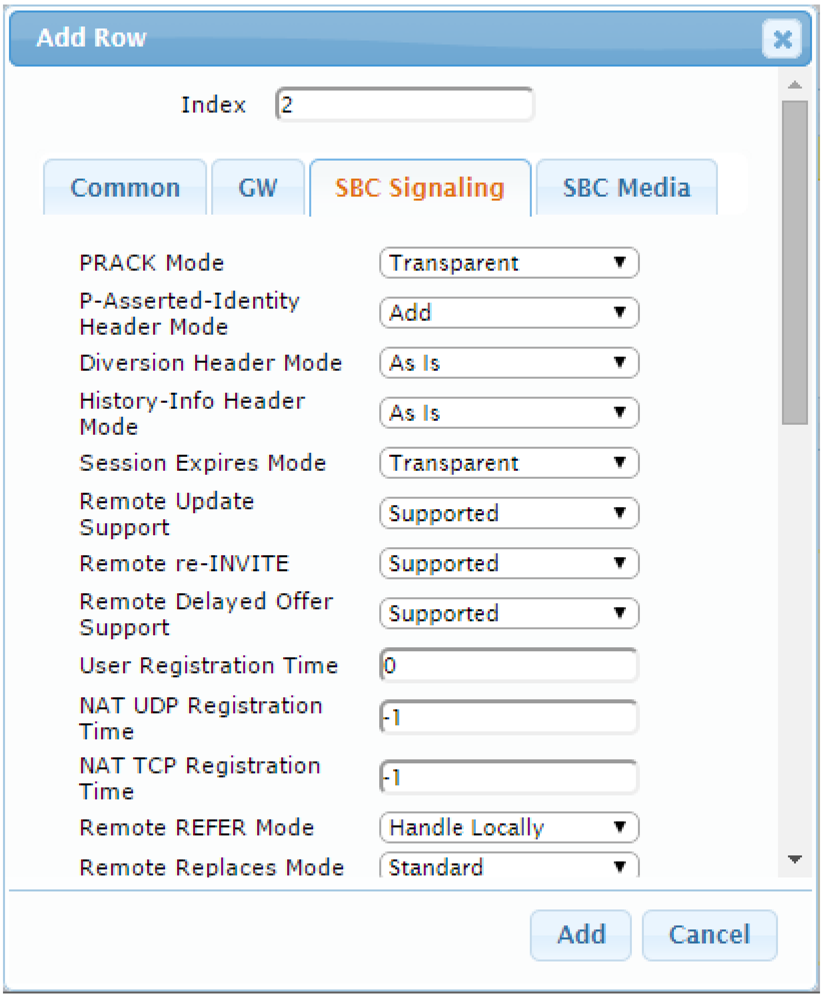
Voice Volume (-32 to 31 dB) 0

Media IP Version Only IPv4

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

Parameter	Value
P-Asserted-Identity Header Mode	**Add** (required for anonymous calls)
Remote REFER Behavior	**Handle Locally** (E-SBC handles / terminates incoming REFER requests instead of forwarding them to SIP Trunk)
Play RBT To Transferee	**Yes** (required for playing ring back tone for transferred calls)

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



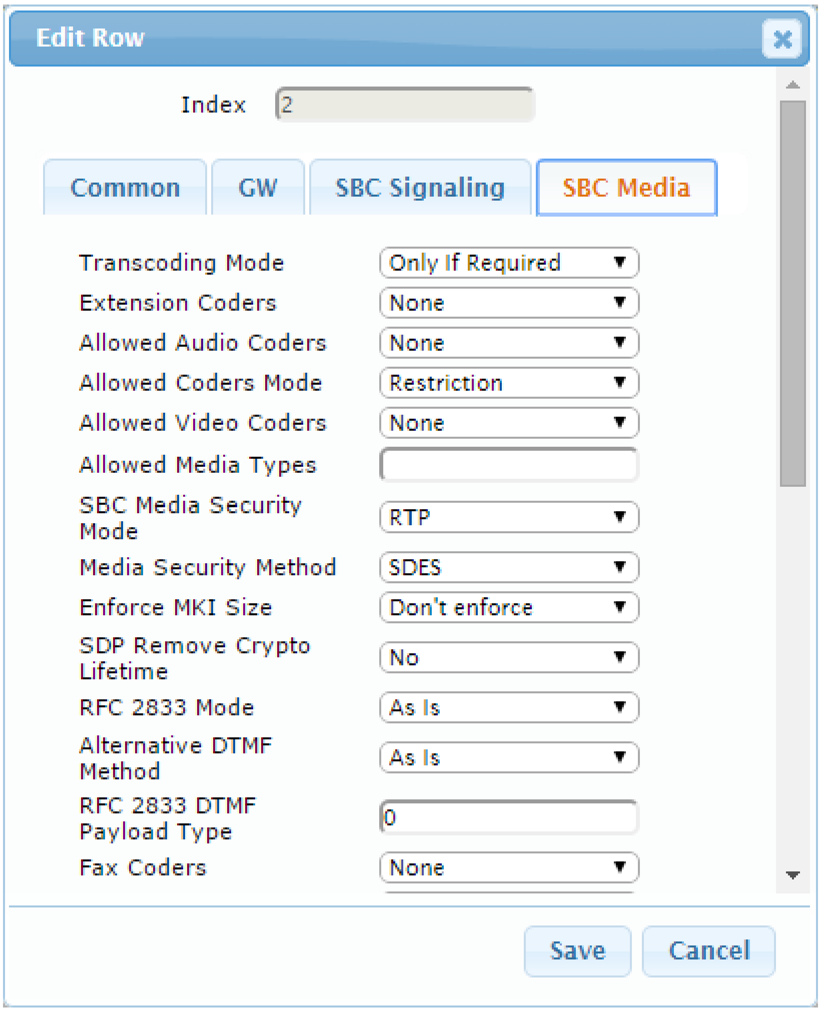
Index: 2

PRACK Mode	Transparent
P-Asserted-Identity Header Mode	Add
Diversion Header Mode	As Is
History-Info Header Mode	As Is
Session Expires Mode	Transparent
Remote Update Support	Supported
Remote re-INVITE	Supported
Remote Delayed Offer Support	Supported
User Registration Time	0
NAT UDP Registration Time	-1
NAT TCP Registration Time	-1
Remote REFER Mode	Handle Locally
Remote Replaces Mode	Standard

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

Parameter	Value
Media Security Behavior	**RTP**

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



Index	2
Transcoding Mode	Only If Required
Extension Coders	None
Allowed Audio Coders	None
Allowed Coders Mode	Restriction
Allowed Video Coders	None
Allowed Media Types	
SBC Media Security Mode	RTP
Media Security Method	SDES
Enforce MKI Size	Don't enforce
SDP Remove Crypto Lifetime	No
RFC 2833 Mode	As Is
Alternative DTMF Method	As Is
RFC 2833 DTMF Payload Type	0
Fax Coders	None

4.7 Step 7: Configure IP Groups

This step shows how to configure IP Groups. The IP Group represents an IP entity on the network with which the E-SBC communicates. This can be a server (e.g., IP PBX or ITSP) or it can be a group of users (e.g., LAN IP phones). For servers, the IP Group is typically used to define the server's IP address by associating it with a Proxy Set. After 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:

- Skype for Business Server 2015 (Mediation Server) located on the LAN
- BroadConnect SIP Trunk located on the WAN

➢ **To configure IP Groups:**

1. Open the IP Group Table page (**Configuration** tab > **VoIP** menu > **VoIP Network** > **IP Group Table**).
2. Add an IP Group for the Skype for Business Server 2015. You can use the default IP Group (Index 0), but modify it as shown below:

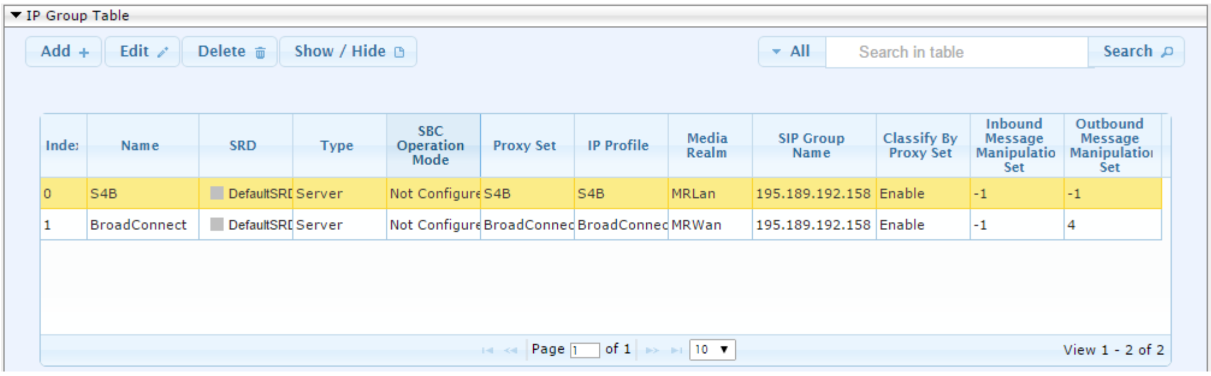
Parameter	Value
Index	**0**
Name	**S4B** (see the Note on page 39)
Type	**Server**
Proxy Set	**S4B**
IP Profile	**S4B**
Media Realm	**MRLan**
SIP Group Name	(According to ITSP requirements)

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

Parameter	Value
Index	**1**
Name	**BroadConnect** (see the Note on page 39)
Type	**Server**
Proxy Set	**BroadConnect**
IP Profile	**BroadConnect**
Media Realm	**MRWan**
SIP Group Name	(According to ITSP requirements)

The configured IP Groups are shown in the figure below:

Figure 4-20: Configured IP Groups in IP Group Table



Inde:	Name	SRD	Type	SBC Operation Mode	Proxy Set	IP Profile	Media Realm	SIP Group Name	Classify By Proxy Set	Inbound Message Manipulatio Set	Outbound Message Manipulatio Set
0	S4B	DefaultSRI	Server	Not Configure	S4B	S4B	MRLan	195.189.192.158	Enable	-1	-1
1	BroadConnect	DefaultSRI	Server	Not Configure	BroadConnec	BroadConnec	MRWan	195.189.192.158	Enable	-1	4

4.8 Step 8: Configure Coders

This step shows how to configure coders (termed *Coder Group*). As Skype for Business Server 2015 supports the G.711 coder while the network connection to BroadConnect 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 BroadConnect SIP Trunk.

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

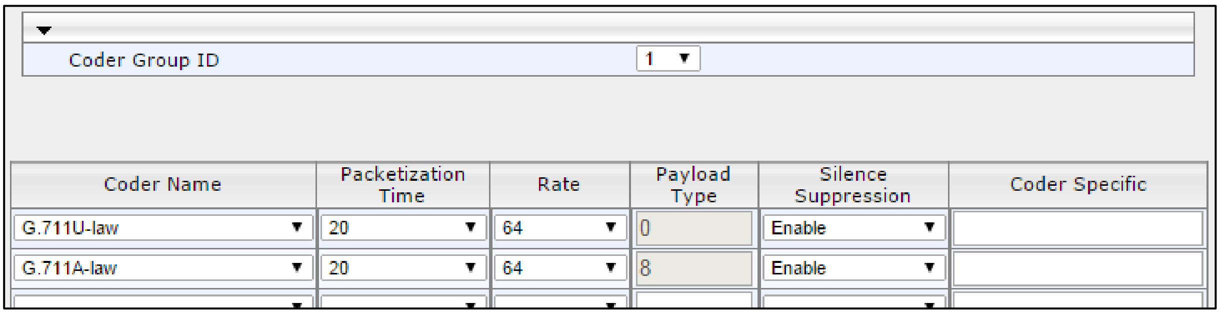
Note: This step is required *only* if operation with a lower bandwidth coder is required.

➢ **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 Skype for Business Server 2015:

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

Figure 4-21: Configuring Coder Group for Skype for Business Server 2015



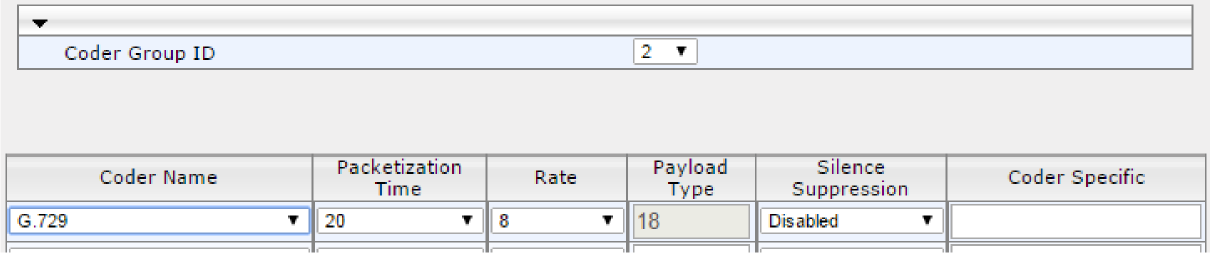
Coder Group ID: 1

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

3. Configure a Coder Group for BroadConnect SIP Trunk:

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

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



Coder Group ID: 2

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

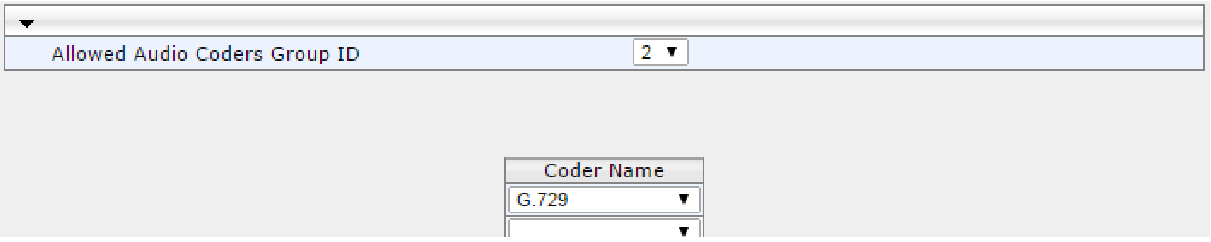
The procedure below describes how to configure an Allowed Coders Group to ensure that voice sent to the BroadConnect SIP Trunk uses the G.729 coder whenever possible. Note that this Allowed Coders Group ID should be assigned to the IP Profile belonging to the BroadConnect SIP Trunk (see Section 4.6 on page 44).

➢ To set a preferred coder for the BroadConnect SIP Trunk:

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

Parameter	Value
Allowed Audio Coders Group ID	**2**
Coder Name	**G.729**

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

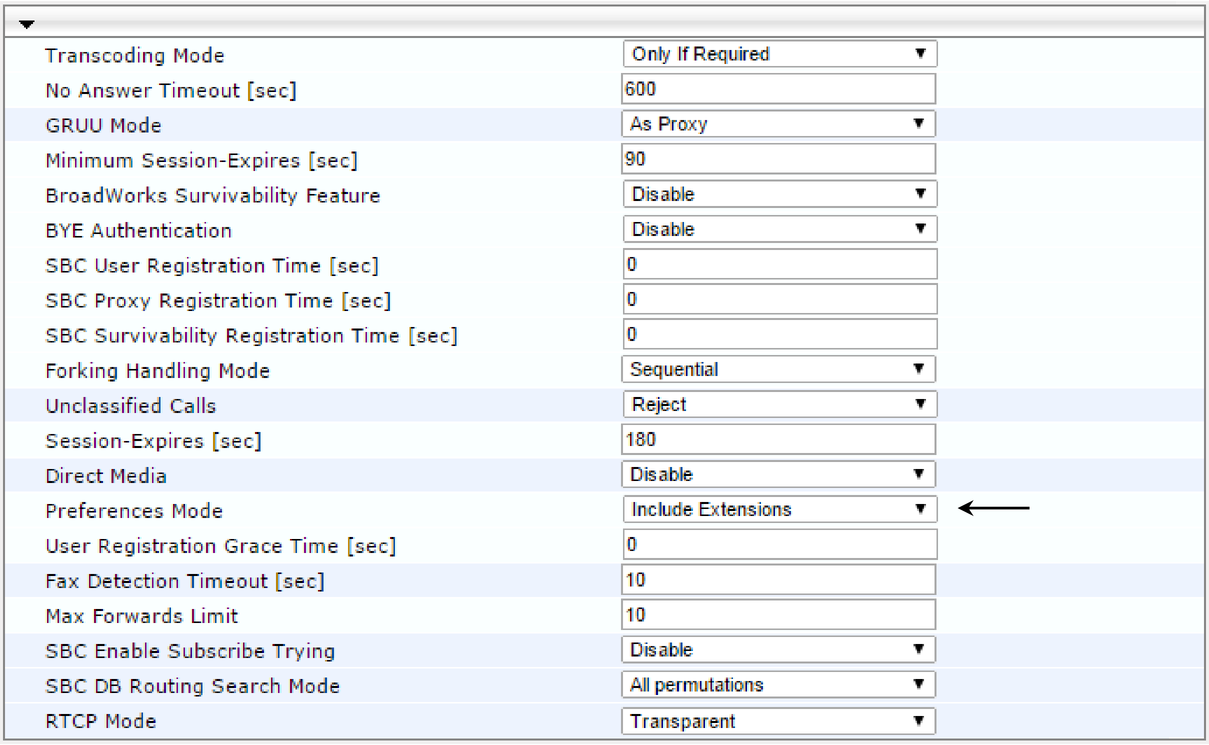


Allowed Audio Coders Group ID: 2

Coder Name
G.729

3. 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
SBC User Registration Time [sec]	0
SBC Proxy Registration Time [sec]	0
SBC 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
Max Forwards Limit	10
SBC Enable Subscribe Trying	Disable
SBC DB Routing Search Mode	All permutations
RTCP Mode	Transparent

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

4.9 Step 9: SIP TLS Connection Configuration

This section describes how to configure the E-SBC for using a TLS connection with the Skype for Business Server 2015 Mediation Server. This is essential for a secure SIP TLS connection.

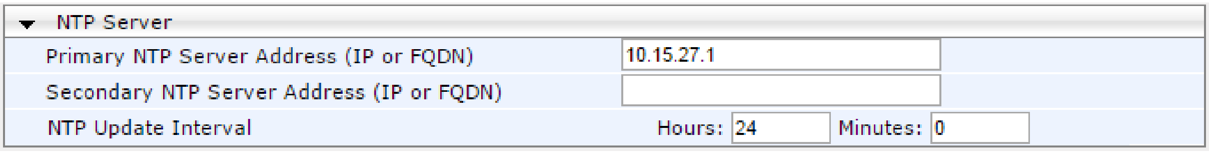
4.9.1 Step 9a: Configure the NTP Server Address

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

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

1. Open the Application Settings page (**Configuration** tab > **System** > **Time And Day**).
2. In the 'NTP Server Address' field, enter the IP address of the NTP server (e.g., **10.15.27.1**).

Figure 4-25: Configuring NTP Server Address



NTP Server

Primary NTP Server Address (IP or FQDN)	10.15.27.1		
Secondary NTP Server Address (IP or FQDN)			
NTP Update Interval	Hours: 24	Minutes: 0	

3. Click **Submit**.

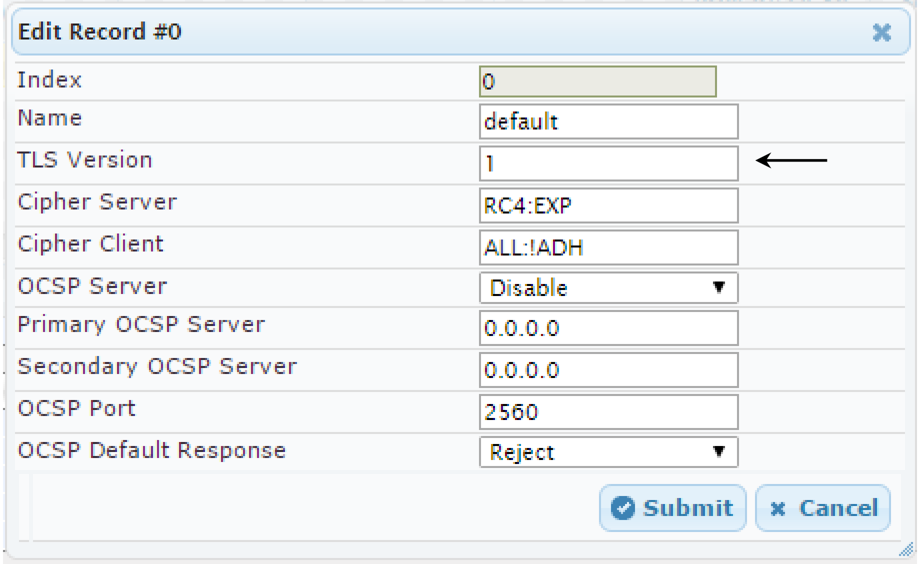
4.9.2 Step 9b: Configure the TLS Version 1.0

This step shows how to configure the E-SBC to use TLS version 1.0 only. AudioCodes recommends implementing only TLS to avoid flaws in SSL.

➢ **To configure the TLS, version 1.0:**

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 (usually default index 0 will be used), and then click 'Edit'.
3. In the 'TLS Version' field, enter **1**.

Figure 4-26: Configuring TLS version 1.0



Index	0
Name	default
TLS Version	1
Cipher Server	RC4:EXP
Cipher Client	ALL:!ADH
OCSP Server	Disable
Primary OCSP Server	0.0.0.0
Secondary OCSP Server	0.0.0.0
OCSP Port	2560
OCSP Default Response	Reject

4. Click **Submit**.

4.9.3 Step 9c: Configure a Certificate

This step shows how to exchange a certificate with Microsoft Certificate Authority (CA). The certificate is used by the E-SBC to authenticate the connection with Skype for Business Server 2015.

The procedure involves the following main steps:

a. Generating a Certificate Signing Request (CSR).

b. Requesting Device Certificate from CA.

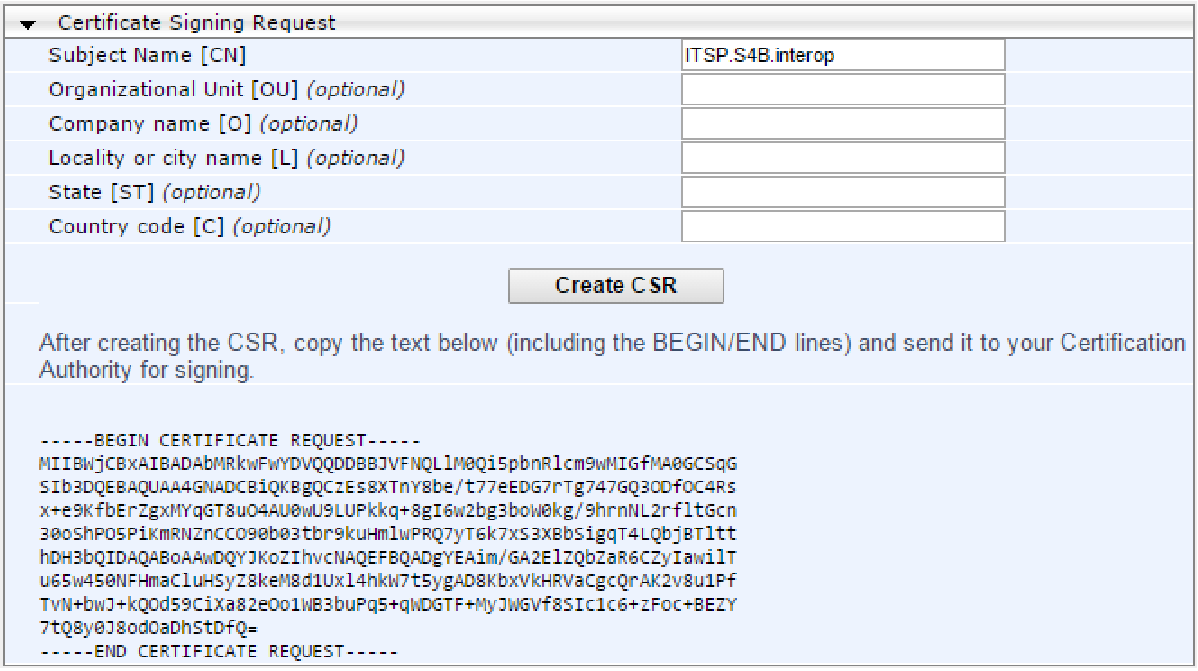
c. Obtaining Trusted Root Certificate from CA.

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 (usually default index 0 will be used), and then click the **TLS 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.S4B.interop**).
   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 4-27: Certificate Signing Request – Creating CSR



▾ Certificate Signing Request

Subject Name [CN]	ITSP.S4B.interop
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.

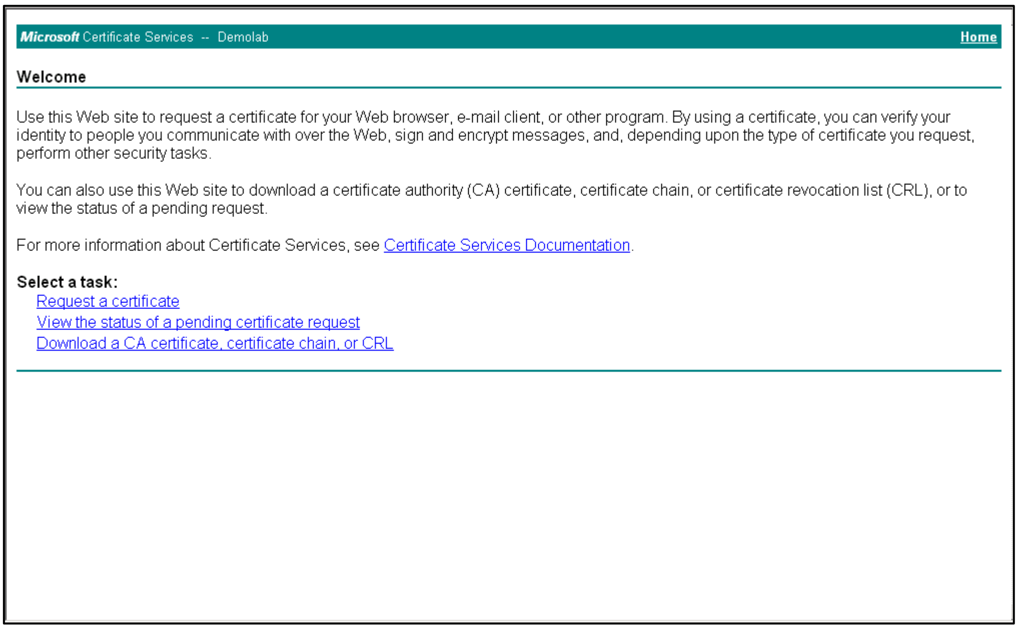
```
-----BEGIN CERTIFICATE REQUEST-----
MIIBWjCBxAIBADAbMRkwFwYDVQQDDBBJVFNQL1M0Qi5pbnRlcm9wMIGfMA0GCSqG
SIb3DQEBAQUAA4GNADCBiQKBgQCzEs8XTnY8be/t77eEDG7rTg747GQ3ODfOC4Rs
x+e9KfbErZgxMYqGT8uO4AU0wU9LUPkkq+8gI6w2bg3boW0kg/9hrnNL2rfltGcn
30oShPO5PiKmRNZnCCO90b03tbr9kuHmlwPRQ7yT6k7xS3XBbSigqT4LQbjBTltt
hDH3bQIDAQABoAAwDQYJKoZIhvcNAQEFBQADgYEAim/GA2ElZQbZaR6CZyIawilT
u65w450NFHmaCluHSyZ8keM8d1Uxl4hkW7t5ygAD8KbxVkHRVaCgcQrAK2v8u1Pf
TvN+bwJ+kQOd59CiXa82eOo1WB3buPq5+qWDGTF+MyJWGVf8SIc1c6+zFoc+BEZY
7tQ8y0J8odOaDhStDfQ=
-----END CERTIFICATE REQUEST-----
```

Note: The value entered in this field must be identical to the gateway name configured in the Topology Builder for Skype for Business Server 2015 (see Section 3.1).

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 4-28: 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.

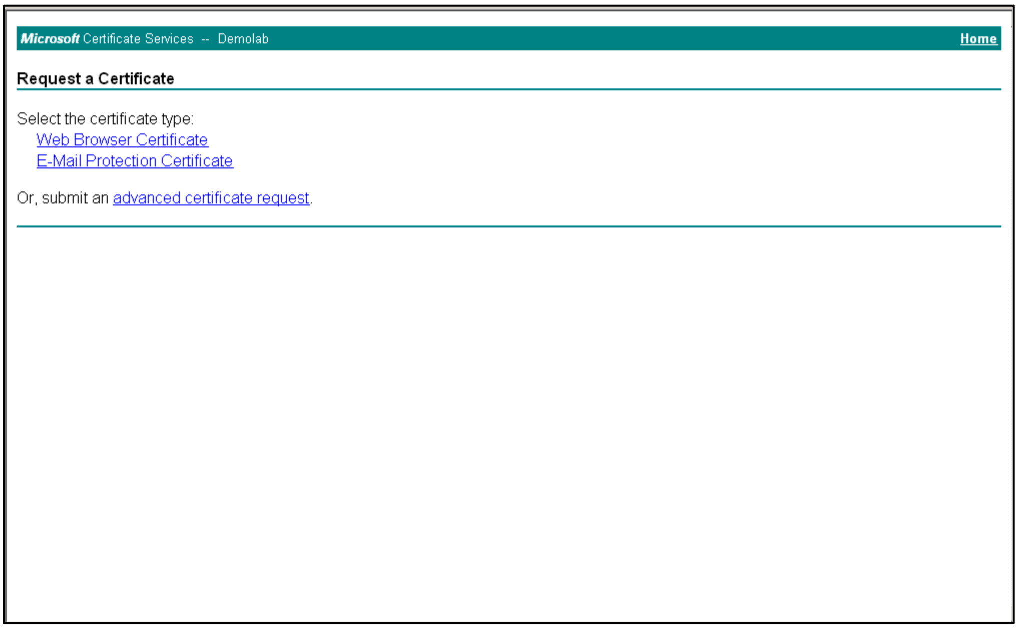
For more information about Certificate Services, see Certificate Services Documentation.

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 4-29: Request a Certificate Page



Microsoft Certificate Services -- Demolab

Home

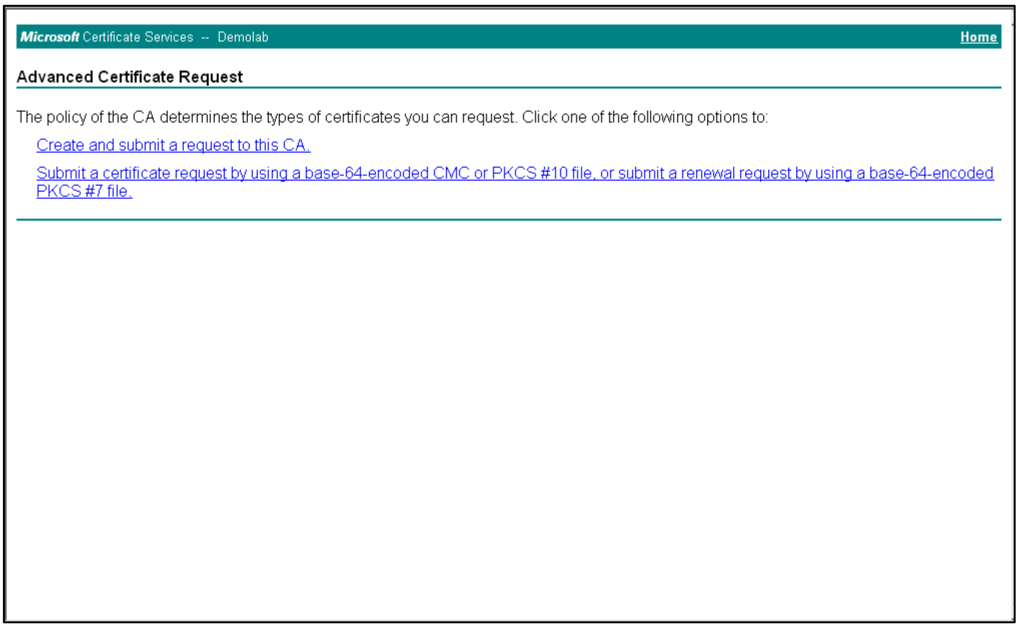
Request a Certificate

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 4-30: Advanced Certificate Request Page



Microsoft Certificate Services -- Demolab **Home**

Advanced Certificate Request

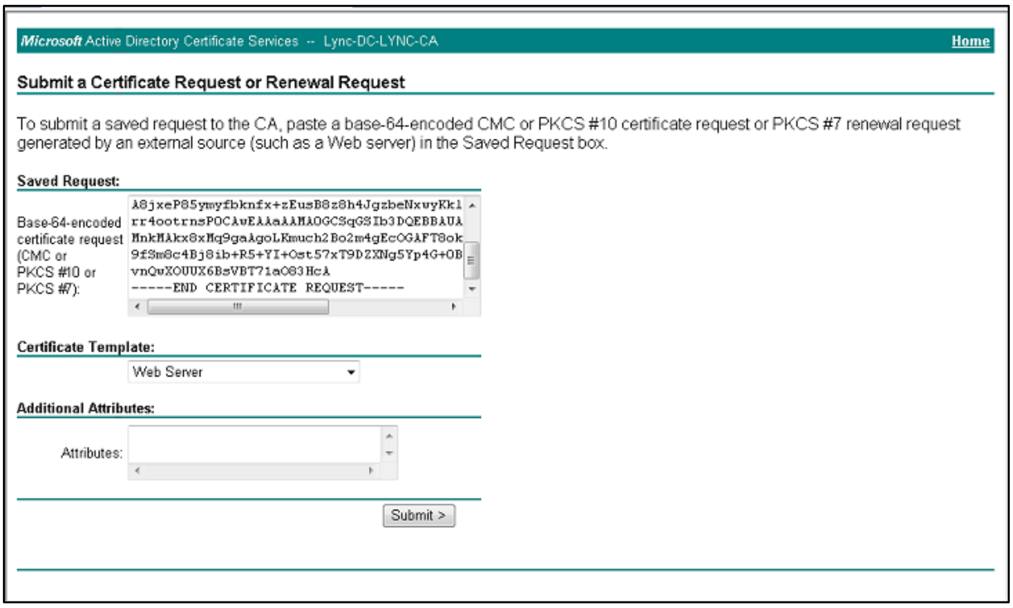
The policy of the CA determines the types of certificates you can request. Click one of the following options to:

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 4-31: Submit a Certificate Request or Renewal Request Page



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

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:

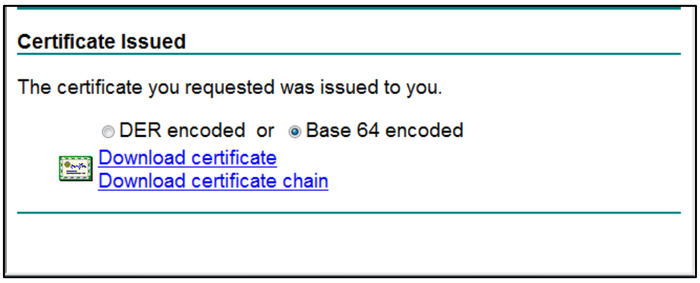
Attributes:

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 4-32: Certificate Issued Page



Certificate Issued

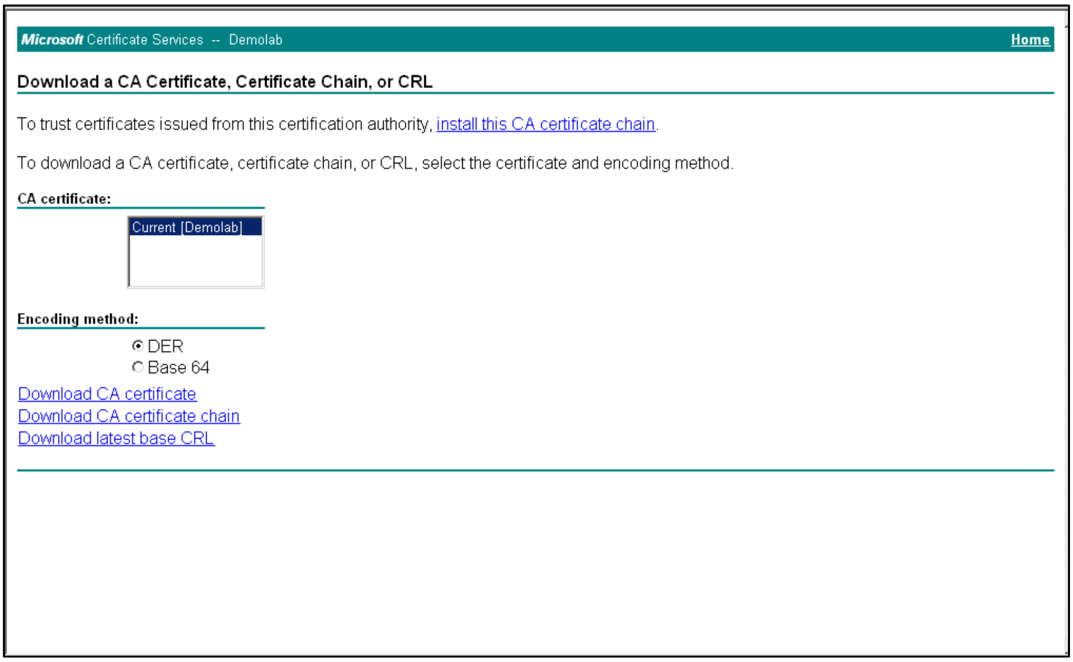
The certificate you requested was issued to you.

○ 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 4-33: 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

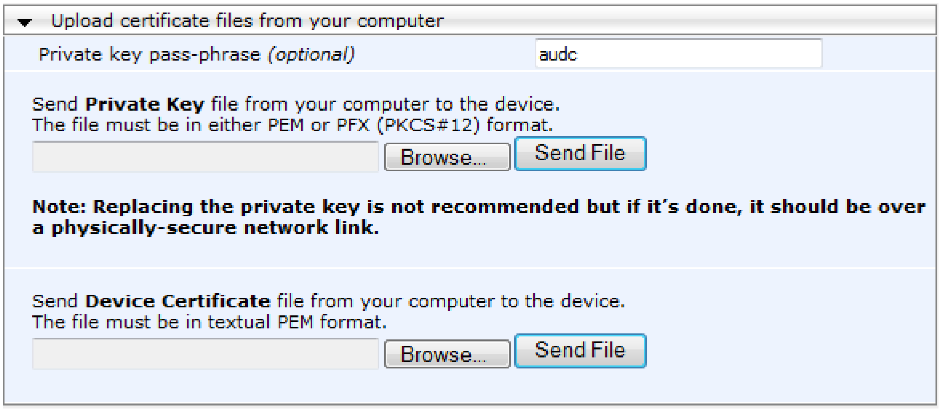
17. Under the 'Encoding method' group, select the **Base 64** option for encoding.
18. Click **Download CA certificate**.
19. Save the file as *certroot.cer* to a folder on your computer.

20. In the E-SBC's Web interface, return to the **TLS Contexts** page and do the following:

    a. In the TLS Contexts table, select the required TLS Context index row (typically, the default TLS Context at Index 0 is used), and then click the **TLS Context Certificates** button, located at the bottom of the TLS Contexts page; the Context Certificates page appears.

    b. 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 4-34: 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.

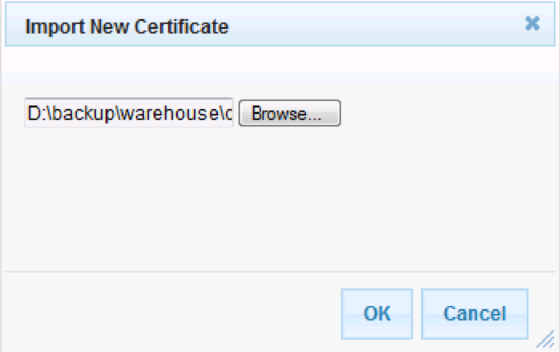
Send **Device Certificate** file from your computer to the device.
The file must be in textual PEM format.

    c. In the E-SBC's Web interface, return to the **TLS Contexts** page.

    d. 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.

    e. Click the **Import** button, and then select the certificate file to load.

Figure 4-35: 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 82).

4.10 Step 10: Configure SRTP

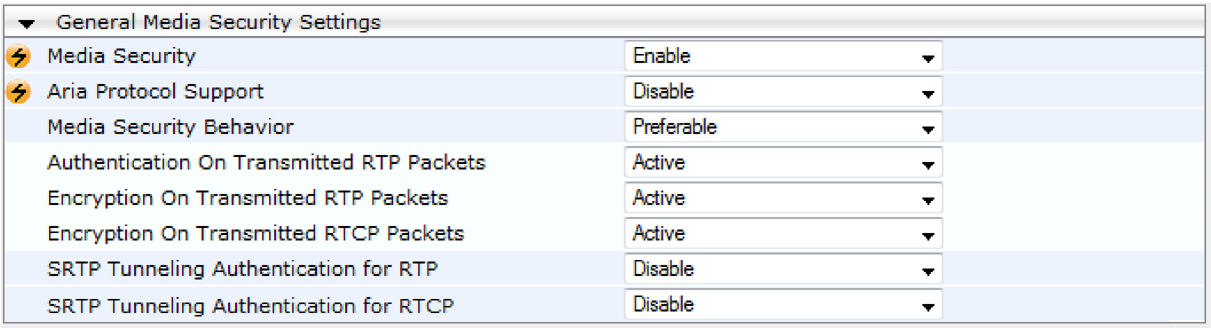
This step shows how to configure media security. If you configure the Microsoft Mediation Server to use SRTP, you need to configure the E-SBC to operate in the same manner. Note that SRTP was enabled for Skype for Business Server 2015 when you configured an IP Profile for Skype for Business Server 2015 (see Section 4.6 on page 44).

➢ **To configure media security:**

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

Parameter	Value
Media Security	**Enable**

Figure 4-36: Configuring SRTP



Media Security	Enable
Aria Protocol Support	Disable
Media Security Behavior	Preferable
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 82).

4.11 Step 11: Configure Maximum IP Media Channels

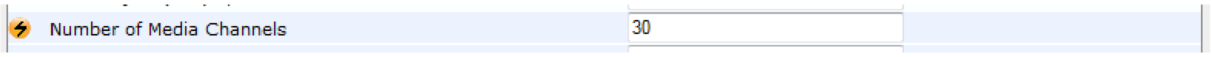
This step shows how to configure the maximum number of required IP media channels. The number of media channels represents the number of DSP channels that the E-SBC allocates to call sessions.

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

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

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

Figure 4-37: Configuring Number of Media Channels



Number of Media Channels 30

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 82).

4.12 Step 12: Configure IP-to-IP Call Routing Rules

This step shows how to configure IP-to-IP call routing rules. These rules define the routes for forwarding SIP messages (e.g., INVITE) received from one IP entity to another. The E-SBC selects the rule whose configured input characteristics (e.g., IP Group) match those of the incoming SIP message. If the input characteristics do not match the first rule in the table, they are compared to the second rule, and so on, until a matching rule is located. If no rule is matched, the message is rejected. The routing rules use the configured IP Groups to denote the source and destination of the call. As configured in Section 4.7 on page 43, IP Group 1 represents Skype for Business Server 2015, and IP Group 2 represents BroadConnect SIP Trunk.

For the interoperability test topology, the following IP-to-IP routing rules need to be configured to route calls between Skype for Business Server 2015 (LAN) and BroadConnect SIP Trunk (WAN):

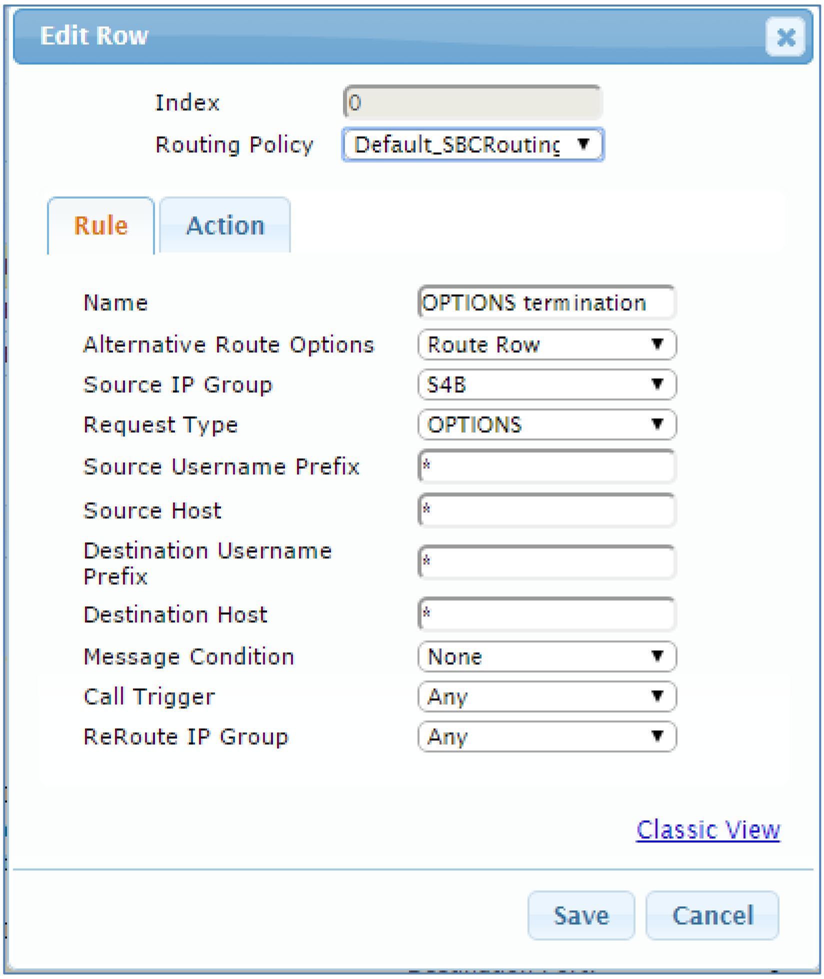
- Terminate SIP OPTIONS messages on the E-SBC that are received from the LAN
- Calls from Skype for Business Server 2015 to BroadConnect SIP Trunk
- Calls from BroadConnect SIP Trunk to Skype for Business Server 2015

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

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

Parameter	Value
Index	**0**
Name	**OPTIONS termination** (arbitrary descriptive name)
Source IP Group	**S4B**
Request Type	**OPTIONS**

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

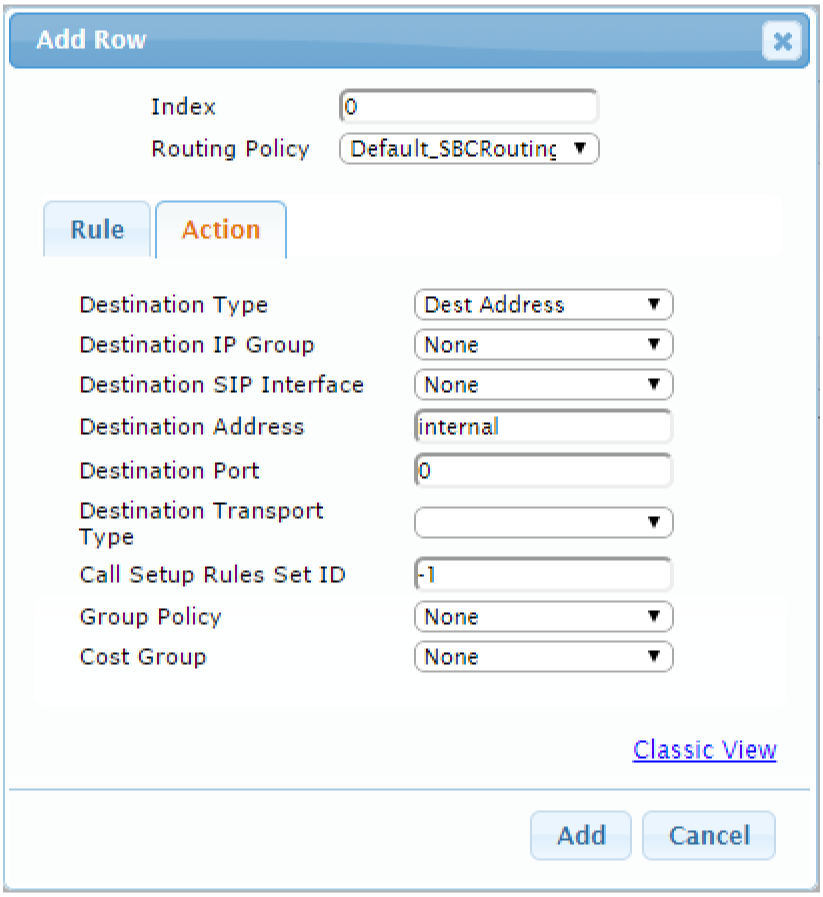


Index	0
Routing Policy	Default_SBCRouting
Name	OPTIONS termination
Alternative Route Options	Route Row
Source IP Group	S4B
Request Type	OPTIONS
Source Username Prefix	*
Source Host	*
Destination Username Prefix	*
Destination Host	*
Message Condition	None
Call Trigger	Any
ReRoute IP Group	Any

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

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

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

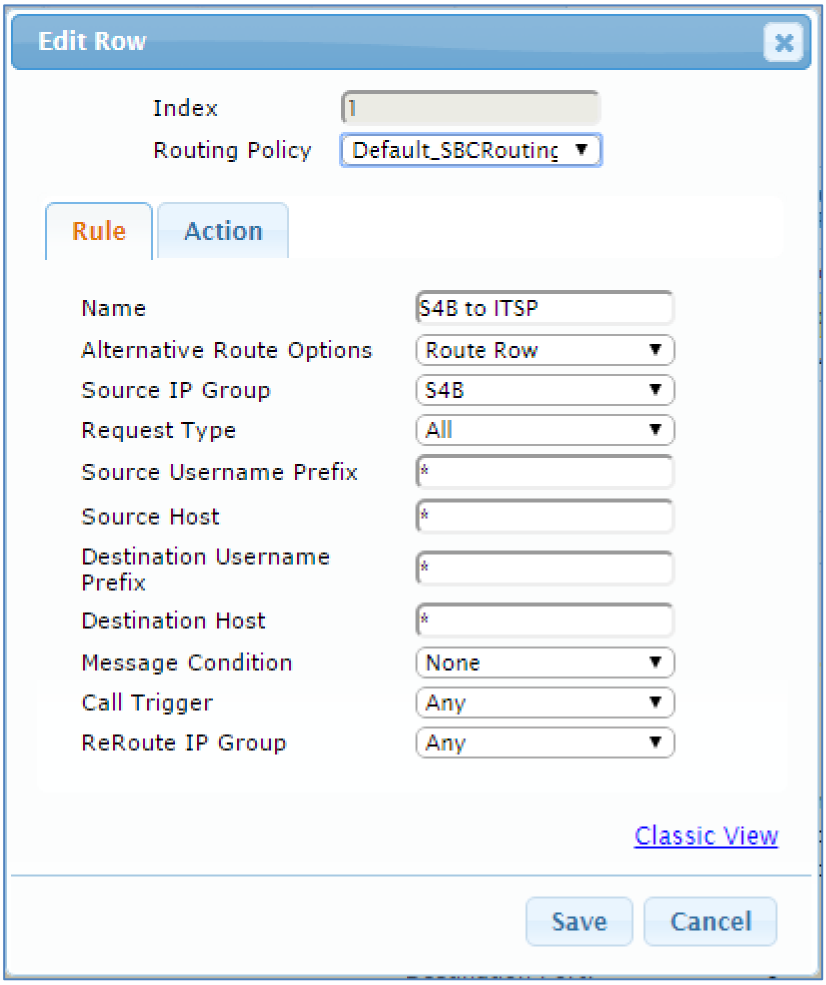


Index	0
Routing Policy	Default_SBCRouting
Destination Type	Dest Address
Destination IP Group	None
Destination SIP Interface	None
Destination Address	internal
Destination Port	0
Destination Transport Type	
Call Setup Rules Set ID	-1
Group Policy	None
Cost Group	None

3. Configure a rule to route calls from Skype for Business Server 2015 to BroadConnect SIP Trunk:
   a. Click **Add**.
   b. Click the **Rule** tab, and then configure the parameters as follows:

Parameter	Value
Index	**1**
Route Name	**S4B to ITSP** (arbitrary descriptive name)
Source IP Group	**S4B**

Figure 4-40: Configuring IP-to-IP Routing Rule for S4B to ITSP – Rule tab



Edit Row

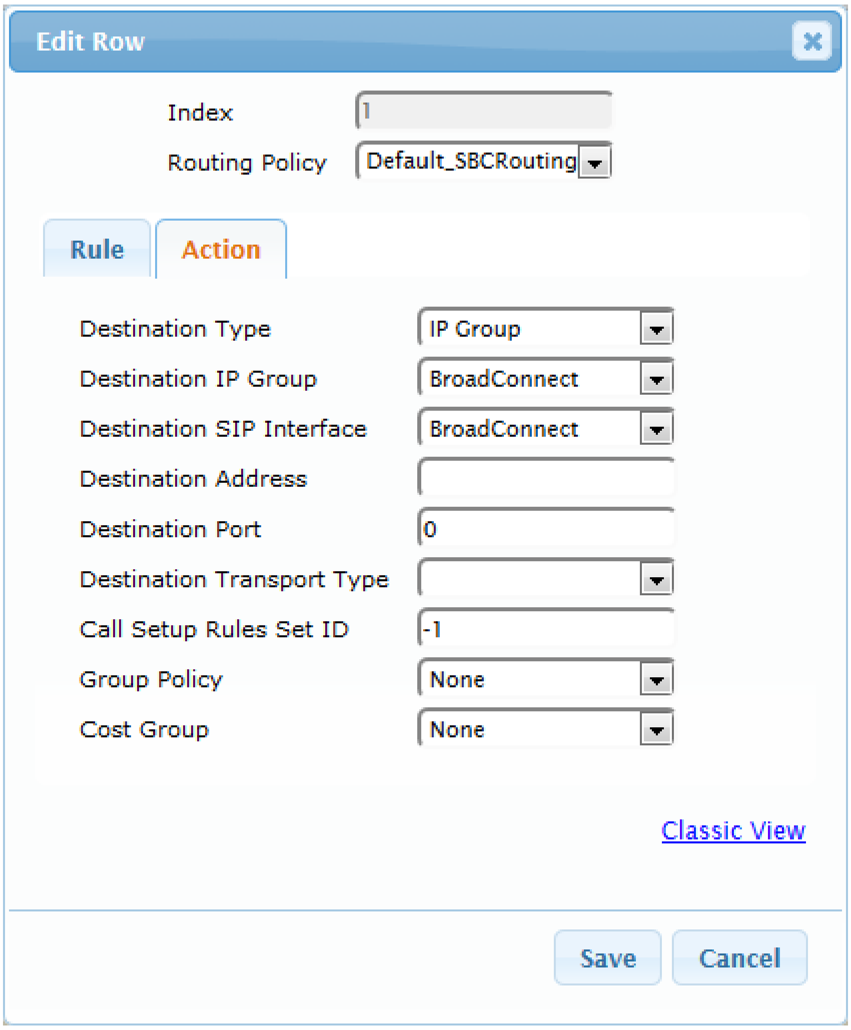
Index	1
Routing Policy	Default_SBCRouting

Name	S4B to ITSP
Alternative Route Options	Route Row
Source IP Group	S4B
Request Type	All
Source Username Prefix	*
Source Host	*
Destination Username Prefix	*
Destination Host	*
Message Condition	None
Call Trigger	Any
ReRoute IP Group	Any

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

Parameter	Value
Destination Type	**IP Group**
Destination IP Group	**BroadConnect**
Destination SIP Interface	**BroadConnect**

Figure 4-41: Configuring IP-to-IP Routing Rule for S4B to ITSP – Action tab



Index	1
Routing Policy	Default_SBCRouting
Destination Type	IP Group
Destination IP Group	BroadConnect
Destination SIP Interface	BroadConnect
Destination Address	
Destination Port	0
Destination Transport Type	
Call Setup Rules Set ID	-1
Group Policy	None
Cost Group	None

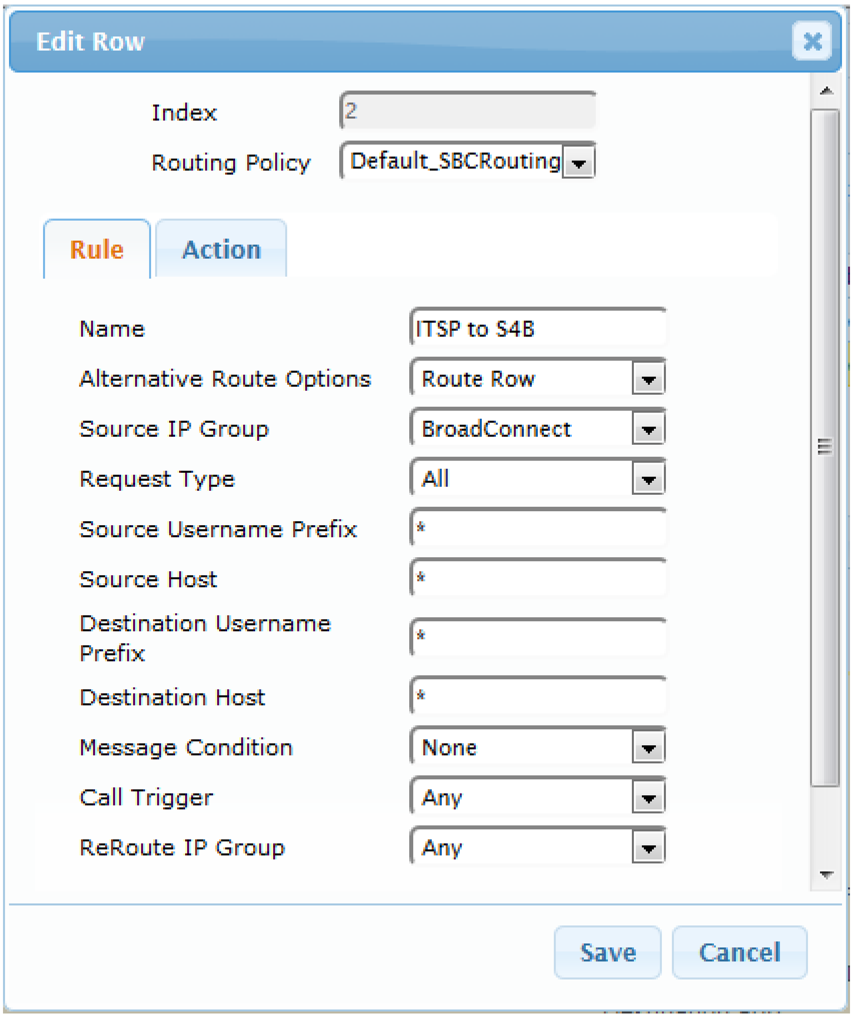
4. To configure rule to route calls from BroadConnect SIP Trunk to Skype for Business Server 2015:

   a. Click **Add**.

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

Parameter	Value
Index	**2**
Route Name	**ITSP to S4B** (arbitrary descriptive name)
Source IP Group	**BroadConnect**

Figure 4-42: Configuring IP-to-IP Routing Rule for ITSP to S4B – Rule tab



Edit Row

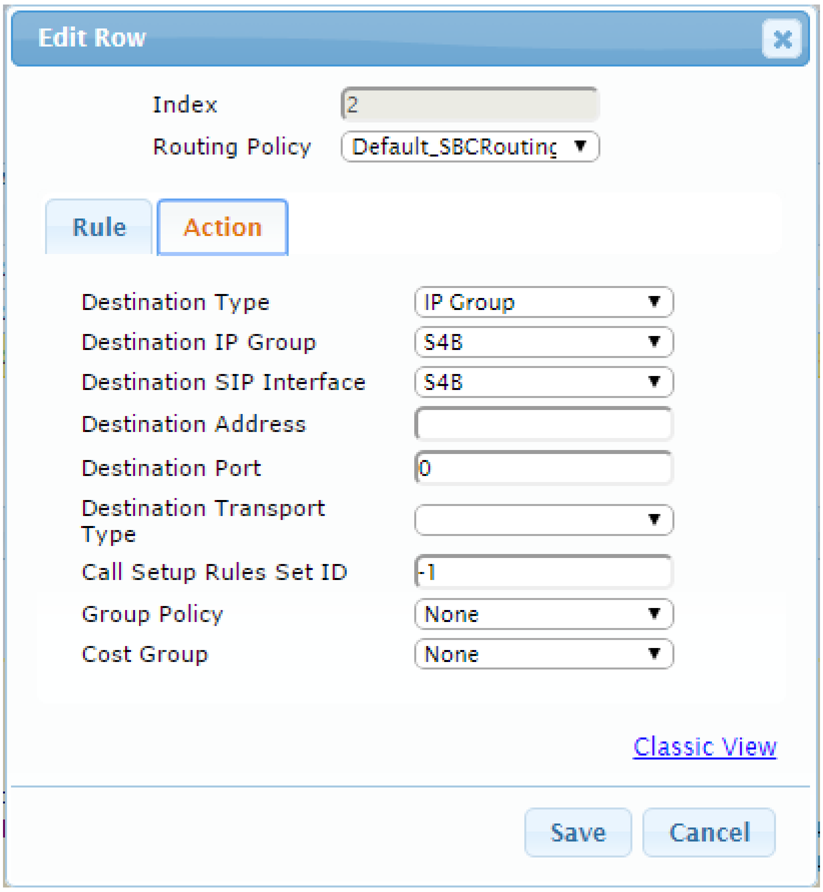
Index	2
Routing Policy	Default_SBCRouting

Name	ITSP to S4B
Alternative Route Options	Route Row
Source IP Group	BroadConnect
Request Type	All
Source Username Prefix	*
Source Host	*
Destination Username Prefix	*
Destination Host	*
Message Condition	None
Call Trigger	Any
ReRoute IP Group	Any

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

Parameter	Value
Destination Type	**IP Group**
Destination IP Group	**S4B**
Destination SIP Interface	**S4B**

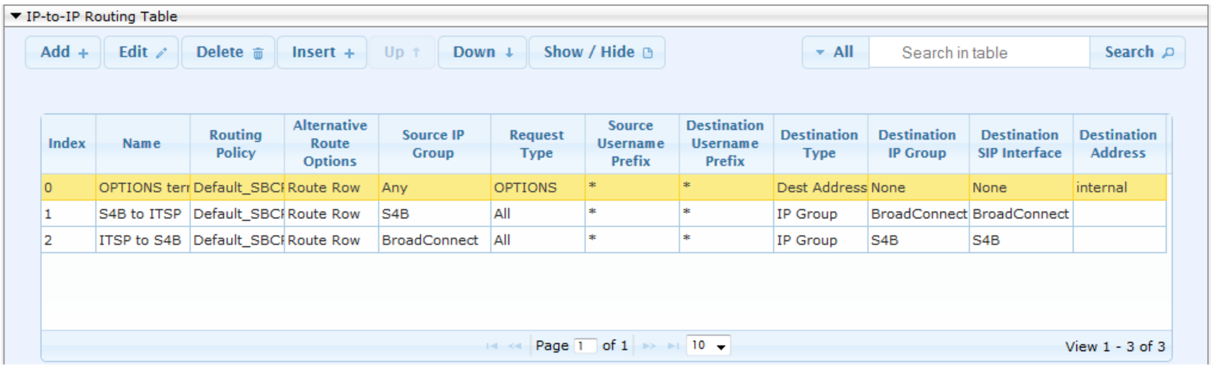
Figure 4-43: Configuring IP-to-IP Routing Rule for ITSP to S4B – Action tab



Index	2
Routing Policy	Default_SBCRouting
Destination Type	IP Group
Destination IP Group	S4B
Destination SIP Interface	S4B
Destination Address	
Destination Port	0
Destination Transport Type	
Call Setup Rules Set ID	-1
Group Policy	None
Cost Group	None

The configured routing rules are shown in the figure below:

Figure 4-44: Configured IP-to-IP Routing Rules in IP-to-IP Routing Table



Index	Name	Routing Policy	Alternative Route Options	Source IP Group	Request Type	Source Username Prefix	Destination Username Prefix	Destination Type	Destination IP Group	Destination SIP Interface	Destination Address
0	OPTIONS terı	Default_SBCI	Route Row	Any	OPTIONS	*	*	Dest Address	None	None	internal
1	S4B to ITSP	Default_SBCI	Route Row	S4B	All	*	*	IP Group	BroadConnect	BroadConnect	
2	ITSP to S4B	Default_SBCI	Route Row	BroadConnect	All	*	*	IP Group	S4B	S4B	

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

4.13 Step 13: Configure IP-to-IP Manipulation Rules

This step shows how to configure IP-to-IP manipulation rules. These rules manipulate the source and / or destination number. The manipulation rules use the configured IP Groups to denote the source and destination of the call. As configured in Section 4.7 on page 43, IP Group 0 represents Skype for Business Server 2015, and IP Group 1 represents BroadConnect SIP Trunk.

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

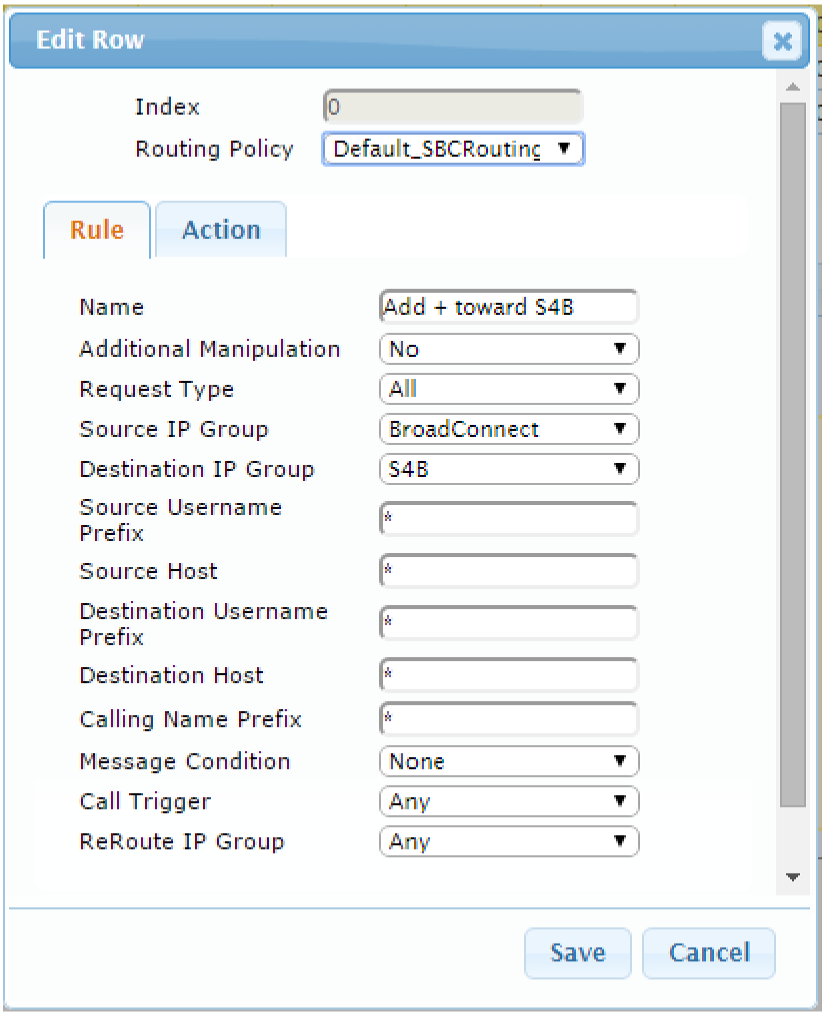
For this interoperability test topology, a manipulation is configured to add the "+" (plus sign) to the destination number for calls from the BroadConnect SIP Trunk IP Group to the Skype for Business Server 2015 IP Group for any destination username prefix.

➢ **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	**0**
Name	**Add + toward S4B**
Source IP Group	**BroadConnect**
Destination IP Group	**S4B**
Destination Username Prefix	***** (asterisk sign)

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



Index: 0

Routing Policy: Default_SBCRouting

Name: Add + toward S4B

Additional Manipulation: No

Request Type: All

Source IP Group: BroadConnect

Destination IP Group: S4B

Source Username Prefix: *

Source Host: *

Destination Username Prefix: *

Destination Host: *

Calling Name Prefix: *

Message Condition: None

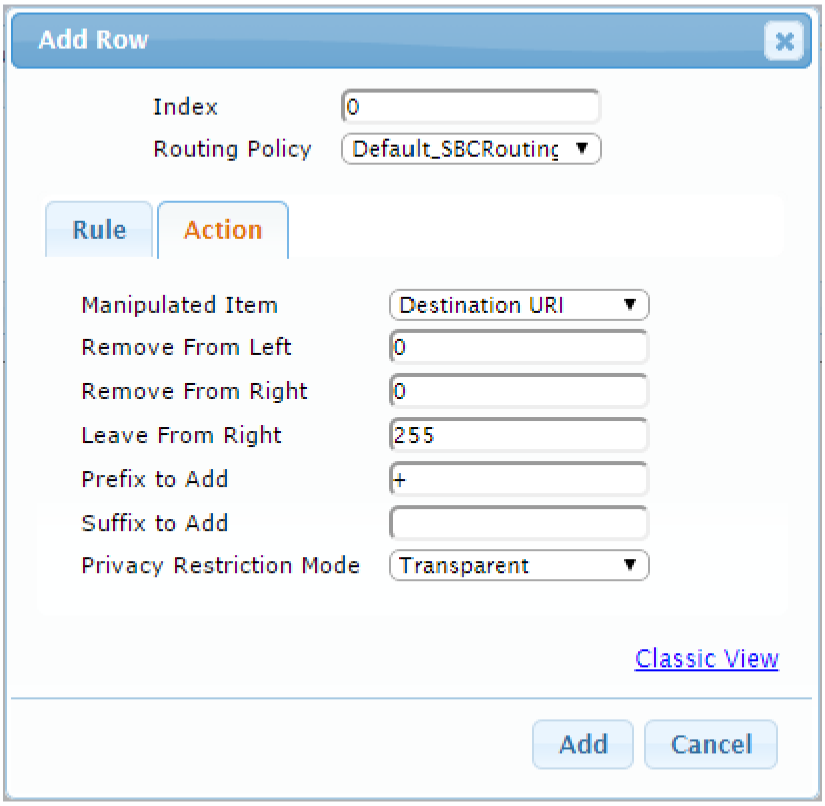
Call Trigger: Any

ReRoute IP Group: 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 4-46: Configuring IP-to-IP Outbound Manipulation Rule - Action Tab

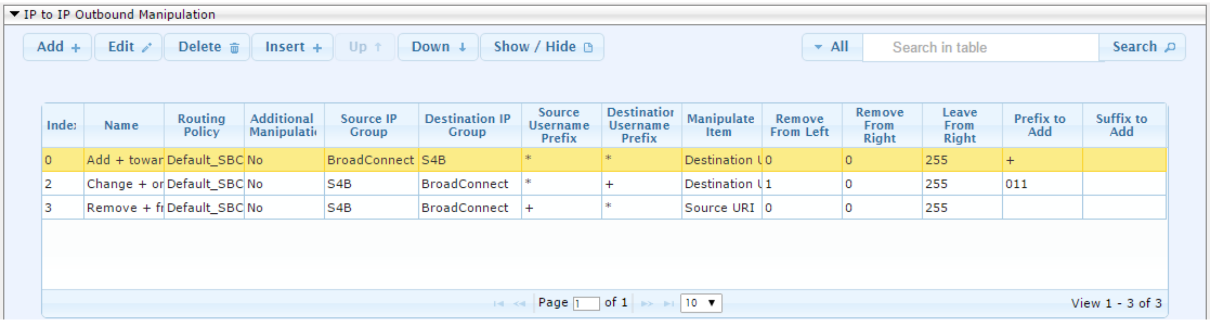


Index	0
Routing Policy	Default_SBCRouting ▼
Manipulated Item	Destination URI ▼
Remove From Left	0
Remove From Right	0
Leave From Right	255
Prefix to Add	+
Suffix to Add	
Privacy Restriction Mode	Transparent ▼

5. Click **Submit**.

The figure below shows an example of configured IP-to-IP outbound manipulation rules for calls between Skype for Business Server 2015 IP Group and BroadConnect SIP Trunk IP Group:

Figure 4-47: Example of Configured IP-to-IP Outbound Manipulation Rules



Inde:	Name	Routing Policy	Additional Manipulati	Source IP Group	Destination IP Group	Source Username Prefix	Destinatior Username Prefix	Manipulate Item	Remove From Left	Remove From Right	Leave From Right	Prefix to Add	Suffix to Add
0	Add + towar	Default_SBC	No	BroadConnect	S4B	*	*	Destination U	0	0	255	+	
2	Change + or	Default_SBC	No	S4B	BroadConnect	*	+	Destination U	1	0	255	011	
3	Remove + fr	Default_SBC	No	S4B	BroadConnect	+	*	Source URI	0	0	255		

Rule Index	Description
1	Calls from ITSP IP Group to S4B IP Group with any destination number (*), add "+" to the prefix of the destination number.
2	Calls from S4B IP Group to ITSP IP Group with the prefix destination number "+", remove "+" from this prefix and add “011” (for international dialing).
3	Calls from S4B IP Group to ITSP IP Group with source number prefix "+", remove the "+" from this prefix.

4.14 Step 14: Configure Message Manipulation Rules

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

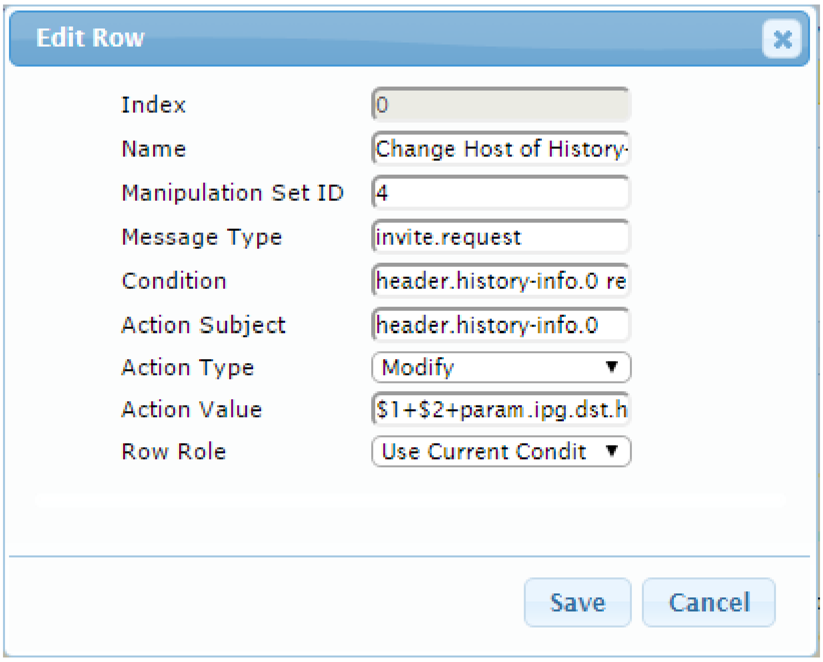
Once you have configured the SIP message manipulation rules, you need to assign them to the relevant IP Group (in the IP Group table) and determine whether they must be applied to inbound or outbound messages.

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

1. Open the Message Manipulations page (**Configuration** tab > **VoIP** menu > **SIP Definitions** > **Msg Policy & Manipulation** > **Message Manipulations**).
2. Configure a new manipulation rule (Manipulation Set 4) for BroadConnect SIP Trunk. This rule applies to messages sent to the BroadConnect SIP Trunk IP Group in a call forwarding scenario. This rule replaces the host part of the SIP History-Info Header with the value that was configured in the BroadConnect SIP Trunk IP Group.

Parameter	Value
Index	**0**
Name	**Change Host of History-Info.0**
Manipulation Set ID	**4**
Message Type	**invite.request**
Condition	**header.history-info.0 regex (.*)(@)(.*)(;user=phone)(.*)**
Action Subject	**header.history-info.0**
Action Type	**Modify**
Action Value	**$1+$2+param.ipg.dst.host+$4+$5**

Figure 4-48: Configuring SIP Message Manipulation Rule 0 (for BroadConnect SIP Trunk)

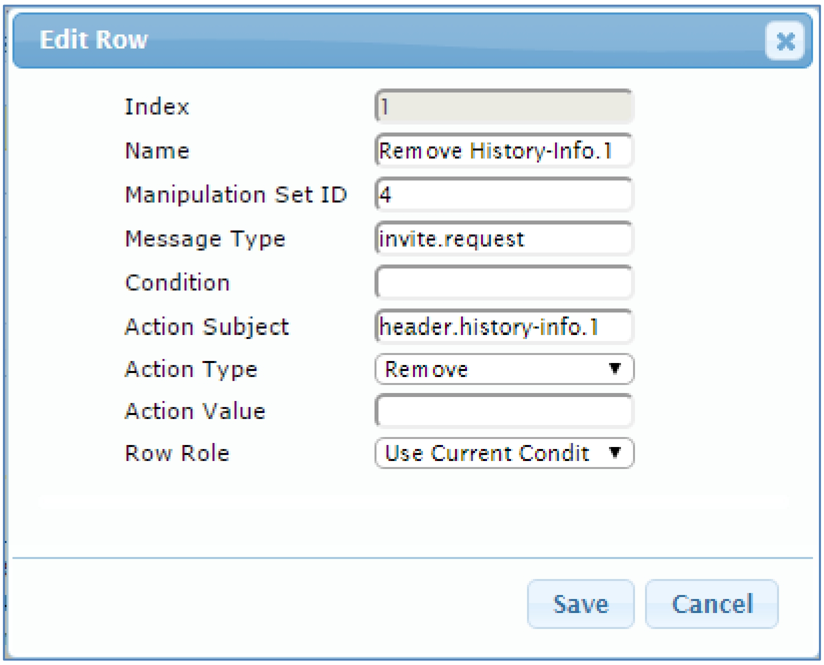


Index	0
Name	Change Host of History-
Manipulation Set ID	4
Message Type	invite.request
Condition	header.history-info.0 re
Action Subject	header.history-info.0
Action Type	Modify ▼
Action Value	$1+$2+param.ipg.dst.h
Row Role	Use Current Condit ▼

3. Configure another manipulation rule (Manipulation Set 4) for the BroadConnect SIP Trunk. This rule also applies to messages sent to the BroadConnect SIP Trunk IP Group in a call forwarding scenario. This rule removes SIP History-Info.1 Header.

Parameter	Value
Index	**1**
Name	**Remove History-Info.1**
Manipulation Set ID	**4**
Message Type	**invite.request**
Action Subject	**header.history-info.1**
Action Type	**Remove**

Figure 4-49: Configuring SIP Message Manipulation Rule 1 (for BroadConnect SIP Trunk)



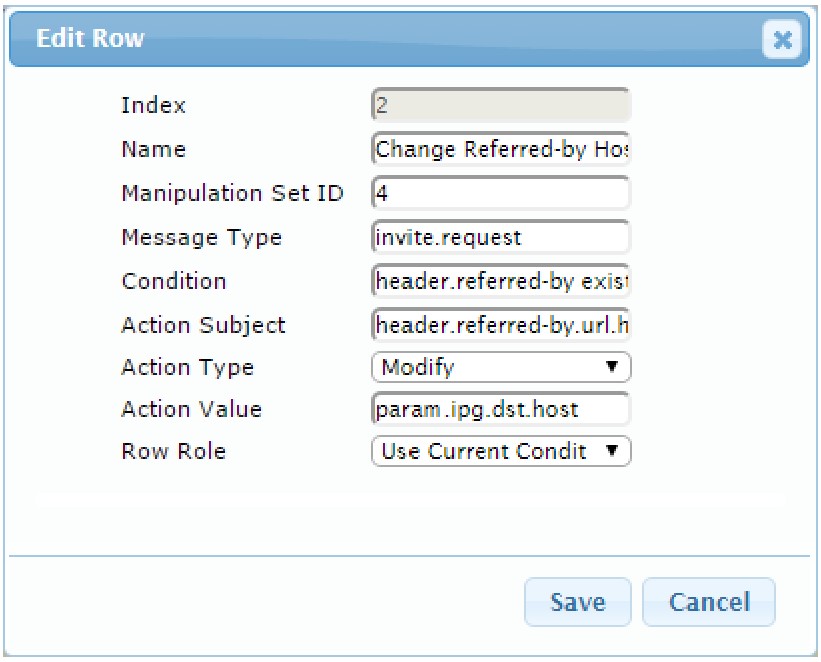
Edit Row

Index	1
Name	Remove History-Info.1
Manipulation Set ID	4
Message Type	invite.request
Condition	
Action Subject	header.history-info.1
Action Type	Remove ▼
Action Value	
Row Role	Use Current Condit ▼

4. Configure another manipulation rule (Manipulation Set 4) for the BroadConnect SIP Trunk. This rule applies to messages sent to the BroadConnect SIP Trunk IP Group in a call transfer scenario. This rule replaces the host part of the SIP Referred-by Header with the value that was configured in the BroadConnect SIP Trunk IP Group.

Parameter	Value
Index	**2**
Name	**Change Referred-by Host**
Manipulation Set ID	**4**
Message Type	**invite.request**
Condition	**header.referred-by exists**
Action Subject	**header.referred-by.url.host**
Action Type	**Modify**
Action Value	**param.ipg.dst.host**

Figure 4-50: Configuring SIP Message Manipulation Rule 2 (for BroadConnect SIP Trunk)



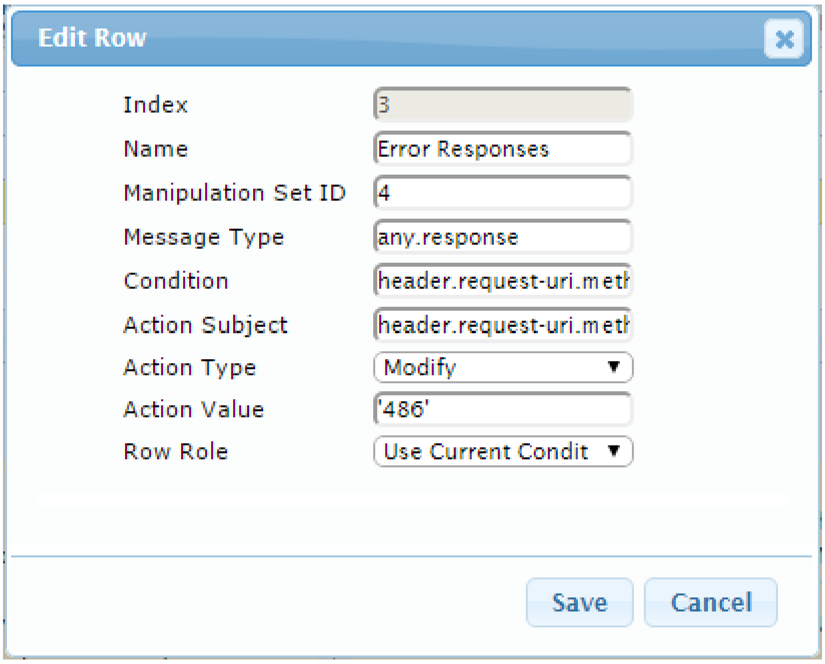
Edit Row

Index	2
Name	Change Referred-by Ho:
Manipulation Set ID	4
Message Type	invite.request
Condition	header.referred-by exist
Action Subject	header.referred-by.url.h
Action Type	Modify ▼
Action Value	param.ipg.dst.host
Row Role	Use Current Condit ▼

5. Configure another manipulation rule (Manipulation Set 4) for BroadConnect SIP Trunk. This rule is applied to response messages sent to the BroadConnect SIP Trunk IP Group for Rejected Calls initiated by the Skype for Business Server 2015 IP Group. This replaces the method type ‘403’ or ‘406’ or ‘480’, with the value ‘486’, because BroadConnect SIP Trunk does not recognize these method types.

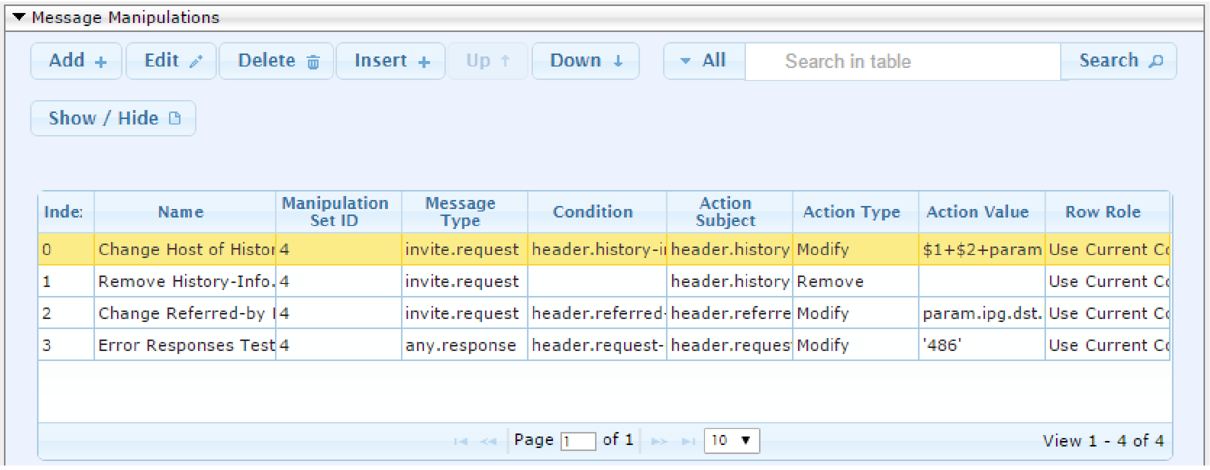
Parameter	Value
Index	**3**
Name	**Reject Cause**
Manipulation Set ID	**4**
Message Type	**any.response**
Condition	**header.request-uri.methodtype=='403' OR header.request-uri.methodtype=='406' OR header.request-uri.methodtype=='480'**
Action Subject	**header.request-uri.methodtype**
Action Type	**Modify**
Action Value	**'486'**

Figure 4-51: Configuring SIP Message Manipulation Rule 3 (for BroadConnect SIP Trunk)



Index	3
Name	Error Responses
Manipulation Set ID	4
Message Type	any.response
Condition	header.request-uri.meth
Action Subject	header.request-uri.meth
Action Type	Modify ▼
Action Value	'486'
Row Role	Use Current Condit ▼

Figure 4-52: Configured SIP Message Manipulation Rules



Inde:	Name	Manipulation Set ID	Message Type	Condition	Action Subject	Action Type	Action Value	Row Role
0	Change Host of Histor	4	invite.request	header.history-i	header.history	Modify	$1+$2+param	Use Current C
1	Remove History-Info.	4	invite.request		header.history	Remove		Use Current C
2	Change Referred-by I	4	invite.request	header.referred-	header.referre	Modify	param.ipg.dst.	Use Current C
3	Error Responses Test	4	any.response	header.request-	header.reques	Modify	'486'	Use Current C

The table below describes SIP message manipulation rules which belong to Manipulation Set ID 4 and which are executed on messages sent to the BroadConnect SIP Trunk IP Group.

These rules are specifically required to enable correct interworking between BroadConnect SIP Trunk and Skype for Business Server 2015. Refer to the *User's Manual* for further details on the full capabilities of header manipulation.

SIP Message Manipulation Rules

Rule Index	Rule Description	Reason for Introducing Rule
0	This rule applies to messages sent to the BroadConnect SIP Trunk IP Group in a call forwarding scenario. This rule replaces the host part of the SIP History-Info Header with the value configured in the BroadConnect SIP Trunk IP Group.	To introduce Topology Hiding in the Call Forward scenarios, the host part of the SIP History-Info Header should be replaced with the value that was configured in the SIP Trunk IP Group.
1	This rule also applies to messages sent to the BroadConnect SIP Trunk IP Group in a call forwarding scenario. This rule removes the SIP History-Info.1 Header.	To introduce Topology Hiding in the Call Forward scenarios, the SIP History-Info.1 Header should be removed.
2	This rule applies to messages sent to the BroadConnect SIP Trunk IP Group in a call transfer scenario. This replaces the host part of the SIP Referred-by Header with the value, configured in the BroadConnect SIP Trunk IP Group.	To introduce Topology Hiding in the Call Transfer scenarios, the host part of the SIP Referred-by Header should be replaced with the value that was configured in the SIP Trunk IP Group.
3	This rule is applied to response messages sent to the BroadConnect SIP Trunk IP Group for Rejected Calls initiated by the Skype for Business Server 2015 IP Group. This replaces the method type ‘403’ or ‘406’ or ‘480’ with the value '486'.	BroadConnect SIP Trunk does not recognize these method types.

6. Assign Manipulation Set ID 4 to the BroadConnect SIP Trunk IP Group:

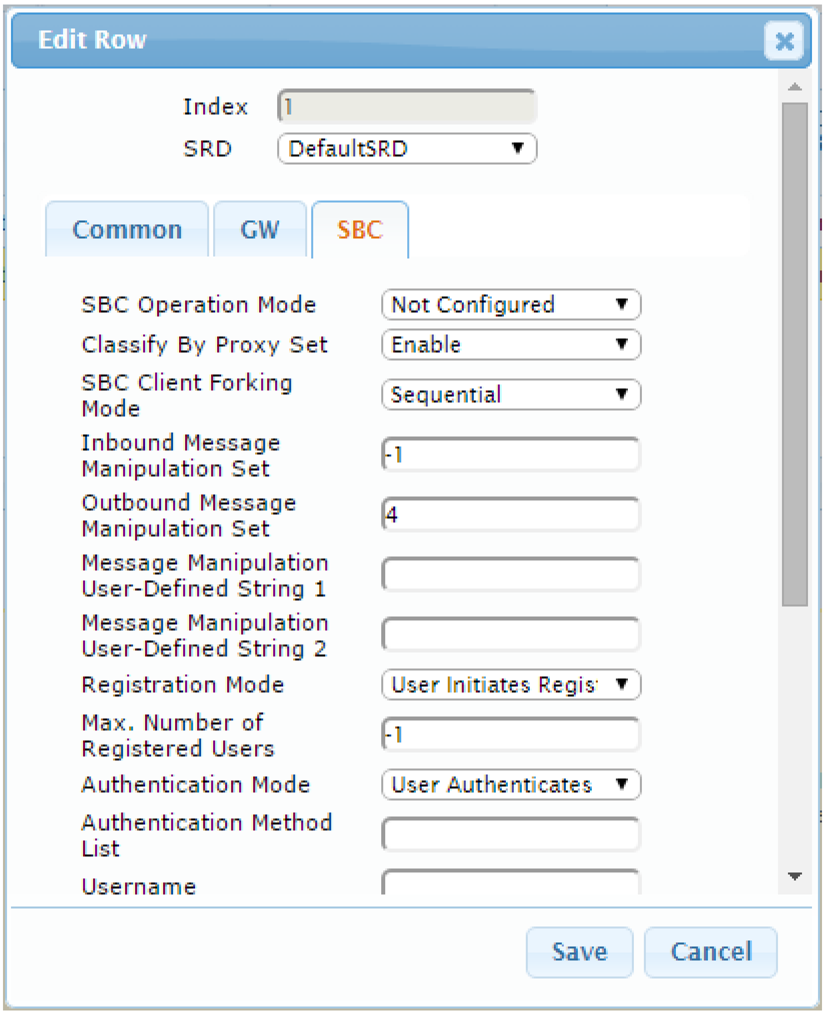
   a. Open the IP Group Table page (**Configuration** tab > **VoIP** menu > **VoIP Network** > **IP Group Table**).

   b. Select the row of the BroadConnect SIP Trunk IP Group, and then click **Edit**.

   c. Click the **SBC** tab.

   d. Set the 'Outbound Message Manipulation Set' field to **4**.

Figure 4-53: Assigning Manipulation Set 4 to the BroadConnect SIP Trunk IP Group



Index	1
SRD	DefaultSRD
SBC Operation Mode	Not Configured
Classify By Proxy Set	Enable
SBC Client Forking Mode	Sequential
Inbound Message Manipulation Set	-1
Outbound Message Manipulation Set	4
Message Manipulation User-Defined String 1	
Message Manipulation User-Defined String 2	
Registration Mode	User Initiates Regis
Max. Number of Registered Users	-1
Authentication Mode	User Authenticates
Authentication Method List	
Username	

   e. Click **Submit**.

4.15 Step 15: Configure Miscellaneous Settings

This section describes configuration of miscellaneous E-SBC settings.

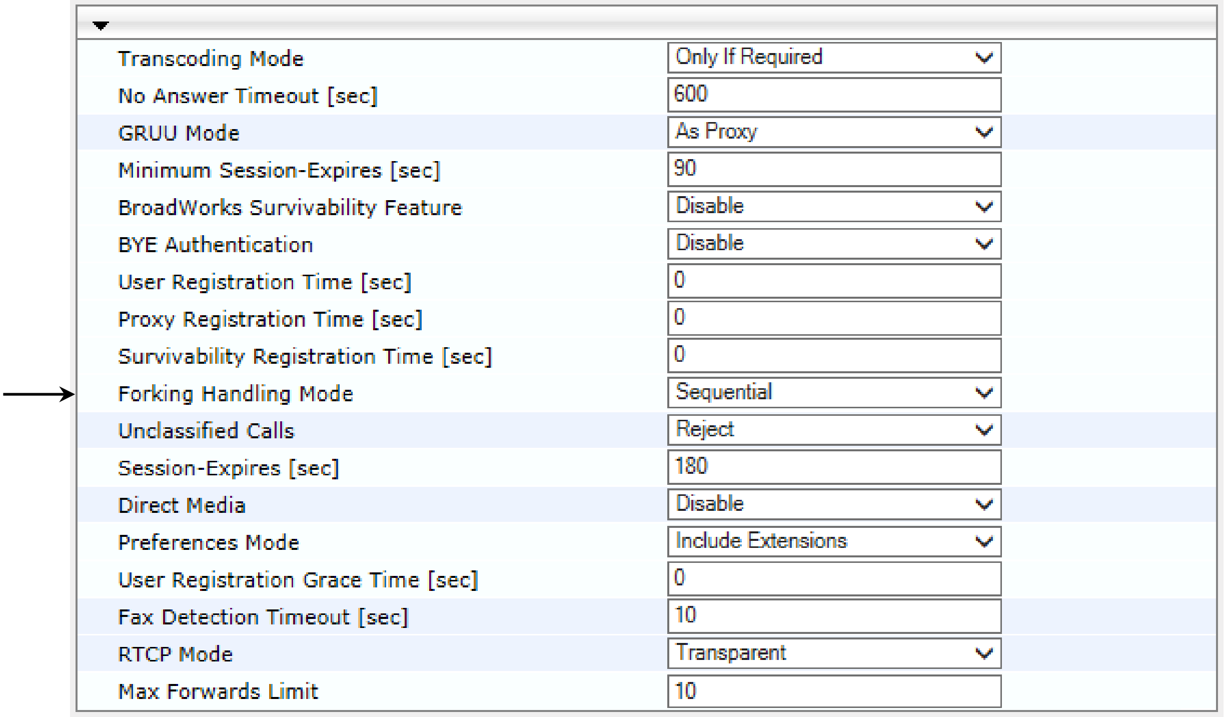
4.15.1 Step 15a: Configure Call Forking Mode

This step shows how to configure the E-SBC's handling of SIP 18x responses received for call forking of INVITE messages. For the interoperability test topology, if 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 a 180 response without SDP is received. It is mandatory to set this field for the Skype for Business Server 2015 environment.

➢ **To configure call forking:**

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

Figure 4-54: 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**.

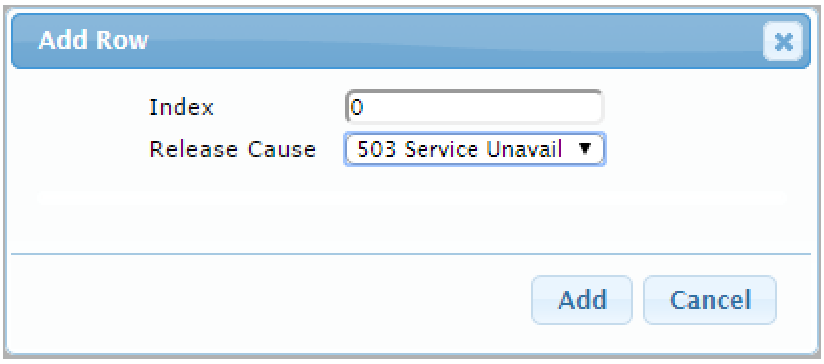
4.15.2 Step 15b: Configure SBC Alternative Routing Reasons

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

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

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

Figure 4-55: SBC Alternative Routing Reasons Table - Add Record



Index: 0

Release Cause: 503 Service Unavail

3. Click **Submit**.

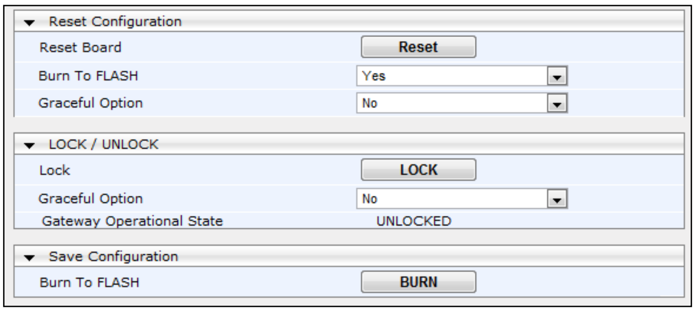
4.16 Step 16: Reset the E-SBC

After you have completed the configuration of the E-SBC as described in this section, 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 4-56: 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 is shown below:

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

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

;Board: Mediant 800 E-SBC
;HW Board Type: 69  FK Board Type: 74
;Serial Number: 5299378
;Slot Number: 1
;Software Version: 7.00A.026.016
;DSP Software Version: 5014AE3_R => 700.38
;Board IP Address: 10.15.17.55
;Board Subnet Mask: 255.255.0.0
;Board Default Gateway: 10.15.0.1
;Ram size: 369M   Flash size: 64M   Core speed: 500Mhz
;Num of DSP Cores: 3  Num DSP Channels: 30
;Num of physical LAN ports: 4
;Profile: NONE
;;Key features:;Board Type: 74 ;IP Media: Conf VXML
VoicePromptAnnounc(H248.9) CALEA TrunkTesting POC ;Channel Type: DspCh=30
IPMediaDspCh=30 ;Coders: G723 G729 G728 NETCODER GSM-FR GSM-EFR AMR EVRC-
QCELP G727 ILBC EVRC-B AMR-WB G722 EG711 MS_RTA_NB MS_RTA_WB SILK_NB
SILK_WB SPEEX_NB SPEEX_WB ;DSP Voice features: IpmDetector RTCP-XR
AMRPolicyManagement ;E1Trunks=1 ;FXSPorts=8 ;FXOPorts=0 ;BRITrunks=5
;Security: IPSEC MediaEncryption StrongEncryption EncryptControlProtocol
;DATA features: ;QOE features: VoiceQualityMonitoring MediaEnhancement
;Control Protocols: MSFT CLI TRANSCODING=30 FEU=100 TestCall=100 MGCP
MEGACO H323 SIP TPNCP SASurvivability SBC=50 ;Default features:;Coders:
G711 G726;

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


[SYSTEM Params]

SyslogServerIP = 10.15.17.100
EnableSyslog = 1
;NTPServerIP_abs is hidden but has non-default value
NTPServerUTCOffset = 7200
;VpFileLastUpdateTime is hidden but has non-default value
NTPServerIP = '10.15.27.1'
```

```
;LastConfigChangeTime is hidden but has non-default value
;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
;PM_gwSBCTranscodingSessions is hidden but has non-default value

[BSP Params]

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

[Analog Params]


[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 = 'BroadConnect'
UseWeblogo = 1
;UseLogoInWeb is hidden but has non-default value
UseProductName = 1
HTTPSCipherString = 'RC4:EXP'
;HTTPSPkeyFileName is hidden but has non-default value

[SIP Params]

MEDIACHANNELS = 30
```

```
REGISTRATIONTIME = 300
GWDEBUGLEVEL = 5
;ISPRACKREQUIRED is hidden but has non-default value
ENABLESBCAPPLICATION = 1
MSLDAPPRIMARYKEY = 'telephoneNumber'
SBCPREFERENCESMODE = 1
MEDIACDRREPORTLEVEL = 1
SBCFORKINGHANDLINGMODE = 1
ENERGYDETECTORCMD = 587202560
ANSWERDETECTORCMD = 10486144
;GWAPPCONFIGURATIONVERSION is hidden but has non-default value

[SCTP Params]


[IPsec Params]


[Audio Staging Params]


[SNMP Params]


[ PhysicalPortsTable ]

FORMAT PhysicalPortsTable_Index = PhysicalPortsTable_Port,
PhysicalPortsTable_Mode, PhysicalPortsTable_SpeedDuplex,
PhysicalPortsTable_PortDescription, PhysicalPortsTable_GroupMember,
PhysicalPortsTable_GroupStatus;
PhysicalPortsTable 0 = "GE_1", 1, 4, "User Port #0", "GROUP_1", "Active";
PhysicalPortsTable 1 = "GE_2", 1, 4, "User Port #1", "GROUP_1",
"Redundant";
PhysicalPortsTable 2 = "GE_3", 1, 4, "User Port #2", "GROUP_2", "Active";
PhysicalPortsTable 3 = "GE_4", 1, 4, "User Port #3", "GROUP_2",
"Redundant";

[ \PhysicalPortsTable ]


[ EtherGroupTable ]

FORMAT EtherGroupTable_Index = EtherGroupTable_Group,
EtherGroupTable_Mode, EtherGroupTable_Member1, EtherGroupTable_Member2;
EtherGroupTable 0 = "GROUP_1", 2, "GE_1", "GE_2";
EtherGroupTable 1 = "GROUP_2", 2, "GE_3", "GE_4";
EtherGroupTable 2 = "GROUP_3", 0, "", "";
EtherGroupTable 3 = "GROUP_4", 0, "", "";

[ \EtherGroupTable ]


[ DeviceTable ]

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

[ \DeviceTable ]


[ InterfaceTable ]

FORMAT InterfaceTable_Index = InterfaceTable_ApplicationTypes,
InterfaceTable_InterfaceMode, InterfaceTable_IPAddress,
InterfaceTable_PrefixLength, InterfaceTable_Gateway,
InterfaceTable_InterfaceName, InterfaceTable_PrimaryDNSServerIPAddress,
InterfaceTable_SecondaryDNSServerIPAddress,
InterfaceTable_UnderlyingDevice;
InterfaceTable 0 = 6, 10, 10.15.17.55, 16, 10.15.0.1, "Voice",
10.15.27.1, 0.0.0.0, "vlan 1";
InterfaceTable 1 = 5, 10, 195.189.192.158, 24, 195.189.192.129, "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 ]


[ WebUsers ]

FORMAT WebUsers_Index = WebUsers_Username, WebUsers_Password,
WebUsers_Status, WebUsers_PwAgeInterval, WebUsers_SessionLimit,
WebUsers_SessionTimeout, WebUsers_BlockTime, WebUsers_UserLevel,
WebUsers_PwNonce;
WebUsers 0 = "Admin",
"$1$LE0VGBxUAQFSUAJXUQANXwoPDwtaeSNwInB2c3B+eihzKSgvfDIzMDI1YGc0YWhub2hlP
GpUVwdVBlNSBgpRXV4=", 1, 0, 2, 15, 60, 200,
"62cabed25276f6d59432fcaf295a1346";
WebUsers 1 = "User",
"$1$fRwcHLO4tOHmvOKy7Oiys7m5vrbzpqfyoKL0r6v7q/iv/P35kpmUwcXBkZWYy5iaz8+Wm
NGBgoPXhdTRi4yDj94=", 3, 0, 2, 15, 60, 50,
"e124fc45691a62316416e055a60edb6f";

[ \WebUsers ]


[ 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", 1, "RC4:EXP", "ALL:!ADH", 0, 0.0.0.0, 0.0.0.0,
2560, 0;
```

```
[ \TLSContexts ]


[ 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_SBCUserBehindUdpNATRegistrationTime,
IpProfile_SBCUserBehindTcpNATRegistrationTime,
IpProfile_SBCSDPHandleRTCPAttribute,
IpProfile_SBCRemoveCryptoLifetimeInSDP, IpProfile_SBCIceMode,
IpProfile_SBCRTCPMux, IpProfile_SBCMediaSecurityMethod,
IpProfile_SBCHandleXDetect, IpProfile_SBCRTCPFeedback,
IpProfile_SBCRemoteRepresentationMode, IpProfile_SBCKeepVIAHeaders,
IpProfile_SBCKeepRoutingHeaders, IpProfile_SBCKeepUserAgentHeader,
IpProfile_SBCRemoteMultipleEarlyDialogs,
IpProfile_SBCRemoteMultipleAnswersMode, IpProfile_SBCDirectMediaTag,
IpProfile_SBCAdaptRFC2833BWToVoiceCoderBW;
IpProfile 1 = "S4B", 1, 0, 0, 10, 10, 46, 40, 0, 0, 0, 0, 2, 0, 0, 0, 0,
-1, 1, 0, 0, -1, 1, 4, -1, 1, 1, 0, 0, "", -1, 0, 0, "", -1, -1, 0, 1, 0,
0, 0, 0, 8, 300, 400, 0, 0, 0, -1, 0, 0, 1, 1, 0, 1, 1, 0, 3, 2, 1, 0, 1,
1, 1, 1, 1, 0, 1, 0, 0, 0, 0, 1, 0, 1, 1, 0, 0, 0, 0, 0, 0, 0, 0, 1, 0,
0, 300, -1, -1, 0, 0, 0, 0, 0, 0, 0, -1, -1, -1, -1, -1, 0, "", 0;
```

```
IpProfile 2 = "BroadConnect", 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, 2, 0, 0, 1, 0, 8, 300, 400, 0, 0, 0, -1, 0, 0, 1, 3, 0, 2, 2, 1, 3, 0,
1, 0, 1, 0, 0, 0, 0, 0, 1, 0, 0, 0, 0, 0, 0, 1, 0, 0, 0, 0, 0, 0, 0, 0,
1, 0, 0, 0, 300, -1, -1, 0, 0, 0, 0, 0, 0, 0, -1, -1, -1, -1, -1, 0, "",
0;

[ \IpProfile ]


[ 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, 100, 6990, 1, "", "";
CpMediaRealm 1 = "MRWan", "WANSP", "", 7000, 100, 7990, 0, "", "";

[ \CpMediaRealm ]


[ SBCRoutingPolicy ]

FORMAT SBCRoutingPolicy_Index = SBCRoutingPolicy_Name,
SBCRoutingPolicy_LCREnable, SBCRoutingPolicy_LCRAverageCallLength,
SBCRoutingPolicy_LCRDefaultCost, SBCRoutingPolicy_LdapServerGroupName;
SBCRoutingPolicy 0 = "Default_SBCRoutingPolicy", 0, 1, 1, "";

[ \SBCRoutingPolicy ]


[ SRD ]

FORMAT SRD_Index = SRD_Name, SRD_BlockUnRegUsers, SRD_MaxNumOfRegUsers,
SRD_EnableUnAuthenticatedRegistrations, SRD_SharingPolicy,
SRD_UsedByRoutingServer, SRD_SBCOperationMode,
SRD_SBCRegisteredUsersClassificationMethod, SRD_SBCRoutingPolicyName;
SRD 0 = "DefaultSRD", 0, -1, 1, 0, 0, 0, -1, "Default_SBCRoutingPolicy";

[ \SRD ]


[ SIPInterface ]

FORMAT SIPInterface_Index = SIPInterface_InterfaceName,
SIPInterface_NetworkInterface, SIPInterface_ApplicationType,
SIPInterface_UDPPort, SIPInterface_TCPPort, SIPInterface_TLSPort,
SIPInterface_SRDName, SIPInterface_MessagePolicyName,
SIPInterface_TLSContext, SIPInterface_TLSMutualAuthentication,
SIPInterface_TCPKeepaliveEnable,
SIPInterface_ClassificationFailureResponseType,
SIPInterface_PreClassificationManSet, SIPInterface_EncapsulatingProtocol,
SIPInterface_MediaRealm, SIPInterface_SBCDirectMedia,
SIPInterface_BlockUnRegUsers, SIPInterface_MaxNumOfRegUsers,
SIPInterface_EnableUnAuthenticatedRegistrations,
SIPInterface_UsedByRoutingServer;
SIPInterface 0 = "S4B", "Voice", 2, 0, 0, 5067, "DefaultSRD", "",
"default", -1, 0, 500, -1, 0, "MRLan", 0, -1, -1, -1, 0;
SIPInterface 1 = "BroadConnect", "WANSP", 2, 5060, 0, 0, "DefaultSRD",
"", "default", -1, 0, 500, -1, 0, "MRWan", 0, -1, -1, -1, 0;
```

```
[ \SIPInterface ]


[ ProxySet ]

FORMAT ProxySet_Index = ProxySet_ProxyName,
ProxySet_EnableProxyKeepAlive, ProxySet_ProxyKeepAliveTime,
ProxySet_ProxyLoadBalancingMethod, ProxySet_IsProxyHotSwap,
ProxySet_SRDName, ProxySet_ClassificationInput, ProxySet_TLSContextName,
ProxySet_ProxyRedundancyMode, ProxySet_DNSResolveMethod,
ProxySet_KeepAliveFailureResp, ProxySet_GWIPv4SIPInterfaceName,
ProxySet_SBCIPv4SIPInterfaceName, ProxySet_SASIPv4SIPInterfaceName,
ProxySet_GWIPv6SIPInterfaceName, ProxySet_SBCIPv6SIPInterfaceName,
ProxySet_SASIPv6SIPInterfaceName;
ProxySet 0 = "S4B", 0, 60, 1, 1, "DefaultSRD", 0, "default", 1, -1, "",
"", "S4B", "", "", "", "";
ProxySet 1 = "BroadConnect", 1, 60, 0, 0, "DefaultSRD", 0, "", -1, -1,
"", "", "BroadConnect", "", "", "", "";

[ \ProxySet ]


[ IPGroup ]

FORMAT IPGroup_Index = IPGroup_Type, IPGroup_Name, IPGroup_ProxySetName,
IPGroup_SIPGroupName, IPGroup_ContactUser, IPGroup_SipReRoutingMode,
IPGroup_AlwaysUseRouteTable, IPGroup_SRDName, IPGroup_MediaRealm,
IPGroup_ClassifyByProxySet, IPGroup_ProfileName,
IPGroup_MaxNumOfRegUsers, IPGroup_InboundManSet, IPGroup_OutboundManSet,
IPGroup_RegistrationMode, IPGroup_AuthenticationMode, IPGroup_MethodList,
IPGroup_EnableSBCClientForking, IPGroup_SourceUriInput,
IPGroup_DestUriInput, IPGroup_ContactName, IPGroup_Username,
IPGroup_Password, IPGroup_UUIFormat, IPGroup_QOEProfile,
IPGroup_BWProfile, IPGroup_MediaEnhancementProfile,
IPGroup_AlwaysUseSourceAddr, IPGroup_MsgManUserDef1,
IPGroup_MsgManUserDef2, IPGroup_SIPConnect, IPGroup_SBCPSAPMode,
IPGroup_DTLSContext, IPGroup_CreatedByRoutingServer,
IPGroup_UsedByRoutingServer, IPGroup_SBCOperationMode,
IPGroup_SBCRouteUsingRequestURIPort;
IPGroup 0 = 0, "S4B", "S4B", "195.189.192.158", "", -1, 0, "DefaultSRD",
"MRLan", 1, "S4B", -1, -1, -1, 0, 0, "", 0, -1, -1, "", "", "$1$gQ==", 0,
"", "", "", 0, "", "", 0, 0, "", 0, 0, -1, 0;
IPGroup 1 = 0, "BroadConnect", "BroadConnect", "195.189.192.158", "", -1,
0, "DefaultSRD", "MRWan", 1, "BroadConnect", -1, -1, 4, 0, 0, "", 0, -1,
-1, "", "", "$1$gQ==", 0, "", "", "", 0, "", "", 0, 0, "", 0, 0, -1, 0;

[ \IPGroup ]


[ SBCAlternativeRoutingReasons ]

FORMAT SBCAlternativeRoutingReasons_Index =
SBCAlternativeRoutingReasons_ReleaseCause;
SBCAlternativeRoutingReasons 0 = 503;

[ \SBCAlternativeRoutingReasons ]


[ ProxyIp ]
```

```
FORMAT ProxyIp_Index = ProxyIp_ProxySetId, ProxyIp_ProxyIpIndex,
ProxyIp_IpAddress, ProxyIp_TransportType;
ProxyIp 0 = "0", 0, "FE.S4B.interop:5067", 2;
ProxyIp 1 = "1", 0, "208.82.88.105:5060", 0;

[ \ProxyIp ]


[ IP2IPRouting ]

FORMAT IP2IPRouting_Index = IP2IPRouting_RouteName,
IP2IPRouting_RoutingPolicyName, IP2IPRouting_SrcIPGroupName,
IP2IPRouting_SrcUsernamePrefix, IP2IPRouting_SrcHost,
IP2IPRouting_DestUsernamePrefix, IP2IPRouting_DestHost,
IP2IPRouting_RequestType, IP2IPRouting_MessageConditionName,
IP2IPRouting_ReRouteIPGroupName, IP2IPRouting_Trigger,
IP2IPRouting_CallSetupRulesSetId, IP2IPRouting_DestType,
IP2IPRouting_DestIPGroupName, IP2IPRouting_DestSIPInterfaceName,
IP2IPRouting_DestAddress, IP2IPRouting_DestPort,
IP2IPRouting_DestTransportType, IP2IPRouting_AltRouteOptions,
IP2IPRouting_GroupPolicy, IP2IPRouting_CostGroup;
IP2IPRouting 0 = "OPTIONS termination", "Default_SBCRoutingPolicy",
"Any", "*", "*", "*", "*", 6, "", "Any", 0, -1, 1, "", "", "internal", 0,
-1, 0, 0, "";
IP2IPRouting 1 = "S4B to ITSP", "Default_SBCRoutingPolicy", "S4B", "*",
"*", "*", "*", 0, "", "Any", 0, -1, 0, "BroadConnect", "BroadConnect",
"", 0, -1, 0, 0, "";
IP2IPRouting 2 = "ITSP to S4B", "Default_SBCRoutingPolicy",
"BroadConnect", "*", "*", "*", "*", 0, "", "Any", 0, -1, 0, "S4B", "S4B",
"", 0, -1, 0, 0, "";

[ \IP2IPRouting ]


[ IPOutboundManipulation ]

FORMAT IPOutboundManipulation_Index =
IPOutboundManipulation_ManipulationName,
IPOutboundManipulation_RoutingPolicyName,
IPOutboundManipulation_IsAdditionalManipulation,
IPOutboundManipulation_SrcIPGroupName,
IPOutboundManipulation_DestIPGroupName,
IPOutboundManipulation_SrcUsernamePrefix, IPOutboundManipulation_SrcHost,
IPOutboundManipulation_DestUsernamePrefix,
IPOutboundManipulation_DestHost,
IPOutboundManipulation_CallingNamePrefix,
IPOutboundManipulation_MessageConditionName,
IPOutboundManipulation_RequestType,
IPOutboundManipulation_ReRouteIPGroupName,
IPOutboundManipulation_Trigger, IPOutboundManipulation_ManipulatedURI,
IPOutboundManipulation_RemoveFromLeft,
IPOutboundManipulation_RemoveFromRight,
IPOutboundManipulation_LeaveFromRight, IPOutboundManipulation_Prefix2Add,
IPOutboundManipulation_Suffix2Add,
IPOutboundManipulation_PrivacyRestrictionMode;
IPOutboundManipulation 0 = "Add + toward S4B",
"Default_SBCRoutingPolicy", 0, "BroadConnect", "S4B", "*", "*", "*", "*",
"*", "", 0, "Any", 0, 1, 0, 0, 255, "+", "", 0;
IPOutboundManipulation 2 = "Change + on 011", "Default_SBCRoutingPolicy",
0, "S4B", "BroadConnect", "*", "*", "+", "*", "*", "", 0, "Any", 0, 1, 1,
0, 255, "011", "", 0;
```

```
IPOutboundManipulation 3 = "Remove + from Source",
"Default_SBCRoutingPolicy", 0, "S4B", "BroadConnect", "+", "*", "*", "*",
"*", "", 0, "Any", 0, 0, 0, 0, 255, "", "", 0;

[ \IPOutboundManipulation ]


[ CodersGroup0 ]

FORMAT CodersGroup0_Index = CodersGroup0_Name, CodersGroup0_pTime,
CodersGroup0_rate, CodersGroup0_PayloadType, CodersGroup0_Sce,
CodersGroup0_CoderSpecific;
CodersGroup0 0 = "g711Ulaw64k", 20, 0, -1, 0, "";
CodersGroup0 1 = "g711Alaw64k", 20, 0, -1, 0, "";

[ \CodersGroup0 ]


[ CodersGroup1 ]

FORMAT CodersGroup1_Index = CodersGroup1_Name, CodersGroup1_pTime,
CodersGroup1_rate, CodersGroup1_PayloadType, CodersGroup1_Sce,
CodersGroup1_CoderSpecific;
CodersGroup1 0 = "g711Ulaw64k", 20, 0, -1, 0, "";

[ \CodersGroup1 ]


[ CodersGroup2 ]

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

[ \CodersGroup2 ]


[ AllowedCodersGroup1 ]

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

[ \AllowedCodersGroup1 ]


[ AllowedCodersGroup2 ]

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

[ \AllowedCodersGroup2 ]


[ MessageManipulations ]

FORMAT MessageManipulations_Index =
MessageManipulations_ManipulationName, MessageManipulations_ManSetID,
MessageManipulations_MessageType, MessageManipulations_Condition,
MessageManipulations_ActionSubject, MessageManipulations_ActionType,
MessageManipulations_ActionValue, MessageManipulations_RowRole;
MessageManipulations 0 = "Change Host of History-Info.0", 4,
"invite.request", "header.history-info.0 regex
(.*)(@)(.*)(;user=phone)(.*)", "header.history-info.0", 2,
"$1+$2+param.ipg.dst.host+$4+$5", 0;
MessageManipulations 1 = "Remove History-Info.1", 4, "invite.request",
"", "header.history-info.1", 1, "", 0;
MessageManipulations 2 = "Change Referred-by Host", 4, "invite.request",
"header.referred-by exists", "header.referred-by.url.host", 2,
"param.ipg.dst.host", 0;
MessageManipulations 3 = "Error Responses Test", 4, "any.response",
"header.request-uri.methodtype=='403' OR header.request-
uri.methodtype=='406' OR header.request-uri.methodtype=='480'",
"header.request-uri.methodtype", 2, "'486'", 0;

[ \MessageManipulations ]


[ GwRoutingPolicy ]

FORMAT GwRoutingPolicy_Index = GwRoutingPolicy_Name,
GwRoutingPolicy_LCREnable, GwRoutingPolicy_LCRAverageCallLength,
GwRoutingPolicy_LCRDefaultCost, GwRoutingPolicy_LdapServerGroupName;
GwRoutingPolicy 0 = "GwRoutingPolicy", 0, 1, 1, "";

[ \GwRoutingPolicy ]


[ ResourcePriorityNetworkDomains ]

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

[ \ResourcePriorityNetworkDomains ]
```

This page is intentionally left blank.

International Headquarters
1 Hayarden Street,
Airport City
Lod 7019900, Israel
Tel: +972-3-976-4000
Fax: +972-3-976-4040

AudioCodes Inc.
27 World's Fair Drive,
Somerset, NJ 08873
Tel: +1-732-469-0880
Fax: +1-732-469-2298

Contact us: www.audiocodes.com/info
Website: www.audiocodes.com

AudioCodes